### Our Gate 2009 Toppers

#### Civil Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Sathish</td>
<td>CE1620834</td>
<td>1</td>
</tr>
<tr>
<td>02</td>
<td>Neta Prasad</td>
<td>CE4920906</td>
<td>2</td>
</tr>
<tr>
<td>03</td>
<td>A. Bhasker</td>
<td>CE1430312</td>
<td>3</td>
</tr>
<tr>
<td>04</td>
<td>Moinul Ahsan Khan</td>
<td>CE1430075</td>
<td>5</td>
</tr>
<tr>
<td>05</td>
<td>Cha, S. B.</td>
<td>CE6090202</td>
<td>10</td>
</tr>
</tbody>
</table>

#### Mechanical Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Desai Haridhikumar G.</td>
<td>ME1580097</td>
<td>7</td>
</tr>
<tr>
<td>02</td>
<td>K. Venkatachalapathi</td>
<td>ME1580093</td>
<td>9</td>
</tr>
</tbody>
</table>

#### Computer Science & Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Jithin Vachary</td>
<td>CS7890420</td>
<td>2</td>
</tr>
<tr>
<td>02</td>
<td>Venkata Satyakrishnan</td>
<td>CS1540909</td>
<td>10</td>
</tr>
</tbody>
</table>

#### Electrical Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Anil Kumar An</td>
<td>EE5240284</td>
<td>2 (Test series)</td>
</tr>
<tr>
<td>02</td>
<td>Girish KumarDasari</td>
<td>EE6760825</td>
<td>7 (Test series)</td>
</tr>
<tr>
<td>03</td>
<td>V. MalladiRamachandru</td>
<td>EE7640633</td>
<td>13</td>
</tr>
<tr>
<td>04</td>
<td>M. NagaRaju Reddy</td>
<td>EE7630195</td>
<td>1a</td>
</tr>
<tr>
<td>05</td>
<td>Thokare Nitin</td>
<td>EE1600530</td>
<td>18</td>
</tr>
</tbody>
</table>

#### Electronics & Communication Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Myeshwantu</td>
<td>EC1440615</td>
<td>9</td>
</tr>
<tr>
<td>02</td>
<td>Sandip Sharath vaiker</td>
<td>EC1460408</td>
<td>11</td>
</tr>
<tr>
<td>03</td>
<td>Venkatesh Sasanalpur</td>
<td>EC1490624</td>
<td>18</td>
</tr>
</tbody>
</table>

#### Instrumentation Engineering Toppers

<table>
<thead>
<tr>
<th>S.No</th>
<th>Student Name</th>
<th>Gate H.T. No</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Mahesh Ravi Varma</td>
<td>IN1570571</td>
<td>2</td>
</tr>
<tr>
<td>02</td>
<td>Md. Abdul Razak</td>
<td>IN970385</td>
<td>3</td>
</tr>
<tr>
<td>03</td>
<td>Maheshwar Rithika</td>
<td>IN2300399</td>
<td>6</td>
</tr>
</tbody>
</table>

### COMMUNICATION SYSTEMS CONTENTS

<table>
<thead>
<tr>
<th>S.No</th>
<th>Chapter</th>
<th>Page No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>Introduction to Signals, Spectra and Communication System</td>
<td>01 - 12</td>
</tr>
<tr>
<td>02</td>
<td>Random Signals and Noise</td>
<td>13 - 20</td>
</tr>
<tr>
<td>03</td>
<td>Analog Communication Systems</td>
<td>21 - 24</td>
</tr>
<tr>
<td></td>
<td>(A) Amplitude Modulation</td>
<td>25 - 47</td>
</tr>
<tr>
<td></td>
<td>(B) Angle Modulation</td>
<td>48 - 62</td>
</tr>
<tr>
<td></td>
<td>(C) Receivers</td>
<td>63 - 70</td>
</tr>
<tr>
<td></td>
<td>(D) Noise in Analog Modulation</td>
<td>70 - 74</td>
</tr>
<tr>
<td></td>
<td>→ Objective Questions</td>
<td>75 - 98</td>
</tr>
<tr>
<td>04</td>
<td>Fundamentals of Information theory and Channel capacity theorem</td>
<td>99 - 101</td>
</tr>
<tr>
<td></td>
<td>→ Objective Questions</td>
<td></td>
</tr>
<tr>
<td>05</td>
<td>Digital Communications</td>
<td>102 - 115</td>
</tr>
<tr>
<td></td>
<td>(A) Introduction to digital communication, sampling,</td>
<td></td>
</tr>
<tr>
<td></td>
<td>PAM, PWM, PPM, PCM, DPCM, and DM</td>
<td></td>
</tr>
<tr>
<td></td>
<td>→ Objective Questions</td>
<td>116 - 127</td>
</tr>
<tr>
<td></td>
<td>(B) ASK, PSK, FSK, SF SK</td>
<td>128 - 133</td>
</tr>
<tr>
<td></td>
<td>(C) Matched filters / Correlation Receiver, B.W., Probability</td>
<td>134 - 143</td>
</tr>
<tr>
<td></td>
<td>of error</td>
<td></td>
</tr>
<tr>
<td></td>
<td>→ Objective Questions</td>
<td>145 - 148</td>
</tr>
<tr>
<td>06</td>
<td>Basics of TDMA, FDMA &amp; CDMA and GMS</td>
<td>149 - 151</td>
</tr>
<tr>
<td></td>
<td>→ Objective Questions</td>
<td></td>
</tr>
</tbody>
</table>

for

GATE, DRDO, & IES

Y. V. Gopala Krishna Murthy
Managing Director
GATE SYLLABUS

COMMUNICATION SYSTEMS

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

**Case (ii):** \( y(t) = e^{-\alpha |t|} \)

**Case (iii):** \( y(t) = 0 \)

**Exponentially Decreasing Function:**

- **A-Exponential Function:**
  - \( x(t) = A e^{-\alpha t} \)

**Sinusoidal Signal:**

- \( x(t) = A \cos(\omega t + \phi) \) or \( x(t) = A \sin(\omega t + \phi) \)
- \( A = \) Max amplitude, \( \phi = \) phase shift
- \( \omega t = \) angular freq = \( 2\pi f \) \((\text{rad/sec})\)

**Rectangular Pulse:**

- Pulse of amplitude \( A \)
- \( T_c \) centered around the origin

**Cosine Signal for \( \phi = 0 \):**

**Sine Signal for \( \phi = 0 \):**

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

8. **Rectangular Pulse:**

- Can be considered as a very very narrow rectangular signal of area = 1

When the pulse width approaches zero, amplitude goes to \( \infty \).

**Basic Operations on signals:**

1. **Amplitude scaling:**
   - Consider a signal \( x(t) \). Using this generate
   - \( y(t) = C x(t) \)
   - \( C = \) amplitude scaling factor
   - \( x(t) \) = Amplifier input
   - \( y(t) = 10 \times x(t) \)

2. **Addition or subtraction of signals:**
   - \( x_1(t) \) and \( x_2(t) \)
   - \( y(t) = x_1(t) \pm x_2(t) \)
3. Multiplication of signals: \( x(t) \) and \( x(-t) \)

\[
\gamma(t) = x(t) \cdot x(-t)
\]

4. Folding or Reflection of Signal: \( y(t) = x(-t) \)

5. Time shifting of signals: \( y(t) = x(t - t_0) \Rightarrow \) delayed by \( t_0 \) sec

6. Time Scaling of signals:

\[
\gamma(t) = x(at)
\]

When \( a = 1 \)
- Signal is compressed
- Signal is expanded

7. Convolution of the \( x(t) \) and \( g(t) \)

\[
\delta(t) = \int_{-\infty}^{\infty} g(-\tau) g(\tau + t) \, d\tau
\]

This operation is useful in system analysis.

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

\[
y(t) = x(t) * h(t)
\]

\( x(t) = u(t) - u(t-5) \), \( u(t) \) is a step function.

\( g(t) * \delta(t) = g(t) \)

\( g(t) * \delta(t-a_1) = g(t-a_1) \)
FOURIER TRANSFORM (SPECTRUM)

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

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

Bandwidth: The range of frequencies occupied by the signal is called bandwidth.

\[ \text{BW} = \delta f - \delta f \]

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

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

\[ G(f) = \int_{-T/2}^{T/2} g(t)e^{-j2\pi ft} \, dt \]

\[ \frac{1}{T} \int_{-T/2}^{T/2} G(f)e^{j2\pi ft} \, df = \frac{AT}{\pi fT} \sin(\pi fT) \]

- While calculating the BW don't consider the -ve side because -ve frequencies do not exist practically
- In order to reduce the BW, eliminate the insignificant frequencies.
- In order to reduce the insignificant high frequencies, pass the signal through LPF (H(f)).
- At the OP we get only main lobe
Properties of Fourier Transform:

01. Linearity property:

\[ g(t) \rightarrow G(f) \quad g(t - t_0) \rightarrow e^{-j2\pi ft_0}G(f) \]

\[ e^{j2\pi ft_0}g(t) \rightarrow G(f - f_0) \]

Example: Consider rectangular signals \( g(t) \) and \( g(t) \)

\[ g(t) \rightarrow 40 \text{sinc}(20t) \]

\[ g(t) \rightarrow 12 \text{sinc}(4t) \]

02. Duality property:

\[ \delta(t) \rightarrow G(0) \]

\[ G(0) \rightarrow \delta(t) \]

\[ \text{AT sinc}(T) \rightarrow G(f) \]

\[ g(t) \rightarrow 1, \quad \delta(t) = \delta(t) \]

\[ 1 \rightarrow \delta(f) \quad \delta(t) \rightarrow \delta(f) \]

03. Time Shifting property:

\[ g(t) \leftrightarrow G(f) \]

\[ g(t - t_0) \leftrightarrow e^{-j2\pi ft_0}G(f) \]

04. Frequency Shifting property (or) Modulation property:

\[ g(t) \leftrightarrow G(f) \]

\[ 1 \rightarrow \delta(t) \]

\[ e^{j2\pi ft_0} \rightarrow \delta(f - f_0) \]

\[ g(t) \text{cos}(2\pi ft_0) \rightarrow \frac{1}{2} [G(f - f_0) + G(f + f_0)] \]

05. Convolution property:

\[ g(t) \rightarrow G(f) \rightarrow g(t) \rightarrow G(f) \]

\[ g(t) \text{cos}(2\pi ft_0) \rightarrow G(f - f_0) + G(f + f_0) \]

Communication systems

Generalized Block Diagram

Irrespective of the form of communication process being considered, there are three basic elements in every communication system, namely, transmitter, channel, and receiver.
Communication Systems

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

Modulation converts:
1. Low frequency signal to a high frequency signal.
2. A wideband signal into a narrowband signal.
3. A baseband signal into a bandpass signal.

Need for Modulation:

1. To reduce the antenna height.
   - The antenna height required to transmit a signal depends on operating wavelength. For efficient radiation, the minimum antenna height should be \( \frac{\lambda}{10} \). To transmit a low frequency signal antenna height required is very high. To reduce the antenna height, the low frequency signal is converted into a high frequency signal by modulation.

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

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

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

Modulation converts a wideband signal into a narrowband signal whose ratio between highest frequency to lowest frequency is approximately one and single antennas will be sufficient to transmit the signal.
To win the ACE join the ACE
Communication Systems  

\[ \int \exp(-x^2) \, dx = 1, \quad x = \sqrt{\pi} \cdot \gamma \left( \frac{1}{2} \right) \]  
\[ \left( 2^{1/4} \pi \right) \int \exp(-x^2) \, dx + \left( 2^{1/4} \pi \right) \int \exp(-x^2) \, dx = 2 \]  
\[ \text{erf}(x) = \frac{1}{2} \exp(1/4) \]  
\[ Q(x) = \frac{1}{2} \exp(1/4) \int \exp(-x^2) \, dx \]

**Common**  
\[ F(x) = \frac{1}{2} \left[ 1 + \text{erf}(x/\sigma) \right] = 1 - \frac{1}{2} \text{erf}(x/\sigma) = 1 - Q(x/\sigma) \]  
\[ P(X > K \sigma) = P(X < -K \sigma) = Q(K) \]

**Rayleigh**  
\[ F(x) = 1 - \exp(-x^2/2\sigma^2), \, x \geq 0 \]  
\[ = 0, \quad x < 0 \]

**Statistical Averages**  
\[ E[X] = \mu_x = \frac{1}{\sigma} \int x f(x) \, dx \]  
\[ \text{Average value of } X \]

\[ \mu_{[g(x)]} = \frac{1}{\sigma} \int g(x) f(x) \, dx \]  
\[ \text{Expected value of } X' \]

\[ \mu_{[Y^2]} = \text{MSV of } X = \text{Expected Value of } X^2 \]  
\[ = \int x^2 f(x) \, dx \]  
\[ \text{Var}(X) = E[(X - \mu_x)^2] = \text{MSV}(X) - (\mu_X)^2 \]

**Standard Deviation**  
\[ \sigma = \sqrt{\text{Var}(X)} \]  
\[ \text{Distance distribution: } \frac{1}{\sqrt{2 \pi} \sigma} \cdot 2 \sigma^2 \]

**Random Process**

\[ \text{Random signals and noise} \]

**Exp. Distribution:**  
\[ f(x) = (\lambda x) \exp(-\lambda x), \quad x > 0 \]  
\[ = 0, \quad x < 0 \]  
\[ F(x) = 0, \quad x < 0 \]

\[ \text{Rayleigh Dist.} \]  
\[ = 1 - \exp(-x^2), \quad x > 0 \]

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

\[ \text{MV: } \mu = 0, \text{ VAR: } 2/4 \]

**Two Random Variables**

\[ X \rightarrow Y \rightarrow Y' \]

\[ \text{Joint Probability Distribution Function: } f(x, y) \]

\[ \text{Joint Probability Density Function: } f(x, y) \]

\[ f(x, y) = f(x) f(y) \]

\[ \text{Volume} = \int f(x, y) \, dx \, dy = 1 \]

\[ \text{Var} = \frac{1}{2} \int f(x, y) \, dy = \frac{1}{2} \int f(x, y) \, dx \]

\[ \text{Standard Deviation: } \text{Var}(X, Y) = \text{Var}(X) = \text{Var}(Y) \]

\[ \text{Statistical Independence: } f(x, y) = f(x) f(y) \]

**Random Processes**

\[ \text{RVs: } X, Y \text{ described by } f(x), f(y), \text{ MSV, VAR, SD} \]

**VARs:**  
\[ \text{RVs: } X, Y \text{ described by } f(x), f(y), \text{ MSV, VAR, SD} \]

\[ \text{RVs: } X, Y \text{ described by } f(x), f(y), \text{ MSV, VAR, SD} \]

**Statistical Independence:**  
\[ f(x, y) = f(x) f(y) \]

\[ \text{Random Processes: } \{ X(t), Y(t), Z(t), \ldots \} \text{ensemble of sample functions shown below} \]

**EX:** Voice/TV signals, Electrical Noise
The RV's observed at $t_k$, $k = 1, 2, 3, \ldots$ are

\[ X(t_k), X(t_{k+1}), X(t_{k+2}), \ldots \]

can be considered as components of a random process $X(t)$ whose statistical properties are described by its cdf, pdf, $M(t)$, $MSV$, VAR, SD.

The RV's $X(t)$ & $X_0(t)$ are described by joint cdf, joint pdf, COR (noise).

Let $X(t)$ be STATIONARY.

Then pdf's of RV's are invariant under a time shift $\tau$.

If pdf's of RV's are invariant under a time shift $\tau$.

\[ \text{E}[X(t_k)] = \text{E}[X(t)] = \text{Cov}[X(t), X(t_0)] = k = 1, 2, \ldots \]

Also the joint PDF of $X(t_k)$ and $X(t)$ is the same as the joint pdf of $X(t_0)$ and $X(0)$.

\[ \text{ACF of } X(t_k) \& X(t_0) = \text{ACF of } X(t_0) \& X(0) = R_x(\tau) \]

Where $\tau = t_k - t_0 = t_0 - t_k = \text{Time difference between the observation instants.}$

\[ \text{ACF: } R_x(\tau) = \text{E}[X(t)X(t+\tau)] = \text{E}[X(t_0)X(t_0+\tau)] \]

Properties:

1. $R_x(0) = \text{E}[X(0)] - \text{MSV} = \text{Power}$
2. $R_x(\tau) = R_x(-\tau)$, EVEN.
3. $| R_x(\tau) | \leq R_x(0)$
4. $R_x(\tau) \rightarrow S_x(t_0)$, $\text{ACF} \rightarrow \text{PSD}$
**Communication Systems**

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

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

- Duration of the pulse = T
- In any pulse duration $T$, the value of $x(t)$ is $\pm 1$ with equal probability. The value $x(t)$ in a given pulse duration $T$ is statistically independent of the value of $x(t)$ in other durations of $T$. The time of occurrence of the first pulse from the origin is shown as $t_0$. It is a random variable, $t_0 \sim T$, $0 \leq t_0 < T$

$$E\left\{X(t)\right\} = 0, \quad E\{X(t)X(h)\} = R_{xx}(t) = \begin{cases} A \left( \frac{|t|}{T} \right), & |t| < T \\ 0, & |t| \geq T \end{cases}$$

Where $t = (t_2 - t_1)$

$$S_x(t) = A^2 T \sin^2(Tt)$$

**THERMAL NOISE**

$$\begin{array}{c}
\begin{array}{c}
E\left\{v_{n}^2\right\} = \frac{KT}{2} \cdot \text{THz}^2 \\
E\left\{v_{m}^2\right\} = \frac{KT}{2} \cdot \text{THz}^2
\end{array}
\end{array}$$

Where $K = \text{Boltzmann Constant} = 1.38 \times 10^{-23} \text{J/K}$ and $T = \text{Abs. Temp. in } ^\circ \text{K}$

**ACE Academy**

**Random signals and noise**

Thermal Noise is Gaussian with $MV = 0$.

- Ave. Noise Power ($\text{max}$) $P_{\text{max}} = \frac{KT}{2} \cdot \text{THz}$
- Ave. Noise PSD $= \frac{\eta}{2}$ W/Hz

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

Thermal Noise is modelled as White Noise.

PSD of white noise $S_w(f)$ and ACF $R_w(t)$ are shown below:

$$S_w(f) = \frac{\eta}{2}$$

$$R_w(t) = \frac{\eta}{2} \delta(t)$$

**Ex:** White noise through Ideal LPF.

**Sketch** $R_w(t)$

**Ex:** White noise through simple practical RC LPF.

$$R(f) = \frac{1}{1 + (f/f_c)^2}$$

Where $f_c = 3 \text{ dB BW} = 1/(2\pi RC)$
Communication Systems

\[ |H(f)|^2 = R_x(t) = \eta \exp \left( -\frac{\| t \|}{4RC} \right) \]

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

A typical P.D.S. \( S_f(f) \) is shown below.

Normally \( f_c \gg 2B \).

\( n(t) \) is referred to as Narrow Band noise.

Similar to a deterministic BP signal \( y(t) \):

\[
\begin{align*}
    n(t) &= n_0(t) \cos(2\pi f_c t) - n_0(t) \sin(2\pi f_c t) - r(t) \cos(2\pi f_c t + \sigma(t)) \\
    n(t) &= n_0(t) \cos(2\pi f_c t) + \hat{r}(t) \sin(2\pi f_c t) \\
    R_n(t) &= R_n(t) \\
    S_n(f) &= S_n(f) = S_n(f - f_c) = S_n(f + f_c), \quad |f| < B \\
    &= 0, \quad |f| > B
\end{align*}
\]

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

\[
S_n(f) = \begin{cases} 
10^4 & |f| < 10^4 \\
0 & |f| > 10^4 
\end{cases}
\]

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

a) Sketch neatly the PSD of the output noise process.

b) Determine the output noise power.

c) Using the bandpass representation for the output noise process, sketch the PSD of the inphase and quadrature noise components, and determine their respective power.

Objective Questions

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

The corresponding cumulative distribution function (CDF) has the form:

\[ F_X(x) = 1 - \exp(-x^2/2) \]

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

\[ (a) K \quad (b) Kx_c \quad (c) Kx \quad (d) K \]

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

- (a) \( R(t) = R(-t) \)
- (b) \( R(t) \leq R(0) \)
- (c) \( R(t) = -R(-t) \)
- (d) The mean square value of the process is \( R(0) \)

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

- (a) \( S(f) \geq S(f') \) 
- (b) \( S(f) > 0 \)
- (c) \( S(f) \geq 0 \) 
- (d) \( S(f) = S(f') \)

05. During transmission over a certain binary communication channel, bit errors occur independently with probability \( p \). The probability of AT MOST one bit error in a block of 10 bits is given by

- (a) \( 1-p \) 
- (b) \( 1-p^2 \) 
- (c) \( p(1-p)^9 + (1-p)^9 \) 
- (d) \( 1-p \)
Statement for Linked Answer Questions 88 & 89.
The following two questions refer to wide sense stationary stochastic processes.

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

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

by driving a

(1) first order lowpass R - L filter
(2) first order highpass R - C filter
(3) tuned L - C filter
(4) series R - L - C lowpass filter

09. The parameters of the system obtained in Q88 would be:

(a) first order R - L lowpass filter would have R = 4 \Omega, L = 4 F
(b) first order R - C highpass filter would have R = 4 \Omega, C = 0.25 F
(c) tuned L - C filter would have L = 4 H, C = 4 F
(d) series R - L - C lowpass filter would have R = 1 \Omega, L = 4 H, C = 4 F

10. Noise with uniform power spectral density of N_0 W Hz is passed through a filter H(\omega) = 2 e^{-\omega/\omega_0} followed by an ideal low pass filter of bandwidth B Hz. The output noise power is Watts is.

(a) 2 N_0 B
(b) 4 N_0 B
(c) 8 N_0 B
(d) 16 N_0 B

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

\[ \int_{-\infty}^{\infty} v^2 f(v) dv \]

\[ (a) 4 \]
\[ (b) 6 \]
\[ (c) 8 \]
\[ (d) 9 \]

12. A white noise process X(t) with noise power spectral density 1 x 10^6 W Hz is input to a filter whose magnitude squared response is shown below.

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

\[ (a) 5 \times 10^7 W \]
\[ (b) 1 \times 10^8 W \]
\[ (c) 2 \times 10^8 W \]
\[ (d) 1 \times 10^9 W \]

13. If the power spectral density of stationary random process is a

\[ R(\tau) \]

The shape of its autocorrelation is

\[ (a) R(\tau) \]
\[ (b) R(\tau) \]
\[ (c) R(\tau) \]
\[ (d) R(\tau) \]

14. If the variance \( \sigma^2 \) of a signal \( y(t) \) is one-third the variance \( \sigma^2_0 \) of a stationary zero-mean discrete-time signal \( x(n) \), then the normalized autocorrelation function \( R_x(k) / \sigma^2_0 \) at

\[ (a) 0.95 \]
\[ (b) 0.90 \]
\[ (c) 0.10 \]
\[ (d) 0.05 \]

15. The PSD of a Gaussian random variable X is given by \( p(x) \) for \( x < 3 \sigma \) and \( 0 \) otherwise. The probability of the event \( X < 4 \) is

\[ (a) 0.12 \]
\[ (b) 0.14 \]
\[ (c) 0.2 \]
\[ (d) 1/4 \]

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

\[ (a) 1 - (1 - p)^n \]
\[ (b) p + (n-1) (1 - p) \]
\[ (c) n p (1 - p)^{n-1} \]
\[ (d) (1 - p)^n + n p (1 - p)^{n-1} \]

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

\[ (a) S_g(\omega) \text{ and } P_g \]
\[ (b) S_g(\omega) \text{ and } P_g \]
\[ (c) S_g(\omega) \text{ and } P_g \]
\[ (d) S_g(\omega) \text{ and } P_g \]

18. The amplitude spectrum of a Gaussian pulse is

\[ (a) \text{uniform} \]
\[ (b) \text{a time function} \]
\[ (c) \text{Gaussian} \]
\[ (d) \text{an impulse function} \]
19. The ACF of a rectangular pulse of duration $T$ is:
   a) a rectangular pulse of duration $T$
   b) a rectangular pulse of duration $2T$
   c) a triangular pulse of duration $T$
   d) a triangular pulse of duration $2T$

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

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

22. The power spectral density of a deterministic signal is given by $S(f) = 1/f^2$ where $f$ is frequency. The auto-correlation function of this signal in the time domain is:
   a) a rectangular pulse
   b) a delta function
   c) a sine pulse
   d) a triangular pulse

23. The auto-correlation function of an energy signal has:
   a) no memory
   b) conjugate symmetry
   c) odd symmetry
   d) even symmetry

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

25. Two resistors $R_1$ and $R_2$ (in ohms) at temperature $T^*$; $K_1$ and $K_2$, respectively, are connected in series. Their equivalent noise temperature is $T^* + (K_1 + K_2)$.

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

27. Zero mean Gaussian noise of variance $N$ is applied to half wave rectifier. The mean squared value of the rectifier output will be:
   a) $0$
   b) $2N$
   c) $\sqrt{2}$
   d) $2N$

28. In a digital communication system, transmissions of successive bits through a noisy channel are assumed to be independent events with error probability $\nu$. The probability of an error in the transmission of an $M$ bit sequence is:
   a) $(1-\nu)^M$
   b) $1 - (1-\nu)^M$
   c) $(1-\nu)^M + (1-\nu)^{M-1}$
   d) $(1-\nu)^{M-1} + (1-\nu)^M$

29. The variance of a random variable $X$ is $\sigma^2$. Then the variance of $-kX$ (where $k$ is a positive constant) is:
   a) $\sigma^2$
   b) $k\sigma^2$
   c) $\sigma^2 + k^2 \sigma^2$
   d) $\sigma^2 / k$
Bandwidth of the A.M signal = 2 W
= 2 × message bandwidth, Bandwidth of USB = W
= Bandwidth of LSB = W

Single tone Modulation of A.M:

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

\[ m(t) = A_m \cos(2\pi f_m t) \]

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

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

\[ s(t) = A_c \cos(2\pi f_c t) + A_m \times \cos(2\pi f_m t) \cdot \cos(2\pi f_c t) \]

and the amplitude of the carrier after modulation is

\[ A_c = A_c \times [1 + \mu \cos(2\pi f_m t)] \],

the maximum value of the positive envelope is

\[ A_c \times [1 + \mu] \],

the minimum value of the positive envelope is

\[ A_c \times [1 - \mu] \].

\[ s(t) = A_c \times \cos(2\pi f_c t + \Delta_c) \cos(2\pi f_m t + \Delta_m) + A_m \times \mu \cos(2\pi f_m t + \Delta_m) \]

carrier   USB   LSB

Power calculations of A.M:

\[ P = \frac{1}{2} \frac{A_m^2}{R} \]

Assuming AM signal is voltage signal, \( V_{ma} = \frac{A_m}{\sqrt{2}} \)

\[ P_{e_m} = \frac{1}{2} \frac{A_m^2}{R} \]

\[ P_{m+m\mu} = \frac{1}{2} \frac{A_m^2 \mu^2}{R} \]

\[ P_{c+m\mu} = \frac{A_c^2 \mu^2}{R} \]

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

\[ \text{Carrier Power} = \frac{A_c^2}{2R} \]

\[ \text{Sidesbands (USB & LSB) Power} \]

\[ \text{Modulation Efficiency} = \frac{\text{Power in the sidesbands}}{\text{Total Power}} \]

\[ \eta = \frac{P_{c+m\mu}}{P_{c+m\mu} + P_{m+m\mu}} = \frac{\mu^2}{2 + \mu^2} \]

B.W. = 2f_m = 2 × Highest frequency of message signal
Multi-tone Modulation:

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

For 2-tone modulation:
\[ m(t) = A_m \cos 2\pi f_{m1} t + A_m \cos 2\pi f_{m2} t \quad (f_{m1} > f_{m2}) \]
\[ s(t) = A_m [1 + K_m m(t)] \cos 2\pi f_0 t \]
\[ = A_m \cos 2\pi f_0 t + A_m K_m \cos 2\pi f_{m1} t + A_m \cos 2\pi f_{m2} t \]
\[ = A_m \left[ \cos 2\pi f_0 t + \mu_1 \cos 2\pi f_{m1} t + \mu_2 \cos 2\pi f_{m2} t \right] \cos 2\pi f_0 t \]
\[ \therefore s(t) = A_m \left[ \cos 2\pi f_0 t + \mu_1 \cos 2\pi f_{m1} t + \mu_2 \cos 2\pi f_{m2} t \right] \cos 2\pi f_0 t \]
\[ = A_m \left[ \cos 2\pi f_0 t + \mu_1 \frac{1}{2} \cos 2\pi (f_{m1} - f_0) t + \mu_2 \frac{1}{2} \cos 2\pi (f_{m2} - f_0) t \right] \]
\[ K_m A_m = \mu_1, K_m A_m = \mu_2 \]

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

\[ \eta = \frac{\mu^2}{2 + \mu^2} \]
\[ \text{Percentage power in sidebands} = \frac{(0.5)^2}{2 + (0.5)^2} \times 100\% = 11.11\% \]

**Generation of AM Signal:**

- **Square Law Modulator:**
  \[ m(t) \rightarrow \text{Non-linear Device} \rightarrow \text{BPF} \rightarrow \text{AM Signal} \]
  \[ A_m \cos 2\pi f_0 t \]
\[ v_0(t) = m(t) + \cos 2\pi f_1 t \]
\[ v_0(t) = K_2 v_0(t) + K_1 v_1(t) \]

If \( K_1 = 0 \), it becomes a square law device.

\[ v_0(t) = K_1 m(t) + K_1 A_0 \cos 2\pi f_1 t + K_1 m(t) + K_2 A_0^2 \cos^2 2\pi f_1 t + K_1 A_0^2 \cos 2\pi f_1 t \]

The other terms are eliminated by the BPF with centre frequency \( f_1 \) and \( BW=2W \).

Amplitude Sensitivity \( K_a = \frac{2K_2}{K_1} \)

Switching Modulator:

Demodulation of AM signals:

(i) Square Law Demodulator (Detector):

[Diagram of demodulator process]

\[ V_1 = A_0 \cdot [1 + K_0 m(t)] \cos (2\pi f_1 t) \]

\[ V_2 = K_2 A_0^2 \cos 2\pi f_1 t + K_2 A_0^2 \cos^2 2\pi f_1 t + K_2 A_0^2 \cos 2\pi f_1 t \]

\[ \frac{V_2}{2} + \frac{2K_2 A_0^2 K_0 m(t)}{2} = \frac{K_2 A_0^2 K_0^2 m(t)}{2} \]

The ratio between the wanted component to unwanted component is

\[ \frac{2K_2 A_0^2 m(t)}{K_0 m(t)} = \frac{2}{K_0 m(t)} \]
The ratio should be as high as possible.

\[ \text{Ratio} = \frac{Z}{K_a A_0 \cos 2\pi f_1}, \text{ for single tone modulation} \]

\[ \text{Ratio (Min.)} = \frac{Z}{\mu} \quad (\text{Ratio is high only when } \mu \ll 1) \]

(2) Envelope Detector:

**AM wave**

\[ v_0(t) = A_0 \cos 2\pi f_1 t \]

Charging time constant \( R_C \) should be very very few. Discharging time constant \( R_C C \) should be very very large.

For an input of \( A_0 (1 + \mu) \cos 2\pi f_1 t \), the output of envelope detector is \( A_0 + A_0 \mu A_0 \). The envelope of the input must be positive to get the exact message signal.

\[ \text{If } \mu > 1 \]

\[ A_0 (1 + \mu) \]

\[ A_0 (1 - \mu) \]

\[ \text{If } \mu < 1 \]

\[ A_0 \cos 2\pi f_1 t \]

\[ A_0 \cos 2\pi f_1 t + A_0 \sin 2\pi f_1 t \]

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

**Synchronoscope Detector**:

\[ v_0(t) = A_0 \cos 2\pi f_1 t + A_0 \sin 2\pi f_1 t \]

\[ A_0 \cos 2\pi f_1 t \]

\[ \text{Oscillator} \]

\[ A_0 \cos 2\pi f_1 t \]

\[ A_0 \cos 2\pi f_1 t + A_0 \sin 2\pi f_1 t \]
DSBSC Modulation

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

\[ s(t) = A_c m(t) \cos 2\pi f_c t + A_c \cos 2\pi f_c t \]  \hspace{1cm} \text{[Time domain Equation of AM]}

\[ s(t) = A_c m(t) \cos 2\pi f_c t \] \hspace{1cm} \text{[Time domain Equation of DSB, carrier is suppressed]}

\[ B(f) = \frac{A_c}{2} \left[ M(f - f_c) + M(f + f_c) \right] \] \hspace{1cm} \text{[Frequency domain Equation]}

\[ B.W. = 2W = 2 \left( \text{Highest frequency component of the message} \right) \]

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

Single-tone modulation of DSB-SC:

\[ m(t) = A_m \cos 2\pi f_t t \]

\[ s(t) = A_c A_m \cos 2\pi f_c t \cos 2\pi f_t t \]

\[ = \frac{A_c A_m}{2} \left[ \cos 2\pi (f_c - f_t) t + \cos 2\pi (f_c + f_t) t \right] \]

\[ B.W. = 2f_t \]

\[ P_{\text{DSB}} = \frac{A_c^2 A_m^2}{8} \quad \frac{P_t}{4} \quad \eta = 1 \]

% Power saving = \frac{P_t}{4} \times 100

Generation of DSB-SC signals:

(1) Balanced modulator

(2) Ring modulator

Balanced modulator:

\[ m(t) \]

\[ \text{Carrier} \]

\[ \text{OP} \]

\[ \text{DSBSC signal} \]

\[ h(t) = A_c \left[ 1 + K_m m(t) \right] \cos 2\pi f_c t - A_c \left[ 1 - K_m m(t) \right] \cos 2\pi f_t t \]

\[ = 2 A_c K_m m(t) \cos 2\pi f_c t \]

\[ = 2 K_m m(t) \]

Ring modulator:

\[ h(t) \]

\[ \text{Carrier} \]

\[ \text{OP} \]

\[ m(t) \cdot c(t) \]
By passing the output of Ring modulator $s_o(t)$ through a BPF with centre frequency $f_c$ and bandwidth $2W$, we can get DSB signal, $v(t) = 4x m(t) \cos (2 \pi f_c t)$.

Demodulation of DSB signals:

Coherent Detection:

\[
\begin{align*}
\text{DSB signal} & \xrightarrow{\text{Multiplier}} v_1 \xrightarrow{\text{LPF}} v_2 \xrightarrow{\text{Oscillator}} s(t) \\
& = A_s \cos \left(2 \pi f_s t + \phi \right) \\
v(t) & = A_s^2 \cos^2 \left(2 \pi f_s t + \phi \right) m(t)
\end{align*}
\]
Single Sideband (SSB) Modulation

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

The general equation is:

\[ s(t) = \frac{A_m}{2} \left( m(t) \cos 2\pi f_c t + \sin 2\pi f_c t \right) \]

\[ P_t = \frac{A_m^2}{8} \]

Power saving \[ \frac{P_s + P_{	ext{USB}}}{P_s} = \frac{P_s + P_{	ext{USB}}}{P_s + P_{	ext{LSB}}} \]

Generation of SSB signals:

1. Frequency Discrimination Method
2. Phase Discrimination Method

Frequency discrimination method:

\[ m(t) \rightarrow \text{Product Modulator} \rightarrow \text{BPF} \rightarrow \text{SSB Signal} \]

The output of PM is a DSB signal. If no DSB signal is passed through a bandpass filter, the upper sideband or lower sideband is suppressed.

If the passband is from \( f_c \) to \( f_c + \Delta f \), we will get the USB.

Phase Discrimination method:

\[ s(t) = \frac{A_m}{2} \left( m(t) \cos 2\pi f_c t + \sin 2\pi f_c t \right) \]

\[ m(t) \rightarrow \text{Product Modulator} \rightarrow \text{Oscillator} \rightarrow \text{SSB (USB / LSB)} \]
Demodulation of SSB signals:

\[ s(t) = \frac{A_m}{2} \cos 2\pi f_c t \left( \cos \theta m(t) + \frac{A_m}{2} \sin \theta s(t) \right) \]

\[ v_c(t) = \frac{A_m^2}{4} \left( 1 + \cos 4\pi f_c t \right) m(t) + \frac{A_m^2}{4} \sin 4\pi f_c t \]

Consider the locally generated signal as \( A_m \cos (2\pi f_c t + \phi) \).

\[ v_c(t) = \frac{A_m^2}{4} \left( \cos \phi m(t) + \frac{A_m^2}{4} \sin \phi s(t) \right) \]

\( \phi = 0^\circ \), \( v_c(t) = \frac{A_m^2}{4} m(t) \)

\( \phi = 90^\circ \), \( v_c(t) = (A_m^2 \cos \phi \hat{m}(t)) \)

So no Quadrature Null effect in the case of SSB, which is a major advantage over DSB.

Vestigial sideband (VSB) modulation

This is mainly used for the transmission of video signals.

1. Video signal has significant low frequency components.
2. Highest frequency varies from 2 MHz to 6 MHz.
Noise affects the video signal especially at low frequencies.

Power required to transmit a VSB signal is same as OSB (ideall).

Generation of VSB signal:

\[ v(t) = A_v \cdot m(t) \cos 2\pi f_v t \]
\[ w(t) = v(t) \otimes h(t) \]
\[ S(f) = V(f) \cdot H(f) \]
\[ S(f) = \frac{A_v}{2} \left[ M(f - f_c) + M(f + f_c) \right] H(f) \]
\[ S(f + f_c) = \frac{A_v}{2} \left[ M(f + 2f_c) + M(f) \right] H(f + f_c) \]

Demodulation of VSB signal:

The output of PM is \( A_v \cdot m(t) \cos 2\pi f_v t \).

\[ A_{PM} = \frac{A_v}{2} \left[ S(f - f_c) + S(f + f_c) \right] \]
\[ = \frac{A_v^2}{4} \left[ M(f - 2f_c) + M(f) \cdot H(f - f_c) + M(f + 2f_c) + M(f) \cdot H(f + f_c) \right] \]

The output of LPF is

\[ = \frac{A_v^2}{4} \left[ M(f) \cdot H(f - f_c) + M(f) \cdot H(f + f_c) \right] \]
\[ = \frac{A_v^2}{4} M(f) \left[ H(f - f_c) + H(f + f_c) \right] \]
To get the exact message signal $H(f - f_c) + H(f + f_c)$ should be a constant ($K_0$).

Output of LPF $= \frac{A_1}{4} K \cdot m(t)$

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

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

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

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

$$v_0(t) = \frac{m(t)}{2} \left[ \cos 2\pi (f_c + f_c) t \cdot \cos 2\pi (f_c - f_c) t \right]$$

When the carrier frequency is increased, it is called upconversion.
If the carrier frequency is decreased, it is said to be downconverted.

\[ f_c > f_c' \]

\[ \text{down conversion} \]

Frequency Division Multiplexing (FDM):

\[ m(t) \]

\[ \text{LPF} \]

\[ f_c \]

\[ \text{Modulator} \]

\[ \text{channel} \]

\[ m(t) \]

\[ \text{LPF} \]

\[ f_c' \]

\[ \text{Modulator} \]

\[ m(t) \]

\[ \text{LPF} \]

\[ f_c'' \]

\[ \text{Modulator} \]

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

To avoid interference,

\[ f_c' > f_c + 20 \text{ KHz} \]

\[ f_c'' > f_c + 20 \text{ KHz} \]

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

\[ \text{B.W.} = 64 \text{ KHz} \]

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

\[ \text{(USB is transmitted)} \]

\[ f \]

FDM Receiver:

\[ \text{Tuning} \]

\[ \text{Carrier} \]

\[ m(t) \]

By tuning we can change the received frequency to carrier frequency of the required channel.
(6) Angle Modulation:

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

1. Frequency Modulation
2. Phase Modulation

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

\[ \phi(t) = \theta(t) + 2\pi f_m t \]
\[ \theta(t) = K_p m(t) \]

\( K_p \) = Phase sensitivity of the modulator (radians/volt)
For single tone modulation

\[ \phi(t) = K_p A_m \cos 2\pi f_m t \]
\[ \theta_{m\text{ax}} = K_p A_m \] = Phase Deviation

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

\[ f_c = \omega_c + K_f m(t) \]
\[ K_f = Frequency \ sensitivity \ (\text{Hz/Volt}) \]
For single tone modulation

\[ f_c = \omega_c + K_f A_m \cos 2\pi f_m t \]
\[ f_{c\text{max}} = \omega_c + K_f A_m \]
\[ f_{c\text{min}} = \omega_c - K_f A_m \]

\( \Delta f = K_f A_m \) = Frequency deviation

Carrier Swing or Total variation of carrier frequency = 2 \( \Delta f \)
Modulation index of FM is \( \beta = \frac{K_v A_v}{f_0} \frac{A_f}{f_0} \)

\( \beta \ll 1 \), Narrow band FM

\( \beta \gg 1 \), Wide band FM

Narrow Band FM:

\[
\begin{align*}
q(t) &= A_v \cos(2\pi f_0 t + \beta \sin 2\pi f_a t) \\
&= A_v \left[ \cos 2\pi f_0 t \cos(\beta \sin 2\pi f_a t) - \sin 2\pi f_0 t \sin(\beta \sin 2\pi f_a t) \right] \\
&\approx A_v \cos 2\pi f_0 t \left( 1 - \frac{A_v}{2} \beta \sin 2\pi f_a t \right), \text{ for } \beta \ll 1 \\
&\approx A_v \cos 2\pi f_0 t - \frac{A_v}{2} \beta \sin 2\pi f_a t \\
\end{align*}
\]

Spectrum of NBFM:

\[
\begin{align*}
S(0) &= A_v \cos 2\pi f_0 t + \frac{A_v}{2} \beta \sin 2\pi f_a t \\
&\approx A_v \cos 2\pi f_0 t + \frac{A_v}{2} \beta \sin 2\pi f_a t \\
&\approx A_v \cos 2\pi f_0 t - \frac{A_v}{4} \beta \sin 2\pi f_a t \\
\end{align*}
\]

B.W of NBFM = 2f_a.

The spectrum of AM and FM are identical except that the spectral component at \( f_0 - f_a \) is 180° out of phase.

\[ P_t = P_i \left( 1 + \beta^2 / 2 \right) \]

Wideband FM:

\[
q(t) = A_v \cos \left( 2\pi f_0 t + \beta \sin 2\pi f_a t \right)
\]

Bessel function of order 'n' is given by

\[ J_n(x) = \left( \frac{1}{2} \right)^{n/2} \frac{x^n}{\Gamma(n+1/2)} \]

Properties:

\[ J_n(x) = (-1)^n J_n(-x) \]

\[
J_n(x) = \sum_{k=0}^{\infty} \frac{(-1)^k}{(k!)^2} \left( \frac{x}{2} \right)^{2k+n}
\]

\[ J_n(x) = \frac{\sin(x) - x J_{n+1}(x)}{\cos(x) - x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^n} \frac{\pi}{\sin(x) + x J_{n+1}(x)} \]

\[ J_n(x) = \frac{(-1)^n}{x^{n+1}} \frac{\pi}{\cos(x) + x J_{n+1}(x)} \]
Characteristics of a WBFM signal:

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

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

2. The amplitudes of the spectral components depend on the Bessel function coefficients \( J_n(\beta) \) which decrease as \( n \) increases. So the amplitudes of the spectral components also decrease on both sides of the carrier.

3. In WBFM spectrum amplitude of carrier component depends on \( I_0(\beta) \) and hence on modulation index \( \beta \).

\[ s(t) = A_c J_0(\beta) \cos(2\pi f_c t) + A_c J_1(\beta) \cos(2\pi f_m t) + A_c J_2(\beta) \cos(2\pi (1 + 2) f_m t) + \cdots \]

The carrier component \( I_0(\beta) \) is zero when \( \beta = 2, 4, 5.5, 8.6, 11.8, \ldots \).

For these values of \( \beta \), the amplitude of the carrier component in the spectrum is zero and the modulation efficiency is 1.

\[ P_c = P_{	ext{carrier}} = \frac{A_c^2 J_0^2(\beta)}{2} \]

\[ P_{s, \text{sidebands}} = \frac{A_c^2 J_1^2(\beta)}{2} \]

\[ P_{s, \text{sidebands}} = \frac{A_c^2 J_2^2(\beta)}{2} \]

\[ P_{s, \text{sidebands}} = \frac{A_c^2 J_3^2(\beta)}{2} \]

Theoretical bandwidth of a WBFM is \( \pi f_m \).
Communication Systems

**Total Power**

\[
\text{Total Power} = \sum \frac{A_i^2}{2} \left( 1 + \frac{A_i^2}{2} \left( 2 - \sum \frac{A_i^2}{2} \right) \right)
\]

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

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

Theoretical bandwidth of FM is \(2\beta + 2\Delta f\) practically the signal is band limited using Carson's rule by passing the FM signal through BPF. Power spectral density gives how the power is distributed among different frequency components. Carson stated that only certain frequencies will have significant amplitudes.

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

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

\[
\text{B} \text{.W} = 2(\beta + 1)\Delta f
\]

Example: An angle-modulated signal with carrier frequency \(\omega_c = 2\pi \times 10^3\) is described by

\[
\phi(t) = 10 \cos(\omega_c t + 5 \sin 3000t + 10 \sin 2000\pi t)
\]

(a) Find the power of the modulated signal.
(b) Find the frequency deviation.
(c) Find the deviation ratio \(\beta\).
(d) Find the phase deviation \(\Delta\phi\).
(e) Estimate the bandwidth of \(\phi(t)\).

**ACE Academy**

**Analog Communication Systems**

Sad: The signal bandwidth is the highest frequency in \(\phi(t)\) (or its derivative). In this case

\[
\text{B} = 2000\pi / 2\pi = 1000 \text{ Hz}
\]

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

\[
P = 10^2 / 2 = 50
\]

(b) To find the frequency deviation \(\Delta f\), find the instantaneous frequency \(\omega(t)\) given by

\[
\omega(t) = \omega_c + 15000 \cos 3000t + 20000t \cos 2000\pi t
\]

The carrier deviation is \(15000 \cos 3000t + 20000t \cos 2000\pi t\). The sinusoids will add in phase at some point, and the maximum value of this expression is \(15000 + 20000\). This is the maximum carrier deviation \(\Delta\omega\). Hence,

\[
\Delta f = \frac{\Delta\omega}{2\pi} = 12.38732 \text{ Hz}
\]

\[
\beta = \frac{\Delta f}{\Delta\omega} = \frac{12.38732}{1000} = 12.387
\]

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

(e) \(B_{\text{FM}} = 2(\Delta f + \beta) = 26.77465 \text{ Hz}\)

**Generation of WBFM signal**

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

**Direct Method**

A voltage-controlled oscillator is used to generate FM signal.
For a Hartley oscillator,

\[ f = \frac{1}{2 \sqrt{L_1 + L_2}} \frac{1}{C} \]

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

C1 is a variable capacitor and depends upon m(t)

Indirect Method:

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

Demodulation of FM signals:

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

In Fig.(a), the tuned circuit (L1 and C3) produces an output voltage that is proportional to the input frequency. The maximum output voltage occurs in the resonant frequency of the tank circuit (L), and its operation decreases proportionately as the input frequency deviates above or below L. The circuit is designed so that the IF center frequency (f0) falls in the center of the most linear portion of the voltage-versus-frequency curve, as shown in Fig.(b). When the intermediate frequency deviates above f0, the output voltage increases; when the intermediate frequency deviates below f0, the output voltage decreases. Therefore, the tuned circuit converts frequency variations to amplitude variations (FM to AM conversion).

D1, C2, and R1 make a simple peak detector that converts the amplitude variations to an output voltage that varies at a rate equal to that of the input frequency changes and whose amplitude is proportional to the magnitude of the frequency changes.

Balanced Slope Detector:

The figure shown below is the schematic diagram for a balanced slope detector.

A single-ended slope detector is a tuned-circuit frequency discriminator, and a balanced slope detector is simply two single-ended slope detectors connected in parallel and fed HIF out of phase. The phase inversion is accomplished by center tapping the tuned secondary windings of transformer T1.

In Fig.(a), the tuned circuits (L1, C1, and L2, C2) perform the FM-to-AM conversion, and the balanced peak detectors (D1, C4, R1, and D2, C5, R2) remove the information from AM envelope. The top tuned circuit (L1 and C2) is tuned to a frequency (f1) that is above the IF center frequency (f0) by approximately 1.33 + Δf (for the FM broadcast and this is approximately 1.33 + 7.5 kHz = 100 kHz). The lower tuned circuit (L2 and C1) is tuned to a frequency (f2) that is below the IF center frequency by an equal amount.

The output voltage from each tuned circuit is proportional to the input frequency, and each output is rectified by its respective peak detector. Therefore, the closer the input frequency is to the tank-circuit resonant frequency, the greater the tank-circuit output voltage. The IF center frequency falls exactly halfway between the resonant frequencies of the two tuned circuits. Therefore, at the IF center frequency, the output voltages from the two tuned circuits are equal in amplitude but oppose in polarity. Consequently, the rectified output voltage across R1 and R2, when added, produces a differential output voltage Vout = 0 V. When the IF deviates above resonance, the top tuned circuit produces a higher output voltage than the
lows, a tank circuit and \( V_{ma} \) goes positive. When the IF deviates below resonance, the output voltage from the lower tank circuit is larger than the output voltage from the upper tank circuit and \( V_{ma} \) goes negative. The output versus frequency response curve is shown in Fig. 3(b).

The slope detector is simplest FM detector. It has several disadvantages, which include poor linearity, difficulty in tuning, and lack of provisions for limiting.

**Foster-Seeley Discriminator:**

A Foster-Seeley discriminator (phase shift discriminators) is a well-distributed frequency discriminator whose operation is very similar to that of the balanced slope detector. The schematic diagram for a Foster-Seeley discriminator is shown in Fig. 4(b).

The capacitance values for \( C_0, C_1, \) and \( C_2 \) are chosen such that they are short circuits for the IF center frequency. Therefore, the output of \( L_1 \) is at ground potential, and the IF signal \( V_{ma} \) is fed directly into the circuit across \( L_1 (V_{ma}) \). The remaining IF is inverted \( 180^\circ \) by transformer \( T_1 \) and divided equally between \( L_2 \) and \( L_3 \). At the resonant frequency of the secondary tank circuit (the IF center frequency), the secondary current \( I_2 \) is in phase with the total secondary voltage \( V_{ma} \) and \( 180^\circ \) out of phase with \( V_{ma} \). Also, due to base coupling, the primary of \( T_1 \) acts as an inductor, and the primary current \( i_p \) is 90° out of phase with \( V_{ma} \) and, because of the inductance. The voltage across the top diode \( V_{D1} \) is the vector sum of \( V_{ma} \) and \( V_{ma} \) and the voltage across the bottom diode \( V_{D2} \) is the vector sum of \( V_{ma} \) and \( V_{ma} \). The corresponding vector diagrams are shown in Fig. 4(b).

**Fig. 2** shows a typical voltage-versus-frequency response curve for a Foster-Seeley discriminator. It can be seen that the output voltage-versus-frequency deviation curve is more linear than that of a slope detector, and because there is only one tank circuit, it is easier to tune. For distortionless demodulation, the frequency deviation should be restricted to the linear portion of the secondary tuned circuit frequency response curve. As with the slope detector, a Foster-Seeley discriminator responds to amplitude as well as frequency variations and, therefore, must be preceded by separate linear circuits.
As with the Foster-Seely discriminator, a ratio detector has a single tuned circuit in the transformer secondary. Therefore, the operation of a ratio detector is similar to that of the Foster-Seely discriminator. In fact, the voltage vectors for D₁ and D₂ are identical. However, with the ratio detector, one diode is reversed (D₃), and current iₚ flows around the outermost loop of the circuit. Therefore, after several cycles of the input signal, shunt capacitor C₁ charges to approximately the peak voltage across the secondary winding of T₁. The reactance of C₁ is low, and R₆ simply provides a path for diode current. Therefore, the time constant for R₆ and C₁ is sufficiently long so that rapid changes in the amplitude of the input signal due to thermal noise or other interfering signals are attenuated and have no effect on the average voltage across C₁. Consequently, C₆ and C₁ charge and discharge proportionally to frequency changes in the input signal and are relatively immune to amplitude variations. Also, the output voltage from a ratio detector is taken with respect to ground, and the diode polarities shown in Fig. (a), the average output voltage is positive. At resonance, the output voltage is divided equally between C₆ and C₁ and distributed as the input frequency is deviated above and below resonance. Therefore, change in V_in are due to changing ratio of the voltage across C₁ and C₆, while the total voltage is clamped by C₆.

Figure below shows the output frequency response curve for the ratio detector. It can be seen that at resonance, V_in is not equal to 0 V but, rather, to one-half of the voltage across the secondary windings of T₁.

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

Phase Modulation:

\[ m(t) = A \cos \left(2\pi f_c t + \phi(t)\right) \]

\[ m(t) = A \cos \left(2\pi f_c t + \phi(t)\right) \]

For an angle-modulated wave:

\[ \phi(t) = K_m m(t) \]

\[ \phi(t) = K_m m(t) \]

For FM:

\[ m(t) = \frac{A}{2K_m} \int m(t) \, dt \]

\[ m(t) = \frac{A}{2K_m} \int m(t) \, dt \]

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

\[
\theta(t) = K_p m(t) + K_o \cos \left( 2\pi f_m t \right)
\]

For sinusoidal signal and exponential, there is no difference between FM and PM except for a phase shift of \(90^\circ\) \(\frac{d}{dt}\) or integration of sinusoidal signal is again a sinusoidal signal but with a phase shift of \(90^\circ\).

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

For single-tone modulation,

\[
\theta(t) = K_p m(t) + K_o \cos \left( 2\pi f_m t \right)
\]

\[
\Delta \beta = K_o A_o = \beta \frac{D}{m}
\]

\[
\theta(t) = A_o \cos \left( 2\pi f_m t + \beta P M \cos 2\pi f_a t \right)
\]

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

\[
BW = 2 \left( \beta m + 1 \right) f_a,
\]

\[
P_{PM} = \frac{A_o^2}{2}
\]

- \(BW\) of PM is independent of message frequency.

**ACE Academy**  
**Analog Communication Systems**  

(C) **Receivers:**

**AM Receivers:**

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

**TRF Receiver:**

![TRF Receiver Diagram]

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

Ionospheric propagation is used for medium wave. No two stations can have same carrier frequency. The received signal strength is of the order of mW or \(\mu\)W. RF amplifier must be low noise amplifier. RF amplifier itself acts as a 00PF. RF amplifier itself consists of a tuned circuit. Thus, it is called tuned RF amplifier.

![RF Amplifier Diagram]

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

**Characteristic parameters of Receiver:**

- **Sensitivity** is defined as the minimum signal that should be applied at the input of a receiver to get a standard output. Sensitivity depends on the gain of the amplifier. If the gain of the RF amplifier is high, the sensitivity is also high.
- **Selectivity** is defined as the ability of the receiver to reject unwanted components (or) Sensitivity is defined as the ability of the receiver to select wanted component. The receiver is having very poor selectivity.
Fidelity is defined as the ability of the receiver to reproduce all frequencies equally in the entire tuning range.

The disadvantage of TRF receiver is poor selectivity.

Superheterodyne Receiver:

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

Intermediate frequency for MW is 455 KHz.

Image frequency:

Frequency before the mixer stage.

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

Image (Frequency) Rejection Ratio:

\[
\text{IRR} = \frac{\text{Gain at } f_1}{\text{Gain at } f_2} \quad \text{Gain at } f_2 \ll 1
\]

By increasing the intermediate frequency, IRR can be increased. By increasing the bandwidth, the gain at \( f_2 \) can be decreased so that IRR increases.

\[
\text{IRR} = \frac{1}{B \cdot W}
\]

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

\[
\text{IRR} = Q \cdot \frac{1 + Q \cdot \rho^2}{1 + Q \cdot \rho^2 + \frac{1}{1 + Q \cdot \rho^2}}
\]

where \( \rho = \frac{f_2}{f_1} \)

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

\[
\text{IRR} = \sqrt{1 + Q \cdot \rho^2} \cdot \frac{1}{1 + Q \cdot \rho^2 + \frac{1}{1 + Q \cdot \rho^2}}
\]

RF amplifier:

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

Mixer: The purpose of the mixer is to down convert the incoming signal. The output of the mixer is always equal to the difference of the input frequencies. If \( f_1 \) is the incoming signal frequency and \( f_2 \) is the local oscillator frequency, the output of the mixer is equal to \( f_1 - f_2 \). Always the local oscillator frequency should be greater than the signal frequency. If the receiver is tuned to \( f_1 \), the local oscillator frequency should be adjusted so that the output of the mixer is equal to IF.

IF amplifier:

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

\[
f_0 - f_c = \text{IF}
\]
Choice of IF:
01. IF is too high poor selectivity and poor adjacent channel rejection.
02. A high value of IF increases tracking difficulties.
03. IF is very low image frequency rejection becomes poorer.
04. If the IF is very low, the frequency stability of the local oscillator should be very high.
05. The IF must not fall within the tuning range of the receiver.

Tracking of a Receiver:

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

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

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

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

Case 1) \( f_l - f_i = IF \)

Case 2) \( f_l - f_i = 450 \)

Tracking error can be minimized using Paddler capacitor & Trimmer circuit.

\[
\frac{1}{2\pi \sqrt{(L_1 + L_2)C}}
\]

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

\[
\frac{1}{2\pi \sqrt{(L_1 + L_2)(E + C_p)}}
\]

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

\[
C_{eq} = \frac{C_1 \cdot C_2}{C_1 + C_2}
\]

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

\[
\begin{align*}
S & = 455 \text{ kHz} \\
\Delta f & = 1465 \text{ kHz} \\
\Delta f & = \frac{\sqrt{1 + Q^2 \rho^2}}{2} \\
\end{align*}
\]

Where \( \rho = 1465 \), \( Q = 555 \), \( Q = 2.2608 \)

\[
\begin{align*}
\text{IRR} & = 113.04
\end{align*}
\]

Automatic Gain Control (AGC):
The purpose of AGC circuits in a receiver is to maintain a constant output irrespective of variations in the input signal strength. If the input signal strength increases the gain of the receiver is reduced or if the input signal strength decreases the gain of the receiver is increased so that the output level is constant.
The following list gives some of the problems that would be encountered in a receiver without this provision:

01. Tuning the receiver would be nightmare. So as to not miss the weak stations, we should have the volume control (in the non-AGC set).
02. The received signal from any given station is constantly changing as a result of changing weather and ionospheric conditions. The AGC allows to listen to a station without constantly monitoring the volume control.
Communication Systems

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

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

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

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

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

Types of AGC:
1. Simple AGC
2. Delayed AGC

Simple AGC:

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

- Strong station
  - Receiver
  - Large output
- Weak station
  - Receiver
  - Small output

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

Delayed AGC:

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

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

The characteristics of AGC are shown below.

![AGC Characteristics Diagram]

- Simple AGC
- Delayed AGC
- No AGC (no gain reduction)

Squelch circuit:

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

Double Spacing:

1600 KHz station is selected when the local oscillator frequency is 1100 KHz as well as 2100 KHz for an IF of 500 KHz. So each station is selected twice on the frequency scale. This is called double spacing.
(D) Noise in Analog Modulation

Noise Analysis of AM:

\[ \frac{S_i}{N_i} \text{ Receiver } \frac{S_o}{N_o} \]

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

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

\[ P = K T_e B \quad ; \quad K = \text{PSD of thermal noise} \]
\[ T_e = \text{Noise equivalent temperature of the receiver} \]
\[ F = \text{Noise figure} \]
\[ K T_e = \frac{N_i}{2} = \text{PSD of a white noise} = \eta \]
\[ \frac{N_o}{2} = \text{PSD} \rightarrow f \]
\[ \text{Total power is uniformly distributed to all frequencies.} \]

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

Central Limit Theorem:

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

\[ Z = X + Y \]

\[ f_z(x) = f_x(x) \ast f_y(x) \]

Uniform \rightarrow Rayleigh

\[ X_t \quad Y \quad X_t \quad Y \quad \text{Receiver} \]

\[ \text{Transmitter} \quad \text{Rayleigh} \]

\[ (S \text{ / } N_i) = \frac{S_i}{N_i} \quad \text{Power of the modulated signal} \]
\[ (S \text{ / } N_o) = \frac{S_o}{N_o} \quad \text{Power of noise in message bandwidth} \]
\[ (S \text{ / } N_{noise}) = \frac{S_o}{N_{noise}} \quad \text{Power of noise in message bandwidth} \]

\[ \text{Figure of Merit} = \frac{(S \text{ / } N_i)}{(S \text{ / } N_o)} = \frac{1}{(S \text{ / } N_{noise})} \]
\[ \text{Noise Figure} = \frac{(S \text{ / } N_o)}{(S \text{ / } N_{noise})} \]

\[ (S \text{ / } N_o) \text{ depends mainly on modulation scheme and receiver characteristics.} \]

\[ \text{Figure of Merit of a DSB system:} \]

\[ S_i = \left( \frac{1}{2} \right)^{\frac{1}{2}} = \frac{A_v^2}{4} \text{, where } A_v^2 = \text{power in the message signal} = P \]
For a single tone modulation

\[ P = \frac{A_m^2}{2} \]

\[ S_n = \frac{A_m^2}{4} \]

\[ N_0 = W N_0 \]

\[ (S/N)_0 = \frac{A_m^2}{2 W N_0} \]

Input to the PM is DSB signal + Narrow band noise

\[ = A_m m(t) \cos 2\pi f_1 t + n(t) \cos 2\pi f_1 t - n(t) \sin 2\pi f_1 t \]

Output of the PM

\[ = A_m m(t) \cos^2 2\pi f_1 t + n(t) \cos^2 2\pi f_1 t - n(t) \cos 2\pi f_1 t \sin 2\pi f_1 t \]

Output of the LPF

\[ S_0 = \frac{A_m m(t)}{2} + \frac{n(t)}{2} \]

\[ (S/N)_0 = \frac{A_m^2}{2 W N_0} \]

\[ N_0 = \frac{(1/4) [ \text{In phase component noise power} ]}{2} \]

\[ = \frac{(1/4) [ 2 W N_0 ]}{4} = \frac{W N_0}{4} \]

\[ (S/N)_0 = \frac{A_m^2}{4 W N_0} \]

\[ \text{Figure of merit} = \frac{A_m^2}{2 W N_0} = 1 \]

Figure of Merit of a SSB system:

\[ S_n = \frac{A_m^2}{4} \]

\[ N_0 = \frac{2 W N_0}{2} = W N_0 \]

\[ (S/N)_0 = \frac{A_m^2}{4 W N_0} \]

Input to the PM = SSB + Narrow band Noise

\[ = [A_m m(t) \cos 2\pi f_1 t + A_m n(t) \sin 2\pi f_1 t] + n(t) \cos 2\pi f_1 t - n(t) \sin 2\pi f_1 t \]

Output of LPF

\[ S_0 = \frac{A_m m(t)}{4} + \frac{n(t)}{2} \]

\[ S_n = \frac{A_m^2}{16} \]

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

\[ (S/N)_0 = \frac{A_m^2}{4 N_0 W} \]

\[ \text{Figure of merit} = \frac{A_m^2}{4 N_0 W} = 1 \]

Noise in AM:

\[ S(t)_{AM} = A_m \cos 2\pi f_1 t + A_m K_m m(t) \cos 2\pi f_1 t \]

\[ S_0 = \frac{A_m^2}{2} + \frac{A_m^2 K_m^2}{2} \]

\[ N_0 = \frac{2 W N_0}{2} = W N_0 \]

\[ (S/N)_0 = \frac{A_m^2 (1 + K_m^2)}{2 W N_0} \]

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

\[ = A_m \cos 2\pi f_1 t + A_m K_m m(t) \cos 2\pi f_1 t + n(t) \cos 2\pi f_1 t - n(t) \sin 2\pi f_1 t \]

Output of Envelope detector

\[ = \sqrt{\left( A_m + A_m K_m m(t) + n(t) \right)^2 + \left( n(t) \right)^2} \]

\[ = A_m + A_m K_m m(t) + n(t) \]
Chapter – 3

Objective Questions Set - A

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

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

03. The modulation index of an AM wave is changed from 0 to 1. The transmitted power is
   a) unchanged b) halved c) doubled d) increased by 100 percent

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

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

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

07. The most suitable method for detecting a modulated signal (2.5 × 10^3 cos 5t) is
    a) envelope detector b) synchronous detector
c) ratio detector d) both a and b

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

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

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

11. A plot of modulation index versus carrier amplitude yields a
    a) horizontal line b) vertical line
c) parabola d) hyperbola
12. A carrier is amplitude modulated to a depth of 40%. The increase in power is a) 10% b) 20% c) 30% d) 40%

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

14. An AM wave is given by $e_{am} = 10^2 \cdot (1 + 0.4 \cos 10^5 t + 0.3 \cos 10^6 t \cos 10^7 t)$. The modulation index is a) 0.4 b) 0.5 c) 0.3 d) 0.9

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

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

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

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

19. In an AM wave, the total power content is 600 W and that of each sideband is 75 W. The modulation index is a) 55.5% b) 60.7% c) 81.6% d) 40.3%

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

21. The AM broadcast band is given by a) 10 kHz to 30 kHz b) 50 kHz to 1500 kHz c) 3 MHz to 30 MHz d) None

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

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

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

25. An audio signal 15 sin 2x (5000 Hz) amplitude modulates 60 sin 2x (10^5 Hz). The modulation index will be a) 20% b) 25% c) 50% d) 75%

26. In the above question, the total bandwidth required to transmit the AM wave in a) 1.5 KHz b) 3 K c) 1010 KHz d) 50 KHz

27. An AM wave is given by $e_{am} = 10^2 \cdot (1 + 0.4 \cos 10^5 t + 0.3 \cos 10^6 t \cos 10^7 t)$. The modulation index is a) 0.4 b) 0.5 c) 0.3 d) 0.9

28. An AM wave is given by $e_{am} = 10^2 \cdot (1 + 0.5 \cos 10^5 t + 0.3 \cos 10^6 t \cos 10^7 t)$. The bandwidth of the AM signal is a) 5 KHz b) 10 KHz c) 20 KHz d) 30 KHz

29. An AM wave is given by $e_{am} = 10^2 \cdot (1 + 0.4 \cos 10^5 t + 0.3 \cos 10^6 t \cos 10^7 t)$. The following frequency will not be present in the spectrum a) 1 MHz b) 2 MHz c) 1010 KHz d) 990 KHz

Key:
- 01.a  02.d  03.d  04.e  05.e  06.d  07.c  08.e  09.a  10.b  11.d  12.d
01. Indicate the false statement regarding the advantages of SSB over double sideband, full-carrier AM.
   a) More channel space is available.
   b) Transmitter circuits must be more stable, giving better reception.
   c) The signal is more noise-resistant.
   d) Much less power is required for the same signal strength.

02. When the modulation index of an AM wave is doubled, the antenna current is also doubled. The AM system being used is:
   a) Single-sideband, full carrier (SSB-SC)
   b) Single-sideband, suppressed carrier (SSB-SC)
   c) Double-sideband, full carrier (DSB-FC)
   d) Double-sideband, suppressed carrier (DSB-SC)

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

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

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

06. Indicate in which one of the following only sideband is transmitted.
   a) IIE
   b) AIE
   c) HIE
   d) CIE

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Key:
01. b  02. c  03. d  04. a  05. c  06. a  07. d  08. b  09. a  10. e  11. b  12. c  13. b  14. e  15. c  16. c  17. b  18. b  19. a  20. e
Chapter 3 - Communication Systems

Objective Questions Set - C

01. In the stabilized frequency modulator AFC system,
   a) the discriminator must have a fast time constant to prevent demodulation
   b) the higher the discriminator frequency, the better the oscillator-frequency stability
   c) the discriminator frequency must be too low, or the system will fail
   d) phase modulation is converted into AM by the equalizer circuit

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

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

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

05. An FM signal with a modulation index
   \( \frac{\text{modulation}}{\text{carrier}} \) is passed through a frequency tripler. The wave in the output of the tripler will have a modulation index of
   a) \( \frac{1}{3} \)
   b) \( \frac{1}{2} \)
   c) 3
   d) 2

06. An FM signal with a deviation \( \delta \) is passed through a mixer, and has its frequency reduced fivefold. The deviation in the output of the mixer is
   a) \( 5 \delta \)
   b) \( 10 \delta \)
   c) \( 25 \delta \)
   d) \( 60 \delta \)

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

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

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

10. Indicate which one of the following is not an advantage of FM over AM:
    a) Better noise immunity is provided
    b) Lower bandwidth is required
    c) More power is required
    d) AM is a simpler circuit

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

12. In an FM stereo multiplex transmission, the
    a) sum signal modulates the 19 kHz subcarrier
    b) difference signal modulates the 19 kHz subcarrier
    c) difference signal modulates the 38 kHz subcarrier
    d) difference signal modulates the 37 kHz subcarrier

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

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

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

16. The output \( V_o \) of the ratio detector is related to the output \( V_1 \) of similar Foster - Seeley discriminator as follows:
    a) \( V_1 = V_o \)
    b) \( V_1 = 2 V_o \)
    c) \( V_1 = 0.5 V_o \)
    d) \( V_2 = 2 V_o \)

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

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

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

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

21. A narrow-band FM does not have the following feature:
    a) It has two sidebands
    b) both sidebands are equal in amplitude
    c) input and output sidebands have same phase
    d) it does not show amplitude modulation

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

23. In a single-tone FM discriminator
    a) \( V_2/V_1 \) is
    b) proportional to deviation
    c) proportional to cube of deviation
Chapter – 3 Additional Objective Questions Set - D

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

02. A sinusoidal voltage, amplitude modulates another sinusoidal voltage of amplitude 1 kV to result in two sideband terms of amplitude 300 volts each. The modulation index is
a) 0.1  
  b) 0.2  
  c) 0.3  
  d) 0.5

07. A carrier voltage of unmodulated carrier power of 1 kW, on being amplitude modulated by an audio sinusoidal voltage to a depth of 100%, has total modulated carrier power of
a) 1255 kW  
  b) 74 kW  
  c) 744 kW  
  d) 4 kW

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

05. A sinusoidal carrier voltage of amplitude 100 volts is amplitude modulated by a sinusoidal voltage of frequency 1 kHz, resulting in maximum modulated carrier amplitude of 130 volts. The modulation index is
a) 0.05  
  b) 0.3  
  c) 0.15  
  d) 0.13

06. An amplitude modulated voltage in volts is given by
   \[ V = 20 \left( 1 + 0.5 \sin 0.02t \right) \sin (6.28 \times 10^3 t) \]. The rms value of the unmodulated carrier voltage in volts is
a) 20  
  b) 20/\sqrt{2}  
  c) 10  
  d) 10/\sqrt{2}

07. An amplitude modulated voltage in volts is given by
   \[ V = 20 \left( 1 + 0.5 \sin 0.02t \right) \sin (6.28 \times 10^3 t) \]. The rms value of the sideband voltage in volts is
a) 5 \sqrt{2}  
  b) 5  
  c) 0.7  
  d) 0.125

08. An amplitude modulated voltage in volts is given by
   \[ V = 20 \left( 1 + 0.5 \sin 0.02t \right) \sin (6.28 \times 10^3 t) \]. The percentage modulation index of the modulated voltage is
a) 25%  
  b) 50%  
  c) 50% - 50%  
  d) 100%

09. An amplitude modulated voltage in volts is given by
   \[ V = 20 \left( 1 + 0.5 \sin 0.02t \right) \sin (6.28 \times 10^3 t) \]. The modulated frequency is
a) 30 Hz  
  b) 100 Hz  
  c) 0.02 Hz  
  d) 200 Hz

10. An amplitude modulated voltage in volts is given by
   \[ V = 20 \left( 1 + 0.5 \sin 0.02t \right) \sin (6.28 \times 10^3 t) \]. The carrier frequency is
a) 2 \times 10^3 Hz  
  b) 10^3 Hz  
  c) 2 \times 10^3 Hz  
  d) 0.5 \times 10^3 Hz

11. In amplitude modulation systems, if modulation index is varied from 1 to 1.2, then
a) Power of the wave increases  
  b) Efficiency of transmission increases  
  c) Bandwidth increases  
  d) The signal gets distorted
12. An amplitude modulated voltage has modulation index of 100%. If the carrier is suppressed, the percentage power saving in
a) 50%  b) 66.6%  c) 75%  d) 25%

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

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

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

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

17. A carrier voltage of rms value 100 volts is amplitude modulated by a sinusoidal audio voltage to a modulation index of 0.2. The rms value of carrier on modulation is
a) 103 volts  b) 102 volts  c) 104 volts  d) 120 volts

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

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

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

21. A carrier is simultaneously amplitude modulated by two sine waves causing individual modulation of 20% and 40%. The overall modulation index is
a) 30%  b) 35%  c) 70%  d) 40%

22. In an AM transmitter, the unmodulated output current is i_0. The modulated current, i, equals
a) i_0  b) i_0 \times m^2  c) i_0 \times m^2  d) i_0 \times m^2/2

23. In an amplitude modulated carrier, the sideband power is 1100 watts while that of each sideband is 80 watts. The unmodulated carrier power is
a) 1000 watts  b) 1000 watts  c) 1000 watts  d) 500 watts

24. In an AM signal, the peak antenna current is 13 Amp and the lowest current is
a) 7 Amp  b) 6.5 Amp  c) 5 Amp  d) 3.5 Amp

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

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

27. An AM signal is detected using an envelope detector. The carrier frequency and modulating signal frequency are 1 MHz and 2 kHz respectively. An appropriate value for the time constant of the envelope detector at 500 μsec is
a) 20 μsec  b) 50 μsec  c) 0.2 μsec  d) 1 μsec

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

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

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

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

32. A frequency-modulated voltage with modulation index 'm' is passed through a frequency doubler. The FM signal in the output of the doubler will have modulation index of
a) 2m  b) m  c) m  d) 4m
33. An FM signal with frequency deviation of 5 kHz is passed through a mixer and has its frequency reduced three fold. The frequency deviation in the output of the mixer is
   a) 15 kHz
   b) 10 kHz
   c) 5 kHz
   d) 3 kHz

34. The bandwidth requirement of a telephone channel is
   a) 3 kHz
   b) 4 kHz
   c) 2 kHz
   d) 5 kHz

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

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

37. A 2.5 volt 500 Hz voltage frequency modulates the carrier to cause frequency deviation of 9 kHz. The modulation index is
   a) 1
   b) 0.25
   c) 0.5
   d) 1

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

39. In FM, the carrier frequency deviation, on having increased the modulating voltage to 10 volts, the modulating index becomes
   a) 1.4
   b) 4
   c) 8
   d) 32

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

41. A 2 volt, 1 kHz signal, frequency modulates a carrier voltage to cause frequency deviation of 5 kHz. If the modulating voltage is changed to 20 volts, 200 Hz, with modulating voltage 28 volts, 200 Hz, with modulating voltage 70 volts, 20 Hz,
   a) 5 kHz
   b) 50 kHz
   c) 0.5 kHz
   d) 100 kHz

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

43. A frequency modulated carrier is represented by \( v = 20 \sin (6.28 \times 10^7 t + 4 \sin 628 t) \). The carrier frequency is
   a) 6.28 \times 10^7 Hz
   b) 3.14 \times 10^7 Hz
   c) 108 Hz
   d) 2 \times 10^7 Hz

44. A frequency modulated carrier is represented by \( v = 20 \sin (6.28 \times 10^7 t + 4 \sin 628 t) \). The modulation frequency is
   a) 628 Hz
   b) 100 Hz
   c) 200 Hz
   d) 314 Hz

45. A frequency modulated carrier is represented by \( v = 20 \sin (6.28 \times 10^7 t + 4 \sin 628 t) \). The frequency deviation is
   a) 400 Hz
   b) 628 Hz
   c) 2112 Hz
   d) 314 Hz

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

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

48. In FM system, if the depth of modulation is doubled, the output power
   a) increases by factor of 4
   b) increases by factor of \(\sqrt{2}\)
   c) increases by factor of 2
   d) remains at unmodulated value

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

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

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

52. In a frequency modulated voltage, the maximum modulating frequency is 15 kHz and the maximum frequency deviation is 75 kHz. If the significant half power bandwidth extends up to \( f_0 \), the theoretical bandwidth required is
   a) 45 kHz
   b) 150 kHz
   c) 180 kHz
   d) 480 kHz

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

54. In a FM system, modulation index is 7 and the practical bandwidth is 160 kHz. The frequency deviation is
   a) 20 kHz
   b) 35 kHz
   c) 70 kHz
   d) 140 kHz
55. In a FM system, the carrier frequency is 200 MHz; maximum modulating frequency is 10 kHz and maximum frequency deviation is 1 MHz. The practical bandwidth requirement is:
   a) 1 MHz  
   b) 2 MHz  
   c) 2.5 MHz  
   d) 4 MHz

56. In the above question, if the modulating signal amplitude is doubled, the practical bandwidth required will be:
   a) 1 MHz  
   b) 2 MHz  
   c) 2.5 MHz  
   d) 4 MHz

57. In which of the following modulation systems does the increase in modulation index result in increase in bandwidth?
   a) amplitude modulation  
   b) frequency modulation  
   c) phase modulation  
   d) both frequency and phase modulations

58. Practical bandwidth of a wideband FM signal (1 < Δ < 100) with modulating frequency f_m and maximum frequency deviation Δ_f approximately equals:
   a) 2f_m  
   b) 2Δ_f  
   c) 2f_m + Δ_f  
   d) 2f_m - Δ_f

59. Practical bandwidth of a narrowband FM signal (Δ < 1) equals:
   a) f_m  
   b) 2f_m  
   c) Δ_f  
   d) 2Δ_f

60. Practical bandwidth of a very wideband FM signal approximately equals:
   a) f_m  
   b) 2f_m  
   c) Δ_f  
   d) 2Δ_f

   where f_m is the modulating frequency and Δ_f is the frequency deviation.

61. From bandwidth point of view, narrow band FM is equivalent to:
   a) AM  
   b) Phase modulation  
   c) SSB  
   d) Suppressed carrier - DSB

62. In a TV system, the modulation methods employed for video and audio signals are:
   a) both amplitude modulation  
   b) both frequency modulation  
   c) respectively amplitude modulation and frequency modulation  
   d) respectively frequency modulation and amplitude modulation

63. Two carriers 40 MHz and 80 MHz respectively are frequency modulated by a signal of frequency 4 KHz, such that the bandwidths of the FM signal in the two cases are the same. The ratio of deviation in the two cases are in the ratio of:
   a) 1:4  
   b) 1:2  
   c) 1:1  
   d) 2:1

64. Pre-emphasis in FM systems involves:
   a) compression of the modulating signal  
   b) expansion of the modulating signal  
   c) amplification of lower frequency components of the modulating signal  
   d) amplification of higher frequency components of the modulating signal

65. As the modulation index of an FM signal with sinusoidal modulation is increased from zero to three, the power in the carrier component will:
   a) increase continuously  
   b) decrease continuously  
   c) first increase, attain a maximum and then decrease  
   d) first decrease, become zero and then increase

66. An FM signal with a deviation Δ is passed through a mixer and has its frequency reduced by fivefold. The deviation in the output of the mixer is:
   a) 6Δ  
   b) intermediate  
   c) (6/5)Δ  
   d) Δ

67. Let m(t) = 5 cos(30t + sin 50t). Its instantaneous frequency (in rad/s) at t = 0 has the value:
   a) 5  
   b) 50  
   c) 55  
   d) 250

68. An FM wave uses a 2.5 V, 500 Hz modulating frequency and has a modulation index of 50. The deviation is:
   a) 500 Hz  
   b) 1000 Hz  
   c) 1250 Hz  
   d) 2500 Hz

69. In the spectrum of a FM wave:
   a) the carrier frequency cannot disappear  
   b) the carrier frequency disappears when the modulation index is large  
   c) the amplitude of any sideband depends on the modulation index  
   d) the total number of sidebands depends on the modulation index

70. In FM reception, amplitude disturbances due to static change:
   a) reduce the signal frequencies  
   b) increase the signal frequencies  
   c) disturb the signal frequencies  
   d) do not affect the signal frequencies

71. The FM modulation index is given by:
   a) Maximum frequency  
   b) Minimum frequency  
   c) Modulating frequency  
   d) Maximum frequency deviation

72. Under identical conditions FM & PM are indistinguishable for a single modulating frequency. Now if the modulating frequency is increased, the FM modulation index will remain constant, where as FM modulation index will increase.
   a) FM modulation index will remain constant, where as FM modulation index will increase  
   b) PM modulation index will remain constant, where as FM modulation index will increase  
   c) FM modulation index will decrease and FM modulation index will increase  
   d) Both FM as well as PM modulation indices will increase

73. The image channel selectivity of a superheterodyne receiver depends upon:
   a) IF amplifiers only  
   b) RF and IF amplifiers only  
   c) Parasitic, RF and IF amplifiers only  
   d) Parasitic, and IF amplifiers only

74. The image channel rejection in a superheterodyne receiver comes from:
   a) IF stages only  
   b) RF stages only  
   c) detector and RF stages only  
   d) detector, RF, and IF stage
Chapter - 5

Additional Objective Questions

1. Consider the amplitude modulated (AM) signal
\[ A_0 \cos \omega t + 1 + 2 \cos \omega t \]
with carrier frequency \( f_c = 10^6 \) Hz. For demodulating the signal using envelope detector, the minimum value of \( A_0 \) should be
(a) 2
(b) 1
(c) 0.5
(d) 0

2. Consider the frequency modulated signal
\[ 10^6 \cos(2 \pi \times 10^3 t + 5 \sin(2 \pi \times 1500 t + 7.5 \sin(2 \pi \times 1000 t)) \]
with carrier frequency of \( 10^6 \) Hz. The modulation index is
(a) 12.5
(b) 10
(c) 7.5
(d) 5

3. The signal \( \cos(\omega t + 0.5 \cos(\omega t)) \) signal is (a) FM only
(b) AM only
(c) both AM and FM
(d) neither AM nor FM

4. The diagonal clipping in Amplitude Demodulation (using envelope detector) can be avoided if RC time constant of the envelope detector satisfies the following condition. (here \( W \) is message bandwidth and \( \omega_m \) in carrier frequency both in rad/s)
(a) \( RC = \frac{1}{W} \)
(b) \( RC > \frac{1}{W} \)
(c) \( RC = \frac{1}{\omega_m} \)
(d) \( RC > \frac{1}{\omega_m} \)

5. A message signal with bandwidth 10kHz is to be transmitted with SSB modulation with carrier frequency \( f_c = 10^6 \) Hz. The resulting signal is then passed through a Bandpass filter with bandwidth of 10kHz. The output would be
(a) \( 2 \times 10^6 \) Hz
(b) \( 2 \times 10^6 \) Hz
(c) \( 2 \times 10^6 \) Hz
(d) \( 2 \times 10^6 \) Hz

6. An Amplitude Modulated signal is given as
\[ x(t) = 100 \left( \cos(\omega_0 t + 0.5 \sin(2 \pi \times 10^3 t)) \right) \]
where \( \omega_0 \) is carrier frequency. The AM and FM
(a) \( 10^6 \) Hz
(b) \( 10^6 \) Hz
(c) \( 10^6 \) Hz
(d) \( 10^6 \) Hz

7. Statement for Multiple Answer Questions

Consider the following Amplitude Modulated (AM) signal, where \( f_c = 300 \) kHz.
\[ x(t) = 10 \cos(2 \pi \times 10^3 t + 5 \sin(2 \pi \times 10^3 t)) \]

(a) \( 25 \)
(b) \( 12.5 \)
(c) \( 6.25 \)
(d) 3.125

8. The average sideband power for the AM signal given above is
8. The AM signal gets added to a noise with Power Spectral Density $S_n(f)$ given in the figure below. The ratio of average sideband power to mean noise power would be:

$\frac{S_c(f)}{S_n(f)} \approx \frac{25}{2}$

9. Which of the following analog modulation scheme requires the minimum transmitted power and minimum channel bandwidth?

(a) VSB  
(b) DSB-SC  
(c) SSBB  
(d) AM

10. A device with input $x(t)$ and output $y(t)$ is characterized by $y(t) = x^2(t)$. An FM signal with frequency deviation of 90 kHz and modulating signal bandwidth of 5 kHz is applied to this device. The bandwidth of the output signal is:

(a) 370 kHz  
(b) 190 kHz  
(c) 380 kHz  
(d) 95 kHz

11. A carrier is phase modulated (PM) with frequency deviation of 10 kHz by a single tone frequency of 1 kHz. If the single tone frequency is increased to 2 kHz, assuming that phase deviation remains unchanged, the bandwidth of the PM signal is:

(a) 21 kHz  
(b) 22 kHz  
(c) 42 kHz  
(d) 44 kHz

12. For a message signal $m(t) = \cos (2\pi f_c t)$ and carrier of frequency $f_c$ which of the following represents a single sideband (SSB) signal?

(a) $\cos (2\pi f_c t) \cos (2\pi f_c t)$  
(b) $\cos (2\pi f_c t)$  
(c) $\cos (2\pi (f_c + f_c) t)$  
(d) $\cos (2\pi (f_c - f_c) t)$

13. A message signal given by $m(t) = (1/2) \cos (\omega_0 t)$ is amplitude modulated to give $a(t) = (1/2) \cos (\omega_0 t) \cos (2\pi f_c t)$ at the receiver.

(a) 33.33%  
(b) 11.11%  
(c) 20%  
(d) 25%

14. A communication channel with AWGN operating at a signal to noise ratio $\text{SNR} > 1$ and bandwidth $B$ has capacity $C$. If the SNR is doubled keeping $B$ constant, the resulting capacity $C'$ is given by:

(a) $C'_2 = 2C$  
(b) $C'_2 = C + B$  
(c) $C'_2 = C + 2B$  
(d) $C'_2 = C + 0.3B$

15. An AM signal is detected using an envelope detector. The carrier frequency and modulating signal frequency are 150 kHz and 2 kHz respectively. An appropriate value for the time constant of the envelope detector is:

(a) 500 μsec  
(b) 20 μsec  
(c) 0.2 μsec  
(d) 1 μsec

16. An AM signal and a narrow-band FM signal with identical carriers, modulating signals and modulation indices of 1 are added together. The resultant signal can be closely approximated by:

(a) broadband FM  
(b) SSB with carrier  
(c) DSB-SC  
(d) SSB without carrier

17. A 1 mW video signal having a bandwidth of 100 MHz is transmitted to a receiver through a cable that has 40 dB loss. If the effective one-sided noise spectral density at the receiver is $10^{-20}$ W/Hz, then the signal-to-noise ratio at the receiver is:

(a) 50 dB  
(b) 30 dB  
(c) 40 dB  
(d) 60 dB

18. A 100 MHz carrier of 1 V amplitude and a 1 MHz modulating signal of 0.1 V amplitude are fed to a balanced modulator. The output of the mixer is passed through an ideal high-pass filter with cut-off frequency of 100 MHz. The output of the filter is added with 100 MHz signal of 1 V to amplitude and 90° phase-shift. The envelope of the resultant signal is:

(a) constant  
(b) $\sqrt{1 + \sin (2\pi \times 10^8 t)}$  
(c) $\sqrt{2 - \sin (2\pi \times 10^8 t)}$  
(d) $\sqrt{2 + \cos (2\pi \times 10^8 t)}$

19. Two sinusoidal signals of same amplitude and frequencies 10 kHz and 10.1 kHz are added together. The combined signal is given to an ideal frequency detector. The output of the detector is:

(a) 0.1 kHz sine wave  
(b) 0.1 kHz sine wave  
(c) a linear function of time  
(d) a constant

20. Choose the correct one from among the alternatives A, B, C, and D after matching an item from Group 1 with the most appropriate item in Group 2.

Group 1

1. FM  
2. AM  
3. SSB  
4. PM

Group 2

1. P: Slope overload  
2. Q: μ-law  
3. R: Envelope detector  
4. S: Capture effect

21. Consider a system shown in the figure. Let $X(f)$ and $Y(f)$ denote the Fourier transform of $x(t)$ and $y(t)$ respectively. The ideal HPF has the cut-off frequency 10 kHz.

The positive frequencies where $Y(f)$ has spectral peaks are:

(a) 1 kHz and 24 kHz  
(b) 2 kHz and 24 kHz  
(c) 1 kHz and 14 kHz  
(d) 2 kHz and 14 kHz
22. A DSB-SC signal is to be generated with a carrier frequency \( f_c = 1 \) MHz using a non-linear device with the input-output characteristic

\[ v_i = a_i v_o + n_i \]

where \( a_i \) and \( n_i \) are constants. The output of the non-linear device can be filtered by an appropriate band-pass filter. Let \( v_i = a_i \cos(2\pi f_c t) + m(t) \) where \( m(t) \) is the message signal. Then the value of \( \Delta f_i \) (in MHz) is

\[ \begin{align*}
\text{a}) & \quad 1.0 \\
\text{b}) & \quad 0.333 \\
\text{c}) & \quad 0.5 \\
\text{d}) & \quad 3.0
\end{align*} \]

The data for Q.23 - 24 is given below:

Let \( m(t) = \cos \left( \frac{2\pi}{5} \times 10^5 t \right) \) be the message signal and \( c(t) = 5 \cos \left( \frac{2\pi}{5} \times 10^5 t \right) \) be the carrier.

23. \( c(t) \) and \( m(t) \) are used to generate an AM signal. The modulation index of the generated AM signal is 0.5. Then the carrier power is

\[ \begin{align*}
\text{a}) & \quad 1/2 \\
\text{b}) & \quad 1/4 \\
\text{c}) & \quad 1/3 \\
\text{d}) & \quad 1/8
\end{align*} \]

24. \( c(t) \) and \( m(t) \) are used to generate an FM signal. If the peak frequency deviation of the generated FM signal is three times the transmission bandwidth of the AM signal, then the coefficient of the term \( 2\pi \) \( (2 \times 10^8 \times 10^7 \times 3) \) in the FM signal (in terms of the constant \( k \)) is

\[ \begin{align*}
\text{a}) & \quad 5k \times 3 \\
\text{b}) & \quad 5(k) \times 3 \\
\text{c}) & \quad 5(k) \times 4 \\
\text{d}) & \quad 5k \times 6
\end{align*} \]

25. Choose the correct one from the alternatives A, B, C, D after matching an item in group I with the most appropriate item in group II.

\[ \begin{align*}
\text{Group I} & \quad \text{Group II} \\
\text{a) raised - cosine} & \quad \text{(a)} \\
\text{b) flat} & \quad \text{(b) parabolic} \\
\text{c) sinusoidal} & \quad \text{(c) Gaussian}
\end{align*} \]

26. A superheterodyne receiver is to operate in the frequency range 550 kHz - 1650 kHz with the intermediate frequency of 450 kHz. Let \( R_{\text{ios}} \) denote the required carrier frequency ratio of the local oscillator and \( f_i \) denote the input frequency (in kHz) of the incoming signal. If the receiver is tuned to 750 kHz, then

\[ \begin{align*}
\text{a}) & \quad R_{\text{ios}} = 4.41, \ I = 1600 \\
\text{b}) & \quad R_{\text{ios}} = 2.10, \ I = 1150 \\
\text{c}) & \quad R_{\text{ios}} = 3.0, \ I = 1600 \\
\text{d}) & \quad R_{\text{ios}} = 9.0, \ I = 1150
\end{align*} \]

27. The input to a coherent detector is DSB-SC signal plus noise. The noise at the detector output is

\[ \begin{align*}
\text{a}) & \quad \text{in - phase component} \\
\text{b}) & \quad \text{quadrature component} \\
\text{c}) & \quad \text{constant term} \\
\text{d}) & \quad \text{envelope}
\end{align*} \]

28. The noise at the input to an ideal frequency detector is white. The detector is operating above threshold. The power spectral density of the noise at the output is

\[ \begin{align*}
\text{a}) & \quad \text{a} \quad \text{m(t)} \quad \text{b) flat} \\
\text{c}) & \quad \text{parabolic} \\
\text{d}) & \quad \text{Gaussian}
\end{align*} \]

29. A linear phase channel with phase delay \( \tau_r \) and group delay \( \tau_g \) must have

\[ \begin{align*}
\text{a}) & \quad \tau_r = \tau_g \quad \text{constant} \\
\text{b}) & \quad \tau_r \propto f \quad \text{and} \quad \tau_g \propto f \\
\text{c}) & \quad \tau_r \text{ constant and} \quad \tau_g \propto f \\
\text{d}) & \quad \tau_r \propto f \quad \text{and} \quad \tau_g \text{ constant}
\end{align*} \]

30. A 1 MHz sinusoidal carrier is amplitude modulated by a symmetrical triangular wave of period 100 µs. Which of the following frequencies will NOT be present in the modulated signal?

\[ \begin{align*}
\text{a}) & \quad 990 \text{ kHz} \\
\text{b}) & \quad 1010 \text{ kHz} \\
\text{c}) & \quad 1020 \text{ kHz} \\
\text{d}) & \quad 1030 \text{ kHz}
\end{align*} \]

31. An angle-modulated signal is given by

\[ x(t) = \cos \left( 2\pi \times 10^8 t + 30 \sin 150 t + 40 \cos 150 t \right) \]

The maximum frequency and phase deviations of \( x(t) \) are

\[ \begin{align*}
\text{a}) & \quad 105 \text{ kHz}, 140 \text{ rad} \\
\text{b}) & \quad 6 \text{ kHz}, 80 \text{ rad} \\
\text{c}) & \quad 105 \text{ kHz}, 100 \text{ rad} \\
\text{d}) & \quad 7 \text{ kHz}, 100 \text{ rad}
\end{align*} \]

32. A bandlimited signal is sampled at the Nyquist rate. The signal can be recovered by passing the samples through

\[ \begin{align*}
\text{a}) & \quad \text{an RC filter} \\
\text{b}) & \quad \text{an envelope detector} \\
\text{c}) & \quad \text{a PLL} \\
\text{d}) & \quad \text{an ideal low-pass filter with the appropriate bandwidth}
\end{align*} \]

33. The amplitude-modulated wave form \( y(t) = A_e [1 + K_m m(t)] \cos \omega t \) is fed to an ideal envelope detector. The maximum magnitude of \( K_m m(t) \) is greater than 1. Which of the following could be the detector output?

\[ \begin{align*}
\text{a}) & \quad A_e m(t) \\
\text{b}) & \quad A_e |1 + K_m m(t)|^2 \\
\text{c}) & \quad |A_e [1 + K_m m(t)]| \\
\text{d}) & \quad A_e |1 + K_m m(t)|^2
\end{align*} \]

34. In an FM system, a carrier of 100 MHz is modulated by a sinusoidal signal of 5 kHz. The bandwidth by Carson's approximation is 1 MHz. If \( y(t) \) (modulated waveform) is then by using Carson's approximation, the bandwidth of \( y(t) \) around 300 MHz and the spacing of spectral components are, respectively

\[ \begin{align*}
\text{a}) & \quad 3 \text{ MHz}, 5 \text{ kHz} \\
\text{b}) & \quad 1 \text{ MHz}, 15 \text{ kHz} \\
\text{c}) & \quad 3 \text{ MHz}, 15 \text{ kHz} \\
\text{d}) & \quad 1 \text{ MHz}, 5 \text{ kHz}
\end{align*} \]

35. A message \( m(t) \) bandlimited to the frequency \( K_m \) has a power of \( P_m \). The power of the output signal in the figure is

\[ \begin{align*}
\text{a}) & \quad P_m \text{ cos}\theta \quad \text{b}) \quad P_m \text{ cos}\theta^2 \\
\text{c}) & \quad P_m \text{ cos}^2\theta \quad \text{d}) \quad P_m \text{ cos}\theta^2
\end{align*} \]
36. A system has a phase response given by \( \Phi (\omega) \) where \( \Phi \) is the angular frequency. The phase delay and group delay at \( \omega = \omega_0 \) are respectively given by:

\[
\begin{align*}
\Phi (\omega) &= a \omega + b \\
\Phi (\omega_0) &= c \\
\frac{d\Phi (\omega)}{d\omega} &= d \\
\frac{d^2\Phi (\omega)}{d\omega^2} &= e
\end{align*}
\]

(a) \( a = \omega_0 \), \( b = 0 \), \( e = -\omega_0 \)
(b) \( a = \frac{\omega_0}{\omega} \), \( b = 0 \), \( c = \omega_0 \)
(c) \( a = \omega_0 \), \( b = 0 \), \( d = e = 0 \)
(d) \( a = \omega_0 \), \( b = 0 \), \( c = \omega_0 \)

37. The input to a channel is a band pass signal. It is obtained by linearly modulating a sinusoidal carrier with a single-tone signal. The output of the channel due to this input is given by:

\[
y(t) = C(100 \cos(100t - 10^{-3}t^2)) + C(100t - 10^{-3}t^2)
\]

(a) \( C = 1 \)
(b) \( C = 0 \)
(c) \( C = 1 \)
(d) \( C = 0 \)

38. A modulated signal is given by:

\[
x(t) = \cos(2\pi f_t) + m(t)
\]

The group delay (\( g \)) in seconds, of the channel are:

\[
a) 10^{-6} \quad b) 10^{-5} \quad c) 10^{-4} \quad d) 10^{-3}
\]

39. A modulated signal is given by:

\[
x(t) = a \cos(b + \Delta t) + s(t)
\]

Where, \( a_0 \) and \( \Delta \) are positive constants, and \( \omega_0 = \pi \). The complex envelope of \( s(t) \) is given by:

\[
\begin{align*}
\exp(-j\omega t) \exp[j\omega_0(t + \Delta t)] &= a \exp(-j\omega t) \\
\exp(-j\omega t) \exp[j\omega_0(t + \Delta t)] u(t) &= b \exp(-j\omega t)
\end{align*}
\]

36. The image channel selectivity of a superheterodyne receiver depends upon:

40. (a) IF amplifiers only
   (b) RF and IF amplifiers only
   (c) Preselector, RF and IF amplifiers
   (d) Preselector, RF and IF amplifiers only

41. A DBSC-SC signal is generated using the carrier \( c(t) \), \( t = 0 \), and the modulating signal \( x(t) \). The envelope of the DBSC-SC signal is:

\[
x(t) \cos(\theta)
\]

42. The image channel selectivity in a superheterodyne receiver comes from:

\[
a) \text{IF stages only} \\
b) \text{RF stages only} \\
c) \text{detector and RF stages only} \\
d) \text{detector, RF, and IF stages}
\]

43. An FM signal with a modulation index \( m \) is applied to a frequency tripler. The modulation index in the output signal will be:

\[
a) 0 \\
b) 3m \\
c) 9m \\
d) 27m
\]

44. The image (second) channel selectivity of a superheterodyne communication receiver is determined by:

\[
a) \text{antenna and preselector} \\
b) \text{the preselector and RF amplifier} \\
c) \text{the preselector and IF amplifier} \\
d) \text{the RF and IF amplifier}
\]

45. A PLL can be used to demodulate:

\[
a) \text{PAM signals} \\
b) \text{PCAM signals} \\
c) \text{FM signals} \\
d) \text{DBSC-SC signals}
\]

51. Which of the following demodulators can be used for demodulating the signal \( x(t) = 5(1 + 2 \cos 200\pi t) \) mV?

\[
a) \text{Envelope demodulator} \\
b) \text{Square-law demodulator} \\
c) \text{Synchronous demodulator} \\
d) None of the above
\]

52. A superheterodyne radio receiver with an intermediate frequency of 455 KHz is tuned to a station operating at 1200 KHz. The associated image frequency is:

\[
a) 255 KHz \\
b) 50 KHz \\
c) 75 KHz \\
d) 100 KHz
\]

54. In commercial TV transmission in India, picture and speech signals are modulated respectively as:

\[
a) \text{PAM and PAM} \\
b) \text{PAM and FM} \\
c) \text{FM and FM} \\
d) \text{FM and VSB}
\]
58. A part of a communication system consists of an amplifier of effective noise temperature 21K, and a gain of 13 dB. Followed by a cable with a loss of 3 dB. Assuming the ambient temperature to be 300 K, we have for this part of the communication system,
(a) effective noise temperature = 30 K.
(b) effective noise temperature = 36 K.
(c) noise figure = 0.49 dB.
(d) noise figure = 1.61 dB.

59. In a superheterodyne AM receiver, the image channel selectivity is determined by:
(a) The preselector and IF stages
(b) The mixer, IF and RF stages
(c) The IF stages
(d) All the stages

60. In a radar receiver, the antenna is connected to the receiver through a waveguide. Placing the preamplifier on the antenna side of the waveguide results in the receiver side having:
(a) A reduction in the overall noise figure.
(b) A reduction in interference.
(c) An improvement in selectivity characteristics.
(d) An improvement in directional characteristics.

61. A carrier \( A \cos \omega_0 t \) is frequency modulated by a signal \( E_\text{c} \cos \omega_m t \). The modulation index is \( m \). The expression for the resulting FM signal is:
(a) \( A \cos \left( \omega_0 t + m \omega_m t \right) \)
(b) \( A \cos \left( \omega_0 t + m \omega_m t \right) \)
(c) \( A \cos \left( \omega_0 t + 2m \omega_m t \right) \)
(d) \( A \cos \left( \omega_0 t + \frac{2m \beta_0}{\omega_m} \cos \omega_m t \right) \)

Chapter 4

Fundamentals of information theory and channel capacity theorem

Objective Questions

01. A memoryless source emits \( n \) symbols each with a probability \( p \). The entropy of the source as a function of \( n \):
(a) increases as \( \log n \)
(b) decreases as \( \log \left( \frac{1}{n} \right) \)
(c) increases as \( n \)
(d) decreases as \( \log n \)

02. Consider a Binary Symmetric Channel (BSC) with probability of error being \( p \). To transmit a bit, say 1, we transmit a sequence of these \( k \). The receiver will interpret the received sequence to represent 1 if at least two bits are 1. The probability that the transmitted bit will be received in error is:
(a) \( p^2 + 3p^2(1-p) \)
(b) \( p^2 \)
(c) \( (1-p)^2 \)
(d) \( p^2 + p^2(1-p) \)

03. A source generates three symbols with probabilities 0.25, 0.25, 0.50 at a rate of 3000 symbols per second. Assuming independent generation of symbols, the most efficient source encoder would have average bit rate as:
(a) 6000 bits/sec
(b) 4500 bits/sec
(c) 3000 bits/sec
(d) 1500 bits/sec

04. A video transmission system transmits 625 picture frames per second. Each frame consists of a 400 x 400 pixel grid with 64 intensity levels per pixel. The data rate of the system is:
(a) 16 Mbps
(b) 100 Mbps
(c) 600 Mbps
(d) 6.4 Gbps

05. Source encoding in a data communication system is done in order to:
(a) enhance the information transmission rate
(b) reduce the transmission errors
(c) conserve the transmitted power
(d) facilitate clock recovery in the receiver.

06. A binary source has symbol probabilities 0.3 and 0.2. If error coding (blocks of 4 symbols) is used, the lower and upper bounds on the average code word length are:
(a) lower
(b) higher

97. An image uses 12 x 512 picture elements. Each of the picture elements can take any of the 8 distinguishable intensity levels. The maximum entropy in the above image will be:
(a) 2097152 bits
(b) 786432 bits
(c) 648 bits
(d) 144 bits

08. A source produces 4 symbols with probabilities 1/2, 1/4, 1/8, and 1/8. For this source, a practical coding scheme has an average codeword length of 2 bits/symbol. The efficiency of the code is:
(a) 1
(b) 7/18
(c) 1/2
(d) 1/4
14. A ternary source produces alphabets A, B and C with probabilities \( P_A = 0.5 \), \( P_B = 0.3 \) and \( P_C = 0.2 \). Which one of the following gives the correct values for the maximum value of the entropy of the source and the corresponding value of \( p \) and the range of \( p \)?

(a) 1.58, 0.33 (b) 0.5, 0.3 (c) 0.3, 0.67 (d) 0.5, 0.67

15. In order to transmit the selection of each of 16 equiprobable events, what is the number of bits required?

(a) 8 (b) 12 (c) 4 (d) 5

16. Match List – I with List – II and select the correct answer using the code given below the lists:

List – I

- A. Entropy coding
- B. Channel capacity
- C. Minimum length code
- D. Equivocation

List – II

1. McMillan’s Rule
2. Redundancy
3. Shannon’s Fano
4. Shannon law

Codes:

<table>
<thead>
<tr>
<th>A</th>
<th>B</th>
<th>C</th>
<th>D</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a)</td>
<td>1</td>
<td>2</td>
<td>3</td>
</tr>
<tr>
<td>(b)</td>
<td>3</td>
<td>4</td>
<td>1</td>
</tr>
<tr>
<td>(c)</td>
<td>1</td>
<td>4</td>
<td>2</td>
</tr>
<tr>
<td>(d)</td>
<td>3</td>
<td>2</td>
<td>1</td>
</tr>
</tbody>
</table>

17. A communication channel has a bandwidth of 100 MHz. The channel is extremely noisy such that the signal power is very much below the noise power. What is the capacity of this channel?

(a) 100 Mbps (b) 50 Mbps (c) 2400 bps (d) Nearly 0 bps

18. A source generates four messages with probability 1/8, 1/8, 1/4 and 1/2.

What is the entropy of the source (bits/message)?

(a) 1 (b) 1.75 (c) 2 (d) 4

19. A source produces 26 symbols with equal probability. What is the average information produced by this source?

(a) < 4 bits/symbol (b) 6 bits/symbol (c) 8 bits/symbol (d) Between 4 and 6 bits/symbol

20. A good line code should have which of the following?

1. Favorable psd
2. Low intersymbol interference
3. Adequate timing control
4. Transparency

Select the correct answer using the code given below:

(a) 1, 3 and 4 (b) 1, 2 and 4 (c) 2, 3 and 4 (d) 1, 2 and 3

21. Which one of the following is correct?

(a) Coding reduces the noise in the signal
(b) Coding deliberately introduces redundancy into messages
(c) Coding increases the information rate
(d) Coding increases the channel bandwidth

22. Which one of the following is the code that is very close to 'trellis code modulation'?

(a) Combination analog and digital modulation
(b) Combination modulation and encoding
(c) Encodes following trellis diagram
(d) Combination amplitude and frequency modulation

23. Source S1 produces 4 discrete symbols with equal probability. Source S2 produces 6 discrete symbols with equal probability. If \( H_1 \) and \( H_2 \) are the entropies of sources S1 and S2 respectively, then which one of the following is correct?

(a) \( H_1 \) is always less than \( H_2 \)
(b) \( H_1 \) is always greater than \( H_2 \)
(c) \( H_1 \) is equal to \( H_2 \)
(d) \( H_1 \) is 1.5 times \( H_2 \) only

24. The entropy of a digital source is 2.7 bits/symbol. It is producing 100 symbols per second. The source is likely to be which one of the following?

(a) A binary source
(b) A quinary source
(c) A octal source
(d) A hexadecimal source
Chapter 5
Digital Communications

(A) Introduction to digital communication, sampling, PAM, PWM, PPM, PCM, DPCM, and DM

Introduction:
Advantages of digital communication over analog communication:

01. Digital communication is more rugged than analog communication because it can withstand channel noise and distortion much better as long as the noise and the distortion are within limits.

02. The greatest advantage of digital communication over analog communication is the viability of regenerative repeaters in the system.

03. Digital hardware is flexible and permits the use of microprocessors, minicomputers, digital switching, and large-scale integrated circuits.

04. Digital signals can be coded to yield extremely low error rates and high fidelity as well as privacy.

05. It is easier and more efficient to multiplex several digital signals.

06. Digital communication is inherently more efficient than analog in realizing the exchange of SNR for bandwidth.

07. Digital signal storage is relatively easy and inexpensive. It also has the ability to search and select information from distinct electronic storhouses.

08. Reproduction with digital messages is extremely reliable without deterioration. Analog messages, such as photos and films, for example, lose quality at each successive stage of reproduction, and have to be transported physically from one distant place to another, often at relatively high cost.

09. The cost of digital hardware continues to halve every two or three years, while performance or capacity doubles over the same period.

Digital Communication is classified into two types:

1. Baseband data transmission

2. Bandpass data transmission

Pulse analog

Pulse digital

Binary

M-ary

PAM

PWM

PPM

PCM

DPCM

DM

ADM

ASK

PSK

QAM

QPSK

16 PSK

32 PSK

QAM

ACE Academy  Digital Communications

The fundamental difference between baseband and bandpass data transmission is with respect to channel. For baseband, the channel is free space and for baseband, the channel is coaxial cable or fiber optic cable or twisted pair.

Whenever the channel is free space, one cannot transmit the digital data directly because the spectrum of digital data is a low frequency spectrum. The height of the antenna required is very large. So the baseband is converted into bandpass or vice versa by using an analog carrier.

Sampling Theorem:
Statement: It states that if the highest frequency in the signal spectrum is \( f_c \), the signal can be reconstructed from its samples taken at a rate not less than 2\( f_c \) samples per second.

By sampling, the continuous time signal is converted into discrete time signal but the amplitude is constant.

The signal is defined only at \( 0, T, 2T, \ldots \)

where \( T \) is the sampling period.

A/D converter performs both sampling and quantization.

Disadvantage in having more number of samples is the pulse width decreases, it increases the bandwidth required to transmit the information.

Analog Signal

A/D converter

Physical Channel

D/A converter

LPF

Condition on \( T \), to get a faithful reproduction of the input analog signal is

\[ T \leq \frac{1}{2f_c} \]

\[ f_s \geq 2W \]

If the number of samples are more, the reconstructed signal is very close to the input signal.

\[ f_s = 2W \] is called the Nyquist rate.

Let us consider the spectrum as

\[ M(0) \]

\[ M(\omega) \]

\[ \omega \]

\[ 0 \]

\[ f \]
Fourier Transform of ideally sampled signal is

\[ F_T(f) = \sum_{n=-\infty}^{\infty} M(f - n/T_s) \]

where

\[ f = \frac{n}{T_s} \quad n = 0, \pm 1, \pm 2, \ldots \]

\[ M(f) = \begin{cases} K & \text{for } f < 0 \quad \text{or} \quad f > 1/T_s \\ 0 & \text{for } 0 \leq f \leq 1/T_s \end{cases} \]

\[ \text{Bandwidth of the signal after sampling is } \frac{1}{T_s}. \]

In this case, ideal LPF is used to get back the message signal.

Practical Sampling:

\[ m(t) \]

\[ T_s \]

\[ T \]

Spectrum of sampling signal

\[ = (\pi/\tau) \sum_{n=-\infty}^{\infty} \text{sinc}(\tau/\tau) M(f - n/\tau) \]

Baseband data transmission:

Pulse Analog Modulation (PAM):

The carrier is high-frequency periodic rectangular pulses. In PAM, the amplitude of the pulses is changed according to the sampled value.

Generation of PAM:

By adding a d.c. level to double-polarity PAM, we can get single-polarity PAM.

Here message signal can be recovered without distortion.

The distortion is a minimum whenever the pulse width is very small. After sampling, the sampled value is held constant using sample and hold circuit.
**Pulse Width Modulation (PWM):**
The width changes according to the sampled value. We have to change the position of leading edge or trailing edge according to sampled value.

**Generation of PWM:**

```
\begin{align*}
m(t) & \rightarrow \text{Slicer} \\
\text{Periodic sawtooth} & \rightarrow \text{Position of Trailing edge is changed} \\
\text{Leading edge is changed} & \rightarrow \text{Position of Leading edge is changed} \\
\text{Both trailing and leading edges are changed.} & \\
m(t) & \rightarrow \text{Periodic sawtooth}
\end{align*}
```

**Pulse Position Modulation (PPM):**

PPM waveform should start at trailing edge of PWM waveform.

PPM is used in satellite communication.

PAM, PPM, PWM are used in telemetry applications. Generation and demodulation is complex in case of PPM.

If leading edges are modulated, we have to use a forward biased diode and multivibrator with positive trigger pulse.
Pulse Digital Communications:

There are four types of pulse digital communication techniques.

1. PCM  2. DPDM  3. DM  4. ADM

Pulse Code Modulation (PCM):

In PCM, a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude. The LPF prior to sampling is included to prevent aliasing of the message signal. The basic operation performed in the transmitter of a PCM sampling system are sampling, quantizing, and encoding. The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter.

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

Quantizer converts a continuous amplitude signal to a discrete-time signal.

The encoding process is used to translate the discrete set of sample values to a more appropriate form of signals.

Source of continuous message signal → LPF → Sample → Quantizer → Encoder → PCM signal applied to channel input

Transmission path:

Distorted PCMsignal produced at channel output → Regenerative repeater → Regenerative repeater → Regenerated PCM signal applied to the receiver

Final channel output → Regenerative repeater → Decoder → Reconstruction filter → Destination

Receiver

Without quantization, it is not possible to have a unique code for each sampled value. Quantizer rounds off each sampled value to nearest quantize level.

Number of quantization levels $L = 2^n$

Step size $\delta = \frac{V_{\text{max}} - V_{\text{min}}}{L}$

Bit rate $(N) = \text{Sampling rate} \times n = \frac{1}{T_s} \times n = \frac{n}{T_s}$

Bit duration $(T_b) = \frac{T_s}{n} = \frac{1}{n}$

Max. B.W $= \frac{n}{T_s}$

Electrical Representation of Binary data:

1. On-off signalling
2. NRZ (Non-return to zero)
3. RZ (Return to zero)
4. Differential encoding
5. Bipolar (split phase) encoding

<table>
<thead>
<tr>
<th>On-off</th>
<th>NRZ</th>
<th>RZ</th>
<th>Bipolar</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 0 0 1 1 0</td>
<td>1 2m 3m 4m 7m 8m</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Regeneration of data is possible in case of digital communications. This is the major advantage over analog communications.

For n-bit PCM,
Differential Encoding:

The binary data is encoded first and is represented as on-off signalling.

1 = previous state
0 = previous state is complemented

\[
\begin{array}{c}
\text{Input} \quad 1 & 0 & 0 & 1 & 1 & 0 \\
\text{Output} \quad 0 & 1 & 0 & 1 & 3 & 0 & 0 & 1
\end{array}
\]

Manchester Coding:

\[
\begin{align*}
1 & : T_s/2 \quad rV_e \\
0 & : T_s/2 \quad -V_e \\
0 & : T_s/2 \quad -V_e \\
1 & : T_s/2 \quad rV_e
\end{align*}
\]

For NRZ signaling, the probability of error is minimum. Thus for PCM, NRZ signaling is generally used.

Regenerative Repeater:

It is used to eliminate the effect of channel noise. It is nothing but one bit quantizer.

\[
5V \quad 0 \quad 0
\]
Synchronization:

Number of signals = N
Number of bits in frame = nN

For each frame, some extra bits are added for synchronization purpose.

In T\_s duration, we get n = \(\frac{N}{a}\) bits, where a = No. of bits used for synchronization.

\[
\text{Bit duration, } T_s = \frac{T_s}{n} \quad \text{without multiplexing}
\]

\[
T_s = \frac{T_s}{nN} \quad \text{with multiplexing}
\]

\[
T_s = \frac{T_s}{nN + a} \quad \text{with multiplexing & synchronization}
\]

Signal to Quantization Noise Ratio:

\[
m(0) = A_m \cos 2\pi f_s t
\]

\[
\text{Signal Power} = \frac{A_m^2}{2}
\]

\[
\text{Noise Power} = \int_{-\infty}^{\infty} x^2(\delta) \, dx
\]

\[
\text{Quantization Noise Power} = \int_{-\delta}^{\delta} \phi^2 \cdot f(\delta) \, d\delta = \frac{\delta^2}{12}
\]

\[
\text{SQNR} = \frac{A_m^2}{\delta^2 / 12}
\]

\[
\delta = 2A_m
\]

\[
\text{Example: In PCM system, the minimum signal-to-quantization noise ratio should be 23 db. Calculate the number of quantization levels required.}
\]

\[
\text{Sol: } \quad \text{SQNR} = 1.8 + 6n = 27
\]

\[
\Rightarrow \quad n = 4
\]

\[
\text{No. of quantization levels} = 2^4 = 16
\]

DPCM (Differential Pulse Code Modulation):

In case of delay, input to the quantizer is present sample and previous sample. In case of predictor, input to the quantizer is sampled value and its predicted value.

**Predictor:** It consists of delay elements by observing the previous sampled values, it predicts the present value.

\[
\text{Bit rate} = \frac{n}{T_s} = \text{BW}
\]
Delta Modulation (DM):

It is used to decrease the bandwidth. Block schematic is similar to DPCM but the encoder is 1 bit encoder.

![Diagram of Delta Modulation](image)

1. Slope Overload error

2. Granular noise

The input to the LPF is in multiples of $\delta$. Slope overload error occurs when the step size is very low. Granular noise occurs when the step size is high.

When the slope of the transmitted waveform and slope of the reconstructed waveform are same then there will be no error.

Slope of reconstructed waveform = $\frac{\delta}{T_s}$

For no slope overload error,

$$\frac{d}{dt} m(t) = \frac{\delta}{T_s}$$

If the slope of the reconstructed signal is low or high compared to $\frac{d}{dt} m(t)$ then slope overload error (or Granular noise) occurs.

$$\frac{\delta}{T_s} \left< \frac{d}{dt} m(t) \right.$$ Slope Overload error

$$\frac{\delta}{T_s} > \frac{d}{dt} m(t)$$

Granular Noise

For $m(t) = A_m \cos(2\pi f_0 t)$

$$\frac{d}{dt} m(t) = -A_m 2\pi f_0 \sin 2\pi f_0 t$$

$$\frac{d}{dt} m(t) = A_m 2\pi f_0$$

**Adaptive Delta Modulation (ADM):** ADM involves additional hardware designed to provide variable step size, thereby reducing slope overload effects without increasing the granular noise.
### Objective Questions

#### SET - A

<table>
<thead>
<tr>
<th>Chapter - 5A</th>
<th>Objective Questions</th>
<th>SET - A</th>
</tr>
</thead>
<tbody>
<tr>
<td>01. The maximum permissible duration between two samples of a 2 KHz signal is</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) 1000 μsec</td>
<td>b) 500 μsec</td>
<td>c) 250 μsec</td>
</tr>
<tr>
<td>c) 250 μsec</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>02. Pick the odd man out</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) PFM</td>
<td>b) PPM</td>
<td>c) PDM</td>
</tr>
<tr>
<td>03. The main advantage of TDM over FDM is that it</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) needs less power</td>
<td>b) needs less bandwidth</td>
<td>c) needs simple circuitry</td>
</tr>
<tr>
<td>d) gives better SN ratio</td>
<td></td>
<td></td>
</tr>
<tr>
<td>04. The PFM needs</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) more power than PPM</td>
<td>b) more samples per second than PPM</td>
<td></td>
</tr>
<tr>
<td>c) more bandwidth than PPM</td>
<td>d) None of the above</td>
<td></td>
</tr>
<tr>
<td>05. The PAM signal can be detected by</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) bandpass filter</td>
<td>b) band-stop filter</td>
<td></td>
</tr>
<tr>
<td>c) high-pass filter</td>
<td>d) low-pass filter</td>
<td></td>
</tr>
<tr>
<td>06. The guard time between pulses increases transmission efficiency</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) True</td>
<td>b) False</td>
<td></td>
</tr>
<tr>
<td>07. Noise can be reduced by increasing sampling rate</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) True</td>
<td>b) False</td>
<td></td>
</tr>
<tr>
<td>08. Pulse stuffing is used in</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) synchronous TDM</td>
<td>b) asynchronous TDM</td>
<td></td>
</tr>
<tr>
<td>c) any TDM</td>
<td>d) None of the above</td>
<td></td>
</tr>
<tr>
<td>09. In communications, the sampling technique leads to</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) Better efficiency</td>
<td>b) Highest speed of communication</td>
<td></td>
</tr>
<tr>
<td>c) Less costly equipment</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>10. The minimum sampling frequency is called</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) Carson frequency</td>
<td>b) Pulse sampling rate</td>
<td></td>
</tr>
<tr>
<td>c) Nyquist sampling rate</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>11. The sampling rate is always between</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) 0 and W</td>
<td>b) W and 2W</td>
<td></td>
</tr>
<tr>
<td>c) 2W and 4W</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>12. As the sampling frequency is increased, the guard band becomes</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) Smaller</td>
<td>b) Remains same</td>
<td></td>
</tr>
<tr>
<td>c) Larger</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>13. Which of the following pulse modulation systems has no carrier-wave equivalent?</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) PPM</td>
<td>b) PDM</td>
<td></td>
</tr>
<tr>
<td>c) PCM</td>
<td>d) None</td>
<td></td>
</tr>
<tr>
<td>14. So far as noise is concerned, as compared to direct baseband transmission, PAM is</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) Better</td>
<td>b) Similar</td>
<td></td>
</tr>
<tr>
<td>c) Worse</td>
<td>d) Same</td>
<td></td>
</tr>
<tr>
<td>15. The sampling in PDM is</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a) Uniform</td>
<td>b) Non-uniform</td>
<td></td>
</tr>
<tr>
<td>c) Dependent on the nature of message signal</td>
<td>d) None</td>
<td></td>
</tr>
</tbody>
</table>

#### Key

16. The PAM signal is converted into PPM with the help of a
   a) Monostable                          
   b) Flip-flop                           
   c) Timer                               
   d) None                               

17. In PBM, the bandwidth requirement is a function of
   a) the position of the pulse           
   b) Maximum pulse width                
   c) Minimum pulse width                
   d) None                               

18. Which of the following circuits cannot be used to generate PAM?
   a) Triangular wave shaper
   b) Square wave shaper
   c) Sine wave shaper
   d) None                               

19. Apart from a dc component, the PPM spectrum consists of
   a) AM and PM sidebands of all multiples of f' 
   b) Spectrum of message derivative
   c) Both of these                       
   d) None                               

20. To generate PDM, the circuit needed is
   a) Sawtooth generator
   b) Chopping and squaring circuit
   c) Both of these                       
   d) None                               

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

1. Adaptive Delta Modulation is preferred over Delta Modulation because its step size changes as per the requirement
   (a) True
   (b) False

2. In PCM, the quantization noise depends on
   a. sampling rate
   b. number of quantization levels
   c. signal power
   d. None of the above

3. Which of the following modulation is digital in nature?
   a. PAM
   b. PPM
   c. D M
   d. None of the above

4. Which of the following modulation is analog in nature?
   a. PCM
   b. DPCM
   c. D M
   d. None of the above

5. In PCM, if the number of quantization levels is increased from 4 to 64, then the bandwidth requirement will
   approximately be
   a. 3 times
   b. 4 times
   c. 8 times
   d. 16 times

6. Quantization noise occurs in
   a. PAM
   b. PPM
   c. D M
   d. None of the above

7. Companding is used in PCM to
   a. reduce bandwidth
   b. reduce power
   c. increase S/N ratio
   d. get almost uniform S/N ratio

8. The main advantage of PCM is
   a. less bandwidth
   b. less power
   c. better S/N ratio
   d. possibility of multiplexing

9. The main disadvantage of PCM is
   a. large bandwidth
   b. large power
   c. complex circuitry
   d. quantization noise

10. The main advantage of DM over PCM is
    a. less bandwidth
    b. less power
    c. better S/N ratio
    d. simple circuit

11. In a DM system, the granular noise occurs when modulating signal
    a. increases rapidly
    b. decreases rapidly
    c. changes within the step size
    d. has high frequency component

12. In DM, slope overload occurs when the
    a. Less rapidly than the encoder can follow
    b. More rapidly than the encoder can follow
    c. None of these

13. In DM, in order to keep S/N ratio, nearly constant, the step size “X” is kept small but increases as the input signal
    a. Increases
    b. Decreases
    c. Remains unaffected
    d. None

14. As compared to message bandwidth, the PCM bandwidth is
    a. much larger
    b. same
    c. much smaller
    d. None

15. In PCM, the high noise immunity is achieved with
    a. increased bandwidth
    b. decreased bandwidth
    c. None

16. In PCM, quantization noise appears as
    a. thermal noise
    b. Shot noise
    c. White noise

17. For noise reduction in PCM, the exchange of bandwidth for S/N ratio is
    a. Linear
    b. Exponential
    c. Uniform

18. To generate PCM, the signal is sampled and converted into
    a. PWM
    b. PPM
    c. PAM

19. In PCM system, if the bandwidth of a channel is increased, the S/N ratio would
    a. Decrease
    b. Remain the same
    c. Increase

20. The bandwidth required for transmitting 4 KHz signal using PCM with 128 quantizing levels is
    a. 8 KHz
    b. 16 KHz
    c. 24 KHz
    d. 28 KHz

Key:
01. a 02. b 03. c 04. d 05. a 06. c 07. d 08. e 09. a 10. d 11. c 12. b 13. a 14. a 15. a 16. e 17. b 18. c 19. e 20. e
01. In a delta modulation scheme, the step height  is 75 mV and step width is 1.5 ms. The maximum slope that the staircase can track is:
   a) 50 V/s  b) 55 V/s  c) 60 V/s  d) 65 V/s

02. The ramp signal \( m(t) = M_0 a t \) is applied to a delta modulator with sampling period \( T_s \) and step size \( \delta \). Slope overload and distortion will occur if:
   a) \( \delta < \frac{a}{T_s} \)  b) \( \delta > \frac{a}{T_s} \)  c) \( \delta < a T_s \)  d) \( \delta > a T_s \)

03. In delta modulation which of the following drawbacks are existing?
   a) 1 & 2 only  b) 2 & 3 only  c) 1 & 3 only  d) 1, 2 & 3

04. The signal-to-quantization noise ratio in an \( n \)-bit PCM system:
   a) depends upon the sampling frequency employed
   b) independent of the value of \( n \)
   c) decreases with increasing value of \( n \)
   d) increases with increasing value of \( n \)

05. A speech signal occupying the bandwidth of 300 Hz to 3 kHz is converted into PCM format for use in digital communication. If the sampling frequency is 8 kHz and each sample is quantized into 256 levels, then the output bit rate will be:
   a) 3 kbps  b) 8 kbps  c) 64 kbps  d) 256 kbps

06. Which of the following pulse communication system is inherently immune to noise?
   a) FFM  b) PCM  c) FWM  d) PAM

07. Companding is used:
   a) to overcome quantizing noise in PCM
   b) in FWM receivers to reduce impulse noise
   c) to protect small signals in PCM from quantizing noise
   d) None of the above

08. In PCM, for 128 standard quantizing levels, the maximum error will be:
   a) \( (1.36) \) of the total amplitude range  b) \( (1/32) \) of the total amplitude range
   c) \( (1/256) \) of the total amplitude range  d) \( (1/528) \) of the total amplitude range

09. Quantizing noise is produced in:
   a) FDM  b) PCM  c) all modulation systems  d) all pulse modulation systems

10. For a 10-bit PCM system, the signal to quantization noise ratio is 62 dB. If the number of bits is increased by 2, then the signal to quantization noise ratio will:
   a) increase by 6 dB  b) increase by 12 dB
   c) decrease by 6 dB  d) decrease by 12 dB

11. Four voice signals, each limited to 4 kHz and sampled at Nyquist rate, are converted into binary PCM signal using 256 quantization levels. The bit transmission rate for the time-division multiplexed signal will be:
   a) 8 kbps  b) 64 kbps  c) 256 kbps  d) 512 kbps

12. A message signal band limited to 5 kHz is sampled at the minimum rate as dictated by the sampling theorem. The number of quantization levels is 64. If the samples are encoded in 8-bit form, the transmission rate is:
   a) 64 kbps  b) 50 kbps  c) 32 kbps  d) 10 kbps

13. The peak-to-peak input to an 8-bit PCM decoder is 2 volts. The signal power-to-quantization noise power ratio (in dB) for an input of 0.5 cos \( \omega_0 t \) is:
   a) 47.8  b) 49.8  c) 95.6  d) 96.6

14. Four independent messages have bandwidths of 100 Hz, 200 Hz, and 400 Hz, respectively. Each is sampled at the Nyquist rate, and the samples are time-division multiplexed (TDM) and transmitted. The transmitted-signal rate (in Hz) is:
   a) 1400  b) 800  c) 600  d) 200

15. A TDM link has 20 channel signals and each channel is sampled 8000 times/sec. Each sample is represented by seven binary bits and contains an additional bit for synchronization. The total bit rate for the TDM link is:
   a) 1180 K bits/sec  b) 1280 K bits/sec  c) 1180 M bits/sec  d) 1280 M bits/sec

16. When the number of quantization levels is increased from 4 to 64, the bandwidth required for the transmission of a PCM signal increases by a factor of:
   a) 3  b) 4  c) 5  d) 6
Chapter - 5A

Objective Questions

Statement for Linked Answer Questions 05 & 06:

An input to a 6-level quantizer has the probability density function f(x) as shown in the figure. Decision boundaries of the quantizer are chosen so as to maximize the entropy of the quantizer output. It is given that 3 consecutive decision boundaries are \( -1', 0', \) and \( 1' \).

01. If the bits 0 and 1 are transmitted using bipolar pulses, the minimum bandwidth required for distortion-free transmission is
(a) 64 kHz  
(b) 32 kHz
(c) 8 kHz  
(d) 4 kHz

02. Assuming the signal to be uniformly distributed between its peak to peak value, the signal to noise ratio at the quantized output is
(a) 16 dB  
(b) 32 dB
(c) 48 dB  
(d) 64 dB

03. The number of quantization levels required to reduce the quantization noise by a factor of 4 would be
(a) 1024  
(b) 512
(c) 256  
(d) 64

04. In delta modulation, the slope overload distortion can be reduced by
(a) decreasing the step size  
(b) decreasing the quantizer noise
(c) decreasing the sampling rate  
(d) increasing the step size

05. The values of a and b are
(a) a = 1/6 and b = 1/12  
(b) a = 1/5 and b = 2/7
(c) a = 1/4 and b = 3/48  
(d) a = 1/3 and b = 1/24

06. Assuming that the reconstruction levels of the quantizer are the mid-points of the decision boundaries, the ratio of signal power to quantization noise power is
(a) 152/9  
(b) 4/3
(c) 76/3  
(d) 32/28

07. A signal m(t) with bandwidth 500 Hz is first multiplied by a signal g(t) where
\[ g(t) = \sum_{k=-\infty}^{\infty} (1 + 0.5 \times 10^{-k}) \]

The resulting signal is then passed through an ideal lowpass filter with bandwidth 1 kHz. The output of the lowpass filter would be
(a) \( \delta(t) \)  
(b) m(t)
(c) 0  
(d) \( \delta(t \div 2) \)

08. The minimum sampling frequency (in samples/sec) required to reconstruct the following signal from its samples without distortion
\[ x(t) = \sum_{n=1}^{\infty} \left( \sin \left( \frac{2 \pi \times 10000}{10 \times n} \right) \right)^2 \]

would be
(a) \( 2 \times 10^3 \)  
(b) \( 4 \times 10^3 \)
(c) \( 6 \times 10^3 \)  
(d) \( 8 \times 10^3 \)

09. The minimum step size required for a Delta Modulator operating at 32 K samples/sec to track the signal (here u(t) is the unit step function)
\[ x(t) = 128 u(t) - u(t - 1) + (250 - 125 u(t - 1) - u(t - 2)) \]

so that slope overload is avoided, would be
(a) \( 2^{-16} \)  
(b) \( 2^{-8} \)
(c) \( 2^{-4} \)  
(d) \( 2^{-2} \)

10. In the following figure the minimum value of the constant "C", which can be added to y1(t) such that y2(t) and y3(t) are different, is

[Diagram showing quantizer and signal flow]

(a) A  
(b) A/2
(c) \( A^2 \)  
(d) A/L

Statement for Linked Answer Questions 11 & 12:

A symmetric three-level first order quantizer is to be designed assuming equiprobable occurrence of all quantization levels.

11. If the input probability density function is divided into three regions as shown in fig. the value of \( 'a' \) in the figure is

(a) 1/3  
(b) 1/2
(c) 1/4  
(d) \( 1/3 \)

12. The quantization noise power for the quantization region between \( -a \) and \( a \) in the figure is

(a) \( 4/81 \)  
(b) 1/9
(c) 5/81  
(d) \( 2/81 \)
14. The positive values of the signal are uniformly quantized with a step size of 0.02 V, and the negative values are uniformly quantized with a step size of 0.01 V. The resulting signal to quantization noise ratio is approximately
(a) 36 dB
(b) 43.8 dB
(c) 42 dB
(d) 40 dB

15. In a PCM system, if the code word length is increased from 6 to 8 bits, the signal to quantization noise ratio improves by the factor
(a) 8/3
(b) 12
(c) 16
(d) 32

16. In the output of a DM speech encoder, the consecutive pulses are of opposite polarity during time interval $t_1 = 0.5 t_0$. This indicates that during this interval
(a) the input to the modulator is
(b) the modulator is going through slope overload
(c) the accumulator is in saturation
(d) the speech signal is being sampled at the Nyquist rate

17. A random variable $X$ with uniform density in the interval $0 \leq x \leq 1$ is quantized as follows:
If $0 \leq x \leq 0.3$, $x = 0$
If $0.3 \leq x \leq 0.5$, $x = 0.7$
Where $x_n$ is the quantized value of $X$. The root-mean-square value of the quantization noise is
(a) 0.573
(b) 0.198
(c) 2.205
(d) 0.266
32. If the number of bits per sample in a PCM system is increased from n to (n + 1), the improvement in signal to quantization noise ratio will be
(a) 3 dB
(b) 6 dB
(c) 7a dB
(d) n dB

33. A 1.0 kHz signal is flat — top sampled at the rate of 1800 samples/sec and the samples are applied to an ideal rectangular LPF with cut-off frequency of 1100 Hz. Then the output of the filter contains:
(a) only 800 Hz component
(b) 400 Hz and 900 Hz components
(c) 800 Hz and 1000 Hz components
(d) 800 Hz, 900 Hz, and 1000 Hz components

34. The signal to quantisation noise ratio in an in – bit PCM system:
(a) depends upon the sampling frequency employed
(b) is independent of the value of 'n'
(c) increases with increasing value of 'n'
(d) decreases with the increasing value of 'n'

Match the following (Q.No. 35, 36, 37)

35. a) AM system
   (1) Correlation detection
   (b) DSB – SC system
   (2) Envelope detection
   (c) FM system
   (3) Amplitude detection
   (d) PLL system
   (4) LPF
   (5) B(6)

36. a) AM
    (1) B(6) bandwidth of the modulating signal
    (b) DSB
    (2) B
    (c) SSB system
    (3) Between B and 2B
    (d) 2B
    (e) PCM (6-bit)
    (4) 2B
    (f) B

37. Increased pulse width in a flat-top sampled signal, leads to
   (a) attenuation of high frequencies in reproduction
   (b) attenuation of low frequencies in reproduction
   (c) generation of errors in reproduction
   (d) no harmful effects in reproduction

38. The bandwidth required for the transmission of a PCM signal increases by a factor of ______ when the number of quantisation levels is increased from 4 to 64

39. Six independent low pass signals of bandwidths 3 W, W, 2 W, 3 W and 2 W Hz are to be time-division multiplexed on a common channel using PAM. To achieve this, the minimum transmission bandwidth of the channel should be ______ Hz

40. A signal has frequency components from 300 Hz to 1.8 KHz. The minimum possible rate at which the signal has to be sampled is

41. A 4 MHz carrier is DSB SC modulated by a low pass message signal with maximum frequency of 2 MHz. The resultant signal is to be slowly sampled. The minimum frequency of the sampling impulse train should be:
   (a) 4 MHz
   (b) 8 MHz
   (c) 8 GHz
   (d) 8.004 GHz

42. A is a BFSK signal detector, the local oscillator has a fixed phase error of 20°. This phase error deteriorates the SNR at the output by a factor of
   (a) cos 20°
   (b) cos 20°
   (c) cos 70°
   (d) cos 70°
Band pass Data Transmission

The output of the encoder has significant low frequency components. If it has to radiate through antennas then antenna height will become a problem. So the encoder output is multiplied by an analog carrier.

Digital Carrier Modulation:

1. Amplitude Shift Keying (ASK):

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

The amplitude of the carrier is modulated according to the message signal.

Demodulation:

\[ 0 \Rightarrow A_c \cos(2\pi f_c t) \]

\[ 1 \Rightarrow A_c \cos(2\pi f_c t + \theta) \]

Output of LPF = \( A_c\sqrt{2} \)

Example:

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

\[ \theta = \pi \]

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

Pass Shift Keying:

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

\[ \theta = \pi \]

\[ s(t) = -A_c \cos(2\pi f_c t) \]
Frequency of the carrier must be a multiple of a bit rate.

\[ T_b = n / f_c \Rightarrow f_c = n f_b \]

**Demodulation**:

PSK

\[ A_c \cos 2\pi f_c t \]

PSK

\[ m(t) \]

\[ A_c \cos 2\pi f_b t \]

\[ -A_c \cos 2\pi f_b t \]

Output of LPF = \( A_c \cos 2\pi f_b t \)

\[ 0 \Rightarrow \text{Output of LPF} = -A_c \cos 2\pi f_b t \]

Bandwidth = 2 \times \text{bit rate}

\[ f_b \]

**Frequency Shift Keying**

\[ x(t) = A_c \cos 2\pi f_c t \]

\[ -A_c \cos 2\pi f_b t \]

\[ f_b > f_c \]

\[ f_b + f_c \]

\[ f_c \]

\[ f_b - f_c \]

\[ f_b + f_c \]

\[ f_b - f_c \]

\[ f_b \]

\[ f_c \]

\[ f_b \]

\[ f_c \]
Demodulation:

\[ f_1 - f_0 = 2 \Delta f \]

PSK signal

\[ A_c \cos 2\pi f_c t \]

\[ A_c \cos 2\pi f_1 t \]

Cut-off frequency < \( f_1 - f_0 \)

1

\[ \frac{A_c}{2} \cos 2\pi (f_1 - f_0) t \]

2

\[ (A_c^2 / 2) \left[ \cos 2\pi (f_1 - f_0) t + \cos 2\pi (2f_1 - f_0) t \right] \]

\[ A_c \cos 2\pi f_2 t \times A_c \cos 2\pi f_1 t \]

1

\[ A_c \cos (2\pi f_0 t) \times A_c \cos (2\pi f_1 t) \]

2

Differential Phase Shift Keying (DPSK):

Binary data

on-off signalling

Differential Encoder

PSK Modulator

DPSK signal

Amplitude level shifter

Multiplier

DPSK signal

Receiver

Advantage of DPSK over PSK in DPSK does not require a coherent carrier for demodulation.
(C) Matched filters / Correlation Receiver, B.W. Probability of error

\[ x(t) \rightarrow \text{filter} \rightarrow \text{sample at } t = T \rightarrow \text{Decision device} \]

Input to the filter \( \rightarrow \) Signal + White noise
\( \rightarrow x_0 + w(t) \)

Output of the filter: signal component
\( = x_0 \cdot h(t) \)
\( = S(t) \cdot H(t) \)

\[ y(t) = \int S(t) H(t) e^{j2\pi f t} \, df \]

After sampling
\[ y(T) = \int S(t) H(t) e^{j2\pi f T} \, df \]

Signal Power
\[ |y(T)|^2 = \left| \int S(t) H(t) e^{j2\pi f T} \, df \right|^2 \]

Schwartz Inequality:
\[ \left| \int g(t) h(t) \, dt \right|^2 \leq \int |g(t)|^2 \, dt \cdot \int |h(t)|^2 \, dt \]

Signal Power \( \leq \int |H(t)|^2 \, df \cdot \int |S(t)|^2 \, df \)

Max. signal Power \( \leq \int |H(t)|^2 \, df \cdot \int |S(t)|^2 \, df \)
\[ H(t) = S^*(t) e^{-j2\pi f t} \]
\[ h(t) = \int S^*(t) e^{-j2\pi f t} \, df \]

Noise PSD at the output of the filter is \( \frac{N_0}{2} \cdot |H(0)|^2 \)
Probability of error:

\[
\begin{align*}
&\text{Binary Nonsymmetric channel} \\
&P_r = P(\hat{0}|0) \quad \text{or} \quad P(\hat{1}|0) \\
\text{Input to the filter} &\rightarrow X(t) = s(t) + n(t) \\
\text{Output of the filter} &\rightarrow Z(t) = a(t) + n(t) \\
\text{Assume the noise as Gaussian noise with} \mu_n = 0, \text{when there is no signal.} \\
Z &\text{is a Gaussian Random variable with} \mu_n = 0
\end{align*}
\]

\[
\begin{align*}
&\text{Binary symmetric channel} \\
&\text{Input to the filter} \rightarrow X(t) = s(t) + n(t) \\
&\text{Output of the filter} \rightarrow Z(t) = a(t) + n(t) \\
&\text{Assume the noise as Gaussian noise with} \mu_n = 0, \text{when there is no signal.} \\
Z &\text{is a Gaussian Random variable with} \mu_n = 0
\end{align*}
\]

\[
\begin{align*}
f(z) &= \frac{1}{\sqrt{2\pi}\sigma^2} \exp\left[-\frac{z^2}{2\sigma^2}\right] \\
\mu_n &= 0 \\
\text{Mean Square Value} &= \sigma_n^2 = \text{N-signal Power} \\
\sigma^2 &= \sigma_n^2 (\text{mean square value})
\end{align*}
\]

When the input is '1', \( s(t) \)

\[
Z = a(t) + n(t)
\]

because of \( n(t) \), \( Z \) is a random variable with mean \( a(t) \).

\[
f(z/n) = \frac{1}{\sqrt{2\pi}\sigma} \exp\left[-\frac{(z-a(t))^2}{2\sigma^2}\right]
\]

When the input is '0', \( s(t) \)

\[
Z = a(t) + n(t)
\]

because of \( n(t) \), \( Z \) is a random variable with mean \( a(t) \).

\[
f(z/n) = \frac{1}{\sqrt{2\pi}\sigma} \exp\left[-\frac{(z-a(t))^2}{2\sigma^2}\right]
\]

'0' is decoded as '1' when \( (a(t) + n(t)) < \frac{a(t) + n(t)}{2} \)

\[
P(X = 1) = P(s(t)) = \int_{-\infty}^{\infty} f(x) dx
\]

Probability that \( 1 \) is decoded as zero is

\[
P_e(0) = P \left( Z < -\frac{a(t) + n(t)}{2} \right)
\]
\[
\frac{1}{\sqrt{2\pi}\sigma^2} \int \exp \left[ -\frac{(z-a)^2}{2\sigma^2} \right] \, dz
\]

Probability of error when \( Z \) is transmitted is

\[
P_e(Z) = \frac{1}{\sqrt{2\pi}\sigma^2} \int \exp \left[ -\frac{(z-a)^2}{2\sigma^2} \right] \, dz
\]

For binary symmetrical channel \( P_{e(1|0)} = P_{e(0|1)} \)

\[
P_e = \frac{1}{\sqrt{2\pi}\sigma^2} \int \exp \left[ -\frac{(z-a)^2}{2\sigma^2} \right] \, dz
\]

Let \( y = \frac{z-a}{\sigma} \)

\[
y = \frac{z-a}{\sigma}
\]

\[
\int \frac{1}{\sqrt{2\pi}\sigma^2} \exp \left[ -\frac{y^2}{2} \right] \, dy
\]

\[
P_e = \frac{1}{\sqrt{2\pi}} \int \exp \left[ -\frac{y^2}{2} \right] \, dy
\]

where \( Q(x) = \frac{1}{\sqrt{2\pi}} \int \exp \left[ -\frac{y^2}{2} \right] \, dy \)

\[
P_e = Q\left( \frac{a_1-a_2}{\sqrt{2}\sigma} \right)
\]

\[
(a_1-a_2)^2 = \text{Difference signal power}
\]

\[
\frac{2E_d}{N_0}
\]

On-off signaling:

\[
E_d = \int_0^\infty (x-a)^2 \, dx
\]

Difference signal energy per bit

\[
= \int_0^\infty (x-a)^2 \, dx
\]

\[
= 2A^2 T_b
\]

NRZ signaling:

\[
\int_0^\infty 4A^2 \, dx = 4A^2 T_b
\]

\[
P_e = Q\left( \sqrt{\frac{2A^2 T_b}{N_0}} \right)
\]

\[
P_{\text{sec}} < P_{\text{sec}}^\text{worst}
\]

Correlation Receiver:

\[
\begin{array}{c}
\text{Filter} \quad h(t) \\
\text{y(t)} \\
\text{Decision device}
\end{array}
\]

\[
y(t) = \int y(x) h(t-x) \, dx
\]

\[
= \int y(x) [T - (t-x)] \, dx
\]

\[
h(t) = s(T-t)
\]

\[
= \int y(t) [T + t - T] \, dt
\]
After sampling,
\[ y(t) = \int_{t-\tau/2}^{t+\tau/2} s(\tau) \cos(2\pi f_c \tau) d\tau \]

In general, correlation receiver and matched filter are same.

\[ P_r = Q \left( \frac{E_s}{2N_0} \right) \]

ASK signal:
\[ s_1 = A_s \cos 2\pi f_c t \]
\[ s_2 = 0 \]
\[ E_s = A_s^2 \int_0^T \cos^2 2\pi f_c t \frac{d}{t} = A_s^2 [T_s/2] = (A_s^2 T_s)/2 \]
\[ P_r = Q \left( \frac{A_s^2 T_s}{4N_0} \right) \]

Signal energy when '1' is transmitted = \[ A_s^2 \frac{T_s}{2} \cos^2 2\pi f_c t \int_0^T = \frac{A_s^2 T_s}{2} \]

Signal energy when '0' is transmitted = 0

Average energy per bit = \[ \frac{A_s^2 T_s}{2} = E_b \]

\[ A_s = \sqrt{\frac{4E_b}{T_s}} \]

\[ s(0) = \sqrt{\frac{4E_b}{T_s}} \cos 2\pi f_c t \quad '0' \]

\[ s(1) = A_s \cos 2\pi f_c t \quad '1' \]

PSK signal:
\[ s_1 = A_s \cos 2\pi f_c t \]
\[ s_2 = -A_s \cos 2\pi f_c t \]
\[ E_s = 4A_s^2 \int_0^T \cos^2 2\pi f_c t \frac{d}{t} = 4A_s^2 T_s \frac{T_s}{2} = 2A_s^2 T_s \]
\[ P_r = Q \left( \frac{2A_s^2 T_s}{2N_0} \right) = Q \left( \frac{A_s^2 T_s}{N_0} \right) \]

This shows PSK is preferred.

\[ P_r = Q \left( \frac{2E_s}{N_0} \right) \]

\[ s(0) = A_s \cos 2\pi f_c t \quad '0' \]

\[ s(1) = -A_s \cos 2\pi f_c t \quad '1' \]

Signal energy when '1' is transmitted = \[ \frac{A_s^2 T_s}{2} \]

Signal energy when '0' is transmitted = \[ \frac{A_s^2 T_s}{2} \]

Average energy per bit = \[ \frac{A_s^2 T_s}{2} \]

\[ A_s = \sqrt{\frac{2E_s}{T_s}} \]
\[
S_d(t) = \sqrt{\frac{2E_b}{T_s}} \cos 2\pi f_d t, \quad '1'
\]
\[
= \sqrt{\frac{2E_b}{T_s}} \cos 2\pi f_d t, \quad '0'
\]

**FSK signal**

\[
S_1(t) = A_c \cos 2\pi f_1 t
\]
\[
S_2(t) = A_c \cos 2\pi f_2 t
\]
\[
E_t = A_c^2 \left( \frac{\cos 2\pi f_1 t - \cos 2\pi f_2 t}{1} \right) \Delta t = A_c^2 T_s
\]
\[
P_t = \frac{A_c^4}{2N_0}
\]

PSK is most preferred compared to ASK and FSK because it has minimum bandwidth and \( P_t \).

**Example**: Consider an AM system with additive thermal noise having a power spectral density \( n_0 = 10^{-10} \text{ W/Hz} \). Assume the baseband message signal \( X(0) \) has a bandwidth of 4 kHz and the amplitude distribution shown figure. The signal is demodulated by envelope detection and appropriate post-detection filtering. Assume \( \mu = 1 \).

(a) Find the minimum value of the carrier amplitude \( A_c \) that will yield \( S/N_0 \geq 40 \) dB.
(b) Find the threshold value of \( A_c \).

-1 \hspace{1cm} 0 \hspace{1cm} 1 \hspace{1cm} \lambda

\[
S_X = \int_0^\infty X^2 \Delta t = \int_0^\infty x^2 f(x) dx = \frac{1}{6} \int_0^\infty x^2 (x+1) dx = \frac{1}{6}
\]
\[
\frac{S_X}{1+S_X} = \gamma = \frac{1/6}{\gamma} = (1/7) \gamma \geq 10^6
\]
11. In ASK, the threshold level is a) Independent of carrier amplitude A_c b) Function of A_c c) None

12. The OAM is a combination of the a) ASK and FSK b) ASK and DS c) FSK and FSK

13. The disadvantage of coherent FSK detection is that a) It leads to high signal fading b) It requires two synchronized oscillators c) Both these d) None

14. FSK is preferred to ASK in applications where a) Fading of signal is prevalent b) Synchronized detection is not feasible c) Both of these d) None

15. The disadvantage of FSK is that a) It does not provide sufficient S/N ratio b) It does not have low error probability c) It is not efficient in use of spectrum space d) None

16. In contrast to analog transmission, digital systems can employ a) Quantization b) Error control c) Both of these d) None

17. In coherent detection in digital transmission, the received waveform is a) Compared with a reference waveform b) Compared with the possible transmitted waveform c) None of these

18. In ASK, the transmission bandwidth is equal to a) Baseband bandwidth b) 1.2 times baseband bandwidth c) Four times baseband bandwidth d) None

19. The total bandwidth required for a raised-cosine spectrum is a) 3W/2 b) W c3/2W d) 4W

20. In a-ary PSK channel, the information rate (per bit/sec) can easily be increased by a) Increasing a b) Decreasing a c) None

Key:

- 01: a 02: c 03: a 04: b 05: b 06: a
- 07: a 08: b 09: c 10: a 11: b 12: a
- 19: a 20: a

Chapter - 5 B & C

Objective Questions

01. Consider a Binary Symmetric Channel (BSC) with probability of error being p. To transmit a bit, any 2, we transmit a sequence of three bits. The receiver will interpret the received sequence to represent 1 if at least two bits are 1. The probability that the transmitted bit will be received in error is:

   (a) p^2 + 3p^3 (1-p)  (b) p^3
   (c) 3p^2 (1-p)  (d) p^3 + 3p^2 (1-p)

02. The raised-cosine pulse p(t) is used for error ISI in digital communications. The expression for p(t) with unity roll-off factor is given by

   \[ p(t) = \frac{\sin 4 \pi W t}{4 \pi W (1 - 16 W^2 t^2)} \]

   The value of p(0.5) at t = 1/4W is

   (a) -0.5  (b) 0  (c) 0.5  (d) \infty

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

   (a) p^n  (b) 1 - p^n  (c) \sum (1-p)^{n-1} (p^n)  (d) 1 - (1-p)^n

Common data for Questions: 04, 05:

Two 4 - ary signal constellations are shown. It is given that \( a_1 \) and \( a_2 \) constitute an orthonormal basis of the two constellations. Assume that the four symbols in both the constellations are equiprobable. Let \( N_0 / 2 \) denote the power spectral density of white Gaussian noise.

Following statements are true?

(a) Probability of symbol error for constellation 1 is lower
(b) Probability of symbol error for constellation 1 is higher
(c) Probability of symbol error is equal for both the constellations
(d) The value of \( N_0 \) will determine which of the two constellations has a lower probability of symbol error
09. Consider a binary digital communication system with equally likely 0's and 1's. When binary 0 is transmitted, the voltage at the detector input can lie between the levels -0.25 V and +0.25 V with equal probability. When binary 1 is transmitted, the voltage at the detector can have any value between 0 and 1 V with equal probability. If the detector has a threshold of 0.5 V (i.e., if the received signal is greater than 0.2 V, the bit is taken as 1), the average bit error probability is:

a) 0.15  
b) 0.2  
c) 0.05  
d) 0.5

10. Choose the correct one from among the alternatives A, B, C, and D which refers to the following as applied to a matched filter. Which of the following does represent the output of this matched filter?

11. A source produces binary data at the rate of 10 Kbps. The binary symbols are represented as shown in fig.

The source output is transmitted using two modulation schemes, namely Binary PSK (BPSK) and Quadrature PSK (QPSK). Let B1 and B2 be the bandwidth requirements of BPSK and QPSK, respectively. Assuming that the bandwidth of the above rectangular pulses is 10 KHz, B1 and B2 are:

a) B1 = 20 KHz, B2 = 20 KHz  
b) B1 = 10 KHz, B2 = 20 KHz  
c) B1 = 20 KHz, B2 = 10 KHz  
d) B1 = 10 KHz, B2 = 10 KHz

12. If E0 is the energy per bit of a binary digital signal, it is 10^-4 watt sec and the one-sided power spectral density of the white noise, N0 = 10^-3 W/Hz. Then, the output SNR of the matched filter is:

a) 26 dB  
b) 10 dB  
c) 0 dB  
d) 13 dB

13. If S represents the carrier synchronization of the receiver and p represents the bandwidth efficiency, then the correct statement for the coherent binary PSK is:

a) p = 0.5, S is required  
b) p = 1.0, S is required  
c) p = 0.5, S is not required  
d) p = 1.0, S is not required

14. At a given probability of error, binary coherent PSK is inferior to binary non-coherent PSK by:

(a) 6 dB  
(b) 3 dB  
(c) 2 dB  
(d) 0 dB

15. For a bit-rate of 4 Kbps, the best possible values of the transmitted frequencies in a coherent binary FSK system are:

a) 16 KHz and 20 KHz  
b) 20 KHz and 32 KHz  
c) 20 KHz and 40 KHz  
d) 32 KHz and 40 KHz

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

a) 1 - (1 - p)^k  
b) p + (n-k) (1 - p)  
c) n p (1 - p)^n-1  
d) (1 - p)^k + n p (1 - p)^n-1

17. In a digital communications system employing Frequency Shift Keying (FSK), the 0 and 1 bit are represented by sine waves of 10 KHz and 25 KHz respectively. These waveforms will be orthogonal for a bit interval of:

a) 45 μsec  
b) 200 μsec  
c) 50 μsec  
d) 250 μsec

18. The input to a matched filter is given by

\[ x(t) = \begin{cases} \sin(2\pi f_0 t), & 0 < t < T_0^{1/2} \\ 0, & \text{otherwise} \end{cases} \]

The peak amplitude of the filter output is:

a) 10 volts  
b) 5 volts  
c) 10 millivolts  
d) 5 millivolts
Communication Systems

19. For a given date rate, the bandwidth $B_o$ of a BPSK signal and the bandwidth $B_p$ of the OOK signal are related as:
   a) $B_o = B_p / 4$
   b) $B_o = B_p / 2$
   c) $B_o = B_p$
   d) $B_o = 2B_p$

20. The line code that has zero dc component for pulse transmission of random binary data is:
   a) Non-return to zero (NRZ)
   b) Return to zero (RZ)
   c) Alternate Mark Inversion (AMI)
   d) None of the above

21. A PAM signal can be detected by using:
   a) an ADC
   b) an integrator
   c) a band pass filter
   d) a high pass filter

22. Match the following:
   a) SSB 1) Envelope detector
      b) AM 2) Integrate and dump
      c) DPSK 3) Hilbert transform
      d) Ratio detector
      e) PLL

23. The bit stream 0100 is differentially encoded using 'Delay and EXOR' scheme for DPSK transmission. Assuming the reference bit is a '1' and assigning phases of '0' and 'π' for '1's and '0' respectively, in the encoded sequence, the transmitted phase sequence becomes:
   a) 0 0 π 0
   b) 0 0 π 0
   c) 0 π π 0
   d) π 0 π 0

24. Coherent demodulation of FSK signal can be effected using:
   a) correlation receiver
   b) bandpass filters and envelope detector
   c) matched filter
   d) discriminator detection

Chapter 6
Basics of TDMA, FDMA, & CDMA and GSM

Objective Questions

01. Four message bands limited to W, W/3 and W/6 respectively are to be multiplexed using Time Division Multiplexing (TDMA). The minimum bandwidth required for transmission of this TDMA signal is:
   a) W
   b) 2W
   c) 6W
   d) 7W

02. In a GSM system, 8 channels can co-exist in 200 kHz bandwidth using TDMA. A GSM based cellular operator is allocated 2 MHz bandwidth. Assuming a reuse factor of $\frac{1}{5}$, i.e. a five-cell repeat pattern, the maximum number of simultaneous channels that can exist in one cell is:
   a) 200
   b) 40
   c) 25
   d) 5

03. In a Direct Sequence CDMA system, the chip rate is 1.2288 x 10^9 chips per second. If the processing gain is desired to be AT LEAST 100, the data rate is:
   a) must be less than or equal to 12.288 x 10^3 bits per second
   b) must be greater than 12.288 x 10^3 bits per second
   c) must be exactly equal to 12.288 x 10^3 bits per second
   d) may be different from FDM and TDM

04. Quadrature multiplexing is:
   a) the same as FDM
   b) the same as TDM
   c) a combination of FDM and TDM
   d) quite different from FDM and TDM

05. Four independent messages have bandwidths of 100 Hz, 100 Hz, 200 Hz, and 400 Hz, respectively. Each is sampled at the Nyquist rate, and the samples are transmitted using frequency division multiplexing (FDM). The transmitted sample rate is 1600 Hz. The possible sample rates are:
   a) 1600 Hz
   b) 800 Hz
   c) 400 Hz
   d) 200 Hz

06. The analog signals, having bandwidths 1200 Hz, 600 Hz and 600 Hz, are sampled at their respective Nyquist rates, encoded with 12 bit words, and time division multiplexed. The bit rate for the multiplexed signal is:
   a) 115.2 kbps
   b) 28.8 kbps
   c) 7.6 kbps
   d) 18.4 kbps

07. In a noisy channel, the probability of an error is p. The probability of at least one error in the transmission of an 8-bit sequence is:
   a) $7(1-p)^7 + p^8$
   b) $7p^7 + 7p^8$
   c) $(1-p)^7 + 7p^8$
   d) $(1-p)^7 + 8p(1-p)^7$
10. Four signals each band—limited to 5 kHz are sampled at twice the Nyquist rate. The resulting PAM samples are transmitted over a single channel in time-division multiplexing. The theoretical minimum transmission bandwidth of the channel should equal to
(a) 20 kHz
(b) 40 kHz
(c) 60 kHz
(d) 80 kHz

11. In a certain '12 channel TDM' system, it is found that channel No. 3 and channel No. 8 are co-hosted to the same input signal. The technique (a) wastes the channel capacity (b) takes care of different sampling rates (c) is required when different bandwidth signals are to be transmitted (d) reduces noise

12. In asynchronous TDM, for n signal sources, each frame contains m 'slots', where m is usually
(a) less than n
(b) 2 n
(c) m = n
(d) greater than 2 n

13. In a cellular communications system, nth least between transmitter and receiver is due to
(a) scattering from buildings, trees, vehicles, and other structures only
(b) reason at (a) above and due to reflection from ground only
(c) reason at (a) and (b) above along with reflection from ionosphere only
(d) reasons at (a), (b), and (c) above along with less due to surface wave phenomenon

14. The number of signalling bits per channel per frame in T1 multiplexer following CCITT hierarchy is
(a) 64000
(b) 128
(c) 4
(d) 400

15. Four signals \( g_1, g_2, g_3, g_4 \) and \( g_5, g_6, g_7, g_8 \) are to be multiplexed and transmitted. \( g_1, g_2, g_3, g_4 \) have a bandwidth of 4 kHz, and the remaining two signals have bandwidth of 8 kHz. Each sample requires 8 bit encoding. What is the minimum transmission bit rate of the system?
(a) 512 kbps
(b) 16 kbps
(c) 192 kbps
(d) 384 kbps

16. A 12 channel TDM system where each channel signal is 4 kHz is sampled at 8 kHz. What is the bandwidth requirement?
(a) 12 kHz
(b) 12 x 4 kHz
(c) 12 x 8 kHz
(d) 12 x 8 x 4 kHz

17. Assertion (A): TDM can be employed to transmit channels having unequal bandwidths.
Reason (R): If sampling theorem is strictly followed, any analog signal can be reconstructed from its samples.
(a) Both A and R are true and R is the correct explanation of A
(b) Both A and R are true but R is NOT the correct explanation of A
(c) A is true but R is false
(d) A is false but R is true

18. In DFM systems used for telephone, which modulation scheme is adopted?
(a) AM
(b) DSB - SC
(c) SSB - SC
(d) FM

19. Which is the most important sub-system for recovering and reconstructing signals in a TDM system?
(a) Envelope detector followed by a low pass filter
(b) Synchronization circuit for proper timing
(c) Bandpass filters to separate channels
(d) Coherent detector to ensure frequency and phase connection

20. Multiplexing is possible if signals are sampled. Two signals have bandwidth A = 0 to 4 kHz and B = 6 to 8 kHz respectively. The sampling frequency chosen is 12 kHz. Which one of the following is correct? This choice of the sampling frequency
(a) is correct since A and B have an integral relationship of 2
(b) will not lead to aliasing
(c) does not obey sampling theorem
(d) can never lead to multiplexing

21. Which one of the following is correct? In a TDM system each signal is allotted in a frame a unique and fixed
(a) frequency slot
(b) time slot
(c) amplitude slot
(d) phase slot
To win the RACE join the ACE
*ALL INDIA 1st RANK 17 TIMES IN GATE*
*ALL INDIA 2nd RANK 10 TIMES IN GATE*

Offers excellent coaching for:
GATE, IES, JTO, APGENCO, APTRANSCO, APPSC, CPWD etc.

For further details contact:

ACE ENGINEERING ACADEMY
Head Office:
36A, 2nd Floor, Saint Mary's School, Neue Taj Mahal Hotel
Opp. Methodist School New Gate,
Fernando Hospital lane,
ABIDS, HYDERABAD - 500 001
Ph: 040 - 24752689, 2475457

Branch Office:
2nd Floor, 314 A & B
Pancom Business Centr,
Anampet,
Ooty Road - 580 016
Ph: 040 - 6974465

www.aceengineeringacademy.com email: ace.gateguru@gmail.com

"All the best"