St.Joseph's College Of Engineering

Department of ECE

CS6304 ANALOG AND DIGITAL COMMUNICATION

Question Bank

UNIT I: ANALOG COMMUNICATION- PART-A

1. Define noise.

Noise is defined as any unwanted form of energy, which tends to interfere with proper reception and reproduction of wanted signal.

2. **Give the classification of noise.**

Noise is broadly classified into two types. They are External noise and internal noise.

3. What are the types of external noise?

External noise can be classified into Atmospheric noise, Extraterrestrial noises,Man –made noises or industrial noises.

4. What are types of internal noise?

Internal noise can be classified into Thermal noise,Shot noise,Transit time noise, Miscellaneous internal noise.

5. What are the types of extraterrestrial noise and write their origin.

The two type of extraterrestrial noise are solar noise and cosmic noise Solar noise is the electrical noise emanating from the sun.Cosmic noise is the noise received from the center part of our galaxy, other distant galaxies and other virtual point sources.

6. Define transit time of a transistor.

Transit time is defined as the time taken by the electron to travel from emitter to the collector.

7. Define flicker noise.

Flicker noise is the one appearing in transistors operating at low audio frequencies. Flicker noise is proportional to the emitter current and junction temperature and inversely proportional to the frequency.

8. Define noise figure.

Noise figure F =S/ N at the output / S/ N at the input ; S/N = Signal power / Noise Power

9. Explain thermal noise.

Thermal noise is the name given to the electrical noise arising from the random motion of electrons in a conductor.

10. Give the expression for noise voltage in a resistor.

The mean –square value of thermal noise voltage is given by Vn2 = 4 K T B R'K – Boltz man constant – resistance – Absolute temperature - , Bandwidth

11. Explain White Noise.

Many types of noise sources are Gaussian and have flat spectral density over a wide frequency range. Such spectrum has all frequency components in equal portion, and is therefore called white noise. The power spectral density of white noise is independent of the operating frequency.

12. Calculate the thermal noise voltage across the simple RC circuit shown with R = 1 kilo Ohm and C = 1 micro Farad at T= 27 degree C.

For a simple RC filter, the equivalent noise bandwidth is:

$$BW_n = \frac{\pi}{2} f_{-3dB} \quad [Hz]$$

$$BW_n = \frac{\pi}{2} \frac{1}{2\pi RC} = \frac{1}{4RC} \quad [Hz]$$

Therefore the total integrated noise equals to:

$$\overline{V^2}_o = \frac{\overline{V^2}_R}{\Delta f} BW_n = 4kTR \cdot \frac{1}{4RC} = \frac{kT}{C}$$

Thermal Noise Voltage $Vn^2 = 4\ K\ T\ R\ (B.W)$
$Vn^2 = 4KTR.(1/4RC)$
$Vn^2 = KT/C$
$=(1.38*10^{-23})(273+27)/(1*10^{-6})$
$=4.14*10^{-23}/(1*10^{-6}) = 4.14*10^{-17}$ volts.

13. What is the need for modulation?

It is extremely difficult to radiate low frequency signals through earth's atmosphere in the form of electromagnetic energy.

At low frequency, the antenna size required becomes impractical.

Information signals often occupy the same frequency band. Signals from two or more sources would interfere if they are not modulated and translated to a different frequency band.

14. With reference to AM ,define modulation index (or) depth of modulation.

It is defined as the ratio of peak amplitude of the message to the carrier signal.

$m = \frac{E_m}{E_c}$ E_m = peak amplitude of modulating signal voltage

E_c = peak amplitude of the unmodulated carrier voltage

15. A broadcast radio transmitter radiates 5 kW power when the modulation percentage is 60%. How much is the carrier power?

$Pt = Pc(1+m^2/2) = 5000/(1+0.6^2/2) = 4237.28w$

16. What is the relationship between total power in AM wave and unmodulated carrier power?

$Pt = Pc(1+m^2/2)$

Pc=unmodulated carrier power; Pt=total power; m=modulation index

17. What is the relationship between total current in AM wave and unmodulated carrier current?

$It = Ic(1+m^2/2)$

Ic= carrier current; It=total current; m=modulation index

18. An unmodulated carrier is modulated simultaneously by three modulating signals with coefficients of modulation m1 = 0.2, m2 = 0.4, m3 = 0.5. Determine the total coefficient of modulation.

$mt = \sqrt{m_1^2 + m_2^2 + m_3^2} = \sqrt{0.2^2+0.4^2+0.5^2} = 0.67$

19. Define amplitude Modulation.(Dec'13)

Amplitude Modulation is the process of changing the amplitude of a relatively high frequency carrier signal in proportion with the instantaneous value of the modulating signal.

20. Define Modulation index and percent modulation for an AM wave.

Modulation index is a term used to describe the amount of amplitude change present in an AM waveform .It is also called as coefficient of modulation. Mathematically modulation index is

$m = Em/Ec$

Where m = Modulation coefficient

Em = Peak change in the amplitude of the output waveform voltage.

Ec = Peak amplitude of the unmodulated carrier voltage.

Percent modulation gives the percentage change in the amplitude of the outputwave when the carrier is acted on by a modulating signal.

21. What is Frequency modulation?

Frequency of carrier is varied in accordance with amplitude of modulating signal.

22. What is Phase modulation?

Phase of carrier is varied in accordance with the amplitude of modulating signal.

23. What is Bandwidth of AM wave?

Band width is difference between highest upper side frequency and lowest lower side frequency. B.W = 2fm(max).

24. What is over,under,critical modulation?

If m >1, has severe distortion.This condition is Over modulation. If m=1, has greatest output and condition is Critical modulation. If m< 1 ,has no distortion and condition is Under modulation.

25. Draw AM envelope with Vmax and Vmin?

26. With reference to FM, define modulation index.
Modulation index is the ratio of frequency deviation and modulating signal frequency
$m = \Delta f \;/\; f_m$
Δf = frequency deviation in Hz
f_m = modulating signal frequency in Hz

27. Define deviation ratio.
It is the worst-case modulation index which is the ratio of maximum permitted frequency deviation and maximum modulating signal frequency.
Deviation ratio = $\Delta f_{(max)} \;/\; f_{m(max)}$

28.State Carson's rule for determining approximate Band Width of FM signal.
Carson rule states that the bandwidth required to transmit an angle modulated wave as twice the sum of the peak frequency deviation and the highest modulating signal frequency.
Band Width = 2 [$\Delta f + f_{m(max)}$ **] Hz**
Δf = frequency deviation in Hz $f_{m(max)}$ = highest modulating signal frequency in Hz

29.A carrier is frequency is frequency modulated with a sinusoidal signal of 2 KHz resulting in a maximum frequency deviation of 5 KHz. Find the approximate band width of the modulated signal.
Δf = frequency deviation in Hz = 5 KHz
$f_{m(max)}$ = highest modulating signal frequency in Hz = 2 KHz
Band Width = 2 [$\Delta f + f_{m(max)}$] Hz = 14 KHz

30. Determine the modulation index of a FM system with a maximum frequency deviation of 75 KHz and maximum modulating frequency of 10 KHz.
$m = \Delta f \;/\; f_m$ = 75 KHz / 10 KHz = 7.5

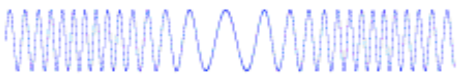
31. Distinguish between narrow band FM and wide band FM.(Dec'13)

Narrow band FM	Wide band FM
Frequency deviation in carrier frequency is very small	Frequency deviation in carrier frequency is large
Band width is twice the highest modulating frequency	Band width is calculated as per Carson's rule

32. What are the advantages of FM over AM?
The amplitude of FM is constant. Hence transmitter power remains constant in FM where as it varies in AM.
Since amplitude of FM is constant, the noise interference is minimum in FM. Any noise superimposing on modulated carrier can be removed with the help of amplitude limiter.
The depth of modulation have limitation in AM. But in FM, the depth of modulation can be increased to any value.
Since guard bands are provided in FM, there is less possibility of adjacent channel interference.
Since space waves are used for FM, the radius of propagation is limited to line of sight(LOS) . Hence it is possible to operate several independent transmitters on same frequency with minimum interference.
Since FM uses UHF and VHF ranges, the noise interference is minimum compared to AM which uses MF and HF ranges.

33. What is the advantage and disadvantage of Angle modulation?
Advantages: 1. Noise Reduction 2. Improved system fidelity 3. More effective use of power
Disadvantage: 1. Require more Bandwidth 2. Use more complex circuits in both transmitter and receiver

34. Draw the FM waveform? (June'13)



35. Define percent modulation?

Percent modulation = [actual frerquency deviation/max allowable frequency deviation] *(100)

36.A Tx supplies 8KW to the antenna when unmodulated. Determine the total power when amplitude modulate to 30%.

$P_t=P_c(1+m_a^2/2)$

$=8x10^3(1+0.3^2/2)$

37. Determine the modulation depth of FM system with a maximum frequency deviation of 75 KHz and the maximum modulating frequency of 10 KHz

$m_f= \Delta f/f_m$

$=75\ x10^3/10\ x10^3$

$= 7.5$

38. Write down the expression for FM signal with sinusoidal modulation

Frequency modulation :It is the form of angle modulation in which instantaneous frequency $f_I(t)$ is varied linearly with the base band signal m(t) Where,$f_I(t)=f_c+k_f\,m(t)$ f_c unmodulated carrier

k_f –Frequency sensitivity of the modulator m(t)-Base band signal

integrating above equation w. r .t time and multiplying with 2π

$\theta_i(t)= 2\pi fc\,t+2\pi K_f\int m(t)\,dt$

$s(t)=A_c\cos\theta_i(t)$

$s(t)= A_c\cos(2\pi fc\,t+2\pi K_f\int m(t)\,dt)$

39.Define instantaneous frequency deviation.

The instantaneous frequency deviation is the instantaneous change in the frequency of the carrier and is defined as the first derivative of the instantaneous phase deviation.

40. What is demodulation?

The process of recovering the message or information signal from the high frequency modulated signal is called demodulation.

<u>PART B</u>:

1. Obtain AM wave equation and explain each term with the help of frequency spectrum and also obtain an expression for its power?

   In amplitude modulation, amplitude of the carrier wave is changed according to the amplitude of the signal. The technique is very much used in the transmission of radio signals.

   Figure shows the process of amplitude

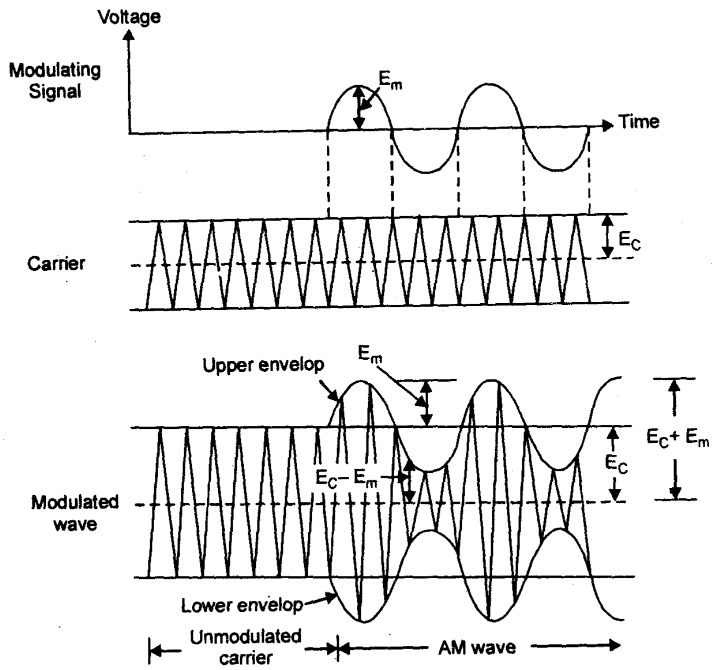


Fig. 4

modulation.

Note that

Only amplitude of the carrier is varied while its frequency and phase remain unchanged.

Whenthere is no signal, the amplitude of the carrier is equal to the unmodulated amplitude. When signal is present, the amplitude of the carrier changes in accordance with the instantaneous value of the signal.

During positive cycle of the signal, the amplitude of the carrier increases to the sum of the amplitudes of the carrier and the signal (E + Em)

(iv) During negative cycle of the signal, the amplitude of the carrier decreases and becomes equal to the difference of the amplitudes of the carrier and the signal.

Ans. Let carrier representation be

$$e_c = E_c \sin \omega_c t$$

and let signal be represented by

$$e_m = E_m \sin \omega_m t$$

Let A be the amplitude of the modulated radio wave. Then

$$A = E_c + e_m$$
$$= E_c + E_m \sin \omega_m t$$
$$= E_c + mE_c \sin w_m t \; (E_m = mE_c)$$
$$= E_c (1 + m \sin \omega_m t)$$

Let the voltage equation of the output modulated wave be :

$$e = A. \sin \omega_c t$$

2.

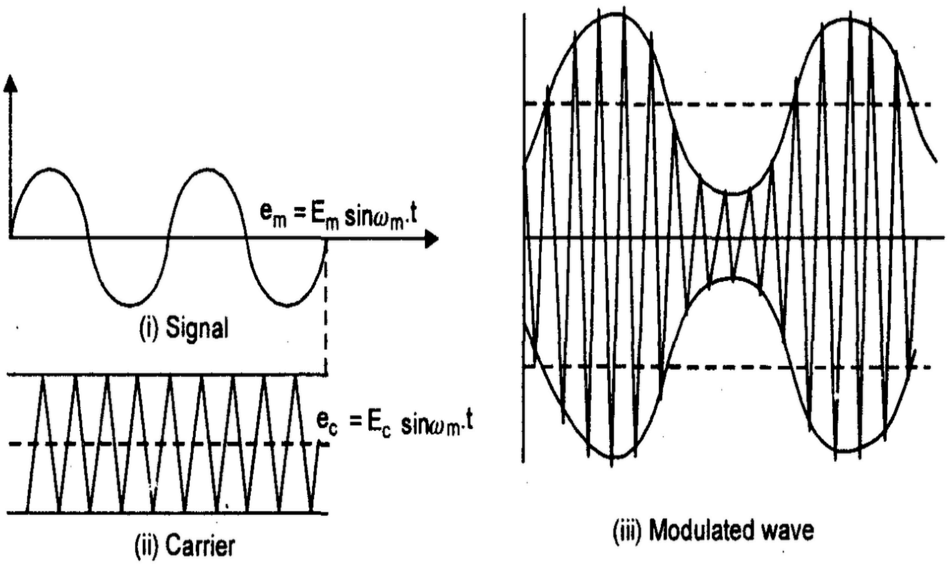


Fig. 5

3.

Putting the value from Equation (*i*)

$$e = E_c(1 + m \sin\omega_m t).\sin w_c t$$

This is the standard equation for the A.M. radio wave.

Solving further $\quad e = E_c \sin \omega_c t + mE_c \sin \omega_c t \sin\omega_m t$

Note that $\quad E_m = mE_c$, where m is the modulating factor.

Solving again, this comes to be

$$e = E_c \sin\omega_c .t + \frac{mE_c}{2}(2 \sin\omega_c .t \sin\omega_m t)$$

[For the bracketed part, use the formula = 2 sin A sin B = cos (A-B)-cos(A + B)]

Then, we get $\quad e = E_c \sin\omega_c t + \frac{mE_c}{2}\cos(\omega_c - \omega_m)t - \frac{mE_c}{2}\cos(\omega_c + \omega_m)t \quad ...(iii)$

This is the equation of the modulated wave.

Note that the equation has three parts:
First part is an unmodulated carrier wave, which remains unchanged in the process. The maximum amplitude is E.
Second part has a maximum amplitude of mE/2 and its frequency is equal to the difference of carrier and the signal frequencies. This is called lower side band (L.S.B.). Recall that angular velocity of the carrier o 2rf, where f is the frequency of the carrier. Similarly tOrn = fm
Third part has also max. amplitude of mE/2 and frequency equal to the sum of carrier and signal frequencies. This is called upper side band (U.S.B.) (See Figure 6b).

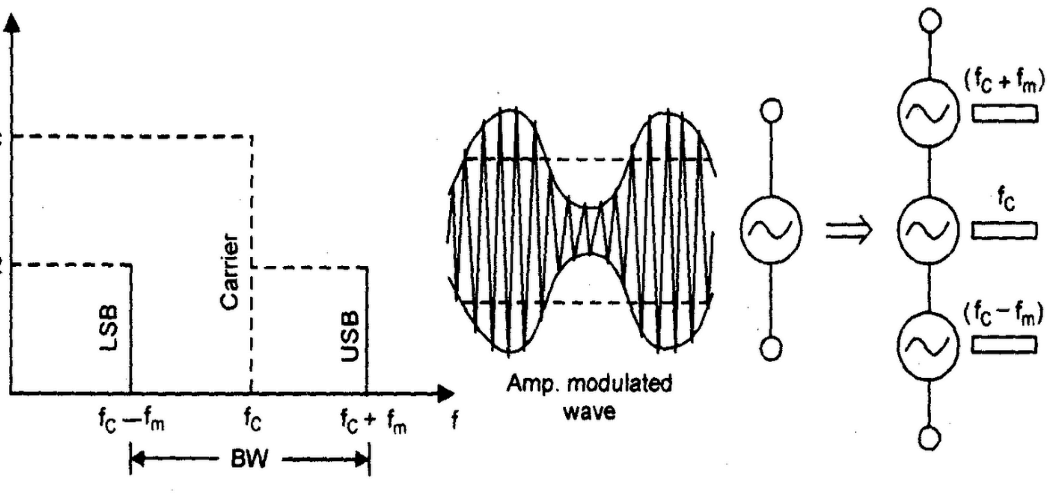


Fig. 6 (b)

Figure 6 shows frequency spectrum of an A.M. wave which is equivalent to three sine waves as shown. Bandwidth (B.W.) of an A.M. wave

$$\text{B.W.} = (f_c + f_m) - (f_c - f_m) = 2f_m$$

[e.g., if $f_c = 100$ kHz, and $f_m = 1$ kHz,

the B.W. $= (100 + 1) - (100 - 1) = 101 - 99 = 2\text{kHz}$ [$= 2f_m$].

2. i)What is the need for modulation? ii) Explain with necessary diagram any one method for generation of AM waves. (June' 13).

Need for Modulation:

Easy transmission
Narrow Banding
Multiplexing
To overcome equipment limitation
Reduce noise & interference.
Improves quality of reception.

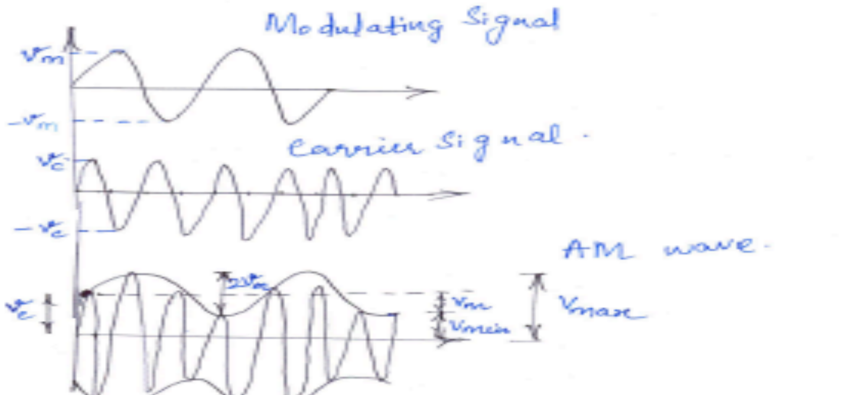
Generation of AM:
Square law modulator------
It consits of summer,non linear element and filter.
Example for non linear devices are diode ,transistor and FET.
It obeys square law equation.

3. Obtain the AM wave equation and draw the AM Envelope. Also explain modulation index and percent modulation in terms of Vmax and Vmin.

Bandwidth: It is defined as the frequency range over which an information signal is transmitted.
∴ It is the difference between the upper and lower frequency limits of the signal.

$$BW = f_H - f_L \quad \text{or} \quad \omega_H - \omega_L$$
$$= 2f_m \quad \text{or} \quad 2\omega_m.$$

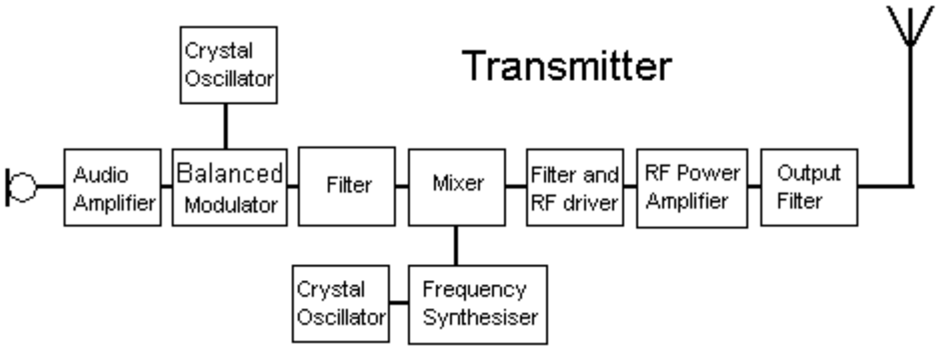
Graphical Representation:



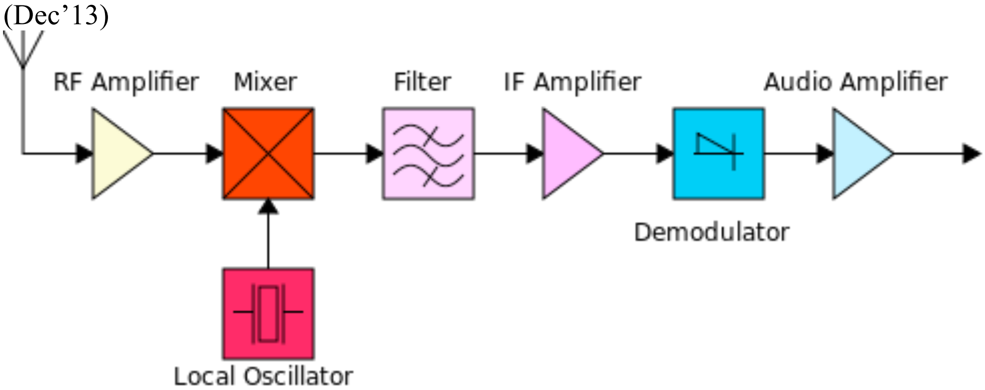
From the diagram,

$$* \; 2V_m = V_{max} - V_{min}$$
$$V_m = \frac{V_{max} - V_{min}}{2}$$
$$* \; V_c = V_m + V_{min} = \frac{V_{max} + V_{min}}{2}$$
$$\boxed{* \; m_a = \frac{V_m}{V_c} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}}$$

4. Explain with block diagram of a Fm transmitter using direct modulation. ii) Discuss about spectral characteristics of FM signal. (June' 13).



5. Draw the block diagram of AM superhetrodyne receiver and explain function of each block. (Dec'13)



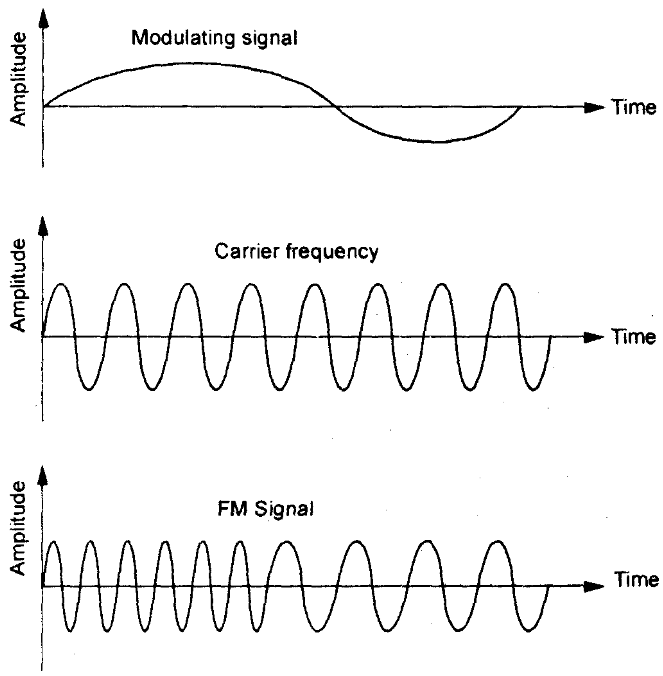
6. An AM modulator has a carrier of 400 KHz with amplitude of 20v; modulating signal of 8 KHz with amplitude of 8.5v is applied. Determine

(a) Upper and lower side frequencies. b)Modulation coefficient and percent modulation
(b) Peak amplitude of the modulated carrier and upper and lower side frequency voltages.
(c) Maximum and minimum amplitude of the envelope.
(d) Expression of modulated wave.
(e) Sketch the output spectrum and envelope.

ANS: fUSB= 408khz fLSB= 392khz
ma=0.425 Peak amplitude of LSB =USB=4.25v
Vmax=28.5 v
Vmin=11.5v

7. Write down the expression for FM and PM waves and draw their frequency spectrum and explain.

Frequency Modulation (FM) In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level amplitude of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.

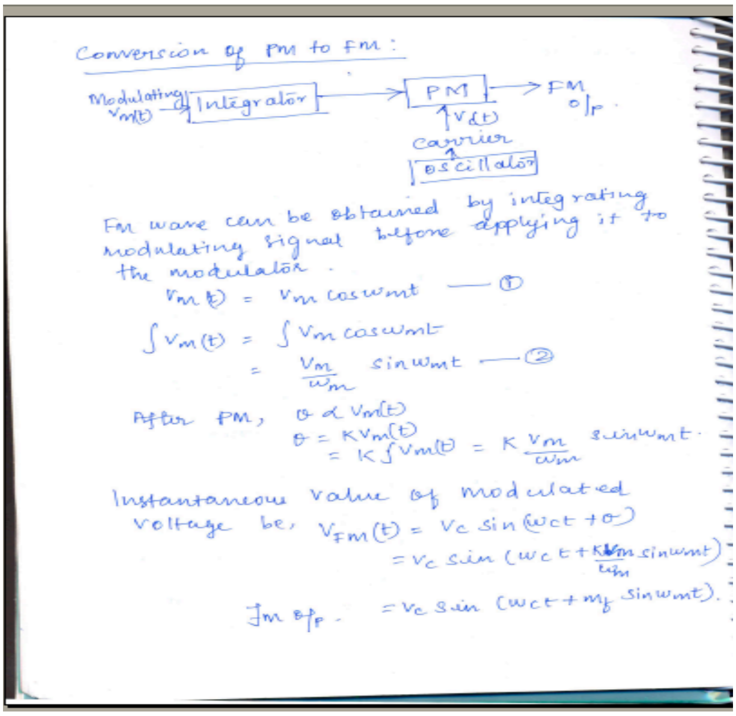


Phase Modulation: PM is used in some systems as an alternative to frequency modulation in PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level of the modulating signal. The peak amplitude and frequency of the carrier signal remains constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly.

8 .Obtain the mathematical expressions for AM & FM modulated waves & draw the necessary waveforms in both cases.
Refer q.No.1

6. Compare AM, FM and PM systems.

7. Describe suitable mechanism that can produce PM from FM Modulator and FM from PM modulator.



FM wave can be obtained by integrating modulating signal before applying it to the modulator.

$v_m(t) = V_m \cos w_m t$ —— ①

$\int v_m(t) = \int V_m \cos w_m t$

$= \frac{V_m}{w_m} \sin w_m t$ —— ②

After PM, $\theta \propto V_m(t)$

$\theta = K V_m(t)$

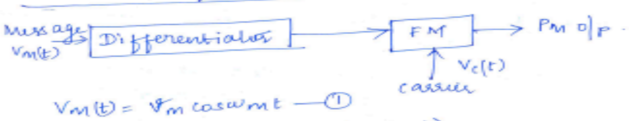
$= K \int V_m(t) = K \frac{V_m}{w_m} \sin w_m t$.

Instantaneous value of modulated voltage be, $V_{FM}(t) = V_c \sin(w_c t + \theta)$

$= V_c \sin(w_c t + \frac{K V_m}{w_m} \sin w_m t)$

Fm o/p. $= V_c \sin(w_c t + m_f \sin w_m t)$.

Conversion of FM to PM:



$V_m(t) = V_m \cos\omega_m t$ — (1)

$\frac{d}{dt} v_m(t) = V_m \omega_m (-\sin\omega_m t)$

$= -V_m \omega_m \sin\omega_m t$

After FM, $\omega_i = \omega_c + \left(\frac{d}{dt} V_m(t)\right) \cdot K$

$= \omega_c - K V_m \omega_m \sin\omega_m t$

$\varphi_i = \int \omega_i \, dt$

$= \int (\omega_c - K V_m \omega_m \sin\omega_m t)\, dt$

$= \omega_c t + K V_m \frac{\cancel{\omega_m}}{\cancel{\omega_m}} \cos\omega_m t$

$= \omega_c t + K V_m \cos\omega_m t$

Instantaneous signal voltage after modulation is $V_{pm}(t) = V_c \sin\varphi_i$

$v_{pm}(t) = v_c \sin(\omega_c t + m_p \cos\omega_m t)$

$m_p \rightarrow$ mod. index of PM.

8. Explain the mathematical analysis of angle modulated wave.

PHASE MODULATION:

Def :- It is defined as the process by which changing the phase of the carrier signal inaccordance with the instantaneous amplitude of the message.

Let the modulating signal be $V_m(t) = v_m \cos\omega_m t$ — (1)

" carrier " $V_c(t) = v_c \sin(\omega_c t + \theta)$ — (2)

θ – phase angle of carrier.

According to def, Phase is changed,

$\theta \propto V_m(t)$

$\theta = K V_m(t)$

$= K V_m \cos\omega_m t$

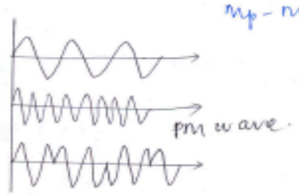
After phase modulation,

$V_{pm}(t) = v_c \sin(\omega_c t + \theta)$

$= v_c \sin(\omega_c t + K V_m \cos\omega_m t)$

$= v_c \sin(\omega_c t + m_p \cos\omega_m t)$

m_p – modulation index of PM.



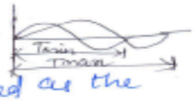
9. Explain the phase deviation and Modulation index of angle modulated wave
   Define all deviations and give their expressions
   Refer Q.10
10. Discuss about frequency deviation and Percentage modulation**.**

Definition :

① Instantaneous Frequency deviation : It is defined as the first time derivative of the instantaneous phase deviation.

② Instantaneous frequency : It is the precise frequency of the carrier at a given instant of time.

③ Frequency deviation : Δf



In FM, the deviation is defined as the amount by which the carrier frequency is varied from its unmodulated value. Magnitude of frequency deviation is proportional to the amplitude of the modulating signal.

④ Phase deviation : (Δθ) : The relative angular displacement of the carrier phase in radians in respect to the reference phase. The change in the carrier phase produces a corresponding change in frequency.

⑤ Instantaneous phase deviation : It is the instantaneous change in the phase of the carrier at a given instant of time & indicates how much phase of the carrier is changing with respect to its reference phase.

11. Write short notes on (i)Shot noise (ii) Thermal noise.
    Def:Shot noise:Arises in electronic devices such as diodes and transistors because of discrete nature of current flow in the devices
    Thermal noise :Arises from the random motion of electrons in a conductor.

Unit II : DIGITAL COMMUNICATION -PART-A

1. Define ASK and FSK. (Dec'13)

ASK: A binary information signal directly modulates the amplitude of an analog carrier.
FSK: The frequency of a sinusoidal carrier is shifted between two discrete values**.**

2. Define bit time and baud rate.

Bit time:It is the reciprocal of the bit rate
Baud rate:The rate of change of a signal on the transmission medium after encoding and modulation have occurred. Baud = 1/ts

3. Define DPSK .

DPSK is an alternative form of digital modulation where the binary input information is contained in the difference between two successive signaling elements rather than absolute phase .It combines two basic operations namely ,differential encoding and phase shift keying.

4. Define QPSK .

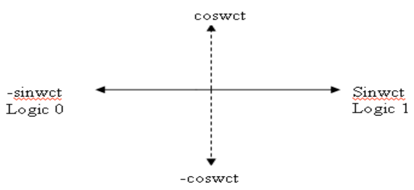
QPSK: The two successive bits in a bit stream ar combined together to form a message and each message is represented by a distinct value of phase shift of the carrier. Each symbol or message contains two bits so the symbol duration Ts =2Tb.
These symbols are transmitted by the same carrier at four different phase shifts as shown below.

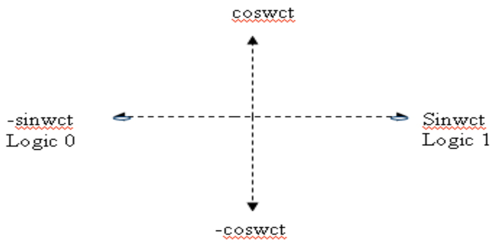
Symbol	Phase
00	-135
01	-45
10	135
11	45

5.What is a constellation diagram? Draw the constellation diagram and phasor diagram for BPSK.

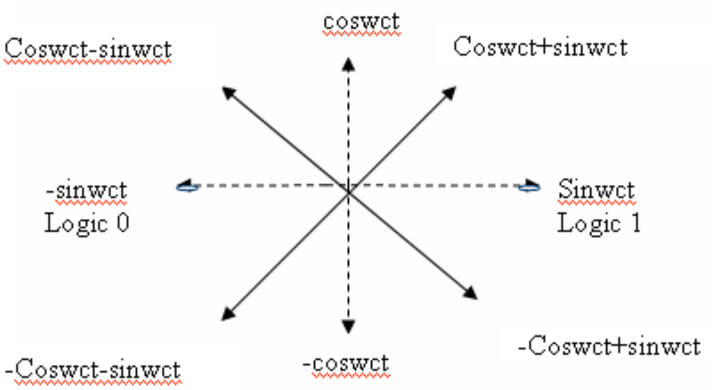
Constellation diagram is used to show the relative positions of the peaks of the phasors.

Phasor diagram: constellation diagram





6.Draw the phasor diagram of QPSK signal.(June'13)



7. What is the minimum bandwidth required for an FSK system?

Bandwidth required=fm-fs+ 2/tb 1/tb =fb=bit rate, fm=mark frequency,fs=space frequency

8. **What is the primary advantage of DBPSK and what is its disadvantage?**

<u>Advantage:</u> simple implementation. No carrier recovery circuit needed for detection.

<u>Disadvantage:</u> It requires between 1 dB and 3 dB more signal to noise ratio to achieve the same BER as that of standard absolute PSK

9. What are the advantages of M-ary signaling schemes?

M-ary signaling schemes transmit multiple bits at a time.

Bandwidth requirement of M-ary signaling schemes is reduced.

10. Compare binary PSK with QPSK.

BPSK Binary Phase Shift Keying	**QPSK** Quadrature Phase Shift Keying
One bit form a symbol	Two bits form a symbol
Two possible symbols	Four possible symbols
Minimum bandwidth required = f_b where f_b is bit rate	Minimum bandwidth required = $f_b / 2$ where f_b is bit rate

11. What are the advantages of QPSK as compared to BPSK?

For the same bit rate, the bandwidth required by QPSK is reduced to half as compared to BPSK. Because of reduced bandwidth, the information transmission rate of QPSK is higher.

12.What happens to the probability of error in M-ary PSK as the value of M increases?

As the value of M increases, the Euclidean distance between the symbols reduces. Hence the symbols are closer to each other. This increases the probability of error in M-ary systems.

13.What is the minimum bandwidth required for BPSK, QPSK, 8-PSK, 8-QAM and16-QAM systems if the bit rate is 10 MBPS?

system	**Minimum band width**	**Minimum band width**

	required if f_b = bit rate	**required** if f_b = 10 Mbps
BPSK	f_b	10 MHz
QPSK	$f_b / 2$	5 MHz
8 – PSK	$f_b / 3$	3.33 MHz
8- QAM	$f_b / 3$	3.33 MHz
16 – QAM	$f_b / 4$	2.5 MHz

14. What is difference between coherent and non coherent detection?

Coherent detection	**Non- Coherent detection**
Carrier which is in perfect coherence with that used in transmitter is used for demodulation. Carrier recovery circuit is needed for detection	No carrier recovery circuit needed for detection.
Relatively complex	Simple implementation

15. Define Bandwidth efficiency. What is the bandwidth efficiency of BPSK and 8-PSK system?

It is the ratio of the transmission bit rate to the minimum bandwidth required for a particular modulation scheme.

$$Bandwidth\,efficiency = \frac{transmission\,rate\ (bps)}{\min imum\,bandwidth\ (Hz)}$$

For BPSK , transmission rate = f_b and minimum bandwidth = f_b

Band width efficiency = 1

For 8-PSK , transmission rate = f_b and minimum bandwidth = $f_b/3$

Band width efficiency = 3

16. What is the difference between probability of error P(e) and bit error rate BER?

P(e) Probability of error is a theoretical (mathematical) expectation of the bit error rate for a given system.BER is an empirical record of a system's actual bit error performance. For Example, if a system has a P(e) of 10^{-5} , this mean that, you can expect one
bit error in every 100,000 bits transmitted. If a system has a BER of 10^{-5} , this mean that, there was one bit error for every 100,000 bits transmitted. BER is measured and then compared to the expected probability of error to evaluate the system's performance.

17. What is the probability of error for (i) non-coherent FSK and (ii) coherent FSK? Compare their error performance.

$$P_e = \frac{1}{2}\exp\left(-\frac{E_b}{2N_0}\right) = probability\,of\,error\,for\,non-coherent\,FSK$$

$$P_e = erfc\left(\frac{E_b}{N_0}\right) = probability\,of\,error\,for\,coherent\,FSK$$

$$\frac{E_b}{N_0} = energy\,per\,bit\,to\,noise\,power\,density\,ratio$$

For a given energy per bit to noise power density ratio, probability of error for non-coherent FSK is greater than that of coherent FSK.

18. Define (E_b / N_0) Energy per bit to Noise power density ratio.

Energy per bit to noise power ratio is used to compare two or more digital modulation systems that uses different bit rates and modulation schemes.

It is the product of carrier to noise power ratio and the noise band width to bit rate ratio. This is equivalent to signal to noise ratio.

$$\frac{E_b}{N_0} = energy per bit to noise power density ratio$$

$$E_b = Energy per bit = C\,T_b$$

$$where\ C = carrier power in watt$$

$$T_b = bit duration in\ sec$$

$$N_0 = Noise power density = \frac{N}{B}$$

$$where\ N = thermal noise power in watt$$

$$B = noise bandwidth in\ Hz$$

$$\frac{E_b}{N_0} = \frac{C}{N} \times \frac{B}{f_b}$$

19. List out the advantages and disadvantages of QPSK.

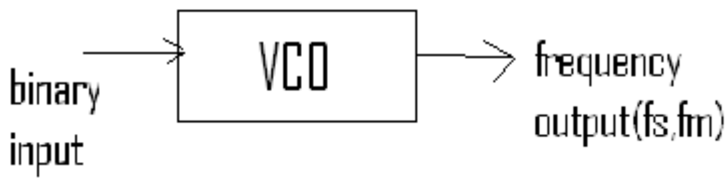
Advantages: low error probability, good noise immunity, baud rate is half of the bit rate

Disadvantages: very complex to generate and detect the signal

20. Define carrier recovery.

It is the process of extracting a phase coherent reference carrier from a received signal.

21. Draw the block diagram of BFSK transmitter. Dec'12



22. Compare QAM and QPSK.

Parameter	QPSK	QAM
Type of modulation	Quadrature phase modulation	Quadrature amplitude and phase modulation
Noise immunity	Better than QASK	Poorer than QPSK
Probabilty of error	Less than QASK	More than QPSK
Type of demodulation	Synchronous	Synchronous
Complexity	Less complex than QASK	More complex than QPSK

23. Why QAM is preferred when compared to other digital to analog modulation technique ?

In all PSK methods, one symbol is distinguished from the other in phase. But all the symbols transmitted are of same amplitude. Noise immunity will improve if the signal vectors differ not only in phase, but also in amplitude. Here, the direct modulation of carriers in quadrature is involved therefore the system is called QAM.

24. List out merits and demerits of ASK.

Merits: Simple technique ,Easy to generate and detect.

Demerits: very sensitive to noise, Used at very low bit rates upto 100 bits/sec.

25. Define frequency deviation.

It is the half difference between mark and space frequencies.

$$\Delta f = \frac{fm - fs}{2}$$

Where Δf—frequency deviation (hz)

fm-fs—absolute difference between mark and space frequencies.

PART B:

1. Explain BPSK Transmitter and receiver with a neat diagram.(Dec'13)

In Binary Phase Shift Keying system, the sinusoidal carrier wave of fixed

frequency f and fixed amplitude A is used to represent binary symbols 0 and 1 except that the phase of carrier for one (1) differs by a phase angle of 180° to the phase of carrier for zero (0).

Let the unmodulated carrier be represented by equation

$$v_c(t) = A_c \cos(2\pi f_c t)$$

Then the binary PSK signal $S(t)$ can be written as

$$S(t) = b(t)\, A_c \cos(2\pi f_c t) \text{ for binary symbol 1}$$
$$= b(t)\, A_c \cos(2\pi f_c t + \pi) \text{ for binary symbol 0}$$

where $b(t)$ is binary data as modulating signal as shown in the figure 36.

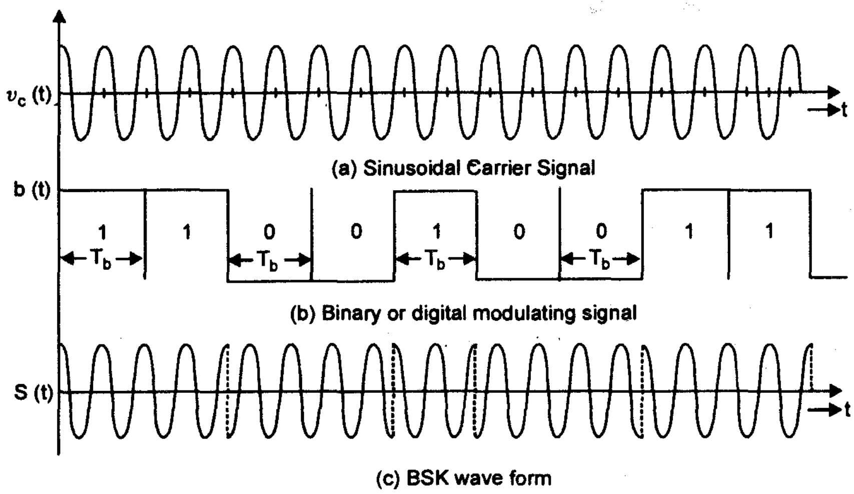


Fig. 36

Here, it may be noted that unlike ASK transmission, the P5K transmission is polar. Polarity changes in the binary signal b(t) are used to produce 1800 changes in the carrier phase. This may be achieved by using double sideband, suppressed carrier modulation (DSBSC), with binary signal as a polar NRZ waveform. The carrier amplitude is multiplied by ± 1, pulsed waveform. When the binary signal b(t) is + 1, the carrier sinusoid is unchanged, and when b(t) is -1, the carrier sinusoid is changed in phase by 180°. Binary phase shift keying is also known as Phase reversal keying (PRK).

2. Explain BFSK Transmitter and receiver with a neat diagram.

Frequency shift keying is the oldest and simplest method of modulation used in modems. Two sinusoidal frequencies are used to represent binary symbols Os and is. For example, a frequency of 1070 Hz for space in data communication is used to represent binary ‘0’ and a frequency 1270 Hz for mark is used to represent binary ‘1’. These two frequencies 1070 Hz and 1270 Hz are used to represent binary signal ‘0’ and ‘1’ respectively for transmission of binary data as shown in figure 34. These two frequency tones are well within the 300 to 3400 Hz band width associated with telephone system.

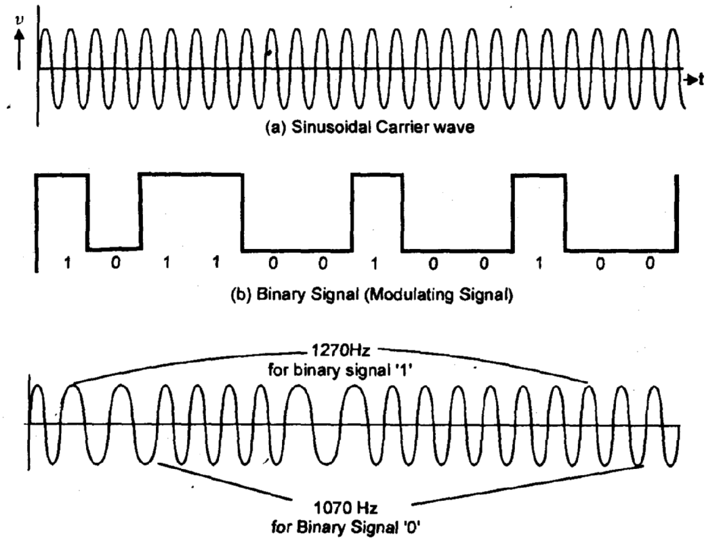


Fig. 34

Another set of frequencies are required to permit simultaneous transmit and receive operation with a modem, known as full duplex operation. A frequency 2025 Hz is used to represent a binary '0' or space and frequency 2225 Hz used to represent binary '1' or mark. These two frequency tones are spaced far enough from the other frequencies (1070 Hz and 1270 Hz) as shown in Fig. 35. To distinguish between these

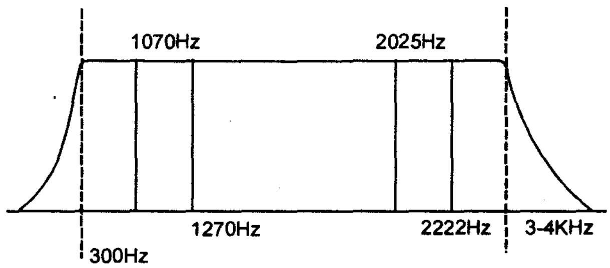


Fig. 35

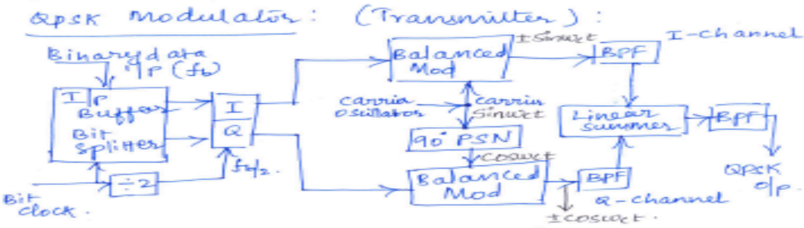
two sets of frequencies selective filters are used. The frequency set consists of 1070 and 1270 Hz tones is used for transmitting and frequency set of 2025 and 2225 Hz tones used for receiving purposes.

3. Explain QPSK Transmitter and receiver with a neat diagram.

QUADRATURE PHASE SHIFT KEYING (QPSK)

* M-ary encoding scheme, four o/p phases are possible. Binary i/p data are combined into group of 2 bits called dibits.
* 4 o/p phases are 45°, 135°, −45°, −135°.
∀ 2 bits I/p produce one o/p phase change. Therefore the rate of change at the o/p (baud rate) is one-half of the I/p bit rate.

QPSK Modulator: (Transmitter):



- Two bits are clocked into Bit splitter.
- Both bits are serially inputted and parallely outputted. One bit to I channel and other to Q-channel.
- I bit modulated a carrier that is inphase with the reference carrier.
Q bit modulates a carrier that is out of phase with the reference carrier.

4. Explain DPSK Transmitter and receiver with a neat diagram. What are the advantages of DPSK over PSK (June'13).
5. With relevant diagram explain the method of synchronous detection of FSK signal. What should be the relationship between bit rate and frequency shift for better performance. (June'13)
Refer Q.2

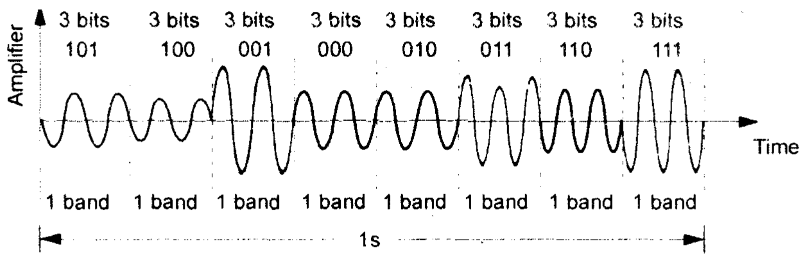
6. Compare the different types of digital modulation techniques.

Parameter	ASK	FSK	PSK
1. Variable Characteristics	Amplitude	Frequency	Phase
2. Bandwidth (Hz)	2R	$\lvert f_1 - f_0 \rvert$ +(1+r) R	(1+r) R
3. Noise Immunity	Low	High	High
4. Error Probability	High	Low	Low

Parameter	ASK	FSK	PSK
5. Performance in presence of noise	poor	Better than ASK	Better than FSK
6. Complexity	Simple	Moderately complex	Very Complex
7. Bit rate	Suitable upto 100 bit/sec	Suitable upto about 1200 bit/sec.	Suitable for high bit rates.
8. Detection method	Envelope	Envelope	Coherent

7. Explain the error performance of different digital modulation techniques.
8. Explain 8 QAM transmitter and receiver diagrams.(Dec'13)

Quadrature Amplitude Modulation (QAM): Quadrature amplitude modulation is a combination of ASK and P so that a maximum contrast between each signal unit (bit, dibit, tribit and so on) is achieved.



The number of amplitude shifts is fewer than the number of the phase shifts. Because amplitude changes are susceptible to noise and require greater shift difference than do phase changes, the number of phase changes used by QAM system is always large the number of amplitude shifts.

The Quadrature Amplitude Modulation (QAM) is called "Quadrature Carrier Multiplexing" (QCM).
Further, this scheme enables two DSBSC (Double side band with supressed carrier) modulated signals topy the same B.W. and therefore it allows the two signals separation at the receiver; therefore, it is also known as "Band width conservation Scheme" (BCS).
(i) QAM Modulator Transmitter. It consists of two separate balanced modulators, which are supplied with two carrier waves of the säme frequency and differing by quadrature (i.e. by 900).

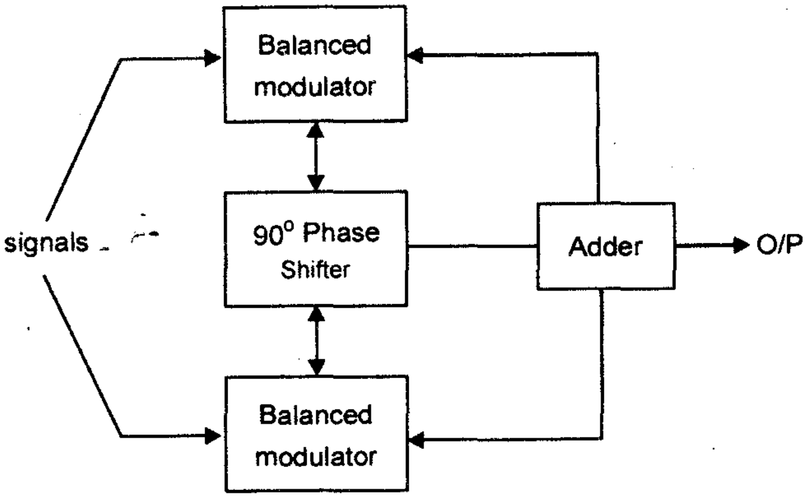
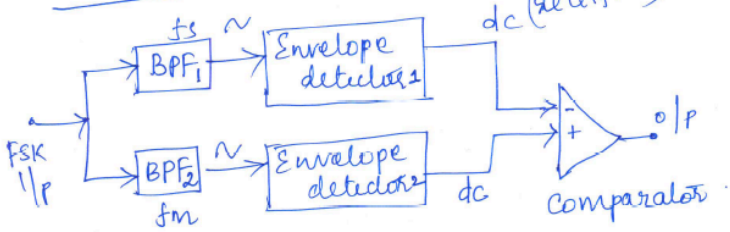


Fig. 43

The output of the balanced modulators is given to an ‘ circuit, and we get the output (multiplexed) signal, which is transmitted. This signal consists of “in phase” and the “quadrature phase components. QAM detector/Receiver. The transmitted (multiplexed) signal is applied simultaneously to two separate coherent detectors, which are supplied with two local carriers of same frequency but differing in phase by 900. The outputs of the detectors are given to low pass filters (LPF) which gives the two original massage signals.

9. Write short notes on the probability error of the following
   (i)FSK (coherent and non coherent)

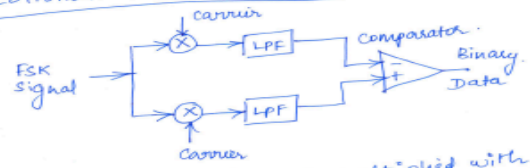
1) Non-Coherent FSK detector:



The respective BPF passes only mark or the space frequency on to its respective envelope detector. Envelope detector indicate total power in each pass band and the Comparator responds to the largest of two powers.
Envelope detector → diode detectors which rectify & filter their i/p's. (dc vtg ∝ to ac i/p)

Received BFSK signal is fs, when it passes to ED1 through BPF 1 and o/p of BPF2 is zero. Then the comparator o/p will be positive representing logic 1. otherwise, BFSK is fm, O/p of BPF1 is zero, comparator o/p be zero represents logic 0. Thus original data is recovered by the receiver.

2) COHERENT FST DEMODULATOR:



The Incoming FSK signal is multiplied with the recovered carrier signals. This signal has exactly the same frequency and phase of the reference carrier at the transmitter. The frequencies fm & fs are not continuous. The filtered o/ps are applied to comparator. Comparator o/p is a digital signal.

10. Write short notes on the Carrier Recovery of the following
(i) Costas loop. ii).Squaring loop.

11.For an 8-PSK modulator with an input data rate equal to 20mbps and a carrier freq of 70Mhz .Determine the minimum double sided Nyquist bandwidth (fn) and baud.
12.For an QPSK modulator with an input data rate (Fb)equal to 20mbps and a carrier freq of 70Mhz .Determine the minimum double sided Nyquist bandwidth (fn) and baud.
13.For an BPSK modulator with an input data rate equal to 20mbps and a carrier freq of 70Mhz . Determine the minimum and minimum upper and lower side freq double sided Nyquist bandwidth (fn) and baud.
14.Expalin the operation of the following (a) 8-PSK (b)QAM.
Refer BPSK and QAM

UNIT III : DATA AND PULSE COMMUNICATION
PART-A

1. What is ASCII code?
ASCII has been recommended by United States as ansi AND BY THE International Standards Organisation as ISO-14962.It is a 7 bit fixed length character set with 2^7.With ASCII code, the LSB is designated b0 and MSB as b7.

2. Define error detection.
It is defined as the process of monitoring the transmission of data and find when a error has occurred.

3. What are the error detection techniques?
Redundancy checking ----- VRC, LRC, CRC,Checksum
Parity coding; Exact count encoding; Echoplex.

4. What is redundancy checking?
Redundancy checking is defined as the process of adding extra bits for detecting errors at the destination.

5. What is meant by CRC?
CRC is a systematic code,where instead of adding bits together to achieve a desired parity ,a sequence of redundant bits called CRC or CRC remainder is added to the end of the data to be transmitted.It can be written as (n,k) cyclic codes.

6. Define FCS.
The group of characters forming a message is called as block or frame of data.Hence,the bit sequence for the LRC is called as block check sequence or frame check sequence(FCS).

7. Name the two error correction method.
Retransmission and forward error correction.

8. Explain briefly about retransmission.

Retransmission is the process of retransmitting the entire message to the receiver,whenever the receiver founds that the received message is in error. If the receiver automatically calls for a retransmission of the entire message,this process of retransmission is called as automatic repeat request(ARQ)

9. What is Forward Error Correction?

It is the type of error correction scheme, where the errors are detected and corrected without retransmission but by adding the redundant bit to the message before transmission.

10. Define DTE and DCE.

DTE is a binary digital device where the information originates or terminates. It contains the hardware and software necessary to establish and control communications between end points within data communication systems. DCE is used to interfaces data terminal equipment to a transmission channel.

11. Define parallel interface.

Parallel interface allows the user to transfer data between two devices with eight or more bits at a same time or simultaneously. It is also called as serial by word transmission.

12. Define UART and USRT.

UART: Universal asynchronous receiver/transmitter is used for asynchronous transmission of data between DTE and DCE. Asynchronous transmission means that an asynchronous data format is used and there is no clocking information transferred between DTE and DCE.

USRT: Universal synchronous receiver/transmitter is used for synchronous data transmission between DTE and DCE .It means that there is clocking information transferred between USRT and modem.

13.Give the primary functions of USRT.

*To perform serial to parallel and parallel to serial conversion of data.

* To perform error detection by inserting and checking parity bits.

*To insert and detect SYN characters.

14. What is meant by RS 232 interface?

RS 232 interface specifies a 25 wire cable with DB25P compatible connector .It is simply a cable and two connectors, the standard also specifies limitations on the voltage levels that the DTE and DCE can output onto or receive from the cable.

15.What is meant by centronics parallel interface?

Centronics parallel interface was originally designed to be used for transferring data between a microcomputer and a printer. Centronics was one of the original companies to design printers especially for desktop computers.

16.Give the primary functions of UART.

*To perform serial to parallel and parallel to serial conversion of data.

* To perform error detection by inserting and checking parity bits.

*To insert and detect start and stop bits.

17.State sampling Theorem.

If a finite energy signal g(t) contains no frequency component higher than W Hz, it is completely determined by specifying its ordinates at a separation of points spaced 1/ 2W seconds apart.

18.Give the methods of Sampling

Ideal Sampling (or) Instantaneous Sampling

Natural Sampling   Flat-top Sampling

19.What is Aliasing or Foldover?

When the continuous time signal g(t) is sampled at the rate less than Nyquist rate, frequencies higher than W takes on the identity of the low frequencies in sampled signal spectrum . This is called aliasing. The use of a low pass reconstruction filter, with its pass band extending from –W to W will not yield an undistorted version of the original signal. Aliasing can be reduced by sampling at a rate higher than Nyquist rate. In other words, Aliasing occurs when the signal is sampled at a rate less than Nyguist rate (2W samples/ sec). It is prevented by using Guard Bands Pre-alias Filter

20. Define Nyquist rate and Nyquist interval.

According to sampling theorem, a continuous time signal can be completely represented in its samples and recovered back if the sampling frequency is $f_S \geq 2W$. Here f_S is sampling frequency and W is the highest frequency component of the signal.

Nyquist rate: The minimum sampling rate of 2W samples per second is called Nyquist rate.

i.e., $f_S = 2W \quad \rightarrow$ Nyquist rate

Nyquist interval: Reciprocal of 2W is called the Nyquist interval.
Nyquist interval = 1/2W.

21. Give the practical procedure for the sampling of a signal whose spectrum is not strictly band limited.

i) Prior to sampling, a low pass pre-alias filter (anti-alias filter) of high enough order is used to attenuate those frequency components of the signal that do not contribute significantly to the information content of the signal.
ii) The filtered signal is sampled at a rate slightly higher than the Nyquist rate 2W, where W is the cutoff frequency of the pre-alias filter.

22. Define aliasing error. Give the upper bound for the aliasing error.

Let $\{g(n/f_S)\}$ denote the sequence obtained by sampling an arbitrary signal g(t) at the rate f_S samples per second. Let $g_i(t)$ denote the signal reconstructed from this sequence by interpolation;
That is, $g_i(t)=\sum g(\ ) \operatorname{sinc}(f_S t - n)$
The absolute error $\varepsilon = | g(t) – gi(t) |$ is called the aliasing error.
The aliasing error is bounded as

$$\varepsilon \leq 2 \quad | G(f) | \, df$$

23. Given the signal m(t)=10 cos (2000π t) cos (8000 π t) , what is the minimum sampling rate based on the low pass uniform sampling theorem?

The equation shows that m(t) is generated by multiplication of two signals. We know that cosAcosB = ½ [cos (A-B)+cos (A+B)]
There fore, m(t) = (10/2) [cos (6000π t) + cos (10000π t)]
The two frequencies in m(t) are 3000 Hz and 5000 Hz and the highest frequency present in m(t) is 5000 Hz. Minimum sampling rate is 2 (5000) = 10000 samples per second.

24. What is Inter symbol Interference (ISI)?

ISI arises because of imperfections in the overall frequency response of the system.
When a short pulse of duration $\mathbf{T_b}$ seconds is transmitted through a band limited system, the frequency components constituting the input pulse are differentially attenuated and, more significantly, differentially delayed by the system. Consequently the pulse appearing at the output of the system is **dispersed over an interval longer than** $\mathbf{T_b}$ seconds. Thus when a sequence of short pulses are transmitted through the system, one pulse every T_b seconds, the dispersed responses originating from different symbol intervals will interfere with each other, there by resulting in ISI.

25. What are the types of pulse modulation systems?

i) PAM ii) PWM iii) PPM

26. What is PAM?

It is a process in which amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of continuous message signal.

27. What do you mean by Aperture Effect?

It is nothing but amplitude distortion occurring at PAM due to the sinc function. It is overcome by using a Equalizer whose transfer function is

$$| H(f)| = T^{-1} \operatorname{sinc}(fT)$$

28. What is PPM?

It is the process in which the position of a pulse relative to its unmodulated time of occurrence is varied in accordance with message signals.

29. What is PWM?

It is the process in which the samples of message signal are used to vary the duration of individual pulses in the carrier.

30. What is Quantization and sampling?

Quantization : It is the process in which the analog sample of the original signal is converted into a digital form.
Sampling :It is the process in which the original analog signal is converted into a discrete time and continuous amplitude signal

31. Classify Quantizers.

Uniform Quanatizer – Representation levels are uniformly spaced
Non-Uniform Quanatizer – Representation levels are non-uniformly spaced

32. What is Quantization Noise?

The difference between the output analog sample and the discrete output quantized signal gives raise to an error called Quantization Noise.

33. What are the advantages of digital transmission?

The advantage of digital transmission over analog transmission is noise immunity. Digital pulses are less susceptible than analog signals to variations caused by noise.

Digital signals are better suited to processing and multiplexing than analog signals.

Digital transmission systems are more noise resistant than the analog transmission systems.

Digital systems are better suited to evaluate error performance.

34. What are the disadvantages of digital transmission?

The transmission of digitally encoded analog signals requires significantly more bandwidth than simply transmitting the original analog signal. Analog signal must be converted to digital codes prior to transmission and converted back to analog form at the receiver, thus necessitating additional encoding and decoding circuitry.

35. Define pulse code modulation.

In pulse code modulation, analog signal is sampled and converted to fixed length, serial binary number for transmission. The binary number varies according to the amplitude of the analog signal.

36. What is the purpose of the sample and hold circuit?

The sample and hold circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal.

37. Define companding, Nyquist sampling rate

Companding is the process of compressing, then expanding. With companded systems, the higher amplitude analog signals are compressed prior to transmission, then expanded at the receiver.

Nyquist sampling rate: Nyquist sampling rate states that, the minimum sampling rate is equal to twice the highest audio input frequency.

38.Find the minimum sampling frequency for a signal having frequency from 10MHz to 10.2MHz, in order to avoid aliasing.

The minimum sampling frequency fs = 2xfm, fs = 2x 10.2=20.4 MHz.

PART B

1.Explain about centronics parallel interface

2.Explain in detail about line control unit in data communication hardware.

3.Write short notes on i)topologies ii)two wire Vs four wire operation.

4.Define and explain ASCII code and Bar codes.

5.Explain the operation of UART.

6. .Explain the operation of USRT.

7. State and explain the various Error Detection methods

Error control can be divided into two categories
1) Error detection
2) Error correction.

ERROR DETECTION:

→ It is the process of monitoring the received data and determining when a transmission error has occured.
→ It is not able to identify that which bit is in error.
→ The purpose of error detection is not to prevent errors from occuring but to prevent undetected errors from occuring.

Error detection techniques:

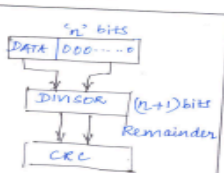
1) Redundancy: It involves transmitting each character twice. If the same character is not received twice in succession, a transmission error has occured.
2) Echoplex: full duplex operation, ie character is transmitted after it has been typed into the transmit terminal, then it has been received at the receiver. Immediately it is transmitted back to the transmitter stream. Manually error can be detected.
3) Exact-count encoding: With exact count encoding, the number of 1's in each character is the same.
4) Parity Coding: It is the simplest error detection scheme used for data communication systems and is used with vertical & horizontal redundancy checking.

CYCLIC REDUNDANCY CHECKING (CRC)

This is the most reliable scheme for error detection. Approximately 99.9% of all transmission errors are detected.
The most common CRC code is CRC-16.
A sequence of redundant bits are called as CRCode.

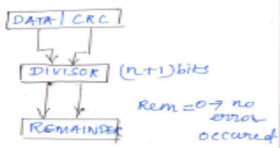
STEPS INVOLVED IN:

SENDER SIDE



* 'n' number of 0's are appended to the data unit. The number n is one less than the number of bits in the predetermined divisor, ie(n+1)
* New data unit is divided by divisor using process called binary division.
* The Remainder is called as CRC.

RECEIVER SIDE



* CRC is appeded to the original data unit.
* New data unit is divided by same divisor.
* If the remainder is zero, then there is no error occured, otherwise there is an error.

8. State and explain the various Error Correction methods.

ERROR CORRECTION

There are three methods to correct an error.

1) Symbol Substitution : It was designed to be used in a human environment - when there is a human being at the receive terminal to analyze the data and make decisions on its integrity.

Ex: $\boxed{BAT} \longrightarrow \boxed{\$AT}$
↑symbol.

An operator could not determine the character and retransmission is required.

ii) Retransmission :

It is the process of retransmitting the entire message to the receiver, whenever the receiver founds that the received message is in error.

Since the receiver automatically calls for a retransmission of the entire message, this process of retransmission is called as automatic repeat request (ARQ).

iii) Forward Error Correction :

It is the only error correction scheme that detects and corrects error without retransmission. The redundant bits are used to detect which bit is in error.

Most popular code is Hamming code.

→ Hamming code is a single bit error correction method using redundant bits.

→ For a data unit of 'm' bits, use the formula, $2^n \geq m+n+1$, to determine the number of redundant bits needed.

The number of hamming bits must be added to a character is determined from the expression.

$2^n \geq m+n+1$, $n \rightarrow$ no/- of hamming bits
$m \rightarrow$ no/- of bits in the character.

9.Explain the procedure of PCM generation and detection with its block diagram.(June'13)

PCM NOTE:

Sampling: A train of narrow rectangular pulses are used to sample the message signal.

Sampling frequency fs must be greater than twice the highest frequency of the message signal i.e $f_s \geq 2f_m$.

Quantization: → Quantizer → approximation or roundingoff

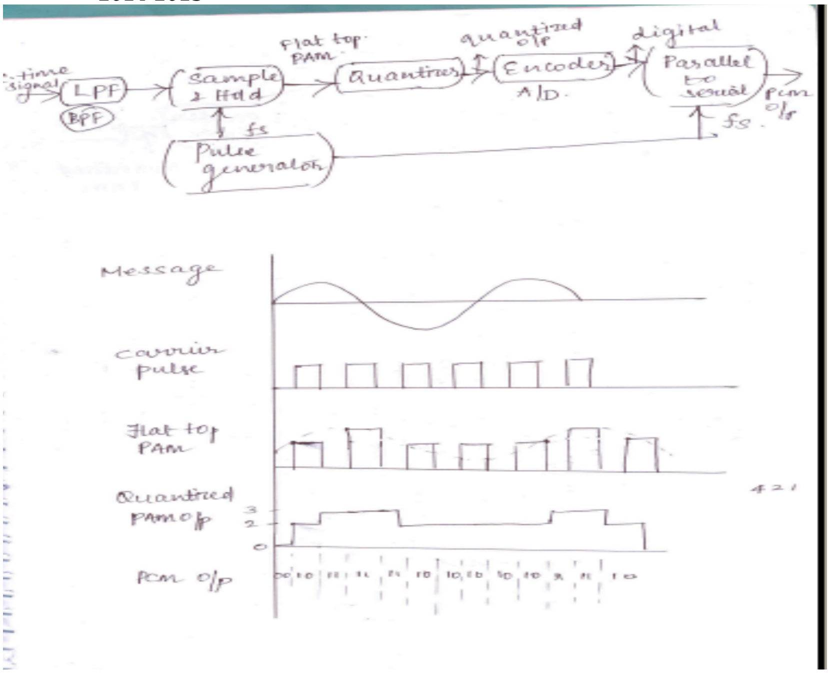
The process of making the signal discrete in amplitude by approximating the sampled signal to the nearest predefined or representation level is called quantization.

When step size between two adjacent level is same throughout the i/p signal range is called uniform quantization.

If the stepsize varies depending on the i/p then ~~the~~ it is known as non-uniform

Encoding: (Encoder:)

The encoder is used to encode the discrete set of samples. The process of allocating some digital code to each level is called encoding.

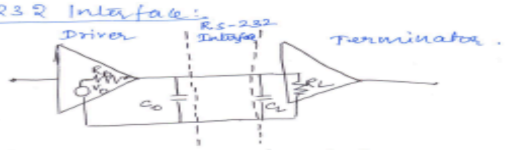


10. Describe On-OFF keying technique in detail.
11.Compare the performance of PAM and PCM.
Calculate the even and odd parity bits for the ASCII character W.
11. Calculate the even and odd parity bits for the ASCII character G and y.
Refer error detection technique
13.for the ASCII encoded message :ALPHA. Find VRCs and LRCs
Refer error detection technique
14.give a detailed note on (i)Retransmission (ii)Forward error control
Refer error correction topic
15.Expalin RS-232 interface in detail .

SERIAL INTERFACES

* A serial interface is used to interconnect DTE & DCE for an orderly flow of data.
* It coordinates, flow of data, control signals & timing information between DTE & DCE.
* RS 232, RS 449, RS-530 serial interface standards are the important std serial interface. RS → Recommended standard.

RS 232 Interface:



* RS 232 interface specifies a 25-wire cable with a DB 25P/DB 25S compatible connector.
* A leveler is called driver if it outputs a signal voltage to the cable and A terminator if it accepts a signal voltage from the cable.
* Eventhough RS-232 is a cable with 2 connectors, the standard specifies

16.Explain IEEE 488 bus in detail.

* Select (SLCT) : This is an active high line which indicates whether the printer is selected or not.
* Error (ERROR) : Active low line, used to indicate a printer in problem.
  - Printer is offline/
  - " is out-of-paper etc.
  - undefined problem.

IEEE - 488 BUS

IEEE 488 bus uses eight bidirectional data lines connected in parallel to interface up to 15 remote devices.

IEEE 488 bus characteristics:

- logic levels → TTL
- No. of data lines → 8
- max. no. of devices → 15
- Distance → 2m
- max total length → 20m
- max speed open collector drivers → 250 kbps
- max speed with tristate drivers → 1 Mbps
- Connector → 24 pin ribbon.

IEEE Bus devices:

i) Controller : It is a device which determines which device can transmit data & which device behave as receivers.

UNIT IV SOURCE AND ERROR CONTROL CODING

PART A

1. Give the factors which influence reliable transmission?

Transmitted signal power. channel bandwidth.

2. List out the advantages of error control coding.

Reduces the required transmitted power; Reduces the size of antennas; Reduces the hardware cost.

3. What are the disadvantages of error control coding?

Increases the transmission bandwidth; Increases the complexity of decoder.

4. Give the types of error control codes.(June'13)

Block codes; Convolutional codes.

5. List the types of block codes.

Linear block codes. Cyclic codes.

6. Define block codes.

The codes which consists of (n-k) parity bits for every k bit message block are known as block codes

7. Define linear block codes.

Block code is the code in which every 'k-bit' message block (n-k) parity bits are appended to produce 'n' bit codeword. If the parity bits are the linear combination of 'k' message bits then the code is referred as linear block codes.

8. What are the systematic codes?

Block codes in which the message bits are transmitted in unaltered form are called systematic codes.

9. Define generator matrix.

Generator matrix Gkxn is used in the encoding operation and its k rows are linearly independent the encoding operation and its k rows are linearly independent

$G_{kxn}=[P_{kx(n-k)} \mid I\ kxk]$;

Where, P-parity matrix

I-identity matrix

10. **Explain Shannon-Fano coding. (June'13)**

An efficient code can be obtained by the following simple procedure, known as Shannon- Fano algorithm. List the source symbols in order of decreasing probability. Partition the set into two sets that are as close to equiprobable as possible, and sign 0 to the upper set and 1 to the lower set. Continue this process, each time partitioning the sets with as nearly equal probabilities as possible until further partitioning is not possible.

11. Define information rate.

If the time rate at which source X emits symbols is r symbols per second. The information rate R of the source is given by R=r H(X) bits/second, H(X)- entropy of the source

12. What is data compaction?

For efficient signal transmission the redundant information must be removed from the signal prior to transmission .This information with no loss of information is ordinarily performed on a signal in digital form and is referred to as data compaction or lossless data compression.

13. Define mutual information and channel capacity.

Mutual information I(X,Y) of a channel is defined by I(X,Y)=H(X)-H(X/Y) bits/symbol
H(X)- entropy of the source, H(X/Y)- conditional entropy of Y.

14. Define entropy.

Entropy is the measure of the average information content per second.

$$\text{Entropy} = -\sum_{i=1}^{n} p_i \log_b(p_i)$$

n = number of different outcomes.

15. Define syndrome.

Syndrome contains information about the error pattern 'e' and may therefore be used for error detection . S is a 1 x (n-k0 vector and is used to decode the vector C from the received vector 'r ' S = r H^T where r = C + e .

16. **Give the properties of syndrome.**

The syndrome depends only on the error pattern and not on the transmitted code word.
All error patterns that differ by a codeword have the same syndrome.
S = e H^T

17. **Define: Cyclic codes**

Cyclic codes are a sub-class of linear block codes .It posses a well defined mathematical structure and which provides efficient decoding.

18.Give any two properties of cyclic codes?

- Linearity and Cyclic property

19. State cyclic porperty.

Cyclic property states that any cyclic shift of a codeword in the code is also a codeword.

20. State linearity property.

Linearity property states that the sum of any 2 code words in the code is also a code.

21. Give the graphical representation of convolutional encoder?

Code tree; Trellis State diagram

22. What is the need for convolution coding?

Convolution coding may be the preferred method in applications where the message bits come in serially rather in large blocks in which case the use of buffer may be undesirable.

23. What is trellis?

Trellis is a tree like structure with remerging branches. The code branch with an input 'o' is drawn by a solid line and a branch by an input '1' is drawn as a dashed line.each input sequence corresponds to a specific path through the trellis.

trellis contains (l+k) levels where
l - length the message and
k - constraint length level
j - is the depth of the trellis

24. What is code tree ?

Each branch of the tree responds an input symbol with the corresponding pair of input binary symbols indicated on the branch.the input 'o' specifies the upper branch of the tree and the input '1' specifies the lower branch of the tree.a specific path is traced from left to right in accordance with the input sequence .the corresponding coded symbols on the branches of that path constitute the output sequence.

25. Distinguish block codes and convolution codes?

Block codes	**Convolution codes**
1) code the block of k Msg bits.	1) code each msg bit individually.

2) needs the buffer to store msg block.	2) does not need the buffer since the bits are arriving in serial fashion.

PART -B

1. Five symbols of the alphabet of DMS and their probabilities are given below. S={So,S1,S2,S3,S4} P(S) = {0.4, 0.2, 0.2, 0.1, 0.1}.Code the symbols using Huffman coding. Find the efficiency of the code.

Step 1: Arrange the codewords in descending order.

Step 2: Represent the least two priority values as 1 and 0.Combine the least priority probabilities and sum them.

Step 3: Arrange the values again. Repeat the process and trace the code value for each value.

2. Find the Shannon –fanno code for the following seven messages with probabilities indicated. S={S1,S2,S3,S4,S5, S6,S7},P(S)= {0.05,0.15,0.2,0.05,0.15,0.3,0.1}.

Step 1	Step 2	Step 3		Code word
0	0 Partition	Stop here		00
0 Partition	1	Stop here		0 1
1	0 Partition	Stop here		1 0
1	1	0 Partition	Stop here	110
1	1	1	Stop here	1 1 1

3. Construct a convolution encoder whose constraint length is 3 and has 3 modulo- 2 adders and an output multiplexer. The generator sequences of the encoder are $g^{(1)}=(1,0,1)$,$g^{(2)}=(1,1,0)$,$g^{(3)}=(1,1,1)$. Draw the block diagram of the encoder. Find the encoder output produced by the message sequence 10111.

STEPS TO BE FOLLOWED:

1.Write the msg polynomial.

2. Write the generator polynomial of each sequence.

3.Compte the code polynomial.

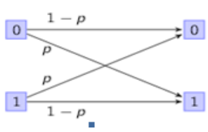
4.Write the code sequence C1= 1001011
C2=1110010
C3=1100101

The codeword is (111,011,010,100,001,110,101)

4. The channel transition matrix is given by

.
$$\begin{pmatrix} 0.9 & 0.1 \\ 0.2 & 0.8 \end{pmatrix}$$

Draw the channel diagram and determine the probabilities associated with outputs assuming equi probable inputs.



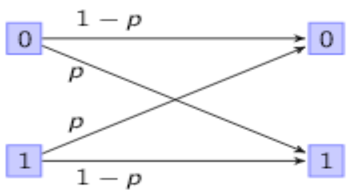
5. Explain coding and decoding process of block codes.

1.Code the block of k message bits.
2. Need the buffer to store message block.

6. Write in detail the procedure of Shannon-fanno coding scheme
7. (i)Derive the channel capacity of Binary symmetric channel.

Binary Symmetric Channel

Binary symmetric channel preserves its input with probability $1-p$ and with probability p it outputs the negation of the input.



Mutual information is bounded by

$$I(X;Y) = H(Y) - H(Y|X) = H(Y) - \sum_x p(x)H(Y|X=x) =$$
$$= H(Y) - \sum_x p(x)H(p, 1-p) = H(Y) - H(p, 1-p) \leq \quad (3)$$
$$\leq 1 - H(p, 1-p).$$

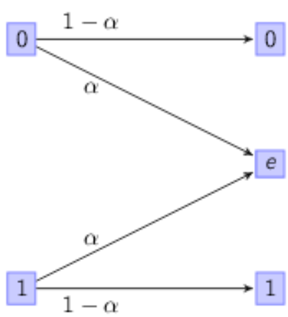
Equality is achieved when the input distribution is uniform. Hence, the information capacity of a binary symmetric channel with error probability p is

$$C = 1 - H(p, 1-p) \text{ bits.}$$

(ii)Derive the channel capacity of Binary erasure channel.

Binary erasure channel

Binary erasure channel either preserves the input faithfully, or it erases it (with probability α). Receiver knows which bits have been erased. We model the erasure as a specific output symbol e.



The capacity may be calculated as follows

$$C = \max_X I(X;Y) = \max_X (H(Y) - H(Y|X)) = \max_X H(Y) - H(\alpha, 1-\alpha). \tag{4}$$

It remains to determine the maximum of $H(Y)$. Let us defined E by $E = 0 \Leftrightarrow Y = e$ and $E = 1$ otherwise. We use the expansion

$$H(Y) = H(Y, E) = H(E) + H(Y|E) \tag{5}$$

and we denote $P(X = 1) = \pi$. We obtain

$$H(Y) = H((1-\pi)(1-\alpha), \alpha, \pi(1-\alpha)) = H(\alpha, 1-\alpha) + (1-\alpha)H(\pi, 1-\pi). \tag{6}$$

8. (i)Derive the channel capacity of a continuous band limited white Gaussian noise channel.
(ii)What are the implications of the information capacity theorem?

Bandlimited continuous-time channel

Definition:

A bandlimited continuous-time channel with white noise :

$$Y(t) = (X(t) + Z(t)) * h(t),$$

where $X(t)$ is the signal waveform, $Z(t)$ is the waveform for the white Gaussian noise, and $h(t)$ is the impulse response of an ideal bandpass filter, which cuts out all frequencies greater than W.

Theorem:

Suppose that a function $f(t)$ is bandlimited to W, namely, the spectrum of the function is 0 for all frequencies greater than W. Then the function is completely determined by samples of the function spaced $1/2W$ seconds apart.

Bandlimited Channel

- Channel bandlimited to $f \in [-W, W]$ and signal duration T
- Nyquist: Signal is completely defined by $2WT$ samples
- Can represent as a $n = 2WT$-dimensional vector space with prolate spheroidal functions as an orthogonal basis
  - White noise with double-sided psd $\frac{1}{2}N_0$ becomes i.i.d. Gaussian $\mathcal{N}(0, \frac{1}{2}N_0)$
  - Signal power constraint $P \Rightarrow$ Signal energy $\leq PT$
    Energy constraint per sample $= \frac{1}{2}P/W$
- Capacity:

$$\begin{aligned} C &= \frac{1}{2}\log(1 + \frac{1}{2}P/W(\frac{1}{2}N_0)^{-1})2W \\ &= W\log(1 + \frac{P}{WN_0}) \ \text{ bits/second} \end{aligned}$$

Summary

- Capacity of discrete-time Gaussian channel:

$$C = \frac{1}{2}\log\left(1 + \frac{P}{N}\right) \quad \text{bits per transmission.}$$

- Capacity per sample:

$$C = \frac{1}{2}\log\left(1 + \frac{P/2W}{N_0/2}\right) = \frac{1}{2}\log\left(1 + \frac{P}{N_0 W}\right) \quad \text{bits per sample.}$$

- Capacity of a bandlimited continuous-time channel with white noise:

$$C = W\log\left(1 + \frac{P}{N_0 W}\right) \quad \text{bits per second.}$$

- $W \to \infty$ gives

$$C = \frac{P}{N_0}\log_2 e \quad \text{bits per second.}$$

9. Explain the properties of mutual information.

The mutual information is defined as the amount of information transferred when x_i is transmitted and y_j is received. It is represented by $I(x_i, y_j)$ and given as,

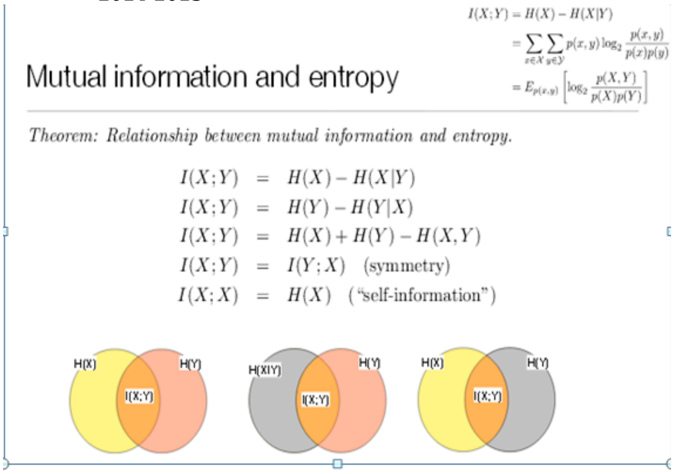
$$\boxed{I(x_i, y_j) = \log\frac{P(x_i / y_j)}{P(x_i)} \text{ bits}} \quad \text{... (5.10.1)}$$

Here $I(x_i, y_j)$ is the mutual information

$P(x_i / y_j)$ is the conditional probability that x_i was transmitted and y_j is received.

$P(x_i)$ is the probability of symbol x_i for transmission.

The average mutual information is represented by $I(X; Y)$. It is calculated in bits/symbol. The average mutual information is defined as the amount of source information gained per received symbol. Here note that average mutual information is different from entropy.



10. Define entropy. Explain the properties of entropy.

Entropy is the measure of the average information content per second. It is given by the expression H(X)=_I P(xi)log2P(xi) bits/sample.

11. Give brief notes on error detection. (June'13)

UNIT V MULTI - USER RADIO COMMUNICATION

PART-A

1. What are the different multiple access techniques used in wire less communication?

TDMA – Time Division Multiple Access
FDMA- Frequency Division Multiple Access
CDMA – Code Division Multiple Access.

2. Compare FDMA and TDMA.(June'13)

FDMA - Frequency Division Multiple Access	TDMA - Time Division Multiple Access
All users access the channel by transmitting simultaneously but using disjoint frequency bands	All users occupy the same RF band width of the channel, but they transmit sequentially in time
Fixed assignment multiple access technique	Fixed assignment multiple access technique
Well suited for analog communication	Well suited for digital communication

3. List the merits of CDMA over TDMA.

CDMA does not require an external synchronization network which is an essential feature of TDMA. CDMA offers a gradual degradation in performance as the number of user is increased. It is there fore relatively easy to add new users to the system. CDMA offers an external interference rejection capability.

4. What is CDMA?

CDMA – Code Division Multiple Access. In CDMA, all users transmit simultaneously and occupy the same RF bandwidth. Each user is assigned a **code** which perform the DSSS or FHSS modulation

5. List out any four features of TDMA.

TDMA shares a single carrier frequency with several users where each user makes use of non overlapping time slots. Data transmission for users is not continuous, but occurs in burst. Because of discontinuous transmission, handoff process is much simpler for a subscriber unit . TDMA uses different time slots for transmission and reception, thus duplexers are not required.

6. What is Satellite?

An artificial body that is projected from earth to orbit either earth (or) another body of solar systems. Types: Information satellites and Communication Satellites

7. Define Satellite Communication.

It is defined as the use of orbiting satellites to receive, amplify and retransmit data to earth stations.

8. State Kepler's first law.

It states that the path followed by the satellite around the primary will bean ellipse. An ellipse has two focal points F1 and F2. The center of mass of the twobody system, termed the barycenter is always centered on one of the foci.e = [square root of (a^2-b^2)]/a

9. State Kepler's second law.

It states that for equal time intervals, the satellite will sweep out equalareas in its orbital plane, focused at the barycenter.

10. State Kepler's third law.

It states that the square of the periodic time of orbit is perpendicular to thecube of the mean distance between the two bodiesa^3= 3 / n^2 Where, n = Mean motion of the satellite in rad/sec.3 = Earth's geocentric gravitational constant.With the n in radians per sec. the orbital period in second is given by, P = 27 / n

11. Define apogee.

The point farthest from the earth.

12. Define Perigee.

The point closest from the earth.

13. What is line of apsides?

The line joining the perigee and apogee through the center of the earth.

14. Define ascending node.

The point where the orbit crosses the equatorial plane going from south to north.

15. Define descending node.

The point where the orbit crosses the equatorial plane going from north to south.

16. Define Inclination.

The angle between the orbital plane and the earth's equatorial plane. It is measured at the ascending node from the equator to the orbit going from east to north.

17.What is declination?

The angle of tilt is often referred to as the declination which must not be confused with the magnetic declination used in correcting compass readings.

18.Write short notes on station keeping.

It is the process of maintenance of satellite's attitude against different factors that can cause drift with time. Satellites need to have their orbits adjusted from time to time, because the satellite is initially placed in the correct orbit,natural forces induce a progressive drift.

19. What is meant by Pitch angle?

Movement of a spacecraft about an axis which is perpendicular to its longitudinal axis. It is the degree of elevation or depression.

20. What is an propellant?

A solid or liquid substance burnt in a rocket for the purpose of producing thrust.

21.What is meant by frequency reuse?

The satellite as a whole to be accessed by earth stations widely separated geographically but transmitting on the same frequency that is known as frequency reuse.

22.What are the limitations of FDMA-satellite access?

a. If the traffic in the downlink is much heavier than that in the uplink, then FDMA is relatively inefficient. b. Compared with TDMA, FDMA has less flexibility in reassigning
channels. c. Carrier frequency assignments are hardware controlled.

23. Write about pre-assigned TDMA satellite access.

Example for pre-assigned TDMA is CSC for the SPADE network. CSC can accommodate upto 49 earth stations in the network and 1 reference station. All bursts are of equal length. Each burst contains 128 bits. The bit rate is 128 Kb / s.

24. Write about demand assigned TDMA satellite access.

The burst length may be kept constant and the number of bursts per frame used by the given station is varied when the demand is varied.

25.Give the generation of cellular systems.

Generations of cellular systems include:

AMPS	1^{st} generation
GSM	2^{nd} generation
W-CDMA	3^{rd} generation

26.Define AMPS.

AMPS is a first-generation cellular technology that uses separate frequencies for each conversation and it uses the 800 MHz frequency band of the spectrum Utilizes FDMA (Frequency division multiple access) to separate users

27.What is GSM?

GSM is the second largest group of mobile communications systems

GSM is a digital cellular communications system. It was developed in order to create a common European mobile telephone standard but it has been rapidly accepted worldwide.

28.What is meant by hand over?

The term **handover** or **handoff** is the process of transferring an ongoing call or data session from one channel connected to another channel. It can be classified into soft and hard hand over.

Is normally performed because the signal level from the current cell is becoming to low, but can also be done for different reasons, such as too much traffic in a cell

PART -B

1. Explain in detail about AMPS.
2. Describe the elements in GSM radio access network.
3. Explain the types of Multiple Access techniques.

Types of multiple access techniques :

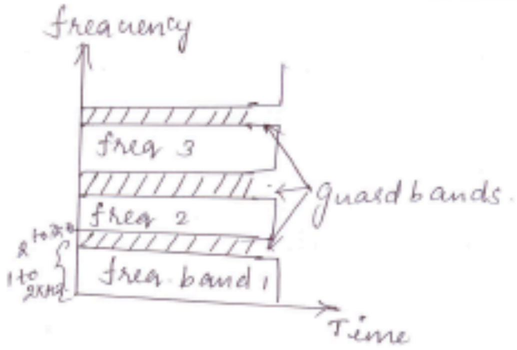
1) Frequency division multiple access (FDMA)
2) Time division multiple access (TDMA)
3) Code " " (CDMA).

(Analog) FDMA - Frequency Division (Voice & data)

* The channel bandwidth is subdivided into a number of subchannels. FDMA assigns individual channel to individual users.
* The channels are assigned on demand to users who request service.
* During call, no other user can share the same channel.

Features of FDMA :

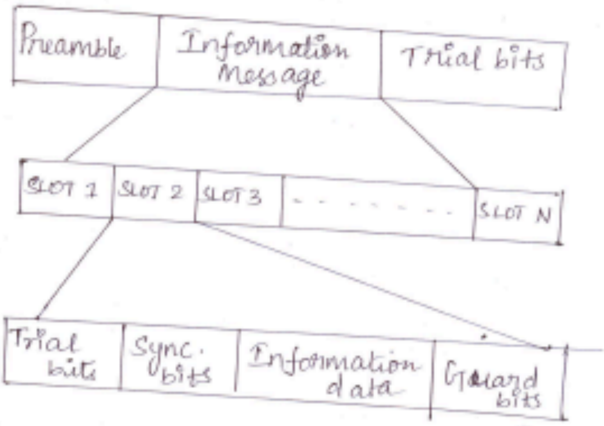
* Overall channel bandwidth is shared by the multiple users ∴ No/. of users can transmit their information simultaneously.
* If the channel is not in use, it is in idle condition. and it cannot be used by the other users.



TDMA - Time division Multiple Access (digital)

* TDMA systems divide the radio spectrum into time slots and in each slot only one user is allowed to either transmit or receive (Ref fig:2)
* Each user occupies a cyclically repeating time slot.
* N time slot comprise a frame.
* It transmit data in a buffer and burst method, so the transmission is not continuous.
* TDMA frame:
  The transmission from various users is interlaced into a repeating frame structure.
  A frame consists of a number of slots.

* Each frame is made up of a Preamble, message and ~~tait~~ trial bits.



Features of TDMA:

* used for the transmission of data and digital signal.
* TDMA shares a single carrier frequency with several users where each user makes use of non-overlapping time slots.
* Data transmission is not continuous but occurs in burst.
* TDMA uses different time slots for transmission and reception.

Code division Multiple Access (CDMA) (Fig:3)

* The Pseudo random codes are used for multiple access. For a particular user, same code is used in the transmitter and receiver for spreading and despreading.
* In CDMA, narrow band message signal is multiplied by a large bandwidth signal called spread signal.
* In CDMA, more than one user is allowed to share a channel or sub-channel with the help of DS-SS signals.
* The sequence or code allows the user to spread the information across the assigned frequency band.
* At the receiver, the signal is recovered by using the same code sequence.
* In CDMA, the users access the channel in a random manner.
* CDMA is also called as spread spectrum multiple access (SSMA).
* It is necessary to introduce guard times and guard bands.
* Does not need any synchronization.
* full power efficiency is possible.
* Hybrid combination of FDMA & TDMA is CDMA.

4. Explain TDMA system with frame structure, frame efficiency and features.
   Refer Q.3
5. Explain CDMA system with its features and list out various problems in CDMA systems.
   Refer Q.3
6. Compare TDMA, FDMA and CDMA

TDMA

Time division multiple access (TDMA) is a channel access method for shared medium (usually radio) networks. It allows several users to share the same frequency channel by dividing the signal into different timeslots. The users transmit in rapid succession, one after the other, each using his own timeslot. This allows multiple stations to share the same transmission medium (e.g. radio frequency channel) while using only the part of its bandwidth they require. TDMA is used in the digital 2G cellular systems such as Global System for Mobile Communications (GSM), IS-136, Personal Digital Cellular (PDC) and iDEN, and in the Digital Enhanced Cordless Telecommunications (DECT) standard for portable phones. It is also used extensively in satellite systems, and combat-net radio systems.

TDMA is a type of Time-division multiplexing, with the special point that instead of having one transmitter connected to one receiver, there are multiple transmitters. In the case of the uplink from a mobile phone to a base station this becomes particularly difficult because the mobile phone can move around and vary the timing advance required to make its transmission match the gap in transmission from its peers

CDMA

Code division multiple access (CDMA) is a form of multiplexing and a method of multiple access that divides up a radio channel not by time (as in time division multiple access), nor by frequency (as in frequency-division multiple access), but instead by using different pseudo-random code sequences for each user. CDMA is a form of "spread-spectrum" signaling, since the modulated coded signal has a much higher bandwidth than the data being communicated.

CDMA also refers to digital cellular telephony systems that make use of this multiple access scheme, such as those pioneered by QUALCOMM, and W-CDMA by the International Telecommunication Union or ITU.

CDMA has been used in many communications and navigation systems, including the Global Positioning System and in the OmniTRACS satellite system for transportation logistics

FDMA

Frequency Division Multiple Access or FDMA is an access technology that is used by radio systems to share the radio spectrum. The terminology "multiple access" implies the sharing of the resource amongst users, and the "frequency division" describes how the sharing is done: by allocating users with different carrier frequencies of the radio spectrum.

This technique relies upon sharing of the available radio spectrum by the communications signals that must pass through that spectrum. The terminology "multiple access" indicates how the radio spectrum resource is intended to be used: by enabling more than one communications signal to pass within a particular band; and the "frequency division" indicates how the sharing is accomplished: by allocating individual frequencies for each communications signal within the band.

In an FDMA scheme, the given Radio Frequency (RF) bandwidth is divided into adjacent frequency segments. Each segment is provided with bandwidth to enable an associated communications signal to pass through a transmission environment with an acceptable level of interference from communications signals in adjacent frequency segments

Code division Multiple Access (CDMA) (fig:3)

* The Pseudo random codes are used for multiple access. For a particular user, same code is used in the transmitter and receiver for spreading and despreading.
* In CDMA, narrow band message signal is multiplied by a large bandwidth signal called spread signal.
* In CDMA, more than one user is allowed to share a channel or sub-channel with the help of DS-SS signals.
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* CDMA is also called as spread spectrum multiple access (SSMA).
* It is necessary to introduce guard times and guard bands.
* Does not need any synchronization.
* full power efficiency is possible.
* Hybrid Combination of FDMA & TDMA is CDMA.

7. State Kepler's three laws of planetary motion. Illustrate in each case their relevance to artificial satellites orbiting the earth.

METHODS

→ For Motion of Satellite some laws should be follow[ed]. Satellite follow the law as the planet around the Sun

→ Jonnes Kepler derived 3 laws describing planetary motion

→ Kepler's law apply quite generally to any two bodies in space which interact through gravitation

two bodies 1. Primary 2. Secondary.

KEPLER'S FIRST LAW:

→ "Path followed by the satellite around the Primary will be an ellipse".

→ centre of mass of the two body system termed as "bary centre" is always centered one of foci.

→ Semi major a

Semi minor b

Eccentricity e

$e = \frac{\sqrt{a^2 - b^2}}{a}$

F1, F2 Focal point

For elliptical orbit $0 < e < 1$

$e > 1$ Hyperbola

$e < 1$ ellipse

$e = 0$ orbit circular.

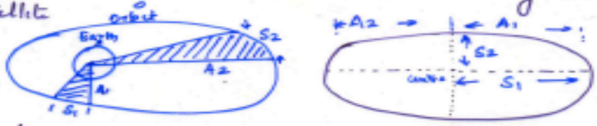
$e = 1$ Parabola

KEPLER'S SECOND LAW:

"For equal time interval, the Satellite will sweep out equal areas in its orbital plane focussed at the bary centre."

[F]rom Fig. satellit[e] ... distance S_2 meter in 1 sec

A_1 & A_2 equal



[vel]ocity S_2 is less than S_1 & take longer time to travel if is far away from earth.

[KEP]LER'S THIRD LAW:

[Sq]uare of the periodic time of the orbit is proportional to the [cube] of the mean distance b/w the two bodies.

[Mea]n distance equal to 'a'

$\therefore a^3 = \mu / n^2$ — gravitational const — Mean Motion

[Or]bital period $P = \frac{2\pi}{n}$ n - rad/sec.

$\mu = 3.986005 \times 10^{14} m^3/s^2$

[Th]is law shows fixed relationship b/w Period & size ⑤

8. Write short notes on Frequency reuse and hand over
9. Explain in detail about types of handover.
10. Explain about Error correction coding in GSM.