SYLLABUS

Academic Session (2015-16)

COMMUNICATION SYSTEMS—[ETEE-309]

UNIT I

Introduction: Overview of Communication system, Communication channels, Mathematical Models for Communication Channels.

Introduction of random Variables: Definition of random variables, PDF, CDF and its properties, joint PDF, CDF, Marginalized PDF, CDF, WSS wide stationery, strict sense stationery, non stationery signals, UDF, GDF, RDF, Binomial distribution, White process, Poisson process, Wiener process.

[T1, T2][No. of Hrs. 11]

UNIT II

Analog Modulation: Modulation- Need for Modulation, Amplitude Modulation theory: DSB-SC, SSB, VSB. Modulators and Demodulators. Angle Modulation, Relation between FM and PM Wave. Generation of FM wave-Direct and Indirect Methods. Bandwidth of FM (NBFM, WBFM)

Pulse Analog Modulation: Sampling-Natural and Flat top. reconstruction, TDM-Pulse Amplitude Modulation (TDM-PAM), Pulse Width Modulation (PWM), Pulse Position Modulation(PPM), Generation and Recovery.

Pulse Digital Modulation: Pulse Code Modulation (PCM), Differential Pulse Code Modulation (DPCM), Delta Modulation (DM), ADPCM.

[T1, T2][No. of Hrs. 11]

UNIT III

Digital Modulation and Transmission: Advantages of digital communication. Modulation schemes: ASK, PSK, FSK. Spectral Analysis. Comparison. Digital Signaling Formats-Line coding.

Information and Coding Theory: Entropy, Information, Channel Capacity. Source Coding Theorem: Shannon Fano Coding, Huffman Coding.

[T1, T2][No. of Hrs. 11]

UNIT IV

Fiber Optical System: Basic Optical Communication System. Optical fibers versus metallic cables, Light propagation through optical fibers. Acceptance angle and acceptance cone, Fiber configurations. Losses in optical fibers. Introduction to Lasers and light detectors. Applications: Military, Civil and Industrial applications.

Advanced Communication Systems: Introduction to cellular radio telephones. Introduction to satellite Communication.

[T1, T2][No. of Hrs. 11]

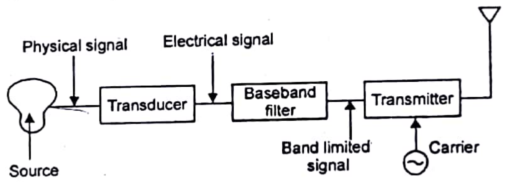
MODEL PAPER–I
FIRST TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

Time : 1.30 hrs. **M.M. : 30**

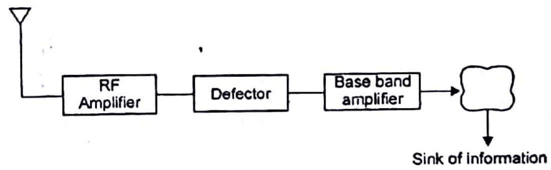
Note: *Attempt Question no.1. which is compulsory and any two more questions from remaining. There is step-marking and use of scientific calculator (non-programmable) is permitted. Assume missing data, if any.*

Q.1. (a) Given a simple block schematic of communication systems.

Ans. Communication Systems may be explained by the following block diagram.



(a) Transmitter



(b) Receiver.

Communication system is divided in two parts.

(*a*) Transmitter section

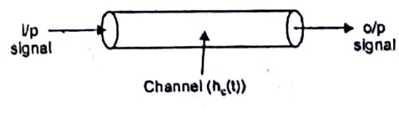
(*b*) Receiver Section.

(*a*) In transmitter section the information is obtained in physical form then with the help of transducer it is converted to the electrical form suitable for amplification band limiting and modulation.

(*b*) In the receiver section, the receiver first amplifies the signal to make it powerful enough to be processed.

Q.1. (b) Define the channel and its capacity to transmit the signal.

Ans. Channel is a medium to transmit an information either in physical or electrical form.



Channel capacity is defined by Shannon Hartley theorem.

Which states that channel capacity is given by:

$$C = B\log_2(1 + S/N)$$

C : Capacity in bits/sec.

B = Bandwidth in Hz

S : Signal Power

N : Noise Power

S/N : Signal to noise ratio in dB.

Q.2. (a) Enlist the different type of channels.

Ans. Different type of channels can be classified on the basis of the following criteria.

(a) Wireline channel: Here a physical medium is in the form of a wire.

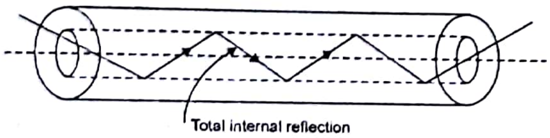


Fig. (Fiber optic Cable)

2. Wireless Electromagnetic Channel: Here the information travels through the free space. Physical size and the configuration depends on the frequency band used. Antenna is most effective device to radiate the signal, and for efficient, radiation of signal antenna should be longer than 1/10th of wavelength.

Q.3. (a) How channel are modelled?

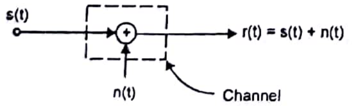
Ans. For the design of communication channel it is better to construct the mathematical model of channel that reflect the important characteristics of transmission medium.

The mathematical model for channel is used in the design of the channel encoder and modulator at the transmitter side and demodulator and channel decoder at the receiver.

Additive noise is considered in the mathematical model, because the transmitted signal $s(t)$ is corrupted by an additive random noise process $x(t)$. Thermal noise is considered statistically as a gaussian noise process when the signal undergoes attenuation in transmission through the channel, the received signal is $s(t) = \alpha.\, s(t) + n(t)$

α is the attenuation factor

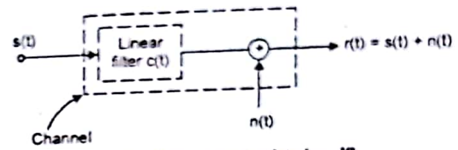
In some physical channels, such as wire-line telephone channels, filters are used to ensure that the transmitted signals do not exceed the specified bandwidth limitations and thus do not interfere with one another.



Such channel are generally characterised mathematically as linear filters channels with additive noise.

$$s(t) = s(t) * c(t) + n(t)$$

$$= \int_{-\infty}^{\infty} c(t)\, s(t-\tau)\, d\tau + n(t)$$



Q.3. (b) How power spectral density is obtained?

Ans. Power spectral density is very important parameter in designing the communication system and deciding the bandwidth of any signal.

Let $S(w)$: PSD (Power Spectral Density)

and $X(t)$: input signal

then $PSD = f[R_{XX}(\tau)]$

$$R_{XX}(T) = \int_{-\infty}^{\infty} x(t)x(t+\tau)dt$$

and $f[R_{XX}(T)]$: Fourier transform of Autocorrelation function (ACF) of the input signal $x(t)$.

Hence:

$$S(w) = \int_{-\infty}^{\infty} R_{XX}(\tau)\, e^{-j\omega t} dt$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} (x(t)\, x(t+\tau)dt) e^{-j\omega t} dt$$

Q.4. Derive the relation for total power transmission in AM.

Ans. In conventional AM the total power transmission can be summerized as

1. Carrier Power
2. Power of Lower side band
3. Power of upper side band.

The modulated signal can be written as:

$$X_{AM}(t) = A_C[1 + m_a\, m(t)]\cos w_c t$$

$$m(t) = A_m \cos w_m t$$

= modulating signal

$$m_a = \text{modulation index} = \frac{A_m}{A_c}$$

then

$$X_{AM}(t) = A_C\left[1 + \frac{A_m}{A_C}\cos\omega_m t\right]\cos w_c t$$

$$= A_C \cos \omega_c t + m_a A_C \cos \omega_c t \cos \omega_m t$$

$$= A_C \cos w_c t + \frac{m_a A_C}{2} [2 \cos \omega_c t \cos \omega_m t]$$

$$X_{AM}(t) = A_C \cos \omega_c t + \frac{m_a A_C}{2} \cos(\omega_C + \omega_m)t$$

$$+ \frac{m_a A_C}{2} \cos(\omega_C - \omega_m)t$$

Total Power = Carrier Power + Power of USB + Power of LSB

$$= \left(\frac{AC}{\sqrt{2}}\right)^2 + \left(\frac{m_a A_C}{\sqrt{2}}\frac{1}{2}\right)^2 + \left(\frac{m_a A_C}{\sqrt{2}\, 2}\right)^2$$

$$= \frac{A_C^2}{2} + \frac{m_a^2 A_C^2}{8} + \frac{m_a^2 A_C^2}{8}$$

$$= \frac{A_C^2}{2} + \frac{m_a^2 A_C^2}{4}$$

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

Where $P_C = \frac{A_C^2}{2}$

MODEL PAPER–I
SECOND TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

Time : 1.30 hrs. **M.M. : 30**

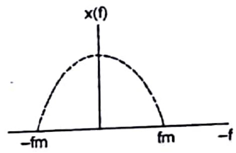
Note: *Attempt Question no. 1. which is compulsory and any two more questions from remaining. There is step-marking and use of scientific calculator (non-programmable) is permitted. Assume missing data, if any.*

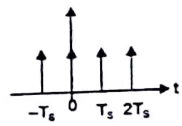
Q.1. (a) State and explain the sampling therem for

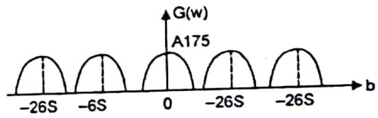
(a) Low Pass signals.

(b) Band pass signals.

Ans. Sampling Theorem for Low Pass Signal: A continuous: Time signal may be completely represented in this sample and recovered back if the sampling freq. is $f_s \geq 2f_m$. Where f_s is the sampling frequency and f_m is the maximum frequency percent in the signal.







(ii) Sampling Theorem (or Band Pass Signal): The sampling theorem for band pass signal may be expressed as under the band pass filter signal $x(t)$ whose maximum band width is f_m can be completely represented into and recovered from its sample it is sample at maximum rate twice the band width. Where 6 m is the maximum frequency. Component present in the signal. Hence the bandwidth is $2f_m$. The minimum sampling rate for band pass signal must be $4f_m$ samples per second.

Q.1. (b) Explain the single tone modulated *fm*.

Ans. Since the *fm* signal can be written as

$$X_{fm}(t) = \cos(w_c t + m_f \sin w_m t)$$

$$= \cos(w_c t) \cos(m_f \sin w_m t)$$

$$- \cos(w_c t) \cos(m_f \sin w_m t)$$

Now $\cos(m_f \sin w_m t)$ will contain only even harmonics.

so $\cos(m_f \sin w_m t) = J_0(m_f) + 2J_2(m_f) \cos 2w_m t + 2J_4(m_f) \cos 4w_m t + ...$

Similarly $\sin(m_f \sin w_m t)$ will contain only the odd harmonics.

so $\sin(m_f \sin w_m t)$

$$= 2J_1(m_f)\sin w_m t + 2J_3(m_f)\sin 3w_m t + \ldots$$

Hence Now in terms of Bessel functions we can write the equation for f_m as:

$$\begin{aligned} v(t) &= J_0(m_f)\cos w_c t - J_1(m_f)[\cos(w_c - w_m)t \\ &\quad - \cos(w_c + w_m)t] \\ &\quad + J_2(m_f)[\cos(w_c - 2w_m)t + \cos(w_c + 2w_m)t] \\ &\quad - J_3(m_f)[\cos(w_c - 3w_m)t - \cos(w_c + 3w_m)t] \\ &\quad + \ldots \end{aligned}$$

Q.2. (a) What is ISI and how to reduce it?

Ans. It arises due to the limitation of bandwidth and the effects of ISI..

1. In the absence of ISI and noise, the transmitted bit can be decoded correctly at the receiver.

2. The presence of ISI will introduce errors in the decisions at the receiver output.

Remedy to Reduce ISI:

1. It has been proved that the function which produces a zero intersymbol interference is a sinc function.

2. This is knowing as Nyquist Pulse shaping. The since pulse transmitted to have a zero ISI.

Q.2. (b) How much bandwidth is required for FM signal where modulating signal frequency is 5 KHz and maximum allowable deviation is 75 KHz.

Ans. As per the carson's rule for approximation of bandwidth for *fm*.

$$(BW)_{fm} = 2(f_m + \Delta f)$$

$$= 2fm\left(1 + \frac{\Delta f}{fm}\right)$$

Here Δf is 75 KHz

and fm is 5 KHz

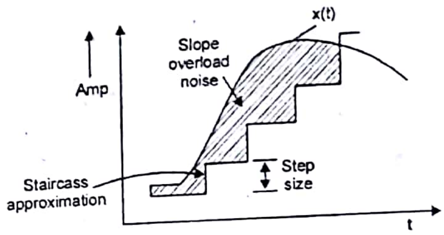
So
$$(BW)_{fm} = 2 \times 5\left(1 + \frac{75}{5}\right)$$

$$= 10\,(1 + 15)$$

$$= 160 \text{ KHz}$$

Q.3. What are the limitations of Delta Modulation and How they can be removed.

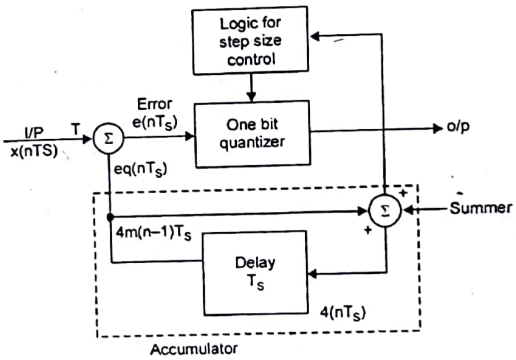
Ans. Slope Overload noise. The noise arrises because of large dynamic range of Input signal $x(t)$, thus so high that the staircase signal cannot approximate it, the step signal size 'Δ' becomes too small for stair case signal $u(t)$ to follow the step segment of $x(t)$. Hence there is a large error between the staircase approximated signal and the orignal input signal $x(t)$. This error is known as slope over load noise.



ADM (Adaptive Delta Modulation): To overcome the quantization due to slope overload noise, the stepwise is made adaptive to variatiuns in input signal $x(t)$. Particularly in the step segment of the signal $x(t)$, the step size is increased. Also if the input is varying slowly. The step size is reduced then the this method is known as ADM.

Show the transmitter of adaptive delta modulator. The logic for step size control is added in the diagram.

The step size control is a increases or decrease according to the specified, if one bit quantizer output is high, then stepwise may be doubled for next sample. If one bit quantizer output is low, then step size may be reduced by one step.



Q.4. Write short notes on following

(a) PWM

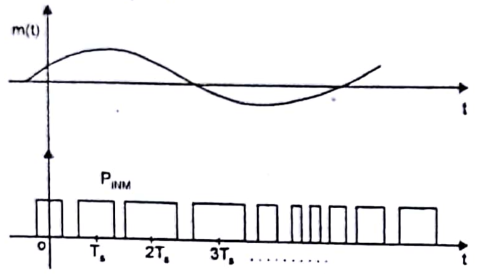
(b) Advantage of digital communication.

(c) FSK

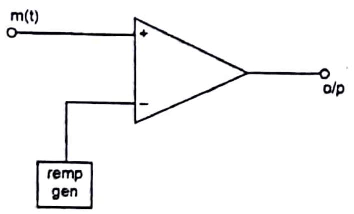
(d) Line Coding

Ans. (a) PWM stands for pulse width modulation or pulse duration modulation (PDM).

Here width of pulse changes in accordance with the amplitude modulating signal.



It can easily be generated by comparator



(b) Digital Communication is advantageous in following aspects:

(a) Noise immunity is better.

(b) System is programmable, hence more reliable.

(c) Regeneration of signal is possible so S/N ratio can be maintained within acceptable limits.

(d) Multiplexing becomes an easy task.

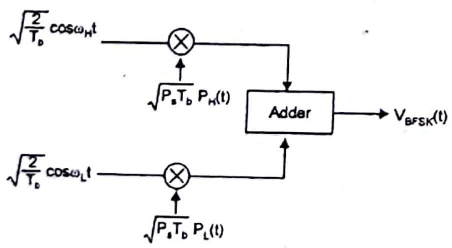
(e) Storage is also feasible.

(c) FSK: It stand for frequency shift keying. Here the frequency shift take place according to the bit pattern. For the binary frequency shift keying.

$$\rho_H(t) = \sqrt{2P_s}\cos(\omega_0 + \Omega)t$$

$$\rho_L(t) = \sqrt{2p_s}\cos(w_0 - \Omega)t$$

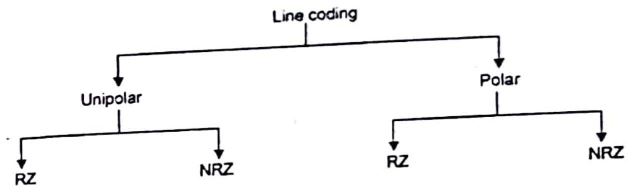
and FSK can easily be generated by the following circuit:



(d) Line Coding: For transmitting the digital information through the transmission line. The main aim of line coding is:

1. To reduce the required bandwidth
2. To reduce the dc component of the signal
3. To reduce the power requirement for signal transmission.
4. To provide the self synchronization.

Line coding can have classification on the basis of the following arrow diagram:



RZ : Return to zero

NRZ : Non Return to zero

MODEL PAPER–I
END TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

Time : 3 hrs hrs. **M.M. : 75**

Q.1.(a) When the input noise is white, the impulse response of the matched filter becomes

$$h(t) = CS(t_o - t)$$

where C is a positive real constant. Prove it.

Ans. Since

$$h(t) = F^{-1}[H(f)]$$

$$= \frac{2K}{No}\int_{-\infty}^{\infty} S^*(f)e^{-jwto}e^{jwt}df$$

$$= \frac{2K}{N_o}\left[\int_{-\infty}^{\infty} s(f)e^{j2\pi f(t_o-t)}df\right]$$

$$= \frac{2K}{N_o}[S(t_o - t)]$$

let

$$C = \frac{2K}{N_o}$$

So

$$h(t) = CS(t_o - t)$$

Hence proved.

Q.1.(b) Write the Properties of Cross-correlation.

Ans. Since cross-correlation of two variables X & Y can be written as

$$R_{XY}(t_1, t_2) = \overline{X(t_1)Y(t_2)}$$

& if $X(t)$ & $Y(t)$ are jointly stationary then

$$R_{XY}(t_1, t_2) = R_{XY}(\tau)$$

where

$$\tau = t_2 - t_1$$

Properties:

(1) $$R_{XY}(-\tau) = R_{YX}(\tau)$$

(2) $$|R_{XY}(\tau)| = \sqrt{R_X(0)R_Y(0)}$$

and

(3) $$|R_{XY}(\tau)| \le \frac{1}{2}[R_X(0) + R_Y(0)]$$

Q.1.(c) Explain the Quadrature Amplitude Shift Keying.

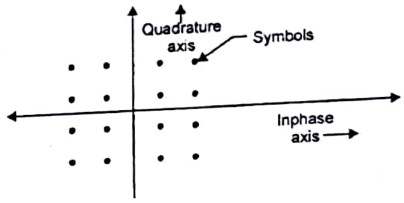
Ans. The General expression for *QASK* is

$$s(t) = x(t)\cos w_c t - y(t)\sin w_c t$$

&

$$g(t) = x(t) + jy(t)$$

in *QAM* Constellation is drawn between (Xi, Yi) & (Ri, Qi) can readily be evaluated. This type of signalling is used by 2400 bits/sec V.22 big computer modems.



I component is given by

$$x(t) = \sum_n x_n h_e\left(t - \frac{n}{D}\right)$$

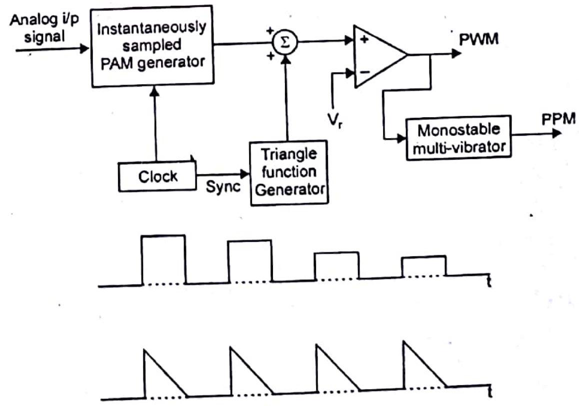
& Q component is given by:

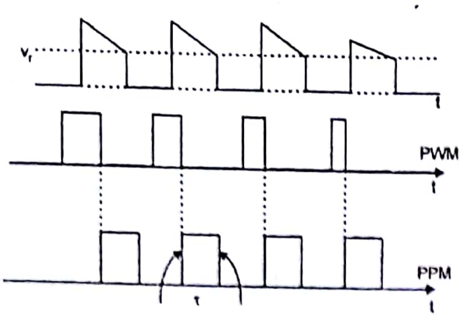
$$y(t) = \sum_n y_n h_e\left(t - \frac{n}{D}\right)$$

Q.1.(d) Explain the generation mechanism of *PTM* signals.

Ans. *PTM* stands for pulse time modulation & it consist of *PWM* & *PPM* signals. In the generation of *PTM* first *PAM* is generated then triangular waveform is generated & is superimposed on the *PAM* signal the combined waveform is compared with the reference voltage V_r.

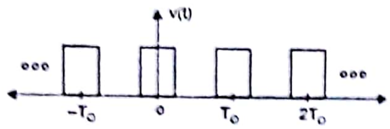
In this comparison PWM waveform is produced this PWM signal further drive the monostable multivibrator & it generates an o/p pulse of fixed duration (τ) which is occurring as a function of trailing edge of PWM signal.

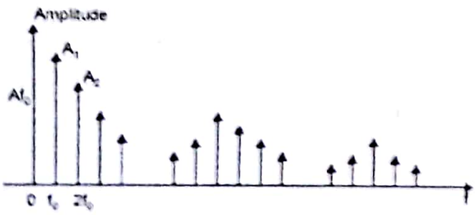


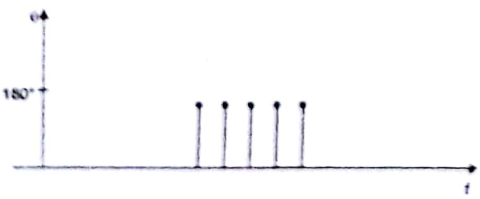


Q.2.(a) Explain the representation of pulse train.

Ans. Let a pulse train is having a duration τ







Here amplitude of n^{th} harmonic is given by

$$a_n = \frac{2V}{n\pi}\sin\left(\frac{n\pi\tau}{T_0}\right)$$

let $$X = \frac{n\pi\tau}{T_0}$$

$$\Rightarrow \qquad a_n = \frac{2V\tau}{T_0}\cdot\frac{\sin x}{x}$$

Here $\frac{\sin x}{x}$ is called of sampling function.

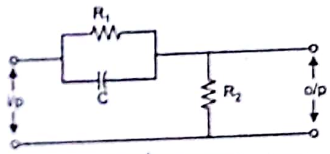
While $$\operatorname{sinc} X = \frac{\sin \pi X}{\pi X}$$

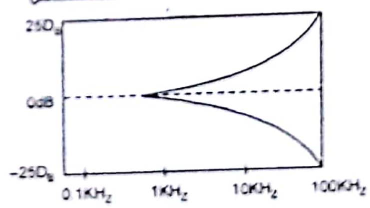
Q.2.(b) Explain the basics of Pre-emphasis & De-emphasis.

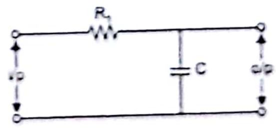
Ans. The low noise level at low frequency & the noise spectral density increases at rate of 6 db/octave & noise performance for *fm* is found to be better than Am.

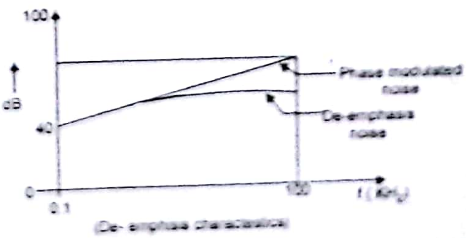
Performance of fm may degrade when we are talking of speech's clarity & this performance will depend on the frequency components & degradation will appear on the higher frequency range.

The situation can be improved by using pre-emphasis & de-emphasis circuits e.g. using pre-emphasis in the transmitter & De-emphasis in the receiver.









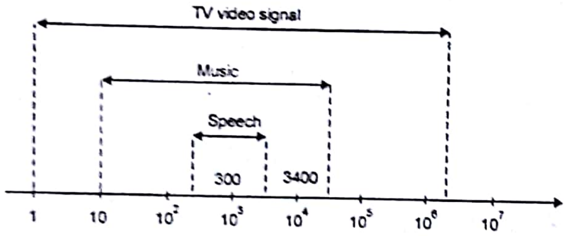
Q.3.(a) How the bandwidth requirement changes from application to application for analog information signal?

Ans. Bandwidth requirement for the speech signal is around 4 KHz. Experimental tests conducted on the speech signals indicate that around 80% of signal's spectral component lies from 300 Hz to 3400 Hz.

$$\text{Hence bandwidth} = (3400 - 300)\text{ Hz}$$
$$= (3100)\text{ Hz}$$

For the high fidelity signal as in music the band width is around 15 to 20,000 Hz.

For video signals around a bandwidth of 4 MHz is required & while a facsimile signal require & bandwidth of around 1000 Hz.



Q.3.(b) A telephone signal with a cut-off frequency of 4 KHz is digitized into 8 bit samples at the Nyquist sampling rate of fs = 2W. Assuming that raised cosine filtering is used with a roll off factor of unit. Calculate

(a) Baseband Transmission bandwidth

(b) Quantization S/N ratio.

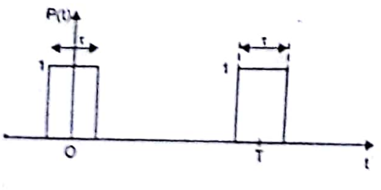
Ans. Transmission bandwidth is

$$B = (1 + P)\,W\,n$$
$$P = \text{roll off factor} = 1$$
$$W = \text{Cut-off freq.} = 4\text{ KHz}$$
$$n = \text{no. of bit required} = 8$$
$$\Rightarrow \quad B = 2 \times 4 \times 8 = 64\text{ KHz}$$
$$(S/N)\text{ dB} = 6n = 6 \times 8 = 48\text{ dB}.$$

Q.4.(a) How rectangular pulses are denoted mathematically in line coding scheme?

Ans. In line coding bit '1' or '0' are indicated by their rectangular pulses. Hence these of functions are indicated as

$$p(t) = \text{rect}\left(\frac{t}{\tau}\right)$$



$$p(t) = 0 \quad \text{for } t < -\frac{\tau}{2}$$
$$= 1 \quad -\frac{\tau}{2} \le t \le \frac{\tau}{2}$$
$$= 0 \quad t > \frac{\tau}{2}$$

Q.4.(b) Explain the Properties of Auto-correlation.

Ans. Auto-correlation is a process of correlating a signal $X(t)$ with itself.

e.g. $\quad R_v(\tau) \triangleq R_{VV}(\tau) \quad \triangleq$ Stands for: By definition

or the expected value of $V(t)$ & $V(t-\tau)$

or $\quad R_v(\tau) = \langle V(t+\tau)\,V^*(t)\rangle$

Properties:

(1) $\quad R_v(0) = P_V \text{ for } \tau = 0$

(2) $\quad |R_v(\tau)| = R_V(0)$

(3) $\quad R_v(-\tau) = R_V^*(\tau)$

e.g. $R_v(\tau)$ has hermitian symmetry & a maximum value at the origin equal to the signal power.

If $V(t)$ is real the $R_v(\tau)$ will be real & even & if $V(t)$ happens to be periodic than $R_v(t)$ will have same periodicity.

Let us consider two signals which result into a signal $z(t)$ as follow:

$$z(t) = V(t) \pm W(t)$$

the $\quad R_Z(\tau) = R_v(\tau) + R_w(\tau) \pm [R_{vw}(\tau) + R_{wv}(\tau)]$

But if $V(t)$ & $W(t)$ happens to be uncorrelated

then $\quad R_{vw}(\tau) = R_{wv}(\tau) = 0$

then $\quad R_Z(\tau) = R_v(\tau) + R_w(\tau)$

Hence the superposition principle holds for the uncorrelated signals.

Q.5.(a) What is intersymbol interference?

Ans. When some information is transmitted over the communication channel a linear distortion take place which indicate that the shape of waveform changes but no view frequency components are added in the spectrum. But it may take the form of ringing in which a pulse for example may have a tail added. This tail is a result of natural build-up & delay of energy in the inductive or capacitive elements in the transmission path.

This tail causes interference in deciding whether '0' or '1' is transmitted.

Q.5.(b) Explain the concept & type of bandwidth.

Ans. Bandwidth: It is that range of frequencies which can appreciable reproduce the signal at the receiving end.

Type:

(1) Absolute Bandwidth

(2) 3 dB bandwidth

(3) Noise equivalent bandwidth

(4) Null to Null bandwidth

(5) Occupied bandwidth

(6) Relative power spectrum bandwidth

Q.6.(a) Explain the phase locked loop.

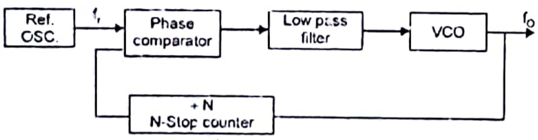
Ans. Heart of the frequency synthesizer is the phased locked loop. A stable oscillator produces a square wave reference frequency fr, which provides one of the i/p to the phase detector circuit. This frequency may be any convenient value. VCO generates a final output frequency f_0 and is designed such that it will tune over a frequency range from minimum to maximum frequency desired.

The o/p frequency is fed to the load & is also used to drive a programmable binary counter that provides the function of binary division.

Phase comparator is a circuit which produces a dc signal whose amplitude in proportional to the phase difference between the ref. signal fr & counter output fo/N.

This dc signal is filtered to smooth out noise & slow the response of the circuit to prevent overshoot or oscillations & is applied as the control input to the VCO.

When the phase difference between fr & fo/N is zero the dc o/p from the phase comparator is just exactly that needed to tune the VCO to the frequency N fr if a phase difference between the two frequency exists then VCO will change the direction of rising or falling of frequency so that the phase difference disappears.



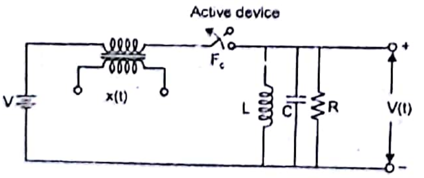
(Basic phase lock loop)

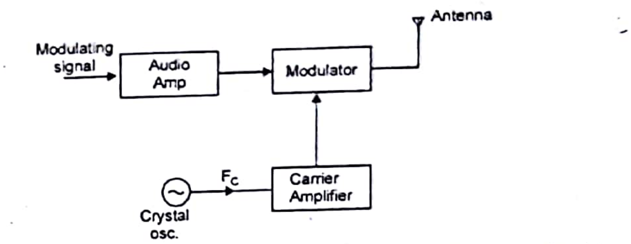
Q.6.(b) Explain the working of switching Modulators.

Ans. Square-law modulators are used for low power levels of modulation e.g. at power level lower than the transmitted values.

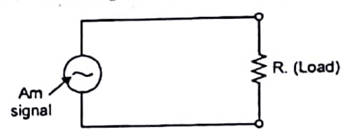
High level modulators are arranged so that undesired products never occur or if occurs in some proportion then there should not be any need to filter then out. This is accomplished with the aid of switching device.

Generally transistor is utilized to accomplish the modulation transistor closes every 1/fs seconds. RLC load called a tank circuit is tuned to resonate at fc so the switching causes tank to resonate (ring) sinusoidally.





Q.7.(a) Find the relation for power of the AM signal for the given circuit.



Ans. Let R be the equivalent resistance & $R = R_{in} + R_L$

$$X_{AM} = A_c \cos w_c t + \frac{mA_c}{2}\cos(w_c + w_m)t + \frac{mA_c}{2}\cos(w_c - w_m)t$$

$$\text{Power in the carrier} = \frac{\left(\frac{A_c}{\sqrt{2}}\right)^2}{R} = \frac{A_c^2}{2R}$$

Power in the upperside band = Power in the lower side band = Px.

$$\text{Hence sideband power} = 2(Px)$$

$$= 2\left[\left(\frac{\frac{mA_c}{2}}{\sqrt{2}}\right)^2\right]$$

$$= 2\left[\frac{m^2 Ac^2}{8R}\right] = \frac{m^2 A_c^2}{4R} = \frac{m^2}{2}\frac{A_c^2}{2R}$$

$$= \frac{m^2}{2} Pc$$

Hence

$$P_t = Pc + \frac{m^2}{2} Pc$$

$$= Pc\left[1+\frac{m^2}{2}\right]$$

For the case of many input modulating signals

$$P_t = Pc\left[1+\frac{m_1^2}{2}+\frac{m_2^2}{2}+\ldots+\frac{m_n^2}{2}\right]$$

where $$M_{eff} = \sqrt{m_1^2+m_2^2+\ldots+m_n^2}$$

Q.7.(b) Explain the Capture effect of FM.

Ans. It is a phenomenon that take place in the FM receiver, when two or more than two or two signals of equal amplitude arrive at the receiver.

Small amplitude variation, then causes the stronger signal to dominate the two & displacing the other signal from demodulation at o/p. It occurs while listening to a distant Fm station & co-channel interference occurs & o/p signal can be expressed as:

$$Y_D(t) = \phi_s(t) + \frac{d}{dt}\left[\arctan\frac{p\sin\phi_s(t)}{1+p\cos\phi(t)}\right]$$

A capture effect occurs when $A_i(t) = Ac$

P = amplitude envelope

$Q(t)$ is the phase information in the modulated signal.

Q.8.(a) What is effective isotropic radiated power (EIRP).

Ans. EIRP is defined as equivalent transmit power & is expressed mathematically as

$$EIRP = P_{in}A_t$$

P_{in} = Antenna i/p power (Watts)

A_t = Transmit antenna gain

EIRP can be expressed in log scale as

$$(EIRP)_{(dBW)} = P_{in(dBw)} + A_{t(dB)}$$

With respect to the transmitter power

$$P_{in} = P_t - L_{bo} - L_{bf}$$

so $$EIRP = P_t - L_{bo} - L_{bf} + A_t$$

P_{in} = antenna i/p power

L_{bo} = Back-off losses of HPA (decibels)

L_{bf} = Total branching & feeder loss (decibels)

A_t = Transmit antenna gain (decibels)

P_t = Saturated amplifier o/p power

Q.8.(b) Find the Geosynchronous Satellite orbital velocity.

Ans. Circumference of Geosynchronous orbit is

$$C = 2\pi r$$
$$= 2\pi(42164 \times 10^3)\ m$$
$$= 264790\ km$$

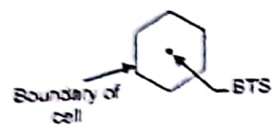
So the velocity of Geosynchronous satellite is

$$V = \frac{264790 \times 1000m}{24 \times 3600\ sec}$$

$$\boxed{V = 11,033\ km/hr.}$$

Q.8.(c) Explain the basic cellular system.

Ans. In a cellular system, mobile terminal unit is called as the mobile station & works inside the cell structure which is theoretically a hexagonal in shape as shown below.



BTS is base transceiver station & is used to code & decode the information & is situated at the centre of cell.

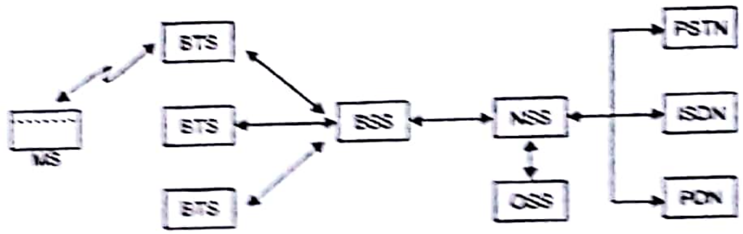
Other important part of the system are:

(1) Base station Subsystem (BSS)

(2) Network switching subsystem (NSS)

(3) Operational support subsystem (OSS)

(4) MSC (Mobile Services Switching Centre)



PSTN: Public Switched Telephone Network
ISDN: Integrated Services Digital Network
PDN: Public Digital Network

MODEL PAPER–II
FIRST TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

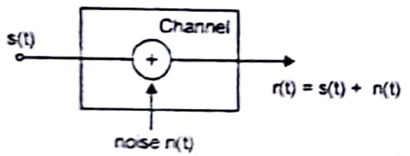
Time : 1.30 hrs. **M.M. : 30**

Note: *Attempt Question no. 1. which is compulsory and any two more questions from remaining. There is step-marking and use of scientific calculator (non-programmable) is permitted. Assume missing data, if any.*

Q.1. (a) On the basis of mathematical modelling, explain the type of channel.

Ans. Communication channels can be classified as follows on the basis of their modelling:

(a) Additive Noise Channels: It is the simplest type of channels. During the transmission of signal through the channel a gaussian noise is added.



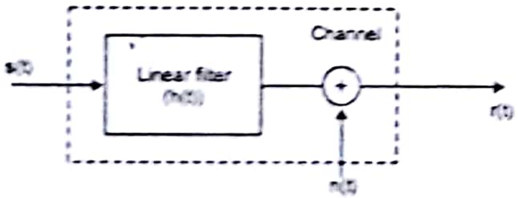
hence the signal $s(t)$ is corrupted by noise of random nature.

When the attenuation take place in the signal while travelling through the medium the signal received can be written as:

$$r(t) = a\,p(t) + n(t)$$

where a represent the attenuation factor.

(b) Linear Filter Channel: In some physical channel such as wireline telephone channels, filters are used to ensure that the transmitted signals do not exceed specified bandwidth limitations and thus do not interfere with each other, such channels are characterised with linear filter with additive noise.

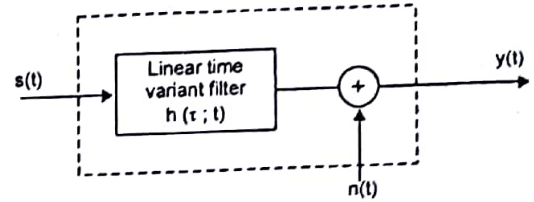


$$r(t) = s(t) \otimes h(t) + n(t)$$

$$= \int_{-\infty}^{\infty} s(\tau) h(t-\tau)\, d\tau + n(t)$$

Q.1. (c) Linear Time-Variant Filter Channel: These are those physical channel which result in time variant channels such channels are water acoustic channels and

ionospheric radio chennels. Such linear filters are characterized mathematically as having impulse response $h(\tau; t)$ where $h(\tau; t)$ is the response of the channel at time t, due to impulse applied at time $t - \tau$ where τ represents the 'age' elapsed time.



where
$$r(t) = s(t) \otimes h(\tau; t) + n(t)$$

$$= \int_{-\infty}^{\infty} h(\tau; t)\, s(t-\tau) d\tau + n(t)$$

Where
$$h(\tau; t) = \sum_{k=1}^{L} a_k(t)\delta(\tau - \tau_k)$$

Where L is the no. of paths available.

Q.2. (a) Explain the significance of binomial distribution function.

Ans. Binomial distribution function is significant only where Bernaulli trial is to be made and is very useful in context of the digital transmission because here two states exist low (0) and high (1).

If probability of correct result is q

Then probability of wrong result is $(1 - q)$

Then out of n trials the probability of K events to be correct N.

$$P_X(K) = {}^nC_K\, q^K (1-q)^{n-k}$$

$$= \frac{n!}{(n-k)!\,k!} q^k (1-q)^{n-k}$$

Q.2. (b) Explain the difference between the binomial and Poisson distribution function.

Ans. Mean Value in the binomial distribution can be calculated as $m = nq$

and
$$\sigma^2 = nq\,(1-q).$$

Poisson distribution can be related to the emission of electron (or shot noise) Arrival of telephone call or data transmission when error rate is low.

Let us take an event where probability of occurrence of an event is very small and occurs in an interval of δT is given by $\lambda\,\delta T$.

If successive occurrences are statistically independent then probability of K occurrences in time T is given by Poisson distribution as

$$P_X(k) = \frac{a^k}{k!} e^{-a}$$

where
$$a = \lambda T$$

Here
$$m = \alpha \text{ and } \sigma^2 = \alpha$$

Binomial distribution is not suitable for large no. of trails and when probability certain event is very small. Then in such cases, this can be approximated by poiss with $\alpha = nq$.

Q.3. (a) Differentiate between AM. FM and PM

Ans.

AM	FM	PM
(Amplitude Modulation)	Frequency Modulation	Phase Modulation
1. In this, amplitude of the carrier wave varies.	In this, frequency of the carrier varies	In this phase of the carri varies.
2. It operates in medium frequency and high frequency ranges.	It operates in VHF and UHF frequency range	It operaties in VHF and UHF frequency range
3. It is dependent on the modulation depth which governs the transmitted power.	It is independent of modulation depth	It is independent of the modulation depth.
4. Envelop of AM wave is dependent on the modulating signal	Envelop of FM wave is constant	Envelop of PM is constan
5. At zero crossing perfect regularity between the spacing of wave	At zero crossing no pefect regularity in their spacing	At zero crossing no perfec regularity in their spacing

Q.3. (b) What is the need of Modulation?

Ans. Modulation is very important while transmitting the information. Important reasons for modulation are as follow:

(a) For decreasing the antenna height

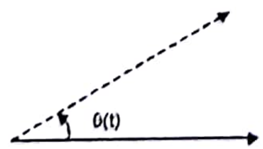
(b) For multiplexing.

(c) For avoiding the attenuation of low frequency signal.

(d) For increasing the speed of transmission

(e) For providing the security to the information.

Q.4. Derive the relation for frequency Modulation.



Ans. Since FM is a type of angle Modulated system so the expression can be written as

$$X_{fm}(t) = A_C \cos(\phi_i)$$
$$\phi_i : \text{Instantaneous phase}$$
$$\phi_i = \text{Constant phase + time varying phase}$$
$$= w_c t + \theta(t)$$

and since $\frac{d\theta(t)}{dt} = w$

Integrating on both the sides

$$\theta(t) = \int_0^t w\,dt$$

Here w represent the change in frequency and $w \propto m(t)$

or
$$w = k_f m(t)$$

k_f is frequency sensitivity factor having dimensions in Hz/volt

So
$$\theta(t) = \int_0^t k_f m(t)\,dt$$

So frequency modulated waveform can be expressed as

$$X_{fm}(t) = A_c \cos\left(w_c t + \int_0^t k_f m(t)\,dt\right)$$

Let
$$m(t) = A_m \cos w_m t$$

$\Rightarrow$
$$X_{fm}(t) = A_c \cos\left(w_c t + \int_0^t A_m \cos \omega_m t\,dt\right)$$
$$= A_c \cos\left(\omega_c t + k_f A_m \int_0^t \cos\omega_m t\,dt\right)$$
$$= A_c \cos\left(w_c t + \frac{K_f A_m}{\omega_m} \sin \omega_m t\right)$$

$$\Rightarrow A_c \cos\left(\omega_c t + \frac{k_f Am}{2\pi f_m} \sin\omega_m t\right)$$

Here
$$\Delta f = \text{frequency deviation} = \frac{K_f A_m}{2\pi}$$

So
$$X_{fm}(t) = A_c \cos\left(w_c t + \frac{\Delta f}{fm} \sin\omega_m t\right)$$

and
$$\frac{\Delta f}{fm} = m_f = \text{modulation index}$$

So
$$X_{fm}(t) = A_c \cos(w_c t + m_f \sin w_m t)$$

Which is the required relation for fm.

MODEL PAPER–II
SECOND TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

Time : 1.30 hrs. **M.M. : 30**

Note: *Attempt Question no.1. which is compulsory and any two more questions from remaining. There is step-marking and use of scientific calculator (non-programmable) is permitted. Assume missing data, if any.*

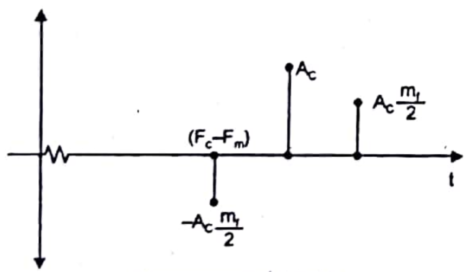
Q.1. (a) Draw the line and phasor diagram of NBFM.

Ans. Since $X_{fm}(t) = A_C \cos w_c t - A_c m_f \sin w_m w_c t$

for *NBfm* (Narrow Band *fm*)

$$X_{fm}(t) = A_C \cos \omega_c t + \frac{m_f A_C}{2} \cos(w_c + w_m)t$$

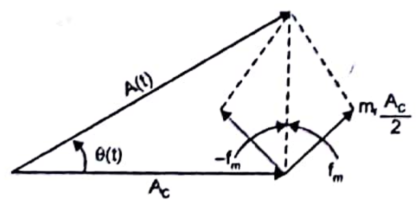
$$-\frac{mf\ A_c}{2} \cos(w_c - w_m)t$$



(Line diagram for NB F_m)

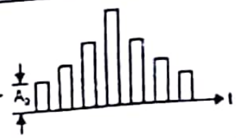
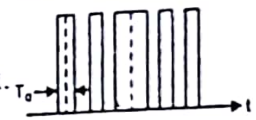
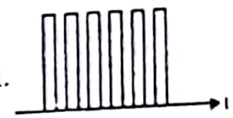
Line Diagram for NBFM

Phasor diagram is as follow:



Q.1. (a) Compare the various parameters of PAM PWM and PPM

Ans.

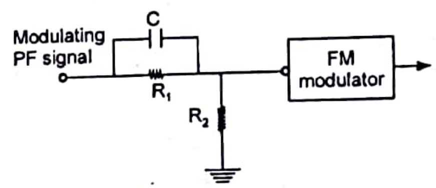
PAM	PWM	PPM
1. 	1. 	1. 
2. Amplitude is proportional to amplitude of modulating signal.	Width is proportional to amplitude of modulating signal.	relative position of pulse is proportional to amplitude of modulating signal.
3. BW depends on width of pulse	BW depends on rise time of pulse	BW depends rise time of pulse.
4. Instantaneous power of transmiter varies	Power of transmitter varies	Power is constant.
5. Noise, interference is high	Noise, Interference is minimum	Noise interference is minimum.

Q.2. Explain the Pre-emphasis and De-emphasis networks used to improve the performance in fm system.

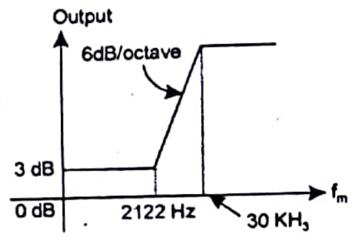
Ans. Pre-emphasis: The artificial boosting of higher modulating frequency is called pre-emphasis. The amount of pre-emphasis in US FM transmission and sound transmission in TV has been standardized at 75 m sec.

The pre emphasis circuit is basically a high pass filter.

The boosting of higher frequency modulating signal is achieved by using pre-emphasis circuit.



Typical Pre-emphasis circuit

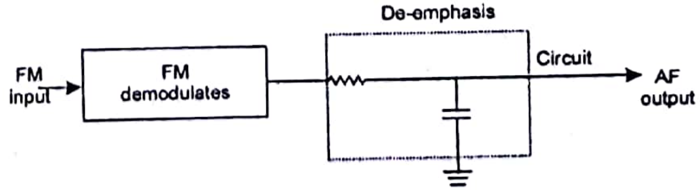


Pre-emphasis characterstic

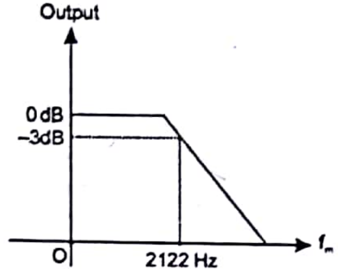
As fm increases, reactance of C decreases and modulating voltage applied to FM modulator goes on increasing. The frequency response characteristic oif RC high pass network is shown in figure. The boosting is done according to this pre arranged curve.

D-Emphasis: The artifical bosting given to the higher modulating frequency in the process of pre-emphasis is compensated at the receiver by a process called D-emphasis.

The demodulated FM is applied to the De-emphasis circuit with increase in FM the creactance of C goes on decreasing and the output of de-emphasis circuit will also reduced.



Typical de-emphasis circuit



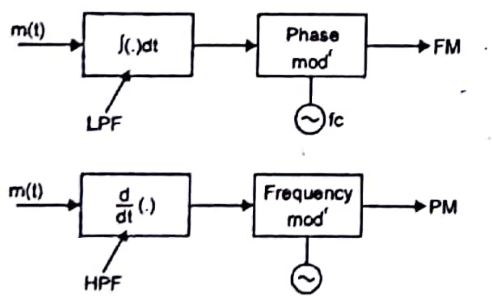
Los pass characteristic

FM improvement Through Pre-Emphasis and D-Emphasis : It has been proved in FM, the noise has a greater effect on the higher modulating frequency. This effect be reduced by increasing the value of modulation index (*mf*) for higher modulating fr (*fm*). This can be done by increasing the deviation Δf and Δf can be increased by increas the amplituted of modulating signal at higher modulating frequencies.

Thus if we boost the amplitude of higher frequency modulating signals artificia then it will be possible to improve the noise immunity at higher modulating frequenc This problem can be solved by using circuit of pre-emphasis and de-emphasis.

Q.2. (*b*) How FM can be generated using PM and vice-versa.

Ans. Both are the two sides of a single coin. If fm is used to transmit the signa can be received on the PM system and if PM is used then it can be received on the F system following block diagrams show the core relation between the two:



Q.3. Draw a block schematic of PCM and explain each.

Ans. In this section we shall discuss the *PCM* generator (*i.e.,* transmitter) from path and receiver). In this section, we discuss the PCM generator (*i.e.,* transmitter fro a practical point of view. Figure shows a practical block diagram of a PCM generator

In PCM generator of figure, the signal $x(t)$ is first passed through the low-pass filt of cutoff frequency f_m *Hz*. This low-pass filter blocks all the frequency components whic are lying above f_m Hz.

The means that now the signal $x(t)$ is bandlimited to f_m Hz. The sample and hol circuit them samples this signal at the rate of f_s. Sampling frequency *fs* is selecte sufficiently above nyquist rate to avoid aliasing *i.e.,*

$$f_s > 2f_m$$

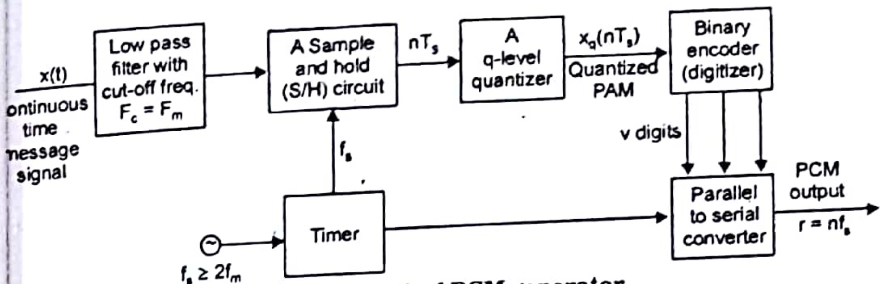
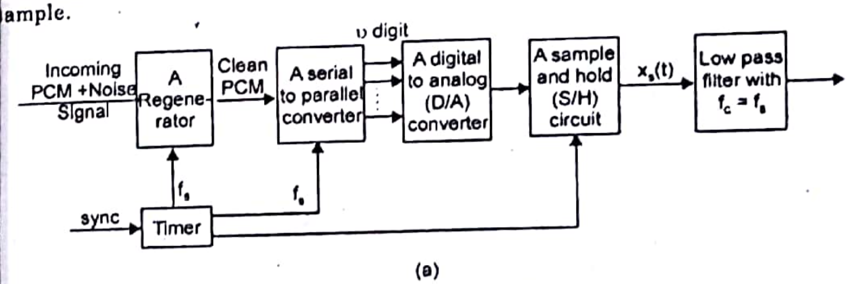


Fig. A practical PCM generator

In figure 1, the output of sample and hold circuit is denoted by $x(nT_s)$. This signal (*nTs*) is discrete in time and continuous in amplitude. A *q*-level quantizer compares nput $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ ith its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which esults in minimum distortion or error. This error is called quantization error. Thus, utput of quantizer is a digital level called $x_q(nT_s)$. Now, the quantized signal level $x_q(nT_s)$ s given to binary encoder. This encoder converts input signal to '*v*' digits binary word. hus $x_q(nT_s)$ is converted to '*v*' binary bits. This encoder is also known as digitizer. Also n oscillator generates the clocks for sample and hold circuit and parallel to serial onverter. In the pulse code modulation generator discussed above, sample and hold, uantizer and encoder combinely form an analog to digital converter (ADC).

PCM Receiver: In this section, we shall discuss a PCM receiver from practical oint of view. Figure (*a*) shows the block diagram of PCM receiver and fig are (*b*) shows he reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulse nd removes the noise. The signal is the converted to parallel digital words for each ample.



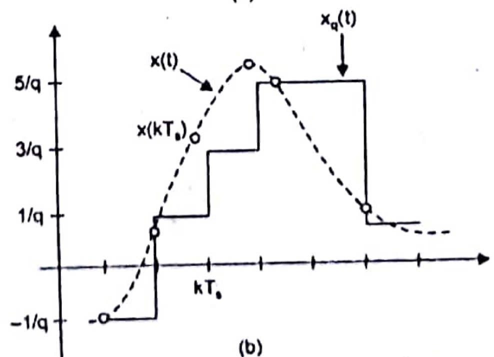


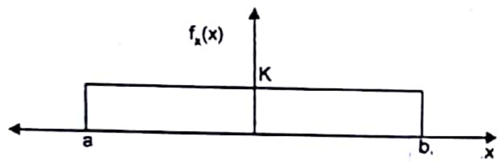
Fig. (*a*) PCM receiver (*b*) Reconstructed waveform.

Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit, is allowed to pass through a lowpass reconstruction filter to get the appropriate original message signal denoted as $y(t)$.

Q.4. Write short notes on the following:

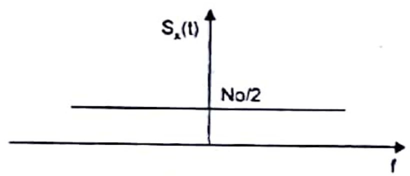
(a) UDF **(b) White Process** **(c) Direct FM Generation.**

Ans. UDF is a uniform distribution function and is a type of probability model. Its response is uniform with respect to the frequency.



Its value remain constant say k with respect to the all frequencies and hence its contains all the frequencies and all components are equally probable.

(b) The term white process is used for those type of random processes in which all frequency components appear with equal power e.g. power spectral density is constant for all frequencies.



Power spectral density of white process

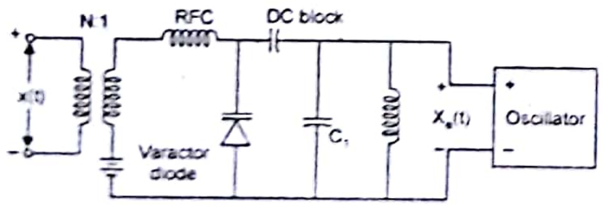
Power content of a white process is

$$s_x(f) = c \text{ then}$$

$$P_x = \int_{-x}^{x} s_x(f)df = \int_{-x}^{x} c df = \infty$$

(c) Direct FM Generation: Here a reactance modulation is used in which the capacitance change in direct proportion of the input applied signal $m(t)$. This change in capacitance further changes the frequency.

Since $$f = \frac{1}{2\pi\sqrt{C'L}}$$



$$c(t) = C_0 - x(t)$$

and if $x(t)$ is small enough and slow enough then oscillator produces $x_c(t) = A_c \cos \theta_c(t)$

and $$\frac{d}{dt}\theta_c(t) = \frac{1}{\sqrt{LC(t)}}$$

Which is having dimensions of frequency and hence frequency is a function of modulating signal $m(t)$.

MODEL PAPER–II
END TERM EXAMINATION
FIFTH SEMESTER (B.TECH)
COMMUNICATION SYSTEMS

Time : 3 hrs. **M.M. : 75**

Q.1.(a) Find out

(a) Power levels in dBm for signal levels of 10 mW & 0.5 mW.

(b) Difference between the two power levels in dB.

Ans. Power level in dBm

$$= 10\log\left(\frac{10\,\text{mW}}{1\,\text{mW}}\right) = 10\,\text{dBm}$$

and $$dBm = 10\log\left(\frac{0.5\,\text{mW}}{1\,\text{mW}}\right)$$

$$= -3\text{ dBm}$$

(b) Difference between the two power levels is

$$10\log(10\text{ mW}) - 10\log(0.5\text{ mW})$$

$$= 10\log\left(\frac{10\,mW}{0.5\,mW}\right) = 13\,dB$$

that mean 10 mW is 13 dB higher than 0.5 mW.

Q.1.(b) Explain the call procedure for making a call from *PSTN* to mobile phone user.

Ans. 1. Telephone goes off-hook & completes the loop receives the dial tone & then user dials the mobile No. of user.

2. The PSTN transfers the mobile No. of user to the MTSO.

3. The cellular network MTSO receives the call from PSTN & translates the received digits & locates the nearest BTS to the Mobile station.

4. If Mobile phone user is available then a positive page response is sent over a reverse channel to the cell-site controller, which is forwarded to the network switch (MTSO).

5. Cell-site Controller assigns an idle user channel to the mobile unit & then instructs the mobile unit to tune to the selected channel.

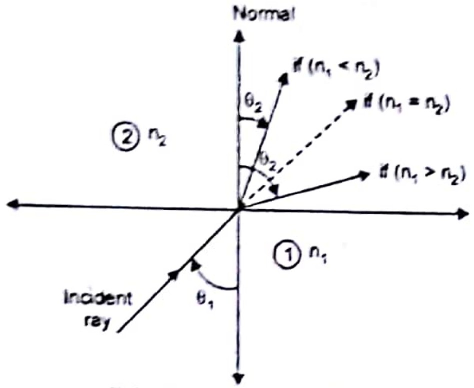
6. Mobile unit sends verification of channel tuning through the cell-site controller.

7. Then cell-site controller sends an audible call progress tone to the subscribers mobile telephone.

8. The mobile answers, the switch terminates the call progress tones & the conversation begins.

Q.1.(c) What is the Snell's Law?

Ans. Snell's law explains how light reacts when it meets the two interface when they are having different refractive index.



Refractive model of snell's low

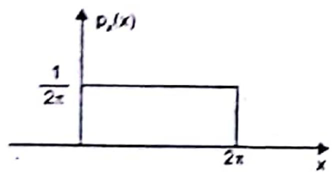
n_1 : refractive index of 1st medium

n_2 : refractive index of 2nd medium

Then mathematically Snell's Law states that:

$$n_1 \sin\theta_1 = n_2 \sin\theta_2$$

Q.1.(d) X is uniformly distributed function as shown below, find $E[x]$, $E[x^2]$, $E[\cos x]^2$ & $E[(x - m_x)^2]$.



Ans. (1) $$E[x] = \int_{-\infty}^{\infty} p_x(x)\,dx = \frac{1}{2\pi}\int_0^{2\pi} x\,dx = \frac{1}{2\pi}\left[\frac{x^2}{2}\right]_0^{2\pi} = \pi$$

(2) $$E[x^2] = \int_{-\infty}^{\infty} x^2 p_x(x)\,dx = \frac{1}{2\pi}\int_0^{2\pi} x^2\,dx = \frac{1}{2\pi}\left[\frac{x^3}{3}\right]_0^{2\pi} = \frac{4}{3}\pi^2$$

(3) $$E[\cos x] = \int_{-\infty}^{\infty} \cos(x)\,p_x(x)\,dx = \frac{1}{2\pi}\int_0^{2\pi} \cos x\,dx = \frac{1}{2\pi}[\sin x] = 0$$

(4) $$E[(x-\pi)^2] = \int_{-\infty}^{\infty} (x-\pi)^2\,p(x)\,dx = \frac{1}{2\pi}\int_0^{2\pi} (x-\pi)^2\,dx$$

$$= \frac{1}{2\pi}\left(\frac{(x-\pi)^3}{3}\right)_0^{2\pi}$$

$$= \frac{1}{2\pi}\left(\frac{\pi^3}{3} - \frac{-\pi^3}{3}\right) = \frac{\pi^2}{3}$$

Q.2.(a) Express the given periodic signal in terms of exponential Fourier series.

Ans. Let the periodic signal be given by

$$V(t) = A_n \cos(2\pi n f_0 t + \phi_n)$$

$$= A_n\left[\frac{e^{j(2\pi n f_0 t + \phi_n)} + e^{-j(2\pi n f_0 t + \phi_n)}}{2}\right]$$

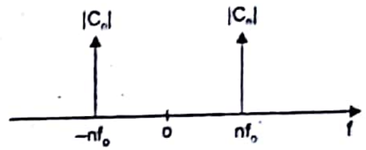
$$= \frac{A_n e^{j\phi_n}}{2} e^{j2\pi n f_0 t} + \frac{A_n}{2} e^{-j\phi_n} e^{-j2\pi n f_0 t}$$

$$= C_n e^{j2\pi n f_0 t} + C_{-n} e^{-j2\pi n f_0 t}$$

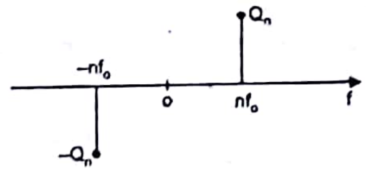
Here C_{-n} is a complex conjugate of C_n. Here $|C_n| = \frac{A_n}{2}$ in combined form $V(t)$ can be expressed as

$$V(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f_0 t}$$

Hence it comes out to be a double sideband spectrum



Amplitude spectrum



Q.2.(b) A PCM signal is to have a signal to noise ratio of 40 dB. The signals are speech & an rms-to-peak ratio of –10dB is allowed for. Find the no. of bits per code word required.

Ans. $$\left(\frac{S}{N}\right)_q = 3K^2 2^{2n}$$

taking $10\log_{10}$ on both the sides:

$$10\log\left(\frac{S}{N}\right)_q = 10\log 3 + 20\log k + 20\,n\log 2$$

$$\left(\frac{S}{N}\right)_q = 4.77 + (k)\,dB + 6.02\,n$$

$$40 = 4.77 - 10 + 6.02n$$

$$\Rightarrow \quad n = \frac{40 - 4.77 + 10}{6.02} = 7.5 \approx 8$$

Q.3.(a) Differentiate between Binits, Bits & Bauds.

Ans. Binary alphabet has only two symbols '0' & '1' which are known as binits a closely related word 'bit' is the unit of information (derived from binary unit).

Since $I = \log_2\left(\frac{1}{P_{sym}}\right) = -\log_2 P_{sym}$

and if some group of symbols are to be enclosed then M-ary encoding scheme is used

$$M = 2^n$$

M = Group of Messages (No. of)

n : no. of bits used to encode

For example if four symbols are used then a group of two bits are used such as 00, 01, 10 & 11 & also referred to as quaternary encoding.

The symbols are modulated into waveforms, each symbol occupies a given time termed as the symbol period denoted by Tsym & the transmission rate is measured in bands. Here 1 baud is referred as 1 symbol per second let T_b be the bit duration then

$$R_B = \frac{1}{T_B}$$

Also $\quad T_{sym} = mT_b$

& it follow that transmission rate are related by

$$R_{sym} = \frac{R_s}{m}$$

Q.3.(b) Find the Hilbert transform of the cosine signal.

Ans. Most of the Hilbert transform praise follow from the property of phase shift of Quadrature filter.

Lt i/p is $\quad x(t) = A\cos(\omega_0 t + \phi)$

then

$$\hat{X}(f) = -j\, sgn\, f X(f)$$

$$= \frac{-j}{2} A\left[\delta(f - f_0)e^{j\phi} + \delta(f + f_0)e^{-j\phi}\right] Sgn\, f$$

$$= \frac{A}{2j}\left[\delta(f - f_0)e^{j\phi} = \delta(f + f_0)e^{-j\phi}\right]$$

$$\Rightarrow \quad \hat{X}(t) = A\sin(\omega_0 t + \phi)$$

$\because$ $\hat{X}(f)$ relation has been taken from Quadrature filter property.

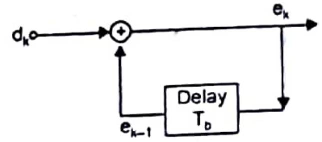
Q.4(a) Explain the concept of differential encoding.

Ans. In differential encoding if two different bits are coming then '1' is transmitted & if at the i/p same bits are coming then '0' is transmitted.

$\Rightarrow$ it is following Ex-Ring rule

$$0 \oplus 0 = 0$$
$$0 \oplus 1 = 1$$
$$1 \oplus 0 = 1$$
$$1 \oplus 1 = 0$$

& logical circuit as follow:



$$e_K = d_K \oplus e_{k-1}$$

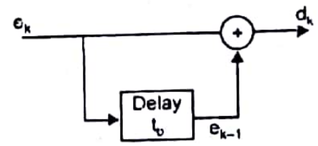
let $\quad d_k$ = data bits

e_k = encoded bits

& $\quad e_{k-1}$ = delay bits

d_K	0	1	0	1	1	0	1
e_{K-1}	0	0	1	1	0	1	1
e_K	0	1	1	0	1	1	0

At the receiver process is different



let $\quad e_k = 0110110$

e_K	0	1	1	0	1	1	0
e_{K-1}	0	0	1	1	0	1	1
d_K	0	1	0	1	1	0	1

Q.4.(b) Explain the Wiener-Kinchine theorm.

Ans. Wiener-Kinchine theorem states that to calculate power spectral density, we first calculate Auto correlation function of the given signal then calculate the Fourier transform of this autocorrelation.

e.g. $\quad G_v(f) = F[R_v(\tau)]$

$$\triangleq \int_{-\infty}^{\infty} R_v(\tau)e^{-j2\pi f\tau}\, d\tau$$

Inverse of this is also true e.g.

$$R_v(\tau) = F^{-1}[G_v(f)]$$

$$\triangleq \int_{-\infty}^{\infty} G_v(f) e^{j2\pi f\tau} df$$

Hence we can write Fourier transform pairs as

$$R_v(f) \leftrightarrow G_v(f)$$

is $V(t)$ is an energy signal then

$$G_v(f) = |V(f)|^2$$

if $V(t)$ is a periodic power signal then

$$V(t) = \sum_{n=-\infty}^{\infty} c(nf_0) e^{j2\pi n f_0 t}$$

Weiner-Kinchine Theorm gives the *PSD* or Power spectrum as

$$G_v(f) = \sum_{n=-\infty}^{\infty} |C(nf_o)|^2 \delta(f - nf_o)$$

Q.5.(a) What is the Pulse shaping?

Ans. The signal when transmitted through the channel to the transmitter can equivalently be represented as follows:

$$V_i(f) \longrightarrow \boxed{H_T(f)} \longrightarrow \boxed{H_C(f)} \longrightarrow \boxed{H_R(f)} \longrightarrow V_o(f)$$

Hence $\quad V_o(f) = V_i(f)[H_T(f) \cdot H_C(f) \cdot H_R(f)]$

& $\quad V_o(t) = F^{-1}[V_o(f)]$

To shape the output pulse, its spectrum is shaped by adjustment of $H_T(f)$ or $H_R(f)$ or both.

Pulse shaping is used to avoid the ISI.

Nyquist pulses are used to avoid ISI. Raised cosine response.

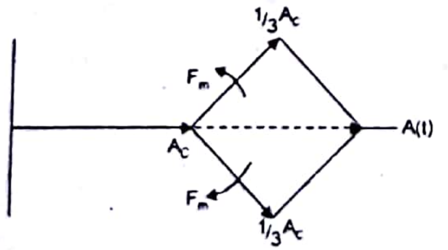
Q.5.(b) Give a phasor relationship of AM signal.

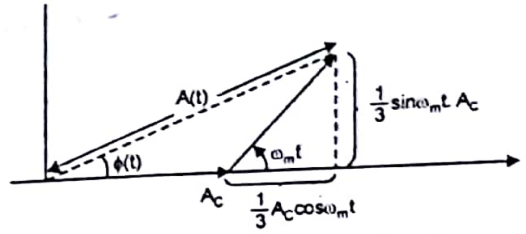
Ans. After modulation the AM signal consist of three major components:

(1) Pure carrier

(2) Upper sideband

(3) lower sideband

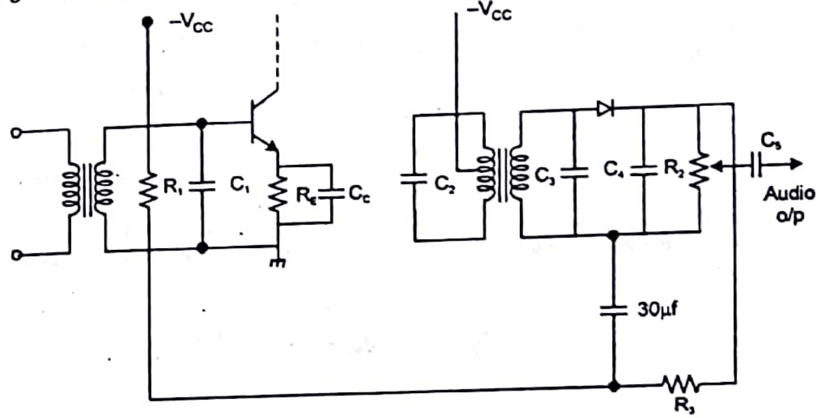


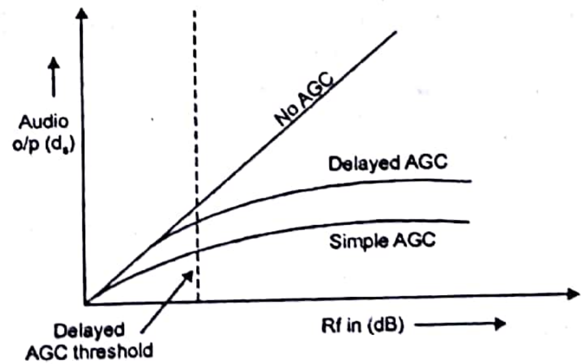


Q.6.(a) Why Automatic Gain Control is needed in the receiver circuits?

Ans. Without AGC circuit the receives may produce blasting sound if sudden increase is applied at the input.

When the input signal increases, the AGC biasing voltage increases which causes the gain of amplifier stages to decreases.





Three type AGC's are used:

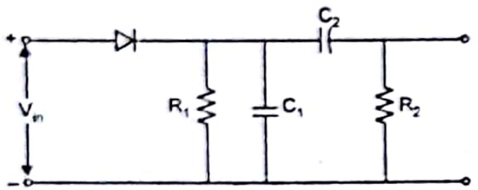
(a) Simple AGC: Used in simplex domestic receivers. Here the AGC bias starts increasing as soon as the i/p received signal starts going beyond the background noise level and the receiver becomes less sensitive.

(b) **Delayed AGC** is used for some better communication receivers. Delayed AGC is obtained until after a certain threshold value is exceeded and after that the gain increases proportionally with the i/p signal.

Maximum sensitivity is utilized to produce the biasing gain for AGC circuit.

Q.6.(b) Explain the relation between carrier frequency & bandwidth of signal in envelope detector.

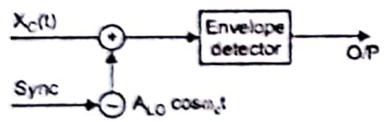
Ans. Envelope detector consist of a rectifying diode followed by the RC low pass filter. It can detect the information only with the modulated signal having carrier.



Here the diode is assumed to be piecewise linear. R_1C_1 acts as a low pass filter, responding only to variations in the peak of V_{in} provided that

$$W << \frac{1}{R_1C_1} << f_c$$

Hence $f_c >> W$, where W is the message signal bandwidth. Variation in the demodulated signal are filtered & DC voltage is proportional to the carrier amplitude. This voltage is fed back to the previous stage to automatically control the voltage to compensate for fading & for demodulating the DSB-SC signal a synchronous carrier is used at the receiver.



Q.7.(a) Find the relation for average powers of FM.

Ans. Let us take the case of sinusoidal FM. Her Peak value of signal spectrum of FM are given by

$$E_{n(max)} = J_n(\beta)E_{C(max)}$$

or
$$E_n = J_n(\beta)E_C$$

or
$$P_n = \frac{E_n^2}{R}$$

$$\text{Total Power} = P_T = P_c + 2(P_1 + P_2 + ...)$$

or
$$P_T = \frac{E_c^2}{R} + \frac{2}{R}(E_1^2 + E_2^2 + ...)$$

In terms of unmodulated carries & the bessel function coefficients, this is:

$$P_T = \frac{E_c^2J_0^2(\beta)}{R} + \frac{2E_c^2}{R}(J_1^2(\beta) + J_2^2(\beta) + ...)$$

$$= \frac{E_c^2}{R}\left[J_0^2(\beta) + 2(J_1^2(\beta)) + J_2^2(\beta) + ...\right]$$

Q.7.(b) Find the relation for noise equivalent bandwidth.

Ans. Filter white noise usually have finite power.

Let N = average noise power

$$N = \frac{N_0}{2}\int_{-\infty}^{\infty}|H(f)|^2\,df = N_0\int_0^{\infty}|H(f)|^2\,df$$

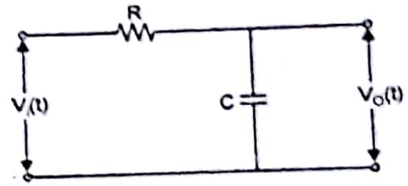
⇒ B_N = Noise equivalent bandwidth

&
$$B_N \triangleq \frac{1}{g}\int_0^{\infty}|H(f)|^2\,df$$

$$g = |H(f)|^2_{max}$$

$$N = gN_0B_N$$

Here B_N equals the bandwidth of an ideal rectangular filter.



at the filter's centre frequency ($f = 0$)

$$|H(0)|^2 = 1$$

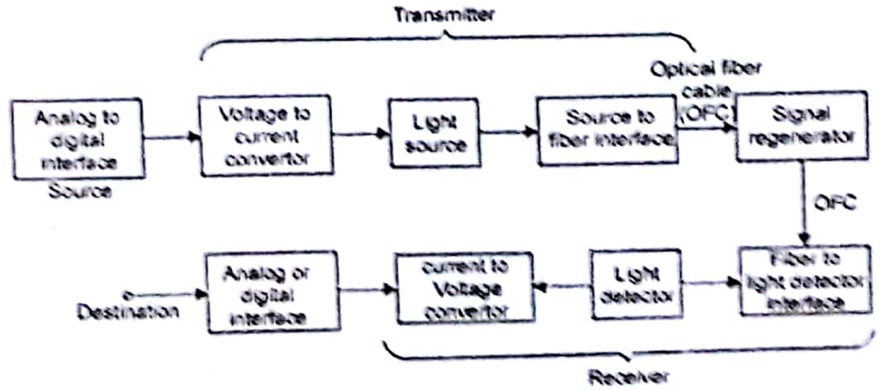
then
$$B_N = \int_0^{\infty}\frac{df}{1+\left(\frac{f}{B}\right)^2} = \frac{\pi}{2}B = \frac{1}{4RC}$$

Q.8.(a) Draw the basic block diagram of optical fiber system.

Ans. Main parts of optical system can be

(a) Transmitter (electrical to optical)

Basic Block Diagram of Optical System



(b) Optical Fiber Cable

(c) Receiver (Optical to electrical)

There is an optical interface between the transmitter to the Fiber cable & also between the optical fiber cable to the receiver.

In the transmitter the light of source can be modulated by either analog signal or digital signal.

at the transmitter LED or LASER can be used of the light source & PIN diode is generally used as the light detector. Avalanche photodiode or phototransistor can be used to detect the light.

Q.8.(b) Explain the Satellite Uplink model.

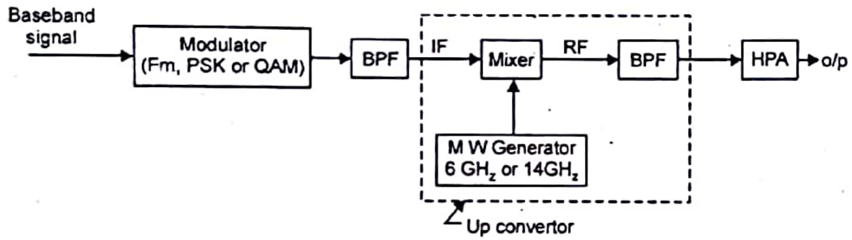
Ans. It is basically an earth station transmitter it typically consists of:

(a) IF modulator

(b) IF to RF microwave up convertor

(c) A high power amplifies (HPA)

(d) Band limiting filter of O/P spectrum



Q.8.(c) Explain the GSM radio subsystem.

Ans. GSM was a name given to Second Generation Mobile Standard (2G) working at 900 MHz & aimed at providing 200 full duplex channels. Later frequencies were allocated at 1800 MHz & was called as DES-1800.

GSM uses two 25 MHz frequency band:

1. Uplink band (from mobile-station to BTS) this is 890 to 915 MHz.

2. Downlink band (from BTS to MS): In this band frequencies from 935 to 960 MHz are used.

GSM was frequency division duplexing & a combination of TDMA & FDMA techniques to provide base station simultaneous access to multiple mobile users. It uses a channel of 200 KHz wide called absolute radio frequency channel number (ARFCN).

Basic parameters of GSM are as follows:

1. GMSK modulation is used.
2. 50 MHz Bandwidth is used (890 to 915 & 935 to 960 MHz).
3. FDMA & TDMA Accessing.
4. 992 Full duplex channel.
5. 200 KHz & Traffic channel.
6. 8, 25 KHz channel within 200 KHz band.
7. Uses supplementary ISDN services.

FIRST TERM EXAMINATION [SEPT. 2015]
FIFTH SEMESTER [B.TECH]
COMMUNICATION SYSTEM [ETEE–309]

Time. 1.30 hrs. **M.M. : 30**

Note: *Q No.1 is compulsory and any two more questions from remaining.*

Q.1. (a) Distinguish between the following (Attempt any four): **(1)**

(i) Narrowband FM and Wideband FM

(ii) Coherent and Non-Coherent Detection

(iii) Wide-sense stationary and Strictly stationary Random Process

(iv) Slope overload noise and Granular noise

(v) PCM and DPCM

Q.1. (i) Distinguish between Narrowband FM and Wideband FM

Ans.

S.No.	Characteristics	Wideband FM	Narrowband FM
1.	Modulation Index	Greater than 1	Less than or slightly greater than one
2.	Maximum deviation	75 KHz	5 KHz
3.	Range of modulation frequency	30Hz to 15 KHz	30 Hz to 3KHz
4.	Maxsimum modulation index	5 to 2500	Slightly greater than 1
5.	Bandwidth	Large, around 15 times greater than narrowband FM	Small, approx, same as that of AM.
6.	Application	Entertainment broadcasting	FM mobile communication
7.	Pre-emphasis and De-emphasis	Needed	Needed

Q.1. (ii) Coherent and non-Coherent Detection

Ans.

Coherent	Non-Coherent
1. Phase synchronized carrier to be generated at the receiver synchronized with that at the transmitter.	1. No phase synchronized local carrier is needed at the reciever.
2. These techniques are complex	2. These techniques are less complex.
3. Yield better performance.	3. Performance is inferior to that of coherent techniques.

Q.1. (iii) Wide Sense stationary and strictly stationary Random Process.

Ans.

Strictly Stationary	Wide-sense stationary
Joint CDF of original set of random variables is equal to that of the new set of the random variable, obtained after shift τ, *i.e.* $F_x(t_1), X(t_2)...X(t_n)$ $F_x(t_1+\tau), x(t_2+\tau)...x(t_n+\tau)$	Mean value $m_x(t)$ and an autocorrelation function which are independent of the shift of time origin. *i.e.* $m_x(t)$ = constant and, $R_x(t_1,t_2) = R_x(t_2-t_1)$

• All the stationary processes are wide sensed stationary but every wide-sense stationary process may not be strictly stationary.

(iv) Slope overload Noise and Granular Noise

Ans.

Slope Overload Noise	Granular Noise
1. An error condition in delta modulation that occurs when the analog signal to be digitized varies too quickly for the the system to follow	1. This noise in delta modulation occurs when the step-size is too large compared to small variations in the input signal.

(v) PCM and DPCM

Ans.

S. No.	Parameter of Comparison	PCM	DPCM
1.	Number of bits	It can use 4, 8 or 16 bits per sample.	Bits can be more than one but are less than PCM.
2.	Levels and step size.	The no. of levels depend on number of bits level size is fixed.	Fixed number of levels are used.
3.	Quantisation error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and quantisation noise is present.
4.	Transmission BM	Highest bandwidth is required	Bandwidth required is lower than PCM.
5.	Feedback	No feedback	Feedback exist
6.	Complexity of implementation.	System complex	Simple.

Q.1. (b) Write short note on any three: **(2)**

(i) Explain Gaussian Distribution.

Ans. It is used for continuous random variables. It perhaps the most important PDF in the area of communication. The majority of noise processes observed in practice are Gaussian.

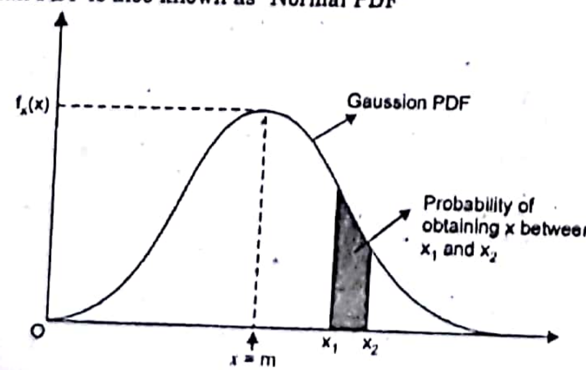
The PDF of a continuous random variables having Gaussian Distribution is

$$f_x(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-(x-m)^2/2\sigma^2} \quad -\infty < x < \infty$$

m = Mean of the random variable

σ^2 = Variance of random variable.

Gausssian PDF is also known as "Normal PDF"



(ii) Why does aperture effect takes place?

Ans. Aperture Effect takes place during the sampling of signals. Specifically during flat-top sampling, to convert varying amplitudes of pulses to flat top pulses we use a sinc function. Because of this, there would be decrease in the amplitude. This distortion is named as "**Aperture effect**" and eliminated by using equilizer in cascade with LPF.

(iii) Properties of Conditional PDF

Ans. Property-1: The conditional PDF is a non-negative function.

i.e. $f_y(y/x) \geq 0$

This is because the conditional PDF is basically a probability density function.

• **Property-2:** The area under a conditisnal PDF is always equal to 1.

i.e. $$\int_{-\infty}^{\infty} f_y(y/x)dy = 1$$

or $$\int_{-\infty}^{\infty} f_x(x/y)dx = 1$$

This is because the conditional PDF is basically a PDF, and therefore it will exhibit the basic property of a PDF.

• **Property-3:** If the random variable x and y are statstically independent then,

$$f_y(y/x) = f_y(y)$$
$$f_x(x/y) = f_x(x)$$

This means that the conditional density functions reduce to the marginal density function.

(iv) Explain what is meant by cross-talk?

Ans. It is the phenomenon by which a signal transmitted on one circuit or channel of a transmission system creates an undesired effect in another circuit. or channel.

It is usually caused by undesired capacitive, inductive or conductive coupling from one circuit channel or another.

Q.2. (a) Determine if the following functions is density function or not:

$f_x(x) = \frac{2+x}{10}$ for $2 \leq x \leq 4$, zero otherwise **(3)**

Ans. $$f_x(x) = \begin{cases} \frac{2+x}{10}, & 2 \leq x \leq 4 \\ 0, & \text{otherwise} \end{cases}$$

Calculating the area under PDF

$$\int_2^4 \frac{2+x}{10}dx \Rightarrow \frac{1}{10}\int_2^4 (2+x)dx$$

$$\Rightarrow \frac{1}{10}[2x]_2^4 + \frac{1}{10}\left[\frac{x^2}{2}\right]_2^4$$

$$\Rightarrow \frac{1}{10}(4) + \frac{1}{10}(6)$$

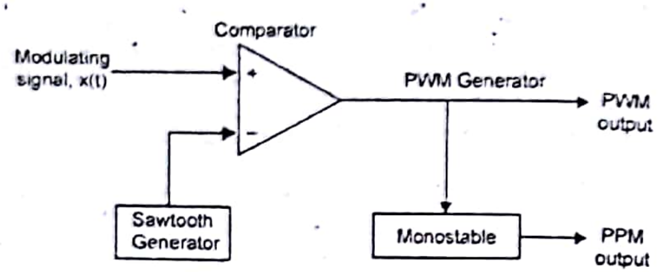
$$\Rightarrow \frac{10}{10} = 1$$

$\because$ area under PDF is 1

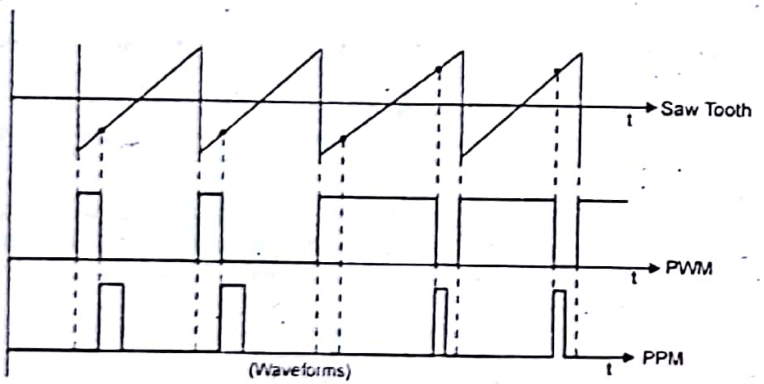
So, it is a density function.

Q.2. (b) Describe generation of PWM signal. (4)

Ans.



The Block diagram is shown above. A sawtooth generates a sawtooth signal of frequency fs, therefore sawtooth signal here is a sampling signal. It is applied to the inverting terminal of the comparator. The modulating signal $x(t)$ is applied to the non-inverting terminal of the same comparator. The comparator output will remain high as long as the instantaneous amplitude of $x(t)$ is higher than that of the ramp signal. This gives rise to a PWM signal.



(Waveforms)

Q.2. (c) An analog signal which is band limited to 3400 Hz is sampled at 8000 Hz. Find the signaling rate if 256 quantization levels are employed in a PCM system. (3)

Ans. Signaling Rate = No. of bits transmitted/second

= No. of samples/sec × No. of bits/sample

Given,

$$fm = 3400 \text{ Hz}$$

$$Q = 256$$

$$fs = 8000 \text{ samples/sec.}$$

and,

$$Q = 2^N$$

No. of bits/sample

$$N = \log_2 Q = \frac{\log_{10} 256}{\log_{10} 2} = \frac{2.408}{0.30}$$

$$= 8.027$$

∴ Signalling Rate = $N.fs$

$$= 8.027 \times 8 \times 10^3$$

$$= 64.219 \text{ Kbps.}$$

Q.3. (a) Compare various parameters of AM, DSB/SC, SSB/SC, VSB/SC. (4)

Ans.

S.No.	Parameter of comparison	AM	DSB/SC	SSB	VSB
1.	Carrier suppression	N.A.	Fully	Fully	N.A.
2.	Sideband suppression	N.A.	N.A.	One S.B completely	One S.B partially
3.	Bandwidth	$2fm$	$2fm$	fm	fm <BW<$2fm$
4.	Transmission efficiency	Minimum	Moderate	Maximum	Moderate
5.	No. of modulating inputs.	1	1	1	2
6.	Application	Radio Broadcasting	Radio Broadcasting	Point to point Mobile communication.	TV transmission

Q.3. (b) Describe DPCM basic Principle with diagrams. (3)

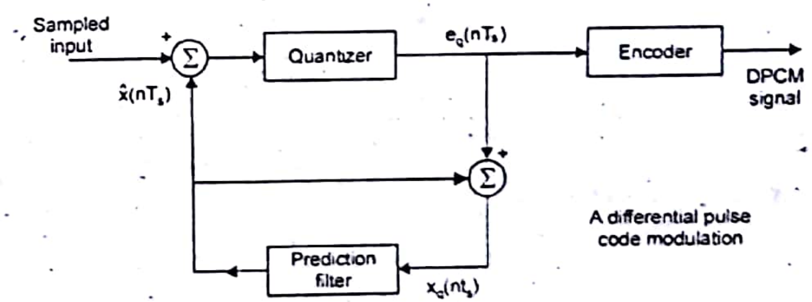
Ans. Principle: DPCM works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value.

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

where,

$$x(nT_s) = \text{Sample signal}$$

$$\hat{x}(nT_s) = \text{Predicted signal}$$



A differential pulse code modulation

Q.3. (c) Determine the modulation index and bandwidth of following FM signal: V(t) = 10 cos (8×10⁸ t + 7 sin 6×10⁴ t) (3)

Ans. We know that the FM wave is given by expression:

$$s(t) = A\cos[2\pi fct + mf\sin(2\pi fmt)]$$

(i) Modulation Index,

$$m_f = \frac{\Delta f}{fm} = 7$$

(On compairing with the given equation)

(ii)

$$\text{B.W.} = 2[\Delta f + fm]$$

∴

$$m_f = \frac{\Delta f}{fm}$$

∴

$$\Delta f = m_f . fm$$

$$= 7.6\times10^4$$
$$= 42 \times 10^4 \text{ Hz}$$
$$= 2\,[42 + 6] \times 10^4$$
$$= 2\,[43] \times 10^4$$
$$= 96 \times 10^4 \text{ Hz.}$$

Q.4. (a) Show that the random process $X(t) = A\cos(w_c t + \theta)$, where θ is a random variable uniformly distributed over the range $(0, 2\pi)$ is a wide-sense stationary process. (6)

Ans. To prove that given Random process $x(t)$ is a wide-sensed stationary process, it is necessary to show that

(i) Ensemble mean of sample function amplitudes at any t is same.

(ii) Autocorrelation $R_\tau(t_1, t_2) = R_\tau(t_2 - t_1)$

(i) Ensemble Average

Ensemble mean of this process is given by:

$$m_x(t) = \int_{-\infty}^{\infty} x\, fx(x,t)\,dx \quad \text{...(i)}$$

here

$$X = X(t) = A\cos(w_c t + \theta)$$

and

$$f_x(x,t) = f_\theta(\theta) = \frac{1}{2\pi}$$

Substituting in (i)

$$m_x(t) = \int_{-\infty}^{\infty} \frac{1}{2\pi} A\cos(w_c t + \theta)\,d\theta$$

$$= \frac{A}{2\pi}[\sin(w_c t + \theta)]_0^{2\pi}$$

$\therefore$ $\boxed{m_x t = 0}$, *i.e.* independent of t.

(ii) Autocorrelation function:

Expression for $R_\tau(t_1, t_2)$ is given by,

$$R_\tau(t_1, t_2) = E[X(t_1).X(t_2)]$$
$$R_\tau(t_1, t_2) = E[A\cos(w_c t_1 + \theta).A\cos(w_c t_2 + 2\theta)]$$
$$R_\tau(t_1, t_2) = E[A^2\cos(w_c t_1 + \theta).\cos(w_c t_2 + \theta)]$$

But, $\cos A \cos B = \frac{1}{2}[\cos(A+B) + \cos(A-B)]$

Thus,
$$R_\tau(t_1, t_2) = E\left[\frac{A^2}{2}\{\cos[w_c(t_1 - t_2)] + \cos[w_c(t_1 + t_2 + 2\theta]\}\right] \quad \text{...(ii)}$$

or

$$R_\tau(t_1, t_2) = \frac{A^2}{2}E\{\cos[w_c(t_1 - t_2)]\} + \frac{A^2}{2}E\{\cos[w_c(t_1 + t_2 + 2\theta]\} \quad \text{...(iii)}$$

The first term on RHS of equation (ii) does not contain random variable θ. Hence

$$E[\cos[w_c(t_1 - t_2)]] = \cos[w_c(t_1 - t_2)] \quad \text{...(iv)}$$

The second term on RHS of equation (ii) is a function of random variable θ. Therefore, it's expected value is given by

$$E[\cos[w_c(t_1 + t_2 + 2\theta)]] = \int_{-\infty}^{\infty} \cos[w_c(t_1 + t_2 + 2\theta].\frac{1}{2\pi}d\theta \quad \text{...(v)}$$

$$\left[\because E[x] = \int_{-\infty}^{\infty} xfx(x)dx\right]$$

or

$$E[\cos[w_c(t_1 + t_2 + 2\theta]] = \int_0^{2\pi} \frac{1}{2\pi}\cos[w_c(t_1 + t_2 + 2\theta)]d\theta$$

$$= \frac{1}{2\pi}\{\sin w_c(t_1 + t_2 + 2\theta)\}_0^{2\pi}$$

$$= 0 \quad \text{(vi)}$$

Putting equation (iii) and (vi) in equation (ii)

$$R_\tau(t_1, t_2) = \frac{A^2}{2}[\cos[w_c(t_1 - t_2)]]$$

This expression shows that the autocorrelation function of time difference $(t_1 - t_2)$. Thus we proved that,

(i) Ensemble mean is independent of time

(ii) Autocorrelation function is function of time difference $(t_1 - t_2)$

• Hence, $X(t)$ is a wide-sensed stationary process.

Q.4. (b) Describe Poisson distribution and its properties. (4)

Ans. This is the standard probability distribution used for the discrete random variables.

As the number 'n' increases, the binomial distribution becomes difficult to handle. If 'n' is very large, probability 'p' is very small and the mean value np is finite, then the binomial distribution can be approximated by the Poisson's distribution. Poisson distribution is thus the limiting case of binomial distribution.

The probability of the random variable having Poisson distribution is given by.

$$P(X = k) = \frac{m^k.e^{-m}}{k!}$$

where, m = mean value

• The mean value of Poisson distribution is given by

$$\boxed{m_X = np}$$

• The variance of Possion distribution is given by

$$\boxed{\sigma_X^2 - np}$$

• The standrard deviation of Poisson distribution is

$$\boxed{\sigma_X \cdot \sqrt{np}}.$$

SECOND TERM EXAMINATION [NOV. 2015]
FIFTH SEMESTER [B.TECH]
COMMUNICATION SYSTEM [ETEE-309]

Time. 1.30 hrs. **M.M.: 30**

Note: Q No.1 is compulsory and any two more questions from remaining.

Q.1. Attempt any five: **(2)**

Q.1. (a) Compare ASK and PSK.

Ans.

ASK	PSK
1. Here amplitude of carrier changes in discrete manner	Here Phase of the carrier changes in the sudden manner
2. ASK is effected by the external noise	It is comparatively more noise resistant
3. It is more simple to generate	This type of circuit is rather more complex

Q.1. (b) Explain channel capacity. **(2)**

Ans. Channel capacity is the amount of the data traffic it can carry in more technical terms it is calculated on the basis of the following relation.

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

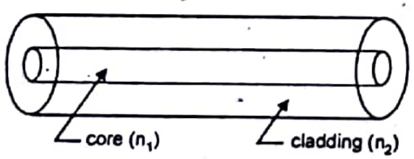
C = Capacity of channels in bits per second (bps)
B = Bandwidth of the given channel in Hz
S = Signal Power (in watts)
N = Noise Power (in watts)

Q.1. (c) Describe acceptance angle in case of optical fibre. **(2)**

Ans. Acceptance angle is that conical part for the entry of light rays for which total internal reflection take place, this angle is given by:

$$\theta_a = \sin^{-1}\left[(n_1^2 - n_2^2)^{\frac{1}{2}}\right]$$

Where n_1 = refractive index of core
n_2 = refractive index of cladding.



Q.1. (d) Briefly explain Handoff. **(2)**

Ans. Handoff is a process of transferring the ongoing call from the site of serving base station to the site of the base station whose radio signal strength is better at that time.

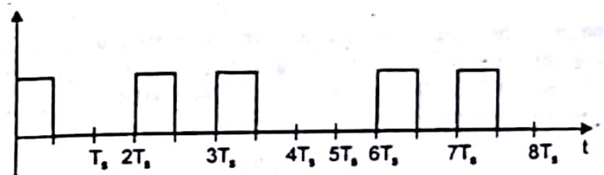
Q.1. (e) What are geostationary satellites.

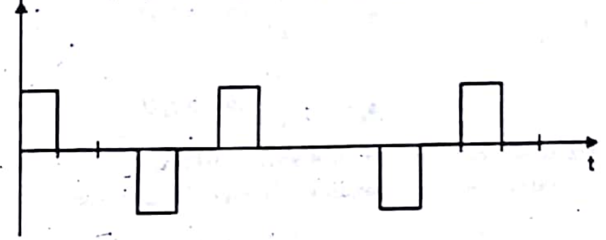
Ans. Geo stationary satellite are so called because their orbital speed is so adjusted by gravitational force as to appear stationary to a geographic location. Hence they are used to superwise the geographic location.

Q.1. (f) Draw the waveforms for the data 10110011 using (i) unipolar RZ and (ii) AMI RZ formats.

Ans. Given data is '10110011'

(i) Unipolar RZ





(ii) AM I R Z

Q.2. (a) Explain FSK transmission using block diagram. Draw FSK signal waveform for binary data 10011. **(4)**

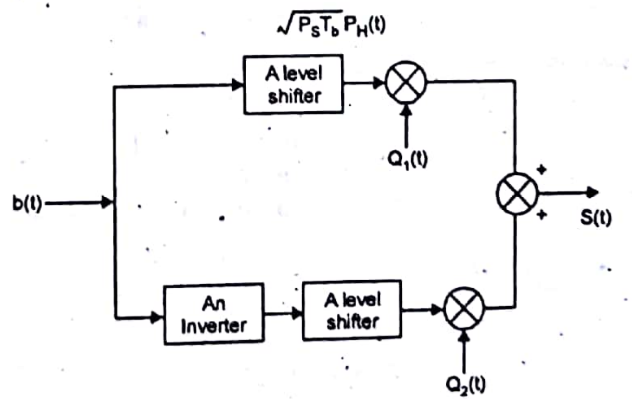
Ans. FSK or frequency shift keying. Here frequency of carrier wave changes in accordance with the binary key pattern when the bit is 1 e.g. $b(t)$ = '1' the frequency of carrier is high and when $b(t)$ is '0' then low carrier frequency is choosen. Mathematically.

$$S_H(t) = \sqrt{2P_s}\cos(2\pi f_c + \Omega)t$$

$$S_L(t) = \sqrt{2P_s}\cos(2\pi f_c - \Omega)t$$

if we use the non-return to zero bipolar scheme, then

$$S(t) = \sqrt{2P_s}\cos[(2\pi f_c + d(t)\Omega)t]$$



where high frequency is

$$f_H = f_c + \frac{\Omega}{2\pi}$$

$$f_L = f_c - \frac{\Omega}{2\pi}$$

in block schematic the input binary signal is first converted to bipolar NRZ schemes Modulator multiplies the shifted version of signal to generate the output FSK signal This can be explained with the help of the following schematic

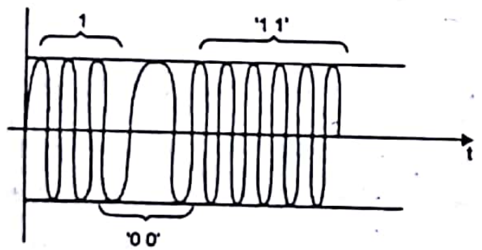
$b(t)$: input binary sequence

$$\phi_1(t) = \sqrt{\frac{2}{T_b}}\cos(2\pi f_H t)$$

$$\phi_2(t) = \sqrt{\frac{2}{T_b}}\cos(2\pi f_L t)$$

$S(t)$ is the output binary frequency shift keying (BFSK)

Given data is '10011' and corresponding FSK signal is given as



Q.2. (b) A transmitter has an alphabet of four letters x_1, x_2, x_3, x_4 and receiver has three letters y_1, y_2, y_3. The joint entropy matrix is:

$$P[X,Y] = \begin{bmatrix} 0.3 & 0.05 & 0 \\ 0 & 0.25 & 0 \\ 0 & 0.15 & 0.05 \\ 0 & 0.05 & 0.15 \end{bmatrix}$$

Find all the entropies.

Ans. Given, the joint entropy is

$$P[X,Y] = \begin{bmatrix} 0.3 & 0.05 & 0 \\ 0 & 0.25 & 0 \\ 0 & 0.15 & 0.05 \\ 0 & 0.05 & 0.15 \end{bmatrix}$$

from the given information

$$P(X) = [0.35 \ \ 0.25 \ \ 0.2 \ \ 0.2]$$

$$P(Y) = [0.3 \ \ 0.5 \ \ 0.2]$$

$$P(X/Y) = \begin{bmatrix} 1 & 0.1 & 0 \\ 0 & 0.5 & 0 \\ 0 & 0.3 & 0.25 \\ 0 & 0.1 & 0.75 \end{bmatrix}$$

$$P(Y/X) = \begin{bmatrix} 0.86 & 0.14 & 0 \\ 0 & 1 & 0 \\ 0 & 0.75 & 0.25 \\ 0 & 0.25 & 0.75 \end{bmatrix}.$$

(i) $H(X)$ is given by:

$$H(X) = -\sum_{i=1}^{4} p(x_i)\log(p(xi))$$

(ii) $H(Y)$ is given by

$$H(Y) = -\sum_{i=1}^{3} p(y_i)\log[p(y_i)]$$

(iii)
$$H(X,Y) = -\sum_{i=1}^{4}\sum_{j=1}^{3} p(x_i\ y_i)\log[p(x_i, y_i)]$$

(iv)
$$H(X/Y) = -\sum_{i=1}^{4}\sum_{j=1}^{3} p(x_i, y_i)\log\left[p\left(\frac{x_i}{y_i}\right)\right]$$

$$H(Y/X) = \sum_{i=1}^{4}\sum_{j=1}^{3} P(x_i, y_i)\log[p(y_i/x_i)]$$

Hence from the given values of the probabilities different and entropies can be calculated.

$H(X) = -[0.35\log(0.35) + 0.25\log(0.25) + 0.2\log(0.2) + 0.2\log(0.2)]$

$= +[0.35(0.456) + 0.25(0.602) + 0.4(0.699)]$

$= 0.1596 + 0.15.5 + 0.2796 = 0.59$

$$H(Y) = -[0.3\log(0.3) + 0.5\log(0.5) + 0.2\log(0.2)]$$
$$= +[0.3(0.523) + 0.5(0.301) + 0.2(0.699)]$$
$$H(Y) = 0.4472$$

Similarly other entropies can be calculated.

Q.3 (a) Apply Shannon-Fano coding procedure for the following message ensemble:

$[X] = [\ x_1 \quad x_2 \quad x_3 \quad x_3 \quad x_5 \quad x_6 \quad x_7\ ]$

$[P] = [\ 0.4 \quad 0.2 \quad 0.12 \quad 0.08 \quad 0.08 \quad 0.08 \quad 0.04\ ]$

Take $M = 2$ **(6)**

Ans.

X	$P(x)$	First Step	Second Step ↓	Third Step ↓	Fourth Step ↓
x_1	0.4	0	0		
x_2	0.2	0	1		
x_3	0.12	1	0	0	
x_4	0.08	1	0	1	
x_5	0.08	1	1	0	
x_6	0.08	1	1	1	0
x_7	0.04	1	1	1	1

Now write the code and code length of the respective symbols

X	Code	Code length
x_1	00	2
x_2	10	2
x_3	001	3
x_4	101	3
x_5	011	3
x_6	0111	4
x_7	1111	4

$$H(x) = -\sum_{i=1}^{7} P(x_i)\log[P(x_i)]$$

$$= [0.4 \log(0.4) + 0.2 \log(0.2) + 0.12 \log(0.12) + 3\times0.08 \log(0.08) + 0.04 \log(0.04)]$$

$$= 0.1592 + 0.1398 + 0.1104 + 0.2633 + 0.056$$

$$= 0.7286 + 0.31 = 2.42$$

$$L = \sum_{i=1}^{7} p(x_i)n_i$$

$$= 0.4\times2 + 0.2\times2 + 0.12\times3 + 0.08\times2\times3 + 0.08\times4 + 0.04\times4$$

$$= 0.8 + 0.4 + 0.36 + 0.48 + 0.32 + 0.16$$

$$= 2.52$$

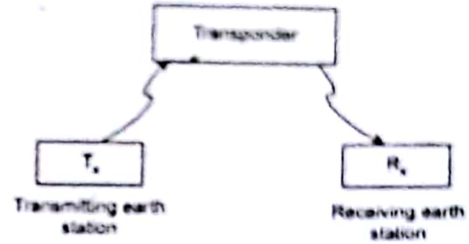
Coding efficiency is given by:

$$\eta = \frac{H(x)}{L}$$

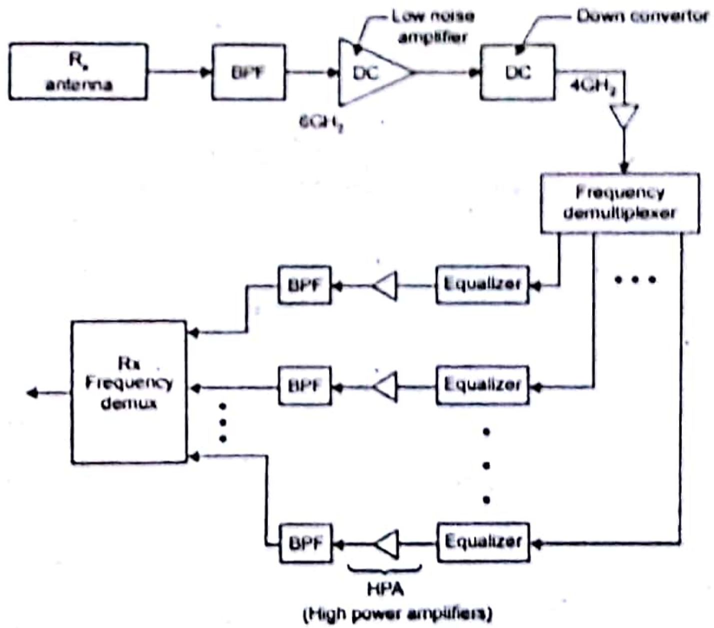
$$= \frac{2.42}{2.52} = 96\%$$

Q.3. (b) Draw block diagram of satellite Transponder and explain function of each block. (4)

Ans. Satellite transponder acts as a relay between the earth stations (transmitting as well as receiving)



Class-C band satellite transponder is given as follow:



The uplink frequency is taken to be 6 GHz. For this a wideband pass filter is used (BPF). The signal is amplified by a low noise Amplifier and down converted by the down convertor from 6 GHz to 4 GHz which is the down link frequency.

At this stage error correction and amplification are also don.

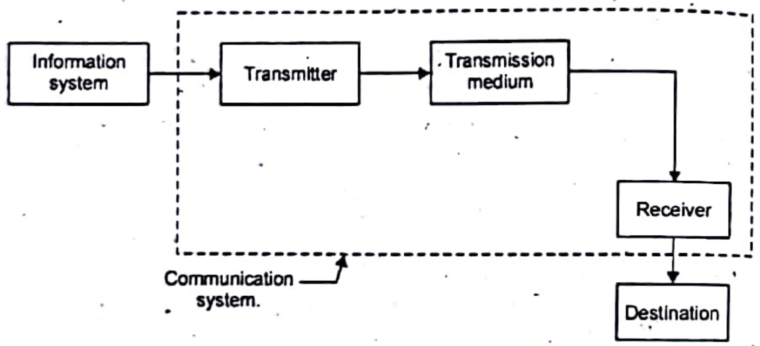
Frequency demultiplexing divides the signal into different frequency bands, the phase error of which is rectified by equalizer. HPA amplifies the signal to adequate limit so as to maintain the Linearity. BPF after the HPA removes the out of band noise, transmission antenna sends combined signal. In 12 channel C band transponder, the

Scanned by CamScanner

input is wideband 500 MHz each of 12 channels is of 36 MHz wide with 4 MHz gaurd band in between.

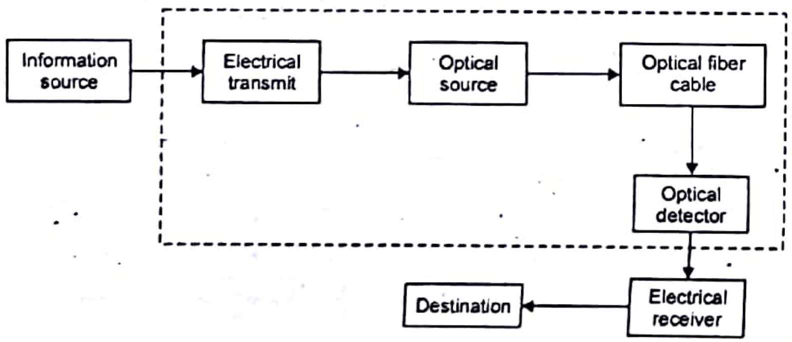
Q.4. (a) Explain elements of an optical fibre transmission link. Give the frequency band for optical communication. **(4)**

Ans. Optical Communication take place around in the range of 10^{14} Hz or 10^{5} GHz yields a for greater potential for the transmission band for the signal. A simple optical system is given as follow:



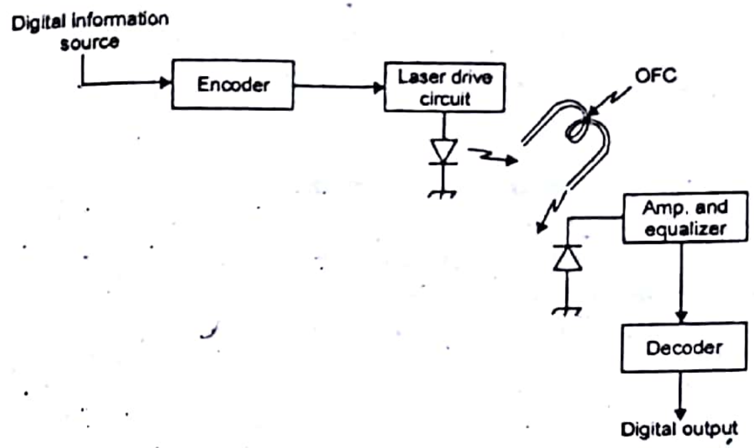
A General view of Communication System

Communication system can be changed to optical system by introducing the following blocks:



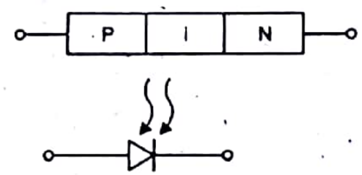
Optical source converts the electrical energy into light energy, then using optical fiber, the information is transmitted and on the receiver side, the optical information is again converted back to electrical energy. Photo diode, PIN diode etc are used as optical detectors.

While LED, LASER are used as the optical source. In case of the digital optical fiber link, encoder is followed by the transducer and then LASER or LED driver circuit is used.



Q.4. (b) Write short note on PIN photo diode. **(4)**

Ans. PIN phto diode is used in optical communication for detection the light intensity variation which produces electrical current variation at the output.

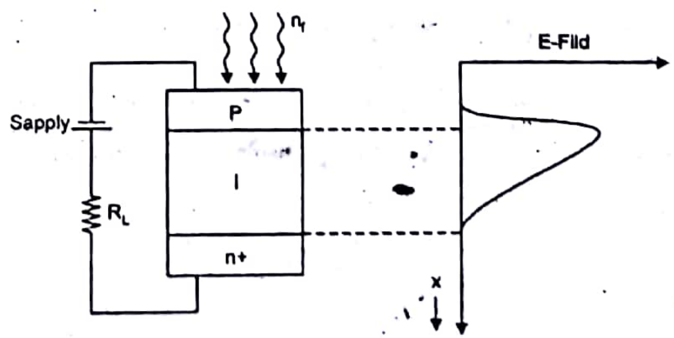


$p \rightarrow p$-type semiconductor

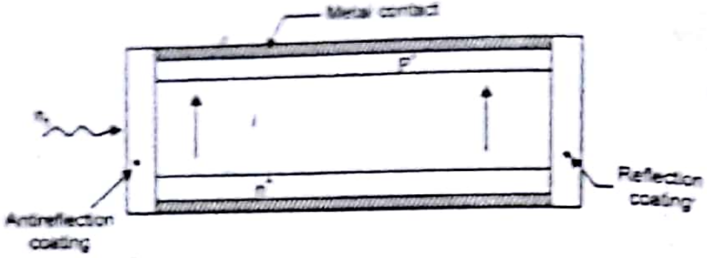
$N \rightarrow N$-type semiconductor

$i \rightarrow$ Intrinsic layer

p-i-n photo diode shows maximum sensitivty around the junction and falls steadily as the distance increases away from the junction.



Intrinsic layer is having more area just to work at longer wavelengths where light penetrates more deeply into the semiconductor material. In the intrinsic layer a lightly doped n-type material is used.

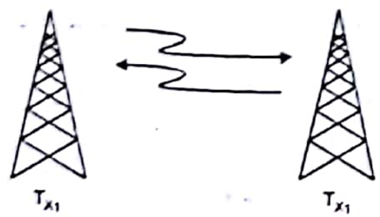


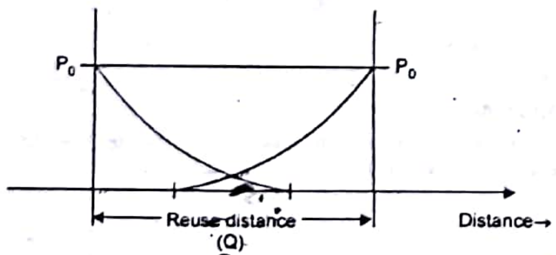
Structure of P-i-N Photo diode

Q.4. (c) Explain frequency reuse concept in brief.

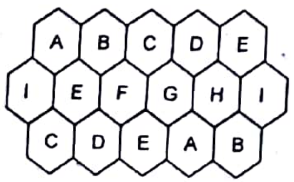
Ans. Frequency reuse is a concept used in the mobile communication to increase the system utilization factor factor. Frequency reuse depends on the following factors

1. Channel interference
2. Population density to be served
3. Grade of the service to be maintained.





After the resuse distance the signal of T_{x1} does not interfere the signal transmitted from the Tx_2 so the same channel frequencies can be used after a distance of Q. Which is also linked with the cluster size. Here the cells using the same frequency are indicated with the same notation.



Repeatation of E, D, A, B indicate the reuse of the same channel frequencies.

END TERM EXAMINATION [DEC. 2015]
FIFTH SEMESTER [B.TECH]
COMMUNICATION SYSTEM [ETEE-309]

Time. 3 Hours **M.M. : 75**

Note: Attempt any five question including Q. no. 1 which is compulsory. Select one question from each unit.

Q.1. (a) Discuss the types, causes and effects of the various forms of noise which may be created within an amplifier.

Ans. There may be the following type of noises which may occur inside the amplifier.

1. Non Linear distortion: As the transistor is a semiconductor device and is having non-linear charactersics, because of which, distortion may occur in the given signal.

2. Partition Effect: As in most of the amplifier or any semiconductor amplifier, transistor is used in one or other form. In this emitter current is divided in base current and collector current which causes distortion.

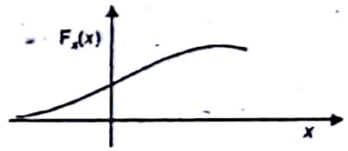
3. Thermal noise: As the temperature increases in the amplifier, the signal is mixed with the thermal electrical equivalent. Hence the main signal becomes noisy.

Q.1. (b) State important properites of CDF and explain them. **(5)**

Ans. CDF is cummulative Distribution function & is defined as $F_x(x \le a) = P(x \le a) = F_x(x)$ Hence it gives the total probabilities in cummulative sense upto a certain value (a)

Properties:

(a) $F_X(x) = \int_{-\infty}^{x} f_X(x)dx$



(b) $F_X(x) \ge 0$

So it is non negative in nature

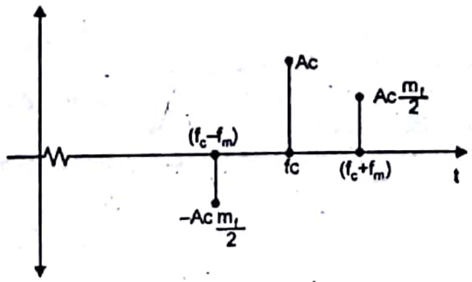
(c) CDF is monotonically increasing function

Q.1. (c) Derive an expression of a Frequency Modulated Wave with spectrum. **(5)**

Ans. Modulation is very important while transmitting the information. Important reasons for modulation are as follow:

(a) For decreasing the antenna height
(b) For multiplexing.
(c For avoiding the attenuation of low frequency signal.
(d) For increasing the speed of transmission
(e) For providing the security to the information.

Spectrum of narrow band *fm* is similar to the *DSB-SC* signal and is as follow



Q.1. (d) What do you mean by line coding. Explain all of its desirable properties. (5)

Ans. Line codes refers to those particular and predefined patterns which are used to encode the incoming bit symbols.

The digital data can be transmitted by various transmission or line codes such as on-off, polar, bipolar and so on. This is called line coding. The different attributes that decide the selection of a particular line coding format are as follows:

(i) **Transmission Bandwidth:** For a line-code, the transmission bandwidth must be as small as possible.

(ii) **Power Efficiency:** For a given bandwidth and a specified detection error probability, the transmitted power for a line code should be as small or possible.

(iii) **Error Detection and Correction Capability:** It must be possible to detect and preferably correct detection errors. For example, in a bipolar case, a signal error will cause bipolar violation and thus can easily be detected.

(iv) **Favourable Power Spectral Density:** It is desired to have zero power spectral density (PSD) at $w = 0$ (*i, e., dc*) since ac coupling and transformers are used in the repeaters. Significant power in low-frequency components cause *dc* wander in the pulse stream when as coupling is used. The a.c coupling is required since the dc paths provided by the cable pairs between the repeater sites are used to transmit the power required to operate repeaters.

(v) **Adequate timing Content:** It must be possible to extract timing or clock information form the signal.

(vi) **Transparency:** It must be possible to transect a digital signal correctly regardes the pattern of 1's and 0's.

Q.1. (e) Explain the significance of Numerical Aperture of fibers. Show that it solely depends on the properties of core and cladding. (5)

Ans. Numrical Aperture basically indicates the region through which light can be incident at the fiber code.

Let NA = Numerical Aperture

then
$$NA = n_0 \sin\theta_a$$
$$= (n_1^2 - n_2^2)^{1/2}$$

So when optical fiber is used in Air where n_o is unity, then it is simply equal to $\sin\theta_a$ where θ_a is acceptance angle Numerical aperture depends on the value of n_1 and n_2 which are refractive indices and depend on the type and structure of material used.

UNIT-I

Q.2. (a) What is the fundamental limitations in Communication system? Why is modulation of signal required for transmission? (6.5)

Ans. Fundamental limitation offered in communication systems are as follow:

1. Capacity of Communication: Any system designed, can not offer arbitrarily large capacity.

2. Noise: Effect of noise keeps on increasing as we increases the distance of seperation between between transmitter and receiver.

For doing communication modulation is one of the most integral part.

Main objectives of doing modulation are as follow:

1. For translating the signal frequency: For translating the signal frequency from the baseband to passband Modulation is required.

2. For ease of radiation: Since the signals having higher frequency are easy to radiate so modulation is done to increase the frequency of the signal.

3. For reducing the height of antenna recquired. Since the height required changes with the wavelength (λ)
$$\lambda = c/f$$

So at higher frequencies, value of λ will decrease which make the antenna to be feasible.

4. For reducing the effect of noise: At higher range of frequencies, effect of noise also reduces So modulation is done to increase the frequency.

Q.2. (b) A three digit message is transmitted over a noisy channel having a probability of error $P(E) = 2/5$ per digit. Find out the corresponding CDF. (6)

Ans. Probability of error $P(E) = \frac{2}{5}$

and Probability of no error
$$= 1 - P(E)$$
$$= 1 - \frac{2}{5}$$
$$= \frac{3}{5}$$

the probability that there is no error in the transmission e.g. all the three bits are received correctly:
$$P_0 = P(c).P(c).P(c)$$

Here $P(c)$ is the probability for correct reception.
$$= \frac{3}{5}\times\frac{3}{5}\times\frac{3}{5} = \frac{27}{125}$$

Probability for one bit in error:
$$P_1 = 3\left[\frac{3}{5}\times\frac{3}{5}\times\frac{2}{5}\right] = \frac{54}{125}$$

For Probability of two bits to be in error
$$P_2 = 3\left(\frac{3}{5}\times\frac{2}{5}\times\frac{2}{5}\right) = \frac{36}{125}$$

Probability of three bits in error
$$\Rightarrow \quad P_3 = \left(\frac{2}{5}\times\frac{2}{5}\times\frac{2}{5}\right) = \frac{8}{125}$$

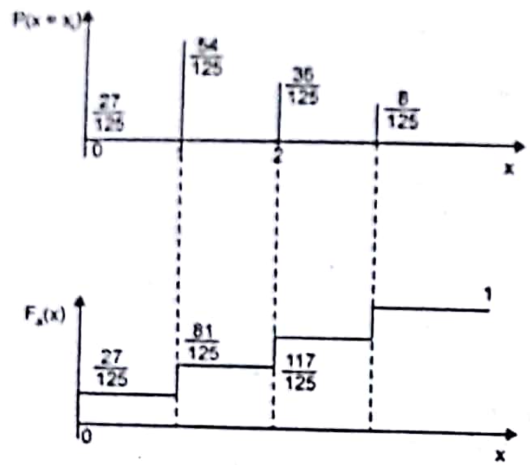
Now CDf can be calculated as follow:

1.
$$F_X(x_0) = P(X \le x_0)$$
$$= \frac{27}{125}$$

2.
$$F_X(x_1) = P(X \le x_1)$$
$$= F_X(x_0) + \frac{54}{125} = \frac{81}{125}$$

3.
$$F_X(x_2) = P(X \le x_2)$$
$$= \frac{27}{125} + \frac{54}{125} + \frac{36}{125} = \frac{117}{125}$$

4.
$$F_X(x_3) = P(X \le x_3)$$
$$= \frac{117}{125} + \frac{8}{125} = 1$$



Or

Q.3. (a) What do you mean by Probability Density Function (PDF)? What is the relation between probability and PDF? (6.5)

Ans. Probability Density Function: This function may be defined as the derivative of the cumulative distribution function

or
$$P_X(x) \triangleq d\,F_X(x)/dx$$

Provided that derivative exists
$$F_X(-\infty) = 0$$

So we can also write it as
$$P(X \le x) = F_X(x) = \int_{-\infty}^{x} P_X(\lambda)\, d\lambda$$

also (1) $P_X(x) \ge 0$

that mean probability will always be non-negative in nature.

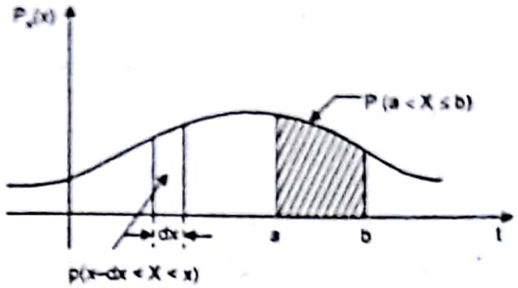
2.
$$\int_{-\infty}^{\infty} P_X(x)\, dx = 1$$

e.g. total area

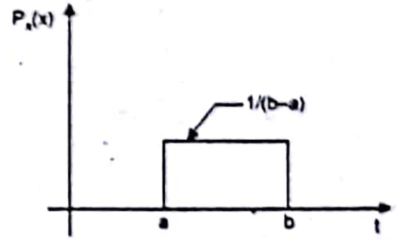
under the curve defined by $P_X(x)$ is unity

3.
$$P(a < x \ge f_x(b) = f_X(b) - f_X(a)$$
$$= \int_a^b P_X(x)\, dx$$

and the concept of *PDF* can also be defined by the following graph



Uniform PDF

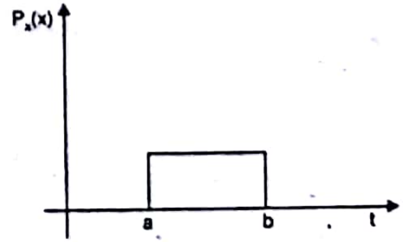


Q.3. (b) What are the needs of using standard probability models? Explain Binomial distribution in detail. (6)

Ans. Standard Probability models are:

1. Uniform *PDX*: It is defined as

$$P_x(x) = \begin{cases} 0 & x < a \\ 1/(b-a) & a \le x \le b \\ 0 & x > b \end{cases}$$



2. Binomial distribution function:
which is defined by

$$P_e(i) = \binom{n}{i} \alpha^i (1-\alpha)^{n-i}$$
$$i = 0, 1, \ldots, n$$

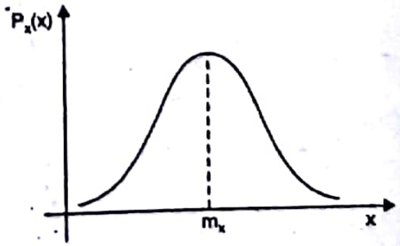
3. Poisson Distribution:
It function is defined as

$$P_e(i) = e^{-\mu T} \frac{(\mu T)^i}{i!}$$

where $m = \mu T$ and $\sigma^2 = m$

4. Gaussian Distribution function: It is defined by

$$P_X(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{\frac{-(x-m_x)^2}{2\sigma^2}}$$



Binomial distribution function is significant only where Bernaulli trial is to be made and is very useful in context of the digital transmission because here two states exist low (0) and high (1).

If probability of correct result is q.

Then probability of wrong result is $(1 - q)$.

Then out of n trials the probability of K events to be correct N.

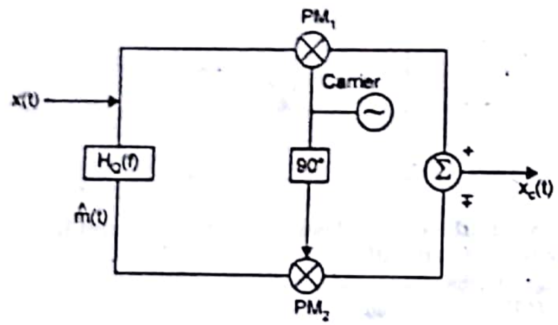
$$P_N(K) = {}^nC_K q^K (1-q)^{n-k}$$

$$= \frac{n!}{(n-k)!k!} q^k (1-q)^{n-k}$$

UNIT II

Q.4. (a) Explain the phase shift method of SSB generation. List its advantages and disadvantages with respect to the other methods of generation. (6.5)

Ans. Phase shift method of SSB generation This method uses hilbert transformer and the product modulators as shown in the fig. given below:



$m(t)$ is input information. Product modulator multiplier $m(t)$ and the carrier as such and product modulator (PM_2) multiplies $\hat{m}(t)$ and 90° phase shifted carrier.

Product modulators generates double side band with suppressed carrier.

Now these two waveforms when added generates single sideband either lower sideband or upper sideband depending on whether +ve sign or – ve sign is used.

Hence the output signal $X_C(t)$ is written as

$$X_C(t) = \frac{A_c}{2} x(t)\cos w_c t \mp \frac{A_c}{2} \hat{X}(t)\cos(w_c t - 90°)$$

Q.4. (b) Explain the Armstrong method for the generation of wideband FM. (6)

Ans. Armstrong Method of FM generation: This method is also known by indirect method of FM generation.

First it generates narrow band FM using phase shift method and then using mixing and mulplier stages, it generates the widebend FM.

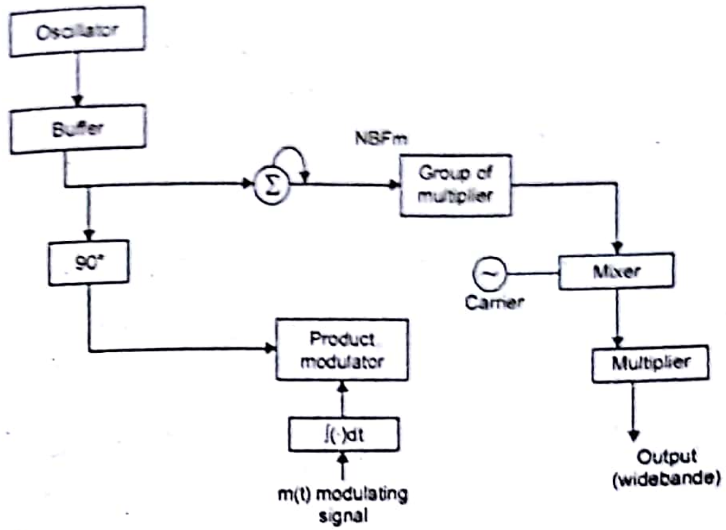
Features of Armstrong Method of FM generation.

1. Crystal oscillator generates the carrier at low frequency typically at 1 MHz, this is applied to a combining network and 90° phase shifter.

2. The modulating signal is applied to the low pass filter to boost the modulating frequencies

FM signal is given by

$$s_1(t) = v_{c_1} \cos[2\pi f_1 t + \phi_1(t)]$$



where the phase $\phi_1(t)$ is given by:

$$\phi_1(t) = 2\pi k_1 \int_0^t x(t)dt$$

Where K_1 represents the frequency sensitivity of the modulator.

For generating NBFM, modulation index (m_f) is kept very small $(m_f << 1)$ because of which $\phi_1(t)$ will be very small so

$$\cos(\phi_1(t)) = 1$$

and

$$\sin(\phi_1(t)) \cong \phi_1(t)$$

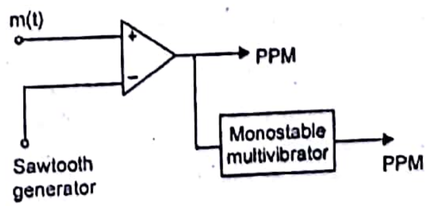
So Narrow band FM can be expressed as

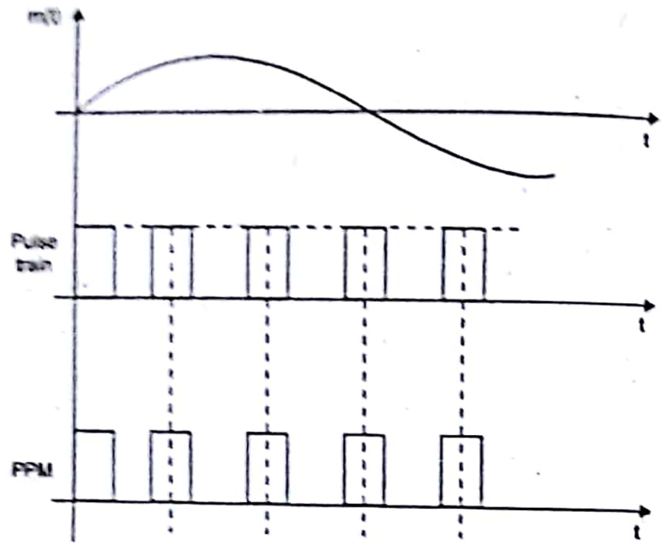
$$s(t) = A_C \cos w_c t - m_f A_c \sin w_c t \sin w_m t$$

Q.5. (a) With the help of neat circuit diagram explain the generation and detection of a PPM signal. (6)

Ans. Pulse position Modulation basically refers to a situation where and neither the pulse width non the pulse height changes but the position of the pulse changes in accordance with the amplitude variation of the input signal.

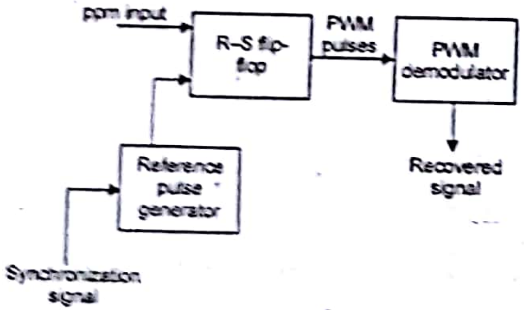
PPM generation: It can be generated with the help of comparator and monostable multivibrator. Here PPM is taking two stages, in which first it is converted to PWM then after that it is changed to PPM.





PPM signal is demodulated by means of R-S flip-flop which generates PWM pulses and then PWM demodulator is used to get the demodulated output.

Block diagram of PPM demodulation is given as follow:



Q.5. (b) What is the slope overload distortion and granular noise in delta modulation and how it is removed in ADM.

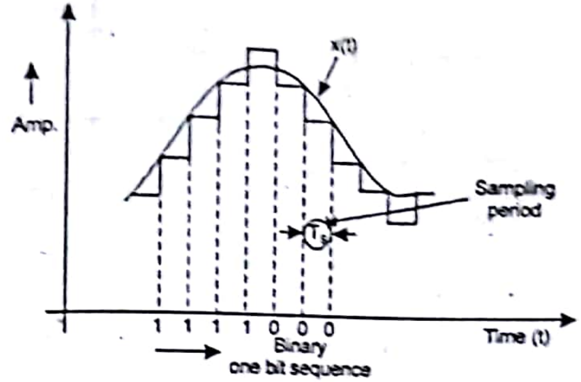
Ans. Delta Modulation: Transmits only ONE bit per sample here, present value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Step size is fixed in Delta Modulation: Input signal $x(t)$ is approximated to the step size by delta modulation difference between $x(t)$ and stair case is confined to two levels

i.e., $+\Delta$ or $-\Delta$

if Δ is positive, then it isencreased by one step and '1' is transmitted.

if Δ is negative then approximated signal is reduced by 1 and '0' is transmitted hence, for each value of sample only '1 bit' is transmitted.



$x(nT_s)$ = sampled signal of $x(t)$

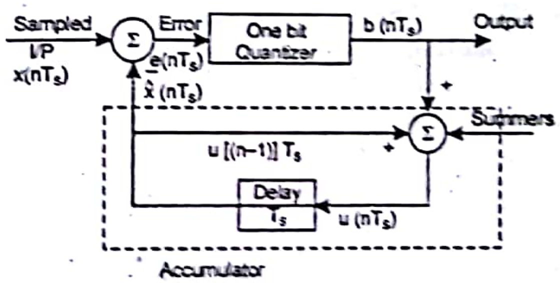
$x_T(nT_s)$ = last sample approximation of staircase waveform.

if we assume $u(nT_s)$ as present sample appl.

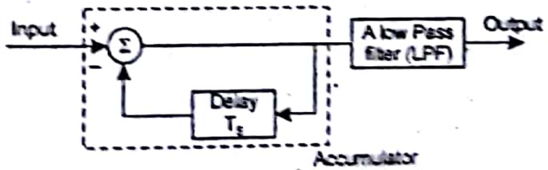
then $\quad u[n-1)T_s]_s = \hat{x}(nT_s)$

$$b(nT_s) = \begin{bmatrix} +\Delta \text{ if } x(nT_s) \geq \hat{x}(nT_s) \\ 1 \text{ is transmitted} \\ -\Delta \text{ if } x(nT_s) < \hat{x}(nT_s) \\ 0 \text{ is transmitted} \end{bmatrix}$$

Modulation:



Receiver:

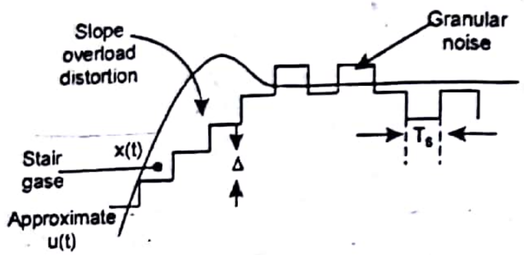


Drawbacks: 1. Slope overload distortion

2. Granular or idle Noise

To avoid slope overload problem,

Slope of pluse > Slope of i/p signal



$$\frac{\Delta}{T_S} \geq \left|\frac{d}{dt}(A_m \cos w_m t)\right|$$

$$\frac{\Delta}{T_S} \geq |A_m(-\sin w_m t).w_m|$$

$$\frac{\Delta}{T_S} \geq A_m w_m$$

or

$$\Delta \geq A_m w_m . T_s$$

$$\boxed{\Delta \geq A_m \frac{w_m}{f_s}} \text{ Required condition.}$$

UNIT III

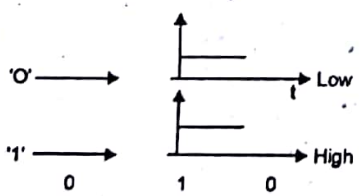
Q.6. (a) Define the following type of line codes or the electrical rperesentation of binary data. (6)

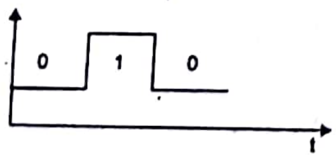
(i) Unipolar NRZ signaling

(ii) Bipolar RZ signaling

Ans. (i) Unipolar NRZ signalling:

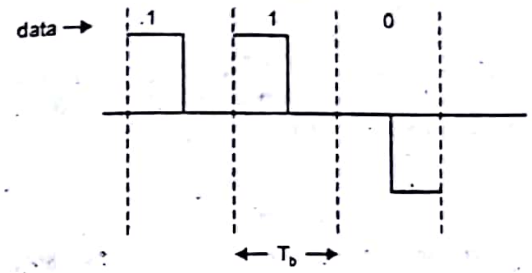
Here we define '0' and '1' as





(ii) Bipolar RZ Signalling: Bipolar Means it is having two polarities, for '0' –ve polarity is used and for '1' +ve polarity is used.

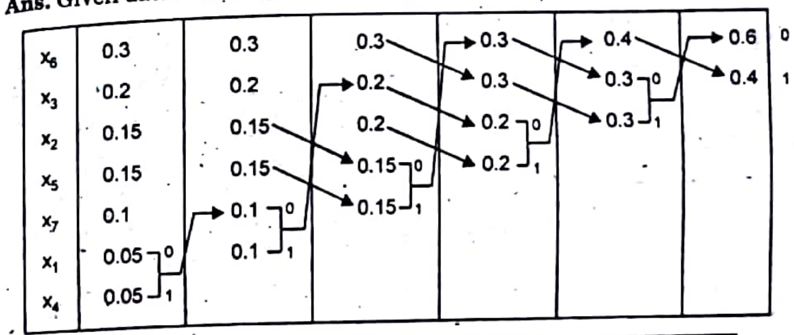
Also during the bit interval there is a transition from high to low for '1' and low to high for '0'.



Q.6. (b) Determine the Huffman code and the efficiency for the following messages with their probabilities given- (6.5)

X1	X2	X3	X4	X5	X6	X7
0.05	0.15	0.2	0.05	0.15	0.3	0.1

Ans. Given data is arranged as follows:



Symbols	Code	Code length
x_1	1 0 0 0	4
x_2	0 0 0	3
x_3	1 0	2
x_4	1 0 0 1	4
x_5	0 0 1	3
x_6	0 1	2
x_7	1 0 0	3

Coding efficiency of Huffman coding is given by

$$\eta = \frac{H(X)}{L}$$

$$H(X) = \sum_{i=1}^{7} p(x_i)\log\frac{1}{p(x_i)}$$

and

$$L = \sum_{i=1}^{7} \eta_i p(x_i)$$

$$H(x) = -[0.05 \log (0.05) + 0.15 \log (0.15) \times 2 + 0.1 \log (0.1) + 0.05 \log (0.05) + 0.3 \log (0.3) + 0.2 \log (0.2)]$$

$= 0.13 + 0.247 + 0.1 + 0.157 + 0.14$
$= (0.774)/0.30 = 2.58$

$L = 0.05\,(4) + 0.15\,(3) + 0.2(2) + 0.05\,(4) + 0.15\,(3) + 0.3(2) + 0.1(3)$
$= 0.2 + 0.45 + 0.4 + 0.2 + 0.45 + 0.6 + 0.3 = 2.6$

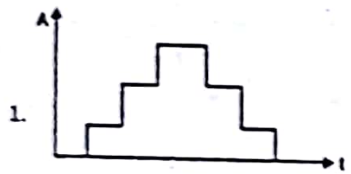
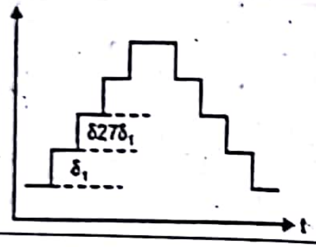
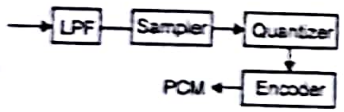
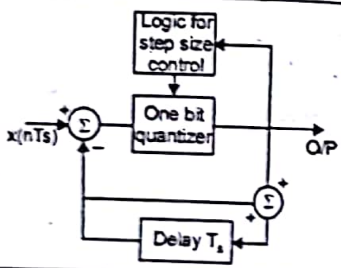
Hence the coding efficiency is given by

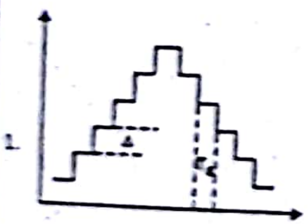
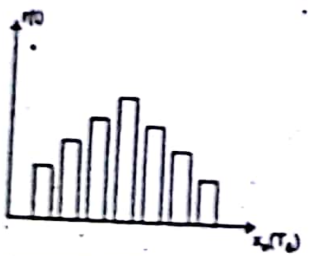
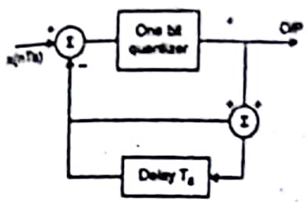
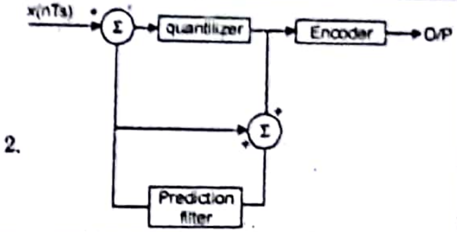
$$\eta = \frac{H(x)}{L} = \frac{2.58}{2.6} = 0.9923 = 99.23\%$$

OR

Q.7. (a) State performance comparison of three basic digital modulation Techniques. (6.5)

Ans. The basic and important digital modulation techniques are PCM, (Pulse code Modulation), Delta Modulation (DM) Adaptive DM and Adaptive PCM. We can compare these techniques as follow:

PCM	ADM
1. 	2. 
2. 	2. 
3. It can use 4, 8, 16, bits sample 4. Level size is fixed 5. No feed back exist 6. Complex	3. One bit/sample is used. 4. Step size varies 5. Feedback exit 6. Simple

DM	DPCM
1. 	1. 
2 	2. 
3. One bit/sample is used 4. Fixed step size 5. Feed back exists. 6. Simple	3. More than one but are less than PCM 4. Fixed No. of levels. 5. Feedback exists 6. Simple.

Q.7. (b) Calculate the capacity of AWGN channel with a bandwidth of 1 MHz and an S/N ratio of 40 dB. (6)

Ans. Given Bandwidth of the channel = 1 MHz

$$(S/N)_{dB} = 40\,dB = 10\log_{10}(P_0/P_i)$$

$$S/N \text{ in linear scale} = 10^4 = 10.000$$

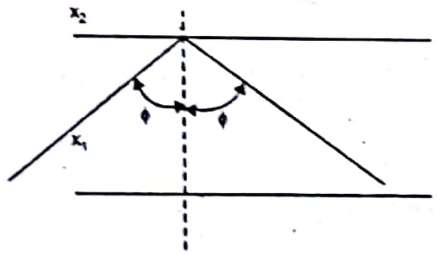
Now the capacity of AWGN channel is

$$C = B\log_2(1 + S/N)$$
$$= 1 \times 10^6 \log_2(1 + 10{,}000)$$
$$= 10^6 \log_2(10001) = 10^6 \times 13.33$$
$$C \approx 13 \text{ MbPS}$$

UNIT-IV

Q.8. Using simple theory describes the mechanism for transmission of light within an optical fiber. Briefly discuss with the help of suitable diagram, what is meant by acceptance angle for an optical fiber. Derive it showing how it is related to the fiber numerical aperture and the refractive indices for the fiber core and cladding. (12.5)

Ans. Let us take the refractive index of the care as x_1 and refractive index of cladding as x_2 then as per total internal reflection, when the input signal ray is incident at some acceptable angle then light ray will travel inside the core



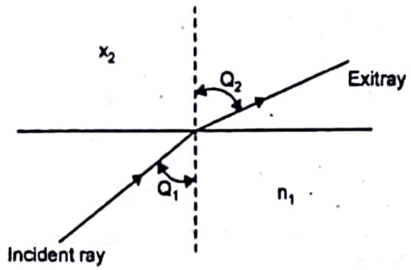
Then as per Snell's Law:

$$x_1 \sin\phi_1 = x_2 \sin\phi_2$$

or
$$\frac{\sin\phi_1}{\sin\phi_2} = \frac{x_2}{x_1} \quad ...(1)$$

where ϕ_1 and ϕ_2 are angles as shown in the fig below:

$x_2 < x_1$

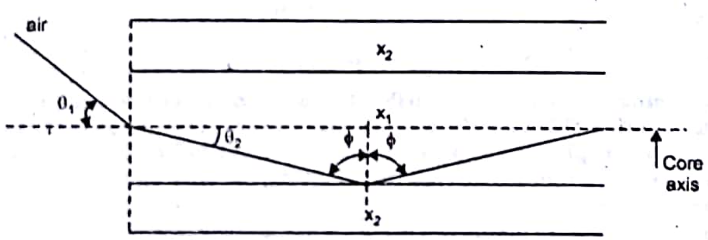


When $\phi_2 = 90$, then refracted ray will travel around the boundary of core, this will happen only at some specific value of ϕ_1 called as critical angle (ϕ_c).

Then equation (1) can be defined as

$$\sin\phi_c = \frac{x_2}{x_1}$$

or
$$\phi_c = \sin^{-1}\left(\frac{x_2}{x_1}\right)$$



from fig.
$$\phi = 90 - 0_2 \Rightarrow 0_2 = 90 - \phi$$
$$x_0 \sin 0_1 = x_1 \sin 0_2$$
$$= x_1 \cos\phi$$

or no
$$x_0 \sin 0_1 = x_1(1 - \sin^2\phi)^{\frac{1}{2}}$$

$\Rightarrow$
$$x_0 \sin 0_a = \left(x_1^2 - x_2^2\right)^{\frac{1}{2}}$$

where 0_a : Acceptance angle

$x_0 \sin 0_a$ basically indicate Numerical aperture (NA)

so
$$\boxed{NA = \left(x_1^2 - x_2^2\right)^{\frac{1}{2}}}$$

Hence NA depends on the value of x_1 and x_2 or on the type of core and cladding structure.

OR

Q.9. (a) The average optical power launched into a 20 km length of the fiber is 700 μW and the average output power is 5 μW. Calculate (i) signal attenuation per km (ii) overall signal attenuation for 12 km optical link using the same fiber with 5 splices each having attenuation of 0.9 dB. **(6)**

Ans. Length of the optical fiber = 20 km

Input incident power = 700 μ watt

output power = 5 μ watt

$$\text{Signal attenuation} = 10\log_{10}\left(\frac{P_i}{P_0}\right)$$
$$= 10\log_{10}\left(\frac{700\times10^{-6}}{5\times10^{-6}}\right)$$
$$= 10\log_{10}(140)$$
$$= 10\,(2.146) = 21.46 \text{ dB}$$

$$\text{Attenuation per KM} = \frac{21.46}{20} = 1.73 \text{ dB/km}$$

(i) Overall attenuation occured in a length of 12 km is simply $\alpha.L$
$$= 1.73 \times 12 = 20.76 \text{ dB}$$

Since there are 5 splices having 0.9 dB loss per splice

then total loss $= 0.9 \times 5 = 4.5$ dB

Hence the total loss occured due to splice as well $= (20.76 + 4.50)$ dB
$$= 25.26 \text{ dB.}$$

Q.9. (b) What do you meant by quantum efficiency and responsivity of a photodiode? Calculate the wavelength at which quantum efficiency and responsivity are equal. **(6.5)**

Ans. Quantum efficiency is defined as the fraction of incident photons which are absorbed by the photodetector and generate electrons which are collected at the detector terminals

$$\eta = \frac{\text{no. of electrons collected}}{\text{no. of incident photons}}$$

or
$$\eta = \frac{r_e}{r_p}$$

Here r_e indicate the rate at which electrons are generated and r_p indicate the rate at which photons are incident or incoming at the photo-detector.

Responsivity: It is generally of more practical use, because it involves the photon energy. Responsivity is defined as the ratio of photo current generated to the optical power incident on the photo detector.

$\Rightarrow$
$$R = \frac{I_P}{P_0}(A/W)$$

Scanned by CamScanner

Photon incoming rate can be defined as

$$r_p = \frac{P_0}{h_f}$$

and $$r_e = \eta r_p$$

so $$r_e = \frac{\eta P_0}{h_f}$$, then output photo

Current is $$I_p = \frac{\eta P_0 e}{h_f}$$

so Responsivity $$= R = \frac{\eta e}{h_f}$$

and in terms of wavelength (λ)

$$\boxed{R = \frac{\eta e \lambda}{h_c}}$$

threshold for detection, Commonly called as long-wavelength cut-off point (λ_C) is

$$\lambda_C = \frac{h_C}{E_g}$$

FIRST TERM EXAMINATION [SEPT. 2016]
FIFTH SEMESTER [B.TECH.]
COMMUNICATION SYSTEM (ETEE–309)

Time : 1.5 hrs. **M.M. : 30**

Note: *Attempt Q. No. 1 which is compulsory and any two more questions from remaining.*

Q.1. (i) What are the benefits of modulation. (2)

Ans. Benefits of Modulation are as follows:

(a) Frequency is increased thereby noise is reduced.

(b) Physical dimension of antenna is of lower value at higher frequency.

(c) Multiplexing can be achieved easily.

Q.1. (ii) Distinguish between Wide-sense stationary and Strictly stationary Random process. (1)

Ans. Wide sense stationary (WSS) process or normally stationary process show a property in which auto-correlation function for given message is independent of time shift. In strict sense stationary process (SSS) PDF of given process does not change with the change in time origin.

Q.1. (iii) Determine the modulation index and bandwidth of following FM signal: v(t) = 10 cos (8 × 10⁸t + 7 sin 6 × 10⁴t) (2)

Ans. From the given eq^n $m_f = 7$

Bandwidth $= 2f_m (1 + m_f)$

Here
$$W_m = 6 \times 10^4$$
$$2\pi f_m = 6 \times 10^4$$
$$f_m = \frac{6 \times 10^4}{2\pi}$$

So
$$BW = \frac{2 \times 6 \times 10^4}{2\pi}(1+7)$$
$$= 15 \times 10^4 \text{ Hz.}$$

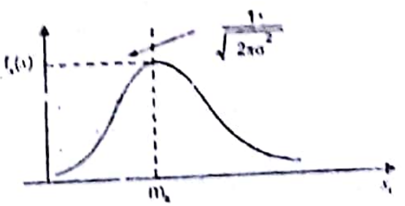
Q.1. (iv) Define Aperture effect. (1)

Ans. Aperture effect: It is the distortion caused due to aperture of input signal wave form while taking the PAM form of Analog modulation.

Q.1. (v) Explain Gaussian Distribution. (2)

Ans. Gaussian distribution is also called normal distribution and mathematical relation is

$$f_X(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-(x-m)^2/2\sigma^2}$$



Q.1. (vi) Calculate the percent power saving for the SSB signal if AM wave is modulated to a depth of 100%. **(2)**

Ans. Power saved = $P_c + P_{SB}$

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

Percentage power saving

$$= \frac{P_c\left(1+\frac{m^2}{4}\right)}{P_t}\times 100$$

for 100% depth, m = 1

$$\text{So \% power saving} = \frac{P_c\left(1+\frac{1}{4}\right)}{P_c\left(1+\frac{1}{2}\right)}\times 100$$

$$= 83.33\%$$

Q.2. (a) A continuous random variable has a probability density function (PDF) expressed as $f(x) = 2e^{-2x}$; for $x \geq 0$.

Determine probability that it will take a value between 1 and 3. **(4)**

Ans. $f(x) = 2e^{-2x}$ for $X \geq 0$

$$P(1 \leq x \leq 3) = \int_1^3 f(x)dx$$

$$= \int_1^3 2e^{-2x}dx$$

$$= 2\left[\frac{e^{-2x}}{-2}\right]_1^3$$

$$= e^{-2} - e^{-6}$$

$$= 0.133.$$

Q.2. (b) Explain Delta modulation and Demodulation with suitable diagrams. **(6)**

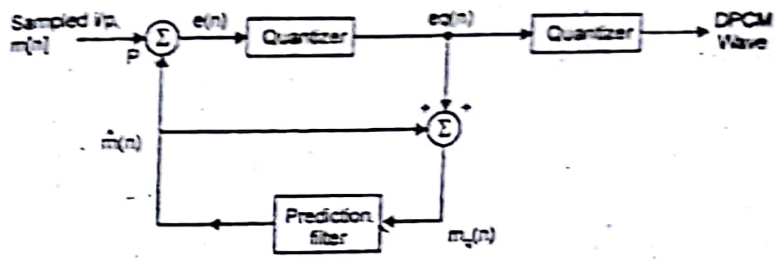
Ans. Delta modulation is a type of one bit modulator depending on the basis of polarity and the changes in the signal. During the quantization process, error occurs and is given by:

$$e[n] = x[n] - \hat{m}[n]$$

and quantization noise is given by:

eq. $$eq[n] = e[n] + q[n]$$

working of delta modulation of different signal flow is given by:



Delta Modulator

$$m_q(n) = \hat{m}(n) + e(n) + q(n)$$

$$m_q(n) = m(n) + q(n)$$

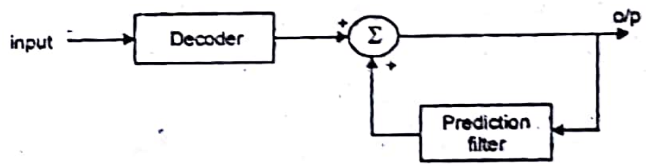
The prediction filter works on processor basis which predicts the next possible incoming signal value and tries to minimise the error between the predicted value and the correct sampled value to be taken in the next sample. It is having two major limitations:

(a) Slope-overload problem

(b) Granular noise distortion

Delta Modulation works at a high frequency which can be much higher as that of PCM system.

Demodulator: It is very much simple as compared to the analog demodulation as shown below:



Hence receiver also works on the prediction as the next incoming signal. So infact this is working like a low pass filter which tries to form the analog signal variation from the incoming digital bit streams

Q.3. (i) Compare PAM, PWM and PPM signals. **(4)**

Ans. PAM:

(1) Here the amplitude of pulse changes as per the analog signal

(2) Pulse width and frequency remains constant.

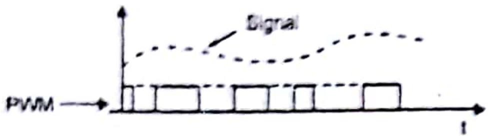
(3) It can be obtained by simply a switch e.g. sample and hold circuit.

(b) PWM:

(1) Here width of the pulse changes as per the modulating signal

(2) Here amplitude and frequency remain same in this type of pulse modulator

(3) This can be achived with the use of comparator.

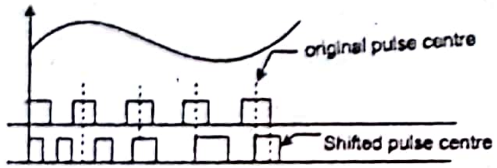


(c) PPM (Pulse Position Modulation)

(1) Here the mean Position of pulse changes as the amplitude variation of the i/p signal

(2) Amplitude and frequency along with the pulse width remains constant.

(3) The wave-shape of PPM is as follow:



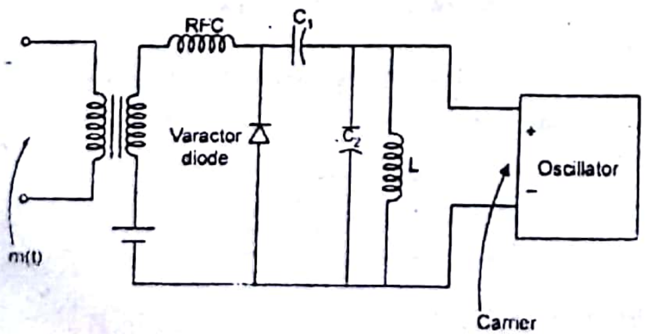
Q.3. (*ii*) Describe Direct method for generation of wide band FM signal. (6)

Ans. In direct method of FM generation the effect of input signal m(t) is directly applied to the carrier signal in terms of frequency variation. For this purpose varactor diode is used in which, Capacitance of the junction changes with respect to the applied voltage and a resonant condition is formed. In such situation the frequency of oscillations is given as:

$$f_0 = \frac{1}{2\pi\sqrt{LC(t)}}$$

Here $C(t)$ indicate the capacitance changing w.r.t. m(t) & since the varactor diode is reverse biased so the capacitance also changes.

For the wide band to be possible from the direct *fm* multipliers are required which changes the deviation and also the modulation index and hence for larger values of *mf*. the wide band *fm* is used Working of wide band *fm* is given as follow:



the capacitance of the diode at any time 't' can be given by

$$C(t) = C_0 - Km(t)$$

Here m(t) = x(t) (modulating signal) it can also be shown easily that

$$\frac{d}{dt}\theta_i(t) = \frac{1}{\sqrt{LC(t)}}$$

and

$$mf = \frac{\Delta f}{fm}$$

Q.4. (*i*) Compare. DSB-FC, DSBC, SSB and VSB modulation techniques. (5)

Ans. Some of the important and comparing features of the given modulation techniques as per question are as follows

(a) DSB- FC

1. Here full carries along with the two side bands is used.
2. Bandwidth is not utilised properly.
3. Efficiency of this system is poor.
4. It is used rarely and for experimental purpose.

(b) DSB-SC

1. Here carrier is suppressed and only two side bands are used.
2. Same signal is transmitted twice.
3. Bandwidth is not ultilised properly.
4. Efficiency improves as compared to DSB-FC.

(c) SSB:

1. Here only single side band is utilized.
2. 83.33% of power is saved.
3. Bandwidth is efficiently utilized.
4. SSB find wide range of applications such as sound transmission, police radios, multiplexing etc.

(d) VSB:

1. It stands for vestigital sideband which means that a part of other side-band and a complete side-band is used for the information modulation and transmission.
2. It is a hybrid of SSB and DSB-SC.
3. Imp. application of VSB are in TV transmission.
4. Bandwidth and power efficiency is also good.

Q.4. (*ii*) Describe Poisson distribution and its properties. (5)

Ans. This is the standard probability distribution used for the discrete random variables.

As the number 'n' increases, the binomial distribution becomes difficult to handle.

If 'n' is very large, probability 'p' is very small and the mean value np is finite, then the binomial distribution can be approximated by the poisson's distribution. Poission distribution is thus the limiting case of binomial distribution.

The probability of the random variable having Poisson distribution is given by

$$P(X=k) = \frac{m^k e^{-m}}{k!}$$

where, m = mean value

- The mean value of Poisson distribution is given by

$$\boxed{M_x = np}$$

- The variance of Poisson distribution is given by

$$\boxed{\sigma_x^2 = np}$$

- The standard deviation of Poisson distribution is

$$\boxed{\sigma_x = \sqrt{np}}.$$

END TERM EXAMINATION [DEC. 2016]
FIFTH SEMESTER [B.TECH.]
COMMUNICATION SYSTEM (ETEE–309)

Time : 3 hrs. **M.M. : 75**

Note: Attempt any five questions including Q. no. 1 which is compulsory. Select one question from each unit. Assume suitable missing data if any.

Q.1. (a) Explain Noise figure and noise temperature in detail. **(5x5=25)**

Ans. Noise figure:

The noise figure is a frequently used measure of an amplifier's goodness, or its departure from the ideal. Thus it is a figure of merit. The noise figure is the noise factor converted to decibel notation:

$$NF = 10 \text{ LOG } F_x$$

where

NF is the noise figure in decibels (dB)

F_x is the noise factor

LOG refers to the system of base-10 logarithms

Noise Temperature:

The noise temperature is a means for specifying noise in terms of an equivalent temperature. Evaluating Equation shows that the noise power is directly proportional to temperature in degrees Kelvin and that noise power collapses to zero at absolute zero (0°K).

Note that the equivalent noise temperature T_e is not the physical temperature of the amplifier, but rather a theoretical construct that is an equivalent temperature that produces that amount of noise power. The noise temperature is related to the noise factor by:

$$T_e = (F_n - 1)\, T_o$$

and to noise figure by:

$$T_e = \left[\text{antilog}\left(\frac{NF}{10}\right) - 1\right] KT_o$$

Now that we have noise temperature T_e, we can also define noise factor and noise figure in terms of noise temperature:

$$F_n = \frac{T_e}{T_o} + 1$$

and

$$NF = 10 \text{ LOG}\left(\frac{T_e}{T_o} + 1\right)$$

The total noise in any amplifier or network is the sum of internally and externally generated noise. In terms of noise temperature:

$$P_{n(total)} = GKB(T_o + T_e)$$

where

$P_{n(total)}$ is the total noise power

Q.1. (b) Define random variable with its types.

Ans. If you have ever taken an algebra class, you probably learned about different variables like x, y and may be even z. Some examples of variables include x = number of heads or y = number of cell phones or z = running time of movies. Thus, in basic math, **a variable** is an alphabetical character that represents an unknown number.

Well, in probability, we also have variables, but we refer to them as random variables. **A random variable** is a variable that is subject to randomness, which means it can take on different values.

As in basic math, variables represent something, and we can denote them with an x or a y or any other letter for that matter. But in statistics, it is normal to use an X to denote a random variable. The random variable takes on different values depending on the situation. Each value of the random variable has a probability or percentage associated with it

1. Discrete Random Variables

We'll start with tossing coins. I want to know how many heads I might get if I toss two coins. Since I only toss two coins, the number of heads I could get is zero, one, or two heads. So, I define X (my random variable) to be the number of heads that I could get.

In this case, each specific value of the random variable - $X = 0$, $X = 1$ and $X = 2$ - has a probability associated with it. When the variable represents isolated points on the number line, such as the one below with 0, 1 or 2, we call it a discrete random variable. **A discrete random variable** is a variable that represents numbers found by counting. For example: number of marbles in a jar, number of students present or number of heads when tossing two coins

2. Continuous Random Variables

Suppose I am interested in looking at statistics test scores from a certain college from a sample of 100 students. Well, the random variable would be the test scores, which could range from 0% (didn't study at all) to 100% (excellent student). However, since test scores vary quite a bit and they may even have decimal places in their scores, I can't possibly denote all the test scores using discrete numbers. So in this case, I use intervals of scores to denote the various values of my random variable.

When we have to use intervals for our random variable or all values in an interval are possible, we call it a continuous random variable. Thus, **continuous random variables** are random variables that are found from measuring - like the height of a group of people or distance traveled while grocery shopping or student test scores. In this case, **X is continuous** because X represents an infinite number of values on the number line

Q.1. (c) Derive an expression of a Frequency Modulated wave with spectrum.

Ans. Recall that a general sinusoid is of the form:

$$e_c = \sin(\omega_c t + \theta)$$

Frequency modulation involves deviating a carrier frequency by some amount. If a sine wave was used to frequency modulate a carrier, the mathematical expression would be:

$$\omega_i = \omega_c + \Delta\omega \sin\omega_m t$$

where

ω_i = instantaneous frequency

ω_c = carrier frequency

$\Delta\omega$ = carrier deviation

ω_m = modulation frequency

This expression shows a signal varying sinusoidally about some average frequency. However, we cannot simply substitute expression in the general equation for a sinusoid. This is because the sine operator acts upon angles, not frequency. Therefore, we must define the instantaneous frequency in terms of angles.

It should be noted that the amplitude of the modulation signal governs the amount of carrier deviation, while the modulation frequency governs the rate of carrier deviation.

The term ω is an angular velocity and it is related to frequency and angle by the following relationship:

$$\omega = 2\pi f = \frac{d\theta}{dt}$$

To find the angle, we must integrate with respect to time, we obtain:

$$\int \omega dt = \theta$$

We can now find the instantaneous angle associated with an instantaneous frequency:

$$\theta = \int \omega_i dt = \int (\omega_c + \Delta\omega \sin\omega_m t) dt$$

$$= \omega_c t - \frac{\Delta\omega}{\omega_m}\cos\omega_m t = \omega_c t - \frac{\Delta f}{f_m}\cos\omega_m t$$

This angle can now be substituted into the general carrier signal to define FM:

$$e_{fm} = \sin\left(\omega_c t - \frac{\Delta f}{f_m}\cos\omega_m t\right)$$

The FM modulation index is defined as the ratio of carrier deviation to modulation frequency:

$$m_{fm} = \frac{\Delta f}{f_m}$$

As a result, the FM equation is generally written as:

$$e_{fm} = \sin(\omega_c t - m_{fm}\cos\omega_m t)$$

This is a very complex expression and it is not readily apparent what the sidebands of this signal are like. The solution to this problem requires knowledge of Bessel's functions of the first kind and order p. In open form, it resembles:

$$J_p(x) = \sum_{k=0}^{\infty} \frac{(-1)^k \left(\frac{x}{2}\right)^{2k+p}}{k!(k+p)!}$$

Scanned by CamScanner

where

$$J_p(x) = \text{mangitude of frequency component}$$
$$p = \text{side frequency number}$$
$$x = \text{modulation index}$$

As a point of interest, Bessel's functions are a solution to the following equation:

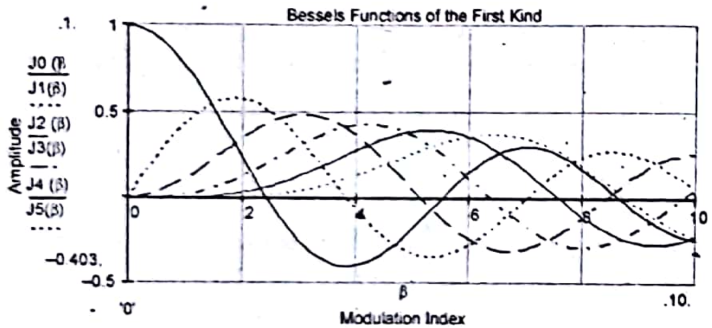
$$x^2 + \frac{d^2y}{dx^2} + x\frac{dy}{dx} + (x^2 - p^2) = 0$$

Bessel's functions occur in the theory of cylindrical and spherical waves, much like sine waves occur in the theory of plane waves.

It turns out that FM generates an infinite number of side frequencies. Each frequency is an integer multiple of the modulation signal. It should be noted that the amplitude of the higher order sided frequencies drops off quickly.

It is also interesting to note that the amplitude of the carrier signal is also a function of the modulation index. Under some conditions, the amplitude of the carrier frequency can actually go to zero. This does not mean that the signal disappears, but rather that all of the broadcast energy is redistributed to the side frequencies.

A plot of the amplitudes of the carrier and first five side frequencies as a function of modulation index resembles:



The open form of the Bessel's function can be used to determine the magnitude of any spectral component.

Q.1. (d) What do you mean by source coding. Explain all of its desirable properties.

Ans. An important problem in communications is the efficient representation of data generated by a discrete source. The process by which this representation is accomplished is called source encoding. The device that performs the representation is called a source encoder. For the source encoder to be efficient, we require knowledge of the statistics of the source. In particular, if some source symbols are known to be more probable than others, then we may exploit this feature in the generation of a source code by assigning short code words to frequent source symbols, and long code words to rare source symbols. We refer to such a source code as a variable_ length code. The Morse code is an example of a variable_ length code. In the Morse code, the letters of the alphabet and numerals are encoded into streams of marks and spaces, denoted as dots "." and dashes "-", respectively.

Our primary interest is in the development of an efficient source encoder that satisfies two functional requirements:

1. The code words produced by the encoder are in binary form.

2. The source code is uniquely decodable, so that the original source sequence can be reconstructed perfectly from the encoded binary sequence.

Source coding does not change or alter the source entropy, i.e. the average number of information bits per source symbol. In this sense source entropy is a fundamental property of the source.

Source coding does, however, alter (usually increase) the entropy of the source coded symbols. It may also reduce fluctuations in the information rate from the source and avoid symbol 'surges' which could overload the channel when the message sequence contains many high probability (i.e. frequently occurring, low entropy) symbols

1. Huffman Coding Algorithm Huffman (1952) devised a variable-length encoding algorithm, based on the source letter probabilities $P(x_i)$, $i = 1, 2, \ldots, L$. This algorithm is optimum in the sense that the average number of binary digits required to represent the source symbols is a minimum, subject to the constraint that the code words satisfy the prefix condition, as defined above, which allows the received sequence to be uniquely and instantaneously decodable.

2. LEMPEL-ZIV CODING

A drawback of the Huffman code is that it requires knowledge of a probabilistic model of the source; unfortunately, in practice, source statistics are not always known a Priori. Moreover, in modeling text we find that storage requirements prevent the Huffman code from capturing the higher-order relationships between words and phrases, thereby compromising the efficiency of the code. To overcome these practical limitations, we may use the Lempel-Ziv algorithm which is intrinsically adaptive and simpler to implement than Huffman coding. Basically, encoding in the Lempel-Ziv algorithm is accomplished by parsing the source data stream into segments that are the shortest subsequences not encountered previously

3. Fano-Shannon Method: This method was suggested by Shannon and Weaver in 1949 and modified by Fano in 1961. This method is used for any source and it involves writing the symbol probabilities in a table in a descending order.

Q.1. (e) Explain the significance of Acceptance angle of fibers.

Ans. Acceptance Angle

The Numerical Aperture (NA) is a measure of how much light can be collected by an optical system such as an optical fibre or a microscope lens.

The NA is related to the acceptance angle θ_a, which indicates the size of a cone of light that can be accepted by the fibre.

Both numerical aperture and acceptance angle are linked to the refractive index via:

$NA = n_a \sin\theta_a = (n_1^2 - n_2^2)^{1/2}$

Where n_1 = refractive index of core

n_2 = refractive index of cladding

n_a = refractive index of air (1.00)

Based on measurements of the critical angle and refractive index of plastic (assume the two plastic blocks are similar) calculate the acceptance angle and numerical aperture for the (2D) block.

The cladding for the Lucite rod is air. Work out the NA using both equations and comment on the results

UNIT-I

Q.2. (a) What do you mean by bandwidth limitation in Communication system? Why is modulation of signal required for transmission? (6.5)

Ans. In the modulation process, two signals are used namely the modulating signal and the carrier In the modulation process, two signals are used namely the modulating signal and the carrier.

Need of Modulation

You may be ask, when the baseband signal can be transmitted directly why to use the modulation ? The answer is that the baseband transmission has many limitations which can be overcome using modulation . It is explained below . In the process of modulation, the baseband signal is translated i.e., shifted from low frequency to high frequency . This frequency shift is proportional to the frequency of carrier. Advantages of Modulation:

→ Reduction in the height of antenna

→ Avoids mixing of signals

→ Increases the range of communication

→ Multiplexing is possible

→ Improves quality of reception

We will discuss each of these advantages in detail below .

1. Reduction in the height of antenna

For the transmission of radio signals, the antenna height must be multiple of $\lambda/4$ where λ is the wavelength , $\lambda = c/f$, where c : is the velocity of light, f: is the frequency of the signal to be transmitted

Now, let us consider a modulated signal at f = 1 MHz . The minimum antenna height is given by,

$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10\times10^6} = 75 \text{ meters}$$

This antenna can be easily installed practically . Thus, modulation reduces the height of the antenna

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver can not separate them from each other . Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals .

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted . They get heavily attenuated . The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance . The modulation process increases the frequency of the signal to be transmitted. Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously . This is possible only with modulation. The multiplexing allows the same channel to be used by many signals . Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time .

5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent . This improves quality of reception.

Q. 2. (b) Explain the difference between probability and PDF. (6)

Ans. The probability distribution function / probability function has ambiguous definition. They may be referred to: Probability density function (PDF) Cumulative distribution function (CDF) or probability mass function (PMF) (statement from Wikipedia)

But what confirm is:

Discrete case: Probability Mass Function (PMF)

Continuous case: Probability Density Function (PDF)

Both cases: Cumulative distribution function (CDF)

Probability at certain x value, $P(X = x)$ can be directly obtained in:

PMF for discrete case and PDF for continuous case.

Probability for values less than x, $P(X < x)$ or Probability for values within a range from a to b, $P(a < X < b)$ can be directly obtained in:

CDF for both discrete / continuous case, Distribution function is referred to CDF or Cumulative Frequency Function

PMF is preferred when Probability at every x value is interest of study. This makes sense when studying a discrete data - such as we interest to probability of getting certain number from a dice roll PDF is preferred when

We wish to model a collected data with a continuous function, by using few parameters such as mean to speculate the population distribution. CDF is preferred when Cumulative probability in a range is point of interest.

Especially in the case of continuous data, CDF much makes sense than PDF - e.g. probability of students' height less than 170 cm (CDF) is much informative than the probability at exact 170 cm (PDF)

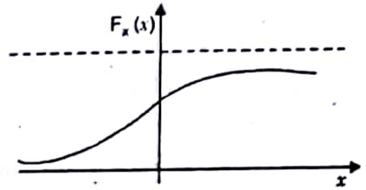
Q.3. Explain cumulative distribution function (CDF). Also explain its properties with proof. (12.5)

Ans. Cumulative distribution function is an important tool to any probabilistic model or to explain the probability behaviour of the system. It can be defined as follow

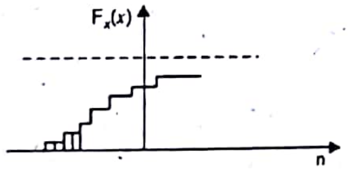
$$F_X(x) = P(X \le x)$$

Here $F_x(x)$ is used to denote CDF. From the given relation it is clear that it will include all those possibilities for which 'X' can take the values upto a certain pre-defined value given by 'x'

It is always a monotonically increasing function and can be shown as follow:



This figure is used for the continuous random variable. For the discrete random variable it can be shown as follow:



This function is having the following properties:

(1) CDF is always bounded between 0 and 1

$\Rightarrow$ $$0 \le F_X(x) \le 1$$

(2) Extreme values of CDF are given by '0' and '1'

$$F_X(\infty) = 1$$

and $$F_X(-\infty) = 0$$

(3) It is always of increasing nature

eg. $$f_X(x_1) \le F_X(x_2)$$

for $$x_1 < x_2$$

UNIT-II

Q.4. (a) A 400 watts carrier is modulated to depth of 75%. Find the total power in AM wave. Assume the modulating signal to be a sinusoidal one. (6.5)

Ans. Carrier power= Pc =400watts

Modulation index=0.75

Total power= $Pc(1+m^2/2)$= 400(1+0.28125) =512.5 watts

Q.4. (b) Explain the significance and applications of vestigial sideband transmission. (6)

Ans. VSB:

The stringent frequency response requirement on the side band filter in SSB-SC system can be relaxed by allowing a part of the unwanted sideband called as Vestige to appear in the output of the modulator. Due to this the design of this sideband filter is simplified to a great extent. But the bandwidth of the system is increased slightly. To generate a VSB signal first the DSB-SC signal and then pass it through a sideband filter.

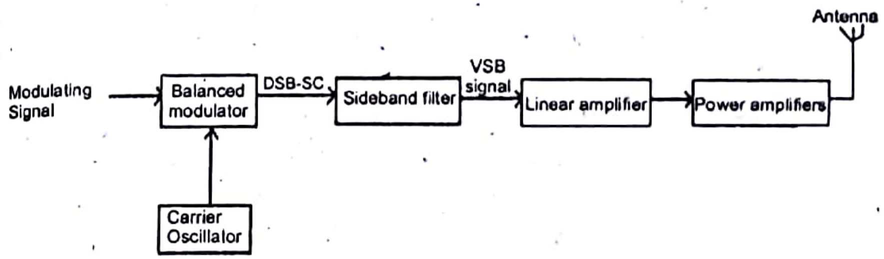
This filter will pass the wanted sideband as it is along with a part of unwanted signal. The VSB signal obtained at the output of filter is applied to a chain of linear amplifiers and raised in power by the power amplifiers. The amplified signal is then applied to the transmitting antenna for transmission of signal.

Advantages of VSB:

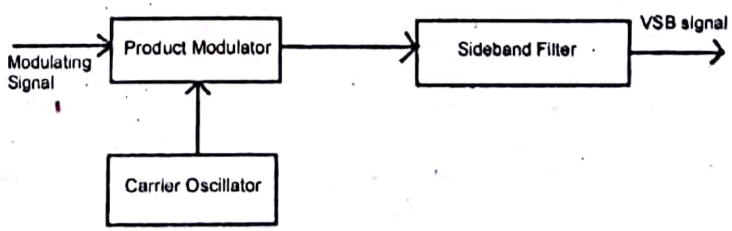
The main advantage of VSB Modulation is the reduction in bandwidth is the reduction in bandwidth. It is almost as efficient as SSB. Due to allowance of transmitting a part of lower sideband, the constraint on the filters have been relaxed. It possesses good phase characteristics and makes the transmission of low frequency components possible Practical filters can be used for partial suppression of LSB.

Applications

VSB modulation has become standard for the transmission of Television signals. Because the video signals need a large transmission bandwidth using DSB-FC or DSB-SC techniques. This is a special type of AM system which is used mainly for the TV transmission all over the world. In the TV transmission it is necessary to transmit the video information and audio information simultaneously. In the VSB transmission the upper sideband of video signal and picture carrier are transmitted without any suppression. Whereas a vestige i.e. a part of lower sideband is transmitted and the remaining part is suppressed.



VSB TRANSMITTER

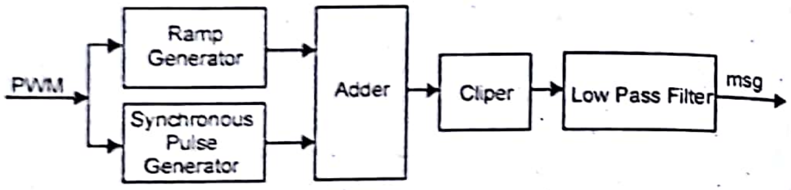


Generation of VSB Signal

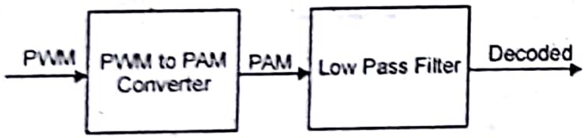
Q.5. (a). With the help of neat circuit diagram explain the generation and detection of a PWM signal. (6)

Ans. How to **detect** or demodulate a **Pulse Width Modulated** Signal? PWM pulse can be detected using **Ramp** generator and some circuit combinations. We have discussed PWM generator circuit using 741 op amp in previous articles. The coded message in the form of PWM can be easily decoded with the help of a **synchronous pulse**. The most simple demodulation principle of PWM signals are describes in this article. The block diagram itself explaining all the decoding principles. Waveforms at different sections of Pulse width demodulation are also given here. PWM signals have wide verities of applications like PWM Inverter circuit, PWM LED brightness control or Dimmer circuit etc.

PWM Detection



PWM coding can be done using 741 op amp that we discussed before. Here the modulated (PWM) wave is applied to the decoder system for getting the message signal. The basic theory behind Pulse width demodulation is that **converting the PWM signal to PAM** (Pulse Amplitude Modulation) signal. PAM can be easily detected by suitable **low pass filter.**



Input PWM wave is applied to Ramp generator and Synchronous'Pulse generator (Fig : 1).

Synchronous pulse generator will generate a pulse waveform such that the pulse will **end at the beginning of each PWM** pulse (See the fig)

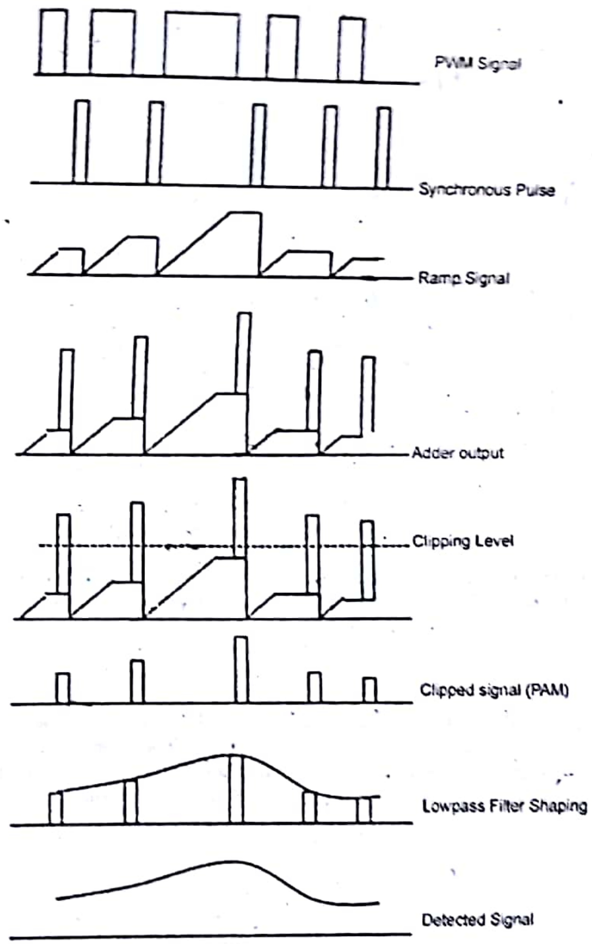
Ramp generator will produce a ramp signal whose **amplitude** is proportional to **width** of the PWM signal

Now apply these Ramp and Synchronous pulse to an **Adder** circuit which adds these signals together.

The next block is a positive Clipper with a specific voltage; Clipper clips the waveform at a particular level as shown in fig. The **output** of clipper will be **PAM** signal, now the PWM signal gets converted to PAM signal.

The PAM can be demodulated by Low Pass filtering method. Thus our next block is Low Pass Filter.

Ramp Generator +Synchronous Pulse + Adder + Clipper = PWM to PAM Converter

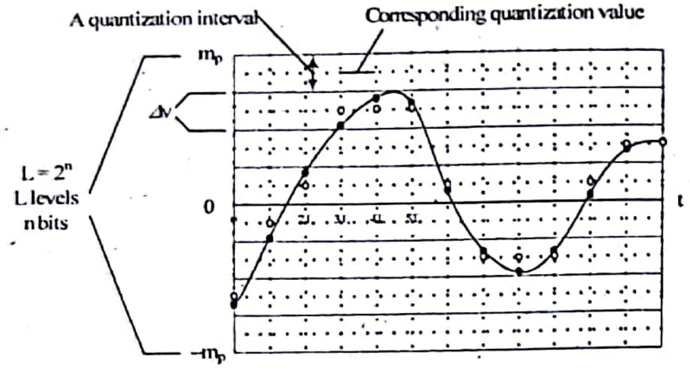


Q.5. (b) What do you mean by Quantization noise/error in PCM? Explain. (6.5)

Ans. Quantization

The process of quantizing a signal is the first part of converting a sequence of analog samples to a PCM code. In quantization, an analog sample with an amplitude that may take value in a specific range is converted to a digital sample with an amplitude that takes one of a specific pre–defined set of quantization values. This is performed by dividing the range of possible values of the analog samples into L different levels, and assigning the center value of each level to any sample that falls in that quantization interval. The problem with this process is that it approximates the value of an analog sample with the nearest of the quantization values. So, for almost all samples, the quantized samples will differ from the original samples by a small amount. This amount is called the quantization error. To get some idea on the effect of this quantization error, quantizing audio signals results in a hissing noise similar to what you would hear when play a random signal.

Assume that a signal with power P_s is to be quantized using a quantizer with $L = 2^n$ levels ranging in voltage from $-m_p$ to m_p as shown in the figure below.

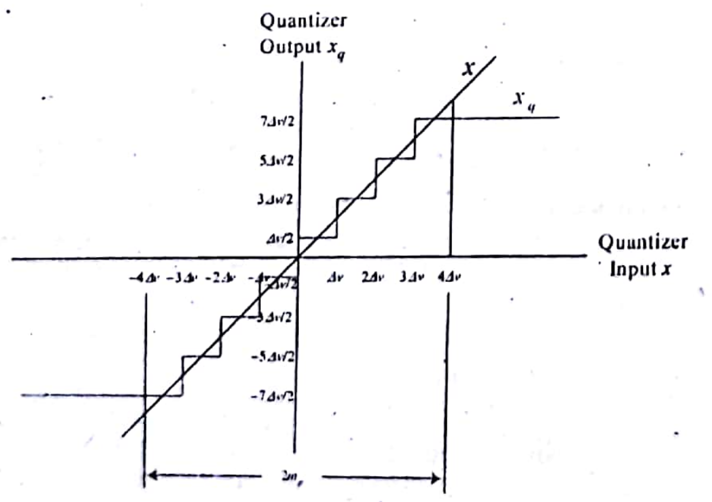


- Quantizer Input Samples x
- Quantizer Output Samples x_q

We can define the variable Δv to be the height of the each of the L levels of the quantizer as shown above. This gives a value of Δv equal to

$$\Delta v = \frac{2m_p}{L}.$$

Therefore, for a set of quantizers with the same m_p, the larger the number of levels of a quantizer, the smaller the size of each quantization interval, and for a set of quantizers with the same number of quantization intervals, the larger m_p is the larger the quantization interval length to accommodate all the quantization range.

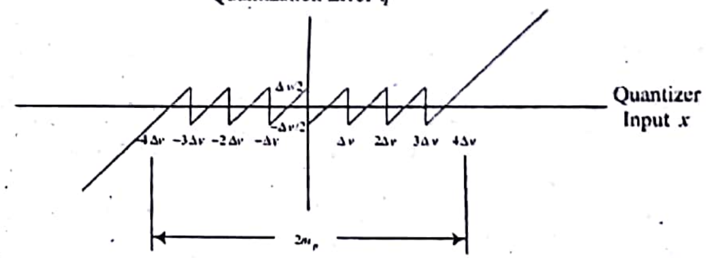


Now if we look at the input output characteristics of the quantizer, it will be similar to the dotted line in the following figure. Note that as long as the input is within the quantization range of the quantizer, the output of the quantizer represented by the dotted line follows the input of the quantizer. When the input of the quantizer exceeds the range of $-m_p$ to m_p, the output of the quantizer starts to deviate from the input and the quantization error (difference between an input and the corresponding output sample) increases significantly.

Now let us define the quantization error represented by the difference between the input sample and the corresponding output sample to be q, or

$$q = x - x_q.$$

Plotting this quantization error versus the input signal of a quantizer is seen next. Notice that the plot of the quantization error is obtained by taking the difference between the blue and red lines in the above figure.



It is seen from this figure that the quantization error of any sample is restricted between $-D\upsilon/2$ and $D\upsilon/2$ except when the input signal exceeds the range of quantization of $-m_p$ to m_p.

UNIT-III

Q.6. (a) Define the following types of channels. **(6)**

(i) Lossless Channel

(ii) Deterministic Channel

Ans. Suppose in the channel matrix, we make the following modifications.

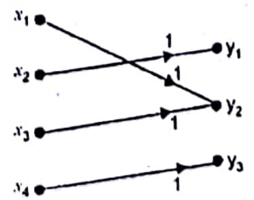
1) Each row of the channel matrix contains one and only one nonzero entry, which necessarily should be a '1'. That is, the channel matrix is symmetric and has the property, for a given k and j, $P(y_j/x_k) = 1$ and all other entries are '0'. Hence given x_k, probability of receiving it as y_j is one. For such a channel, clearly,

$$H(Y/X) = 0 \text{ and } I(X,Y) = H(Y)$$

Notice that it is not necessary that $H(X) = H(Y)$ in this case. The channel with such a property will be called a 'Deterministic Channel'.

Example

Consider the channel depicted in Fig 3.8. Observe from the channel diagram shown that the input symbol x_k uniquely specifies the output symbol y_j with a probability one. By observing the output, no decisions can be made regarding the transmitted symbol



$$P(Y/X) = \begin{bmatrix} 0 & 1 & 0 \\ 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix}$$

Fig.

Scanned by CamScanner

(2) Each column of the channel matrix contains one and only one nonzero entry. In this case, since each column has only one entry, it immediately follows that the matrix P(X|Y) has also one and only one non zero entry in each of its columns and this entry, necessarily be a '1' because:

If $p(y/x_k) = \alpha, p(y/x_r) = 0, r \neq k, r = 1,2,3...m.$

Then $p(x_k, y_j) = p(x_k) \times p(y/x_k) = \alpha \times p(x_k).$

$p(x_r, y_k) = 0, r \neq k, r = 1,2,3...m.$

$$\therefore p(y_j) = \sum_{i=1}^{m} p(x_r, y_j) = p(x_k, y_j) = \alpha p(x_k)$$

$$\therefore p(x_k/y_j) = \frac{p(x_k, y_j)}{p(y_j)} = 1. \text{ and } p(x_r/y_j) = 0. \forall\ r \neq k, r = 1,2,3,...m.$$

It then follows that H(X/Y) = 0 and I(X.Y) = H(X)

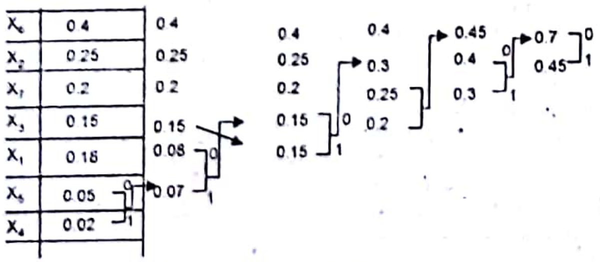
Notice again that it is not necessary to have H(Y) = H(X). However in this case, converse of (a) holds. That is one output symbol uniquely specifies the transmitted symbol, whereas for a given input symbol we cannot make any decisions about the received symbol. The situation is exactly the complement or mirror image of (a) and we call this channel also a deterministic channel (some people call the channel pertaining to case (b) as 'Noiseless Channel', a classification can

be found in the next paragraph). Notice that for the case (b), the channel is symmetric with respect to the matrix P(X|Y)

Q.6. (b) Determine the Huffman code and the efficiency for the following messages with their probabilities given **(6.5)**

X1	X2	X3	X4	X5	X6	X7
0.08	0.25	0.15	0.02	0.05	0.4	0.2

Ans. Arrange the given probabilities in the decreasing order as proceed as follow:



The respective codes of the symbols are as follow:

	Code	Code length
X_1 →	0100	4
X_2 →	01	2
X_3 →	011	3
X_4 →	01011	5
X_5 →	01010	5
X_6 →	00	2
X_7 →	11	2

Average code length

$$\bar{L} = \sum_{i=1}^{7} P_i n_i$$

$$= 0.08\,(4) + 0.25\,(2) + 0.15\,(3) + 0.02\,(5)$$
$$+ 0.05\,(5) + 0.4\,(2) + 0.2\,(2)$$
$$= 0.32 + 0.5 + 0.3 + 0.1 + 0.25 + 0.8 + 0.4$$
$$= 2.67$$

Entropy H(x) is as follow

$$H(x) = -\sum_{i=1}^{7} P_i \log P_i$$

$$= 0.08 \log_2 (0.08) + 0.25 \log_2 (0.25) + 0.15 \log_2 (0.15)$$
$$+ 0.02 \log_2 (0.02) + 0.05 \log_2 (0.05) + 0.4 \log_2 (0.4)$$
$$+ 0.2 \log_2 (0.2)$$
$$= 0.08\,(1.097) + 0.25\,(0.602) + 0.15\,(0.82)$$
$$+ 0.02\,(1.7) + 0.05\,(1.3) + 0.4\,(0.4) + 0.2\,(0.7)$$
$$= 2.509$$

$$\text{Now efficiency} = \frac{H(X)}{\bar{L}} = \frac{2.509}{2.67} \times 100$$
$$= 94\%$$

Q.7. (a) Explain Binary phase shift keying (BPSK) in detail. **(6.5)**

Ans. Generation

To generate the BPSK signal, we build on the fact that the BPSK signal is a special case of DSB-SC modulation. Specifically, we use a product modulator consisting of two components.

***(i)* Non-return-to-zero level encoder,** whereby the input binary data sequence is encoded inpolar form with symbols 1 and 0 represented by the constant-amplitude.

***(ii)* Product modulator,** which multiplies the level encoded binary wave by the sinusoidal carrier of amplitude to produce the BPSK signal. The timing pulses used to generate the level encoded binary wave and the sinusoidal carrier wave are usually, but not necessarily, extracted from a common master clock.

***(iii)* Detection:** To detect the original binary sequence of 1s and 0s, the BPSK signal at the channel output is applied to a receiver that consists of four sections

(a) Product modulator, which is also supplied with a locally generated reference signal that is a replica of the carrier wave

(b) Low-pass filter, designed to remove the double-frequency components of the product modulator output (i.e., the components centered on) and pass the zero-frequency components.

(c) Sampler, which uniformly samples the output of the low-pass filter at where; the local clock governing the operation of the sampler is *synchronized* with the clock responsible for bit-timing in the transmitter.

(d) Decision-making device, which compares the sampled value of the low-pass filters output to an externally supplied *threshold*, every seconds. If the threshold is exceeded, the device decides in favor of symbol 1; otherwise, it decides in favor of symbol 0 levels.

Now the loss occured along the 5Km length of fiber will be

$$\alpha_{dB} \times L = 3.98 \times 5$$
$$= 19.89 \text{ dB}$$

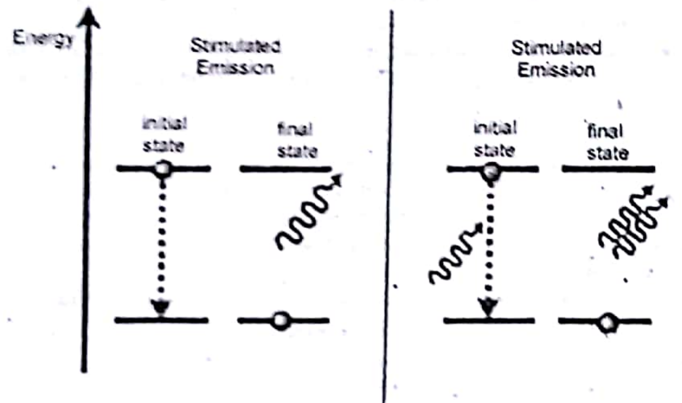
Since

$$\frac{P_i}{P_o} = 10^{19.89/10}$$

$$P_o = \frac{P_i}{10^{19.89/10}} = \frac{1 \times 10^{-3}}{97.5}$$
$$= 10.25 \ \mu\text{watt}$$

Q.2. (b) Discuss the mechanism of optical feedback to provide oscillation and distinctive spectral output from the device. (6)

Ans. LASER stands for **light amplification by stimulated emission of radiation.** I'm not sure what you mean by laser amplification, but I suppose you are asking **how light is amplified inside** a LASER. For this you have to understand the "stimulated emition of radiation" part.



At the left hand side of the image you see an electron that was excited to an upper energy level and after some time naturally returns to the most stable lower energy level with the emission of a photon. At the right hand side of the image you also have an electron at an upper energy level that decays when a photon interacts with it, emiting two **coherent** photons. **This is stimulated emission.**

Coherent light means the **phase** of every photon that comes out of the laser is the same and they can **interfere** with each other in a predictable way. This happens for example if the beam is divided in two, each "child-beam" travels a different distance and they are superimposed again. Or, in other example if a laser beam is **refracted** through **a slit** and shows as an **interference figure** on the target. With **incoherent light** each photon **does interfere** with the others, but the interference evens out because the photons don't have the same starting phase and interfere with each other in constructive and destructive ways for the same distance **traveled**.

What I want to say is that stimulated emission is responsible for coherent light.

Laser pumping is what keeps putting the electron on an upper initial state, so that another photon can come and "pull" the electron down again. You can see laser pumping as the laser power source, because once all electrons are in a upper state, one **spontaneous emission** of a photon will result in a chain of **stimulated emissions** when the spontaneous photon gives birth to two **coherent photons** by stimulated emission.

FIRST TERM EXAMINATION [SEPT. 2017]
FIFTH SEMESTER [B.TECH]
COMMUNICATION SYSTEMS [ETEE-309]

Time : 1½ hrs. **M.M. : 30**

Note: Q. Attempt Q.No. 1 which is compulsory and any two more questions from remaining.

Q.1. (a) In case of FM if the message frequency is 15 kHz and the frequency deviation is 75 kHz, then the bandwidth required for transmission is ——. (2)

Ans. $f_m = 15$ kHz

$\Delta f = 75$ kHz

$$m_f = \frac{\Delta f}{f_m} = \frac{75 \text{ kHz}}{15 \text{ kHz}} = 5$$

As per carson's Rule

$$(BW)_{f_m} = 2f_m(1 + m_f)$$
$$= 2 \times 15 \times 10^3 (1 + 5)$$
$$= 180 \times 10^3 \text{ Hz}$$
$$= 180 \text{ kHz.}$$

Q.1. (b). Explain the difference between ergodic process and stationary process. (2)

Ans. A stationary process is a stochastic process whose statistical properties do not change with time. For a strict-sense stationary process, this means that its joint probability distribution is constant; for a wide-sense stationary process, this means that its 1st and 2nd moments are constant

A *stochastic process* (SP) is a collection of random variables (RV) indexed by time, k. A *stationary* SP is one in which the statistics of the RV, which is "tossed" at each time instant k, are invariant along the time; these "statistics" are usually the average, the variance and the cross-term properties (covariance, correlation) of the RV.

In theory, for a given SP, call it x[k], where k is the discrete time index (discrete SPs are simpler to understand than continuous SPs) you can produce a very high (even infinite) number of "realizations", that is, "instances" of that SP: call them x1[k],x2[k],.../xr[k] where r is the index of the *realization*.

So, there are two indices playing the game here: k for time; and r for realization. In the picture below, related with a *continuous SP*, the time variable is t(horizontal axis) and the realizations are indexed with A,B,C,D,..(vertical axis) and X is the denomination of the SP. This illustrates the meaning of the two axes associated with a SP.

To get the statistics of the RV x[k0] at a certain instant, k0, we have to grab the value of that RV taken at several realizations, xr[k0],r=1,2,3.

Ergodicity Principle: If the time averages converge to the corresponding ensemble averages in the probabilistic sense, then a time-average computed from a large realization can be used as the value for the corresponding ensemble average

Q.1. (c) Derive the mathematical expression for an Amplitude Modulated wave and discuss its generation using square law device. (3)

Ans. Generation of AM Waves:

The circuit that generates the AM waves is called as amplitude modulator and in this post we will discuss two such modulator circuits namely:

1. Square Law Modulator 2. Switching Modulator

Both of these circuits use a non-linear elements such as a diode for their implementation. Both these modulators are low power modulator circuits.

Square Law Modulator

Generation of AM Waves using the square law modulator could be understood in a better way by observing the square law modulator circuit shown in fig.1

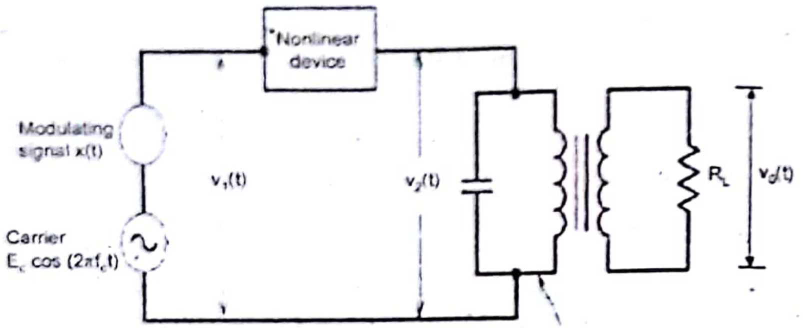


Fig. 1

It consists of the following : A non-linear device, A bandpass filter, A carrier source and modulating signal

The modulating signal and carrier are connected in series with each other and their sum $V_1(t)$ is applied at the input of the non-linear device, such as diode, transistor etc.

Thus,

$$v_1(t) = x(t) + E_c \cos(2\pi f_c t) \quad ...(1)$$

The input output relation for non-linear device is as under

$$v_2(t) = a v_1(t) + b v_1^2(t) \quad ...(2)$$

where a and b are constants.

Now, substituting the expression (1) in (2), we get

$v_2(t) = a[x(t) + E_c \cos(2\pi f_c t)] + b[x(t) + E_c \cos(2\pi f_c t)]^2$

Or,

$$v_2(t) = \underbrace{ax(t)}_{(1)} + \underbrace{aE_c \cos(2\pi f_c t)}_{(2)} + \underbrace{bx^2(t)}_{(3)} + \underbrace{2bx(t)E_c \cos(2\pi f_c t)}_{(4)} + \underbrace{bE_c^2 \cos^2(2\pi f_c t)}_{(5)}$$

Or,

$$v_2(t) = \underbrace{ax(t) + bx^2(t) + bE_c^2 \cos^2(2\pi f_c t)}_{\text{Unuseful Terms}} + \underbrace{aE_c \cos(2\pi f_c t) + 2bx(t)E_c \cos(2\pi f_c t)}_{\text{Useful Terms}}$$

The five terms in the expression for $V_2(t)$ are as under :

Term 1: $ax(t)$: Modulating Signal

Term 2 : $a E_c \cos(2\pi f_c t)$: Carrier Signal

Term 3 : $b x^2(t)$: Squared modulating Signal

Term 4 : $2b x(t) \cos(2\pi f_c t)$: AM wave with only sidebands

Term 5 : $b E_c^2 \cos^2(2\pi f_c t)$: Squared Carrier

Out of these five terms, terms 2 and 4 are useful whereas the remaining terms are not useful.

Let us club terms 2, 4 and 1, 3, 5 as follows to get .

The LC tuned circuit acts as a bandpass filter. Its frequency responce is shown in fig.2 which shows that the circuit is tuned to frequency f_c and its bandwidth is equal to $2f_m$. This bandpass filter eliminates the unuseful terms from the equation of $v_2(t)$.

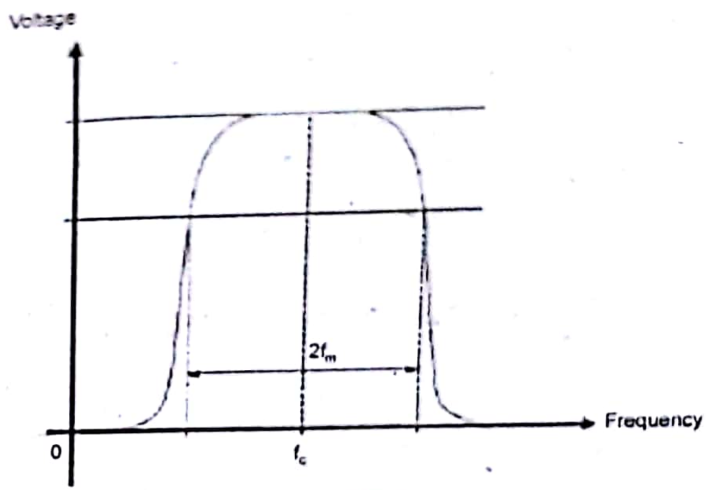


Fig. 2

Hence the output voltage $v_o(t)$ contains only the useful terms .

$$V_o(t) = aE_c \cos(2\pi f_c t) + 2bx(t) E_c \cos(2\pi f_c t)$$

Or,

$$V_o(t) = aE_c\left[1 + \frac{2b}{a}x(t)\right]\cos(2\pi f_c t)$$

Therefore,

$$s(t) = E_c[1 + mx(t)] \cos(2\pi f_c t) \quad ...(3)$$

Q.1. (*d*) What are the drawbacks of Delta Modulation? (3)

Ans. Slope overload distortion: This distortion arises because of large dynamic range of input signal. To reduce this error, the step size must be increased when slope of signal x(t) is high. Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is known as Linear Delta Modulator (LDM).

Granular noise: Granular noise occurs when step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount because of large step size. The error between the input and approximated signal is called granular noise. The solution to this problem is to make step size small.

Adaptive Delta Modulation

To overcome the quantization error due to slope overload distortion and granular noise, the step size (Δ) is made adaptive to variations in input signal x(t). Particularly in the step segment of the x(t), the step size is increased. Also, if the input is varying slowly, the step size is reduced. Then this method is known as Adaptive Delta Modulation (ADM). The adaptive delta modulators can take continuous changes in the step size or discrete changes in the step size

Q.2. (a) Consider the random process $X(t) = A\cos(\omega t + \varphi)$. Where A and ω are constants and φ is a random variable distributed on $[-\pi, \pi]$. Check whether $X(t)$ is Ergodic. **(5)**

Ans. The Ergodic processes are the one which are having, the time average and esemble average, the same the esemble average of the given process is given by:

$$m_x = \int_{-\infty}^{\infty} x\, f_x(x)\, dx$$

or since it is also a function of time so

$$m_x(t) = \int_{-\infty}^{\infty} x\, f_x(x,t) dx$$

$$x(t) = A\cos(\omega_c t + \theta) \text{ and } f_x(x,t) = f_\theta(\theta) = \frac{1}{2\pi}$$

$$m_x(t) = \int_{-\pi}^{\pi} \frac{1}{2\pi} A\cos(\omega_c t + \theta)\, d\theta$$

$$= \frac{A}{2\pi}[\sin(\omega_c t + \theta)]_{-\pi}^{\pi}$$

$$= \frac{A}{2\pi}[\sin(\omega_c t + \pi) - \sin(\omega_c t - \pi)]$$

$$= \frac{A}{2\pi}[-\sin\omega_c t + \sin\omega_c t] = 0$$

Similarly the time average also comes out to be the same.

So the given process is said to be ergodic in nature.

Q.2.(b) The a.c. r.m.s. antenna current of an AM transmitter is 6.2 A when unmodulated and rises to 6.7 A when modulated. Calculate the percentage of modulation. **(5)**

Ans. $I_{r.m.s} = 6.2\,A$ (un.modulated current)

$I_{r.m.s} = 6.7A$ (Modulated)

$$I_t^2 = I_c^2\left(1 + \frac{m^2}{2}\right)$$

so,

$$\frac{I_t^2}{I_c^2} = 1 + \frac{m^2}{2}$$

$$-\frac{m^2}{2} = 1 - \frac{I_t^2}{I_c^2}$$

$$= \frac{I_c^2 - I_t^2}{I_c^2}$$

or

$$\frac{m^2}{2} = \frac{I_t^2 - I_c^2}{I_c^2}$$

$$m = \sqrt{2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]}$$

$$\sqrt{0.3356} = 0.5793 = 57.93\%.$$

Q.3. (a) Derive the expression in dB for maximum signal to noise ratio in PCM. **(5)**

Ans. Signal to noise ratio in PCM is given by

$$\frac{S}{N_q} = \frac{\text{Normalized Signal Power}}{\text{Normalized noise Power}}$$

$$= \frac{\text{Normalized Signal Power}}{\frac{\Delta^2}{12}}$$

Also $q = 2^v$

and since the step size (Δ) is given by

$$\Delta = \frac{2X_{max}}{q} = \frac{2X_{max}}{2^v}$$

$$\frac{S}{N_q} = \frac{\text{Normalized Signal Power}}{\left(\frac{2X_{max}}{2^v}\right)^2 \cdot \frac{1}{12}}$$

and normalized signal Power $= \frac{X_{max}^2}{2}$

so

$$\left(\frac{S}{N_q}\right) = \frac{X_{max}^2(2^{2v})12}{2(4X_{max}^2)} = \frac{12(2^{2v})}{8}$$

$$\left(\frac{S}{N_q}\right) = \frac{3}{2}(2^{2v})$$

$$10\log\left(\frac{S}{N_q}\right) = 10\log\left(\frac{3}{2}2^{2v}\right)$$

$$= 10\log\frac{3}{2} + 10\log(2^{2v})$$

$$= 1.76 + 20\,V\log 2$$

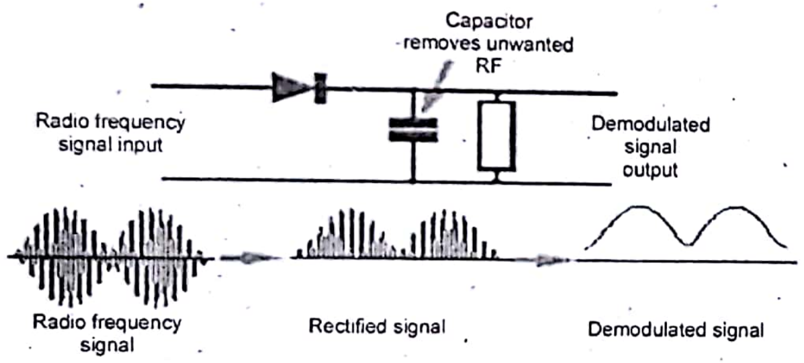
$$\left(\frac{S}{N_q}\right)_{dB} = 1.76 + 6.02\,V$$

Q.3. (b) Explain the noncoherent detection of amplitude modulated wave. Derive the mathematical expression to avoid diagonal clipping in envelope detector. **(5)**

Ans. Diode detector basics: A number of methods can be used to demodulate AM, but the simplest is a diode detector.

It operates by detecting the envelope of the incoming signal which it does by rectifying the signal. Current is allowed to flow through the diode in only one direction, giving either the positive or negative half of the envelope at the output. If the detector is to be used only for audio detection it does not matter which half of the envelope is used, either will work equally well. Only when the detector is also used to supply the automatic gain control (AGC) circuitry will the polarity of the diode matter.

The AM detector or demodulator includes a capacitor at the output. Its purpose is to remove any radio frequency components of the signal at the output. The value is chosen so that it does not affect the audio base-band signal. There is also a leakage path to enable the capacitor to discharge, but this may be provided by the circuit into which the demodulator is connected.



Simple AM diode detector circuit: For a better detection of the modulated signal with the diode detector below, one requirement is that the time constant of the RC filter conforms to:

$(1/\omega c) \le RC \le (1-\mu^2)^{1/2}/_{\omega m \mu}$

where: ωc is the angular frequency of the carrier, ωm is the angular frequency of the information, and μ is the modulation index. The formula is derived from practical experiences and not from mathematical 1st principles. It is unprovable other than by being practical and thinking what a diode detector has to achieve. Firstly, the formula states that RC has to be equal to or greater than 1/ωc. If the RC time constant were too short there would be significant levels (ripple) of the carrier frequency on the output - this is not what is wanted from a diode detector (or an AC rectifier in a power supply) BUT, it's never going to be a perfect brick wall filter and so carrier ripple has to be acceptable (to some degree). Personally, I would like to see the RC time constant 5 times greater than 1/ωc. At the other end of the scale, RC cannot be too big or it will start to significantly attenuate high frequencies in the "detected" analogue waveform that is represented by 1/ωm. Here is a picture that hopefully explains:

If RC is too small, the carrier wave. Product at the output is shown in fig.

The main characteristic to have the best detection is $RC < \frac{\sqrt{1-m_a^2}}{\omega_m m_a}$

If RC is too small, the carrier wave produced at the output is shown in fig. (a)

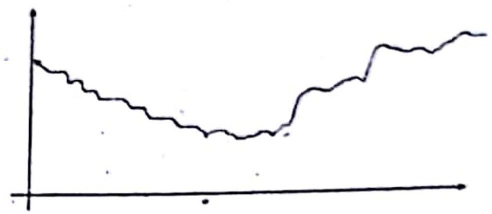


Fig. (a)

If RC is too large, the voltages or the signal at the output does not follow the envelope as is fig. (b)

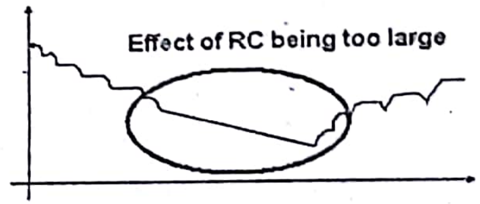


Fig. (b)

If RC is too cange, the voltages or the signal at the output does not follow the envelope as is fig.

This picture was taken from here and basically is saying, if the modulation index is too high for the value of RC chosen there will come a point in the detection of the signal that the RC time constant is too long.

You should also note that as the modulation index approaches 1, the RC time constant has to theoretically be very small and this will make it likely clash with the requirement for it to be significantly greater than 1/ωc.

Q.4. (a) An FM wave is represented by v(t) = 10 sin(6π 10⁶t + 6 sin 1250πt). Find- **(5)**

(i) The carrier and modulating frequencies.

(ii) The modulation index.

(iii) Is it a narrow band or wideband FM ?

(iv) The maximum deviation of FM wave.

(v) What power will this FM dissipate in a 10-ohm resistor ?

Ans. The given expression for FM wave is

$$V(t) = 10\sin(6\pi\, 10^6 t + 6\sin 1250\pi t)$$

(i) Since $\omega_c t = 2\pi f_c t = 6\pi\, 10^6 t$

$$= f_c = \frac{6\pi 10^6 t}{2\pi t} = 3 \times 10^6$$

$$2\pi f_m t = 1250\pi t$$

$$f_m = \frac{1250\pi t}{2\pi t} = 625 \text{ Hz}$$

(ii) $m_f = 6$

(iii) Since $m_f = 6$, so $m_f > 1$

Hence the given equation is of wide band f_m.

(iv) Maximum deviation is

$$\frac{\Delta f}{f_m} = m_f$$

$$\Delta f = f_m \cdot m_f$$

$$= (625)6 = 3750$$

(v) Power dissipation in 10Ω resistor

$$P = \frac{V^2}{R}$$

Since $V_m = 10$

$$V^2 = \left(\frac{V_m}{\sqrt{2}}\right)^2 = \frac{(10)^2}{2 \times 10}$$

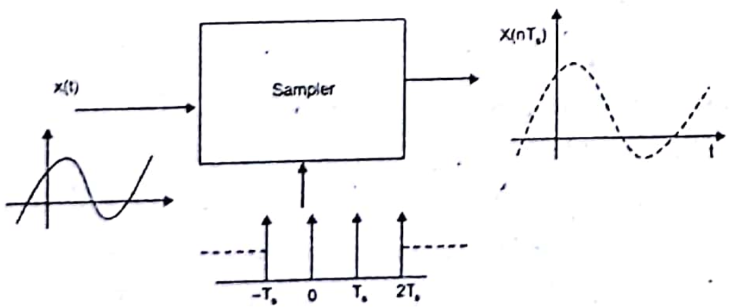
$$= \frac{10 \times 10}{2 \times 10} = 5 \text{ watt}$$

Q.4.(b) (i) State and prove sampling theorem. **(3)**

(ii) Find the Nyquist rate of the signal sin c² 100t **(2)**

Ans. (i) Let $x(t)$ be any continous signal and band limited to f_m Hz

Let us take any sampler, having sampling period to be T_s.



In frequency domain the output samples can be represented as follow:

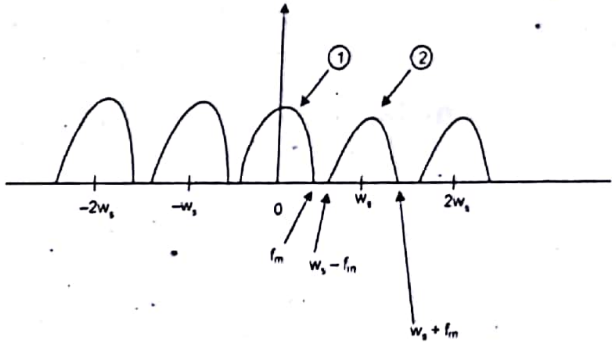


Fig. 1

Take $w_s = f_s$

The fourier transform version of the above pulses can be represented as follow:

$$G(w) = \frac{1}{T_s}\begin{bmatrix} X(w) + X(w - w_s) + X(w + w_s) \\ +.... + X(w - nw_s) + X(w + nw_s) \end{bmatrix}$$

$$G(w) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} X(w - nw_s)$$

Here $w_s = \frac{2\pi}{T_s}$ rad/sec

As shown in fig.1, to avoid the interference of pulse (1) from the pulse (2), we should have the following condition:

$$w_s - f_m \geq f_m$$

or $w_s \geq 2f_m$

Taking $w_s = f_s$

We have

$$f_s \geq 2f_m$$

This indicates that sampling frequency should be two times that of message signal frequency.

Ans. (ii) Let $x(t) = \text{sinc}^2\, 100t$

$$= \left(\frac{\sin 100t}{100t}\right)^2 = \frac{\sin^2 100t}{100t \times 100t}$$

Since $\sin^2\theta = \frac{1 - \cos 2\theta}{2}$

Here $\theta = 100t$

So $\frac{1 - \cos 200t}{2} = \sin^2 100t$

So $x(t)$ Contains $\frac{1}{2} - \frac{\cos 200t}{2}$ Component.

$$200t = 2\pi ft$$

$$f = \frac{200}{2\pi}$$

$$\text{Nyquist rate} = 2f$$

$$= \frac{400}{2\pi} = \frac{200}{\pi}.$$

END TERM EXAMINATION [DEC. 2017]
FIFTH SEMESTER [B.TECH]
COMMUNICATION SYSTEMS
[ETEE–309]

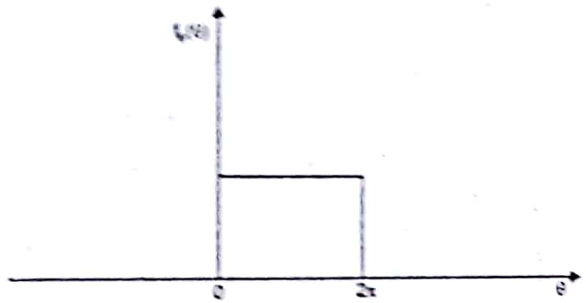
Time : 3 hrs. **M.M. : 75**

Note: Attempt any five questions including Q.No. 1 which is compulsory. Select one question from each unit.

Q.1.(a) Determine the PSD and the mean square value of a random process $x(t) = A\cos(\omega_c t + \theta)$ where θ is an RV uniformly distributed in the range $(0, 2\pi)$. **(5)**

Ans. The given random Process is $x(t) = A\cos(w_c t + \theta)$

θ is Random variable uniformly distributed.



$$E[x^2] = \int_{-\infty}^{\infty} x^2 f_X(x)dx$$

$$= \int_0^{2\pi} x^2 f_\theta(\theta)d\theta \Rightarrow = \int_0^{2\pi} A^2\cos^2(\omega_c t + \theta) f_\theta(\theta)d\theta$$

$$= \int_0^{2\pi} A^2\cos^2(\omega_c t + \theta)\frac{1}{2\pi}d\theta \Rightarrow = \frac{A^2}{2\pi}\int_0^{2\pi}\frac{1+\cos 2(\omega_c t + \theta)}{2}d\theta$$

$$= \frac{A^2}{4\pi}\int_0^{2\pi}[1+\cos 2(\omega_c t + \theta)]d\theta$$

$$= \frac{A^2}{4\pi}\left\{[\theta]_0^{2\pi} + [\sin 2(\omega_c t + \theta)]_0^{2\pi}\right\}$$

$$= \frac{A^2}{4\pi}2\pi = \frac{A^2}{2}$$

Q.1.(b) What is phase deviation and frequency deviation in angle modulated system ? **(5)**

Ans. Frequency deviation : It is the change in the frequency from the original value either in Positive direction and in negative direction it is denoted by Δf

Let if max frequency achieved in frequency modulated system be f_{max} then $\Delta f = f_{max} - f_c$

or if minimum frequency achieved in angle modulated signal is f_{min} then $\Delta f = f_c - f_{min}$

so $$f_c - f_{min} = f_{max} - f_c$$

$\Rightarrow$ $$2f_c = f_{max} + f_{min}$$

$$f_c = \frac{f_{max} + f_{min}}{2}$$

$$\Delta f \propto |m(t)|$$

$$\Delta f = K_f|m(t)|$$

Where k_f is the frequency sensitivity constant.

Phase deviation : Let the fixed phase be θ_c and variable phase be y

then $$\theta_{max} = \theta_c + \psi_{max}$$
$$\theta_{min} = \theta_c - \psi_{max}$$

Then phase deviation can be thought of an angle which changes from the fix value. Let $\Delta\theta$ be the phase deviation

then $$\Delta\theta \propto |m(t)|$$
$$\Delta\theta = K_p m(t)$$

Where K_p is the phase sensitivity constant.

Q.1.(c) What is the necessity of non-uniform quantization and explain companding? **(5)**

Ans. Non Uniform Quantization:

Although uniform quantization is straight forward and appears to be a natural approach it may not be optimal. Suppose f is much more likely to be in one region than in others. It is reasonable to assign more reconstruction levels to that region. If f falls rarely between t_1 and t_2, the reconstruction level r_1 is rarely used. Rearranging reconstruction levels r_1, r_2, r_3, r_4 so that they all lie between t_1 and t_4 makes more sense. Quantizers in which reconstruction and transition levels do not have even spacing is called non-uniform quantization. The notion that uniform quantizer is the optimal MMSE when $P_f(f)$ is uniform suggests another approach. Specifically we can f to g in such a way that $P_g(g)$ is uniform. Quantize g with a uniform quantization and then perform the inverse nonlinearity. The non linearity is called companding. One choice of nonlinearity or companding is given by $g = [u] = \int_{x_0=-\infty}^{u} P_f(x)dx - \frac{1}{2}$. The resulting $P_g(g)$ is uniform in the interval (–1/2, 1/2). The nonuniform quantization by g = companding minimizes the distortion. $D' = E[(\hat{g} - g)^2]$

The data rate is important in telecommunication because it is directly proportional to the cost of transmitting the signal. Companding is a common technique for reducing the data rate of audio signals by making the quantization levels unequal.

The loudest sound that can be tolerated (120 dB SPL) is about one-million times the amplitude of the weakest sound that can be detected (0 dB SPL). However, the ear cannot distinguish between sounds that are closer than about 1 dB (12% in amplitude) apart. In other words, there are only about 120 different loudness levels that can be detected, spaced logarithmically over an amplitude range of one-million. This is important for digitizing audio signals. If the quantization levels are equally spaced, 12 bits must be used to obtain telephone quality speech. However, only 8 bits are required if the quantization levels are made unequal, matching the characteristics of human hearing. Therefore if the signal is small, the levels need to be very close together; if the signal is large, a larger spacing can be used.

Companding can be carried out in three ways as follows:

(1) run the analog signal through a nonlinear circuit before reaching a linear 8 bit ADC,

(2) use an 8 bit ADC that internally has unequally spaced steps, or

(3) use a linear 12 bit ADC followed by a digital look-up table (12 bits in, 8 bits out).

Each of these three options requires the same nonlinearity, just in a different place: an analog circuit, an ADC, or a digital circuit. Two nearly identical standards are used for companding curves:

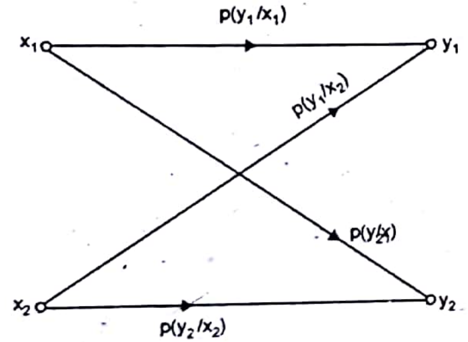
μ255 law (also called mu law), used in North America, and "A" law, used in Europe.

Both use a logarithmic nonlinearity, since this is what converts the spacing detectable by the human ear into a linear spacing.

Q.1.(*d*) Explain various types of channels with their corresponding matrix. **(5)**

Ans. There can be different channels like

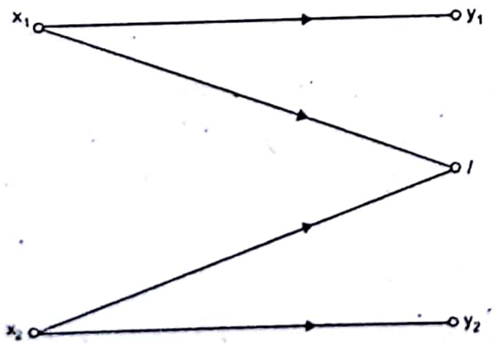
(1) Binary symmetric channel



Channel matrix of such a channel:

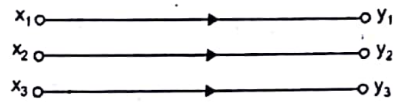
$$P(Y/X) = \begin{bmatrix} p(y_1/x_1) & p(y_2/x_1) \\ p(y_1/x_2) & p(y_2/x_2) \end{bmatrix}$$

(2) Binary Erasure channel



$$P(Y/X) = \begin{bmatrix} p(y_1/x_1) & p(l/x_1) & p(y_2/x_1) \\ p(y_1/x_2) & p(l/x_2) & p(y_2/x_2) \end{bmatrix}$$

(3) Noiseless Channel



Here each input is connected to the single output.

$$P(Y/X) = \begin{bmatrix} p(y_1/x_1) & 0 & 0 \\ 0 & p(y_2/x_2) & 0 \\ 0 & 0 & p(y_3/x_3) \end{bmatrix}$$

$$= \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix}$$

Q.1.(*e*) What do you meant by quantum efficiency and responsivity of a photodiode? Calculate the wavelength at which quantum efficiency and respectively are equal. **(5)**

Ans. Please ref. to Q. No. 9(b) of End Term. ETEE-309, DEC-2015 at page No. 31

UNIT-I

Q.2.(*a*) Show that the random procces $x(t) = A\cos(w_c t + \theta)$ where θ is an RV uniformly distributed in the range $(0, 2\pi)$ is a wide - sense stationary process. **(6.5)**

Ans. The given random Process is

$$x(t) = A\cos(w_c t + \theta)$$

and θ is a random variable of $(0, 2\pi)$

Any random process is said to be of wide. Sense stationary nature if the mean comes out to be constant and Auto-Correlation do not depend on 't'.

$$E[X(t)] = \int_0^{2\pi} \frac{1}{2\pi} A\cos(w_c t + \theta)\, d\theta$$

$$= \frac{1}{2\pi}\int_0^{2\pi} A\cos(w_c t + \theta)\, d\theta$$

$$= \frac{A}{2\pi}\left[\sin(w_c t + \theta)\right]_0^{2\pi}$$

$$= 0$$

Hence the mean value comes out to be fix.

The auto-correlation of this process is given by:

$$R_{XX}(t, t+\tau) = E[X(t)\, x(t+t)]$$

$$= \frac{A^2}{2}\cos w_c\tau + \frac{A^2}{2} E[\cos(2w_c t + w_c\tau + 2\theta)]$$

Since $$E[(x(t)x(t+\tau))] = E[A\cos(w_c t+\theta)A\cos(w_c t+\theta+t)]$$

so $$R_{XX}(t, t+\tau) = \frac{A^2}{2}\cos w_c\tau$$

$$= R_{XX}(\tau)$$

Hence the Auto-Correlation does not depend on the 't'.

Hence the given process is said to be wide-sense stationary process.

Q.2.(b) Mean and variance of sum of two statistically independent random variables is sum of individual means and variance. Justify your answer. (6)

Ans. Let the two independent random variables be X, Y

Let, $$Z = X+Y$$

$$m_z = E[Z] = \int_{-\infty}^{\infty} z f_z(z)dz$$

$$= \int_{-\infty}^{\infty} (X+Y)f_{XY}(x,y)\,dxdy$$

$$= \int_{-\infty}^{\infty} X f_X(x)dx + \int_{-\infty}^{\infty} Y f_Y(y)dy$$

$$= m_X + m_Y$$

Since $$\sigma_X^2 = \text{variance}$$

$$= (x-m_X)^2$$

$$= E(X^2) - m_X^2$$

$$\sigma_Y^2 = E[Y^2] - m_Y^2$$

$$\sigma_Z^2 = \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} (Z-m_z)^2 f_{XY}(x,y)\,dxdy$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} (Z^2 + m_z^2 - 2zm_z) f_{XY}(x,y)\,dxdy$$

$$\sigma_Y^2 = \sigma_X^2 + \sigma_Y^2$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} Z^2 f_{XY}(x,y)\,dxdy$$

$$+ \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} m_z^2 f_{XY}(x,y)\,dxdy + 2\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} 2m_z^2 f_{XY}(x,y)\,dxdy$$

OR

Q.3. Explain cumulative distribution function (CDF). Also explain its properties with proof. (12.5)

Ans. Refer to Q. 3. End Term Examination 2016.

CDF (Cumulative Distribution Function)/PDF(Probability Distribution Function)

Definition: If X is a real random variable, then the function F: R→R defined by

$$F_X(x) = P(X < x), \text{ where } \infty < x < \infty$$

is called distribution function (d.f) of the random variable X. It is also known as Cumulative Distribution Function (c.d.f). $F_X(x) = \Sigma pi$ where i is such that $xi \le x$, $pi \ge 0$ and $\Sigma Pi = 1$

In probability theory and statistics, the cumulative distribution function (CDF, also cumulative density function) of a real-valued random variable X, or just distribution function of X, evaluated at x, is the probability that X will take a value less than or equal to x

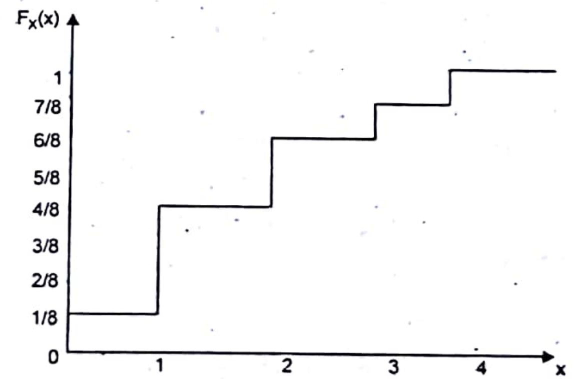
If X is a discrete random variable then the distribution function can be obtained by adding the probabilities successively. Suppose X takes values x1,x2,.....,xn...with probabilities p(x1),p(x2)....p(xn)...such that p(xi)=1, then by the definition of the distribution function $F_X(x)$ we have

X	x_1	x_2	x_3	...	...	
$F_X(x)$	$p(x_1)$	$p(x_2)$	$p(x_3)$	..	...	..

$F_X(xi) = p(X \le xi)$

$F_X(xi) = p(x1)+p(x2)+...+p(xi)$

So for discrete random variable X, $F_X(x)$ is a monotonically increasing, right continuous, step function.



Hence the probability distribution function is defined on the entire real line and not just on the range of the random variable. Indeed, P(X=xi) represents the step size or jump at xi of the F_X curve i.e.

$P(X=xi) = F_X(xi) - F_X(xi-1)$

For the CDF following properties hold:

(1) $F(-\infty) = 0$ and $F(+\infty) = 1$.

(2) F is a non-decreasing function, i.e. if $x1 \le x2$, then $F(x1) \le F(x2)$.

(3) If $F(x0) = 0$, then $F(x) = 0$ for every $x\ d \le x0$.

(4) P{X > x} = 1 – F(x).

(5) P({x1 < X ≤ x2}) = F(x2) – F(x1).

Proof. Property 1: By definition, FX(a) = P(A) where A = {X ≤ a}. Then F(= +∞) = P(A), with A = {X ≤ + ∞}. This event is the certain event, since A = {X ≤ +∞} = {X ε R} = S. Therefore F(= + ∞) = P(S) = 1. Similarly, Then F(= –∞) = P(A), with A = {X ≤ – ∞}. This event is the impossible event, RV X is a real variable, so F(= –∞) = P(ϕ) = 0.

Proof. Property 2: This is a consequence of the fact that probability is a nondecreasing function, i.e. if A ⊆ B then P(A) ≤ P(B). Indeed, recall that if A ⊆ B then the set B can be partitioned into B = A+ B ∩ A⁻·, so, using the additivity property of probability, P(B) = P(A) + P B ∩ A⁻ ≥ P(A). If x1 ≤ x2, then the corresponding events A = {X ≤ x1} and B = {X ≤ x2} have the property that A ⊂ B. Therefore P({X ≤ x1}) ≤ P({X ≤ x2}), which by the definition of a CDF means F(x1) ≤ F(x2)

Proof. Property 3: This property is related to the nondecreasing property of the CDF: going backwards, to the left, toward –∞, the CDF can only decrease or stay constant. If F(x0) = 0, to the left of x0 it can only stay constant (=0) since it cannot take negative values. For x ≤ x0 the events A = {X ≤ x} and B = {X ≤ x0} have the property that A ⊂ B, therefore 0 ≤ F(x) = P ({X ≤ x}) ≤ P ({X ≤ x0}) = F (x0) = 0 therefore F(x) = 0

Proof. Property 4: This is a consequence of the properties of probability of complementary events: recall that P A⁻ = 1 – P(A). SetA = {X ≤ x}; then its complement (the set of all elementary events which are not in A) is A⁻ = {X > x}, i.e. {X ≤ x} + {X > x} = S = R. As above, P A⁻ = 1 – P(A), but with from the definition of the CDF, P A⁻ = P({X > x}) = 1 – P({X d ≤ x}) = 1 – F

Proof. Property 5: This property shows how to use the CDF to calculate the probability that RV X returns a value in a given interval [x1, x2]. The CDF is defined on semi-infinite intervals of the form (–∞, x1] and (–∞, x2]. Note that the events A = {X ∈ (–∞, x1]}, B = {X ∈ (–∞, x2]} and C = {X ∈ (x1, x2]} are in the relationship B = A + C since A and C are disjoint and B = A ∪ C. Therefore P(B) = P(A) + P(C), or P(C) = P(B) – P(A). Hence the property.

UNIT–II

Q.4.(a) Derive an expression of a frequency Modulated wave with spectrum. (6)

Ans. Refer to Q. 1. (c) of End Term Examination 2016.

Q.4.(b) Prove that signal to quantization noise ratio for linear quantization is (S/N)dB ≤ (4.8 + 6n). (6.5)

Ans. Let us take the signal peak amplitude to be X_{max}

Signal power $= X^2_{max}$

Quantization noise $= \dfrac{\Delta^2}{12}$

and $\Delta = \dfrac{2X_{max}}{2^n}$

When n = *number* of bits used in code

so $\Delta^2 = \dfrac{(2X_{max})^2}{2^{2n}}$

$$\frac{S}{N} = \frac{X^2_{max}\, 2^{2n}\cdot 12}{4X^2_{max}} = 3\times 2^{2n}$$

$$\left(\frac{S}{N}\right)_{dB} = 10\log_{10}(3\times 2^{2n})$$

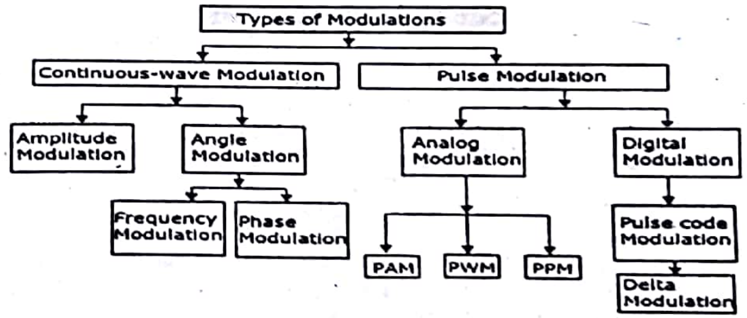
$$= 10\log_{10}(3) + 10\log_{10}(2^{2n}) = 4.77 + 6n.$$

OR

Q.5.(a) Explain the performance comparison of various pulse analog modulation methods. (6.5)

Ans. In a communication system, the modulation is an important step. Modulation is the process of transmitting a message signal (Baseband signal with low frequency) from transmitter to receiver without changing its characteristics (like amplitude, frequency, phase) by using a carrier signal (high frequency) which varies in accordance with the instantaneous values of the low frequency wave by keeping its frequency and phase constant.

The modulation techniques are classified into two major types: analog and digital or pulse modulation. We have discussed previously the different types of modulation techniques, let us understand the basic difference between PAM, PWM, and PPM.



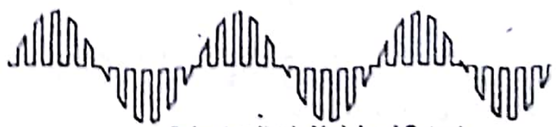
Before going to discuss the difference between PAM, PWM, and PPM, let us discuss individually each. All these are pulse analog modulation techniques

Pulse Amplitude Modulation

By varying the amplitude of the pulses (the carrier signal) in proportion to the instantaneous values of the analog signal (the message signal).

Carrier Pulses Train Used for Modulation

Analog Signal for Modulating



Pulse Amplitude Modulated Output

Pulse Amplitude Modulation (PAM) Signals

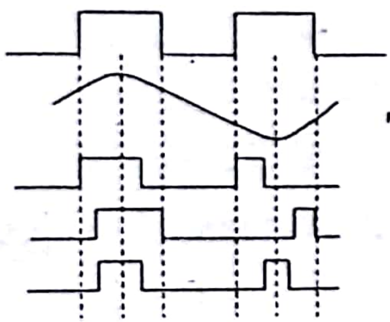
The above figure illustrates the time domain representation of the PAM technique which mentions analog message and PAM modulated signal as an output.

Pulse amplitude modulation is used in the popular Ethernet communication standard. The PAM modulator and demodulator circuits simple compared to other kinds of modulation and demodulation techniques.

There are two categories of PAM techniques, one is the pulses have the same polarity and the other in which the pulses can have both positive and negative polarities according to the amplitude of the modulating signal.

Pulse Width Modulation

The Pulse width Modulation– By varying the width of the pulses (the carrier signal) in proportion to the instantaneous values of the analog signal (the message signal).



The width of the pulse varies, but the amplitude of the pulse remains constant. Amplitude limiters are used to make the amplitude constant. These circuits clip-off the amplitude, to a preferred level and hence the noise is limited.

There are three types of PWM. They are:

The leading edge of the pulse being constant, the trailing edge varies according to the message signal.

The trailing edge of the pulse being constant, the leading edge varies according to the message signal.

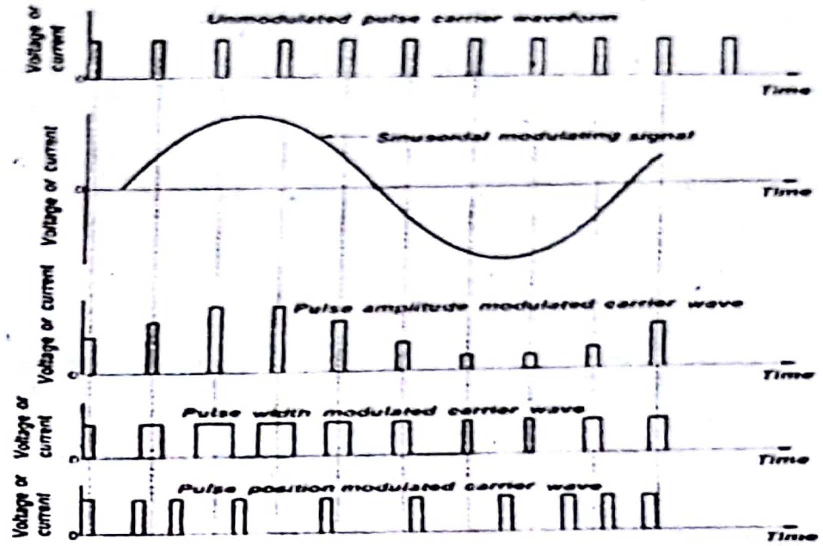
The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal.

Pulse Position Modulation

By varying the position of the pulses (the carrier signal) in proportion to the instantaneous values of the analog signal (the message signal).

Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal.

Hence, the position of these pulses is proportional to the width of the PWM pulses. But the main disadvantage of the PPM modulation technique is, The synchronization between transmitter and receiver must be needed.



Difference Between PAM, PWM, and PPM

In all the above cases, we detect the message of the pulse modulated signal and reconstruct the original analog signal.

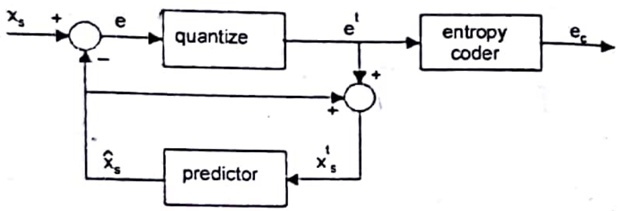
Difference Between PAM, PWM, and PPM

The below table gives the detailed difference between PWM, PAM, and PPM.

S.No.	Parameter	PAM	PWM	PPM
1.	Type of Carrier	Train of Pulses	Train of Pulses	Train of Pulses
2.	Variable Characteristic of the Pulsed Carrier	Amplitude	Width	Position
3.	Bandwidth Requirement	Low	High	High
4.	Noise Immunity	Low	High	High
5.	Information Contained in	Amplitude Variations	Width Variations	Position Variations
6.	Power efficiency (SNR)	Low	Moderate	High
7.	Transmitted Power	Varies with amplitude of pulses	Varies with variation in width	Remains Constant
8.	Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
9.	Bandwidth depends on	Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
10.	Transmitter power	Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneou transmitter Power remains constant with the width of the pulses
11.	Complexity of generation and detection	Complex	Easy	Complex
12.	Similarity with other Modulation Systems	Similar to AM	Similar to FM	Similar to PM

Q.5.(b) Explain differential Pulse code modulation (DPCM) in detail. (6)

Ans. Differential pulse code modulation (DPCM) is a procedure of converting an analog into a digital signal in which an analog signal is sampled and then the difference between the actual sample value and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value. DPCM code words represent differences between samples unlike PCM where code words represented a sample value. Basic concept of DPCM - coding a difference, is based on the fact that most source signals show significant correlation between successive samples so encoding uses redundancy in sample values which implies lower bit rate. Realization of basic concept (described above) is based on a technique in which we have to predict current sample value based upon previous samples (or sample) and we have to encode the difference between actual value of sample and predicted value (the difference between samples can be interpreted as prediction error). Because it's necessary to predict sample value DPCM is form of predictive coding. DPCM compression depends on the prediction technique, well-conducted prediction techniques lead to good compression rates, in other cases DPCM could mean expansion comparing to regular PCM encoding.



Communication is the main transmission means of modern network, its development history is only one hundred and twenty years, but has experienced three generation: short wavelength multimode optical fiber, long wavelength multimode optical fiber and long wavelength single-mode fiber.

Using optical fiber communication is a major change in the history of communication , take United States, Japan, Britain, France for example, more than 20 countries have announced that it would no longer construct cable communication line, and committed to the development of optical fiber communication. China optical fiber communication has entered the practical stage.

Design of DPCM system means optimizing the predictor and quantizer components, because the quantizer is included in prediction loop there is complex dependancy between the prediction error and quantizaton error so joint optimization should be performed to assure optimal results. But, modeling such optimization is very complex so optimization of those two components are usually optimized separately. It has been shown that under the mean-squared error optimization criterion, apart constructions of quantizor and predictor are good approximations of joint optimization. Same as in the previous paragraph, facts in this paragraph are also applicable to signals in general.

A typical example of a signal good for DPCM is a line in a continuous-tone (photographic) image which mostly contains smooth tone transitions. Another example would be an audio signal with a low-biased frequency spectrum.

For illustration, we present two histograms made from the same picture which were coded in two ways. The histograms show the PCM and DPCM sample frequencies, respectively

UNIT-III

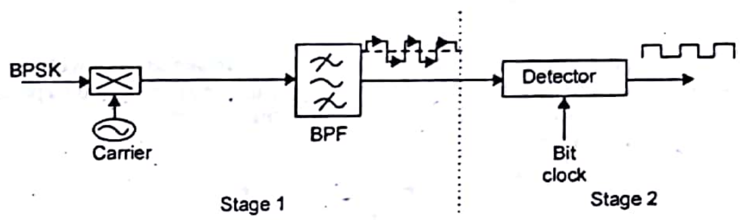
Q.6.(a) Explain the generation and detection of BPSK signal. Also draw its spectrum and geometrical representation. (6.5)

Ans. Refer to Q. 7. (a) of End Term Examination 2016.

BPSK Demodulator

The block diagram of BPSK demodulator consists of a mixer with local oscillator circuit, a bandpass filter, a two-input detector circuit. The diagram is as follows.

BPSK Demodulator



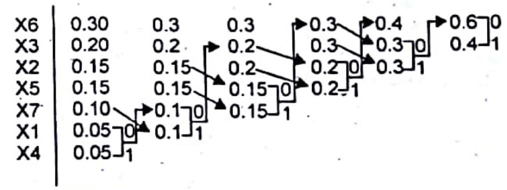
By recovering the band-limited message signal, with the help of the mixer circuit and the band pass filter, the first stage of demodulation gets completed. The base band signal which is band limited is obtained and this signal is used to regenerate the binary message bit stream.

In the next stage of demodulation, the bit clock rate is needed at the detector circuit to produce the original binary message signal. If the bit rate is a sub-multiple of the carrier frequency, then the bit clock regeneration is simplified. To make the circuit easily understandable, a decision-making circuit may also be inserted at the 2nd stage of detection.

Q.6.(b) Determine the Huffman code and the efficiency for the following messages with their probabilities given. (6)

X1	*X2*	*X3*	*X4*	*X5*	*X6*	*X7*
0.05	0.15	0.2	0.05	0.15	0.3	0.1

Ans.



X	*P*(*x*)	Code(*x*)	Code length (*n*)
X1	0.05	1000	4
X2	0.15	000	3
X3	0.20	10	2
X4	0.05	1001	4
X5	0.15	001	3
X6	0.30	00	2
X7	0.10	101	3

$$H(X) = -[0.05 \log (0.05) + 0.15 \log (0.15) + 0.20 \log (0.20) + 0.05 \log (0.05) + 0.15 \log (0.15) + 0.3 \log (0.3) + 0.1 \log 0.1]$$

$$= -[0.1 \log (0.05) + 0.3 \log (0.15) + 0.3 \log (0.3) + 0.2 \log (0.2) + 0.1 \log 0.1]$$

$$0.13 + 0.247 + + 0.157 + 0.14 + 0.1$$

$$= \frac{0.774}{0.307} = 2.57$$

$$L = 0.05(4) + 0.15 (3) + 0.20 (2) + 0.05 (4) + 0.15 (3) + 0.30 (2) + 0.10 (3)$$

$$= 0.2 + 0.45 + 0.4 + 0.2 + 0.45 + 0.6 + 0.3 = 2.6$$

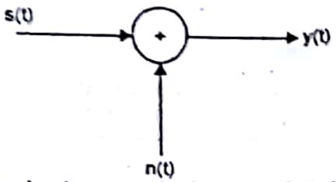
Now efficiency of coding :

$$\eta = \frac{H(X)}{L} = \frac{2.57}{2.6} = 98.84\%.$$

Q.7.(a) Find the capacity of the additive white Gaussian noise channel. (6)

Ans. Additive white Gaussian noise (AWGN) consists of all the frequencies and the noise is additive in nature throughout the length of the channel.

The channel can be modelled as



The channel can be Gaussian in nature in the sense that the PDF of noise is Gaussian. Capacity of such channel in given by

$$f_N(n) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-(n-m_n)^2/2\sigma^2}$$

Capacity of such channel is given by

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

Here
S : Signal Power
N : Noise Power
B : Bandwidth

$N = \eta B$, Where is the noise spectral density let us find the capacity of a this channel when bandwidth is theoretically increased to ∞

$$C|_{B\to\infty} = \underset{B\to\infty}{\text{lt}}\ B\log_2\left(1+\frac{S}{\eta B}\right) = \frac{B}{\beta\eta}\left(\frac{\eta BS}{S}\right)\log_2\left(1+\frac{S}{\eta B}\right)$$

$$= \frac{S}{\eta}\log_2\left(1+\frac{S}{\eta B}\right)^{\eta B/S}$$

$$C|_{B\to\infty} = \frac{S}{\eta}\log_2\left[\lim_{x\to\infty}(1+x)^{1/x}\right]$$

where $$x = \frac{S}{\eta B} = \frac{S}{\eta}\log_2 e = 1.44\ S/\eta$$

Q.7.(b) Verify the following expression:
$H(X, Y) = H(X/Y) + H(Y)$ **(6.5)**

Ans. Since

$$H(X,Y) = -\sum_{j=1}^{m}\sum_{k=1}^{n} P(x_j y_k)\log P(x_j y_k)$$

$$= -\sum\sum P(x_j y_k)\log\left[P(x_j/y_k)\cdot P(y_k)\right]$$

$$= -\sum\sum P(x_j y_k)\{\log P(x_j/y_k) + \log P(y_k)\}$$

$$= -\sum\sum P(x_j y_k)\log P(y_k) - \sum\sum P(x_j y_k)\log P\left(x_j/y_k\right)$$

$$= -\sum_{j=1}^{m} P(x_j)\sum_{k=1}^{n} P(y_k)\log P(y_k) - \sum_{j=1}^{m}\sum_{k=1}^{n} P(x_j y_k)\log(P(x_j/y_k)) = H(Y) + H(X/Y)$$

Since $$\sum_{j=1}^{m} P(x_j) = 1$$

Hence $$H(X,Y) = H(X/Y) + H(Y)$$

UNIT-IV

Q. 8. Using simple ray theory, describe the mechanism for transmission of light within an optical fiber. Briefly discuss with the help of suitable diagram, what is meant by acceptance angle for an optical fiber. Derive it showing how it is related to the fiber numerical aperture and the refractive indices for the fiber core and cladding. (12.5)

Ans. Refer Q.8. of End Term Examination Dec- 2015

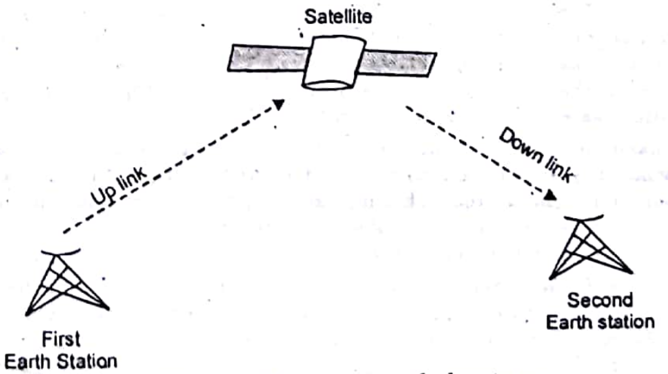
Q. 9 (a) The average optical power launched into a 20 km length of the fiber is 700 µω and the average output power is 5 µW. Calculate (i) signal attenuation per km (ii) overall signal attenuation for 12 km optical link using the same fiber wit 5 splices having attenuation of 0.9 dB. (6)

Ans. Refer to Q. 9. (a) of End Term Examination Dec- 2015

Q.9. (b) Why is there a need for satellite communication? List the various applications of satellite. (6.5)

Ans. Satellites are able to fulfil a number of roles. One of the major roles is for satellite communications. Here the satellite enables communications to be established over large distances - well beyond the line of sight. Communications satellites may be used for many applications including relaying telephone calls, providing communications to remote areas of the Earth, providing satellite communications to ships, aircraft and other mobile vehicles, and there are many more ways in which communications satellites can be used.

Need of satellite Communications: Satellites are able to provide communications in many instances where other forms of communications technology may not provide a feasible alternative.



Communications satellites provide a number of advantages

1. Flexibility: Satellite systems are able to provide communications in a variety of ways without the need to install nw fixed assets.

2. Mobility: Satellite communications are able to reach all areas of the globe dependent upon the type of satellite system in use, and the ground stations do not need to be in any one given location. For this reason, many ships use satellite communications.

3. Speedy deployment: Deployment of a satellite communications system can be very speedy. No ground infrastructure may be required as terrestrial lines, or wireless base stations are not needed. Therefore many remote areas, satellite communications systems provide an ideal solution.

4. Provides coverage over the globe: Dependent upon the type of satellite communications system, and the orbits used, it is possible to provide complete global coverage. As a result, satellite communications systems are sued for providing communications capabilities in many remote areas where other technologies would not be viable..

When considering the use of a satellite some disadvantages also need to be taken into consideration.

1. Cost: Satellites are not cheap to build, place in orbit and then maintain. This means that the operational costs are high, and therefore the cost of renting or buying space on the satellite will also not be cheap.

2. Propagation delay: As distances are very much greater than those involved with terrestrial systems, propagation delay can be an issue, especially for satellites using geostationary orbits. Here the round trip from the ground to the satellite and back can be of the order of a quarter of a second.

3. Specialised satellite terminals required: Even though the operator will operate all the required infrastructure, the user will still need a specialised terminal that will communicate with the satellite. This is likely to be reasonably costly, and it will only be able to be used with one provider.

Communications satellite applications

There are many different ways in which communications satellites can be used:

1. Telecommunications: Satellite systems have been able to provide data communications links over large distances. They were often used in place of intercontinental submarine cables which were expensive and unreliable in their early days. Nowadays cable technology has significantly improved to provide much higher levels of capacity especially as a result of fibre optic technology and their reliability has also greatly improved. As a result satellites are less frequently used to replace terrestrial cables, although in some instances this remains the case.

2. Satellite phones: The concept of using a mobile phone from anywhere on the globe is one that has many applications. Although the terrestrial cellular network is widely available, there are still very many areas where coverage is not available. In these situations satellite phones are of great use.

As an example satellite phones are widely used by the emergency services for situations when they are in remote areas, even of countries that might have a good cellular network, but not in remote areas. They may also be for communications in rural areas where no cellular coverage may be available. They also find uses at sea, and in developing countries, or in uninhabited areas of the globe.

Direct broadcast : While terrestrial broadcasting is well established it has a number of limitations: namely the coverage, especially in hilly areas where the hills may shade the signals from receivers, and also the bandwidth which is prime spectrum in the lower end of the UHF portion of the spectrum.

Direct broadcast satellite, DBS, technology enables both these issues to be overcome. The high angle of the satellites means that for most latitudes a high angle of signal direction means that hills do not provide a major coverage issue. Also operating around 12 GHz, more bandwidth is generally available enabling more stations - both television and radio - to be accommodated.