CBCS SCHEME

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18EC53

Fifth Semester B.E. Degree Examination, Jan./Feb. 2021
Principles of Communication System

Time: 3 hrs. Max. Marks: 100

Note: Answer any FIVE full questions, choosing ONE full question from each module.

Module-1

1 a. Explain in detail the working of switching modulator with diagram and necessary derivations. **(10 Marks)**

b. Explain the generation of DSBSC modulated waves using ring modulator. **(10 Marks)**

OR

2 a. Illustrate the amplitude modulation process and draw the waveform for modulation index $M > 1$ & $M < 1$. **(08 Marks)**

b. Explain with relevant block diagram and working of FDM system. **(08 Marks)**

c. A 400 W carrier is modulated on a depth of 75 percent. Calculate the total power in the modulated wave of following form AM.

(i) Double Side Band with Full Carrier (DSBFC)

(ii) Double Side Band Suppressed Carrier (DSBSC) **(04 Marks)**

Module-2

3 a. Derive the equations for frequency modulated wave. Define modulation index and frequency deviation. **(12 Marks)**

b. A 93.2 MHz carrier is frequency modulated by 5 kHz sine wave the resultant FM signal has frequency deviation of 40 kHz:

(i) Find the carrier swing of FM signal

(ii) What are highest and lowest frequencies of FM signal?

(iii) Calculate the modulation index of FM

(iv) B.W of FM signal **(08 Marks)**

OR

4 a. Explain the Narrow band FM with relevant expressions and phasor diagrams. **(10 Marks)**

b. Discuss the nonlinear effects in FM system. **(06 Marks)**

c. Assume that the maximum value of frequency deviation Δf is fixed at 50 kHz for a certain FM transmission. Given that the maximum modulating frequency is 15 kHz. Calculate the necessary transmission bandwidth. **(04 Marks)**

Module-3

5 a. Derive the expression for figure of merit for DSB-SC receiver. **(10 Marks)**

b. Find figure of merit for single tone FM. **(06 Marks)**

c. Write short notes on:

(i) Shot Noise

(ii) White Noise **(04 Marks)**

OR

6 a. With FM receiver model, derive the expression for figure of merit. (10 Marks)

b. Briefly explain the following as application to FM:

(i) Pre-emphasis

(ii) De-emphasis (06 Marks)

c. An AM receiver operating with a sinusoidal modulating signal has a following specifications: m = 0.8 and $(SNR)_0 = 30$ dB. What is carrier to noise ratio? (04 Marks)

Module-4

7 a. State sampling theorem and explain same with neat sketches and equation. (10 Marks)

b. With neat block diagram, explain the TDM. (06 Marks)

c. A Compact Disc (CD) audio signals digitally using PCM. Assume the audio signal bandwidth to be 20 kHz.

(i) What is the Nyquist rate?

(ii) If the Nyquist samples are quantized to L = 65, 536 levels and then binary coded, determine the number of bits required to encode a sample. (04 Marks)

OR

8 a. What are advantages digitizing the analog signals? (06 Marks)

b. With a block diagram, explain the generation and detection of PPM. (10 Marks)

c. Discuss Bandwidth – Noise trade off. (04 Marks)

Module-5

9 a. With a neat diagram, explain the basic elements of a PCM. (08 Marks)

b. Discuss the concept and operation of delta modulation in detail. (08 Marks)

c. PCM system uses uniform quantizer followed by a 7 bit binary encoder. The bit rate of the system is 50×10^6 bps. What is minimum message bandwidth? (04 Marks)

OR

10 a. Write a note on MPEG + Video. (10 Marks)

b. Draw the resulting waveform for 01101001 using unipolar NRZ, polar NRZ, unipolar Z2, Bipolar RZ. (06 Marks)

c. A TV signal with a bandwidth of 4.2 MHz is transmitted using binary PCM. The number of representation level is 512. Calculate:

(i) Codeword length

(ii) Final bit rate

(iii) Transmission bandwidth (04 Marks)

* * * * *

MODULE 1

1 a) Switching Modulator

1.3 : SWITCHING MODULATOR :-

It is a Diode circuit used to generate AM-Signal.

Q) Explain the operation of switching Modulator with circuit diagram and waveforms. Dec/Jan 2016/2017

↳ Switching Modulator is used to generate AM signal.

Circuit diagram :-

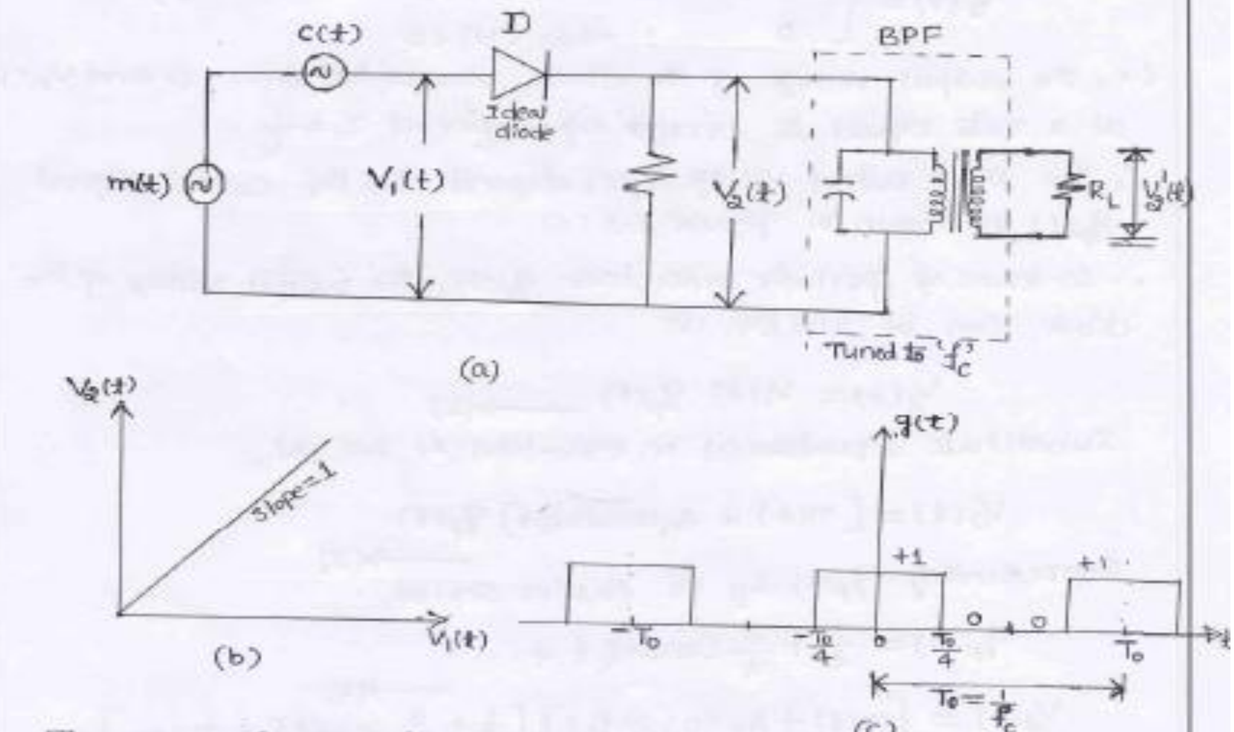


Figure 1.3 : (a) Switching Modulator circuit diagram
(b) Idealized input ($v_1(t)$) and output $V_2(t)$ relation of Diode
(c) periodic pulse-train of c(t).

Explanation :- Switching modulator consists of an Ideal diode which is used as a switch, followed by Band pass Filter (BPF) tuned to frequency 'f_c' as shown in figure 1.3 (a).

↳ Message signal m(t) and Carrier signal c(t) are simultaneously applied as input signal for ideal diode 'D', as shown in figure 1.3(a).

∴ The total input '$V_1(t)$' to the diode is given by

$$V_1(t) = m(t) + c(t)$$

$$\therefore \boxed{V_1(t) = m(t) + A_c \cos 2\pi f_c t} \longrightarrow (1)$$

It is assumed that $|m(t)| \ll A_c$. Therefore ON & OFF of Diode 'D' is Controlled by $c(t)$.

∴ The output voltage of Diode 'D' is,

$$V_2(t) = \begin{cases} V_1(t) & ; \text{ when } c(t) > 0 \Rightarrow \text{shown in figure 1.3(b)} \\ 0 & ; \text{ when } c(t) < 0 \end{cases}$$

i.e., the output voltage of the diode varies between 0 and $V_1(t)$ at a rate equal to carrie signal period $T_0 = \frac{1}{f_c}$.

∴ The Diode output Voltage $V_2(t)$ depends on the control signal $g_p(t)$ as shown in figure 1.3 (c).

∴ In terms of periodic pulse train $g_p(t)$, the output voltage of the diode can be written as

$$V_2(t) = V_1(t) \cdot g_p(t) \longrightarrow (2)$$

Substitute equation (1) in equation (2) we get,

$$V_2(t) = [m(t) + A_c \cos 2\pi f_c t]\, g_p(t) \longrightarrow (3)$$

Representing $g_p(t)$ by its Fourier Series,

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \cos 2\pi f_c t + \cdots \longrightarrow (4)$$

$$\therefore V_2(t) = [m(t) + A_c \cos 2\pi f_c t]\left[\frac{1}{2} + \frac{2}{\pi} \cos 2\pi f_c t + \cdots\right]$$

$$V_2(t) = \frac{1}{2} m(t) + \frac{2}{\pi} m(t) \cos 2\pi f_c t + \frac{A_c}{2} \cos 2\pi f_c t + \frac{2A_c}{4}\left[\underbrace{\cos^2 2\pi f_c t}_{\left(\frac{1+\cos 2\theta}{2}\right)}\right] + \cdots$$

$$V_2(t) = \frac{1}{2} m(t) + \frac{2}{\pi} m(t) \cos 2\pi f_c t + \frac{A_c}{2} \cos 2\pi f_c t + \underbrace{\frac{A_c}{4}}_{(DC)} + \frac{A_c}{4} \cos 4\pi f_c t + \cdots \longrightarrow (5)$$

The required AM wave Centered at 'f_c' is obtained by passing $V_2(t)$ through an ideal BPF having Center frequency 'f_c' and $BW = 2f_m$ Hz

∴ The output of the BPF is

$$V_2'(t) = \frac{2}{\pi} m(t) \cos(2\pi f_c t) + \frac{A_c}{2} \cos 2\pi f_c t$$

$$V_2'(t) = \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c} \cdot m(t)\right] \cos 2\pi f_c t$$

$$V_2'(t) = \frac{A_c}{2}\left[1 + k_a m(t)\right] \cos 2\pi f_c t \quad \Leftarrow \text{AM-Wave} \longrightarrow (6)$$

where $k_a = \frac{4}{\pi A_c}$ = Amplitude Sensitivity parameter

Equation (6) is the standard AM signal produced by the switching modulator with carrier amplitude scaled down to '$\frac{A_c}{2}$'

* * * * *

1 b) Ring Modulator

1.6. RING MODULATOR ***

Q) Explain the generation of DSBSC wave using Ring Modulator and also sketch the necessary waveforms.

↳ Ring Modulator is a product modulator used for generating DSBSC-Modulated signal.

Circuit diagram:

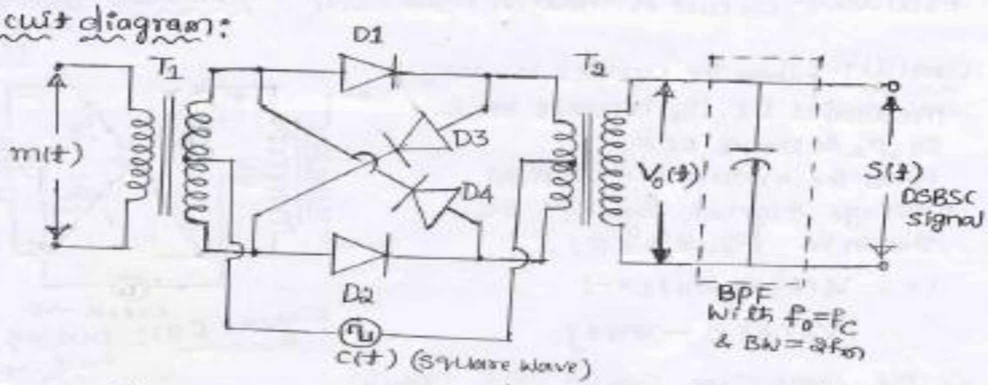


Figure 1.6(a): Circuit diagram of Ring Modulator

↳ The circuit diagram of Ring modulator is shown in figure 1.6(a) consists of two center-tapped transforms T1, T2 and four diodes D1, D2, D3 and D4 connected in bridge circuit and a BPF with center frequency 'f_c', BW = $2f_m$.

↳ the Carrier signal is applied to the Center taps of the input (T_1) and output (T_2) transformers. Modulating signal is applied to the input transformer T_1.

↳ The output voltage appears across the secondary of the transformer, T_2 (After passing through BPF).

↳ The Diodes Connected in the bridge Circuit (Ring) acts like switches and their switching is Controlled by carrier signal (square wave).

Circuit operation :-

Case(i) : When the Carrier is +ve, the Diodes D1, D2 becomes ON & Diodes D3, D4 becomes OFF.
Hence the Modulator Multiplies message signal m(t) by +1.

i.e., $V_0(t) = m(t) \times (+1) = m(t)$

Equivalent circuit is shown in Figure 1.6(b)

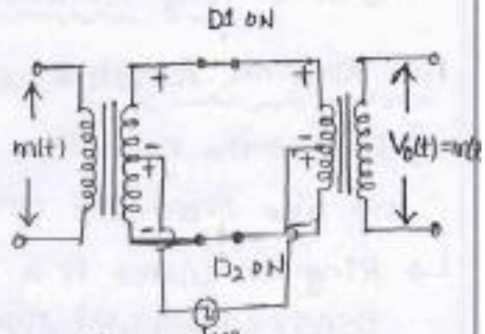


Figure 1.6(b): During +ve half cycle of c(t)

Case(ii) :- When the Carrier is −ve, the Diodes D3, D4 becomes ON & D1, D2 becomes OFF.
Hence the modulator multiplies message signal by "−1" as shown in figure 1.6(c).

i.e., $V_0(t) = m(t) \times -1$

$V_0(t) = -m(t)$

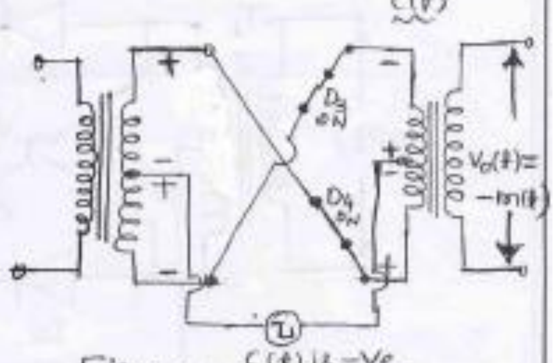


Figure 1.6 (c): During −ve half cycle of c(t)

∴ By combining Case(i) and Case(ii)
The Ring Modulator output at the Secondary of transformer T_2 is given by

$$V_0(t) = m(t) \times c(t) \quad \text{—(1)}$$

The square wave Carrier $C(t)$ can be represented by a fourier Series as:

$$C(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c t(2n-1)]$$

$$\therefore C(t) = \frac{4}{\pi}\left[\cos 2\pi f_c t - \frac{1}{3}\cos 6\pi f_c t + \cdots\right] \quad \text{---(2)}$$

$\therefore$ Substitute equation(2) in $V_0(t)$ equation(1) we get

$$V_0(t) = m(t) \times \frac{4}{\pi}\left[\cos 2\pi f_c t - \frac{1}{3}\cos 6\pi f_c t + \cdots\right] \quad \text{---(3)}$$

When $V_0(t)$ is passed through BPF having Center frequency 'f_c' and Bandwidth '$2f_m$' we get DSBSC signal,

$$\boxed{S(t) = \frac{4}{\pi} m(t) \cos 2\pi f_c t} \Leftarrow \text{DSBSC Wave generated from RING Modulator}$$

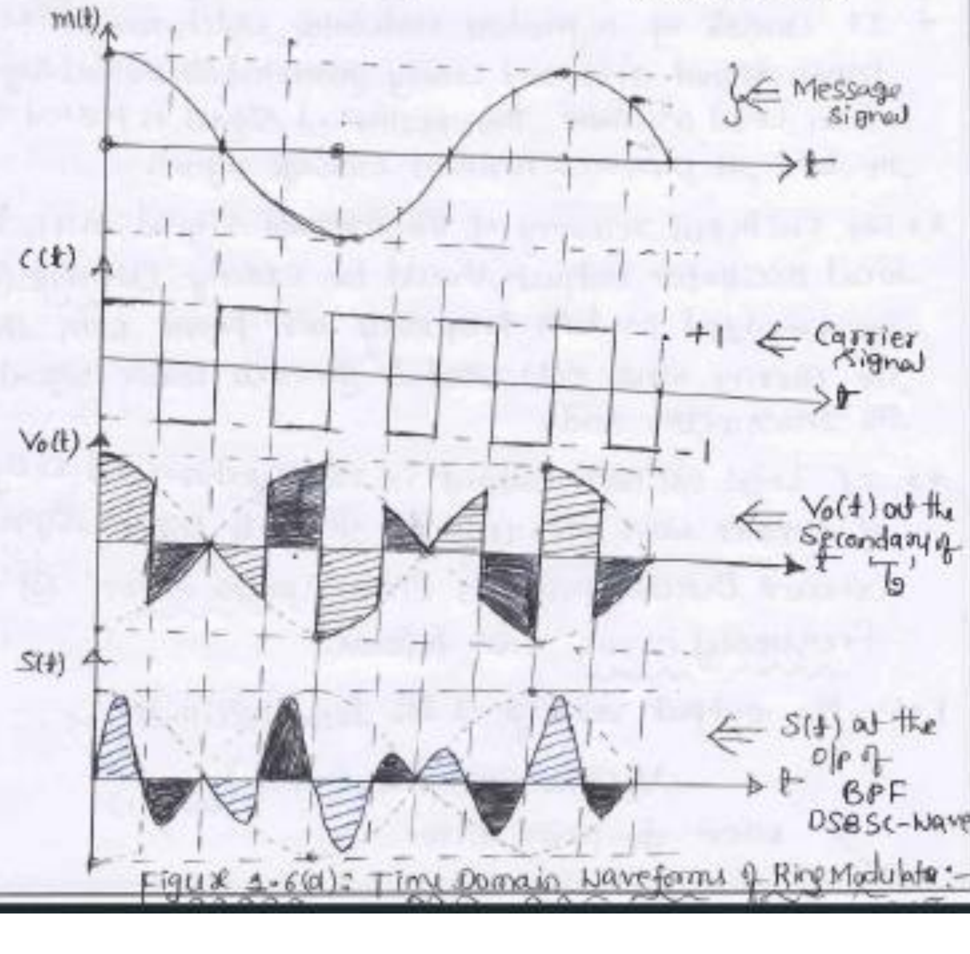


Figure 1.6(d): Time Domain Waveforms of Ring Modulator:-

Time and Frequency domain description of AM-signal:

Q) Define Amplitude Modulation. Obtain the expression for AM by both time domain and frequency domain representation with necessary waveforms.

VTU Sep

↳ Amplitude Modulation:-

Defn:- It is a process of altering the amplitude of carrier signal in accordance with the instantaneous values of message signal by keeping frequency and phase of carrier signal constant.

Expression for AM signal:-

- The instantaneous value of message signal is given by,

$$m(t) = A_m \cos(2\pi f_m t) \quad \text{—— (1)}$$

Where, $A_m \Rightarrow$ Amplitude of message signal.

$f_m \Rightarrow$ frequency (or) Bandwidth of message signal.

- The instantaneous value of carrier signal is given by,

$$c(t) = A_c \cos(2\pi f_c t) \quad \text{—— (2)}$$

Where, $A_c \Rightarrow$ Amplitude of carrier signal.

$f_c \Rightarrow$ frequency of carrier signal.

- We know that the standard equation of AM signal is given by,

$$s(t) = A_c \left[1 + k_a m(t)\right] \cos(2\pi f_c t) \quad \text{—— (3)}$$

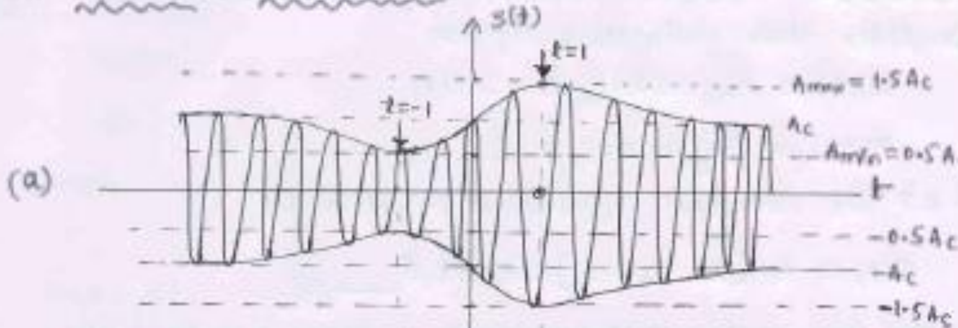
where, K_a = Amplitude sensitivity parameter.

Substitute $m(t) = A_m \cos 2\pi f_m t$ in equation (3)

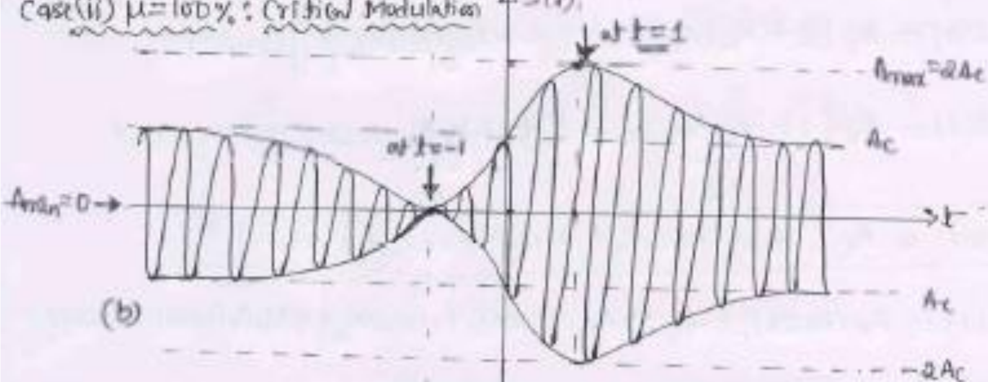
$$s(t) = A_c \left[1 + \underbrace{K_a \cdot A_m}_{} \cos(2\pi f_m t)\right] \cos(2\pi f_c t)$$

$$\therefore \; s(t) = A_c \left[1 + \mu \cos(2\pi f_m t)\right] \cos(2\pi f_c t) \quad \text{—— (4)}$$

Case(i) : $\mu = 50\%$: Under Modulation :-



Case(ii) $\mu = 100\%$: Critical Modulation



Case(iii) : $\mu = 125\%$: Over Modulation :-

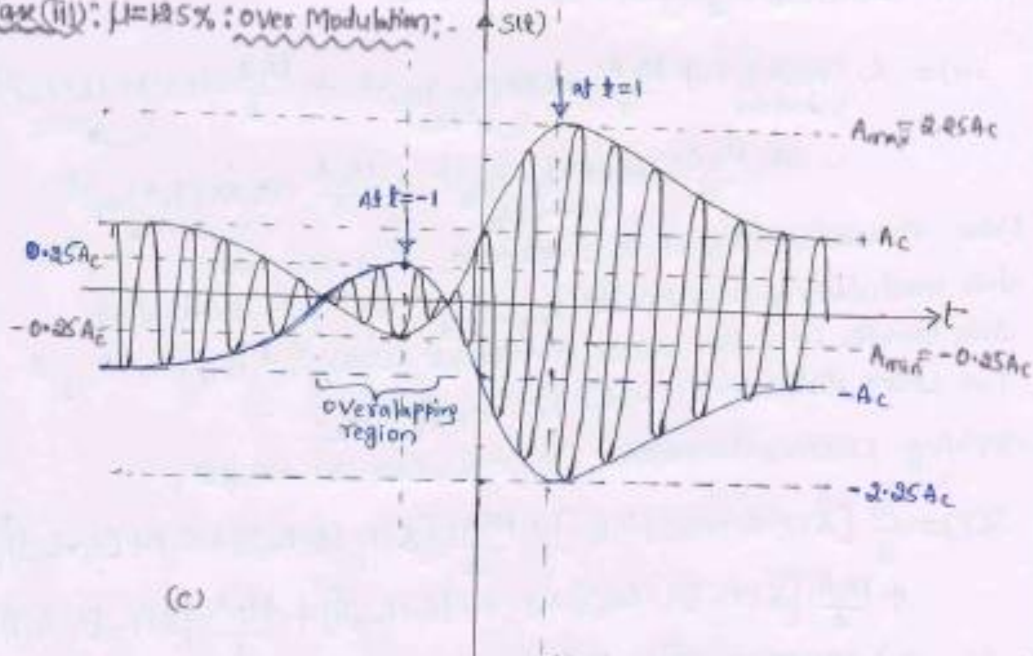


Figure 1.2: AM signal for $m(t) = \frac{t}{1+t^2}$ for (a) $\mu = 50\%$ (b) $\mu = 100\%$ and (c) $\mu = 125\%$

* Frequency Division Multiplexing <FDM>

↳ Multiplexing is a process of combining N-independent message signals into a composite signal suitable for transmission over a common channel

↳ Multiplexing is accomplished by seperating the signals either in frequency (or) time.

↳ The technique of separating the signals in frequency domain is referred to as "Frequency Division Multiplexing" (FDM)

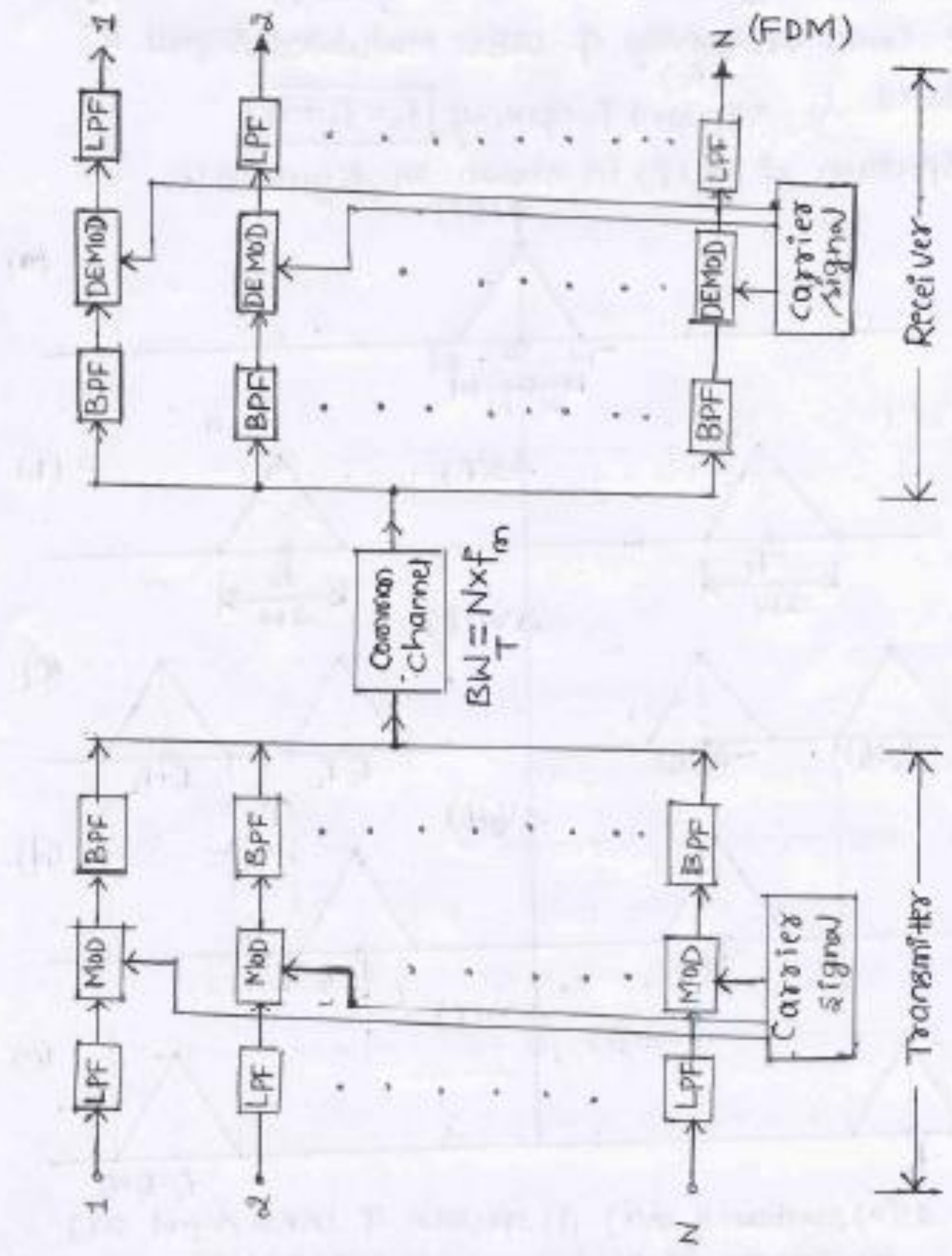


Figure 1: Block diagram of FDM system

The block diagram of FDM-system is shown in figure1.

↳ N-Incoming independent message signals are modulated by mutually Exclusive carriers supplied from carrier source at each modulator. The Modulated signals are passed through the BPF to select any one side band. Therefore BPF's produces SSB-signals and are separated in frequency and combined into a composite signal. and this process is called Frequency division multiplexing.

↳ Multiplexed signal is transmitted over the communication channel.

↳ Total Bandwidth required to N-SSB Modulated signals without any Guard band is

$$BW_T = N \times f_m \;; \; N = \text{number of Input signals}$$

↳ At the receiver side N-independent message signals are recovered by passing the composite signal through the BPF followed by Demodulator and LPF.

Advantages of FDM:-

1. A Large Number of signals can be transmitted simultaneously.
2. FDM does not requires synchronization between Transmitter & receiver.
3. Demodulation of FDM is easy

Dis advantages of FDM:-

1. Communication channel must have Large Bandwidth i.e., $BW_T = N \times f_m$
2. Large Numbers of Modulators & Filters are required.
3. Cross talk occurs in FDM

3 a) FM

→ Frequency Modulation is a process of altering the frequency of Carrier signal in accordance with the instantaneous values of message signal by keeping amplitude & phase of Carrier Constant.

Time domain expression:-

- Let the instantaneous value of Carrier signal is

$$c(t) = A_c \cos 2\pi f_c t \longrightarrow (1)$$

- Let the instantaneous value of message signal is

$$m(t) = A_m \cos 2\pi f_c t \longrightarrow (2)$$

- We know that the standard equation of Angle modulated wave is given by,

$$S(t) = A_c \cos \theta_i(t) \longrightarrow (3)$$

where $\theta_i(t)$ = Angle of FM wave (modulated wave)

- We know that the instantaneous frequency $f_i(t)$ of FM signal is given by

$$f_i(t) = f_c + K_f m(t) \quad \text{—— (4)}$$

where, K_f = frequency sensitivity

$m(t)$ = message signal

- We know that the Angular frequency,

$$w_i(t) = \frac{d}{dt}\theta_i(t)$$

$$\Downarrow$$

$$2\pi f_i(t) = \frac{d}{dt}\theta_i(t)$$

$$\therefore f_i(t) = \frac{1}{2\pi}\frac{d}{dt}\theta_i(t) \quad \text{—— (5)}$$

Substitute $f_i(t) = f_c + K_f m(t)$ in equation (5) we get,

$\therefore \quad f_c + K_f m(t) = \frac{1}{2\pi} \frac{d}{dt} \theta_i(t)$

$\therefore \quad \frac{d}{dt} \theta_i(t) = 2\pi f_c + 2\pi K_f m(t) \quad — (6)$

⇒ Apply Integral on both sides of equation (6) we get

$$\int \frac{d}{dt} \theta_i(t) = \int [2\pi f_c + 2\pi K_f m(t)]\, dt$$

$$\Downarrow$$

$$\theta_i(t) = 2\pi f_c t + 2\pi K_f \int m(t)\, dt \quad — (7)$$

$\therefore$ The General equation of FM signal is

$S(t) = A_c \cos \theta_i(t)$ — using equation (7)

$$\boxed{S(t) = A_c \cos[2\pi f_c t + 2\pi K_f \int m(t)\, dt]} \quad — (8)$$

Equation (8) is the general equation of FM signal for any message signal $m(t)$.

for, $m(t) = A_m \cos 2\pi f_m t$

$$\int m(t)\, dt = \int A_m \cos 2\pi f_m t\, dt \qquad \left(\because \int \cos mx\, dx = \frac{\sin mx}{m}\right)$$

$$= \frac{A_m}{2\pi f_m} \cdot \sin 2\pi f_m(t) \quad — (9)$$

$\therefore$

$$S(t) = A_c \cos\left[2\pi f_c t + 2\pi K_f \times \frac{A_m}{2\pi f_m} \cdot \sin(2\pi f_m t)\right]$$

$$= A_c \cos\left[2\pi f_c t + \frac{K_f A_m}{f_m} \sin 2\pi f_m t\right]$$

$$\boxed{S(t) = A_c \cos[2\pi f_c t + \beta \sin 2\pi f_m t]} \quad — (10)$$

Equation (10) is the standard equation of FM signal for $m(t) = A_m \cos 2\pi f_m t$; where $\beta = \frac{K_f A_m}{f_m} = \frac{\Delta f_{max}}{f_m} \Leftarrow$ Modulation Index of FM signal.

3 b)fc= 93.2MHz, fm=5kHz, deviation =40kHz

1. Carrier Swing = 2* deviation =80k
2. Higher freq = fc+ deviation= 93.24Mhz, Lower =93.16MHz
3. Modulation Index = Deviation/fm = 40k/5k= 8
4. BT = 2(deviation+fm) =90kHz

4 a) Narrow Band FM

↳ Narrow band FM signals are characterized by modulation index, β less than 1.

↳ Narrow band FM signal equation can be derived from general FM equation for $m(t) = A_m \cos 2\pi f_m(t)$, as follows

$$S(t) = A_C \cos[2\pi f_c t + \beta \sin(2\pi f_m t)] \quad \text{——— (1)}$$

equation (1) is general FM equation for $m(t) = A_m \cos(2\pi f_m t)$ obtained in section 1-2.

W.K.T $\cos(A+B) = \cos A . \cos B - \sin A . \sin B$

$$(1) \overset{\therefore}{\Rightarrow} \quad S(t) = A_C \cos 2\pi f_c t \cos(\beta \sin 2\pi f_m t) - A_c \sin(2\pi f_c t) \times \sin(\beta \sin 2\pi f_m t) \quad \longrightarrow (2)$$

For Narrow band FM-signals, $\beta < 1$

$\therefore$ The value of $\beta \sin 2\pi f_m t$ becomes less than 1-degree. and It approaches almost $0°$. Therefore

$$\cos(\beta \sin 2\pi f_m t) \simeq 1 \quad \text{——— (3)} \quad \left(\because \lim_{\theta \to 0°} \cos\theta \simeq 1\right)$$

$$\sin(\beta \sin 2\pi f_m t) \simeq \beta \sin 2\pi f_m t \quad \text{——— (4)} \quad \left(\because \lim_{\theta \to 0°} \sin\theta = \theta\right)$$

By substituting equations (3) & (4) in equation (2) we get Narrow band FM signal

$$\boxed{S(t) = A_C \cos 2\pi f_c t - A_C \beta \sin 2\pi f_c t \cdot \sin 2\pi f_m t} \quad \text{——— (5)}$$

$\therefore$ Narrow band FM signal consists of 3- frequency components

↳ $f_c \Rightarrow$ Carrier signal

$f_c - f_m \Rightarrow$ Lower side band

$f_c + f_m \Rightarrow$ Upper side band

} Same as that of Standard AM signal.

$\therefore$ Total transmission Bandwidth of Narrow band FM $\Rightarrow \boxed{BW_T = 2f_m}$

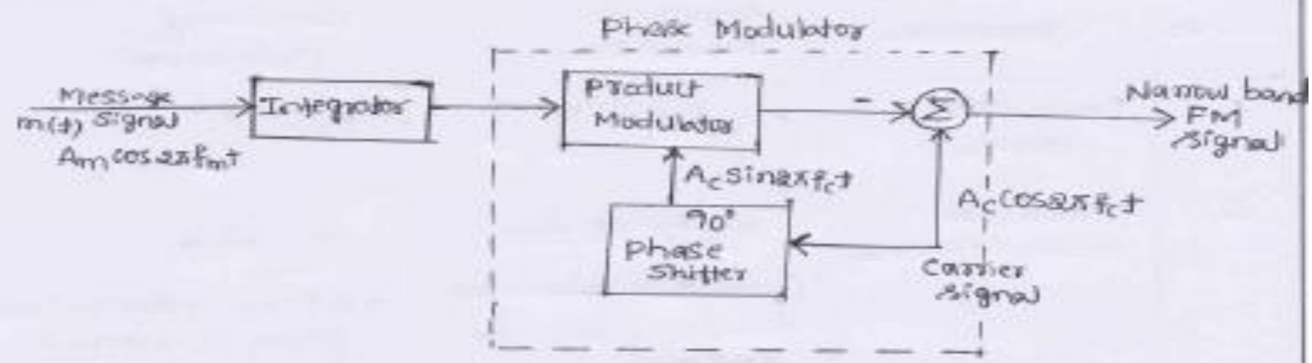


Figure 1: Indirect Method of Generating FM signal using (Narrow band) Phase modulator :-

Figure 1. shows the indirect method of generating Narrow band FM signal shown in figure ⑤, using phase modulator.

Note: The Bandwidth required to transmit Narrow band FM signal is same as that of AM-signal transmission channel bandwidth "$2f_m$".

4 b) Nonlinear effects in FM

↳ Non-linear effects can be of two-types
(i) strong (ii) weak.

* Non-linearity is said to be strong, if it is intentionally introduced into the circuit in a controlled manner
Ex: square law devices.

* Non linearity is said to be weak, when it is inherently present in the circuit.

The effect such non-linearities will limit m(t) levels in the system.

In FM-generation system, weak non-linearity is present. the effect of weak non-linearity in FM-systems can be by considering the input and output relation of the memoryless - non linear device used in the frequency multiplier.

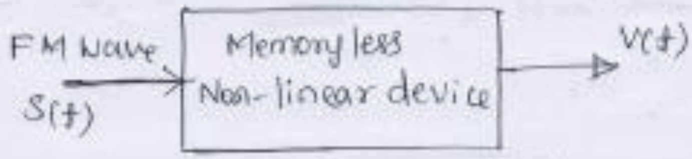


Fig 1: Non-linear device used in FM System:-

Consider a memoryless Non-linear device as shown in fig-1

W.K.T, the relation between input & output signal is

$$V(t) = a_1 S(t) + a_2 S^2(t) + a_3 S^3(t) + \cdots + a_n S^n(t) \quad \text{—} (1)$$

Let us consider upto 3rd order

i.e., $V(t) = a_1 S(t) + a_2 S^2(t) + a_3 S^3(t)$ — (2)

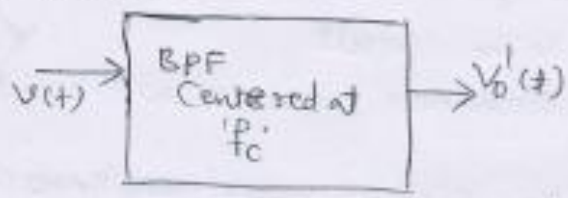
W.K.T the expression for FM-wave is

$$S(t) = A_c \cos[2\pi f_c t + \phi(t)] \quad \text{where } \phi(t) = 2\pi k_f \int_0^t m(t)\,dt \quad \text{—} (2)$$

$$\therefore V(t) = \underset{(1)}{a_1 A_c \cos[2\pi f_c t + \phi_1(t)]} + \underset{(2)}{a_2 A_c^2 \cos^2[2\pi f_c t + \phi_1(t)]} + \underset{(3)}{a_3 A_c^3 \cos^3[2\pi f_c t + \phi_1(t)]} \quad \text{—} (3)$$

$\therefore$ The the output voltage consists of DC components and three FM-signals with carrier frequencies f_c, $2f_c$ & $3f_c$ having frequency deviations Δf, $2\Delta f$ and $3\Delta f$ respectively.

The Desired FM signal can be separated using a BPF as shown in fig.2.



We get the FM-system o/p after passing through BPF is

$$V_0'(t) = a_1 A_c \cos[2\pi f_c t + \phi(t)] \quad \text{—} (4)$$

equation (4) is same as that of FM-input signal

$S(t) = A_c \cos[2\pi f_c t + \phi(t)]$ Except for change in amplitude.

$\therefore$ Amplitude Non-linearities of The FM-S/m does not affect FM-Signal

4 c) fm=15kHz, deviation =50kHz

BT = 2(deviation+fm) =130kHz

Module 3

5c) Noise

3.11: Shot Noise :-

(Q) Write a short note on shot Noise. 4-Marks.

→ Shot Noise appears in active devices due to random behaviour of charge carriers (electrons and holes).

Example:

- In Vaccum tubes, shot noise is generated due to random emission of electrons from Cathode-plate.
- In semiconductor devices, shot noise is generated due to random diffusion of minority charge carriers (or) random generation and recombination of electron-hole pairs.
- In Photo detectors/LED's, it is generated due to random emission of photons.

Consider a Vaccum diode shown in fig.1. Let 'I' be the current and shot noise component generated is $I_{SN}(t)$ as shown in fig.2.

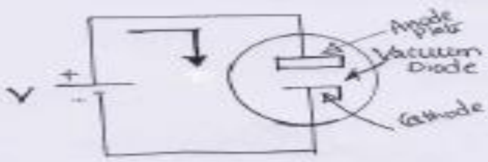


Fig.1: Vaccum diode

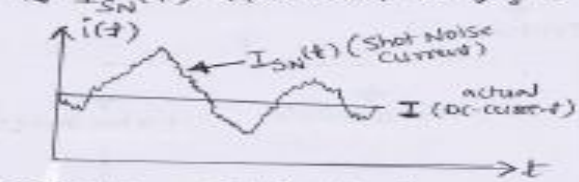


Fig2: Current Variation in Vaccum diode

With respect to Fig.2 the total current in vaccum diode is,

$$i(t) = I + I_{SN}(t)$$

; $I_{SN}(t) \Rightarrow$ shot noise current generated due to random Emission of electrons from Cathode.

→ ∴ The Mean square value of fluctuating shot noise current in Vaccum diode is given by

$$E[I_{SN}^2] = 2qIB_N$$

; where $q = 1.6 \times 10^{-19}$ Coulombs

B_N = Noise Equivalent Bandwidth

I = required DC current in Vaccum diode.

→ for PN junction diodes,

$$E[I_{SN}^2] = 2q(I+I_s)B_N$$

; I_s = Reverse Leakage Current of D.L

3.12: Thermal Noise :- (VTU Q.P)

a) Write a short note on Thermal Noise.

↳ Thermal Noise is generated due to random moment of thermally induced carriers (electrons) in a conductor.

↳ The random motion of thermally induced electrons produces electric current which is random in nature. This random current is called "thermal noise" (or) "Johnson Noise"

↳ Figure 1 shows noise-model using resistor and its equivalent thevenin's circuit in fig.2.

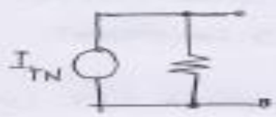


Fig 1: Noise model

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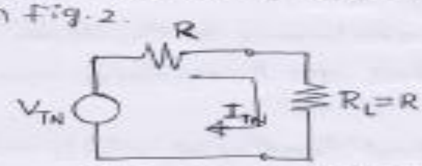


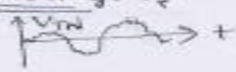
Fig2: Equivalent circuit (To find Max Noise Power) $R_L = R$

I_{TN} = Thermal Noise Current

$V_{TN} = I_{TN} \times R$ ↳ Thermal Noise voltage.

↳ The Mean value of thermal noise current is always zero.

↳ Mean square value of the thermal noise voltage is



$$E[V_{TN}^2] = 4KB_NTR$$

where K = Boltzmann Constant $= 1.38 \times 10^{-28}$ J/K

T = temperature at which resistor is operating $= 290$ K (standard)

B_N = Noise equivalent Bandwidth in Hz.

R = Resistor in Ω

↳ The Maximum noise power, produced across noisy resistor model shown in fig.2 is

$$P_N = \frac{E[V_{TN}^2]}{4R} = \frac{4KB_NTR}{4R} = \underline{KTB_N} \text{ Watts.}$$

Note: Max power delivered to load when $R_L = R$ &

$$P_{max} = \frac{V^2}{4R} \;//.$$

5 a) FOM DSBSC

Q) Show that Figure of merit for DSBSC System is unity.

June/July-2017

(8-Marks)

↳

Let m(t) be the message signal and 'P' be the average power in m(t).

c(t) be the carrier signal, then time domain expression for DSBSC-signal is given by the product of m(t) and c(t).

∴ DSBSC-modulated signal, S(t) is

$$S(t) = m(t) \cdot c(t)$$

$$S(t) = m(t) \cdot [A_c \cos(2\pi f_c t)] \longrightarrow (1)$$

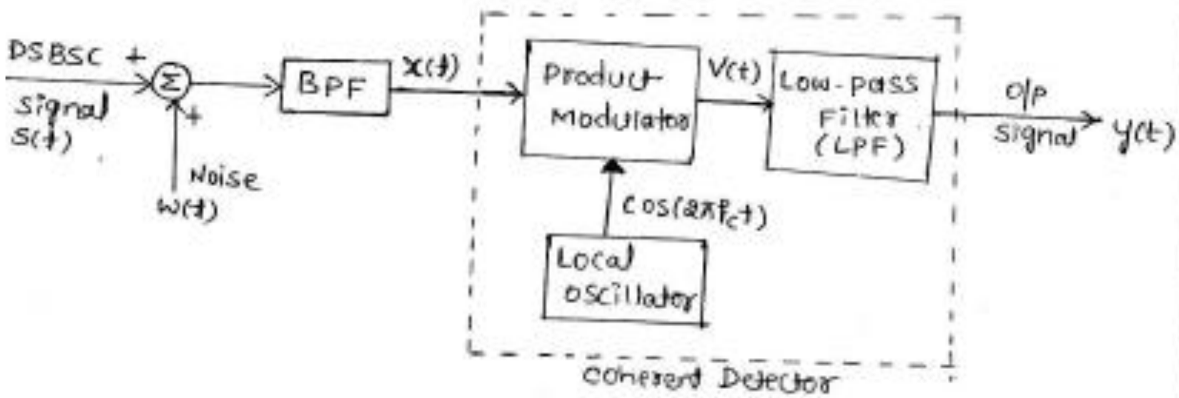


Figure 1: Model of DSBSC receiver using Coherent Detector

Figure 1 shows the model of a DSBSC receiver using a Coherent Detector.

In figure 1, the filtered signal $[x(t) = s(t) + n(t)]$ is applied to product modulator.

The product modulator multiplies the filtered signal $x(t)$ with locally generated carrier "$\cos 2\pi f_c t$" & produces the product signal

$V(t) = x(t) \cdot \cos(2\pi f_c t)$

$V(t)$ is applied to low-pass-filter it eliminates all higher frequency components & produces output signal $y(t) = m_d(t) + n_d(t)$.

To find channel SNR $(SNR)_c$:-

↳ The DSBSC signal is given by

$$s(t) = m(t) \times A_c \cos(2\pi f_c t) \quad \text{——} (*)$$

• Therefore, the average power of the modulated signal $s(t)$ is

$$E\left[(s(t))^2\right] = E\left[(m(t))^2 \cdot (A_c \cos 2\pi f_c t)^2\right] = \frac{A_c^2}{2} \cdot P$$

Where P_m = Average power of message signal $= E\left[(m(t))^2\right]$.

• Average power of the noise in message bandwidth is given by "$N_0 W$". where W = Bandwidth of message signal, $m(t)$.

∴ channel signal to noise ratio is

$$(SNR)_C = \frac{\text{Average power of the modulated signal}, s(t)}{\text{Average power of the noise in message bandwidth}}$$

$$(SNR)_C = \frac{A_c^2 P}{2 N_0 W} \quad \text{——} (A)$$

To find output SNR $(SNR)_0$:-

Total signal at the input of Coherent detector is

$$x(t) = s(t) + n(t) \longrightarrow (2)$$

We know that the narrow band noise signal $n(t)$ in its Canonical form is represented by

$$n(t) = n_I(t) \cdot \cos(2\pi f_c t) - n_Q(t) \sin(2\pi f_c t) \longrightarrow (3)$$

Where $n_I(t)$ = Inphase noise Component and $n_Q(t)$ = Quadrature phase noise Component, measured with respect to Carrier signal $\cos(2\pi f_c t)$.

Substitute equation (3) in equation (2) we get

$$x(t) = s(t) + n_I(t) \cdot \cos(2\pi f_c t) - n_Q(t) \cdot \sin(2\pi f_c t) \longrightarrow (4)$$

Therefore, the output of product Modulator is given by,

$$v(t) = x(t) \times \cos(2\pi f_c t) \quad \text{——} (5)$$

Substitute $x(t)$ equation (4) in (5) we get

$$V(t) = \left[s(t) + n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)\right]\cos(2\pi f_c t)$$

$$V(t) = s(t).\cos(2\pi f_c t) + n_I(t)\cos^2(2\pi f_c t) - n_Q(t).\sin(2\pi f_c t).\cos(2\pi f_c t)$$

DSBSC signal, $s(t) = m(t).A_c\cos(2\pi f_c t)$, Therefore.

$$V(t) = A_c.m(t).\cos^2(2\pi f_c t) + n_I(t)\cos^2(2\pi f_c t) - n_Q(t).\sin(2\pi f_c t).\cos(2\pi f_c t) \longrightarrow (6)$$

Using Trignometric Identities

$$\cos^2\theta = \frac{1+\cos 2\theta}{2} \text{ and } \sin\theta.\cos\theta = \frac{\sin 2\theta}{2}$$

$$V(t) = \frac{A_c.m(t)}{2}\left(1+\cos(4\pi f_c t)\right) + \frac{n_I(t)}{2}\left(1+\cos(4\pi f_c t)\right) - \frac{n_Q(t)}{2}\sin(4\pi f_c t)$$

$$V(t) = \frac{A_c m(t)}{2} + \frac{A_c.m(t)}{2}.\cos(4\pi f_c t) + \frac{n_I(t)}{2} + \frac{n_I(t)}{2}.\cos(4\pi f_c t) - \frac{n_Q(t)}{2}\sin(4\pi f_c t) \longrightarrow (7)$$

The output of product Modulator $V(t)$, is applied to low pass filter it allows only $\frac{m(t).A_c}{2}$ & $\frac{n_I(t)}{2}$ components & eliminates all other higher frequency terms.

∴ The output signal of coherent detector is

$$y(t) = \underbrace{\frac{A_c m(t)}{2}}_{\substack{\text{demodulated}\\ \text{signal}}} + \underbrace{\frac{n_I(t)}{2}}_{\substack{\text{output}\\ \text{noise}}}$$

Therefore Average power of demodulated Output signal $E\left[\frac{A_c^2}{4}.m^2(t)\right]$ $\Big\} = \frac{A_c^2}{4}.P$

Average power of output Noise $E\left[\left(\frac{n_I(t)}{2}\right)^2\right]$ $\Big\} = \frac{N_0 W}{2}$ $\Leftarrow$ Half of input noise power.

∴ output signal to Noise ratio is

$$(SNR)_0 = \frac{\text{Average power of the demodulated signal}}{\text{Average power of output noise}}$$

$$(SNR)_0 = \frac{(A_c^2/4)P}{(N_0 W/2)} = \frac{A_c^2 P}{2N_0 W} \longrightarrow (B)$$

∴ Figure-of-Merit for DSBSC-receiver system is

$$\text{Figure of Merit} = \frac{(SNR)_0}{(SNR)_c} \longrightarrow (8)$$

Substitute equation (A) and equation (B) in equation(8) we get

$$FoM = \frac{(SNR)_0}{(SNR)_c} = \frac{\left(\frac{A_c^2 P}{2N_0 W}\right)}{\left(\frac{A_c^2 P}{2N_0 W}\right)} = \underline{1}$$

∴ Figure-of-Merit (FoM) for DSBSC receiver is Unity.

5 b)Single tone FM FOM

Note:

$$(\mathrm{SNR})_{O,\mathrm{FM}} = \frac{3A_c^2 k_f^2 P}{2N_0 W^3}$$

$$(\mathrm{SNR})_{C,\mathrm{FM}} = \frac{A_c^2}{2WN_0}$$

$$\left.\frac{(\mathrm{SNR})_O}{(\mathrm{SNR})_C}\right|_{\mathrm{FM}} = \frac{3k_f^2 P}{W^2}$$

For sinusoidal modulating signal, $P = A_m^2/2$ and $\Delta f = K_f A_m$

$K_f^2 P = K_f^2 A_m^2/2 = (\Delta f)^2/2$

Figure of merit $= 3*(\Delta f)^2/(2W^2) = 3*(\Delta f)^2/(2f_m^2)$

6a) FOM FM

Q) Prove that Figure-of merit for single tone Frequency modu-
-lated signal is $1.5\beta^2$.

→ The single-tone Frequency modulated wave $s(t)$ is given by,

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right) \longrightarrow (1)$$

where $m(t)$ = message signal.

Let $\phi(t) = 2\pi k_f \int_0^t m(t)\,dt$, then

$$s(t) = A_c \cos\left(2\pi f_c t + \phi(t)\right) \longrightarrow (2)$$

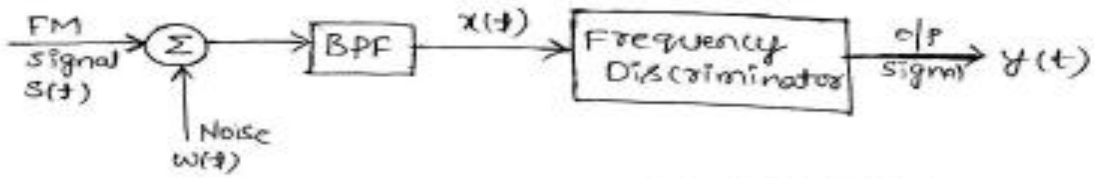


Fig1: Model of FM receiver using Frequency Discriminator

Fig1. shows the model of FM receiver using frequency discriminator.

To determine Channel SNR $(SNR)_c$:-

W.K.T the FM signal is

$$s(t) = A_c \cos\left(2\pi f_c t + \phi(t)\right) \text{ --- } (*)$$

∴ Average power of Modulated signal $s(t)$ $\Big\} = \dfrac{A_c^2}{2}$

Average power of noise in message band width is $\Big\} = N_0 \times W$

$$\therefore \quad (SNR)_C = \frac{A_c^2}{2 N_0 W} \longrightarrow (A)$$

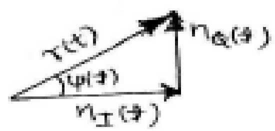
- To Determine output SNR $(SNR)_0$:-

The total signal at the input of frequency discriminator is,

$$x(t) = s(t) + n(t) \longrightarrow (3)$$

For output SNR, analysis let us express $n(t)$ in terms of its magnitude $[r(t)]$ and phase $[\psi(t)]$ given by the equation

$$n(t) = r(t)\cos(2\pi f_c t + \psi(t)) \longrightarrow (4)$$



where $r(t) = \sqrt{n_I^2(t) + n_Q^2(t)} \longrightarrow (5)$

$$\psi(t) = \tan^{-1}\left(\frac{n_Q(t)}{n_I(t)}\right) \longrightarrow (6)$$

∴ Total signal at the input of demodulator is

$x(t) = s(t) + n(t)$ becomes

$$x(t) = A_c\cos(2\pi f_c t + \phi(t)) + r(t)\cos(2\pi f_c t + \psi(t)) \longrightarrow (7)$$

The relative phase $\theta(t)$ can be expressed as

$$\theta(t) \simeq \phi(t) + \frac{n_Q(t)}{A_c} \longrightarrow (8) \quad ; \quad \text{where } n_Q(t) = r(t)\cdot\sin\psi(t)$$

With an ideal phase discriminator, the output $y(t)$ is proportional to the phase deviation $\frac{d}{dt}\theta(t)$.

i.e., the output signal,

$$y(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt} \longrightarrow (9)$$

Substitute $\theta(t)$ from equation (8) in equation (9) we get

$$y(t) = \frac{1}{2\pi}\frac{d}{dt}\left[\phi(t) + \frac{n_Q(t)}{A_c}\right] \quad \because \phi(t) = 2\pi K_f\int_0^t m(t)dt$$

$$= \frac{1}{2\pi}\frac{d}{dt}\left[2\pi K_f\int_0^t m(t)dt + \frac{n_Q(t)}{A_c}\right]$$

$$y(t) = \underbrace{K_f m(t)}_{\text{demodulated signal}} + \underbrace{\frac{1}{2\pi A_c}\frac{d}{dt}n_Q(t)}_{\text{O/P Noise}}$$

$\therefore$ Average power of demodulated O/P signal $\Big\} = K_f^2 P \longrightarrow (10)$

where P = power in message signal $m(t) = \frac{A_m^2}{2}$

$$\left.\begin{array}{l}\text{Average power of}\\ \text{output Noise}\end{array}\right\} = \frac{N_0}{A_c^2}\int_{-W}^{W} f^2 df = \frac{N_0}{A_c^2}\left(\frac{f^3}{3}\right)_{-W}^{W}$$

$$= \frac{2N_0W^3}{3A_c^2}$$

$\therefore$ output signal to noise ratio is

$$(SNR)_0 = \frac{K_f^2 P}{\left(\frac{2N_0W^3}{3A_c^2}\right)} = \frac{3A_c^2K_f^2P}{2N_0W^3} \longrightarrow (B)$$

$\therefore$ Figure-of-Merit for FM receiver is

$$FOM = \frac{(SNR)_0}{(SNR)_C} \longrightarrow (C)$$

Substitute equation (A) & equation (B) in equation (C) we get,

$$FOM = \frac{\left(\frac{3A_c^2K_f^2P}{2N_0W^3}\right)}{\left(\frac{A_c^2}{2N_0W}\right)} = \frac{3K_f^2P}{W^2} \longrightarrow (D)$$

$\therefore$ Substitute $P = \frac{A_m^2}{2}$; Average power of message signal in equation (D) we get

$$FOM = \frac{3 K_f^2 A_m^2}{2 W^2} = \frac{3}{2}\left(\frac{K_f A_m}{W}\right)^2 \longrightarrow (E)$$

We know that the modulation Index of FM-Signal

$$\beta = \frac{\Delta f}{f_m} = \frac{K_f A_m}{W}$$

∴ using the value of 'β' in FOM equation (E) we get Figure-of-Merit of FM receiver

$$\boxed{FOM = \frac{3}{2}\beta^2 = 1.5\,\beta^2} \quad \text{where } \beta = \frac{K_f A_m}{W}$$

6 b) Pre Emphasis, De-Emphasis

Pre-emphasis and De-emphasis in FM :-

Q> With circuits and characteristics, Explain the importance of pre-emphasis and De-emphasis in FM-systems.
VTU - 8M -

↳ pre-emphasis and De-emphasis methods are commonly used in FM-transmitter and FM-receiver respectively to improve the Threshold.

→ pre-emphasis and De-emphasis are simple RC networks used to improve threshold upto 13dB to 16dB.

→ Figure 1 shows the FM transmitter with pre-emphasis filter having transfer function $H_{pe}(f)$.

↳ Figure 1, shows the pre-emphasis filter used before FM-transmitter.

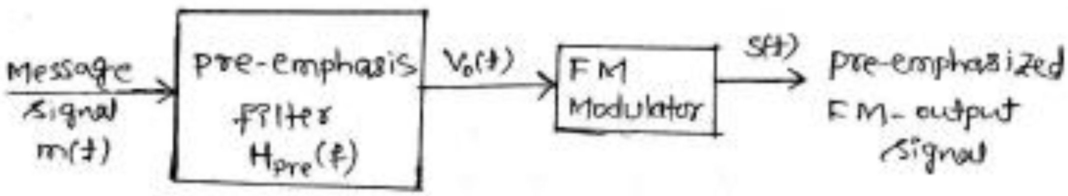


Fig 1: Use of pre-emphasis filter in FM transmitter

* pre-emphasis filter or circuit :-

↳ pre-emphasis circuit is a High-pass-Filter (HPF) with transfer function shown in figure 2.

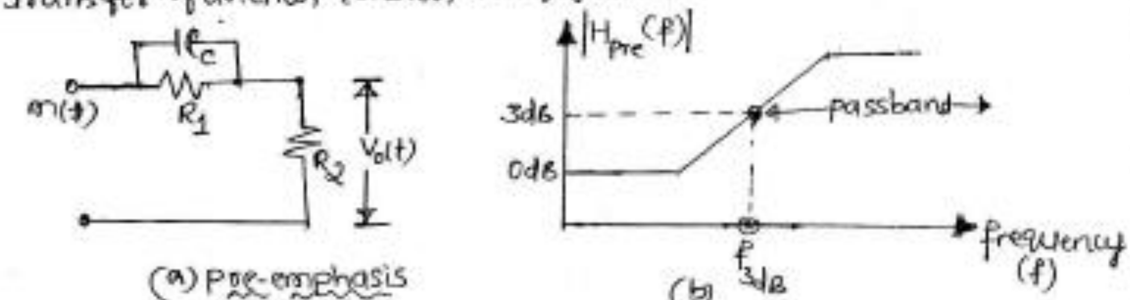


Fig.2: pre-emphasis filter circuit diagram with frequency response

↳ To improve $(SNR)_0$ at the FM-Modulator output, the high freque--ncy components of the message signal m(t) are artificially emphasized at the transmitter, before the Modulation-process & is shown in fig.1.

↳ After pre-emphasis, m(t) occupies entire range of bandwidth assigned. Then at the frequency discriminator (FM-demodulator) of FM receiver, inverse operation of pre-emphasis called De-emphasis is performed.

- Figure 3, shows the FM receiver with de-emphasis filter having frequency response $|H_{de}(f)|$.
- De-emphasis filter/circuit is used after FM-demodulator as shown in figure.3.

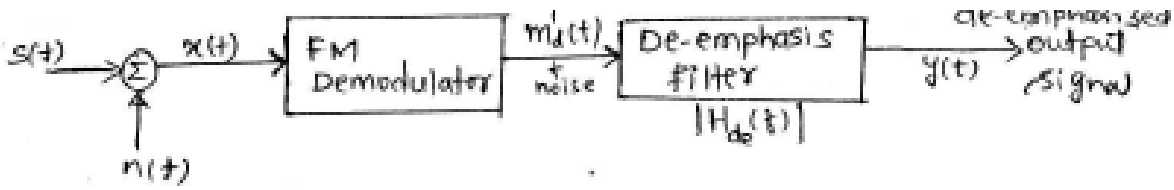
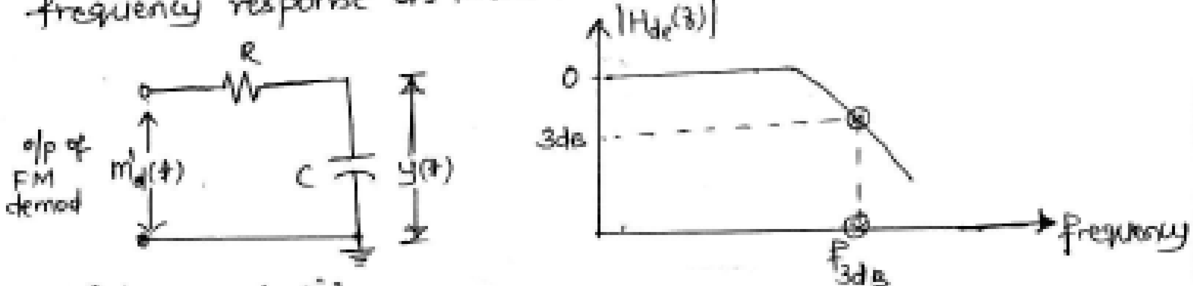


Figure 3: FM receiver with De-emphasis filter

De-emphasis filter (or) Circuit :-

↳ De-emphasis circuit is a simple RC-Lowpass filter with frequency response as shown in Figure 4.



(a) De-emphasis Circuit

(b) Frequency response

Fig. 4: De-emphasis filter circuit diagram & its frequency response

MODULE 4

7 a) Smapling

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

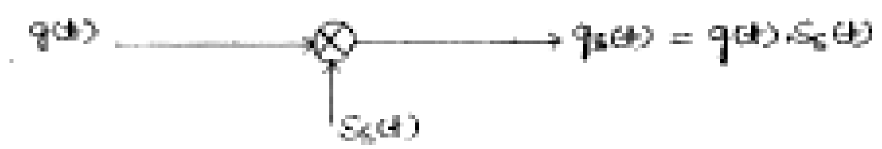
That is Sampling frequency, $f_s \geq 2W$.

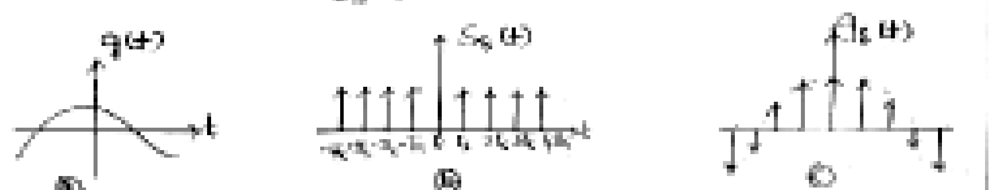
Where W= Highest frequency in base band continuous time signal.

This condition is also called Nyquist condition for sampling process.

Explanation and Proof:

* Consider an arbitrary signal g(t) of finite energy, which is specified for all time. A segment of the signal g(t) is shown in fig(1)(a). suppose that we sample the signal g(t) instantaneously and at a uniform rate, once every 'T_s' seconds. Consequently, we obtain an infinite sequence of samples spaced 'T_s' seconds apart and denoted by $\{g(nT_s)\}$, where 'n' takes on all possible integer values. We refer to 'T_s' as the sampling period, and to its reciprocal $f_s = 1/T_s$ as the sampling rate. This ideal form of sampling is called instantaneous sampling.





Fig(1): (a) Analog signal (b) Periodic signal ($\delta_s(t)$) (c) Sampled signal $g_\delta(t)$

* Let '$g_\delta(t)$' denote the signal obtained by individually weighting the elements of a periodic sequence spaced T_s seconds. Therefore, sampled output $g_\delta(t)$ is given by,

$$g_\delta(t) = g(t) \cdot \delta_\delta(t) \qquad \text{——— (1)}$$

* Let $\delta_\delta(t)$ denotes the periodic impulse train and is represented as,

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

Substituting Eq.(2) in Eq.(1) we get

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

Using shifting property of impulse function

WKT,
$$g(t) \cdot \delta(t - nT_s) = g(nT_s)\, \delta(t - nT_s)$$

$$\therefore \boxed{g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\, \delta(t - nT_s)} \qquad \text{——— (3)}$$

For frequency domain consider,

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

Taking Fourier Transform on both sides, we get

$$G_\delta(f) = G(f) * \delta_\delta(f) \qquad \text{——— (4)}$$

where,

$$S_\delta(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \qquad (5)$$

Substituting Eqⁿ (5) in Eqⁿ (4) we get.

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

From convolution property of impulse function

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

$$\therefore \quad G_s(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s) \qquad (6)$$

Eqⁿ (6) can be rewritten as,

$$G_s(f) = f_s\, G(f) + f_s \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} G(f - nf_s) \qquad (7)$$

When the spectrum of $G_s(f)$ is passed through an LPF, then the 2nd term in RHS of Eqⁿ (7) is eliminated resulting in

$$G_s(f) = f_s \cdot G(f)$$

$$\therefore \quad G(f) = \frac{1}{f_s} \cdot G_s(f) \qquad (8)$$

where $f_s = 2W$

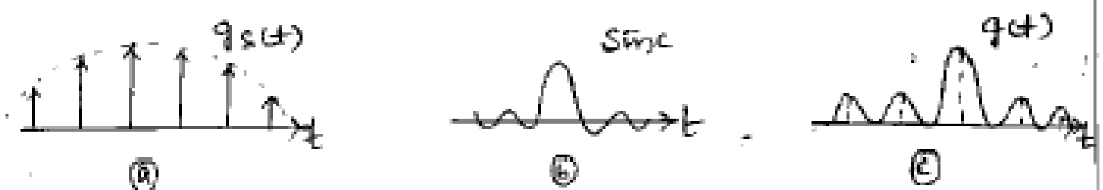


Fig : Recovering g(t) signal from sequence of samples g(nTs).

Now, we may state the sampling theorem for strictly band-limited signals of finite energy into two equivalent parts :

1) A bandlimited signal of finite energy, which only has frequency components less than "W" Hertz, is completely described by specifying the values of the signal at instants of time separated by $\frac{1}{2W}$ seconds.

2) A bandlimited signal of finite energy, which only has frequency components less than "W" Hertz, may be completely recovered from a knowledge of its samples taken at the rate of 2W samples per second.

The sampling rate of 2W samples per second, for a signal bandwidth of 'W' Hertz is called the Nyquist rate; its reciprocal $1/2W$ (measured in seconds) is called the Nyquist interval.

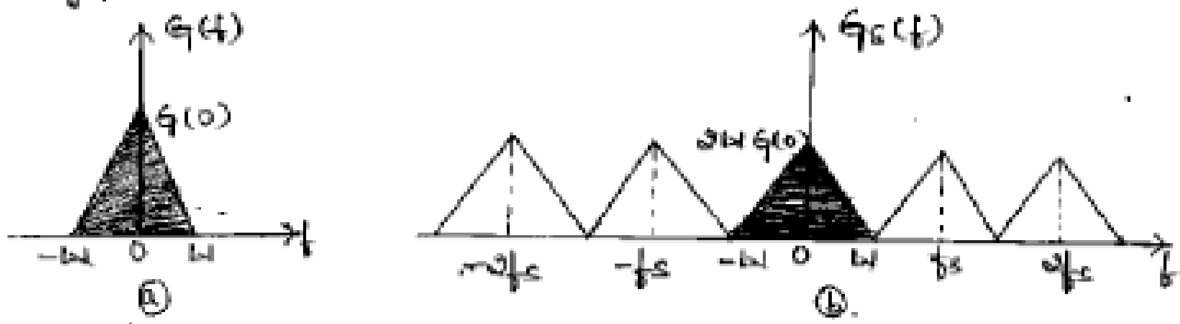


Fig : (a) Spectrum of a strictly band limited signal g(t).
(b) Spectrum of a sampled version of g(t) for $T_s = \frac{1}{2W}$.

7 b) TDM

* TIME DIVISION MULTIPLEXING : [TDM]

Time Division Multiplexing is a method of transmitting and receiving independent signals over a common channel by means of synchronised switches at each end of transmission line so that each signal appears on the line only a fraction of time in an alternating pattern.

* Fig(5) shows the block diagram of TDM system.

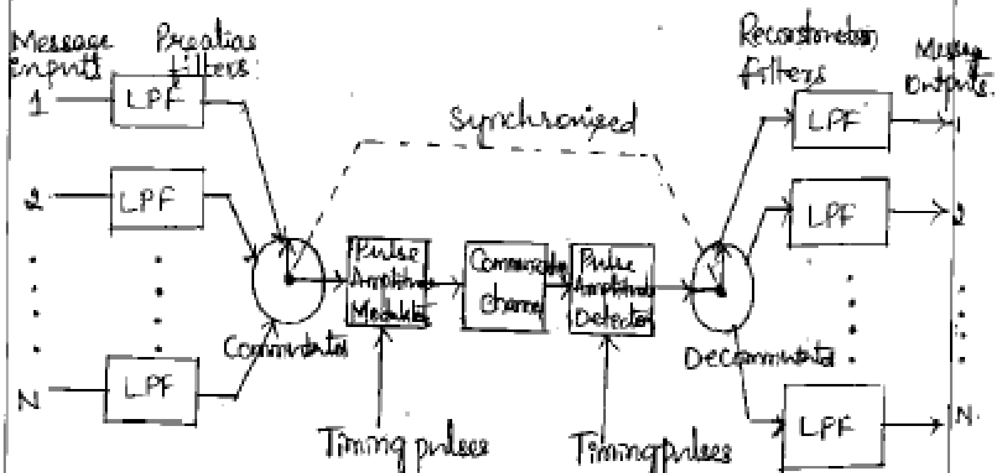


Fig 5 : Block Diagram of TDM system.

* The concept of TDM is illustrated in the fig(5). The Lowpass filters are used to remove high frequency components present in the message signal. The output of the pre-alias filters are then fed to a commutator, which is usually implemented using electronic switching circuitry.

* The function of commutator is as follows:

i) To take a narrow sample of each of the 'N' samples of input at a rate of $f_s \geq 2W$.

ii) To sequentially interleave (multiplex) these 'N' samples inside a sampling interval $T_s = 1/f_s$.

* The multiplexed signal is then applied to a pulse amplitude modulator whose purpose is to transform the multiplexed signal into a form suitable for transmission over a common channel.

* At the receiving end, the pulse amplitude demodulator performs the reverse operation of PAM and the decommutator distributes the signals to the appropriate low pass reconstruction filters. The decommutator operates in synchronisation with the commutator.

7 C

Note:

1) Levels= 2^R
Word length=R

2) Nyquist Rate= 2W

Soln:

a) Nyquist Rate 2*20K = 40K Hz

b) L=65, 536 =2^R

R= 16

MODULE 5

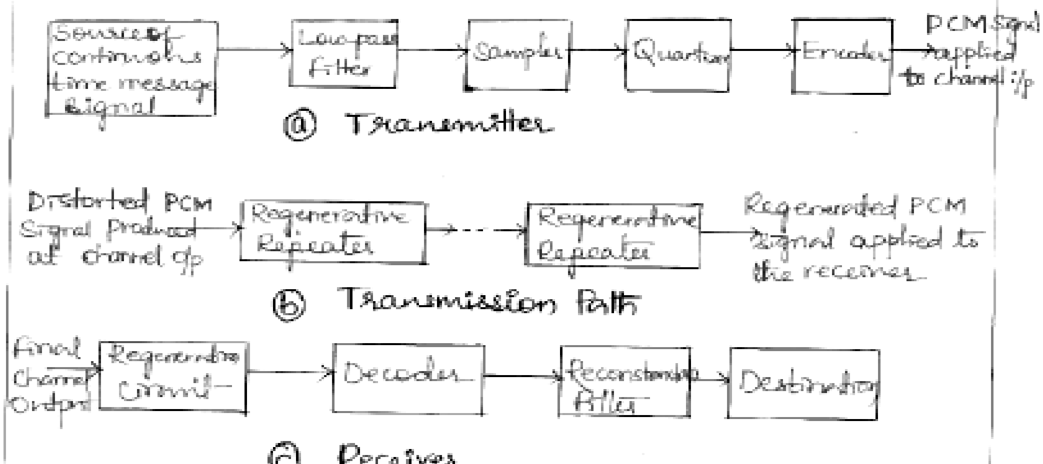
9 a) PCM

* PULSE CODE MODULATION :

* In pulse code Modulation (PCM), a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude.

* The basic operations performed in the transmitter of a PCM system are sampling, quantizing and encoding as shown in fig 6(a). The Lowpass filter prior to sampling is included to prevent aliasing of the message signal. The quantizing and encoding operations are usually performed in the same circuit, which is called an analog to-digital converter.

* The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction of the train of quantized samples as shown in fig 6(c). Regeneration also occurs at intermediate points along the transmission path as necessary as indicated in fig 6(b).



Fig(6) : The basic elements of a PCM system.

* SAMPLING :

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

$$f_s \geq 2W.$$

* Quantization :

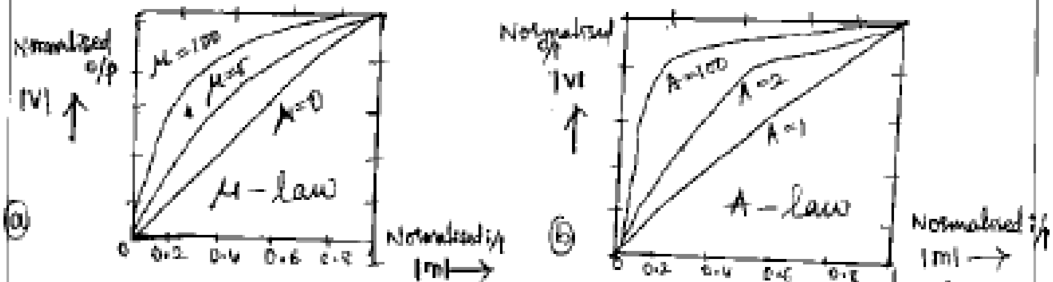
The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude.

* For uniform quantization, we have mid-tread and mid-rise quantizer and for non-uniform quantization, we have two compression laws μ-law and A law.

* The use of a non-uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer. A particular form of compression law that is used in practice is the so called μ-law, defined by

$$|v| = \frac{\log(1+\mu|m|)}{\log(1+\mu)} \qquad \text{——— (1)}$$

where m and v are normalized input & output vtg and 'μ' is positive constant.



* Another compression law that is used in practice is the so called 'A'-law as shown above.

$$|V| = \begin{cases} \dfrac{A|m|}{1+\log A}, & 0 \le |m| \le \dfrac{1}{A} \\ \dfrac{1+\log(A|m|)}{1+\log A}, & \dfrac{1}{A} \le |m| \le 1 \end{cases}$$

* Encoding :

* In combining the processes of sampling and quantizing the specification of a continuous message (baseband) signal becomes limited to a discrete set of values, but not in the form best suited to transmission over a line or radio path.

* In a binary code, each symbol may be either of two distinct values or kinds, such as the presence or absence of a pulse. The two symbols of a binary code are customarily denoted as 0 and 1.

* Line code : It is a line code that a binary stream

* REGENERATION :

The distorted PCM wave obtained from the transmitter is sent to the amplifier equilizer. The output of equilizer device is passed to the Decesion making device to decide the signal interms of 1 or 0 (coded o/p).

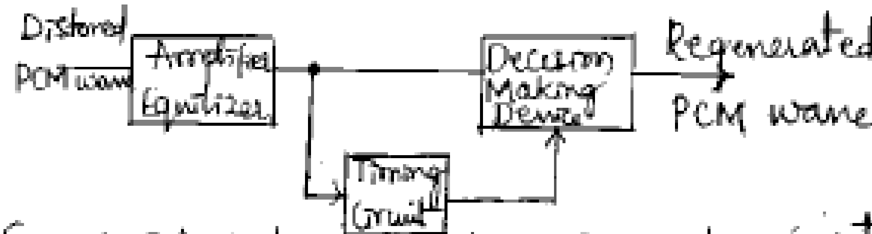


Fig : Block diagram of a regenerative repeater.

* Decoding :

The decoding process involves generating a pulse the amplitude of which is the linear sum of all the pulses in the codeword, with each pulse being weighted by its place value ($2^0, 2^1, 2^2 \ldots 2^{R-1}$) in the code, where 'R' is the number of bits per sample.

* FILTERING :

The final operation in the receiver is to recover the message signal wave by passing the decoder output through a lowpass reconstruction filter whose cutoff frequency is equal to the message bandwidth 'W'.

9 b)

Increased BW is a concern for PCM

Dif

$$e(nT_s) = m(nT_s) - m_q(nT_s - T_s)$$

BW

In d ... pled to purposely increase the

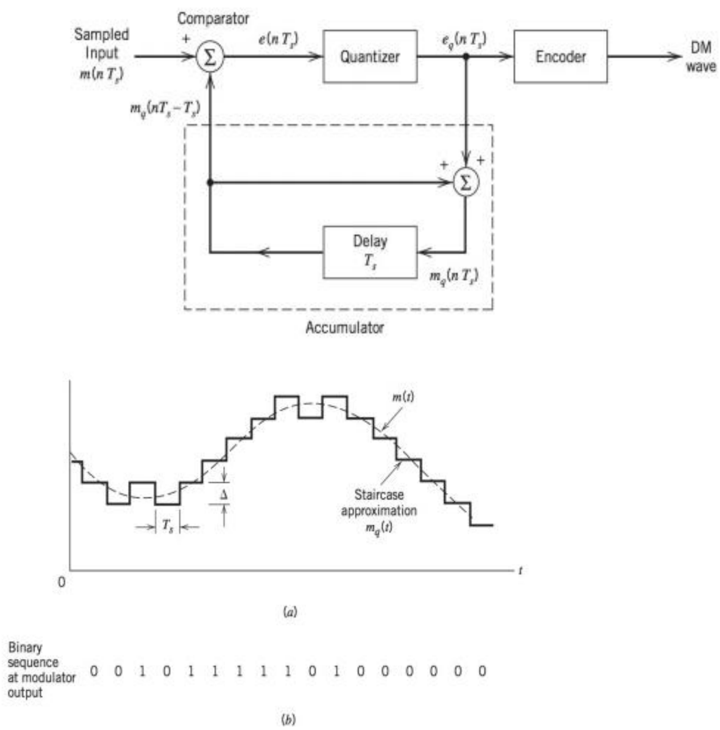
$$e_q(nT_s) = \Delta \operatorname{sgn}[e(nT_s)]$$

Use ... gnal.

DM ... of the message signal

$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$

The ... ed into only two levels, nan



$$e(nT_s) = m(nT_s) - m_q(nT_s - T_s)$$

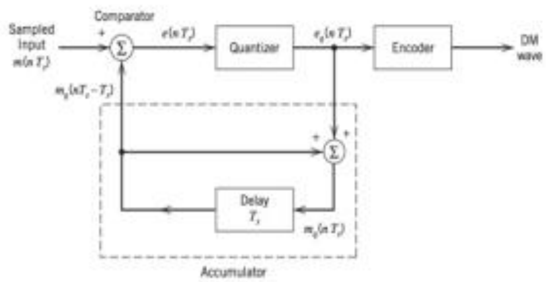
$$e_q(nT_s) = \Delta \operatorname{sgn}[e(nT_s)]$$

$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$

Thus, if the approximation falls below the signal at any sampling epoch, it is increased by Δ. If, on the other hand, the approximation lies above the signal, it is diminished by Δ.

DM(A/D) Encoder

- Comparator
- Quantizer
- Accumulator

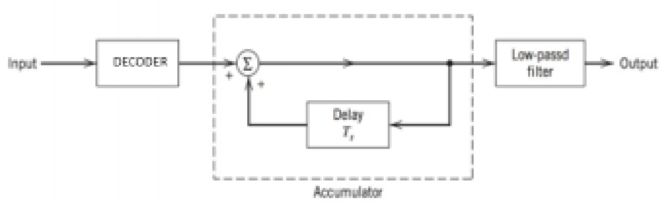


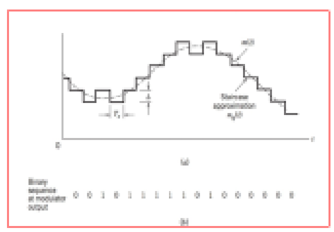
The quantizer consists of a hard limiter with an input-output relation that is a scaled version of the signum function

Accumulator

$$m_q(nT_s) = \Delta \sum_{i=1}^{n} \operatorname{sgn}[e(iT_s)]$$
$$= \sum_{i=1}^{n} e_q(iT_s)$$

DM(D/A) decoder





$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$

$$m_q(nT_s) = \Delta \sum_{i=1}^{n} \operatorname{sgn}[e(iT_s)]$$
$$= \sum_{i=1}^{n} e_q(iT_s)$$

- Staircase approximation mq(t) is reconstructed by passing the sequence of positive and negative pulses, produced at the decoder output, through an accumulator in a manner similar to that used in the transmitter.
- The out-of-band quantization noise in the high-frequency staircase waveform mq(t) is rejected by passing it through a low-pass filter.

10 c)

1) Levels= 2R

Word length=R

2) Transmission bandwidth of PCM >= R* W, R bit per sample, W bandwidth of message signal.

3) Bit Rate =R*2W {Nyquist Rate= 2W}

Given: W=4.2MHZ, Levels=512

i) 2R=512 , R=9: Code length=9bits

ii) Final Bit Rate: R*2W = 9*2*4.2M bits/Sec = 75.6Mbps

iii) Min Transmission bandwidth of PCM = R* W =9*(4.2 M)Hz = 37.8 MHz