(~~AM~~)

Non-linear modulation / Angle modulation

(1) Freq modulation

(2) Phase modulation

$$c(t) = A_c \cos(2\pi f_c t + \phi)$$

radians

radians

$$\cancel{2\pi f_c(t)}\ 2\pi f_c t + \phi(t) = \theta(t)$$

$$c(t) = A_c \cos(\theta(t))$$

In angle modulation, this changes in accordance to message signal.

Angle modulation

↳ angle of carrier changes in accordance to message signal $m(t)$

when we change freq. of angle part
=) frequency modulation

if we change phase (ϕ) ~~with~~
=) angle modulation

II- dependence of f_c is on $m(t)$, then it is called FM (frequency modulation)

PM (phase modulation)

$$c(t) = A_c \cos(2\pi f_c t + \phi)$$

$$\boxed{\phi = k_p m(t)}$$

↳ phase sensitivity constant

* ϕ changes in accordance to $m(t)$

$$\boxed{S_{pm}(t) = A_c \cos(2\pi F_c t + k_p m(t))}$$

general expression

↳ if $m(t) = 0$, $\phi = 0 = k_p m(t)$,

no modulation,

$S_{pm}(t) = A_c \cos(2\pi f_c t)$

(modulated signal = carrier signal)

FREQUENCY MODULATION

$f_c \Rightarrow$ freq. of carrier before modulation.

$f_i \Rightarrow$ freq. of carrier after modulation

$$\boxed{f_i = f_c + K_f\, m(t)}$$

"frequency sensitivity constant"

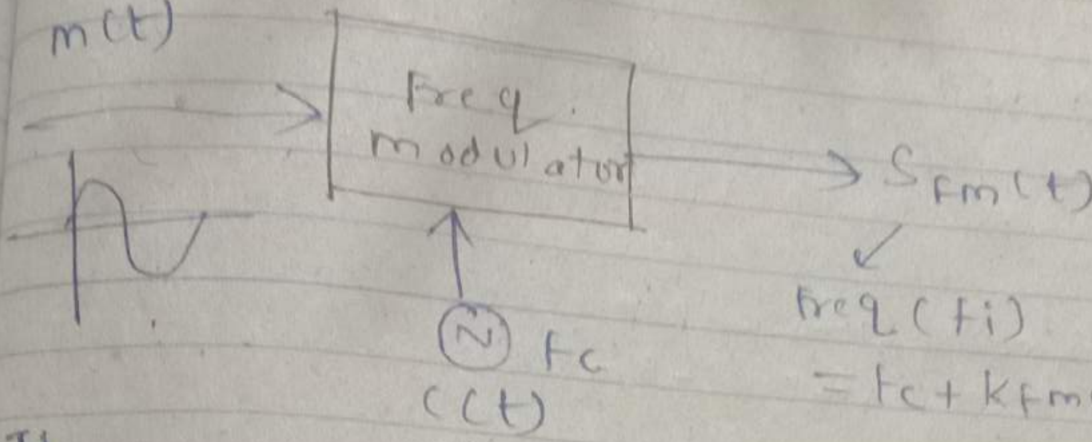
$$\boxed{\text{Unit of } K_f = \text{HZ}/\text{volts}}$$

If $m(t) = 0$, $f_i = f_c$, "NO MODULATION"

If $K_f = 25\,\text{kHz}/\text{V}$

↳ For 1 V of change in m(t), frequency changes by 25 kHz in carrier signal

→ It specifies the amount of freq. change in carrier for 1V of change in message signal.



freq(f_i) $= f_c + k_f m(t)$

If,

$m(t)$, $f_i = f_c$

$\boxed{k = 25 \text{ kHz}}$

$m(t) = 5V$, $f_i = f_c + 125K$ freq. of carrier increases

If $m(t) = -5$,

$f_i = f_c - 125K$ freq. of carrier decreases

i) $m(t) = 0$, $f_i = f_c$

ii) $m(t) > 0$, $f_i > f_c$

iii) $m(t) < 0$, $f_i < f_c$

FM is also called "voltage to frequency conversion"

"since change in m(t) in V changes freq. of carrier signal"

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

$$f_i = f_c + K_f\, m(t)$$

max. amp $= A_m$

min. amp $= -A_m$

max. freq of FM signal

$$\boxed{f_{i_{max}} = f_c + K_f A_m}$$

min. freq of FM signal

$$\boxed{f_{i_{min}} = f_c + K_f(-A_m)}$$

MAX. FREQ DEVIATION IN FM

"f_c" => carrier frequency.

$$\boxed{\Delta f = K_f A_m}$$

TOTAL FREQ. SWING OF FM SIGNAL

$$f_c - K_f A_m \longleftarrow f_c \longrightarrow f_c + K_f A_m$$

$$\boxed{\text{Total freq. swing} = 2 K_f A_m}$$

Q1) A sinusoidal carrier of 20V amplitude and 2MHz of frequency is FM by msg signal of $10\sin(4\pi\times10^3 t)$

$K_f = 50\,KHz/volt$.

Find ΔF, f_{max}, f_{min} ?

Soln:

MS

$$\Delta f = K_f A_m = 50\,KHz/V \times 10\,V$$

$$= 500\,KHz$$

$$f_{max} = f_c + K_f (A_m)$$

$$= 2\,MHz + 50\,KHz/V \times 10\,V$$

$$= 2\,MHz + \cancel{50\,KHz}\ 500\,KHz$$

$$2500\,KHz$$

$$= 2000 + 500 = \cancel{2050\,KHz}$$

$$f_{min} = 2\,MHz - 50\times 10\,KHz$$

$$= 2\,MHz - 500\,KHz$$

$$= 2000\,KHz - 500\,KHz$$

$$= 1500\,KHz$$

SS

$A_c = 20V$

$f_c = 2MHz$

$m(t) = 10\sin(4\pi \times 10^3 t)$

$A_m = 10$

$f_m = 2 \times 10^3 Hz$

$K_f = 50 kHz/V$

$f_{max} = f_c + K_f A_m$

$= 2000 + 50 \times 10$

$= 2500 kHz$

$f_{min} = f_c - K_f A_m$

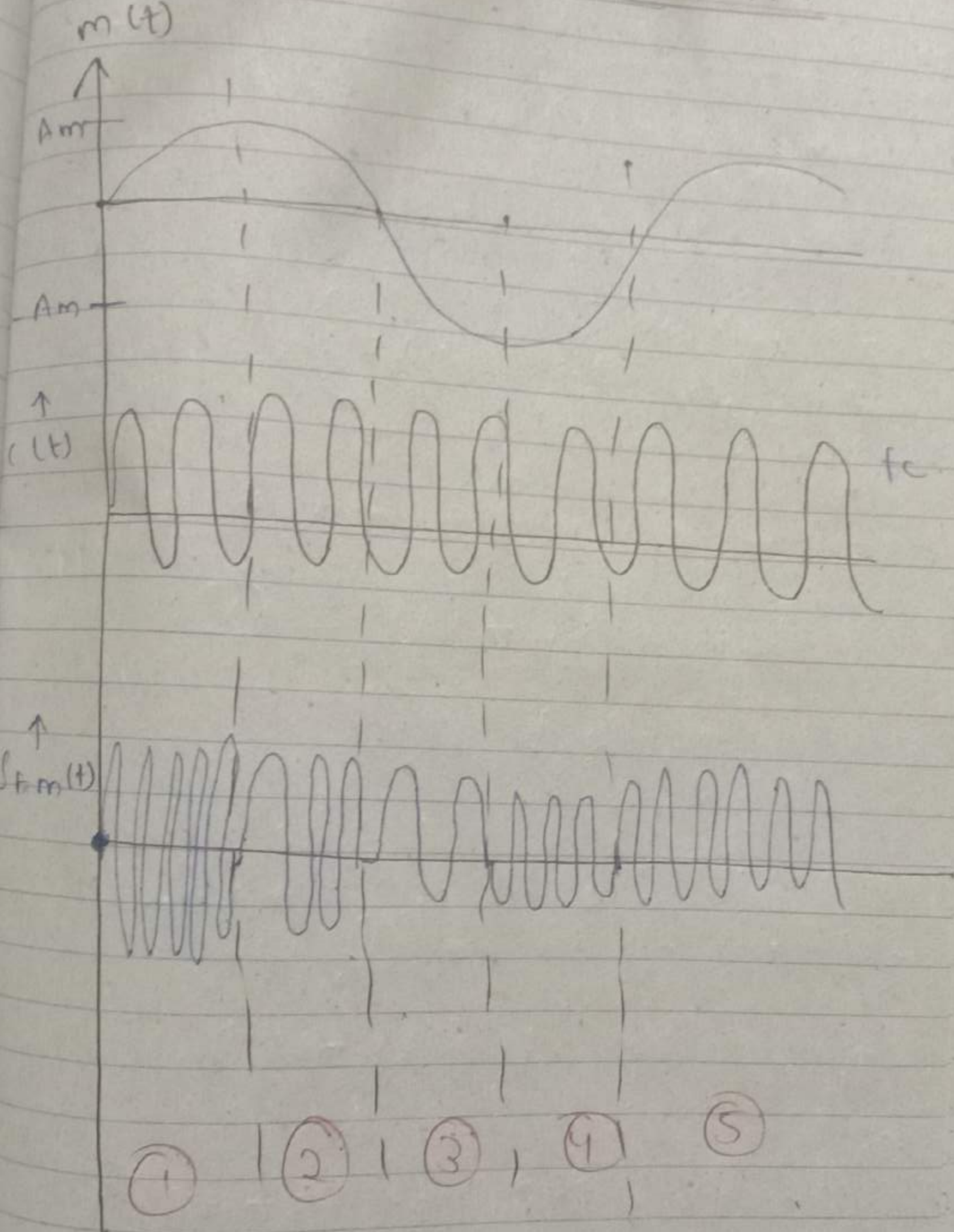
$= 2000 - 50 \times 10$

$= 1500 kHz$

$\Delta F = K_f A_m$

$= 500 kHz$

GRAPHICAL ANALYSIS OF FM



1 → m(t) Amp. ↑, Freq ↑

2 → m(t) Amp. ↓, Freq ↓

3 → m(t) Amp. ↓, Freq ↓

4 → m(t) Amp ↑, Freq ↑.

GENERAL EXPRESSION OF FM SIGNAL

$m(t)$ = message signal

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

$$\left(\overset{\text{angular freq}}{\omega} = \frac{d\overset{\text{angle}}{\theta}}{dt}, \quad \theta = \int \omega \, dt \right)$$

f_i = modulated signal freq.

$$f_i = f_c + k_f m(t) \quad \text{— (1)}$$

$$\theta(t) = \theta_c + \phi$$

$$\theta_c \rightarrow 2\pi f_c t$$

multiplying 2π on (1) both sides,

$$2\pi f_i = 2\pi (f_c + k_f m(t))$$

$$\int 2\pi f_i \, dt = \int 2\pi (f_c + k_f m(t)) \, dt$$

$$= 2\pi f_i t = 2\pi f_c t + 2\pi k_f \int m(t) \, dt$$

SIGNAL

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

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

$$\boxed{s_{FM}(t) = A_c \cos\left(2\pi f_c t + 2\pi K_f \int m(t)\,dt\right)}$$

SINGLETONE FM

(single freq. component in m(t))

$$m(t) = A_m \cos(2\pi f_m \cancel{t}\,t)$$

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

$$s_{FM}(t) = A_c \cos\left(2\pi f_c t + 2\pi K_f \int A_m \cos(2\pi f_m t)\,dt\right)$$

$$= A_c \cos\left(\cancel{2\pi f_c t}\; 2\pi f_c t + 2\pi K_f \times \frac{A_m}{2\pi f_m} \times \sin(2\pi f_m t)\right)$$

$$= \boxed{A_c \cos\left(2\pi f_c t + \frac{K_f A_m}{f_m} \sin(2\pi f_m t)\right)}$$

t

dt

MODULATION INDEX OF FM SIGNAL $\rightarrow (\beta)$

$$\boxed{\frac{K_f A_m}{f_m} = \beta = \frac{\Delta F}{f_m}}$$

$$\boxed{S_{FM}(t) = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))}$$

$\boxed{\beta : \text{modulation index}}$

FM

narrow band FM $(\beta \leq 1)$

wide band FM $(\beta > 1)$

NARROW BAND FM (NBFM)

$$s_{FM}(t) = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$$

$\beta << 1$ (Narrow band)

$$= \cancel{A_c \cos(2}$$

$$= A_c[\cos(2\pi f_c t)\cos(\beta \sin(2\pi f_m t)) - \sin(2\pi f_c t)\sin(\beta \sin(2\pi f_m t))$$

~~θ very small~~

~~θ since β very small~~

For small values of θ,

$$\cos\theta \approx 1$$

$$\sin\theta \approx \theta$$

$$= A_c[\cos(2\pi f_c t) \cdot 1 - \sin(2\pi f_c t) \cdot \beta \sin(2\pi f_m t)]$$

$$\boxed{= A_c \cos(2\pi f_c t) - A_c \beta \sin(2\pi f_c t) \sin(2\pi f_m t)}$$

$S_{NBFM}(t)$]

$$= A_c \cos(2\pi f_c t) - \frac{A_c \beta}{2}\left[\cos(2\pi(f_c - f_m)t)\right]$$

$$+ \frac{A_c \beta}{2}\left[\cos(2\pi(f_c + f_m)t)\right]$$

$\updownarrow$ compare

$$S_{AM}(t) = A_c \cos(2\pi f_c t)$$

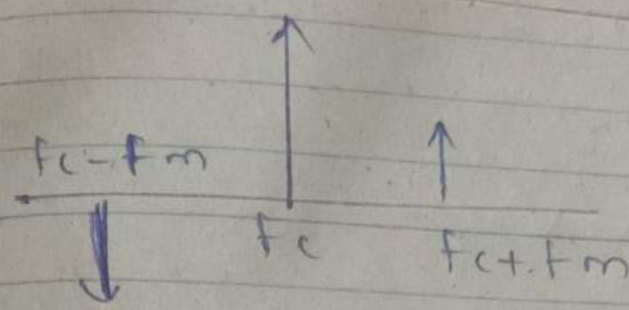
$$+ \frac{A_c \mu}{2}\cos(2\pi(f_c + f_m)t)$$

$$+ \frac{A_c \mu}{2}\cos(2\pi(f_c - f_m)t)$$

* NBFM many ~~[illegible]~~ similarities to AM

* ~~carrier~~ lower band of NBFM is inverted in respect to lower band of AM

SPECTRUM OF NBFm



* $\boxed{BW = f_c + f_m - f_c + f_m = 2f_m}$

* $\boxed{P_{NBFm} = \cancel{P/Add} = P_c\left[1 + \beta^2/2\right]}$

* NBFm is not a significantly used version of Fm since BW & power requirements same as Am. (Not getting much improve--ment).

GENERATION OF NBFM signal

$S_{NBFM}(t)$

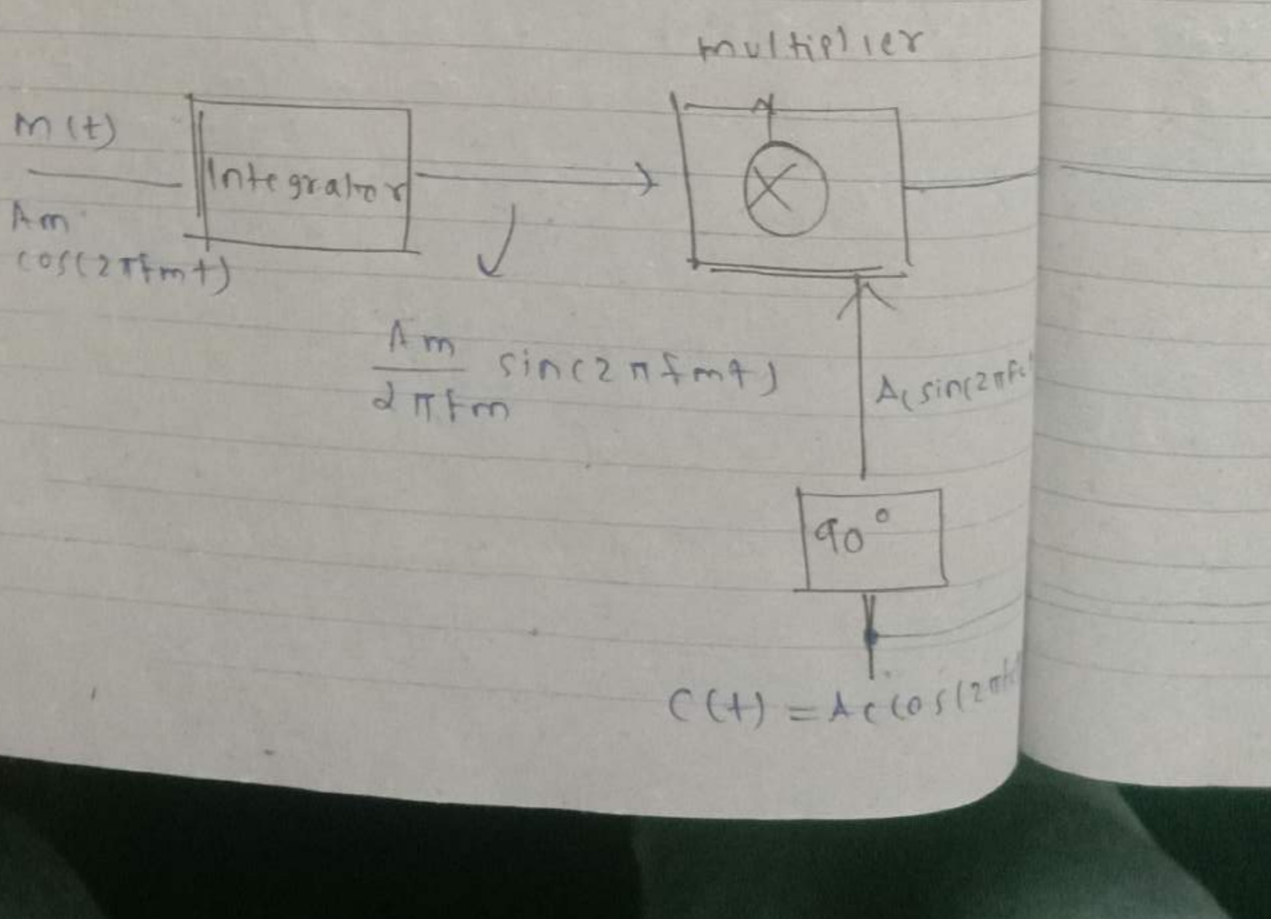
$= A_c \cos(2\pi f_c t)$

$- \frac{A_c \beta}{2} \sin(2\pi f_c t) \sin(2\pi f_m t)$

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

$- \frac{A_c k_f A_m}{f_m} \sin(2\pi f_c t) \sin(2\pi f_m$

CIRCUIT



GENERATION OF NBFM Signal

$S_{NBFm}(t)$

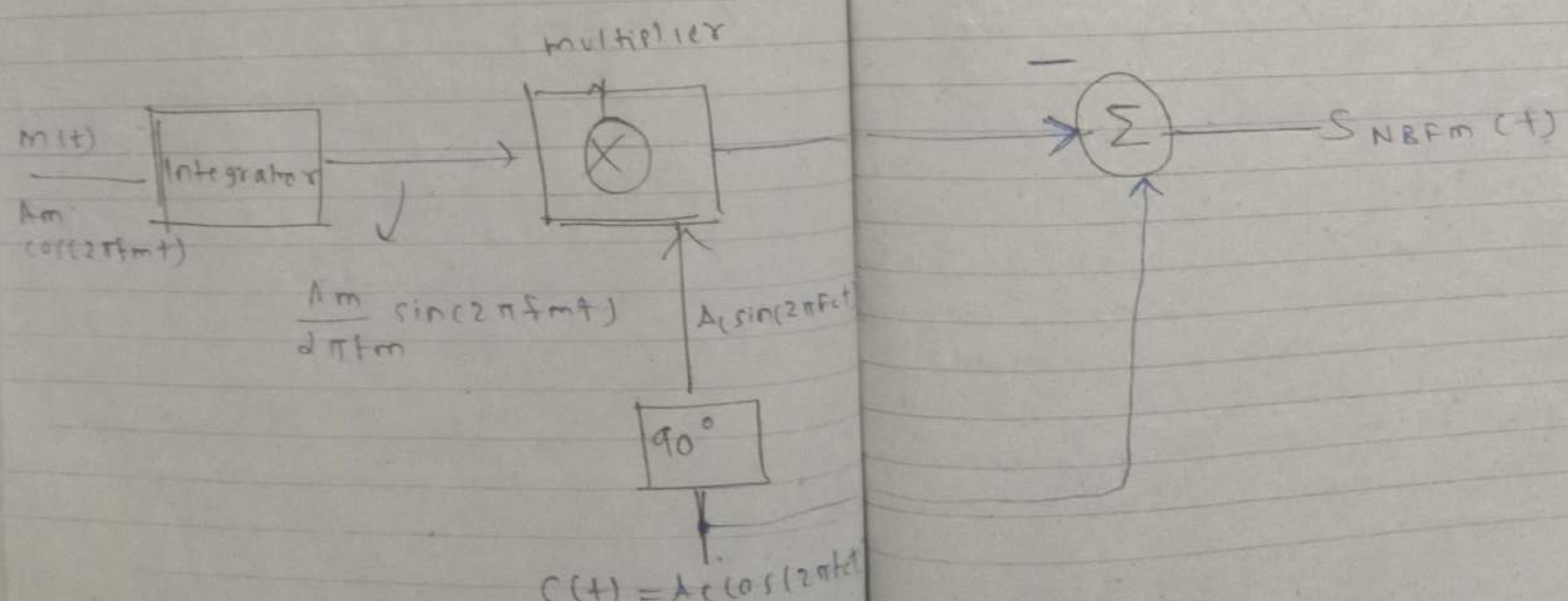
$= A_c \cos(2\pi f_c t)$

$- \frac{A_c \beta}{2} \sin(2\pi f_c t) \sin(2\pi f_m t)$

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

$- \frac{A_c k_f A_m}{f_m} \sin(2\pi f_c t) \sin(2\pi f_m t)$

CIRCUIT



POC - Lec 2 (After mid.)

RECAP

$f_i = f_c + k_f m(t)$

↳ freq. sensitivity constant

$= Hz/V$

$S_{FM}(t) = \cancel{A_c \cos(2\pi f_c t}$

$= A_c \cos(2\pi f_c t + 2\pi k_f \int m(t)\,dt)$

Single tone

$S_{FM}(t) = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$

β = mod. index

NBFM ($\beta \leq 1$)

WBFM ($\beta > 1$)

$\beta = \frac{\Delta f}{f_m}$ → freq. deviation

$= \frac{k_f A_m}{f_m}$

NBFM $\xleftrightarrow[\text{similar}]{}$ AM

but ↳ lower side band inverted !!

WIDE BAND FM

$$\beta > 1$$

BESSEL FUNCTION

$$J_n(x) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j(x\sin\theta - n\theta)} \, d\theta$$

n = order of Bessel.

θ = dummy variable

PROPERTIES:

(1) $J_n(x) \downarrow,\ n \uparrow$

$$J_0(x) > J_1(x) > - - - -$$

(2) $\sum_{n=-\infty}^{\infty} J_n^2(x) = 1$

summation of square of coefficients

(3) Bessel function always a real quantity

GENERAL EXPRESSION OF WBFM

$$S_{FM}(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$$

$$\cos\theta = Re[e^{j\theta}]$$

$$S_{FM}(t) = A_c Re[e^{j(2\pi f_c t + \beta \sin(2\pi f_m t))}]$$

$$= A_c Re[e^{j2\pi f_c t} \cdot \underbrace{e^{j\beta \sin(2\pi f_m t)}}]$$

continuous periodic signal (we find Fourier series of this component)

$$\boxed{S_{WBFM}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[2\pi(f_c + nf_m)t]}$$

(1) ∞ terms

(2) $A_c J_0(\beta)\cos(2\pi f_c t)$

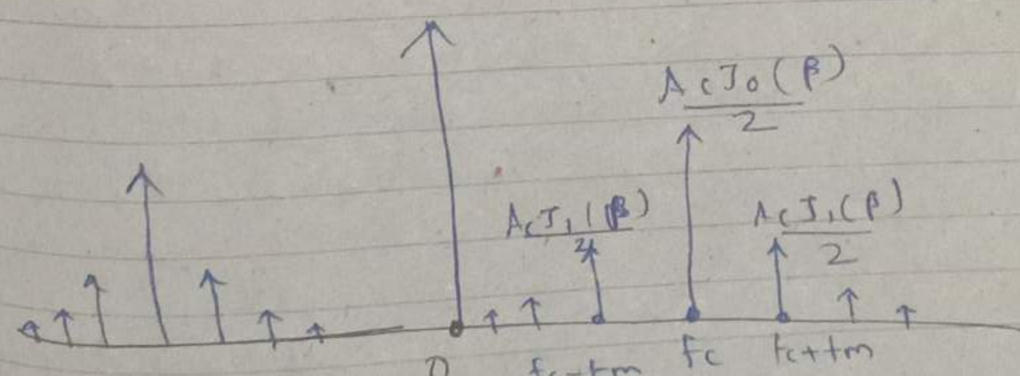
$+ A_c J_1(\beta)\cos(2\pi(f_c+f_m)t)$

$+ A_c J_{-1}(\beta)\cos(2\pi(f_c-f_m)t))$

$+ A_c J_2(\beta)\cos(2\pi(f_c+2f_m)t)$

$+ - - -$

SPECTRUM OF WBFM



* WBFM consists of carrier component & ∞ no. of side bands

* Actual BW of wideband FM is ∞

* For WBFM, lower order sideband are significant sidebands and higher order sidebands not significant

POWER OF WBFM

$$P_t = P_c + (P_{USB_1} + P_{USB_2} - - \infty)$$
$$+ (P_{LSB_1} + P_{LSB_2} - - \infty)$$

$$P_c = \frac{A_c^2 J_0^2(\beta)}{2} \qquad P_{LSB_1} = \frac{A_c^2 J_{-1}^2(\beta)}{2}$$

$$P_{USB_1} = \frac{A_c^2 J_1^2(\beta)}{2} \qquad P_{LSB_2} = \frac{A_c^2 J_{-2}^2(\beta)}{2}$$

$$P_{USB_2} = \frac{A_c^2 J_2^2(\beta)}{2}$$

$$P_t = \frac{A_c^2}{2}\left[\underbrace{\left(\cdots + J_{-2}^2(\beta) + J_{-1}^2(\beta) + J_0^2(\beta) + J_1^2(\beta) + J_2^2(\beta) \cdots\right)}_{=1 \text{ (according to prop. of Bessel function)}}\right]$$

$$\boxed{P_t = \frac{A_c^2}{2} = P_c}$$

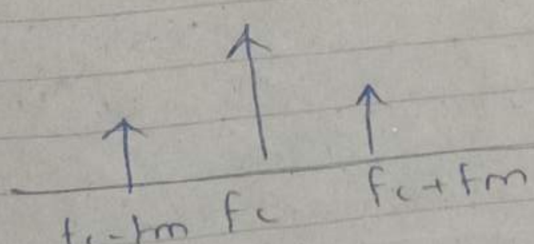
* Total power of WBFM = power of carrier before modulation

PRACTICAL BW OF WBFM

derived using {CARSON'S RULE}

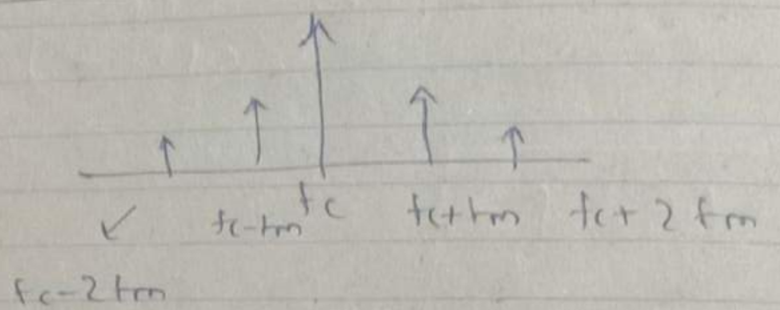
$$\boxed{BW_{WBFM} = \infty \text{ (Actual)}}$$

CASE 1: WBFM consists of significant sidebands upto 1st order



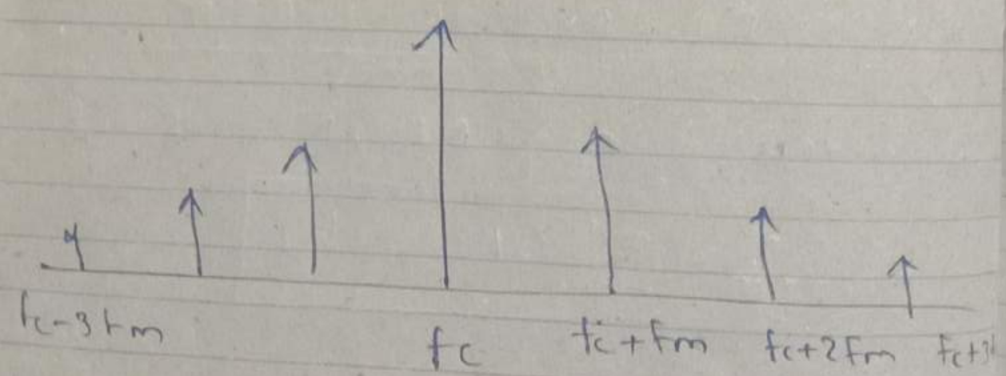
$BW = 2f_m$

CASE 2: Sidebands upto 2nd order



$$BW = 4f_m = 2 \times (2f_m)$$

CASE 3: Sidebands upto 3rd order



$$BW = 6f_m = 3 \times (2f_m)$$

Upto nth order

$$BW = n \times (2f_m)$$

CARSON

WBFm
sideband
where

BW =

BW = (

BW =

Q1) A
for an
modulat
of 10V
$P_{av} + 1$ kf = 2
(i) Find
(ii) Find
(iii) Find
(iv) Find P

CARSON'S RULE

WBFM consists of significant sidebands upto "$(\beta + 1)$" order where β = modulation index

$$\boxed{BW = (\beta + 1) \cdot 2f_m}$$

$$\downarrow$$

$$BW = \left(\frac{\Delta f}{f_m} + 1\right) \cdot 2f_m$$

$$\therefore \boxed{BW = 2(\Delta f + f_m)}$$

Q1) A sinusoidal carrier of 20V and 2 MHZ is frequency modulated by a sinusoidal msg. of 10V & 50 KHZ,

$K_f = 25\ KHZ/V$

Part 1

(i) Find Δf

(ii) Find β

(iii) Find BW

(iv) Find Power (P)

Part 2

(v) Repeat above (i–iv) if Amp. of msg. signal is doubled.

Soln:

$A_m = 10V$

$A_c = 20V$

$f_c = 2MHz$ $\quad f_m = 50 kHz$

$k_f = 25 kHz/V$

* $\beta = \frac{\Delta f}{f_m} = \frac{k_f A_m}{f_m} = \frac{25 \times 10}{50} = 5$

" $\beta \geq 1$, WBFM "

* $BW = (\beta + 1) \cdot 2 f_m$

$= (5+1) \times 2 \times 50 kHz$

$= 600 kHz$

* $P_t = \frac{A_c^2}{2} = \frac{20 \times 20}{2} = 200 W$

Part 2:

$A_m = 10 \times 2 = 20V$

$\uparrow \Delta f = k_f A_m \uparrow$

A_m = doubled, ΔF doubled

$\Delta f = 250 \times 2 = 500 kHz$

$$\beta = \frac{\Delta F}{F_m} = \frac{K_f A_m}{f_m}$$

$\beta = 10$ ($A_m = 2$ times, $\beta = 2$ times)

WBFM"

$$BW = (\beta + 1) \cdot 2 f_m$$

$$= (10+1) \times 2 \times 50 = 11 \times 100$$

$$= 1100 \text{ kHz}$$

$$P_t = P_c = \frac{A_c^2}{2} = 200 \text{ W}$$

~~NOTE~~ NOTE:

~~Amplitude of msg signal~~

Amp. of message signal

→ changes

① ΔF

② BW

③ β

→ not changes

① P_t

$2\pi f_c t = \overline{2\pi \times 10^6 t}$

S POC lec 3 (After midsem)

Q1) $c(t) = 5\cos(2\pi \times 10^6 t)$
$m(t) = \cos(4\pi \times 10^3 t)$

a) $c(t)$ & $m(t)$ are used to generate AM with $\mu = 0.707$ find bandwidth & power?

b) $c(t)$ & $m(t)$ are used to generate FM with max. freq deviation as 3 times the BW of AM. Find coefficient of $\cos(2\pi \times (1016 \times 10^3))t$ in FM expression?

Soln: MS

(i) $A_c = 5$, $f_c = 10^6$ Hz
$A_m = 1$ $f_m = 2 \times 10^3$ Hz

a) $\mu = 0.707$
$BW = 2f_m = 4 \times 10^3$ Hz
$= 4$ kHz

(ii) $P = \frac{A_c^2}{2}\left(1 + \frac{\mu^2}{2}\right)$
$= \frac{5 \times 5}{2}\left(1 + \frac{0.707^2}{2}\right) = 15.62$

b) Δf

$= \Delta f$

$\beta =$

$\beta > 1$

$f_c +$

$10^3 \times$

coeff

$J_n(\beta)$

b) $\Delta f = 3 . BW_{Am}$

$= \Delta f = 3 \times 4 \text{ kHz} = 12 \text{ kHz}$

$\beta = \frac{\Delta f}{f_m} = \frac{12 \cancel{\times 000} \text{ kHz}}{2 \text{ kHz}}$

$= 6$

$\beta > 1$, WBFm

$f_c + n f_m = 1016 \times 10^3$

$10^3 \times 1\cancel{0^3} + n \times 2 \times \cancel{10^3} = 1016 \times \cancel{10^3}$

$1000 + 2n = 1016$

$2n = 1016 - 1000 = 16$

$n = 8.$

$\cancel{\text{coefficient} = A_c \times J_n(8)}$

$\cancel{J_n(8) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j(8 \sin\theta -}}$

SS

(Ans)

a) $m(t) = \cos(4\pi \times 10^3 t)$

$A_m = 1$

$f_m = 2\ kHz$

* $c(t) = 5\cos(2\pi \times 10^6 t)$

$A_c = 5$

$f_c = 1000\ kHz$

* $BW = 2f_m = 2 \times 2 = 4\ kHz$

$$P = P_c\left(1 + \frac{\mu^2}{2}\right)$$

* $$= \frac{A_c^2}{2}\left(1 + \frac{\mu^2}{2}\right)$$

$$= \frac{25}{2}\left(1 + \frac{0.707^2}{2}\right)$$

$$= \frac{25}{2}\left(1 + \frac{1}{4}\right) = \frac{125}{8}\ W$$

b) $\Delta f = 3\, BW_{AM}$

$= 3 \times 4\, kHz = 12\, kHz$

$\beta_{FM} = \frac{\Delta f}{f_m} = \frac{12\, kHz}{2\, kHz}$

$= 6$

$\beta > 1$

$\Rightarrow WBFM$

$\cos [2\pi (1016 \times 10^3) t]$

$S_{WBFM} = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos 2\pi (f_c + n f_m) t$

$\cos 2\pi (f_c + n f_m) t$

$f_c + n f_m = 1016$

$1000 + n f_m = 1016$

$n = 16/2 = 8$

order of Bessel func $= 8$

$\beta = 6$

$\text{coeff} = A_c J_8(6)$

$= 5 J_8(6)$ (Ans)

Q2) A sinusoidal carrier of freq "f_c" is used for both AM & FM transmitter. msg signal freq is given by 5kHz. Max. freq deviation = 2×BW Am, find modulation index of both AM & FM such that strength of freq component $f_c + 5k$ is same in both AM & FM spectrum.

$J_1(0) = 1$

$J_1(2) = 0.57$

$J_1(4) = 0.37$

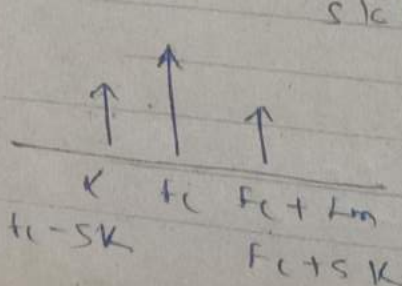
Soln:

$f_m = 5\,kHz$

$$BW_{Am} = 2 \times f_m$$
$$= 2 \times 5\,kHz$$
$$= 10\,kHz \text{ (Ans)}$$

$$\Delta F = 2 \times 10\,kHz = 20\,kHz$$

$$\beta = \frac{\Delta f}{f_m} = \frac{20\,kHz}{5\,kHz} = 4 \text{ (Ans)}$$



...er of freq
AM & FM
...nal freq
freq
...nd
...th AM
of freq
in

12

(Ans)

"Strength => Power"

Power of sideband in AM $= \dfrac{A_c^2 \mu^2}{8}$

Power of SB wrt FM $= \dfrac{A_c^2}{4} J_n^2(\beta)$

$$\frac{A_c^2 \mu^2}{\cancel{8}\,2} = \frac{\cancel{A_c^2}}{\cancel{4}} J_n^2(\beta)$$

$$\frac{\mu^2}{2} = J_n^2(\beta)$$

$n = 1$, $\beta = 4$

$$\frac{\mu^2}{2} = J_1^2(4)$$

$$\mu^2 = 2J_1^2(4)$$

$$= 2 \times (0.37)^2$$

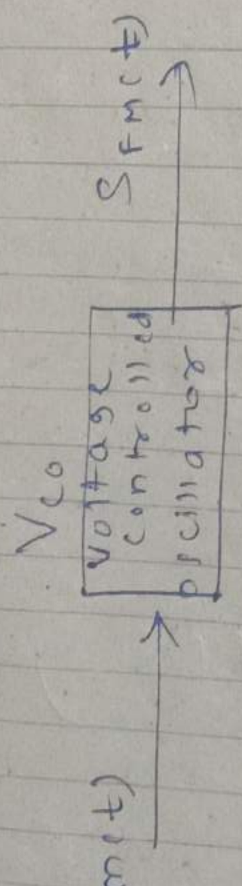
$$\mu = \sqrt{2 \times (0.37)^2}$$

$$= 0.52 \quad \text{(Ans)}$$

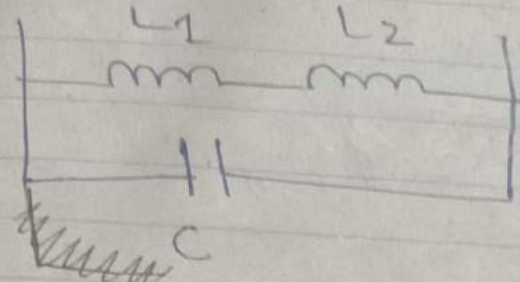
GENERATION OF ~~PM~~ FM

1. Direct method
2. Indirect method / Armstrong method

DIRECT METHOD



VCO :

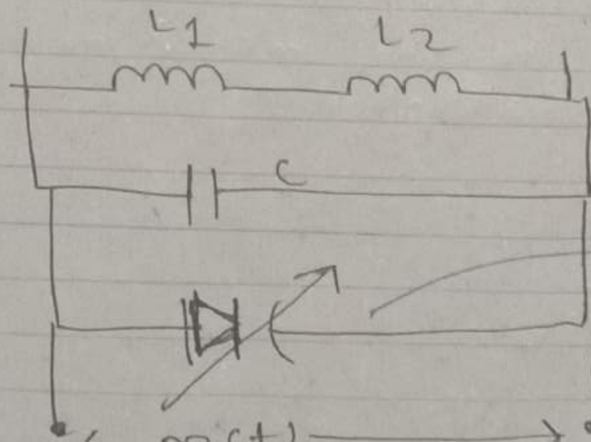


⇒ Hartlay oscillator

$$f_R = \frac{1}{2\pi\sqrt{(L_1+L_2)C}}$$

resonating freq.

VCO from hartlay occillator :



→ varactor diode (variable ~~[illegible]~~ voltage capacitor)

* In appliance to m(t), capicitance of varactor diode change

* let C' the capicitance of varactor diode

* C' changes, overall capicitance changes, f_r changes

$$f_r = \frac{1}{2\pi\sqrt{(L_1 + L_2) \cdot C''}}$$

$$C'' = (C + C')$$

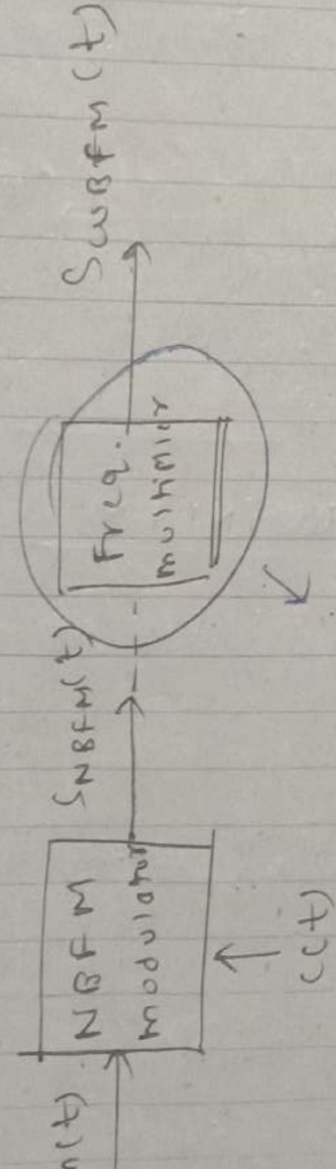
* Hence f_r varies in accordance to $m(t)$ signal.

NOTE:

Here the freq. is varied in accordance to the msg. signal voltage variations.

INDIRECT METHOD (ARMSTRONG METHOD)

→ WBFM is generated from NBFM



It is a square law device followed by proper bandpass filter

v_i — SLD — $v_2 = a_1 v_1 + a_2 v_2^2 + a_3 v_3^3$ — — —

"square law device"

Ex)

Let $a_1 = 1$

$V_i = \cos(2\pi f_i t) \rightarrow$ Squaring device $\rightarrow V_o = V_i^2 = \cos^2(2\pi f_i t)$

freq $= f_i$ (n = 2) multiplying by 2

$$= \frac{1 + \cos(4\pi f_i t)}{2}$$

$$\boxed{\text{freq} = 2 f_i}$$

$\cos\theta \longrightarrow$ Freq. multiplier $\longrightarrow \cos(n\theta)$

(n) = order

NOW PASSING NBFM :

NBFM $= A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$

$\downarrow$

Freq. multiplier (order = n) → multiplication order

$\downarrow$

$A_c \cos(2\pi n f_c t + n\beta \sin(2\pi f_m t))$

WBFM such that "$n\beta > 1$".

I/P of freq. multiplier	O/P of freq. multiplier
f_c	nf_c
β	$n\beta$
f_m	f_m (No change in message signal frequency)
$\Delta f = \beta f_m$	$n\Delta F$

Q3) An FM is given as

$$s(t) = 10\cos(2\pi \times 10^6 t + 0.2\sin(2\pi \times 2 \times 10^3 t$$

It is passed through cascaded freq. multipliers having multiplication order as 4 & 5 respectively. Find all parameters of FM signal at the o/p of each multiplier?

Soln:

$$s(t) = 10\cos(2\pi \times 10^6 t + 0.2\sin(4\pi \times 10^3 t))$$

$$= A_c \cos(2\pi f_c t + \cancel{n f_m}\ \beta \sin(2\pi f_m t))$$

$A_c = 10$

$\beta = 0.2$

$f_c = 1\ MHz$

$f_m = 2\ kHz$

$\Delta f = \beta f_m = 0.4\ kHz$

After pa

$A_c = 1$

$f_c =$

$n\beta =$

$f_m =$

$\Delta f = n$

$BW = 2$

$P_t = \frac{A}{2}$

$= \frac{10}{2}$

After pas

$A_c = 10\ V$

$n f_c = 5$

$\beta = n\beta =$

$f_m = 2k$

$\Delta f = n\Delta$

After passing through $n=4$

$A_c = 10\,V$

$f_c = n f_c = 4 \times 1\,MHz = 4\,MHz$

$n\beta = 4 \times 0.2 = 0.8$ (NBFM)

$f_m = 2\,kHz$

$\Delta f = n \Delta f_{old} = 4 \times 0.4 = 1.6\,kHz$

$BW = 2 f_m = 4\,kHz$

$P_t = \frac{A_c^2}{2}\left(1 + \beta^2/2\right)$

$= \frac{100}{2}\left(1 + \frac{0.64}{2}\right) = 66\,W$

After passing through consecutive $n=5$

$A_c = 10\,V$

$n f_c = 5 \times 4 = 20\,MHz$

$\beta = n\beta = 5 \times 0.8 = 4$ (WBFM)

$f_m = 2\,kHz$

$\Delta f = n \Delta f = 5 \times 1.6 = 8\,kHz$

$$BW = (\beta + 1) \cdot 2f_m$$

$$= 20\,kHz$$

$$P_t = \frac{A_c^2}{2} = 50\,w$$

POC lec 4 (After midterm)

FM Detectors

Procedure

FM to AM conversion

↓

AM demodulators
(Envelope detector)

↓

m(t)

→ Information in FM signal is contained in the instances frequency.

$$\boxed{\omega_i = \omega_c + K_f\, m(t)}$$

→ FM demodulators are implemented by generating an AM signal whose amplitude is proportional to instantaneous freq of FM signal.

S. $x_{FM}(t) = A\cos[\omega_c t + k_f \int_{-\infty}^{t} m(\tau)\,d\tau]$

$\Downarrow$

AM

$\Downarrow$

AM demodulator

$\Downarrow$

m(t)

FM to AM conversion

* (1) Differentiator

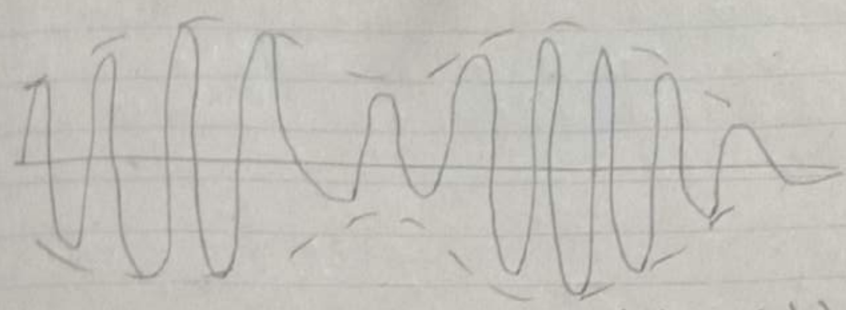
$$x_{FM}(t) \longrightarrow \boxed{\text{diff.}} \longrightarrow ?$$

P. $$\frac{dx_{FM}(t)}{dt} = \frac{d}{dt}\left[A\cos(\omega_c t + k_f \int_{-\infty}^{t} m(\tau)\,d\tau)\right]$$

$$= A[\omega_c + k_f m(t)] \sin\left[\omega_c t + k_f \int_{-\infty}^{t} m(\tau)\,d\tau\right]$$

It is both frequency & Amplitude modulated

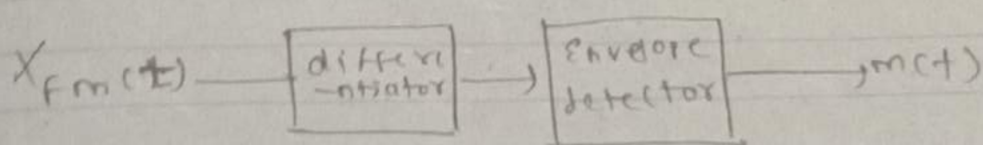
PLOT OF THE above SIGNAL (Freq. & Amp. modulated)



$$\frac{d x_{Fm}(t)}{dt}$$

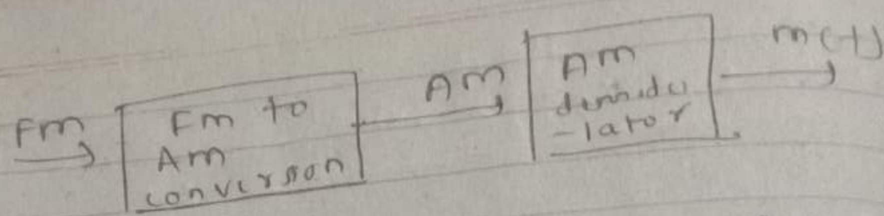
envelope of signal

$= A(\omega_c + k_F m(t))$



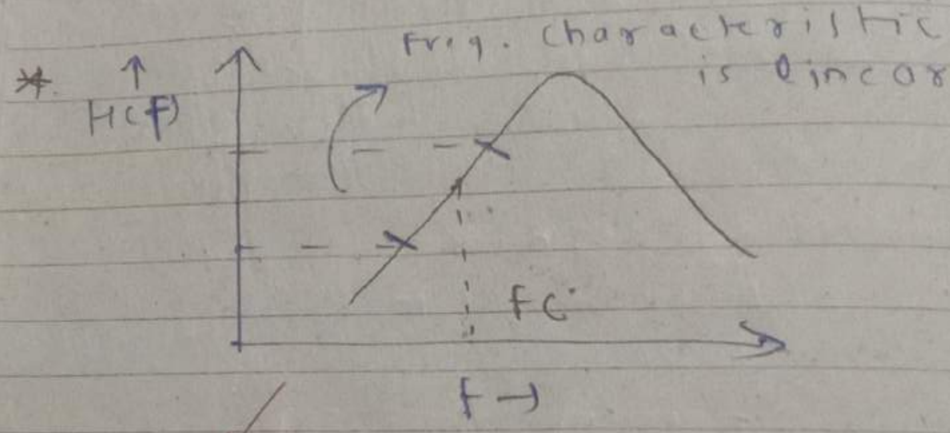
* The envelope of this wave is $A(\omega_c + k_F m(t))$

constants, If we are able to remove this, we can get back the m(t) signal

S 

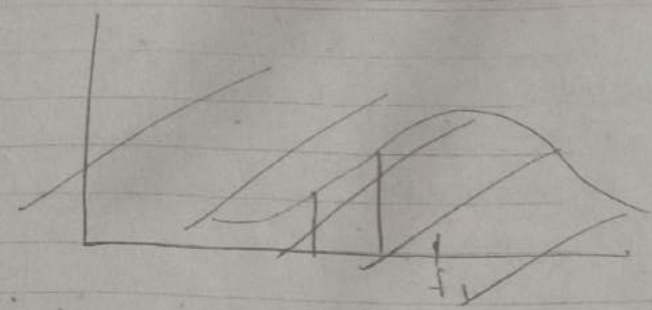
* A simple tuned circuit followed by an Am demodulator/envelope detector can serve as Fm demodulator

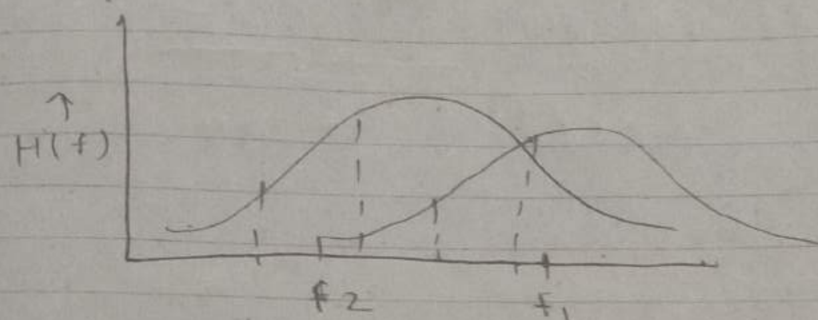
*

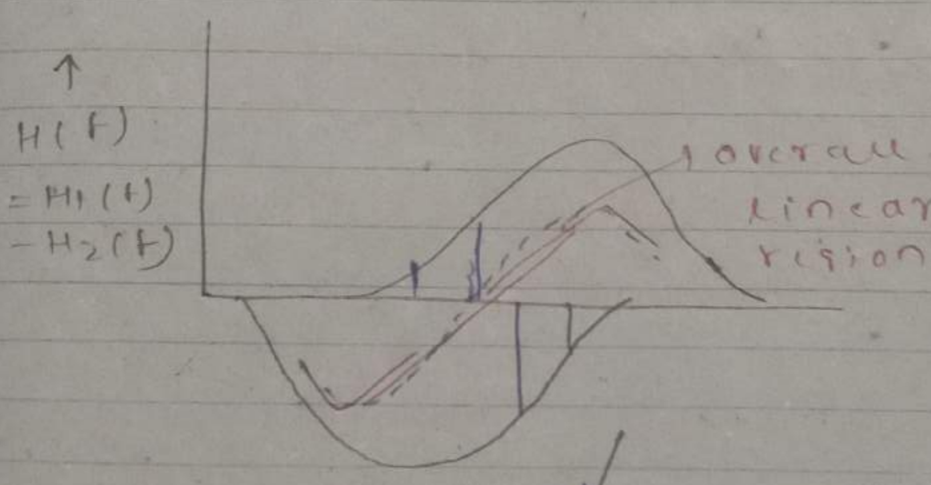
* 

"Slope detection" = since detection on slope of freq. characteristic

P To get linear characteristic over a wide range of frequencies we used two tuned circuits at frequency f_1 & f_2. connected in a configuration ⇒ Balanced discriminator







Balanced demodulator

"explain the working of demodulator"

RATIO DETECTOR

It provides
(better protection against
carrier amplitude variation)

PHASE LOCKED LOOP (PLL)

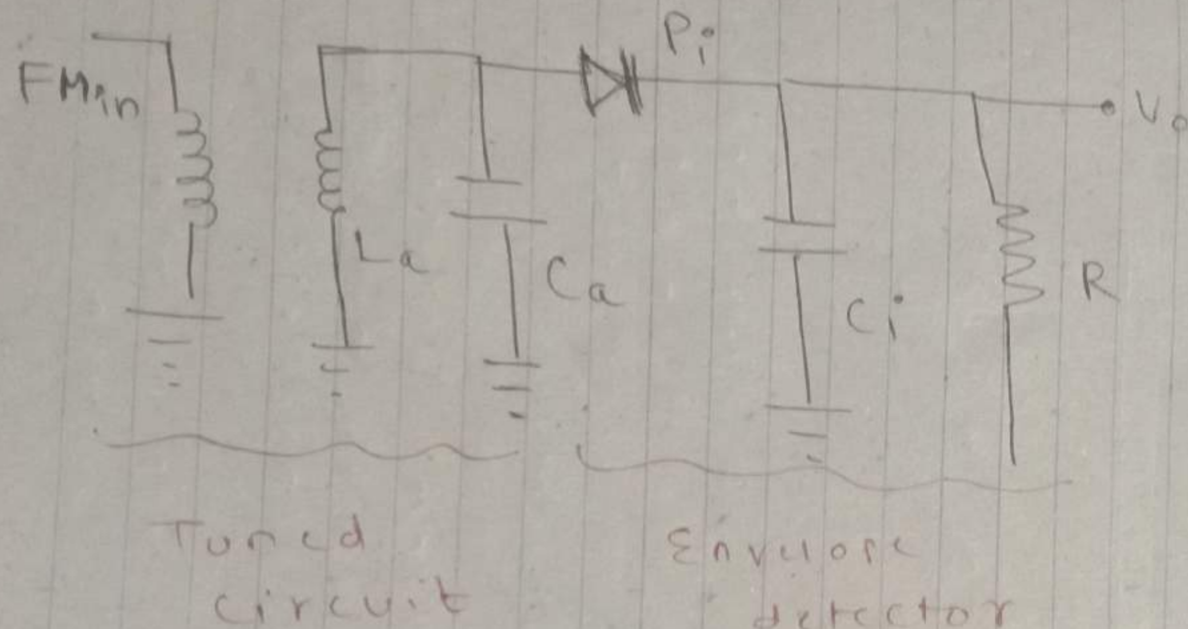
* most widely used FM demodulator

low cost

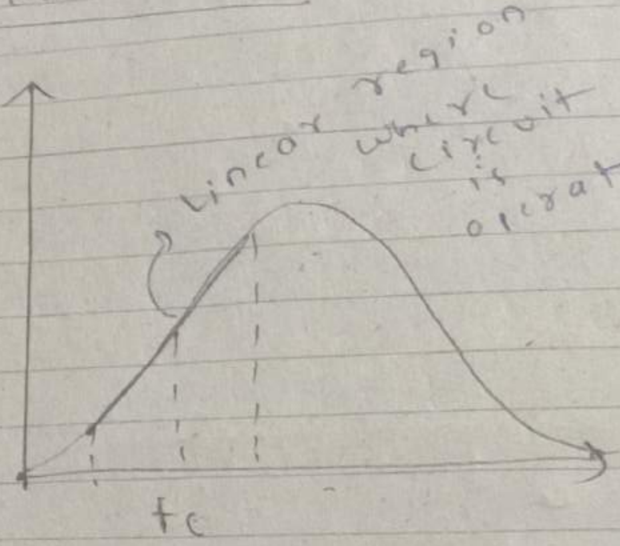
superior performance

~~SLOPE DETECT~~

SLOPE DETECTOR



S FREQ. CHARACTERISTIC OF TUNED CIRCUIT



* fc : resonating freq. of Tuned circuit

* Tuned circuit consisting of La & Ca produced an O/P voltage that is proportional to I/P frequency.

* At the resonant frequency fc of tuned circuit, max. ~~voltage~~ O/~~P occurs~~ O/P voltage occurs.

* Freq. variation is mapped to Amplitude variation

Freq. ↑ by Δf ⇒ O/P voltage ↑
(f_c)

Freq ↓ by Δf ⇒ O/P voltage ↓
(f_c)

envelope detector
↓
m(t) = Final O/P

DISADVANTAGE

linear region very less, with most non-linear frequency characteristic

If 2 signals have constant phase diff over a period of time
=> both signals have same freq.

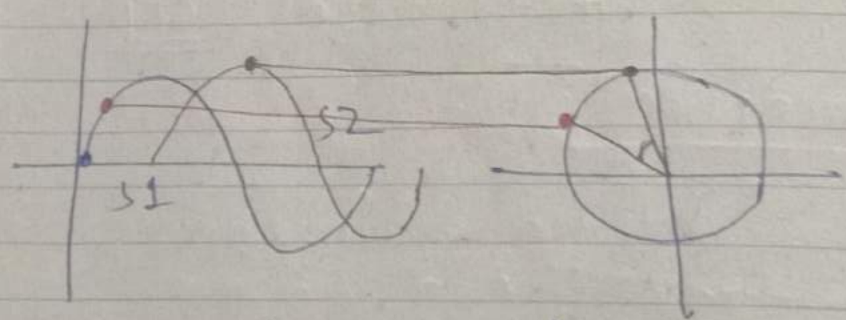
POC lec 5 (After midsem)

based on this fact

PHASE LOCKED LOOP (PLL)

* most extensively used type of demodulator

* Track the phase & frequency of incoming signal



Phase plane

If both have same freq. both move with same pace, & any point mapped onto phase circle by both signals S1 & S2 will have the "same phase difference"!!

USES OF

1. Fm
2. Sy
3. Clo
4. Ti

USES OF PLL

(1) Fm dimodulation

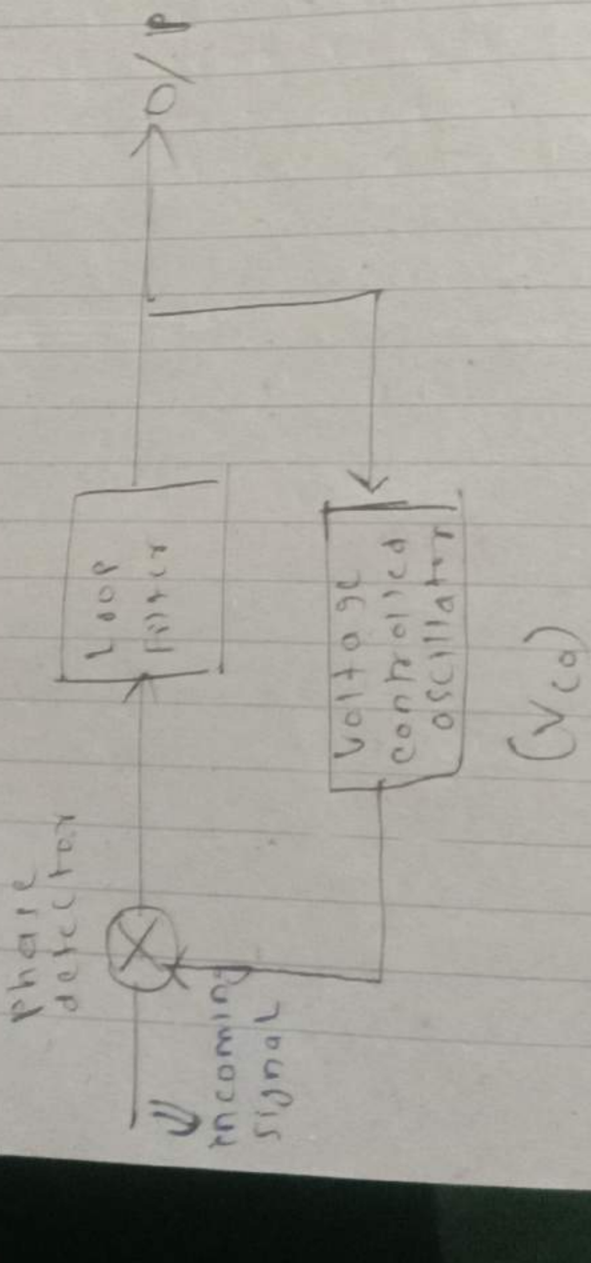
(2) Synchronous dimodulation of Am signal

(3) clock recovery system

(4) Timing recovery system

BASIC STRUCTURE OF PLL

PLL has 3 basic components



BASIC OPERATION OF PLL

* Typical feed back system, feedback tends to follow I/P signal.

* If diff. in feedback & I/P signal then FB signal brought close to I/P signal.

* We compare the phase of feedback signal with phase of incoming signal, If there is phase difference, then given to feedback circuit that changes the feedback signal so that phase difference minimized.

* V_{CO} adjusts its own frequency such that frequency & phase can track those of the I/P signal

Freq. of V_{CO}:

constanst of V_{CO}

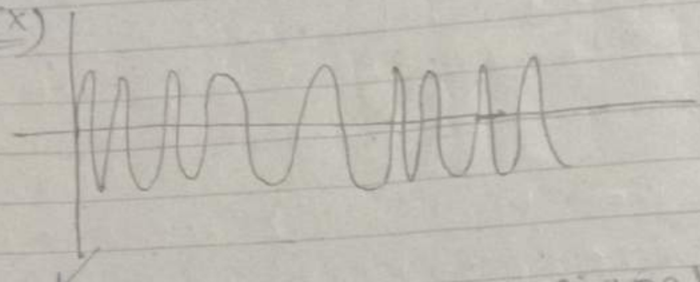
$$\omega = \omega_c + c\, \underbrace{e(t)}$$

free running frequency

error signal after loop filtering process

When $e(t) = 0$, $\omega = \omega_c$

Ex)



Let the incoming signal be FM signal

Let the loop be in locked condition,
→ freq of incoming & freq. of feedback signal is same

When the freq. of incoming signal changes, feedback circuit
→ becomes operational, change in frequency ⇒ "new frequency captured & locked"

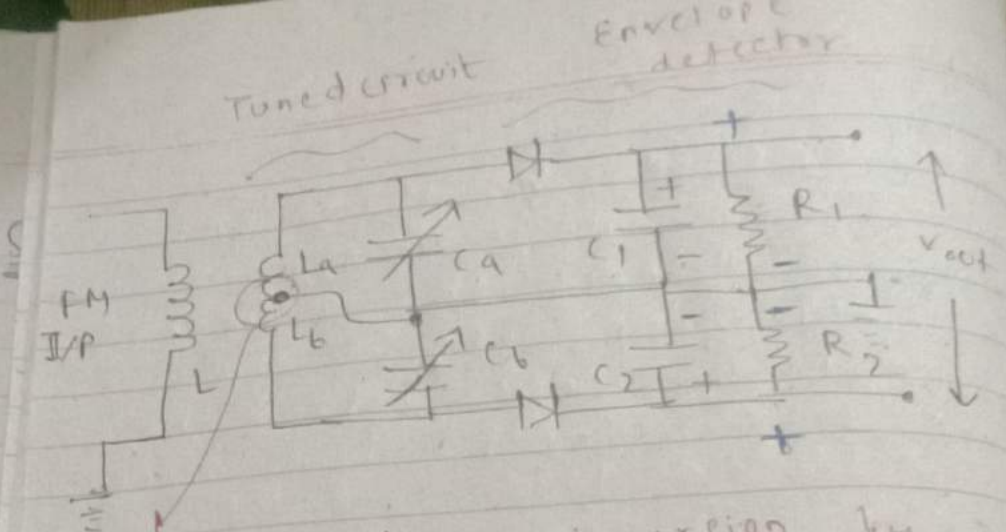
LOCK RANGE : V_{co} tracks the freq & Phase of incoming signal. A PLL can track the incoming frequency only over a finite range of frequency shift & the range of freq. shift = Lock range.

CAPTURE RANGE : If initially, the I/P and the O/P freq. are not close enough, then the loop may not acquire lock ~~the~~. ~~very~~

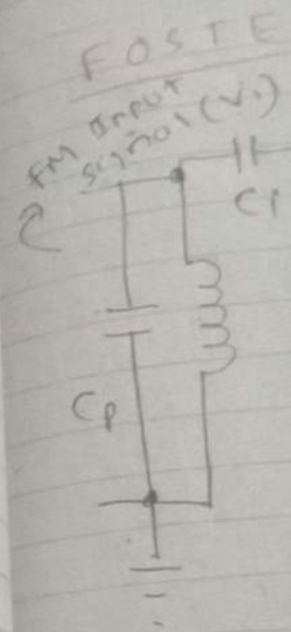
The range in which the lock can be achieved initially is called capture range.

BALANCED SLOPE DETECTOR

This circuit combines 2 circuits of individual slope detectors in parallel & FM I/P signal is fed 180° out of phase



Provides phase inversion by center tapping



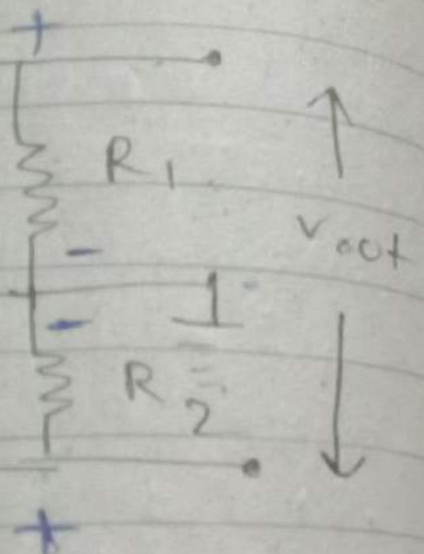
ADVANTAGE

- wider ~~...~~ range of frequency over which frequency characteristic linear

DISADVANTAGE

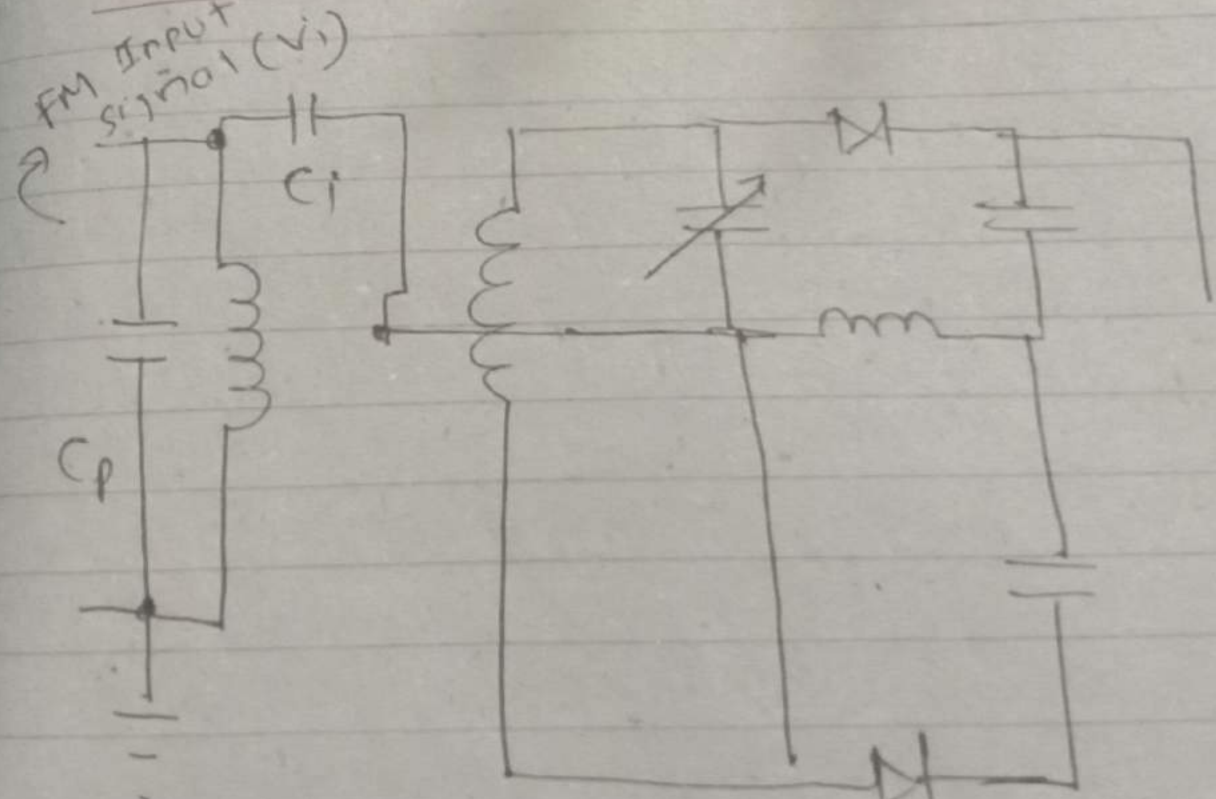
- difficulty in tuning
- Poor limiting (Amplitude variations cannot be limited with much efficiency)

lope
tector



n by

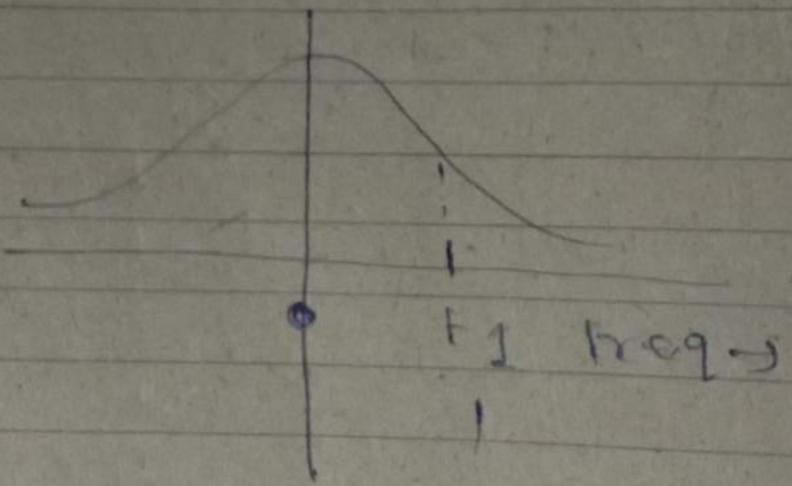
FOSTER - SEALEY DISCRIMINATOR



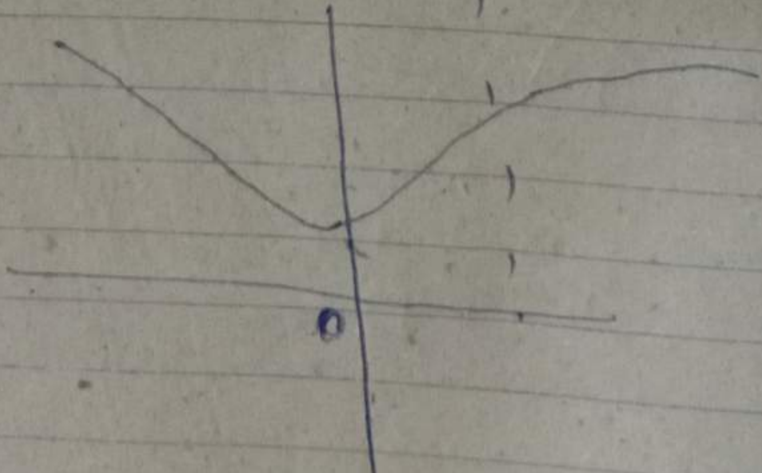
POC-Lec 6 (PRE EMPHASIS & DE-EMPHASIS)

→ Power spectral density

PSD of Audio signal



PSD of noise



upto f_1, $\frac{S}{N} > 1$ (desired)

→ beyond f_1, $\frac{S}{N} < 1$ (not desired)

For high freq

(1) Either signal power 'S' has to be increased

OR

(2) decrease the noise power

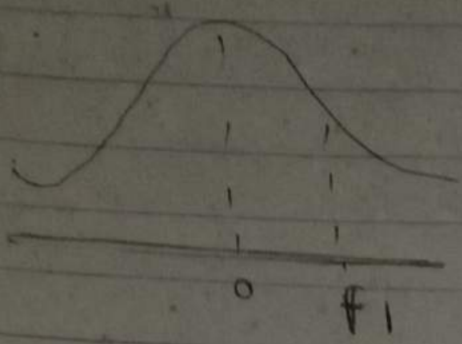
⇒ (The signal strength is increased artificially at high frequency
↳ PRE-EMPHASIS

PRE-EMPHASIS

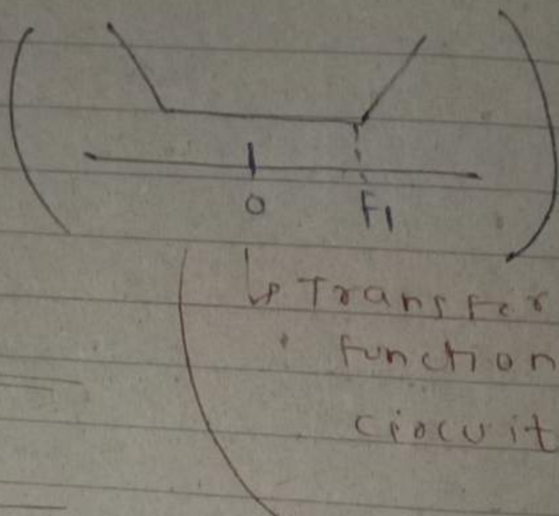
↳ process of increasing the strength of high frequency components of audio signal is called pre-emphasis

→ Pre-emphasis is done at the transmitter before modulation

CIRCUIT FOR PRE-EMPHASIS :

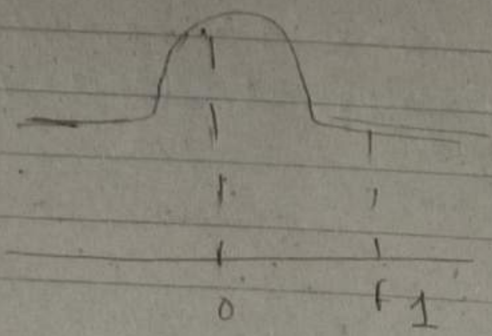


Pre-emphasis circuit

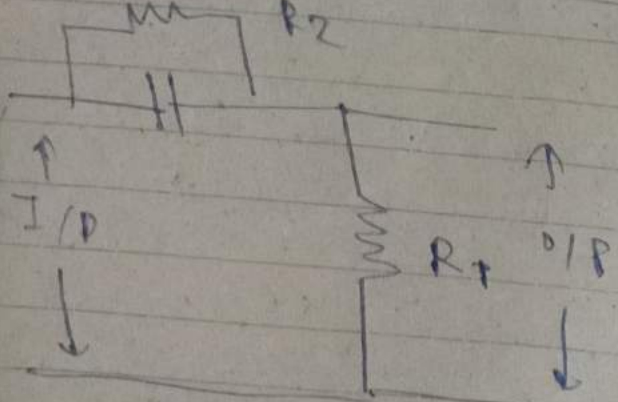


↳ Transfer function of circuit

→ Lead compensator circuit



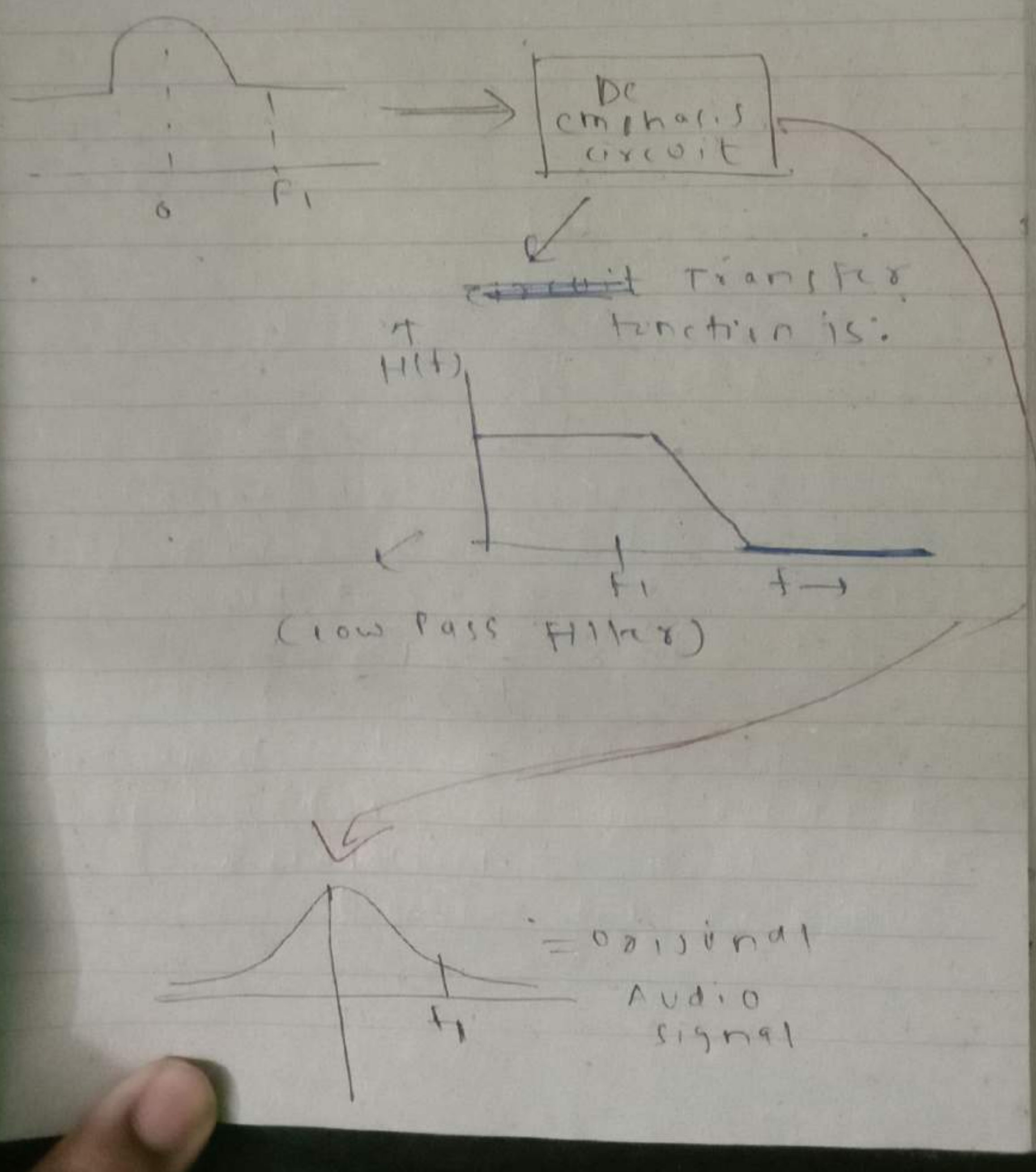
Lead compensator circuit



DE-EMPHASIS

At the receiver end after demodulation

→ Decreasing the strength of the signal of high frequency components of Audio signal



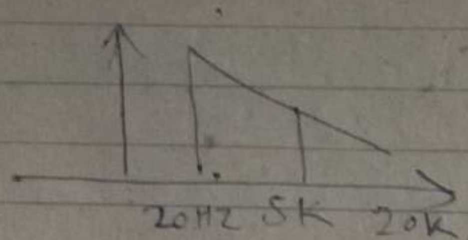
Q)
Why preemphasis & deemphasis are only needed in FM & not in AM?

Soln:

For AM:

BW = 10 kHz

fm = 5 kHz



~~signal~~

* Signal strength decreasing with frequency

* message signal lies in vicinity of 5kHz, the signal strength is appropriate in case of AM, Signal strength $\frac{S}{N} > 1$

* So in AM, only the low frequency of Audio signal is considered for transmission, so preemphasis & deemphasis not required

For fm:

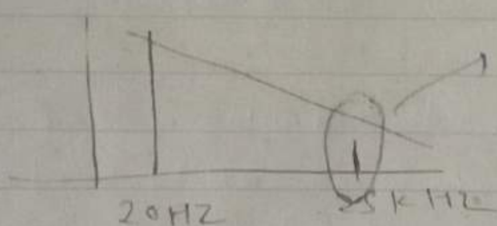
BW = 200 kHz

&

$2(\Delta f + f_m) = 200$ kHz

$\Delta f = 70$ kHz

$f_m = 25$ kHz



$\frac{S}{N} < 1$, signal strength less

&

therefore preemphasis & demphasis required in case of FM.

→ (FDM)

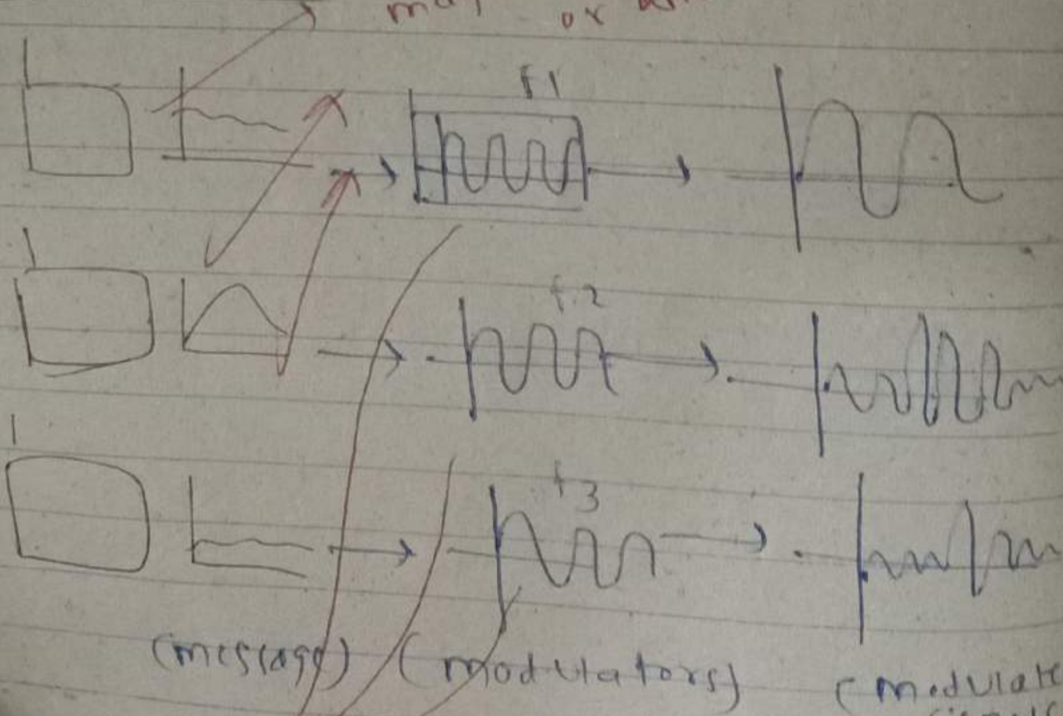
FREQUENCY DIVISION (MULTIPLEXING)

transmission of several signals simultaneously using/sharing the same resource

* In FDM, several signals are transmitted simultaneously by sharing a band of channel.

FDM Process (At transmitter end)

may be digital or analog



subcarriers for individual modulation

(FDM)

FREQUENCY DIVISION (MULTIPLEXING)

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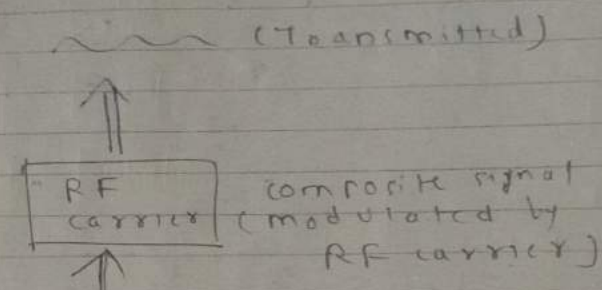
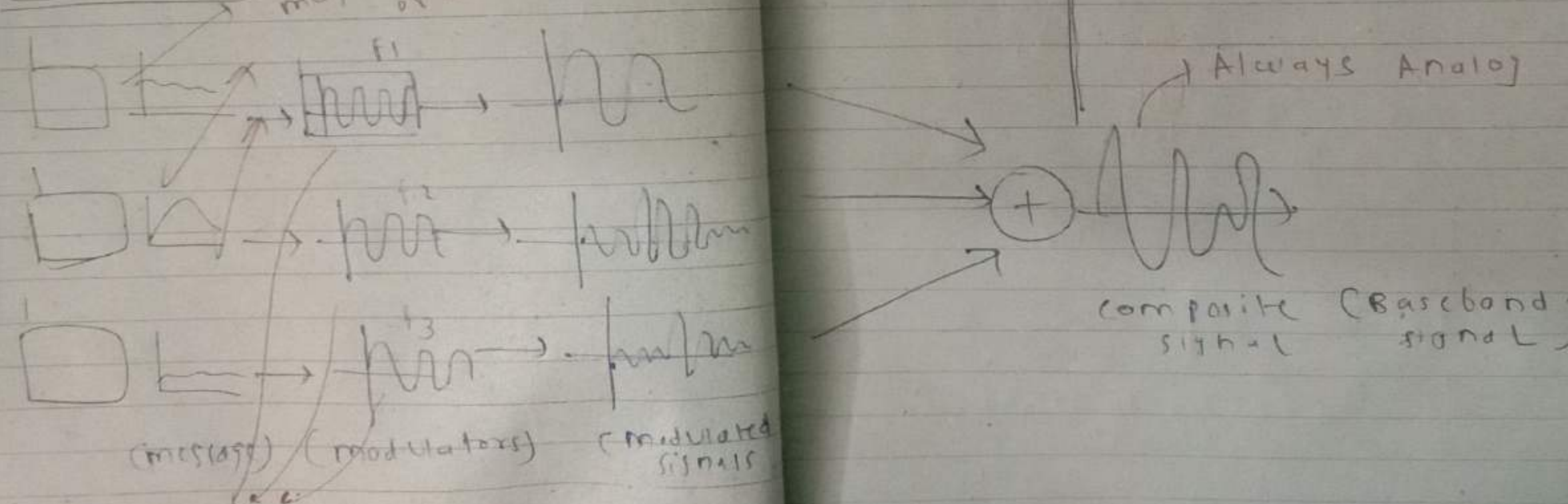


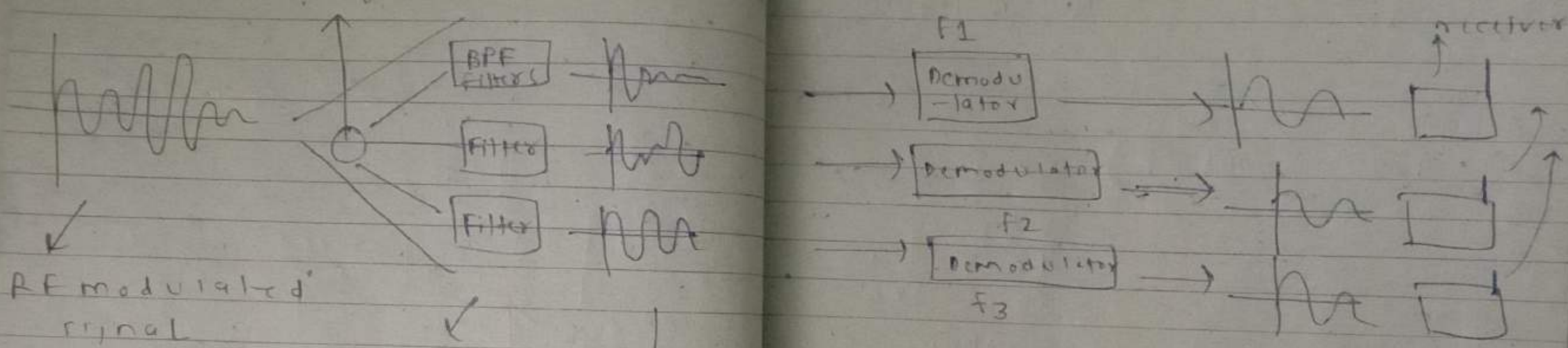


subcarriers for individual modulation

* subcarriers are adequately separated to avoid overlap/interference

At receiver end

RF carrier demodulation



RF modulated signal

individual BPF filter

modulated signal

individual demodulators with same carrier frequency as at the transmitter end

original message signals

POC lec 7

PHASE MODULATION

Phase of carrier signal is varied in accordance to the message signal

$c(t) = A_c \cos(2\pi F_c t)$ carrier before Phase modulation

$S_{PM}(t) = A_c \cos(2\pi F_c t + \phi)$ Phase deviation

Phase modulated signal (carrier signal after Phase modulation)

$\boxed{\phi(t) = K_p m(t)}$

Phase sensitivity constant

$$S_{pm}(t) = A_c \cos(2\pi f_c t + k_p m(t))$$

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

Maximum Phase deviation

$$\Delta\phi = \max(k_p m(t))$$

$$= \max(k_p A_m \cos(2\pi f_m t))$$

$$\boxed{\Delta\phi = k_p A_m}$$

Standard expression of PM

$$S_{PM}(t) = A_c \cos(2\pi f_c t + k_p A_m \cos(2\pi f_m t))$$

$$\boxed{k_p A_m = \text{modulation index } (\beta)}$$

also called "maximum phase deviation ~~[illegible]~~"

$\beta = K_p A_m = \Delta\phi$

modulation index

max. phase deviation

For FM:

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

dependent on message signal frequency

For PM:

$$\beta = K_p A_m = \Delta\phi$$

independent of ~~m(t)~~ message signal frequency

In PM,

"β & $\Delta\phi$ are independent of the message signal frequency variations".

$$S_{PM}(t) = A_c \cos(2\pi f_c t + \beta \cos(2\pi f_m t))$$

$$S_{FM}(t) = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$$

" general expression of FM & PM same, except 90° phase shift at message frequency component

" Bandwidth & power requirement in FM & PM are exactly same "

$$\boxed{BW_{PM} = 2(\beta+1) f_m = 2(\Delta f + 1) f_m}$$

$$\boxed{Power_{PM} = \frac{A_c^2}{2}}$$

$\Longrightarrow$

Q1) $S_{PM}(t) = 10\cos\left(2\pi\times10^{6} t + 6\sin 6\pi\times10^{3} t\right)$

(1) Find all the parameters of PM

(2) repeat above ~~for~~ after doubling the message signal frequency

Soln:

$$S_{PM}(t) = 10\cos\left(2\pi\times10^{6} t + 6\sin(6\pi\times10^{3} t)\right)$$

$$S_{PM}(t) = A_c\cos\left(2\pi f_c t + \cancel{D_1}\,\beta\sin(2\pi f_m t)\right)$$

(1)

$A_c = 10$

$f_c = 1\,MHz = 1000\,kHz$

$f_m = 3\,kHz$

$\beta = 6$

$BW = 2(\beta+1)f_m$

$= 42\,kHz$

$P_t = \dfrac{A_c^2}{2} = 50W$

(2) If $f_m = 6\,KHZ$

$$BW = 2(\beta+1) f_m$$

$$= 84\,KHZ$$

$$\beta = K_f A_m = 6$$

$$P_t = \frac{A_c^2}{2} = 50w$$

~~m(t) freq~~

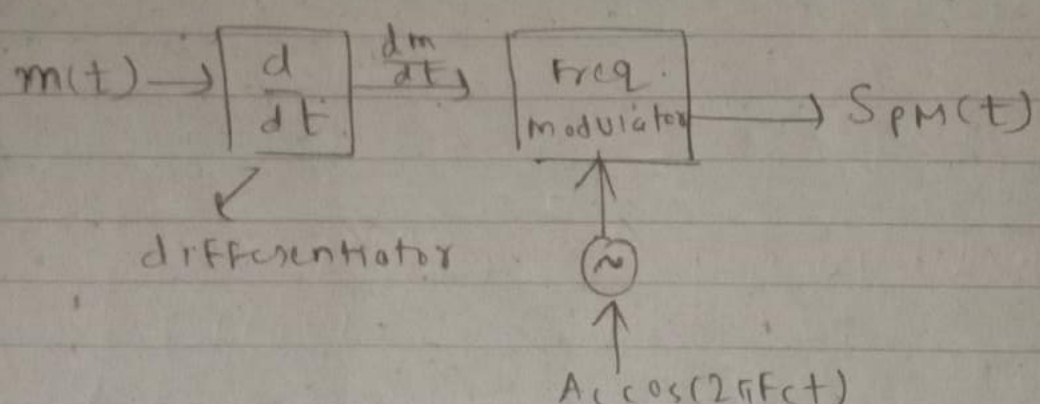
$$\boxed{m(t)\ \text{frequency} \underset{\text{doubles}}{\uparrow} = BW \underset{\text{doubled}}{\uparrow}}$$

GENERATION OF PM

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

$$S_{FM}(t) = A_c\cos(2\pi f_c t + 2\pi k_f \int m(t)\,dt)$$

we can generate from FM



Q2) An angle modulated signal is given as:

$$s(t) = \cos(2\pi(10^6 \times 2t + 30\sin 150t + 40\cos 150t))$$

Find max freq deviation & max phase deviation

Soln:

$$s(t) = \cos(2\pi\{ \quad \})$$

$$\phi_i(t) = 2\pi[2\times10^6 t + 30\sin 150t + 40\cos 150t]$$

↳ modulated angle

For FM: FM modulated

$$f_i = \frac{1}{2\pi}\frac{d}{dt}\phi_i(t) = f_c + K_f m(t)$$

$$= \frac{1}{2\pi}\frac{d}{dt}[2\pi(2\times10^6 t + 30\sin 150t + 40\cos 150t]]$$

$$f_i = 2\times10^6 + 4500\cos 150t - 6000\sin 150t$$

$$f_i = f_c + \underbrace{K_f m(t)}$$

↳ freq deviation

freq deviation

$= 4500\cos(150t) - 6000\sin(150t)$

max freq deviation

$= \max(A\cos(2\pi f_1 t) + B\sin(2\pi f_1 t))$

$\hookrightarrow \sqrt{A^2+B^2}$

$= \sqrt{4500^2+6000^2} = 7500\,Hz$

$\boxed{= 7.5\,kHz}$

For PM

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

phase deviation

$s(t) = A_c\cos 2\pi(2\times10^6 t + 30\sin(150t + 40\cos 150t))$

$k_p m(t) = 60\pi\sin 150t + 80\pi\cos 150t$

$\Delta\phi = \max(k_p m(t))$

$= ((60\pi)^2 + (80\pi)^2)^{1/2}$

$\boxed{\Delta\phi = 100\pi}$

PAM = Pulse amplitude modulation
PPM = Pulse position modulation
PWM = Pulse width modulation

lec 8 (UNIT-4)

DIGITAL REPRESENTATION OF ANALOG SIGNAL

modulation

- Analog (continuous)
  - Amplitude modulation
    - AM
    - DSB-SC
    - SSB-SC
    - VSB
  - Angle modulation
    - FM
    - PM
- Digitial
  - Basebandl Pulse modulation
    - Pulse Analog
      - PAM
      - PPM
      - PWM
    - Pulse digitial modulation
      - PCM
      - DPCM
      - DM & ADM
  - Band pass modulation
    - ASK
    - PSK
    - FSK

PCM: Pulse code modulation
DPCM: differential PCM
DM: Delta modulation
ADM: Adaptive DM

In Analog mod,

$$\left.\begin{matrix} m(t) \\ c(t) \end{matrix}\right\} \rightarrow \text{both Analog / continuous}$$

In digital mod,

Pulse modulation

↳ $c(t)$ = pulse train
(pulse wave).

PULSE ANALOG MODULATION

* $c(t)$ = periodic pulse train

* Amp., position & width of the carrier pulse train is varied in accordance to samples of continuous $m(t)$.

PULSE DIGITAL MODULATION

$m(t)$ = discrete in both time & Amplitude

$c(t)$ = periodic pulse train
↳ sequence of coded pulses

* This type of modulation does not have any continuous wave counter part.

* PCM is most common.

4 predominat methods of pulse modulation

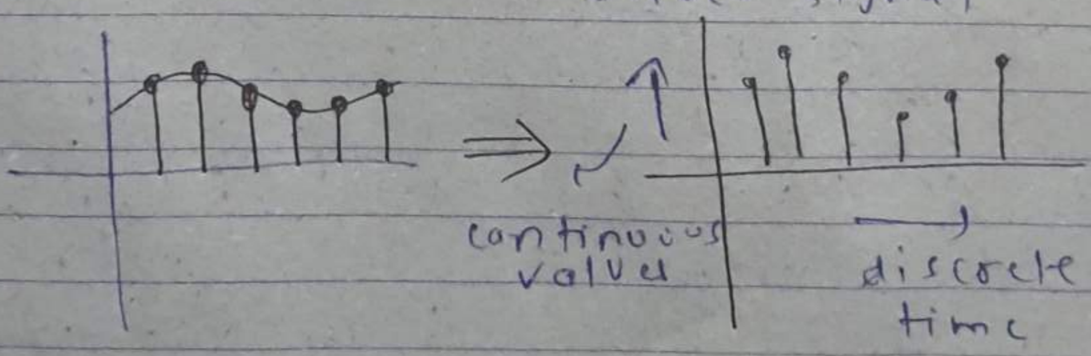
↳ PWM
↳ PPM
↳ PAM
↳ PCM

ANALOG TO DIGITAL SIGNAL

3 step process:

(1) Sampling ⟹ discrete time continuous valued signal

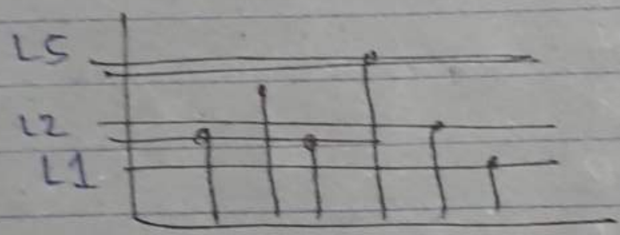
(2)



(2) **Quantization**

(rounding of infinite sampled values to a finite no. of values)

→ Discrete time & discrete Amplitude



} no. of levels fixed

(3) **Encoding**

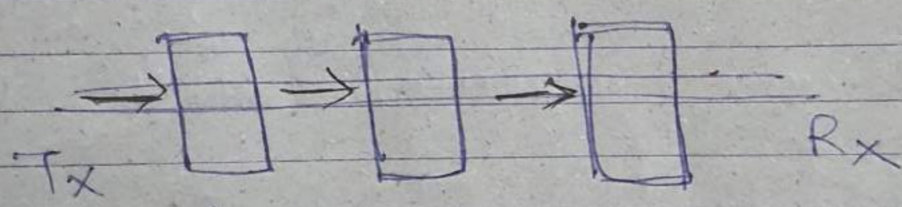
sequence (ones & zeros) is assigned to different O/P values of quantizer

(DC)

ADVANTAGES OF DIGITAL COMMUNICATION

(1) Noise immunity ⇒ can withstand channel noise & distortion ~~th~~ much better than Analog ~~Communic~~ communication

(2) Regenerative repeaters in DC



* receive the signal
* regenerate digital signal.
* retransmit

↳ advantage
noise overcome

(3) Digital hardware implementation is flexible.

(4) coding can be done to reduce the error rate

(5) multiplexing of digital signals is easier

(6) Digital signal storage is relatively easy & inexpensive

(7) Reproduction of digital signal is reliable

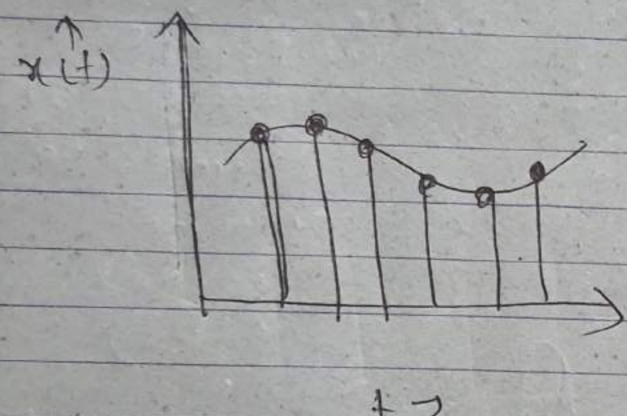
(8) cost of digital hardware reduces to $\frac{1}{2}$th in every 2-3 years and

performance is getting doubled

Lec 9

SAMPLING THEOREM

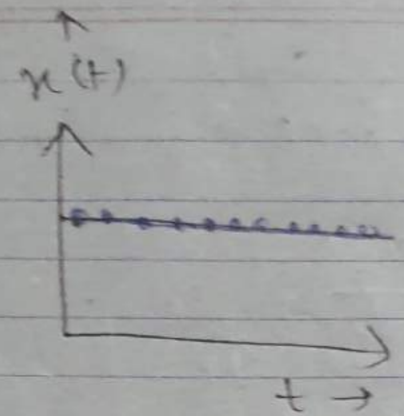
All the latest signal processing & digital comm. concepts are based on the validity of sampling theorem



→ A continuous time signal can be sampled & recovered back from the knowledge of the samples only if sampling frequency $\boxed{f_s \geq 2 f_m}$

where f_m is max. frequency is present in the message signal

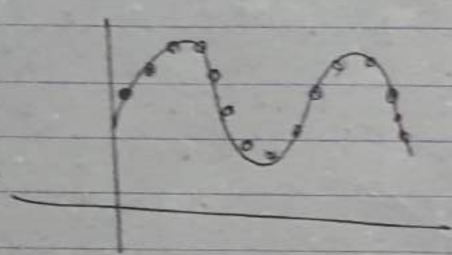
Ex1)



DC signal

"0 frequency"

Ex2)



max freq $= f$

let $f_s = 11f$

↓

11 samples in 1 time period (cycle)

↓

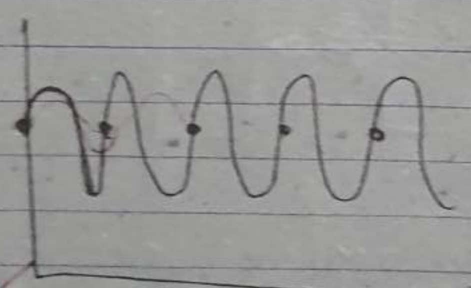
if $f = 1\,kHz$

$f_s = 11\,kHz$

↓

11000 samples/sec

Ex3)



↓

Let $f = 3\,kHz$

$f_s = f = 3\,kHz$

↓

1 sample per cycle

→ not desirable scenario, since hard to reconstruct back the message signal

<u>Nyquist frequency</u> $= 2 f_m$

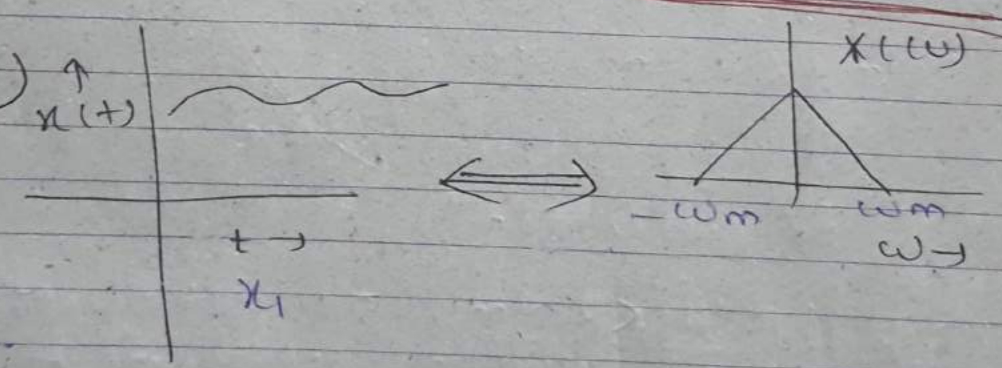
<u>Nyquist rate</u> = minimum sampling rate

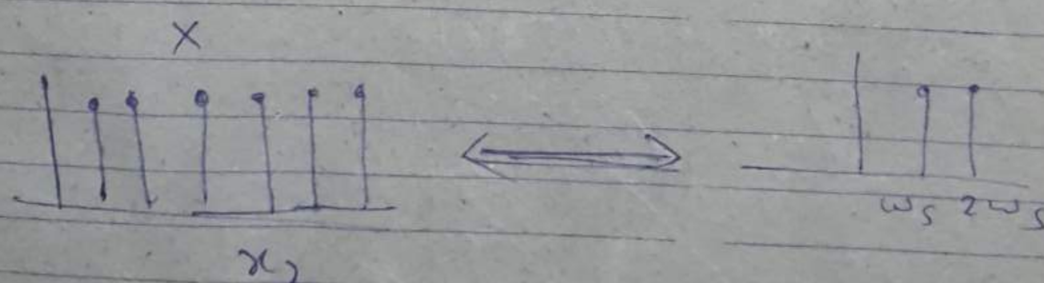
$= \frac{1}{2}$ (sampling rate)

Q) What happens when $f_s < 2 f_m$?

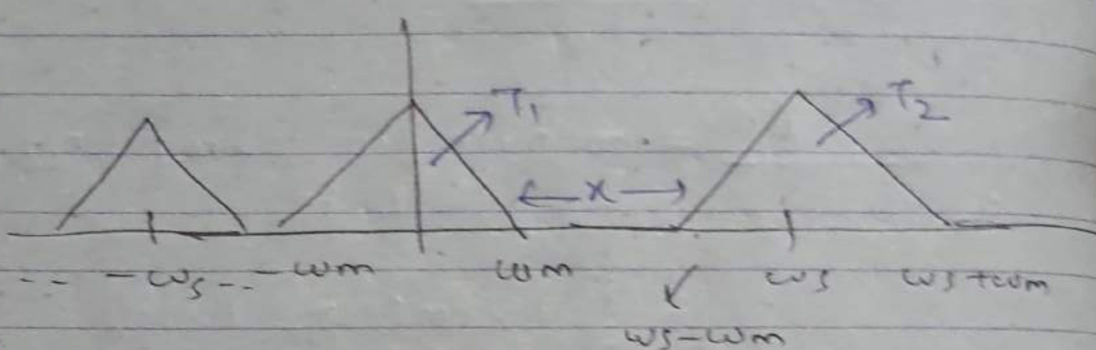
contributes to "<u>Aliasing Error</u>"

Ex)





$$x_1 \cdot x_2 = X_1(\omega) \overset{\text{convolution}}{*} X_2(\omega)$$



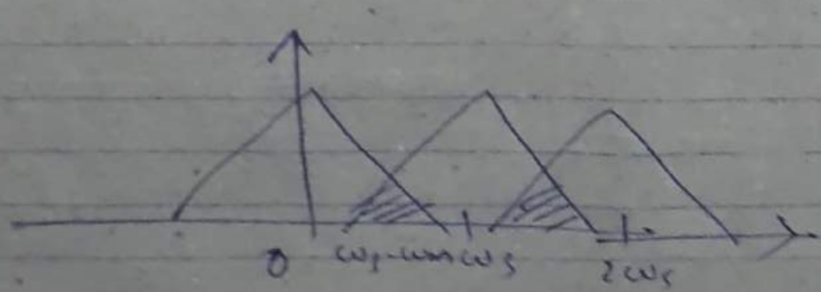
If $\omega_s \gg 2\omega_m$, sufficient guard band between T_1 & T_2

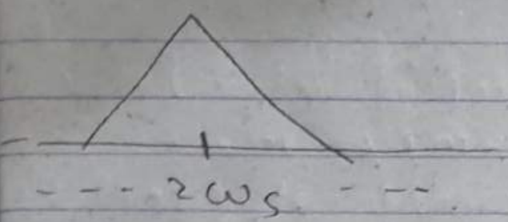
that is,

$\omega_m + x + \omega_m = \omega_s$

" no distortion "

Now if $\omega_s < 2\omega_m$, no sufficient guard band b/w the spectrums & they will overlap.





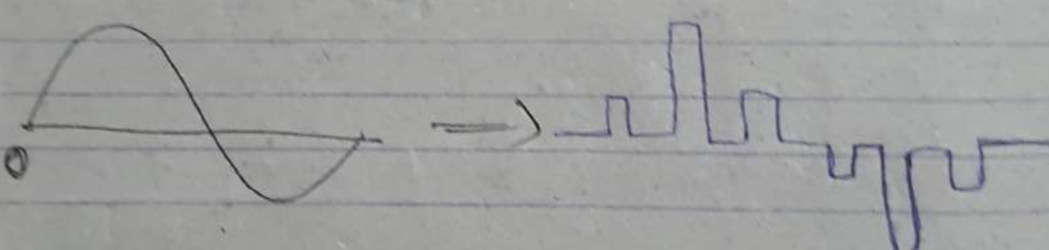
→ distortion impossible to get back the m(t))

→ this type of signal is called aliasing error (if $f_s < 2f_m$)

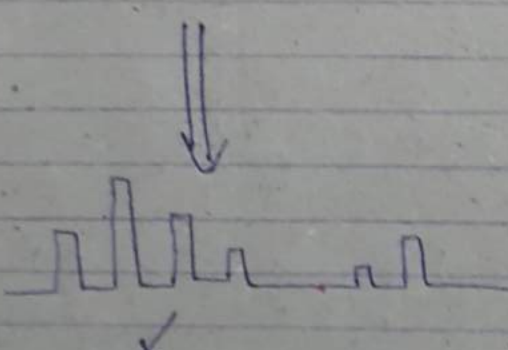
PULSE AMPLITUDE MODULATION (PAM)

* Simplest form of pulse modulation

* In PAM, signal sampled at regular intervals & each sample made proportional to amplitude of signal at the instant of sampling.



(Double polarity PAM signal)



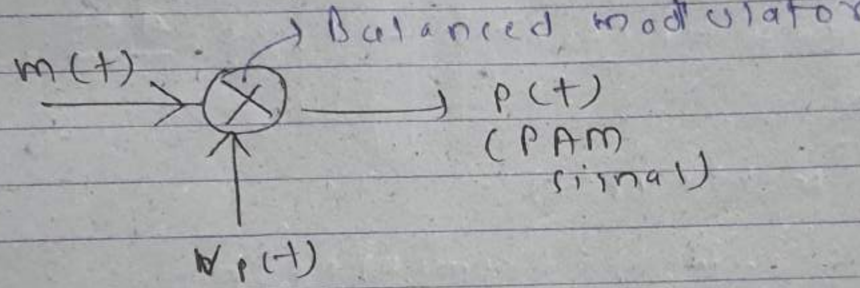
Single polarity PAM

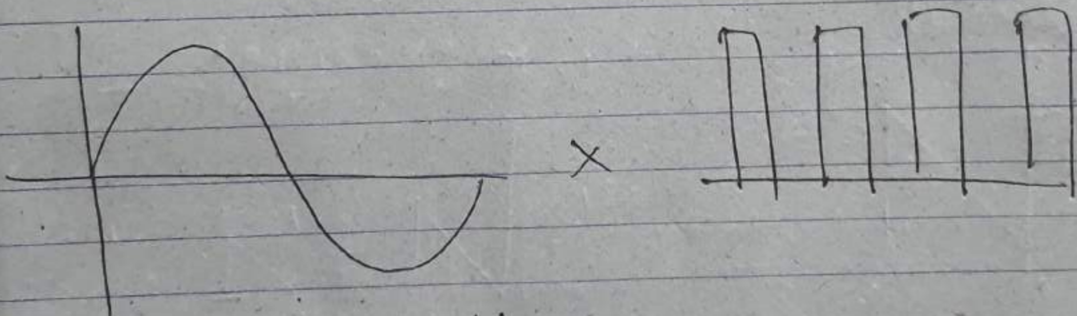
"A fixed DC ~~signal~~ signal given to Double polarity PAM to make negative pulses positive" modulating signal = message signal

NATURAL SAMPLING

m(t)

carrier pulse train $V_p(t)$ multiplied with m(t) to get PAM.

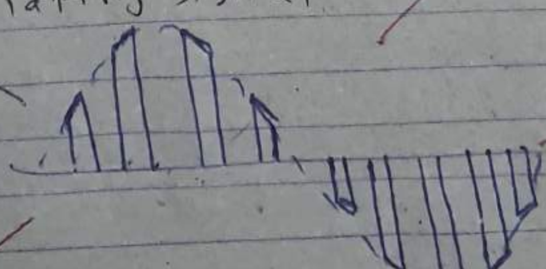




⇓

Top follows the shape of modulating signal

top part of pulses not flat



top part ~~not~~ same as natural m(t)

Also called "natural PAM"

Fourier series representation of pulse train

$$= a_0 + \sum_{n=1}^{\infty} a_n \cos\left(\frac{2\pi n t}{T_s}\right)$$

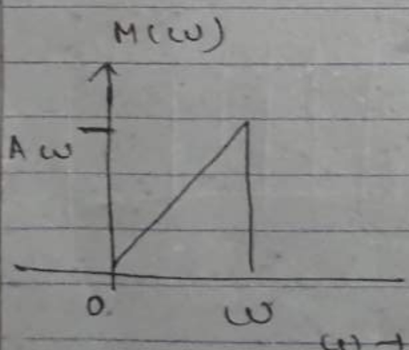
$$V_p(t) = a_0 + a_1 \cos\frac{2\pi t}{T_s} + a_2 \cos\frac{4\pi t}{T_s} + \dots$$

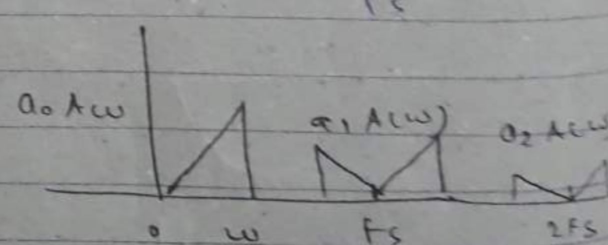
PAM

$$P(t) = m(t) \times v_p(t)$$

$$= a_0 m(t) + a_1 m(t) \cos\frac{\pi t}{T_s}$$

$$+ a_2 m(t) \cos\frac{4\pi t}{T_s} - -$$





spectrum of m(t)

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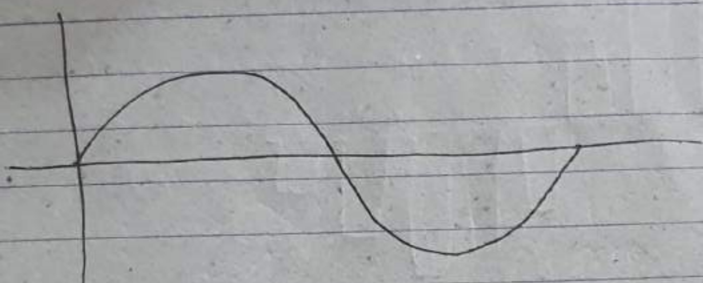
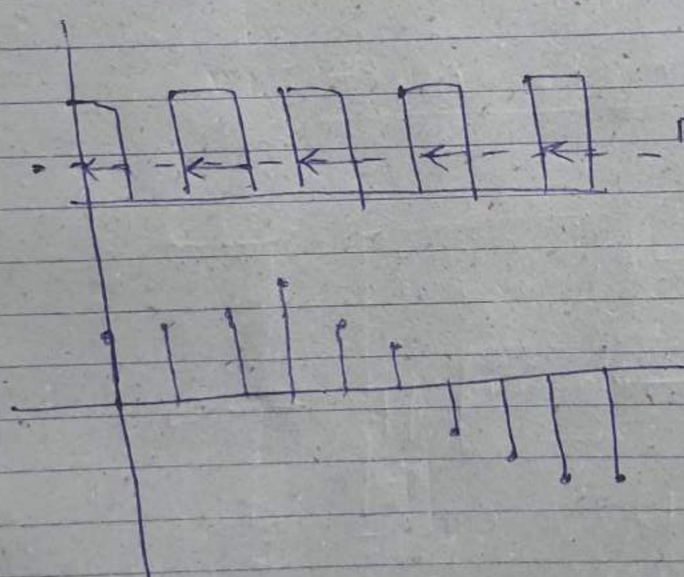
FLAT TOP ~~SIG~~ SAMPLING (~~FA~~ FLATTOP PAM)

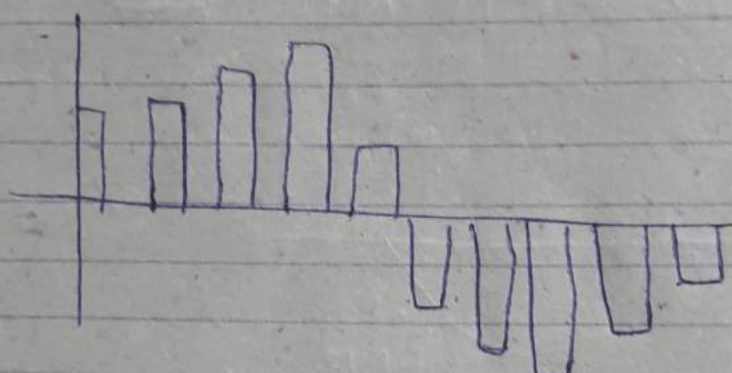




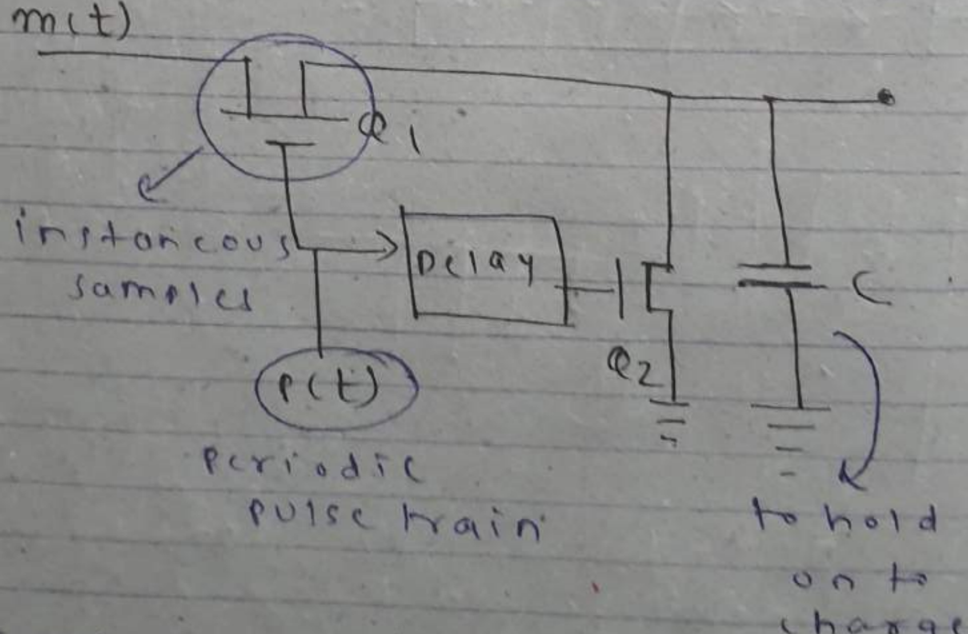
pulse width reduced to impulse, we get flat top

pulse energy will be small, hence cannot be used for communication.

first generate instantaneous sampling then hold it for certain time period to generate flat-top sampling

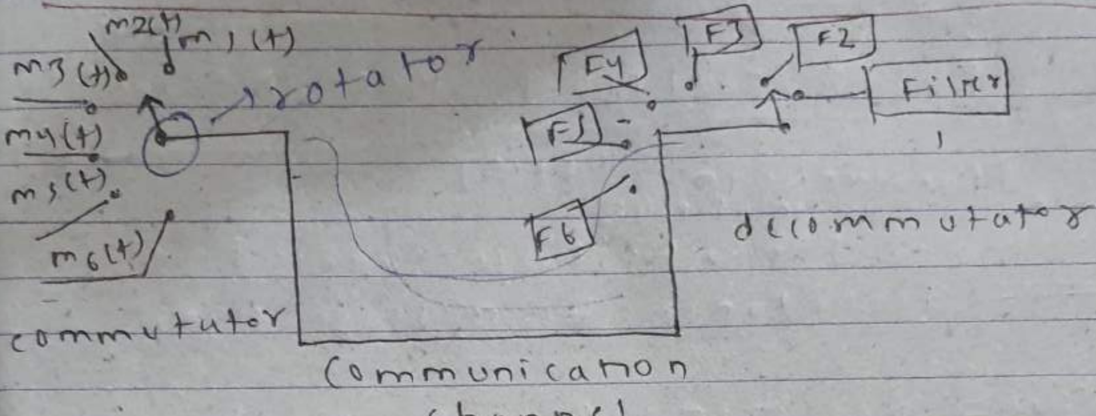


Sample & hold circuit

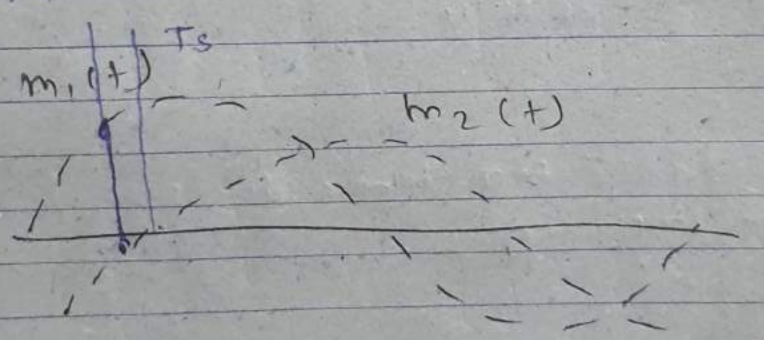


First sample, then hold charge for some instant

PAM & TIME DIVISION MULTIPLEXING



~~rotator~~ rotator 1 & rotator 2 ~~s~~ synchronized



First we sample $m_1(t)$, $m_2(t)$ at "Ts intervals"

PULSE WIDTH MODULATION (PWM)

& PULSE POSITION MODULATION (PPM)

↳ Randomness in width of pulse

PWM = width of pulse varied in accordance to message signal

PPM = position of each fixed width pulse is varied in accordance to message signal

↳ randomness in position of pulse

↳ not suitable for time division multiplexing

Application:

① PWM finds in controlling speed of DC motor

PWM &
Pulse tim

PWM Gen

Input signal

Sawtooth signal

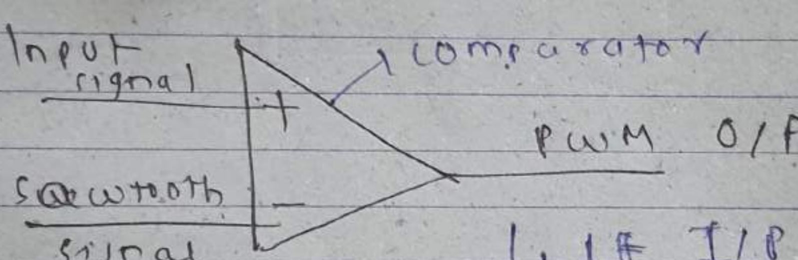
PPM Gener

PWM

Inverter

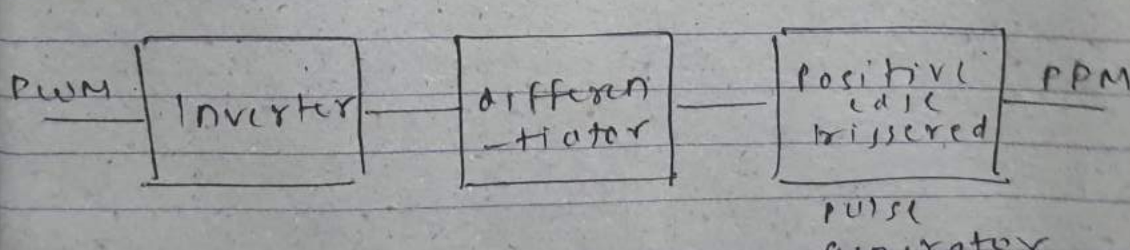
PWM & PPM together called Pulse time modulation.

PWM Generation



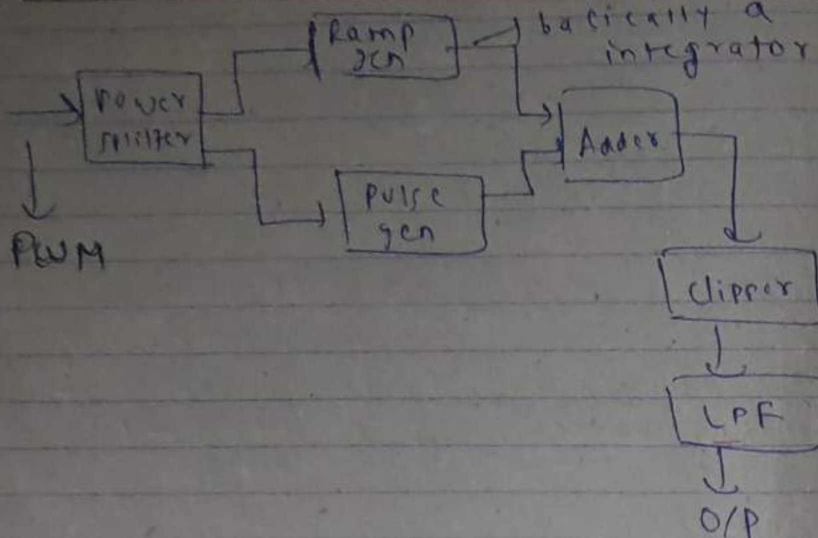
↳ If I/P signal more, O/P = +V
Sawtooth signal more, O/P = −V

PPM Generation



ELC II

DETECTION OF PWM & PPM

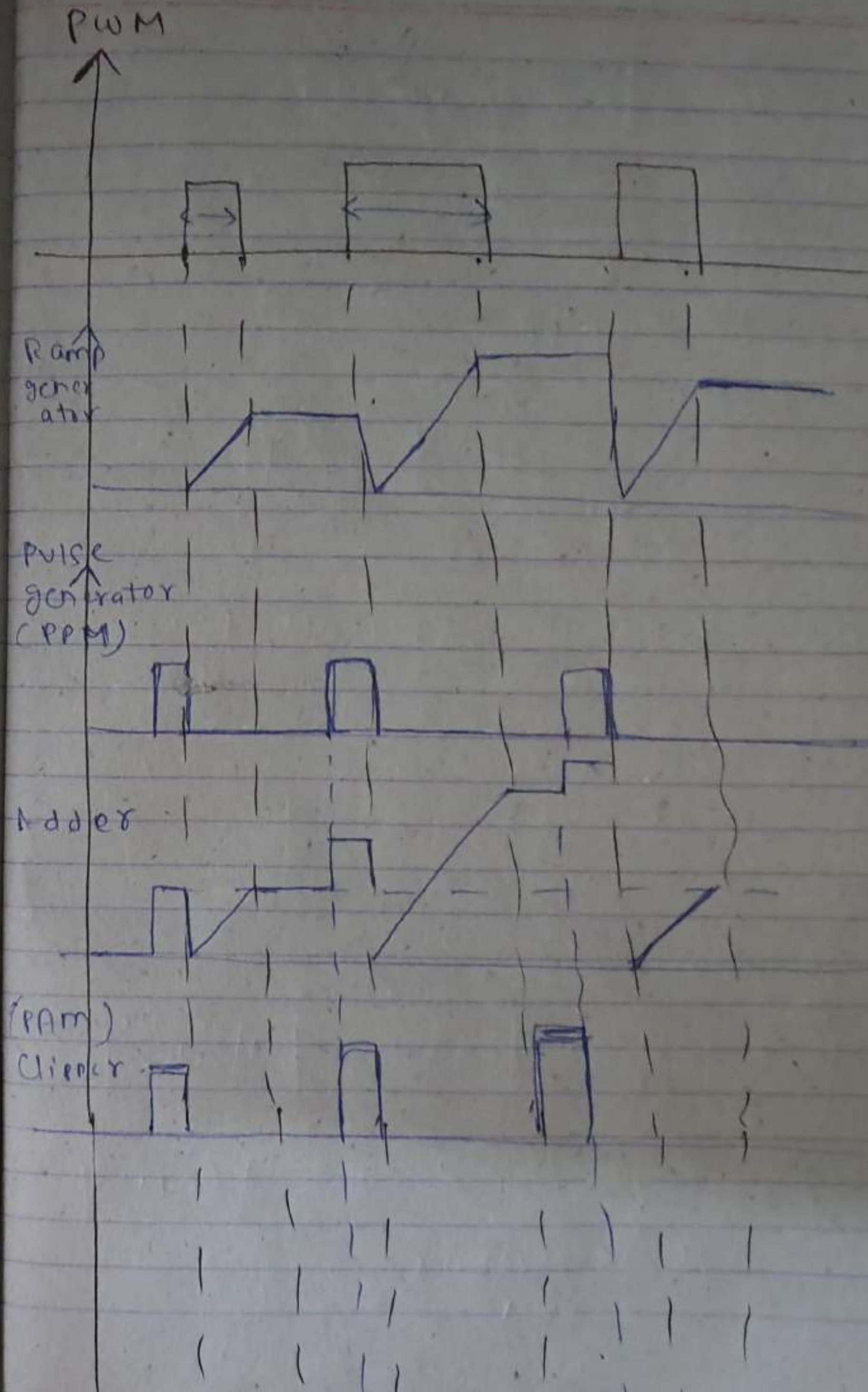


Working:

Pulse gen = to generate PPM

generate PPM at each rising edge

Clipper = clips the DC component & convert this into a PAM signal



PAM → LPF → m(t)

* Overall summary

PWM/PPM → PAM ↓ LPF → m(t)

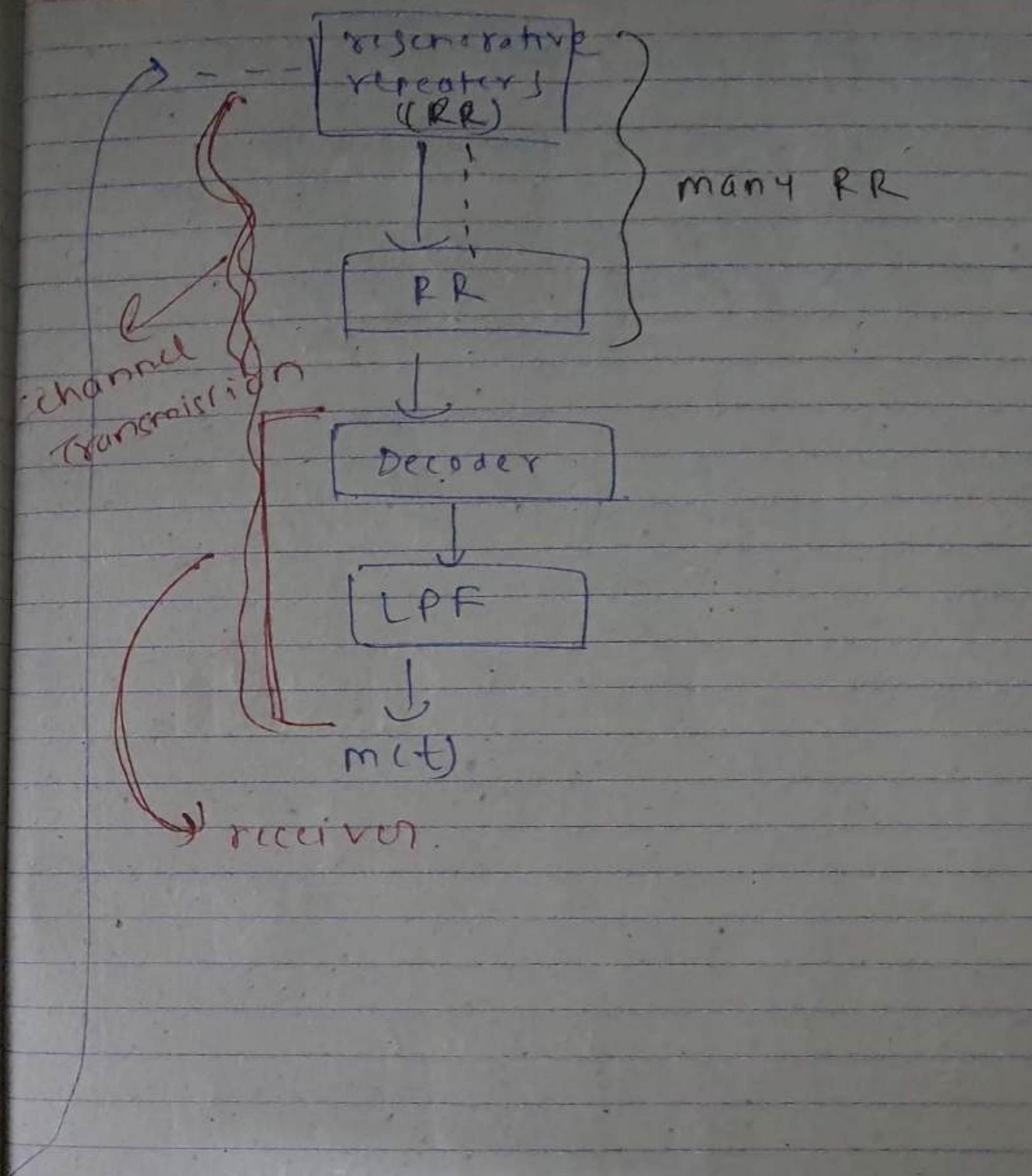
PULSE CODE MODULATION (PCM)

purpose of PCM is to convert analog information into binary information.

Block Diagram

m(t) → LPF → sampler → Quantizer → encoder ~~coder~~ ↓ PCM

Transmitting end.



LPF: (1) Band limit the signal
(2) avoids very high freq
(3) Avoids aliasing
(Anti-aliasing filter before the sampler)

Sampler: To obtain the instanteous sample values of m(t)

Quantizer: rounding of the sampled value to the nearest quantization level

encoder: To represent each quantized ~~level~~ value to a unique binary code.

Importance of quantizer

$f_m = 1 MHz$

$F_s = 2 \times 10^6$ samples/sec

these many large samples are to be taken to encoder ~~quantizer~~, and encoding them very different if quantizer does not used & quantize (limit) the samples

$\boxed{L = 2^n}$ n bits for the sample

no of quantization levels

Q1) if $L = 4$, $n = 2$
2 bits

$$\begin{pmatrix} 0\ 0 \\ 0\ 1 \\ 1\ 0 \\ 1\ 1 \end{pmatrix}$$

RR: eliminate the channel noise and regenerate fresh copy of the original signal.

Decoder: ~~encoding process~~ decode the pulse coded waveform

LPF: / Reconstruction filter:

To filter & read back the original signal

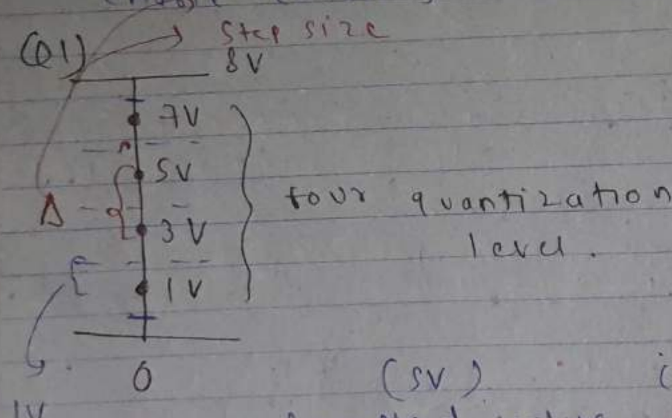
Bit rate of PCM

= no. of bits/sample × sampling rate

$= \boxed{n_b \times f_s}$

QUANTIZATION PROCESS

* rounding of the sample value to limit no. of samples to enable encoding process.



(SV) Sampled value		(QE) Quantization Value
0.8 V	⟶	1V
2.2 V	⟶	3 V
7.9 V	⟶	7 V
7 V	⟶	7 V

S.V = Sampled value
Q.E = Quantized value

Quantization error (S.V − Q.E)

1 − 0.8 = 0.2

0.8

0.9

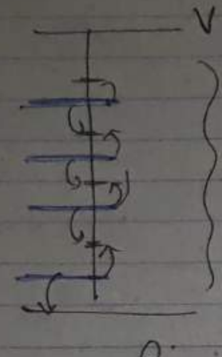
0

* maximum value of Quantization Error

$\boxed{\max(Q_E) = \Delta/2}$ } Δ = step size

lec 12

Quantization



range divided into L no. of equal of steps & middle point of each step = quantized level

L = no. of steps = no. of quantization level

$max(Q_E) = \Delta/2$

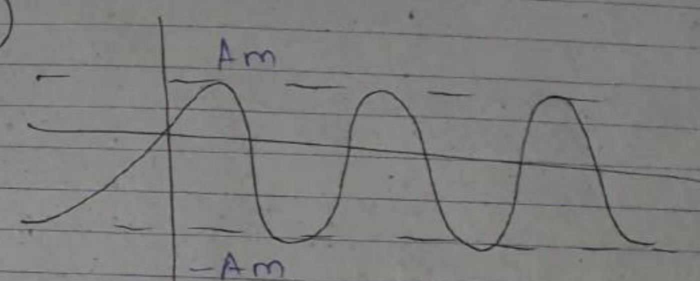
If n = no. of bits per sample

L = no. of quantization level

$\boxed{L = 2^n}$

$\Delta = \left(\frac{V_{max} - V_{min}}{L}\right)$ → dynamic range (range in which samples = distributed)

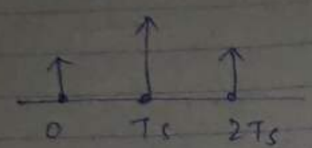
Ex)



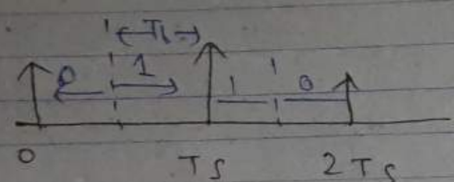
* $\Delta = \frac{Am + Am}{L} = \frac{2Am}{2^n}$

* $\boxed{Q_E = \text{Sampled valued} - \text{Quantized Value}}$

T_s = sampling duration



Bit duration



$n = 2$ bits / sample

$$\boxed{T_b = T_s/2}$$

For n, no. of bits,

$$\boxed{T_b = \text{Bit duration} = T_s/n}$$

Bit rate (Bits/second) $= \dfrac{\text{bits}}{\text{sample}} \times \dfrac{\text{sample}}{\text{second}}$

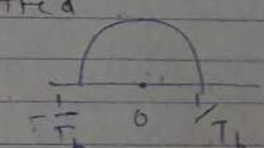
$$\boxed{R_b = n \cdot f_s}$$

$$\boxed{R_b = \frac{n}{T_s}}$$

MAX TRANSMITTER BW

① 1 bit is Transmitted

$BW = 1/T_b$



② 2 bits are transmitted

$BW = 1/2T_b$

③ 3 bits are transmitted

$BW = \dfrac{1}{3T_b}$

* $BW = \frac{R_b}{2}$, In practical scenario, atleast 2 bits are transmitted in a bunch,

It can easily transmit signals having bandwidth $\frac{R_b}{3}, \frac{R_b}{4}, \ldots \frac{R_b}{n}$

Q1) $m(t) = 10\cos(8\pi \times 10^3 t)$ is transmitted through 4 bit PCM. Find all the parameters?

Soln

$n = 4$ bits/sec

$m(t) = 10\cos(8\pi \times 10^3 t)$

$A_m = 10$

$f_m = 4kHz$

$L = 2^n = 16$ quantization level

Step size $(\Delta) = \frac{V_{max} - V_{min}}{L}$

$= \frac{10 - (-10)}{L}$

$= \frac{2 \times 10}{16}$

Sampling freq $(f_s) = 2f_m$

$= 2 \times 4k = 8k$

$(Q_e)_{max} = \pm \Delta/2 = \pm 20/32$ volts

bit rate $(R_b) = nf_s$

$= 4 \times 8 = 32k$

bit duration $(T_b) = 1/R_b$

max. $BW = R_b/2 = 16k$

Q2) $m(t)$ of $A_m = 20V$ & $f_m = 5kHz$ is transmitted through 256 level PCM system.

Sampling rate = 25% higher than Nyquist rate. Find all the parameters?

Soln $A_m = 20V$

$f_m = 5kHz$

$L = 256$

$f_s = NR + 25\,NR$

$L = 2^n$

$n = 8$ bits / sample

$NR = 2f_m = 10\,kHz$

$f_s = 10K + \frac{25}{100} = 12.5K$

$= 1.25 \times 10 = 12.5\,kHz$

$\Delta = \frac{V_{max} - V_{min}}{L}$

$= \frac{2 \times 20}{256} = 0.15625\,V$

$(Q_e)_{max} = \pm \Delta/_2$

$R_b = n F_s$

$T_b = 1/R_b$

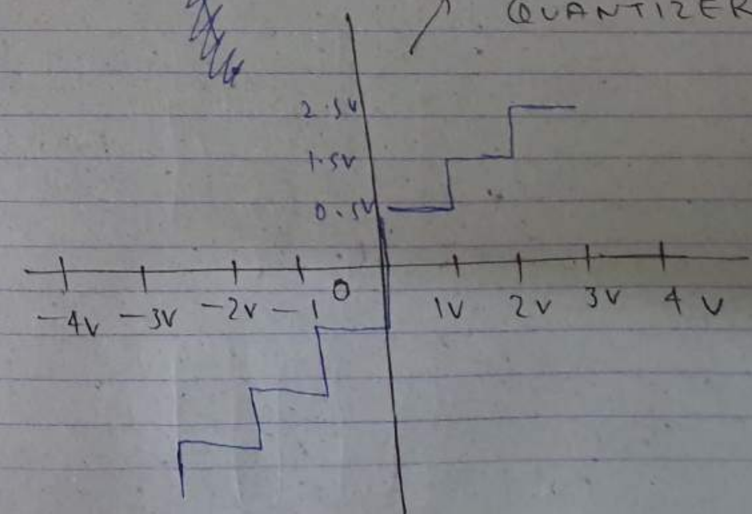
$BW = R_b/_2$

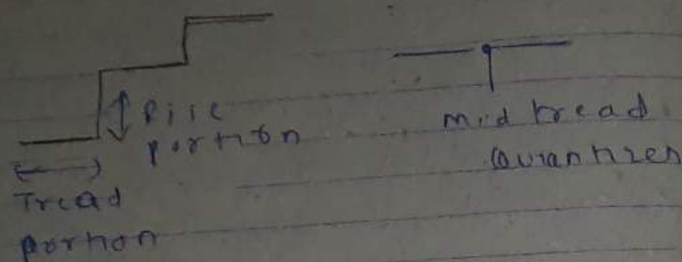
Q3) A sinusoidal msg signal is transmitted through PCM where $(Q_e)_{max}$ can be atmost 2% of peak to peak amplitude of msg signal. Find the no. of bits / sample required?

ECC13

QUANTIZER CHARACTERISTICS

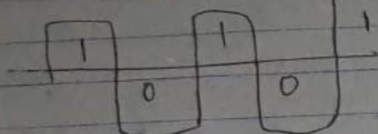
MID-RISE QUANTIZER



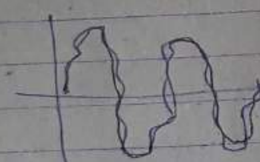


REGENERATIVE REPEATERS

→ threshold comparator



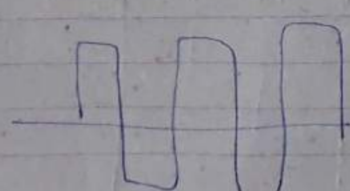
⇒

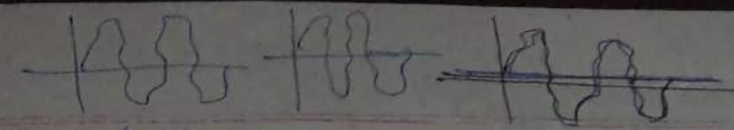


↓

Threshold comparator (RR)

↓

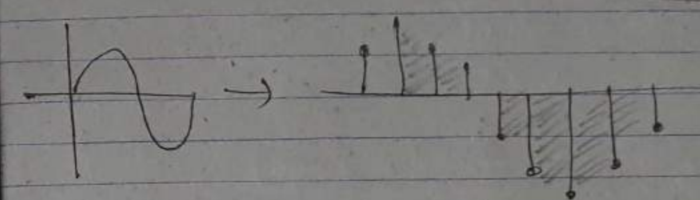




no. of regenerative repeaters

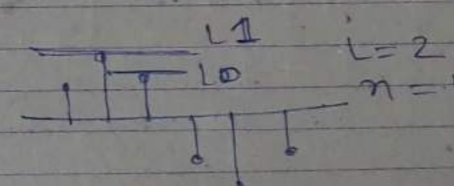
1. ~~decreases~~ depend on distance b/w the transmitter & receiver
2. Quality of channel

After



↓

L0 = level 0
L1 = level 1



↓ electrical representation in electrical waveforms of binary signals

* Line coding Techniques

Line coding techniques

① ON-OFF

0 → 0V
1 → +ve

② NRZ (non return to zero)

0 → -ive
1 → +ve

③ RZ (return to zero)

0 → 0V

$1 \rightarrow \begin{cases} T_b/2 \ (+ve) \\ T_b/2 \ 0V \end{cases}$

④ differential encoding

0 → complement of prev. o/p

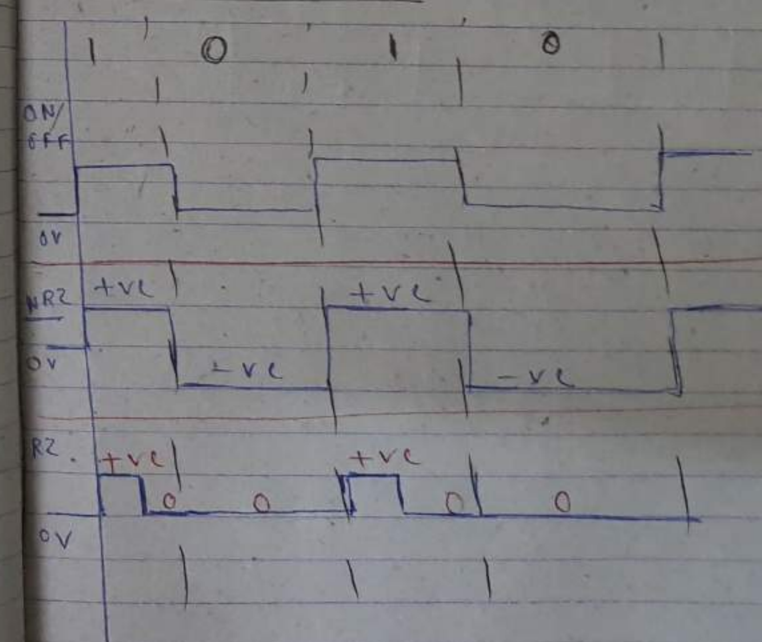
1 → same as previous o/p

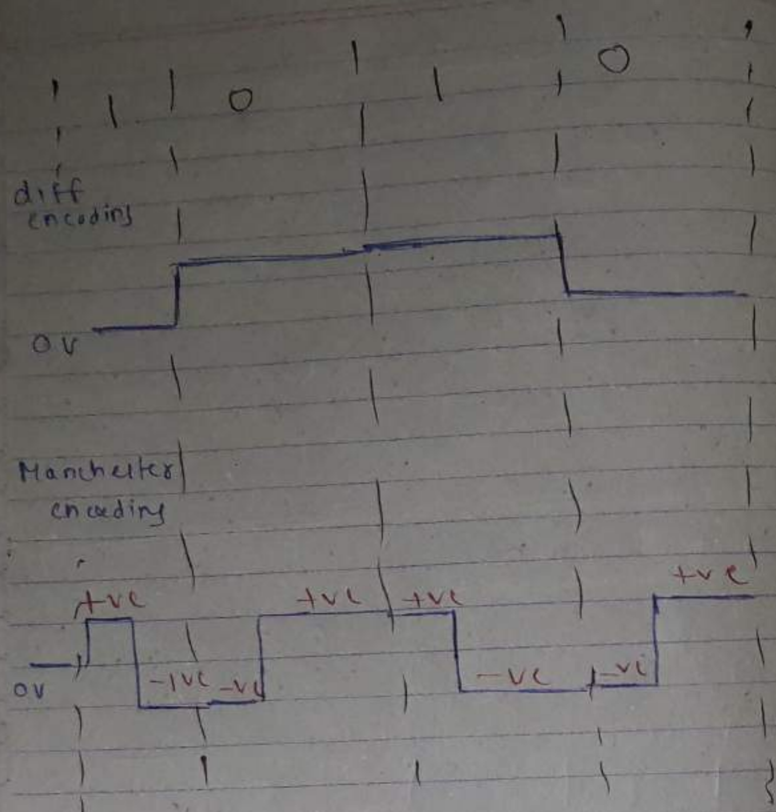
⑤ Manchester encoding

$0 \rightarrow \begin{cases} T_b/2 \ (-ive) \text{ first half} \\ T_b/2 \ (+ive) \text{ for next half} \end{cases}$

$1 \rightarrow \begin{cases} T_b/2 \ (+ve) \\ T_b/2 \ (-ive) \end{cases}$

let Bit sequence:





NOISES IN PCM System

① channel noise

② Quantization noise

Channel noise can be eliminated with help of regenerative repeaters

Quantization noise

$$[Q_e]_{max} = \Delta/2$$

By reducing step size (Δ), the quantization error will reduce.

→ more no. of bits / sample

if $L = 4$, $n = 2$ bits/sample

$L = 256$, $n = 8$ bits/sample

if $n \uparrow$, BW reqd $\uparrow$

To decrease Δ, value of n has to be ~~decreased~~ increased

$$n\uparrow \Rightarrow \Delta\downarrow \Rightarrow Q_e\downarrow$$

But value of n cannot be increased to a very large value.

$$n\uparrow \Rightarrow BW\uparrow = n\,f_s/2$$

n should be such that BW is not high, & Q_e is minimum.

lec 14

Quantization noise power

$(Q_e)_{max} = \frac{\Delta}{2}$

↳ undeterministic, so a RV

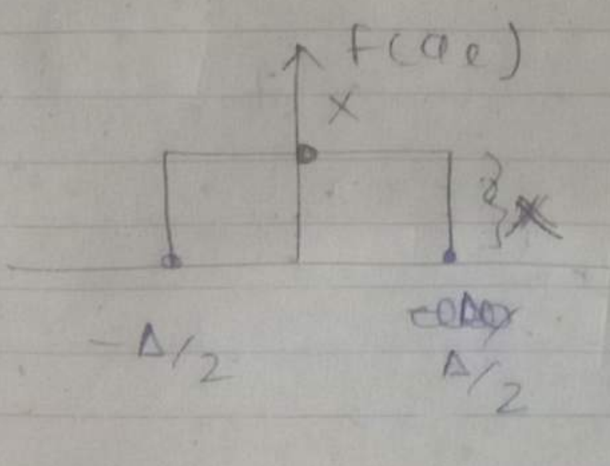
N_q = power (Q_e)

= mean square (Q_e)

$= E(Q_e^2)$

$= \int_{-\infty}^{\infty} Q_e^2 \, f_{Q_e}(Q_e) \, dQ_e$

Assume Q_e satisfied uniform probability density fn



Area (PDF) = 1

$x \cdot \Delta = 1$

$\boxed{x = 1/\Delta} = \text{PdF} = f_{Qe}(Qe)$

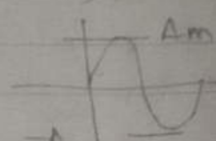
$$N_q = \int_{-\Delta/2}^{\Delta/2} Qe^2 \times \frac{1}{\Delta}\, dQe$$

$$= \frac{1}{\Delta} \frac{Qe^3}{3} \Big|_{-\Delta/2}^{\Delta/2}$$

$$\boxed{N_q = \frac{\Delta^2}{12}}$$, Δ = step size

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

$$\Delta = \frac{A_m + A_m}{L}$$



$$\boxed{\Delta = \frac{2A_m}{2^n}}$$

$$N_q = \frac{4A_m^2}{(2^n)^2} \times \frac{1}{12}$$

$$\boxed{N_q = \frac{1}{3} \frac{A_m^2}{2^{2n}}}$$

SIGNAL TO QUANTIZATION NOISE RATIO

$$SQNR = \frac{\text{Signal power}}{\text{Quantization noise power}}$$

$$= \frac{A_m^2/2}{\frac{1}{3}\frac{A_m^2}{2^{2n}}}$$

$$\boxed{SQNR = \frac{3}{2} 2^{2n}}$$ — should be as high as possible

$n\uparrow, L\uparrow, \Delta\downarrow, Qe\downarrow, N_q\downarrow$

$SQNR\uparrow$

$n\uparrow$, $BW\uparrow$

$$\boxed{BW = \frac{R_b}{2} = \frac{n f_s}{2}}$$

$$\boxed{SQNR = \frac{3}{2} 2^{2n}}$$

$n \to n+k$,

SQNR inc. by 2^{2k} times

$$SQNR_{dB} = 10\log_{10}\left(\frac{3}{2} 2^{2n}\right)$$

$$= 10\log\left(\frac{3}{2}\right) + 10\log_{10} 2^{2n}$$

$$\boxed{SQNR_{dB} = 1.8 + 6.02n}$$

(Q1) m(t) & sin $(8\pi \times 10^3 t)$ is transmitted through PCM system. Sampling rate is 50% higher than nyquist rate & min SQNR should be 22 dB. Find

(1) Transmission BW

(2) $SQNR_{dB}$

(ii) $(SQNR)_{dB} = 1.8 + 6n$

$= 1.8 + 6 \times 4$

$= 25\,dB$

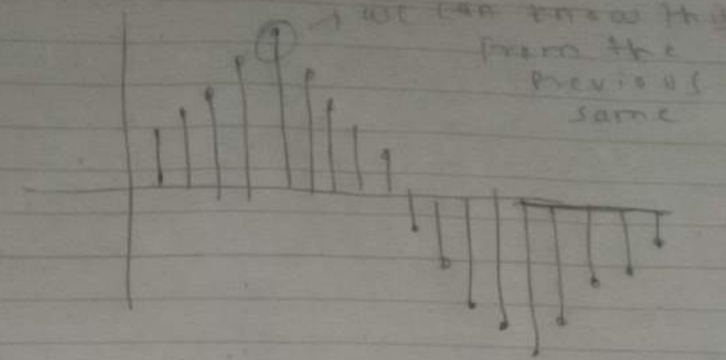
DPCM (differential ~~pulse code~~ PCM)

$(Q_e)_{max} = \frac{\Delta}{2} \Rightarrow \Delta = \frac{V_{max} - V_{min}}{2^n}$

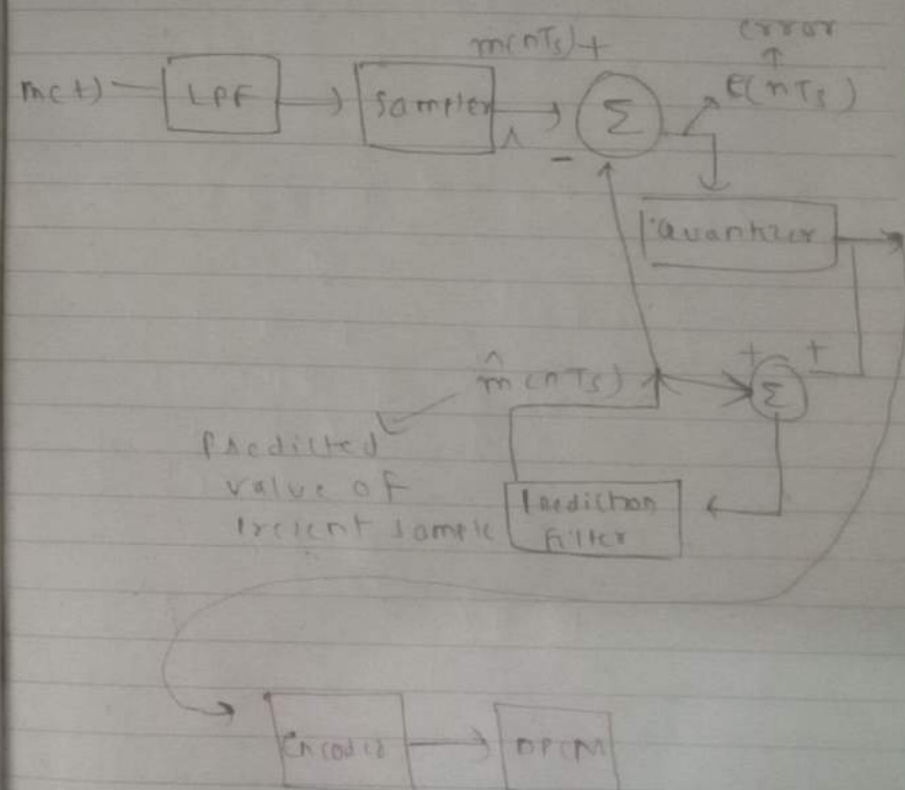
to decrease Δ, $V_{max} - V_{min} \downarrow$

dynamic range $\downarrow$

In DPCM, aim is to reduce the dynamic range



more the samples are nearer, more nearer their amplitude



In PCM, original samples were quantized. Dynamic range more

In DPCM, error b/w the sample & predicted value of sample and now this error is quantized, Dynamic range less

Prediction Filter ⟹ By analyzing the past behaviour of the signal, it predicts the present sample value

$$e(nT_s) = m(nT_s) - \hat{m}(nT_s) \quad — ①$$

$m(nT_s)$ → Present sample value

$\hat{m}(nT_s)$ → Predicted value of present sample

$e(nT_s)$ = Prediction ~~error~~ error

~~Present~~ all quantized values stored in prediction filter and by analyzing the previous quantized values, signal prediction value place & present sample is detected

$$q(nT_s) = e(nT_s) - e_q(nT_s)$$

$q(nT_s)$ → quantization error

This is ~~[illegible]~~ DPCM transmitter

$$\text{Prediction filter I/P} = \hat{m}(nT_s) + e_q(nT_s)$$

from eqn ②

$$e_q(nT_s) = e(nT_s) - q(nT_s)$$

$$= \hat{m}(nT_s) + e(nT_s) - q(nT_s)$$

$q(nT_s)$ → error

$$= m(nT_s) - q(nT_s) \quad (\text{Using eqn ①})$$

$m(nT_s)$ → Original sample value

$$\boxed{QV = SV - QE}$$

QV → Quantized value

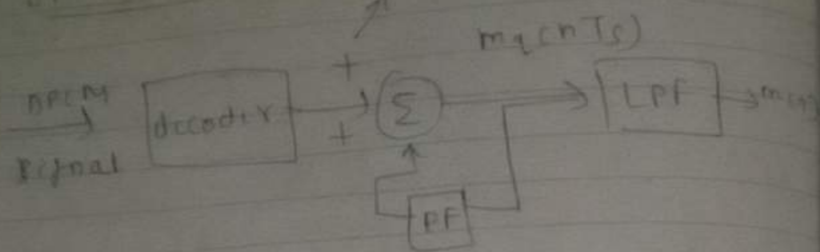
SV → Sampled value

QE → Quantization error

$$= m_q(nT_s)$$

$m_q(nT_s)$ → Quantized value of $m(nT_s)$

DPCM receiver



$$LPF_{o/p} = \underbrace{e_q(nT_s)}_{\text{quantized value}} + \underbrace{\hat{m}(nT_s)}_{\text{predicted value}}$$

$= m_q(nT_s)$, Quantized signal

⇓

LPF

⇓

m(t)

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DELTA MODULATION

* Special case of DPCM
* 1 bit DPCM
* Prediction error less
* 2 Quantization level are required & 1 bit encoding
* Transmission BW very less

$$BW = \frac{n f_s}{2}$$

If $n=1$, "BW very less"

$n=1$, levels(L) $=2$

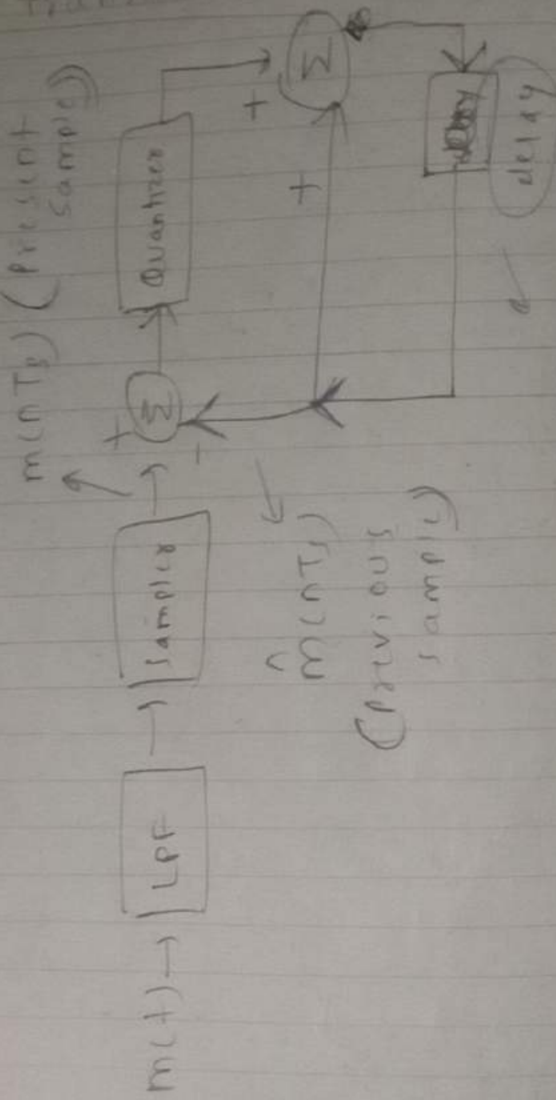
↳ these 2 levels can be represented by $+\Delta$ & $-\Delta$

Δ = step size

Δ ≈ 1 encoded

−Δ ≈ 0 encoded

DM Transmitter



in DPCM, there was prediction filter

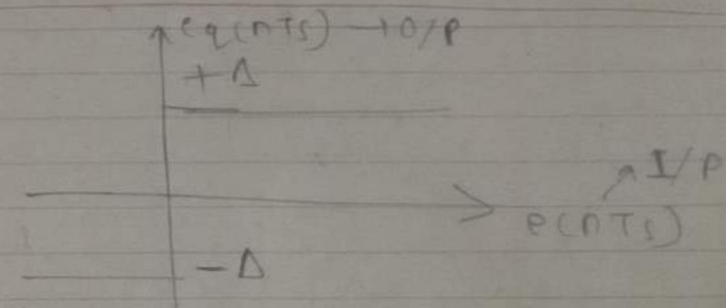
$e(nT_s)$ = "very less"

$= \underbrace{m(nT_s) - \hat{m}(nT_s)}_{\text{Very small}}$ (present sample, previous sample)

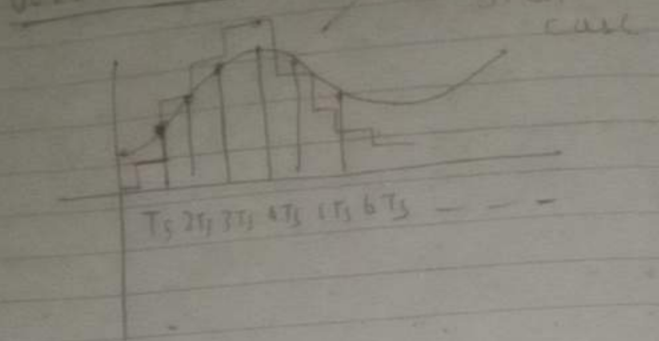
"results in small dynamic range"

therefore only 2 levels required & 1 bit.

CHARACTERISTICS OF QUANTIZER



WORKING



$e(nT_s) = +\text{ive} = +\Delta$ (mapped to 1)

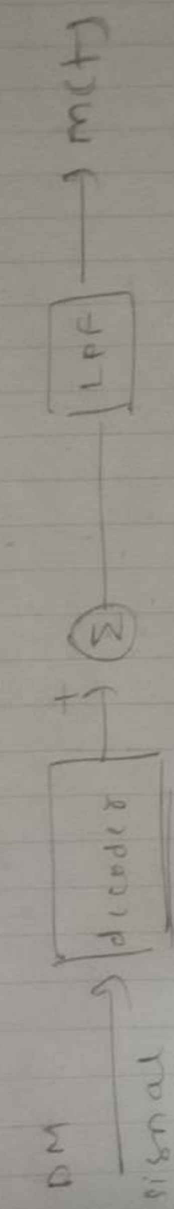
OR

$-\text{ive} = -\Delta$ (mapped to 0)

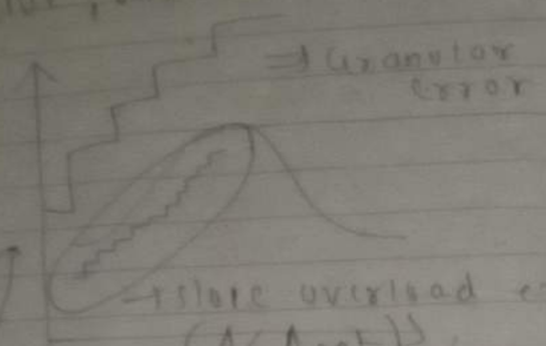
DM receiver

Similar to DPCM receiver, instead of prediction filter, use "delay block"

In DM, sampling freq very very high (oversampling), & diff between the present & previous sample very low; two levels of quantization required



Δ should have an optimal value, otherwise issues:



if $\Delta < \Delta_{optimal}$ distortion and DM signal may not be able to track the original message signal.

If ($\Delta > \Delta_{optimal}$), distortions in receiving end

ADAPTIVE DELTA MODULATION

(to overcome previous discussed errors)

* step size changes continuously according to change of message signal

$\rightarrow e(nT_s) = +ive \ (+\Delta)$

$e(nT_s) = -ive \ (-\Delta)$

DM is limited for only transmission of signal which are having constant rate of change

Adaptive DM ⇒ no slope overload error & granular error

$$\boxed{m(t) = A_m \cos(2\pi f_m t)}$$

$$\boxed{\frac{\Delta_{opt}}{T_s} = 2\pi f_m A_m}$$

$$\boxed{\frac{\Delta}{T_s} < 2\pi f_m A_m}$$

condition for slope overload error

$$\boxed{\frac{\Delta}{T_s} > 2\pi f_m A_m}$$

condition for granular error to occur

Q1) continuous signal of $8\sin(8\pi\times10^3 t)$ is passed through DM whose pulse rate = 4000 pulses/sec

Find Δ_{opt}?

Soln $f_s = 4000$ pulses/sec

$m(t) = 8\sin(8\pi\times10^3 t)$

$A_m = 8$, $f_m = 4$ kHz

$$\frac{\Delta_{opt}}{T_s} = 2\pi f_m A_m$$

in DM, $\boxed{\text{pulse rate} = \text{sampling rate}}$

$n = 1$

$R_b = n f_s = f_s$

$f_s = 1/T_s = 4000$ samples/sec

$\Delta_{opt} \times 4000 = 2\pi \times 4 \times k \times 8$

$$\boxed{\Delta_{opt} = 16\pi \text{ Volts}}$$