* Communication

* It is transmisn of information from 1 place to another.

* Voice → 300 Hz to 3.5 kHz eg. Telephone

Audio → 20 Hz to 20 kHz eg Radio.

Video → 0 to 4.5 MHz TV.

* Modulator converts low frequency signal into high frequency signal as freq^n ↑se λ ↓ses. (i.e.) length of antena ↓ses.

$\because \lambda = \frac{c}{f}$

* modulator used to
① To Reduce size of antenna
② Multiplexing
③ To reduce effect of noise ($\because S/N >> 1$ is cond^n to have good transmission)

* Bandwidth of signal should as low as possible. & B.W. of channel should be high for multiplexing.

* To reduce the BW of signal insignificant frequencies should eleminated $\because$ max energy (i.e. 99%) is concentrated in first three lobe of the spectrum.

$\therefore$ we use use LPF To have low frequencies having main lobe. B.W. $= \left(\frac{1}{T}\right)$ Hz

practical bandwidth of Rectangular pulse is inversely proportional to pulse width.

$\therefore$ B.W. ↑es Pulse width decreases.

Modulation property is nothing but frequency shifting property.

$$\underbrace{g(t)}_{\text{signal}} \; \underbrace{\cos 2\pi f_c t}_{\text{Carrier signal}} \longrightarrow \frac{\overbrace{G(f-f_c)}^{\text{Right side}} + \overbrace{G(f+f_c)}^{\text{left side}}}{2}$$

f_c: carrier frequency

Carrier is very high frequency signal FM → 88 M - 108 MHz
Radio - GHz.

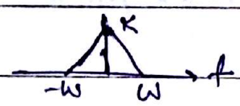
* If a signal is multiplied by ~~[illegible]~~ carrier the signal is shifted to the left side & rightside by f_c & the amplitude become half.
* hence modulation is nothing but frequency translation ∵ signal with low frequency range is translated to high freqⁿ range.

$$1 \cdot \cos(2\pi f_c t) \longleftrightarrow \frac{\delta(f-f_c) + \delta(f+f_c)}{2}$$

Concept of modulation & demodulation

* At transmitter side

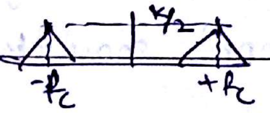
(original signal) $m(t) \longleftrightarrow$



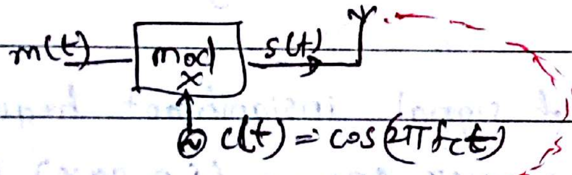
Low freqⁿ ∴ problem with antenna while transmitting $m(t)$.

$\downarrow \times \cos(2\pi f_c t)$ Tx modulation to remedise Antena problem.

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

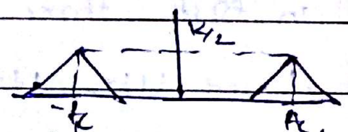


high freqⁿ ∴ Now we can transmitt.



* At reciever side.

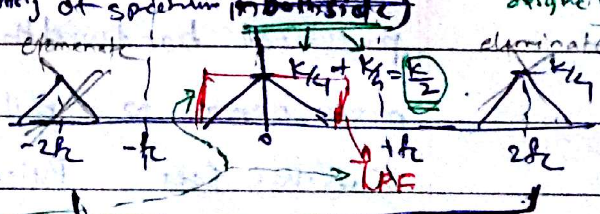
$s(t) = m(t)\cos 2\pi f_c t \longleftrightarrow$



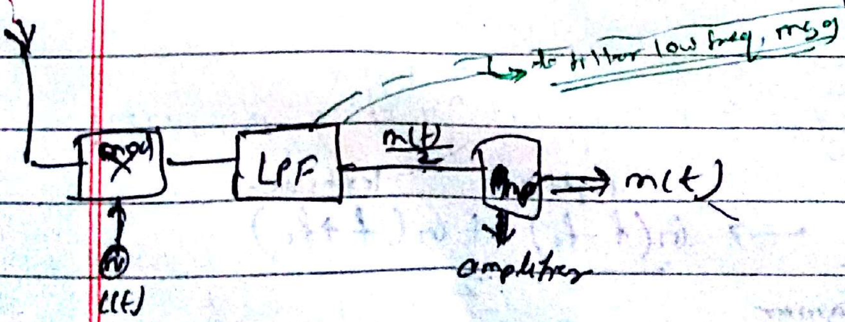
$\downarrow \times \cos 2\pi f_c t$ Rx modulation (to have spectrum to shift at origin) (again shifting of spectrum ~~in both side~~)

$= m(t)\cos^2 2\pi f_c t$

$= \frac{m(t)}{2} + \cancel{\frac{m(t)\cos 4\pi f_c t}{2}}$



* eliminated when this spectrum is passed ~~through~~ through LPF.
So we get $\frac{m(t)}{2}$ back.
now by amplifying we get $m(t)$



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* In modulation characteristic parameters of a carrier is varied according to msg signal.

3 parameter can vary (1) amplitude (2) freq^n (3) phase.

(1) Amplitude modulation
⇓
(1) AM
(2) DSB
(3) SSB
(4) VSB

(2) Frequency modulation
⇓
(1) N.B.F.M.
(2) WBFM

(3) Phase modulation.
⇓
(1) N.B.PM.
(2) W.B.P.M.

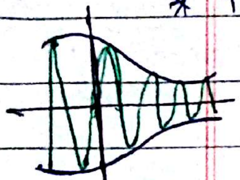
Carrier Signal. ⟹ $c(t) = \underset{\downarrow\ \text{Am}}{A_c} \cos(\underset{\downarrow\ \text{FM}}{2\pi f_c t} + \underset{\downarrow\ \text{P.M}}{\phi_m})$

* Amplitude Modulation

*. Peak amplitude of the carrier is ~~created~~ varied according to msg signal.

amplitude sensitivity.

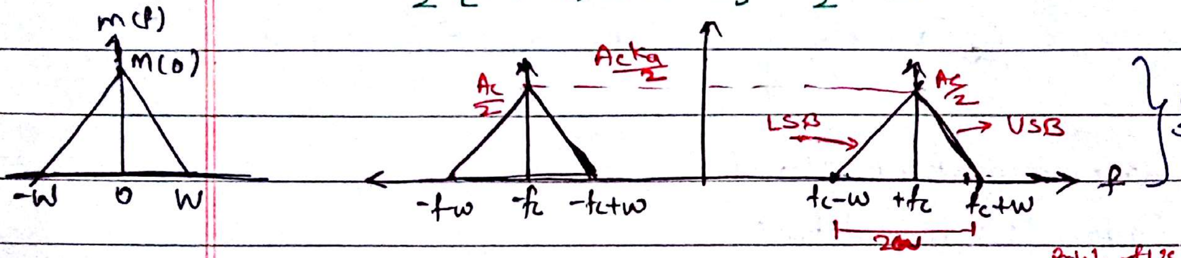
* time domain eqn of AM $\quad s(t) = A_c \cos(2\pi f_c t) + A_c K_a m(t) \cos(2\pi f_c t)$



Standard form ⇒ $\therefore s(t) = \underbrace{A_c}_{\text{Peak amplitude of carrier}} \underbrace{[1 + K_a m(t)]}_{\text{Peak amplitude after modulation}} \cos(2\pi f_c t)$

* Modulated signal in frequency domain

$$S(f) = \frac{A_c}{2}[\delta(f-f_c) + \delta(f+f_c)] + \frac{A_c}{2} K_a [M(f-f_c) + M(f+f_c)]$$



} AM spectrum.

B.W. of USB = W
B.W. of LSB = W.

B.W. = 2W. or 2× bigest frequency of m(t)

* ∴ AM consists of carrier freq^n cies, USB frequencies & LSB freq^n cies.

* Singletone Modulation of AM. (i.e. m(t) has only one frequency)

→ the msg signal is sinusoidal signal ∵ only one (single) freqncy }

$\therefore s(t) = A_c \cos 2\pi f_c t + A_c k_a \underbrace{A_m \cos 2\pi f_m t}_{m(t)} \cos 2\pi f_c t$

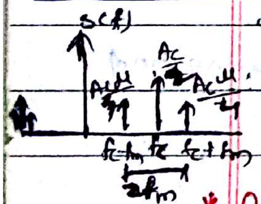
$\boxed{\mu = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}}$

let $K_a A_m = \mu$ = modulation index. $0 < \mu \le 1$

for the carrier $A_c[1+\mu] \to$ max amp.

$\therefore s(t) = \underbrace{A_c[1 + \mu \cos 2\pi f_m t]}_{\text{amp. after modulation}} \cos 2\pi f_c t$

$A_c[1-\mu]$ min amp



$s(t) = \underbrace{A_c \cos 2\pi f_c t}_{\text{carrier}} + \underbrace{\frac{A_c \mu}{2} \cos 2\pi (f_c + f_m)t}_{\text{USB}} + \underbrace{\frac{A_c \mu}{2} \cos 2\pi (f_c - f_m)t}_{\text{LSB}}$

* Power Calculations $\quad P = I_{rms}^2 R$ or V_{rms}^2 / R.

* $V_{RMS} = \frac{A_c}{\sqrt{2}}$ $\quad \therefore P_{fc} = \frac{A_c^2}{2R}, \; P_{USB} = \frac{A_c^2 \mu^2}{8R}, \; P_{LSB} = \frac{A_c^2 \mu^2}{8R}$.

(if R not given take R=1)

* $P_t = P_{fc} + P_{USB} + P_{LSB} = P_{fc} \boxed{1 + \frac{\mu^2}{2}}$ ⇒ $\frac{P_{fc} \mu^2}{2}$ is SB power

* modulatn efficiency $\quad \eta = \frac{P_{SB}}{P_t} = \frac{\text{side band power}}{\text{total power}} = \frac{\mu^2}{2+\mu^2}$

$V_{max} = A_c[1+\mu]$

$V_{min} = A_c[1-\mu]$

$V_{max} + V_{min} = 2A_c$

$\mu = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$ $\quad \frac{V_{max}}{V_{min}} = \frac{1+\mu}{1-\mu}$

antena current after modulatn $\boxed{I_t = I_c \sqrt{1 + \frac{\mu^2}{2}}}$

antena current before modulatn

* If μ is not given in problem take $\mu = \frac{A_m}{A_c}$

* carrier is not modulated means $\mu = 0$ $\quad P_{fc} = P_t$

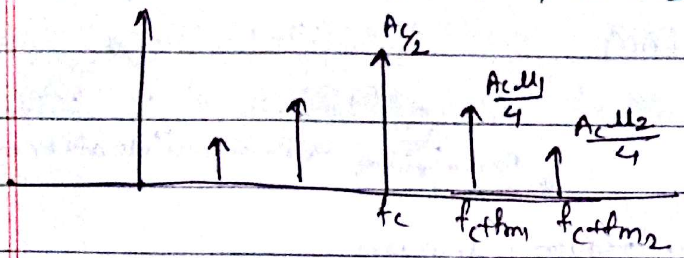
* when μ changes from 0 to 1 power increased by 50% of P_{fc}

* Multitone modulation:- (m(t) contains more than one frequency)

$$m(t) = A_{m_1}\cos 2\pi f_{m_1} t + A_{m_2}\cos f_{m_2} t.$$

$$s(t) = A_c[1+\mu_1\cos 2\pi f_{m_1} t+\mu_2 \cos 2\pi f_{m_2} t]\cos 2\pi f_c t$$

$$K_a A_{m_1} = \mu_1 \quad , K_a A_{m_2} = \mu_2$$



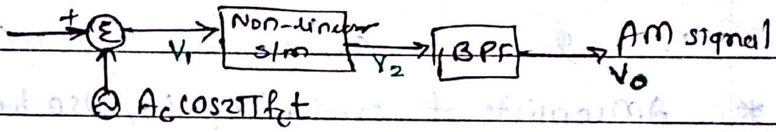
B.W. = (2 × Highest of B.B freq. available)
$= 2 \times f_{m_2}$

$$\mu_t = \sqrt{\mu_1^2+\mu_2^2}$$

$$\therefore P_t = P_c\left[1+\frac{\mu_t^2}{2}\right] \qquad P_t = P_c + P_{USB} + P_{LSB}.$$

* Generation of AM signal.

① Square law modulator :-



mod o/p ① $V_1 = [m(t) + A_c\cos 2\pi f_c t]$

Non lin. o/p ② $V_2 = aV_1 + bV_1^2$ ⇒ This contain All the term but we have terms of m(t) only so BPF is used to attenuate high freq^n.

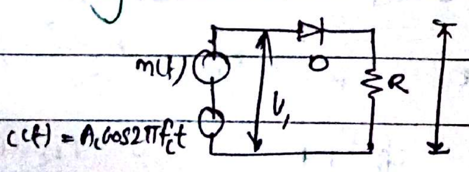
*(BPF freq. range ~~varies according to problem~~)

By BPF excluding unwanted freq^n. ③ $\therefore V_0 = aA_c\cos 2\pi f_c t + 2bA_c m(t)\cos 2\pi f_c t$

* amp. sensitivity $(S_{amp}) = \frac{2b}{a}$

④ Then rearrang the V_0 eq^n in standard form & comparing find μ, A_c, f_m, f_c etc to find power.

② Switching modulator:-

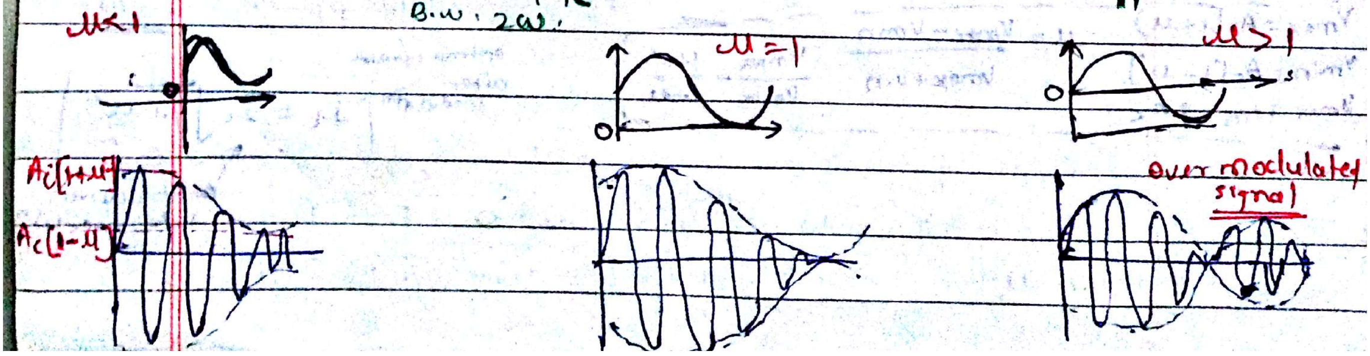


$V_1 = m(t) + A_c\cos 2\pi f_c t$

$V_2(t) = V_1(t)$ for $c(t) > 0$.
$= 0$ for $c(t) < 0$.

(Passing $V_2(t)$ through BPF) ⇒ $V_2(t) = \frac{A_c}{2}\cos(2\pi f_c t) + \frac{2}{\pi} m(t)\cos 2\pi f_c t$.

Center freq. f_c
B.W. 2ω.



* Demodulation of AM Signal.

① Square law modulator :- (used for $\mu \le 1$) ($\because$ can not detect over modulated signal)

AM v_1 → Non-Linear s/m → LPF → v_0

$v_1 = aA_c \cos 2\pi f_c t + A_c K_a m(t) \cos 2\pi f_c t$

$v_2 = av_1 + bv_1^2$ then v_2 passed through LPF to select low frequency

$\therefore$ Now $V_0 = \underbrace{b A_c^2 K_a m(t)}_{\text{Signal component}} + \underbrace{\frac{b A_c^2 K_a^2}{2} m^2(t)}_{\text{noise component}}$ * $\because$ o/p of LPF consist of m(t) & $m^2(t)$ component

$m^2(t)$ causes interference with m(t).

$$\frac{\text{Signal component}}{\text{Noise component}} = \frac{S}{N} = \frac{2}{K_a m(t)}$$

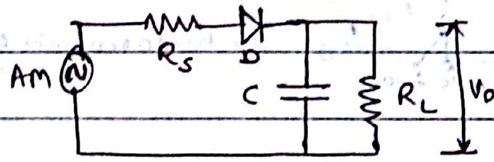
if $\frac{S}{N} = \frac{2}{K_a m(t)} < 1$ demodulation not possible.

$\therefore \frac{S}{N} >> 1 \Rightarrow \frac{2}{K_a A_m \cos 2\pi f_m t} >> 1$

$\therefore \left(\frac{S}{N}\right)_{min} = \frac{2}{K_a A_m} = \frac{2}{\mu}$ if $\mu = 10$ $S/N = 0.2$
if $\mu = 0.1$ $S/N = 20$

$\therefore \mu$ should be < than 1 so that S/N is high.

② Envelop detector (used for $\mu \le 1$) (can not detect over modulated sig)



* envelope ~~of~~ of modulated signal is same as msg signal
* so peak amplitude of AM signal Represent msg signal.

i/p	o/p
$10 \cos 2\pi f_c t$	$V_0 = 10$
$A_c \cos 2\pi f_c t$	$V_0 = A_c$
$e^{t} \cos 2\pi f_c t$	$V_0 = e^{-t}$
$A \cos 2\pi f_c t + B \sin 2\pi f_c t$	$V_0 = \sqrt{A^2 + B^2}$

$\therefore A_c[1 + K_a m(t)] \cos 2\pi f_c t$ $\boxed{V_0 = A_c[1 + K_a m(t)]}$ as demodulator ckt not consider DC component

$\therefore V_0 = A_c K_a m(t)$

③ Synchronous Detector → multiplexer → $v_1(t)$ → LPF → $v_2(t)$ → m(t)
oscilator

(for synchronous detector modulation efficiency is 100% but complexity is very high)

$v_2(t) = \frac{A_c^2}{2}[K_a m(t)]$

* Single side band (SSB)

* Only USB or LSB is transmitted.
* for single tone modulat^n $m(t) = A_m \cos 2\pi f_m t$.

time domain eq^n $\Rightarrow \frac{A_c A_m}{2} \cos 2\pi (f_c \pm f_m) t$ $\quad + \rightarrow$ USB, $- \rightarrow$ LSB.

* for multitone modulation

time domain eq^n $\rightarrow \frac{A_c}{2} m(t) \cos 2\pi f_c t \mp \frac{A_c}{2} \hat{m}(t) \cos 2\pi f_c t$ $\quad - \rightarrow$ USB, $+ \rightarrow$ LSB

$\hat{m}(t)$ is helbert Transform of $m(t)$ which give 90° phase shift

Power

$$\text{Total power in DSB} = \overbrace{\frac{A_c^2 A_m^2}{8R}}^{\text{LSB power}} + \overbrace{\frac{A_c^2 A_m^2}{8R}}^{\text{USB Power}}$$

Total power in SSB = LSB power or USB power.

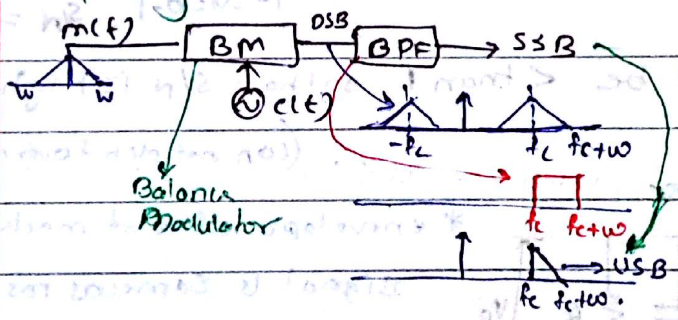
in SSB BW & PW is ↓sed by 50% compared to DSB

$$\% \text{ power saving in DSB} = \frac{2}{2+\mu^2} \times 100 = \frac{\text{power saved in DSB}}{P_t} \quad \text{around 80\%}$$

$$\text{—— " —— SSB} = \frac{1+\mu^2/4}{1+\mu^2/2} \times 100 = \frac{\text{Power saved in SSB}}{P_t} \quad \text{around 90\%}$$

* Generation of SSB signal.

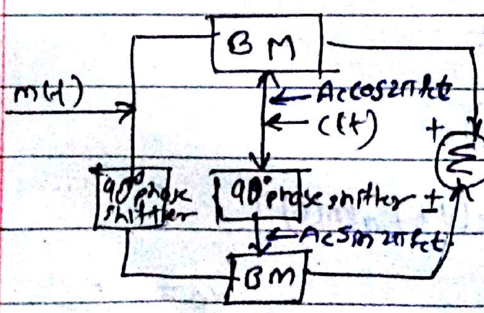
① frequency Descrimination method :-



$^{||}$ly we can generate for LSB by BPF range of $f_c - w$ to f_c

② Phase Descrimination method.

we know time domain eq^n of SSB $\Rightarrow \underbrace{\frac{A_c}{2} m(t) \cos 2\pi f_c t}_{\substack{\text{it's multiplicat}^n\text{ of carrier} \\ \text{\& msg signal by } \boxed{BM} \text{ block}}} + \underbrace{\frac{A_c}{2} \hat{m}(t) \sin 2\pi f_c t.}_{\substack{\text{it is multiplication of carrier} \\ \text{signal and msg signal with are} \\ \text{shifted by 90° phase shift.}}}$



this method is complex as compared to first method.

* Demodulation of SSB

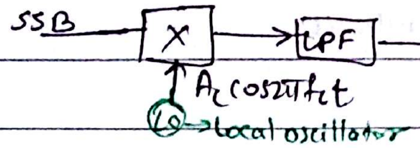
*** Enevelop detector** ✗ — SSB → [ED] → o/p

o/p of envelop D. $= \sqrt{\left[\frac{A_c}{2} m(t)\right]^2 + \left[\frac{A_c}{2} \hat{m}(t)\right]^2}$

if $m(t) = \cos 2\pi f_m t \cdot \Rightarrow o/p = \frac{A_c}{2} \sqrt{\cos^2 2\pi f_c t + \sin^2 2\pi f_c t} = \frac{A_c}{2}$

⇒ o/p is constant ∴ Enevelop detector cannot used to demodulate of DSB & SSB signal ∴ Only Synchronous modulation is the option

(1) Synchronous Modulation



o/p $= \frac{A_c^2}{4} m(t)$ when $\phi = 0°$ → L.O without phaseshift

$= \pm \frac{A_c^2}{4} \hat{m}(t)$ when $\phi = 90°$.

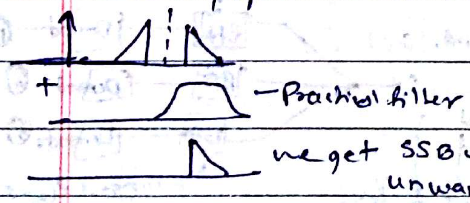
$o/p = \frac{A_c^2}{4} m(t)\cos\phi \pm \frac{A_c^2}{4} \hat{m}(t)\sin\phi$ ⇒ if ϕ is not 0 & 90° i.e. L.O. have some phase shift

Ad-vantage of SSB. – BW ↓ by 50% ; power ↓ by 50% compared to DSB.
No quadrature Null effect.

Dis-adv. of SSB – since filters are used & No practical filter has sharp cut. (ideal filter not possible) ∴ Not possible to suppress side band which is not required.

* ∴ SSB modulatⁿ is suitable only for the transmission of voice only

* there is gap of 600Hz betⁿ USB & LSB in voice signal spectrum

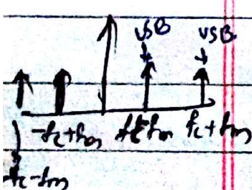


* Doubleside band (DSB)

* both sidebands are transmitted LSB & USB.

single tone modulatⁿ DSB-SC.

$s(t) = \frac{A_c A_m}{2} [\cos 2\pi(f_c - f_m)t + \cos 2\pi(f_c + f_m)t]$



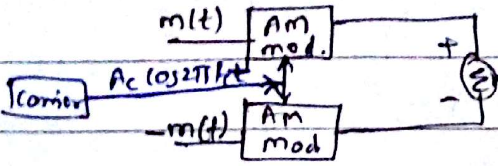
$S(f) = \frac{A_c A_m}{4}[\delta(f - f_c + f_m) + \delta(f + f_c - f_m)] + \frac{A_c A_m}{4}[\delta(f - f_c - f_m) + \delta(f + f_c + f_m)]$

$BW = 2f_m$

$Power = \frac{A_c^2 A_m^2}{4}$ $P_{USB} = P_{LSB}$.

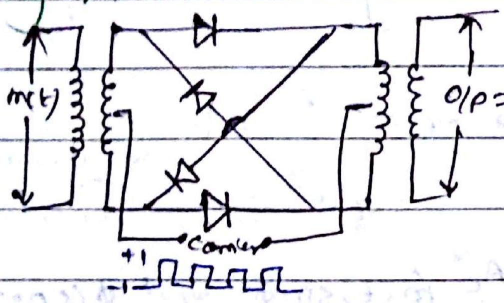
* Generation of DSB-SC signal

(1) Balance modulator



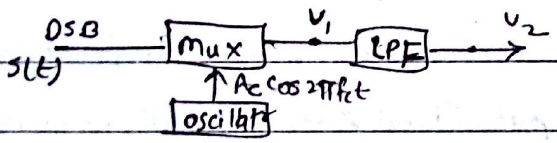
$O/P = 2k_a c(t) m(t)$

(2) Ring Modulator.



* Demodulation of DSBSC signal

coherent Detection : (Synchronous Detection)



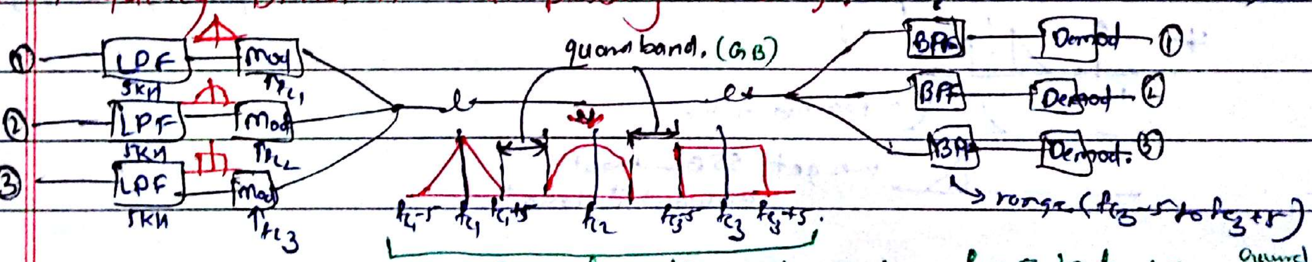
$V_2(t) = \frac{A_c^2 m(t)}{2} \cos\phi$

$\phi = 0.\ V_2(t) = \frac{A_c^2 m(t)}{2}$ ⇒ info not lost.

$\phi = 90^\circ\ V_2(t) = 0$ ⇒ info. totally lost

*Quadrature null effect.

* Frequency Division Multiplexing (FDM).



B.W. of multiplexed signal = $f_{c_1}-5$ to $f_{c_3}+5$ = 2GB + 10+10+10 (guard band)

* Carrier freq^{n}cies carefully selected to avoid Interference of signal.
* For multiplexing B.W. of the channel should be as high as possible & B.W. of the signal is as low as possible to have multiplexing.
* here 3 signal passed through one channel.!

twisted pair cable → 500KHz (BW) → telephone comm.

coaxial cable → 500MHz (BW) → T.V. transmission (require 4.5MHz)

fiber optical cables → in range of GHz (BW)

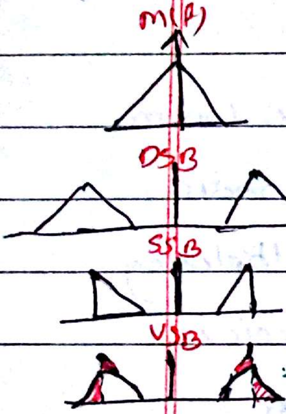
* Vestigial Sideband (VSB) modulation

→ Used in transmission of video signal. (∵ significant rang is upto 4.5 MHz)

→ AM & DSM not used ∵ BW is 9 MHz ∴ multiplexing is not possible

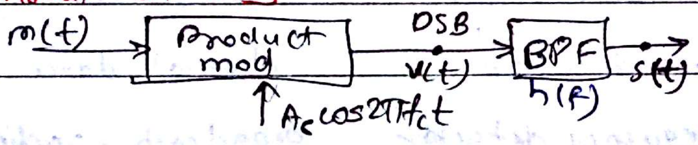
→ SSB used only for voice.

→ ∴ VSB is used for video trsn.



⇒ B.W. = $f_m + f_v$. [power required is same as SSB (ideal)]

* Generatⁿ of VSB



$s(t) = v(t) \otimes h(t)$

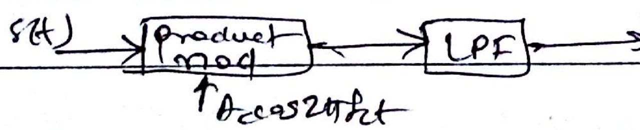
$S(f) = V(f) \cdot H(f)$

time domain eqⁿ of VSB $s(t) = \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t - m_s(t)\sin 2\pi f_c t\right]$

$m_s(t)$ ⇒ o/p of filter with i/p m(t)

filter has $H_s(f) = j[H(f-f_c) + H(f+f_c)]$

* Demodulation of VSB



SSB signal. o/p = $\frac{A_c^2}{4} m(f) \underbrace{[H(f-f_c) + H(f+f_c)]}_{k}$

o/p = $\frac{A_c^2}{4} m(t)\, k$

Scanned by CamScanner

* figure of merit of DSB signal = 1
* — " FOM SSB signal = 1
* — " FOM Noise in AM = $\frac{\mu^2}{2+\mu^2} = \eta$ [modulatⁿ efficiency]

Name	Detection process	Applicatⁿ
Baseband	No modulatⁿ	short distance comm.
AM	Envelope detector	Broadcasting radio.
DSB-SC	Synchro. Detector	Analog data, multiplexing
SSB-SC	— " —	point to point voice
VSB-SC	— " —	Digital data.
SSB-FC.	Envelope Detector	T.V. video.
PM	Phase detector	digital data.
FM.	Frequency detector	Broadcasting radio, Microwave relay sattelite system.

* Angle modulation.

Angle of carrier signal (eigth frequency or phase) is varied according msg signal.

1) Frequency Modulation (FM)

2) Phase modulation (PM)

* $\theta = \omega t = 2\pi f t \Rightarrow \frac{d\theta}{dt} = 2\pi f \rightarrow f = \frac{1}{2\pi}\frac{d\theta}{dt} \Rightarrow \theta(t) = 2\pi \int f_i \, dt.$

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

* $s(t) = A_c \cos[2\pi f_c t + 2\pi k_f \int^t m(t)\,dt] \Rightarrow$ FM signal

↳ frequency sensitivity. $\boxed{\text{carrier swing} = 2\Delta f}$ $\boxed{\Delta f = k_f A_m = \text{freq deviatn}}$

* $s(t) = A_c \cos[2\pi f_c t + k_p m(t)] \Rightarrow$ PM signal

↳ phase sensitivity.

*[for single tone $m(t) = A_m \cos 2\pi f_m t$] $\boxed{k_p A_m \rightarrow \text{phase deviation}}$

* FM in case of single tone modulation $m(t) = A_m \cos 2\pi f_m t$

$$s(t) = A_c \cos\left[2\pi f_c t + \underbrace{\frac{k_f A_m}{f_m}}_{\beta} \sin 2\pi f_m t\right].$$

$$\therefore \beta = \frac{k_f A_m}{f_m} \rightarrow \text{Modulation Index.}$$

$$\beta = \frac{\Delta f}{f_m} - \frac{\text{frequency deviation}}{\text{msg frequency}}.$$

① $\beta << 1$, Narrow Band FM (NBFM)

② $\beta >> 1$, Wide Band FM (WBFM)

$f_i = f_c + k_f m(t)$ f_i - instantaneous frequency.

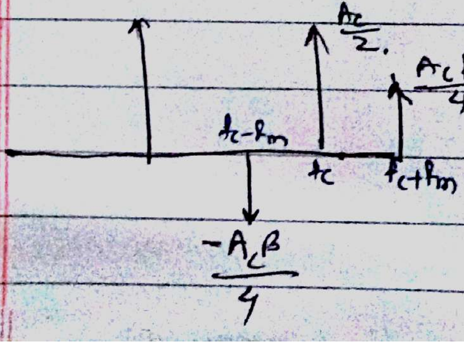
$k_f \rightarrow$ indicates change in carrier frequency per volt

FM can be considered as voltg to freq. converter i.e. voltg variation to freq. variation ∴ as value $m(t)$ ↑ses f of carrier ↑s for max voltage f is max i.e. f_{max} & for minimum f_{min}.

$f_{max} = f_c + \Delta f$ $f_{min} = f_c - \Delta f$

* NBFM ($\beta << 1$) ∴ $s(t) = A_c \cos 2\pi f_c t - A_c \beta \sin 2\pi f_c t \sin 2\pi f_m t.$

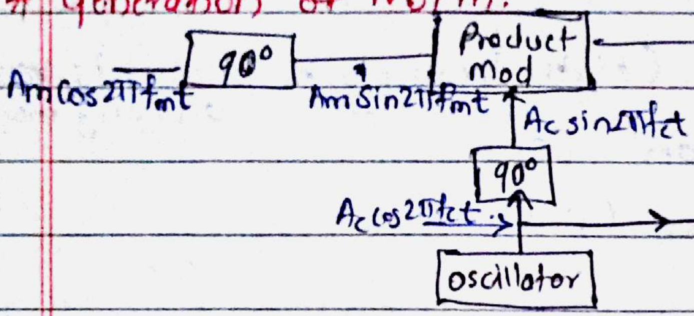
$$s(t) = A_c \cos 2\pi f_c t + \frac{A_c\beta}{2}\cos 2\pi (f_c + f_m)t - \frac{A_c\beta}{2}\cos 2\pi (f_c - f_m)t$$



B.W. = $2f_m$

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

* generation of NBFM.



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

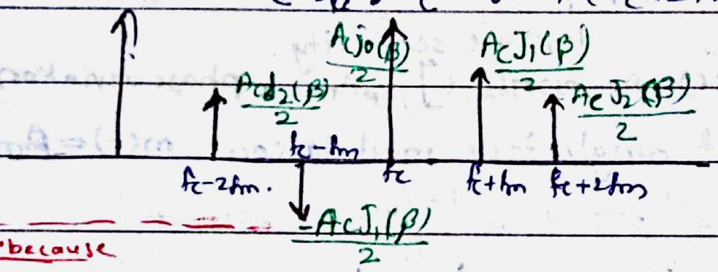
$\beta = \frac{A_m k_f}{f_m}$

* WideBand FM (WBFM).

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

$n = 0, \pm 1, \pm 2, \ldots$

$$\therefore s(t) = A_c J_0(\beta) \cos 2\pi f_c t + A_c J_1(\beta) [\cos 2\pi(f_c + f_m)t - \cos 2\pi(f_c - f_m)t]$$
$$+ A_c J_2(\beta) [\cos 2\pi(f_c + 2f_m)t + \cos 2\pi(f_c - 2f_m)t]$$



Theoretical BW = ∞

Practical by carson's rule

$BW = 2(\beta + 1) f_m$ $\quad \beta = \frac{\Delta f}{f_m}$

$= 2\Delta f + 2 f_m$

minus because

Properties:-

*(only 3 consider upto $\beta = 2$) $\therefore BW = 6 f_m$

→ WBFM spectrum consist of carrier & infinite number of sidebands seperated by f_m.

→ amplitude of the sideband depend on Bessel f^n coefficient $J_n(\beta)$ which is ↓ses as n ↓ses.

→ WBFM spectrum amp. of carrier depends on $J_0(\beta)$ and hence modulation index β.

P_{total} (total power) $= \frac{A_c^2}{2} \sum_{n=-\infty}^{\infty} J_n^2(\beta)$, $=1$

$P_{f_c} = P_{carrier} = \frac{A_c^2 J_0^2(\beta)}{2}$

$\therefore \underline{P_t} = \frac{A_c^2}{2} = \underline{P_c}$

$P_{f_c + f_m} = P_{f_c - f_m} = \frac{A_c^2 J_1^2(\beta)}{2}$

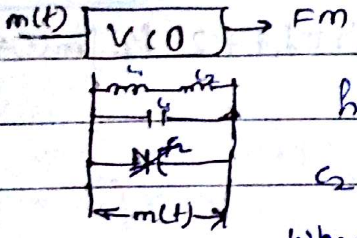
Bessel f^n $J_n(\beta) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j(x \sin\theta - n\theta)} d\theta$.

① $J_n(x) = (-1)^n J_{-n}(x)$

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

* Generation of WBFM.

① Direct Method.



VCO – voltage controlled oscillator.

freq. of oscillatⁿ $= \dfrac{1}{2\pi\sqrt{(L_1L_2)(C+C_2)}}$

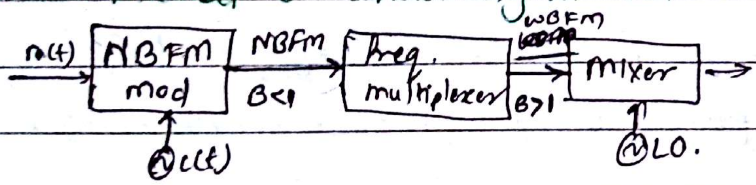
C_2 – varactor diode used as voltg variable cap.

when m(t) is i/p capacitance of C_2 changes. Hence o/p signal freq. changes.

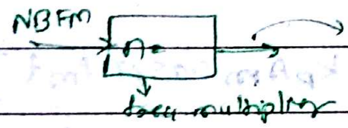
* This method is used in short distance communicatⁿ)

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

② Indirect method or armstrong method.



* square law device multiply frequency by 2

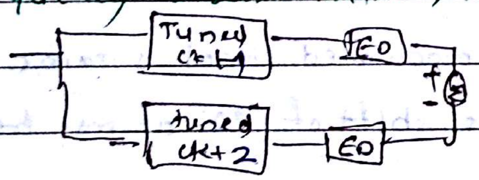


$f_o = nf_c$

$\beta_o = n\beta_o$

* Demodulation of WBFM.

① frequency discrimination method. using balance slope detector



$V_o = V_1 - V_2$

converts frequency variation to voltage variation.

i) when $f_i = f_c$ as we know $f_i = f_c + K_f m(t)$

∴ $m(t) = 0$ ∴ voltage $m(t) = 0$.

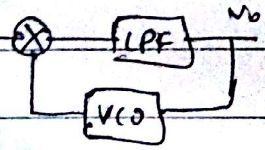
ii) when $f_i > f_c$ $V_o > 0$ m(t) should greater than 0

iii) $f_i < f_c$ $V_o < 0$ m(t) should be less than 0,

Resonant frequency of tuned ckt1 $= f_{r_1} > f_c + \Delta f$

tuned ckt 2 $= f_{r_2} < f_c - \Delta f$.

② Phase Discrimination method using Phase lock loop (PLL)



1st order PLL.

Ilyr to synchronous detector.

LO is replaced by VCO

whe i/p to PLL $\cos(2\pi f t + \phi)$ $V_o \propto \frac{d}{dt}[\phi]$

when i/p to PLL is FM $A_c \cos(2\pi f_c t + 2\pi k_f \int m(t)dt$

$V_o \propto 2\pi k_f\, m(t)$ $V_o = \dfrac{k_f}{k_v} m(t)$

k_f → freq. sensitivity at Tx

k_v → freq. sensitivity at Rx.

$\phi = k_p\, m(t)$ rad

* ① $m(t) \rightarrow \boxed{\text{PM modulator}} \rightarrow A_c\cos[2\pi f_c t + k_p m(t)]$ PM phase deviatn

$k_p A_m = \Delta\phi$

② $m(t) \rightarrow \boxed{\int} \xrightarrow{\int m(t)} \boxed{\text{PM mod}} \rightarrow A_c\cos[2\pi f_c t + 2\pi k_f \int m(t)\,dt]$ FM

∴ FM can be generated using phase modulator by prior $\int$ of m(t)

IIly PM can generated from FM

$m(t) \rightarrow \boxed{\text{Diff.}} \xrightarrow{\frac{d\,m(t)}{dt}} \boxed{\text{FM modulator}} \rightarrow A_c\cos[2\pi f_c t + k_p m(t)]$

* Bandwidth and power of PM are same as that of FM.

$BW = 2(\beta_{pm}+1) f_m \qquad P_{pm} = \frac{A_c^2}{2}$

β of pm is independent of frequencies.

* figure of merit $m_{FM} = \frac{3}{2}\beta^2$ (for single tone modulatn)

* $s(t)_{PM} = A_c\cos[2\pi f_c t + k_p A_m \cos 2\pi f_m t]$

$s(t)_{PM} = A_c\cos[2\pi f_c t + \beta \cos 2\pi f_m t]$

modulatn index $\Rightarrow \Delta\phi = k_p A_m = \beta$

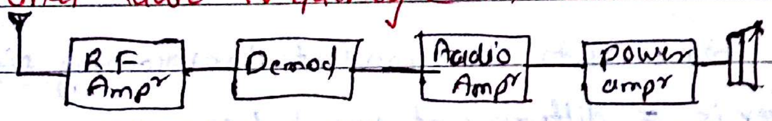
time domain eqn of phase modulated signal is same as the FM signal except a phase shift of 90° at msg frequency.

* Radio Receivers.

characteristics

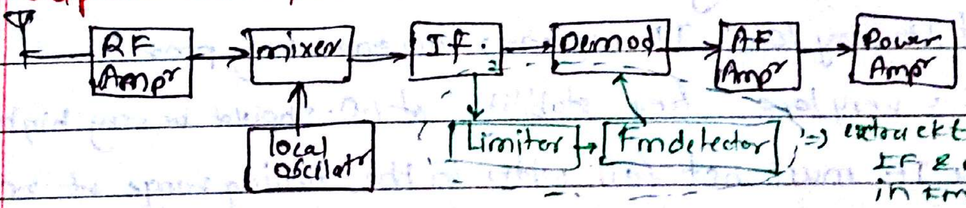
* Sensitivity: minimum signal strength to get std o/p.
  sensitivity depends on overall gain of s/m.
* Selectivity: ability to select the required frequencies. & Reject unwanted. $B.W. = \frac{f_r}{Q}$ → f_r → resonant freq., Q → quality factor
  tuned ckt tuned to resonant freq. f_r & has B.W. 10KHz then $f_r - 5$ & $f_r + 5$ frequensies are selected.
* Fidelity: ability to reproduce all audio frequencies equally in the range of entire tuning

* Tuned Radio Frequency (TRF) Reciever



* carrier frequencies alloted from FM = 88MHz – 108MHz.
  — " — from AM = 550KHz – 1650KHz
* RF amp's are tuned ampr's which allotes only selected frequencies. ~~which~~ depends on f_r (resonant frequency)
  f_r can change by changing the capacitor value by tuning knob.

* Super heterodyne Reciever.



⇒ extra ckt b/n block IF & demod in FM Recievers.

AM Range → 550 – 1650KHz & IF = 455KHz

FM Range → 88 to 108 MHz & IF → 10.7 MHz.

i) RF converted to IF then to AF

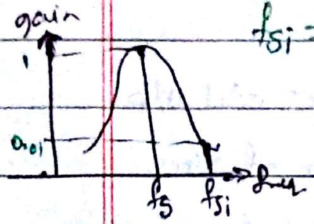
ii) IF freq. is always equal to 455KHz & LO is tunned such that IF obtained = 455KHz called tunning.

iii) IF ampr tuned circuit always tunned to 455K.

IF → intermidiate frequency

f_{si} → image frequency.

* Image frequency (IF & its suppression)



$f_{si} = f_s + 2IF$

gain at f_{si} should be as minimum as possible.

then $S/N \gg 1$

To measure suppression factor Image Reject Ratio (IRR) is used. $IRR = \alpha = \frac{G_{fs}}{G_{fsi}} = \frac{1}{0.01} = 100.$

$\alpha = \sqrt{1 + Q^2 \rho^2}$ $\quad \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$

α should as high as possible in cascade $\alpha = \alpha_1 \cdot \alpha_2$

* RF ampr

(1) IRR ↑ses (2) over all gain ↑ses (3) sensitivity ↑ses (4) S/N at o/p ↑ses.

* Mixer

→ Purpose of mixer is to downconvert the incoming signal

→ o/p of mixer is = difference of input frequencies.

→ f_s – incoming signal freq. f_l – local oscillator frequency.

then o/p. of mixer $= \boxed{f_l - f_s}$ $\quad f_l$ always $> f_s$.

→ if reciever tuned at f_s, f_l adjusted so that o/p would be IF

i.e. $f_l - f_s = IF$

* choise of IF.

→ IF IF is too high poor selectivity poor adjustent channel Reject

→ high value IF ↑ses tracking difficulties ↑ses.

→ If IF very low IF rejection becomes very poor.

→ IF is very low freq. stobility of L.O. should be very high.

→ The IF must not fall with in the tuning range of reciever.

* Tracking of Reciever.

→ ganged capacitor adjusted for proper operation at highest freq.

→ padder capacitor in ~~parallel~~ series with tank inductor adjusted to provide tracking at low frequency.

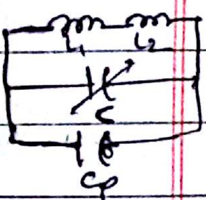
→ final adjustment made at mid frequencies by slight adjusment of the inductance in each tanks.

adjust LO frequency so that mixer o/p = IF $\quad f_l - f_s = IF$

$f_l - f_s \neq IF \Rightarrow$ tracking error.

tracking error can be minimised using padder capacitor & trimmer circuit. $f_2 = \frac{1}{2\pi\sqrt{(L_1+L_2)C}}$

changing C nothing but tunning of Reciever adjusting padder capacitor is called tuning of reciever. $\therefore f_2 = \frac{1}{2\pi\sqrt{(L_1+L_2)(C+C_p)}}$



for fine tuning to reduce tracking error trimmer capacitor is placed in series with C.

$\therefore C_{eq} = \frac{C C_T}{C+C_T} + C_p$.

* Automatic Gain Control

→ to maintain constant o/p irrespective of ~~the~~ variation in i/p signal strength

→ if i/p strength ↑es gain ↓ses ~~drop~~ & vice versa. so that o/p gain is constant

Fading:- variation in signal strength at the i/p of the Reciever due to atmospheric conditions & multiple path Reflections

→ Recievers changes gain according to signal strength to have constant o/p.

① simple AGC. ② Delayed AGC.

→ simple AGC has disadvantage of gain reduction even for weak signal

→ Delayed AGC not provide gain reduction until some arbitarily signal level is attained.

* Squelch circuit

→ switch on the reciever when signal is present switch off the reciever when signal is absent.

* Double spotting

→ 1600 KHz station is selected when L.O. frequency is 1100 KHz as well as 2100 KHz. for an IF 500 KHz. so each station is selected twice.

* Digital communications.

→ more rugged than analog ∵ it can withstand channel noise & distortion.
→ Possibility of introducing regenerative repeaters in the former.
→ high fidelity & high privacy.

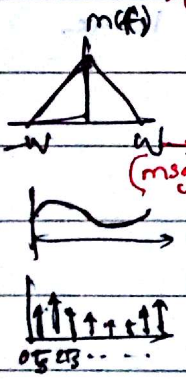
① BaseBand data Tx
- Pulse Analog
  - PAM
  - PWM
  - PPM
- Pulse digital
  - PCM
  - DPCM
  - DM
  - ADM

② Bandpass data Tx
- Binary
  - ASK
  - FSK
  - PSK
- M-array.
  - 8 PSK
  - 16 PSK
  - 32 PSK
  - QAM.

* for bandpass channel is freespace & for passband channel is co-axial cable or fiber optic cable or twisted pair

* We can not Tx digital data (Binary data) in freespace. ∵ its low freq. spectrum, hight of antena should be very large so baseband signal is converted to ~~pass~~ bandpass signal or digital data converted to analog then modulate & Txed.

* Sampling :- Analog s/g $\xrightarrow{to}$ digital s/g. (ADC)



$f_s \geq 2W \Rightarrow T_s \leq \frac{1}{2W}$; W → fm ; W – highest freq. of s/g.

No. of samples more reconstruction is very close to i/p.

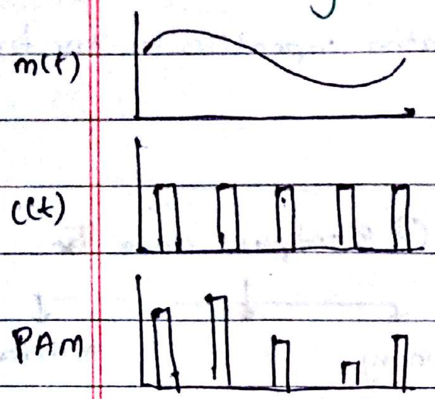
∴ $\boxed{f_s = 2W}$ → fm is called Nyquist rate

if $f_s < 2f_m$ aliasing occurs loss of data.

* Baseband data Tx

① * Pulse Analog

*1) Pulse Analog modulation (PAM)

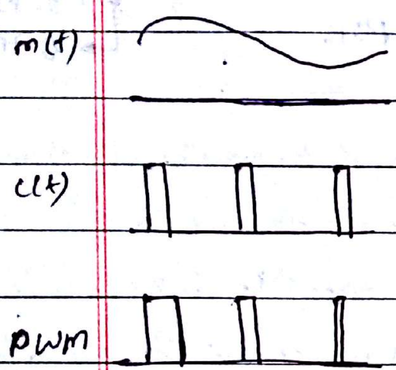


* carrier is high freq: rectangular pulse
* amp. is change according to sampled value

generation:-

m(t) → multiplier → PAM signal
c(t) ↑ (into multiplier)

*2) Pulse width Modulation (PWM)



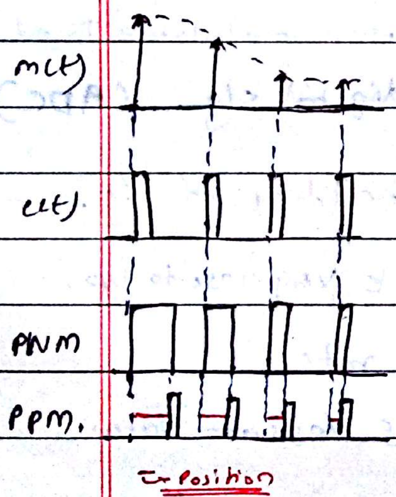
* Pulse duration modulation or pulse length modulation
* width is changes according to sample value.

Generation

m(t) → (Σ) → Slicer → PWM signal

Periodic sawtooth waveform. ↑ (into Σ)

*3) Pulse position Modulation (PPM)



* Pulse time modulation.

* position of pulse change according to m(t) value.

② * Pulse Digital.

*1) PCM (Pulse code modulation.

Source of continuous time signal → LPF → Sampler → Quantizer → Encoder → PCM signal

(Quantizer, Encoder: done by ADC)

Tx channel — Regenerative Repeaters.

PCM signal → Decoder → Reconstruction filter → Destiny.

* Quantiser converts continuously varying amp. signal to discrete amp. sig.
* encoding use to translate discrete set of sample in 1's & 0's.

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* For n bit PCM No. of quantisation levels $= 2^n$.

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

$$\text{Bit rate } (R_b) = \text{sampling Rate} \times n = \frac{1}{T_s} \times n = \frac{n}{T_s}$$

$$\text{Bit duration } (T_b) = \frac{T_s}{n} = \frac{1}{R_b} \qquad \text{Max. B.W.} = \frac{n}{T_s} = \frac{1}{T_b}$$

$$\text{min B.W.} = \frac{1}{2T_b}$$

* Regenerative Repeaters reproduce original sig free from noise
* in quantizer each sample is rounded of to the nearest quantization levels. present in quantizer depend on encoder.

8V
} 11→7V
6V
} 10→5V
4V
} 01→3V
2V
} 00→1V
0V
here 4 levels

Let samples from 0 to 8V. if encoder is 2bit ADC.

$\therefore$ levels $= 2^2 = 4$. devide 0 to 8V in 4 equal intervals.

or step size = 2V.

* sample between 0–2 is rounded off to 00 at receiver. 00 value is used so original value can not be recovered. The diff betn that original value & quantised level value is called Quantiser error. & distortion due to this is quantisatn noise.

Quantised error q_e = sampled value − quantised value.

$$(q_e)_{max} = \frac{\Delta}{2} \quad \left(q_e \text{ varies from } -\frac{\Delta}{2} \text{ to } +\frac{\Delta}{2}\right)$$

* To reduce Quantisation error Reduce step size (Δ) or ↑se leves (L)

* level increases by ↑sing 'n' (bits)

Companding.

Passing the baseband sig through compressor then compressed sig sent to uniform quantizer. At the reciever this is compensated by expander this process is called companding

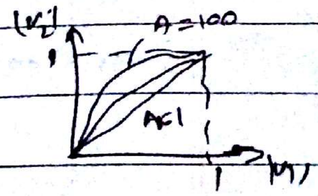
Two methods are used ① μ-Law ② A-law.

(L.S.)

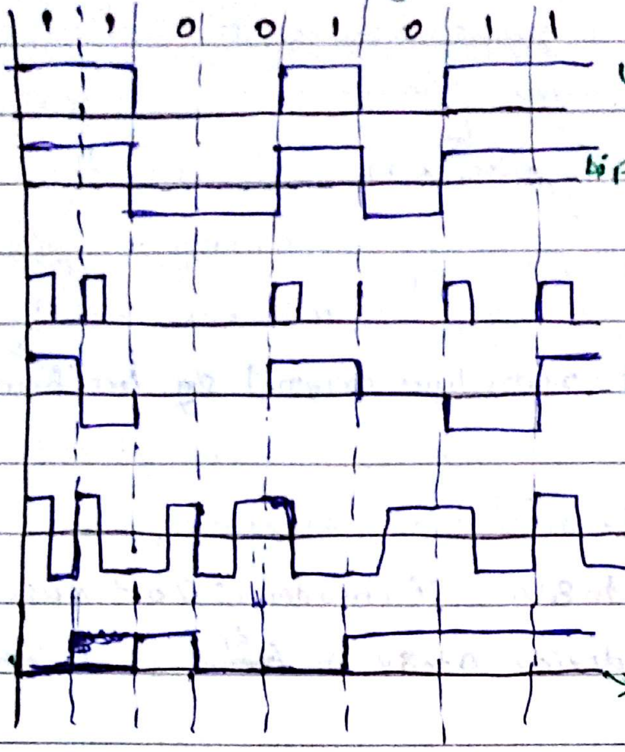
① μ law → $|V_2| = \dfrac{\log(1+\mu|V_1|)}{\log(1+\mu)}$

② A law → $|V_2| = \dfrac{A|V_1|}{1+\log A} \qquad 0 \le |V_1| \le \dfrac{1}{A}$

$$\frac{1+\log(A|V_1|)}{1+\log A} \qquad \frac{1}{A} \le |V_1| \le 1$$



* Line codes (electrically representat^n of 1's & 0's)



Unipolar ON/off signaling - high for 1

Bipolar/NRZ signaling +V for 1, -Ve for 0

RZ signaling ½ pulse for 1, nothing for 0

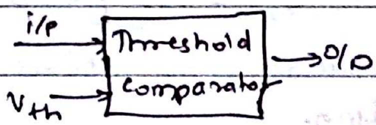
Bipolar signaling ~~pulse for 1~~ ~~depulse~~ for 0, alternate + & - pulses for 1

Manchester code (split phase) for 1 → +ve & -ve half pulses, for 0 → -ve & +ve half pulses

→ Differential encoding changes at zero, continue at ~~one~~.

* most widely used NRZ & bipolar signalling ∵ effect of noise is less ∴ used in PCM, DPCM, DM, FSK, PSK.
* On/off signalling (unipolar) used in ASK.
* Differential encoding is used in DPSK.
* RZ (Return to zero) used in fiberoptics cable.

* Regenerative Repeaters :-



If $i/p > V_{th}$ o/p high
If $i/p < V_{th}$ o/p low.

∴ noise voltage eleminated.

for NRZ threshold voltage is 0.

* Signal to quantisation Noise Ratio :-

Quantisation Noise power = mean square value of the noise

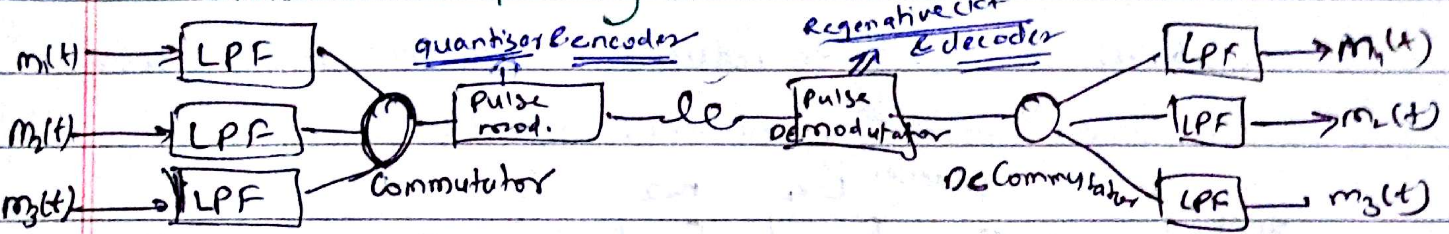
$$= \int_{-\infty}^{\infty} x^2 f(x)\, dx$$

$$= \frac{\Delta^2}{12}$$

∴ signal to quantisation Noise Ratio (SQNR) = $\frac{A_m^2/2}{\Delta^2/12}$ but $\Delta = \frac{2A_m}{2^n}$

$$\boxed{\therefore SQNR = 1.8 + 6n \text{ dB}}$$

* Time division Multiplexing (TDM)



* time taken by commutator or decommutator for one full revolutⁿ is T_s

Total no. of channels $= N$, max freq. of every channel $= f_m$.

no. of quantisatⁿ levels $= L = 2^n$, sampling freq. $= 2f_m$

sampling period $= T_s = \frac{1}{2f_m}$

one bit is used to separate every channel & one bit is used for synchronisation for one period.

$\therefore$ Total no. of bits per sampling period $= Nn + N + 1$

bit rate $= R_b = \frac{1}{T_b} = \frac{nM}{T_s}$ → (No. of signals per sample)

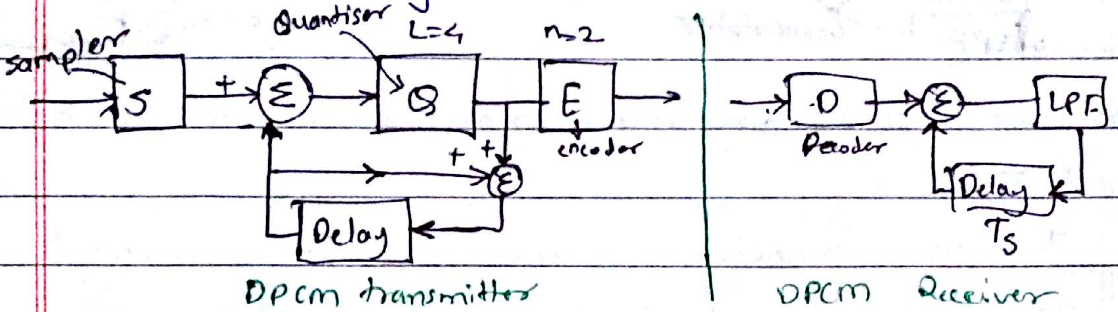
bit duration $(T_b) = T_s = \frac{\text{sampling period}}{\text{total no. of bits}} = \frac{1/2f_m}{Nn + N + 1}$

minimum B.W. $= \frac{1}{T_b} = 2f_m(Nn + N + 1)$

FDM	TDM
* carrier freq. are alloted	* time slots are alloted
* All signals Tx at same rate	* signals Tx in alloted time rate
* synchronisatⁿ not required	* synchronisation Required

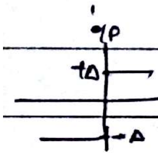
* DPCM(Differential pulse code modulation

→ DPCM is used to reduce the quantisation error without increasing number of bits.

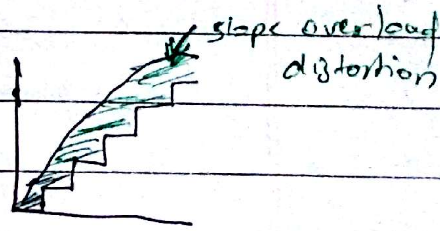


DPCM transmitter   DPCM Receiver

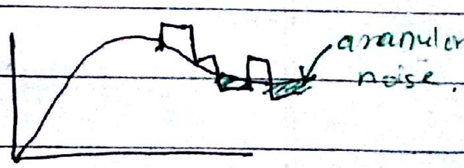
* In PCM samples are applied directly to quantiser so dynamic range is high
* In DPCM difference betn two successive samples is applied as i/p to quantiser so dynamic range is reduced.
& hence dynamic range varies with sampling rate.
* bit Rate of DPCM is same as PCM signal. $= \frac{n}{T_s} =$ B.W.
* Delta Modulation
Block diagram same as DPCM. but the encoder & Decoder is 1 bit encoder.



* It is used to decrease B.W. i.e. to reduce bit rate
⇒ characteristics of Q (i/p-o/p)
* When i/p to $\boxed{Q}$ is +ve o/p is +Δ if -ve o/p is -Δ
In DM as n=1 bit rate is same as sampling rate
$R_b = \frac{1}{T_s}$ $T_b = T_s$ $BW \propto \frac{1}{\text{pulse width}}$
at reciever end 1 is decoded as Δ & 0 as -Δ
The signal Reconstructn depends on Δ.
* Quantisatn errors in DM:-
① slope overload distortion
when step size Δ is too small for the stare case approximation.



② Granular noise
step size Δ is longer.



condition for no slope overload distortion for single tone modulatn is $\frac{\Delta}{T_s} = A_m 2\pi f_m$

Scanned by CamScanner

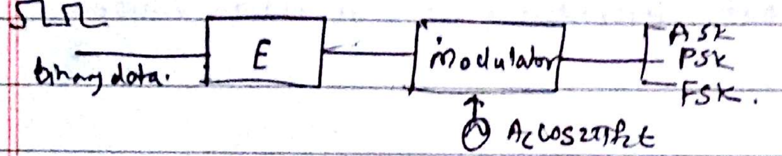
* Adaptive delta modulation. (ADM)

→ In ADM additional hardware is designed to provide variable step size thereby reducing slope overload effects without increasing the granular noise.

[illegible]

* Bandpass communication system.

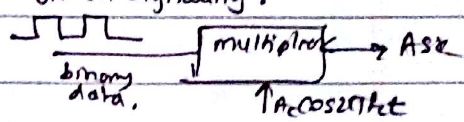
msg signal is in form of binary data but having significant low frequencies. so to Tx antina problem occures so msg signal is multiplied by anolog carrier through modulator.



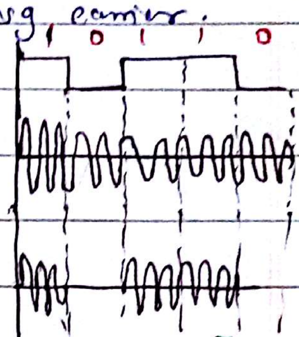
* Digital carrier modulation

① Amplitude shift keying (ASK):

*on-off signaling.



Amplitude is modulated according to msg, carrier.



$s(t) = A_c \cos 2\pi f_c t$ if $m(t)=1$

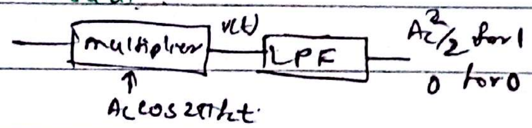
$= 0$ if $m(t)=0$

$B.W. = 2 \times \frac{1}{T_b} = 2 \times$ bit rate.

$A_c = \sqrt{\frac{2E_b}{T_b}}$

$s_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_c t$

Demodulatn:

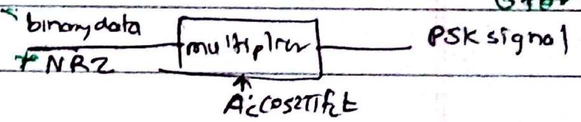


energy per bit $E_b = \int_0^{T_b} s_1(t)\, dt = P \times T_b$

$= \frac{A_c^2}{2} T_b$ for 1

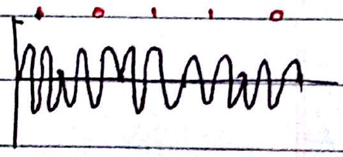
0 for 0

② phase shift keying (PSK)



$s(t) = A_c \cos 2\pi f_c t$ if $m(t)=1$

$= -A_c \cos 2\pi f_c t$ if $m(t)=0$.



$T_b = \frac{n}{f_c} \Rightarrow f_c = \frac{n}{T_b} = n f_b$

$E_b = \frac{A_c^2 T_b}{2}$

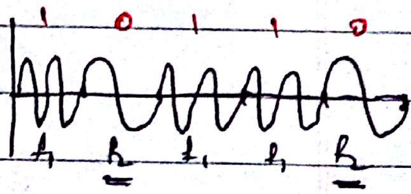
Demodulation: B.D. same as ASK.

$B.W. = 2 \times$ Bitrate $= 2 \times f_b$.

o/p $= A_c^2/2$ for 1

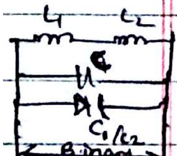
$= -A_c^2/2$ for 0.

③ Frequency shift keying (FSK):

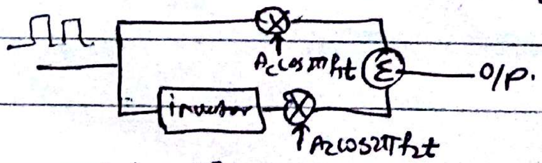


$s(t) = A_c \cos 2\pi f_1 t$ if $m(t)=1$

$= A_c \cos 2\pi f_2 t$ if $m(t)=0$.



generation:-



$f_1 = f_c + K_f V \Rightarrow$ Mark freq.

$f_2 = f_c - K_f V \rightarrow$ space freq

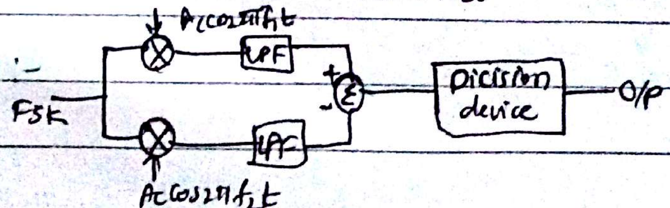
$B.W. = 2\Delta f + 2f_b$

$= f_1 - f_2 + 2f_b$

$f_1 = \frac{1}{2\pi\sqrt{L_1 (C_1+C_2)}}$

$f_2 = \frac{1}{2\pi\sqrt{(L_1+L_2)C(C_1+C_2)}}$

Demodulation:-

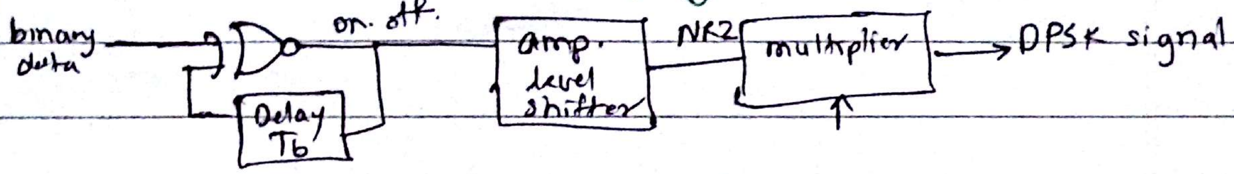


if $s(t) = A_c \cos 2\pi f_1 t$ o/p is $\frac{A_c^2}{2}$

if $s(t) = A_c \cos 2\pi f_2 t$ o/p is $-\frac{A_c^2}{2}$

cut off frequency of LPF should be less than $(f_1 - f_2)$ so that $f_1 - f_2$ terms is also supressed.

Differential Phase shift keying (DPSK).



→ amplituck level shifter use to change on off signaling to NRZ by adding negative dc level

→ DPSK not require a coherent carrier for modulation.

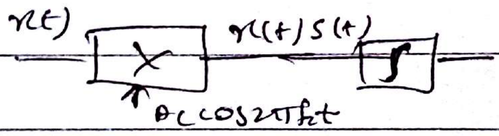
Matched filter.

→ Optimum filter is an integrator that gives min. probability of error.

→ Matched filter is the optimum filter with white noise as i/p.

$H(f) = k g(T-t)$ $g(t)$ is i/p signal.

Correlatⁿ Receiver



$$y(T) = \int_0^{T_b} r(\tau)\, s(\tau)\, d\tau$$

In general correlation receiver & match filter are same.

$$P_e = Q\left[\sqrt{\frac{E_b}{2N_0}}\right]$$