Princide of communication system 11/1/23

- **Source of information**: It will create some message, it can be audio signal. or video signal, data, picture etc

↓

- **Input Transducer**: Input Transducer is a device that will convert form of signal to some electrical signal e.g (microphone)

↓

- **Transmitter**: To process the signal so that it can transmitt (receive some electrical path) i) amplification of the signal
ii) modulation } application

↓

- **Channel**: Connect transmeter with reciever
Some unwanted signal

- **Reciever** → • **Users**

• Baseband signal

Signal from each source is called Base band signal.

modulation

- continuous modulation (Am, Fm, Pm)
- pulse modulation (PAM, PWM, PPM)

• Need of modulation

i) To reduce the antenna height $\left[l = \frac{c}{4f} \right]$

[l = Antenna length]

[Practicality of Antenna height]

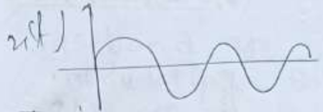
ii) To remove interference

iii) Reduction of noise

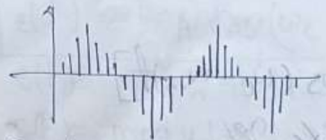
(transmitt analog signal) [continuous]

• Analog communication and Digital communication (transmitt digital signal)

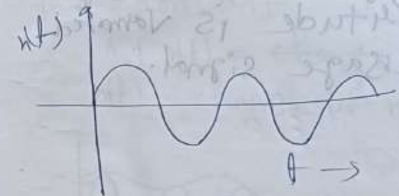
• $\boxed{\text{Analog signal} = 1}$



It is continuous in time and amplitude.

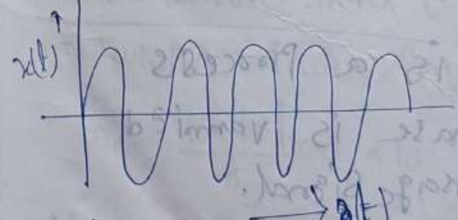


It is discrete in time & amplitude.



• $\boxed{\lambda = \frac{c}{f}}$ [c = speed of light, f = frequency] [λ = wavelength]

• $\boxed{\omega = 2\pi f}$



• $\boxed{l = \frac{\lambda}{4} = \frac{c}{4f}}$

[l = Antenna length]

• modulation

$A_m \cos(\omega_m t) \cdot A_c \cos(\omega_c t)$

$\frac{A_m A_c}{2}\left[2\cos(\omega_m t)\cos(\omega_c t)\right]$

$\frac{A_m A_c}{2}\left[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t\right]$

• Amplitude modulation

Amp modulation is a Process by which carrier amplitude is varried according to the message signal.

• Frequency modulation:

Frequency modulation is a Process by which carrier frequency is varried according to the message signal.

• Phase modulation:

Phase modulation is a Process by which carrier phase is varried according to the message signal.

• Amplitude modulation (Am)

• Defination-

Amplitude modulation is a Process by which carrier amplitude is varried according to the message signal.

• Ex Pression (•)

Baseband or message signal or modulating signal

$x(t)$ - message signal

$\boxed{e(t) = A\cos(\omega_c t)}$

$e(t)$ = High frequency of carrier signal

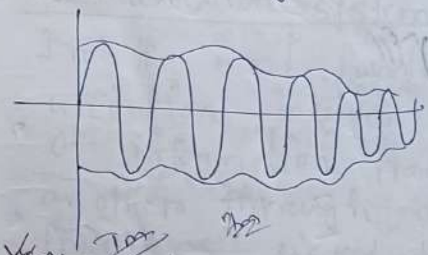
$s(t)$ = modulated signal (•)

$s(t) = [A + x(t)]\cos(\omega_c t)$

[ω_c = Constant

$A + x(t)$ = ~~change~~ $E(t)$ = Envelop

$x(t)$ = change of message signal]



• modulation Index (•)

modulation index for amplitude modulation = m_a

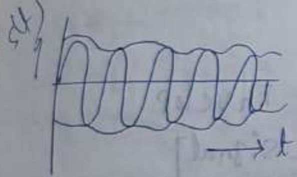
$m_a = \frac{|x(t)|_{max}}{A} \times 100$

Also known as depth of modulation, degree of modulation or modulation factor.

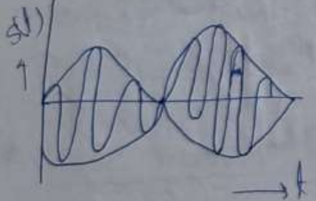
Amplitude modulation

- under modulation ($m_a < 1$)
- perfect modulation ($m_a = 1$)
- over modulation ($m_a > 1$)

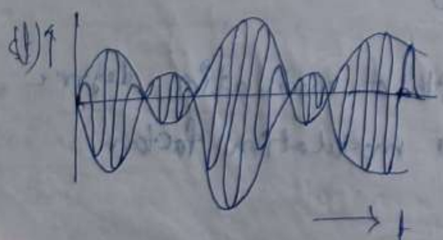
• under modulation (•)



• Perfect modulation (•)



• over modulation (•)



• Introduction

Communication is the Process of establishing connection or link between two Points for information exchange.

(OR)

Communication is simply the Process of conveying message at a distance or Communication is the Basic Process of exchanging information.

• The communication Process : Elements of a communication system

In the most fundamental sense, communication involves the transmission of information from one point to another through a succession of Processes as listed below:

1) The generation of a thought Pattern or image in the mind of an originator.

2) The description of that image, with a certain measure of Precision, by a set of oral visual symbols.

3) The encoding of these symbols in a form that is suitable for transmission

over a physical medium of interest.

4) The transmission of the encoded symbols to the desired destination.

5) The decoding and reproduction of original symbols.

6) The recreation of the original thought pattern or image with a definable degradation in quality in the mind of a reapent.

Figure below shows the block diagram of a general communication system in which different functional elements are represented by blocks. The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.

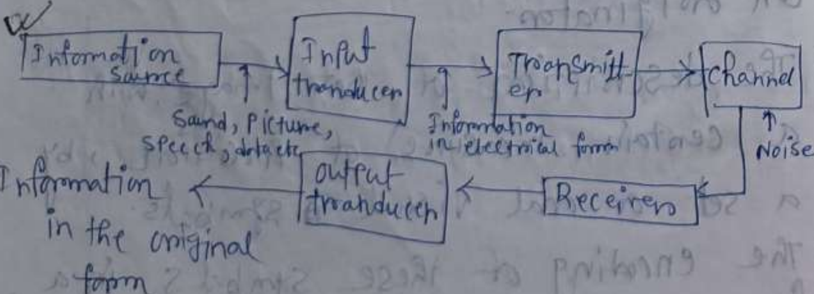


Fig:- Block diagram of a communication system

1) Information Source:- The function of information source is to produce required message which has to be transmitted.

2) Input Transducer:- It is used to convert the original message signal into a time varying electrical signal.

3) Transmitter- The function of transmitter is to process the electrical signal from different aspects. For example in radio broadcasting the electrical signal obtained from sound signal, is processed to restrict its range of audio frequencies and is often amplifiers modulation is the main function of it.

4) Channel:- The function of channel is to provide a physical connection between the transmitter and receiver. These are the types of channel point to point and broad cast channels.

5) Receiver:- The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal.

e) Destination :- It is the final stage which is used to convert an electrical message signal into its original form.

• Baseband signal :- The message signal generated from the information source is known as baseband signal. This baseband signal may be a combination of two or more message signals. If the baseband signal is transmitted directly, then it is known as baseband transmission.

The baseband signal cannot usually be transmitted through space by radio because the antennas required are too long and multiple baseband signals ~~transmitted~~ transmitting simultaneously would interfere with one another.

• Modulation process - The purpose of communication system is to deliver a message signal from an information source in a recognizable form to a user destination, with ~~a recognit~~ the source and the user being physically seperated from each other.

To do this the transmitted modifies the message signal into a form which is suitable for transmission over channel. Then modification is achieved by means of a process known as modulation.

Modulation may be defined as the process by which some characteristics of a signal called carrier is varied in accordance with the instantanious value of another signal called modulating signal. Signals containing information or intelligence are referred as modulating signal. This information bearing signal is also called baseband signal.

The carrier signal frequency is greater than modulating signal frequency. The signal resulting from the process of modulation is known as modulated signal.

Type of modulation – modulation is basically two types:

(i) Continuous wave modulation – when the carrier wave is continuous in nature, the modulation process is known as continuous wave modulation or analog modulation.

Examples of continuous wave modulation are AM and FM.

(ii) Pulse modulation – when the carrier wave is a pulse-type waveform, modulation process is known as pulse modulation.

The analog pulse modulation may be of following three types:

(i) Pulse amplitude modulation (PAM)

(ii) " " duration (PDM)

(iii) " " position (PPM)

Need for modulation or Benifits of modulation

We need modulation for the following reason –

(i) Practicality of antenna – we know that in case when free space is used as transmitting medium (i.e. channel) messages are transmitted and received with the help of antennas. For efficient radiation and reception the transmitting and receiving antennas must have lengths comparable to a quarter-wavelength of the frequency used. For example in AM broadcast systems, the maximum audio frequency transmitted from a radio station is of order of 5 kHz. If this message audio signal is transmitted without modulation, then the height of the antenna required for an effective radiation and reception will be 1/4th of the wavelength given as

$$l = \lambda/4 = \frac{c}{4f} = \frac{3\times10^8}{4\times5\times10^3} = 15\,km$$

obviously, it will be totally impracticable to construct and install an antenna of such a height. However this height of the antenna may be ~~radiation and reception is achieved~~. ~~In modulation radiation~~ reduced by modulation technique and yet effective radiation and reception is achieved. In modulation process audio signal at radio stations are translated to higher frequency spectrum i.e radio frequency range. These higher radio frequencies with the small wavelength act as for audio frequency (i.e modulating signal). Thus the height of the antenna required is much reduced and becomes practical.

As an example, if an audio frequency is translated to a radio frequency carrier of frequency 3 MHz, the antenna height required would be

$l = \lambda/4 = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 3 \times 10^6} = 25$ m

this antenna height may be achieved practically.

(ii) To remove interference: we know that the frequency range of audio signal is from 20 Hz to 20 kHz. In radio broadcasting there are several radio stations. In case there is no modulation all these stations transmit audio or sound signal in the range of 20 Hz to 20 kHz. Due to this transmission over same range the programmes of different station will get mixed up.

Hence in order to keep the various signals seperate, it is necessary to translate or shift them to different portions of the electromagnetic spectrum. Thus each station is allocated a band of frequency. This also overcomes the drawback of poor radiation efficiently at low frequency

(iii) Reduction of noise:- Noise is the major limitation of any communication. Although noise can not be eliminated completely, but with the help of several modulation technique, effect of noise can be minimized.

(b) Analog and Digital Communication:-

Depending upon the message signal communication may also be classified as under

(i) Analog Communication
(ii) Digital Communication

(i) Analog Communication:- It is that type of communication in which the message or information signal to be transmitted is analog in nature.

(ii) Digital Communication:- In digital communication the message signal to be transmitted is digital in a nature. this means that digital communication involves the transmission of information in digital form.

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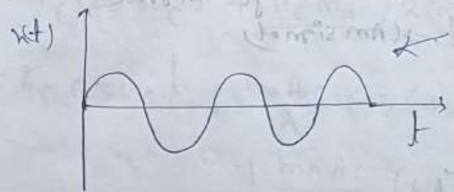
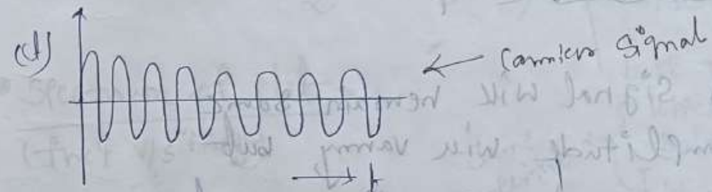
Single tone Amplitude modulation

$\omega_c >> \omega_m$

$c(t) = A_c \cos \omega_c t$
$x(t) = A_m \cos \omega_m t$ } A_m (is) needed for long distance communication





Anteena height/length $\propto$ frequency

$l = \lambda/4$ [λ = wave length of a signal]

$= \frac{c}{4f}$ [λ = wavelength]

[l = Anteena length]

Ex-

If $f = 3KHz$, $l = \frac{3\times10^8 \ m}{4\times3\times10^3}$

$= 75\times10^4 \ m$

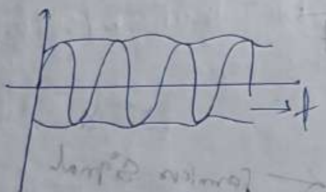
For Increase the length we have to decrease frequency.

multitone frequency modulation -
$x(t) = (A_{m_1}\cos\omega_{m_1}t + A_{m_2}\cos\omega_{m_2}t)$

~~Sin~~ singletone modulation -
frequency

the merge / overlap of message signal
or carrier signal. → (Am signal)

$s(t)$



Carrier signal will remain same
(may amplitude will varry but
frequency will remain constant).

modulating signal $x(t) = A_m \cos\omega_m t$ [$\omega_m = 2\pi f_m$]
Carrier signal $c(t) = A_c \cos\omega_c t$
[ω_c = frequency]

According to the defination of ~~m~~ amplitude modulation,

Am signal → $\boxed{s(t) = [A_c + x(t)]\cos\omega_c t}$ (i)

$= [A_c + A_m \cos\omega_m t]\cos\omega_c t$

~~$= A_c \cos\omega_c t + A_m \cos$~~

$= A_c\left[1 + \left(\frac{A_m}{A_c}\right)\cos\omega_m t\right]\cos\omega_c t$ $\boxed{m_a = \frac{|x(t)|_{max}}{A_c}}$

$= A_c[1 + m_a \cos\omega_m t]\cos\omega_c t$

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

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

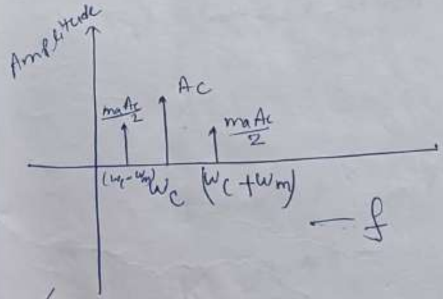
$= A_c \cos\omega_c t + \frac{m_a A_c}{2}\cos(\omega_c + \omega_m)t$
$+ \frac{m_a A_c}{2}(\omega_c - \omega_m)t$

[As $x(t) = A_m \cos\omega_m t$
A_m = max amplitude of message signal]

[$E(t) = A_c + x(t)$
Envelope]
That contains message info

[$\omega_c + \omega_m$ = Upper side Balance = USB]
[$\omega_c - \omega_m$ = Lower Side Balance = LSB]
same amplitude value

- Spectrum of plot =
(freq v/s Amplitude)



[$s(t)$ contains 3 frequency term = carrier freq, USB, LSB]

- $\boxed{\text{Band width} = USB - LSB}$

Range of freq. present in Am = Band width (BW)

$BW = USB - LSB$
$= (\omega_c + \omega_m) - (\omega_c - \omega_m)$
$\boxed{BW = 2\omega_m}$ [for Am signal]
[ω_m = message signal frequency]

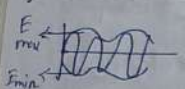
[For transmitting message ~~the~~ with an wireless medium antenna is used] for Am,

$\boxed{BW = 2 \times \text{message signal frequency}}$

mathematical calculation of modulation index

$$E(t) = A_c(1 + m_a \cos\omega_m t)$$

Find interms of Amplitude(Amp) (•)



$$E(t) = A_c(1 + m_a \cos\omega_m t)$$

$\boxed{E_{max} = A_c(1 + m_a)}$ — (i) [when, $\cos\omega_m t = 1$]

$\boxed{E_{min} = A_c(1 - m_a)}$ — (ii) [when, $\cos\omega_m t = -1$]

$$\frac{E_{max}}{E_{min}} = \frac{1 + m_a}{1 - m_a}$$

$\Rightarrow E_{max} = E_{max} m_a + E_{min} + E_{min} m_a$

$\Rightarrow m_a(E_{max} + E_{min}) = E_{max} - E_{min}$

$$\Rightarrow \boxed{m_a = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}}$$

• Power Content of Am signal

$$E(t) / s(t) = A_c \cos\omega_c t + \frac{A_c m_a}{2} \cos(\omega_c + \omega_m)t + \frac{A_c m_a}{2} \cos(\omega_c - \omega_m)t$$

If,

$x(t) = A \cos\theta$

then, $P(\text{Power}) = \dfrac{\left(\frac{A}{\sqrt{2}}\right)^2}{R}$

[A = Rms value of amplitude]

[R = Resistance through which we will transmitt signal power]

$$P_t = P_c + P_{USB} + P_{LSB}$$

[P_t = total power]

[P_{USB} = Power of USB, P_{LSB} = " " LSB]

[$A_c m_a$ = Rms value of USB and LSB amp (GRms or RRms)]

$$= \frac{\left(\frac{A_c}{\sqrt{2}}\right)^2}{R} + \frac{\left(\frac{A_c m_a}{2\sqrt{2}}\right)^2}{R} + \frac{\left(\frac{A_c m_a}{2\sqrt{2}}\right)^2}{R}$$

$$= \frac{A_c^2}{2R} + \frac{A_c^2 m_a^2}{4 \times 2R} \times 2$$

$$= \frac{A_c^2}{2R} + \frac{A_c^2 m_a^2}{4R}$$

$$= \frac{A_c^2}{2R} + \frac{A_c^2 m_a^2}{8R} + \frac{A_c^2 m_a^2}{8R}$$

$$= \frac{A_c^2}{2R}\left(1 + \frac{m_a^2}{4} + \frac{m_a^2}{4}\right)$$

$$\boxed{P_t = \frac{A_c^2}{2R}\left(1 + \frac{m_a^2}{2}\right)}$$

$m_a > 1$, Above modulation

$m_a = 1$, Perfect modulation

$m_a < 1$, under modulation

$$P_t = P_c\left(1 + \frac{1}{2}\right) = P_c \times 1.5$$

$\boxed{P_t = 1.5 P_c}$ [P_c = perfect modulation]

[Total Power = 1.5 × Perfect modulation]

Ex-(1)

Suppose we have a modulating signal

$x(t) = 10 \sin(2\pi \times 10^3 t)$

$c(t) = 20 \sin(2\pi \times 10^4 t)$

i) find ma?

ii) Percentage of ma?

iii) frequency of side band USB LSB?

iv) Amp of USB, LSB?

v) Band width?

i) $m_a = \frac{10}{20} = \frac{1}{2} = 0.5$, under modulation

ii) $m_a \times 100 = 50\%$. $[\because m_a = \frac{A_m}{A_c}]$

iii) $USB = \cancel{\omega_c + \omega_m}$

$f_c + f_m$ [$\because$ linear frequency in "Hz" f_c, f_m]

$= (10^4 + 10^3)$ Hz

$= 11 \times 10^3$ Hz

$LSB = f_c - f_m$

$= (10^4 - 10^3)$ Hz

$= 9 \times 10^3$ Hz

(iv) $LSB = USB = \frac{m_a A_c}{2} = \frac{0.5 \times 20}{2}$

$= 5$

(v) Band with $= 2 \times f_m$ (for am)

$= 2 \times 10^3$ Hz

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Ex-(2)

A 400 watts carrier is modulated to a depth of modulation of 75%. Find the total power in Am signal. Assume the modulating signal to be sinusoidal one.

$P_t = P_c\left(1 + \frac{m_a^2}{2}\right)$ → EXP = Imp

$= 400\left(1 + \frac{(0.75)^2}{2}\right)$ [P_c = Carrier Power]

$= 512.5$ watt [$\because 75\% = 0.75$]

(3) Same Problem, P_t (total power) given is 60%. Find the depth of modulation or carrier power?

$P_c = \frac{P_t}{\left(1 + \frac{m_a^2}{2}\right)}$

Transmission efficiency (•)

[P_c = Carrier power
P_S = Side band power]

$P_t = P_c + P_S$ [P_S will contain upper side band power and lower side band power]

[P_S = side band powers]

$\eta = \frac{P_S}{P_t} \times 100\%$

[Exp of $s(t) = A(1 + m_a \cos\omega_m t)\cos\omega_c t$

$= A\cos\omega_c t + \frac{A m_a}{2}\cos(\omega_c+\omega_m) + \frac{A m_a}{2}\cos(\omega_c - \omega_m)$]

$A\cos\omega_c t$: that is wastage of power as it has no message information; transmit this power then.

This side band signal part carries message information.

$= \frac{P_c \times \frac{m_a^2}{2}}{P_c(1+\frac{m_a^2}{2})} \times 100$

$$\boxed{\eta = \frac{m_a^2}{2+m_a^2} \times 100\%}$$

[$\because P_t = P_c(1+\frac{m_a^2}{2}) = P_c + (P_c \frac{m_a^2}{2})$]

for perfect modulation,

$P_c = 1$

$\therefore \eta = \frac{(1)}{2+(1)^2} \times 100\%$

$= \frac{1}{3} \times 100$

$\boxed{\eta = 33.33\%}$ [For perfect modulation]

✓

Bandpass filter = tuned circuit

Current calculation (•)

I_c = rms value of carrier current

I_t = rms value of total current

Antenna resistance = R (current will flow through R)

$P_t = I_t$

$\boxed{P_t = I_t^2 R}$

$\boxed{P_c = I_c^2 R}$

Now,

$P_t = P_c(1 + \frac{m_a^2}{2})$

$\Rightarrow \frac{P_t}{P_c} = 1 + \frac{m_a^2}{2}$

$\Rightarrow \frac{I_t^2 R}{I_c^2 R} = 1 + \frac{m_a^2}{2}$

$\Rightarrow I_t^2 = I_c^2(1 + \frac{m_a^2}{2})$

$\Rightarrow \boxed{I_t = I_c\sqrt{1+\frac{m_a^2}{2}}}$

For multitone amplitude modulation signal (•)

$$\boxed{P_t = P_c\left[1 + \frac{m_1^2}{2} + \frac{m_2^2}{2} + \frac{m_3^2}{2}\right]}$$

[Suppose we our message signal contains 3 mess frequency]

for n frequency

$$\boxed{P_t = P_c\left[1 + \frac{m_1^2}{2} + \frac{m_2^2}{2} + \frac{m_3^2}{2} + \cdots\right]}$$

we can write,

$P_t = P_c(1 + \frac{m_t^2}{2})$ where $t = 1, 2, \ldots n$ [$\because \frac{m_1^2}{2} + \cdots + \frac{m_n^2}{2}$ will go to $\frac{m_n^2}{2} = \frac{m_t^2}{2}$]

so, from comparison,

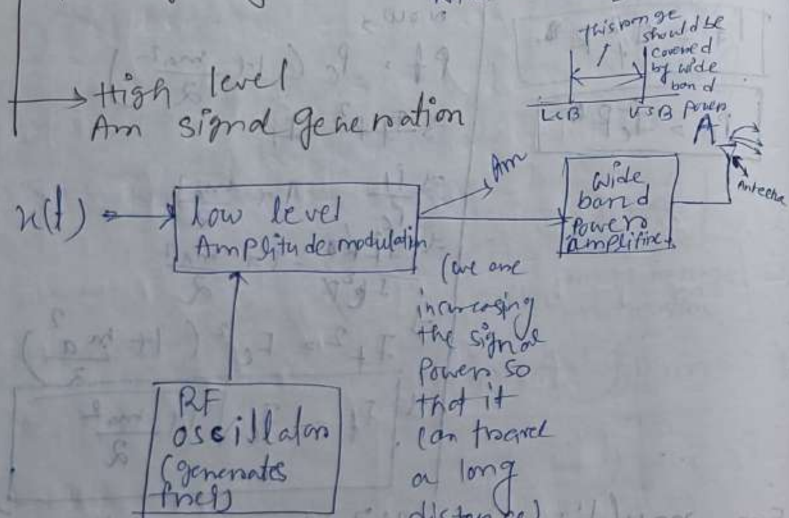
$$\boxed{m_f = \sqrt{m_1^2 + m_2^2 + m_3^2 + \cdots + m_n^2}}$$

[m_f = Equivalent modulation index]

Generation of Am signal

→ low level Am signal generation (carrier power will be low)

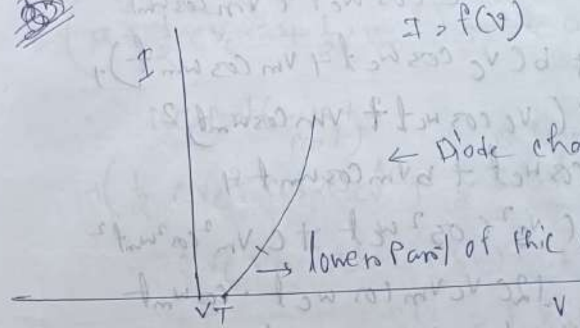
→ High level Am signal generation



(we are increasing the signal power so that it can travel a long distance)

• Square law Diode modulation

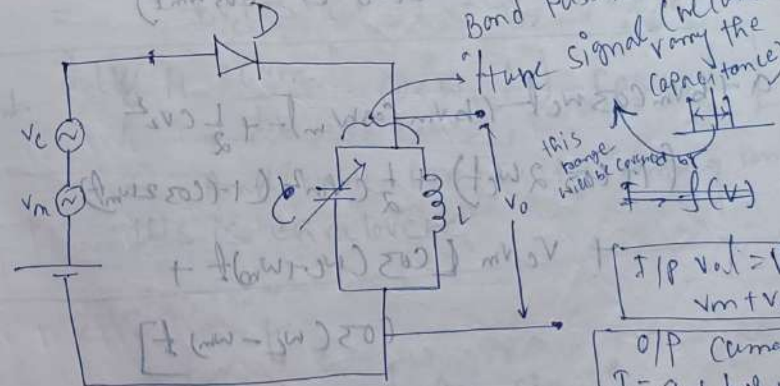
$I = f(V)$



← Diode characteristics

→ lower part of the char is very much non linear that helps to generate carrier signal

Ckt



Band pass filter

→ tune signal (we can vary the capacitance)

this range will be covered by $I = f(V)$

$\boxed{I/P \text{ val} = V = V_m + V_c}$

$\boxed{O/P \text{ Current } I = a + bV + cV^2 \text{ (non-linear eqn)}}$

V_{AA} is used to set the Operating point = Biasing (fixed operating point) voltage, So that the diode can be Operated, this should not considered as I/P val.

(0)

$v_c = V_c \cos w_c t$

$v_m = V_m \cos w_m t$

$\therefore v$ (i/p val) $= v_c + v_m$

$= V_c \cos w_c t + V_m \cos w_m t$

$\therefore I = a + b(V_c \cos w_c t + V_m \cos w_m t) +$
$c(V_c \cos w_c t + V_m \cos w_m t)^2$

$= a + bV_c \cos w_c t + bV_m \cos w_m t +$
$c(V_c^2 \cos^2 w_c t + c V_m^2 \cos^2 w_m t^2$
$+ 2c V_c V_m \cos w_c t \cdot \cos w_m t$

$= a + bV_c \cos w_c t + bV_m \cos w_m t +$
$\frac{1}{2} \cdot cV_c^2 (2\cos^2 w_c t) + \frac{1}{2} cV_m^2 (2\cos^2 w_m t)$
$+ cV_c V_m (2\cos w_c t \cdot \cos w_m t)$

$= a + bV_m \cos w_c t + bV_m \cos w_m t + \frac{1}{2} cV_c^2 (1 + \cos 2w_c t) + \frac{1}{2} cV_m^2 (1 + \cos 2w_m t)$
$+ V_c V_m [\cos(w_c + w_m)t + \cos(w_c - w_m)t]$

$[\because 2\cos a \cos b = \cos(a+b) + \cos(a-b)]$

$= a + bV_c \cos w_c + bV_m \cos w_m + \frac{1}{2} cV_c^2 (1 + \cos 2w_c t)$
$+ \frac{1}{2} cV_m^2 (1 + \cos 2w_m t) + cV_c V_m \cos(w_c + w_m)t + cV_c V_m \cos(w_c - w_m)t$

$= \left(a + \frac{1}{2} cV_c^2 + \frac{1}{2} cV_m^2\right) + bV_c \cos w_c t + bV_m \cos w_m t$

① DC part (constant part) ② carrier s. ③ modulating s.

$+ \left(\frac{1}{2} cV_c^2 \cos 2w_c t + \frac{1}{2} cV_m^2 \cos 2w_m t\right) +$ ④ harmonics of carrier s. & m.s. (modulating)

$cV_c V_m \cos(w_c + w_m)t + cV_c V_m \cos(w_c - w_m)t$ ⑤ ⑥

[Rest of the signals are filtered/removed by tune filter]

$I_{out} = bV_c \cos w_c t + cV_c V_m \cos(w_c + w_m)t + cV_c V_m \cos(w_c - w_m)t$

$= bV_c \cos w_c t + 2cV_c V_m \cos w_c t \cos w_m t$

$$\boxed{I_{out} = bV_c\left(1 + \frac{2cV_m}{b} \cos w_m t\right) \cos w_c t}$$

Amplitude part ∴ it is varying part
this is envelope

⇓

low level AM signal generation

AM → DSB - FC
F = Full

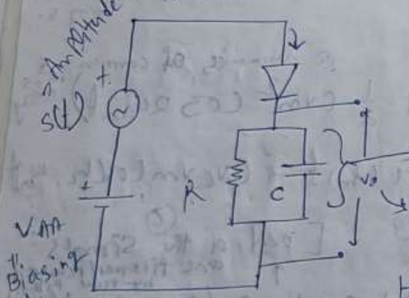
Types

• ~~Demodulation~~ of Am Demodulators

→ ① Square law detector (for low level amp modulated sig. detection or demodulation)

→ ② Envelope detector (for Am signal with large carrier detection)

① Square law detector



Here, $v = A(1+m_a\cos\omega_m t)\cos\omega_c t$

$I = f(v)$

$I = av + bv^2$

$= aA(1+m_a\cos\omega_m t)\cos\omega_c t + b\{A(1+m_a\cos\omega_m t)\cos\omega_c t\}^2$

$= (aA + aAm_a\cos\omega_m t)\cos\omega_c t + b\{A^2(1+m_a^2\cos^2\omega_m t + 2m_a\cos\omega_m t)\cos^2\omega_c t\}$

Final exp. x(t) cos ω_m t. Have to find low pass messages.

$= aA\cos\omega_c t + aAm_a\cos\omega_m t\cos\omega_c t + b(A^2\cos^2\omega_c t + A^2 m_a^2\cos^2\omega_m t\cos^2\omega_c t + 2m_a\cos\omega_m t\cdot\cos^2\omega_c t\}$

$= aA\cos\omega_c t + \frac{1}{2}aAm_a(2\cos\omega_m t\cdot\cos\omega_c t) + \frac{1}{2}bA^2\, 2\cos^2\omega_c t + \frac{1}{4}bA^2 m_a^2\cdot 4\cos^2\omega_m t\cos^2\omega_c t + \frac{1}{2}bm_a\cdot(2\cos^2\omega_c t)\cdot\cos\omega_m t$

$= aA\cos\omega_c t + \frac{1}{2}aAm_a[\cos(\omega_m+\omega_c)t + \cos(\omega_m-\omega_c)t] + \frac{1}{2}bA^2(1+\cos 2\omega_c t) + \frac{1}{4}bA^2m_a^2(2\cos\omega_m t\cdot\cos\omega_c t)^2 + \frac{1}{2}bm_a\cdot(1+\cos 2\omega_c t)\cdot\cos\omega_m t$

$= aA\cos\omega_c t + \frac{1}{2}aAm_a[\cos(\omega_m+\omega_c)t + \cos(\omega_m-\omega_c)t] + \frac{1}{2}bA^2(1+\cos 2\omega_c t) + \frac{1}{4}bA^2m_a^2[\cos(\omega_m+\omega_c)t + \cos(\omega_m-\omega_c)t]^2 + \frac{1}{2}bm_a(\cos\omega_m t + \cos\omega_m t\cdot\cos 2\omega_c t)$

$= aA\cos\omega_c t + \frac{1}{2}aAm_a\cos(\omega_m+\omega_c)t + \frac{1}{2}aAm_a\cos(\omega_m-\omega_c)t + \frac{1}{2}bA^2 + \frac{1}{2}bA^2\cos 2\omega_c t + \frac{1}{4}bA^2m_a^2[\cos^2(\omega_m+\omega_c)t + \cos^2(\omega_m-\omega_c)t + 2\cos(\omega_m+\omega_c)t\cdot\cos(\omega_m-\omega_c)t] + \frac{1}{2}bm_a\cos\omega_m t + \frac{1}{2\cdot 2}bm_a\cdot 2\cos\omega_m t\cdot\cos 2\omega_c t$

Other types of Am

• Double side Band Suppress Carrier method (DSB-SC)

$s(t) = A(1+m_a\cos\omega_m t)\cos\omega_c t$

$= A\cos\omega_c t + Am_a\cos\omega_m t\cos\omega_c t$

$\boxed{\text{SSB-SC}}$ = Single Side Band Suppress Carrier where 1 carrier & 1 side band will be suppressed

VSB: vestigial side band = we are transmitting one complete side band, part of another side band and suppressing the carrier

Am

DSB-FC DSB-SC SSB-SC VSB

Disadvantage of Am → wastage of Power

Application of Am = The TV signal transmission etc.

(+) = $aA \cos w_c t + \frac{1}{2} aA m_a \cos(w_m + w_c) t + \frac{1}{2} aA m_a \cos(w_m - w_c) t + \frac{1}{2} bA^2 +$

$\frac{1}{2} bA^2 \cos 2w_c t + \frac{1}{4} m_a^2 \cdot bA^2$

$\left[\frac{1}{2} \cdot 2\cos^2(w_m + w_c) t + \frac{1}{2} \cdot 2\cos^2 (w_m - w_c) t + \cos(w_m + w_c + w_m - w_c) t + \cos(w_m + w_c - w_m + w_c) t\right]$

$+ \frac{1}{2} \cdot bm_a \cos w_m t + \frac{1}{4} bm_a \cdot$

$\left[\cos(w_m + 2w_c) t + \cos(w_m - 2w_c) t\right]$

$= aA \cos w_c t + \frac{1}{2} aA m_a \cos(w_m + w_c) t +$

$\frac{1}{2} aA m_a \cos(w_m - w_c) t + \frac{1}{2} bA^2 +$

$\frac{1}{2} bA^2 \cos 2w_c t + \frac{1}{8} m_a^2 bA^2 \cdot$

$\left[1 + \cos 2(w_m + w_c) t\right] +$

$+ \frac{1}{8} m_a^2 bA^2 \left[1 + \cos 2(w_m - w_c) t\right] +$

$\frac{1}{4} m_a^2 \cdot bA^2 \cdot \cos 2w_m t +$

$\frac{1}{4} m_a^2 \cdot bA^2 \cos 2w_c t + \frac{1}{2} bm_a \cos w_m t$

$+ \frac{1}{4} \cdot bm_a \cdot \cos(w_m + 2w_c) t +$

$\frac{1}{4} bm_a \cdot \cos(w_m - 2w_c) t$

$= \left(\frac{1}{2} bA^2 + \frac{1}{8} m_a^2 bA^2 + \frac{1}{8} m_a^2 \cdot bA^2\right) +$

$\frac{1}{2} bm_a \cos w_m t + \left[\frac{1}{2} aA m_a \cos(w_m + w_c) t + \frac{1}{2} aA m_a \cos(w_m - w_c) t + \frac{1}{2} bA^2 \cos 2w_c t + \frac{1}{8} m_a^2 \cdot bA^2 \cos 2(w_m - w_c) t + \frac{1}{4} m_a^2 bA^2 \cdot \cos 2w_m t + \frac{1}{4} m_a^2 \cdot bA^2 \cos 2w_c t + \frac{1}{4} bm_a \cdot \cos(w_m + 2w_c) t + \frac{1}{4} bm_a \cdot \cos(w_m - 2w_c) t\right]$

~~Iout~~ =

$= \left(\frac{1}{2} bA^2 + \frac{1}{8} m_a^2 bA^2 + \frac{1}{8} m_a^2 \cdot bA^2\right) + \frac{1}{2} bm_a \cdot \cos w_m t$

$+ \left[\frac{1}{2} \cdot bA^2 \cos 2w_c t + \frac{1}{4} m_a^2 \cdot bA^2 \cos 2w_m t + \frac{1}{4} m_a^2 \cdot bA^2 \cos 2w_c t\right]$

(+) $\frac{1}{2} aA m_a \cos(w_m + w_c) t + \frac{1}{2} aA m_a \cos(w_m - w_c)$

$\frac{1}{4}bma \cdot \cos(w_m - 2w_c)t + \left[\frac{1}{8}ma^2 \cdot bA^2 \cdot \cos 2(w_m - w_c)t\right] + \frac{1}{4}bma \cdot \cos(w_m + 2w_c)t$

$I_{out} = \frac{1}{2}aAma\cos(w_f + w_c)t + \frac{1}{2}aAma$

$\cos(w_f - w_c)t +$

$\frac{1}{4}bma \cdot \cos(w_m - 2w_c)t + \frac{1}{4}bma \cdot$

$\cos(w_m + 2w_c)t$

$= \frac{1}{2}aAma[2\cos w_f t \cdot \cos w_c t] +$

$\frac{1}{4}bma \cdot [2\cos w_m t \cdot \cos 2w_c t]$

$= 1/2 A$

(1)

30/1/23

Double side band - suppressed carrier (DSB-SC)

$s(t) = A(1 + m_a \cos w_m t)\cos w_c t$

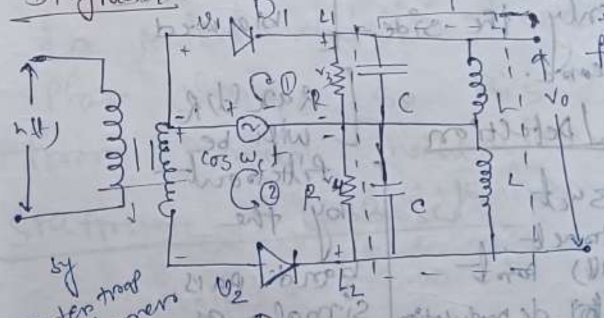
$= A\cos w_c t + \frac{Ama}{2}\cos(w_c + w_m)t + \frac{Ama}{2}\cos(w_c - w_m)t$

$s(t) = Ama\cos w_c t \cos w_m t$

$\boxed{s(t) = K\cos w_c t \cos w_m t}$

$\boxed{s(t) = Kx(t)\cos w_c t}$ $[\because x(t) = \cos w_m t]$

Diagram



→ Band Pass filter the band width freq will be pass through (2 elements) (Non linear behaviour)

By center tap transformers

$[I_2$ o/p current$]$

$I = av + bv^2$

$[$I/p voltage $= x(t)]$

$I_1 = av_1 + bv_1^2$

$[v_1 = \cos w_c t + x(t)]$

$= a\cos w_c t + ax(t) + b\cos^2 w_c t + 2b\cos w_c t\, x(t) + bx^2(t)$ — (1)

$I_2 = aV_2 + bV_2^2$

$[\because V_2 = \cos w_c t - x(t)]$

$= a\cos w_c t - ax(t) + b\cos^2 w_c t - 2bx(t)\cos w_c t$

$= b_2 + b x^2(t)$ — (2)

$v_0 = (v_3 - v_4)$ ← input of band pass filter.

$= I_1 R - I_2 R$

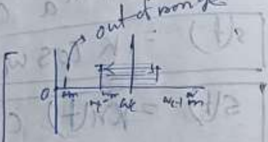
$= (I_1 - I_2) R$

$= 2a\,x(t) \cdot R + 4b\,x(t) \cos \omega_c t \cdot R$

$V_0 = 4b\,x(t) \cos \omega_c t$

$= K\,x(t) \cos \omega_c t$

There is no carrier term therefore there is no carrier, only the side band is here.



[$\because \omega_m << \omega_c$
$\therefore \omega_m$ will be blocked]

[$2a\,x(t)R$ will be filter out by the band pass signal as band pass signal is desired in that way]

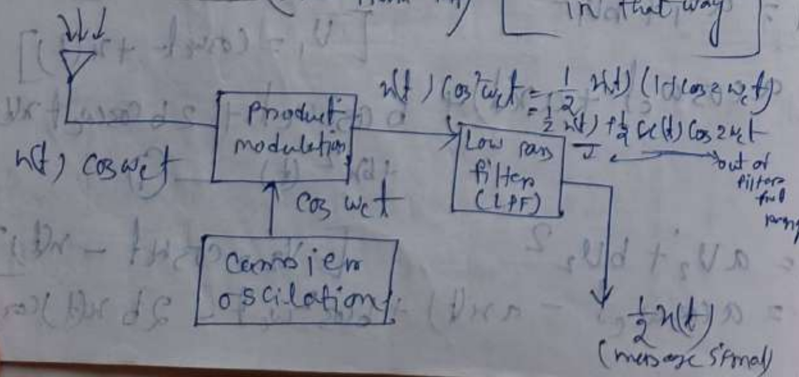
Demodulation / Detection

$s(t) = x(t) \cos \omega_c t$

↓ extract $x(t)$ part by demodulation

Synchronous demodulation of (DSB-SC) (Intro from pdf)

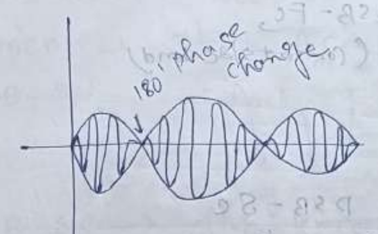
$x(t)\cos^2 \omega_c t = \frac{1}{2} x(t)(1 + \cos 2\omega_c t)$
$= \frac{1}{2} x(t) + \frac{1}{2} x(t) \cos 2\omega_c t$ → out of filters freq range



~~$s(t) = x(t) \cos \omega_c t$~~

• $x(t) = V_m \cos \omega_m t$ Plot that in all maths in notes.

Waveform of (DSB-SC)

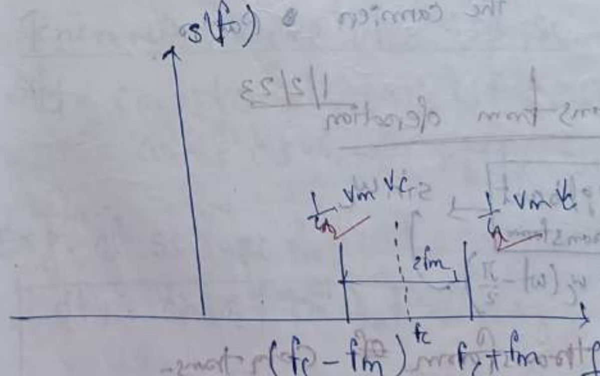


180° phase change

[the cutoff freq of LPF $= \omega_m$, if the signal freq is greater than ω_m than the signal will be ~~cut of~~ blocked]

In perfect modulation there is no phase change so this is different from perfect modulation.

Spectrum of (DSB-SC)



Here USB and LSB are identical.

$\boxed{\text{Band width here} = 2f_m}$

Transmitter power eqn $\rightarrow P_t = P_c\left(1+\frac{ma^2}{2}\right)$

$P_{t(DSB-SC)} = P_c \cdot \frac{ma^2}{2}$

if $m = 1$

Power saving, for DSB-FC, (carrier + side band)

$P_t = P_c\left(1+\frac{1}{2}\right)$

$\boxed{P_t = 1.5\, P_c}$

Power saving for DSB-SC

% of power saving (suppressing carrier power)

$= \frac{P_c}{1.5 P_c} \times 100$

$= 66.66 \times 100$

$= 66.66\,\%$ value of power is saved if we suppress the carrier power.

CW 1/2/23

Hilbert transform operation

$\cos \omega_t \rightarrow$ [Hilbert transform] $\rightarrow \sin \omega_c$

$\cos \omega_c\left(\omega t - \frac{\pi}{2}\right)$

why Hilbert transform of ~~Ga~~ transform only one ~~two~~ signal?

single ~~inf~~ i/p and o/p (points)

CW 1/2/23

single side band suppressed carrier (SSB-SC) modulation

(Introduction part) $\rightarrow$ from pdf

Exp. of double side band suppress carrier (DSB-SC) $\rightarrow s(t) = A_m \cos \omega_m t \cdot A_c \cos \omega_c t$

$= \frac{A_m A_c}{2}\left[\underbrace{\cos(\omega_c + \omega_m)t}_{USB} + \underbrace{\cos(\omega_c - \omega_m)t}_{LSB}\right]$

SSB-SC $\longrightarrow s(t) = K \cos(\omega_c + \omega_m)t \longrightarrow$ transmitting USB

$= K\left[\cos \omega_c t \cos \omega_m t \mp \sin \omega_m t \sin \omega_c t\right] \rightarrow$ USB

[You can ~~consi~~ consider $\frac{A_m A_c}{2} = K$]

[Considering K as unit : $K = 1$]

$s(t) = \cos \omega_c t \cdot \cos \omega_m t + \sin \omega_c t \sin \omega_m t \rightarrow$ LSB

Generalized exp of SSB-SC

$s(t) = \cos \omega_c t \cos \omega_m t \pm \sin \omega_c t \sin \omega_m t$ [$x(t)$ = modulating signal] (message signal)

$= \cos \omega_c t \cos \omega_m t\, x(t) \pm x_h(t) \cdot \sin \omega_c t$ [$x(t) = \cos \omega_m t$, $x_h(t) = \sin \omega_m t$]

$x_h(t)$ $\rightarrow$ Hailbert transformation

Exp. of SSB-SC (Final)

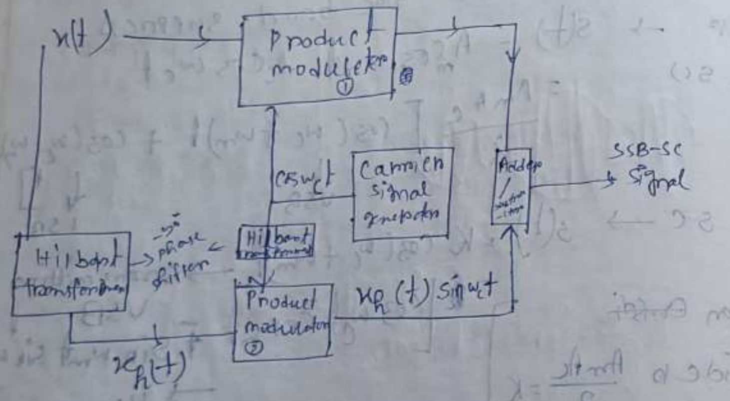
$$\boxed{s(t) = \cos \omega_c t \cos \omega_m t\, x(t) \pm x_h(t) \cdot \sin \omega_c t}$$

'+' indicates the transmission of ~~USB~~ LSB

'−' ~~LSB~~ USB

Generation of SSB-SC

$x(t) \rightarrow$ message signal (taken from signal source)



Block diagram

$$s(t) = x(t)\cos w_c t \pm x_h(t)\sin w_c t$$

$$v(t) = \cos w_c t \cdot s(t)$$

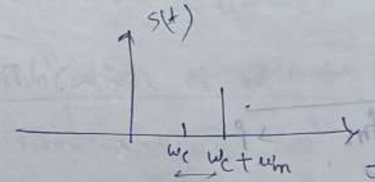
$$= \cos w_c t\,[x(t)\cos w_c t \pm x_h(t)\sin w_c t]$$

$$= x(t)\cos w_c t * \cos t \pm x_h(t)\cos w_c t * \sin w_c t$$

$$= \frac{x(t)}{2}[1+\cos 2w_c t] \pm \frac{x_h(t)}{2}\sin 2w_c t$$

$$= \frac{1}{2}x(t) \quad (o/p)$$

~~Action~~ Spectrum



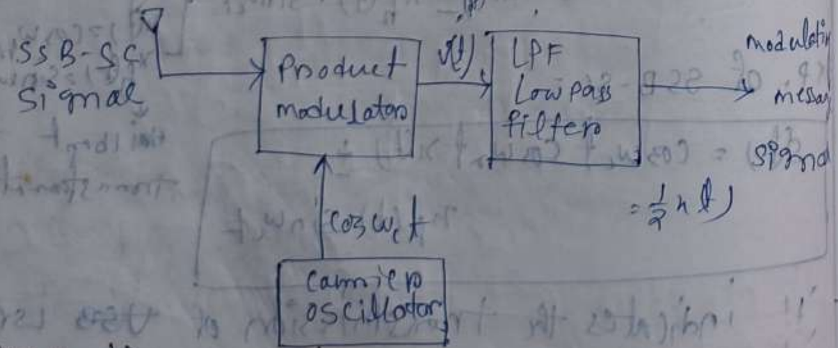
$w_m = 2\pi f$

Demodulation of SSB-SC

Extraction or detection of SSB-SC →

Demodulation. It is synchronous detection process. (Extra) → from pdf

LPF = Allow the message signal to the o/p.



Block diagram of recovering the message signal

Advantage of SSB-SC

i) Less band width, as less side band → less money.

ii) we are saving lots of power as we are saving carrier signal, 1 side band.

iii) Reduce of interference of noise.

Disadvantage of SSB-SC

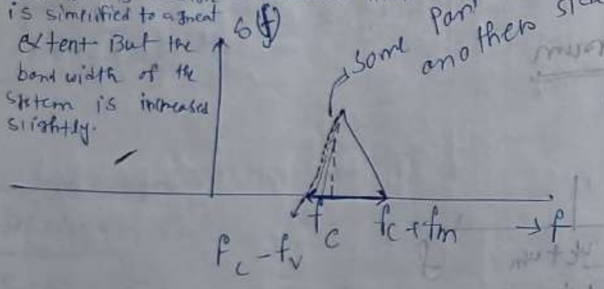
i) we need to design a filter that will pass only one side band, so filter design

is complex
(ii) Here reciever and transmitter is in synchronised form, it don't allow any unwanted signal.

VSB (vestigeal sideband transmission) (•)

[Intro] → The strict frequency response requirements on the sideband filter in SSB-SC system can be relaxed by allowing a part of the unwanted sideband (called vestige) to appear in the output of the modulator.

Due to this design of sideband filter is simplified to a great extent But the band width of the system is increased slightly.

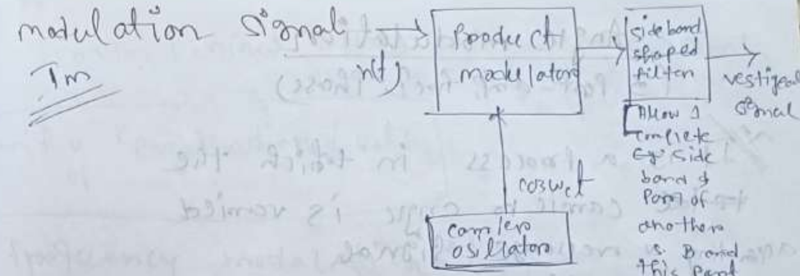


$f_v < f_m$

f_v is the part of another signal side band

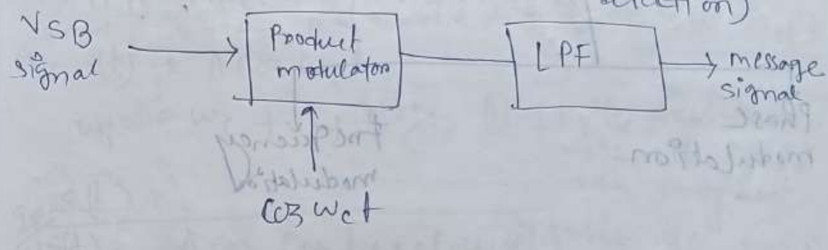
$\boxed{BW = (f_m + f_v)_{Hz}}$ (vestigeal part)

[message signal is combination of 2 or more signals → multitone (multi freq)
message signal is combination of 1 signal → singletone (single freq)]



[Allow a complete of side band & part of another s. band this part is called vestige]

Demodulation (Here also we use synchronous detection)



Application of VSB (•)

1> Transmission of TV signal.

VSB modulation has become standard for the transmission of Television signals. But the video signals need a large transmission bandwidth if transmitted using DSB-FC or DSB-SC technique.

Angle modulation
(3 Part- amp, freq, phase)

It is a process in which the total carrier angle is varied according to the message signal.

Types of Angle modulation

- Phase modulation
- Frequency modulation

Phase modulation (•)

$c(t) = A\cos(w_c t + \theta_0)$ [carrier sig]

If we assume θ_0 is in initial phase, i.e. $\theta_0 = 0$ [phase part]

then, $c(t) = A\cos w_c t$ [w_c = frequency part]

$= A_c \cos\varphi$ [$w_c t = \varphi$ = phase angle]

$\boxed{\varphi_i = \varphi + K\, x(t)}$ [φ_i = Instantinious phase angle]

$= w_c t + K_p x(t)$

$\boxed{s(t) = A\cos(w_c t + K_p x(t))}$ P.M [K_p = phase constant / phase sensitivity]

Phase modulation is a process in which the ins. phase of the carrier signal will vary according to the instantinious values of message signal.

$\boxed{\text{unit of } K_p = \text{radian / volt}}$

Frequency modulation (•)

$c(t) = \boxed{A\cos w_c t}$

After frequency modulation, F.M signal, $s(t) = A\cos w_i t$

$w_i = w_c + K_f\, x(t)$ [w_i is the function of $x(t)$, if $x(t)$ will vary w_i will also vary]

$s(t)$

$\boxed{s(t) = A\cos[w_c t + K_f \int x(t)\,dt]}$

[w_i = instantinious frequency]

derivation

$fm \xrightarrow[s(t)]{} A\cos\varphi_i$

$\varphi = w_c t$

$\Rightarrow \frac{d\varphi}{dt} = w_c$

$\Rightarrow d\varphi = w_c\, dt$

$\Rightarrow \varphi = \int w_c\, dt$

for phase modulation derivation detail of

$\boxed{s(t) = A\cos(w_c t + K_p x(t))}$

So, $w_c \longrightarrow w_i$

$\varphi \longrightarrow \varphi_i$, as φ is dependent on w_c

$\phi_i = \int \omega_i \, dt$

$= \int (\omega_c + k_f x(t)) \, dt$

$= \omega_c t + k_f \int x(t) \, dt$

for the modulation the derivation detail

∴ the exp of FM signal will be

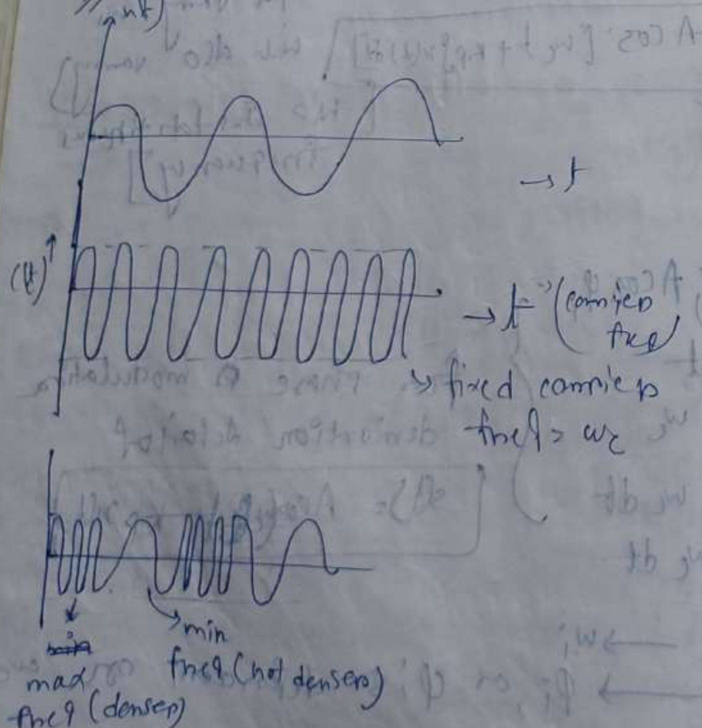
$$\boxed{s(t) = A\cos[\omega_c t + k_f \int x(t)\, dt]}$$

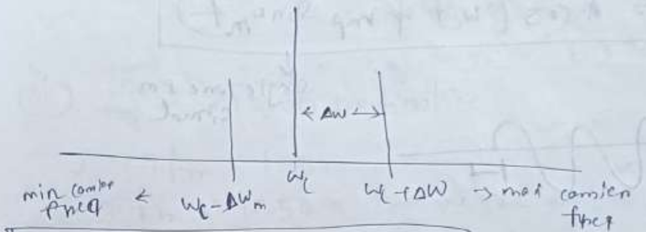
Q. ① using P.M can you make F.M? [From this we have to pass pm into some integrator]

② Diff between exp of Fm & Pm? [For reverse Fm → Pm add a derivator]

Freq deviation (•)

$$\boxed{\omega_i = \omega_c + k_f x(t)}$$





$$\boxed{\Delta\omega = \text{frequency deviation}}$$

$$\boxed{\text{Carrier swing} = 2\Delta\omega}$$

6/2/23

Mathematical expression of a single tone frequency modulated signal (•)

Let, the carrier signal is $c(t) = A\cos\omega_c t$ and modulating signal is, $x(t) = V_m \cos\omega_m t$. The expression of frequency modulated signal will be $s(t) = A\cos\phi_i$

$\omega_i = \omega_c + k_f x(t)$

$= \omega_c + k_f \cos\omega_m t$

$$\boxed{|k_f V_m| = \Delta\omega}$$

$$\boxed{\frac{\Delta\omega}{\omega_m} = m_f}$$

[k_f = frequency sensitivity]

[V_m = maximum amplitude of modulating signal]

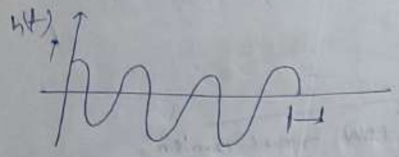
$\phi_i = \int \omega_i \, dt$

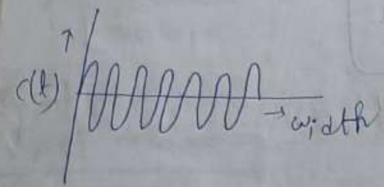
$= \int (\omega_c + k_f V_m \cos\omega_m t) \, dt$

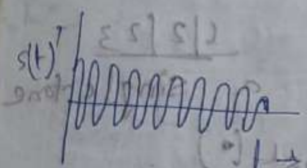
$= (\omega_c t + \frac{k_f V_m}{\omega_m} \sin\omega_m t) = (\omega_c t + m_f \sin\omega_m t)$

$$\boxed{s(t) = A\cos(w_c t + m_p \sin w_m t)}$$

singletone FM signal







Ex-① $v(t) = 12\cos(6\times10^8 t + 5\sin 1250 t)$

Determine i) carrier freq
ii) modulating freq
iii) modulation index
iv) max freq deviation
v) what Power will this FM wave dissipate in 10Ω resistor?

$$\boxed{s(t) = A\cos[w_c t + m_f \sin w_m t]}$$

$v(t) = 12\cos[6\times10^8 t + 5\sin 1250 t]$

(i) Carrier freq $= w_c = 6\times10^8$ radian/sec

$\Rightarrow 2\pi f_c = 6\times10^8$

$\Rightarrow f_c = \frac{6\times10^8}{2\pi}$

$= 95.49 \times 10^6$ Hz

(ii) $w_m = 1250$ radian/sec

$\Rightarrow 2\pi f_m = 1250$

$\Rightarrow f_m = \frac{1250}{2\pi}$

$= 198.94$ Hz

(iv) $m_f = \frac{K_f V_m}{w_m}$

or $m_f = \frac{\Delta f}{f_m}$ $\left[m_f = \frac{\Delta w}{w_m}\right]$

$\Rightarrow 5 = \frac{\Delta f}{198.94}$

$\Rightarrow \Delta f = 994.7$ Hz

(iii) $m_f = \frac{K_f V_m}{w_m} = 5$

(v) $P = \frac{V_{rms}^2}{R} = \frac{\left(\frac{12}{\sqrt{2}}\right)^2}{10}$

$= \cancel{0.648}\ 7.2$ W

Power dissipation

Phasor representation of Angle modulated signal (•)

$x(t) = A\cos\theta$
↓
phasor representation
Phase representation → $Ae^{j\theta}$

$$\boxed{s(t) = A\cos[\omega_c t + k_f \int x(t)\,dt]}$$

↓ fm signal

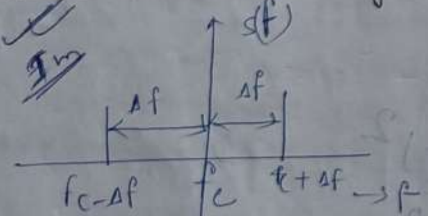
$$\boxed{pm : \rightarrow s(t) = A\cos[\omega_c t + k_p x(t)]}$$

$C_{Fm} = Ae^{j(\omega_c t + k_f \int x(t)\,dt)}$
$= Ae^{j(\omega_c t + k_f y(t))}$ [where $y(t) = \int x(t)\,dt$]

$C_{pm} = Ae^{j(\omega_c t + k_p x(t))}$

Types of frequency modulation

Fm



$B\cdot W = (f_c + \Delta f) - (f_c - \Delta f)$

$\boxed{B.W = 2\Delta f}$ → Angular frequency then $\boxed{BW = 2\Delta\omega}$

$\Delta\omega = |k_f \cdot x(t)|_{max}|$ $\boxed{k_f \text{ unit} : \frac{Hz}{Volt}}$

If $\Delta\omega$ = high then B.W high as BW $\propto \Delta\omega$.

On depending $\Delta\omega$ there is 2 types of Fm –
→ NBFm (Narrow band Fm) [less b.w]
→ WBFm (wide band Fm) [large "]

In case of NBFm, k_f is small, B.W = less
" " WBFm, k_f is large, BW large

For large freq daviation then the k_f should be large [Freq daviation $\propto k_f$]

Im (•)

If k_f, less (NBFm)
then, in $s(t) = A\cos[\omega_c t + k_f \int x(t)\,dt]$

$[k_f y(t) \ll 1]$
$[y(t) = \int x(t)\,dt]$

So, $C_{Fm} = Ae^{j[\omega_c t + k_f y(t)]}$

$= A[\cos(\omega_c t + k_f y(t)) + j\sin(\omega_c t + k_f y(t))]$
[$A[\cos\theta + j\sin\theta]$ → Euler's exp.]

$= Ae^{j\omega_c t} \cdot e^{jk_f y(t)}$

$= A[\cos(\omega_c t) + j\sin\omega_c t] \cdot [\cos k_f y(t) + j\sin k_f y(t)]$
[Euler's expression]

$= A[\cos\omega_c t + j\sin\omega_c t] \cdot [1 + jk_f y(t)]$ → [If θ → very small then, $\cos\theta \simeq 1$, $\sin\theta \simeq \theta$]

$$\therefore C_{NBFm} = A[\cos w_c t + j\sin w_c t][1 + jK_f y(t)]$$

$$= A[\cos w_c t + jK_f y(t)\cos w_c t + j\sin w_c t + j^2 K_f y(t)\sin w_c t]$$

$$= A[\cos w_c t + jK_f y(t)\cos w_c t + j\sin w_c t - K_f y(t)\sin w_c t]$$

$$= [A\cos w_c t - AK_f y(t)\sin w_c t] + j[AK_f y(t)\cos w_c t + A\sin w_c t]$$

Real part will be represented as FM signal.

(Phasor representation) $[Ae^{j\theta} \;|\; x(t) \to A\cos\theta$ real part; $A\cos\theta + jA\sin\theta]$

$$\therefore \boxed{s(t)_{NBFm} = A\cos w_c t - AK_f y(t)\sin w_c t} \quad \text{(Generalized Eqn)}$$

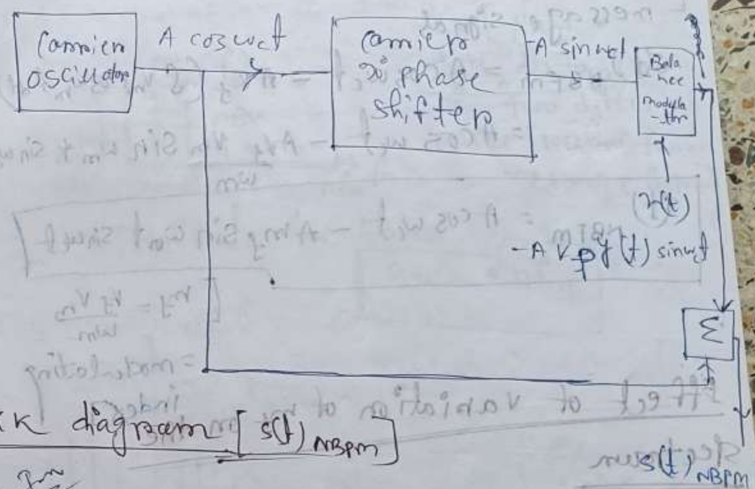
$$s(t)_{NBFm} = A\cos w_c t - AK_f \int x(t)\,dt \,\sin w_c t$$

$$= A\cos w_c t - \left(\frac{AK_f}{w_m}\right)\sin w_m t \cdot \sin w_c t$$

$$\boxed{s(t)_{NBFm} = A\cos w_c t - m_f \sin w_m t \,\sin w_c t}$$

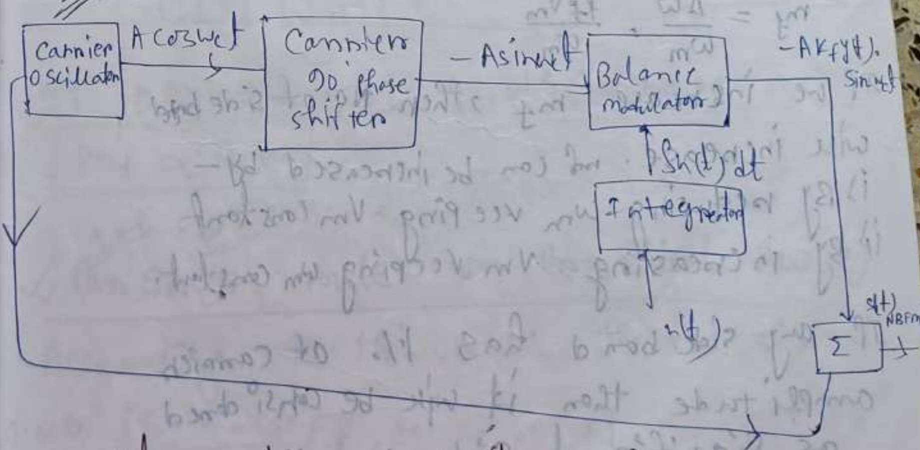
For PM,

$$\boxed{s(t)_{NBPM} = A\cos w_c t - AK_p x(t)\sin w_c t}$$



Block diagram [s(t) NBPM]

Imp



Block diagram [s(t), NBFm]

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Expression for singletone narrowband of fm signal

$$s(t)_{NBFm} = A\cos w_c t - AK_f \int x(t)\,dt \cdot \sin w_c t$$

$x(t) = V_m \cos w_m t$
[message signal

$$s(t)_{NBFm} = A\cos w_c t - AK_f\left(\int V_m \cos w_m t\,dt\right)\sin w_c t$$
$$= A\cos w_c t - AK_f \frac{V_m}{w_m}\sin w_m * \sin w_c t$$

$$\boxed{s(t)_{NBFm} = A\cos w_c t - Am_f \sin w_m t \sin w_c t}$$

[$m_f = \frac{K_f V_m}{w_m}$ = modulating index]

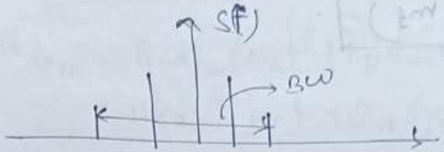
Effect of variation of m_f on the spectrum

$$m_f = \frac{\Delta w}{w_m} = \frac{K_f V_m}{w_m}$$

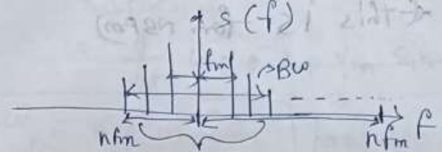
If we increase m_f, then No of side band will increase. m_f can be increased by—
i) By reducing w_m keeping V_m constant
ii) By increasing V_m keeping w_m constant.

If any side band has 1% of carrier amplitude then it will be considered as significant side band.

$m_f = 0.5$



$m_f = 1$

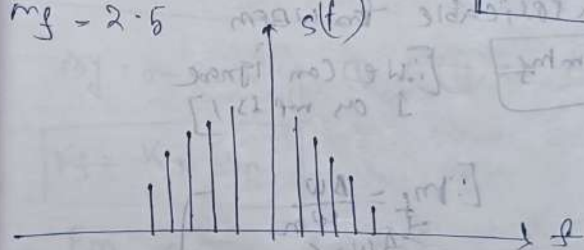


[fixed this distance of two different spectrum: fm = message frequency]

significant side band

$\boxed{B\cdot w = 2nfm}$

$m_f = 2.5$



m_f increases, B·w also increases

$\boxed{m_f \propto B\cdot w}$

Carson's rule or Practical BW

According to this rule the fm signal B·w is =

$$\boxed{B\cdot w = 2(fm + \Delta f)}$$

$$B\cdot w = 2(wm + \Delta w)$$
(Angular frequency)

$= 2\omega_m \left(1 + \frac{\Delta\omega}{\omega_m}\right)$

$$\boxed{B\omega = 2\omega_m(1 + m_f)}$$

Case - I

$\Delta\omega << \omega_m$

$\therefore m_f << 1$ [from, $B\cdot\omega = 2\omega_m(1+m_f)$]

$\therefore \boxed{B\omega \simeq 2\omega_m}$ & this is for NBFM)

Case - II

$\Delta\omega >> \omega_m$

$\therefore m_f >> 1$

this is applicable for WBFM

$\therefore \boxed{B\cdot\omega = 2\omega_m m_f}$ [i.e we can ignore 1 as $m_f >> 1$.]

$\boxed{B\cdot\omega = 2\Delta\omega}$ [$\because m_f = \frac{\Delta\omega}{\omega_m}$

$\Rightarrow \Delta\omega = (m_f \cdot \omega_m)$]

Phase modulation: An analytical view (•)

$\varphi_i = \omega_c t + k_p x(t)$ [φ_i → instantaneous phase]

$x(t) = V_m \cos\omega_m t$

$\varphi_i = \omega_c t + k_p V_m \cos\omega_m t$

$\Rightarrow |\varphi_i - \omega_c t| = k_p V_m \cos\omega_m t = k_p V_m$ [$\because (\cos\omega_m t)_{max} = 1$]

max phase change / phase variation / phase deviation

as $\omega_c t = \varphi$ (phase angle)

$\Rightarrow \boxed{\theta_d = k_p V_m}$

$s(t)_{pm} = A\cos(\omega_c t + k_p V_m \cos\omega_m t)$

$= A\cos(\omega_c t + \theta_d \cos\omega_m t)$

$\omega_i = \left(\frac{d\varphi_i}{dt}\right)$ [Rate of change of instantaneous phase is equal to angular freq.]

$= \frac{d}{dt}\ \omega_c - k_p V_m \sin\omega_m t \cdot \omega_m$

$\boxed{\Delta\omega_{pm} = k_p V_m \omega_m}$ [$\Delta\omega = \omega_i - \omega_c$] [$(\sin\omega_m t)_{max} = 1$] — (I)

$\boxed{\Delta\omega_{Fm} = k_f V_m}$ — (II)

By compearing (I) & (II) we can write

$\boxed{k_f = k_p \omega_m}$

Fm Generator(•) (2 ways to generate a Fm signal)

i) The direct method of Fm signal generation.

ii) Indirect

(Also known as Arom strong method)

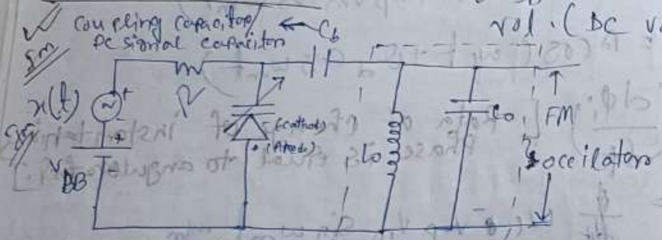
→ (I) Reactance modulator

→ (II) Varactor diode modulator

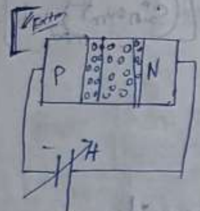
Varactor diode method of Fm signal Generator (•)

Circuit diagram (tank ckt) [V_{BB} - fixed biasing vol. (DC val)]

coupling capacitor / dc signal capacitor ← C_b



varactor diode



[oscillating freq of this oscillator]

oscillating freq is denoted as carrier freq ← $f_c = \frac{1}{2\pi\sqrt{L_0 C_0}}$

Reverse bias condition

Diplition layer is increased

if we increased the reverse bias, the capacitance will also increased.

Eqn of varactor diode capacitance (•)

$$\boxed{C_d \propto V_d^{-1/2}}$$

[V_d = val. accross the varactor diode
K = some constant]

$$\boxed{C_d = K(V_{BB} + x(t))^{-\frac{1}{2}}}$$

$$\left[V_d \approx V_{BB} + x(t)\right]$$

[As if R is negligible then all V_d will be occure accross diode]

Equivalent Capacitance $= \boxed{C_0 + C_d}$

↑ This is variable

after applying we can also write, $\boxed{f_c = \frac{1}{2\pi\sqrt{L_0 C_0}}}$ [L_0 is constant and C_0 is variable]

After applying eqv capacitance, Equivalent frequency,

$$f_i = \frac{1}{2\pi\sqrt{L_0(C_0 + C_d)}}$$

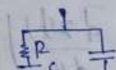
$$= \frac{1}{2\pi\sqrt{L_0\left[C_0 + K\{V_{BB} + x(t)\}^{-\frac{1}{2}}\right]}}$$

Instantinious freq is the function of modulating frequency.

Before we are giving $x(t)$ & C_d in this ckt C_0 is constant, after giving it the C_0 is not constant, but L_0 is constant.

Christal oscillator - A crystal oscillator is an electric oscillator type circuit that uses a piezoelectric resonator, a crystal, as its frequency-determining element. It is a wafer of quartz crystal or ceramic with electrodes connected to it.

 → symbol of crystal



C_m ← Eqv. ckt of a crystal

Drawbacks of this varactor diode - (•)

i) Due to the use of tank ckt modulating signal is directly controlling tank ckt so this ckt isn't stable.

ii) Distortion in Fm signal.

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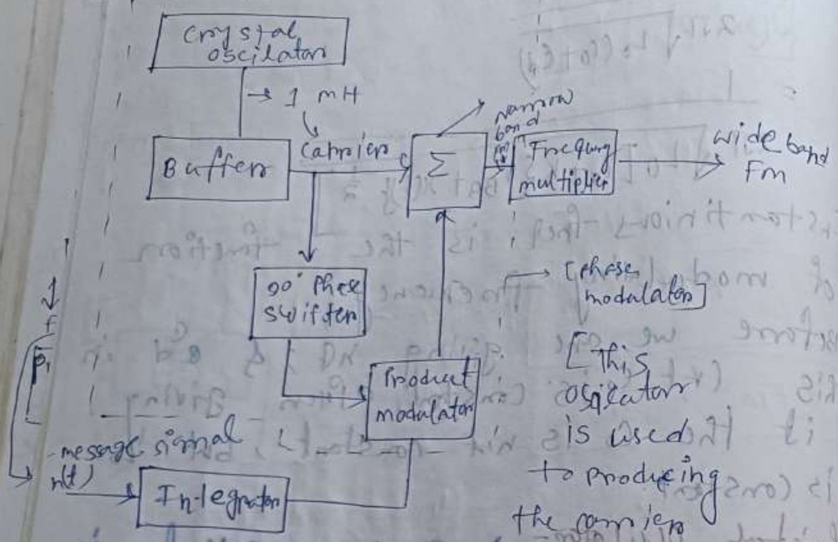
Indirect method (Armstrong method)



[Phase modulator]

[This oscilator is used to producing the carrier signal]

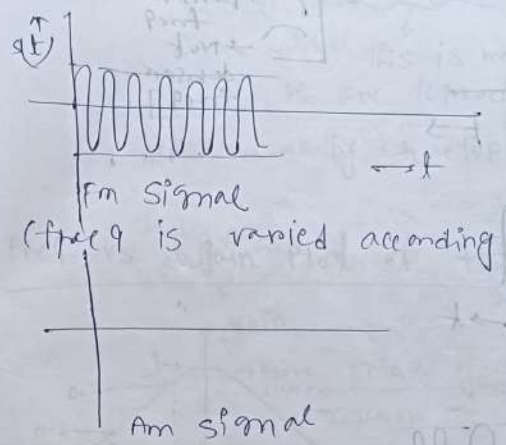
[This is indirect method we are using phase modulation]

Block diagram

Freq mult. $\int m(t) \left(\frac{dx(t)}{dt} \right)$ $x(t)$

Demodulation of FM waves

FM → AM → Demodulation

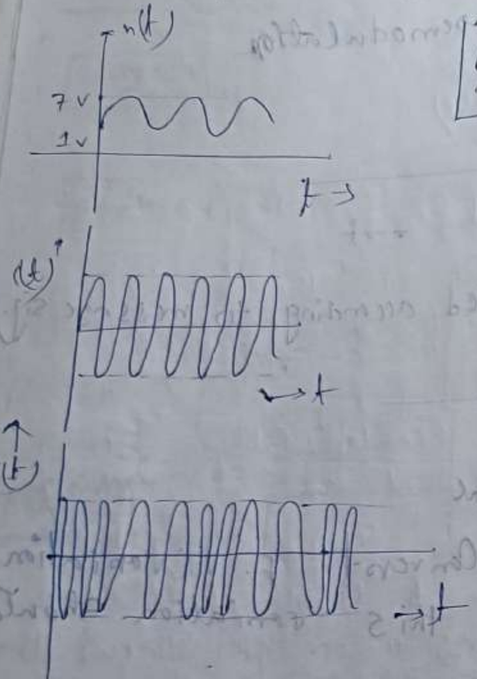


Demodulators convert F.V (variation) to A.V and this variation should be linear.

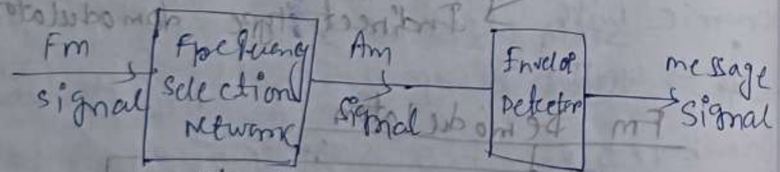
Demodulator → Direct type demodulator
→ Indirect type demodulator

FM Demodulator
- Direct
  - Freq Discrimination
  - Zero crossing Detector
- Indirect
  - Phase locked loop (PLL)

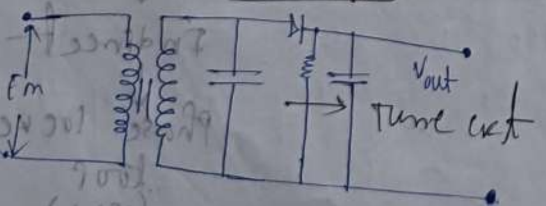
slope Detector (o) Imp



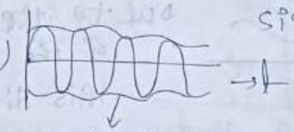
[→ Denser freq
→ Not denser freq]



Block diagram

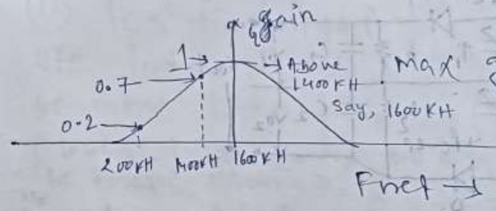


[Envelop Detector → to detect the message signal



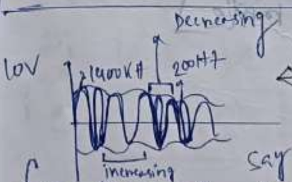
this is message signal we are demodulating it by envelop detector]

Freq vs gain plot of tune ckt



Max gain = 1

max freq of Fm signal	= 1400 KH
min freq " " "	= 200 KH

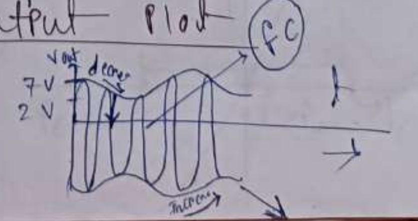


← Input Plot

say
for 1400 KH, gain = 0.7, o/p = 0.7 × 10 = 7 V

for 200 KH, say gain = 0.2, o/p = 0.2 × 10 = 2 V

Output Plot



as, freq ∝ gain

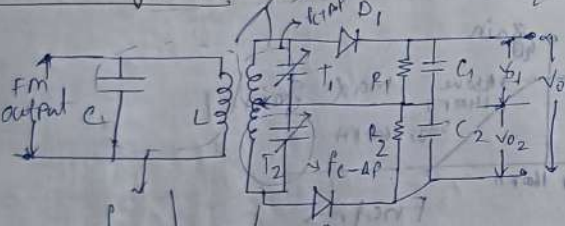
$$\text{Gain} = \frac{o/p}{i/p} = \frac{\text{gain} \times V}{i/p}$$

Distortion as there is a variance in amplitude due to the non-linearity of the f vs 'J' graph this distortion is occurring.

cw 15/2/23

Balanced slope detector (Balanced Frequency detector)

ckt diagram → tune ckt (the parallel combination of L and C)



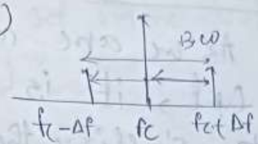
(This transformer is tune to carrier sig.) (center tap transformer)

[R1 & C1 are use to ripple]

Here,

$\boxed{(o/p)\ V_0 = V_{01} - V_{02}}$

Frequency region (3 types)



(i) $f_{in} = f_c$ [f_{in} = Input freq]

It will couple the fm sig. from primary side to secondary side. But the sig. which will couple the tune ckt 1 & ckt 2 will be equal but their phase will different. ∴ the input in D_1 & D_2 is same, i.e the o/p of D_1 & D_2 is also same, ∴ $\boxed{V_0 = V_1 - V_2 = 0}$

(ii) $f_c < f_{in} < f_c + \Delta f$

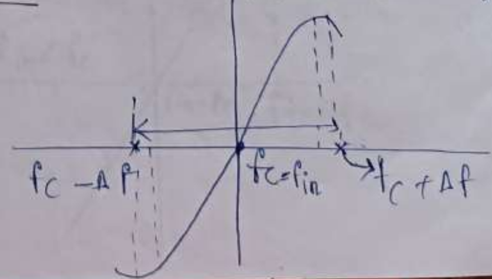
The induced val at tune ckt 1 T_1 will be more than induced val at tune ckt 2 (lower tune ckt). So the i/p val. to D_1 will be more than D_2, ∴ $V_{01} > V_{02}$ $\boxed{\text{i.e } V_0 > 0}$

(iii) $f_c - \Delta f < f_{in} < f_c$

The induced val. at tune ckt 1 (upper tune ckt) T_1 will be less than induced val at tune ckt 2. So, the i/p val. to D_1 will be less than D_2, ∴ $V_{01} < V_{02}$,

i.e $\boxed{V_0 < 0}$

Plot



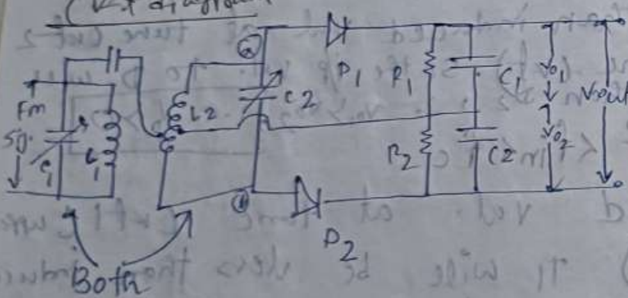
(In comparison with slop detector this graph is more linear)

Drawbacks

1) As we are controlling 3 type ckt, it is complex. In order to simplify the tuning process we can use phase discriminator or Foster seely Discriminator.

Phase Discriminator or Forster seeley discriminator

Ckt diagram



(2 tune ckt)

$$\boxed{O/P\ val = V_0 = V_{01} - V_{02}}$$

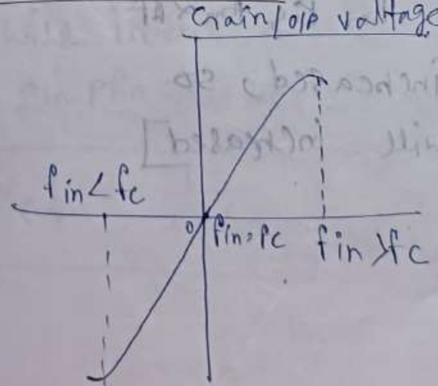
3-fel region(s)

(i) $f_{in} < f_c$ - D_2 will get more vol. than V_1, so, $\boxed{V_0 < 0}$

(ii) $f_{in} = f_c$ - This both diode get equal val. i/p and it will produce the o/p same. So the effective vol (output voltage) will be, $\boxed{V_0 = V_{01} - V_{02} = 0}$

(iii) $f_{in} > f_c$ - Diode D_1 will get more val than diode D_2, so, $\boxed{V_0 > 0}$

Plot (Graph)



Advantages

(i) The tuning process becomes simplified compare to the earlier ckt and linearity is get greater than earlier graph.

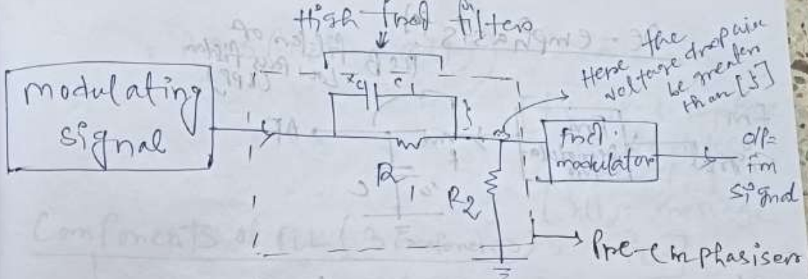
Pre-emphasis (·)

[In order to remove the noise we can increase the mf (modulating index) for high modulating signal freq. mf can be increased by increasing the amplitude of A.m signal. This artificially boosting process is known as pre-emphasis.]

[$m_f \propto$ Amp of f_m, as

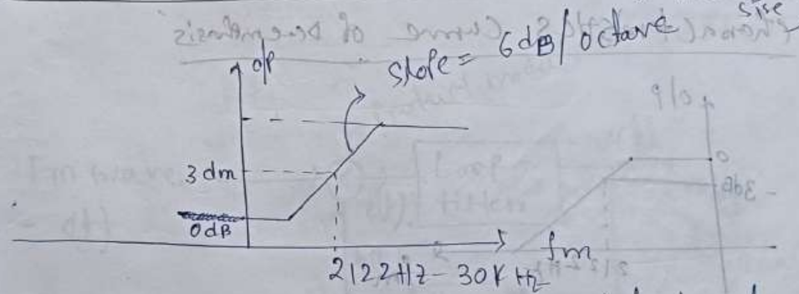
$$m_f \uparrow = \frac{\Delta f}{f_m}$$

if we be increasing ~~Am~~ ~~bs~~ $A_m \propto \Delta f$ then Δf will be increased, so finaly the mf will increased]



Implimentation diagram

~~Diagram~~ of characteristics curve (Application of Pre-emphasise)

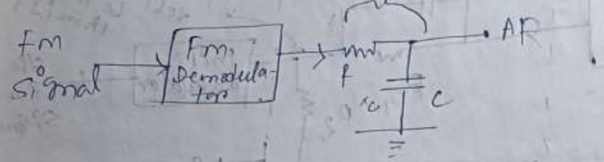


If it is in between 2122Hz - 30KHz, it isn't high freq, so noise will not induced. If it is ~~more~~ than this the noise will induced then we need to pre-emphasis it.

$$\boxed{X_C \downarrow = \frac{1}{2\pi f_m \uparrow}}$$ $\Rightarrow$ so more high fm signal

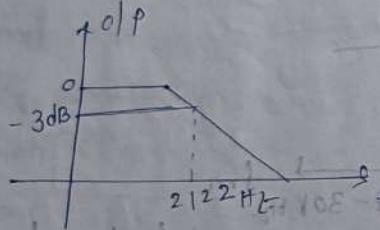
De-emphasis (•)

RCB filter or Low pass filter (LPF)



$X_C \downarrow = \frac{1}{2\pi f_m T}$ so less amount of voltage will appear as o/p. So fm signal will decompose.

characteristics curve of De-emphasis



PLL (Phase locked loop) (•)

negative feedback system, it is primarily used to track the freq of the incomming signal.

$s(t) = A\cos(w_c t + k_f \int x(t)\, dt)$

↓

our purpose is to detect this $x(t)$

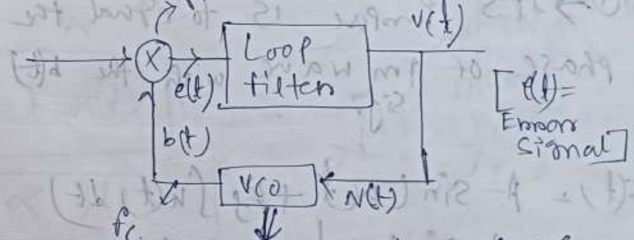
$= A\cos(w_c t + \phi_1(t))$

[i.e, it's purpose is to detect that $\phi_1(t)$]
[$x(t)$ = message info.]

Components of PLL (3 components)

- VCO (kind of freq. modulator) (Voltage control oscillator)
- multiplier
- loop filter

product modulator / multiplier



[fc will change according to the vol. signal (v(t))] message. It will recieve some signal and produce some fm signal.

Block diagram

If phase of fm wave isn't equal to phase of VCO than the operation will be continued. when phase of s(t) and b(t) are equal than phase locked condition will be achieved.

Initially vco is set to carrier freq f_c, and the o/p of this signal $b(t)$ and $s(t)$ will be a 90° phase shift [∵ 90° phase difference]

$$\boxed{\begin{aligned} s(t) &= A \sin(w_c t + K_f \int x(t)\, dt) \\ b(t) &= A_v \sin(w_c t + K_v \int v(t)\, dt) \end{aligned}}$$

[K_v = freq sensitivity of vco]

(x) → Its purpose is to equal the phase of Fm wave with the $b(t)$ sig.

$s(t) = A \sin(w_c t + K_f \int x(t)\, dt)$

$\Rightarrow A \sin(w_c t + \phi_1(t))$

$b(t) = A_v \sin(w_c t + K_v \int v(t)\, dt)$

$= A_v \sin(w_c t + \phi_2(t))$

$e(t) = s(t) \cdot b(t)$

$= A_c A_v \sin(w_c t + K_f \int x(t)\, dt) \sin(w_c t + K_v \int v(t)\, dt)$

$$= \frac{A_c A_v}{2} \Big[\underbrace{\sin[2 w_c t + K_f \int x(t)\, dt + K_v \int v(t)\, dt]}_{\text{blocked}} + \sin[\underbrace{K_f \int x(t)\, dt}_{\phi_1(t)} - \underbrace{K_v \int v(t)\, dt}_{\phi_2(t)}] \Big]$$

when phase locked is achieved or near to achieve, then we can consider $\sin(\phi_1(t) - \phi_2(t)) = \phi_1(t) - \phi_2(t)$ it is a very less value, so, we can say $e(t) \simeq 0$ and this is our purpose.

$\therefore e(t) \simeq 0$

$\Rightarrow \frac{A_c A_v}{2} (\phi_1(t) - \phi_2(t)) \simeq 0$ [Another part will be blocked by the loop filter & this term will be passed through loop filter]

$\Rightarrow \phi_1(t) = \phi_2(t)$

$\Rightarrow K_f \int x(t)\, dt = K_v \int v(t)\, dt$

$\Rightarrow v(t) = \frac{K_f}{K_v} \cdot x(t)$

$\boxed{v(t) = K\, x(t)}$ [$K = \frac{K_f}{K_v}$ constant]

i.e. $\boxed{v(t) \propto x(t)}$

i.e, using phase lock we can demodulate the message signal.

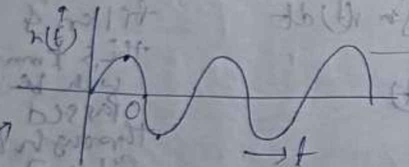
Fm signal is more efficient than Am signal & how it is in the notes.

Comparison between Fm & Am signal is also in notes.

Sampling Theory(•)

✓✓Imp
✓Imp Imp

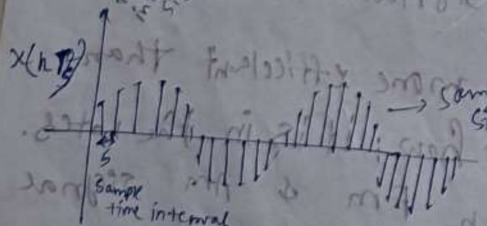
A band ~~widthed~~ limited continuous time signal of finite energy may be completely represented in its sample and recovers back, If sampling frequency $f_S \geq 2f_m$ where f_m is the max. signal frequency.



for every time instance we will get some value, so it's a continuous time signal.

↓ after sampling

we can also recover this sig. into that signal



$f_S \geq 2f_m$

Ex - 20 K Hz > 10 K Hz

20 × 1000 = 20000 Samples/sec(min)

Sample time interval - [It is the frequency of data collection. For Event-based Sampling (EBS), the sampling interval is used to calculate the target number of samples and the sample after the value]//

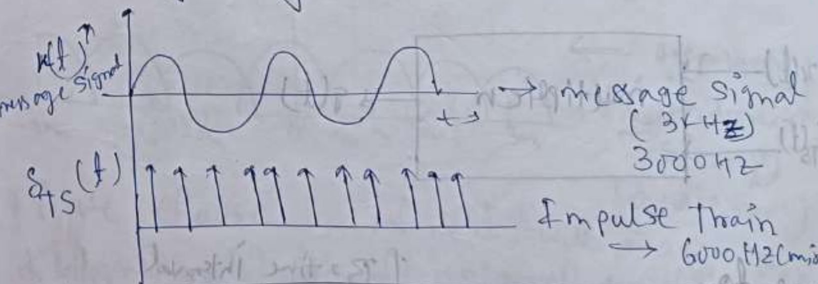
$\boxed{f_S = 2f_m}$ → Neiquist rate (Hz)

$\boxed{T_S = \frac{1}{2f_m}}$ → neiquise interval (Tm)

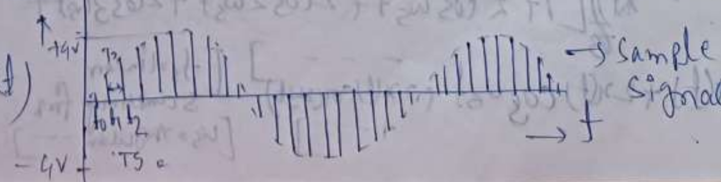
Sample time interval - It is the distance or time between which measurements are taken, or data is recorded.

CW
Imp **Proof of Sampling Theorem(•)** 22/2/23

Sampling theory should be $f_S \geq 2f_m$



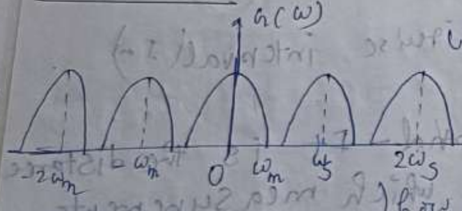
$\boxed{\text{frq of Imple Train} = 2 \times \text{message freq}}$



$x(t)$ = This signal is continuous with time.

$g(t)$ = This signal (sample signal) is discrete time continuous amplitude signal.

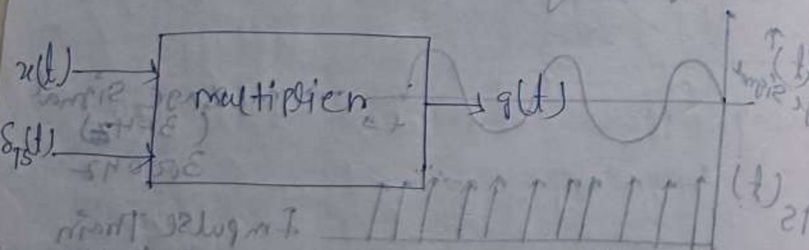
Spectrum:



how we are getting that spectrum

$$\boxed{g(t) = x(t) \times \delta_{Ts}(t)}$$

Block diagram



Derivation

$g(t) = x(t) \cdot \delta_{Ts}(t)$

$= x(t) \frac{1}{Ts}[1 + 2\cos w_s t + 2\cos 2w_s t + 2\cos 3w_s t + \ldots]$

$= \frac{1}{Ts}[x(t) + 2x(t)\cos w_s t + 2x(t)\cos(2w_s t) \ldots]$

[Ts = time interval between 2 instant]

[f_s = linear sampling frn]

[w_s = Angular ---]

$G(w) = \frac{1}{Ts}[\underbrace{X(w)}_{\text{Base to Pass}} + X(w - w_s) + X(w + w_s) + X(w - 2w_s) + X(w + 2w_s)]$

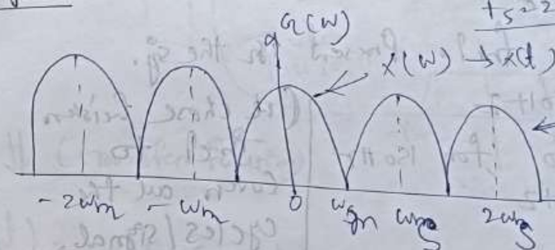
$\left[w_s = \frac{2\pi}{Ts} = 2\pi f_s\right]$

$[\because x(t) \longleftrightarrow X(w)$

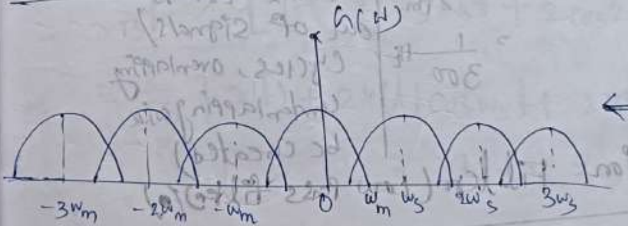
$2x(t)\cos w_s t \longleftrightarrow X(w - w_s) + X(w + w_s)]$

[when we chose $f_s = 2f_m$, then the cycles(←) will touch each others]

Diagram



While, $f_s < 2f_m$



If we choose $f_s < 2f_m$ it will make a distortion at out signal. So we must have to choose $f_s \geq 2f_m$. So from this we can say $f_s \geq 2f_m$ or $f_s = 2f_m$

Ex-① Analogue ~~Another~~ signal is expressed by this eqn

$x(t) = 3\cos(50\pi t) + 10\sin(300\pi t) - \cos(100\pi t)$

find the Nyquist rate.

$f_1 = \frac{50\pi}{2\pi} = 25Hz$ $\quad 2\pi f_2 = 300\pi \Rightarrow f_2 = \frac{300}{2} = 150Hz$ $\quad 2\pi f_3 = 100\pi \Rightarrow f_3 = 50Hz$

f_m = max freq present in the sig.

$\therefore f_m = 150Hz$

$\therefore f_s = 2\times150$, for 150 Hz

Nyquist rate $= 300Hz$

Nyquist interval $= T_s = \frac{1}{2f_m} = \frac{1}{300}$

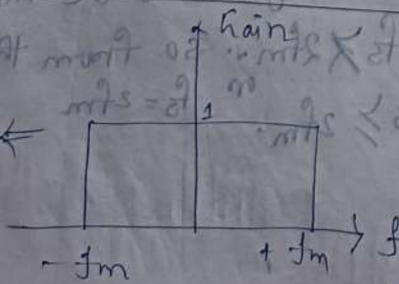
(We chose the highest freq to cover all the cycles/signal, otherwise we can't cover all of signals/cycles, overlapping, underlapping will be created)

Reconstruction Filter (Low Pass filter) (or)

⇓

Interpolation filter

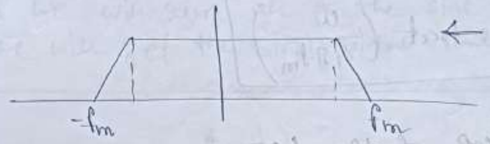
Freq response curve of Low Pass filter / freq characteristics



Ideal LPF ←

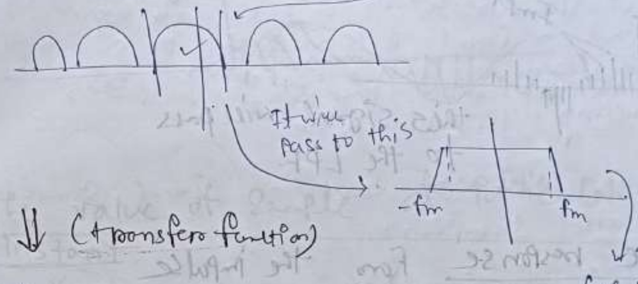
[It's cutoff freq is f_m (upto f_m it is allowed). The freq above f_m will be blocked by this filter]

[the gain will be multiplied at o/p]



← Practical LPF (It will never gain '0')

Our purpose is to pass this



It will pass to this

We will get our o/p = i/p × gain of

⇓ (transfer function)

$g(t) = x(t) \cdot \delta_{Ts}(t)$

$= \frac{1}{T_s} x(t) \left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t \ldots\right]$

$= \frac{1}{T_s}\left[x(t) + 2x(t)\cos\omega_s t + 2x(t)\cos 2\omega_s t + 2x(t)\cos 3\omega_s t + \ldots\right]$

$\frac{1}{T_s} x(t) \times T_s \rightarrow x(t)$ [The gain of the LPF $= T_s$]

Transfer function: The ratio of fourier transformers of output and input. (It is a mathematical function that theoretically models the system's output for each possible input).

$$\boxed{H(\omega) = T_s \times rect\left(\frac{\omega}{4\pi f_m}\right)}$$

$\Downarrow$

freq domain rep of the LPF

this signal will pass to the LPF.

(Fourier transform) $\Downarrow$

Impulse response for the impulse

what should be the response is called impulse response.

$$\boxed{h(t) = F^{-1}[H(\omega)]}$$

[F^{-1} = inverse Fourier]

$$= F^{-1}\left[T_s \, rect\left(\frac{\omega}{4\pi f_m}\right)\right]$$

$$\boxed{h(t) = 2 f_m T_s \, sinc(2\pi f_m t)}$$

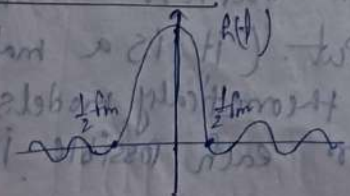
(sinc function)

$= 1 \cdot sinc(2\pi f_m t)$

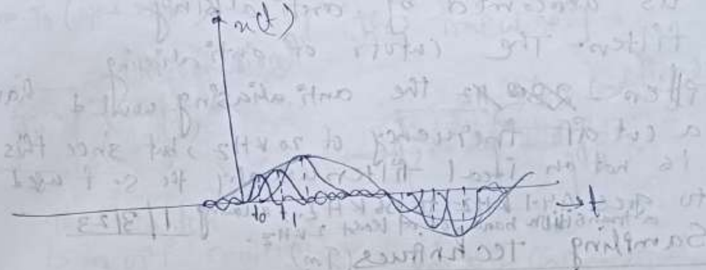
[If we pass impulse func through LPF we will get sinc func as an O/P]

[$\because 2 f_m T_s = 1$]

Sinc function



If we will sum all of the sinc function we will get the sinusoidal function (curve)



The value of sample at particular time instant

$k^{th} \Rightarrow x(kT_s) \cdot \delta(t - kT_s)$

for 1, $x(T_s) \cdot \delta(t - T_s)$, for k^{th} s. If we want whole sig then we have to sum it/integrate it

$$x(t) = \sum_k x(kT_s) \cdot h(t - kT_s)$$

$$= \sum x(kT_s) \cdot sinc[2\pi f_m (t - kT_s)]$$

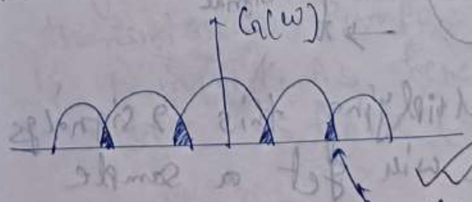
[for k^{th} sample it will shifted (phase shift)]

$$\boxed{x(t) = \sum_k x(kT_s) \, sinc(2\pi f_m - k\pi)}$$

$\hookrightarrow$ Interpolation formula

$$\left[\begin{array}{l} f_s = 2 f_m \\ \Rightarrow T_s = \frac{1}{f_s} \\ = \frac{1}{2 f_m} \\ T_s f_m = \frac{1}{2} \end{array}\right]$$

Effect of under sampling



$f < 2 f_m$

Imp << Imp

Aliasing effect $\}$ → Imp

(∴ it's geometry)

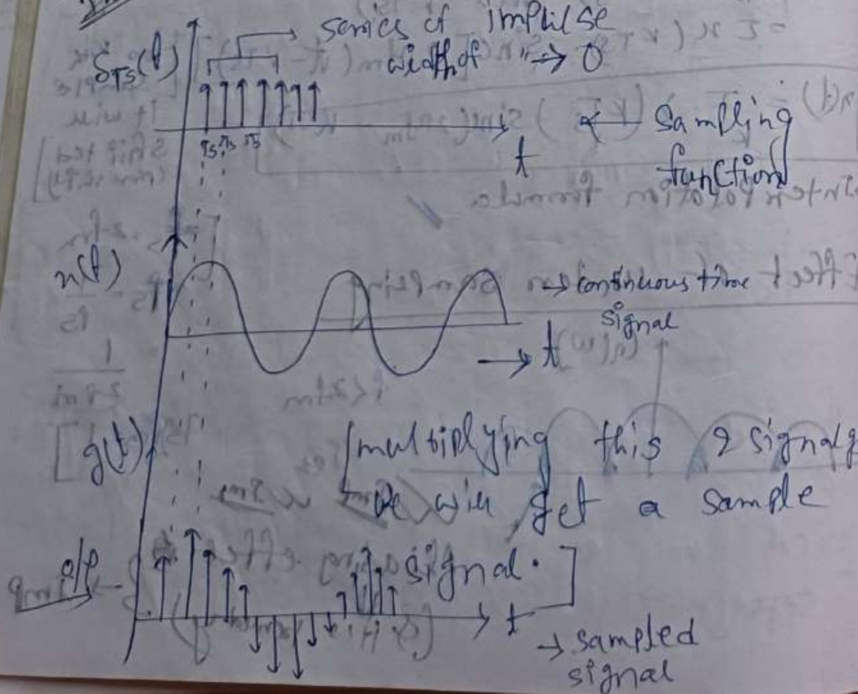
To overcome aliasing effect we have to ensure $f_s \geq 2f_m$. It is overcomed by anti aliasing filter. The cutoff of anti aliasing filter - 22 KHz the anti aliasing would have a cut-off frequency of 20 KHz, but since this is not an ideal filter usually the s.f. used to goes 44.1 KHz to 96 KHz, allowing a transition band of at least 2 KHz.

1/3/23

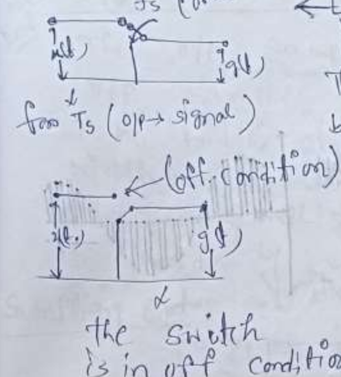
Sampling techniques (Im)

1) Instantaneous Sampling / Ideal Sampling
2) Natural Sampling
3) Flat top Sampling

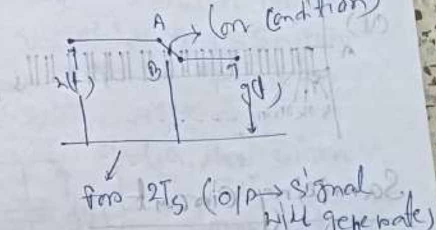
1) Instantaneous Sampling / Ideal Sampling -



Sampling ckt



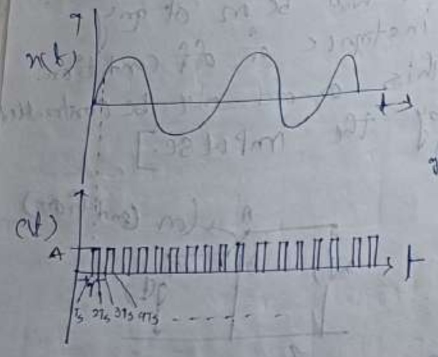
[It will be on at any instance of off condition. This on-off will be controlled by the impulse.]

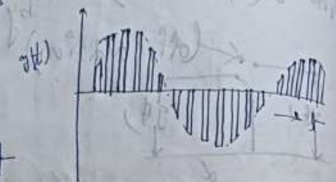


While impulse will be present it will pass input to the output, when it is not present then no signal generation. (Dependence on impulse). The amp of $g(t)$ will be same as impulse signal.

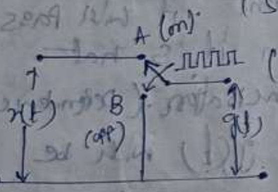
2) Natural Sampling

For Ideal Sampling, ~~Here,~~ the width of impulse (τ) tensed to zero ($\tau \to 0$) [it is negligible] {For ideal width → 0} so the power contained is also very negligible, so it is not very suitable for transmission purpose. Here this problem is removed. As here, the width of impulse is finite (finite width).





Sampling Circuit



(switch will be operated based on the applied impulse) (this switch can also be made by transistor)

[switch will be at pos A at high i/p and remain there as long as the i/p is high, at the end of the pulse the switch will be pass to B]. (at $\tau \to 0$, zero width, ie no impulse)

[the o/p is following the input for τ time duration]

$g(t) = x(t)$, when $c(t) = A$ (c(t) will be A for τ time duration)

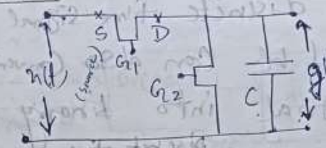
$g(t) = 0$, when $c(t) = 0$ (switch is in off condition)

3) Flat-top sampling Technique

Imp

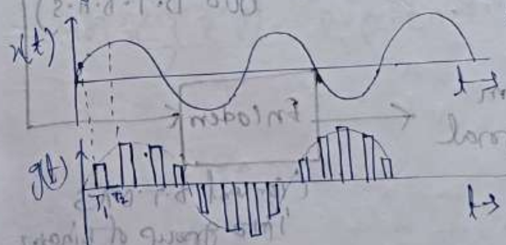
The o/p sample pulse, the top of o/p sample pulse will be flat. Here width of pulse = finite (finite width) we will use 2 switch (discharge switch, sampling switch). This switch is made by fide effect Transistor.

Sampling Ckt



(G_1, G_2 switch → flide effect transistor) (they will be on & off at different time instant)

G_1 on, G_2 off → high → capacitor will be full with charge & o/p will be follow the pattern so like T_1. At reverse the reverse thing will happen of o/p -- like T_2.



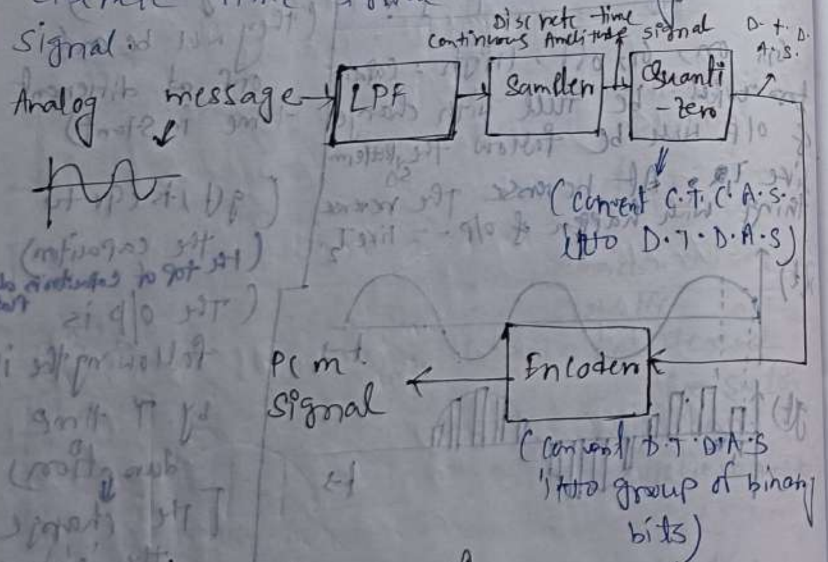
(g(t) is equal to the capacitor) (the top of capacitor is also flat) (The o/p is following the i/p by τ time duration) [The charge of the C will be equal to the value of the signal for τ time duration]

Wave form coding techniques

(Advantages of digital transmission mode)
(Analogue → Am, Fm, sampling etc all)

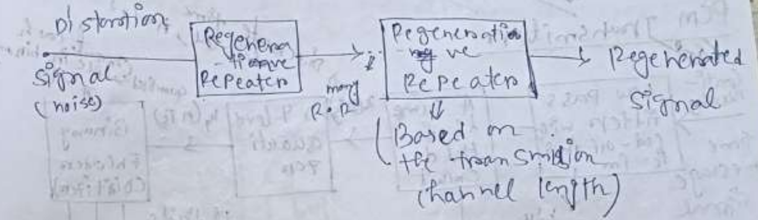
PCm = Pulse code modulation (digital transmitters)
(uses →)

PCm → we can convert a continuous time signal into discrete time signal by using PCm. If we can also convert discrete time signal into binary signal

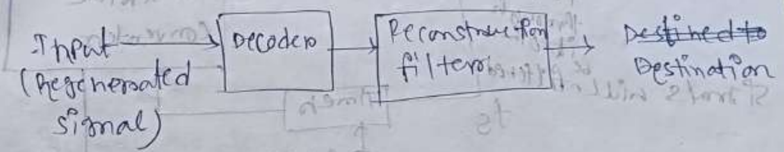


Block diagram of PCm transmitter

Components in the transmission path



Block diagram of transmission path

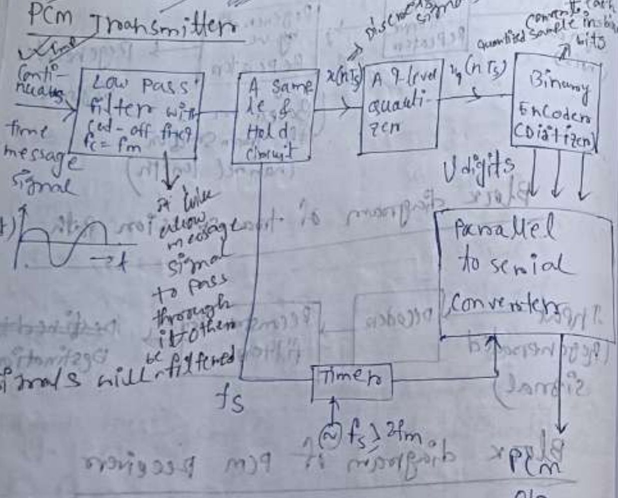


Block diagram of PCm Receiver

We use R.P to reduce the effect of the noise (R.P) = Regenerative Repeaters)

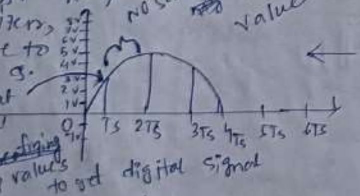
20/3/23

PCM Transmitter



Block diagram of PCM Transmitter

If 3 bits, 2^3, 8V quantizers, we have to fix 8V in that

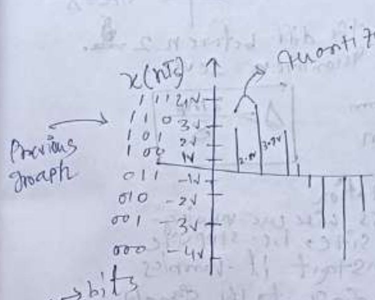


← Discrete time signal

digitizing amp values to get digital signal

[To get digital Signal we have to make amplitude axis digitize]

- Peak to peak amplitude (abbreviated P-P) is the change between peak (highest amp value) and trough (lowest amp value), which can be negative)



We are fixing discrete signal

$2^3 = 8$ → binary bits

8 quantizer level

000
001
010
011
100
101
110
111

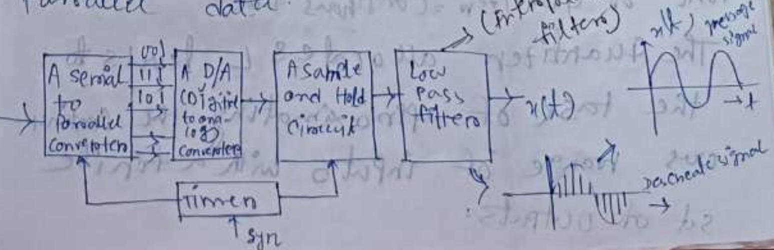
1st o/p = 110 (3V)
2nd o/p = 111 (4V)
3rd o/p = 110 (3V)

By this way we will get all the binary bits

If the bits are parallel then we have to convert it in series as the channel can be narrow.

PCM Receiver

Here series data should be converted into parallel data.



Block diagram of PCM Receiver

(m)

Quantization

Uniform Quantization (Quantization process will use the same step size (constant))

Non-uniform Quantization (Quantization process will use varying step sizes, i.e. stepsize not constant it varries according to the signal nature.)

Uniform Quantization → mid-tread quantization process; mid-rise quantization process

Step size 

1v = step size = Δ

the diff between 2 quantizer levels.

$$\boxed{\Delta = \frac{2m_p}{8}}$$

Uniform Quantizator (Im) 22/3/23

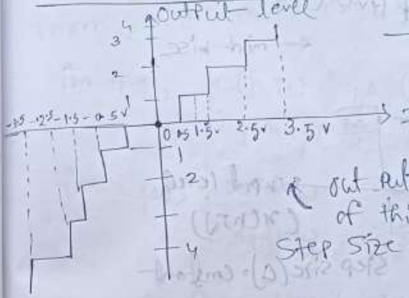
→ mid tread type ~~quantiz~~ Symmetric quantization of

→ Symmetric " " mid-rise type

func of quantizer = continuous time

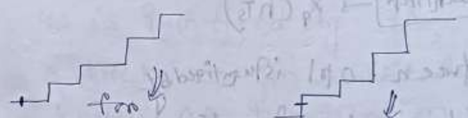
The quantizer allocates 4 levels to the task of approximating the continuous range of inputs with a finite set of outputs.

Transfer char. of this 2 type of quantizer



→ mid-tread

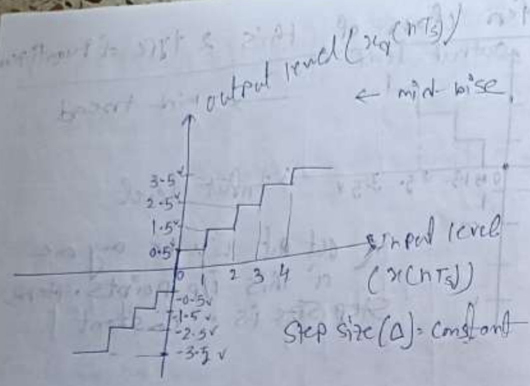
output will be any one of this i/p points. Here, step size is = constant



for mid-tread type

for mid-rise type

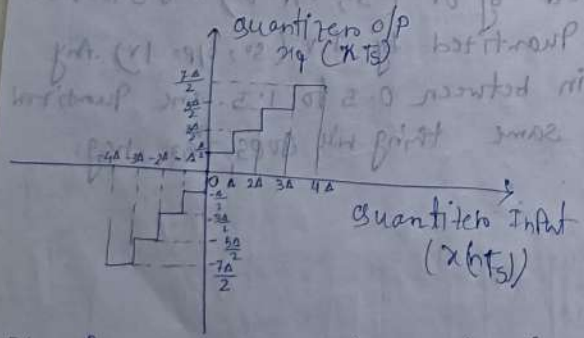
If your sample signals (i/p) are quantized from 0 to 0.5v then, it is quantized by 0v. (So, o/p = 0). 0.5 to 1.5v will be quantized by 1v (so, o/p = 1v). Any values in between 0.5 to 1.5 are quantized by 1v. same thing will goes for neg. axis.



$x(nT_s) \rightarrow \boxed{\text{quantizer}} \rightarrow x_q(nT_s)$

Any value in between 0 to 1 is quantized by 0.5V, from 1 to 2 are quantized by 1.5V. Same thing will goes for negative axis.

Working principle of uniform quantizer



I/P signal range = -4Δ to $+4\Delta$, i/p signal can take any value in this range.

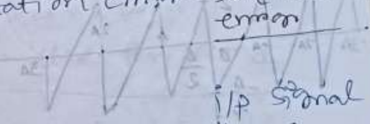
But the o/p of this quantizer can be from one of this 8 o/p's.

for $+4\Delta$, $x(nT_s) = 4\Delta$ (i/p)
$x_q(nT_s) = \frac{7\Delta}{2}$ (o/p)

for -4Δ, $x(nT_s) = -4\Delta$ (i/p)
$x_q(nT_s) = -\frac{7\Delta}{2}$ (o/p)

$\pm 4\Delta$ is quantized by the fixed value $= \pm\frac{7\Delta}{2}$

4Δ is quantized by $\frac{7\Delta}{2}$, so there is a difference in amplitude, this diff. is known as quantization error.



$\boxed{\text{max error} = \frac{\Delta}{2}}$

At 0 the error is minimum, the i/p is increasing gradually, and the error as well. The error will be max at 'Δ'. 'Δ' is quantized by $\frac{\Delta}{2}$, so max error = $\frac{\Delta}{2}$.

$\boxed{\text{Quantization error} = \varepsilon = x_q(nT_s) - x(nT_s)}$

Ex = ① $= \frac{7\Delta}{2} - 4\Delta$
$= -\Delta/2$

Ex = ② $= -\frac{7\Delta}{2} - (-4\Delta)$
$= \frac{\Delta}{2}$

$\boxed{\text{max error } (\varepsilon) = \left|\frac{\Delta}{2}\right|}$ ← only magnitude not sign

$\boxed{\text{If, } 0 < x(nT_S) < \Delta \text{, then } x_q(nT_S) = \frac{\Delta}{2}}$

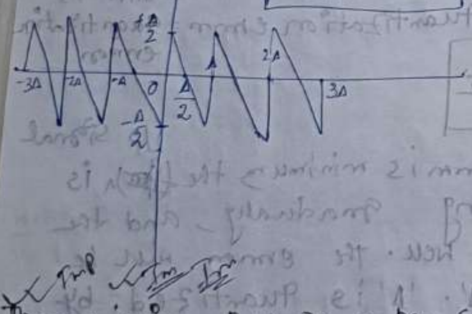
$\boxed{\text{If, } \Delta < x(nT_S) < 2\Delta \text{, then } x_q(nT_S) = \frac{3\Delta}{2}}$

Variation of q.e (ε) with i/p signal

$\left(\frac{\Delta}{2} - \frac{\Delta}{2}\right) = 0$

$\left(\Delta - \frac{\Delta}{2} - \Delta\right) = -\frac{\Delta}{2}$

$\boxed{\varepsilon_{max} = \left|\frac{\Delta}{2}\right|}$



Transmission B.W. of a PCM system

With in BW PCM can transmitt the signal.

Let, a quantizer is using 'v' binary bits to represent each quantization level.

Let say q is the no. of quan. level used in binary bits.

then, $\boxed{q = 2^v}$

In encoding operation after quantization process each quantized level or sample is converted to v no. of binary bits.

$\boxed{\text{In PCM, the quantization rate} = f_s}$

∵ we are generating f_s no of samples per seconds. So, per second f_s no. of samples are transmitted. We are transmitting v no. of binary bits. So, we ~~En the we~~ are transmitting $v \times f_s$ no. of binary bits in per second.

No of bits transmitted per second

we know no. of quantized sample transmission per second is f_s & we transmitt v no. of bits, for each of f_s sample, ~~no~~

$\boxed{\text{No of bits transmitted per second} = vf_s}$

This vf_s is known as signalling rate

$\boxed{\text{Signalling rate} = r = vf_s \text{ in PCM}}$

[Here, f_s = signal freq, $f_s \geq 2f_m$]

∴ Here, $f_s \geq 2f_m$

$\Rightarrow \boxed{r \geq 2vf_s}$

The transmission BW of PCM signal is $= \frac{1}{2}$ of signalling rate

Therefore, transmission BW of PCM signal,

$$\boxed{BW \text{ of } PCM = r/2}$$

$$\geq \frac{2 v f_m}{2}$$

$$\boxed{\text{Transmission BW of PCM signal} \geq v f_m}$$

Quantization Noise/errors in PCM

(mse square error)

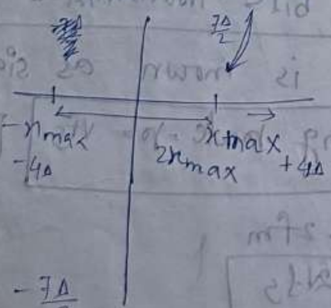
$$\varepsilon = x(nT_s) - x_q(nT_s)$$

signal i/p $= x(nT_s) \rightarrow (-x_{max} \text{ to } +x_{max})$

Peak to peak amp $2x_{max}$

Step size/value $(\Delta) = \frac{2x_{max}}{q}$

$= \frac{\text{Peak to peak value}}{\text{no. of quan. levels}}$

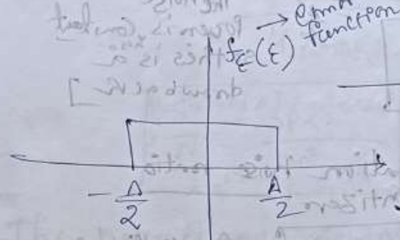


The way of reducing the quantization errors (ε)

If we will more no. of quantization the error will be reduced, i.e. we have to reduce the step size.

as ~~$\Delta \propto 2$~~ $\varepsilon \downarrow \propto \frac{\Delta \downarrow}{2}$ & $\Delta \downarrow = \frac{2x_{max}}{q \uparrow}$ $\boxed{\varepsilon = \text{error}}$

Error function

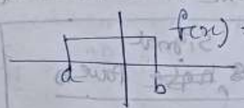


$$\boxed{\frac{1}{\Delta} = \frac{1}{\frac{\Delta}{2} - \left(-\frac{\Delta}{2}\right)}}$$

(error function value)

error function is also known as PDF of error (Probability density function)

Ex –



$$f(x) = \frac{1}{b-a}$$

$$\boxed{\begin{aligned} f_\varepsilon(\varepsilon) &= 0, \text{ for } \varepsilon \leq -\frac{\Delta}{2} \\ &= \frac{1}{\Delta}, \text{ for } -\frac{\Delta}{2} \leq \varepsilon \leq \frac{\Delta}{2} \\ &= 0, \text{ for } \varepsilon > \frac{\Delta}{2} \end{aligned}}$$

MSE (mean square error) / quantization noise power / Error power

Imp

$$E[\epsilon^2] = \int_{-\infty}^{\infty} \epsilon^2 f_\epsilon(\epsilon)\, d\epsilon$$

$$E[\epsilon^2] = \int_{-\Delta/2}^{\Delta/2} \epsilon^2 \cdot \frac{1}{\Delta}\, d\epsilon$$

$$= \frac{1}{3\Delta}\left[\epsilon^3\right]_{-\Delta/2}^{\Delta/2}$$

[as Δ is fixed the noise power is constant also so this is a drawback]

$$\boxed{E[\epsilon^2] = \frac{\Delta^2}{12}}$$

Imp

Signal to quantization noise ratio for linear quantizer

Imp. To find the performance of a quantization signal parameter

$$\left(\frac{S}{N}\right) = \frac{\text{Normalized signal power}}{\text{Normalized noise power}}$$ Imp

SNR (signal to noise ratio)

SNR should be high for an efficient communication signal. We should increase the normalization signal power & decrease the normalization noise power.

$$\boxed{SNR = \frac{\text{Normalized Signal Power}}{\frac{\Delta^2}{12}} \quad —(1)}$$ Imp

now we know, $q = 2^v$

& $\Delta = \frac{2x_{max}}{q}$

So, $\Delta = \frac{2x_{max}}{2^v}$

So, from (1) $SNR = \frac{S}{N} = \frac{P \times 12}{\left(\frac{2x_{max}}{2^v}\right)^2}$ [P = Normalized signal power]

$$= \frac{12P \times 2^{2v}}{4x_{max}^2}$$

$$\boxed{\frac{S}{N} = \frac{3P \times 2^{2v}}{x_{max}^2}}$$ relation between signal to noise rate for linear quantizer

The normalized signal amplitude $= \frac{x_{max}}{x_{max}} = 1$

$\therefore \frac{S}{N} = 3P \times 2^{2v}$

If the power is also normalized (the value is divided by its max value) then P=1,

$$\boxed{\frac{S}{N} = 3 \times 2^{2v}}$$

the max value of P will be, $P \leq 1$

$$\therefore \boxed{\frac{S}{N} \leq 3 \times 2^{2v}}$$

SNR in dB

$$\left(\frac{S}{N}\right)_{dB} \leq 10 \log_{10}(3 \times 2^{2v})$$

$$\leq 10 \left[\log_{10} 3 + \log_{10} 2^{2v}\right]$$

$$\boxed{\frac{S}{N} \le (4.8 + 6v)\ dB}$$ [v → the no. of binary bits]

$\frac{S}{N} \propto v$, If we increase v, SNR will also increase. We can increase v by increasing q, we can increase q by decreasing v.

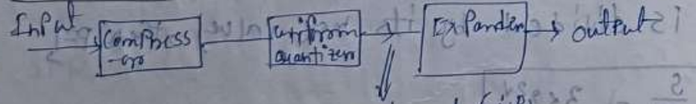
27/3/23

Non-uniform Quantization

Companding = Compression + Expanding

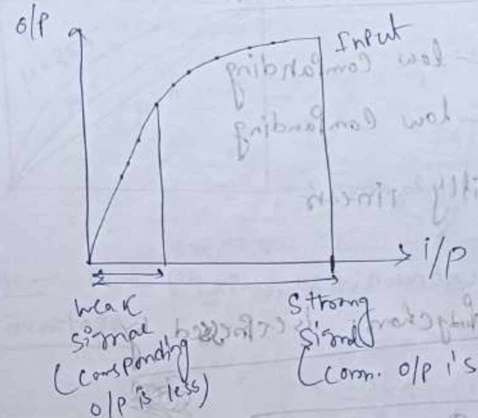
weak sig. is amplified before passing it through a quantizer. ~~for~~ strong signal it will remain strong.

Block diagram



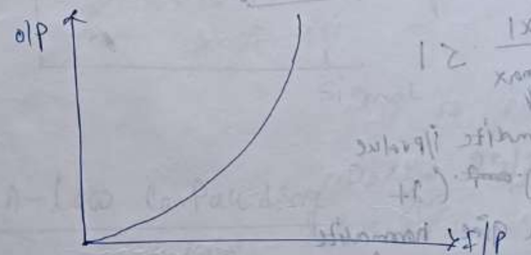
the signal (i/p) which ~~is atlat~~ has been atunated has to be amplified, & the signal (i/p) which has been amplified has to be atunated

Compressor char

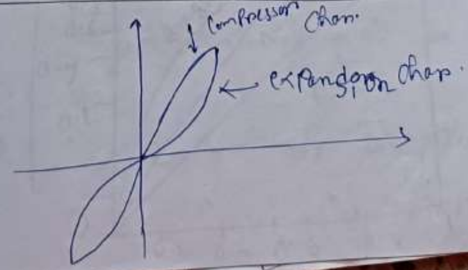


~~Compressor uniform qua~~

Expansion char



Combination of two char. curve



Non uniform quantizer

→ μ-law Companding

→ A-law Companding

Approximitly linear

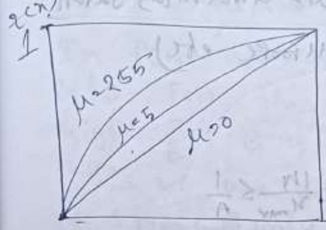
μ-law Companding is expressed by this eqn —

$$\boxed{z(x) = (\text{sign of } x)\,\frac{\log_e\left(1+\mu\frac{|x|}{x_{max}}\right)}{\log_e(1+\mu)}}$$

where, $0 \le \frac{|x|}{x_{max}} \le 1$

↓

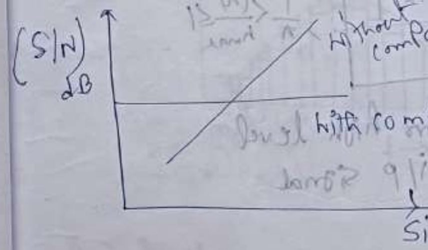
Normalize i/p value of the sig. (It will give normalize signal amp.)



→ normalized input = $\frac{|x|}{x_{max}}$

Use — PCM (Japan, USA, Canada)

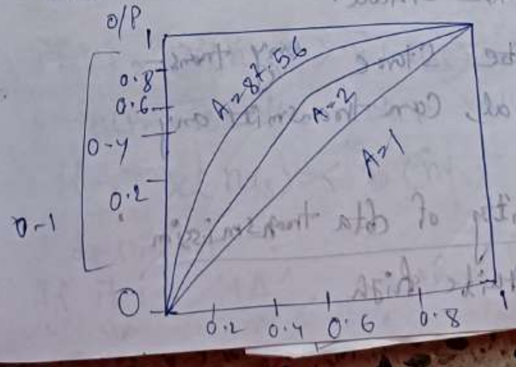
Advantage



A-law Companding

29/3/23

Piecewise linear



A=1, straight line → uniform quantizer.

Practical val. of A-law Companding = 87.56

use – PCM system (South america, Japan, Europe etc)

mathematically

$$\boxed{Z(x) = \frac{A\frac{|x|}{x_{max}}}{1+\log_e A}}, \quad 0 \le \frac{|x|}{x_{max}} \le \frac{1}{A}$$

→ for low level i/p signal

Normalized max i/p value = 1

$$\boxed{Z(x) = \frac{1+\log\left[A\frac{|x|}{x_{max}}\right]}{1+\log_e A}}, \quad \frac{1}{A} < \frac{|x|}{x_{max}} \le 1$$

↓ for high level i/p signal

Application of PCM

1) telephone system

2) Space communication – Transmitted power is quite small.

limiter = can be store any transmitted signal, can transmit any signal.

3) security of data transmission

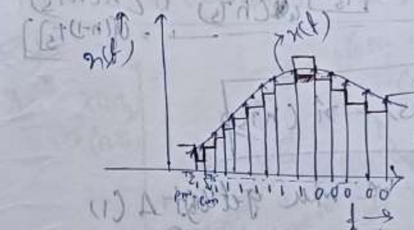
Drawback

1) Diameter is quite high

Delta modulation

In PCM we transmit the entire bits. For this B.W requirement is high. [T.B.W of PCM → nf_m]. For this we use delta modulation. Here we transmit only one bits per ~~level~~ sample. So, we can reduce the B.W in DM.

Working Principle –



→ amp of Samples.
→ we have to convert those sample sigs into ~~Staircase~~ staircase signals.

Previously approximated Sample value of Staircase Sig → $\boxed{\hat{x}(nT_s)}$

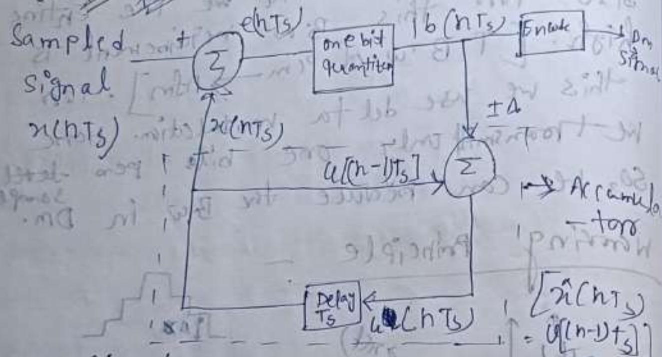
Current Sig. – $\boxed{x(nT_s)}$

if, $x(nT_s) > \hat{x}(nT_s)$, then we have to increase the step value by 'Δ'. $\Delta\uparrow$

if, $x(nT_s) < \hat{x}(nT_s)$ decrease 'Δ' $\Delta\downarrow$

If it is '$+\Delta$', we shall transmit 1
'$-\Delta$' → 0

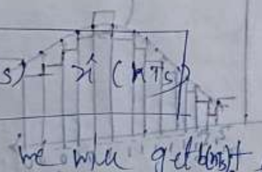
Block diagram of transmitting DM signal



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

Error signal

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



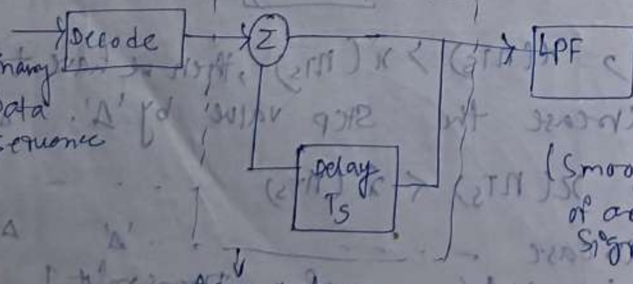
if error is (+)ve we will get $b(nT_s) = +\Delta$ (1)

 " " " (-)ve " " " $b(nT_s) = -\Delta$ (0)

$b(nT_s) = +\Delta$ when $x(nT_s) > \hat{x}(nT_s)$

$= -\Delta$, " $x(nT_s) < \hat{x}(nT_s)$

Receiver



Demodulated signal (smooth version of analog signal)

Accumulator (works as feedback loop to receive signal)

Advantages

1) Signalling rate of transmission value will be reduces, this is high B.W requirement.

2) Here the transmission is quite easy.

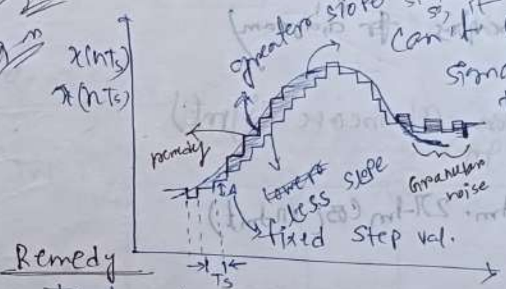
Drawbacks

1) Slope overload destortion

2) Granular noise

1) Slope Overload destortion

Here the signal < i/p signal



staircase slope, so it can't catch i/p signal properly, there will be some diffe, this is called slope overload destortion. (error)

Remedy – To reduce this error we can't ~~fix~~ send ~~a~~ fixed step size, we need to increase the step values. i.e we have to make the step size variable according to the signal slope.

2) Granular noise

Here the staircase signal > i/p signal, ~~so~~

Here the signal slope is almost constant. there will be some small error, this is known as granular noise.

Remedy – we have to make the step size variable according to the signal slope.

Bitrate for DM

$$\boxed{B_R = f_s \times n = f_s\left(\frac{1}{n}\right)}$$

Prove

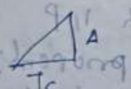
A signal is given, this is a sinusoidal sig.

f_m, A_m

$x(t) = A_m \cos(2\pi f_m t)$

We need to prove S.d.O (error) will occur for this signal, if $A_m > \frac{\Delta}{2\pi f_m T_s}$

T_s = Sampling period.



(previous for diagram)

$\text{slope} = \frac{\Delta}{T_s}$ (for staircase signal)

According to the f_m,

$$\frac{d\,x(t)}{dt} = A_m \cdot 2\pi f_m \cos(2\pi f_m t)$$

$$\left.\frac{d\,x(t)}{dt}\right|_{max} = A_m \cdot 2\pi f_m \cdot 1$$

the ~~distortion~~ S.d.O will occur while,

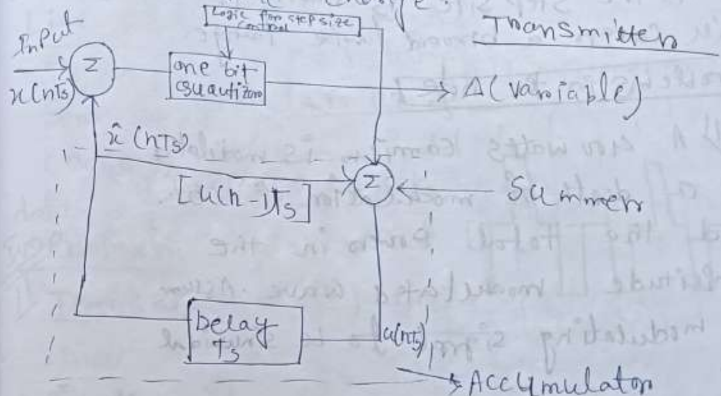
$$A_m \cdot 2\pi f_m > \frac{\Delta}{T_s}$$

$$\Rightarrow A_m > \frac{\Delta}{2\pi f_m T_s} \text{ (proved)}$$

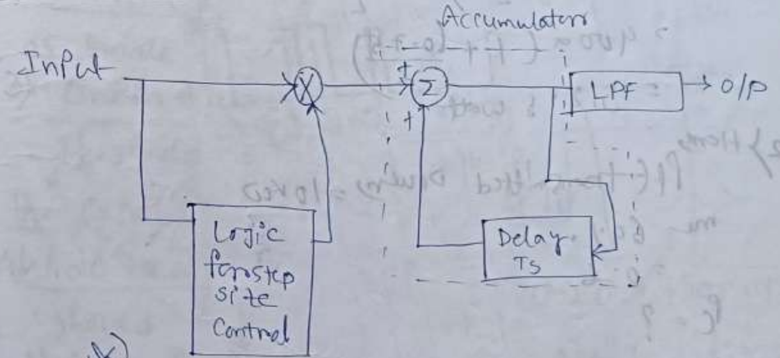
Adaptive Delta modulation (ADM) 12/4/23

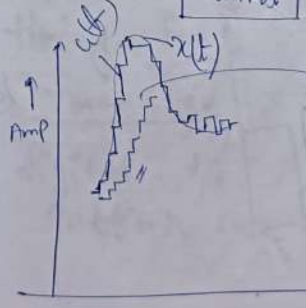
It is ~~used to introduce~~ varried according to the signal change.

Transmitter



Receiver





At delta modulation step size is fixed.

Advantage

1) Adaptive signal change

2) As the step size is varying it will cover a broad / wide range.

Problems in the note

Q.1) A 400 watts carrier is modulated to a depth of modulation of 75%. Find the total power in the amplitude modulated wave. Assume the modulating signal to be sinusoidal one.

$$P_t = P_c\left(1+\frac{m_a^2}{2}\right)$$

$$= 400\left(1+\frac{(0.75)^2}{2}\right)$$

$$= 512.5 \text{ watt}$$

2) Here,

P_t (transmitted power) $= 10 kW$

$m_a = 60\%$

$= 0.6$

$P_c = ?$

$$P_t = P_c\left(1+\frac{m_a^2}{2}\right)$$

$$\Rightarrow 10\times10^3 = P_c\left(1+\frac{(0.6)^2}{2}\right)$$

$$\Rightarrow P_c = \frac{10\times10^3}{\left(1+\frac{(0.6)^2}{2}\right)}$$ Ans

3) ~~AM~~ Angle modulation

26/4/23

Line coding and its properties

It is necessary to represent the datas. Some physical data in waveform = line coding. Use to represent some digital data.

Properties -

1) Transmission bandwidth - It should be as small as possible. Tx BW should be as small as possible.

2) Power efficiency - Tx power should be as small as possible.

3) Error detection & correction capability - It should have error detection & correction capability. Line code

4) Adequate timing content - Line code should have the capability to extract timing from the signal.

5) Transparency - It should transmit a digital data correctly regardless the pattern of 1's or 0's.

Different types of line code

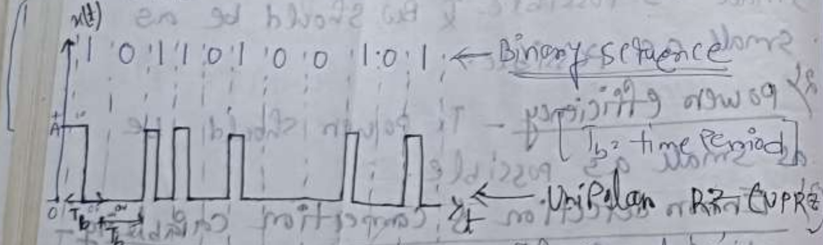
Unipolar RZ & NRZ

26/9/23

(RZ = Return to zero
NRZ = Not return to zero)

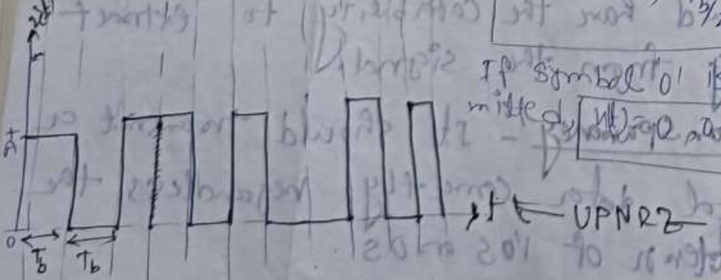
Unipolar RZ & NRZ : 3m 4m

waveform should have one polarity. when i/p data is high. 0 is represented by 0V. That is called on-off (1 → on, 0 → off) line volt. This is it's another name.



If symbol '1' is transmitted then

$$x(t) = +A,\quad 0 < t \le T_b/2$$
$$= 0,\quad T_b/2 < t \le T_b$$



If symbol '0' is transmitted, $x(t) = 0,\ 0 \le t \le T_b$

If the symbol '1' is transmitted,

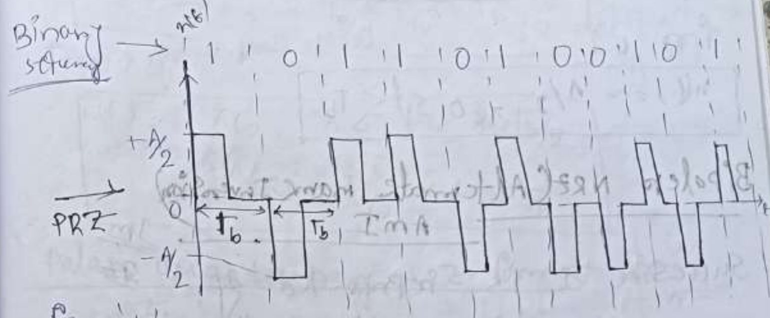
$$\boxed{x(t) = +A,\quad 0 < t \le T_b}$$

If symbol is '0' is transmitted, $\boxed{x(t) = 0,\ 0 \le t \le T_b}$

we have two polarity

Polar RZ

5m 2m



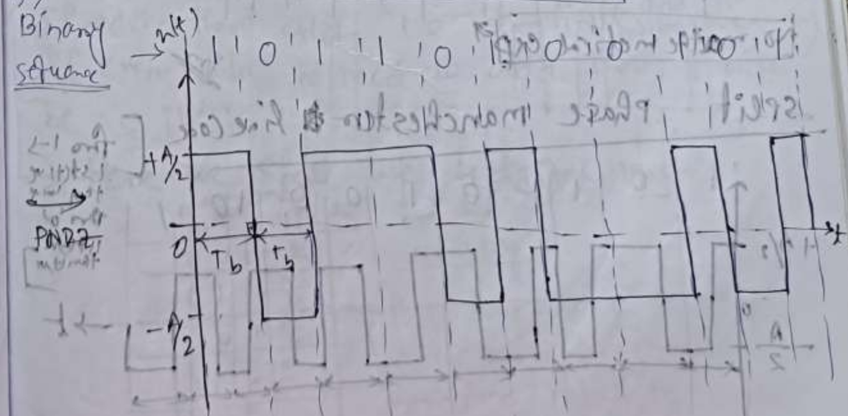
For '1',

$$x(t) = +A/2,\quad 0 \le t \le T_b/2$$
$$= 0,\quad T_b/2 \le t \le T_b$$

For '0',

$$x(t) = -A/2,\quad 0 \le t \le T_b/2$$
$$= 0,\quad T_b/2 \le t \le T_b$$

3m 4m



for '1',

$$\boxed{x(t) = +A/2, \quad 0 \le t \le T_b}$$

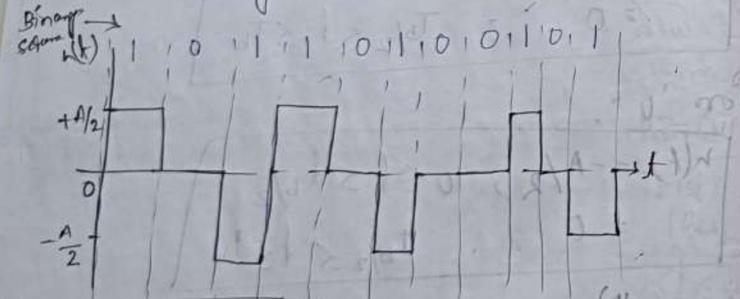
for '0'.

$$\boxed{x(t) = -A/2, \quad 0 \le t \le T_b}$$

Bipolar NRZ (Alternate mark Inversion) AMI

Imp 2m

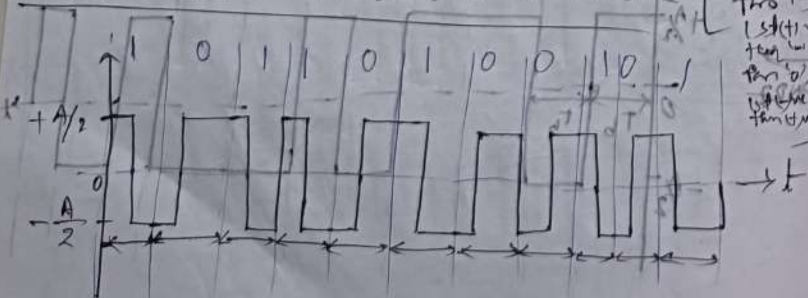
Successive '1' is represented by same pulses with opposite polarity. '0's represented by '0 volt'.



No mathematical exp

Split phase manchester line code

[for 1 → 1st half +ve then '0']



↑ ~~PolRZ~~ SPMLC

for '1',

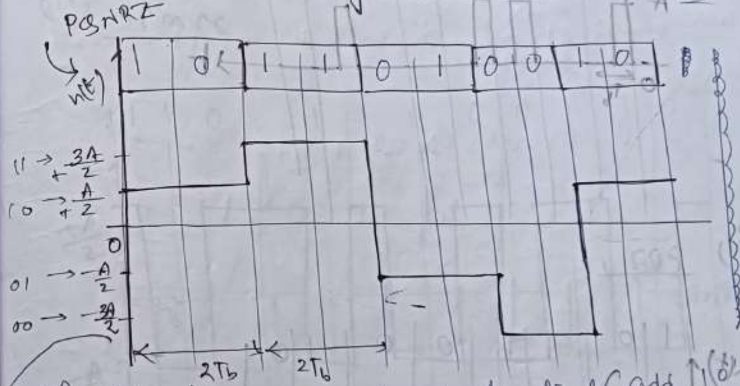
$$\boxed{\begin{aligned} x(t) &= +A/2, \quad 0 \le t \le T_b/2 \\ &= -\frac{A}{2}, \quad T_b/2 \le t \le T_b \end{aligned}}$$

for '0',

$$\boxed{\begin{aligned} x(t) &= -A/2, \quad 0 \le t \le T_b/2 \\ &= +A/2, \quad T_b/2 \le t \le T_b \end{aligned}}$$

Polar Quaternary NRZ format

Imp 2m



If odd then add '0' at first (add 1 '0' for odd binary bits 1, 3, 5...)
If even data no need to add those, otherwise be as it is.

→ Here we shall use 4 digital level.

message bit combination	$x(t) = a_n$
0 0	$-\frac{3A}{2}$
0 1	$-\frac{A}{2}$
1 0	$+\frac{A}{2}$
1 1	$+\frac{3A}{2}$

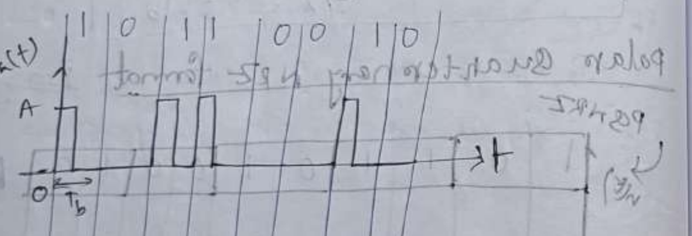
m-ary

For our case m → 4 i.e we have to create 4 positive polarity levels.

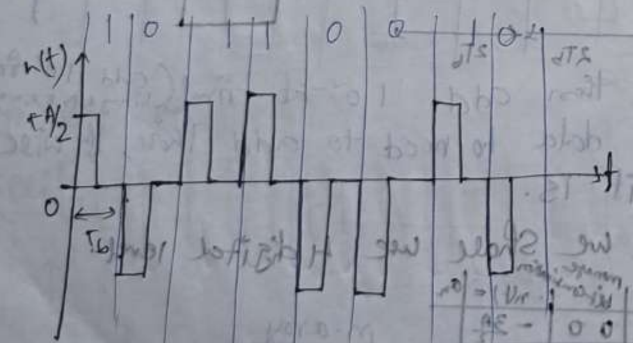
① given, the bit sequence 1 0 1 1 0 0 1 0 . Draw the resultant form, if bit is transmitted –

i) uni polar RZ iv) SRMCC
ii) polar RZ v) polar quaternary
iii) AMI

(i) URZ,

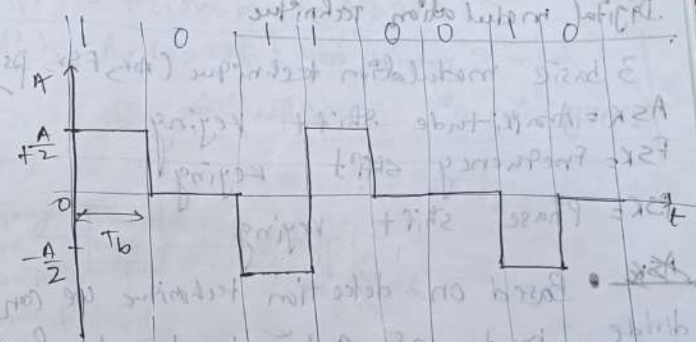


(ii) PRZ,

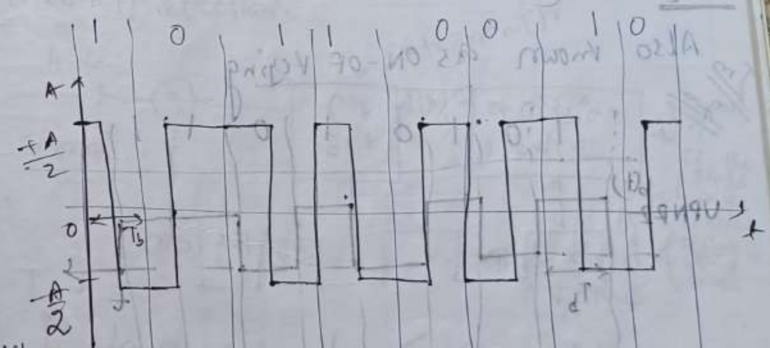


$-\frac{3A}{2}$	0 0
$-\frac{A}{2}$	1 0
$+\frac{A}{2}$	0 1
$+\frac{3A}{2}$	1 1

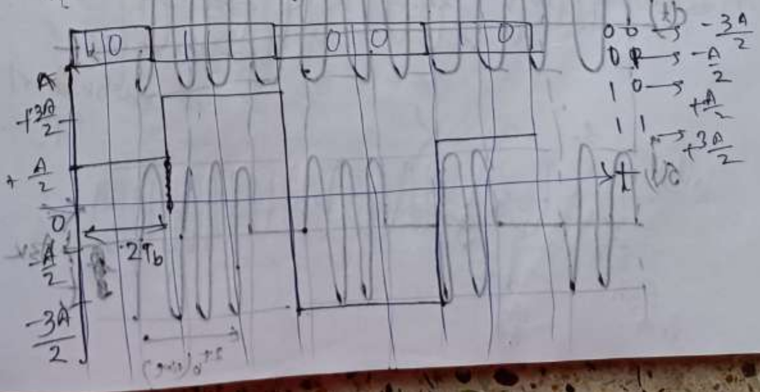
(iii) AMI



(iv) SRMCC



V) Polar quaternary



3/5/23

Digital modulation technique

3 basic modulation technique (ASK, FSK, PSK)

ASK = Amplitude shift keying

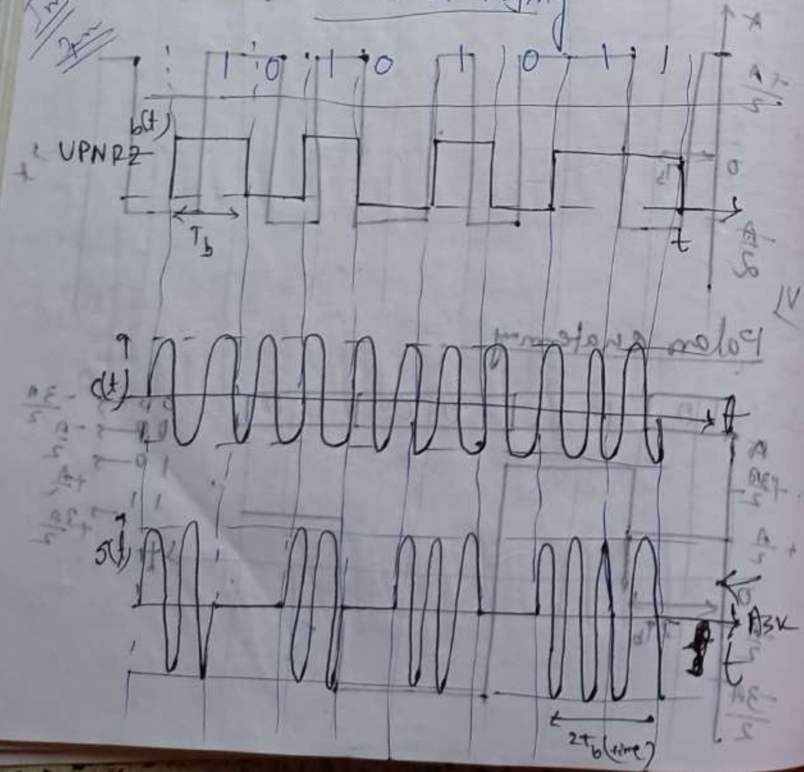
FSK = frequency shift keying

PSK = Phase shift keying

ASK • Based on detection technique we can divide m.t. as Coherent d.m.t. & Non-coherent digital modulation technique.

ASK

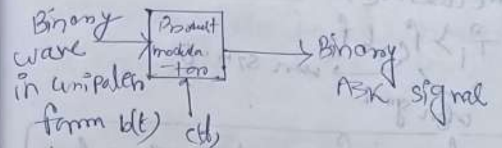
Also known as ON-OF keying



Generation of ASK

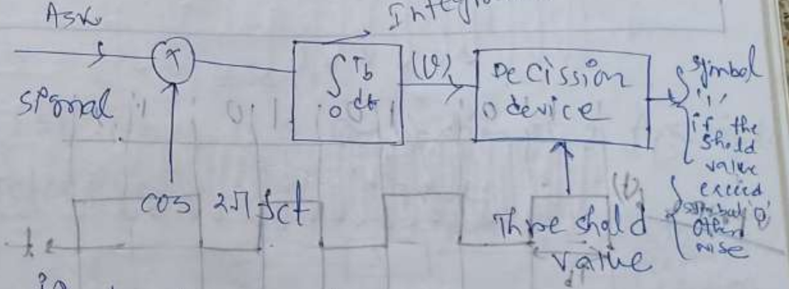
$c(t) = A_c \cos w_c t = A_c \cos 2\pi f_c t$

$$\boxed{\begin{aligned} s(t) &= A_c \cos 2\pi f_c t \text{ for binary symbol '1'} \\ &= 0 \end{aligned}}$$



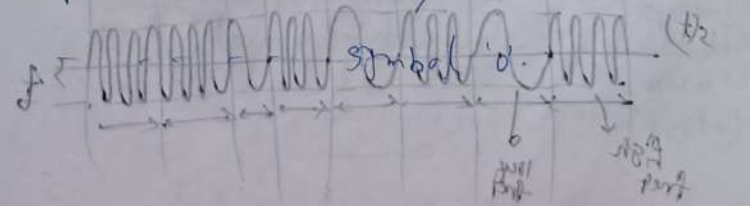
Detection of ASK signal

(i) Coherent detection,



If o/p of Integrator > Threshold value then the D.D will take value/decission in favour of symbol '1'

If - - - - - < - - symbol '0'.



FSK

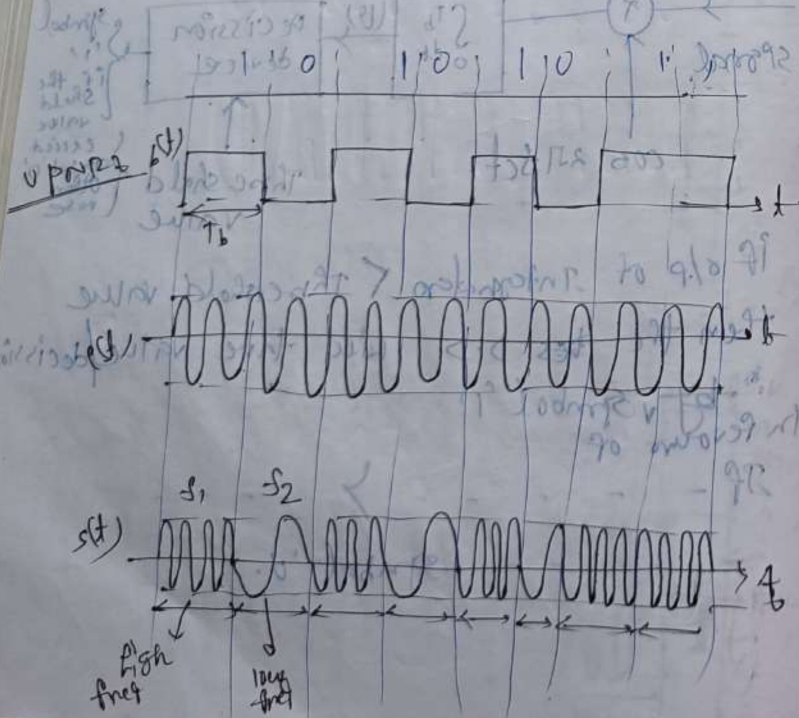
we will use 2 carrier

Generation of FSK/BFSK (Binary FSK)

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

$f_1 > f_c > f_2$ when sym 0, when sym 1

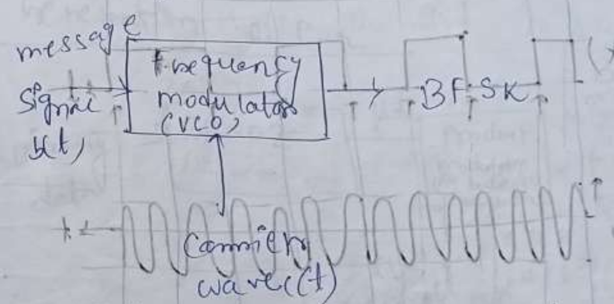
$$\boxed{\begin{aligned} S(t) &= A_c \cos 2\pi f_1 t, \quad \text{for binary symbol 1} \\ &= A_c \cos 2\pi f_2 t, \quad \text{for binary symbol 0} \end{aligned}}$$



the carrier f will low when the message / base band signal is '0', it will be high when the message / base band signal is 1.

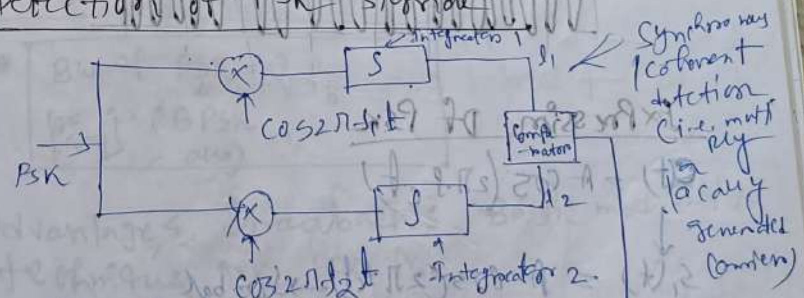
Generation of FSK signal

We have to vary the carrier freq according to the baseband signal (F.M concept)



FSK signal is a combination of 2 ASK signal.

Detection of FSK signal



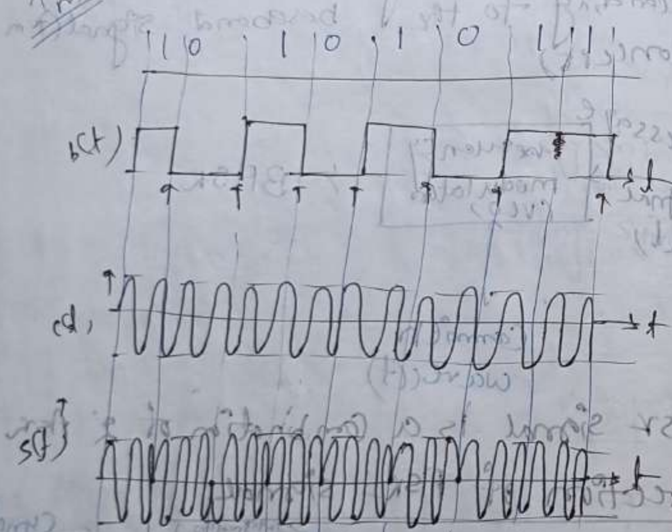
When $l_1 > l_2$, symbol 1

$l_1 < l_2$, symbol 0

symbol 1 when $l_1 > l_2$, otherwise ($l_1 < l_2$) 0

- $\boxed{\text{ASK signal BW} = 2f_b}$
- $\boxed{\text{PSK Signal BW} = 4f_b}$

PSK
Tm



Expression of PSK

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

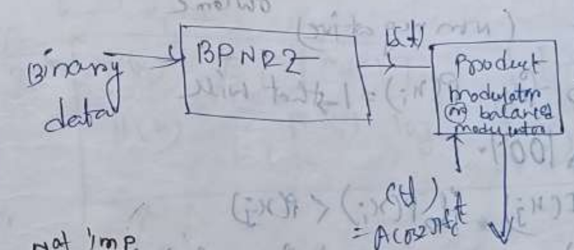
$s_1(t) = A\cos(2\pi f_c t) \rightarrow$ symbol '1'

$s_2(t) = A\cos(2\pi f_c t + 180^\circ) \rightarrow$ symbol '0'
$= -A\cos(2\pi f_c t)$

$s(t) = b(t)\cdot A\cos 2\pi f_c t$ [where $b(t) = +1$ → symbol 1]

$= b(t)\, A\cos 2\pi f_c t$ [where $b(t) = -1$ → symbol 0]

Generation of PSK



[BP = Bipolar as ±1 is present, b(t) can value be ±1]

not imp
Reception of PSK

- $\boxed{\text{BW of PSK} = 2f_b}$ [B = Binary] · (BPSK also)

Advantages, Drawbacks, Basic modulation technique, ASK, FSK, PSK

8/5/23

Information theory

$I(x_i) \propto \frac{1}{P(x_i)}$

Probability of occurance of the event

$\boxed{I(x_i) = \log_b \frac{1}{P(x_i)}}$ b may be = 2, 10, e

Properties of ~~Information~~ $I(x_i)$

outcome

i) $I(x_i) \geq 0$ (Non negative)

ii) $I(x_i) = 0$ for $P(x_i) = 1$ that will happen / 100%

iii) $I(x_i) > I(x_j)$ if $P(x_i) < P(x_j)$

iv) $I(x_i, x_j) = I(x_i) + I(x_j)$ if x_i & x_j are independent

$b = 2 \rightarrow$ bit
$b = 10 \rightarrow$ Decit
$b = e \rightarrow$ nat

Basic of Information

If we have n no. of equiprobable outcomes then it the prob of each outcomes

No. of equiprobable outcome

Ex Im

Source $\rightarrow P(x_1) = \frac{1}{2}$
$P(x_2) = \frac{1}{4}$
$P(x_3) = 1/8$

$\log_2 \frac{1}{1/2} = \log_2 2 = 1$ bit

$\log_2 \frac{1}{1/4} = \log_2 2^2 = 2$ Bit

$\log_2 \frac{1}{1/8} = \log_2 2^3 = 3$ bit

Im Qn

Entropy $\rightarrow$ Avg info. ~~contain per message~~ content per source symbol / message / event.

$x \rightarrow x_i$

Im Qn

$$\boxed{\begin{aligned} H(x) &= E[I(x_i)] \\ H(x) &= \sum_{i=1}^{m} P(x_i) \log_2 \frac{1}{P(x_i)} \end{aligned}}$$

[m = no. of symbols / outcomes]

[P(x) = Probability of Particular message or symbol]

Expression for entropy.

[$I(x_i)$ = Information content]

Binary Source $\rightarrow$ 0, 1 $\frac{1}{2}$

$0 \rightarrow \frac{1}{2}$

$1 \rightarrow \frac{1}{2}$

$H(x) = P(x_1) \log_2 \frac{1}{P(x_1)} + P(x_2) \log_2 \frac{1}{P(x_2)}$

$= \frac{1}{2} \log_2 \frac{1}{1/2} + \frac{1}{2} \log_2 \frac{1}{1/2}$.

$\frac{1}{2} + \frac{1}{2}$

$= 1$ bit / symbol

Information Rate

Source $X \rightarrow r$ symbol/sec [r = no. of symbols]

$$\boxed{R = rH} \text{ bit/sec}$$

[bit/sec = R: Information rate]

[R = Information rate]

[$r \times H$ = $\frac{\text{symbol}}{\text{sec}} \times \frac{\text{bit}}{\text{symbol}}$]

Ex: DMS → r symbol

$P_1 = \frac{1}{2}, P_2 = \frac{1}{4}, P_3 = \frac{1}{8}, P_4 = \frac{1}{16}, P_5 = \frac{1}{16}$

Find entropy & R if there are 16 outcomes/sec.

r = 16 outcome/sec

$$H(X) = \sum_{i=1}^{5} p(x_i) \log_2 \frac{1}{p(x_i)}$$

$$= P_1 \log_2 \frac{1}{P_1} + P_2 \log_2 \frac{1}{P_2} + P_3 \log_2 \frac{1}{P_3} + P_4 \log_2 \frac{1}{P_4} + P_5 \log_2 \frac{1}{P_5}$$

$$= \frac{15}{8} = 1.875 \text{ bit/outcome}$$

(bit/whatever is mentioned in r is it outcome/symbol etc)

$R = r \times H(X)$
$= 16 \times 1.875$
$= 30$ bit/sec

Channel Representation

$$x_i \begin{Bmatrix} x_1 \\ x_2 \\ \vdots \\ x_m \end{Bmatrix} \rightarrow \boxed{P(y_j|x_i)} \rightarrow \begin{Bmatrix} y_1 \\ y_2 \\ \vdots \\ y_n \end{Bmatrix} y_j$$

← Conditional Probability

$\boxed{\text{The no. of Prob.} = m \times n}$

Channel matrix

$$P(Y|X) = \begin{array}{c} \\ x_1 \\ x_2 \\ \vdots \\ x_m \end{array} \begin{bmatrix} P(y_1/x_1) & P(y_2/x_1) & \cdots & P(y_n|x_1) \\ P(y_1/x_2) & P(y_2/x_2) & \cdots & P(y_n/x_2) \\ \vdots & \vdots & & \vdots \\ P(y_1|x_m) & P(y_2|x_m) & \cdots & P(y_n/x_m) \end{bmatrix}$$

Transitional Prob. – obtaining a particular o/p signal by obtaining a particular i/p signal.

i/p = 2×3 → transitional probability

$\boxed{\text{If we add the prob. s of one row it will be = 1}}$

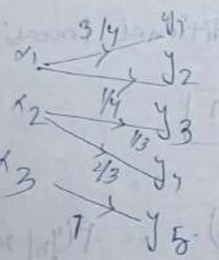
Types of channel

1. Lossless channel – A channel described by a channel matrix having only 1 non-zero element in each column.

$$P(Y/X) = \begin{bmatrix} 3/4 & \frac{1}{4} & 0 & 0 & 0 \\ 0 & 0 & \frac{1}{3} & \frac{2}{3} & 0 \\ 0 & 0 & 0 & 0 & 1 \end{bmatrix}$$

Row = i/p, o/p = col

channel diagram



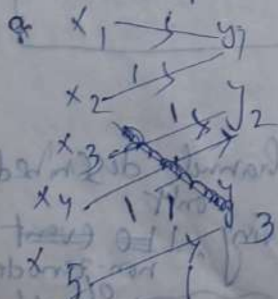
→ channel diagram

ii) Deterministic channel – –

Ans – – – 1 non-zero element in each row.

$$
\begin{array}{cc}
 & \text{o/p} \\
P(y/x) = \text{i/p} & \begin{bmatrix} 1 & 0 & 0 \\ 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix}
\end{array}
$$

iii) Channel diagram



iii) Noiseless channel – Combination of 1st 2.

Ans

$$P(y/x) = \begin{bmatrix} 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 1 \end{bmatrix}$$

Channel diagram

$x_1 \xrightarrow{1} y_1$

$x_2 \xrightarrow{1} y_2$

$x_3 \xrightarrow{1} y_3$

$x_4 \xrightarrow{1} y_4$

1) suggestions (F.E) — Have to attain — FM = 80

12 MCQ → 10 MCQ (compulsory) — 10×1 = 10

[can write, Q.n No. option no. & ans]

[No need to write all ans. Just write the correct option]

2) 10 broad qns → Have to attain 7 — 7×10 = 70

Imp qns

(1) Draw the block diagram of communication system & explain the funtion of each block briefly.

(2) Base band signal → what do you mean by Baseband signal.

(3) modulation Process → what is the needs of modulation. (Explain all the points according to marks (brief & clean))

from chap-1 (Am) → (1) qn from 1st chapter

from chap-2 (Am) → (2) qns from 2nd "

(4) Defination of Amplitude modulation.

(5) modulation index.

(6) Am modulation signal based on modulating index or types of modulation.

(7) Derive an exp for single tone Am signal (Am = Amplitude modulation)

(8) Power content of Am signal & the expression — only for numericals. (2 or 3 ex)

(9) Transmission efficiency concept. (the expression includes)

(10) mathematical ex based on transmission efficiency exp.

(12) Square law diod modulation method (generation of Am)

(13) Square law dectector

(14) DSB-SC carrier generation (Draw the ckt, math. exp) [Balaced modulator produce only two side band powers & show that]

(15) Demodulation Process (synchronous detection)

(16) single tone Am → maths ex [Percentage of Power saving for 50% & 100% modulation]

(17) Demodulation of SSB-SC explain with diagram.

(18) single tone Am generation (same as 12)

(19) vastigious side band → generation process & advantages & disadvantages. → VP

chap-3 [Am] — (2) qns

(20) Defi - Angle modulation mathematical represen tation, freq deviation, carrier swing, math exp for single tone Angle frequency modulation — all numericals in the notes.

(21) NBFm → exp, generation of NBFm, ckt diagram, single tone NBFm's exp & generation

(22) carson's rule.

(23) "Varactor diode method for FM signal generation.

(24) Drawbacks of direct method.

(25) Armstrong method (indirect method) → P

(26) Demodulation method of FM. (slope detector, ratio detector) → P

(27) Pre-emphasis, its application → P

chap 4 [Sampling]

(28) Sampling theory, prove it, nyquist rate & interval, aliasing effect & it's remedy.

Calculation of sampling freq & sampling rate.

(29) Read - reconstruction part.

(30) Any sampling technique. (Natural → P)

chap - 5 (waveform coding)

(31) Explain how analog signal is converted into digital signal using PCM, include block diagram.

(32) Quantization process, uniform quantizer

(33) Transmission BW in PCM system (derive the exp).

(34) Quantization error, noise power calculation

(35) Exp of SNR. (bit/sec in 2nd rely, 1st portion)

(36) Non uniform, Companding, Normal feature, A law companding

(37) Describe the block diagram of delta modulator transmitter (include block diagram)

ECE Prev yr Qn Soln (2022)

1) a) The function of the input transducer in a communication system is – to convert message signal into electrical signal.
sound

b) If the radiated power of AM transmitter is 10kW, the power in the carrier signal for modulation index 0.6 is nearly – 8.47 kW

$$\left[P_t = P_c\left(1+\frac{m_a^2}{2}\right) \right.$$

$$\Rightarrow 10\times10^3 = P_c\left(1+\frac{0.6^2}{2}\right)$$

$$\Rightarrow P_c = 8.47\text{ kW} \Big]$$

c) A 1000 kHz carrier is simultaneously amplitude modulated with 300 Hz & 2 kHz audio sine wave. The frequency which will not present in the output is – 700 kHz

$$\Big[f_c = 1000\times10^3$$

$$f_{m_1} = 300,\ f_{m_2} = 2\times10^3$$

$$f_{USB1} = f_c + f_{m_1} = 1000\times10^3 + 300$$

$$= 1.0003 * 10^6$$

$$= 100.3\times10^3 = 1000.3\text{ kHz} \Big]$$

$$f_{LSB1} = f_c - f_{m_1} = 1000\times10^3 - 300$$

$$= 999.7\text{ kHz}$$

$f_{USB2} = f_c + f_{m2} = 1000 \times 10^3 + 2 \times 10^3$

$= 1002\ KHz$

$f_{LSB2} = f_c - f_{m2} = 1000 \times 10^3 - 2 \times 10^3$

$= 998\ KHz$

$\therefore$ Ans = 700 KHz

d) ~~A It~~ In Commerical Fm broadcasting the maximum frequency deviation is normally - 75 KHz

e) Pre-emphasis in Fm ~~broad~~ System involves - amplification of higher frequency components of the modulating signal.

f) If a signal band-limited to f_m Hz is sampled at a rate less than $2f_m$ the reconstructed signal will be distorted

g) Companding is used in PCM for improve signal to noise ratio for low level input signal.

h) In a DM system the granular noise occurs when the modulating signal changes very slowly

i) How many bits would be required to represent 256 quantization levels in PCM?
8 bits

$2^n = 256$
$\Rightarrow 2^n = 2^8$
$\Rightarrow n = 8$ bits

j) Which modulation technique is known as ON-OFF keying - ASK

2) a) Draw the block diagram of a Communication system & briefly explain the function of each block.
done

b) What is baseband transmission. Done

c) Explain the needs of modulation in communication system. Done

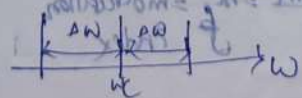
3) a) What is meant by amplitude modulation? Done

b) Define the term modulation index for Am. Done

c) Derive an expression for single tone amplitude modulation wave. Done.

d) Draw the waveform of overmodulated Am wave and write the condition for overmodulation. Done

5) a) What do you mean by frequency deviation & carrier swing? We know that the instantaneous freq of Fm wave is given as $\omega_i = \omega_c + k_f x(t)$
The instantaneous freq. of Fm signal varies with time around the carrier freq ω_c according to the modulating signal. The maximum change in instantaneous frequency from the avg frequency ω_c is called frequency deviation. Freq deviation $= \Delta\omega = \pm |k_f\, x(t)|_{max}$



Carrier swing = The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the carrier swing. Carrier swing = 2 × freq. deviation = $2\Delta\omega$

b) Derive an expression for single tone narrow band fm signal.

We know that the general exp. for NBFM is given as,

$s(t)_{fm} = A\cos\omega_c t - A k_f\, y(t) \sin\omega_c t$ — ①

Here, $y(t) = \int x(t)\,dt$

where $x(t)$ is modulating signal & $A\cos\omega_c t$ is a carrier signal.

For a single tone narrow band fm, we take modulating signal as consisting of a single frequency.

i.e., $x(t) = V_m \cos\omega_m t$ where ω_m is the single modulating frequency.

Thus, $y(t) = \int x(t)\,dt$

$= \int V_m \cos\omega_m t\,dt$

$= \frac{V_m}{\omega_m} \sin\omega_m t$

Putting this value of $y(t)$ in the eqn ① we get, $s(t)_{fm} = A\cos\omega_c t - A k_f \frac{V_m}{\omega_m} \sin\omega_m t \sin\omega_c t$

But we know that $\frac{k_f V_m}{\omega_m} = m_f$ = modulation index

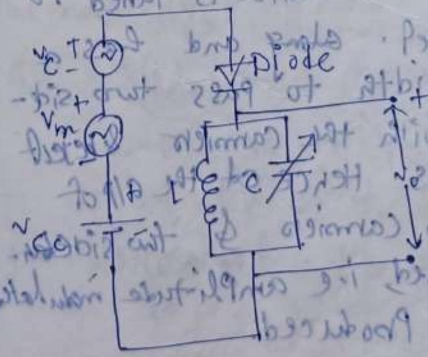
Hence, $s(t)_{fm} = A\cos\omega_c t - A m_f \sin\omega_m t \sin\omega_c t$ which is the required expression for a single-tone narrow band fm.

c) A single tone frequency modulated wave is denoted by following expression:

$s(t) = 12\cos[6\times10^8 t + 5\sin(1250t)]$. Determine carrier frequency, modulating frequency, modulation index & maximum frequency deviation.

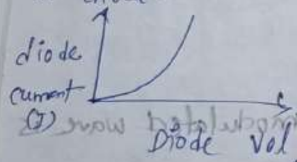
Done (notes) (frequency means = linear frequency)

4(a) Describe the generation of Am signal using square law diode modulation technique.



← Square law diode modulation

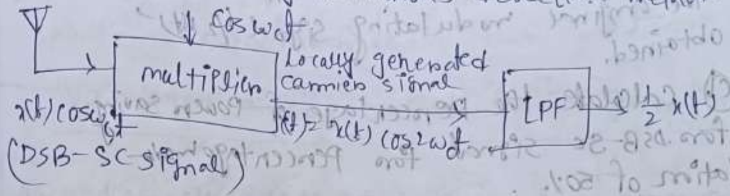
It may be observed from the above fig. that the carrier and modulating signals are applied across the diode. A dc battery V_{cc} is connected across the diode to get a fixed operating point on the V-I characteristics of diode.



As the carrier and modulating frequency are applied at the input of diode, the different freq. terms will appear at the o/p of the diode. These different freq. term are applied across a tuned ckt, which is tuned to the carrier freq. along and has a narrow bandwidth to pass two side-bands along with the carrier & reject other frequencies. Hence at the o/p of the tuned ckt, carrier & two sidebands are obtained, i.e amplitude modulated wave/signal is produced.

(b) How is a DSB-SC signal demodulated? :- The DSB-SC signal may be demodulated by synchronous detection method.

Block diagram :- Fig below shows the block diagram of synchronous detection method.



Synchronous Detection method

In synchronous detection method, the received modulated or DSB-SC signal is first multiplied by a locally generated carrier signal $\cos \omega_c t$ & then passed through a low-pass filter (LPF). At the o/p of a LPF, the modulating signal is recovered.

mathematically,

$$e(t) = \underbrace{x(t)\cos\omega_c t}_{\text{DSB-SC signal}} \cdot \underbrace{\cos\omega_c t}_{\text{locally generated carrier signal}}$$

$$e(t) = x(t)\cos^2\omega_c t = \frac{1}{2}x(t)[2\cos^2\omega_c t]$$

$$= \frac{1}{2}x(t)[1+\cos 2\omega_c t]$$

$$= \frac{1}{2}x(t) + \frac{1}{2}x(t)\cos 2\omega_c t$$

Now it may be observed that when multiplied signal $e(t)$ is passed through a LPF, then the term $\frac{1}{2}x(t)\cos 2\omega_c t$ centred at $\pm 2\omega_c$ is suppressed by LPF and thus at the o/p of LPF the original modulating signal $\frac{1}{2}x(t)$ obtained.

Q) Calculate the percentage of power saving for DSB-SC signal for percentage modulation of 50%.

The total power in Am wave.

$P_t = P_c\left(1 + \frac{m_a^2}{2}\right)$

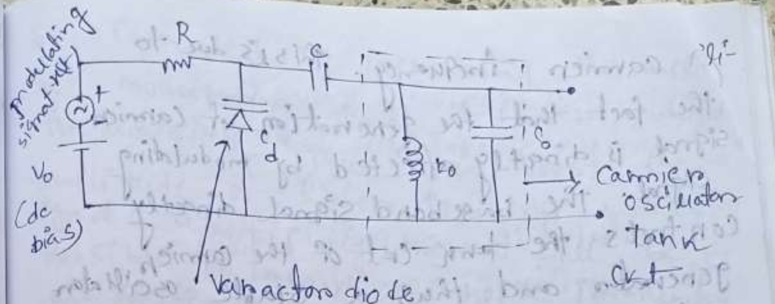
At 50% depth of modulation (m_a) $m = 0.5$

$P_t = 1.125 P_c$

$\therefore$ % of power saving $= \frac{P_c \times 100}{1.125 P_c} = 88.88\%$.

Q) a) Explain the varactor diode method for generation of Fm signal.

The varactor diode is a semiconductor diode whose junction capacitance changes with dc bias voltage. This varactor diode is connected in shunt with the tuned ckt of the carrier oscillator. This arrangement is shown in fig. below.



Varactor diode method of Fm generation

In varactor diode Fm generation arrangement the value of capacitor C is made much smaller at the operating frequency so that its reactance is very low. As a result when C is connected in series with the lower capacitance of varactor diode, the effect is as if diode is connected directly across the tuned ckt. Then the total effective ckt capacitance is the capacitance C_d of the varactor diode in parallel with C_0. This fixes the center carrier frequency.

(b) What are the drawbacks of direct method for Fm generation?

Following are the drawbacks of the direct method

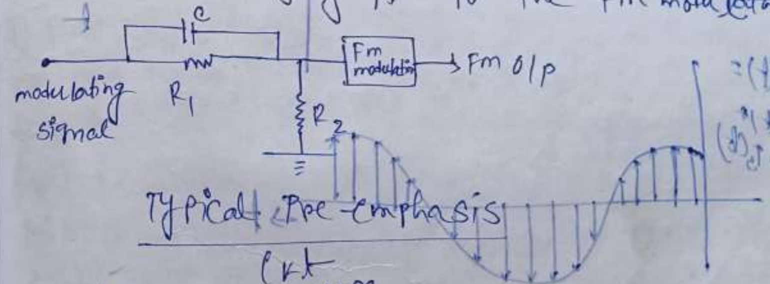
i) In direct method of Fm generation, it is not easy to get a high order stability in carrier frequency. this is due to the fact that the generation of carrier signal is directly affected by modulating signal. the baseband signal directly controls the tank ckt of the carrier generator and thus a stable oscillator ckt (i.e crystal oscillator) cannot be used. This means that the carrier generation cannot be of high stability which is necessary requirement.

~~cannot be of high stability which is necessary~~

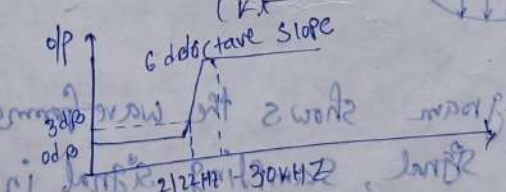
ii) The non-linearity of the varactor diode produces a frequency variation due to harmonics of the modulating signal and therefore the FM signal is distorted.

(c) Explain Pre-emphasis in FM signal generation.

It has been proved that in FM, the noise has a greater effect on the higher modulating frequencies. This effect can be reduced by increasing the value of modulation index (m_f) for higher modulating frequencies (f_m). This can be done by increasing the deviation Δf & Δf can be increased by increasing the amplitude of modulating signal at higher modulating frequencies. Thus if we boost the amplitude of higher frequency modulating signals artificially then it will be possible to improve the noise immunity at higher modulating frequencies. The artificial boosting of higher modulating frequencies is called pre-emphasis. Boosting of higher frequency modulating signal is achieved by using pre-emphasis ckt of fig below. The modulating AF signal is passed through a high pass RC filter, before applying it to the FM modulator.



Typical Pre-emphasis ckt



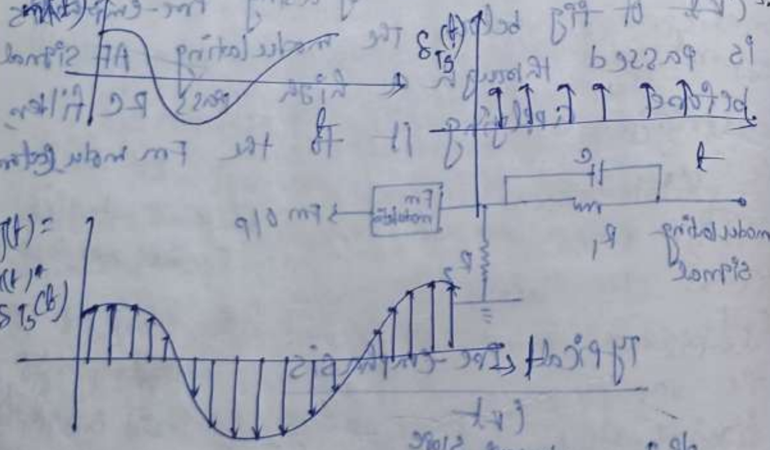
Pre-emphasis characteristics

7) (a) State sampling theorem. Done

(b) What is aliasing? How is it prevented? Done.

(c) Explain ideal sampling technique.

Ideal Sampling is also known as instantaneous sampling or impulse sampling. The sampling process multiplies the i/p signal and the carrier signal, which is present in the form of train of pulses.



The above diagram shows the waveforms of the message signal, sampling signal in the form of train of pulses, and the sampled signal. The working principle that multiplies the i/p signal and the sampling signal is known as multiplication principle.

mathematically,

$$g(t) = x(t) * \delta_{T_s}(t)$$

(sampled signal) (message signal) (sampling signal that is present in form of train of pulses)

(d) Find the Nyquist rate & the Nyquist interval for the signal $x(t) = \cos(4000\pi t)\cos(1000\pi t)$. Done.

8) (a) Explain the operation of a PCM transmitter with suitable block diagram. (Digital to analog signal conversion (same)). ADC. Done

(b) Derive the expression for transmission bandwidth in a PCM system. Done

(c) A TV signal having a b.w of 4.2 MHz is transmitted using binary PCM system. Given that number of quantization level is 512. Determine transmission bandwidth and output signal to quantization noise ratio. Done

(5) a) Explain Delta modulation in detail with suitable block diagram.

Reason to use Delta modulation –

We have observed in PCM that it transmits all the bits which are used to code a sample. Hence signaling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, Delta modulation is used.

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Mathematical expression –

The principle of delta modulation can be explained with the help of few eqn as under –

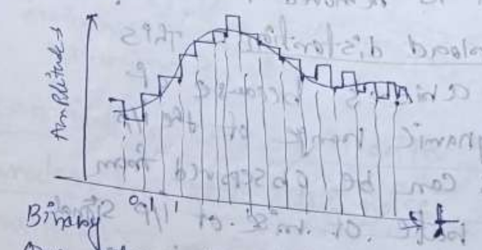
The error between the sampled value of $x(t)$ & last approximated sample is given as

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

Where $e(nT_s)$ = error at present sample

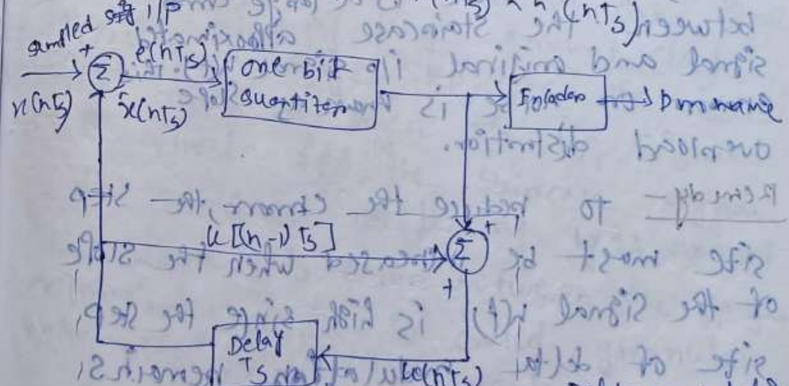
$x(nT_s)$ = sample signal of $x(t)$

$\hat{x}(nT_s)$ = last sample approximation of the staircase waveform

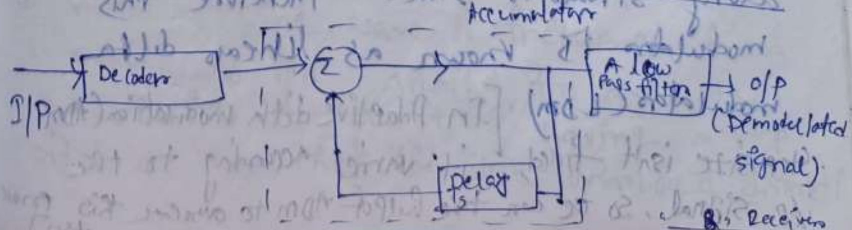


Delta modulation waveform

$$b(nT_s) = \begin{cases} +\Delta, & \text{if } x(nT_s) \geq \hat{x}(nT_s) \\ -\Delta, & \text{if } x(nT_s) < \hat{x}(nT_s) \end{cases}$$



Delta modulation transmitter

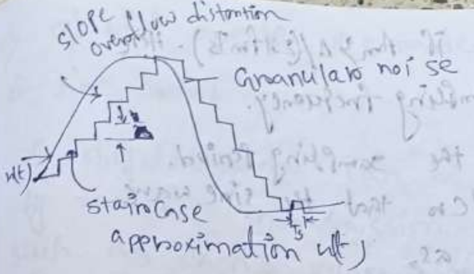


B) Receiver

(b) What are slope overload distortion & granular noise in delta modulation & how it is removed ~~for~~ in ADM?

Slope overload distortion :- This distortion arises because of large dynamic range of the i/p signal. As can be observed from fig, the rate of rise of i/p signal x(t) is so high that the staircase signal can't approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t). Hence, there is a large error between the staircase approximated signal and original i/p signal x(t). This error or noise is known as slope overload distortion.

Remedy - To reduce the error, the step size must be increased when the slope of the signal x(t) is high. Since the step size of delta modulation remains fixed, its max or minimum slope occurs along straight lines. Therefore this modulator is known as linear delta modulator (LDM). [In Adaptive delta modulation (ADM) step size isn't fixed, it varies according to the i/p signal, so we can take help of ADM to overcome this problem also]



Quantization errors in delta modulation

Granular or ~~Ideal~~ Idle noise :-

Granular or idle noise occurs when the step size is too large compared to small variations in the input signal. This means that for very small variation in the i/p signal the staircase signal is changed by large amount (Δ) because of large step size. The above fig shows that when the i/p signal is almost ~~flat~~ flat, the staircase signal u(t) keeps on oscillating by ±Δ around the signal. This error between the i/p & staircase approximated signal is called granular noise. The soln of the problem is to make step size small or we can take help of ADM also.

Remedy

(c) Given a sine wave of frequency f_m & amplitude A_m applied to a delta modulation ~~& how it is~~ having a step size Δ. Show that the slope overload distorti

-on will occur if $A_m > \Delta/(2\pi f_m T_s)$. Here T_s is the sampling frequency.

Here T_s is the sampling period.

Let us consider that the sine wave is represented as,

$x(t) = A_m \sin(2\pi f_m t)$

It may be noted that the slope of $x(t)$ will be maximum when derivative of $x(t)$ with respect to t will be maximum. The maximum slope of delta modulator may be given as

maximum slope = $\frac{\text{Step size}}{\text{Sampling period}} = \frac{\Delta}{T_s}$

We know that, slope overload distortion will take place if slope of sine wave is greater than slope of delta modulator.

$\max \left| \frac{d}{dt} x(t) \right| > \frac{\Delta}{T_s}$

$\Rightarrow \max \left| \frac{d}{dt} A_m \sin(2\pi f_m t) \right| > \frac{\Delta}{T_s}$

$\Rightarrow \max \left| A_m 2\pi f_m \cos(2\pi f_m t) \right| > \frac{\Delta}{T_s}$

$\Rightarrow A_m 2\pi f_m > \frac{\Delta}{T_s}$ $\left[\because \cos(2\pi f_m t)_{max} = 1 \ (\because t = 0) \right]$

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

10) a) What are the desirable properties of line code?

The digital data can be transmitted by various transmission or line code such as on-off, polar, bipolar and so on. This is called line coding. The desirable properties of line code—

1) **Transmission bandwidth** – For a line code, the transmission bandwidth must be as small as possible.

2) **Power efficiency** – For a given bandwidth & a specified detection error probability the transmitted power for a line code should be as small as possible.

3) **Error detection & correction capability** – It must be possible to detect & preferably to correct detection errors. For ex, in a bipolar case, a signal error will cause bipolar violation & thus can easily be detected.

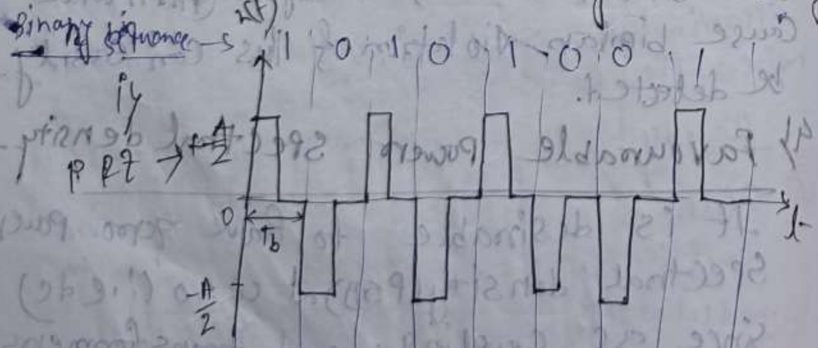
4) **Favourable power spectral density** – It is desirable to have zero power spectral density (PSD) at $\omega = 0$ (i.e. dc) since ac coupling and transformers

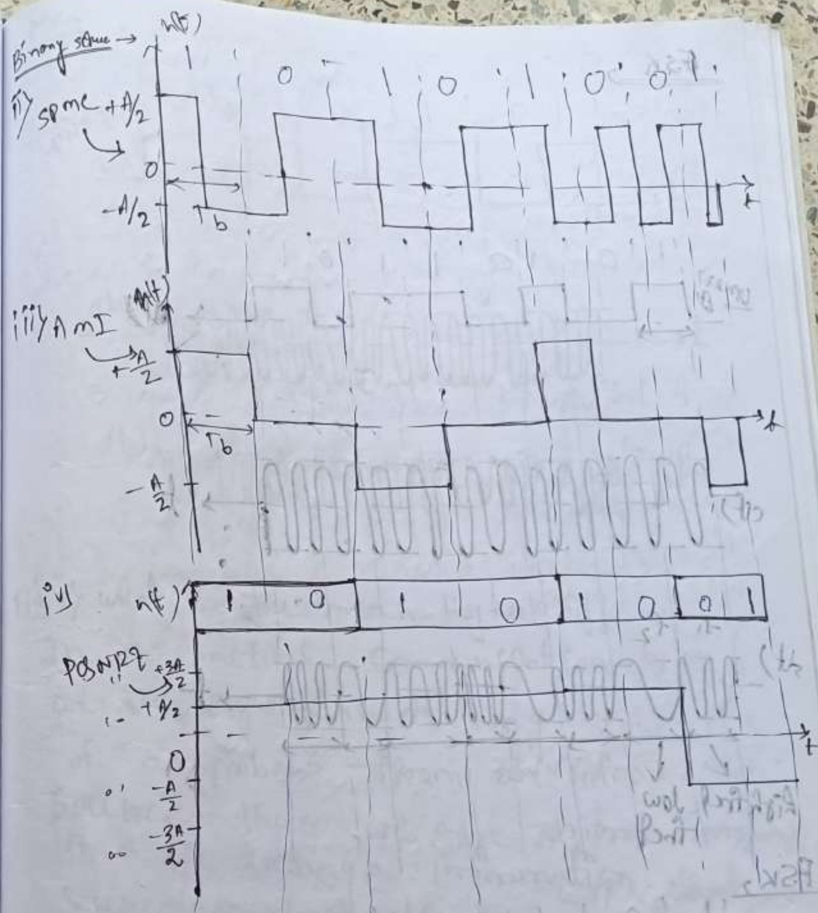
are used at the repeaters significant power in low-frequency components causes dc wander in the pulse stream when ac coupling is used.
The a.c coupling is required since the dc paths provided by the cable pairs between the repeater sites are used to transmit the power required to operate the repeaters.

5) Adequate timing content – It must be possible to extract timing or clock information from the signal.

6) Transparency – It must be possible to transmit a digital signal correctly regardless the pattern of 1's and 0's.

b) The binary data 10101001 is transmitted over a baseband channel. Draw the waveform for the transmitted data using following format. i) Polar RZ ii) Split phase manchester coding. iii) AMI iv) Polar quaternary NRZ signaling

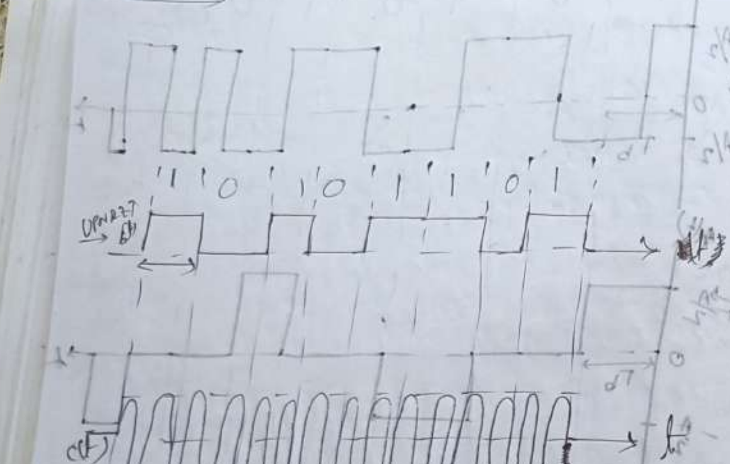


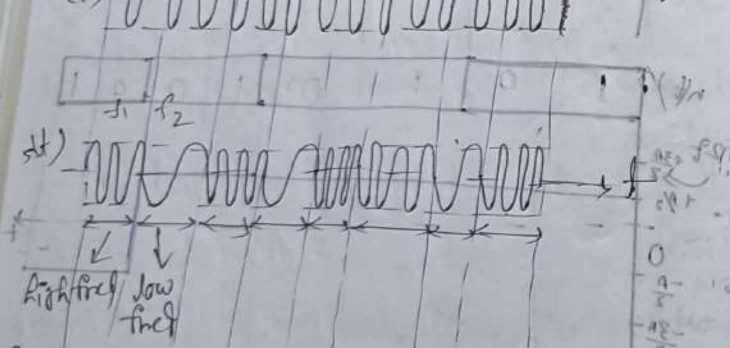


00 → $-\frac{3A}{2}$
01 → $-\frac{A}{2}$
10 → $\frac{A}{2}$
11 → $\frac{3A}{2}$

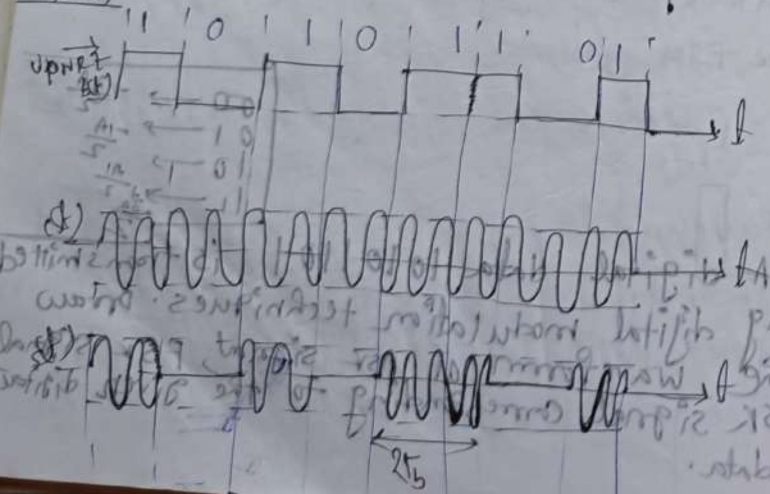
c) A digital data 10101101 is transmitted using digital modulation techniques. Draw the waveform of ASK signal, FSK signal, PSK signal corresponding to the given digital data.

. ASK ;

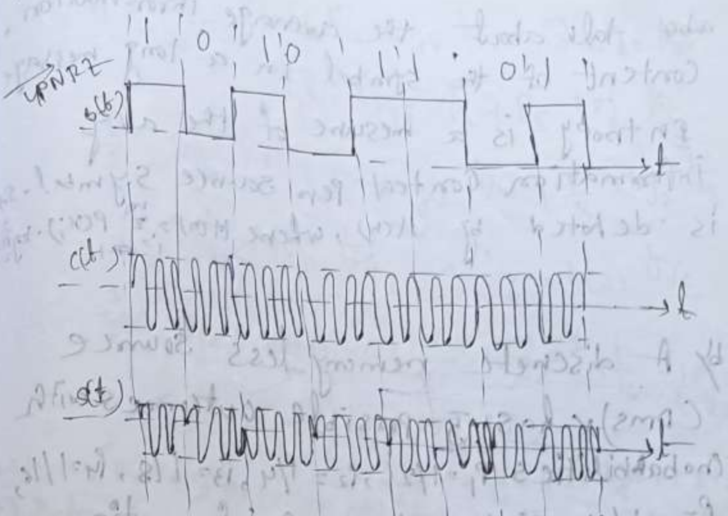




FSK ;



PSK ;



11) a) what do you mean by entropy?

In a practical communication system, we usually transmit long sequences of a symbols from an information source. Thus we are more interested in a the average information that a source produces than the information content of a source single symbol. In order to get information content of the symbol, we take notice of the fact that the flow of information in a system can fluctuate widely because of randomness involved into the selection of the symbols. Thus we request to

~~abo~~ talk about the average information content of the symbol in a long message.

Entropy is a mesure of the avg information content per source symbol, & it is denoted by H(n), where $H(n) = \sum_{i=1}^{m} P(x_i) \cdot \log_2 \frac{1}{P(x_i)}$

b) A discrete memory less source (DMS) x has 5 possible outcomes with probabilities $P_1 = 1/2$, $P_2 = 1/4$, $P_3 = 1/8$, $P_4 = 1/16$, $P_5 = 1/16$. Find entropy & information rate if there are 16 outcomes per second.

$P_1 = 1/2$, $P_2 = 1/4$, $P_3 = 1/8$, $P_4 = 1/16$, $P_5 = 1/16$

Total 5 possible outcomes

given that, $r = 16$ outcomes/s

we know that, $H(X) = \sum_{i=1}^{m} P(x_i) \cdot \log_2 \frac{1}{P(x_i)}$

So, here, $H(X) = \sum_{i=1}^{5} P(x_i) \cdot \log_2 \frac{1}{P(x_i)}$

$= P(x_1) \cdot \log_2 \frac{1}{P(x_1)} + P(x_2) \cdot \log_2 \frac{1}{P(x_2)} + P(x_3) \cdot \log_2 \frac{1}{P(x_3)} + P(x_4) \cdot \log_2 \frac{1}{P(x_4)} + P(x_5) \cdot \log_2 \frac{1}{P(x_5)}$

[m = no. of outcomes
P(x_i) = Probability of particular message outcome
H(x) = Entropy]

$= P_1 \cdot \log_2 \frac{1}{P_1} + P_2 \cdot \log_2 \frac{1}{P_2} + P_3 \cdot \log_2 \frac{1}{P_3} + P_4 \cdot \log_2 \frac{1}{P_4} + P_5 \cdot \log_2 \frac{1}{P_5}$

$= \frac{1}{2} \cdot \log_2 \frac{1}{1/2} + \frac{1}{4} \cdot \log_2 \frac{1}{1/4} + \frac{1}{8} \cdot \log_2 \frac{1}{1/8} + \frac{1}{16} \cdot \log_2 \frac{1}{1/16} + \frac{1}{16} \cdot \log_2 \frac{1}{1/16}$

$= \frac{1}{2} \cdot \log_2 2 + \frac{1}{4} \log_2 2^2 + \frac{1}{8} \cdot \log_2 2^3 + \frac{1}{16} \cdot \log_2 2^4 + \frac{1}{16} \cdot \log_2 2^4$

$= \frac{1}{2} + \frac{1}{4} \times 2 + \frac{1}{8} \times 3 + \frac{1}{16} \times 4 + \frac{1}{16} \times 4 = \frac{15}{8} = 1.875$ bit/outcome

we know, $R = r \times H(x)$ [R = info. rate] $\Rightarrow R = 16 \times \frac{15}{8} = 30$ bit/s

c) Determine the Huffman code for the following messages with their probabilities given.

x_1	x_2	x_3	x_4	x_5	x_6
0.3	0.25	0.2	0.12	0.08	0.05

$\boxed{\text{not in syllabus}}$

(38) Errors of DM, remedy of them (how are they eliminated)

(39) $A_m \geq \frac{\Delta}{2\pi f_c T_s}$ (prove)

(40) Adaptive delta modulation, modulator ckt.

(11) 9th

Chap-6 (WC-11)

(1) Given binary pattern, convert it into NRZ, SPL, AMI format

modulation technique (theory) chap-7

(42) ASK, FSK, PSK - transmitter & receivers (generation & detection) merged

(43) coherent & non coherent detection of signal

chap-8 (Info. Theory)

(43) How info. is calculated (exp)

(42) math. ex.

(44) Entropy (related maths)

(45) Channel diagram, types, matrix (Discuss the noiseless channel & diagram & draw channel diagram)