SWAMI VIVEKANANDASCHOOL OF ENGG. & TECH.

MADANPUR, BBSR

LECTURE NOTES

ON

ANALOG & DIGITAL COMMUNICATION

Year & semester: 3RD Year, 5TH Semester

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Sub-ADC   Dt:- 01/09/2020

(Analog & digital communication)

Unit-1

Elements of Communication System   NO-1

Communication process:- concept of elements of communication system It's block dig-

• What is communication?

Communication can be defined as the process of ex-change of information through means suches, words, action, ~~signals~~ between two or more Individuals.

Parts of a communication system:

Any system which provides communication consists of the three Important and basic parts as shown on the fig

$\boxed{\text{Sender}}$ - - - Channel - - $\boxed{\text{Reciver}}$

Sender:-

Sender is the person who sends a messge It could be a transmiting station from where the signal is transmitted.

Channel:-

message

It the medium through which the message signals treavel to reach the dissinata

Reciver :

It the person who receive the messages. it could be a reciving station where the transmitted signal is being recived. Deepending on their characteristics signals are mainly clasified in to two types Analog & digital.

Types of signals

- Analog
  - Periodic
  - Aperiadic
- Digital
  - Periodic
  - Aperiodic

Analog signal :-

A contineous time verying signal which represents time varying quantity can be termed as an Analog signal. This signal keeps on varying respect to time according to the instantaneous value of the quantity which represents

The communication which is based on

anlog signal & analog values is called as
freq communication.

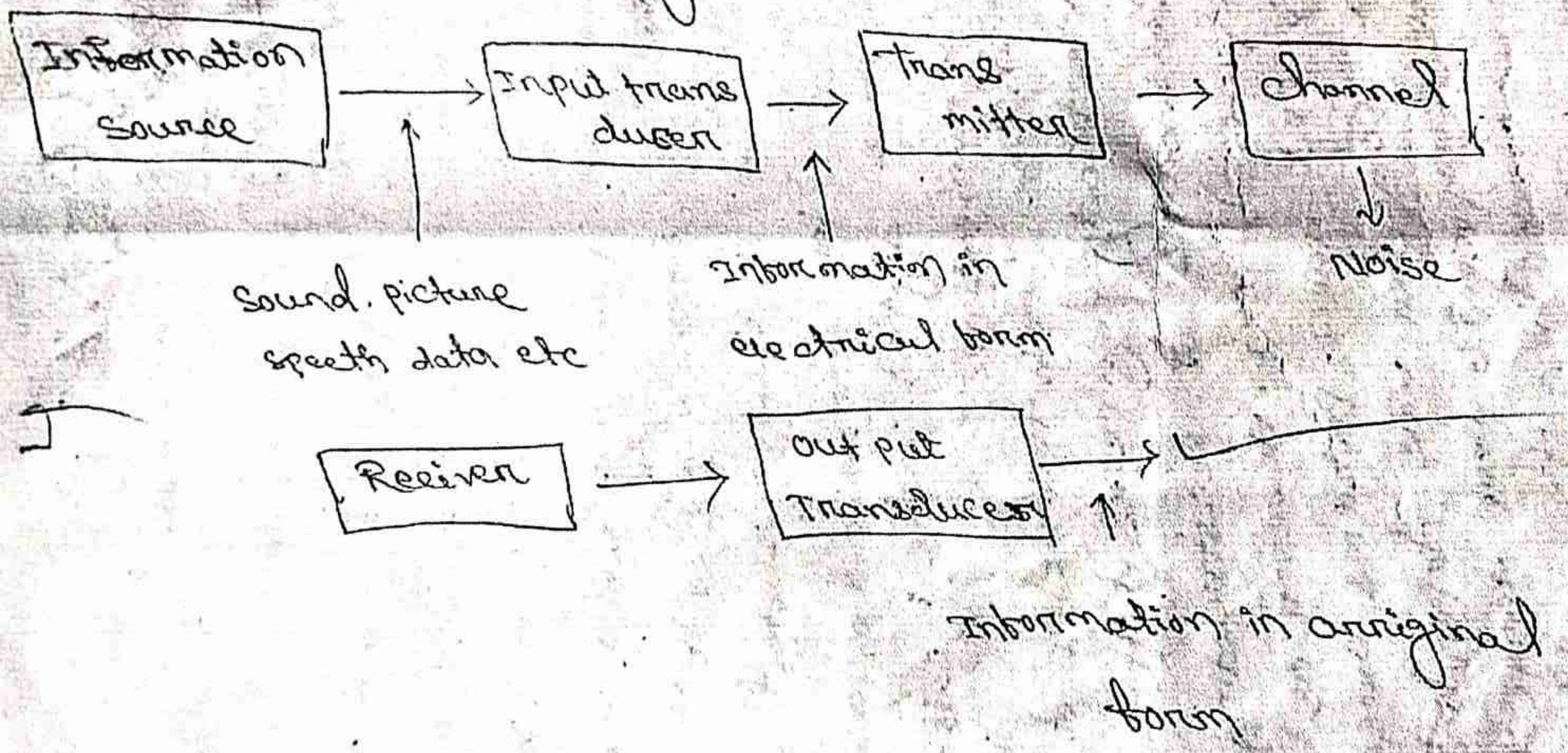
Digital signal:-

A signal which is discrete in nature or which is non contenous in form can be termed as a Digital signal. This signal has Individ values denoted separately which are not based on the previous value as it they are derived at they particular insted of time

The communication based on digital signals digital value i called as digital communication.

with the small transistor of their hands electrical Engineers of the 1950 saw the Possibilites of constructing ber more advanced ckt. However as the complexity of ckt grew. problems arose. one ckt like a computer was dependint on ckt Ib the components were large the cuires Interconnecting then most being the electric signals took time to go through the ckt thus slowing the computer.

block. diagram of communication system



Dt:- 01/04/2020

Information Source :- 5

As we know in know a communication system serves to communicat a message or Information

> This Information originited in the information source

> In general their can be varyous message in the form of words group of words code, symbles, sound, Signal etc.

> However out of this message only the dizied message is secured in communicated thin for We can say that the function of Information Source is to produced reqgired message which has to be transmeted.

Input transducer :-

> A transducer is the divice which converts one form of energy into another form.

> The message form the Information Source may or may not be electrical & in nature In a case when the message produce by the Information source not electrical the Information naiture and I/p transducer is used to convert it into a time-varying electrical signal.

For Ex:- In case of radio broadcasting a micro phone converts the Information or Message which is in the form of sound waves

Into cor-responding electrical signal.

- Transmitter:-

The function of the transmitter is to process the electrical signal from different aspects for ex:- In radio broadcasting the electrical signal obtained from sound signal, it processed to restrict it's range of audio frequencies. All these processing of the message signal are done just to ease the transmission of the signal through channel.

* The Channel :-

The term channel means the medium through which the message travels from the transmitter to the receiver. During the process of transmission & reception the signal gets distorted due to noise introduced in the system.

Reciver:-

The main function of the receiver is to reproduce the message signal in electrical form the distorted received signal. The reproduction of the original signal is accomplished by a process is know as the demodulation.

Destination :~

Destination is the final stage which is used to convert an electrical message signal into its original form

For ex:-
In radio broadcasting the destination is a loudspeaker which works as a transducer i.e convert the electrical signal in the form of original sound signal

Communication Channels :- DT = 05/09/2020

> A communication channel refers either to a physical transmission medium such as wire or to a logical connection over a multiplexed medium such as a radio channel in the communication & computer networking

> A channel is used to convey an information signal; for ex :- A digital bit stream, from one or several senders to one or several receivers.

> A channel has a certain capacity for transmitting information, often measured by its bandwidth in HZ or its data rate in bits per second

> communicating data from one location to another requires some form of pathway or medium. These pathways called communication channels. use two types of media: cable (twisted - pair wire, cable & Fibre optic cable)

Broadcasting microwave, statellite, radio, &
Inform.
> cable or wire line media use physical wire cable
to transmit data & Information
> Twisted - pair wire & co-axial cables are made
of copper & fiber-optic cables made of glass

Modulation:- Defination
modulation is the process of changing the para
metter of the carrier signal, in accordance
with the Instantaneous values of the
modulating signal.

Need for modulation:-
Base band signals are Incompatiable for
direct transmission. for such a signal to
travel Longer distances. its strength has
to be Increased by modulating with a high
frequency carrier wave. which doesn't
affect the parameters of the modulating
signal.

There are 3 basic types of modulation:
> Amplitude modulation
> Frequency modulation
> Phase modulation

Amplitude modulation

It's a type of modulation where the amplitude of the carrier signal is modulated in proportion to the message signal while the freq & phase are kept const.

✓ Frequency modulation

It's a type of modulation where the freq. of the carrire signal is modulated in proparction to the message signal whire the amplitude & phase are kept const.

Phase modulation

It's a type a modulation where the phase of the carrier signal is varied accordance to the law frequency of the message is knows as phase modulation

Advantages of modulation:-

- Reduction of antenna size
- no signal mixing
- Increased communication
- Mutiprexing of signals
- possiblity of bandwidth adjustments
- Improved reception Quality

* What is Signal:-

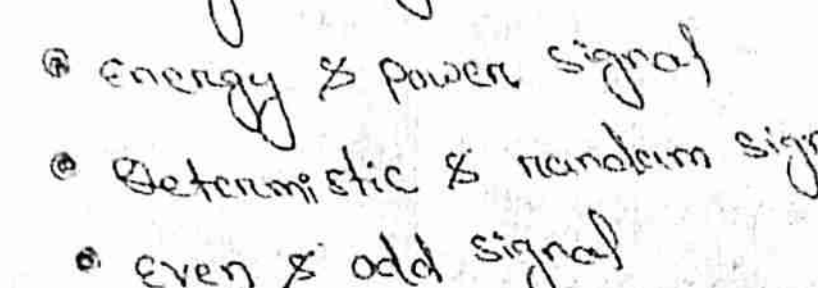
Signal is a function that conveys information about a phenomenon. It refers to any time varying voltage, carrier or electromagnetic wave that carries information. Ex:- Image signal

Signal classifications :-

Signal can be classified according to many crit ria.
- Analog & Digital signal
- Energy & power signal
- Deterministic & random signal
- Even & odd signal

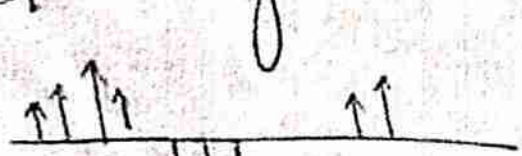
Contineous Time & Discrete Time signals

A signal said to be continious when it is defined for all instant of time



II Discrete time:-

A signal is said to be discrete when it is defined at only discrete instants of time



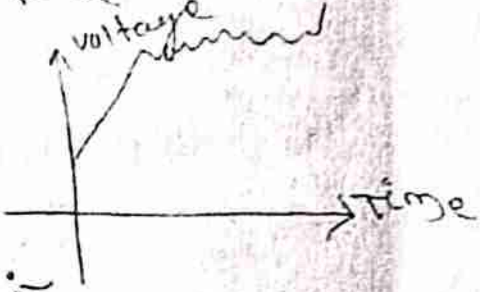
discrete time

Deterministic & Non-deterministic signal :-

A signal is said to be deterministic if there is no uncertainty with respect to its value at any instant of time

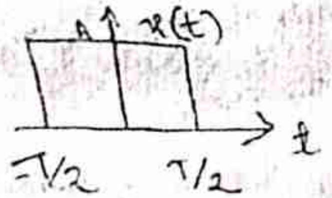
Random signal

A signal is said to be Random signal if there is uncertainty with respect to it's value at some instant of time



Even & odd signal :-

A signal is said to be even when it satisfies the condition $x(t) = x(-t)$



odd signal :-

signal is said to be odd when is satisfies the condution $x(t) = -x(-t)$

Periodic & Aperiodic signal :-

A signal is said to be periodic if it satisfies the condition $x(t) = x(t+T)$ or $x(n) = x(n+N)$

where

T = fundamental time period

$1/T = F$ = Fundamental frequency

The above signal will repeat for every time interval T_0 hence it is periodic with period T_0

Energy & Power signals :-

⇒ A signal is said to be energy signal when it has Finite energy.

⇒ A signal is said to be power signal when it has finite power.

Note: A signal can't be both energy & power simultaneously. Also a signal may be neither energy nor power signal.

Power of energy signal = 0

Energy of power signal = ∞

Real & Imaginary signals :-

⇒ A signal is said to be real when it satisfies the condition $x(t) = x^*(t)$

⇒ A signal is said to be odd when it satisfies the condition $x(t) = -x^*(t)$

ex:- If $x(t) = 3$ then $x^*(t) = 3^* = 3$ here $x(t)$ is a real signal

If $x(t) = 3j$ then $x^*(t) = 3j^* = -3j = -x(t)$

here $x(t)$ is odd signal

Note: for a real signal, Imaginary part should be zero. similarly for an imaginary signal, real part sould be zero

Bandwidth Limitation :-

Bandwidth refers to how much digital information we can send or receive across a connection in a certain amount of time some times its called data transfer rate.

Bandwidth Limits & Issues :-

The term bandwidth limit exceeded means that the amout of bandwidth that was allocated to the hosting plan has been reached Band width is consumed when dat is retrived from the sever & delivered to the end user (out bound traffic) well as when the End user uploads data to the web server (Inbound traffic) All data sent from the site to the client & vice versa gets accounted for and bandwidth usage.

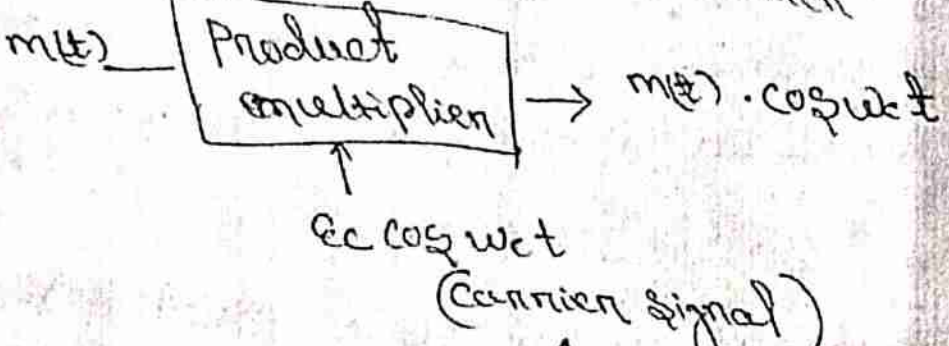
If the amount of bandwidth used exceeds the amount of bandwidth allocated by the web host. The browser will return a bandwidth limit exceed error.

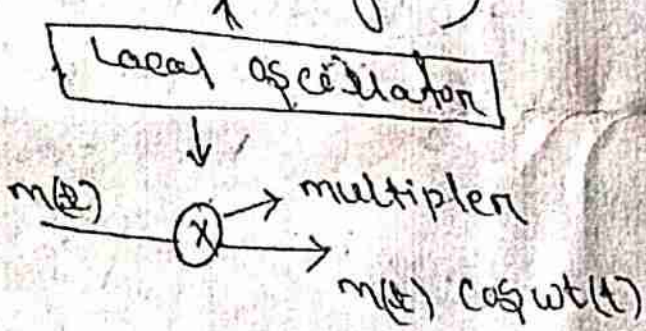
ch = 2nd

Q/ why AM is not efficient modulation:- $\alpha l = 1/10/20$

Ans:- As we know that the max^m efficiency $\eta = 33\%$ for $(m=1)$ which is practically very less values. This means 33% of the transmitted power is used for carry the message & rest 67% of the total power used by the carrier is not usable.

Generation of DSBSC (Double side band suppressed carrier)





m (t) → modulating signal Em cos Wmt

Ec cos wct → High frequency carrier generate from local oscilator & the Product modulator function is to multiply the signals. so it can be replaced by a multiplier

So, now, $m(t)$ & $\cos\omega_c t$ are the two inputs which are multiplyed at the multiplyer.

So, the multiplyer o/p is

$$m(t)\, E_c \cos \omega_c t \quad \text{(i)}$$

$$E_m E_c (\cos \omega_m t \cdot \cos \omega_c t) \quad \text{(ii)}$$

$$= \frac{E_m E_c}{2}\left[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t\right] \quad \text{(iii)}$$

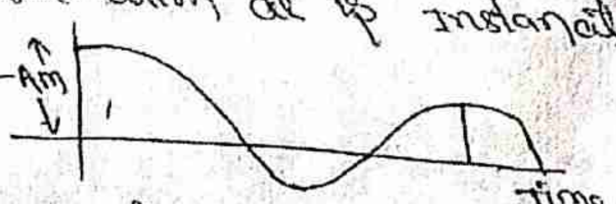
The eqn (3) expressed DSB = SC

As there are two side bands

⇒ $(\omega_c+\omega_m)$ & $(\omega_c-\omega_m)$ so it is double side band supressed carrier.

Amplitude (linear) modulation system:-

→ The amplitude of the carrier signal varrying in accordance with the instantaneous amplitude of the modulation signal which means the amplitude of the carrier signal contening know information varries as for the amplitude of the signal contening information at is instanat.



→ The 1st fig shows the modulating which is the message signal → the next one is the carrier wave which is a high frequency signal & constrise no information.
- while the last one is the resultant and modulation wave.

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Derivation of Amplitude modulation:—

> modulation index is also know as modulation is defined for the carrier wave to describe the modulated variable of carrier signal varing with respected to its unmodulated level is respectivily

$$M = \frac{A_m}{A_c}$$

> Consider maximum & minimum amplitude of the waves as amplitud and A_{max} & A_{min}.
> depending open $\cos(2\pi f_m t)$ following to equation are derived with maximum & minimum amplitude of the modulated waves

$$A_{max} = A_c + A_m$$

$$A_{min} = A_c - A_m$$

$$A_{max} + A_{min} = A_c + \cancel{A_m} + A_c + \cancel{A_m}$$

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

② $A_{max} = A_c + A_m$

$$A_{min} = A_c - A_m$$

$$A_{max} + A_{min} = (A_c + \cancel{A_m}) + (A_c - \cancel{A_m})$$

$$A_{max} + A_{min} = 2A_c \quad \text{------ (i)}$$

$$A_c = \frac{A_{max} + A_{min}}{2}$$

$$A_{max} - A_{min} = (\cancel{A_c} + A_m) - (\cancel{A_c} - A_m)$$

$$A_{max} - A_{min} = 2A_m \quad \text{---- (ii)}$$

$$\mu = \frac{\frac{A_{max} + A_{min}}{\cancel{2}}}{\frac{A_{max} - A_{min}}{\cancel{2}}}$$

$$\mu = \frac{A_m}{A_c}$$

The Power relation Amplitude modulation :-

> Consider the following eqn Amplitude modulation wave

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

> Power of Am wave is equal to the sum of power of Am carrier upper side band & lower side band frequency component

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

> we know that the standard formula for power of cos signal is

$$P = \frac{V^2_{rms} \text{(rating mean square)}}{R} = \left(V_m / \sqrt{2}\right)^2$$

> where

V_{rms} is the rms value of cos signal had

> 1st let us find the upper power carrier power $\frac{1}{1}$

Carrier power:-

$$P_c = \frac{\left(\frac{A_c}{\sqrt{2}}\right)}{R}$$

$$P_c \Rightarrow \frac{(A_c/\sqrt{2})^2}{R} \Rightarrow \frac{A_c^2/2}{R} \Rightarrow \frac{A_c^2}{2} \times \frac{1}{R} \Rightarrow \frac{A_c^2}{R}$$

Upper side band power:-

$$P_{USB} = \frac{\left(\frac{A_c\mu}{2\sqrt{2}}\right)^2}{R}$$

$$= \frac{A_c^2\mu^2/8}{R}$$

$$= \frac{A_c^2 m^2}{8R}$$

③ Similarly we will get the lower side band power that of the upper side band lower side band power.

$$P_{LSB} = \frac{A_c^2\mu^2}{8R}$$

Now let us add their power in order to get the power in order get power the AM wave.

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

$$\Rightarrow P_t = \frac{A_c^2}{2R} + \frac{A_c^2\mu^2}{8R} + \frac{A_c^2\mu^2}{8R}$$

$$P_t = \frac{A_c^2}{2R}\left(1 + \frac{\mu^2}{\cancel{4}\,2} + \frac{m^2}{\cancel{4}\,2}\right)$$

Dt = 01/09/2020

$P_t = P_c\left(\frac{1+\mu^2}{2}\right)$ 19

Band width of Amplitude modulation wave:~ Dt = 25/10/20
(frequency range)

→ Band width is the difrence the highest and lowest frequency of signal mathmaticaly we can write is as

$$\boxed{BW = f_{max} - f_{min}}$$

→ consider the following eqn of amplitude modulated wave.

$$\boxed{s(t)\ A_c\left[1+\mu\cos(2\pi f_m t)\right]\cos(2\pi f_c t)}$$

$\Rightarrow s(t)\ \left[A_c + A_c\ \mu\cos(2\pi f_m t)\right]\cos(2\pi f_c t)$

$\Rightarrow A_c\cos(2\pi f_c t) + A\mu\cos(2\pi f_m t)\cos(2\pi f_c t)$

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

Hence the amplitude modulated has three frequ- those are carrier frequency fc, opper side fc+fm & lower side band fc-fm.

Hence

$$\boxed{\begin{aligned} f_{max} &= f_c + f_m\ \& \\ f_{min} &= f_c - f_m \end{aligned}}$$

Substitute

Fmax & Fmin value in band with formula.

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$\Rightarrow BW = f_c + f_m - (f_c - f_m)$

$f_{max} - f_{min} = (f_c + f_m) - (f_c - f_m)$

$= \cancel{f_c} + f_m - \cancel{f_c} - f_m$

$= 2f_m$

Thus it can be said that the bandwidth reqquised for amplitude modulated wave is there the frequency of the modulating signal.

Frequency division multiplexer :- Dt=3/10/20

It is a techique in which several message signal combine into a composite signal for transmission over a common channel

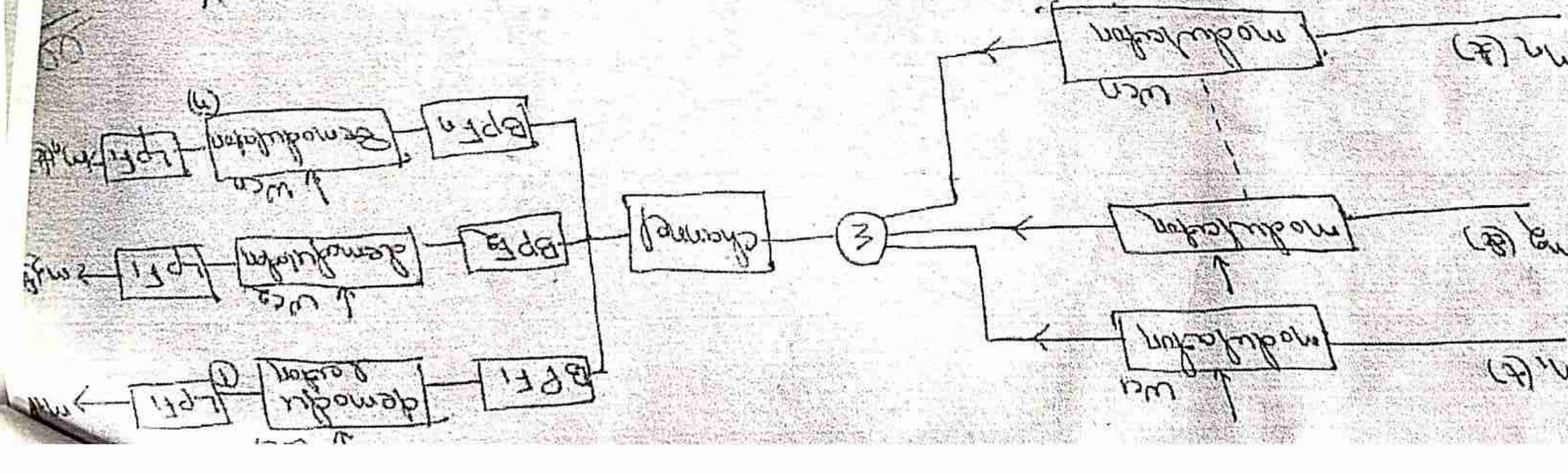
$\Rightarrow$ But to transmit those signal in a same channel the signal must be kept a path from each other Interms of thir frequency rangs. So that these signals do not Interfear among eachother

Ex $\Rightarrow$ Consider $m_1(t)$ is a message signal & these modulated by carrier frequency ω_{c1} in terms of KHz (Assming) 100 KHz). m_2 is another message signal & it modulated by a carrier freq ω_{c2} (300 KHz)

Like this the resultant modulation signal are added & Transmitted through a common channel & which will not be Interfearn. due to their is a

Tu - 01/09/2020

huge gap b/w to modulated signal



→ Then the modulated signal passes through band pass filter corresponding to the freq range of the modulating signal. So $m_1(t)$ modulating signal passes through BPF_1 & demodulated with the help of again ω_{c1} & passes through LPF_1. So the final result will be $m_1(t)$.

→ Similarly for $m_2(t)$ BPF_2, demodulator 2 so the final result will be $m_2(t)$.

→ FDM is basically used to in telephone system & cable TV

→ In case of AM broadcasting the carrier frequency is 10 KHz & in FM broadcasting the carrier freq is 200 KHz

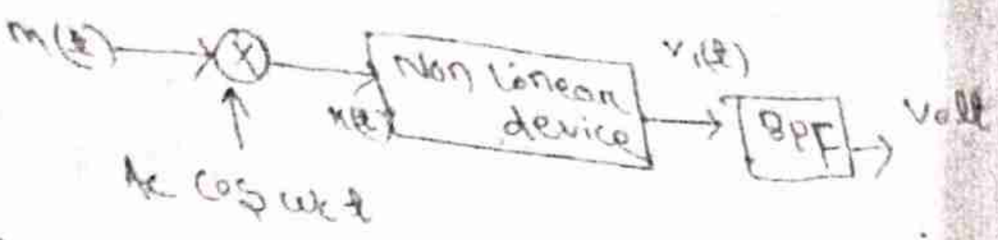
Generation of Am or DSB signal :-

> Two methods are there

(1) power law Am modulator

(2) switching Am modulator

* Power law Am modulator



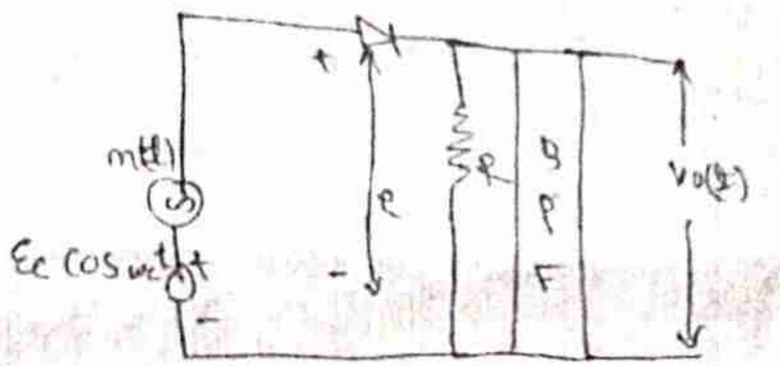
> output of the adder is

$m(t) + A_c \cos \omega_c t$

for non-linear device

$y = ax + bx^2$

$v_1(t) = a\left[m(t) + A_c \cos \omega_c t\right] + b\left[m(t) + A_c \cos \omega_c t\right]^2$



Now passing the signal to the BPF the analysis is shown below

> Here R is only multiplied as a type of constant

> In this diode is used as a non-linear device

from the above ckt

$e = \left[m(t) + E_c \cos \omega_c t\right]$ by applying KVL

--/09/2020

We know that

$i = ae + be^2$

$= a[m(t) + E_c \cos w_c t] + b[m(t) + E_c \cos w_c t]^2$

We know that

$V_0(t) = iR$

$= R[a(m(t) + E_c \cos w_c t) + b(m(t) + E_c \cos w_c t)^2]$

$= Ra[m(t) + E_c \cos w_c t] + bR[m(t) + E_c \cos w_c t]^2$

$= Ra\, m(t) + RaE_c \cos w_c t + bR\, m(t)^2 + bR\, E_c^2 \cos^2 w_c t$

$= Ra\, E_m \cos w_m t + Ra\, E_c \cos w_c t + bR\, E_m^2 \cos^2 w_c t$

→ When $V_0(t)$ is input to the BPF which has a center frequency w_c which means it will pass $(w_c + w_m)$ & $(w_c - w_m)$. So in both the upper end & the lower end the passes frequency should be high

(Since both the ends contain w_c which means high freq)

Note And the supressed part the $RE_m \cos w_m + bRE_m^2 \cos wt$ & these two signal contain low freq (w_m)

& finally the output is i.e $aRE_c \cos w_c t + 2bR \cos w_c t \cos w_m t$

$= aRE_c \cos w_c t + bR(\cos w_c t + \cos w_m t)$

$= \underbrace{K_1 E_c \cos w_c t}_{\text{carrier}} + \underbrace{K_2 \cos(w_c + w_m) t \; \cos(w_c - w_m) t}_{\text{DSB-SC signal}}$

$\because K_1 = aR$
$K_2 = bR$

→ which is AM (or carrier + DSB-SC signal)

SSB - SC : Dt = 28/9/2020 24

for SSB-SC single side Band & single carrier

BW = Fm (for single side Band)

→ when $m < 1$ $\frac{A_m}{A_c} < 1$

$\frac{E_m}{E_c} < 1$

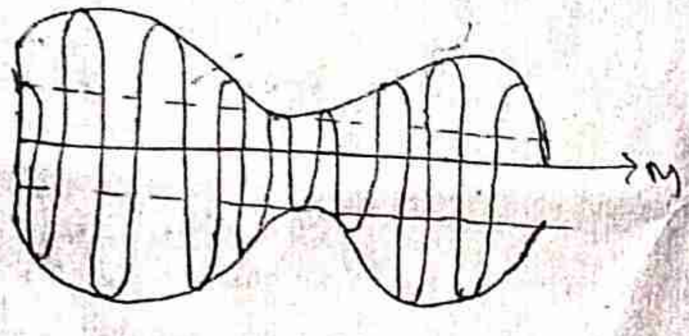
$A_m < A_c$

let $m = 0.5 < 1$

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

$$\frac{(0.5^2)}{2+(0.5)^2} \times 100$$

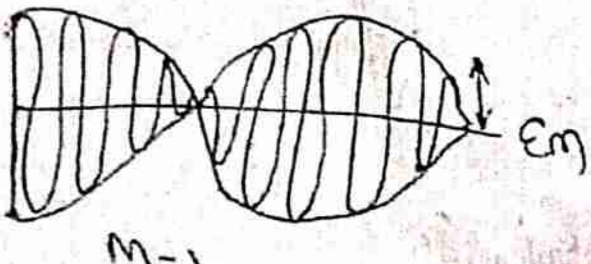
$= 11.11\%$



for $m=1$

$A_m = A_c$

$$\eta = \frac{m^2}{2+m^2} = \frac{1}{3} \times 100 = 33.33\%$$



$m=1$

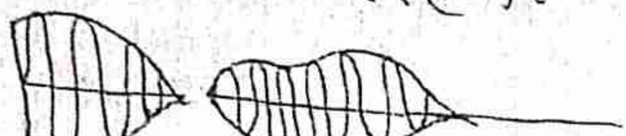
for ~~a~~ $M>1$

$= \frac{A_m}{A_c} > 1$

$= A_m > A_c$

* Here let $m=1.5$

$$\eta = \frac{m^2}{2+m^2} = \frac{(1.5)^2}{2+(1.5)^2} \times 100 = 57\%$$



> In This case $M>1$ some part of the Information are lost so the exact Information is not possible to get so modulation Index cant not be taken for $M>1$ & it should be.

dt-2/1/ SSB-SC (single side band supressed carrier signal)

> If carrier & one side band is supressed no Information is lost so it makes the advantage of Transmission as less band width.

Generation:-

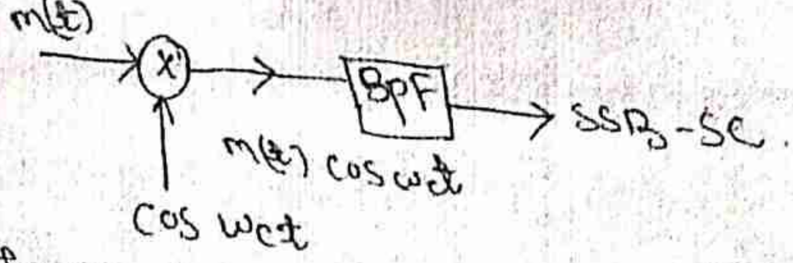
1- By filter method

2- By phase shift method

1- By filter method :-

> It is most commonly used

> in this method a the DSB-SC signal is passed through a sharp cut-off filter to eleminate any one side band



⇒ The o/p of the multiplier is $m(t) \cdot \cos w_c t$

$= E_m \cos w_m(t) \cdot \cos w_c(t)$

$= \frac{E_m}{2}\left[\cos(w_c + w_m)t + \cos(w_c - w_m)t\right]$

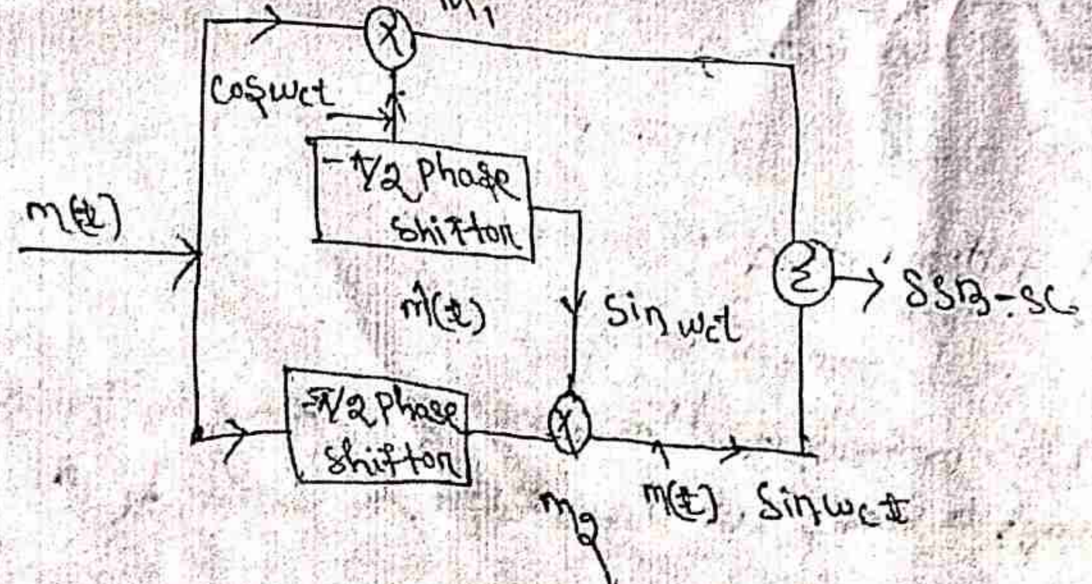
$= \frac{E_m}{2}\cos(w_c + w_m)t + \frac{E_m}{2}(\cos w_c - w_m)t$

⇒ Here the band pass filter is a special filter frequency (So it is called sharp cut off BPF)

→ If the Band pass filter has a cut-off freq $(w_c + w_m)$ then USB (Upper side Band) will be passed i.e $\frac{E_m}{2}\cos(w_c + w_m)t$ will be passed.

→ But it is very difficult to pass completely by a signal

2. <u>Phase shift method:</u>~



→ for m_1 multiplier the I/p signals are $m(t)$ & $\cos w_ct$ and the o/p is $m(t)\cos w_ct$.

→ For m_2 multiplier the I/p signals are $\sin w_ct$ & $\hat{m}(t)$

$\cos w_c(t)$ is phase shifted by $-\pi/2$

which means $\cos w_ct \cdot \pi/2 = \sin w_ct$

But Here the $m(t) = E_m \cos w_mt$ after $\pi/2$ Phase shifter it becomes $\hat{m}t$ i.e $E_m \sin w_mt$. so the final o/p for $m_2 = \hat{m}t \cdot \sin w_ct$

$= E_m \sin w_mt \cdot \sin w_ct$

→ Finally o/p of m_1 & m_2 is

$m(t) \cdot \cos w_ct \pm \hat{m}(t) \cdot \sin w_ct$

$= E_m \cos w_ct \cdot \cos w_ct \pm E_m \sin w_mt \cdot \sin w_ct$

We know that :-

$\cos(A \mp B)$

$= \cos A \cdot \cos B \pm \sin A \cdot \sin B$

$= E_m \cos(w_m(t) \mp w_c(t))$

$$\boxed{\begin{array}{l} \text{It } E_m \cos(w_ct - w_mt) \rightarrow \text{lower side band which is a SSB-SC signal} \\ E_m \cos(w_ct + w_mt) \rightarrow \text{upper sid band} \end{array}}$$

→ I/p to the multiplier are $m(t)\cos w_ct \pm \hat{m}(t) \sin w_ct \cdot \cos w_ct$. so the o/p of the multiplier is

$m(t)\cos^2 w_ct + \hat{m}(t) \sin w_ct \cdot \cos w_ct$

$\Rightarrow \frac{m(t)}{2}(\cos 2w_ct) + \frac{\hat{m}t}{2}(\sin 2w_ct)$

$= \frac{E_m \cos\omega_m t}{2}\left(\cos 2\omega_c t + \frac{E_m \sin\omega_m t}{2} \sin 2\omega_c t\right)$

$= \frac{E_m}{2}\left(\cos^2 \omega_m t \cdot \cos 2\omega_c t + \sin\omega_m t \ \sin 2\omega_c t\right)$

when this is passed through low pass filter it has cut-off freq W_m so only $\cos W_m t$ will be passed & rest of the signal are supressed by the low pass filter.

VSB-SC (vestigeal side band supressed carrier.)

> when a signal contain high frequency component the USB & LSB tends to need the carrier frequency under such condition is is very difficult to isolate one side band from the other so to over come this difficulty VSB technique is used. which in state of a rejecting on side band completely, gradual cutoff of one side band is allowed, this cut off is compensated by vestige / or some portion of the other side band.

Angle modulation:

> Angle modulation comprises of phase modulation & frequency modulation. & this refers to the process by which the phase angle of the carrier signal is varies in accordance with the modulation signal or message signal

Angle modulation:- The process of changing phase modulation means both the phase & frequency modulation of the carrier signal is varied in accordance to the Instantaneous value of modulation signal. keeping the amplitude constant is a Angle modulation.

Let us consider a unmodulated carrier wave.

$$\phi(t) = E_c \cos(\omega_c + \phi_0)$$

$$= E_c \cos \phi t$$

$$\Rightarrow \phi(t) = \omega_c t + \phi_0$$

Now differenciating with respect

$$\frac{d\,\phi(t)}{dt} = \omega_c$$

$\omega_c \Rightarrow$ Here is the frequency which is the dependent.

Now time dependent angular velocity.

is called Instantaneous angular frequency.

$$\omega_i(t) = \frac{d\,\phi(t)}{dt}$$

$$\int \omega_i(t)\, dt = \phi(t)$$

 1/11/20

Phase modulation :-

The process of changing the Instantaneous phase Linearly with base band signal about an unmodulated phase $\omega_c(t)$ keeping the Amplitude & frequency constant.

Again let us conside a carrier signal

$E_c \cos \omega_c t$ we know that

$$\varphi(t) = (\omega_c t + \varphi_0) + K_p\, m(t) \quad \text{---- (i)}$$

Assming the angle of φ_0 at particular Instant

$$\varphi_0 = 0 \text{ so}$$

$$\varphi(t) = \omega_c t + K_p\, m(t)$$

Here k_p represents the phase sensitivity constant.

So the expression for phase modulation

$$\boxed{e_{pm}(t) = E_c \cos \varphi(t)}$$

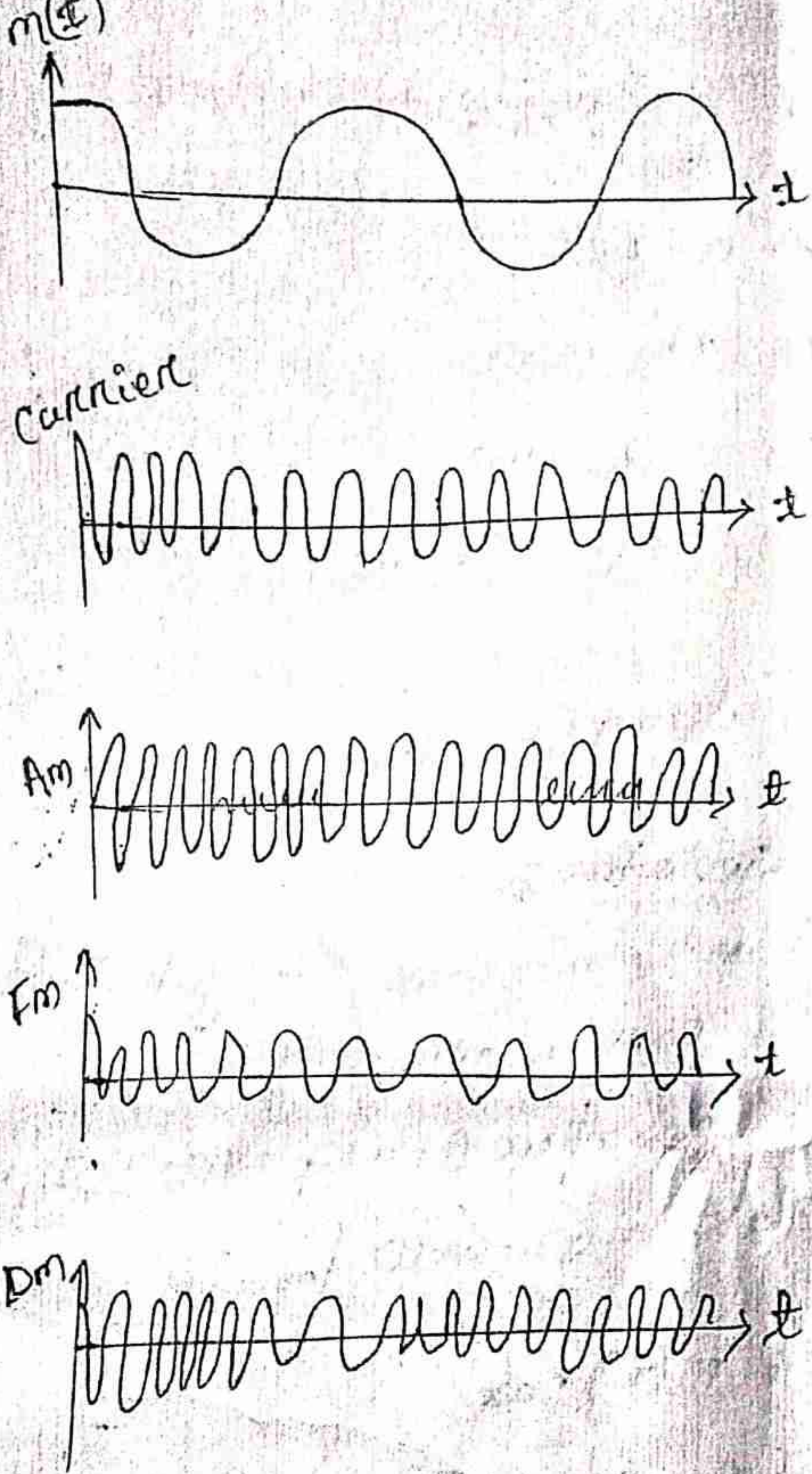
$$= E_c \cos\left[\omega_c(t) + K_p\, m(t)\right] \quad \text{---- (ii)}$$

Frequency Modulation :-

→ it is defined of the Instantneous respect to base band signal un modulated wt carrier frequency ω_c

The frequency of the carrier anlaysis means some times compression takes place in the +ve cycle of $m(t)$ & some time expansion takes place of the changes According to the modulating.

Diagram



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The instantaneous angular freq in freq modulation is $\omega_i(t) = \omega_c t + k_f\, m(t)$

$k_F \Rightarrow$ the frequency sensitivity constant

but we know that

$$\psi_i(t) = \int \omega_i(t)\, dt$$

$$= \int \omega_c(t) + k_f\, m(t)$$

$$= \omega_c(t) + \int k_F\, m(t)$$

$$= E_c \cos \omega_c(t) + \int k_f\, m(t)$$

$+ e_f\, m(t)$

$$E_c \cos\left[\omega_c(t) + \int k_F\, m(t)\right]$$

frequency deviation:~

It is defined as the maximum change in instantaneous C_1 carrier frequency

$$\Delta\omega = |\omega_i - \omega_c|_{max} = |\omega_c(t) + k_F\, m(t) - \omega_c(t)|_{max}$$

$$= |\omega_c(t) + k_F\, m(t) - \omega_c(t)|_{max}$$

$$\Delta\omega_1 = k_F\, m(t)_{max}$$

$$= k_f\, E_m \cos \omega_m t_{max}$$

So to have max^m value take $\cos \omega_m t = k_F E_m 1$

So it will be

$$\boxed{\Delta\omega = k_f \cdot E_m}$$

Modulating Index :-

> It is the ratio of frequency deviation to the frequency of modulation signal.

> It is donoted by $\boxed{\beta = \frac{\Delta\omega}{\omega m}}$

> β is always greater than standard effect.

Q/

* $\varphi i(t) = \omega c(t) + K_F \int m(t)\, dt$
* $\varphi i t = \omega c(t) + K_F \int Em \cos \omega m(t)\, dt$
* $\omega c(t) + K_F Em \int \cos \omega m(t)\, dt$
* $\omega c(t) + K_F \sin \omega m(t)$
* $\omega c(t) + K_F \frac{Em}{\omega m} \sin \omega m t$
* $\omega c(t)\ \beta \sin \omega m t$

> Since β = the modulating Index of Fm

$$e_{Fm}(t) = E \cos\left[\omega c t + \beta \sin \omega m t\right]$$

types of frequency modulation:-

① wide Band frequency modulation (WBFm)

② Narrow Band frequency modulation (NBFm)

WBFm

when the value of modulating Index the large number of side band are produce hence the Bandwidth produce.

IF β >1

* ΔF >1

$$\therefore \beta = 2\left(F_m + \Delta F\right)$$

$$\Rightarrow 2\Delta F$$

$\boxed{F_m = 0}$ wide BFm

$\boxed{\Delta F = 0}$ NBFm

$$\frac{\Delta F}{F_m} > 1 \quad \beta > 1$$

NBFm:-

When the value of modulation Index is small hence it is called frequency modulation Then the band width is

$$\Rightarrow \frac{\Delta F}{F_m} > 1$$

$$\therefore BW = 2\left(F_m + \Delta F\right)$$

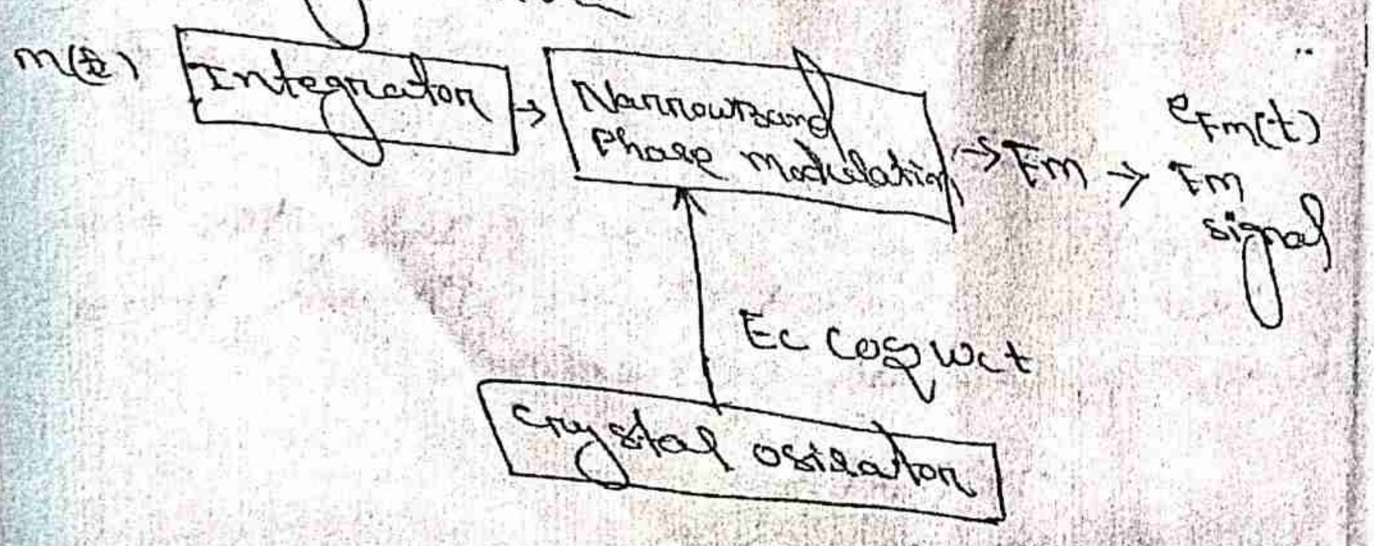
$$= > 2F_m$$

Due to small BW deviation in NBFm is less.

Generation of frequency modulation:-

1. Armstrong method
2. parameter variation method

① Armstrong method:-



(33)

this is an indirect method of generation of wide Band Fm is was propanse by armstong.
but in this method we have to generate NBFm first.
Here the i/p is m(t) & is passed through Integration which massage signal at ...

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Frequency modulation & Amplitude modulation & Advantage & Dis Advantage:~

Advantage of Fm over Am:-

The Amplitude of an fm wave remains cons over time this allows the encoders & decoders the freedom to remove the noise from the received signal This is done with the assistance of a filter that Removes signal of wave length greater than that of the transmitted signal there by removing the noise Immune systems, since the Am signal transferred Information of the wave form cant be altered

> In Am systems the power consumption for signal transmission is Higher when compared to Fm systems In Am systems the power depends on the modulation Index also called "mn" when mn reaches unity the power consumation is 100%

in Fm system the power of the transmitted signal is proportion to the Amplitude of the unmodulated Carrier signal & it is constant there fore fm is usually more power efficient than Am system.

> In fm systems the frequency deviation of the signal is related to the Noise ratio A higher freq deviation means that the base band signal Can be easily retrived from the fm signals.

where as less deviation means it is harder to separate the signal from the noise. In Am systems the only method of reducing noise in the transmission is the increase in the transmitted power of the signal which increase the cost of operation of the Am system.

→ In an Am system there is no guard band in betn two adjacent channels this seriously increases the occurrence of interference of Am radio station unless one signal is strong enough to very power the other where as in the case of Fm signal. The adjacent fm channel are separated by guard bands which result in very little inter-ference betn adjacent fm channels.

Dis Advantages of Fm over Am:-

→ The equipment needed for fm & Am system is different. The equipment cost to an fm channel is more complex & involves complicated circuiting As a result fm systems are costlier than Am systems.

→ Fm system works using a line of sight propagation ~~Am systems~~ where as Am systems use sky wave propagation consequently the receiving area of an fm system is much smaller than that of Am & fm antenna for Fm systems need to be close by where as Am systems can communication with other system through the world by reflected signals off the Ionosphere.

→ In an fm system there is infinite no of side bands resulting in a theoretical band width on fm signal being infinite this band width is limited by Carson's rule but it is still much larger than this of an Am system. In an Am system the bandwidth is only twice the modulation frequency this is another reason why fm system are costly than Am systems.

4(v):- NBFM Armstrong

Then the o/p of the integration of $\int m(t)$

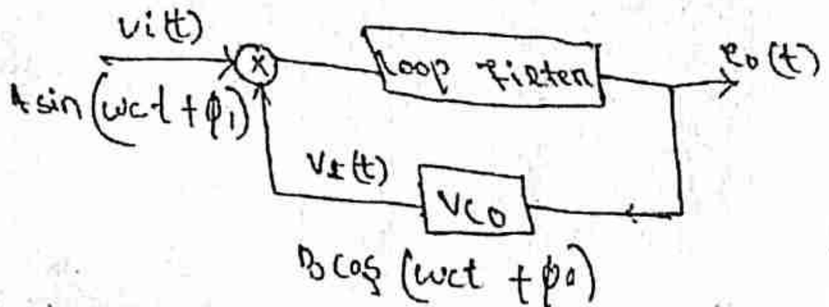
* ⇒ The $\int m(t)$ is the I/P to the phase modulation. Its funⁿ is to phase modulation the signal $\int m(t)$ with the help of crystal oscillator whose o/p is $E_c \cos \omega t$ so the final o/p of NBPM $E_c \cos(\omega_{c1} t + K_f \int m(t)\, dt)$
* ⇒ This can also be written as $E_c \cos(\omega_{c1} t + \beta_1 \sin \omega_m t)$
⇒ This signal is called NBFM signal and here the modulation index is β_1 which is very low.
* ⇒ To get WBFM on standard which so the signal is passed through the frequency multiplier. The funⁿ of frequency multiplier used to increase the carrier freq as well as the frequency deviation.
* ⇒ so it frequency modulation is of order 2 which means $[\times 2]$ is used as the carrier frequency deviation will be $2\Delta\omega_1$
* ⇒ By taking nth order frequency multiplier $n\omega_{c1}$ so the final o/p will be $E_c \cos(n\omega_{c1} t + n\beta_1 \sin \omega_m t)$

Demodulation of Fm :-

There is the 3 methods demodulation of Fm

1) PLL method (Phase locked)
2) Limiter discriminator method
3) Slope detector method

PLL method :-



The Incoming signal is fm signal which is

$$A \sin [w_c t + \phi_i]$$

$$\phi_i = k \int m(t)\, dt$$

→ PLL cosist 3 main components :-

① multipier
② loop filter
③ voltage contral oscillator

① Multiplier :-

Its function used to multiply Incoming signal $A \sin(w_c t + \phi)$ with the feedback signal $B \cos[w_c t + \phi]$ which is the o/p of VCO

→ it is also used a phase comparator

② Loop filter :-

It is a Narrow wind LPF which supress the high frequency component.

③ voltage contral oscillator :-

It is a sine wave generator whose frequency is determined by the applied voltage to it from the external source. $e_o(t)$ $w_i(t) = w_c + k_v e_o(t)$

where ω_i = o/p freq. of Vco when applied $V_o(t)$ voltage.

ω_c = it is the Initial frequency when no voltage is applied.

⇒ One Important property of the o/p of vco is 90° phase shift of I/p to it.

Description:-

→ Initially $m(t) = 0$ & $e_o(t) = 0$ at the time vco has been adjusted such that the frequency of Vco is set at carrier frequency ω_c.

→ Vco o/p i.e $V_f(t)$ is phase quadrature of carrier - wave.

→ Now the Incoming I/p signal

$$A\sin(\omega_c t + \phi_0) = A\sin\left[\omega_c t + k\int m(t)\,dt\right]$$

which means FM signal and the o/p of Vco is

$B\cos\left[\omega_c t + \phi_0(t)\right]$ so the o/p of amplifier is

$$A\sin\left[\omega_c t + k\int m(t)\right] \cdot B\cos\left[\omega_c t + t\phi_0\right] \Rightarrow a\sin x\cdot\cos y$$

$$\Rightarrow \frac{AB}{2}\sin\left[2\omega_c t + \phi_i + \phi_0\right] + \sin\left[\phi_i - \phi_0\right]$$

$$\frac{\sin x+y}{2} + \frac{\sin(x-y)}{2} = \sin x\cdot\cos y$$

$$\Rightarrow \sin(x+y) + \sin(x-y)$$

$$\Rightarrow 2\sin x\cdot\cos y$$

After passing the o/p of multiplier through which supresses high freq. component so the o/p of the filter is $\frac{AB}{2}\left[\phi_i - \phi_0\right]$

→ Let $(\phi_i - \phi_o) = \phi_e$

Where ϕ_e = Phase error & this should be very less

→ here we consider

$$\phi_e = 0$$

$\Rightarrow 0 = \phi_i - \phi_o$

$$\phi_i = \phi_o$$

$\Rightarrow \phi_i(t) = \phi_o(t)$

We know that $\boxed{\phi_o(t) = k_f \int m(t)\, dt}$

Therefore $A\sin[\omega_c(t) + \phi_i(t)] \cdot B\cos[\omega_c(t) + \phi_o(t)]$

→ The o/p of VCO means the o/p frequency

i.e $\boxed{\omega_i(t) = \omega_c + k_c e_o(t)}$ --- ①

Which is also called as the instantaneous freq. of FM

→ Therefore the o/p of VCO is

$$B\cos(\omega_c t + \phi_o t)$$

Let $\psi(t) = \omega_c t + \phi_o(t)$

$$\frac{d\phi(t)}{dt} = \omega_c + \frac{d\phi_o(t)}{dt}$$

$$\boxed{\omega_i(t) = \omega_c \frac{d\phi_o(t)}{dt}} \quad \text{---} ②$$

→ These two eqn ① & ② are both o/p s for VCO

$$\omega_c + k_1 e_o(t) = \omega_c + \frac{d\phi_o(t)}{dt}$$

Here, $k_e e_o(t) = \dfrac{d\phi_o(t)}{dt}$

$$k_1 e_o(t) = \frac{d k_f \int m(t)\, dt}{dt}$$

$K_1 e_0(t) = K_F\, m(t)$

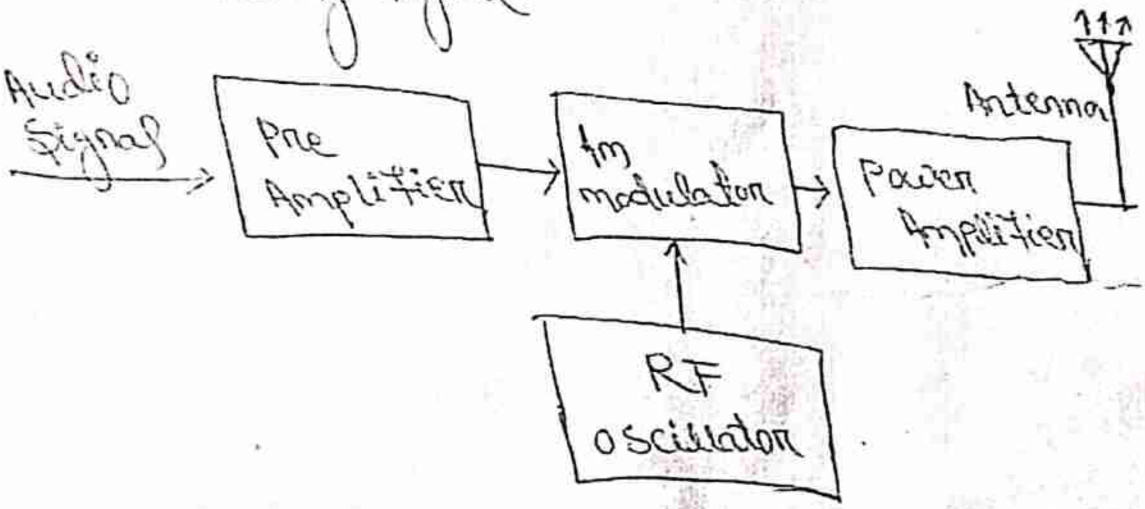
M1

Chapter = 3

AM Transmitter :-

AM transmitter takes the audio signal as an I/P and delivers amplitude modulated wave to the antenna as an O/P to be transmitted.

The block diagram of AM transmitter is shown in the following figure



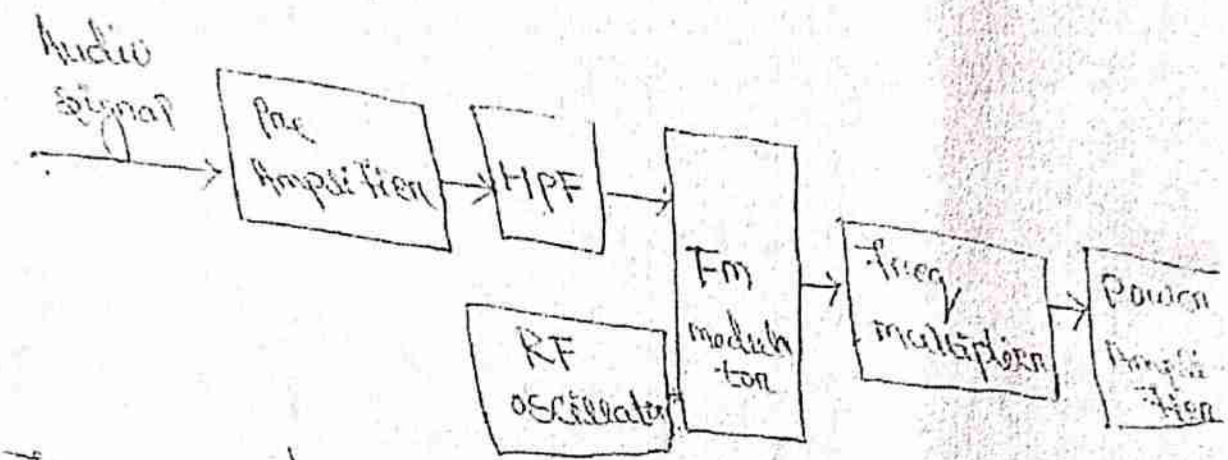
The working of AM transmitter can be explained as follows:

- The audio signal from the O/P of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal
- The RF oscillator generates the carrier signal
- Both the modulating & the carrier signal is sent to AM modulator.
- Power amplifier is used to increase the power levels of AM wave. This wave is finally passed to the antenna to be transmitted.

FM Transmitter

FM Transmitter is the whole unit which takes the audio signal as an I/P and delivers FM wave to the antenna as an O/P to be transmitted.

* The block diagram of FM transmitter is shown in the following figure.



The working of FM transmitter can be explained as follows.

- The audio signal from the O/P of the microphone is sent to the pre-amplifier which boosts the level of the modulating signal.
- This signal is then passed to high pass filter which acts as a pre-emphasis network to filter out the noise & improve the signal to noise ratio.
- This signal is further passed to the FM modulator circuit.
- The oscillator circuit generates a high freq carrier, which is sent to the modulator along with the modulating signal.
- Several stages of frequency multiplier are used to increase the operating freq.

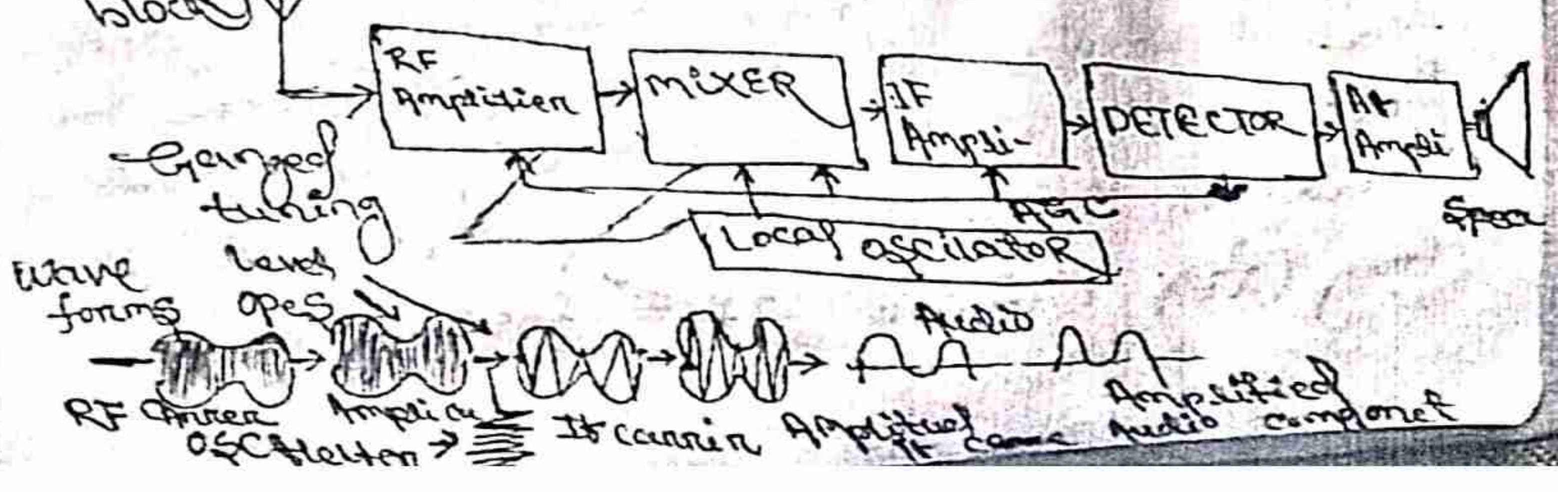
Even then the power of the signal is not enough to transmit. Hence a RF power amplifier is used at the end to increase the power of the modulated signal. This FM modulated o/p is finally passed to the antenna to be transmitted.

Superheterodyne FM

Superheterodyne is basically a process of designing and constructing wireless communications such as radio receivers by mixing two frequencies together in order to produce a diffn frequency component called as intermediate freqy (IF) so as to reduce signal frequency prior to processing.

A superheterodyne receiver usually consists of an antenna RF amplifier, mixer, local oscillator, IF amplifier, detector, AF amplifier and a speaker.

* The working of a superheterodyne receiver is explained with the help of the block diagram given below in fig 1 Along with the wave form at the o/p of each block.



VVI

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Fig-1. Superheterodyne receiver

- In the superheterodyne receiver, the Incoming Signal through the antenna is filtered to reject the Image frequency and then amplified by the RF amplifier.
- RF amplifier can be tuned to select and amplify a particular carrier freq within the AM broadcast range only the select frequency and it two sidebands are allowed to pass through the amplifier
- The carrier of the received signal is called rad frequency carrier and its frequency is radio freq ~ f_{RF} and the local oscillator signal operates at f_{osc}. The amplified RF frequency is then mixed with the local oscillator freq.
- The combining of these two signals is done at the mixer which produces sum & difference freq signals of the Incoming carrier signal and local oscillator signal, which are

$$f_{osc} + f_{RF} \ \& \ f_{osc} - f_{RF}$$

- The sum frequency $(f_{osc} + f_{RF})$ is rejected by the filter & the remaining difference freq $(f_{osc} - f_{RF})$ signal which is a down converted frequency signal is called as Intermediate freq (IF) carrier $(f_{IF} = f_{osc} - f_{RF})$

This Idea of the superheterodyne receiver is to reduce the high frequency radio components of the Incoming carrier to a fairly low, fixed value such as to be processed at the different stages of the receiver, & also to provide good stability, gain & proper selectivity and fidelity.
কর performance

- The modulation of the IF carrier signal is same as that of the original carrier signal & it has a fixed frequency of $\boxed{455}$ KHz which is amplified by one or more stages of amplification.

- The IF signal is amplified with the help of IF amplifier which raises its level for the Information extraction process. Also the IF amplifier fulfills most of the gain and bandwidth requirements of the receiver.

- IF amplifier operations are independent to the frequency at which receiver is tuned, Maintaining the selectivity & sensitivity of the superheterodyne receiver considerably constant throughout the tuning range of the receiver.

- This amplified IF signal is applied to the detector to detect the Information signal component from 455 KHz IF, to reproduce the original Information data, which is generally in the form of audio signal.

- The detector stage eliminates one of the side bands which is still present & separates the RF from the audio components of the other

- The RF component is filtered out and audio is supplied to the audio stages for amplification.
- The generated audio signal is then applied to the AF amplifier in increase the audio freq level of the signal & to provide enough gain to drive the speaker or headphones.
- A speaker is connected to the AF amplifier to play the audio information signal.
- An important part of superheterodyne receiver is Automatic gain control (AGC) which is given to the RF, IF & mixer stages in order to generate constant O/P irrespective of the varying I/P signal.
- Superheterodyne radio receiver in spite of being more complicated than some of the other receiver offers many advantages in term of performance, most importantly the selectivity. it is more efficiently able to remove unwanted and distorting signals than other forms like TRF & regenerative receivers.
- Due to the enormous advantages provided by the superheterodyne receivers compared to the other radio receivers, they are widely used in all broadcast radio receivers, commercial radio as well as televisions operate on the basis of the superheterodyne principle.

4-1

Superheterodyne is basically a process of designing & constructing wireless communications such as radio receivers by mixing two frequency together in order to produce a diff frequency component called as intermediated frequency (IF), so as to reduce signal frequency prior to processing.

- A superheterodyne receiver usually consists of an antenna RF amplifier, mixer, local oscillator, IF amplifier, detector AF amplifier & a speaker.

- The working of a superheterodyne receiver is explained with the help of the block diagram given below in Fig 1 along with the wave forms at the o/p of each block.

BLOCK Diagram F.M. Receiver 23/4/20

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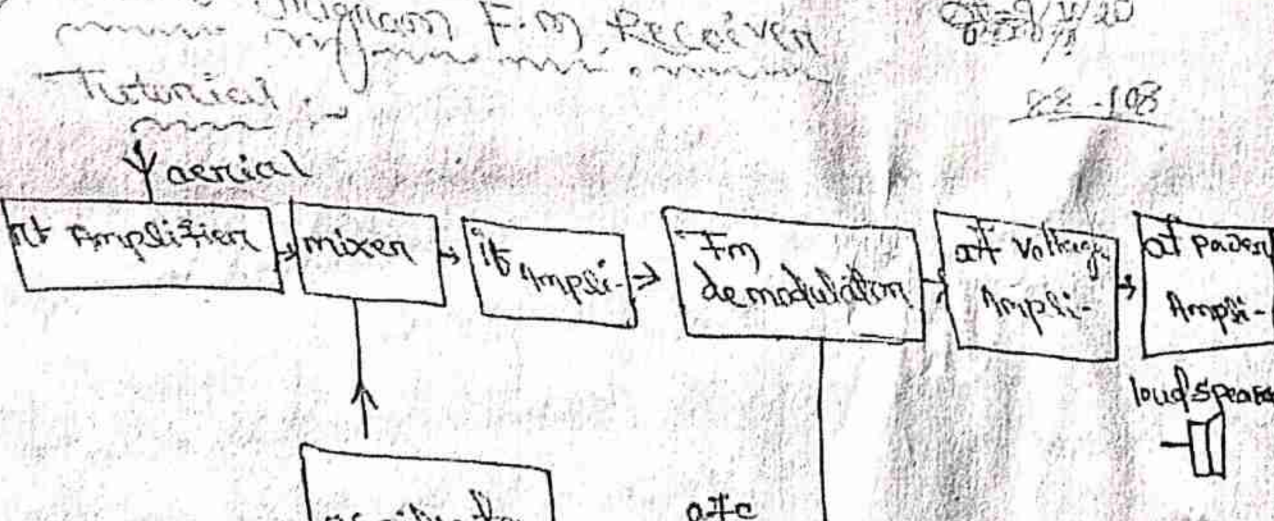

most of these blocks are discussed individually and in more detail on other pages.

*see filters, mixers, frequency changers am modulation & amplifi...

⇒ The f.m band covers 88-108 MHz
⇒ There are signals from many radio transmitters in this band inducing signal voltages in the Aerial.
⇒ The RF amplifier selects & amplifies the desired station from the many.
⇒ It is adjustable so that the selection frequency can be altered.
⇒ This is called TUNING.
- This makes the design and operation of the amplifier much simpler.
- The amplified I.F signal is fed to the demodulator
- This ckt recovers the audio signal & discard the I.F. carrier.
- Some of the audio is fed back to the oscillator as an AUTOMATIC FREQUENCY CONTROL voltage. This ensures that the oscillator frequency is stable in spite of temperature changes.
- The audio signal voltage is increased in amplitude by a voltage amplifier.
- The power level is increased sufficiently to drive the loud speaker by the power amplifier.

* The cheaper receivers the tuning is fixed and
* the tuning filter is wide enough to pass all
* signals in the f.m. band.

* The selected frequency is applied to the mixer

- The O/p of an oscillator is also applied to the mixer
* The mixer & oscillator form a FREQUENCY CHANGER circuit.
* The O/p from the mixer is the Intermediate freq (I.f)
* The i.f is a fixed frequency of 10.7 MHz
* No matter what the frequency of the selected radio station is the i.f. is always 10.7 MHz
* The i.f signal is fed into the i.f Amplifier.
* The advantage of the I.f amplifier is that its frequency & bandwidth are fixed, no matter what the frequency of the Incoming signal is.

AGC Concept :-

Concept of frequency conversion, Rf amplifier & If Amplifier, Tuning S/N ratio.

Frequency Conversion:-

- A freq. changer an electronic device that conversion alternating current (AC) of one freq to alternating current of another frequency
- A Heterodyne is used in signal electronic to convert freq.

RF Amplifier:-

A RF amplifier is a type of electronics amplifier that converts a low-power radio freq signal into a higher power signal.

Typically RF power amplifiers drive the antenna of a transmitter.

IF Amplifier :-

Intermediate freq. amplifiers are amplifier stages used to raise signal levels in radio & television receivers, as freq intermediate to the higher radio freq signal from the antenna & the power audio or video freq that the receiver is receiving.

Tuning :-

To adjust (an electronic receiver) to a desired freq. To adjust (an electronic (or)) so as to make it resonant with a given I/p signal " To adjust (an engine for ex) for max usability or performance.