Adv

(1) High sensitivity & selectivity
(2) High adjacent channel rejection
(3) No variation in BW.

Image signal and image signal rejection

→ Superheterodyne receiver suffer from a major drawback known as image frq problem.

→ Image signal is a signal whose frq. is as higher with f_o as the f_{ds} is smaller than the f_o.

If

f_o → local osc^n frq. f_{ds} → desired signal frq.

f_i → intermediate frq.

then, $\boxed{f_o - f_{ds} = f_i}$ —(1) ⇒ { • f_{ds} is lesser than f_o with an amount 'f_i'

By definition of image signal

$\boxed{f_{im} = f_o + f_i}$ —(2) f_{im} → image frq.

$= f_i + f_{ds} + f_i$

$= f_{ds} + 2f_i$ —(3) → from this eq^n it is clear that image signal is $2f_i$ higher than with desired signal

• If this signal is reached to antenna mixer by antenna then it will produce same intermediate frq 'f_i' which will be produced by desired signal f_{ds}.

$\underbrace{f_i = f_o - f_{ds}}$ If produced by f_{ds} $\underbrace{f_i = f_{im} - f_o}$ If produced by image frq.

$$f_i = f_{im} - f_o$$
$$= f_{ds} + 2f_i - f_o = f_{ds} + 2(f_o - f_{ds}) - f_o$$
$$= f_{ds} + 2f_o - 2f_{ds} - f_o$$
$$= \underbrace{f_o - f_{ds}} \quad \text{—(4)}$$

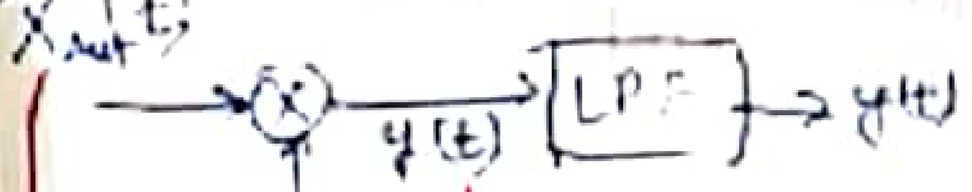
Acc^n to eq^n (1) & eq^n (4) desired signal and image signal produce same IF which will further amplify by IF amp^r and interfere with the signal. so the image signal must be rejected. which depends on selectivity

$\alpha = \sqrt{1 + Q^2 \rho^2}$ ⇒ Image frq. rejection ratio   where $\rho = \frac{f_{im}}{f_{ds}} - \frac{f_{ds}}{f_{im}}$

Demodulation (AM)

Synchronous Demodulation

DSB-C

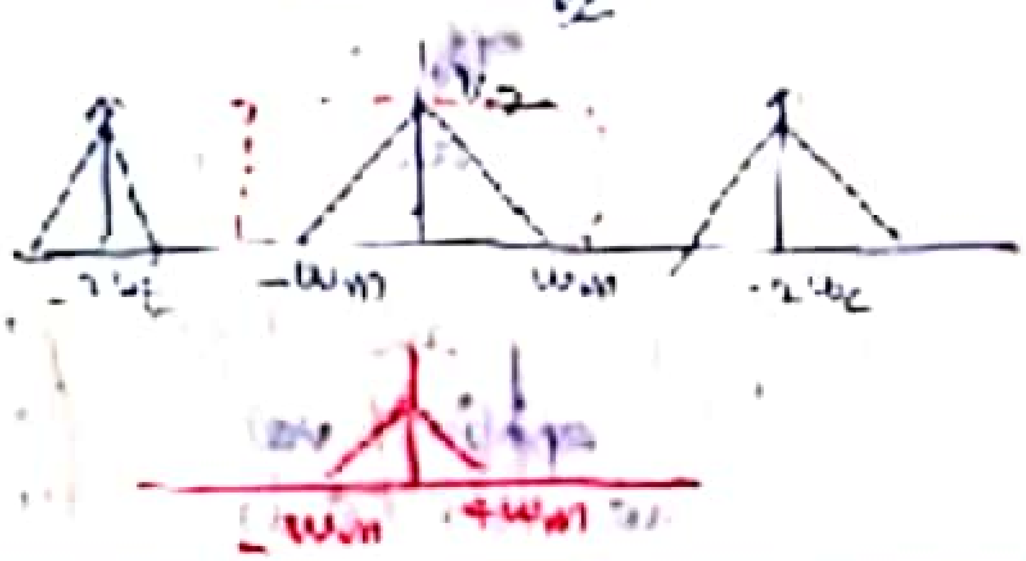


$X_{AM}(t) = [A_c + m(t)]\cos\omega_c t$

$y(t) = (A_c + m(t))\cos^2\omega_c t$

$= (A_c + m(t))\left(\frac{1+\cos 2\omega_c t}{2}\right)$

$= \frac{A_c}{2} + \frac{A_c}{2}\cos 2\omega_c t + \frac{m(t)}{2} + \frac{m(t)\cos 2\omega_c t}{2}$



DSB-SC

$y_1(t) = A_c m(t)\left[\frac{1+\cos 2\omega_c t}{2}\right]$

$= \frac{A_c m(t)}{2} + \frac{A_c m(t)}{2}\cos 2\omega_c t$ — filtered out

SSB-SC

$X_{SSB}(t) = m(t)\cos\omega_c t + \hat{m}(t)\sin\omega_c t$

$y_1(t) = m(t)\cos^2\omega_c t - \hat{m}(t)\sin\omega_c t\cos\omega_c t$

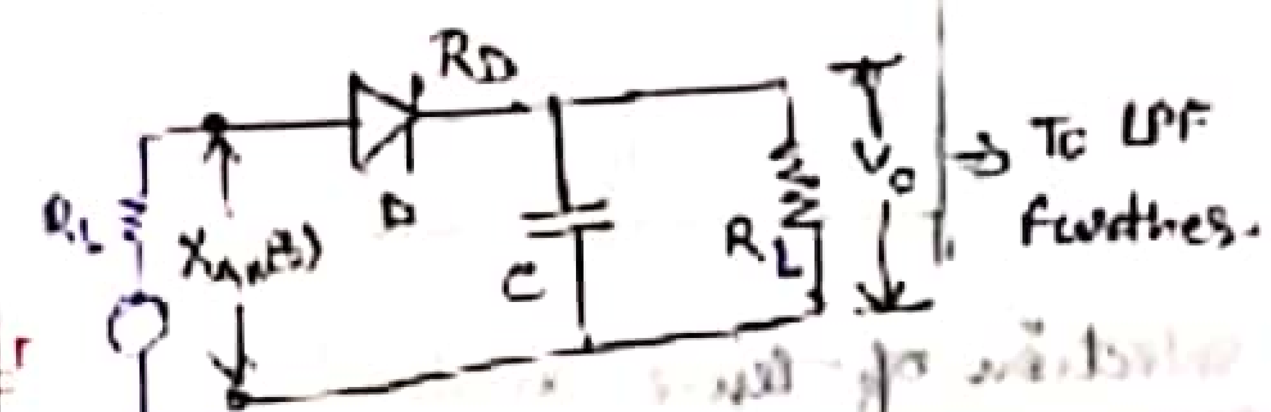
$= m(t)\left(\frac{1+\cos 2\omega_c t}{2}\right) - \hat{m}(t)\left(\frac{\sin 2\omega_c t}{2}\right)$

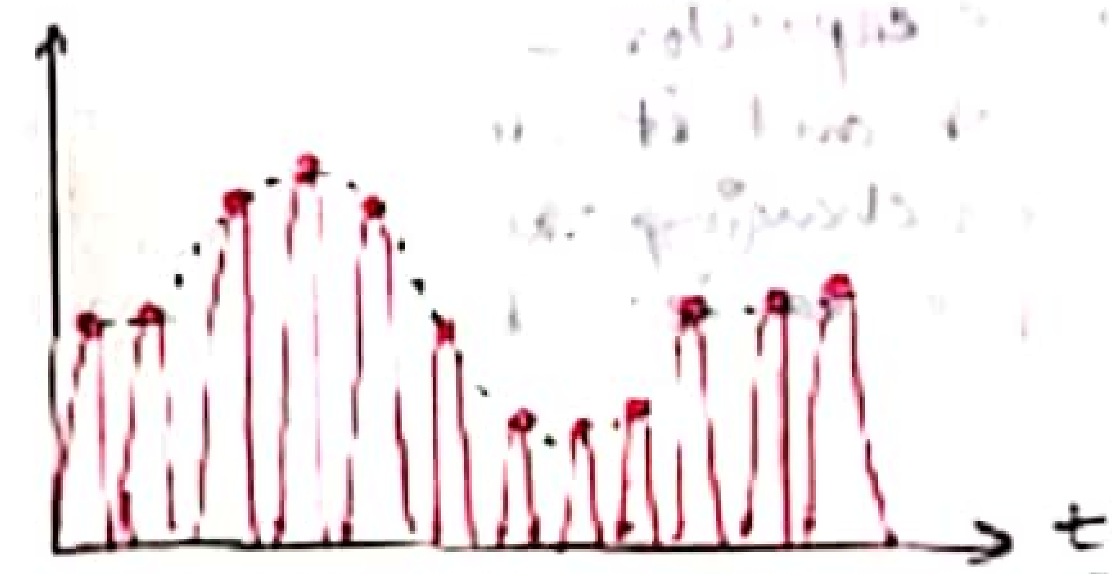
$= \frac{m(t)}{2} + \frac{m(t)}{2}\cos 2\omega_c t + \frac{\hat{m}(t)}{2}\sin 2\omega_c t$ — filtered out

Asynchronous Demodulation

Envelope Detector

→ Also called Diode Detector

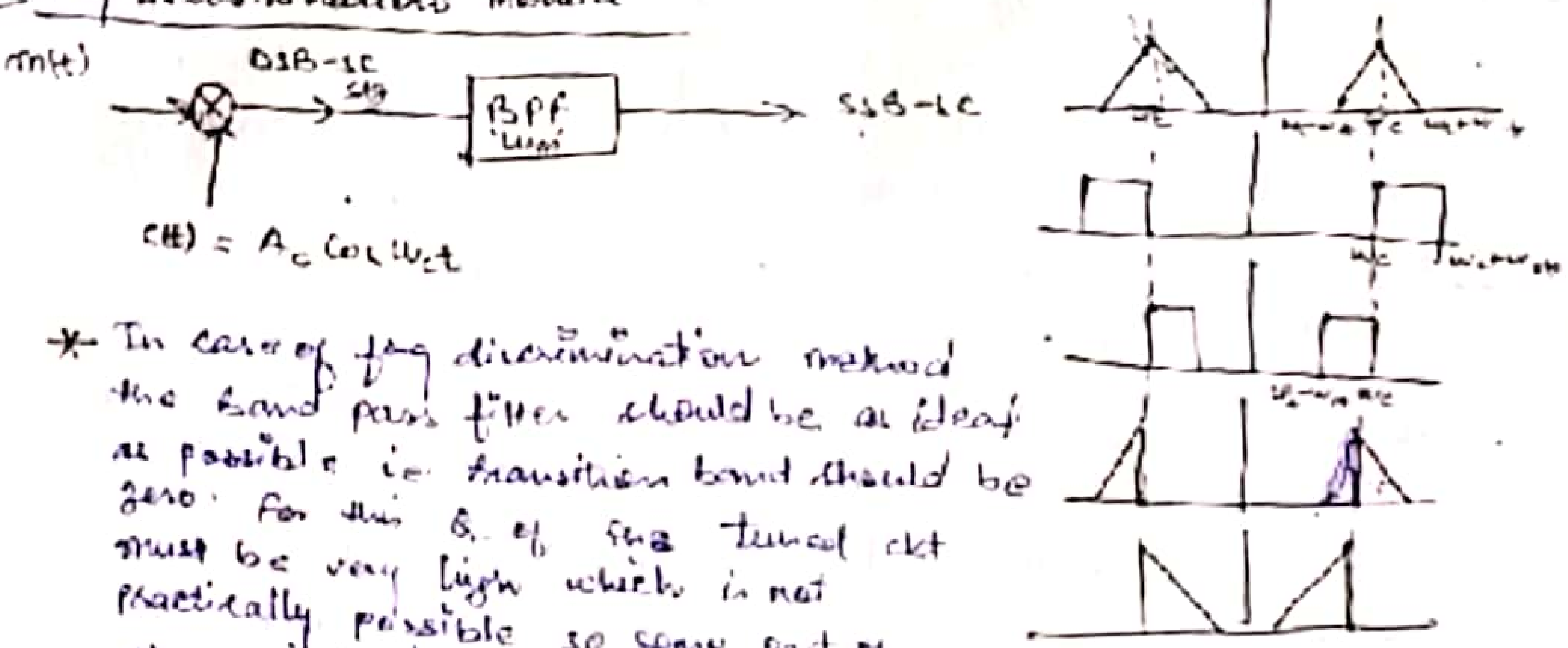




- During the +ve half cycle of the i/p sig., the diode is forward biased and the cap^r C charges up rapidly to the peak of the i/p signal.
- As soon as cap^r charges to the peak value, the diode stops conducting. The cap^r will discharge through R between the positive peaks.
- When i/p sig falls below the peak value, the diode becomes reverse biased and the cap^r C discharges slowly through the load resistor R_L.
- This discharging process continues until the next positive half cycle, when the i/p sig becomes greater than the voltage across the cap^r, the diode conducts again and process is repeated.

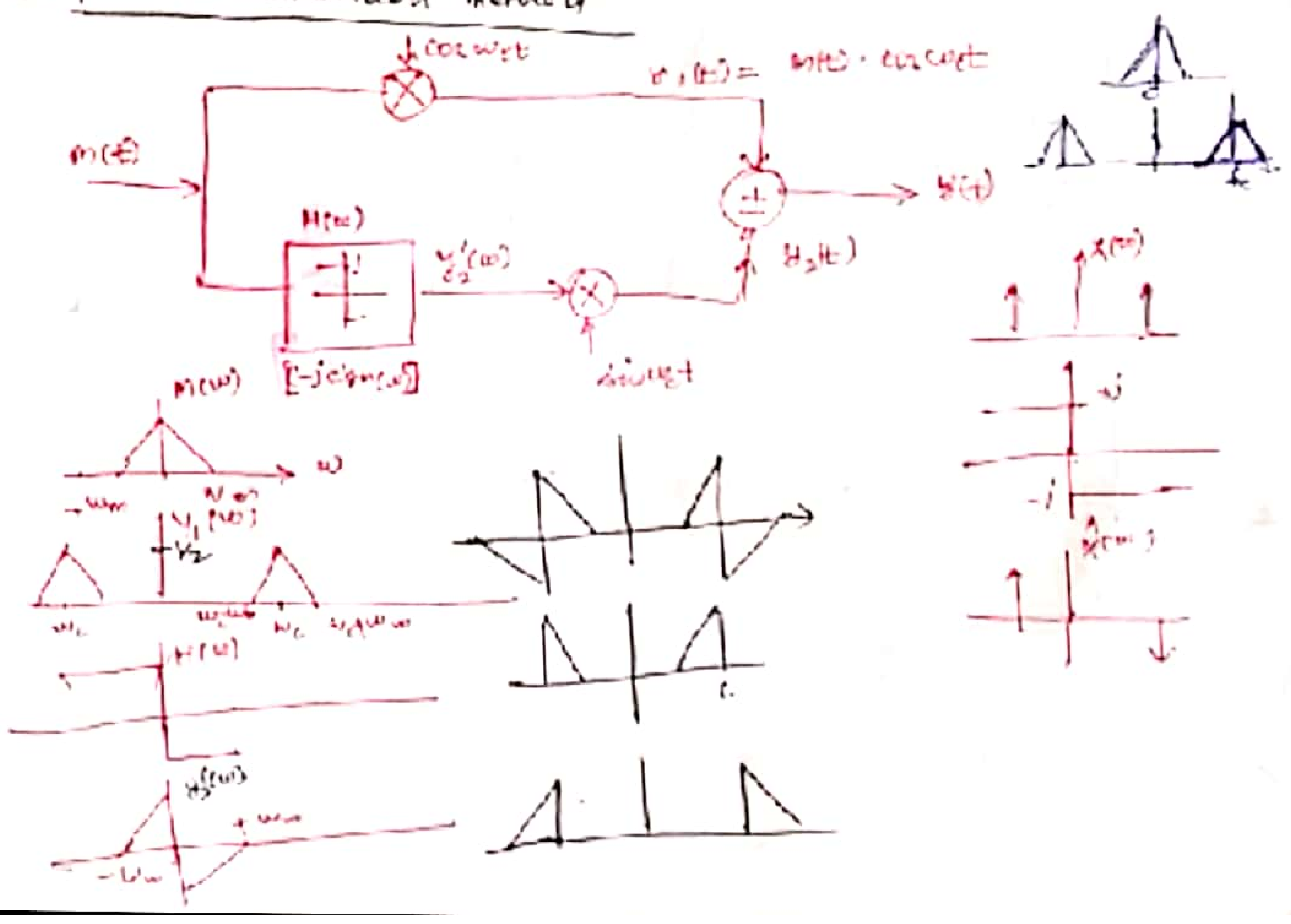
Generation of SSB-SC → freq discrimination method
→ Phase discrimination method.

1) Freq Discrimination Method



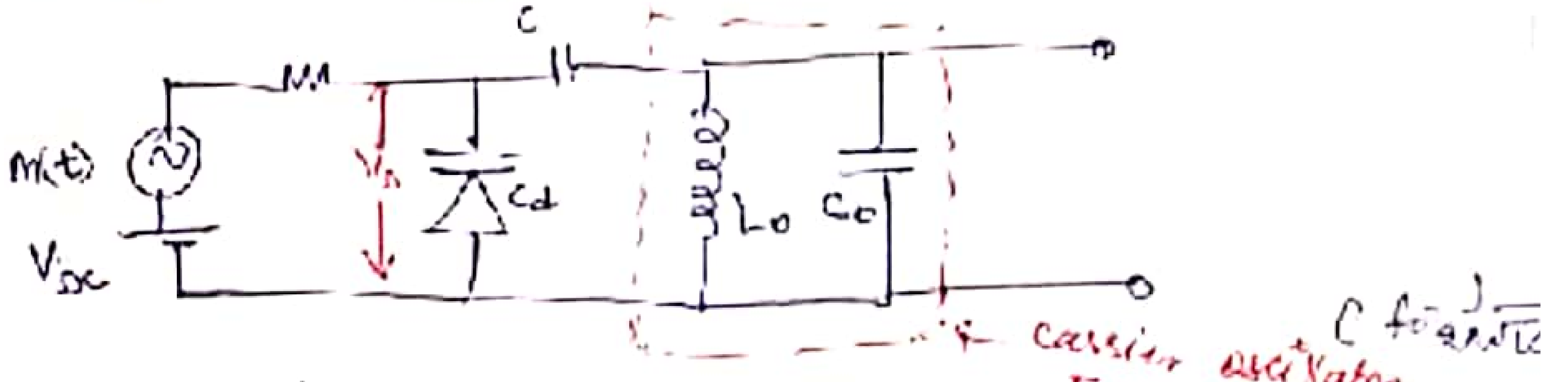
* In case of freq discrimination method the band pass filter should be as ideal as possible i.e. transition band should be zero. For this Q of the tuned ckt must be very high which is not practically possible so some part of other side band is also transmitted that's why this types of SSB generation is not used.

2) Phase discrimination method



Varactor Diode Modulator

→ A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied bias. It must operates in reverse biased.



→ The varactor diode is connected in shunt with the tuned ckt of the carrier oscr.

The capacitance C_d of varactor diode

$$C_d = \frac{K}{\sqrt{V_D}} = K (V_D)^{-1/2}$$

where $V_D = V_{DC} + m(t)$

→ The total capacitance of the oscr tank ckt will be $C_0 + C_d$ and thus the instantaneous frq of oscillation f_i is

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

$$f_i = \frac{1}{2\pi\sqrt{L_0(C_0 + K V_D^{-1/2})}}$$

→ The instantaneous frq f_i of FM signal depends upon V_D which in turn depends upon the value of modulating signal m(t). Thus f_i also depends upon m(t) and FM is generated.

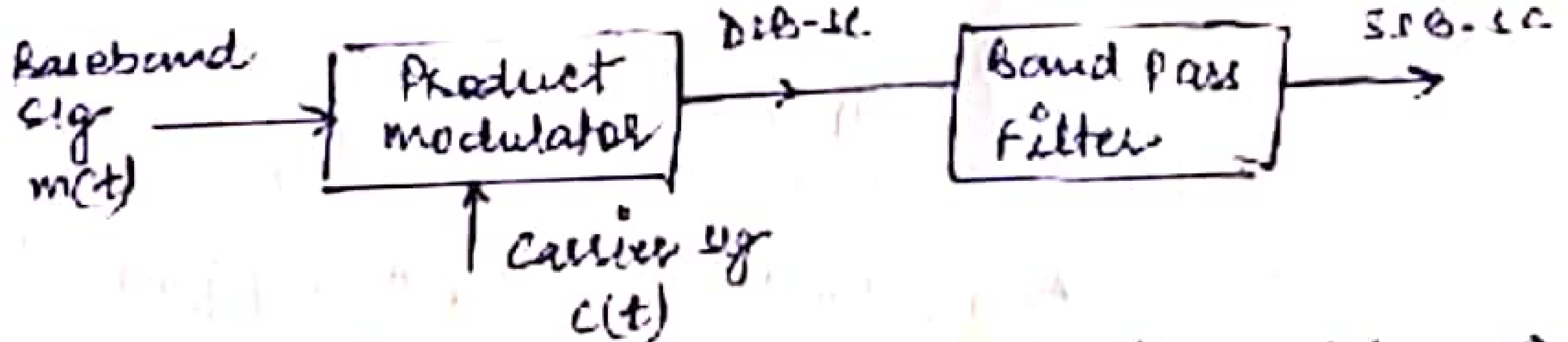
Disadv

In Direct method of FM generation, it is not easy to get a high order stability in carrier frq.

Single Sideband (SSB-SC)

→ Single sideband supressed carrier modulation is the process in which only upper side-band or lower sideband is transmitted.

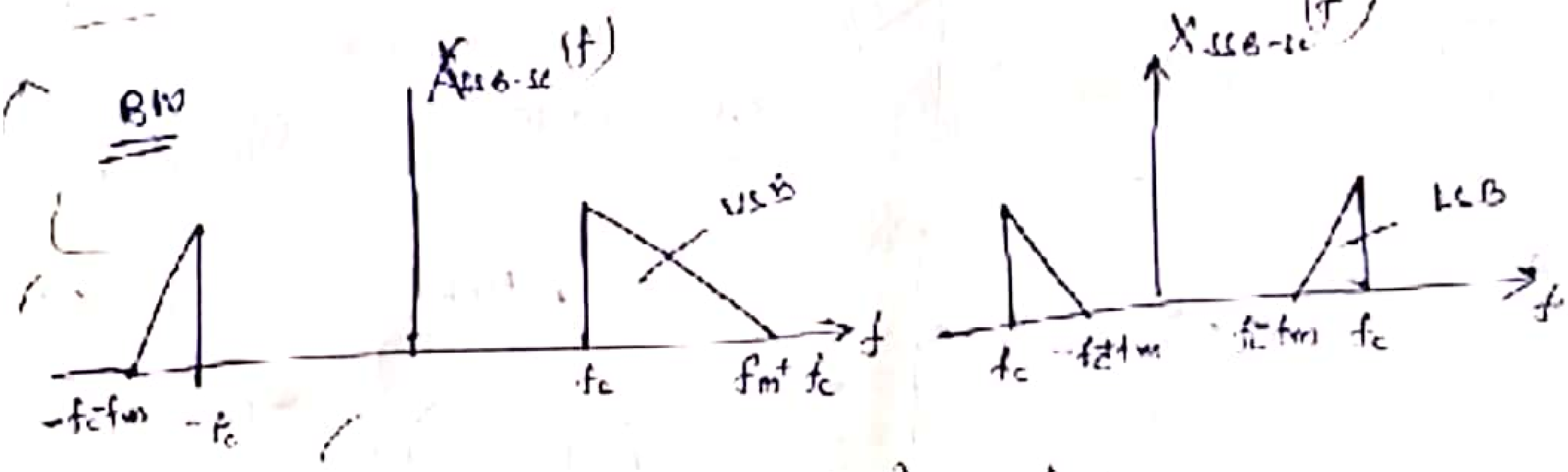
→ Since both the side band contains same infⁿ so instead sending both we can supress one sideband.



$$X_{SSB-SC}(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c + f_m) t \quad \Rightarrow \text{For USB}$$

$$X_{SSB-SC}(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c - f_m) t \quad \Rightarrow \text{For LSB}$$

BW



So $B_T = (f_c + f_m) - f_c$

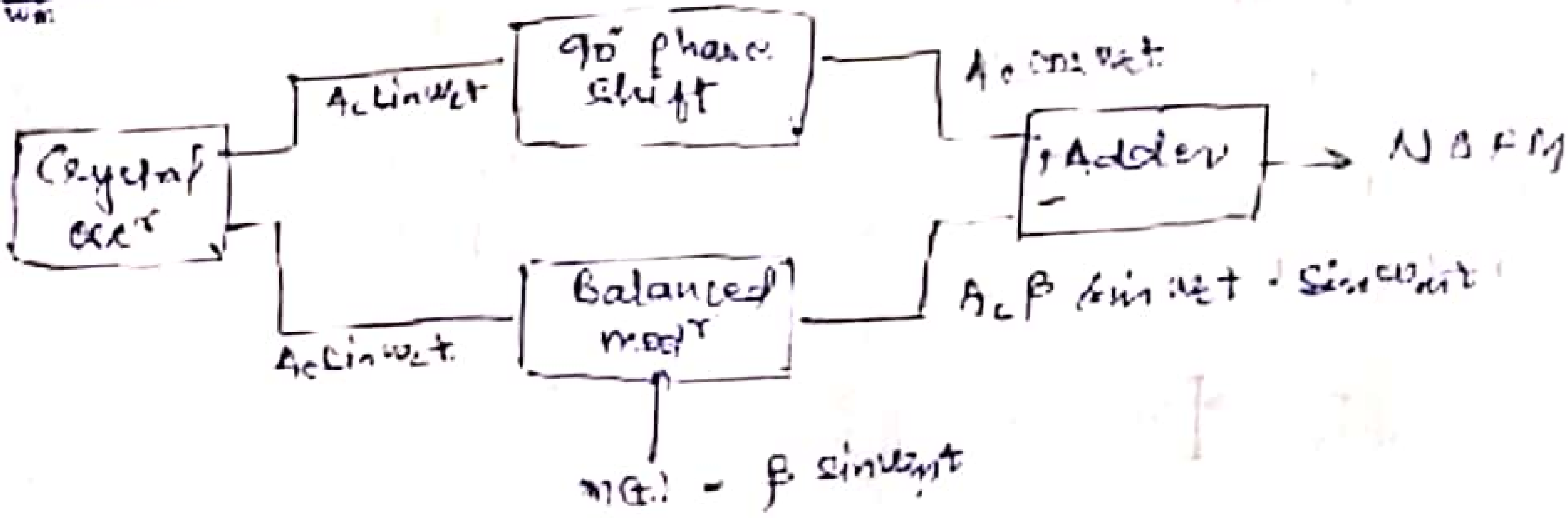
$$\boxed{BW = f_m}$$

Drawback of SSB

It's filter is very difficult to design (Accuracy is difficult to achieve and costly also).

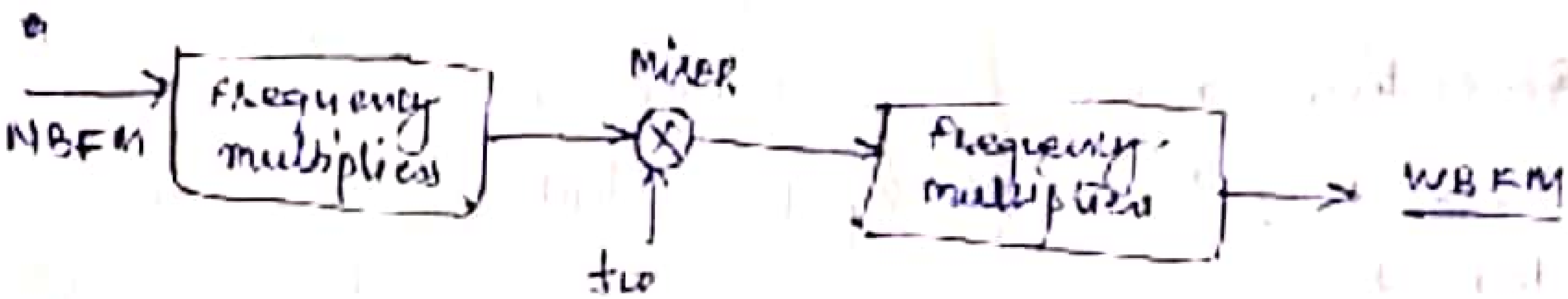
$m(t) = A_m \cos w_m t$

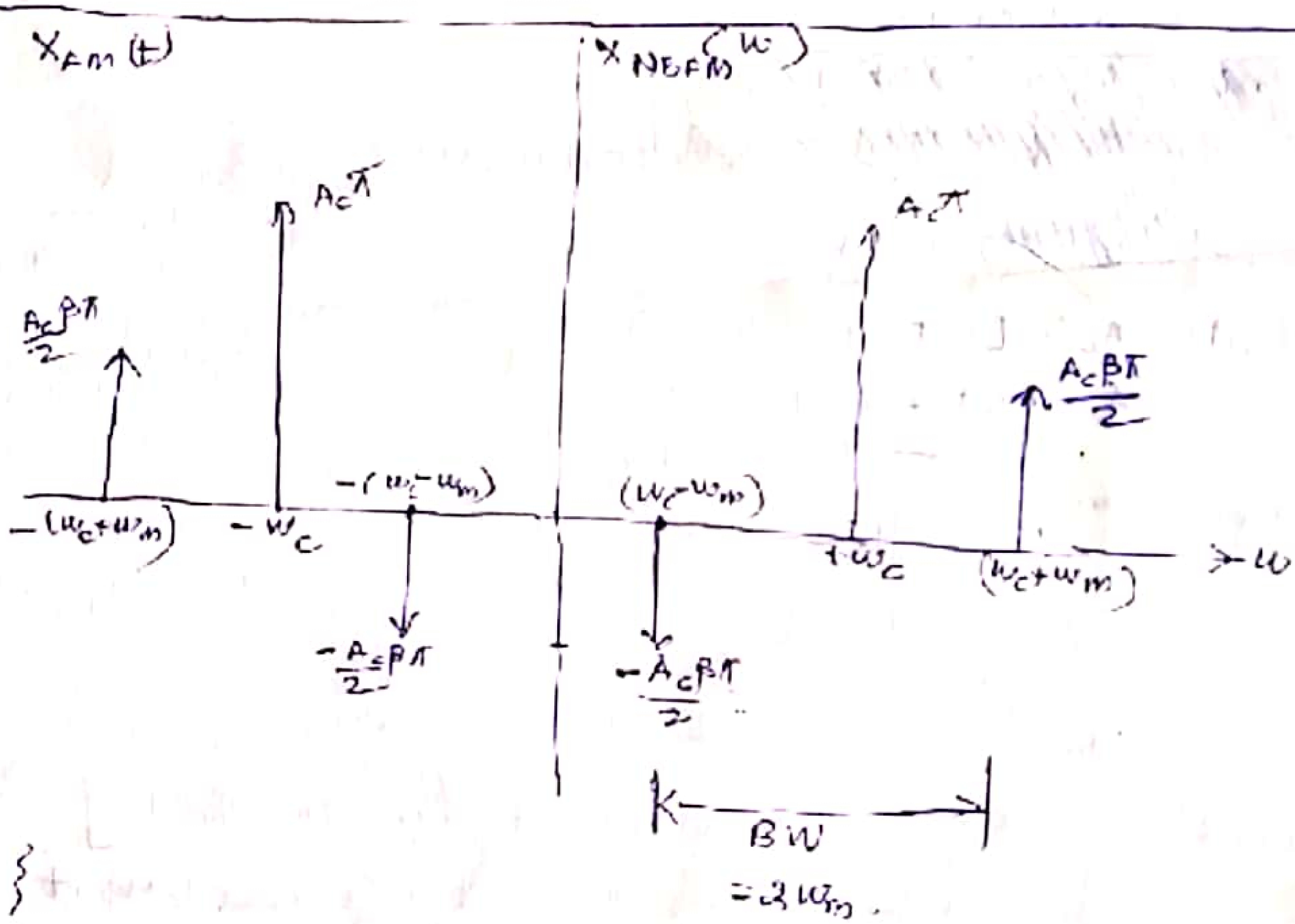
$= \frac{A_m}{w_m} \sin w_m t$

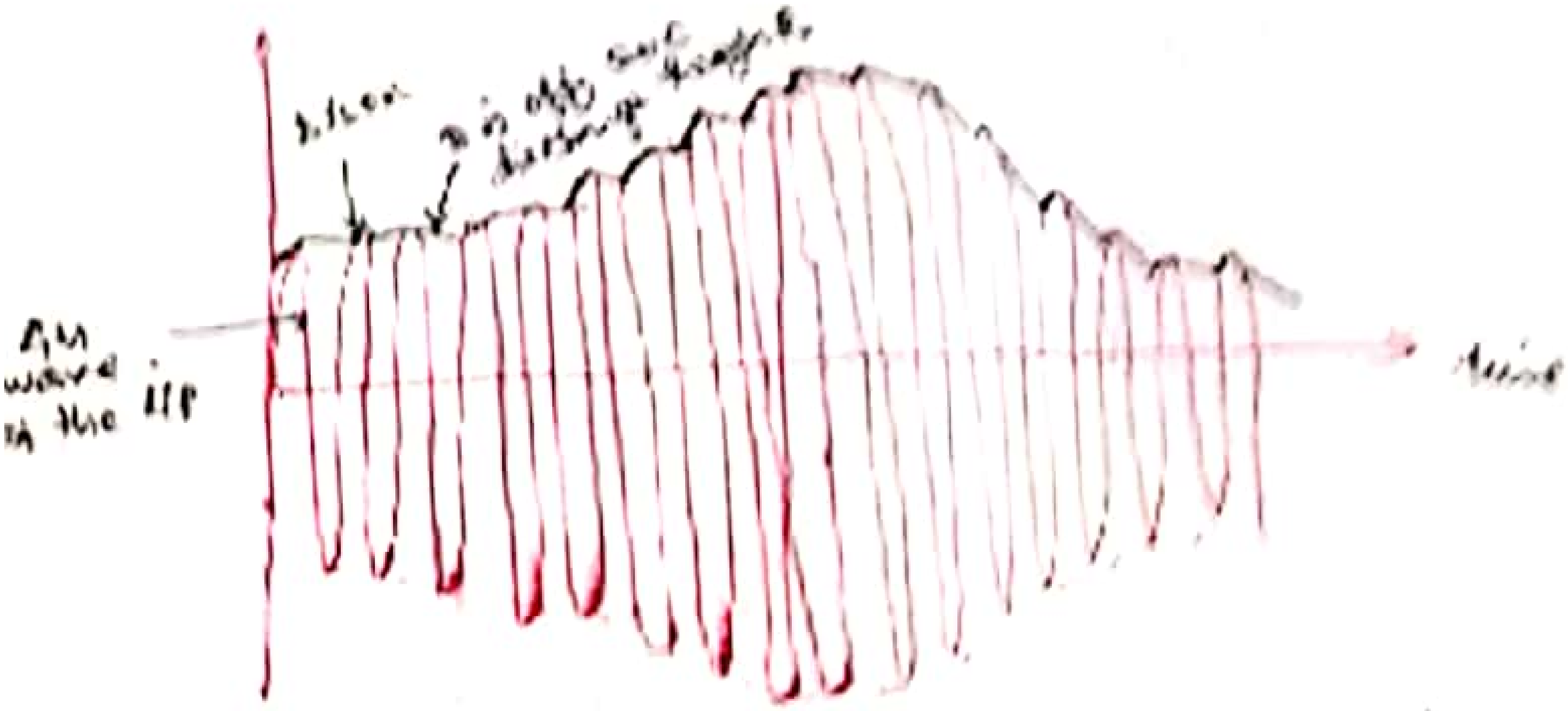


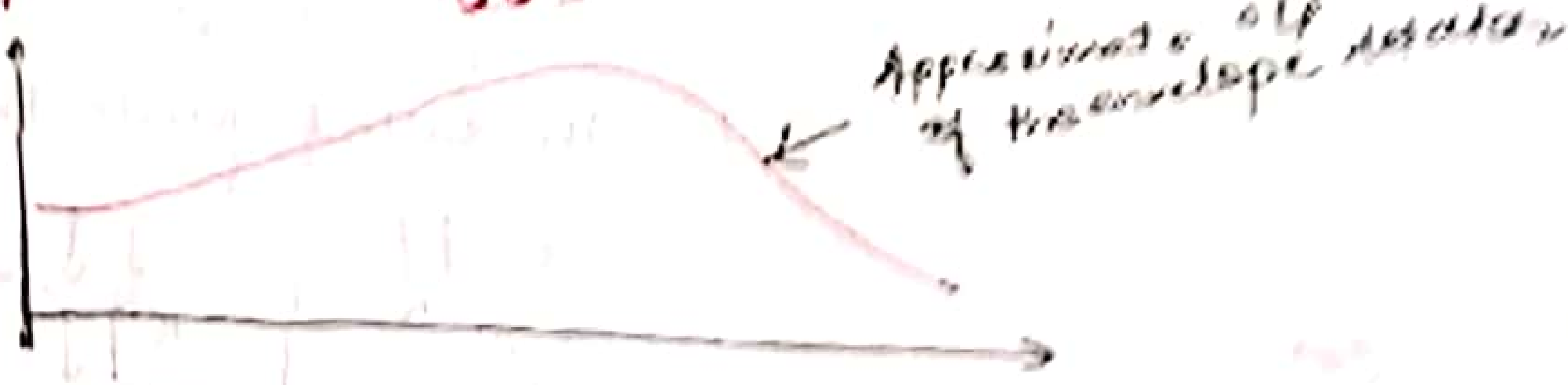
$$y(t)_{NBFM} = A_c \cos w_c t + A_c \beta \sin w_c t \sin w_m t$$

This expression represents a narrow band FM









Selection of time constant

→ The capacitor C charges through D and Rs when the diode is ON and it discharges through R_L when D is off.

→ The charging time constant $\tau_c = R_s C$ must be short as compared to the carrier time period i.e. (1) $\tau_c << \frac{1}{f_c}$ so that cap charges rapidly and thereby follow the applied voltage when the diode is conducting.

→ The discharging time constant $\tau_L = R_L C$ must be long enough to ensure that the cap discharges slowly through the [illegible] between positive peaks of the carrier wave. However it should not be so long that the capr voltage does not discharge at the [illegible] rate of change of the modulating wave. i.e.

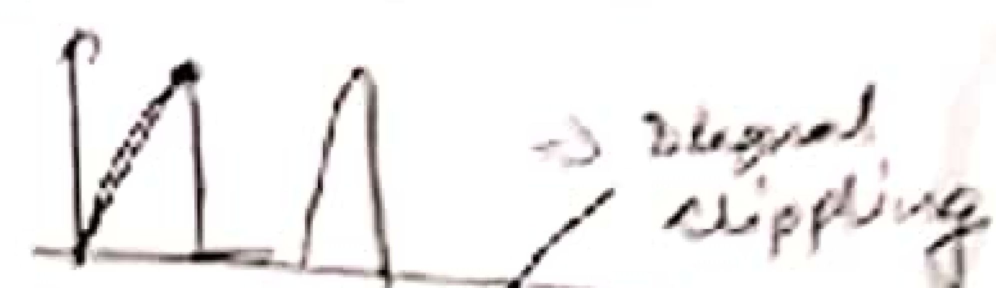
$$\frac{1}{f_c} << R_L C << \frac{1}{W} \rightarrow \text{max msg freq.}$$

(2) (3)

(1) if $R_L C \not\gg \frac{1}{f_c}$

(2) if $\frac{1}{f_c} \not\ll R_L C$

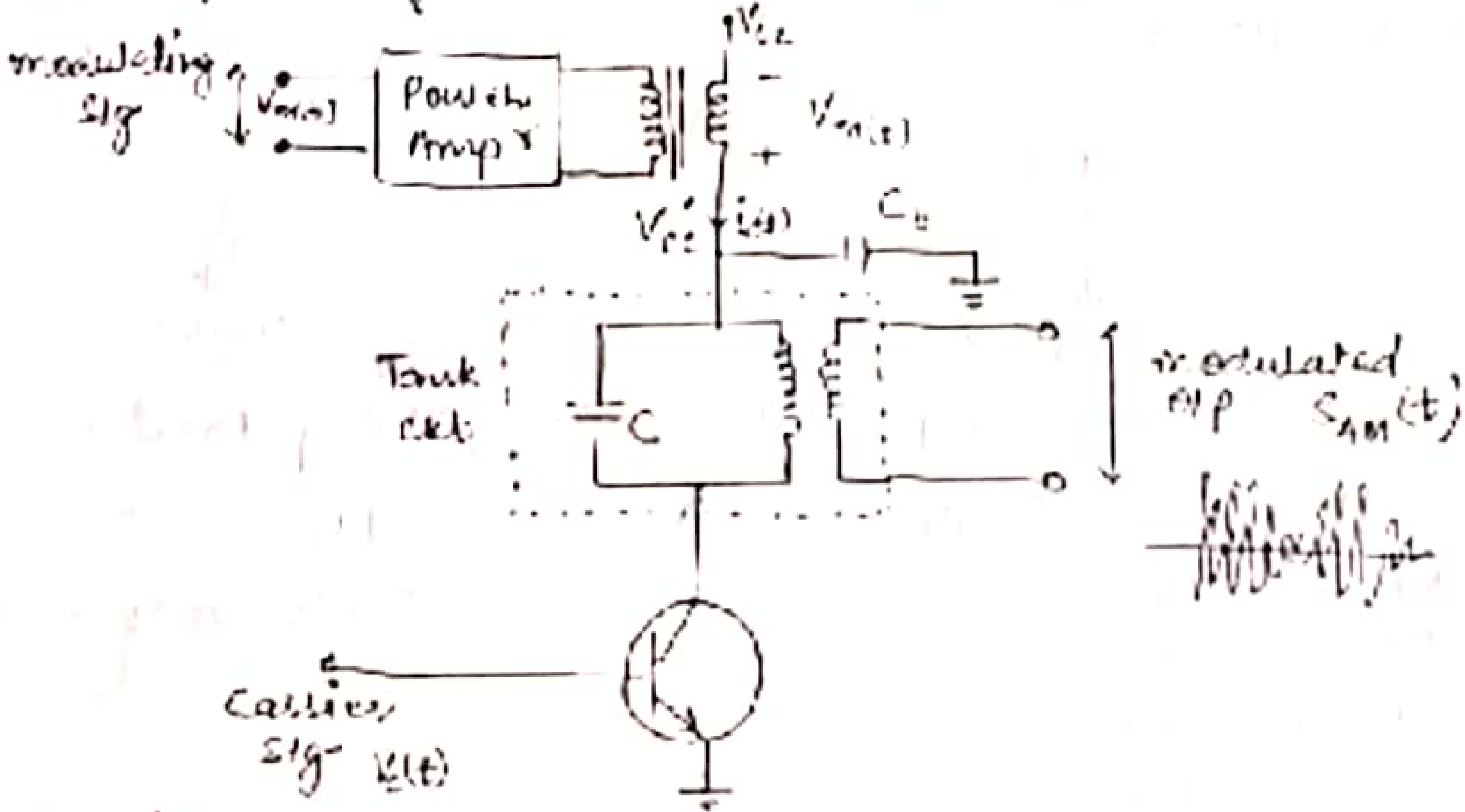
(3) $R_L C \not< \frac{1}{W}$



If $R_L C \gg \frac{1}{W}$ then discharging will be Horizontal and the rectified o/p voltage miss out many peaks bcz of that Diagonal clipping occurs.

Collector Modulator

→ It is a linear type of modulator. In this modulation, linear region of ch^c of T^r or Diode is used.



Construction

→ Carrier sig is applied at base and modulating sig is applied at collector terminal at secondary AF transformer.

→ The modulating sig is amplified through power Ampr then applied to primary of AF T/F.

→ Tank ckt is connected at collector as a load from which modulated sig is received.

→ Bypass capr is used for protection of AF T/F from high freq signal.

Working

→ In this method, the modulating voltage is amplified first then applied to collector for modulation that is why it is known as high level modulation.

→ Bias voltage V_{cc} is superimposed on modulating sig so V'_{cc} bias voltage changes acc. to modulating sig $m(t)$

$$V'_{cc} = V_{cc} + m(t)$$

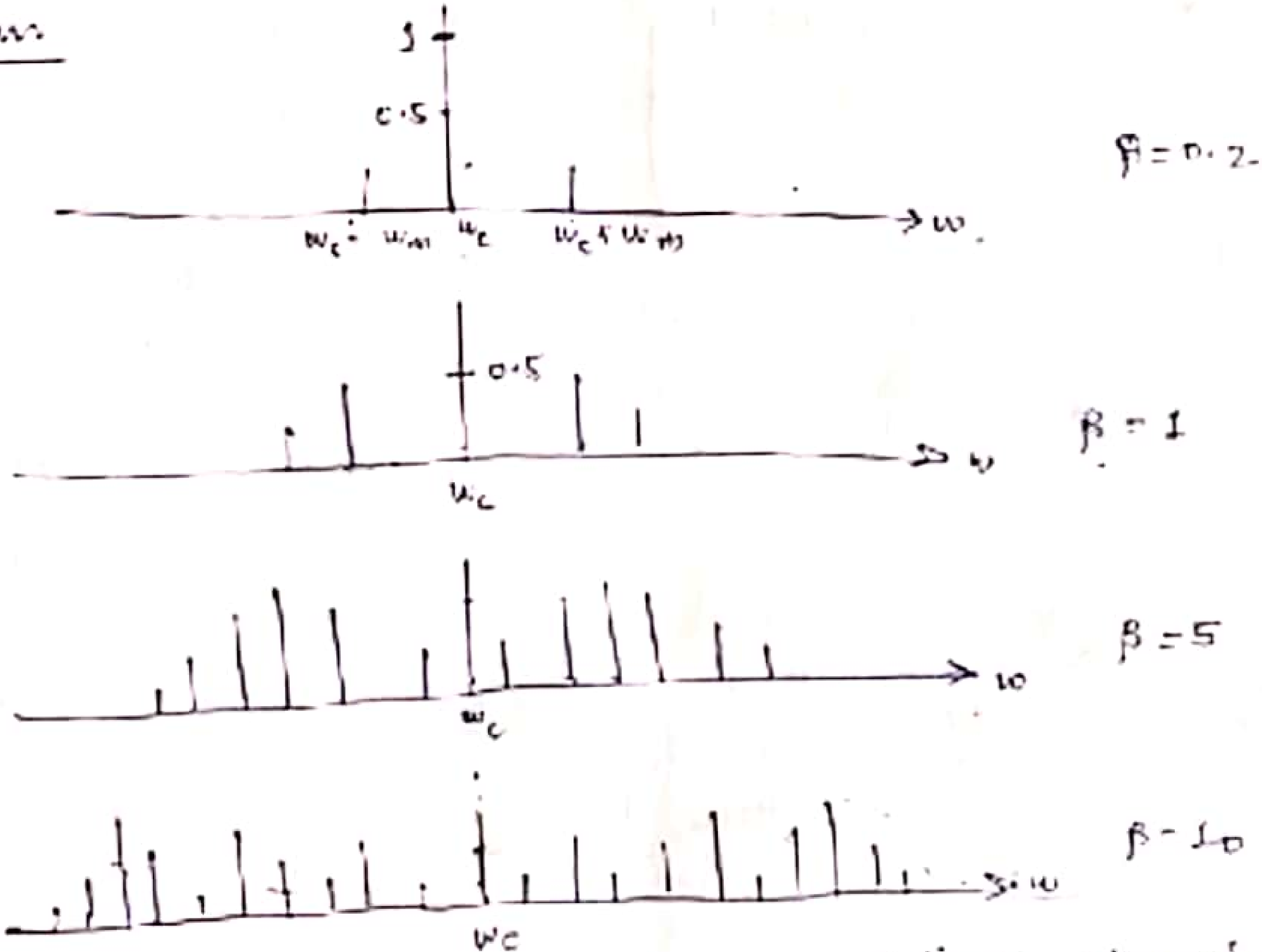
$$S_0(t) = A_c \cos[\omega_c t + \beta \sin \omega_m t]$$

$$S_0(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos(\omega_c + n\omega_m)t$$

→ $J_n(\beta)$ is the Bessel f^n of first kind of order n and argument β.

This expression is not simple due to sine of sine f^n. The only way to solve this eq^n is by Bessel f^n.

Spectrum



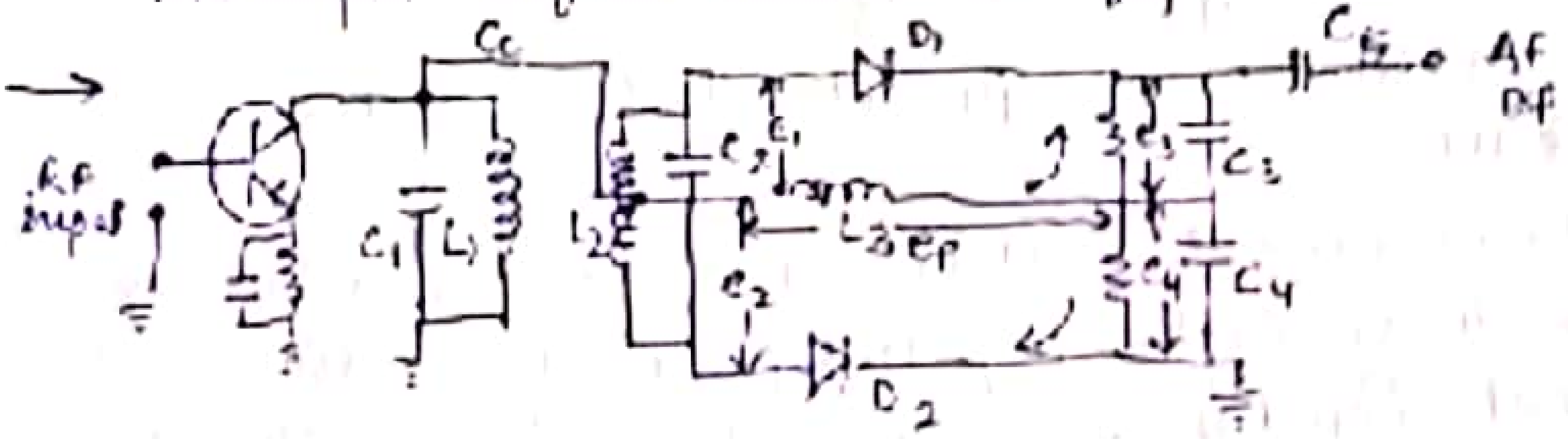
→ The relative amplitudes of the spectral lines depend on the values of $J_n(\beta)$ and value of $J_n(\beta)$ becomes very small for large value of n

⇒ FM
- → Narrowband FM {β is small compared to 1}
- → wide band FM {large values of β}

FOSTER SEELY DISCRIMINATOR

→ The foster seely discriminator is also known as the phase shift discriminator. It uses a double tuned rf T/F to convert frq variations in the received fm signal to amplitude variations. These amplitude variations are then rectified and filtered to provide a dc output voltage. This voltage varies in both amplitude and polarity as the input signal varies in frq

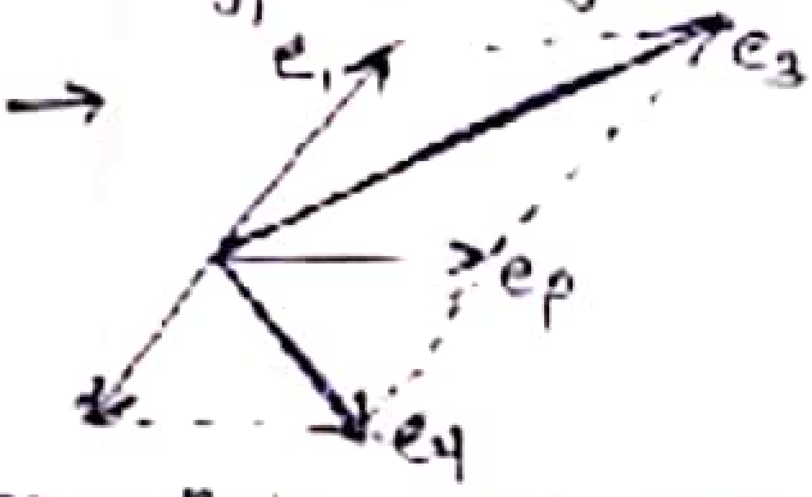
→



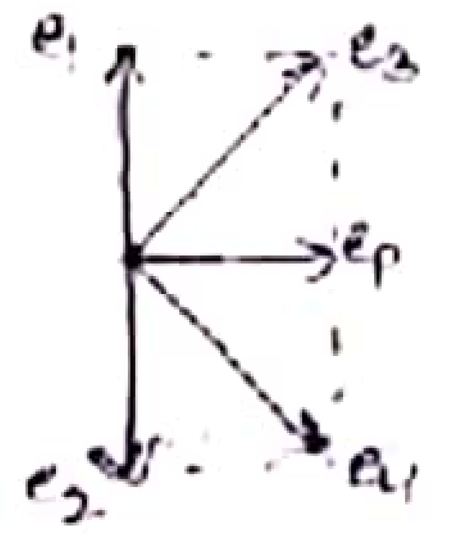
→ The collector ckt tank consists of C_1 L_1. C_2 & L_2 form the secondary tank ckt. Both tank ckts are tuned to the center frq of incoming fm signal.

→ L_3 is the dc return path for D_1 & D_2. R_3 & R_4 are load and bypassed by C_3 & C_4 to remove rf. C_c is o/p coupling capt

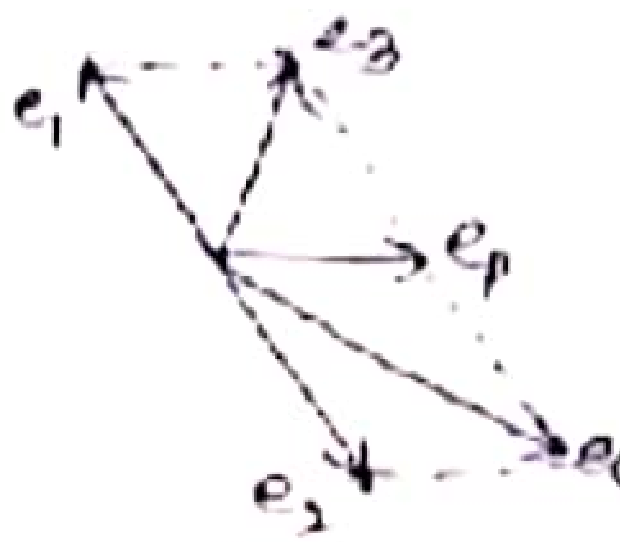
→



(1) opn above resonance



(2) opn at resonance



(3) opn below resonance

(2) opn at resonance

The input signal applied to the primary tank ckt as e_p. Since coupling capt C_c has negligible reactance at i/p frq, rf choke L_3 is effectively in parallel with primary tank ckt. Also, because L_3 is effectively in parallel with primary tank ckt, input voltage e_p also appears across L_3. With voltage e_p applied to the primary of T_1, a voltage is induced in the secondary which causes current to flow in secondary tank ckt

The current flowing in tank ckt causes voltage drops across each half of the secondary. These voltage drops are equal amplitude and opposite phase.

let $m(t) = A_m \cos w_m(t)$

→ then in PM

$\phi(t) = k_p\, m(t) = K_p A_m \cos w_m t$

⇓

Modulation index $\beta_{PM} = K_p A_m$

→ for FM, $\phi(t) = K_f \int m(t)dt = K_f \int A_m \cos w_m(t)\, dt$

$= \frac{K_f A_m}{w_m} \sin w_m(t)$

⇓

Modulation index $\beta_{FM} = \frac{K_f A_m}{w_m}$

So,

$$\beta = \begin{cases} K_p A_m & \text{for PM} \\ \frac{K_p A_m}{w_m} & \text{for FM} \end{cases}$$

⇒ Modulation index or Max^m value of phase deviation

Important point about β

- In FM β decide the BW of FM.
- β_{FM} can be greater than 1
- β also decides no. of side bands having significant amplitu

Deviation Ratio

→ The modulation index corresponding to the max deviation and max freq is called as deviation ratio.

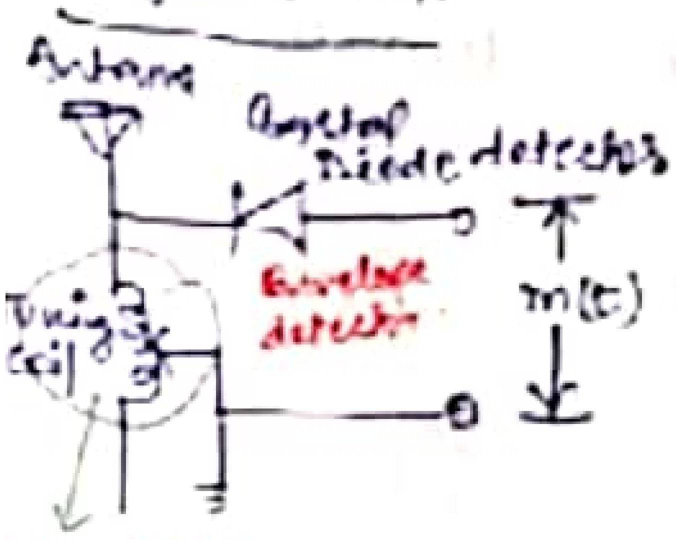
$$D = \frac{\text{Max}^m \text{ deviation}}{\text{Max}^m \text{ modulating freq}} = \frac{\Delta w}{w_m}$$

$$\boxed{D = \frac{\Delta w}{w_m}}$$

Types of Radio Receivers

- → (1) Crystal radio Rx
- → (2) Tuned radio freq. Rx
- → (3) Super heterodyne radio Rx

Crystal Rx

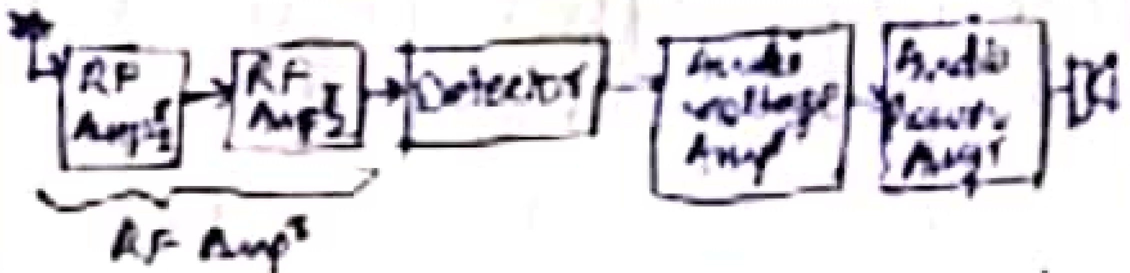


Desired freq can be selected by changing turns of coil

→ Selectivity & sensitivity is very poor

→ Ampr is not used so weak demodulated signal

TRF Rx



→ Good only for medium wave freq but poor performance at higher freq.

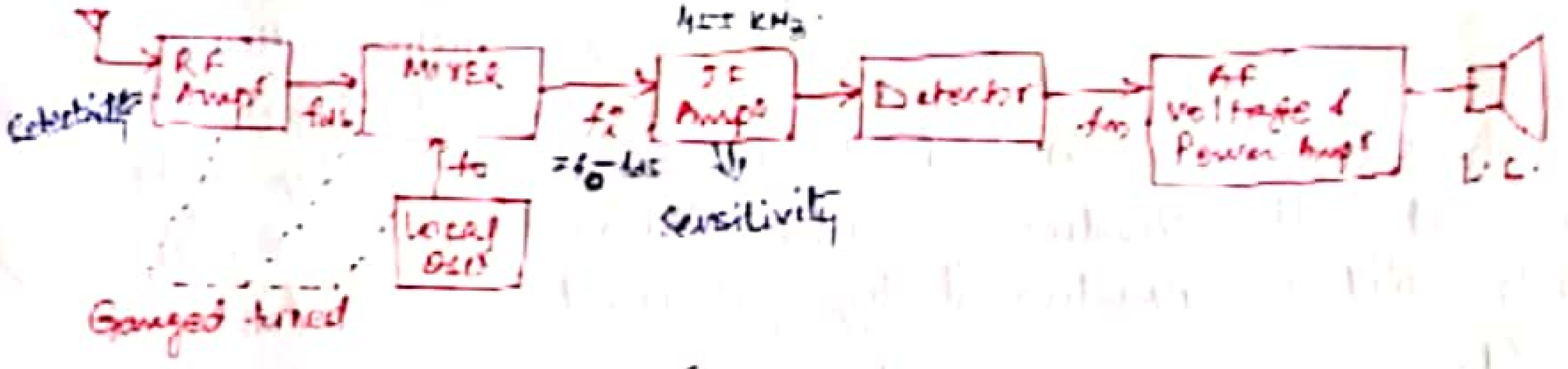
→ Instable due to multiple RF ampr. variation in BW. Poor selectivity

→ Simple ckt & good sensitivity.

SHRR

A superheterodyne Rx, often shortened to super het, is a type of Radio Rx that uses freq mixing to convert a received signal to a fixed Intermediate freq which can be more conveniently processed than the original carrier freq.

Super Heterodyne radio Receiver



$$f_o = f_s + f_i$$

$$f_i = f_o - f_s \rightarrow \text{is selected}$$

Working

→ The received signal is amplified by rf ampr which is a tuned Ampr ckt. Amplified signal is passed on to a mixer. here rf carrier is mixed with 'f_o' generated by local oscr. The process of mixing is called heterodyning. f_o is selected to be above f_s.

→ Mixer generates sum and difference of freq $(f_o + f_s)$ and $(f_o - f_s)$. $(f_o + f_s)$ eliminated by a filter. The difference freq $(f_o - f_s)$ is called intermediate freq carrier.

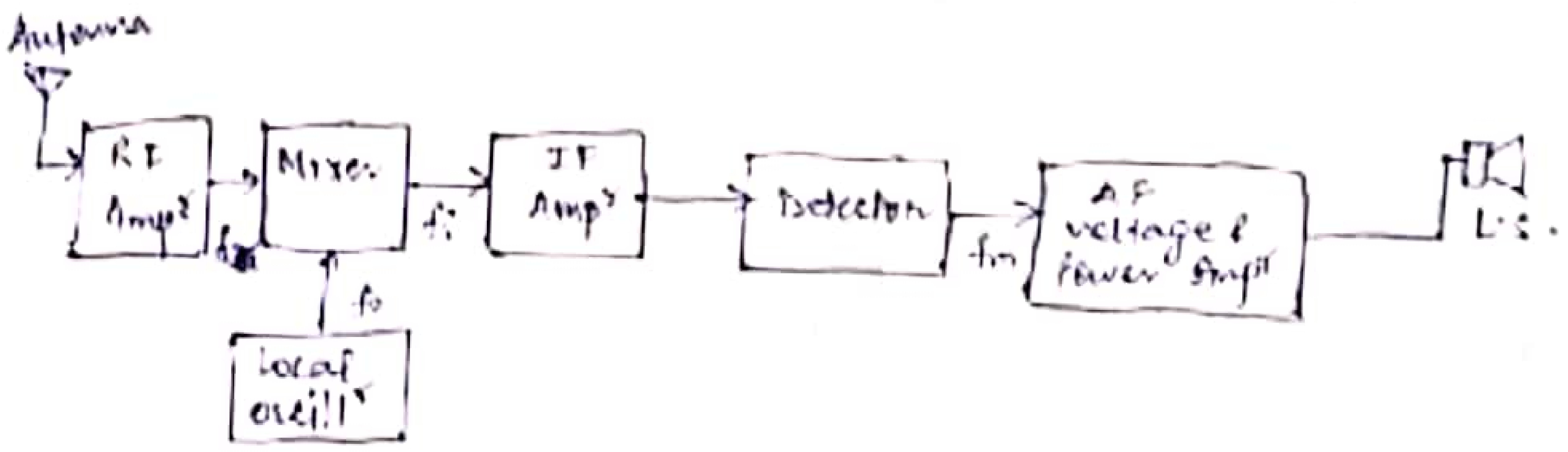
Tuned Radio frq Rx

→ works well on medium waves. but perform poor on higher radio frq.

Super heterodyne Receiver

The process of mixing two sig having diff frq to produce a new frq.

→ Superheterodyne is based on principle of heterodying.



- Mixer generates sum and difference of freq i.e. the mixer o/p consists of $f_s + f_o$ (eliminated by filter) and $f_o - f_r$ (is called intermediate frq "f_i")

- Super heterodyne Rx suffers from major drawback known as image frq problem.
- If a frq f_{si} manages to reach the mixer

$$f_{si} = f_{osc} + f_i \qquad (f_{osc} = f_r + f_i)$$
$$= f_r + f_i + f_i$$
$$\boxed{f_{si} = f_r + 2f_i}$$

The rejection of an image frq by a single tuned ckt may be defined as the ratio of the gain at sig frq to the gain at image frq

$\alpha = \sqrt{1 + Q^2 \rho^2}$ where $\rho = \frac{f_{si}}{f_r} - \frac{f_r}{f_{si}}$ Q → Quality factor

Transmission BW of FM :

→ Theoretically, an FM wave contains an infinite no of side frequencies and consequently BW required is also infinite. Most of the power of FM wave resides in a finite no of sidebands.

→ Carson generalized the BW formula for an FM wave as

$$B_T = 2(\beta+1) f_m \quad \text{in Hz}$$

$$B_T = 2(\beta+1) \omega_m \quad \text{in rad/s}$$

→ For Narrowband FM $\{ \beta << 1 \}$

$$\boxed{B_T = 2 f_m}$$

For wideband FM $\{ \beta >> 1 \}$

$$\boxed{B_T = 2 \beta f_m}$$

→ In terms of deviation ratio the BW is

we know $\beta = \frac{\Delta f}{f_m}$ so $\Delta f = \beta f_m$

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

$$= 2\Delta f + 2 f_m$$

$$\boxed{B_T = 2(\Delta f_m + f_m)}$$

Because the winding is inductive, voltage is 90° out of phase with current in it.

→ The voltage applied to the anode of D_1 is the vector sum of voltages e_p and e_1

$$e_3 = e_p + e_1$$

→ The voltage applied to D_2 is vector sum of e_p & e_2

$$e_4 = e_p + e_2$$

→ At resonance $e_3 = e_4$

→ The diodes conduct on opposite half cycles of the input waveform

<u>Operation above resonance</u>: A phase shift occurs when an input freq higher than center freq is applied and current & voltage phase relationship changes. When tuned ckt operates at above resonance, the inductive reactance of coil ↑ & capacitive reactance of capr ↓. Above resonance the tank ckt acts like an inductor. Secondary current lags e_p. e_1 & e_2 still out of phase

Collector Modulators }
Base Modulators } DSB-C

Balanced Modulator → DSB-SC

AM

- **DSB-C**
  - Linear modulator
    1) Collector
    2) Base
    3) Collector-Base
  - Non-linear modulator
    1) Square law modulator
- **DSB-SC**
  1) Balanced
  2) Ring
- **SSB-SC**
  1) Freq discriminator
  2) Phase "
  3) Weaver's method

Collector Modulator

Generation of AM Signal

- **Low level modulation** — In this technique the generation of AM wave takes place in initial stage of ampl^n i.e. at low power level. The generated AM sig is then amplified using no. of amplifier stages.
- **High level modulation** — In high level mod^n the generation of AM wave takes place in the final stage of ampl^n & therefore mod^n circuitry has to handle high power.

Disadvantages of AM (DSB-C)

- The carrier of AM wave does not convey any information and we know

$$P_T = P_c + \frac{m^2}{4} P_c + \frac{m^2}{4} P_c$$

In AM for 100% modulation about 67% of the total power is consumed in transmitting carrier.

→ So power is wastage

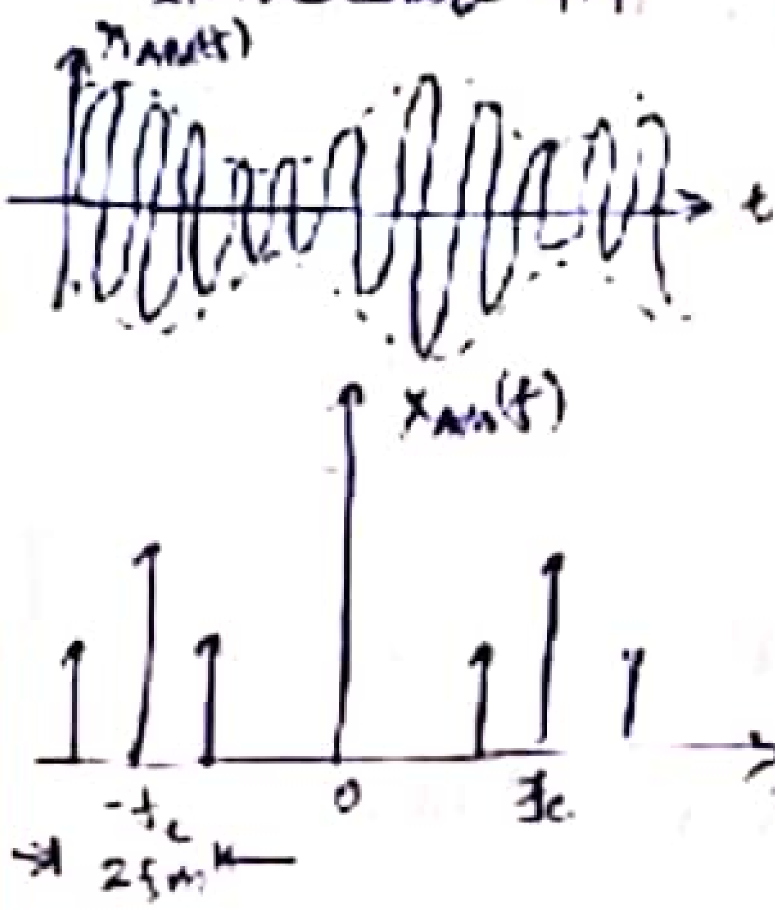
- We know that two side bands are mirror image of each other and all the infn can be conveyed by one of the sideband only.

→ Hence DSB-C system is BW inefficient system.

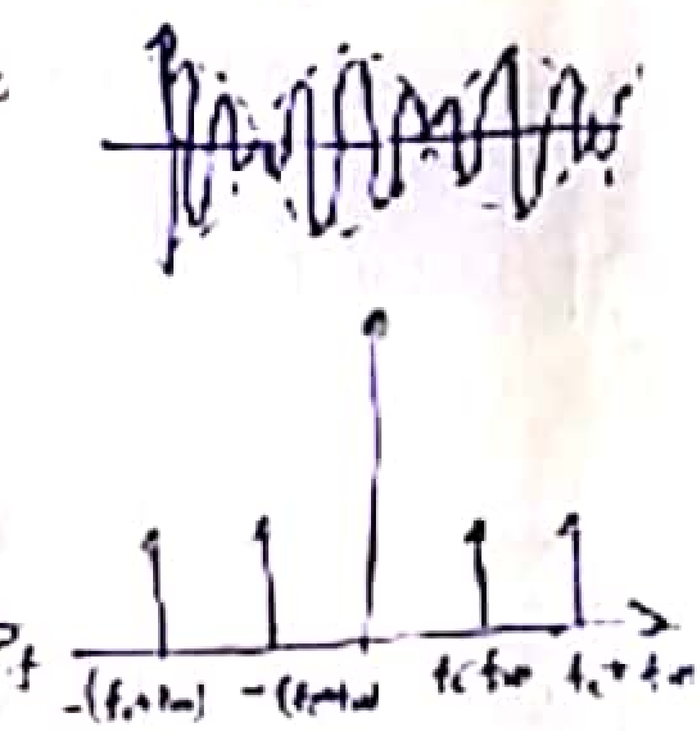
AM

- DSB-C
- DSB-SC
- SSB-SC
- VSB

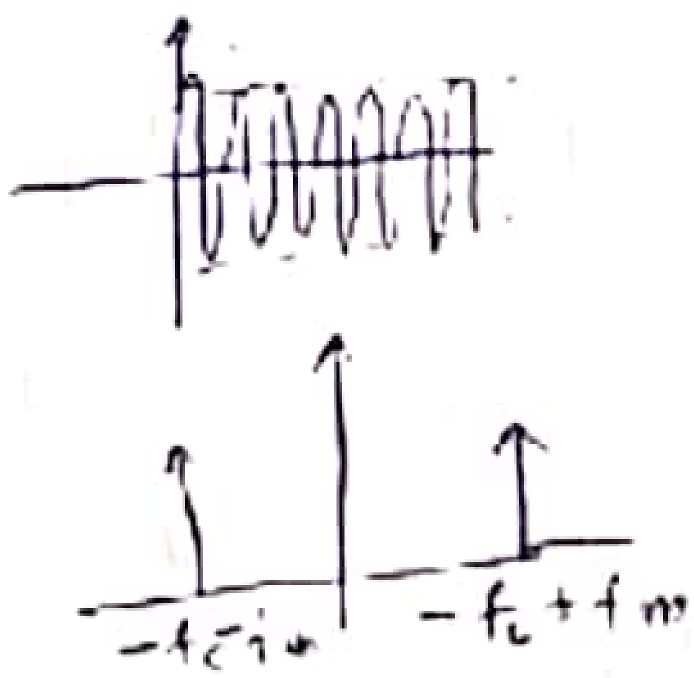
DSB-C: • Standard AM



DSB-SC: • carrier is suppressed

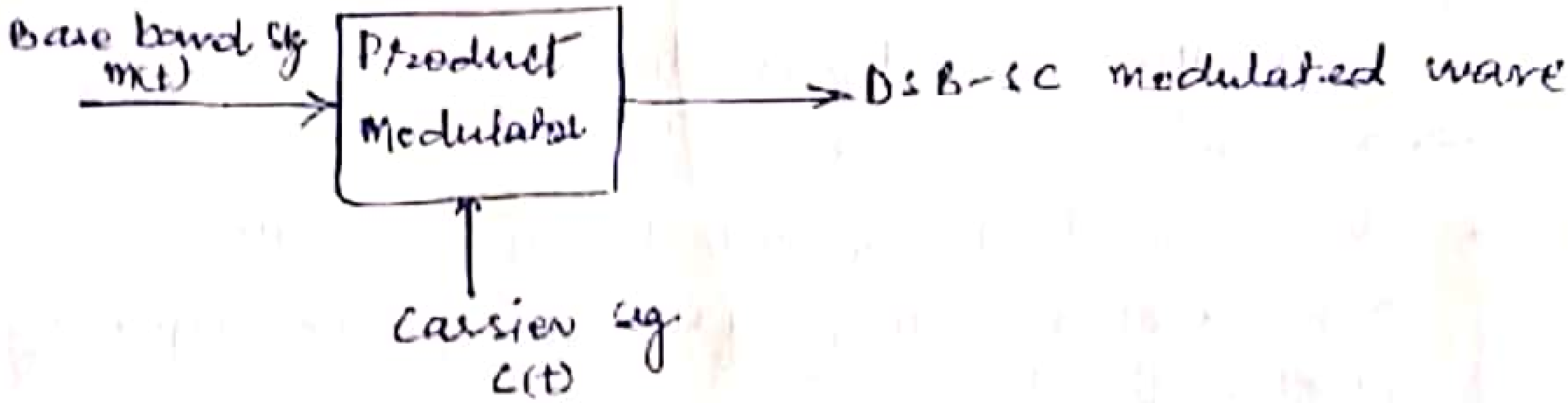


SSB-SC: • carrier + one side band is suppressed



DSB-SC → In this modulation, only upper side band and lower side bands are transmitted.

→ We can obtain DSB-SC modulation by taking the product of the carrier c(t) and modulating sig m(t) which is known as product modulator.



if $\begin{cases} c(t) = A_c \cos 2\pi f_c t \\ m(t) = A_m \cos 2\pi f_m t \end{cases}$

then, $X_{DSB-SC}(t) = c(t) \cdot m(t)$

$= (A_c \cos 2\pi f_c t) \times (A_m \cos 2\pi f_m t)$

$= A_c A_m \cos 2\pi f_c t \cdot \cos 2\pi f_m t$

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

$(\because 2\cos A \cos B = \cos(A+B) + \cos(A-B))$

So, $\boxed{X_{DSB-SC}(t) = \frac{A_c A_m}{2} \left\{ \cos 2\pi (f_c + f_m) t + \cos (f_c - f_m) t \right\}}$

We can see from this eqn that in DSB-SC there is only upper and lower side bands and carrier is absent.

frequency domain description of DSB-SC

$$x_{DSB-SC}(t) = m(t) \cdot c(t)$$

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

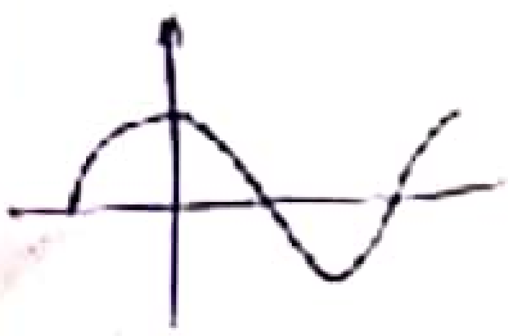
↓ F.T.

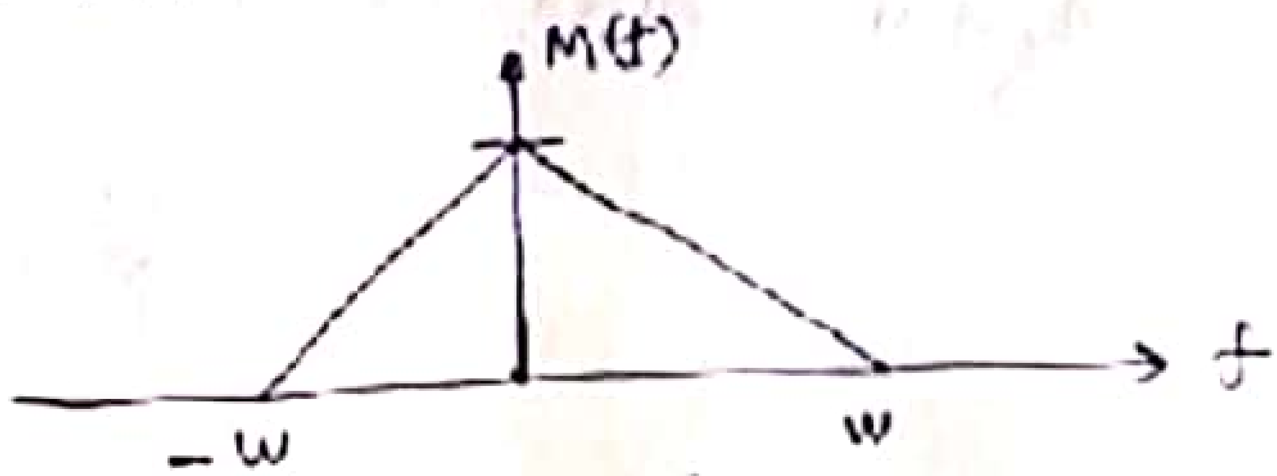
$$X_{DSB-SC}(f) = F.T. \left\{ A_c\, m(t) \cos 2\pi f_c t \right\}$$

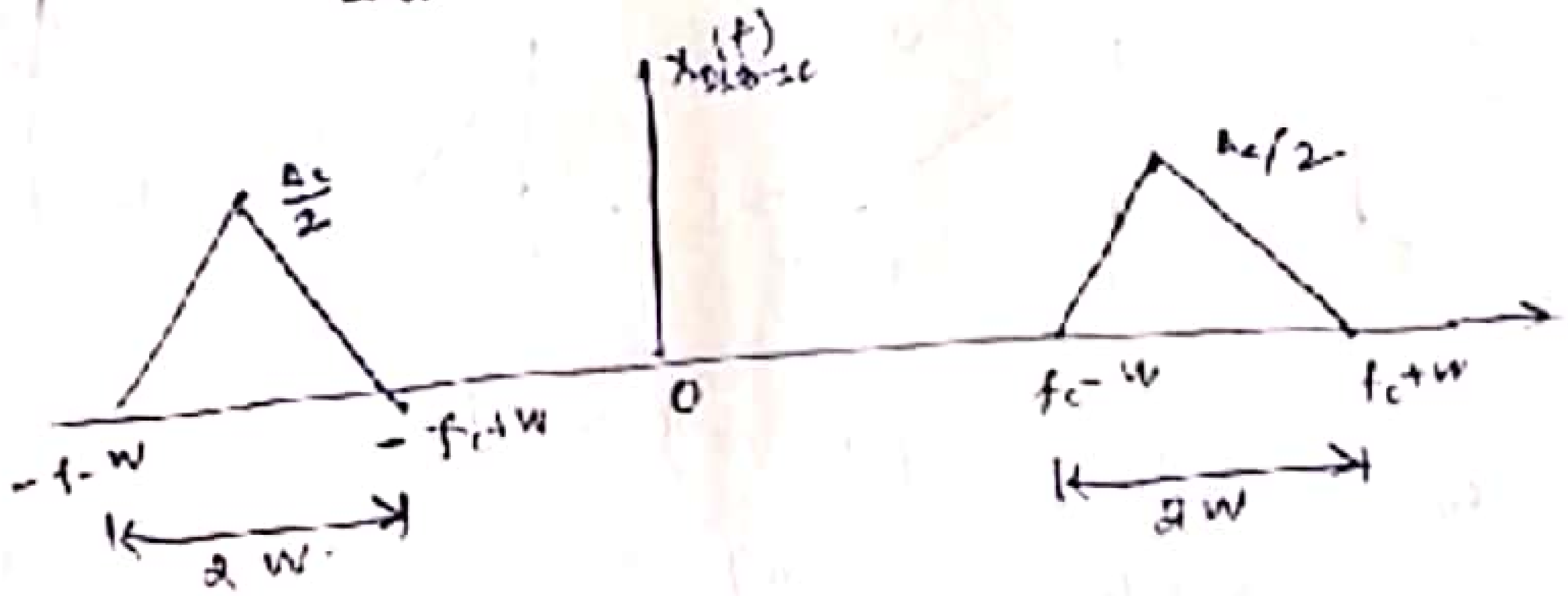
$$= A_c \left\{ m(t) \left(\frac{e^{j2\pi f_c t} + e^{-j2\pi f_c t}}{2} \right) \right\}$$

$$= \frac{A_c}{2} FT\left[m(t)\, e^{j2\pi f_c t} \right] + \frac{A_c}{2} FT\left[m(t)\, e^{-j2\pi f_c t} \right]$$

$$X_{DSB-SC}(f) = \frac{A_c}{2} M(f - f_c) + \frac{A_c}{2} M(f + f_c)$$







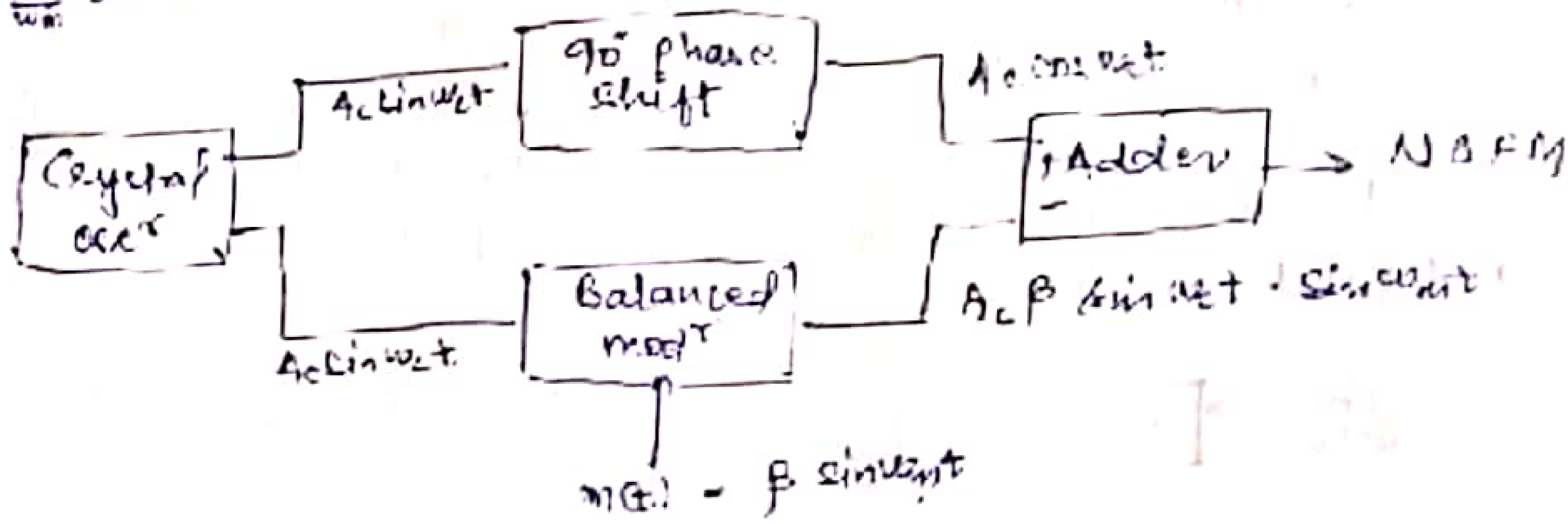
Transmission BW

$$B = (f_c + W) - (f_c - W) = 2W \text{ Hz}.$$

→ Thus the transmission BW of DSB-SC is same as that of the standard AM.

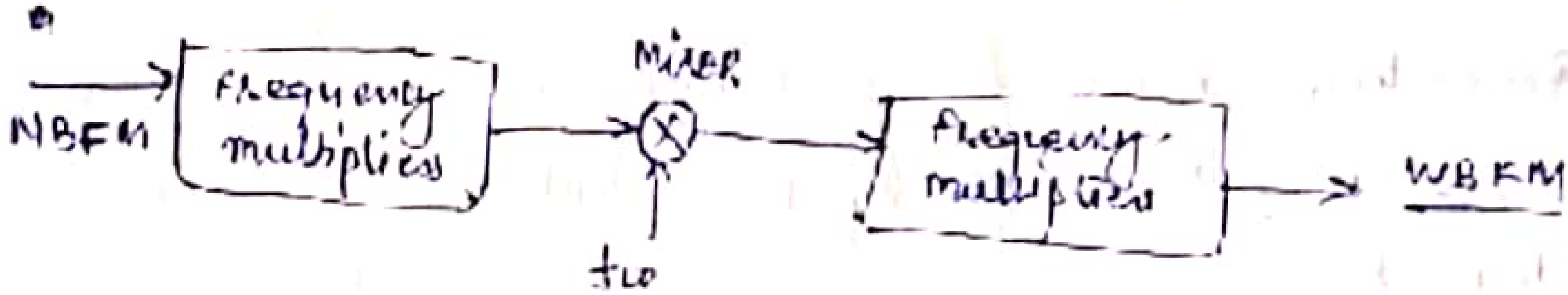
$m(t) = A_m \cos w_m t$

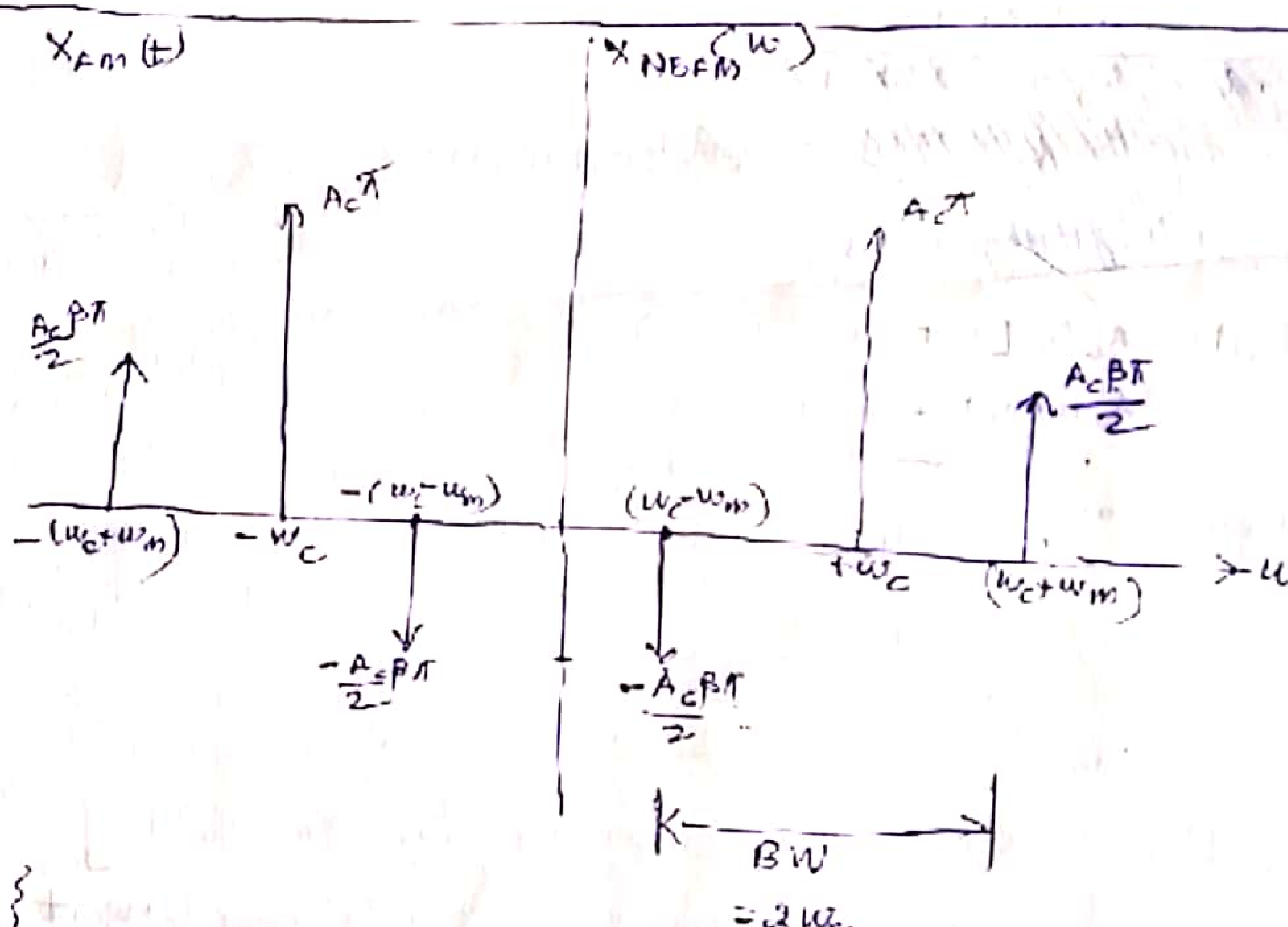
$= \frac{A_m}{w_m} \sin w_m t$



$$y_{NBFM}(t) = A_c \cos w_c t + A_c \beta \sin w_c t \sin w_m t$$

This expression represents a narrow band FM





Operation

→ SSB signal can be generated by assuming two separate simultaneous DSB modulation and combining them suitably depending on the desired sideband.

→ The incoming msg sig m(t) and o/p of carrier osc^n are applied to the balanced modulator 'A'. This balanced modulator produces a DSB-SC wave that translates the frq spectrum of m(t) symmetrically spaced about the carrier frq f_c.

→ The Hilbert transform of m(t) i.e. $\hat{m}(t)$ is applied to second balanced modulator, producing a DSB-SC wave that contains sidebands having identical amplitude spectrum as those of modulator 'A' but of a different relative phase

$$\text{Adder o/p} = m(t) A_c \cos\omega_c t \pm \hat{m}(t) \sin\omega_c t$$

In this way both forms of SSB wave can be generated.

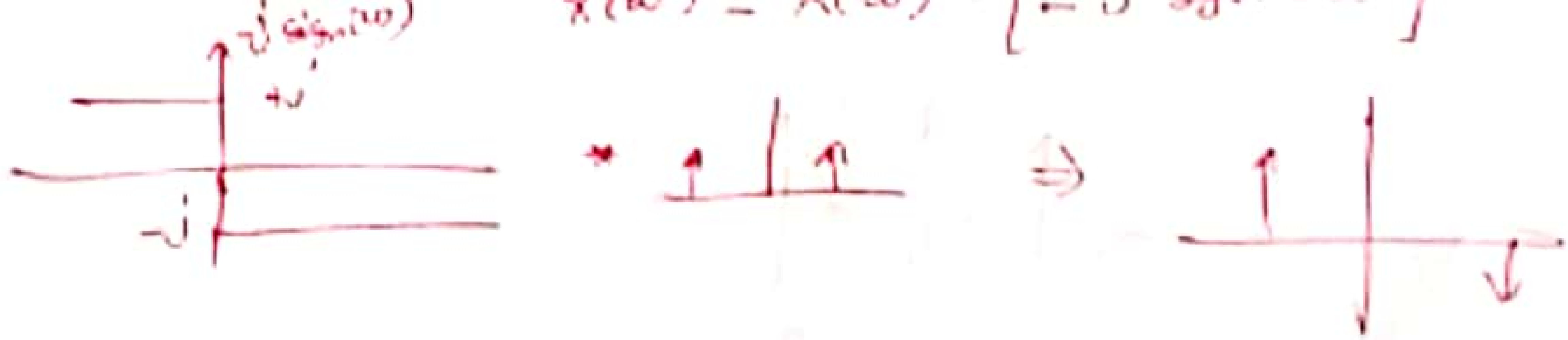
⇒ Hilbert T/F is the method of seperating the sig w.r.t. respect to their phase contents & not with their frq.

⇒ Hilbert transform seperates the sig with a phase diff of 90° b/w them.

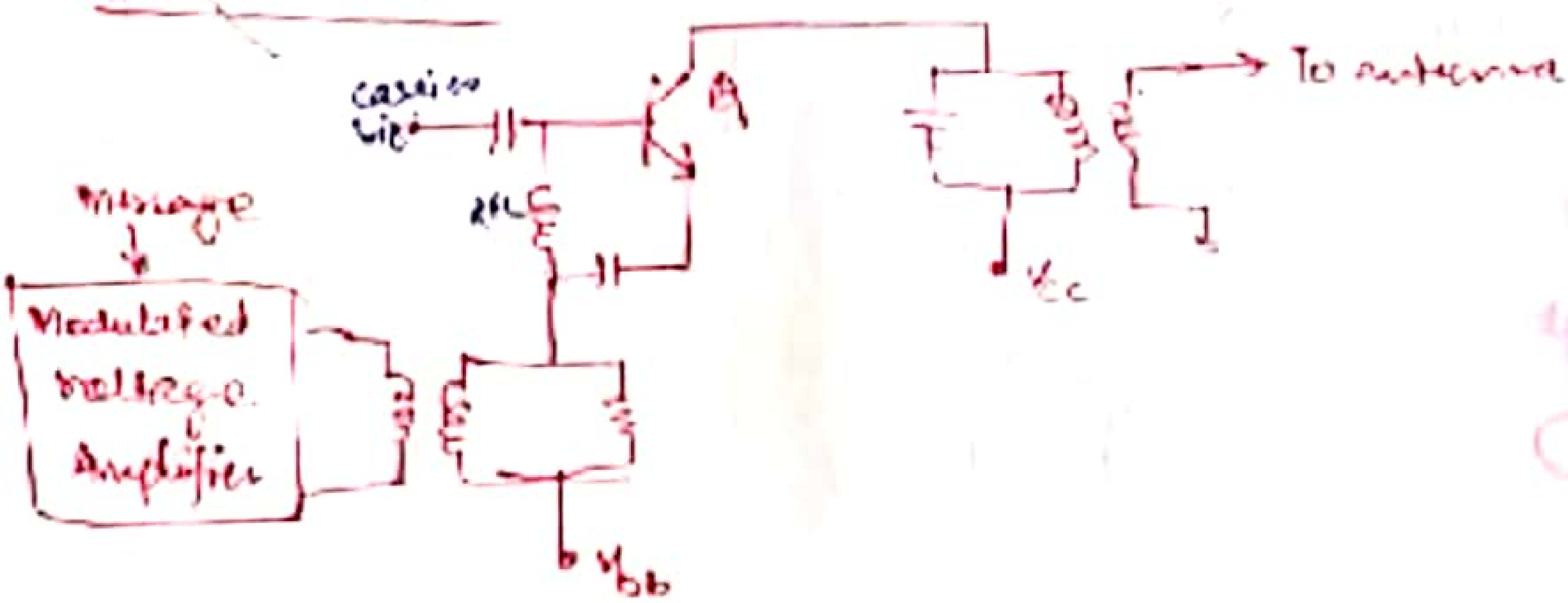
$x(t) \rightarrow sig \Rightarrow \hat{x}(t) = x(t) * \frac{1}{\pi(t)}$

$$= \int_{-\infty}^{\infty} \frac{x(\tau)}{\pi(t-\tau)} d\tau$$

$$\hat{X}(\omega) = X(\omega) \cdot [-j \, \text{sgn}(\omega)]$$



Base Modulation



→ In this carrier signal is also applied at base.

→ RFC is used to isolate high freq signal with audio T/F.

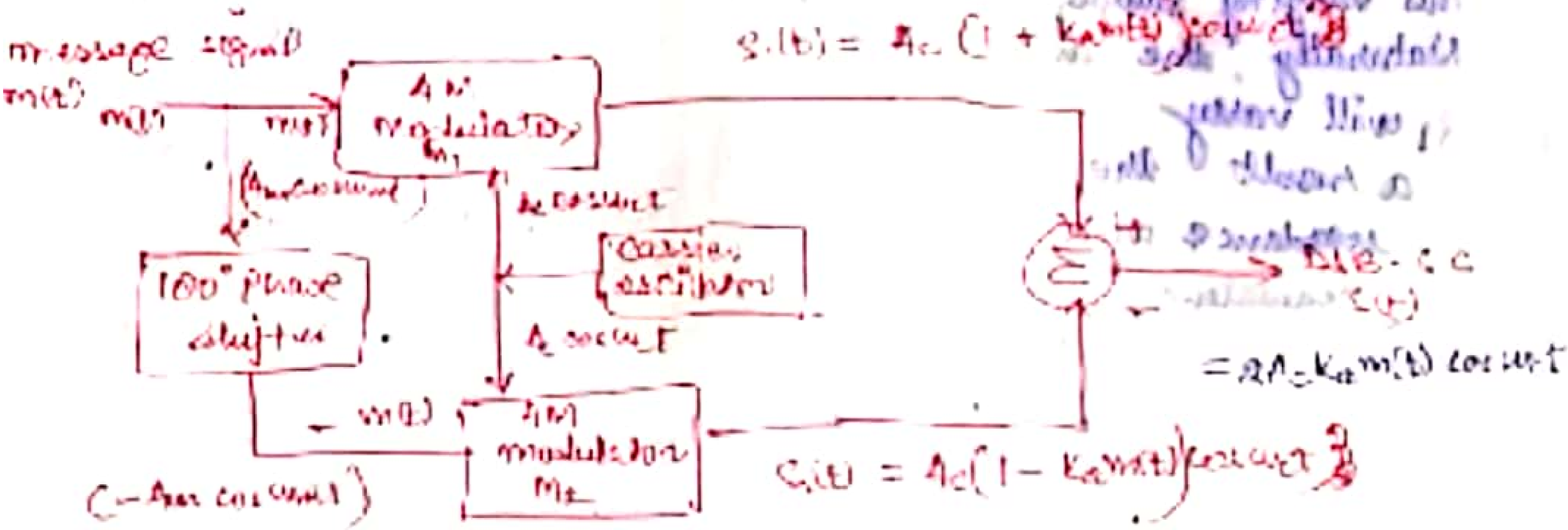
→ Modulation is done at base. At collector amplified o/p is collected.

→ Since amplification process is done after modulation so it is a low level modulation system.

Balanced Modulator

→ A DSB-SC wave is basically the multiplication of the modulating sig m(t) with carrier wave c(t), so we need to use a device called product modulator.

① Balanced Modulator ② Ring Modulator

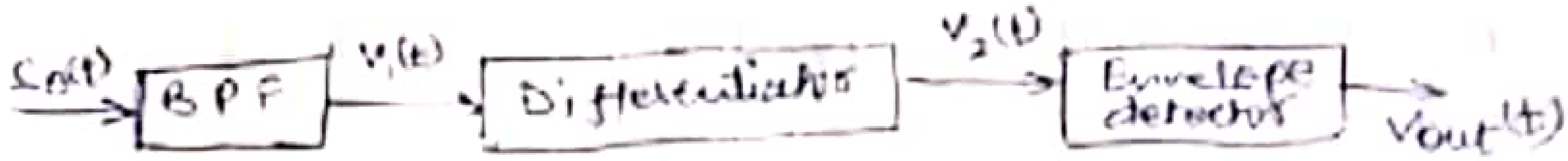


$s_1(t) = A_c(1 + k_a m(t))\cos\omega_c t$

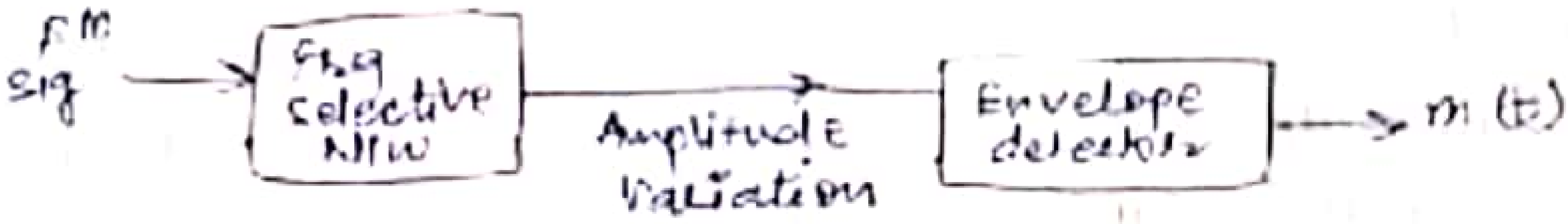
$s(t) = 2A_c k_a m(t)\cos\omega_c t$

$s_2(t) = A_c(1 - k_a m(t))\cos\omega_c t$

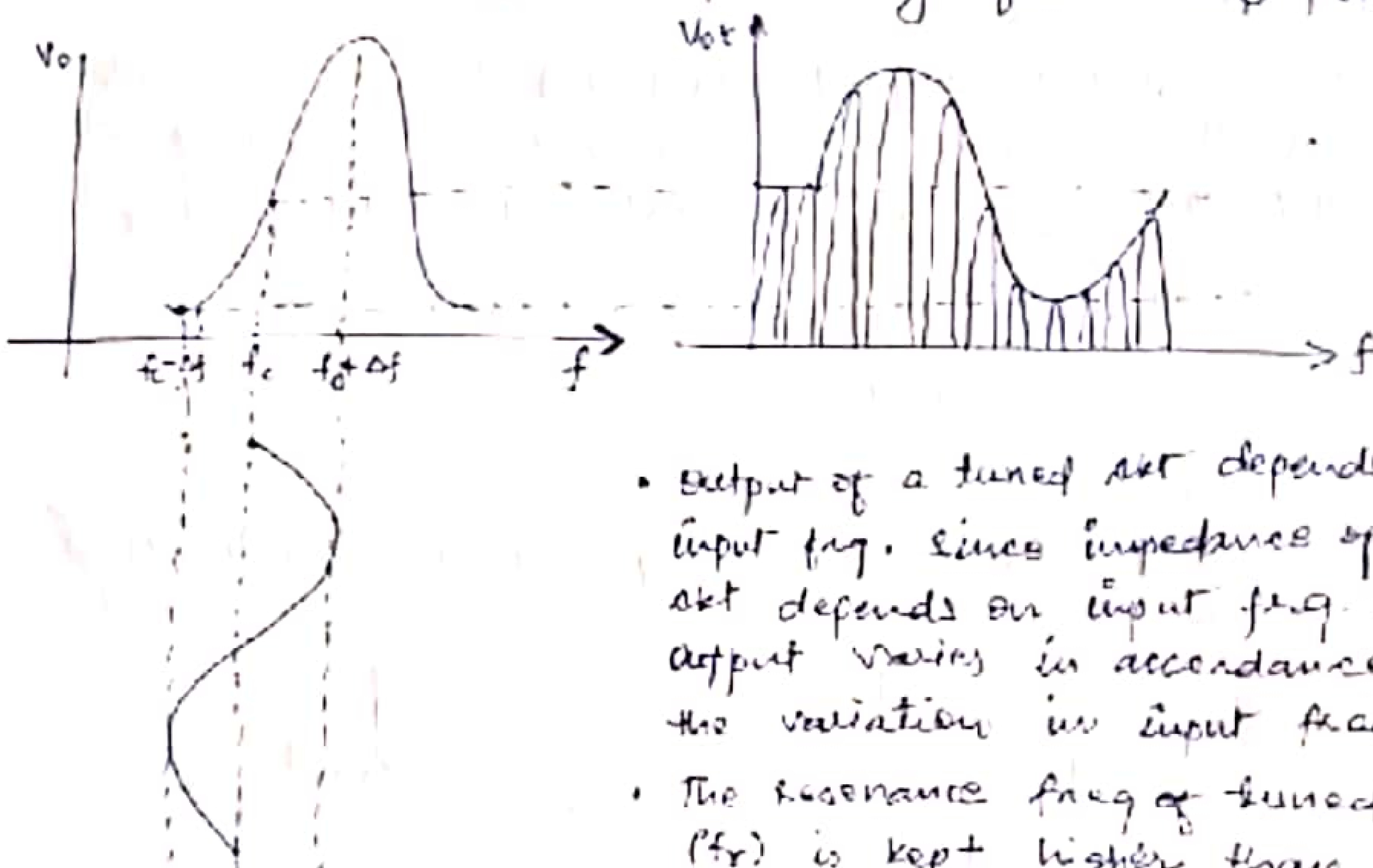
Thus, except for a scaling factor $2k_a$, the o/p of the balanced modulator is equal to the product of the modulating sig m(t) and the carrier wave c(t). It is the DSB-SC signal.



→

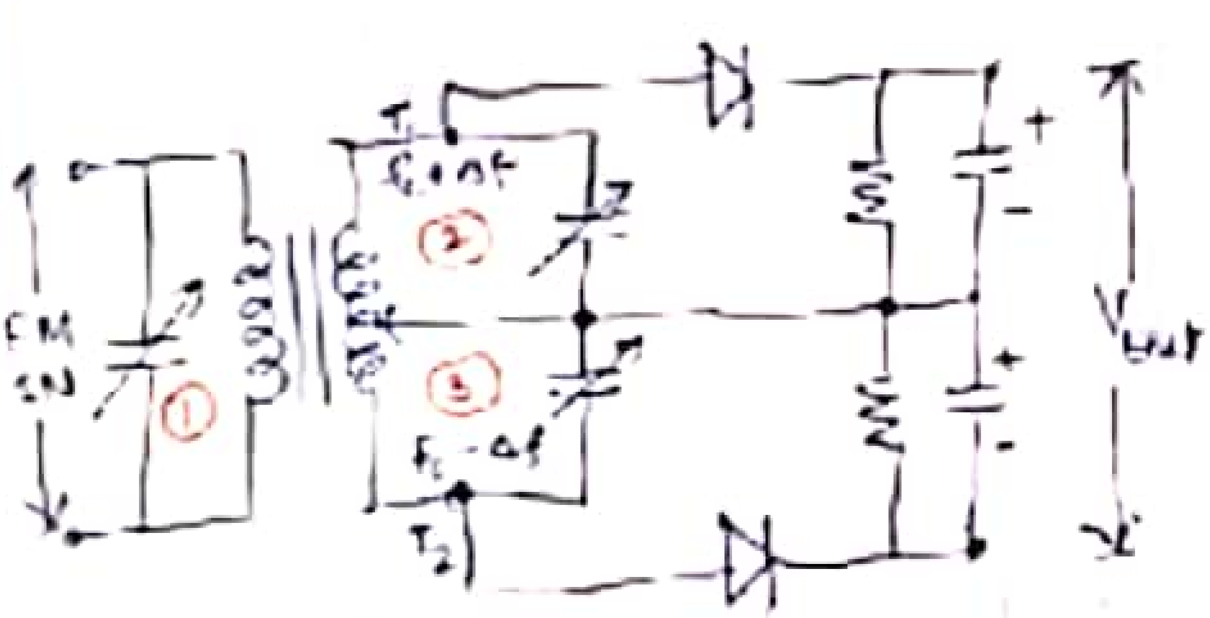


→ In slope detection a series RLC resonant ckt is used as freq selective n/w. The resonant freq of the RLC n/w is kept at $(f_c + \Delta f)$. The 3db point is kept at f_c & $(f_c - \Delta f)$ is kept away from 3db point.

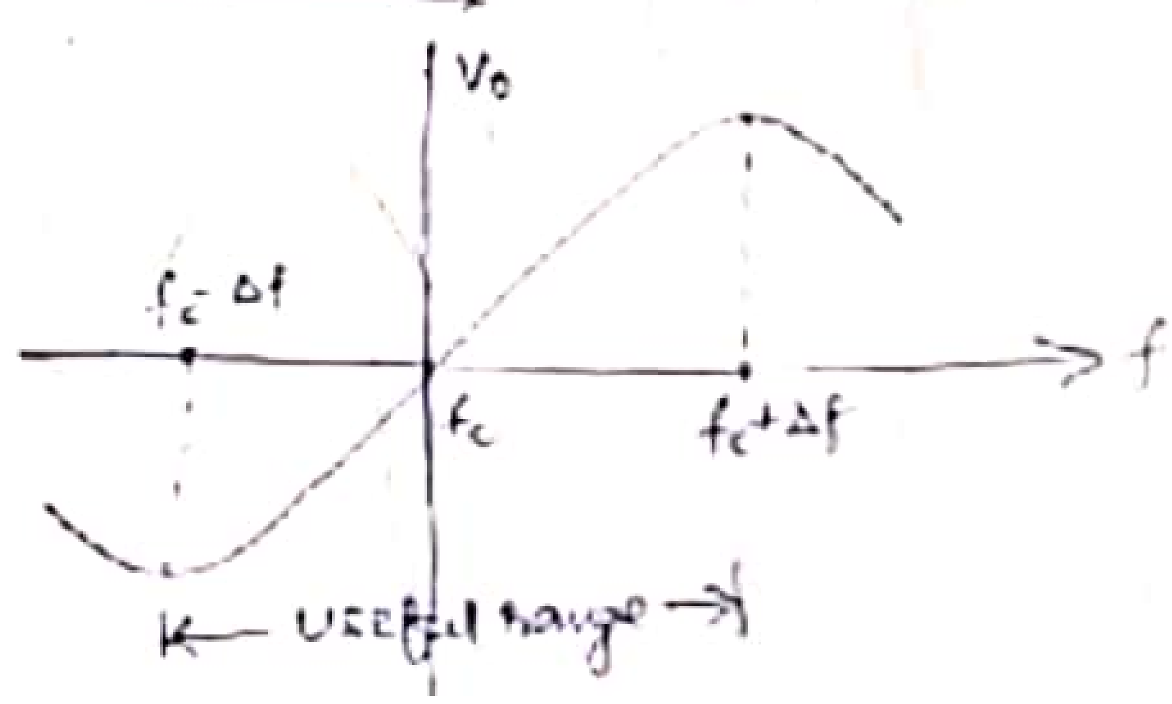


- Output of a tuned ckt depends on input freq. Since impedance of tuned ckt depends on input freq so the output varies in accordance with the variation in input frequency.
- The resonance freq of tuned ckt (f_r) is kept higher than the carrier freq 'f_c'.

$$f_r > f_c$$

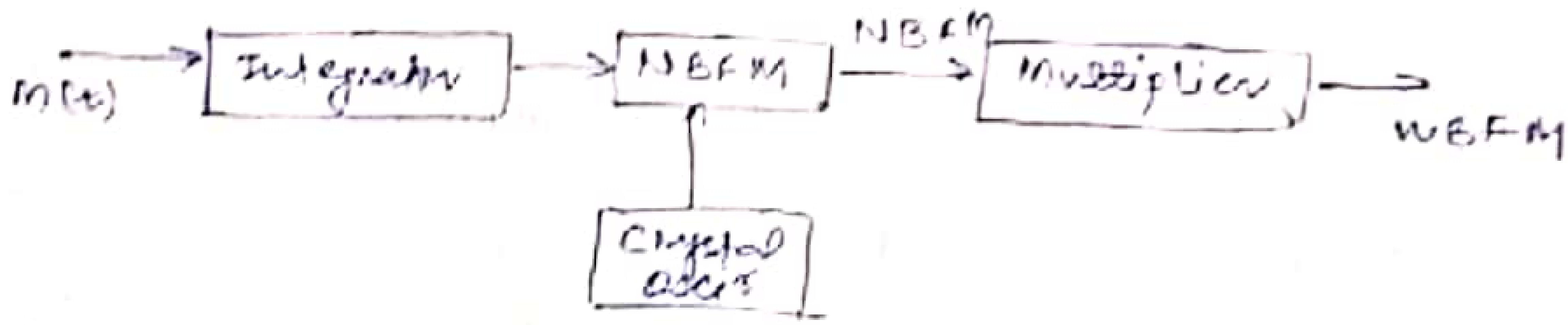


- Triple Tuned ckt



Indirect method (Armstrong Method) of FM Generation

Most LC osc^n are not stable enough to provide a carrier frq. The carrier frq usually vary due to temp. variation, humidity and aging effect so instead of using LC osc a crystal osc^n must be use.



→ In this method the FM is obtained through phase modulation
→ The operation of Armstrong method can be understood by dividing it into two parts as follows:

Part 1: Generating a narrow band FM using a phase modulator

Part 2: Using the Mixer and frq multipliers to obtain the desired values of frq deviation, modulation index and carrier.

$$X_{FM}(t) = A_c \cos[\omega_c t + \beta \sin \omega_m t]$$

$$= A_c \cos \omega_c t \cdot \cos \beta \sin \omega_m t - A_c \sin \omega_c t \sin \beta \sin \omega_m t \quad \text{--- (1)}$$

NBFM: when (β is very small)

$\cos\theta = 1$ when $\theta \to 0$; $\sin\theta = \theta$ when $\theta \to 0$

$\cos \beta \sin \omega_m t = 1$
$\sin \beta \sin \omega_m t = \beta \sin \omega_m t$ $\} \Rightarrow \beta \to 0$

$$x_{NBFM}(t) = A_c \cos \omega_c t - A_c \sin \omega_c t \, \beta \sin \omega_m t$$

$$X_{NBFM}(t) = A_c \cos \omega_c t - \frac{A_c \beta}{2}\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right]$$

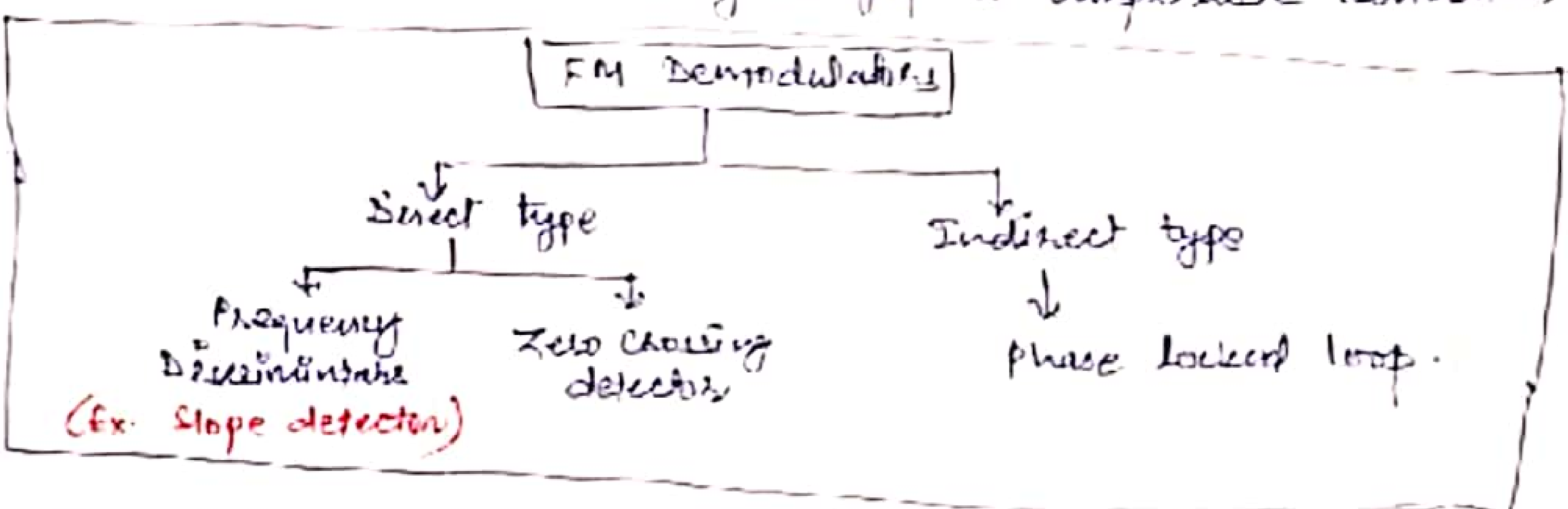
$$= A_c \cos \omega_c t - \frac{A_c \beta}{2} \cos(\omega_c - \omega_m)t + \frac{A_c \beta}{2} \cos(\omega_c + \omega_m)t$$

Demodulation of FM waves

→ Freq demodulation is the process that enables us to extract or recover the original modulating signal from freq modulated wave.

→ Demodulation of an FM wave requires a system that produces an O/P proportional to the instantaneous freq deviation of i/p this is called freq discriminator.

→ FM demodulator is basically a freq to amplitude converter.



→ There are many ways to build FM demodulator but almost all of them are based on one of three principle.

1. FM to AM conversion (slope detector
2. Phase shift or quadrature detection (Foster Seely)
3. Zero crossing detection

Let $m(t) = A_m \cos \omega_m(t)$

→ then in PM

$\phi(t) = K_p\, m(t) = K_p A_m \cos \omega_m t$

⇓

Modulation index $\beta_{PM} = K_p A_m$

→ For FM, $\phi(t) = K_f \int m(t)dt = K_f \int A_m \cos \omega_m(t)\, dt$

$= \frac{K_f A_m}{\omega_m} \sin \omega_m(t)$

⇓

Modulation index $\beta_{FM} = \frac{K_f A_m}{\omega_m}$

So,

$$\beta = \begin{cases} K_p A_m & \text{for PM} \\ \frac{K_p A_m}{\omega_m} & \text{for FM} \end{cases}$$

⇒ Modulation index or Max^m value of phase deviation

Important points about β

- In FM β decides the BW of FM.
- β_{FM} can be greater than 1
- β also decides no. of side bands having significant amplitude

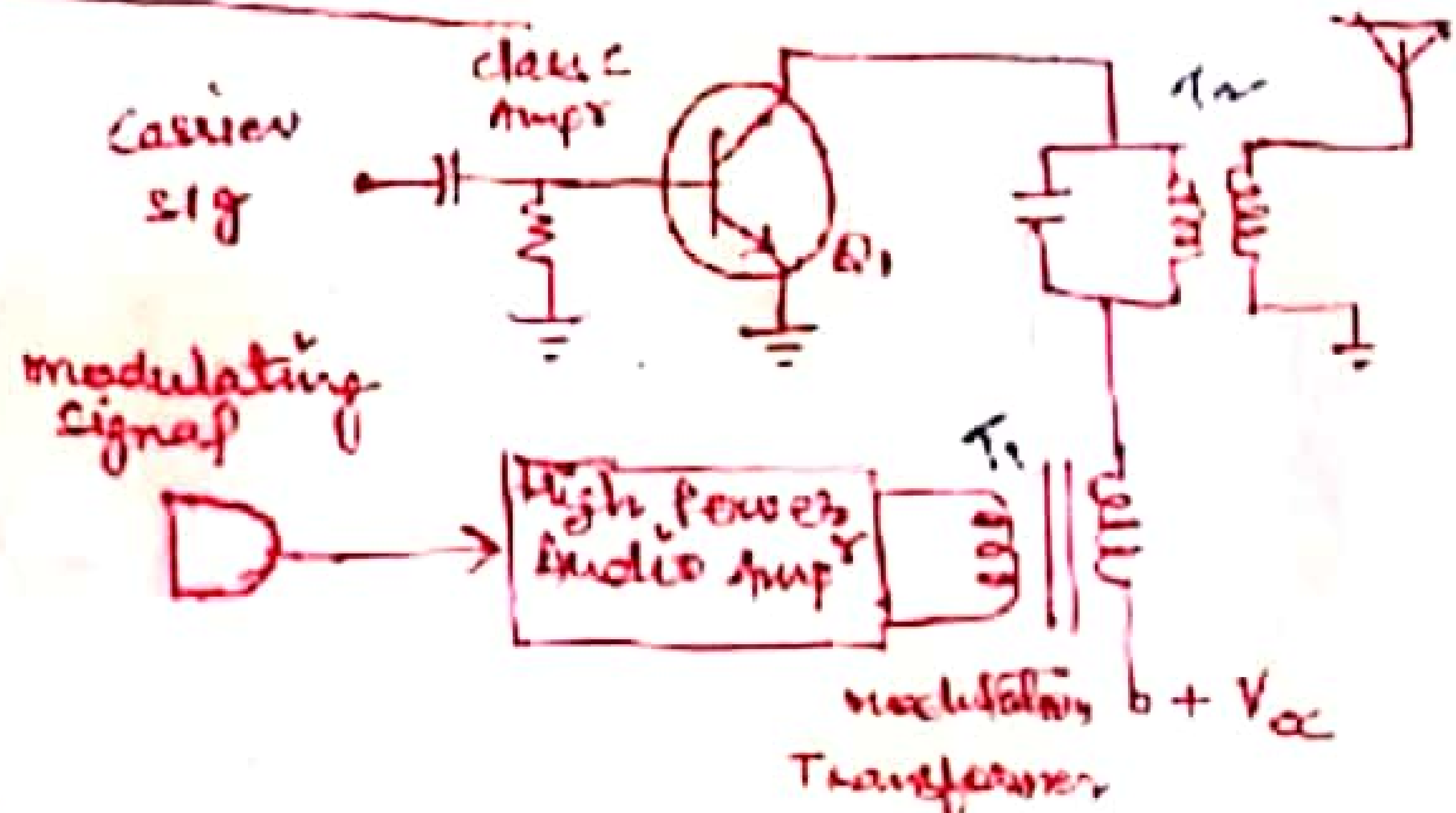
Deviation Ratio

→ The modulation index corresponding to the max deviation and max freq is called as deviation ratio.

$$D = \frac{\text{Max}^m\ \text{deviation}}{\text{Max}^m\ \text{modulating freq.}} = \frac{\Delta\omega}{\omega_m}$$

$$D = \frac{\Delta\omega}{\omega_m}$$

Collector Modulator



→ The modulator is a linear power Ampr that takes the low level modulating signal and amplifies it to a high power level. The modulating o/p signal is coupled through modulation T/F T_1 to the class C Ampr. The secondary winding of the modulation T/F is connected in series with the collector supply voltage V_{cc} of the class C Ampr.

→ With zero modulation input signal, there will be zero modulⁿ voltage across the secondary of T_1. Therefore, the collector supply voltage will be applied directly to the class C Ampr and the o/p carrier will be a steady sine wave.

→ When the modulation signal occurs, the AC voltage across the secondary of the modulation T/F will be added to and substracted from the collector supply voltage.

→ This varying supply voltage is applied to the class C Ampr, Naturally the amplitude of the current pulses through Transistor Q_1 will vary.

→ As a result the amplitude of the carrier sine wave varies in accordance with the modulated signal. For example, when the modulating signal goes positive, it adds to the collector supply voltage, thereby increasing its value and causing higher current pulses and a higher amplitude carrier. When the modulating signal goes negative, it substracts from the collector supply voltage making it less. For that reason, the class C ampr current pulses are smaller, thereby causing a lower amplitude carrier o/p.

→ Hence Amplitude modulated wave is obtained which is then transmitted through antenna.

Angle Modulation

→ Angle modⁿ is a modulation in which the angle of a sine wave carrier is varied in accordance with the infⁿ contained in the modulating sig. while amplitude is constt.

$$S_\theta(t) = A_c \cos[\omega_c t + \phi(t)]$$

↳ is a fⁿ of msg s/g

→ Angle Modⁿ → FM (Freq. of carrier is varied)
→ PM (Phase angle of carrier is varied)

→ Angle Modⁿ is being used for:

1. cellular radio
2. Radio broadcasting
3. Satellite commⁿ
4. μwave commⁿ
5. TV sound transⁿ

→ For Angle modⁿ $S_\theta(t) = A_c \cos\theta(t)$

where $\theta(t) = \omega_c t + \phi(t)$

$\phi(t)$ → Instantaneous phase deviation

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

$\frac{d}{dt}\phi(t)$ → Instantaneous frq deviation

Demodulation of FM waves

→ Frequency demodulation is the process that enables us to extract or recover the original modulating signal [m(t)] from frequency modulated wave.

→ Demodulation of an FM wave requires a system that produces an output proportional to the instantaneous freq. deviation of the input signal. Such a system is called freq. discriminator.

→ The FM demodulator operates on an altogether different principle than the AM detector. AM detector is basically a rectifier. But FM demodulator is basically a freq to amplitude converter.

→ There are many ways to build FM demodulator but almost all of them are based on one of three principles -

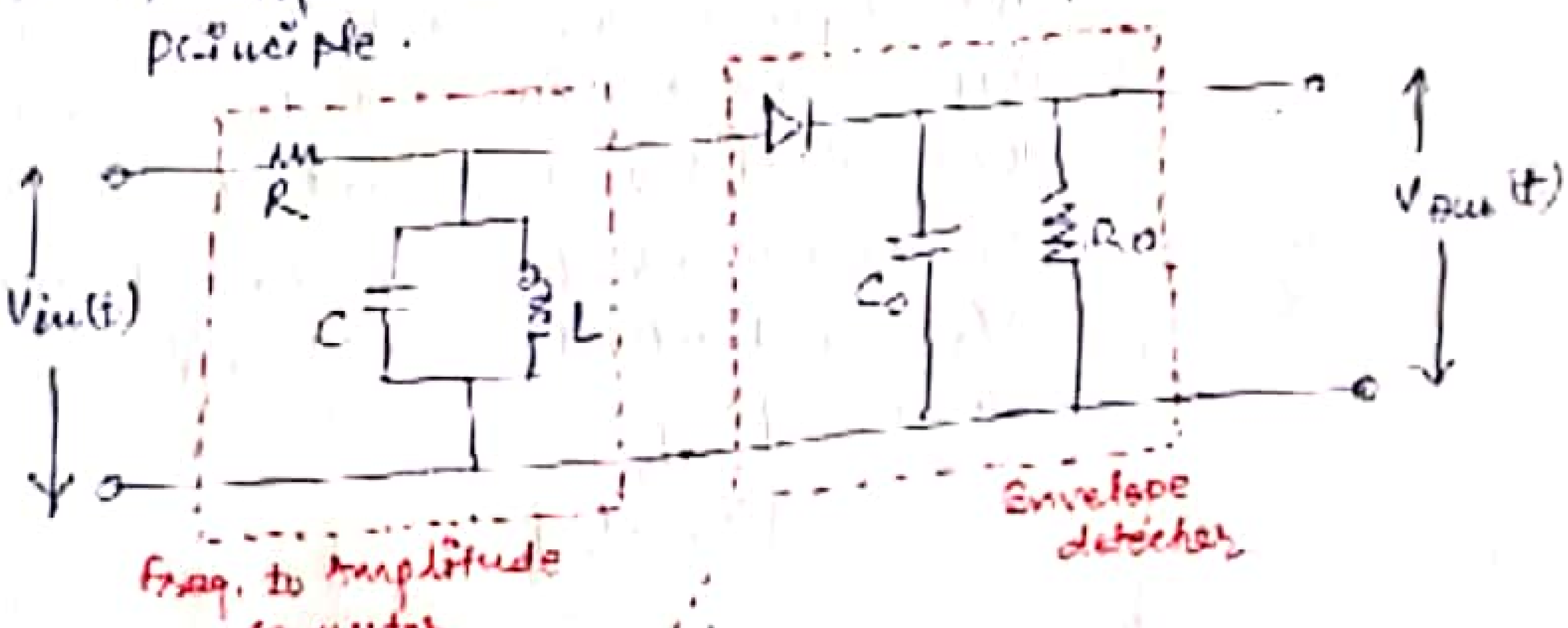
1. FM to AM conversion   Ex: slope detector

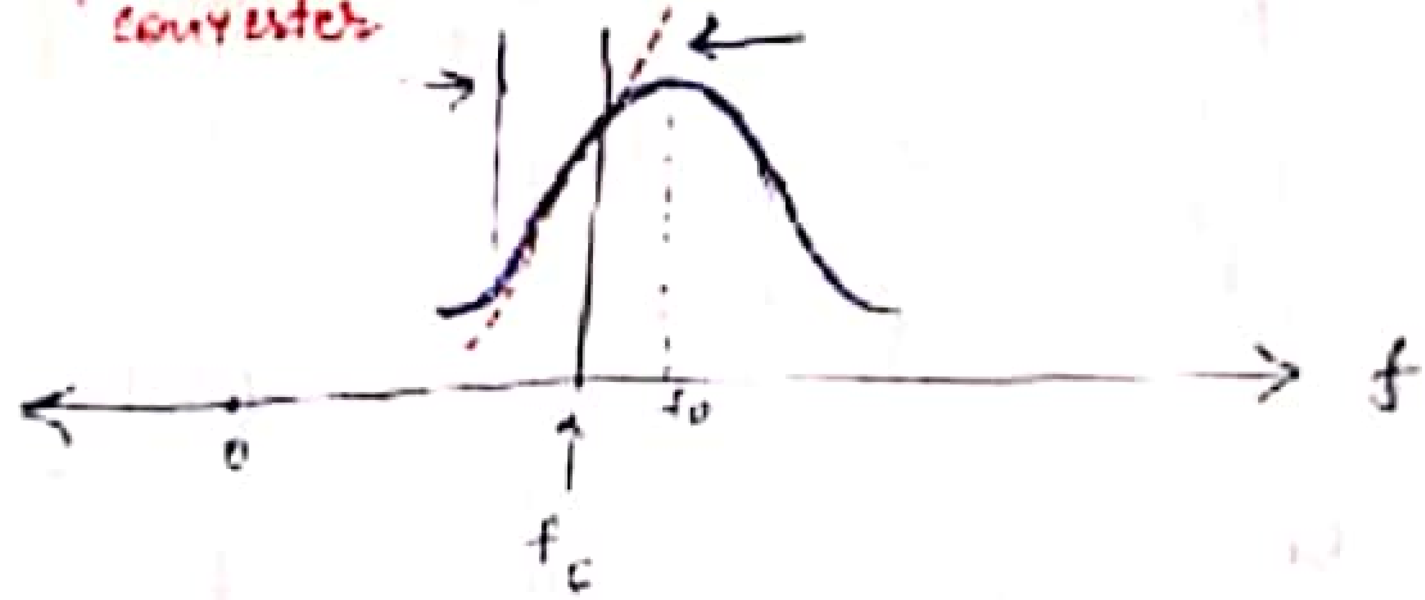
2. Phase shift or quadrature detection

3. Zero crossing detection

Slope detection

→ A slope detector is an example of the FM to AM conversion principle.





Phase Modulation

→ Phase mod^n is a form of modulation that represents in a variations in the instantaneous phase of a carrier wa. Phase mod^n is not very widely used for radio transm^n

→ In PM,

$$\phi(t) \propto m(t)$$ (Phase deviation is proportional to msg sig)

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

$\phi(t)$ → Instantaneous phase deviation; K_p → phase deviation const.; $m(t)$ → msg sig

→ So,

$$S_{PM}(t) = A_c \cos[\omega_c t + K_p\, m(t)]$$

⇓

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

$$\boxed{\omega_i = \frac{d}{dt}\theta(t) = \omega_c + K_p \frac{d}{dt} m(t)}$$

** {Thus in PM, the instantaneous freq ω_i varies linearly with the derivative of msg sig}

Freq Modulation : FM conveys info^n over a carrier wave by varying its instantaneous freq. FM is widely for broadcasting of music and speech and in two way radio system.

→ In FM, the instantaneous freq deviation $\frac{d}{dt}\phi(t)$ is proportional to msg sig. i.e.

$$\frac{d}{dt}\phi(t) \propto m(t)$$

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

$$\Rightarrow \phi(t) = K_f \int_{t_0}^{t} m(t)\, dt$$

K_f → freq deviation const.

→ So,

$$S_{FM}(t) = A_c \cos\left[\omega_c t + K_f \int m(t)\, dt\right]$$

$$\theta(t) = \omega_c t + K_f \int m(t)\, dt$$

$$\boxed{\omega_i = \frac{d}{dt}\theta(t) = \omega_c + K_f\, m(t)}$$

Adv

(1) High sensitivity & selectivity
(2) High adjacent channel rejection
(3) No variation in BW.

Image signal and image signal rejection

→ Superheterodyne receiver suffer from a major drawback known as image freq problem.

→ Image signal is a signal whose freq. is as higher with f_o as the f_{ds} is smaller than the f_o.

If

f_o → local osc^n freq. f_{ds} → desired signal freq.

f_i → intermediate freq.

then, $f_o - f_{ds} = f_i$ —(1) ⇒ { f_{ds} is lesser than f_o with an amount 'f_i'

By definition of image signal

$f_{im} = f_o + f_i$ —(2) f_{im} → image freq

$= f_i + f_{ds} + f_i$

$= f_{ds} + 2f_i$ —(3) → from this eq^n it is clear that image signal is $2f_i$ higher than with desired signal

* If this signal is reached to antenna mixer by antenna than it will produce same intermediate freq 'f_i' which will be produced by desired signal f_{ds}.

$f_i = f_o - f_{ds}$ (If produced by f_{ds}) $f_i = f_{im} - f_o$ (If produced by image freq.)

$f_i = f_{im} - f_o$

$= f_{ds} + 2f_i - f_o = f_{ds} + 2(f_o - f_{ds}) - f_o$

$= f_{ds} + 2f_o - 2f_{ds} - f_o$

$= f_o - f_{ds}$ —(4)

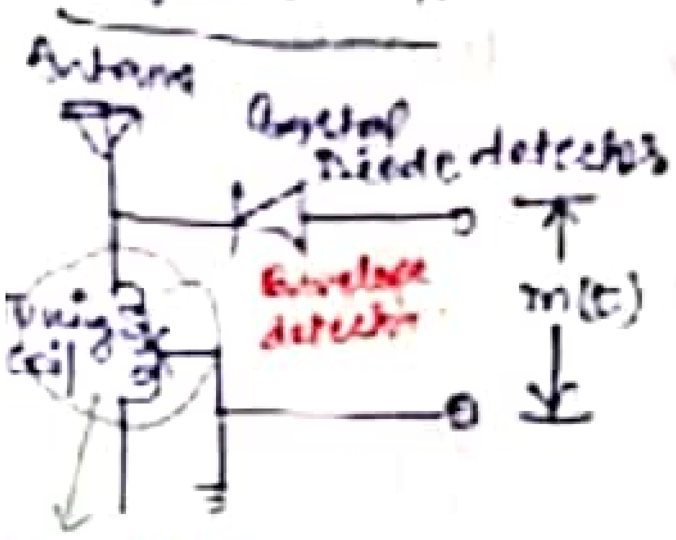
Acc^n to eq^n (1) & eq^n (4) desired signal and image signal produce same IF which will further amplify by IF amp^r and interfere with the signal. so the image signal must be rejected. which depends on selectivity

$\alpha = \sqrt{1 + Q^2 \rho^2}$ ⇒ Image freq. rejection ratio where $\rho = \frac{f_{im}}{f_{ds}} - \frac{f_{ds}}{f_{im}}$

Types of Radio Receivers

- (1) Crystal radio Rx
- (2) Tuned radio freq. Rx
- (3) Super heterodyne radio Rx

Crystal Rx

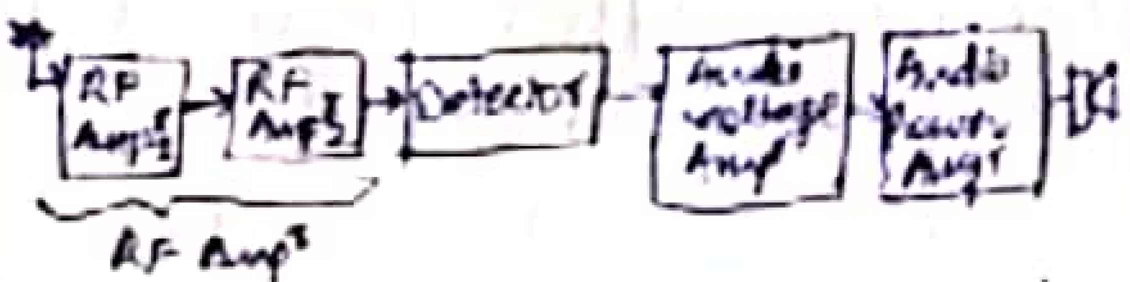


Defined freq can be selected by changing turns of coil

→ Selectivity & sensitivity is very poor

→ Ampr is not used so weak demodulated signal

TRF Rx



→ Good only for medium wave freq but poor performance at higher freq.

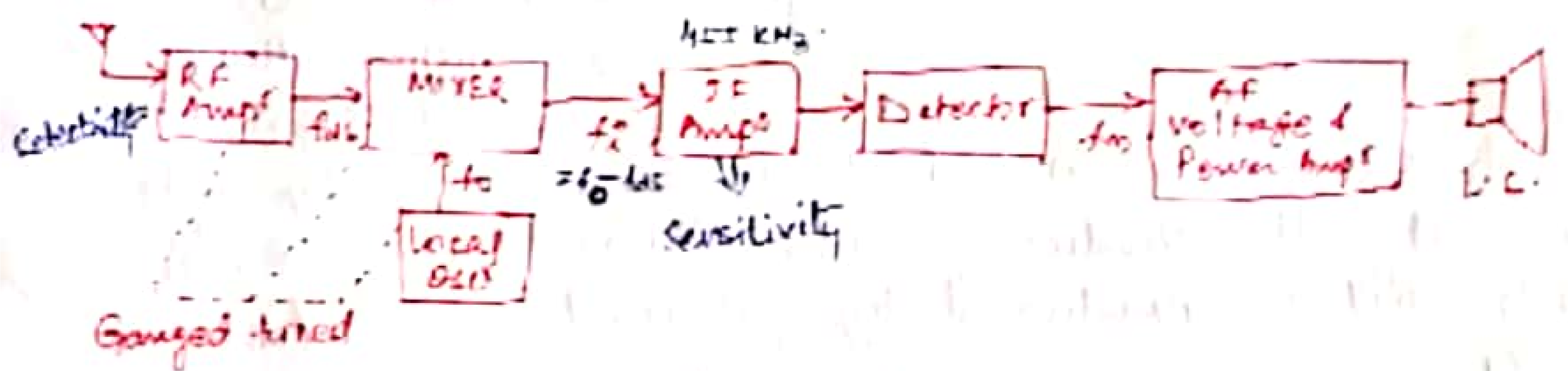
→ Instable due to multiple RF ampr. variation in B.W. Poor selectivity

→ simple ckt & good sensitivity.

SHRR

A superheterodyne Rx, often shortened to super het, is a type of Radio Rx that uses freq mixing to convert a received signal to a fixed Intermediate freq which can be more conveniently processed than the original carrier freq.

Super Heterodyne radio Receiver



$$f_o = f_s + f_i$$

$$f_i = f_o - f_s \rightarrow \text{is selected}$$

Working

→ The received signal is amplified by rf ampr which is a tuned Ampr ckt. Amplified signal is passed on to a mixer. here rf carrier is mixed with f_o generated by local oscr. The process of mixing is called heterodyning. f_o is selected to be above f_s.

→ Mixer generates sum and difference of freq $(f_o + f_s)$ and $(f_o - f_s)$. $(f_o + f_s)$ eliminated by a filter. The difference freq $(f_o - f_s)$ is called intermediate freq carrier.

Operation

→ SSB signal can be generated by assuming two separate simultaneous DSB modulation and combining them suitably depending on the desired sideband.

→ The incoming msg sig $m(t)$ and o/p of carrier osc^r are applied to the balanced modulator 'A'. this balanced modulator produces a DSB-SC wave that translates the frq spectrum of $m(t)$ symmetrically spaced about the carrier frq f_c.

→ The Hilbert transform of $m(t)$ i.e. $\hat{m}(t)$ is applied to second balanced modulator, producing a DSB-SC wave that contains sidebands having identical amplitude spectrum as those of modulator 'A' but of a different relative phase

$$\text{Adder o/p} = m(t)\, A_c \cos\omega_c t \pm \hat{m}(t) \sin\omega_c t$$

In this way both forms of SSB wave can be generated

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⇒ Hilbert transform seperates the sig with a phase diff of 90° b/w them.

$x(t) \to \text{sig} \Rightarrow \hat{x}(t) = x(t) * \frac{1}{\pi(t)}$

$$= \int_{-\infty}^{\infty} \frac{x(\tau)}{\pi(t-\tau)}\, d\tau$$

$$\hat{X}(\omega) = X(\omega) \cdot [-j \operatorname{sgn}(\omega)]$$

Comparison AM and Angle Mod^n

Angle modulation has following advantages over AM

(1) The amplitude of FM is constt. It is independent of depth of modulation. Hence Tx power remains constant in FM whereas it varies in AM.

(2) Since amplitude of FM is constt, the noise interference is minimum in FM. Any noise superimposing on amplitude can be removed with the help of amplitude limits, whereas it is difficult to remove amplitude variations due to noise in AM.

(3) The depth of modulation have limitation in AM. But in FM the depth of modulation can be increased to any value by increasing the deviation. This does not cause any distortion in FM sig.

(4) Since guard bands are provided in FM, there is less probability of adjacent channel interference.

(5) Since space waves are used for FM, the radius of prop^n is limited to line of sight. Hence it is possible to operate several independent Tx on same frq with min^m interference.

(6) Since FM uses UHF and VHF ranges, the noise interference is min^m compared to AM which uses MF and HF ranges.

Disadv:

(1) BW requirement is much higher than AM.

(2) The FM transmitting & receiving equipment is more complex and costly.

(3) Since FM uses UHF & VHF range of frq, its area of reception is limited only to line of sight. This is much lower than area covered by AM.