UNIT-1

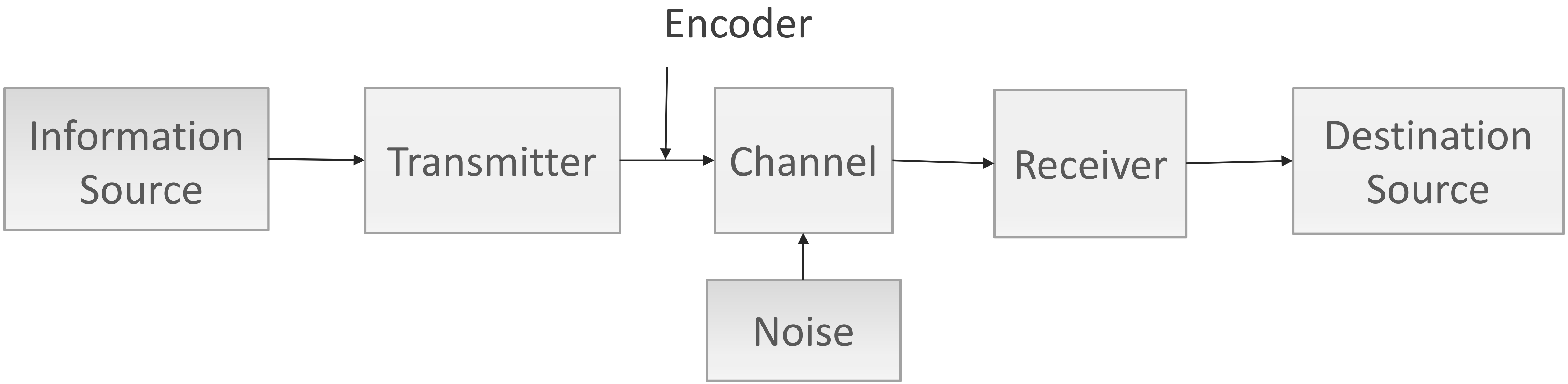
AMPLITUDE MODULATION

CONTENT

- Need for modulation
- Amplitude modulation
- Time and Frequency domain description
- Single tone modulation
- Power relations in AM waves
- Generation of AM waves
- Switching modulator
- Detection of AM waves
- Envelope detector
- DSBSC modulation
- Time and frequency domain description
- Generation of DSBSC waves
- COSTAS Loop
- SSB modulation
- Time and Frequency domain description
- Frequency discrimination and phase discrimination methods for generating SSB
- Demodulation of SSB waves
- Principle of Vestigial side band modulation

BLOCK DIAGRAM

- <u>Basic Element of Communication</u>



- The above block diagram is basic element of communication.
- The communication is transmitting/transfer the data to the source to destination

MODULATION

- Modulation is a simply widely used process in the communication system.
- When a very high frequency carrier signal is used to transmit very low frequency message signal, So that the transmitted signal continuous to have all the information contained in the original message signal.

TWO NEEDS FOR MODULATION

- In the modulation process two signals are used namely,
  1. Modulating signal M(t)
  2. Carrier signal C(t)

- Modulating signal is nothing but have base band signal (or) Information source.
- Where as the carrier signal is nothing but a very high frequency signal.

- **Need for modulation:-**
- we know that the base band signals are in Compatible for the direct transmission to the medium and therefore , we have to use modulation techniques for the communication of base band signal.
- The Advantages of using modulation techniques are

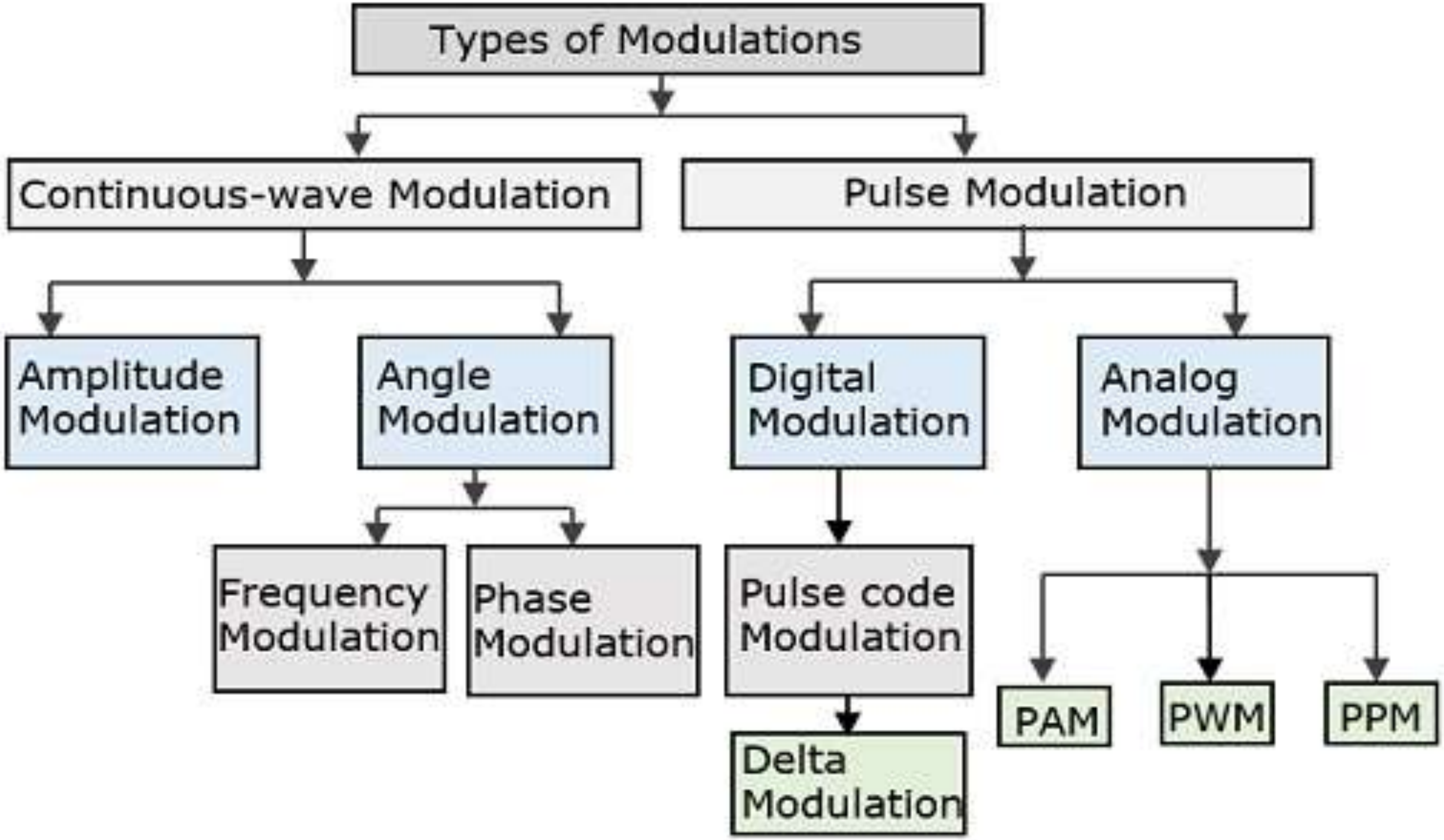
  1) Reduce the height of Antenna.

  2)Increases the range of Communication.

  3) Avoids mixing of signal

  4) Allows multiplexing of signals.

  5) Allows Adjustments in Bandwidth.


Types of Modulations
Continuous-wave Modulation
Pulse Modulation
Amplitude Modulation
Angle Modulation
Digital Modulation
Analog Modulation
Frequency Modulation
Phase Modulation
Pulse code Modulation
Delta Modulation
PAM
PWM
PPM

AMPLITUDE MODULATION

- In Amplitude modulation the amplitude of a Carrier signal is varied by the modulating signal.
- Here the modulating signal is Nothing but base band signal (or) information signal while carrier signal is very high frequency signal.
- In amplitude modulation we are having two domains,

  1) time domain.

  2) frequency domain

TIME DOMAIN:-

- The instantaneous values of modulating signal and carrier signal are given below!

→ Instantaneous values of modulating signal m(t):

$M(t) = Am \cos (2\pi fmt)$ ---(1)

→ Instantaneous values of carrier signal C(t):-

- $C(t)= AC \cos (2\pi fct)$ ----(2)
- Where AM,AC are amplitude of modulating Signal and carrier signal.
- The standard form of Amplitude modulation (Am)is defined by
- $S(t) = AC (1+ kam(t)] \cos (2\pi fct)$
- where ka = constant
- The figure below shows amplitude modulated waves for different magnitude are ka.m(t).

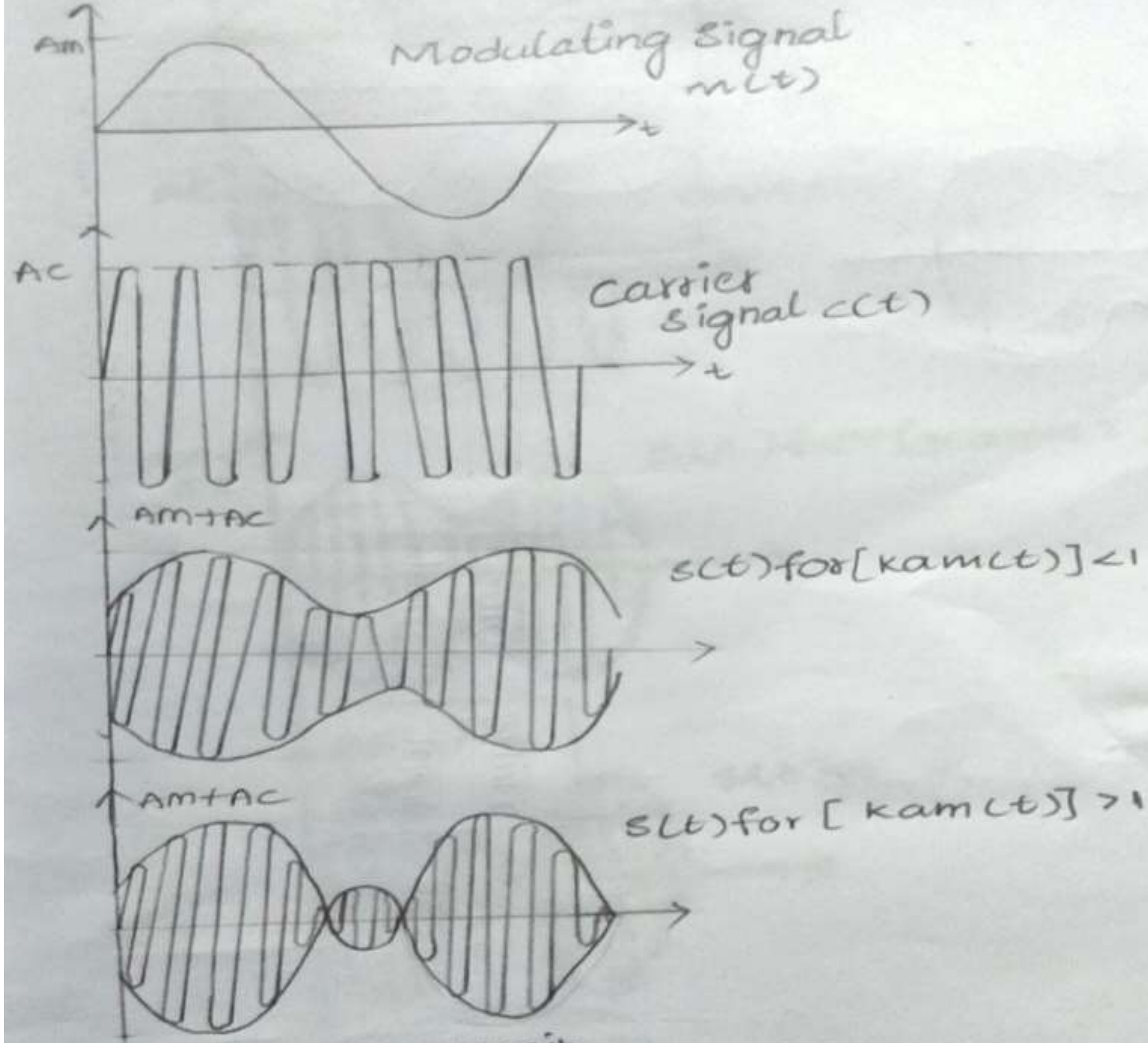


fig - Time Domain

Percentage modulation (or) modulation factor(μ) :-

- $s(t) = AC\ [1 + kam(t)] \cos(2\pi fc(t))$
- Substitute the values of m(t) eqn(1) in eqn(3)
- we get
- $s(t) = AC\ [1 + ka\ Am\cos(2\pi fmt)] \cos 2\pi fct$----(4)
- eqn(4) can be written as
- $s(t) = AC\ [\ 1+\mu \cos(2\pi fmt)].\cos 2\pi fct$ -----(5)

  ($\mu \rightarrow$dimensionless constant)
- when μ is multiplied by 100 then μ is expressed Numerically as percentage modulation.
- when μ > I then over-modulation takes place
- To avoid envelop distortion (or) over modulation factor μ must be kept below unity.

FREQUENCY DOMAIN

- The modulates carrier has new signals at different frequency called side bands. Occurs in the frequency sprectrum directly above and below the carrier frequency.
- Now will see the extenstions of side band frequencies in amplitude modulation where as in mathematical expression
- FUSB $\rightrightarrows$ FLSB
- FUSB:-----upper side band frequency(fc+fm)
- FLSB:------lower side band frequency(fc-fm)
- From eqn(5)
- S(t)=AC[1+μcos(2πfm(t))]cos(2πfc(t)-------(5)
- S(t)=Accos(2πfc(t)+μAccos(2πfc(t)cos(2πfm(t)------(6)
- Cosa.cosb=1/2[cos(a-b)-cos(a+b)]
- S(t)=Accos(2πfc(t)+μAC/2[cos(2πfc(t)-2πfm(t)-cos(2πfc(t)+2πfm(t)

BANDWIDTH OF AM WAVE

- Bandwidth of the Am wave can be calculated by subtracting th frequency of lower side Band frequency from upper side band frequency.
- BW=FUSB-FLSB

=(fc+fm)-(fc-fm)

=(fc+fm-fc+fm

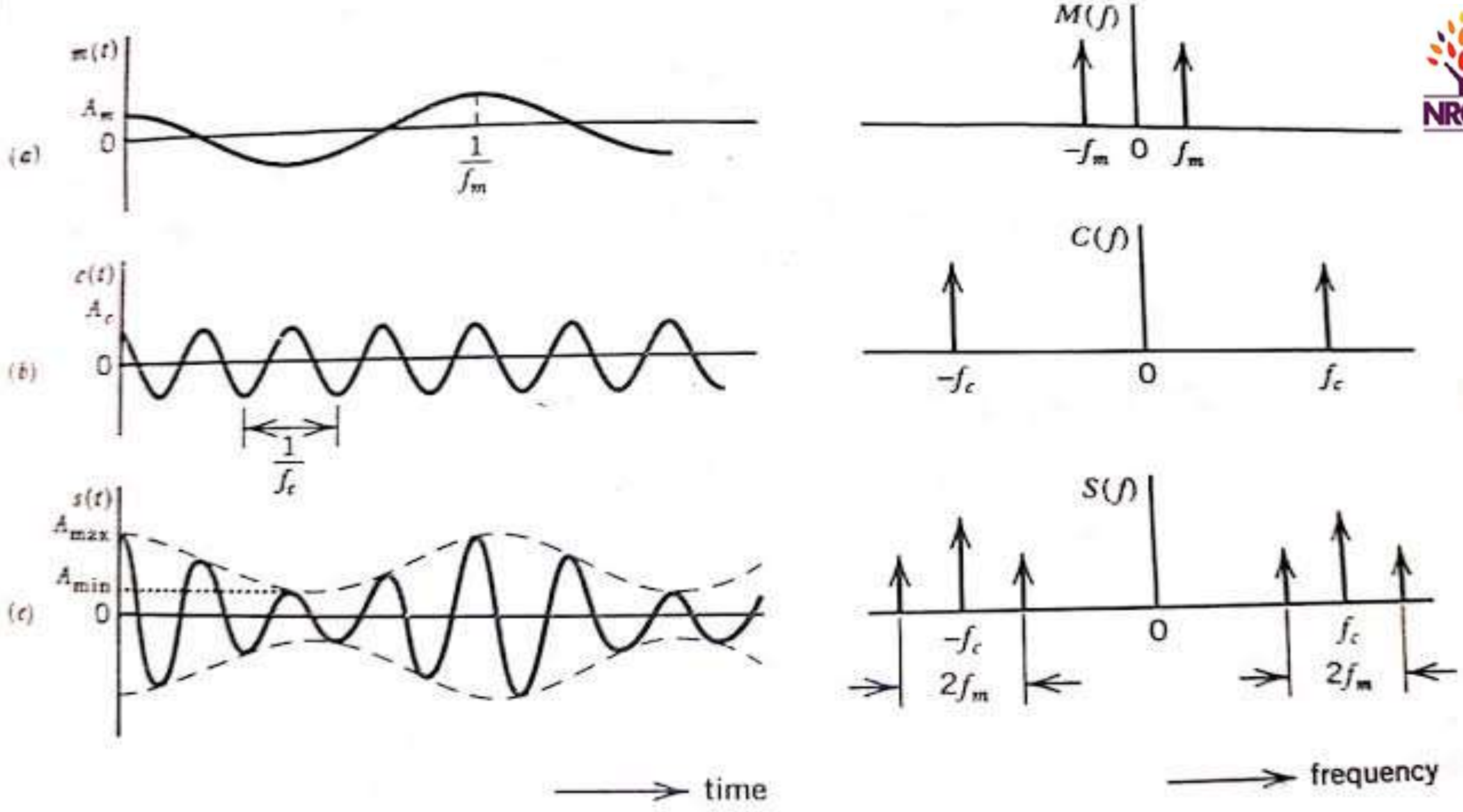
=2fm

BW 2fm

- Bandwidth of amplitude modulated wave is twice the frequency of modulating signal.

SINGLE TONE MODULATION

- We know that
- S(t)=ACcos(2πfc(t)+μAC/2 (cos(2πfc(t)-2πfm(t))-μAC/2(cos(2πfc(t)+2πfm(t))
- Apply fouerier transform for above eqn then
- S(f)=AC/2[∂(f-fc)+∂(f+fc)]+μAC/4[∂(f-fm)+∂(f+fc+fm)]-μAC/4(∂(f-fc+fm)+∂(f+fc+fm)]
-



m(t)
A_m
0
(a)
$\frac{1}{f_m}$
c(t)
A_c
0
(b)
$\frac{1}{f_c}$
s(t)
A_{max}
A_{min}
0
(c)
time
M(f)
$-f_m$ 0 f_m
C(f)
$-f_c$ 0 f_c
S(f)
$-f_c$ 0 f_c
$2f_m$
$2f_m$
frequency

- Let us See the time domain waveform for a amplitude waveform with $\mu<1$.
- As shown in the fig the positive and Negative peakes of the Carrier waveform one inter connected with imaginary Lines to form a exact shape of the signal (modulating signal).
- Then imaginary Line in the carrier waveform called envelope and it is same as modulating Signal.

- Amax and Amin are maximum and minimum values of envelope then from eqn(5) substitute in cos=1
- We get
- $A_{max}=A_C/2(1+\mu)$
- $A_{min}=A_C/2(1-\mu)$
- $A_{max}/A_{min}*(1+\mu)/(1-\mu)$
- $A_{max}(1-\mu)=A_{min}(1+\mu)$
- $-\mu(A_{max}+A_{min})=A_{min}-A_{max}$
- $\mu=A_{max}-A_{min}/A_{max}+A_{min}$

POWER RELATIONS IN AM WAVE

- The power relations in AM wave is three components they are,
- 1) unwanted
- 2)PLSB
- 3)PUSB

Total power Pt=Pc+PLSB+PUSB

- $PC=(AC/\sqrt{2})^2=AC^2/2R$
- $PC=AC^2/2R$
- PLSB=PUSB
- $PLSB=(AC/2/2)^2/R$
- $=AC^2/8*1/R$
- $=AC^2/8R$
- PUSB=PLSB

$$PUSB=AC^2/8R$$

$$PT=Pc+PLSB+PUSB$$

$$PT=AC^2/2R+AC^2/8R+AC^2/8R$$

$$PT=AC^2/2R[1+\mu/4+\mu/4]$$

$$PT=AC^2/2R[1+\mu/4]$$

$$PT/Pc=1+\mu^2/2$$

$$PT/Pc-1=\mu^2/2$$

$$\mu^2/2=PT/Pc-1$$

$$\mu^2=2[PT/Pc-1]$$

$$\mu=\sqrt{2}(PT/Pc-1)$$

GENERATION OF AM WAVE

- They are techniques to generate amplitude modulator waves.

✓ Low level modulation

✓ High level modulation

1)LOW LEVEL MODULATION:-

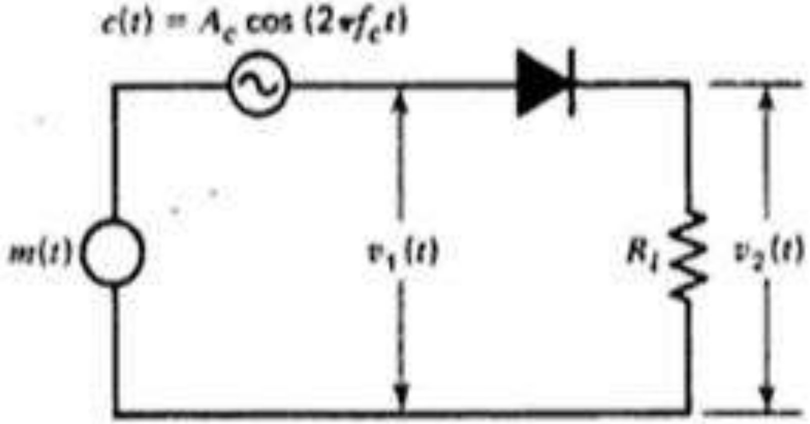
Low level modulation techniques is used in the initial stage of amplification ie., at low power level.

2)HIGH LEVEL MODULATION:-

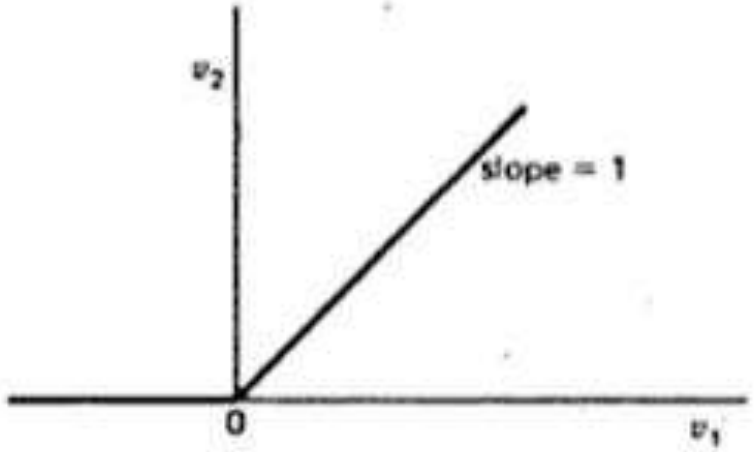
In high level modulation the modulator takes place of final stage of amplification . Therefore modulation circuit has to handle high power.

=> SWITCHING MODULATOR

- **In this modulator multiplication operation is deplaced by or similar Switching operation The figure shows the Switching modulator Ct)**



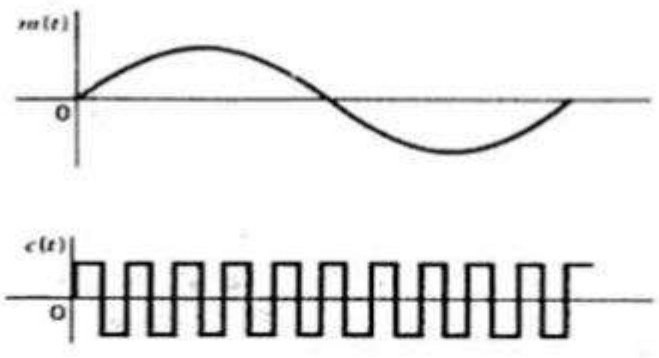
CIRCUIT DIAGRAM



Idealized Input Output Characteristic Curve

-> Here diode D is assumed to act as an ideal switch i.e it presents zero impedanc when it is forward biase and infinite impedore when it is reverse blas .

-> The diode Switch is controlled by carrier wave C(t), when c(t) is greater than zero, diode is forward bias (on Switch) when c(t) <0 then diode is Reverse bias (off Switch) . because , in this circuit amplitude of c(t) applied to the diode is large .



The above figure shows the resultant wave form RL

we know that $V1(t) = C(t) + M(t)$.

$V1(t) = Ac \cos(2\pi fct) + m(t)$

where m(t) is less than AC. ($M(t) << AC$)

$M(t) << C(t)$

$Am << Ac$

The o/p voltage $V_2(t)$ can be represented as

$$V_2(t) = \begin{cases} v_1(t) & ; c(t) > 0 \\ 0 & ; c(t) < 0 \end{cases} \quad \text{— (1)}$$

The load voltage $V_2(t)$ varies periodically b/w the values V1(t) and Zero.

$V_2(t) = c(t) + m(t) + gp(t)$.

$= Accos(2\pi fct) + m(t) + gp(t)$...... (eq-2)

where gp(t) is a periodical pulse train of duty cycle is equal to 1½ and period t0 = 1/fc

Representing this gp (t) by its fourier Series, we have

we have

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos\left(2\pi f_c t\,(2n-1)\right)$$

$$\boxed{g_p(t) = \frac{1}{2} + \frac{2}{\pi}\cos(2\pi f_c t) + \text{all harmonic component}} \quad (3)$$

Substitute eq (3) in eq (1) we get

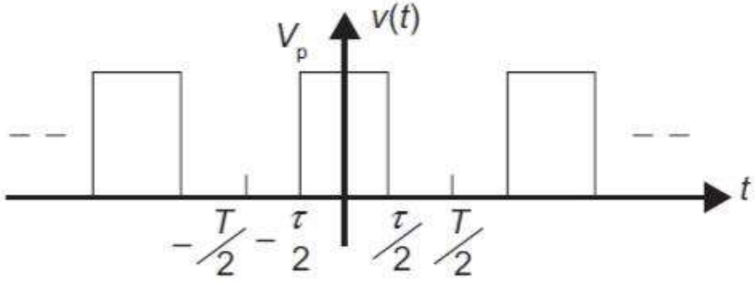
$$\boxed{V_2(t) = \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c}m(t)\right]\cos(2\pi f_c t) + \text{unwanted terms}} \rightarrow (4)$$

The first term in eq4 is desired Am wave with Amplitude Sensitivity

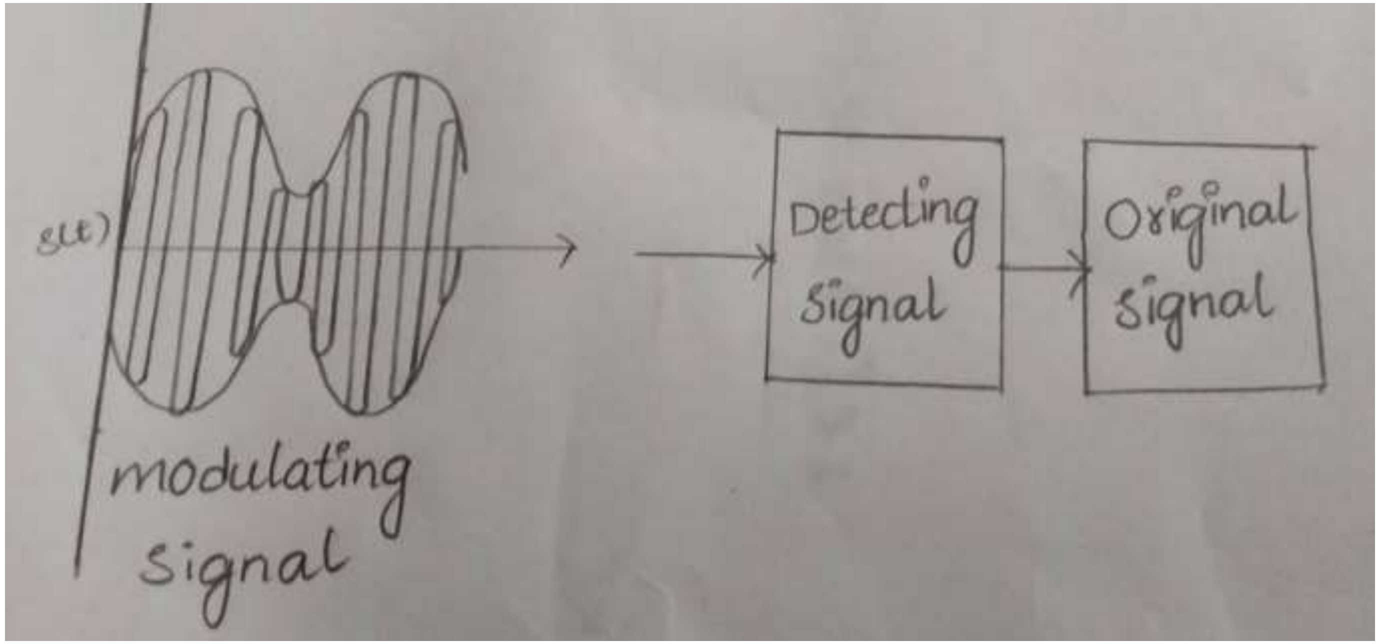
Ka=4/πAc

The unwanted terms are removed from the lood vottage v2(t) by using bandpass filters.

PERIODIC PULSE TRAIN:



DETECTION OF AM WAVES



- A demodulator circuit accepts a modulator signal and recover the original modulating information.
- These circuits are known as detector(or) Demodulator.

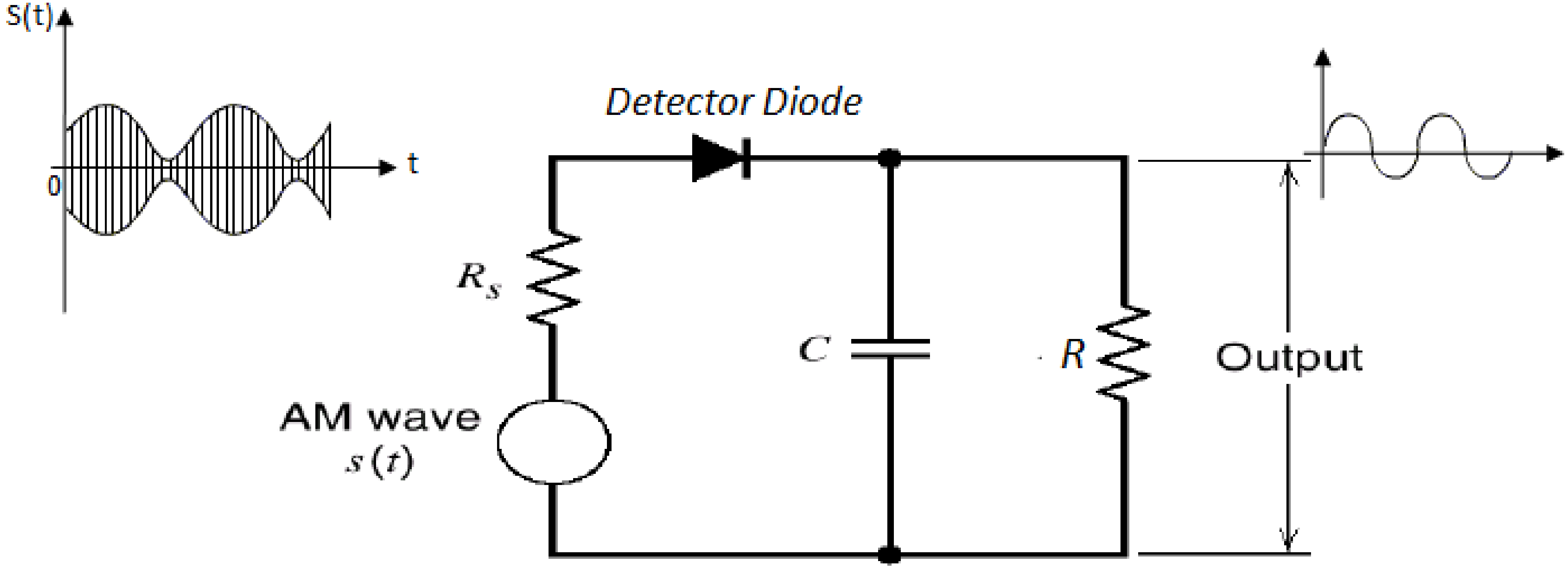
Most widely used amplitude detectors (or) demodulators are,

- ✓ Square-law detector
- ✓ Envelope detector

1.ENVELOPE DETECTOR:-

- An envelope detector is simple and highly effected device that is well suited for the demodulation of narrow band AM waves for which the percentage modulation is lessthan 100%.
- In an envelope detector, the output of detector follows the envelope the envelope of modulating signal.

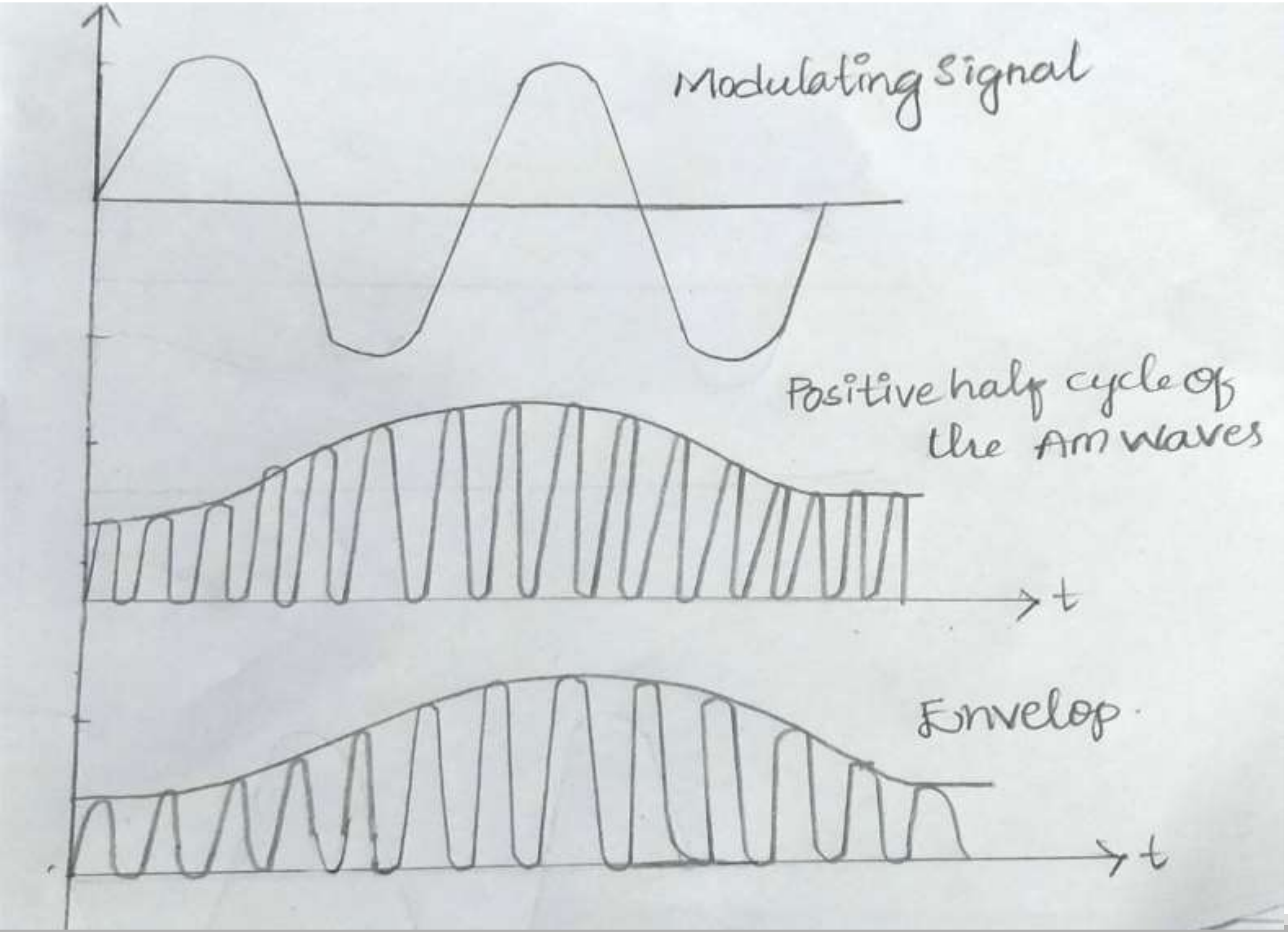
CIRCUIT DIAGRAM



The above fig shows Envelop detector. Circuit diagram where Resistor & capacitors RC filter.

* This circuit is also known as diode detectors.
* In the positive half cycle of the Am signal, diode conducts (ON switch) and current flows through R.
* when as the in Negative half cycle of the Am signal the diode is reverse Bias and diode is OFF state then No current will flow in the R .
* As a Result only positive half of the AM wave appears across the RC AS shown in fig .

* Let us See how RC filter response to the positive half of the Am wave



Charging and discharging of capacitors

*As shown in the above fig we have assumed that the Am waves applied to the envelope detector is supplied by a voltage source internal impedance Rs.

-> To Rapidly charge the capacitor to the peak value of input signal , charging time constant RSxC must be short compared with carrier period 1/fC

RSC<<1/fc

-> on the other hand the discharging Time constant must long enough to ensure that the capacitor discharges Slowely to the load Resistance. Blw positive peak of the carrier wave but not so long that the capacitor voltage will not discharge at the maximum rate charge of modulating wave .

i.e $1/fc \ll RLC \ll 1/\omega$

.

$\# \omega$ = message bandwidth

→ **Distortions in envelop detectors:**

They are two types of distortions in envelop.

1) Negative Peak clipping
2) diagnand clipping.

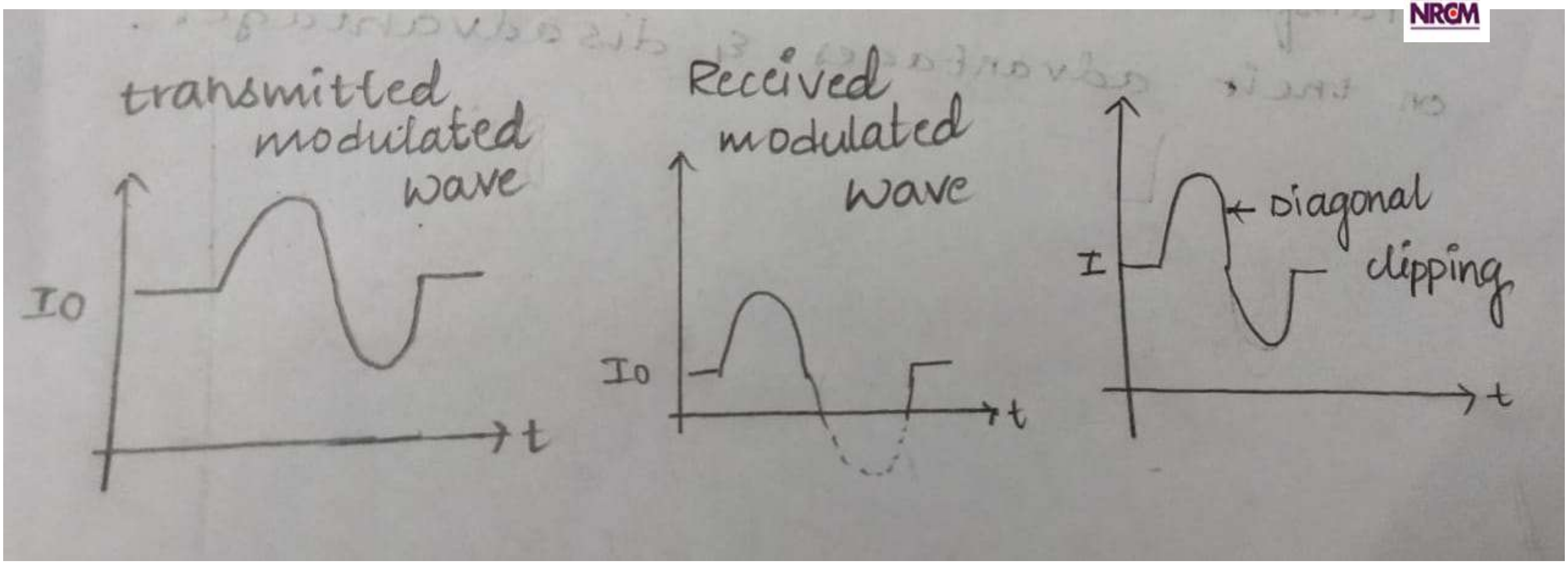
1) **Negative Peak clipping :**

-> modulation index has Em/EC , If it is defined as IM/IC

. IM=CM/ZM. And IC= EC/RC

Where ZM= Audio diode load impedance.

. RC=dc diode resistance.



transmitted modulated wave
Io
t
Received modulated wave
Io
t
Diagonal clipping
I
t

2.Diagonal clipping

Mmax=Mdzm/Rc

Where Md=IM/IC = MRC/ZM

Mmax = 1* ZM/RC

Mmax= ZM/RC

Generation of DSBSC signal waves

- We have seen a double side band suppressed carrier modulated and wave consists in the simple product modulating signal and carrier signal.
- The desire product can be achieved by device called as product modulation or balanced modulator.

Balanced modulator

- Balanced modulator is used to supressed the modulator in carrier signal from AM signal

- The inputs to the balanced modulator. The output to the balanced modulator DSBSC and upper and lower side bands

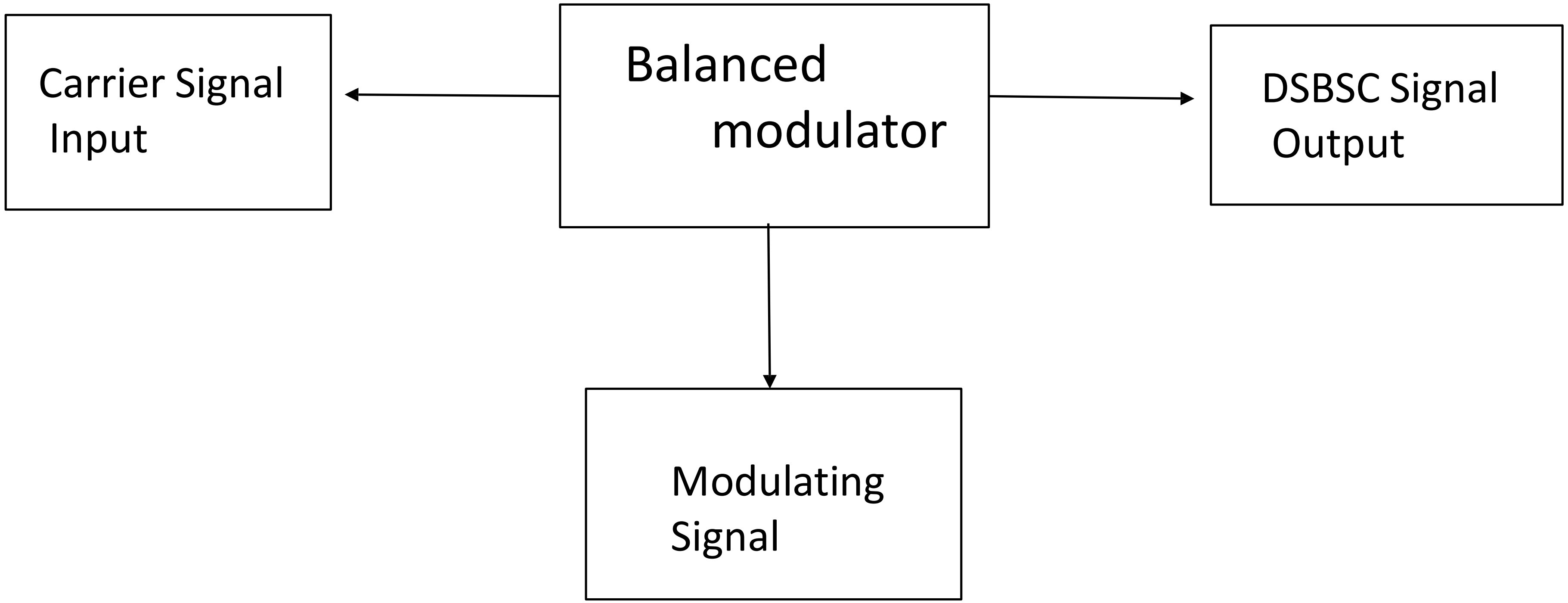
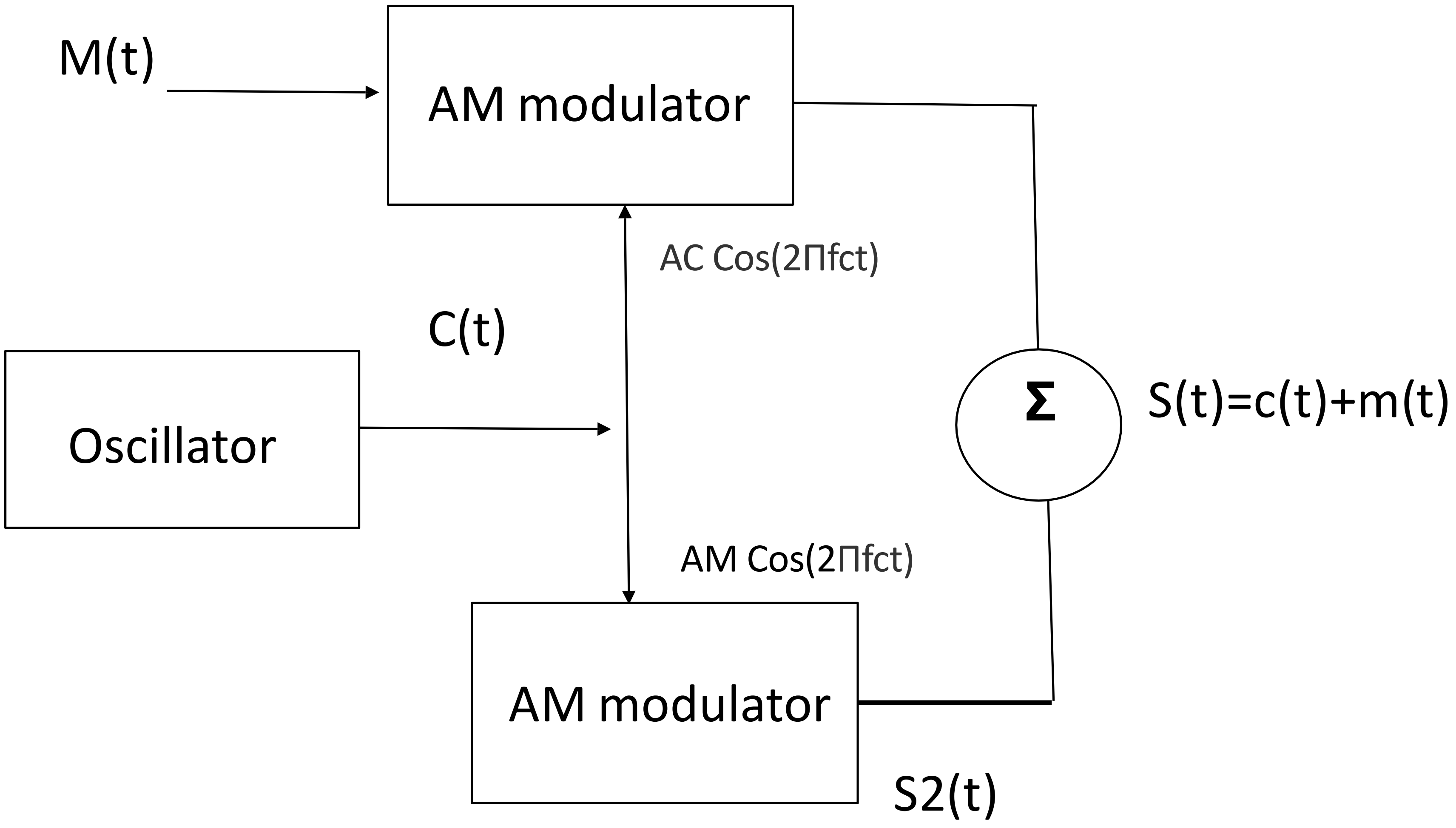


Fig: Balanced modulator with input and output

Principle used in the balanced modulator



- when two signal that different frequencies carrier and modulating signal are passed through a non- linear signal, Am Agnal is generated with suppressed signal.

- A device having non- linear signal diode, resistances JFET are transistor are used in the balance modulator to generate AM signal with suppressed.

- Figure shows the balanced modulator it consists of two balanced modulator Amplitude modulator that are arranged in the balance modulator so that suppressed in the carrier wave.

- Here two balance modulator are identical and expect the sine reversal of the modulating wave apply to the input of the one of them.

- Therefore, output of the two modulation can be given as below.

$$S1(t)= AC\,[1+K(a)m(t)]\cos(2\Pi fct) \longrightarrow \text{3 Equation}$$

$$S2(t)=AC\,[1-K(a)m(t)]\cos(2\Pi fct) \longrightarrow \text{4 Equation}$$

- Subtracting S1(t) and S2(t) from equation 1 and 2 then we get the equation.

$$S(t)=S1(t)-S2(t)$$

$$S(t)=2KaAC\cos(2\Pi fct)-m(t) \longrightarrow \text{5 Equation}$$

- The balanced output is equal to the product of modulating and the carrier.

- The methods used for suppression of carrier can be classified as two types:

1. Using the diode, ring or lattice modulator.

2. Using JFET modulator.

- **Balanced modulator using the diode, ring or lattice modulator.**

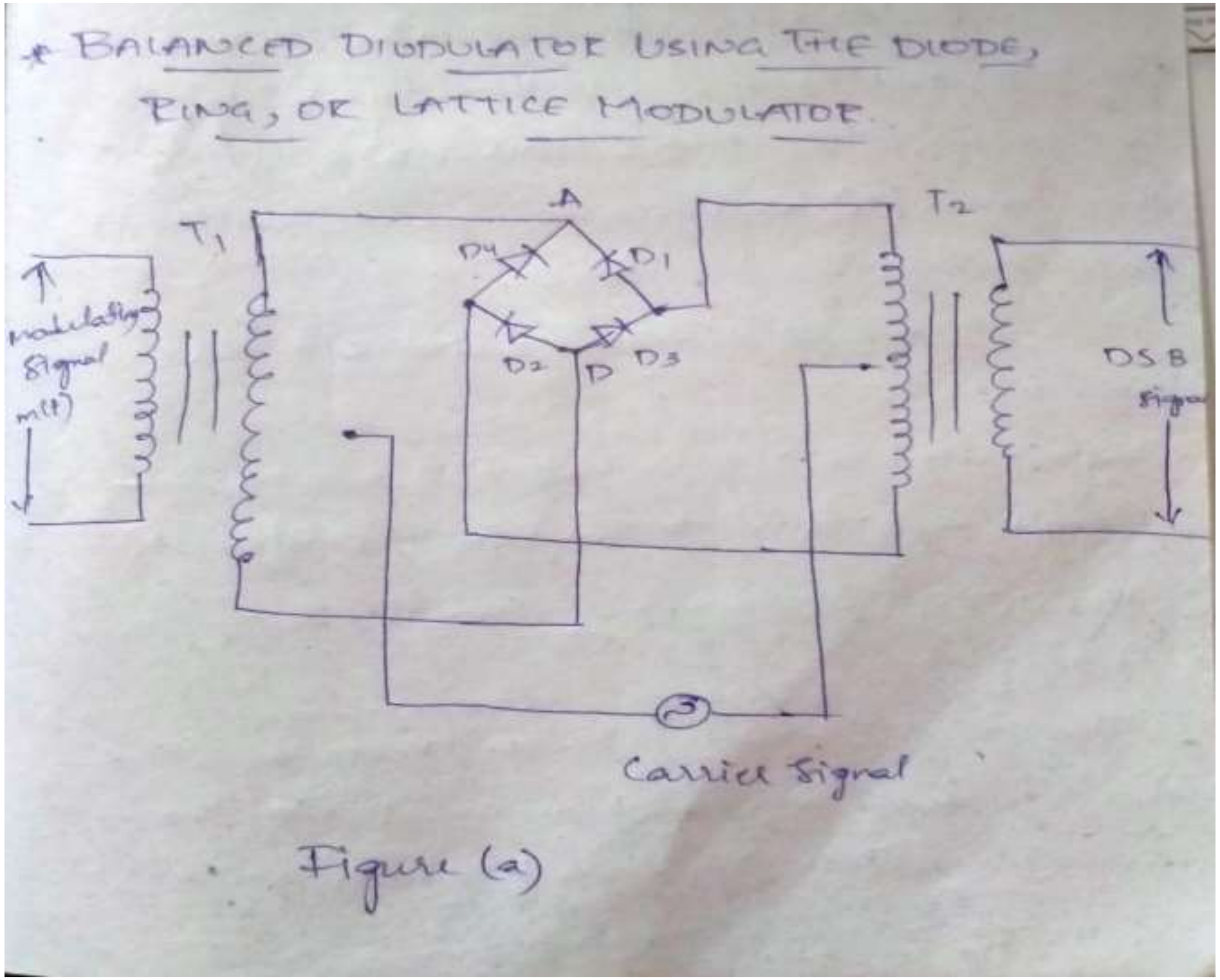
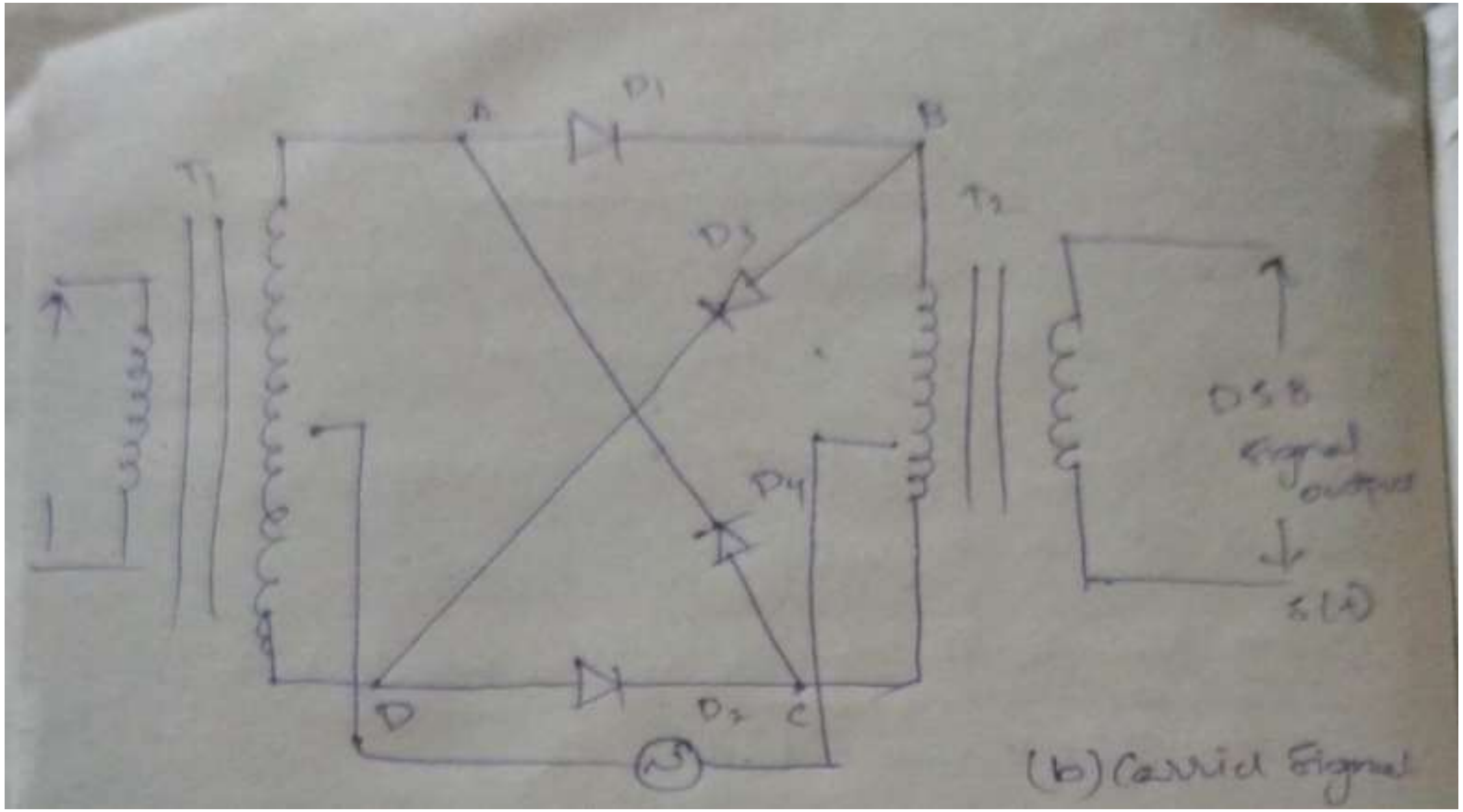


Figure (a)



Figure(b): carrier signal

- Figure(a) and (b) shows lattice type modulator consists of an input transformer (D1) and output transformer (T2) and four diodes are connected in bridge circuit.

- Input and output transformers and modulating signal is applied to the input transformer (T1).

- The output appears at across the secondary transformers.

- The diodes connected in the bridge as acts switches and switching is operated by the carrier as it is used in the high frequency and amplitude then modulating signal.

Positive half cycle of the carrier signal :-

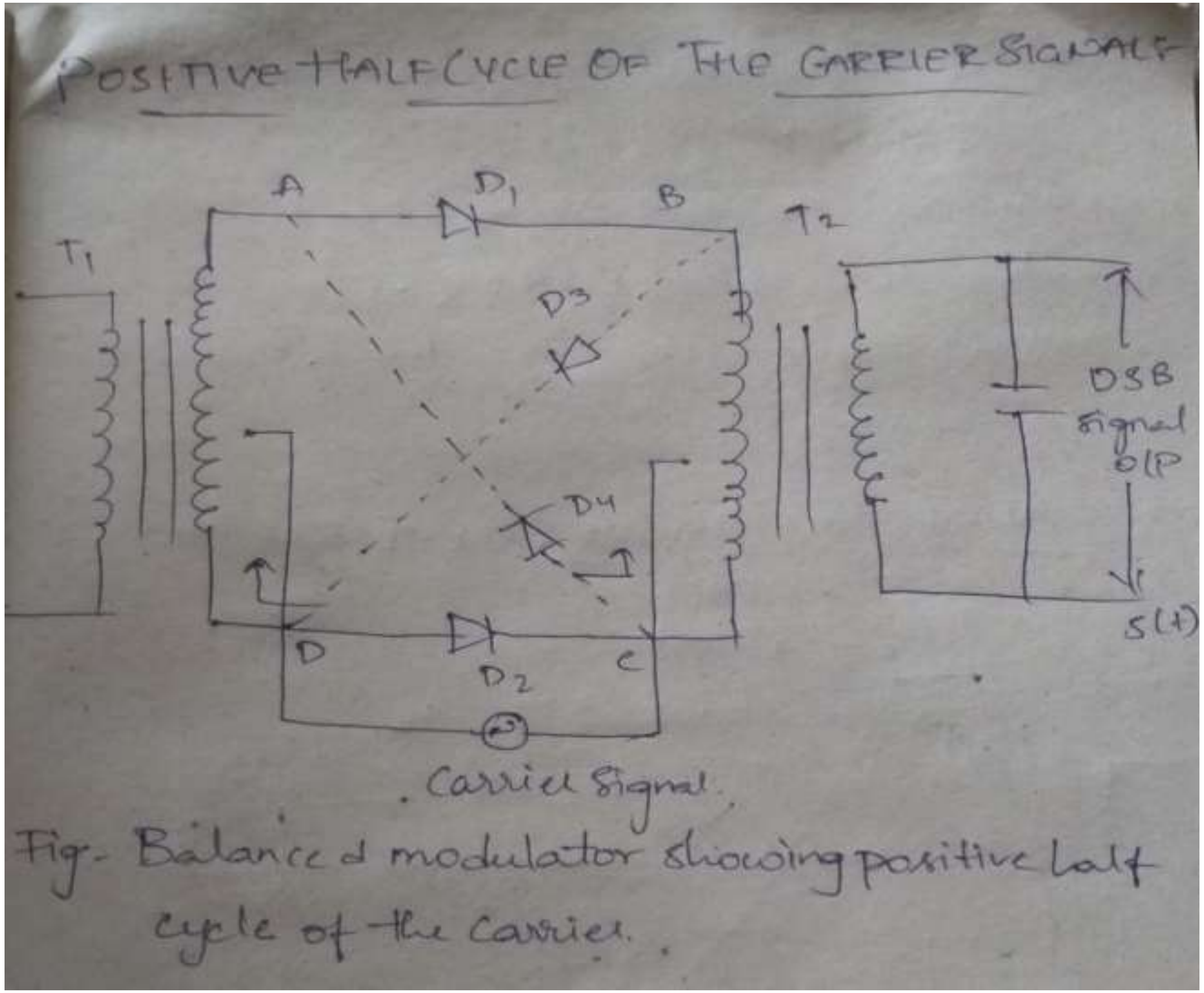
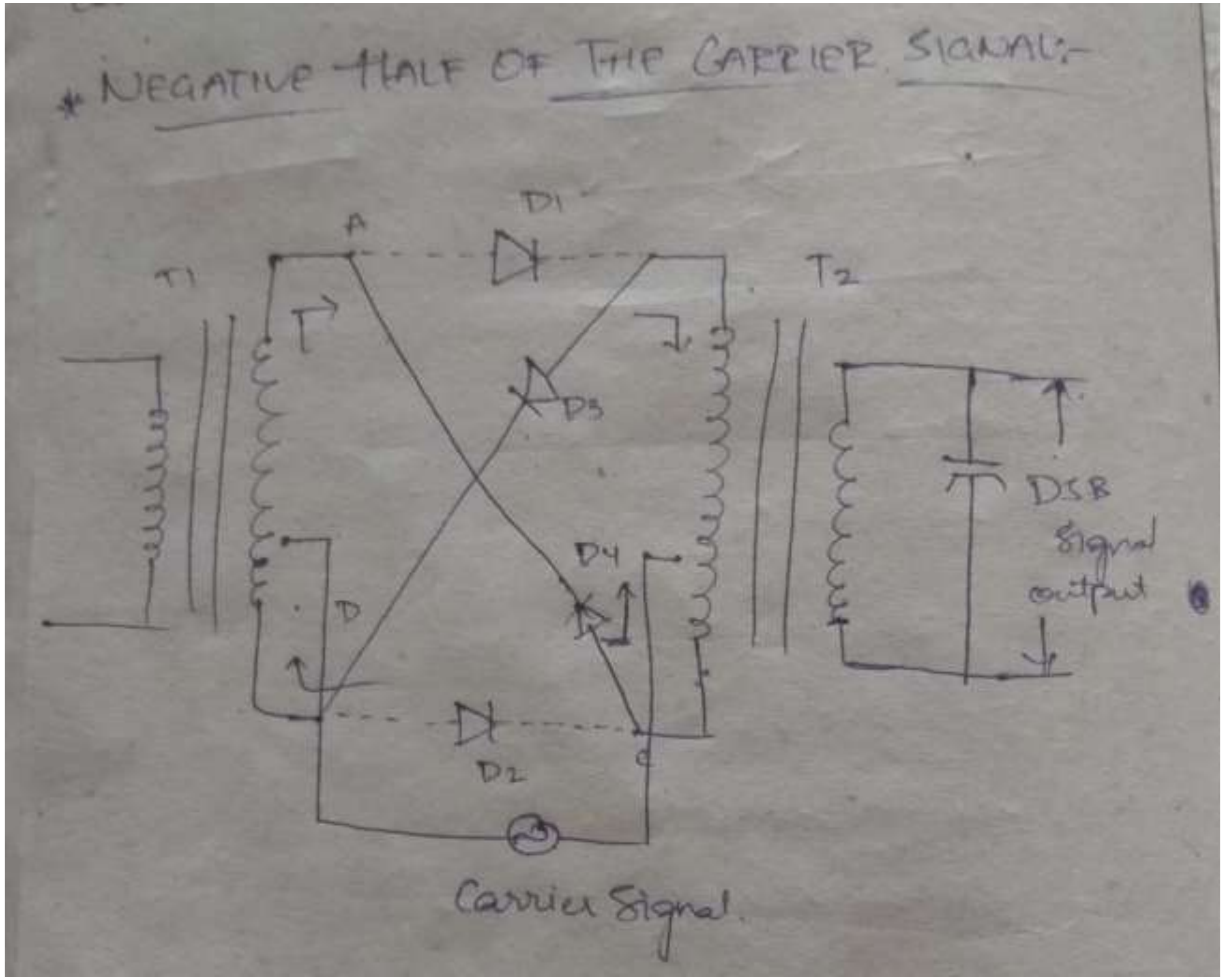


Fig.- Balanced modulator showing positive half cycle of the carrier.

- We first assume that m(t)=0.
- In the positive half cycle of carrier signal D1 and D2 are forward bias and D3 and D4 are in reverse bias as shown in the figure.
- We can see that the current divides equally upper and lower primary portions of the primary winding of T2.
- The current in the upper winding produces the magnetic field that is equal and opposite to the magnetic field produced by the current in the lower half of the secondary.
- As the magnetic field are equal and opposite and each other they cancel each other producing.
- No output at the secondary transformerT2does is suppressed.

Negative half of the carrier signal



- In the negative half cycle of a carrier signal diodes D1 and D2 are reverse biased and diodes D3 and D4 are forward biased as shown in the figure.

- Similarly, the positive half cycle and here also primary winding of T1 and T2 are equal to each other and they cancel each other.

- Therefore, T2 at secondary is zero.

- The suppression of carrier depends upon the matching of the diode characteristics and the precision of the center transformation to give equal upper and lower current and magnetic fields.

With modulating signal :-

- Now assume a low frequency sine wave is applied to the primary of T1 as the modulating signal.
- This signal will appear across T1 secondary.
- In the positive half cycle, the diodes D1 and D2 are forward biased and they will connect the secondary of T1 to the primary of T2.
- As a result, the modulating signal at the Secondary of T1 is applied to the primary of T1 to the diodes D1 and D2.
- In the negative half cycle, D3 and D4 are forward biased they will connect the Secondary of T1 to the primary of T2 with reverse connections.

- This results 180° phase shift in the modulating signal.
- The ring modulator is more ideal when carrier is a square wave.
- In case of square wave signal, the carrier c(t) is represented by a fourier series as

$$C(t)=4\backslash\pi \sum_{N=1}^{\infty} (-1)^{\wedge}n-1\backslash 2n-1(\cos[2\pi fct(2n-1)]) \longrightarrow \text{1 Equation}$$

- We know that for ,

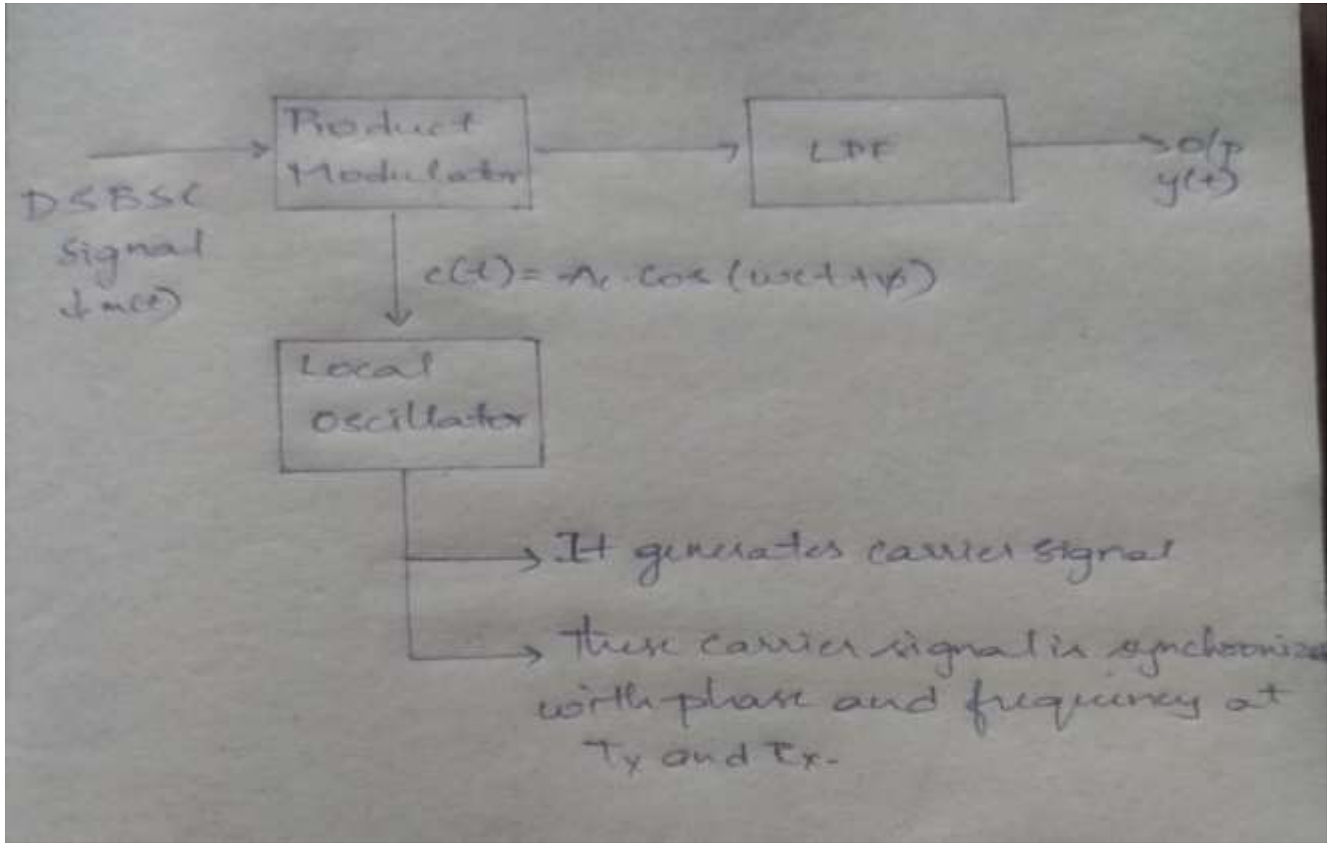
$$S(t)=C(t).m(t) \longrightarrow \text{2 Equation}$$

- Value of C(t) in above equation,

$$S(t)= =4\backslash\pi \sum_{N=1}^{\infty} (-1)^{\wedge}n-1\backslash 2n-1(\cos[2\pi fct(2n-1)]).m(t)$$

Coherent detection of DSBSC signal:-

- Demodulation of DSBSC signal:- Demodulation of DSBSC signal after passing through the circuit at the output side we will get the original information.



- The output of the product modulator is
  $S(t)=m(t).c(t)$

$$=X(t).\cos(\omega t).A_c \cos(\omega t+\phi)$$
$$=A_c.X(t)[\cos(\omega t).\cos(\omega t+\phi)$$

- Multiply and divide by '2'

$$=A_c.X(t)/2[2\cos(\omega t).\cos(\omega t+\phi)$$

- After solving we get,

$$S(t)=\underbrace{A_c.X(t)/2[\cos\phi]}_{\text{(Low frequency original signal)}} + \underbrace{A_c.X(t)/2[\cos(2\omega t+\phi)]}_{\text{(Unwanted term)}}$$

- The product modulator output is sent to the lowpass filter.
- The working of the low pass filter is it takes the low frequency signal and rejects the unwanted signal.
- The output of this lowpass filter is

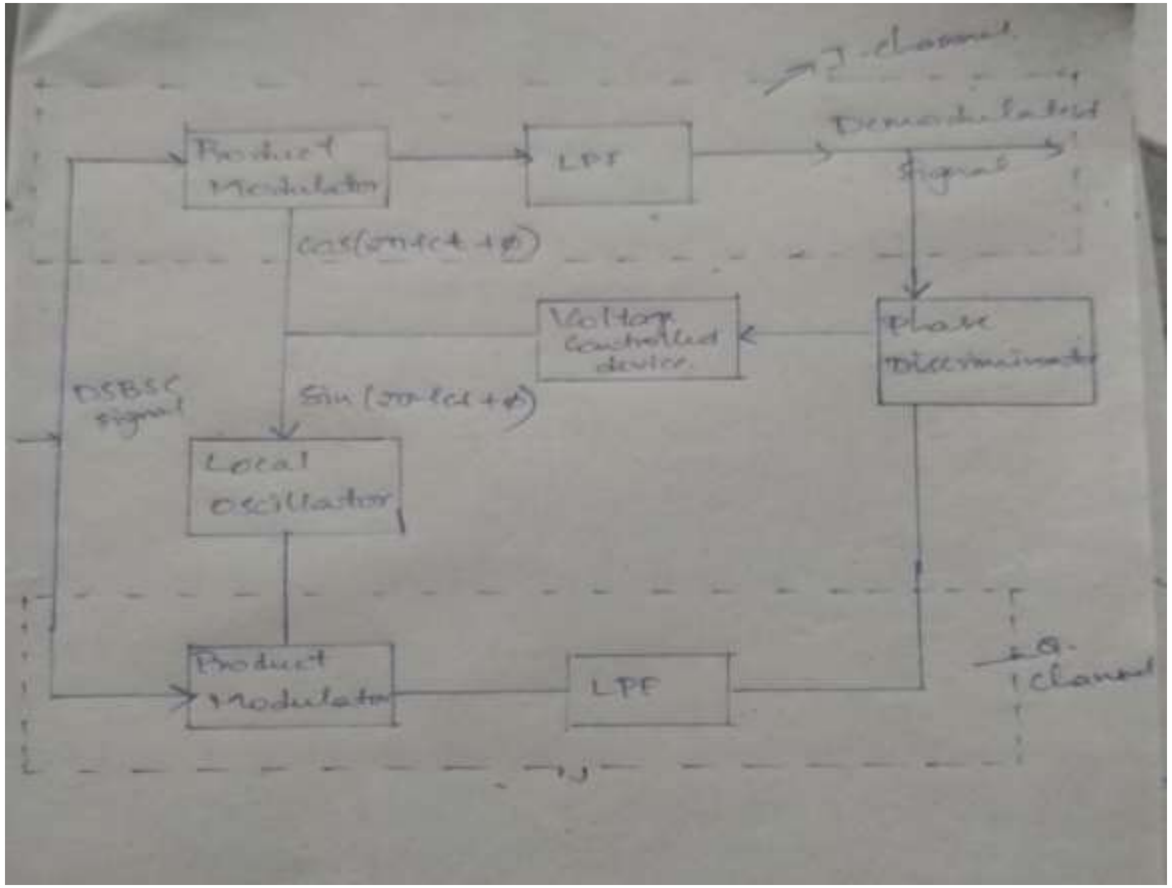
$$Y(t)=Ac.X(t)/2.[\cos\phi]$$

Where '$\cos\phi$' is phase error

- The value of 'ϕ'is Maximum when $\phi=0$,
  Minimum if $\phi= +\backslash- \pi/2$

COSTAS Loop:-

Costas loop is a phase-locked loop (PLL) based circuit which is used for carrier frequency recovery from suppressed-carrier modulation signals and phase.



- The figure shows the COSTAS loop in this loop receive consists of two coherent modulator supplying with the same input signal.
- However, the local oscillator signal supplied to the product modulator are 90° auto phase.

SSB-SC Generation of phase discrimination

- The phase shift method of SSB generation uses the phase shift technique that cases one of the side bands to be cancelled out.
- It consists of two balanced modulators instead of one and two phase shifting network.

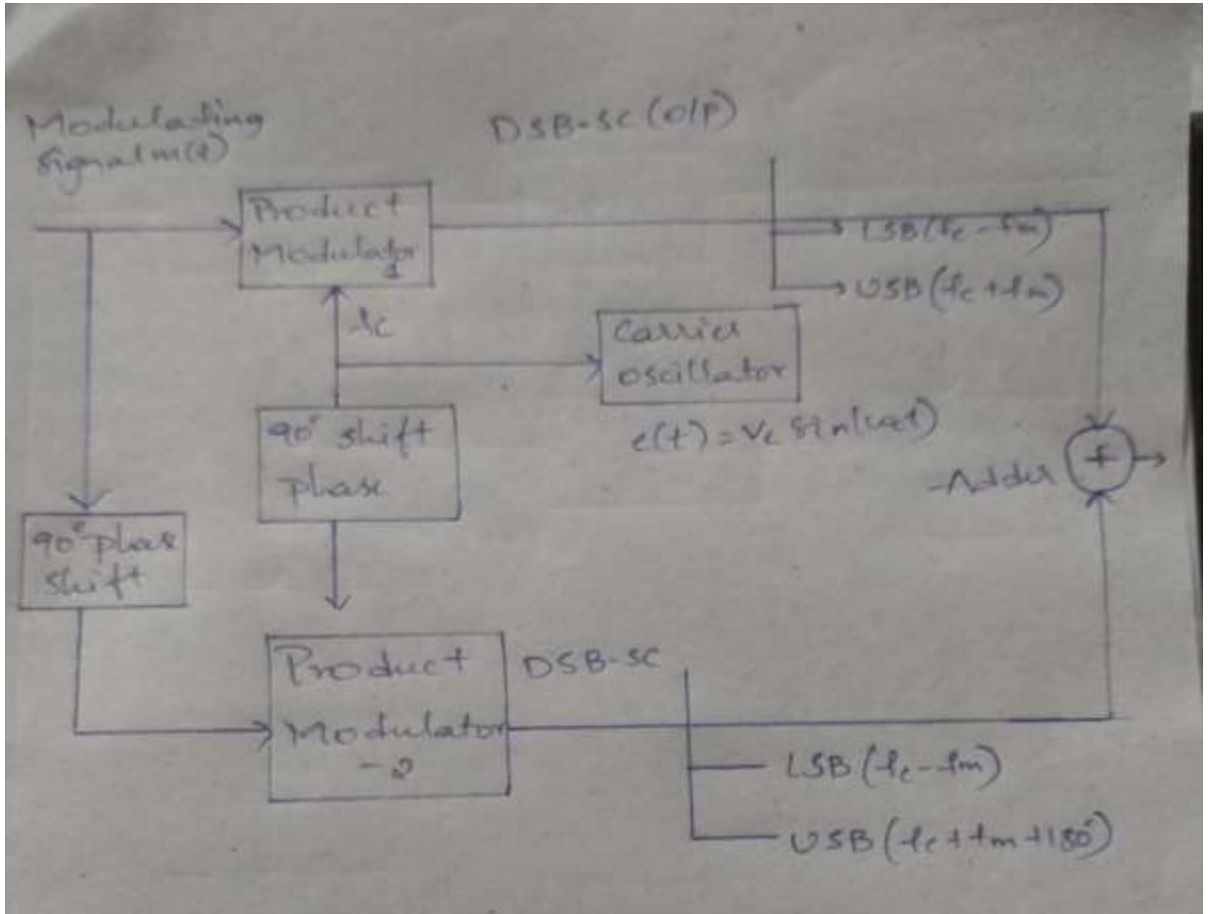


Fig: PHASE DISCRIMINATION.

- As a result we will get low frequency original signal LSB(fc-fm) is the decreased output of SSBSC phase discrimination.

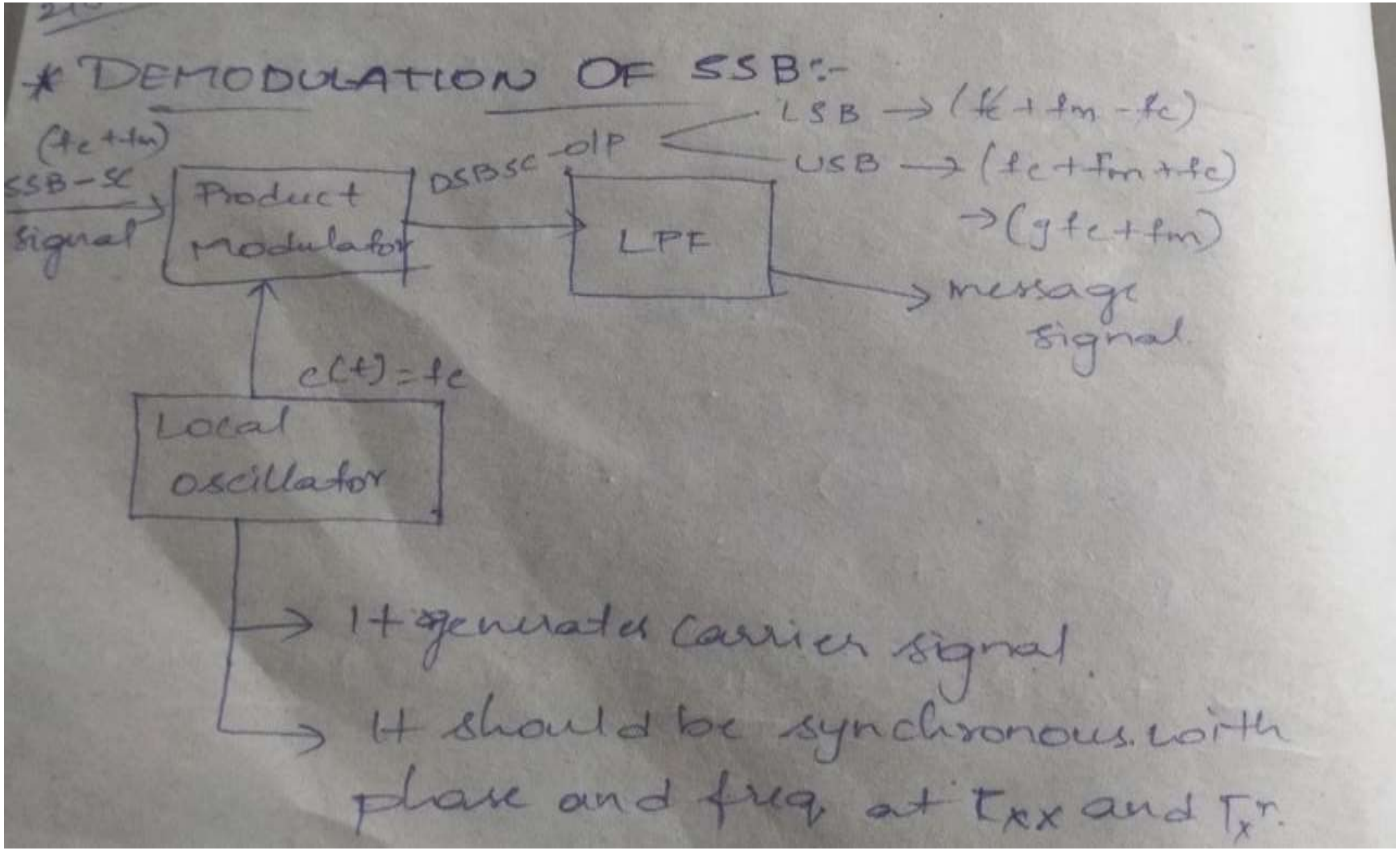
Advantages of phase shift methods:-

- It can generate SSB at any frequency.
- Easy switching from one side band to off side band is possible.

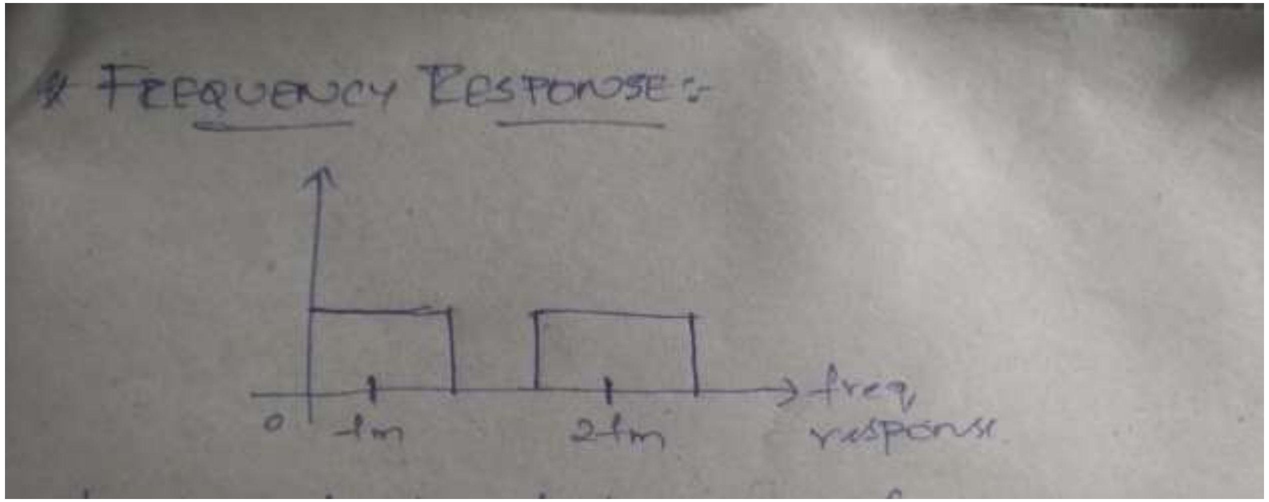
Disadvantages:-

- If phase shifter provides a phase change other than 90° at any audio frequency that particular frequency will not completely from the sidebands.
- The output two product modulator should be same otherwise cancellation will be incomplete.

Demodulation of SSB:-



Frequency response:-

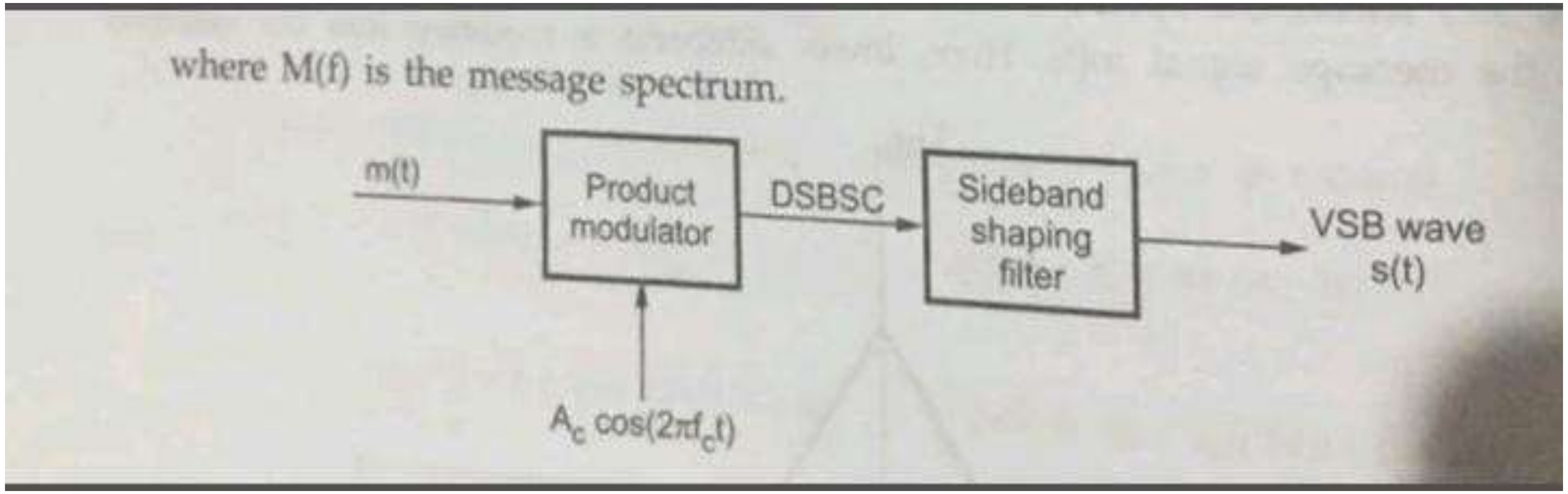


- The input signal of SSB-SC(fc+fm) single sideband is given to product modulation.
- Carrier signal c(t) is given to product modulator.
- The output of product modulator is DSB-SC.

  LSB=(~~fl~~+fm-~~fc~~)

- LSB=fm
- USB=2fc+fm

Generation of VSB modulated wave:-



- The relation between the transfer function H(f) of the filter and the spectrum S(f) of the VSB modulated wave s(t)is defined by

$$S(f)=Ac/2\left[M(f - fc) + M(f + fc)\right]H(f)$$

Vestigial sideband modulation:-

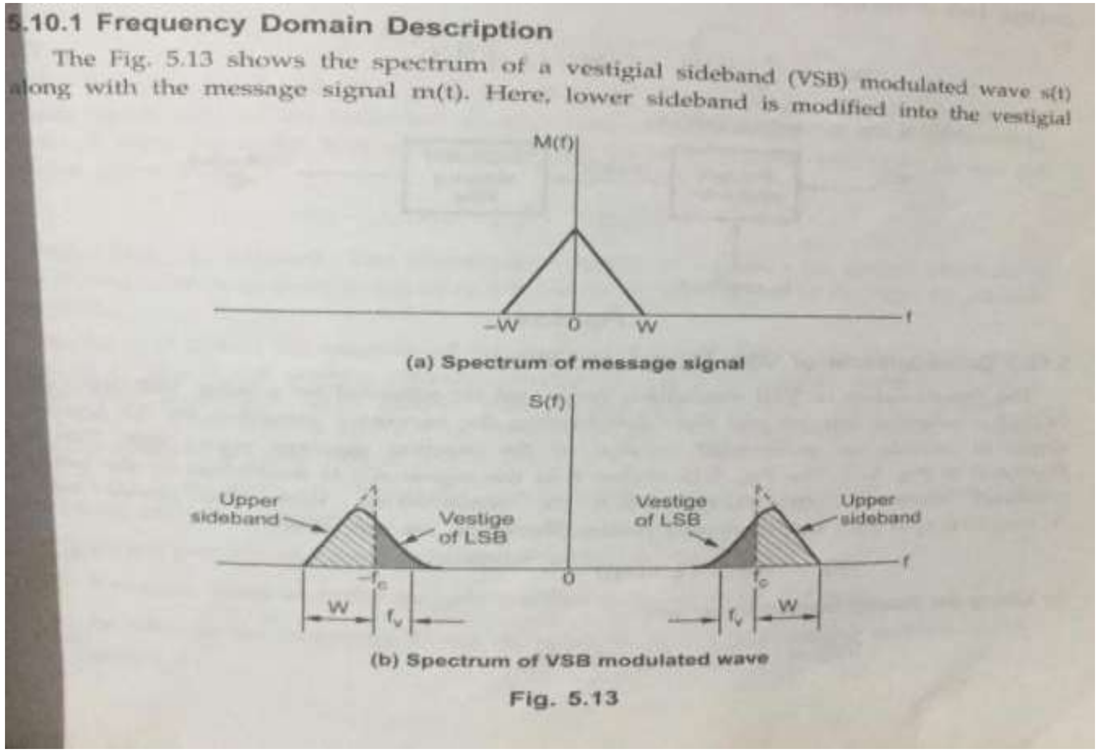
- The SSB modulation is not appropriate way of modulation when the message signal contains significant components at extremely low frequencies.

- Because in such cases the upper and lower sidebands meet at the carrier frequency and it is difficult to isolate one sideband.

- This is the compromise between SSB modulation and DSBSC modulation.

- The television signals contain significant components at extremely low frequencies and hence vestigial sideband modulation is used in television transmission.

Frequency Domain:-


5.10.1 Frequency Domain Description

The Fig. 5.13 shows the spectrum of a vestigial sideband (VSB) modulated wave s(t) along with the message signal m(t). Here, lower sideband is modified into the vestigial

- The above figure shows the spectrum of a vestigial sideband modulated wave s(t) along with the message signal m(t).

UNIT-3

TRANSMITTERS AND RECEIVERS

CONTENT

TRANSMITTERS

1.Classification of Transmitters
2.AM Transmitters
3.FM Transmitters

RECEIVERS

1.Radio Receiver-Receiver types-Tuned radio fre[illegible]y receiver
2.Superhetrodyne receiver
3.RF Section and Characteristics-Frequency changing and tracking
4. Intermediate frequency
5.Image frequency
6.AGC
7.Amplitude limiting
8.FM Receiver
9.comparison of AM and FM Receivers.

TRANSMITTERS:-

- A Transmitter is an electronic device , which modifies the incoming message signal to make it suitable for transmission over the communication channel.

(Or)

- A Transmitter is an electronic device used in telecommunications to produce radio waves in order to transmit or send data with the aid of an antenna.

The transmitter is able to generate a radio frequency alternating current that is then applied to the antenna, which,in turn , radiates this as radio waves.

CLASSIFICATION OF TRANSMITTERS:-

Based on the types of modulation used radio transmitters are classified as,

- ✓ AM Transmitters
- ✓ FM Transmitters
- ✓ PM Transmitters

1. AM TRANSMITTERS:-

AM Transmitters use amplitude modulation technique for modulating the carrier . These transmitters can be used for radio broadcast on long , medium and short waves.

2.FM TRANSMITTERS:-

FM Transmitters use frequency modulation techniques for modulating the carrier wave. These transmitters can be used for radio broadcasting in VHF and UHF ranges.

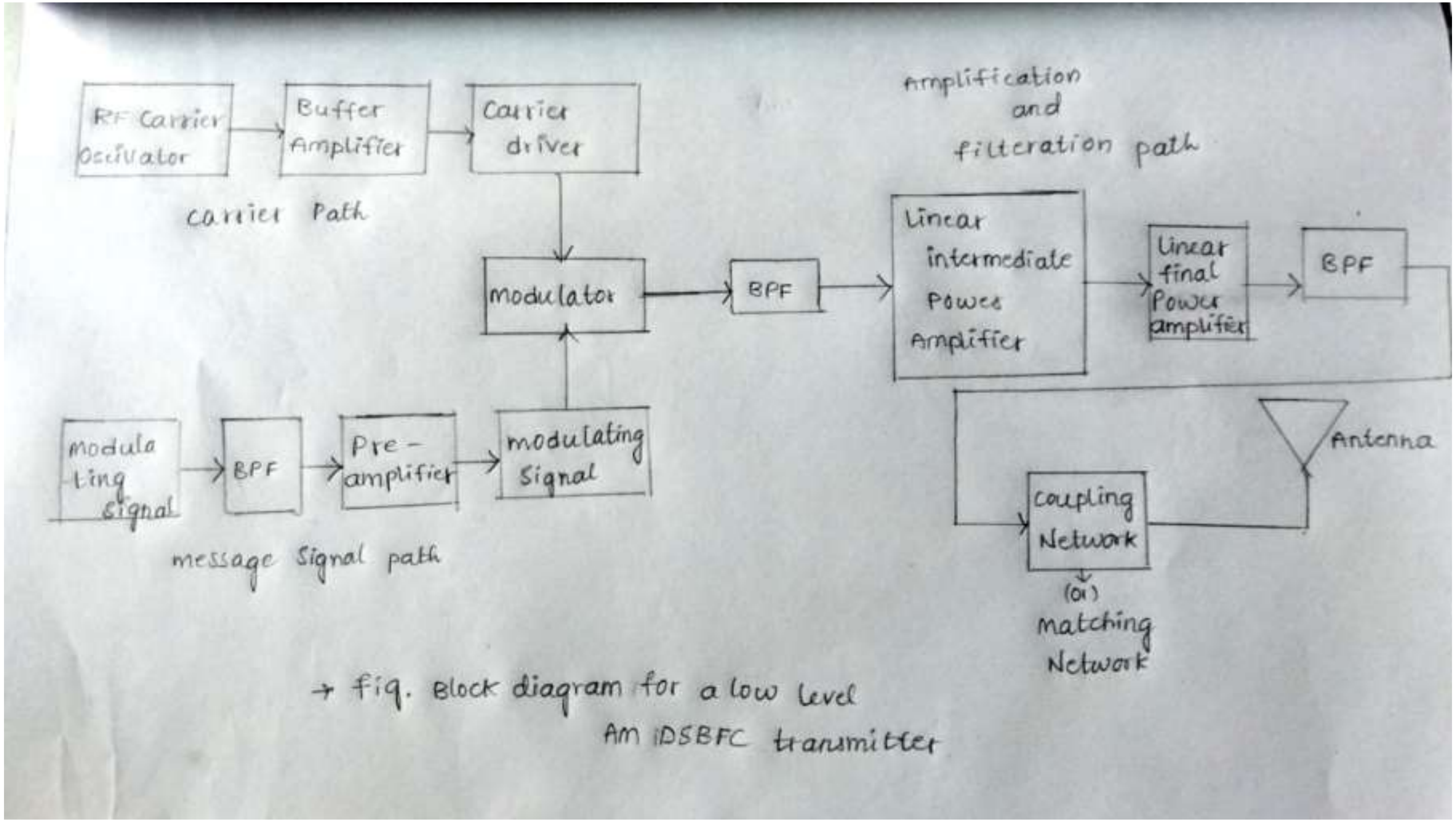
3.PM TRANSMITTERS:-

PM Transmitters use pulse modulation technique for modulating the amplitude, width and positive of the pulse carrier

For this it uses pulse amplitude modulation pulse width modulation and pulse position modulation respectively.

AM TRANSMITTERS:-

- 1.Low level transmitters
- Block diagram for a low level AM DSBFC Transmitter
-



Pre amplifier:-

→ Linear voltage amplifier with high input impedance.

→ To raise Source Signal amplitude to a usable level with minimum nonlinear distortion and as littl thermal noise as possible.

Modulating signal driver:-

Amplifies the information signal to an adequate level to sufficiently drive the modulator.

RF carrier oscillator:-

- To generate the carrier Signal.
- Usually a crystal controlled oscillator in used.

Buffer amplifier:-

- Low gain high input Impedance linear amplifier.
- To isolate the oscillator from the high Power amplifiers.

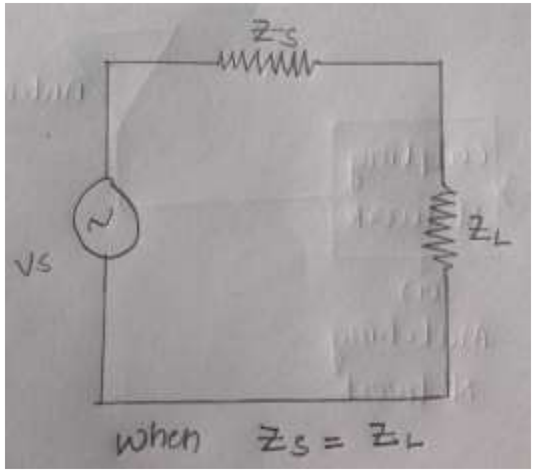
Modulator:-

- Can use either emitter collector modulation.
- Intermediate and final power amplifiers (pull-push modulators).
- Required with low-level transmitters to maintain symmetry in the AM envelope.

COUPLING NETWORK:-

- Matches output impedance of the final amplifier to the transmission line/antenna.

  Maximum power input



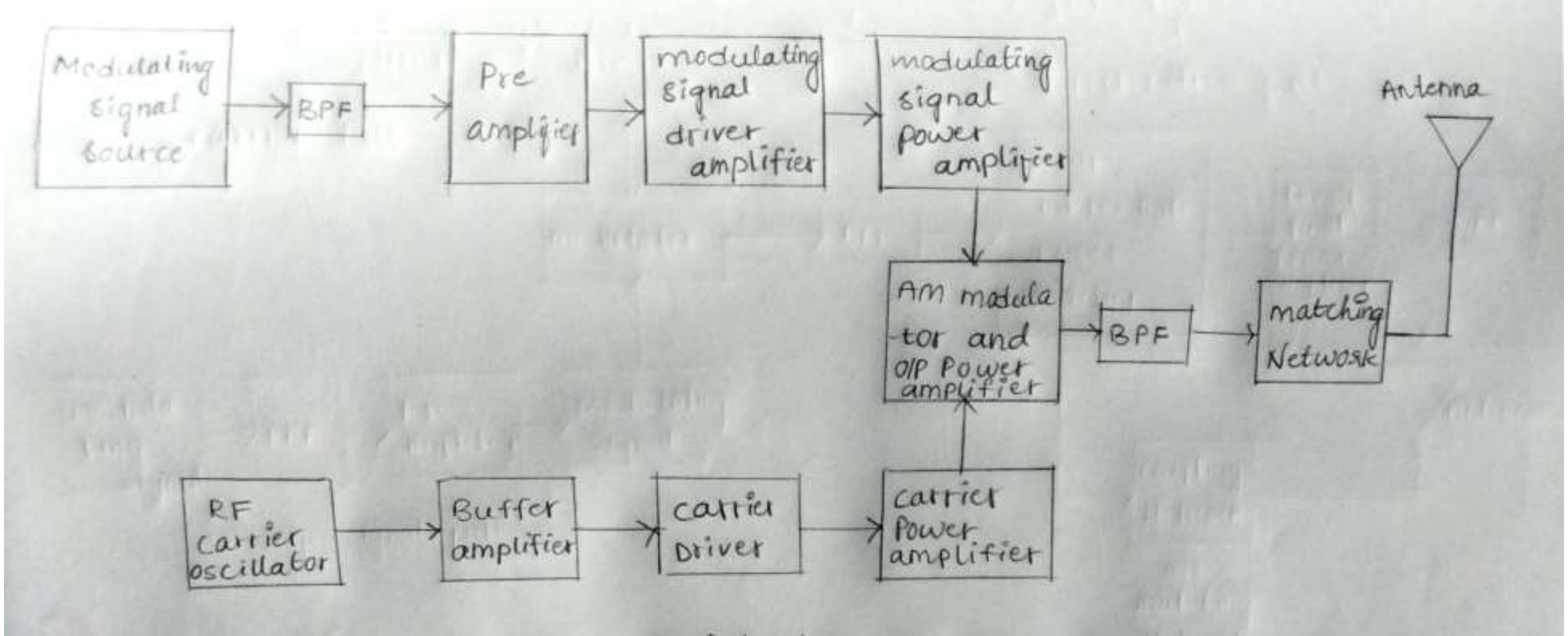
- The maximum power to be transferred to the load from the source if and only if the source impedance is Equals to load impedance. Ie., Zs=Zl
- Therefore a matching network / coupling network is required in every transmitter to match the impedances.

POWER AMPLIFIER:-

- To provide high power modulating signal necessary to achieve 100% modulation.
- Same circuit as low level transmitter for carrier oscillator , buffer and driver but with addition of power amplifier.

HIGH LEVEL TRANSMITTER

- Block diagram for a high level of AM DSBFC Transmitter



→ fig. Block diagram for a high level of AM DSBFC transmitter.

FM TRANSMITTERS

- Block diagram of a FM transmitter

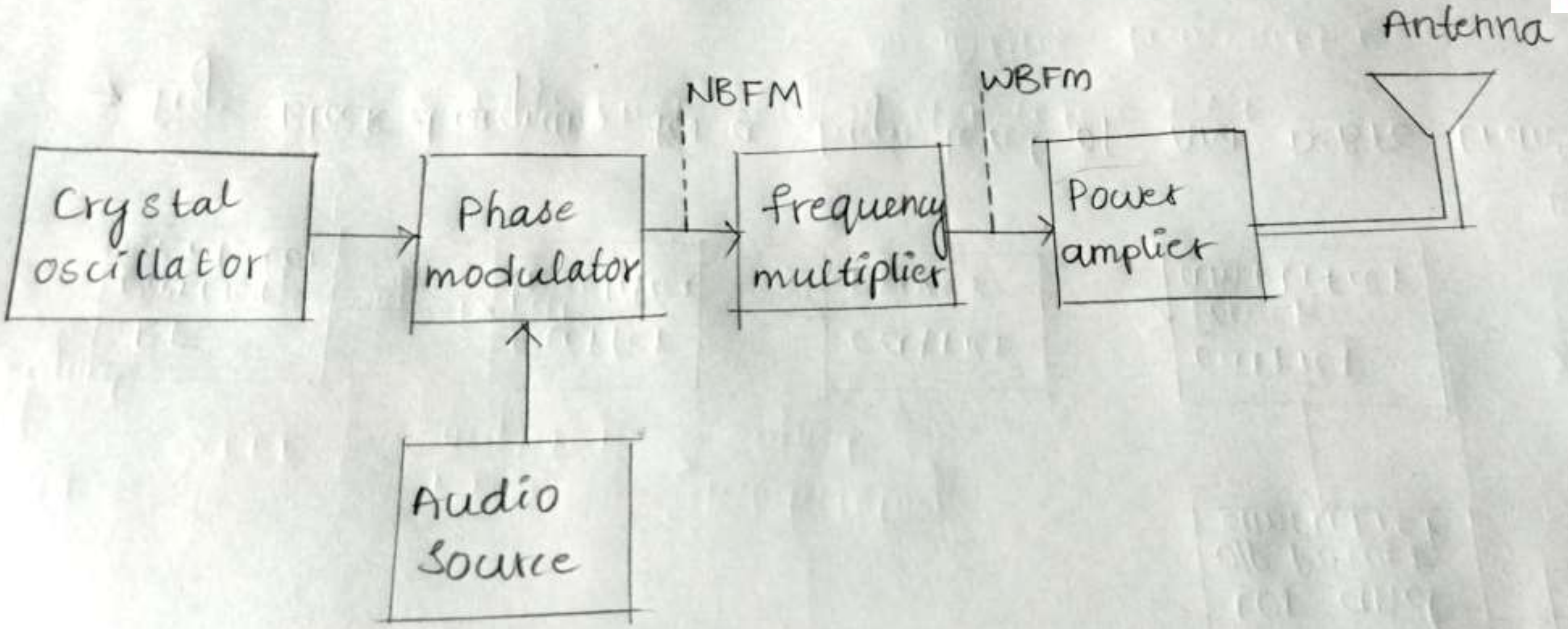


fig. FM transmitters

Crystal Oscillator:-

Crystal oscillator generates the stable carrier Signal

Phase modulator:-

The phase modulator modulates the carrier Signal and the message signal in the low power range to generate a NBFM.

Frequency multiplier:-

The frequency multiplier is used to increases the frequency deviation and Carrier Sigral frequency to a derived level.

Power amplifier:-

The power amplifier gives the required power level to the signal which passes through the antenna.

Antenna:-

Antenna is a device which is used for sending and receiving the information.

RECEIVERS

- A Receiver is an electronic device, which recovers the original message signal from the modified message signal.

➢ RADIO RECEIVER:-

- A radio receiver is an electronic device which picks up a desired modulated radio frequency signal and recovers the base band signal from the modulated signal.

➢ TYPES

- The radio receiver can be classified as,

1.AM Receivers

2.FM Receivers

➢ AM Receivers are further classified as,

- Tuned radio frequency (TRF) receivers
- Superheterodyne(SHR) receivers.

TUNED RADIO FREQUENCY (TRF) AMPLIFIER:-

- Block diagram of a tuned radio frequency amplifier.

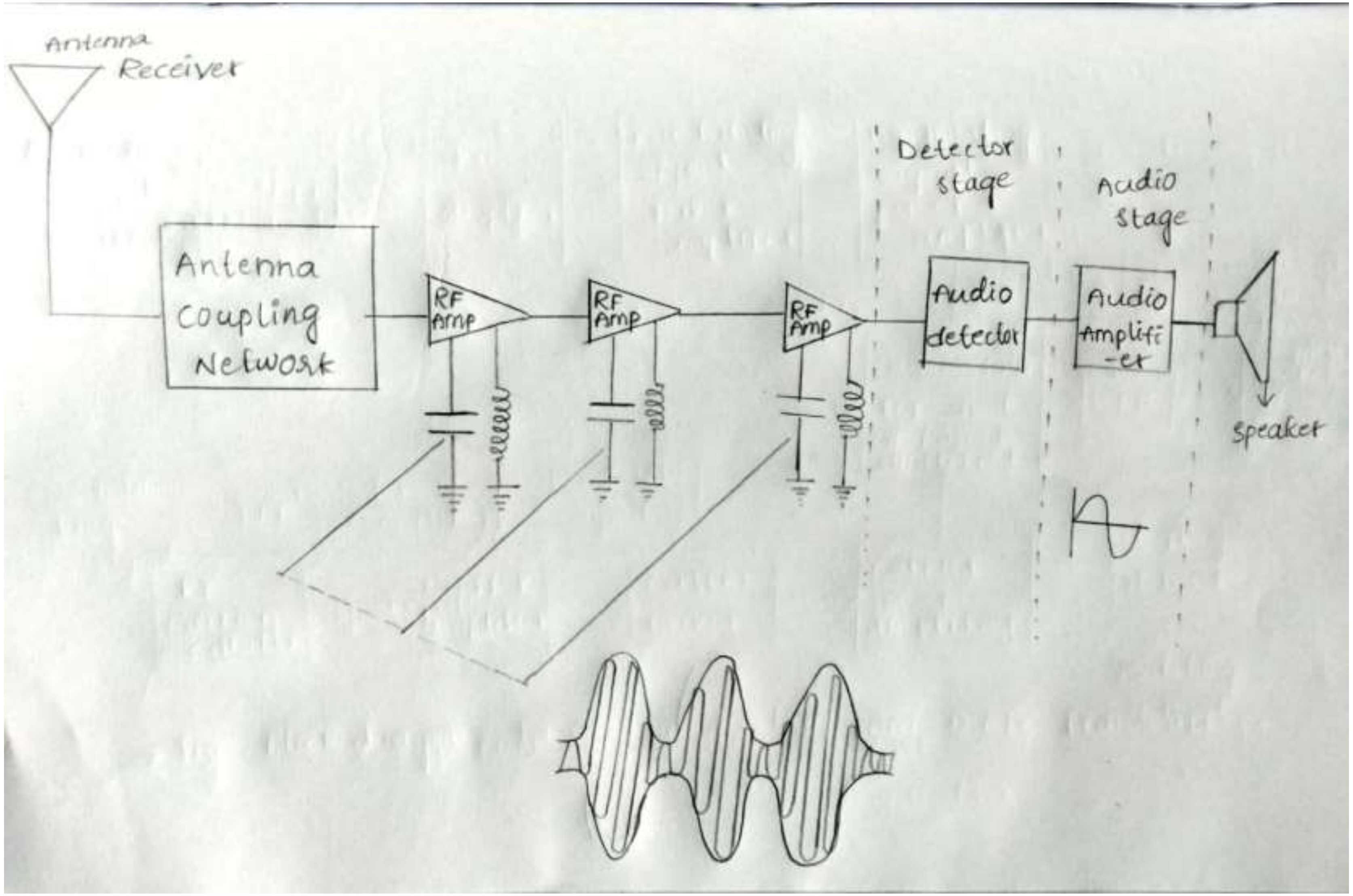


Fig. TRF Receiver

TUNED RF RECEIVER(TRF):-

- It is the earliest and simpliest Receivers design.

TRF Consist of RF amplifiers stages, detector and audio amplifier stages.

•The received signal is turned by LC circuit to a passband centered at Carrier frequency.

selectivity pass only the desired signal, others are rejected.

- The tuned signal is boost up by an amplifier for better information detection.

Signal info detection is made at the demodulator & further amplified for the Speaker olp.

Disadvantages:-

- BW is in consistent and varies with center frequency when turned over a wide range of i/p frequencies selectivity changes, (means the extent to which a Rx can differentiate between the derived signal and other signal)
- Instability due to the large no.of RF amplifier all turned to the same center.
- Frequency:- oscillation
- Gain is not uniform over a wide range of frequency.

SUPERHETRODYNE RECEIVER:-

- Block diagram of superhetrodyne receiver.

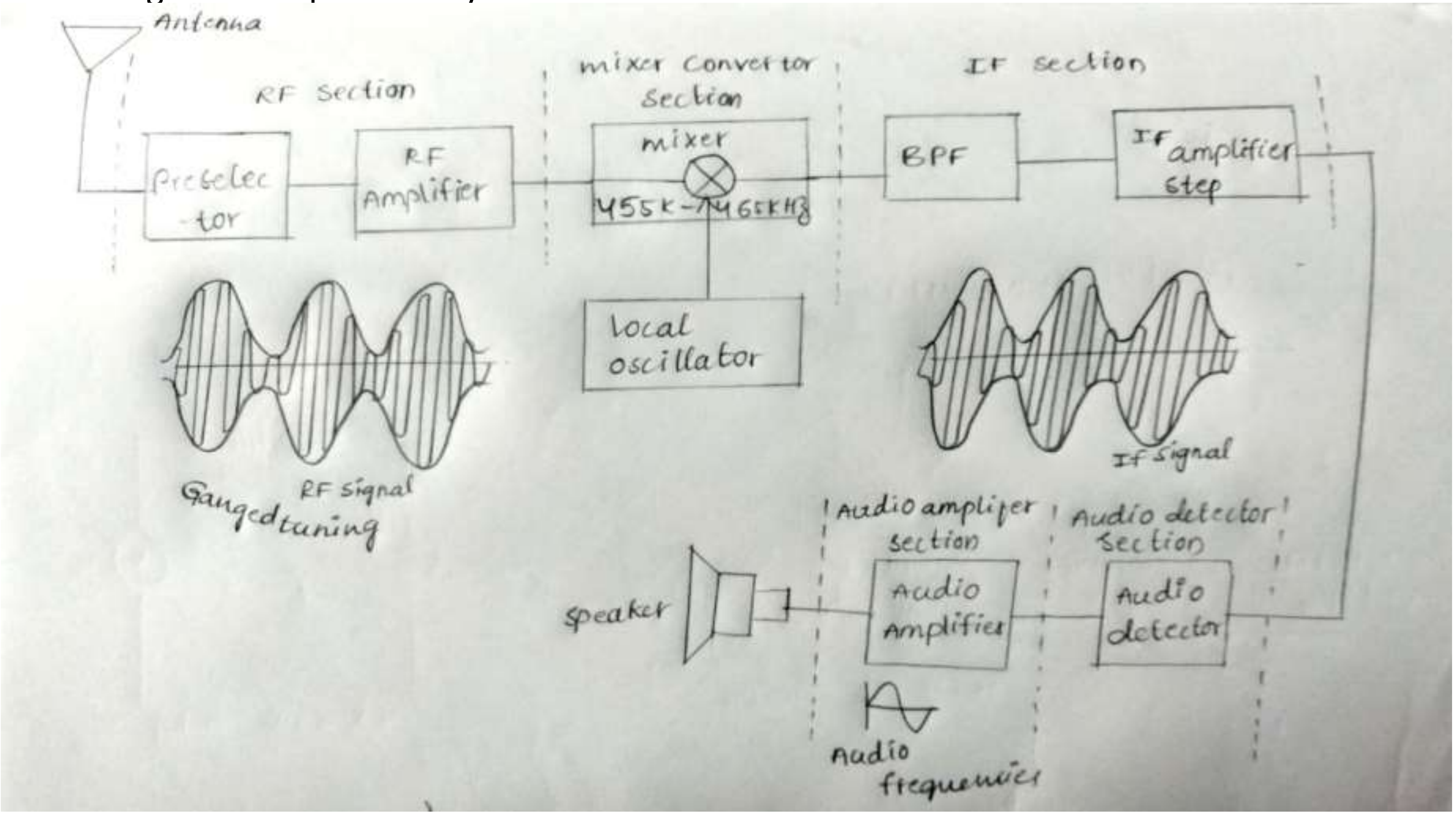


Fig.SHR(Super heterodyne receiver)

Super heterodyne receiver:-

- Superhets was designed to overcome the problems in TRF
- Complex circuitry compared to TRF but excellent performance under many conditions.

HETERODYNE means :-

- To mix two frequencies together in a non linear device or to translate one frequency to another using nonlinear device.

SUPERHETS concept:-

- It tunes into desired signal and converts the signal to intermediate frequency via a signal.
- Then IF signal is optimized to fully received the modulated into signal.

Stages in superhets:-

1. RF stage:-

- Which takes the signal from the antenna and amplifies it to a level large enough to be used to the following stages.

Mixer and local oscillator:-

- Converts the RF signal to IF signal .

IF signal:-

- Further amplifies the signal and has Bandwidth and passband shaping appropriate for the received signal .

Detector stage:-

- Recovers (demodulates) the information signal from the carrier signal.

AF stage:-

- The received signal is amplified for loudspeaker or inter connection to common systems.

RF section and characteristics-frequency changing and tracking:-

- RF section is a pre- requisite for radio receiver . It is usually a tunable circuit connected to the terminals of antenna. Its function is to select the desired frequencies and reject the undesired ones from the receiver. A receiver with RF section need not require any RF amplifier. If at all an RF amplifier is used , its output is fed to the mixer for which the input is another tuned circuit .
- RF amplifier offers several advantages to RF section of the receiver. Some of them are,

1. High gain and good signal to noise ratio
2. Good sensitivity and selectivity
3. High reliability and long life
4. Low cost
5. Low power consumption

AUTOMATIC GAIN CONTROL(AGC):-

- Automatic gain control is a circuit which maintains constant output by adjusting the overall gain of the receives according to the strength of the input signal received (ie.., if RF input signal is weak AGC automatically increases the gain of receiver and vice versa).
- Such adjustments in gain of the receiver are done by providing a DC bias voltage to the RF or IF or MIXER .
- TYPES OF AGC
- There are two types of AGC circuits in use. They are,
- 1.Simple AGC
- 2.Delayed AGC

1.SIMPLE AGC

- The AGC circuit in which bias of the receiver increases with increases in received signal level (to an amount greate than the thermal noise) is called SIMPLE AGC.
- This type of AGC circuit is used for controlling the AC gain of amplifiers. A simple AGC circuit is shown in figure.

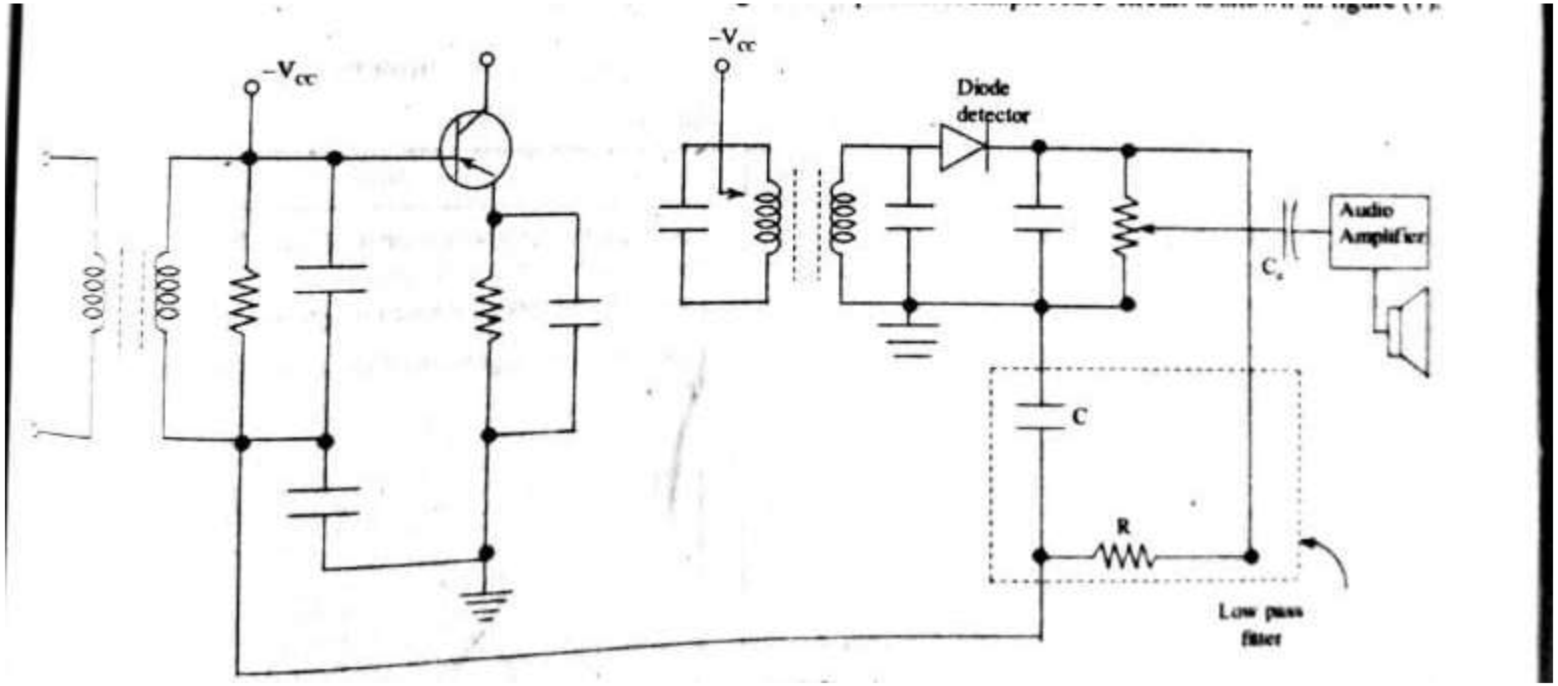
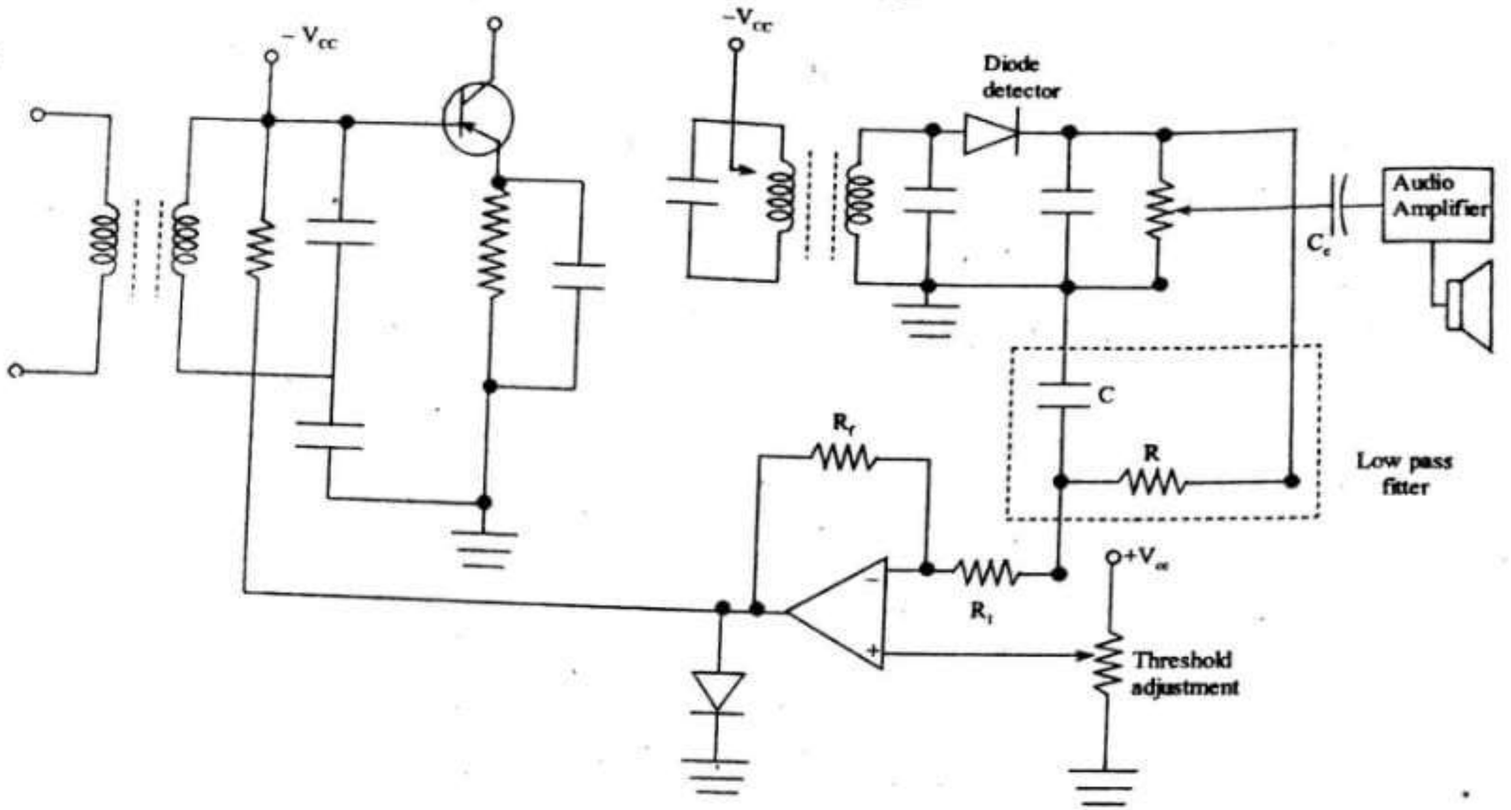


Fig. simple AGC

- In this circuit arrangement , half –wave rectifier is used to provide DC bias . The DC bias is then passed through a low pass filter for removing the AC content from it. The low pass filter (LPF) is so chosen that its time constant is 10 times greater than the period of received signal. The resultant output of LPF is fed to RF or IF stages.

2.DELAYED AGC

- The AGC circuit in which AGC is applied only after the signal strength reaches a predetermined threshold delayed AGC.
- A delayed AGC circuit prevents the AGC feedback voltage from reaching the RF or IF amplifier until the RF level exceeds the predetermined magnitude . A delayed AGC circuit is shown in figure.



AMPLITUDE LIMITING :-

In an FM receiver , when the amplitude variations present in the IF signal are fed directly to be demodulator, noise produced in the circuit. To prevent such noise from entering into the receiving section, the FM receiver employs an amplitude limiter circuit in front of the demodulator. This process is know as amplitude limiting.

The amplitude noise limiter circuit removes the amplitude variations from the signal and allows only frequency variations of signals to reach the demodulator circuit . The circuit diagram of an amplitude limiter (or) FM noise limiter is as shown in fig.

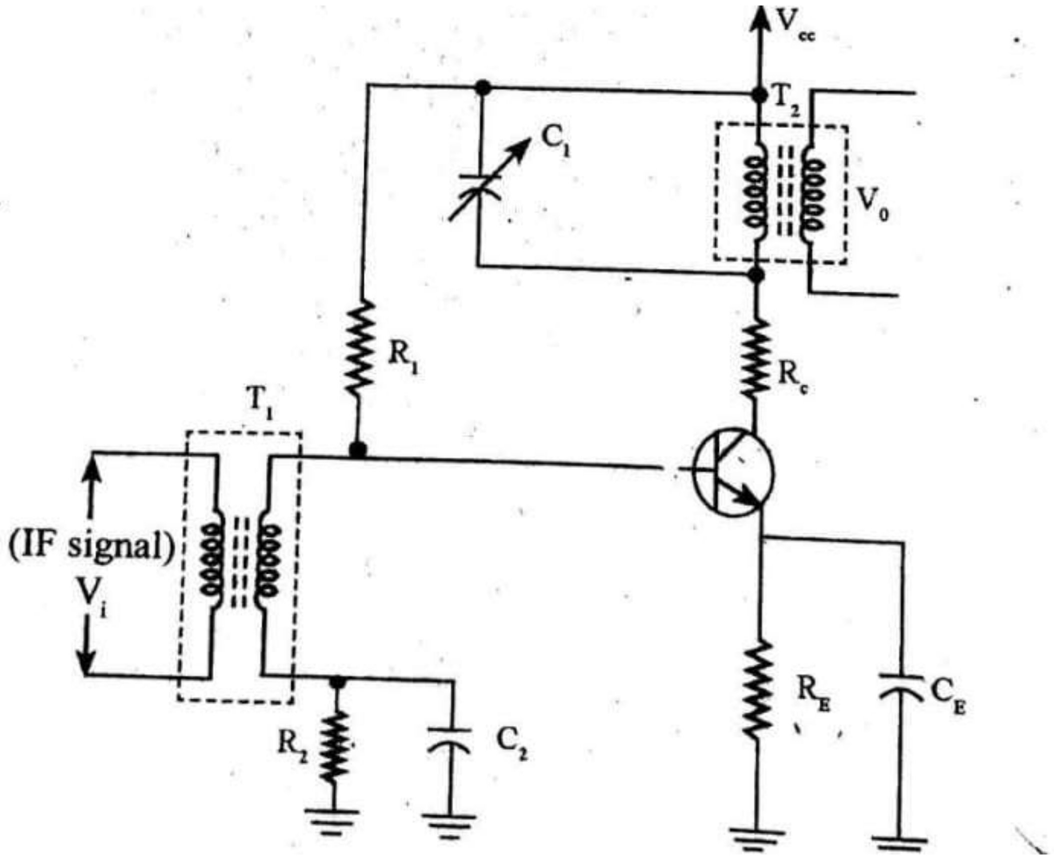


Fig.Amplitude limiter (or) FM noise limiter circuit

- The limiter circuit consists of a bipolar transistor, two double tuned IF transformers al with R_1, R_2, R3, resistors. These resistors maintain the transistor in active region, by providing D.C bias under zero signal conditions. The main function of a limiter is to maintain the transistor in active region throughout the receiving operation. If the level of the input signal voltage is low, the limiter acts as a normal class A amplifier. Whenever the input signal voltage level becomes larger than the active range. the transistor tends to be driven from cut-off to saturation region.
  - The point forms the threshold of limiter as it is the point where actual limiting action begins. The Rc resistor connected across the collector, provides effective supply voltage and reduces the collector current, responsible for driving the transmitter to saturation region.

FM RECEIVER

- The block diagram of an FM receiver is as shown in figure.

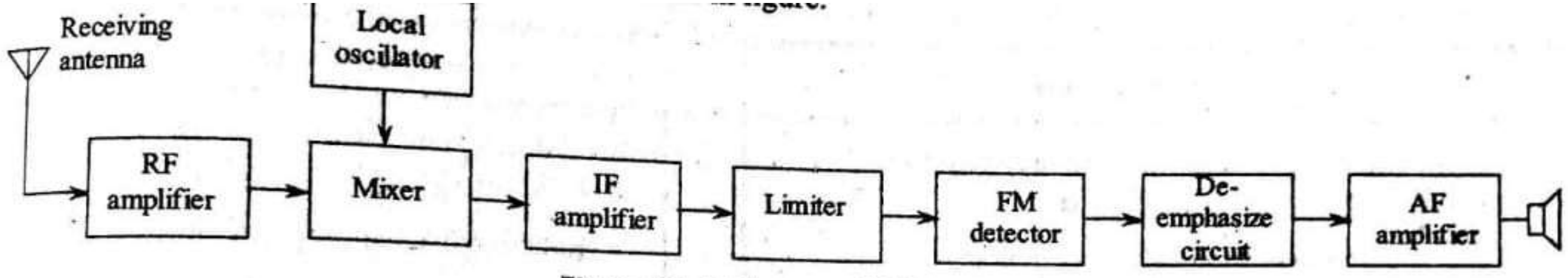


Figure: Block Diagram of FM Receiver

1. RF AMPLIFIERS:-

- RF amplifiers are used in FM receivers to minimize the noise figure and to match the input impedance of the receiver with the impedance of antenna. These amplifiers increase the strength of the signal to a satisfactory level and feeds the amplified output to the mixer.

2. LOCAL OSCILLATOR:-

- The local oscillator used in FM receivers generates carrier waves of frequency lower than the input signal frequency.

3.MIXER:-

- A mixer is used to combine the RF amplified output with the output of local oscillator to produce a high intermediate frequency(IF). This high IF helps in attaining effective image rejection capability of a receiver.

4.IF AMPLIFIERS:-

- IF amplifiers are used for amplifying intermediate frequencies. These amplifiers provide high gain and larger bandwidths of the order of 150 kHz.

5.LIMITER:-

- Limiter is a form of clipping device, in which the output remains constant irrespective of the variations in the input signal. FM receivers uses an amplitude limiter to clip off the amplitude variations present in the signal. As a result, noise gets reduced without affecting the information content of the signal. The constant frequency modulated carrier is then applied to a discriminator circuit.

6.DISCRIMINATOR:-

- A discriminator or an FM detector, applied next to the limiter circuit, extracts the original audio frequency from the Frequency Modulated (FM) carrier.

7.DE-EMPHASIS NETWORK:-

- A de-emphasis circuit is employed to reduce the high audio frequencies which are directly proportional to the frequencies of transmitter. These circuits also help in reducing the frequency-modulated noise which enters the front-end of the receiver.

8.AF POWER AMPLIFIER:-

- The audio frequency power amplifier input from the de-emphasis network , amplifies the audio signal to a desired level. This amplified output is then fed to the loudspeaker at the receiving end.

COMPARISON OF AM AND FM RECEIVERS IS AS FOLLOWS

AM RECEIVERS

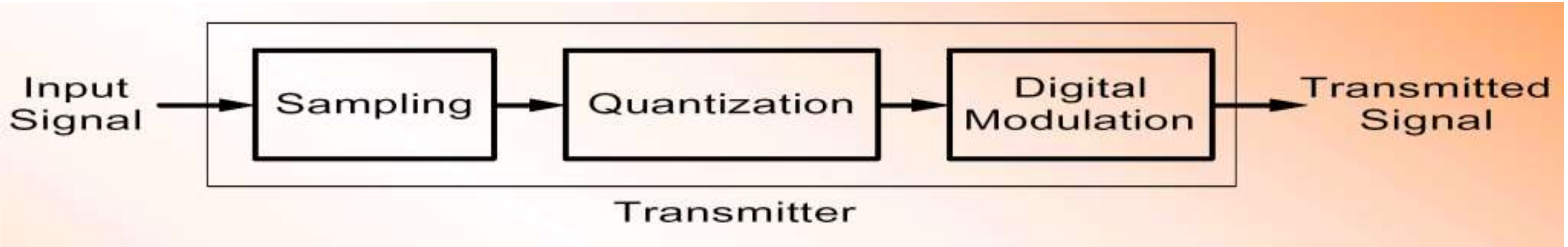
1. An AM receiver operates within a frequency ranges of 540 kHz to 1600 kHz.
2. It requires less bandwidth (i.e., 8 kHz).
3. It is highly effected by noise.
4. It is a superheterodyne type receiver.
5. Special circuits such as limiters, AGC and beat frequency oscillators are not found in AM.
6. It provides less gain.
7. The tuning range of AM receivers is about 2:1.
8. Signal to noise ratio is less in AM receivers.
9. It does not require de-emphasis circuit.
10. It provides less selectivity.
11. It offers low fidelity.
12. Interference is more at the receiving section.

FM RECEIVERS

1. An FM receiver operates at UHF and VHF of 88MHz to 108 MHz.
2. It requires more bandwidth upto 15 kHz.
3. It is less effected by noise.
4. It is also a superheterodyne receiver.
5. FM receiver uses limiters, AGC and beat. frequency oscillators for its operation.
6. It provides comparatively more gain than AM receivers.
7. The tuning range of FM receivers is nearly 1.23:1.
8. Signal to noise ratio is more in FM receivers.
9. It requires de-emphasis circuit to recover the original signal.
10. Selectivity is more in FM receivers.
11. It offers high fidelity.
12. Interference is less at the receiving section.

UNIT-4
PULSE MODULATION

Digital Representation of Analog Signals



Sampling Theorem

Sampling Theorem (or Nyquist Criterion):

Statement:

"If a signal is sampled at a rate at least, but not exactly equal to twice the max frequency component of the waveform, then the waveform can be exactly reconstructed from the samples without any distortion"

$$f_s \geq 2 f_{\max}$$

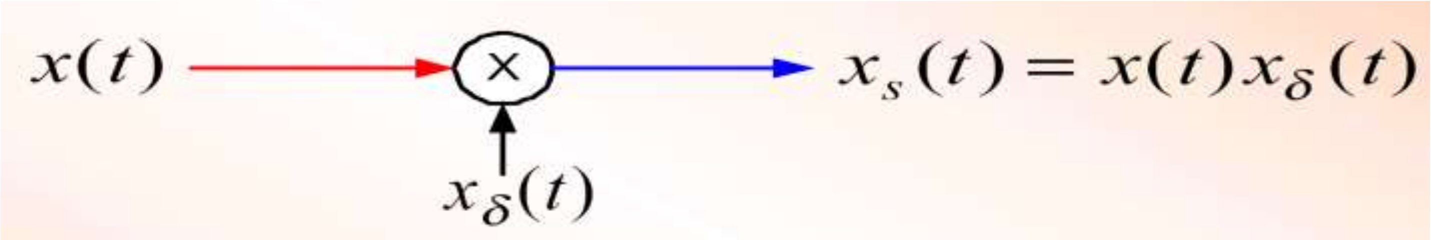
Sampling

Types of sampling

- Ideal Sampling
- Natural Sampling
- Flat-Top Sampling

Ideal Sampling (or Impulse Sampling)

Is accomplished by the multiplication of the signal *x(t)* by the uniform train of impulses (comb function)
Consider the instantaneous sampling of the analog signal *x(t)*



- Train of impulse functions select sample values at regular intervals

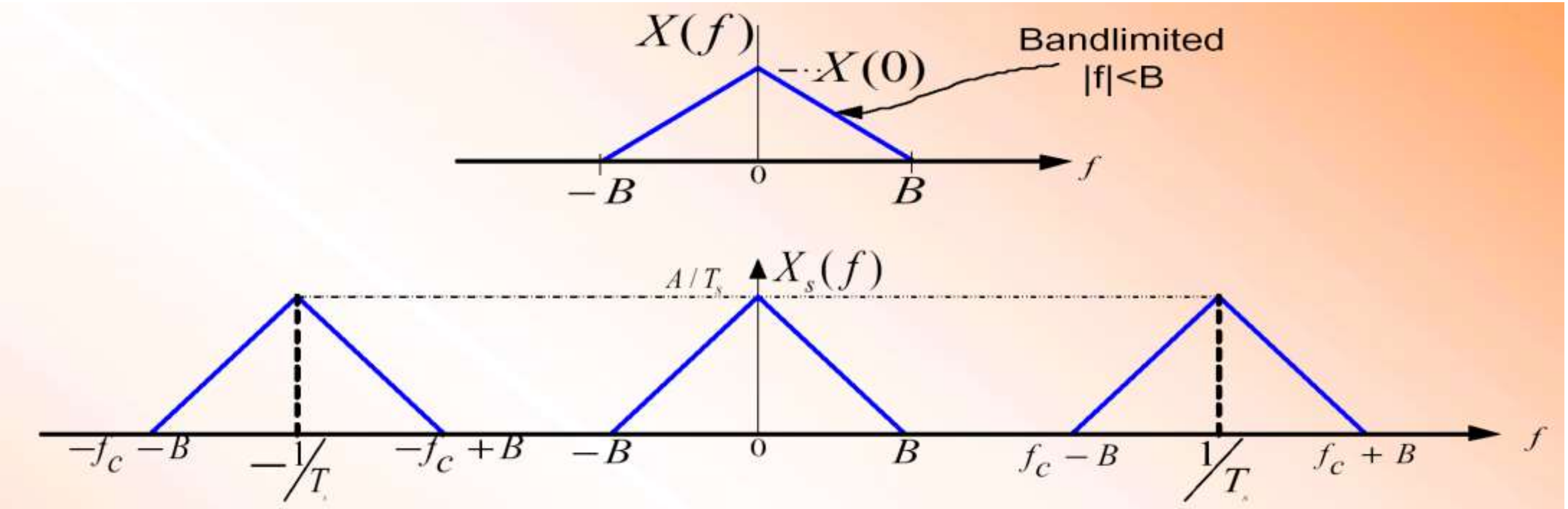
$$x_s(t) = x(t)\sum_{n=-\infty}^{\infty}\delta(t-nT_s)$$

- Fourier Series representation:

$$\sum_{n=-\infty}^{\infty}\delta(t-nT_s) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty}e^{jn\omega_s t}, \qquad \omega_s = \frac{2\pi}{T_s}$$

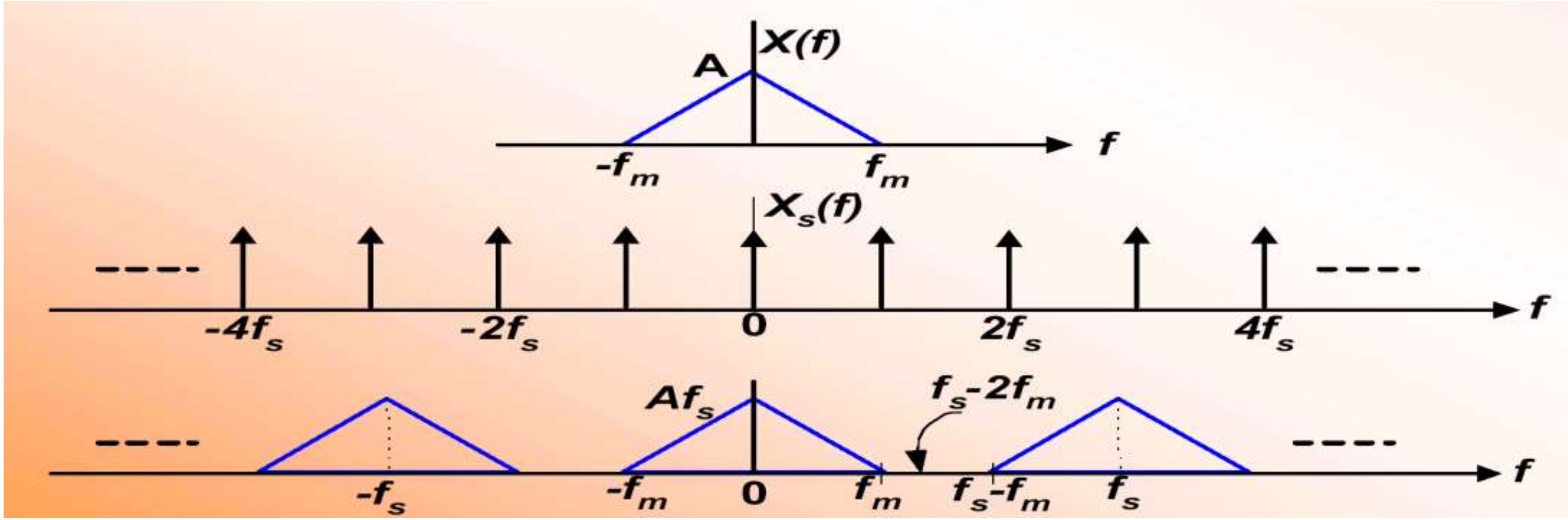
Ideal Sampling (or Impulse Sampling)

- This shows that the Fourier Transform of the sampled signal is the Fourier Transform of the original signal at rate of *1/T_s*



Ideal Sampling (or Impulse Sampling)

- This means that the output is simply the replication of the original signal at discrete intervals, e.g



- T_s is called the ***Nyquist interval:*** It is the longest time interval that can be used for sam bandlimited signal and still allow reconstruction of the signal at the receiver without on

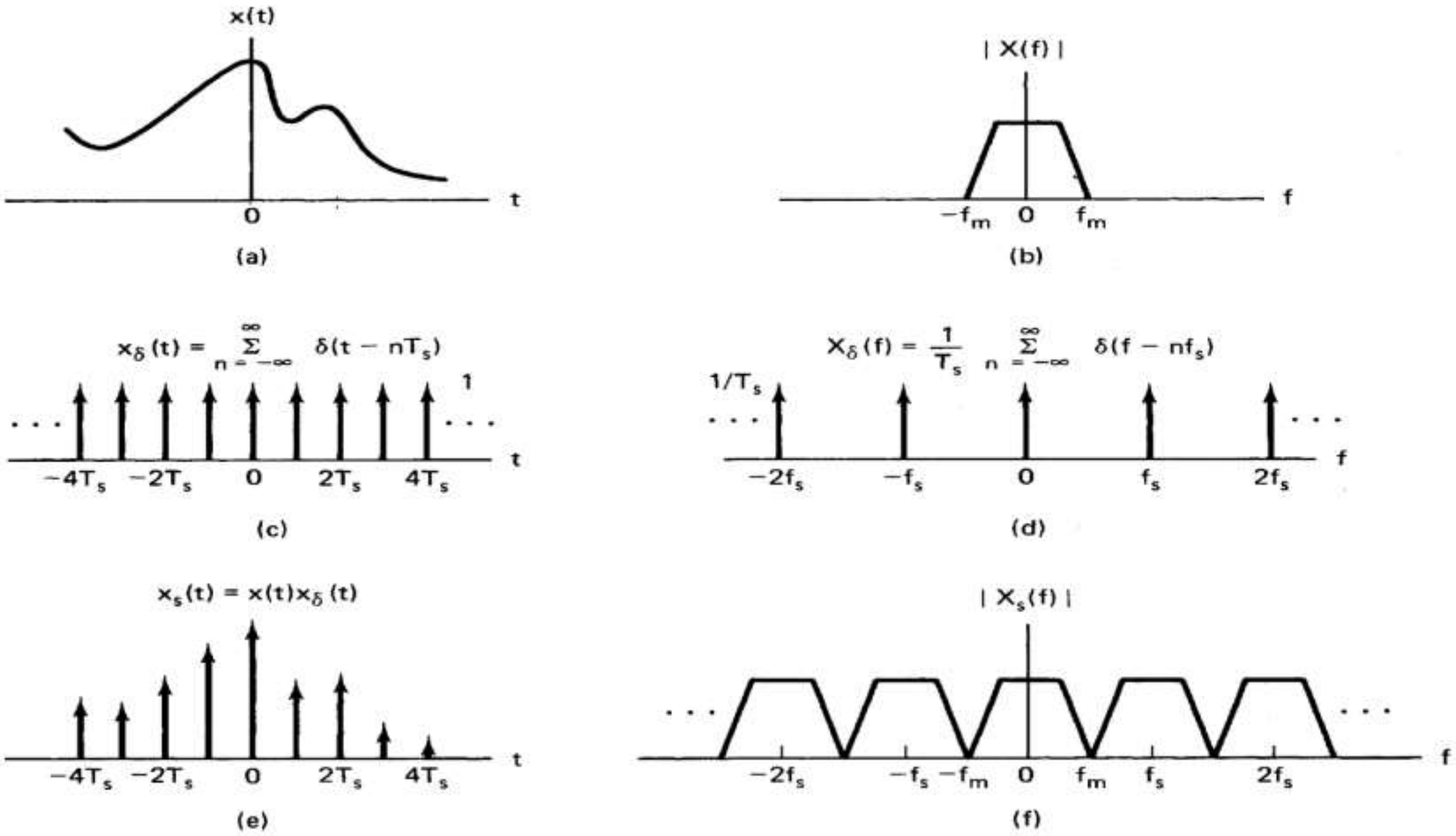
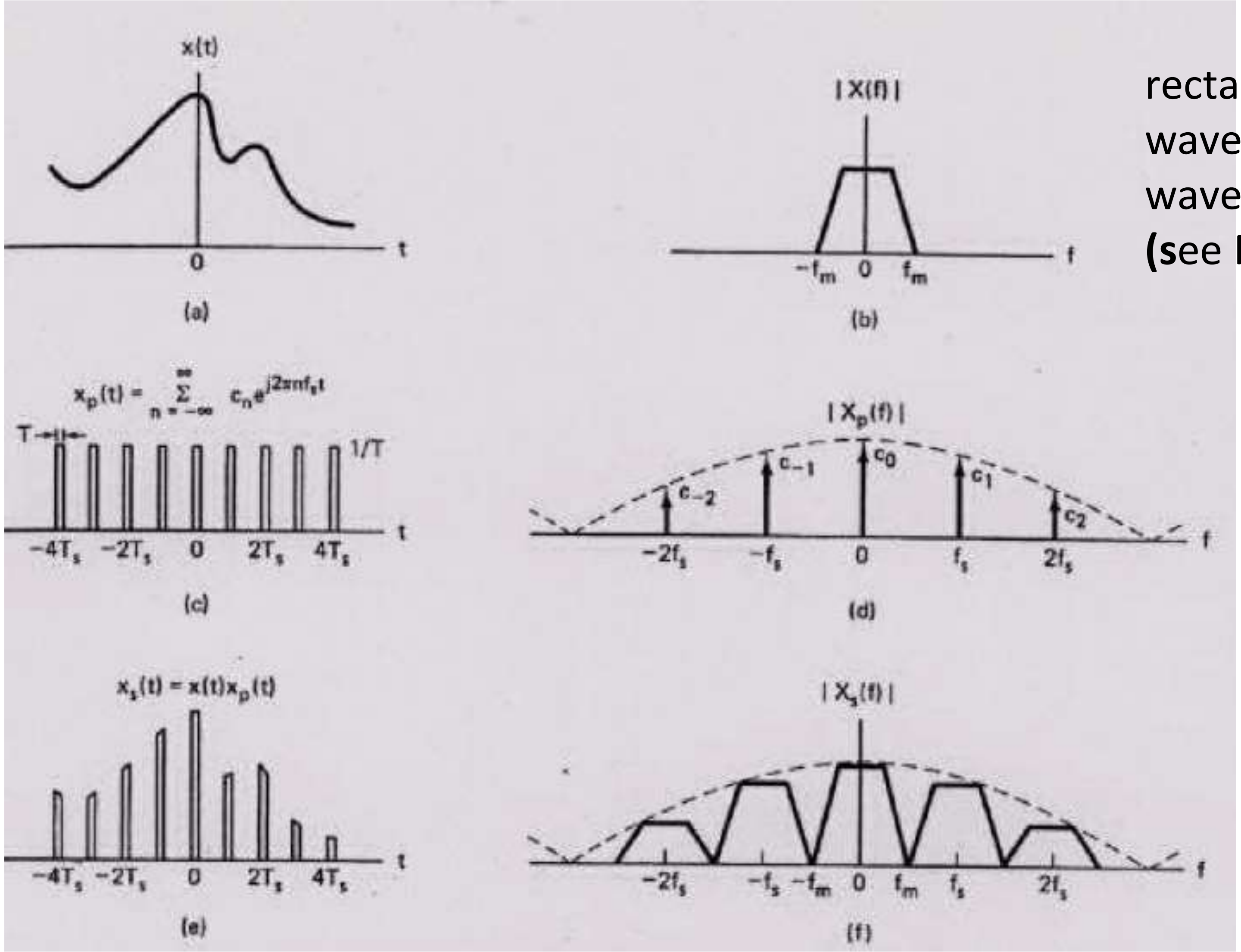


Figure 2.6 Sampling theorem using the frequency convolution property of the Fourier transform.

Natural Sampling



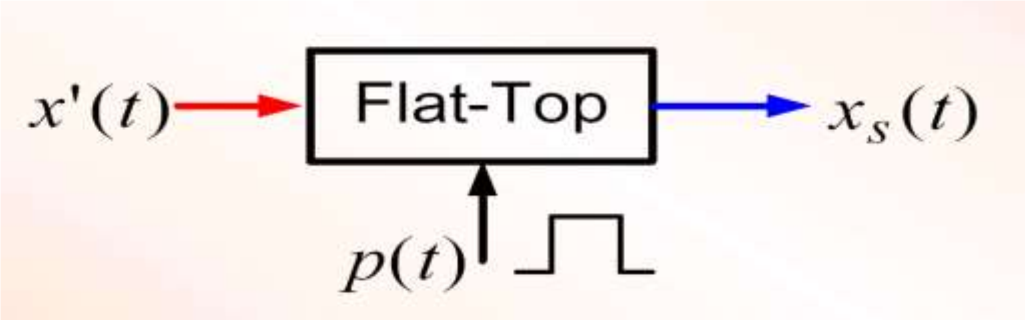
If we multiply *x(t)* by a train of rectangular pulses $x_p(t)$, we obtain a gated waveform that approximates the ideal sampled waveform, known as **natural sampling** or **gating** (see Figure 2.8)

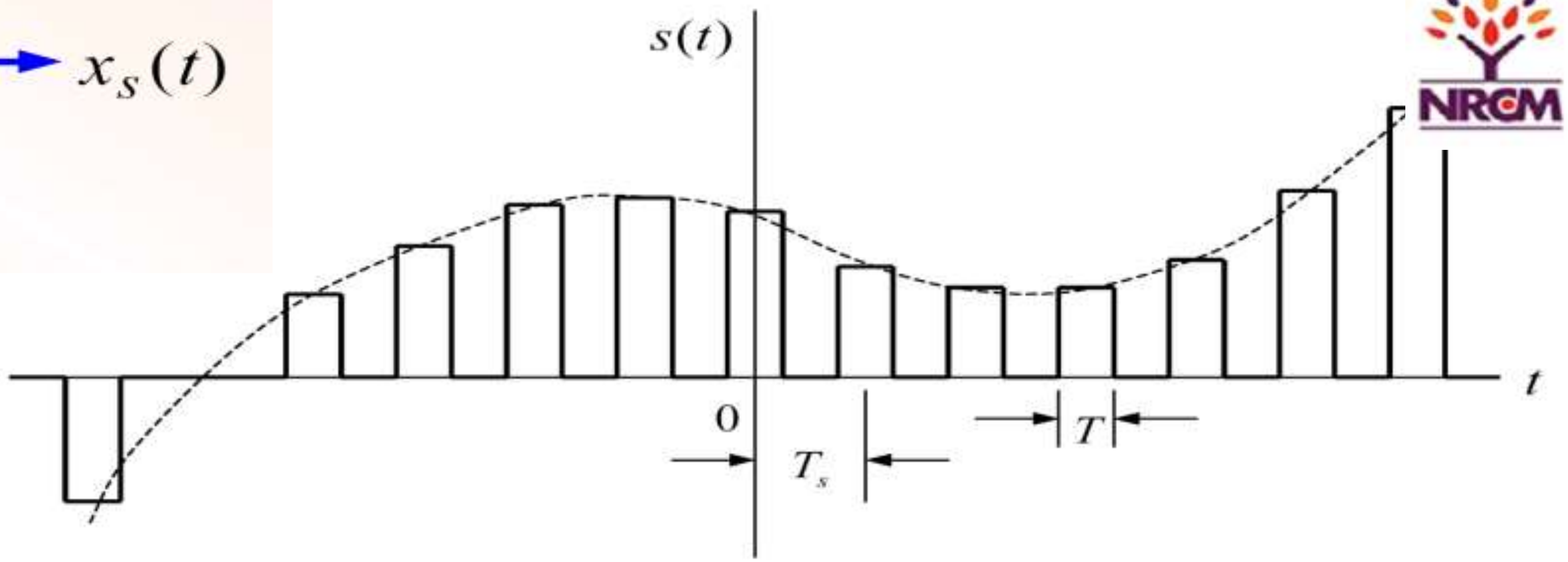
$$x_s(t) = x(t)x_p(t)$$
$$= x(t)\sum_{n=-\infty}^{\infty} c_n e^{j2\pi n f_s t}$$
$$X_s(f) = \Im[x(t)x_p(t)]$$
$$= \sum_{n=-\infty}^{\infty} c_n \Im[x(t)e^{j2\pi n f_s t}]$$
$$= \sum_{n=-\infty}^{\infty} c_n X[f - nf_s]$$

Each pulse in $x_p(t)$ has width T_s and amplitude $1/T_s$
The top of each pulse follows the variation of the signal being sampled
$X_s(f)$ is the replication of $X(f)$ periodically every f_s *Hz*
$X_s(f)$ is weighted by $C_n \leftarrow$ Fourier Series Coeffiecient
The problem with a natural sampled waveform is that the tops of the sample pulses are not flat
It is not compatible with a digital system since the amplitude of each sample has infinite number of possible values
Another technique known as ***flat top sampling*** is used to alleviate this problem

Flat-Top Sampling

Here, the pulse is held to a constant height for the whole sample period
Flat top sampling is obtained by the convolution of the signal obtained after ideal sampling with a unity amplitude rectangular pulse, *p(t)*
This technique is used to realize ***Sample-and-Hold*** (S/H) operation
In S/H, input signal is continuously sampled and then the value is held for as long as it takes to for the A/D to acquire its value





Flat top sampling (Time Domain)

$$x'(t) = x(t)\delta(t)$$

$$x_s(t) = x'(t) * p(t)$$

$$= p(t) * x(t)\delta(t) = p(t) * \left[x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \right]$$

Taking the Fourier Transform will result to

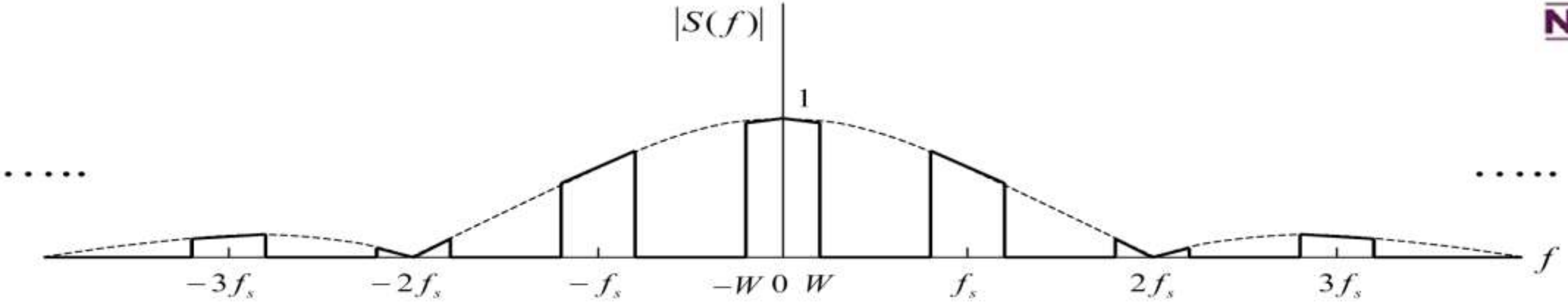
$$X_s(f) = \Im[x_s(t)]$$

$$= P(f)\,\Im\left[x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \right]$$

$$= P(f)\,\Im\left[X(f) * \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \right]$$

$$= P(f) \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

where *P(f)* is a *sinc* function

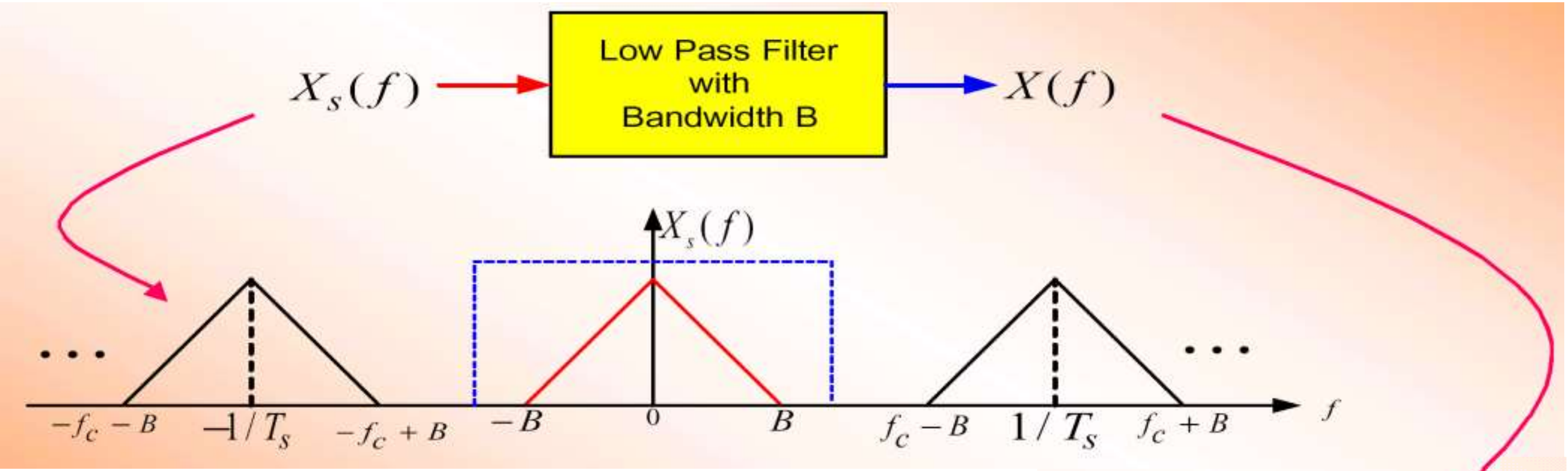


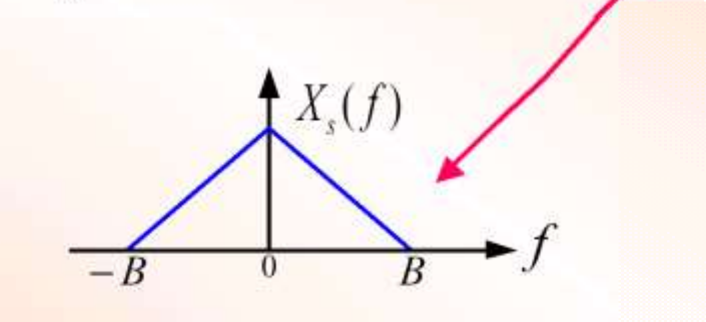
Flat top sampling (Frequency Domain)

- Flat top sampling becomes identical to ideal sampling as the width of the pulses become shorter

Recovering the Analog Signal

One way of recovering the original signal from sampled signal $X_s(f)$ is to pass it through a Low Pass Fi... (LPF) as shown below

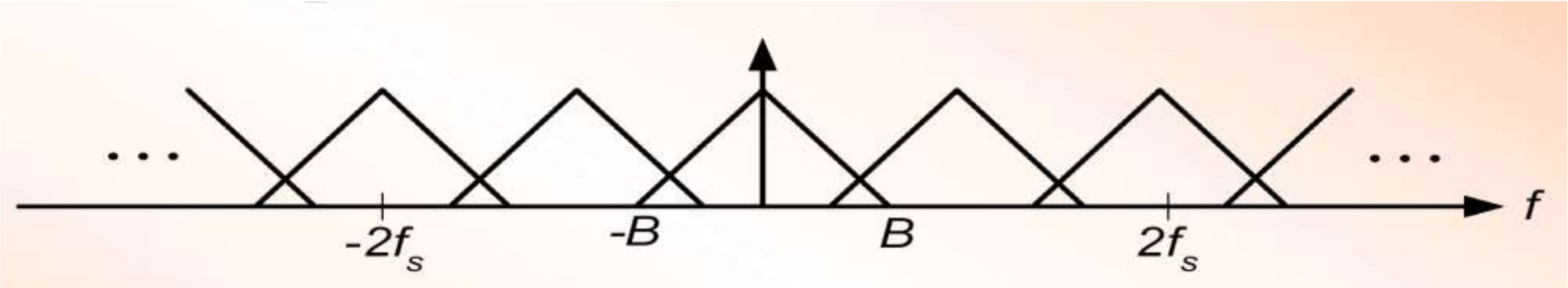




- If $f_s > 2B$ then we recover $x(t)$ ***exactly***
- Else we run into some problems and signal is not fully recovered

Undersampling and Aliasing

If the waveform is ***undersampled*** (i.e. $fs < 2B$) then there will be ***spectral overlap*** in the sampled signal



Solution 2: Over Sampling and Filtering in the Digital Domain

The signal is passed through a low performance (less costly) analog low-pass filter to limit the bandwidth.

Sample the resulting signal at a high sampling frequency.

The digital samples are then processed by a high performance digital filter and down sample the resulting signal.

Practical Sampling Rates

Speech

- Telephone quality speech has a bandwidth of 4 kHz (actually 300 to 3300Hz)
- Most digital telephone systems are sampled at 8000 samples/sec

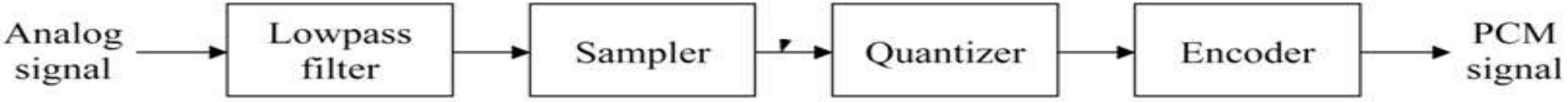
Audio:

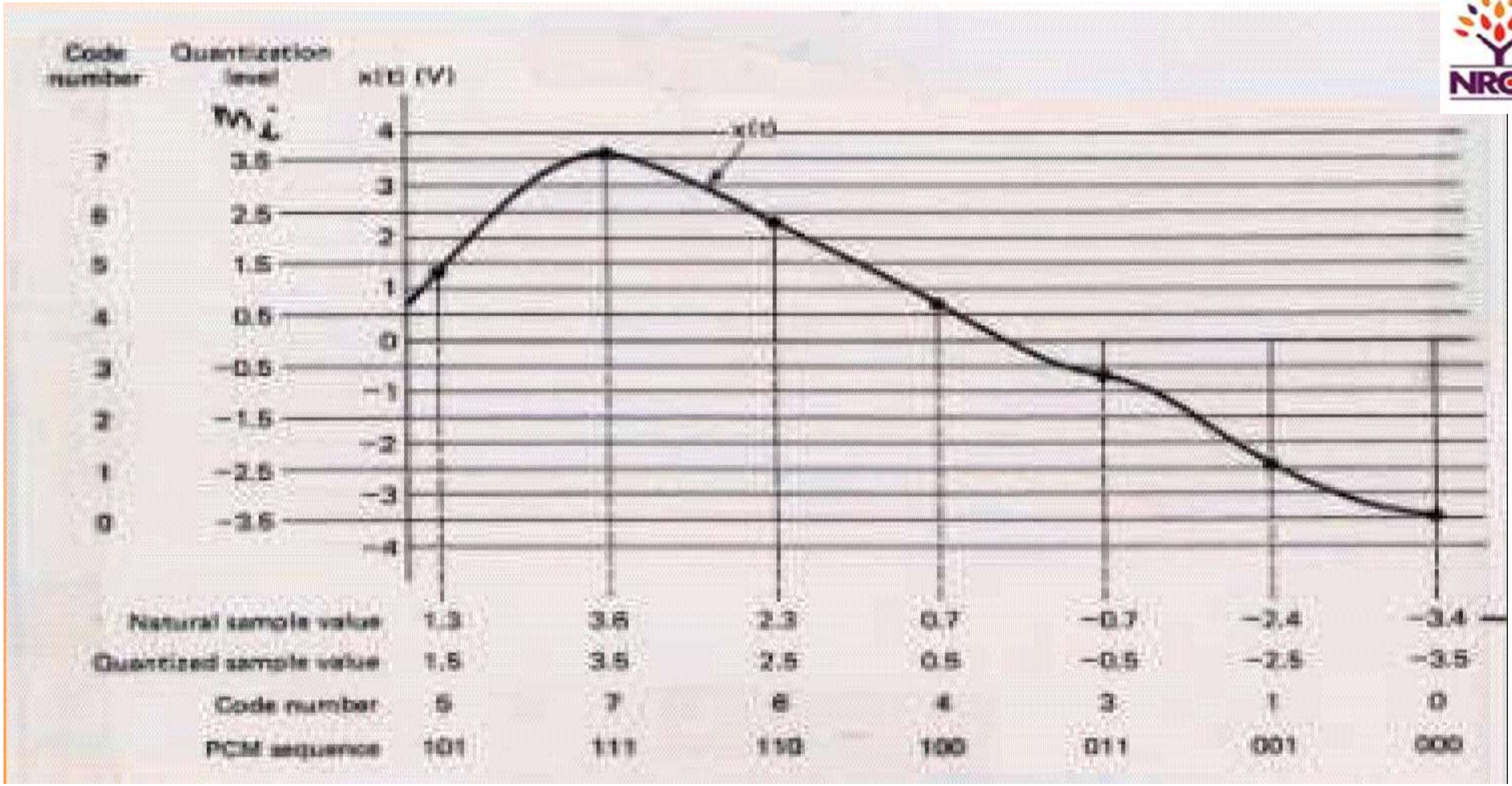
- The highest frequency the human ear can hear is approximately 15kHz
- CD quality audio are sampled at rate of 44,000 samples/sec

Video

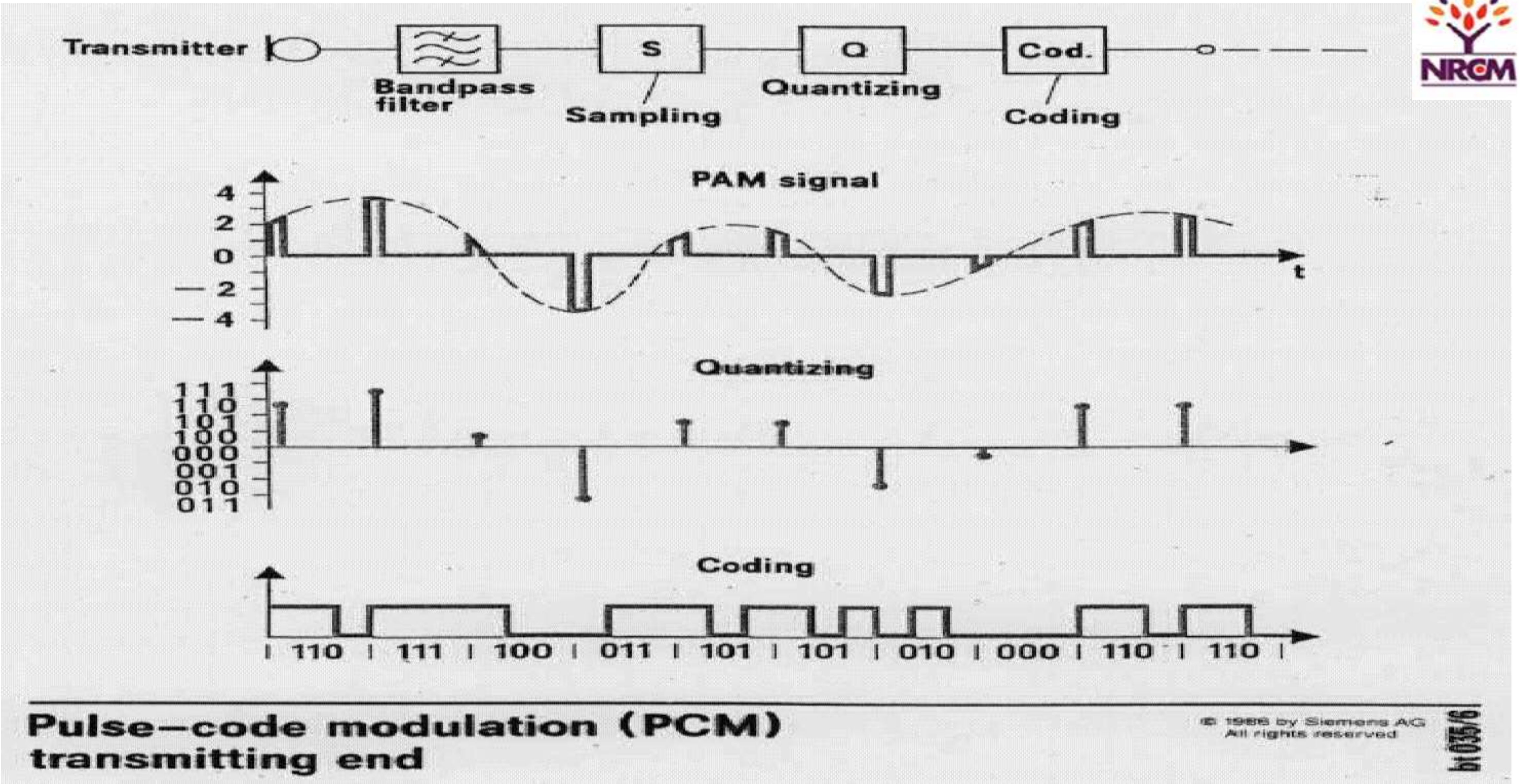
- The human eye requires samples at a rate of at least 20 frames/sec to achieve smooth motion

Pulse Code Modulation (PCM)





See Figure 2.16 (Page 80)


Transmitter
Bandpass filter
S
Sampling
Q
Quantizing
Cod.
Coding
PAM signal
4
2
0
— 2
— 4
t
Quantizing
111
110
101
100
000
001
010
011
Coding
110
111
100
011
101
101
010
000
110
110
Pulse—code modulation (PCM) transmitting end
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bt 035/6

Advantages of PCM:

Relatively inexpensive

Easily multiplexed: PCM waveforms from different sources can be transmitted over a common digital channel (TDM)

Easily regenerated: useful for long-distance communication, e.g. telephone

Better noise performance than analog system

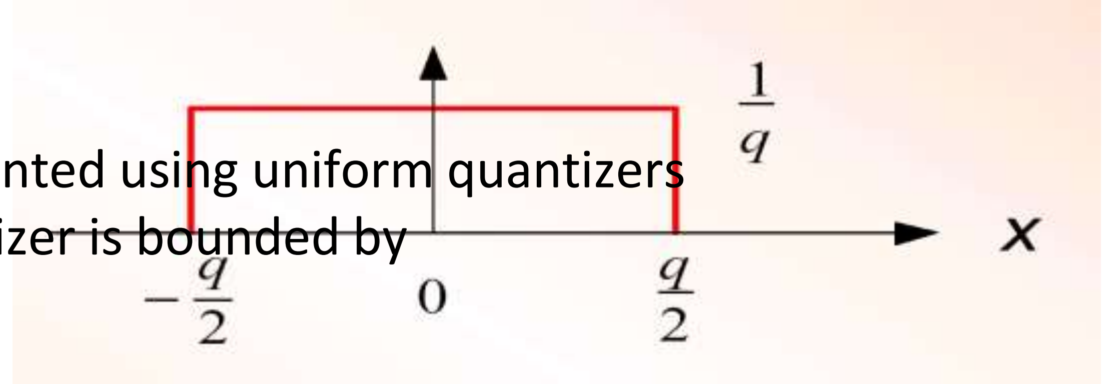
Signals may be stored and time-scaled efficiently (e.g., satellite communication)

Disadvantage:

Requires wider bandwidth than analog signals

Uniform Quantization

A quantizer with equal quantization level is a Uniform Quantizer
Each sample is approximated within a quantile interval
Uniform quantizers are optimal when the input distribution is uniform



Most ADC's are implemented using uniform quantizers
Error of a uniform quantizer is bounded by

$$-\frac{q}{2} < e \leq \frac{q}{2}$$

Signal to Quantization Noise Ratio

The mean-squared value (noise variance) of the quantization error is given by:

$$\sigma^2 = \int_{-q/2}^{q/2} e^2 p(e)de = \int_{-q/2}^{q/2} e^2 \left(\frac{1}{q}\right) de = \frac{1}{q} \int_{-q/2}^{q/2} e^2 de$$

$$= \frac{1}{q} \frac{e^3}{3} \bigg|_{-q/2}^{q/2} = \frac{q^2}{12}$$

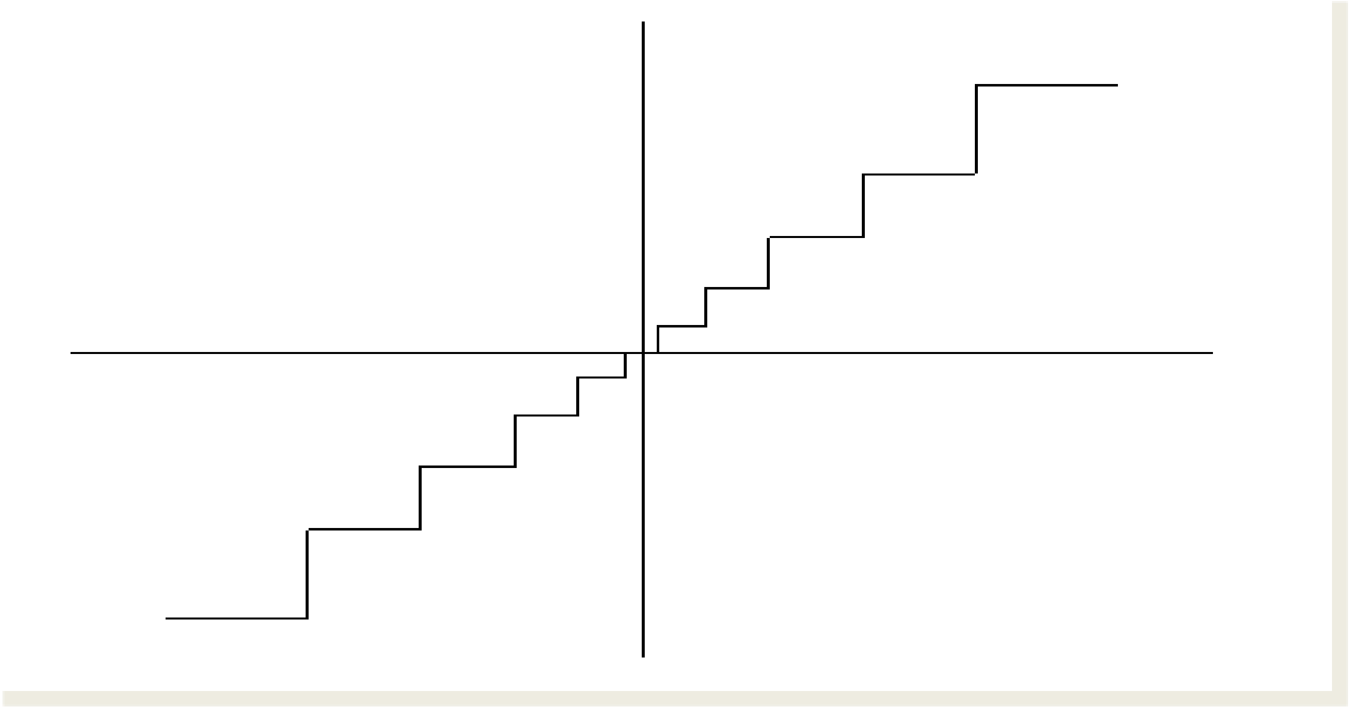
Nonuniform Quantization

Nonuniform quantizers have unequally spaced levels

It is characterized by:

Variable step size

Quantizer size depend on signal size



- Many signals such as speech have a nonuniform distribution
- use fine quantization (small step size) for weak signals and large quantization (large step size) for strong signals

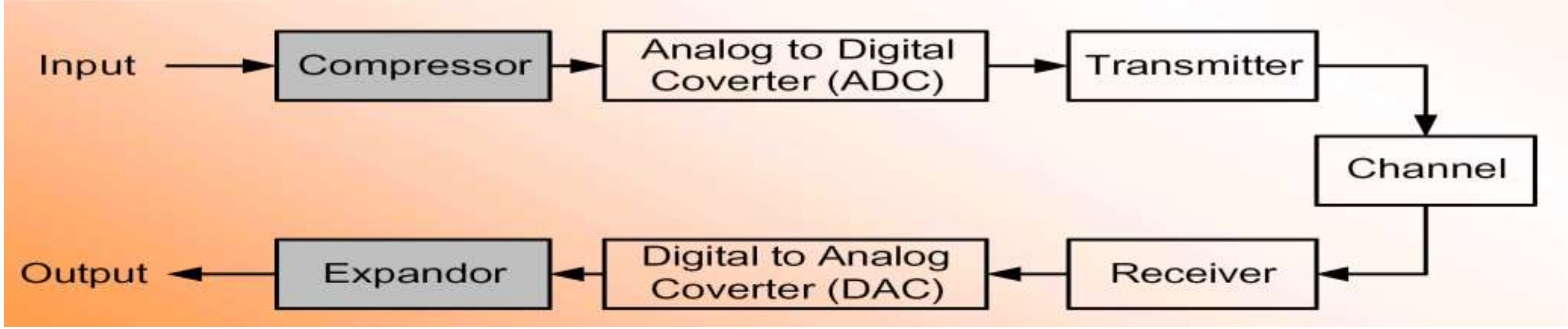
Nonuniform quantization using companding

Companding is a method of reducing the number of bits required in ADC while achieving an equivalent dynamic range or SQNR

In order to improve the resolution of weak signals within a converter, and hence enhance the SQNR, the ***weak signals*** need to be ***enlarged***, or the ***quantization step size decreased***, but only for the weak signals

But *strong signals* can potentially be *reduced* without significantly degrading the SQNR or alternatively increasing quantization step size

The compression process at the transmitter must be matched with an equivalent expansion process at the receiver



There are in fact two standard logarithm based companding techniques

US standard called ***μ-law companding***

European standard called ***A-law companding***

Types of Companding

μ -Law Companding Standard (North & South America, and Japan)

$$y = y_{\max} \frac{\log_e \left[1 + \mu(|x| / x_{\max}\right]}{\log_e(1+\mu)} \operatorname{sgn}(x)$$

where

x and y represent the input and output voltages

μ is a constant number determined by experiment

In the U.S., telephone lines uses companding with $\mu = 255$

- Samples 4 kHz speech waveform at 8,000 sample/sec
- Encodes each sample with 8 bits, $L = 256$ quantizer levels
- Hence data rate $R = 64$ kbit/sec

$\mu = 0$ corresponds to uniform quantization

Delta Modulation

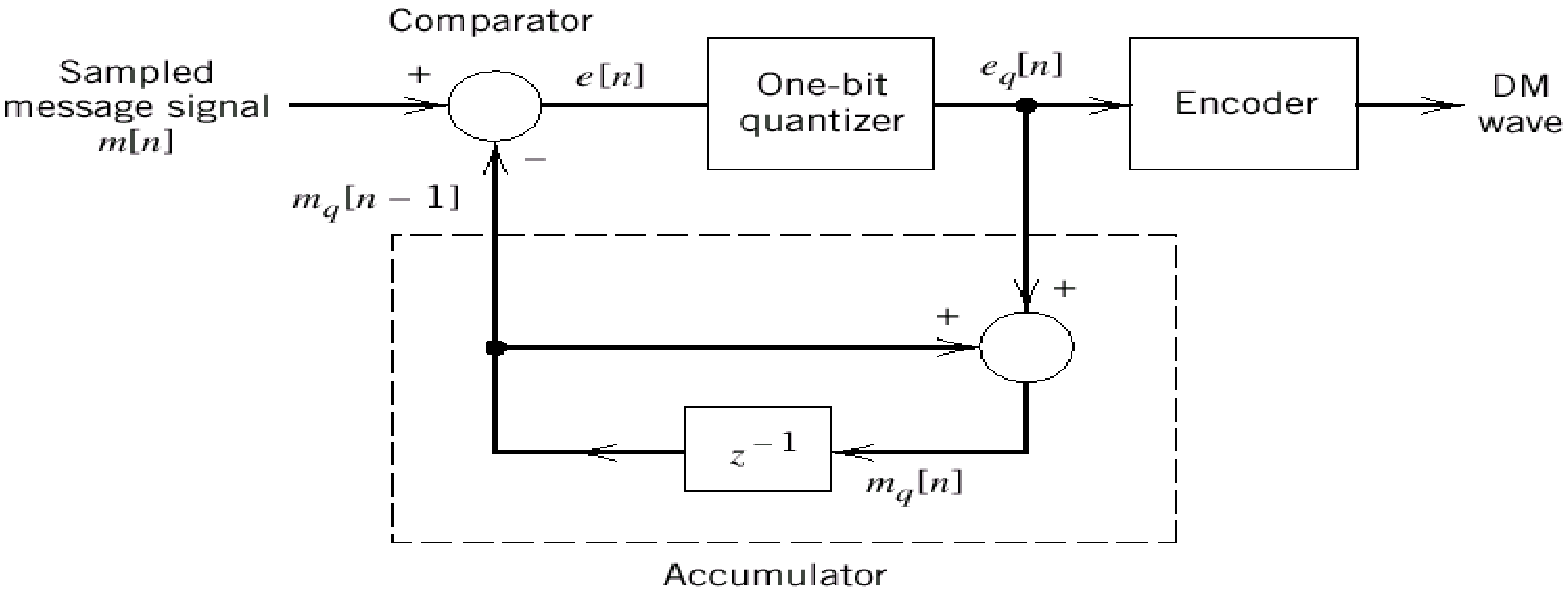
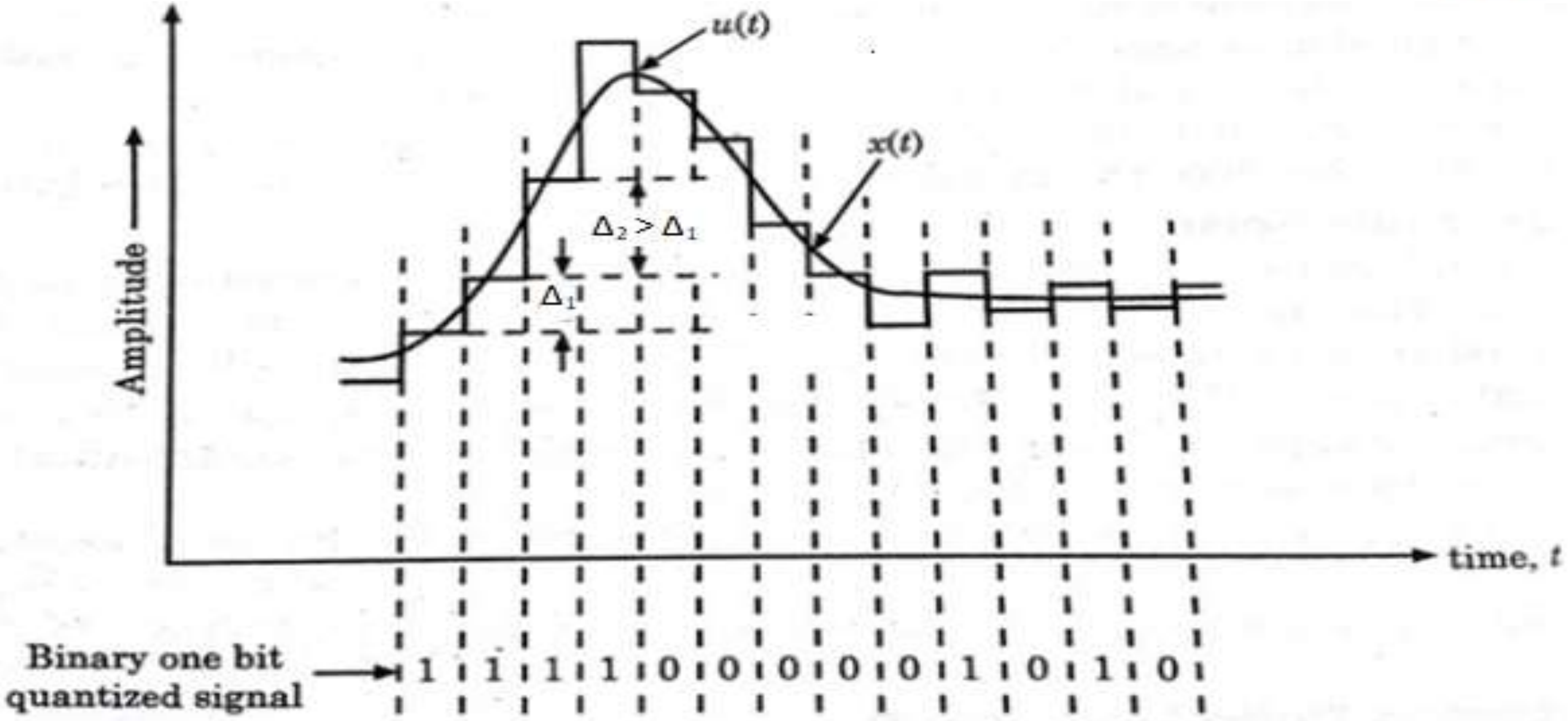


Fig: The Transmitters of a DM System


u(t)
x(t)
Amplitude
$\Delta_2 > \Delta_1$
Δ_1
time, t
Binary one bit quantized signal
1 1 1 1 0 0 0 0 0 1 0 1 0

Delta Demodulation

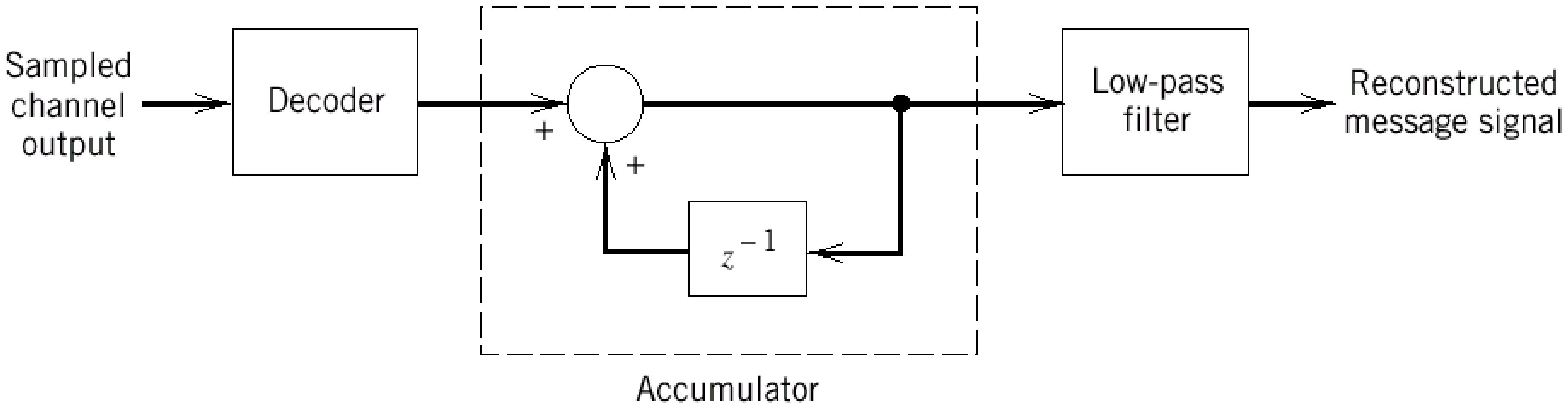


Fig:The Receiver of a DM System

Noise in Delta Modulation

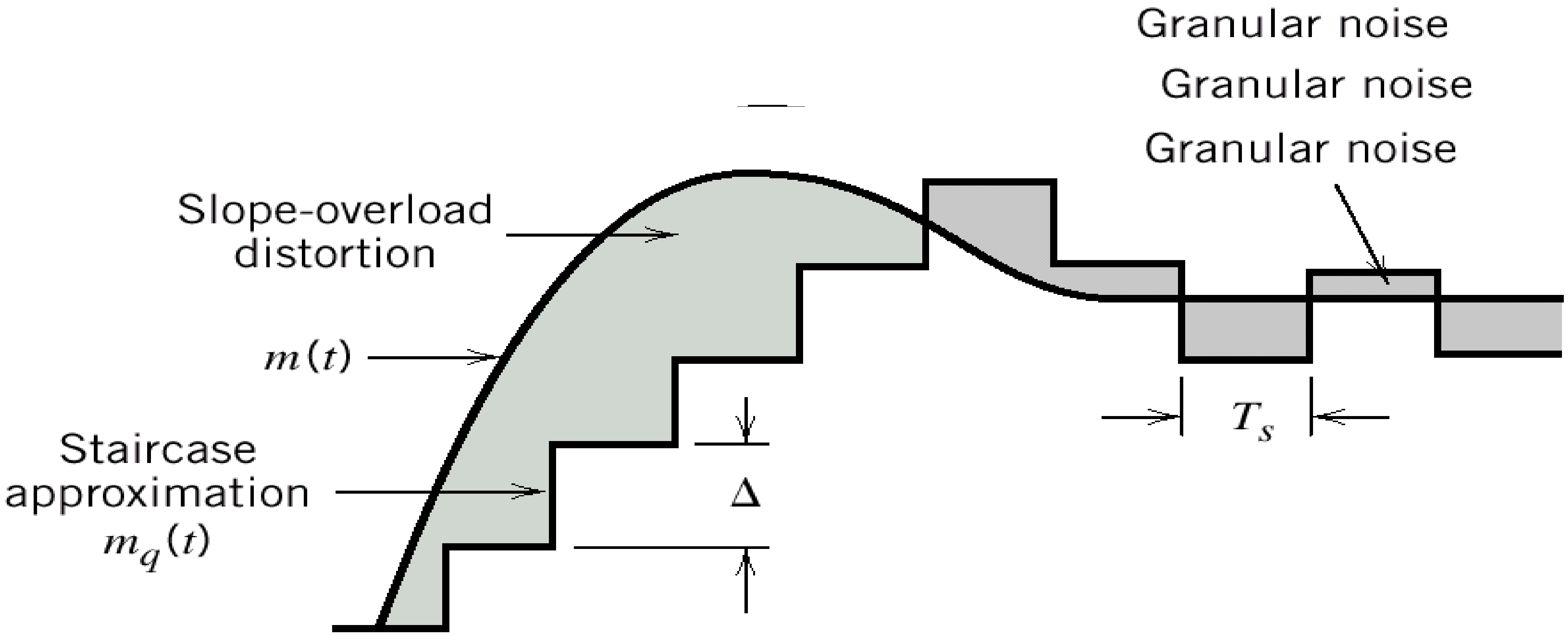
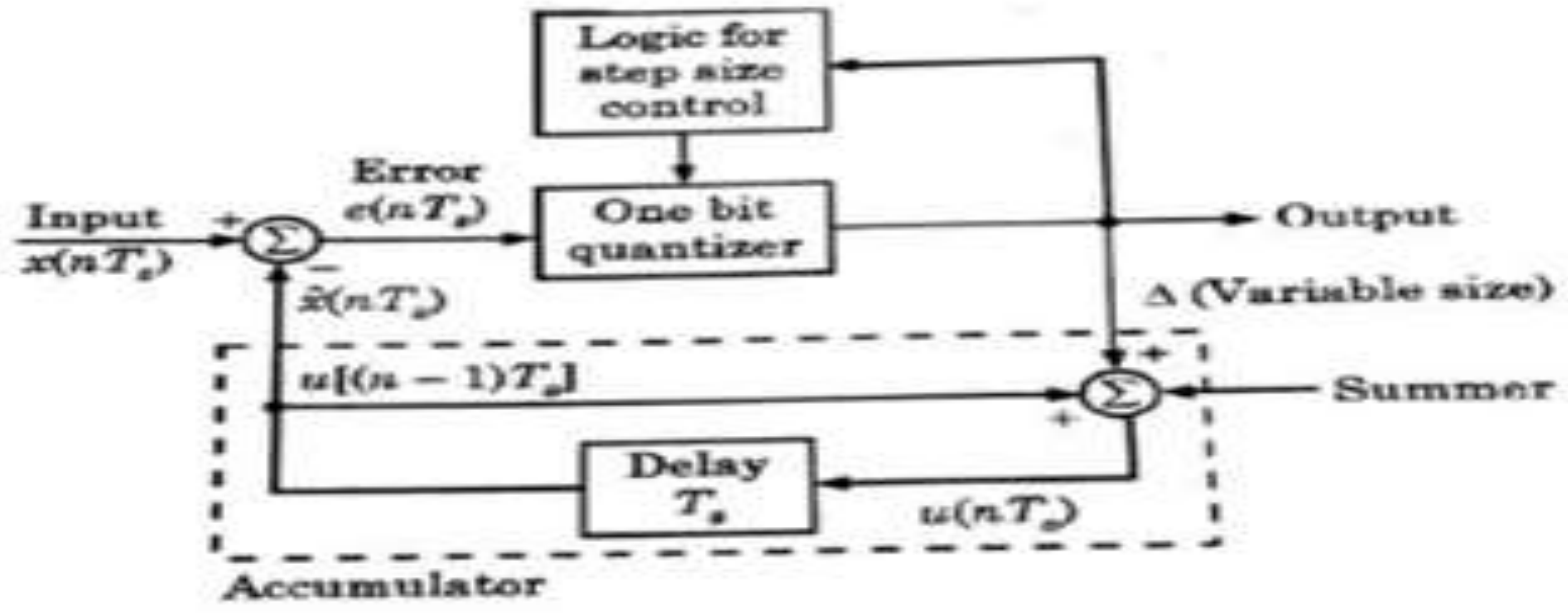
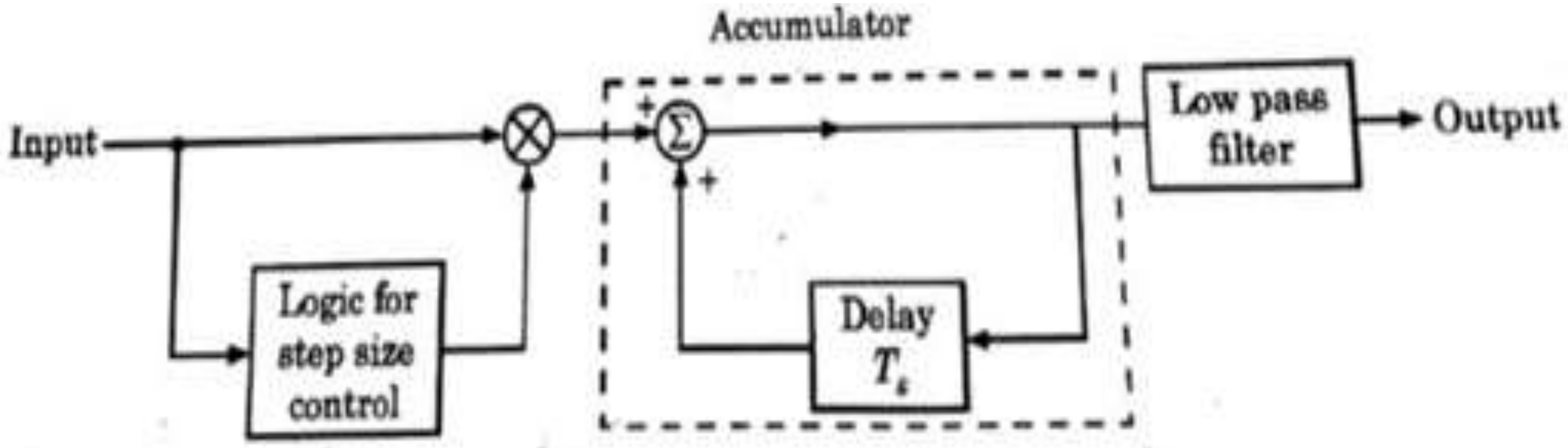


Fig: Noise in Delta modulation

Adaptive Delta Modulation



Adaptive Delta Demodulation



Advantages of Adaptive Delta Modulation

Adaptive delta modulation has certain advantages over delta modulation as under :

1.The signal to noise ratio of ADM is better than that of DM because of the reduction in slope overload distortion and idle noise.

2.Utilization of bandwidth is better in ADM than DM.