PRINCIPLES OF COMMUNICATION

BRANCH : IT

SUBJECT CODE : 4IT3-04

RAJASTHAN TECHNICAL UNIVERSITY, KOTA

SYLLABUS

II Year- IV Semester: B.Tech. (Information Technology)

4IT3-04: Principles of Communication

Credit: 3
3L+0T+0P

Max. Marks: 150(IA:30, ETE:120)
End Term Exam: 3 Hours

SN	Contents	Hours
1	**Introduction:** Objective, scope and outcome of the course.	1
2	ANALOG MODULATION: Concept of frequency translation. Amplitude Modulation: Description of full AM, DSBSC, SSB and VSB in time and frequency domains, methods of generation & demodulation, frequency division multiplexing (FDM). Angle Modulation: Phase and frequency modulation. Descriptions of FM signal in time and frequency domains, methods of generation & demodulation, pre- emphasis & de-emphasis, PLL.	7
3	PULSE ANALOG MODULATION: Ideal sampling, Sampling theorem, aliasing, interpolation, natural and flat top sampling in time and frequency domains. Introduction to PAM, PWM, PPM modulation schemes. Time division multiplexing (TDM)	8
4	PCM & DELTA MODULATION SYSTEMS: Uniform and Non-uniform quantization. PCM and delta modulation, Signal to quantization noise ratio in PCM and delta modulation. DPCM, ADM, T1 Carrier System, Matched filter detection. Error probability in PCM system.	8
5	DIGITAL MODULATION: Baseband transmission: Line coding (RZ, NRZ), inter symbol interference (ISI), pulse shaping, Nyquist criterion for distortion free base band transmission, raised cosine spectrum. Pass band transmission: Geometric interpretation of signals, orthogonalization. ASK PSK, FSK, QPSK and MSK modulation techniques, coherent detection and calculation of error probabilities.	8
6	SPREAD-SPECTRUM MODULATION: Introduction, Pseudo-Noise sequences, direct sequence spread spectrum (DSSS) with coherent BPSK, processing gain, probability of error, frequency-hop spread spectrum (FHSS). Application of spread spectrum: CDMA.	8
	Total	**40**

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UNIT 1 : ANALOG MODULATION

Review of Signals and Systems :→

A Signal may be a function of time, temperature, position, pressure, distance etc. Some signals in our daily life are music, speech, picture and video signals.

Defination :- "A function of one or more independent variables which contains some information is called a signal."

In electrical sense, the signal can be voltage or current. The voltage or current is the function of time as an independent variable.

In daily life, we come across several electric signals such as Radio signal, TV signal, computer signal etc.

Classification of Signals :→

Signals may be classified as under-

i) Continuous-time and Discrete time signals.

ii) Real and complex signals

iii) Deterministic and Random signals

iv) Periodic and Non-periodic signals

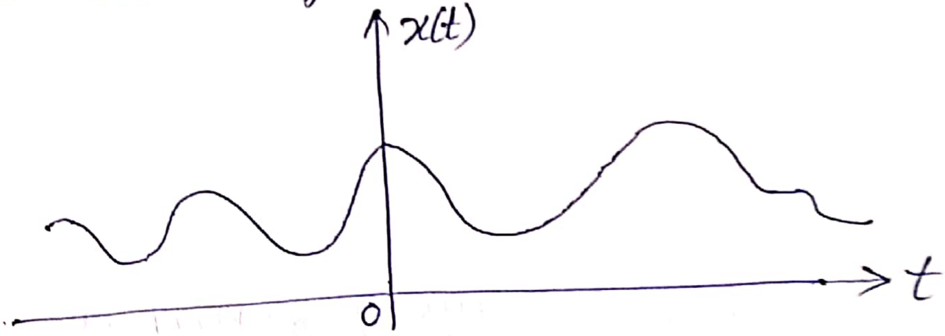
v) Even and odd signals

vi) Energy and power signals

vii) Analog and Digital signals

1.) Continuous-time and Discrete-time signals— ②

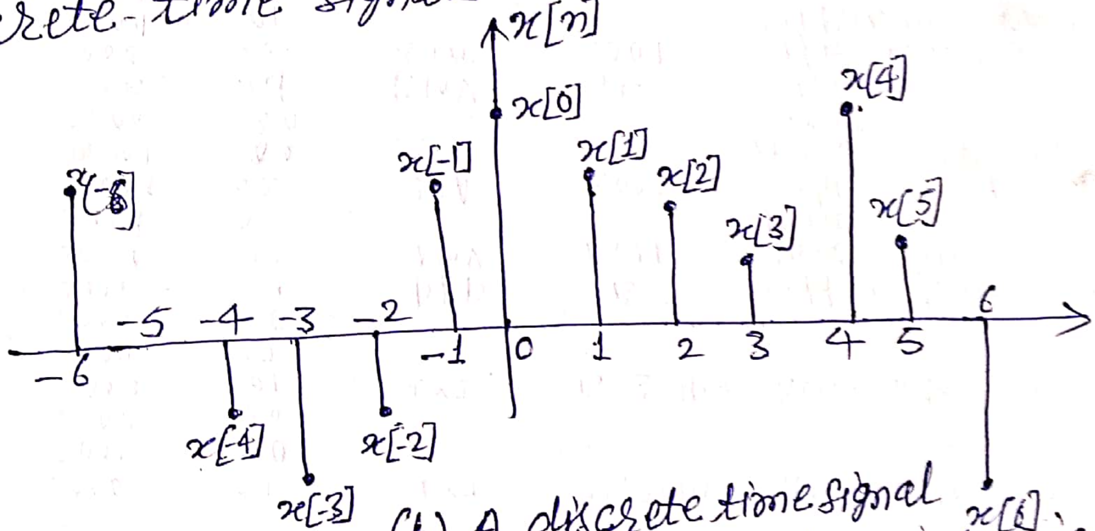
Continuous time signal—



(a) A Continuous-time signal

A signal $x(t)$ is a continuous-time signal if t is a continuous variable. This means that a continuous time signal is defined continuously in the time domain.

Discrete-time signals—



(b) A discrete time signal

On the other hand, if time t is a discrete variable, i.e. $x(t)$ is defined at discrete times, then $x(t)$ is a discrete time signal. Since a discrete time signal is defined at discrete times, it is often identified as a sequence of numbers and is denoted by $x[n]$, where n = integer

2. Real and Complex signals— A signal $x(t)$ is a real signal if its value is a real number. Similarly, a signal $x(t)$ is a complex signal if its value is a complex number.

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3.) Deterministic and Random signals—

Deterministic signals are those signals which can be completely specified in time

for examples — $x(t) = bt$

This is a ramp signal whose amplitude increases linearly with time and the slope is b.

$$x(t) = a \sin \omega t$$

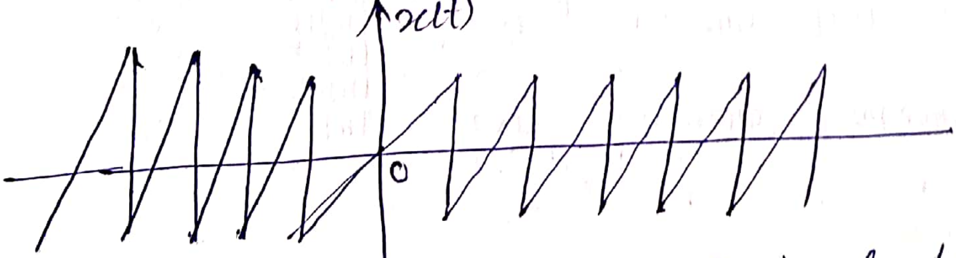
For this signal, the amplitude varies sinusoidally with time and its maximum amplitude is A

On other words a random signal is one whose occurrence is always random in nature. The pattern of such a signal is quite irregular. Random signals are also called non deterministic signals.

Example — Thermal noise, generated in an electric circuit.

4.) Periodic and Aperiodic signals :→

A periodic signal —



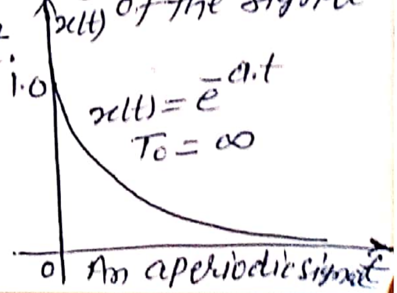
A periodic signal is that type of signal which has a repeats over and over with a repetition period of T.

In other words, a signal is called periodic if it exhibits periodicity as follow:-

$$x(t+T) = x(t), \quad -\infty < t < \infty,$$ T is the period of the signal

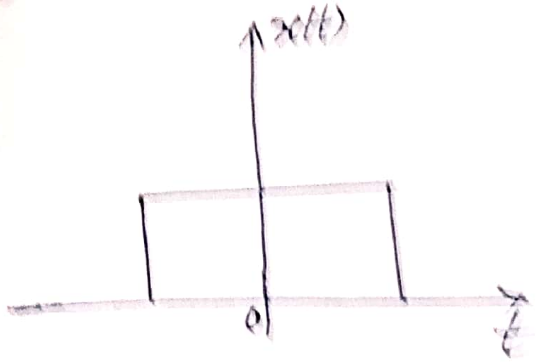
Aperiodic signal — If it does not repeat.

Sometimes aperiodic signals are said to have a period equal to infinity



An aperiodic signal

5) Even and odd Signals :-



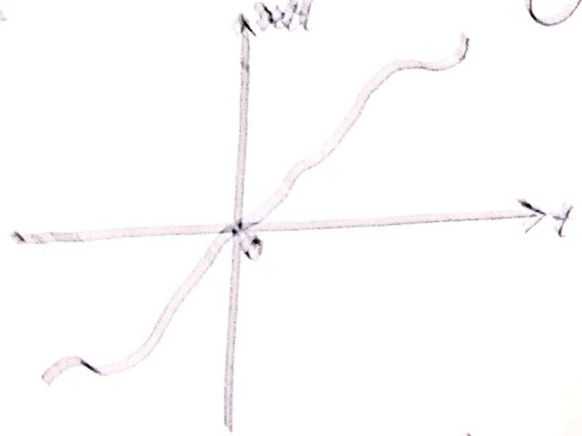


Fig.(a) :- Even Signal

Fig.(b) :- odd Signal

Even Signal - It is that type of signal which exhibits symmetry in the time domain. Mathematically, an even signal must satisfy the following condition-

$$x(t) = x(-t)$$

Odd Signal :- Similarly, an odd signal is that type of signal which exhibits anti symmetry. This type of signal is not identical about the origin. Actually, The signal is identical to its negative.

6.) Energy and Power Signals :-

The energy signal is one which has finite energy and zero average power.

Hence, $x(t)$ is an energy signal, if: $0 < E < \infty \ \& \ P = 0$ (Energy, power)

Power Signal - The power signal is one which has finite average power and infinite energy.

Hence, $x(t)$ is a power signal, if: $0 < P < \infty \ \& \ E = \infty$

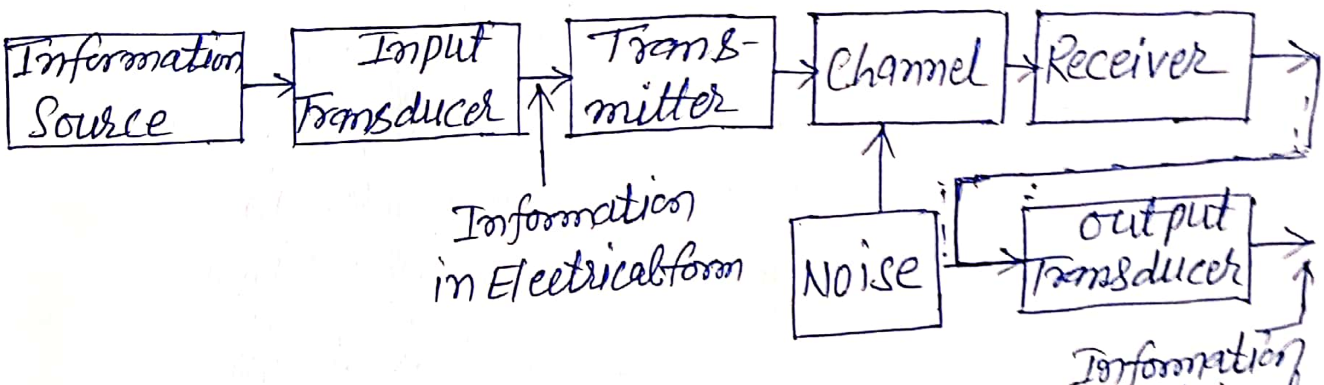
Communication :→ Communication is the process of establishing connection or link between two points for information exchange.

OR

Communication is simply the process of conveying message at a distance or communication is the basic process of exchanging information.

Examples of communication system are Line Telephony and line telegraphy, radio telephony and line-telegraphy, radio broadcasting, point-to-point communication and mobile communication, computer communication, radar communication, Television broadcasting, radio telemetry, radio aids to navigation, radio aids to aircraft landing etc

BLOCK diagram of a communication system OR Analog Comm. sys.



In Analog Communication, the analog message in the original form or signal (Audio signal OR Information signal) modulates some High carrier frequency (OR carrier signal) inside the Tx-mitter to produce modulated signal (i.e, base band signal). This modulated signal is then transmitted with the help of a transmitting antenna to travel through the Transmission channel

At the receiver, this modulated signal is received and processed to recover the original message signal.

Examples- All the AM, FM radio transmission and T.V. transmission. Base band Signal :→

* The message signal generated from the information source is known as base band signal. This baseband signal may be a combination of two or more message signal. The base band signal may be both Analog and digital.

Analog base band signal varies continuously with time and has continuous amplitude. The digital base band signal is discrete in both time and amplitude.

Classification of communication :→

They are two classifications as

i) Line Communication ii) Wireless or Radio Commu.

↳ In line communication, the medium of transmission is a pair of conductors called transmission line. This is also called as line channel. This means that in line communication, the transmitter and the Receiver are connected through a wire or line.

ii) Wireless or Radio Communication — In wireless, a message is transmitted through open space by Electromagnetic waves called as radio waves. Radio waves are radiated from the transmitter in open space through a device called Antenna. A receiving antenna intercepts the radio waves at the receiver.

Examples- All the radio, T.V. and satellite broadcasting

MODULATION :→ It may be defined as the process by which some characteristic OR parameter of a signal called carrier is varied in accordance with the instantaneous value of another signal called modulating signal. The carrier frequency is greater than the modulating frequency. The signal resulting from the process of modulation is called modulated signal.

Types of Modulation :→ It is basically of two types-

i> Continuous wave modulation — When the carrier wave is continuous in nature, the modulation process is known as continuous wave modulation or Analog modulation.

Examples — i> Amplitude Modulation (AM), ii>
ii> Frequency Modulation (FM)] Angle Modu.
iii> Phase Modulation (PM)

AM :- When the amplitude of the carrier is varied in accordance with the message signal, it is known as AM.

FM :- FM in which the instantaneous frequency of the carrier, respectively, are varied in accordance with the message signal

PM :→ PM in which the instantaneous phase of the carrier, respectively, are varied in accordance with the message signal.

2> Pulse Modulation OR Analog Pulse Mod. :→ They are three types as—
i> PAM (Pulse ~~Mod~~ Amplitude Modulation)
ii> PWM OR PDM (Pulse Width OR duration Modulation)
iii> PPM (Pulse Position Modulation)

Advantage of ANALOG Communication :→

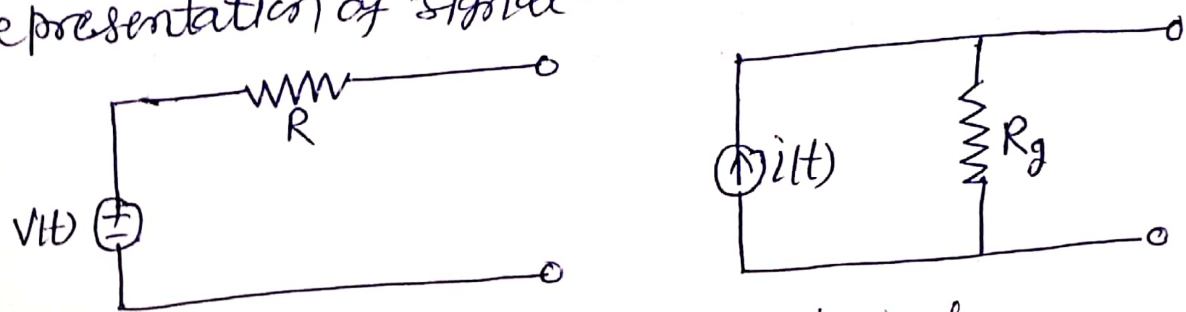
1.) Transmitted modulated signal is analog in nature.
2.) Amplitude, frequency or phase variations in the transmitted signal represent the information or message.
3.) Noise immunity is poor for AM, but improved for FM and PM.
4.) It is not possible to separate out noise and signal. Therefore, repeaters cannot be used.
5.) Coding is not possible
6.) Bandwidth required is lower than that for the digital modulation methods
7.) Frequency Division Multiplexing (FDM) is used for multiplexing.
8.) Not suitable for transmission of secret information in military applications.
9.) Analog modulation systems are AM, FM, PM, PAM, PWM and PPM etc.

REPRESENTATION OF SIGNALS :→

Now, an electrical signal, either a voltage signal or a current signal, may further be represented in two forms:-

i) Time domain representation
ii) Frequency domain representation

Representation of signal—



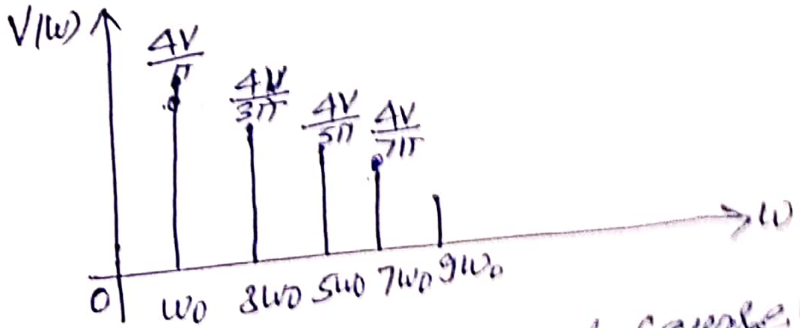
(a) Thevenin's form (b) Norton's form

i) TD Representation – In time domain representation, a signal is a time varying quantity

ii) Frequency domain representation – In frequency domain a signal is represented by its frequency spectrum.

To obtain frequency spectrum of a signal, Fourier series and Fourier transform are used.

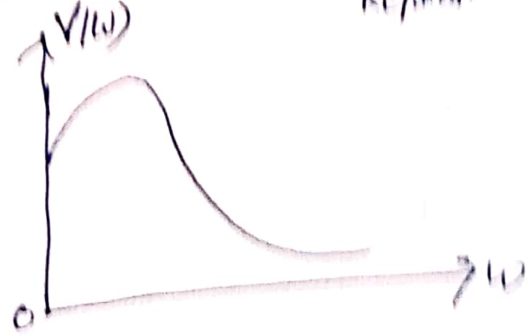
Fig. :- An arbitary time-domain signal



Frequency spectrum of square wave signal (freq^n domain Representation)

Fourier Transform–

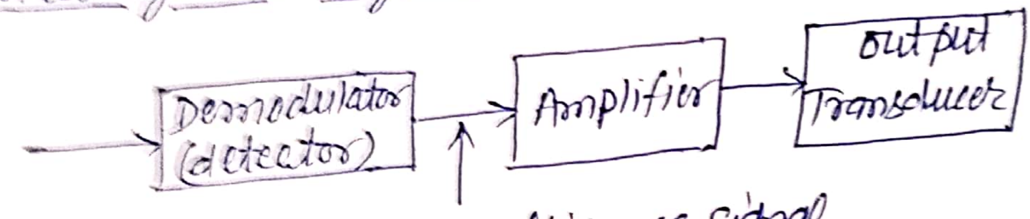
Time domain



Freq^n domain

Amplitude Modulation (AM):-

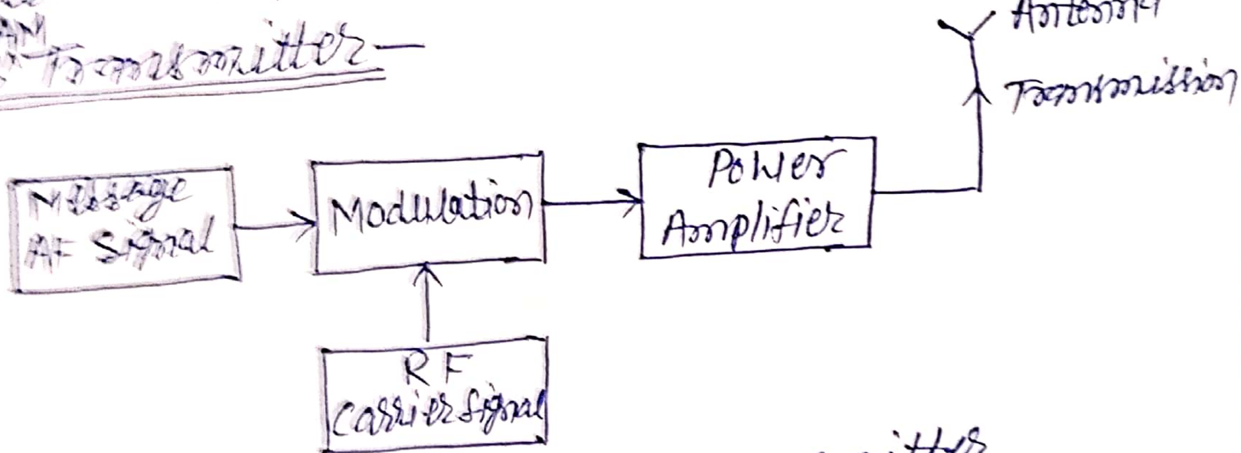
Modulation is the process of putting information onto a high frequency carrier for transmission. The transmission takes place at the high frequency (the carrier) which has been modified to "carry" the lower frequency information. The low frequency information is often called the intelligence signal.



Modulating signal or baseband signal also known as

Once this information is received the intelligence signal must be removed from the high freq. carrier, a process known as Demodulation.

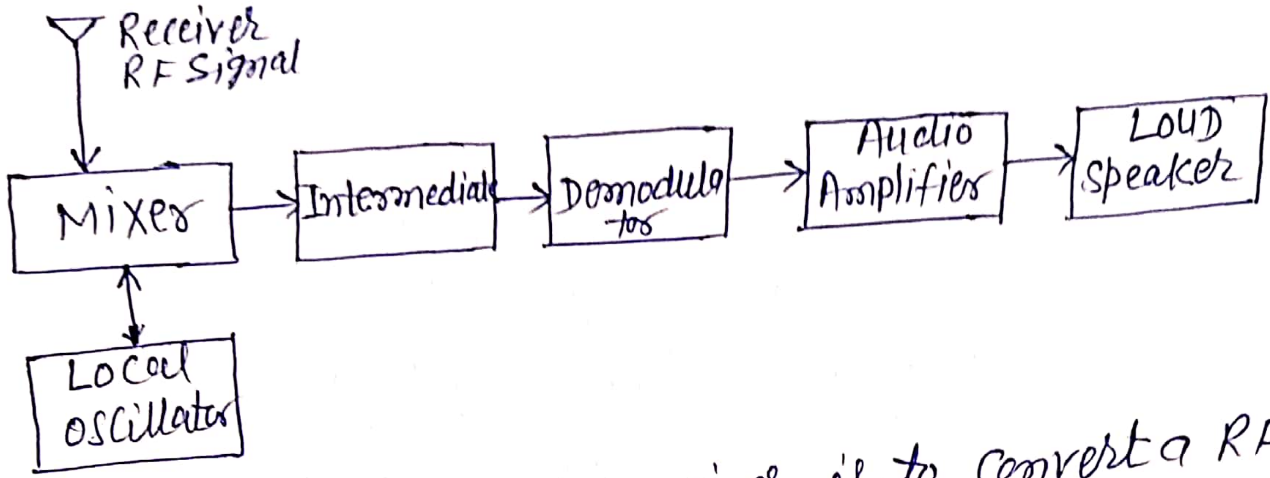
The AM Transmitter-



BLOCK diagram of AM Transmitter

The Channel-

The AM Receiver :→



The first stage of receiver is to convert a RF signal into a comparatively low frequency called "Intermediate Frequency (IF)". This conversion is done with of a Local oscillator and mixer.
In the mixer, a locally generated RF signal is mixed with received RF signal

Modulation basically is defined in two ways:-

1) Linear Mod.

2) Non linear Mod.

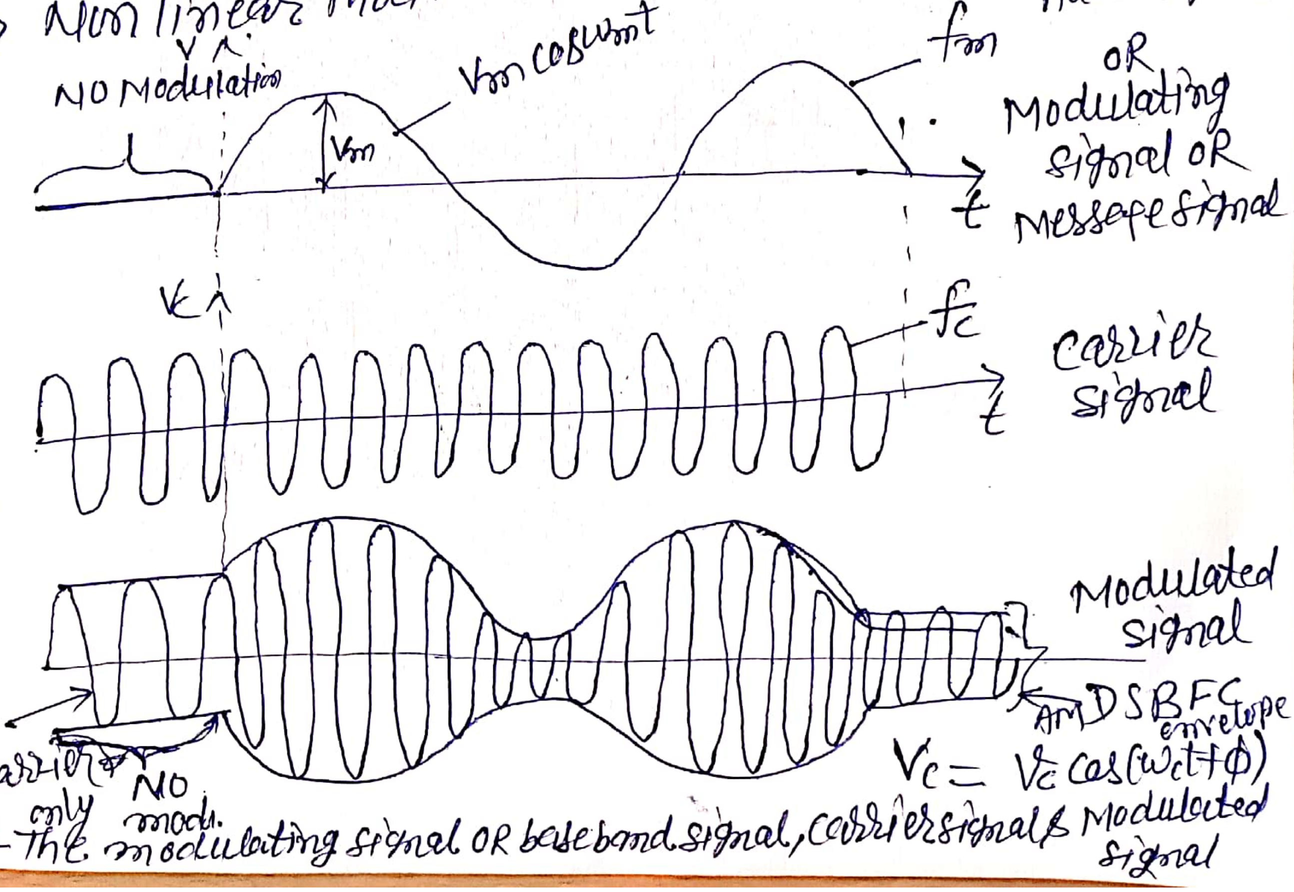


Fig :- The modulating signal OR baseband signal, carrier signal & Modulated signal

Modulation Index :→

Let us consider a sinusoidal carrier wave $c(t)$ given as $c(t) = A\cos(w_c t + \phi)$ ($\because \phi = 0$)

$$c(t) = A\cos w_c t \quad \text{---- (1)}$$

Here A is the maximum amplitude of the carrier wave and w_c is the carrier frequency. We have assumed that the phase of the carrier wave is zero in eqn (1)

Let $x(t)$ denotes the modulating or baseband signal, then according to amplitude modulation, the maximum amplitude A of the carrier will have to be made proportional to the instantaneous amplitude of modulating signal, $x(t)$. Modulating signal OR single tone amplitude

The standard equation for a amplitude modulated (AM) wave may be expressed as

$$s(t) = \underbrace{x(t)\cos w_c t}_{DSB\cdot SC} + A\cos w_c t \quad \text{---- (2)}$$

where $x(t)$ is single tone modulating signal $\quad \because x(t) = V_m \cos w_m t$ --- 2(i)

$$\text{OR} \quad s(t) = [A + x(t)]\cos w_c t \quad \text{---- (3)}$$

$$\text{OR} \quad s(t) = E(t)\cos w_c t \qquad \text{where } E(t) = A + x(t)$$

$E(t)$ is called the envelope of AM wave. This envelope consists of the baseband signal $x(t)$.

From eqn (3),

$$s(t) = A\cos w_c t + x(t)\cos w_c t$$

putting the value of $x(t)$, we get (from eqn 2(i))

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

$$\text{OR} \quad s(t) = A\cos w_c t\left[1 + \frac{V_m}{A}\cos w_m t\right] \quad \text{---- (4)}$$

$$m = \frac{|x(t)|_{max}}{A}, \quad \text{where } m_a \text{ is modulation index}$$

Where $|x(t)|_{max}$ denotes the maximum amplitude of modulating signal and A is the max. amplitude of the carrier signal.

In this case, we have

$$|x(t)|_{max} = V_m$$

Therefore, $\boxed{m_a = \frac{V_m}{A}}$ OR $m_a = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100\%$

Putting this value of m_a in Eq.ⁿ (4), we get

$$\boxed{s(t) = A \cos w_c t [1 + m_a \cos w_m t]} \quad \text{-----} (5)$$

This is the desired Expression for single tone modulated signal

To observe the frequency components in AM signal.

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

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

OR $s(t) = A \cos w_c t + \frac{A.m_a}{2} [2 \cos w_c t \cos w_m t]$

$s(t) = A \cos w_c t + \frac{A.m_a}{2} [\cos (w_c + w_m)t \cos (w_c - w_m)t]$

OR $s(t) = A \cos w_c t + \frac{A.m_a}{2} \cos (w_c + w_m)t + \frac{A.m_a}{2} \cos (w_c - w_m)t$ ----- (6)

Eq.ⁿ (6) reveals that the AM signal has three frequency components as follow—

i) Carrier frequency w_c having amplitude A.

ii) Upper sideband $(w_c + w_m)$ having amplitude $\frac{m_a.A}{2}$

iii) Lower sideband $(w_c - w_m)$ having amplitude $\frac{m_a.A}{2}$

* Infact, the difference between these two extreme frequencies is equal to the Bandwidth of the AM wave

Therefore, Bandwidth $B = (w_c + w_m) - (w_c - w_m)$ OR $\boxed{B = 2w_m}$

Thus, it is clear that the bandwidth of the amplitude modulated wave is twice the highest frequency present in the base band or modulating signal.

We can plot the frequency spectrum of single tone amplitude modulated wave.

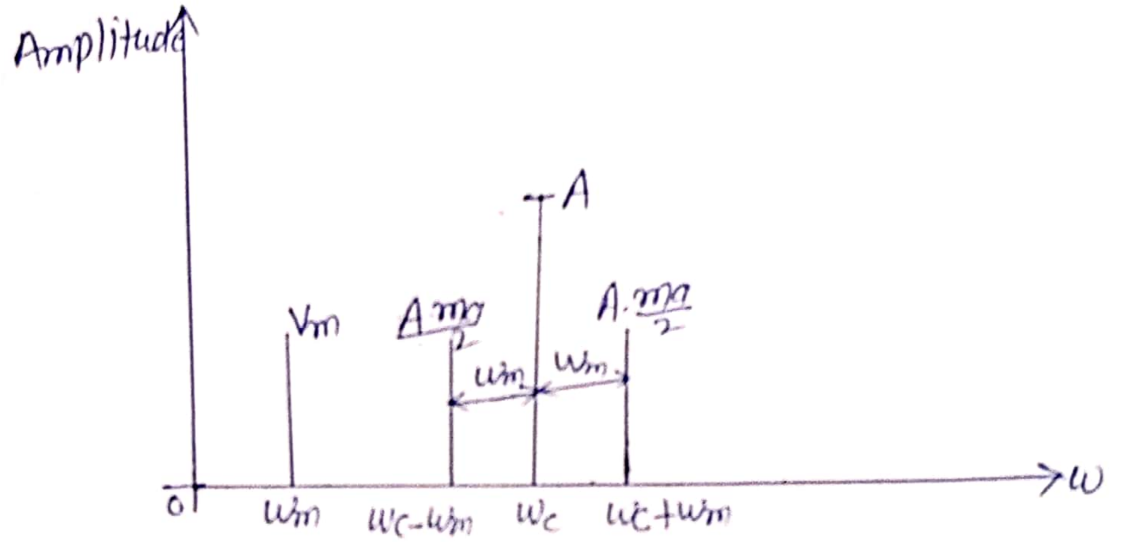


Fig.:- Single sided frequency spectrum of single tone AM wave.

Types of AM :→ It is also called as the Double side band full carrier system (DSB-FC)

- i) DSB-FC
- DSB-SC — Double sideband suppressed carrier system
- SSB-SC — Single sideband suppressed carrier system
- ISB — Independent sideband system
- VSB — Vestigial sideband system

DSB-FC :→ The disadvantages of DSB-FC AM are as-

i) Power wastage takes place in DSB-FC transmission

ii) DSB-FC system is bandwidth inefficient system. (because the BW of DSB-FC system is $2f_m$.)

DSBSC System :- We know that the modulated signal, which contains no carrier but two side bands is called Double sideband suppressed carrier (DSB-SC) modulation.

This means that the term $x(t) \cos w_c t$ represents a DSB-SC signal. Therefore, a DSB-SC signal is obtained by simply multiplying modulating signal $x(t)$ with carrier signal $\cos w_c t$. This is achieved by a product modulator.

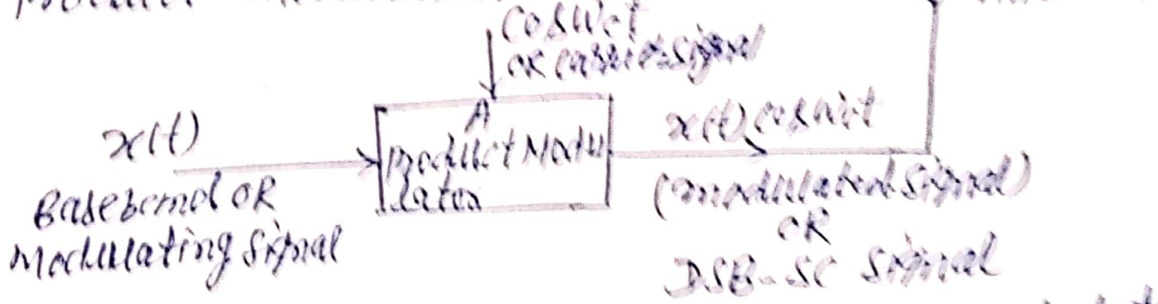


Fig. :- The Block diagram of a product modulator

Generation of DSB-SC Signal :-

Generation of DSB-SC signal is called a Product modulator. The product modulator is two types name as i) The Balanced Modulator (2) The Ring Modulator

i) Modulation using Non linear Devices (Balanced Modulator) -

A Semiconductor diode is a good Example of a non-linear devices.

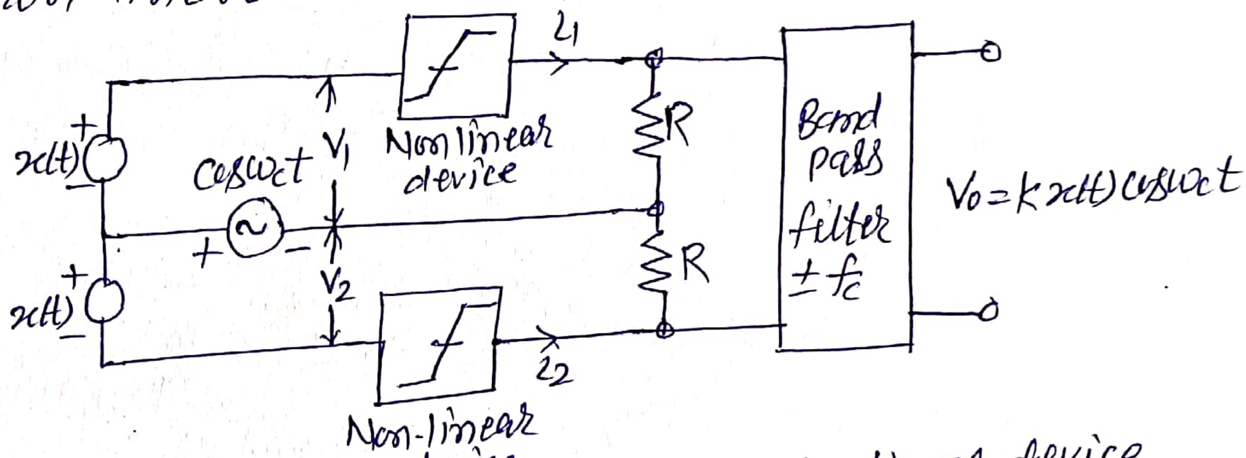
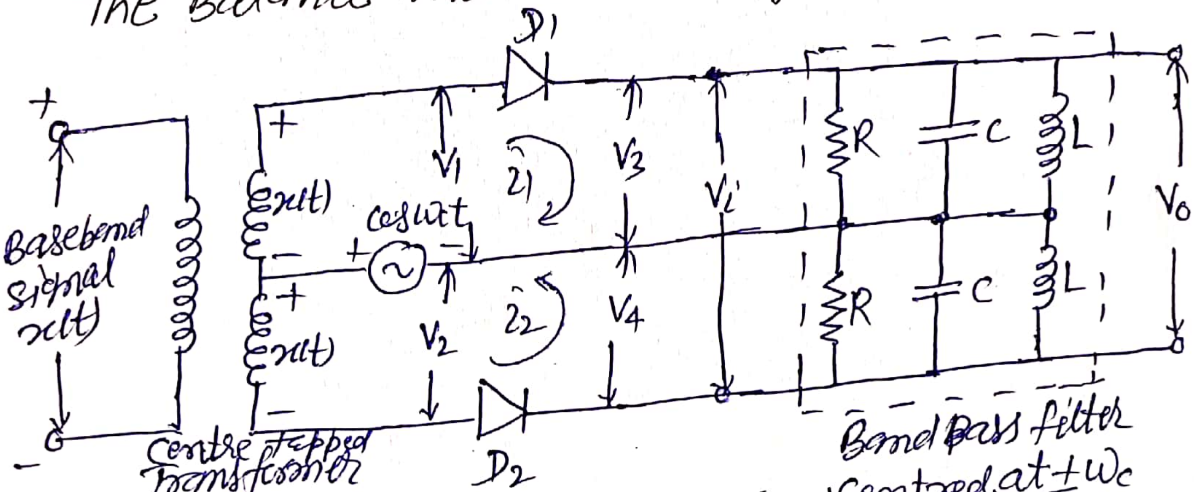


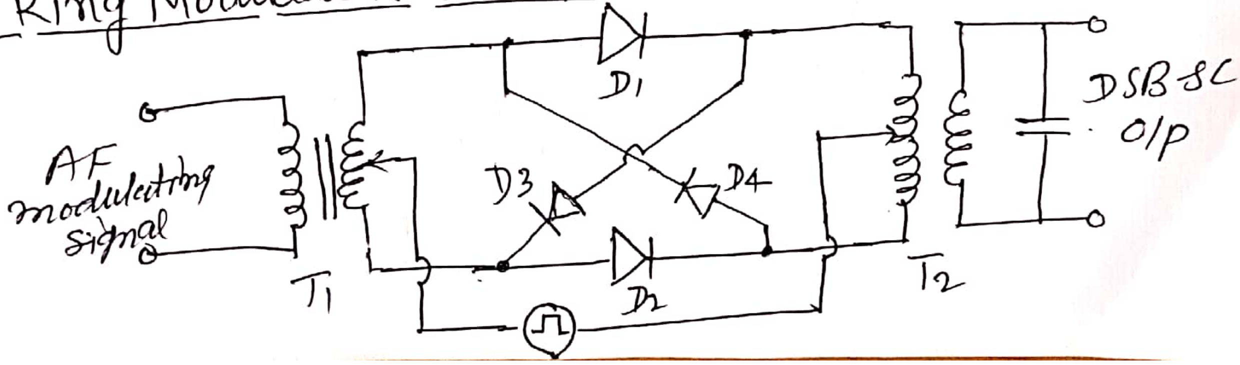
Fig.:- DSB-SC Modulator using Non linear device

Such an arrangement is called as Balanced Modulator

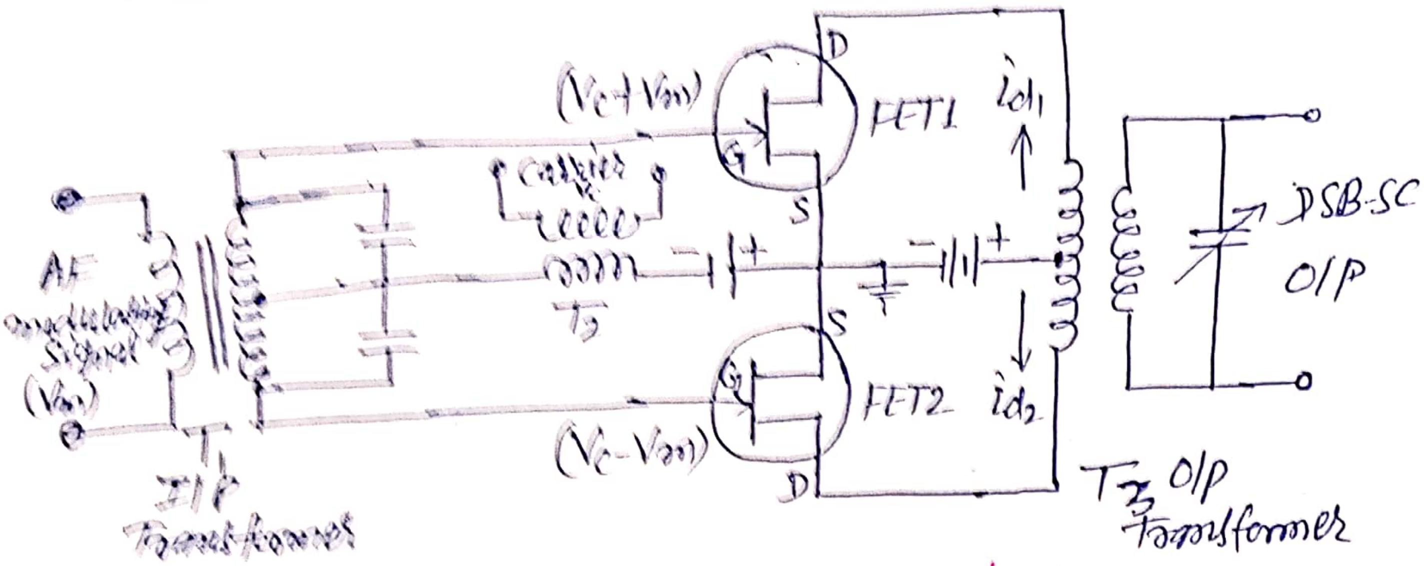
The Balance modulator using Diodes -



The Ring Modulator OR Chopper Mod.



Balanced Modulator using FETs :→



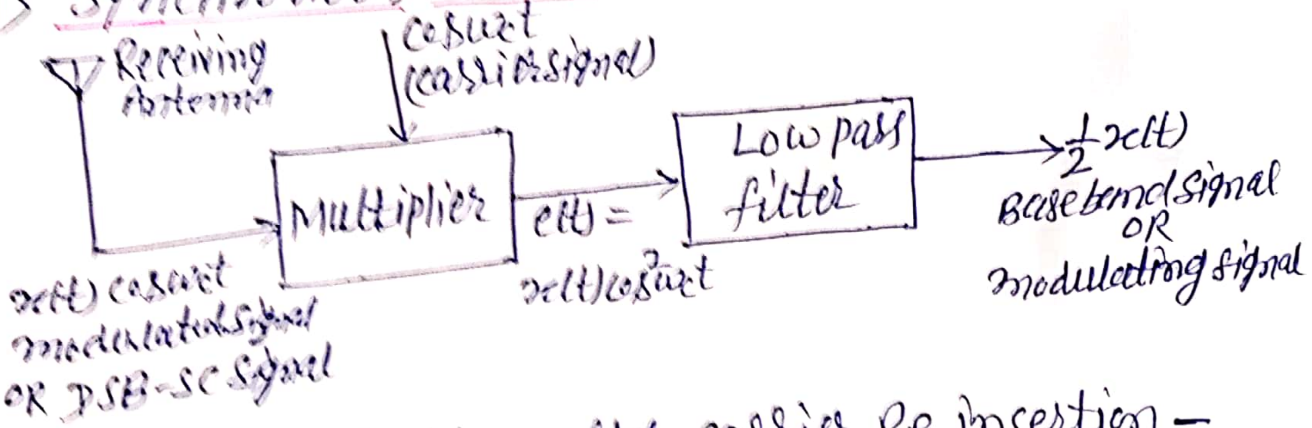
Demodulation of DSB-SC Signals :→

The DSB-SC signal may be demodulated by following two methods-

i) Synchronous detection method

ii) Using envelope detector after carrier Re-insertion.

i) Synchronous detection method-



ii) Envelope Detection after carrier Re-insertion -

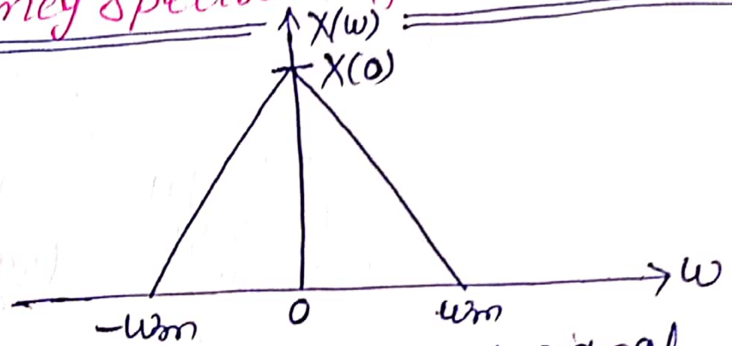
The other method of demodulation of DSB-SC signal is by inserting a carrier generated at the receiver end with the help of a Local oscillator. We know that if we insert a carrier of same freq^n & phase to DSB SC signal, it converts DSB-SC signal into a conventional AM wave. Now, this AM wave is demodulated by an envelope detector.

Single side band suppressed carrier (SSB-SC) Modulation :->

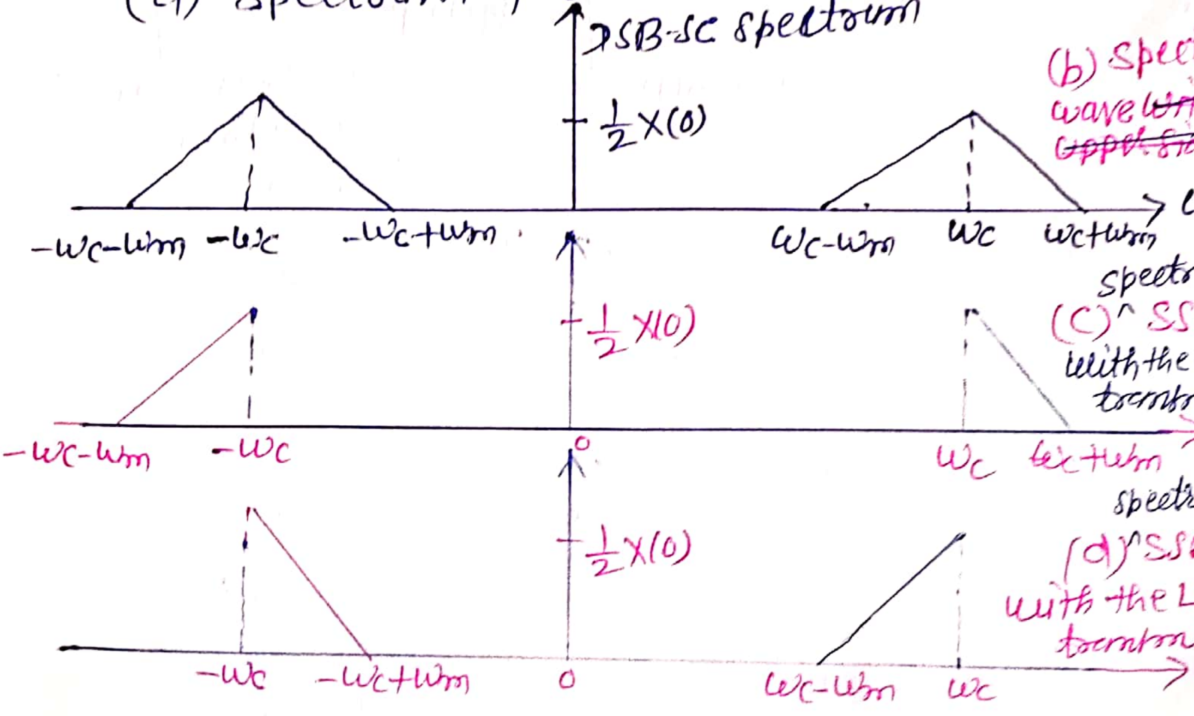
This means that as far as the transmission of information is concerned, only one sideband is necessary. Thus, if the carrier and one of the two sidebands are suppressed at the transmitter, no information is lost. Modulation of this type which provides a single sideband with suppressed carrier is known as SSB-SC system.

Thus, SSB-SC system reduces the transmission bandwidth by half.

Frequency Spectrum of SSB-SC system-



(a) spectrum of base band signal



(b) Spectrum of DSB-SC wave ~~with the upper sideband omitted~~

(c) spectrum of SSB-SC wave with the upper sideband transmitted

(d) spectrum of SSB-SC wave with the lower sideband transmitted.

Figure (b) shows the frequency spectrum of DSB-SC modulation which contains no carrier but two sideband, i.e., Lower sideband and upper sideband.

Figure (c) shows the frequency spectrum of single sideband suppressed carrier modulation consisting of upper side band only, i.e., lower side band and carrier signal are suppressed.

Figure (d) shows the frequency spectrum of single sideband suppressed carrier modulation consisting of lower sideband only, i.e., upper sideband and carrier signal are suppressed.

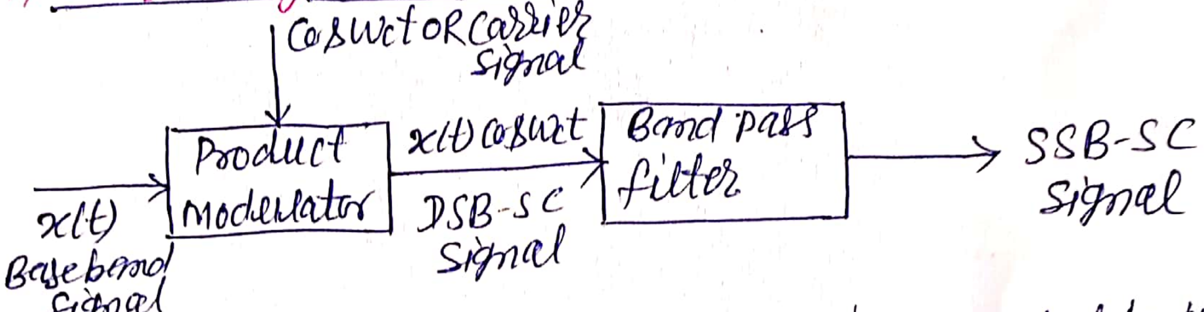
Generation of SSB-SC signal :-

SSB-SC signals may be generated by two methods as -

i> Frequency discrimination method OR Filter method

ii> Phase discrimination method OR Phase-shift method

i> Frequency Discrimination method -



In a filter method, a DSB-SC signal is generated by using an product modulator OR a balanced modulator. After this, from the DSB-SC signal one of the two sidebands is filtered out by a band pass filter.

2) Phase shift method or phase discrimination method -

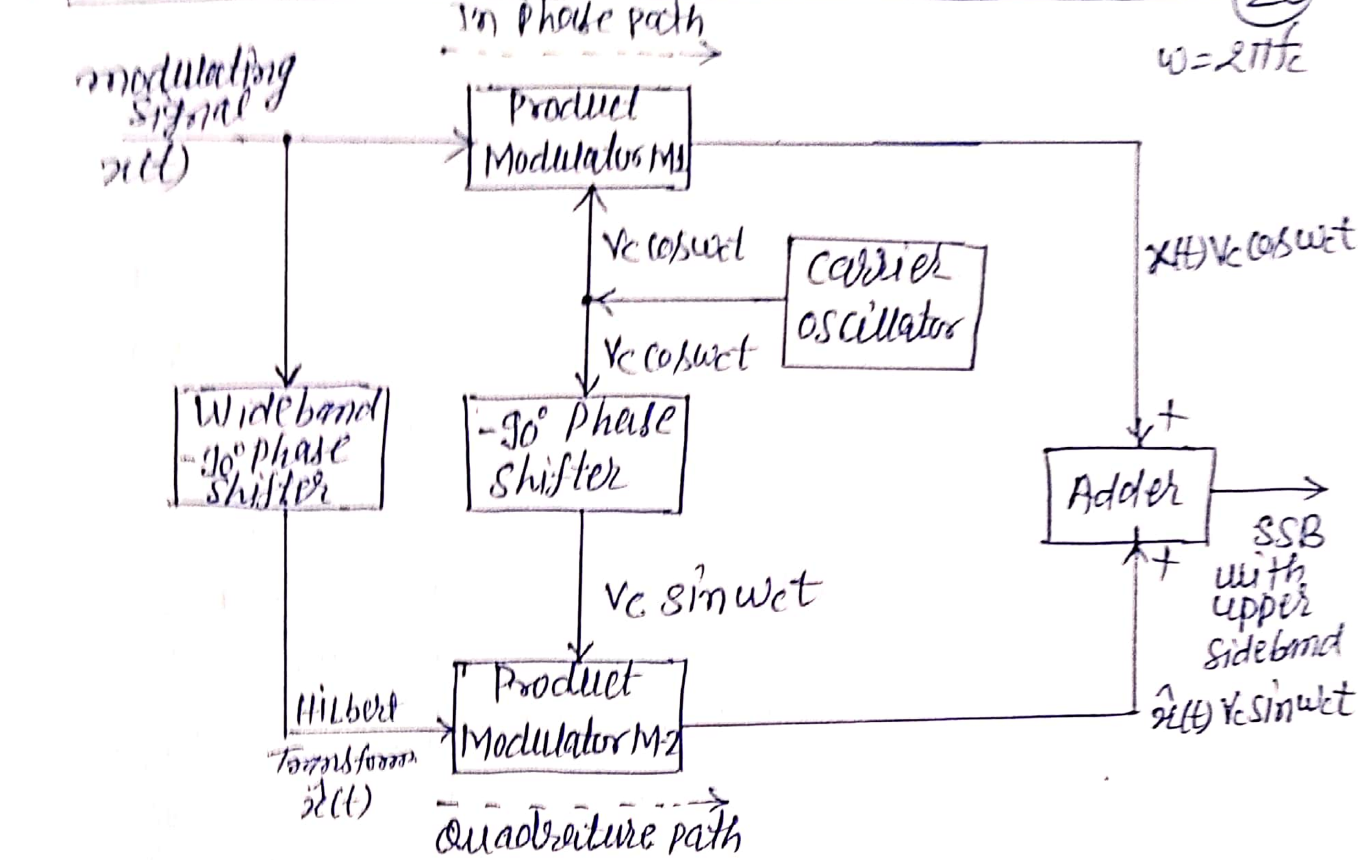


Fig. :- Block diagram of phase Shift method

This system is used for the suppression of lower sideband

* HILBERT TRANSform :- It may be observed that the function $x_1(t)$ obtained by providing $(-\pi/2)$ phase shift to every frequency component present in $x(t)$, actually represents the Hilbert transform of $x(t)$.

The O/Ps of M_1 & M_2 are applied to an adder. Note the (-) sign for the Quadrature path.

Adder output $= V_c[x(t) \cos\omega_c t + \hat{x}(t) \sin\omega_c t]$

This expression represents the SSB signal with only USB i.e. it rejects the LSB.

SSB With LSB :- Same block diagram of above But little changess as $\boxed{\text{90 phase Shifter}}$

Advantages of phase shift method :-

i) It can generate the SSB signal at any frequency, so the frequency up converter stage is not required.

ii) It can use the low audio frequencies as modulating signal (In filter method, this is not possible)

iii) It is easy to switch from one sideband to the other

Disadvantage of phase shift method -

i) It is that the design of the 90° phase shifting n/w for the modulating signal is extremely critical.

ii) This n/w has to provide a correct phase shift of 90° at all the modulating frequencies which is practically difficult to achive.

DEMODULATION OF SSB System :->

Requirements of SSB Receiver -

1) High reliability
2) Excellent suppression of adjacent signals
3) High signal to noise ratio
4) Ability to demodulate SSB

Coherent SSB Demodulation — The Product Mod. is a type of coherent SSB demodulator.

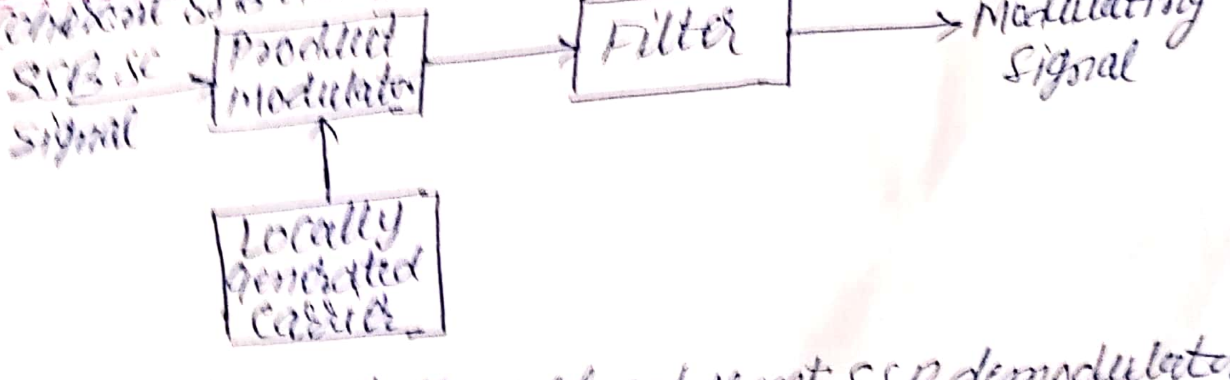
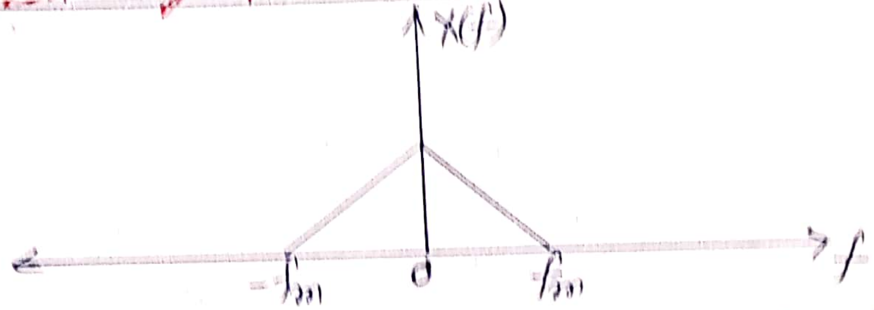


Fig.:- Block diagram of coherent SSB demodulator

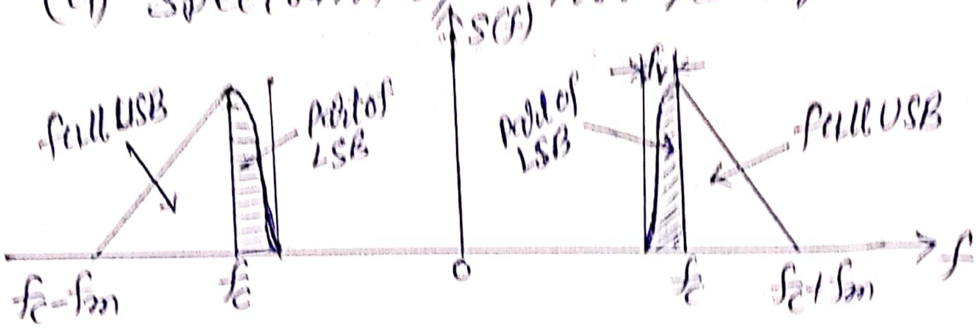
VESTIGIAL SIDE BAND TRANSMISSION (VSB) :-

The stringent frequency response requirements on the sideband filter for SSB SC system can be relaxed by allowing a part of the unwanted sideband (called as vestige) to appear in the o/p of the modulator. Due to this, the design of the sideband filter is simplified to a great extent. But the bandwidth of the system is increased slightly.

Frequency spectrum of VSB :-



(a) Spectrum of message signal



[b] Spectrum of VSB signal

In the frequency spectrum, it is assumed that the upper sideband is transmitted as it is and the lower sideband is modified into vestigial sideband.

Transmission bandwidth :-

$$B = (f_m + f_v)\ Hz$$

where f_m = message bandwidth

f_v = width of the vestigial sideband

Advantage of VSB —

i) The VSB modulation is the Reduction in bandwidth. It is almost as efficient as the SSB.

ii) Due to allowance of transmitting a part of lower sideband, the constraint on the filters have relaxed. So practically, easy to design filters can be used.

iii) It makes the transmission of low freq^n components possible.

Application of VSB :→

It has become standard for the transmission of Television signals. Because the Video signals need a large transmission bandwidth if transmitted using DSB-FC OR DSB-SC techniques.

Generation of VSB Modulation :→

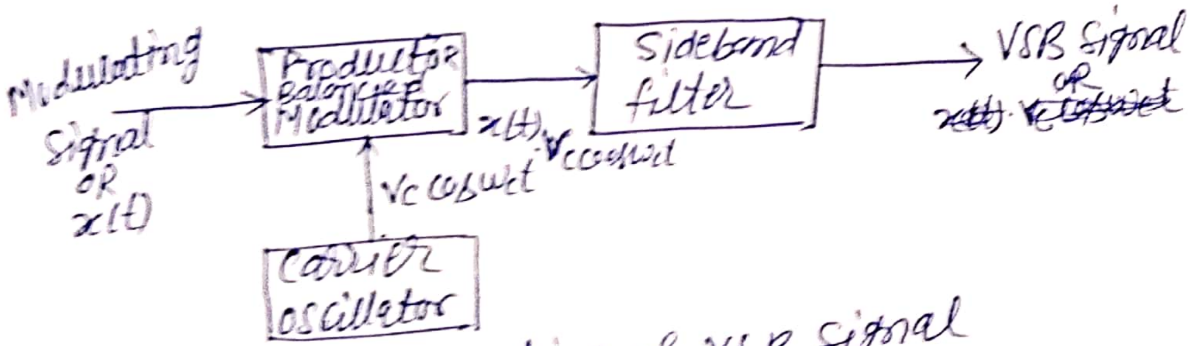


Fig. :- Generation of VSB Signal

The O/P of product modulator is as

$$m(t) = x(t) \cdot c(t) = x(t) \cdot V_c \cos \omega_c t$$

This represents a DSB-SC modulated wave. This DSB-SC signal is then applied to a sideband shaping filter. The design of this filter depends on the desired spectrum of the VSB modulated signal. This filter will pass the wanted sideband as it is and the vestige of the unwanted sideband. The spectrum of the VSB Modulated signals —

$$S(f) = \frac{V_c}{2}\left[X(f-f_c) + X(f+f_c)\right] H(f)$$

$H(f)$ — Transfer function of the filter

Demodulation of VSB system :-

The synchronous detector —

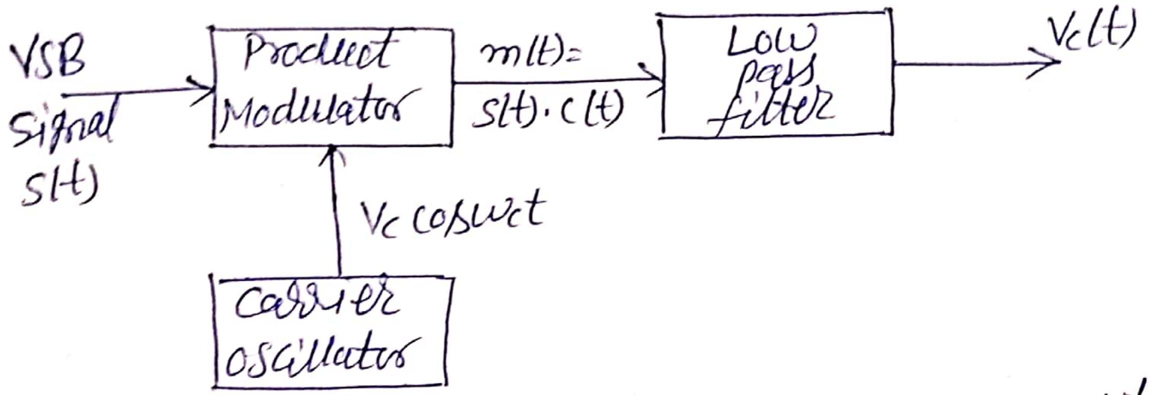


Fig. 1 - Synchronous detector of VSB demodulation

Hence, the o/p of the product modulator as

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

Taking the Fourier transform of both sides, we get

$$M(f) = S(f) * \left[\tfrac{1}{2}\delta(f+f_c) + \tfrac{1}{2}\delta(f-f_c)\right]$$

$$M(f) = \frac{1}{2} S(f+f_c) + \frac{1}{2} S(f-f_c) \quad \text{--- (1)}$$

But $S(f) = \frac{V_c}{2}\left[X(f-f_c) + X(f+f_c)\right] H(f)$

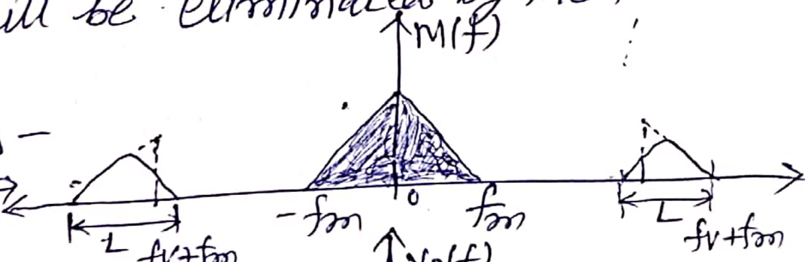
Hence, we have

$$M(f) = \frac{V_c}{4}\left[X(f-2f_c)\,H(f-f_c) + X(f+2f_c)\,H(f+f_c)\right]$$

$$+ \frac{V_c}{4} X(f)\left[H(f-f_c) + H(f+f_c)\right] \quad \text{--- (2)}$$

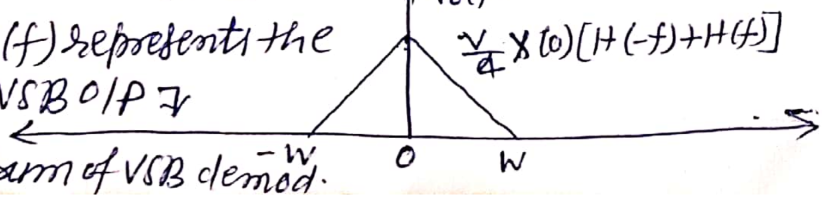
The first term in the above equation represents the carrier-free VSB modulated wave, corresponding to a carrier freq of $2f_c$. This term will be eliminated by the filter to product o/p $V_o(t)$

The spectrum $m(f)$ is as -



(a) Spectrum of product Modulator o/p

The IInd term in Eqn (2) of M(f) represents the spectrum of demodulated VSB O/P :-



(b) Spectrum of VSB demod.

Time domain Description of VSB modulated wave:→

$$S(t) = \frac{V_c}{2}\left[x(t)\cos\omega_c t - x_Q(t)\sin\omega_c t\right] \quad \text{---} (1)$$

This is the VSB modulated wave in time domain.
It represents the VSB wave with full USB & a vestige of LSB

$$S(t) = \frac{V_c}{2}\left[x(t)\cos\omega_c t + x_Q(t)\sin\omega_c t\right] \quad \text{---} (2)$$

The time domain description for the VSB modulated wave with full LSB and vestige of USB.

Comparison of AM Techniques:→

S.NO.	Parameter of Comparison	DSBFC (Standard A.M)	DSB-SC	SSB	VSB
1	carrier suppression	N.A.	Fully	Fully	N.A.
2	Side band suppression	N.A.	N.A.	One S.B. Completely	One S.B. suppressed partially
3	Bandwidth	$2f_m$	$2f_m$	f_m	$f_m < BW < 2f_m$
4	Transmission efficiency	Minimum	Moderate	Maximum	Moderate
5	No. of modulating I/Ps	1	1	1	2
6	Application	Radio Broadcasting	Radio Broadcasting	Point to point mobile Communication	T.V.

ANGLE MODULATION :-

Angle modulation may be defined as the process in which the total phase angle of a carrier wave is varied in accordance with the instantaneous value of the modulating or message signal while keeping the amplitude of the carrier constant.

Mathematical representation -

Let us consider that an unmodulated carrier signal is as (1)

$$c(t) = A\cos(\omega_c t + \theta_0)$$

Where A = Amplitude of carrier

ω_c = carrier frequency

θ_0 = some phase angle

OR $c(t) = A\cos\phi$, where $\phi = \omega_c t + \theta_0$ is the total phase angle of the carrier signal.

Now, if this angle ϕ is varied according to the instantaneous value of the message or modulating signal, the carrier signal is then said to be angle modulated.

Types of angle modulation :-

Frequency Modulation (FM) — Narrowband FM, Wideband FM

Phase modulation (PM)

FM :-> Frequency of the carrier is varied according to the message signal.

PM :-> Phase angle of the carrier is varied according to the message signal.

Advantage of Angle modulation -

1) Noise Reduction
2) Improved system fidelity
3) And more efficient use of power

disadvantage of Angle Modulation -

1) Increased Bandwidth
2) And use of more complex circuits

Application of Angle Modulation -

These are following applications as -

1) Radio broadcasting
2) Two way mobile radio
3) Microwave Communication
4) TV sound transmission
5) cellular radio
6) Satelite communication

Phase Modulation [PM] :- PM is that type of angle modulation in which the phase angle ϕ is varied linearly with a base band or modulating signal $x(t)$ about an unmodulated phase angle $(\omega t + \theta_0)$.

Mathematical Representation -

We know that unmodulated carrier signal is as $c(t) = A\cos(\omega t + \theta_0)$ OR $c(t) = A\cos\phi$

where $\phi = \omega t + \theta_0$, Neglecting θ_0, we get total phase angle of unmodulated carrier is

$$\phi = \omega t$$

Now according to phase modulation, this phase angle 'ϕ' is varied linearly with a baseband or modulating signal $x(t)$

Therefore,

$$\phi_i = \omega_c t + K_p \cdot x(t) \quad \text{---- (2)}$$

Where K_p is the proportionality constant and is known as phase sensitivity of the modulator. This is expressed in radians/volts.

Since, the unmodulated carrier wave is

$$c(t) = A\cos\phi$$

Therefore, the phase modulated wave will be

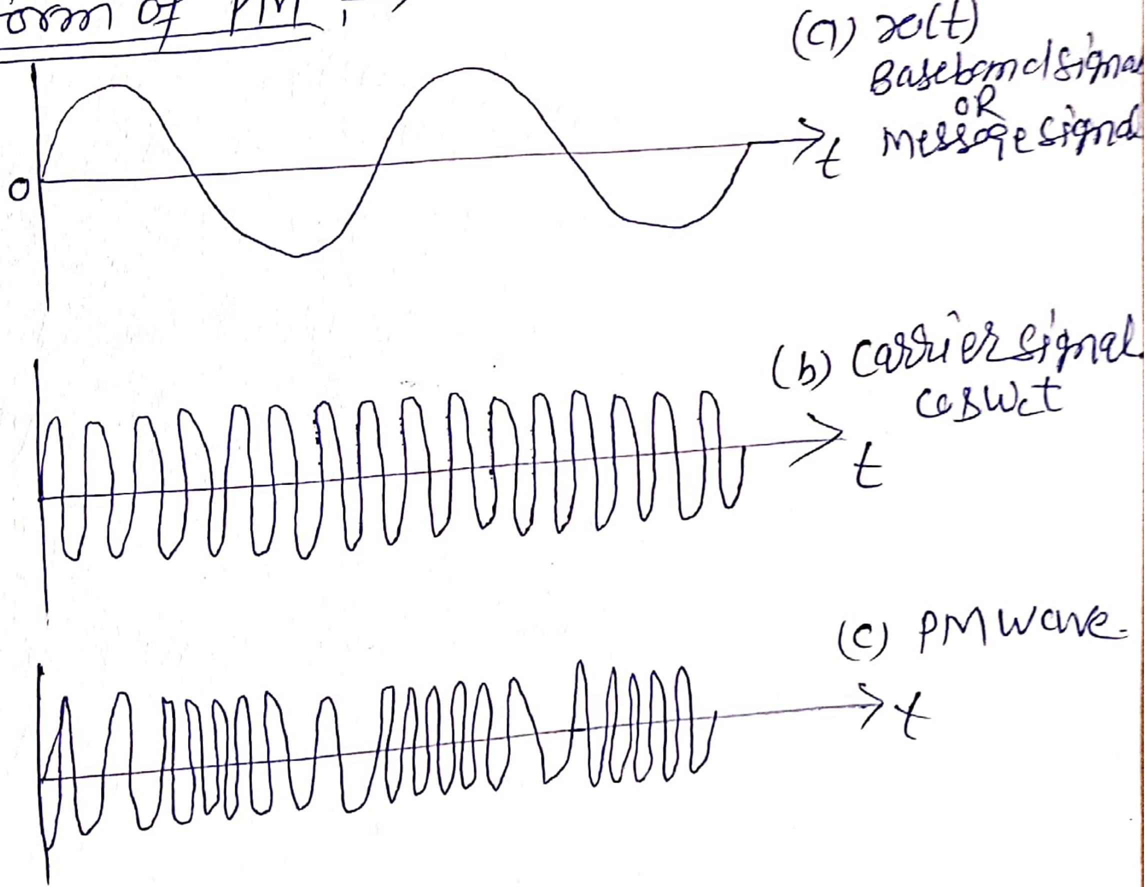
$$s(t) = A\cos\phi_i \quad \text{-------- (3)}$$

putting value of ϕ_i (from Eqn. (2))

OR $\boxed{s(t) = A\cos[\omega_c t + K_p \cdot x(t)]}$

which is the required mathematical expression for a phase modulated wave.

Wave form of PM :→



Frequency Modulation (FM) :→

Definition — FM is that type of angle modulation in which the instantaneous frequency ω_i is varied linearly with a message or baseband signal $x(t)$ about an unmodulated carrier frequency ω_c.

This means that the instantaneous value of the angular frequency ω_i will be equal to the carrier frequency ω_c plus a time varying component proportional to the baseband signal $x(t)$.

Mathematical Representation :→

We know that the instantaneous frequency is as

$$\omega_i = \omega_c + k_f \cdot x(t) \quad \text{......} \quad (1)$$

Where k_f is proportionality constant and is known as the frequency sensitivity of the modulator. Unit is Hz/Volt.

Now, let the expression for unmodulated carrier signal as

$$c(t) = A\cos(\omega_c t + \theta_0)$$

$$\text{OR} \quad c(t) = A\cos\phi \;, \quad \text{where } \phi = \omega_c t + \theta_0 \quad (2)$$

'ϕ' is the total phase angle of the unmodulated carrier.

Let ϕ_i be the instantaneous phase angle of the modulated carrier is

$$c(t) = A\cos\phi$$

On FM A must remain constant & only angle ϕ will change.

Hence, the freq^n modulated wave will be

$$s(t) = A\cos\phi_i \quad \text{......} \quad (4) \;, \quad \text{where } \phi_i = \text{Instantaneous phase angle}$$

From eq^n ②, on differentiation,

$$\frac{d\phi}{dt} = w_c$$

$$\phi = \int w_c dt \quad \text{......} ③$$

We may write this eq^n for instantaneous phase angle ϕ_i as

$$\phi_i = \int w_i dt \quad \text{......} ④$$; w_i = instantaneous freq^n of freq^n modulated wave.

putting value of w_i from eq^n ①

$$\phi_i = \int [w_c + K_f x(t)] dt = w_c t + K_f \int x(t) dt$$

putting value of ϕ_i in eq^n (A)

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

The FM wave will be

$$\boxed{s(t) = A\cos\left[w_c t + K_f \int_0^t x(t)\, dt\right]}$$

Which is the required general expression for FM wave.

<u>Modulation index, (mf)</u> = $\frac{\text{Freq}^n \text{ deviation}}{\text{Modulation freq}^n} = \frac{\Delta w}{w_m}$

for A.see B.

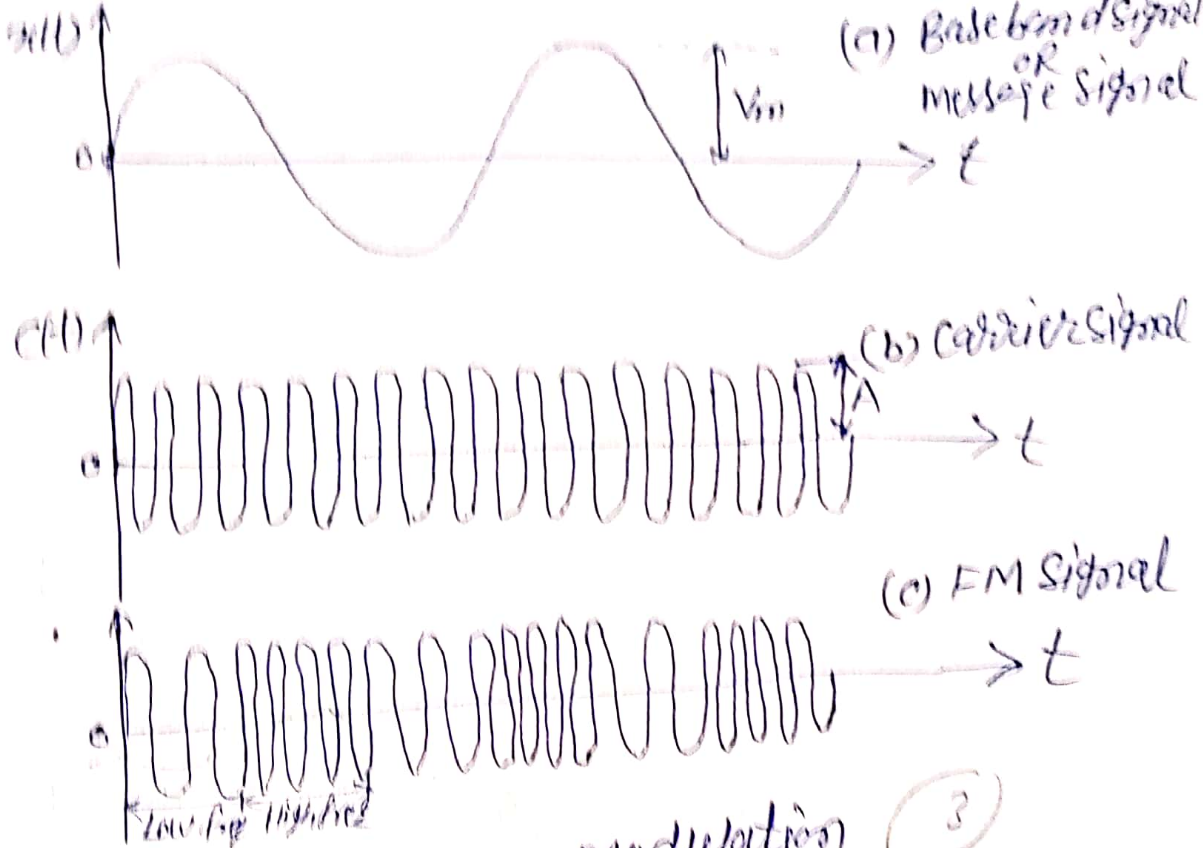
this mf may be > 1 OR $\boxed{m_f = \frac{\Delta f}{f_m}}$

* Depending upon the value of K_f –

i) Narrow band FM :- In this case, K_f is small and hence the bandwidth of FM is Narrow

ii) Wide band FM – In this case, K_f is Large & hence the FM signal has a wide bandwidth.

FM wave ↦



FM Generation → OR FM Modulation

- Direct methods
  - Reactance modulators
  - Varactor diode Modulators
- Indirect methods
  - Armstrong method

Demodulation of FM wave OR FM

- Direct type
  - Frequency discriminators
  - Phase discriminator
    - Foster Seeley Discriminator
    - Ratio Detector
  - Zero crossing detector
- Indirect type
  - Phase locked loop (PLL)

We know that the Angle modulated carrier is as $s(t) = A\cos[\omega_c t + \phi(t)]$

or $s(t) = A\cos[2\pi f_c t + \phi(t)]$ —— (1)

Where A is the amplitude of unmodulated carrier and f_c is the frequency of unmodulated carrier and $\phi(t)$ is the instantaneous phase angle.

Further, instantaneous phase angle is as

$\phi(t) = k_p x(t)$ for phase modulation (PM)

$\phi(t) = 2\pi k_f \int_0^t x(t)dt$ for frequency modultion (FM)

Where $x(t)$ is the modulating or message signal,

k_p is the sensitivities of the phase modulator

k_f is the sensitivities of the frequency modulator

Now channel noise $n_i(t)$ at the I/P of the demodulator is a bandpass noise with power spectral density (PSd) $S_n(f)$ & band with $2(\Delta f + f_m)$ Hz

Thus, noise $n_i(t)$ may be written as

$$n_i(t) = n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t$$

Let $n_c(t) = r(t)\cos\theta(t)$

$n_s(t) = r(t)\sin\theta(t)$

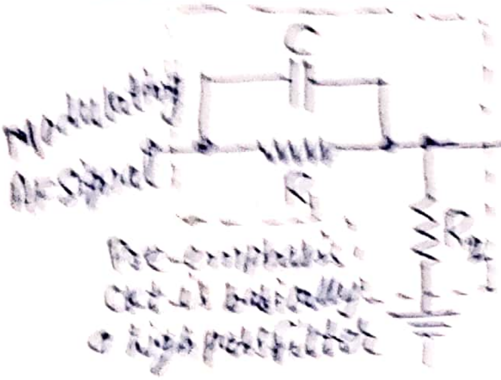
OR $n_i(t) = r(t)\cos\theta\cos\omega_c t - r(t)\sin\theta\sin\omega_c t$

$n_i(t) = r(t)[\cos 2\omega_c t + \theta]$ —— (2)

where $r(t) = \sqrt{n_c^2(t) - n_s^2(t)}$

and $\theta = \tan^{-1}\dfrac{n_s(t)}{n_c(t)}$

PRE-EMPHASIS AND DE-emphasis :→



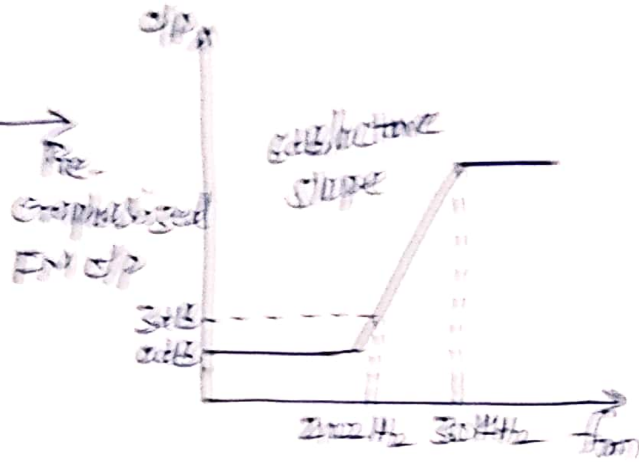
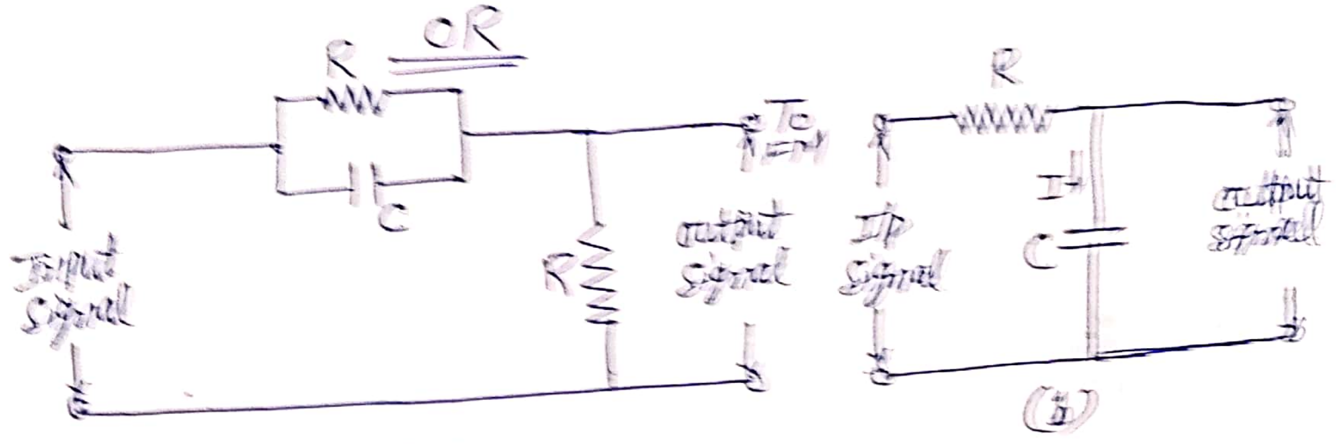


Fig (a):- Pre-emphasis Circuit



(a)

Fig (a) Pre emphasis Circuit (b) De-emphasis Circuit

In the Pre emphasis and de-emphasis method, simple RC networks are used to improve the threshold.

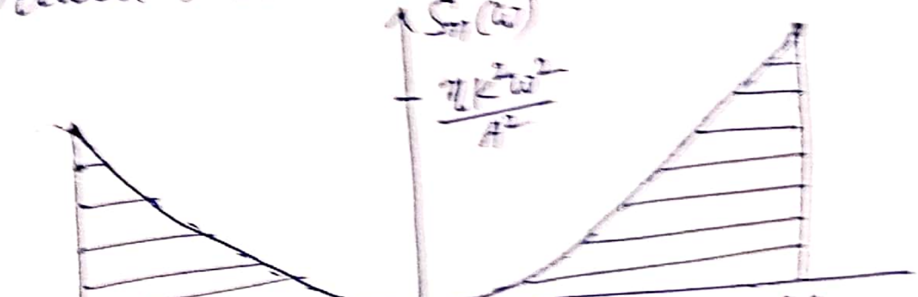


Fig 2- Noise-power Density Spectrum at the Discriminator o/p

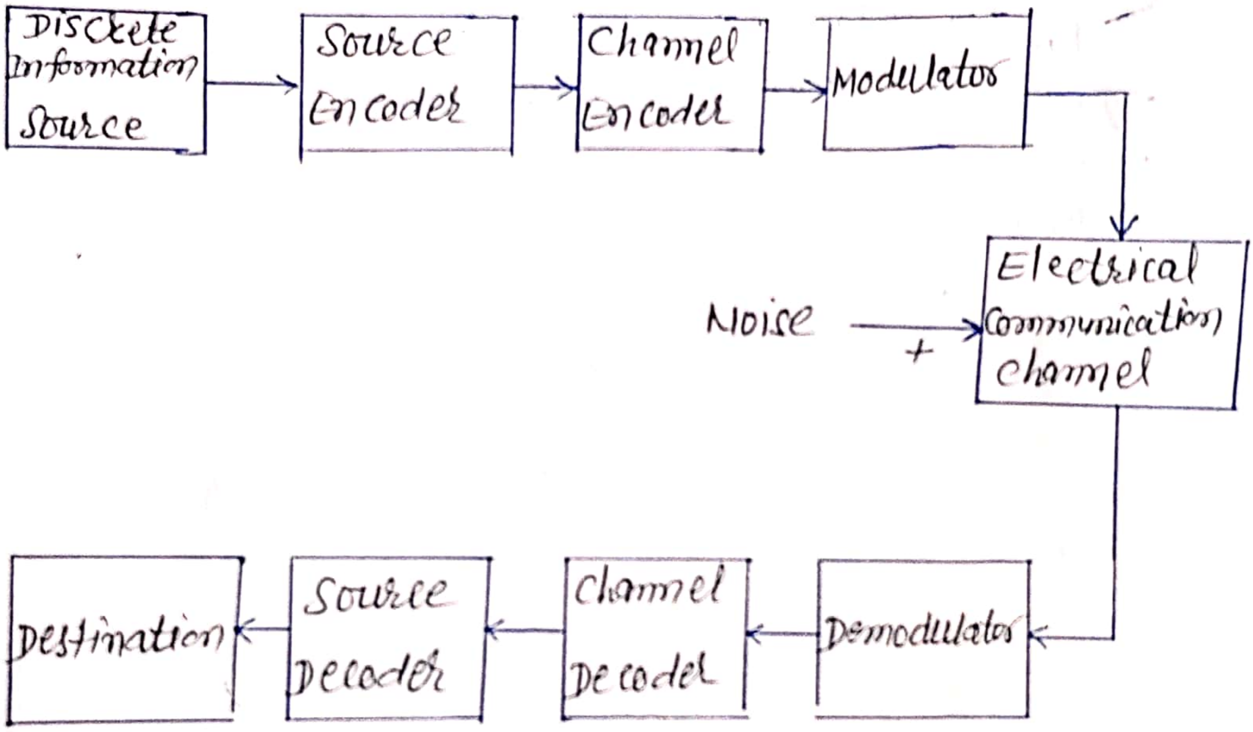
It has been observed in fig 2 that the noise power density spectrum increases as a parabolic function with frequency. Also, we have observed in previous that a message signal has a decaying fourier spectrum with frequency. The result is that signal to noise ratio at the output of the detector becomes very low towards the higher edge of the message band and may cause threshold effect, in spite of the fact that (S_o/N_o) is large enough at lower edge of the message band

Scanned with CamScanner

UNIT-3 : PCM & DELTA MODULATION SYSTEMS

Digital Communication :- In digital communication, the message signal to be transmitted is digital in nature. This means that digital communication involves the transmission of information in digital form.

Block diagram of a Digital Communication system :-



There are many blocks as 1) Discrete information source 2) Source Encoder and Decoder 3) Channel Encoder and Decoder 4) Digital Modulators and Demodulators. 5). Communications channel

1) Discrete information source :- In case of digital communication, the information source produces a message signal which is not continuously varying with time

2) Source Encoder and Decoder :-

The symbols produced by the information source are given to the source encoder. These symbols cannot be transmitted directly. They are first converted into digital form (i.e., binary sequence of 1's and 0's) by the source encoder. Each binary '1' and '0' is known as a bit. The group of bits is called a codeword. The codeword can be of 4, 8, 16 or 32 bits length. As the number of bits are increased in each codeword, the symbols that may be represented are also increased. As an example, 8 bits would have 2^8 i.e., 256 symbols.

Some typically source encoders are pulse code modulators (PCM), delta modulators, vector quantisers etc. Source encoders must have following parameters:-

i) Block size - As an example, the block size of 8 bits source encoder will be 2^8 i.e., 256 codewords.

ii) Codeword length - As an example, if 8 bits are assigned to each codeword, then the codeword length will be 8 bits.

iii) Average data rate ⇒ Data rate = Symbol rate × codeword length

$$= 10 \times 8$$

$$\text{Data rate} = 80 \text{ bits/seconds}$$

iv) Efficiency of the Encoder - The efficiency of the encoder is the ratio of minimum source information rate to the actual output data rate of the source encoder.

3. Channel Encoder and Decoder :-

A channel encoder must have the following important parameters -

i) The coding rate that depends upon the redundant bits added by the channel encoder.

ii) The coding method used

iii) coding efficiency which is the ratio of data rate at the input to the data rate at the o/p of the encoder

iv) Error control capabilities

v) Feasibility of the encoder and decoder

4) <u>Digital Modulation and Demodulators</u> :→

Now, if the modulating signal is digital then digital modulation techniques are used. The carrier signal used by digital modulators is always continuous sinusoidal wave of High frequency.

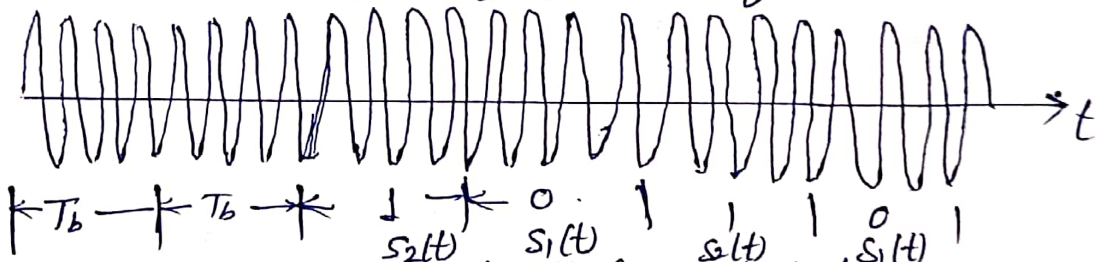


Fig ; The o/p of a digital modulator

For example, if one bit at a time is transmitted, then digital modulator signal is $S_1(t)$ to transmit binary '0' and $S_2(t)$ to transmit binary '1' as shown in fig.

Here the signal $S_1(t)$ has low frequency compared to signal $S_2(t)$. Hence, here, even though the modulated signal seems to be continuous, the modulation is discrete. This means that a signal carrier is converted into two waveforms $S_1(t)$ & $S_2(t)$ because of digital modulation.

Examples of Digital modulation as ASK (Amplitude shift keying, FSK (Frequency shift keying) PSK (Phase shift keying) DPSK (Differential phase shift keying), MSK (Minimum shift keying.

A digital modulation method must have following parameters –

1) Bandwidth needed to transmit the signal
2) Probability of symbol or bit error
3) Synchronous or asynchronous method of detection
4) Complexity of implementation.

5. Communication channel :–

The connection between transmitter and receiver is established through a communication channel, the communication can take place through wirelines, wireless or fiber optic channels.

Advantages and Disadvantages of Digital Communication –

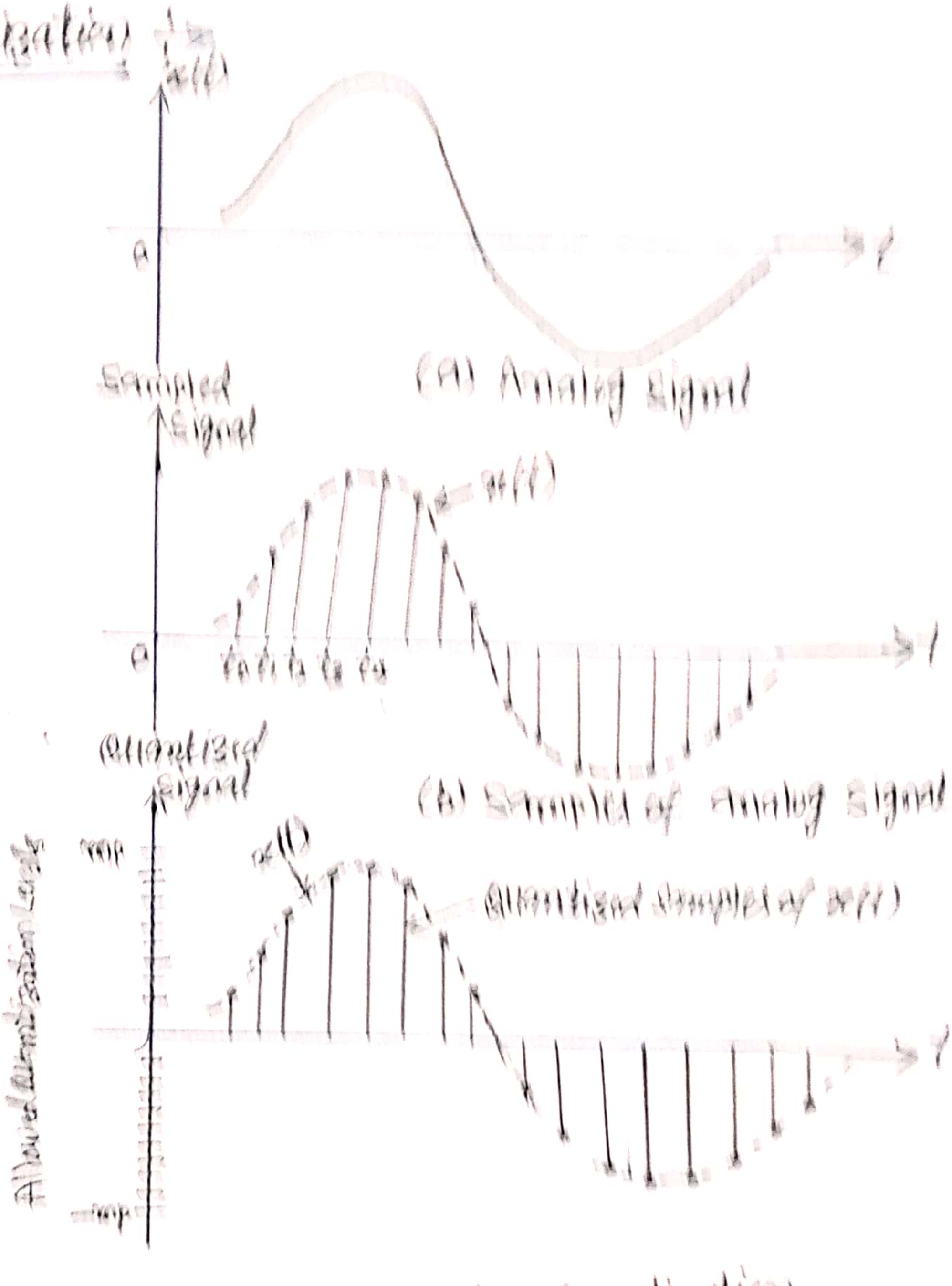
Advantages –

1) It systems are simpler and cheaper
2) In D.C. the speech, video and other data may be merged and transmitted over a common channel using multiplexing.
3) Since the transmission is digital & the channel encoding is used therefore the noise does not accumulate from repeater to repeater in long distance communications.
4). Since the transmitted signal is digital in nature, therefore a large amount of noise interference may be tolerated
5) Since in D.C, channel coding is used, therefore the errors may be detected and corrected in the receivers.

Disadvantages :–

1) Due to analog to digital conversion, the data rate becomes high. Therefore more transmission bandwidth is required for DC
2) Digital Comm. needs synchronization in case of synchronous modulation.

Quantization



(c) Quantization

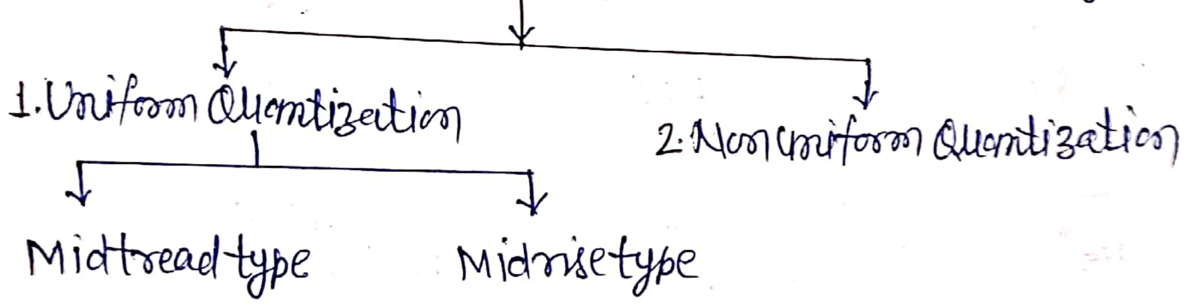
Let us consider an analog signal as shown in fig.(a). First of all, we get samples of this signal according to sampling theorem. For this purpose, we mark the time instants t_0, t_1, t_2 and so on, at equal time intervals along the time axis. At each of these time instants, the magnitude of the signal is measured and thus samples of the signal are taken.

Now, we can say that the signal in figure (b) is defined only at the sampling instants. This means that it no longer is a continuous function of time, but rather, it is a discrete time signal.

However, ~~it is~~ a since the magnitude of each sample can take any value in a continuous range, the signal in figure (b) is still an analog signal.
This difficulty is neatly resolved by a process known as Quantization. In Quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels.

Each of magnitude $\delta = \frac{2m_p}{L}$ ← intervals.

<u>Types of Quantization</u> :→ There are two types as

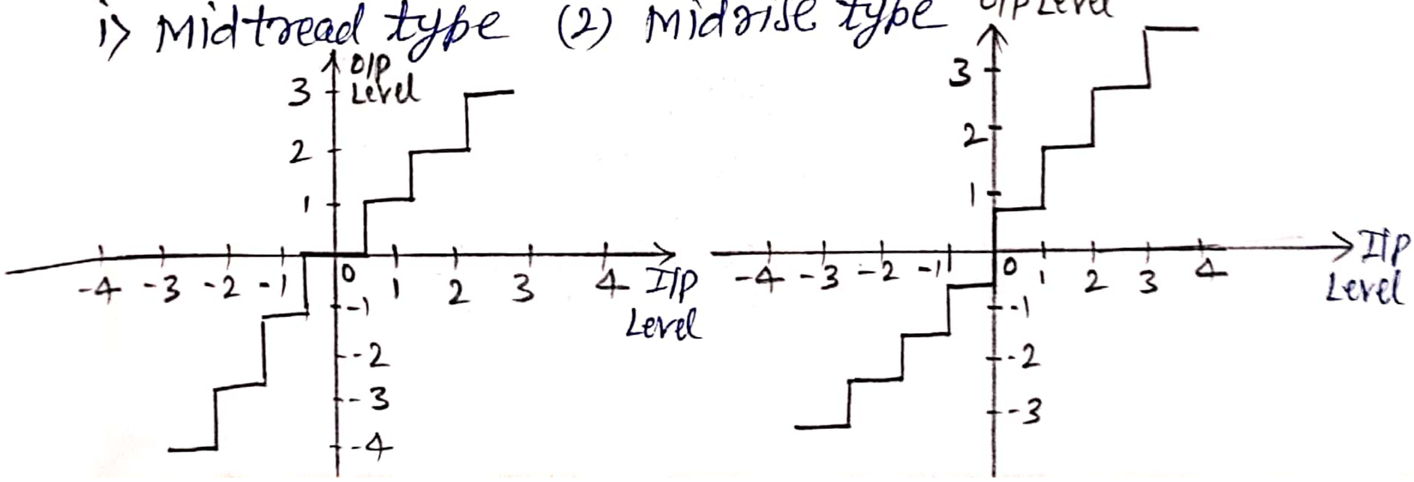


1) <u>Uniform Quantization</u> :→ A Uniform Quantization is that types of Quantizer in which the 'step size' remains same throughout the input range

2) <u>Non Uniform Quantization</u> :→ A Non uniform Quantisation is that type of Quantizer in which the 'step size' varies according to the input signal values.

<u>Types of Uniform Quantization</u> :- There are two types as

i) Midtread type (2) Midrise type



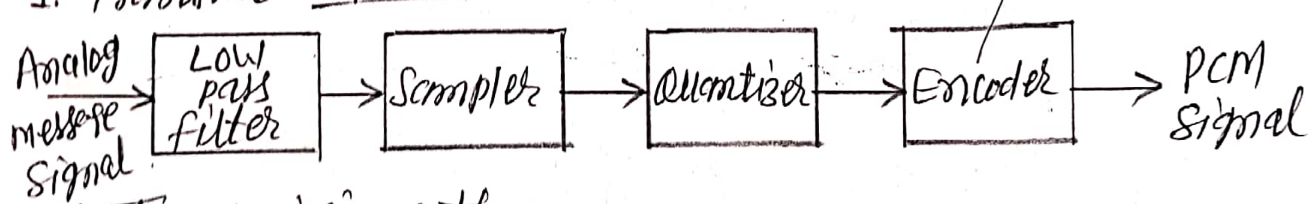
PULSE CODE MODULATION (PCM) :-

Definition:- Pulse code modulation is known as a digital pulse modulation technique. Infact, the PCM is quite complex compared to the analog pulse modulation i.e., PAM, PWM & PPM) in the sense that the message signal is subjected to a great number of operations.

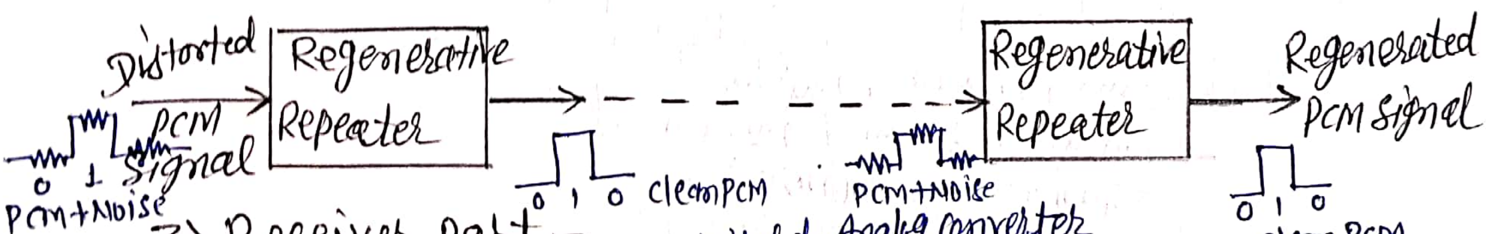
Elements of a PCM system –

It consists of three main parts as transmitter, 2) transmission path 3) Receiver. The transmitter consists of four operations as Low pass filter, sampling, quantizing and Encoding as shown in figure(a). The transmission path consists of two regenerative repeaters, 1st is I/p repeat ers and o/p repeater. The receiver part consists of four blocks or operations as Regeneration circuit, Decoder, Reconstruction filter and Destination.

1. Transmitter part –



2.) Transmission path –



3.) Receiver part –

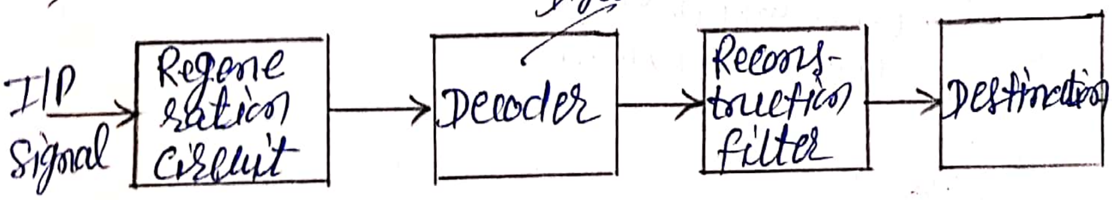


Figure:- The basic elements of a PCM system (a) Tx. (b) Transmission path (c) Rx.

Few Important Points—

- PCM is a digital pulse modulation system but there is an important difference between them. Pulse modu. like PAM, PWM & PPM are Analog pulse modulation system.
- This means that the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position.
- The information is transmitted in the form of code words. A PCM system consists of a PCM encoder (Transmitter) & a PCM decoder (Receiver).
- The essential operations in the PCM transmitter are sampling, quantizing and encoding.
- PCM Generator or Transmitter—

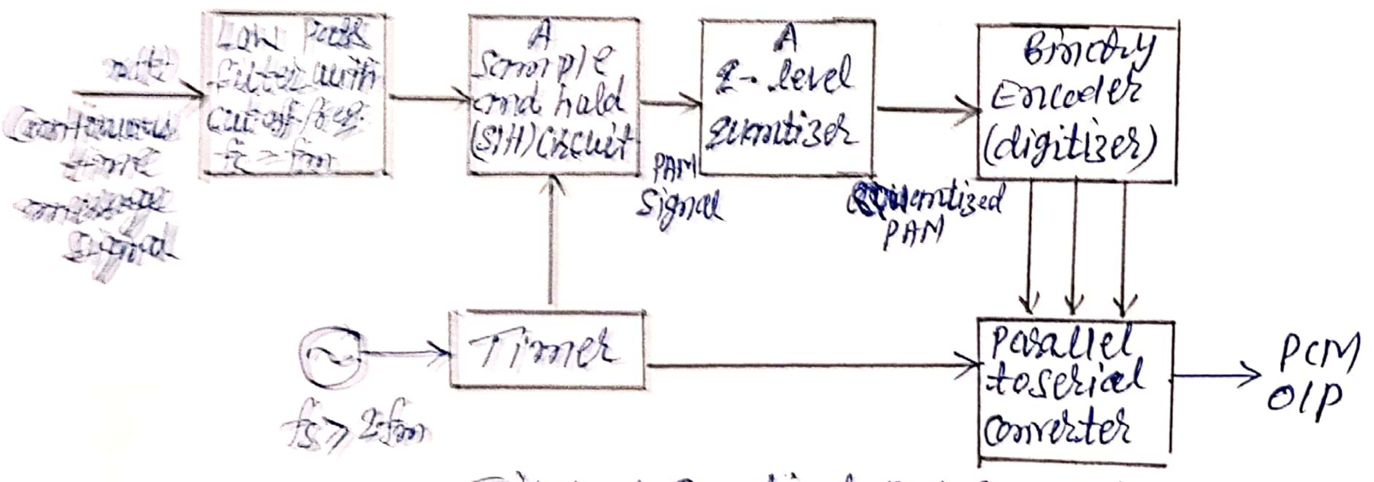


Fig:- A Practical PCM Generator

Black diagram of a Repeater—

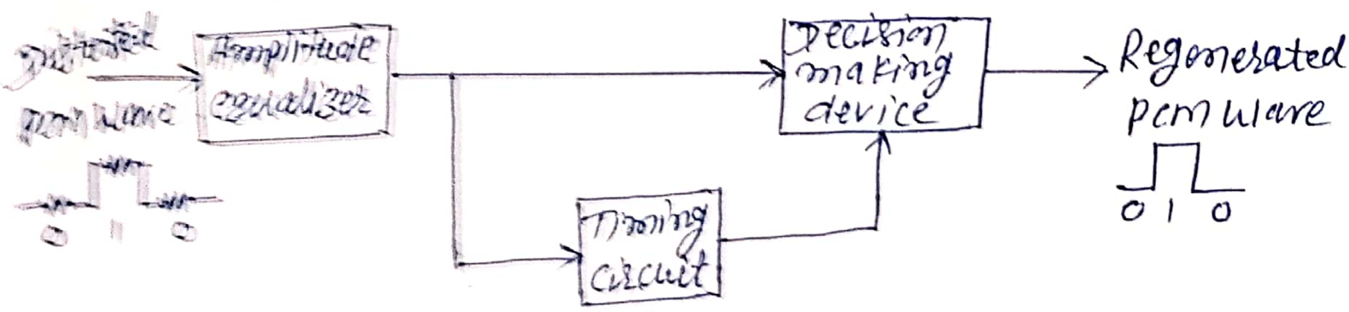


Fig:- Block diagram of a Regenerative repeater

Application of PCM :→

1) With the advent of fibre cables, PCM is used in telephony.

ii) In space communication, space craft transmits signals to earth. Here, the transmitted power is quite small (10 or 15W) and the distances are very large (i.e., a few million Km). However, due to the high noise immunity, only PCM systems can be used in such applications.

Advantages of PCM

i) PCM provides high noise immunity.

2) Due to digital nature of the signal, we can place repeaters between the tx. and Rxs. Infact, the repeaters regenerate the received PCM signal. This can not be possible in analog systems. Repeaters further reduce the effect of noise.

3) We can store the PCM signal due to its digital nature.

4) We can use various coding techniques so that only the desired person can decode the received signal

Drawbacks of PCM or disadvantages of PCM –

i) The Encoding, decoding and quantizing circuitry of PCM is complex.

ii) PCM requires a large bandwidth as compared to the other systems.

Delta Modulation :-

i) Reason to use Delta modulation—

We have observed in PCM that it transmits all the bits which are used to code a sample. Hence, signaling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, Delta Modulation is used.

Working :-

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Input signal x(t) is approximated to step signal by the delta modulator. This step size is kept fixed. The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e., $+\Delta$ and $-\Delta$. Now, if the difference is positive, then approximated signal is increased by one step, i.e. 'Δ'. If the difference is -ive, then approximated signal is reduced by 'Δ'.

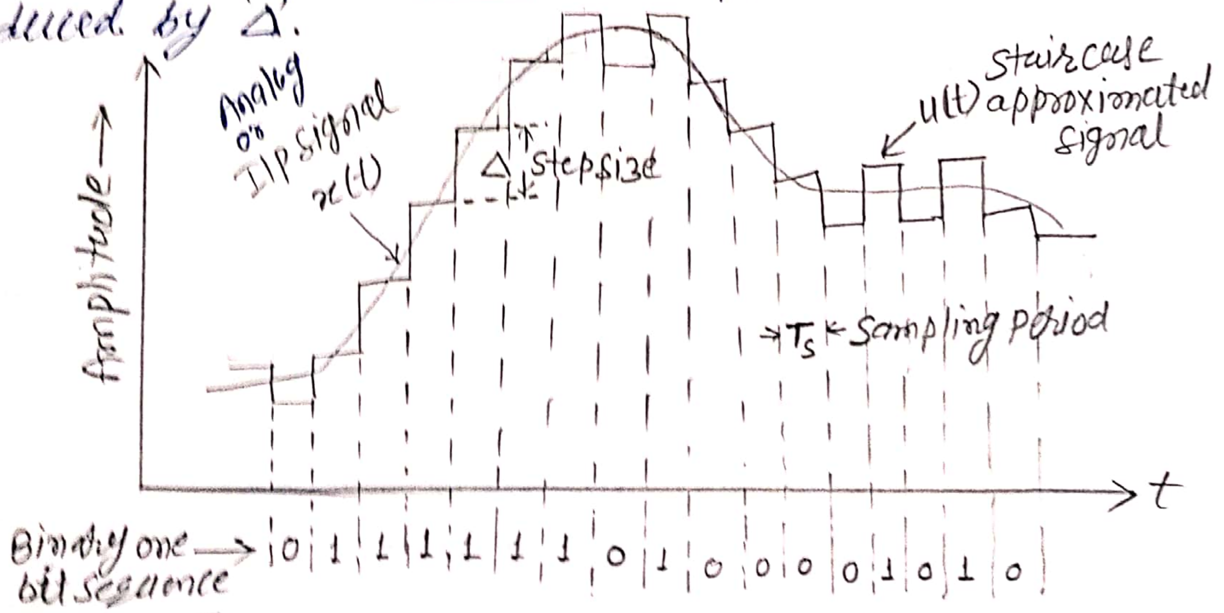


Fig. :- Delta modulation waveform

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Hence, for each sample, only one binary bit is transmitted. Figure shows the analog signal $x(t)$ and its staircase approximated signal $u(t)$ by the delta modulator.

Transmitter part –

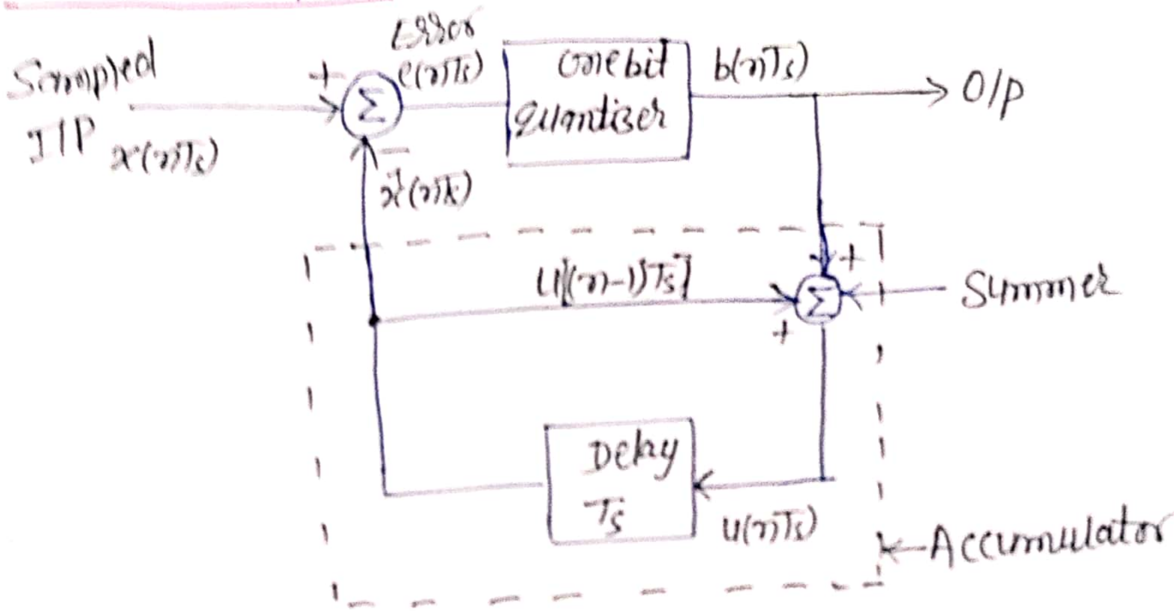


Fig. (a) :- A Delta modulation Transmitter

Receiver part :→

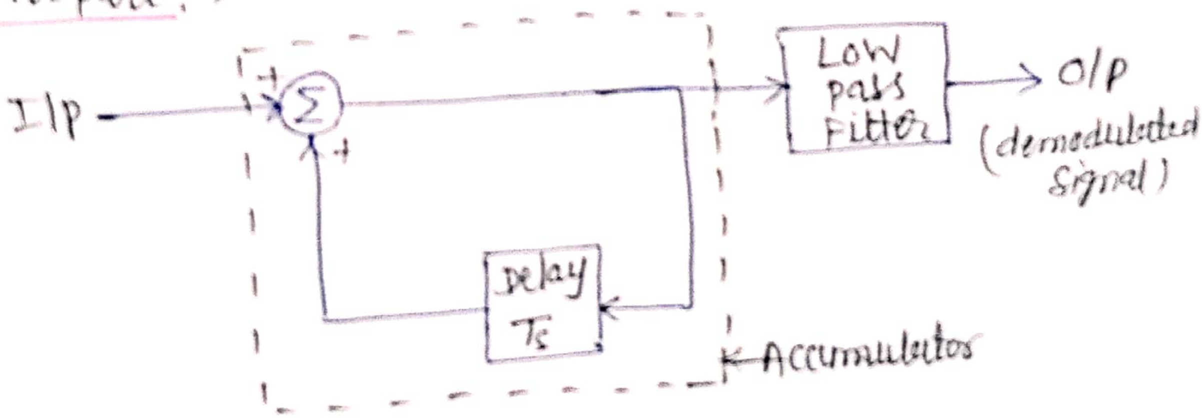


Fig. (b) :- A Delta modulation Receiver

Transmitter of Deltamodulation consists of two blocks as one bit quantizer and accumulator. The summer in the accumulator adds quantizer output ($\pm\Delta$) with the previous sample approximation. This gives present sample approximation.
i.e.,

$$u(nT_s) = u(nT_s - T_s) + [\pm\Delta]$$

OR $$u(nT_s) = u[(n-1)T_s] + b(nT_s)$$

The previous sample approximation $u(n-1)T_s$ is Restored by delaying one sample period T_s. The sampled input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get Error signal $e(nT_s)$.
Thus depending on the sign of $e(nT_s)$, one bit quantizer Generates an o/p of $+\Delta$ or $-\Delta$. If the step size is $+\Delta$, then binary '1' is transmitted and if it is $-\Delta$, then binary '0' is transmitted.

Receiver Part of Deltamodulation :→

The Receiver part of DM consists of two blocks as the accumulator and Lowpass filter. The accumulator Generates the staircase approximated signal o/p and is delayed by one sampling period T_s. It is then added to the I/p signal. If I/p is '1' then it adds $+\Delta$ step to the previous o/p. If I/p is '0' then one step 'Δ' is subtracted from the delayed signal. the LPF has the cutoff frequency equal to Highest frequency in $x(t)$

Advantages of Delta Modulation :-

1) Since, the DM transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for DM compared to PCM.

2) The transmitter and receiver implementation (or circuitry) is very simple. there is no analog to digital converter required in DM.

Disadvantages of Delta Modulation :-

The DM has two major disadvantages or drawbacks

i) Slope overload distortion 2) Granular or idle noise

i) Slope Overload Distortion :-

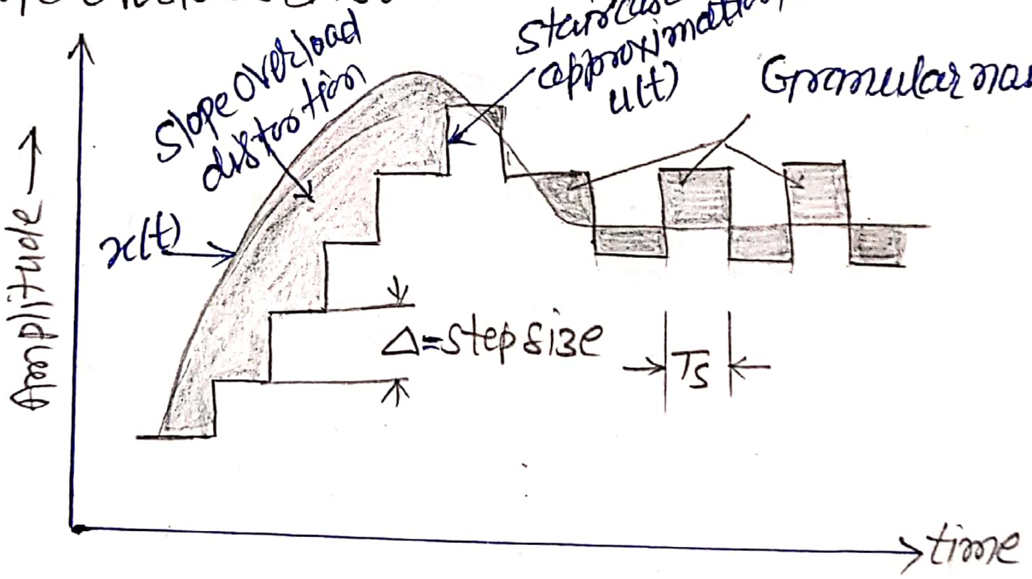


Fig.4:- Quantization Errors in Delta modulation.

According to figure or waveform, the rate of rise of I/P signal x(t) is so High that the staircase approximated signal cannot approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t). Hence, there is a Large Error between the staircase approximated signal & the original I/P signal x(t). This Error or noise is known as slope overload distortion.

2) Granular OR Idle Noise :-

Granular or idle noise occurs when the step size is too large compared to small variations in the I/p signal. This means that for very small variations in the I/P signal, the staircase signal is changed by large amounts (Δ) because of large step size.

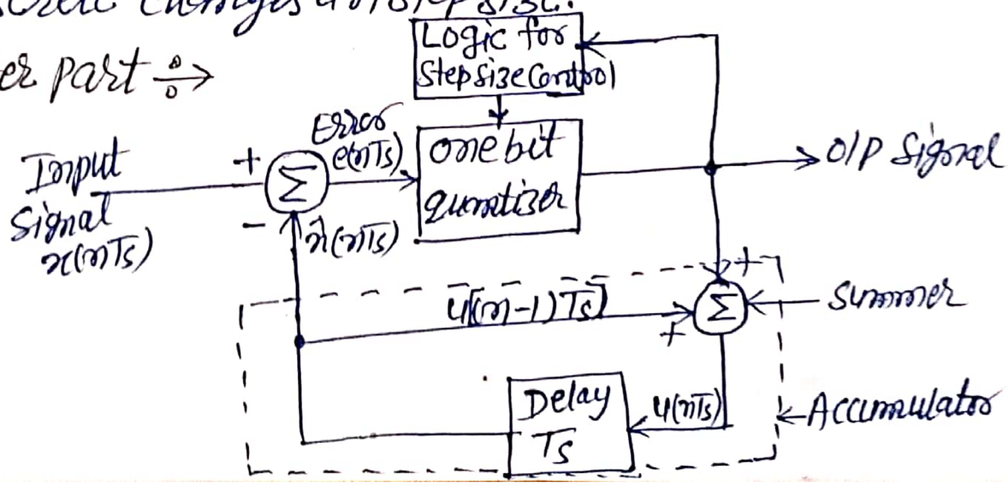
According to the figure, ~~that~~ when the I/p signal is almost flat, the staircase signal u(t) keeps on oscillating by $\pm\Delta$ around the signal. The Error between the I/p and staircase signal is called Granular noise. [The solution to this problem is to make step size small]

Adaptive Delta Modulation (ADM) :-

Reason to use Adaptive D.M :-

To overcome the quantization Errors due to slope overload and Granular noise, the step size (Δ) is made adaptive to variations in the I/p signal x(t). Particularly in the steep segment of the signal x(t), the step size is increased. Also, if the I/p is varying slowly, the step size is reduced. Then this method is known as Adaptive Delta modulation. The ADM can take continuous changes in step size or discrete changes in step size.

Transmitter part :-



According to block diagram, the Transmitter consists of three blocks as i) one bit quantizer ii) Logic for stepsize control iii) Accumulator. The step size increases or decreases according to a specified rule depending on one bit quantizer o/p. As an example, if one bit quantizer o/p is High (1), then step size may be doubled for next sample. If one bit quantizer o/p is low, then step size may be reduced by one step.

Receiver part of ADM :-

According to block diagram, the Receiver part of ADM consists of three blocks as i) Logic for stepsize control ii) Accumulator iii) low pass filter. There are two portions, the 1st portion produces the step size from each incoming bit. Exactly the same process is followed as that in Transmitter. The previous I/P and present I/P decides the step size. It is then applied to an accumulator which builds up staircase waveform.

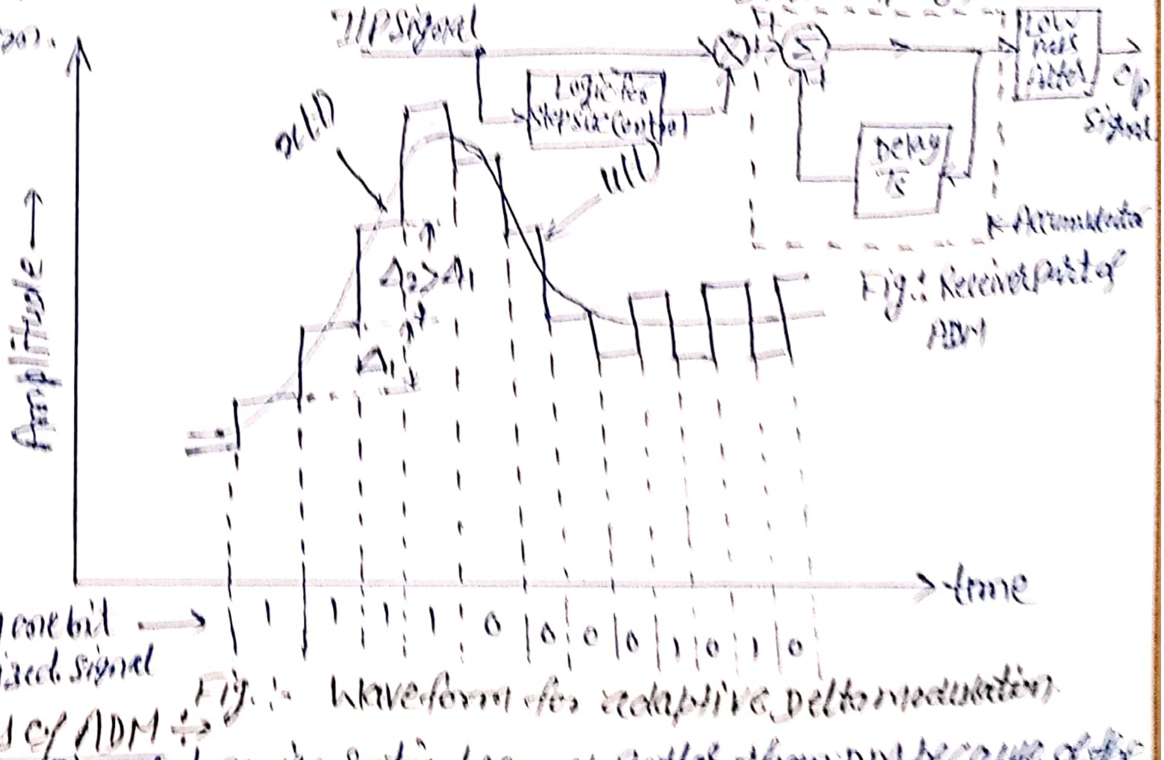


Fig.: Receiver part of ADM

Fig.:- Waveform for adaptive delta modulation

Advantages of ADM :-

i) The signal to noise ratio becomes better than DM because of the reduction in slope overload distortion and idle noise.

ii) Because of the variable step size, the dynamic range of ADM is wider than DM.

iii) Utilization of bandwidth is better than delta modulation.

Differential pulse code Modulation (DPCM) :-

Reason to use DPCM :- It may be observed that the Samples of a Signal are Highly correlated with each other. This is due to the fact that any Signal does not change fast: this means that its value from present sample to next sample does not differ by large amount. the adjacent samples of the signal carry the same information with a little difference. When these samples are encoded by a standard PCM system. The resulting encoded signal contains some Redundant information.

Redundant information in PCM -

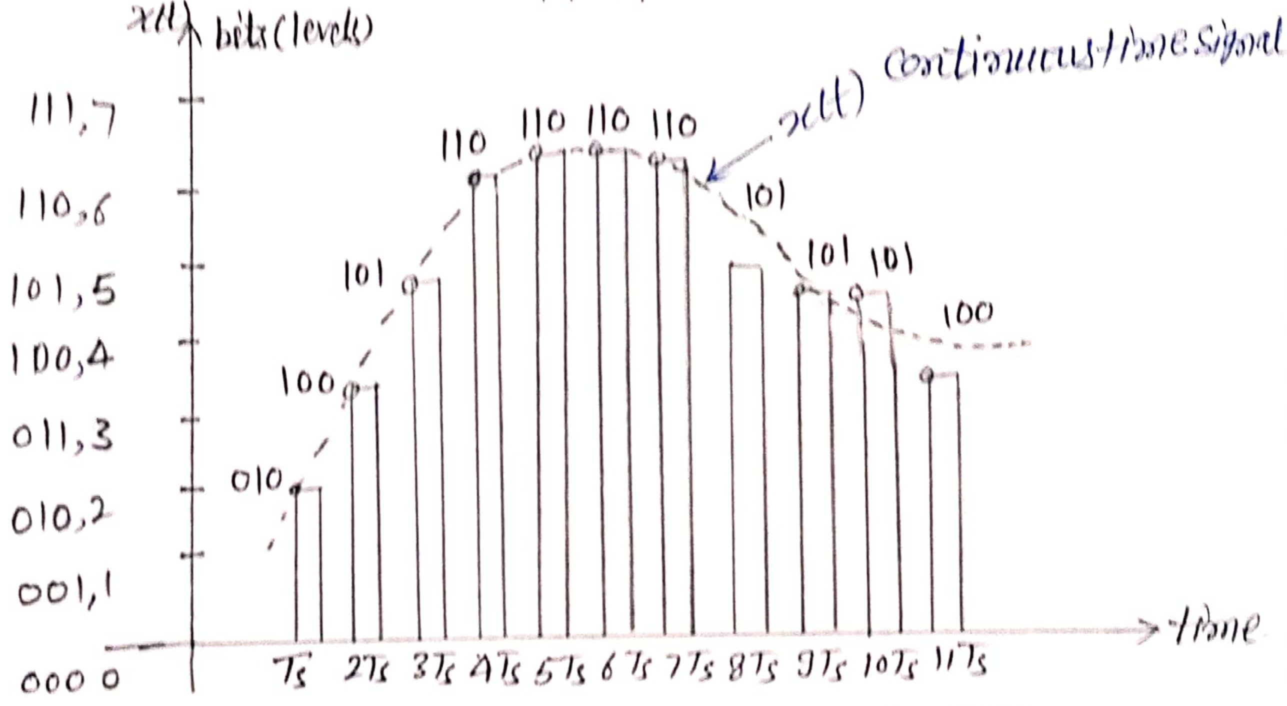


Fig. :- Redundant information in PCM

The samples are encoded by using 3 bit (7 levels) PCM. The sample is quantized to the nearest digital level as shown by small circles in the figure. The encoded binary value of each samples is written on the top of the samples. We can observe from figure that the samples taken at 4Ts, 5Ts & 6Ts are encoded to same value of 110. this information can be carried only by one sample. But three samples are carrying the same information means that it is Redundant

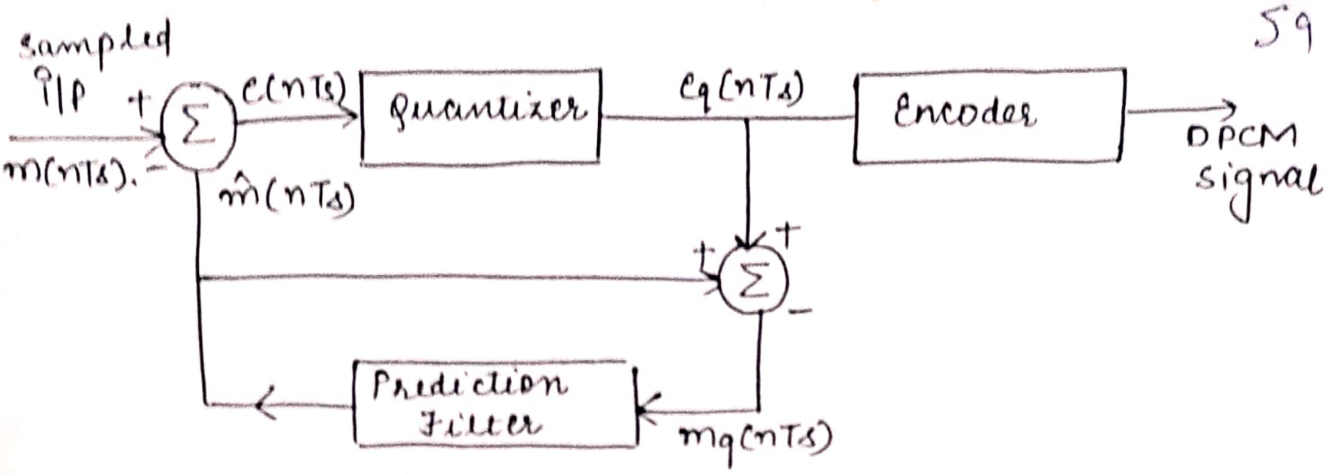
Consider another example of samples taken at $10T_s$ and $11T_s$. The difference between these samples only due to last bit and 1^{st} two bits are redundant, since they are not change.

If this redundancy is reduced then overall bit rate will decrease & no. of bits required to transmit one sample will also be reduced. This type of digital pulse modulation scheme is known as DPCM

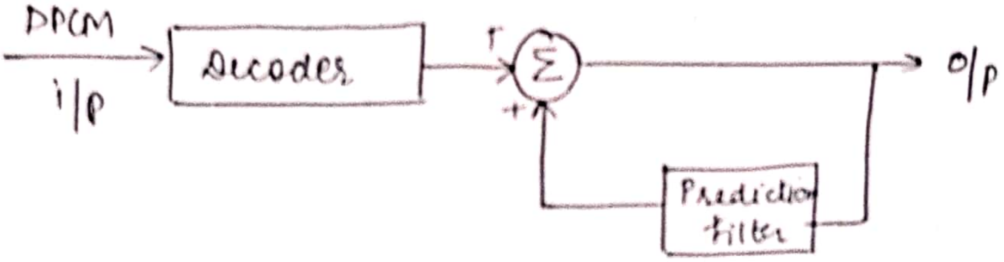
Transmitter part of DPCM :→ In fact the DPCM works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may be not be exact but it is very close to the actual sample value. According to the block diagram of DPCM Transmitter, the sampled signal is denoted by $x(nT_s)$ and the predicted signal is denoted by $\hat{x}(nT_s)$. The comparator $\boxed{\Sigma}$ finds out the difference between the $x(nT_s)$ & $\hat{x}(nT_s)$. This is known as prediction error ($e(nT_s)$).

Receiver part of DPCM :→

The decoder 1^{st} reconstructs the quantized error signal from incoming binary signal. The prediction filter o/p and quantized error signals are summed up to give the quantized version of the signal. Thus the signal at receiver differs from actual signals by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.



DPCM Transmitter.



DPCM Reciever.

Advantages:

i) DPCM system is very easy

ii) Co-relation b/w successive are very good so the o/p samples is far better than that of PCM

iii) Very less number of levels are required to encode.

Disadvantages:

idea is to encode the message signal in digital pulses form & at reciever side to sense them then we may skip the need of predict filter that is not easy to design & also cause of error.

Q5: Comparison b/w DM, PCM, ADM, DPCM

Parameters ~~Numbers~~	DM	ADM	PCM	DPCM
No. of bits	uses only 1 bit for sample	uses only 1 bit to encode 1 sample	uses 4, 8, or 16 bit per sample	Bits can be more
Step size and Levels	fixed cannot be varied.	step size varies acc. to signal	No. of levels depend upon no. of bits level size is kept fixed	fixed no. of levels are used
Quantisation error & Distortion	slope over load distortion & granular noise	quantisation noise present No other errors	quantisation error. depence upon no. of levels used.	slope overload distortion & quantistn is used
Transmission BW	lowest B.W req.	lowest B.W req.	highest B.W req. as no. of bits req. are high	B.W lower than PCM
Feedback	feedback exists in transmitter	Feedback exists	No feedback	Feedback exists
Complexity of implementation	simple	simple	complex	Simple.

Unit-4: DIGITAL MODULATION

Base Band Transmission

A Base Band Digital Communication System :→

A Base band Digital communication system consists of many elements as
1) Source 2) Multiplexer 3) Line coder
4) Regenerative Repeater.

1) **Source** – as a data set, a computer, a digitized voice signal (PCM or DM), a digital facsimile or television or telemetry equipment.

2) **Multiplexer** – To utilize this capacity effectively, we combine several sources through a digital multiplexer using the process of interleaving. Thus a channel is time shared by several message simultaneously.

3) **Line coder** – The o/p of a multiplexer is coded into electrical pulses or waveforms for the purpose of transmission over the channel. This process is known as line coding or transmission coding.

Example – Line code is on-off (RZ), Polar (RZ), Bipolar, also known as Alternate mark inversion (AMI)
1) NRZ (Non Return to zero) and RZ (Return to zero) Unipolar
2) NRZ and RZ polar
3) NRZ Bipolar
4) Manchester
5) Polar quaternary NRZ

Unit # 2 ; "Base band transmission"

When the signal is transmitted over channel, without any modulatio of high frequency carrier. This is called baseband signal transmissio

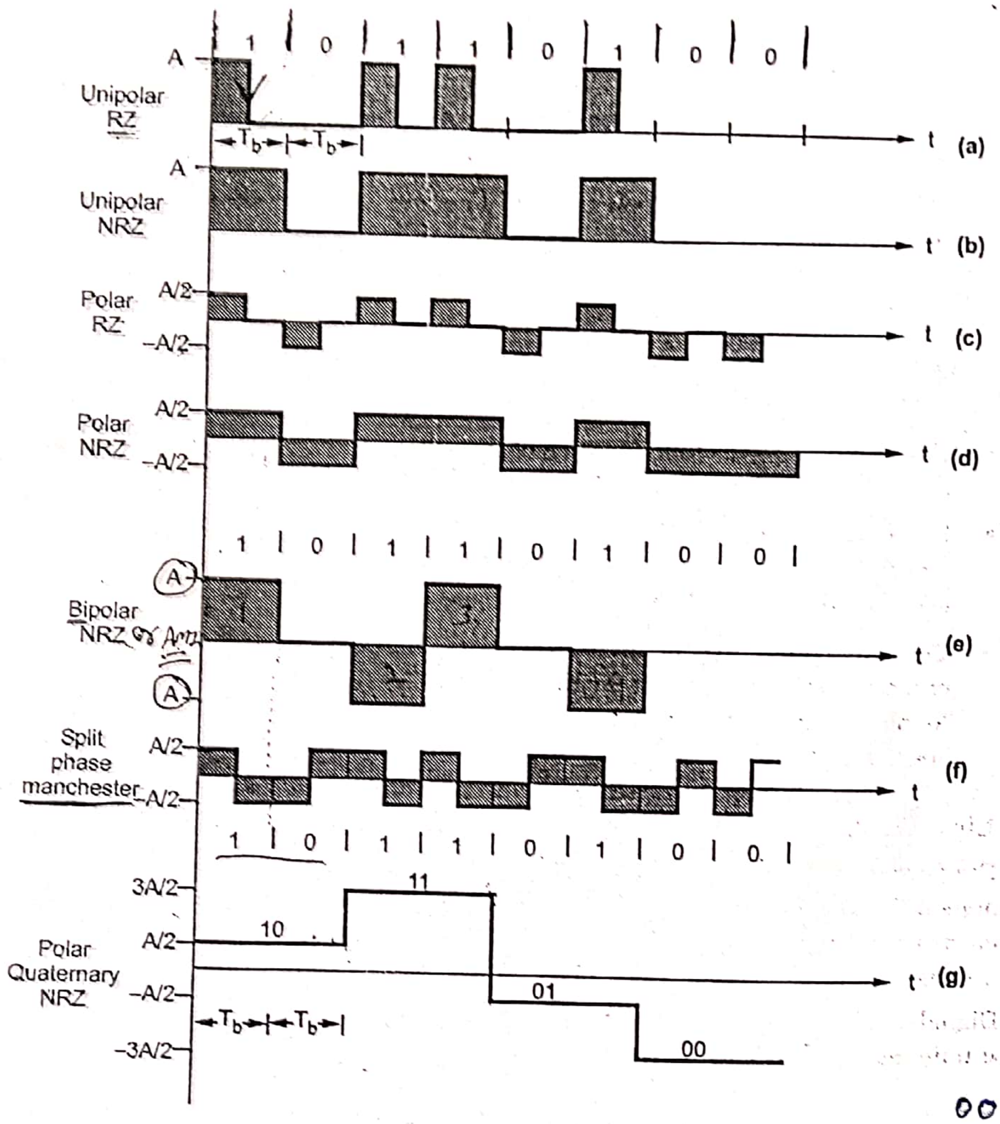


Fig. 1 Various digital PAM signals formats
(a) Unipolar RZ (b) Unipolar NRZ (c) Polar RZ (d) Polar NRZ
(e) Bipolar NRZ (f) Split phase manchester (g) Polar quaternary NRZ

00 = $-3A/2$
01 = $-A/2$
10 = $+A/2$
11 = $+3A/2$

Ref: J.S. Katre

Problems occurred in baseband transmission: major problem is "Inte symbol interference". This is due to dispersive nature of the chan

Corrective measures to minimize errors in baseband transmission :-
"Nyquist criterion" gives a condition for distortionless baseban transmission. It is possible to reduce the effect of intersymbol interference with the help of "raised cosine spectrum."

[3] Digital Data formats :-

(a) Unipolar RZ and NRZ -

In unipolar format the waveform does have a single polarity. The waveform is simple on-off. In the unipolar RZ form, the waveform has zero value when symbol '0' is transmitted and waveform has 'A' volts when '1' is transmitte and for remaining $T_b/2$ waveform returns to zero value, i.e for unipolar RZ form. [see fig. 1(a)]

If symbol '1' is transmitted,

$$x(t) = A \quad \text{for } 0 \le t < T_b/2 \text{ (Half interval)}$$
$$= 0 \quad \text{for } T_b/2 \le t < T_b \text{ (" ")}$$

And if symbol '0' is transmitted, $x(t) = 0$ for $0 \le t \le T_b$ (Complete i

Example -

various PAM formats are shown in Fig. 1. All the formates are shown for a binary message "10110100".

for unipolar NRZ form (see fig 1(b)) -

If symbol '1' is transmitted, $x(t) = A$ for $0 \le t < T_b$ (complete interv

If " '0' " , $x(t) = 0$ for $0 \le t < T_b$ " ,

the pulse does not return to zero on its own.

Imp points :-

(1) As compared to RZ format, NRZ pulse width (pulse to pulse interval is same) is more. Hence energy of the pulse is more.

(2) Unipolar format has some average DC value. This DC value does not carry any information.

(b) Polar RZ and NRZ -

In the polar RZ format, symbol '1' is represented by positive voltage polarity and '0' is represented by negative voltage polarity. Since this is RZ format, the pulse is transmitted only for half duration. If symbol '1' is transmitted,

$$x(t) = +A/2 \quad \text{for } 0 \le t < \tfrac{T_b}{2} \quad \text{and}$$
$$= 0 \quad \text{for } T_b/2 \le t < T_b \qquad \text{(see fig. 1.c)}$$

If symbol '0' is transmitted,

$$x(t) = -\frac{A}{2} \quad \text{for } 0 \le t < \frac{T_b}{2}$$
$$= 0 \quad \text{for } \frac{T_b}{2} \le t < T_b.$$

And for polar NRZ, (see fig 1(d))

If symbol '1' is transmitted

$$x(t) = +\frac{A}{2} \quad \text{for } 0 \le t < T_b$$ (for complete interval)

And if symbol '0' is transmitted,

$$x(t) = -\frac{A}{2} \quad \text{for } 0 \le t < T_b.$$ (for complete interval)

Some Imp. points –

① Since polar RZ and NRZ formats are bipolar, the averge DC value is minimum.

② If probabilities of occurence of symbol '1' and '0' are same, then averge DC components of the waveform will be zero.

(c) Bipolar NRZ (Alternate mark inversion; AMI)

In this format successive 1's are represented by pulses with alternate polarity and 0's are represented by no pulses. (see Fig 1.e).

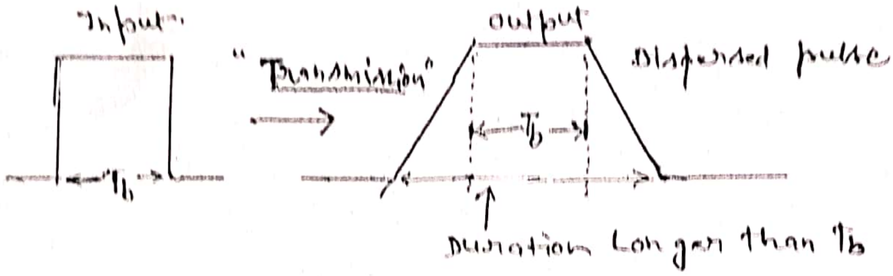
If there are even no. of 1's, the DC component of the waveform will be zero. The advantage of this format is that the ambiguities due to transmission sign inversion are eliminated.

(d) Split phase Manchester – [fig. 1(f)].

Here if symbol '1' is to be transmitted, then a positive half interval pulse is followed by a negative half interval pulse. If symbol '0' is to be transmitted, then a negative half interval pulse is followed by a positive half interval pulse. Thus for any symbol the pulse takes positive as well as negative values ie.

(1) Intersymbol Interference (ISI) :-

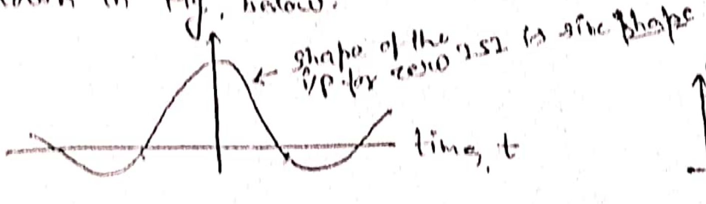
The ISI arises due to the imperfections in the overall freq. response of the system. Due to differentially attenuated or differentially delayed, the input pulse of duration T_b seconds will be dispersed at the output, over an interval which is longer than T_b seconds. Due to this dispersion, the symbols each of duration T_b will interfere with each others when transmitted over the communication channel.

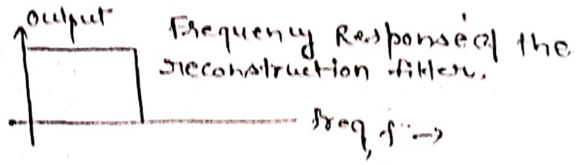


The presence of ISI will introduce errors in the decision at the receiver output.

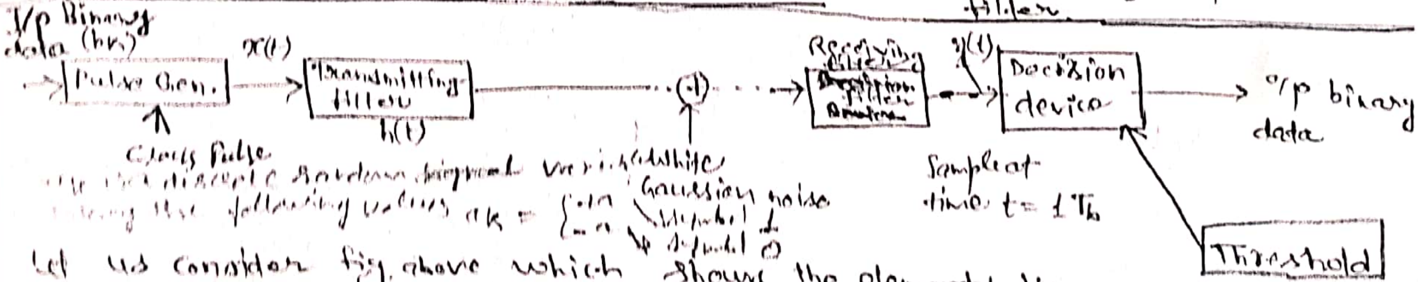
To Reduce ISI :→

Instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero. This is known as Nyquist pulse shaping. The sinc pulse transmitted to have a zero "ISI" As shown in Fig. below.





(a) Ideal pulse shape for zero ISI, (b) Frequency response of the filter.



Let us consider fig. above which shows the elements binary PAM system. The input signal consists of a binary data sequence $\{b_k\}$ with a bit-duration of T_b seconds.

This sequence is applied to a pulse generator to produce a discrete PAM signal which given by.

$$x(t) = \sum_{k=-\infty}^{\infty} a_k v(t - kT_b).$$

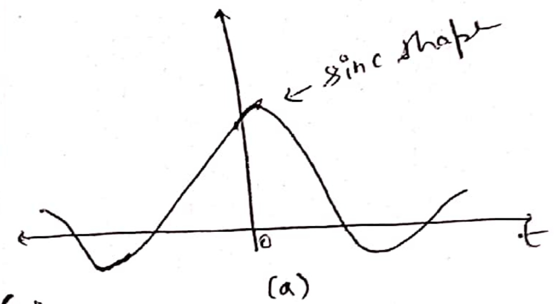
Nyquist's criterion for distortionless baseband Binary transmission -

Basic concept -

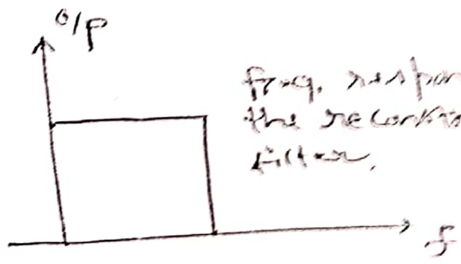
1) In order to minimize the effects of ISI, we have to design the transmitting and receiving filters properly.

(2) The transfer function of the channel and the shape of transmitted pulse are generally specified.

(3) It has been proved that the function which produces a zero ISI is a sinc function. Hence, instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero. This is known as Nyquist pulse shaping.



(a)



(b)

(4) Further, we know that Fourier transform of a sinc pulse is a rectangular function. Hence, to preserve all the frequency components, the frequency response of the filter must be exactly flat in the pass band and zero in the attenuation band as shown in above fig. (b).

(5) We know the o/p of the receiving filter is,

$$y(t) = \mu \sum_{k=-\infty}^{\infty} a_k \, p(t - kT_b).$$

o/p y(t) is dependent on the received pulse p(t) & the scaling factor μ.

(6) To reconstruct the transmitted data sequence $\{b_k\}$, First "extracting" & then "decoding" the corresponding sequence of weights from the output $y(t)$.

(iii) Extraction - is basically the process of sampling. The signal $y(t)$ is sampled at $t = iT_b$.

(iv) Decoding - The decoding should be such that the contribution of the weighted pulse i.e. $a_k p(iT_b - kT_b)$, for $i = k$ be free from ISI. This can be stated mathematically as under:

$$p(iT_b - kT_b) = \begin{cases} 1 & \text{for } i = k \\ 0 & \text{for } i \neq k \end{cases}$$

where $p(0) = 1$ due to normalizing.

If $p(t)$ i.e, received pulse satisfies the above expression, then the receiver o/p given by.

$$y(t_i) = \mu a_i \quad \text{(for all } i\text{)}$$ ("which indicates zero ISI in the absence of noise. The noise term is ignored.")

The eye pattern -

The eye pattern is an experimental tool that can be used in the field for evaluating the combined effect of ISI and channel noise in the case of baseband data-transmission system under operational conditions. It is the pattern that is displayed on the screen of a 'high-persistence' screen oscilloscope showing synchronized superposition of the pulses in successive time slots, or symbol intervals, with the sweep time and synchronization of the oscilloscope appropriately adjusted. It is called an eye pattern because it resembles the human eye for binary baseband signalling. Figure below shows the formation of an eye pattern on the screen of oscilloscope.

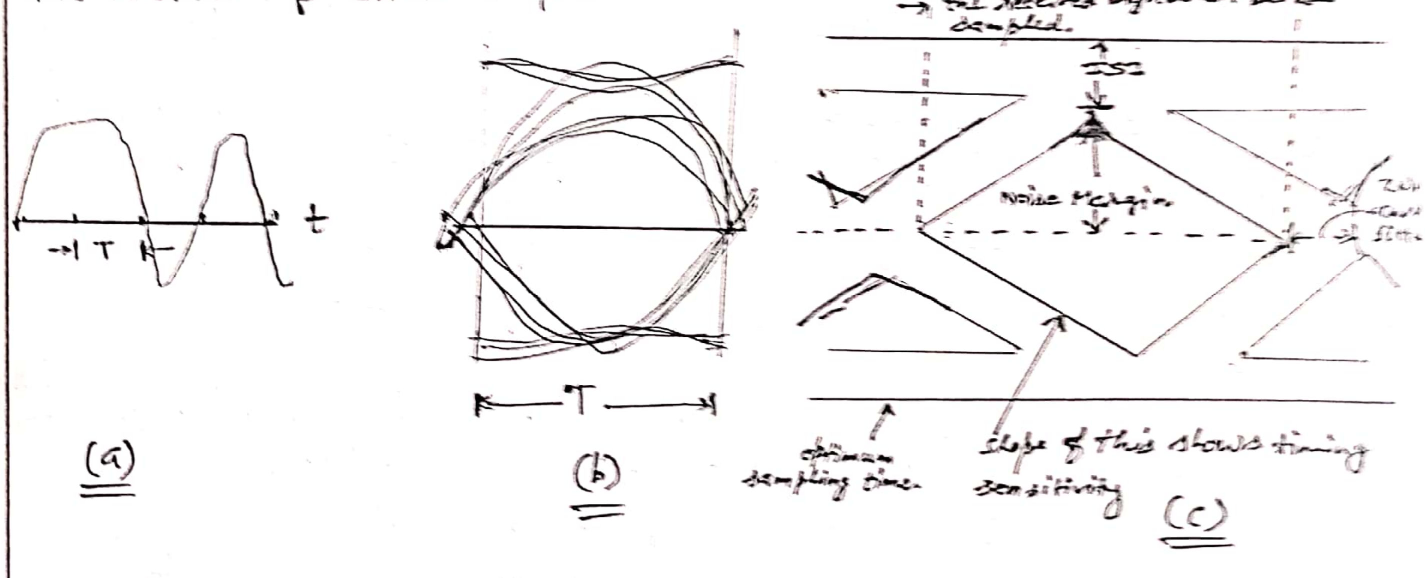


Fig. (a) Polar binary signal distorted by transmission through the channel (b) The eye pattern, (c) Generalized binary eye pattern and its interpretations

Quite a lot of useful information is provided by the eye pattern as to how the baseband transmission system is performing. One can, by observing the pattern, draw inference regarding the extent of ISI, the extent of zero crossing jitter, the noise margin available etc.

Introduction :→ Modulation is defined as the process by which some characteristics of a carrier is varied in accordance with a modulating signal.

In digital communications, the modulating signal consists of binary data or an M-ary encoded version of it. This data is used to modulate a carrier wave (usually sinusoidal) with fixed frequency.

In digital communication, the modulation process involves switching or keying the Amplitude, frequency or phase of the carrier in accordance with the I/p data.

Thus, there are three basic modulation techniques

1) ASK (Amplitude shift keying) OR ON OFF Keying
2) FSK frequency shift keying
3) PSK Phase shift keying.

Digital Modulation formats.

1) ASK Waveform –

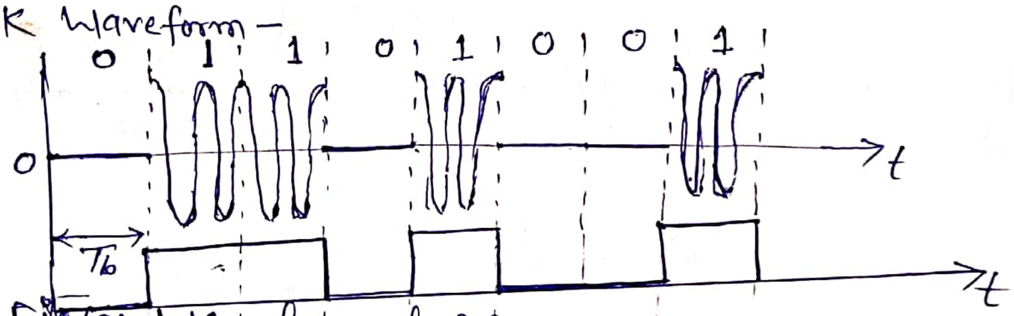


Fig.(a) Waveform of ASK

2) PSK Waveform. –

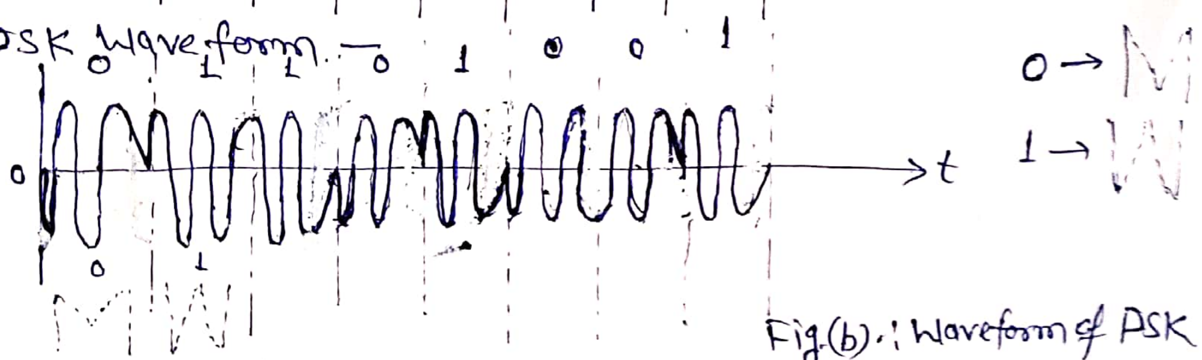


Fig.(b).: Waveform of PSK

3) FSK Waveform –

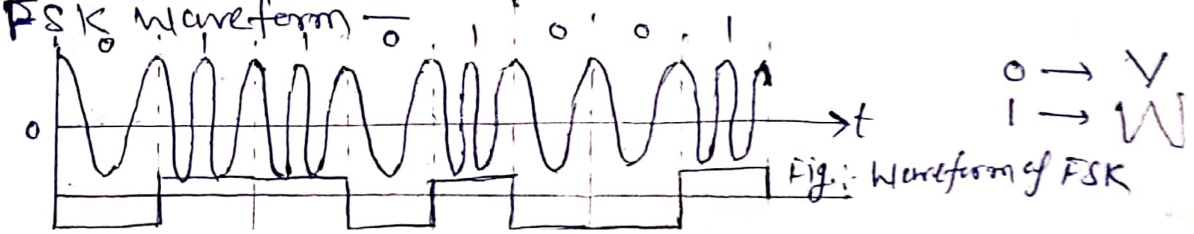


Fig.: Waveform of FSK

Digital Modulation or Switching or Signaling :-

When it is required to transmit digital signal on a bandpass channel, the amplitude, frequency or phase of the sinusoidal carrier is varied in accordance with the incoming digital data. Since the digital data is in discrete steps, the modulation of the bandpass sinusoidal carrier is also done in discrete steps. This type of modulation is digital modulation or is also known as switching or signaling.

Bandpass channel :- This means that the channel will transmit some range or band of frequencies. Such type of transmission is known as bandpass transmission and the communication channel is known as bandpass channel.

ASK :- If an amplitude of the carrier is switched depending on the I/p digital signal, then it is called amplitude shift keying (ASK).

FSK :- If the frequency of the sinusoidal carrier is switched depending upon the I/p digital signal then it is known as the frequency shift keying (FSK).

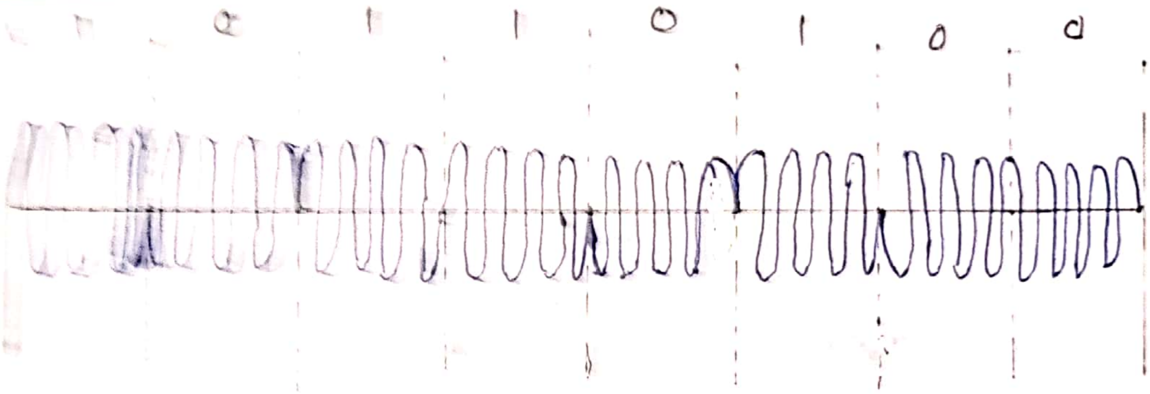
PSK :- If the phase of the carrier is switched depending upon the I/p digital signal.

* This is similar to phase modulation. Since the phase & frequency modulation has constant amplitude envelope therefore FSK and PSK also has a constant amplitude envelope. Because of constant amplitude of FSK & PSK the effect of non-linearities, noise interference is minimum on signal detection.

M-ary transmission :- In digital modulation instead of transmitting one bit at a time, we transmit two or more bits simultaneously. This is known as M-ary transmission.

Quadrature modulation — We use two quadrature carriers for modulation. This process is known as quadrature modulation.

ASK or BASK ⇒ For example 10110100



Binary Amplitude Shift Keying OR on-off keying :→ (BASK) or digital

Amplitude Shift Keying (ASK) or ON-OFF Keying (OOK) is the simplest digital modulation technique. In this method, there is only one bit unit energy carrier and it is switched on or off depending upon the I/P binary sequence.

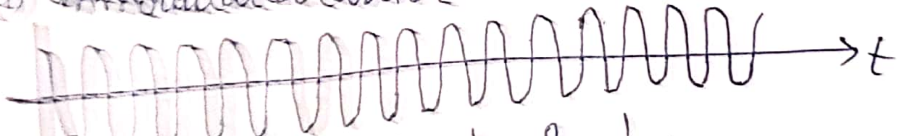
Expression and waveforms —

The ASK waveform may be represented as

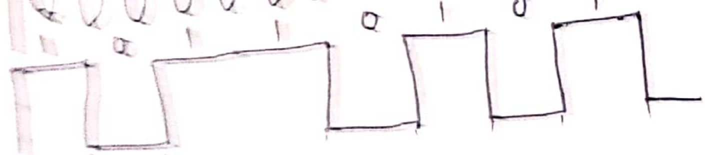
$$s(t) = \sqrt{2P_s}\cos(2\pi f_c t) \quad (\text{To transmit '1'}) \; ---- \; (1)$$

To transmit symbol '0', the signal $s(t) = 0$ i.e., no signal is transmitted. Signal $s(t)$ contains some complete cycles of carrier frequency f_c.

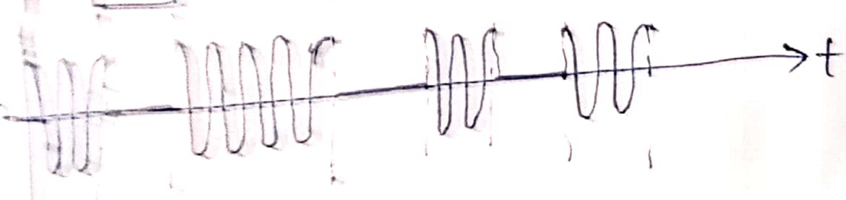
Waveforms — (a) unmodulated carrier



(b) NRZ unipolar bit sequence



(c) ASK waveform



Signal Space Diagram of ASK —

The ASK waveform of eq^n $s(t) = \sqrt{2P_s}\cos(2\pi f_c t)$ for symbol '1' can be represented as,

$$s(t) = \sqrt{P_s T_b}\cdot\sqrt{2/T_b}\cos(2\pi f_c t) = \sqrt{P_s T_b}\,\phi_1(t) \quad \text{-----} (2)$$

This means that there is only one carrier function $\phi_1(t)$. The signal space diagram will have two points on $\phi_1(t)$. One will be at zero and other will be at $\sqrt{P_s T_b}$.

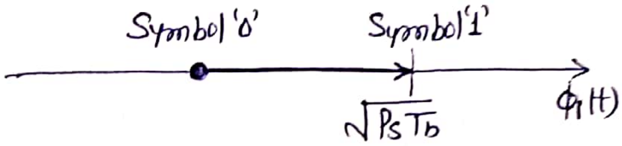


Fig. signal space dia. of ASK

Thus, the distance between the two signal points is

$$d = \sqrt{P_s T_b} = \sqrt{E_b} \quad \text{-----} (3)$$

Generation or Modulation or Transmission of ASK Signal :→

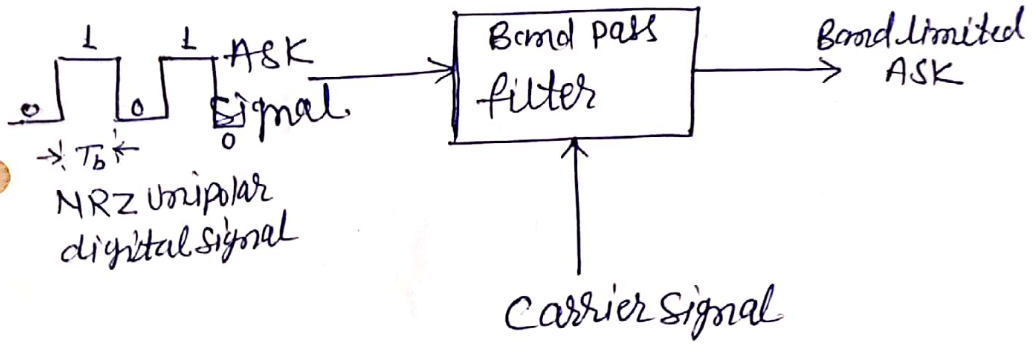
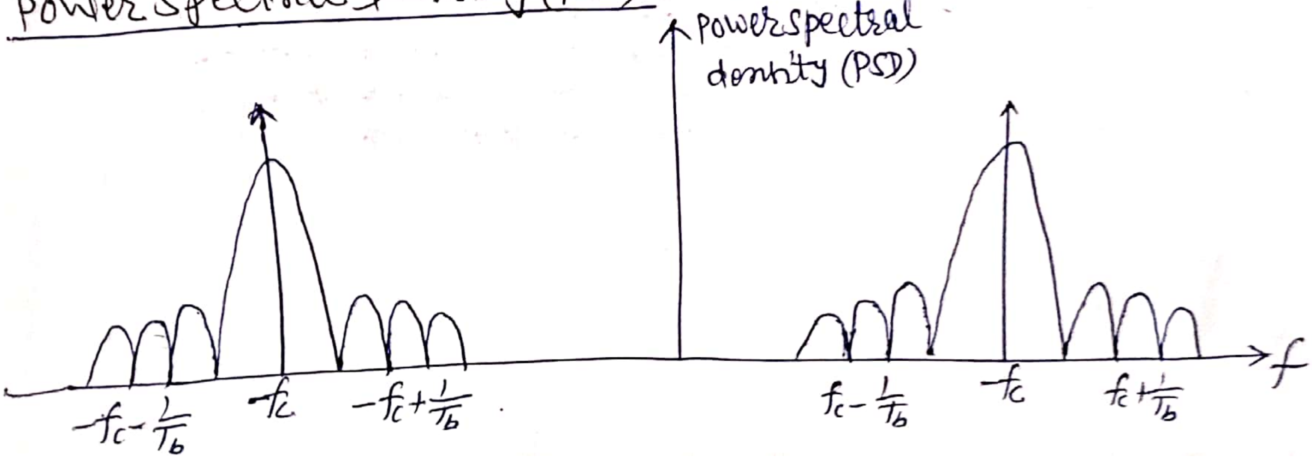


Fig. :- Generation of binary ASK waveform

Power Spectral Density (PSD) :- of ASK Signal



ASK (Amplitude shift keying) :→

This is the simplest digital modulation technique, where a binary information signal directly modulates the amplitude of an analog carrier.
ASK is one of the digital modulation techniques to modulate the digital waveform such as PCM output.
ASK sometimes called digital amplitude modulation (DAM)
ASK is similar as that if the information signal is digital and the amplitude of the carrier is varied proportional to the information signal, a digitally modulated signal called amplitude shift keying (ASK)

ASK Reception or Demodulation :-> or Coherent detection

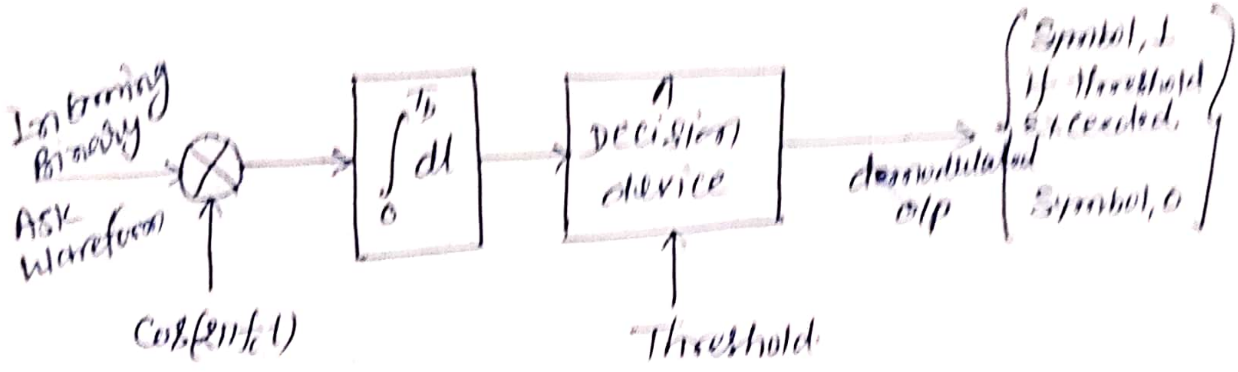


Fig.:- Coherent detection of binary ASK signals.

Binary Frequency Shift Keying (BFSK) or FSK :->

Generation of BFSK :->

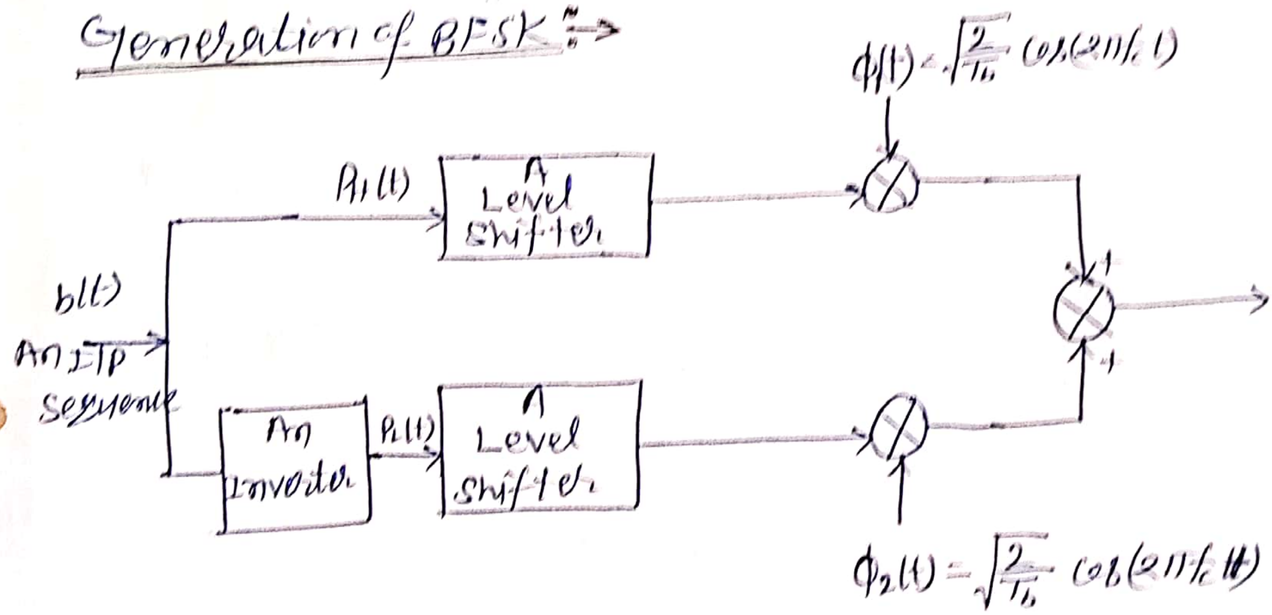


Fig.:- Block diagram of BFSK generation.

BFSK Receiver :- OR Coherent Detection of BFSK :->

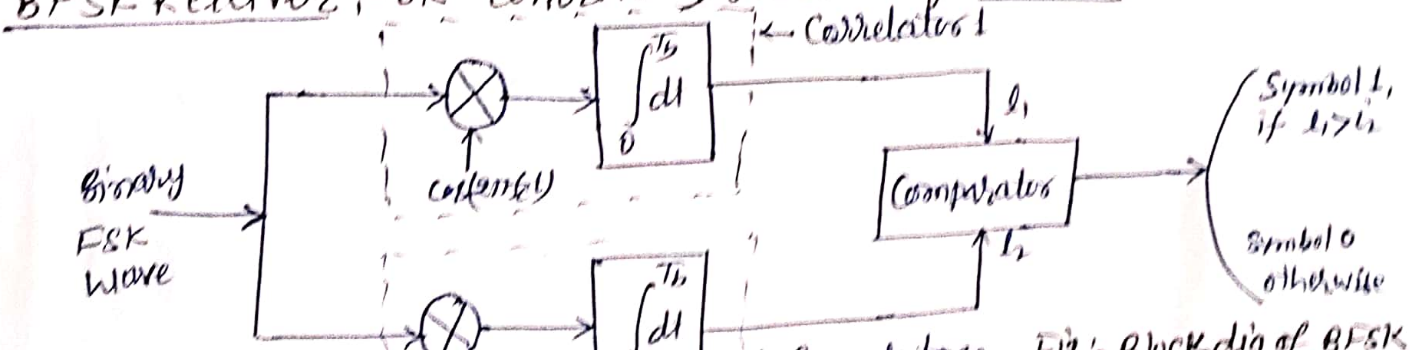


Fig.:- Block dia of BFSK receiver.

Salient features of BFSK —

i) BFSK is relatively easy to implement

ii) It has better noise immunity than ASK.

Draw back of BFSK :→

The major draw back is its High bandwidth requirement.

Expression for BFSK — Then we can write following equations. Let there be a freq^n shift by Ω

If $b(t) = '1'$, Then $S_H(t) = \sqrt{2P_s}\cos(2\pi f_c + \Omega)t$ --- ①

$b(t) = '0'$, then $S_L(t) = \sqrt{2P_s}\cos(2\pi f_c - \Omega)t$ --- ②

Hence, there is increase or decrease in frequency by Ω.

The equation ① \$ ② combinely may be written as

$$S(t) = \sqrt{2P_s}\cos[(2\pi f_c + d(t)\Omega)t] \quad \text{--- ③}$$

* Hence, if symbol '1' is to be transmitted, the carrier freq^n will be $f_c + \left(\frac{\Omega}{2\pi}\right)$ and is represented by f_H

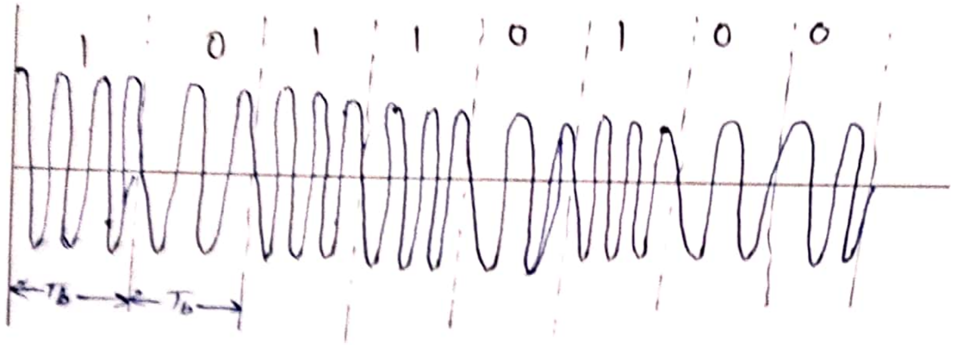
* If symbol '0' is to be transmitted, then the carrier freq^n. will be $f_c - \left(\frac{\Omega}{2\pi}\right)$ and is represented by f_L

Therefore, we have

Thus $f_H = f_c + \frac{\Omega}{2\pi}$ for symbol '1'

$f_L = f_c - \frac{\Omega}{2\pi}$ for symbol '0'

Waveform of FSK :-



Power Spectral density (PSd) of a BFSK Signal :-

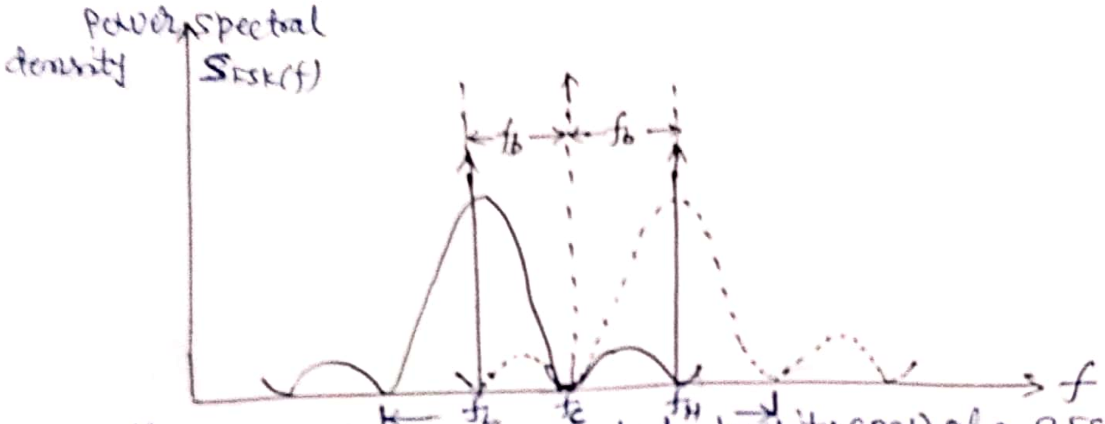


Fig. Power Spectral density (PSd) of a BFSK Signal.

BW = $4f_b$

Practically FSK Ckt dia.

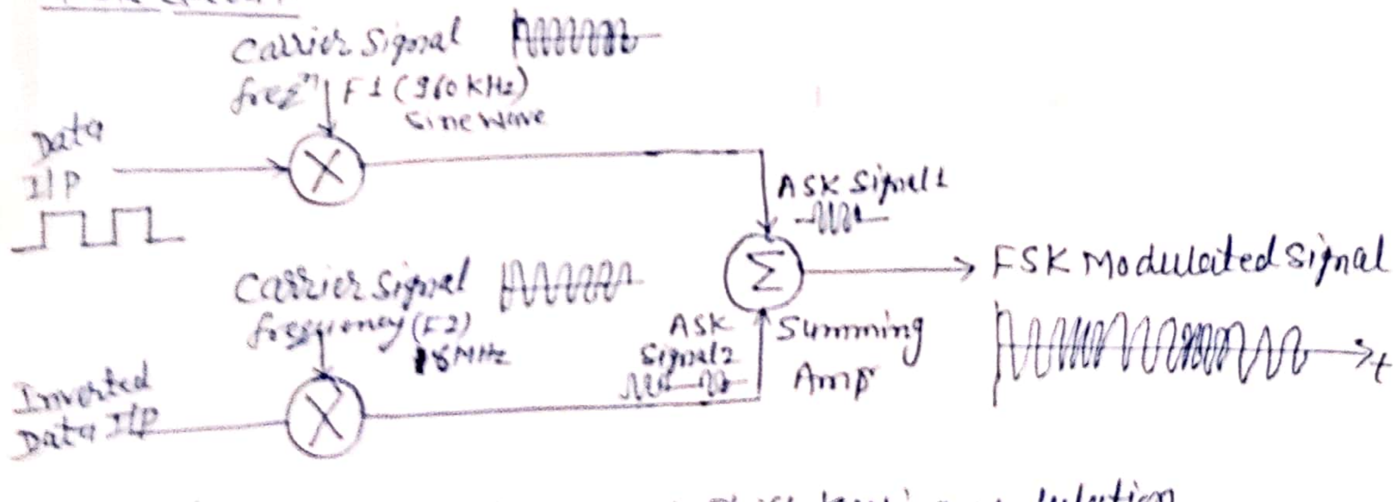
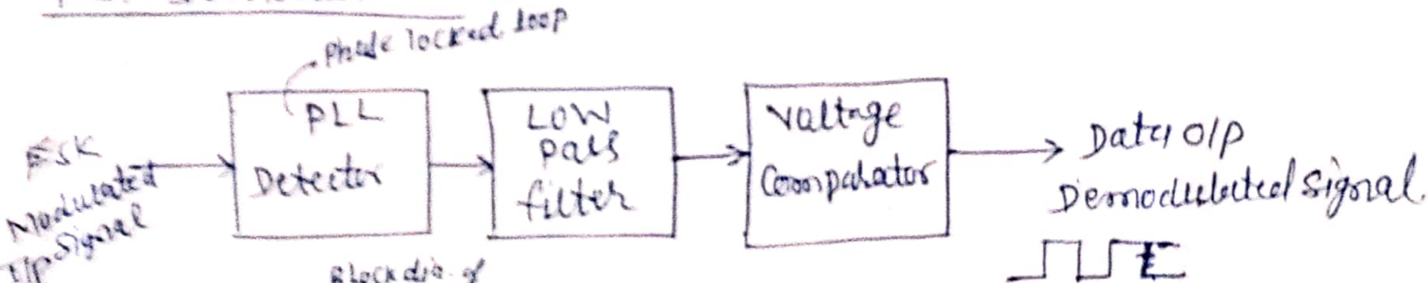


Fig :- Frequency Shift Keying Modulation.

FSK Demodulation :-



Block dia. of
Fig :- FSK Demodulation

Binary phase shift keying (BPSK) :→

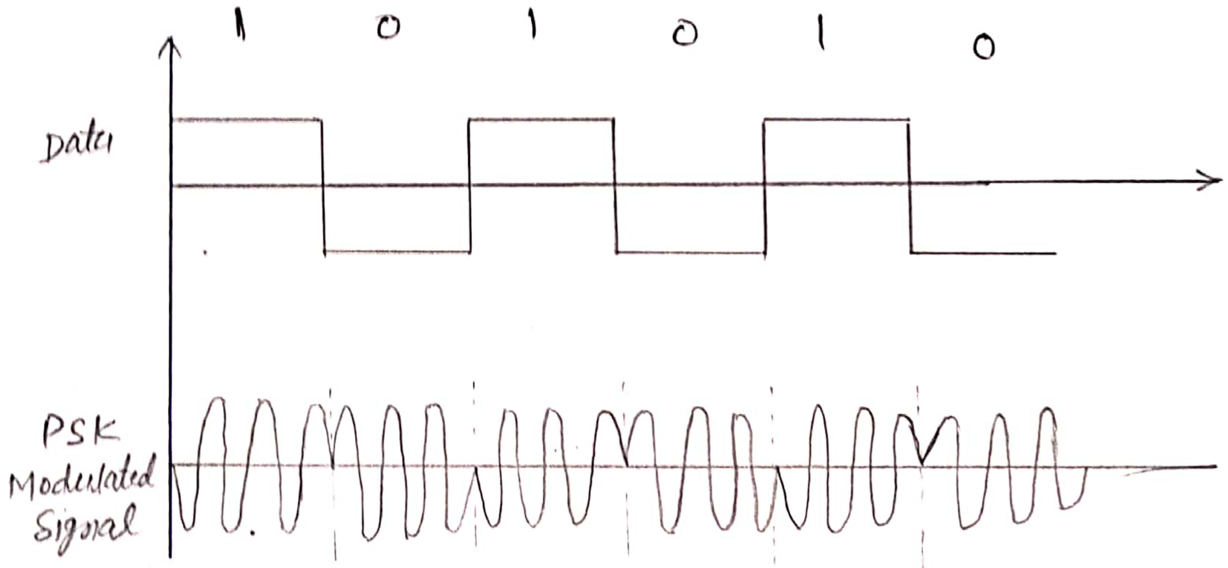
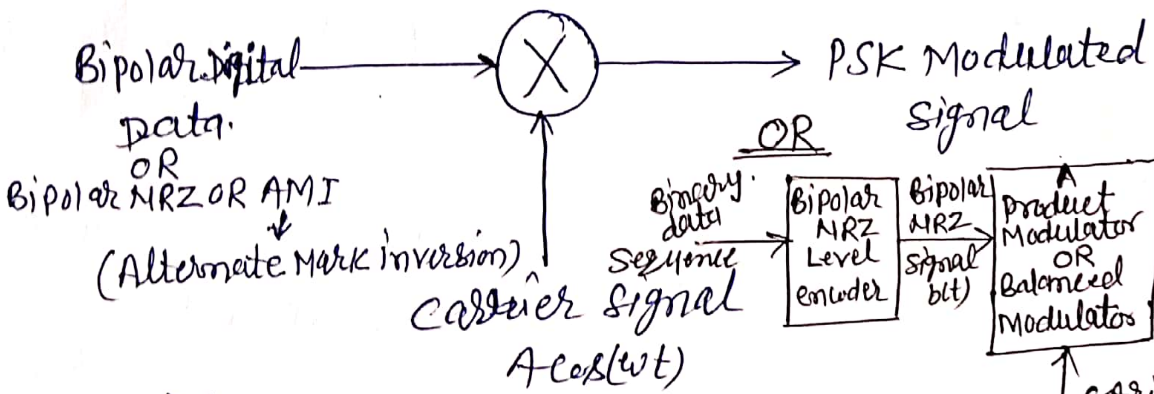


Fig.:- Phase shift keying waveform

The functional block representation of the PSK modulator—



OR

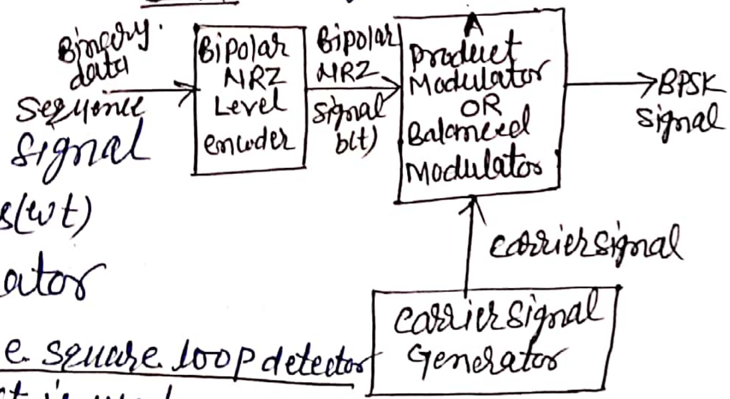
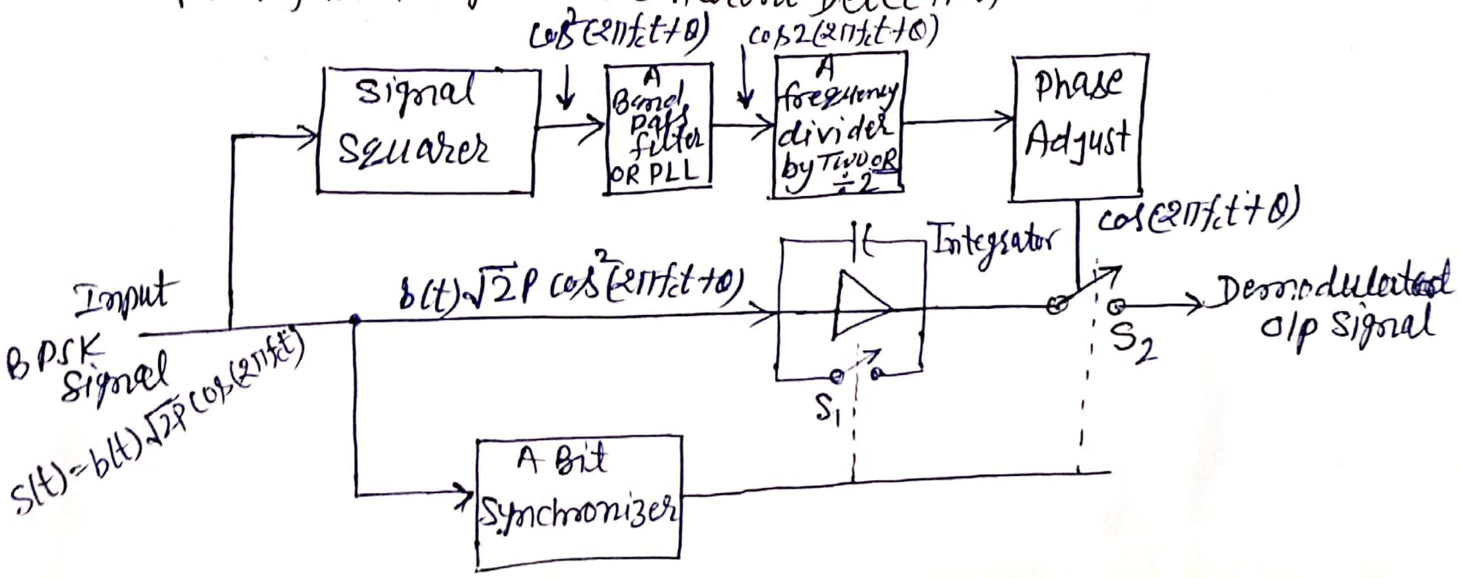
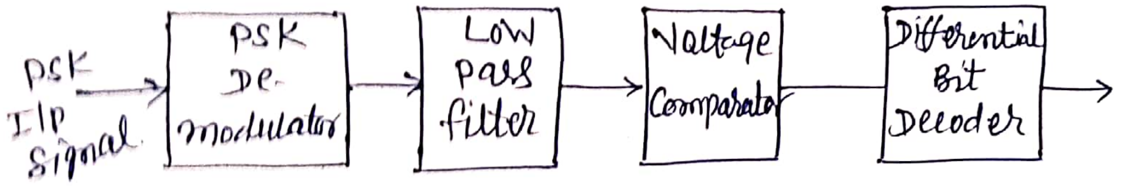


Fig.:- PSK Modulator

PSK Demodulation :→ the square loop detector ckt is used.
OR
Reception of BPSK signal OR coherent Detection



PSK Receiver system :→



Power spectral density (PSd) :→ of the NRZ waveform –

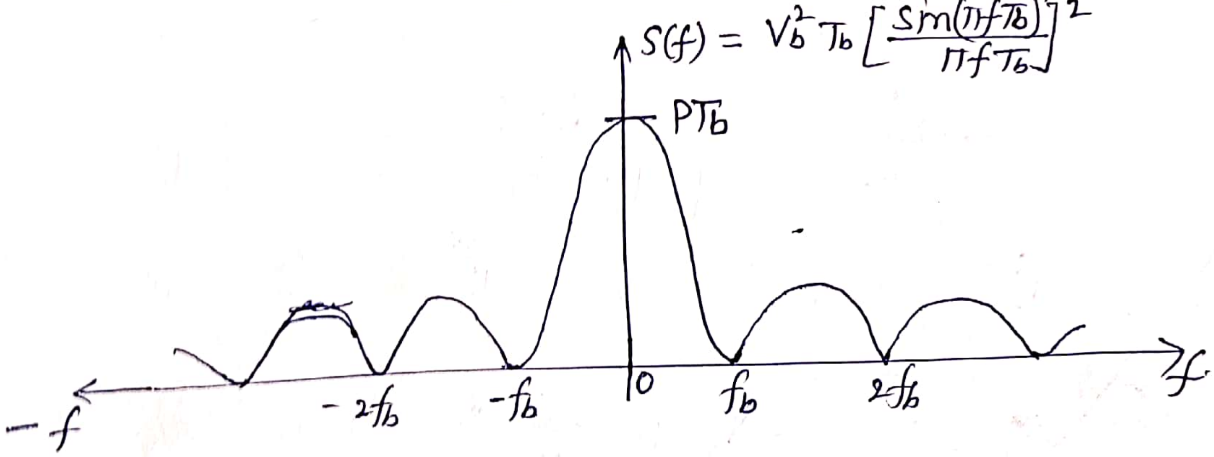


Fig.:- Power spectral density (PSd) of NRZ baseband signal.

The transmitted BPSK signal is given by $S(t) = b(t)\sqrt{2P}\cos(2\pi f_c t)$

~~Expression for BPSK~~ :→ This signal undergoes the phase change depending upon the time delay from transmitter end to receiver end. This phase change is, usually, a fixed phase shift in the transmitted signal.

Let us consider that this phase shift is θ. Because of this, the signal at the I/p of the receiver ~~can be~~ as

$$S(t) = b(t)\sqrt{2P}\cos(2\pi f_c t + \theta)$$

Now, from this received signal, a carrier is separated because this is coherent detection. The o/p of the signal squarer device is the signal $\cos^2(2\pi f_c t + \theta)$.

$\left\{ \because \cos^2\theta = \frac{1+\cos 2\theta}{2} \right.$

$$\Rightarrow \cos^2(2\pi f_c t + \theta) = \frac{1}{2} + \frac{1}{2}\cos 2(2\pi f_c t + \theta)$$

This signal is applied to a BPF whose passband is centered around $2f_c$. BPF removes the DC level of $\frac{1}{2}$ & at the o/p we get $\cos 2(2\pi f_c t + \theta)$.

BPSK :→ Binary Phase Shift keying is the most efficient of the three digital modulation i.e., ASK, FSK and PSK. Hence, binary phase shift keying is used for high bit rate. In BPSK, phase of the sinusoidal carrier is changed according to the data bit to be transmitted. Also, a bipolar NRZ signal is used to represent the digital data coming from the digital source.

Expression for BPSK :-

In a binary phase shift keying (BPSK), the binary symbols '1' and '0' modulate the phase of the carrier. Let us assume that the carrier is as

$$s(t) = A\cos(2\pi f_c t) \text{ - - - - - } (1) \quad \because \omega = 2\pi f_c$$

Here 'A' represents peak value of sinusoidal carrier. For $1\,\Omega$ load resistor, the power dissipated would be,

$$P = \frac{1}{2}A^2$$

or $$A = \sqrt{2P} \text{ - - - - - - - } (2)$$

Now, when the symbol is changed, then the phase of the carrier will also be changed by an amount of $180°$.

then $$\boxed{s(t) = \sqrt{2P}\cos(2\pi f_c t)}$$

For Example -

For Symbol '1', we have

$$\boxed{s_1(t) = \sqrt{2P}\cos(2\pi f_c t)}$$

~~If sent~~ For Symbol '0', we have

$$s_2(t) = \sqrt{2P}\cos(2\pi f_c t + \pi) \qquad \left\{ \because \cos(\theta+\pi) = -\cos\theta \right.$$

or $$\boxed{s_2(t) = -\sqrt{2P}\cos(2\pi f_c t)}$$

Generation of BPSK signal :→

BPSK signal may be generated by applying carrier signal to a balanced modulator. The Binary data signal (0s & 1s) is converted into a NRZ bipolar signal by an NRZ encoder. Here, the bipolar signal b(t) is applied as a modulating signal to the balanced modulator.

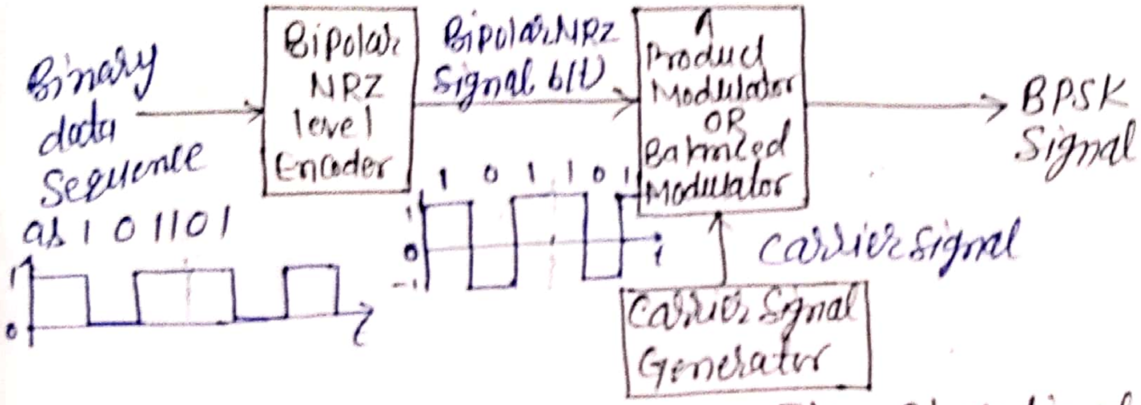


Fig. :- Block dia. of a BPSK Signal Gen^r.

* When I/p digital signal Binary 0, Then Bipolar NRZ signal b(t) is −1 and BPSK o/p signal is $-\sqrt{2P}\cos\omega t$
* When I/p digital signal Binary 1, Then Bipolar NRZ signal b(t) is +1 and BPSK o/p signal is $+\sqrt{2P}\cos\omega t$.

Now this signal is passed through a frequency divider by two. At the o/p of frequency divider, we get a carrier signal whose frequency is f_c i.e. $\cos(2\pi f_c t+\theta)$.
The o/p of the multiplier is $b(t)\sqrt{2P}\cos^2(2\pi f_c t+\theta)$.
This signal is then applied to the bit synchronizer and Integrator. The integrator integrates the signal over one bit period. The starting and ending times of a bit are watched upon by the bit synchronizer. At the end of bit duration T_b, the bit synchronizer closes switch S_2 temporarily. Thus the o/p of the integrator is connected to the decision device, so that the o/p of integrator is sampled.

The synchronizer then opens switch S_2 and S_1 is closed temporarily. Due to this the integrator voltage is reset to zero.
The integrator then integrates next bit. This means that the phase change occurs in the carrier only at zero crossing.

<u>Advantages of BPSK :-</u>

1) BPSK has lower bandwidth as compared to a BFSK signal
2) BPSK has the best performance of all the three digital modulation techniques in presence of noise.
3) BPSK has a very good noise immunity.

<u>Drawbacks of BPSK :-</u>

Figure shows the block dia. of BPSK receiver. To regenerate the carrier in the receiver. We start by squaring $b(t)\sqrt{2P}\cos(2\pi f_c t+\theta)$. If the received signal is [illegible] (cos ωt ...), then the squared signal remains same as before. Hence, the recovered carrier is unchanged even if the I/P signal has changed its sign.

Therefore, it is not possible to determine whether the received signal is equal to $b(t)$ or $-b(t)$. Infact, this results in ambiguity in the O/P signal.

Remark :- This problem can be removed if we use differential phase shift keying (DPSK)

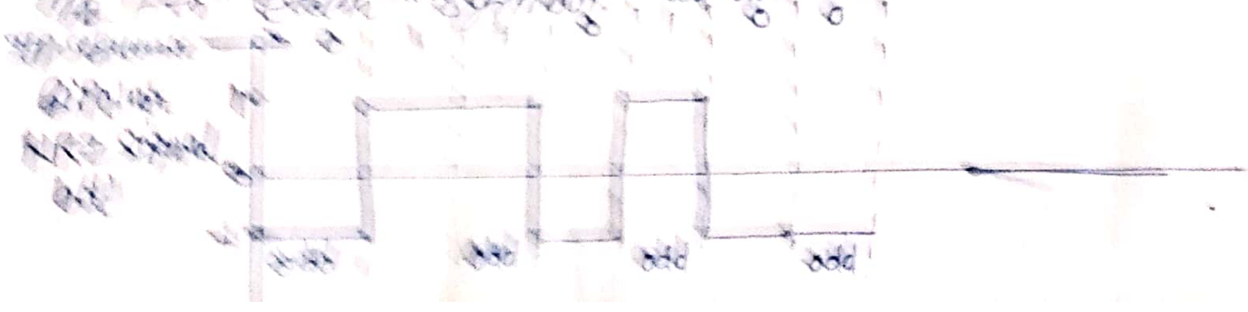
Quadrature phase shift keying (QPSK) :-

However, in QPSK, two successive bits are combined. Infact, this combination of two bits forms four distinct symbols. When the symbol is changed to next symbol, then the phase of the carrier is changed by 45° ($\pi/4$ radians).

Symbol and corresponding phase shifts in QPSK -

S.No.	I/P successive bits		Symbol	phase shift in carrier
1	1 (1V)	0 (-1V)	S_1	$\pi/4$
2	0 (-1V)	0 (-1V)	S_2	$3\pi/4$
3	0 (-1V)	1 (1V)	S_3	$5\pi/4$
4	1 (1V)	1 (1V)	S_4	$7\pi/4$

There are four symbols and phase is shifted by $\pi/4$ [illegible]



Generation of QPSK :-

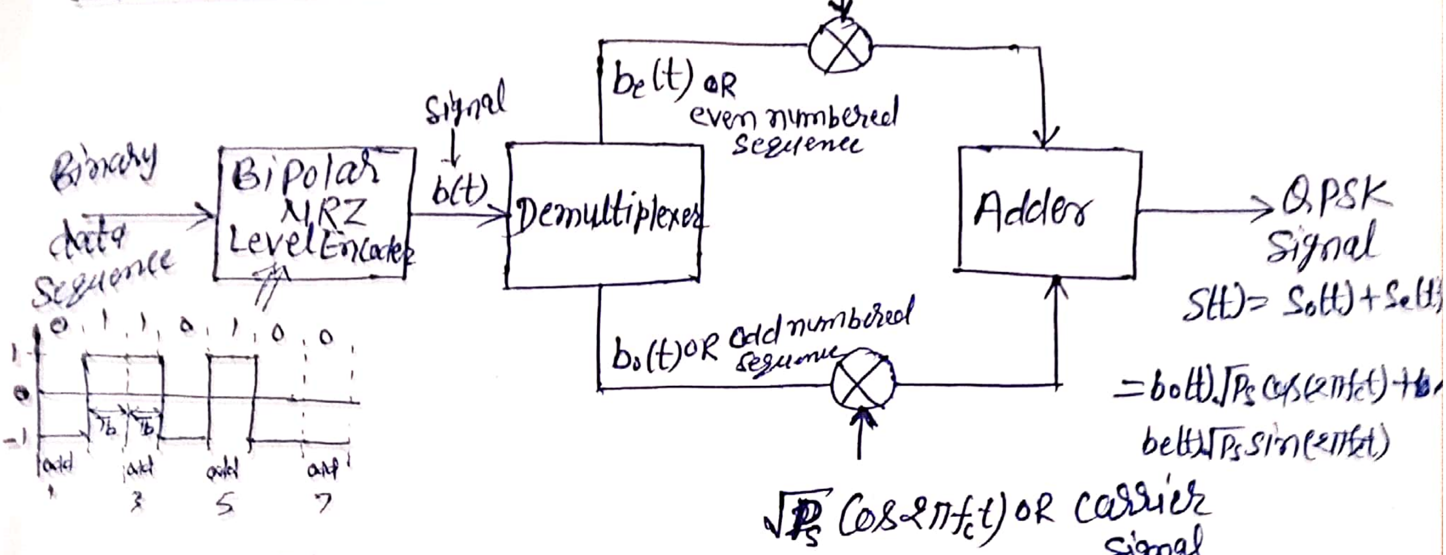
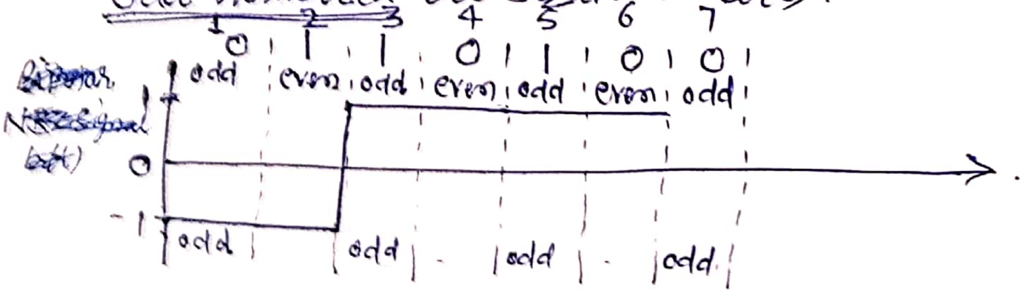


Fig. (a) Generation of QPSK

Here, the I/P binary sequence is 1st converted to a bipolar NRZ type of signal. This signal is denoted by b(t). It represents binary '1' by +1V and binary '0' by −1V. The demultiplexer divides b(t) into two seperate bit streams of the odd numbered $(b_o(t))$ and even numbered bits $(b_e(t))$. The symbol duration of both of these odd & even numbered sequences is $2T_b$.

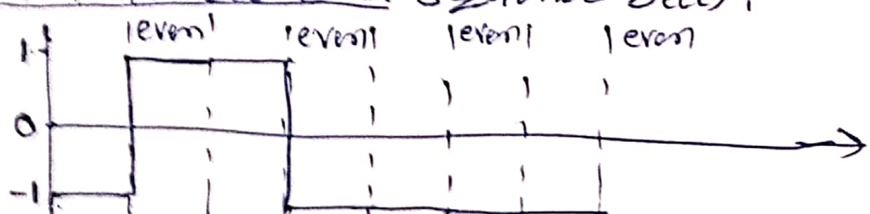
Hence, each symbol consists of two bits.

Odd numbered bit sequence $b_o(t)$:→

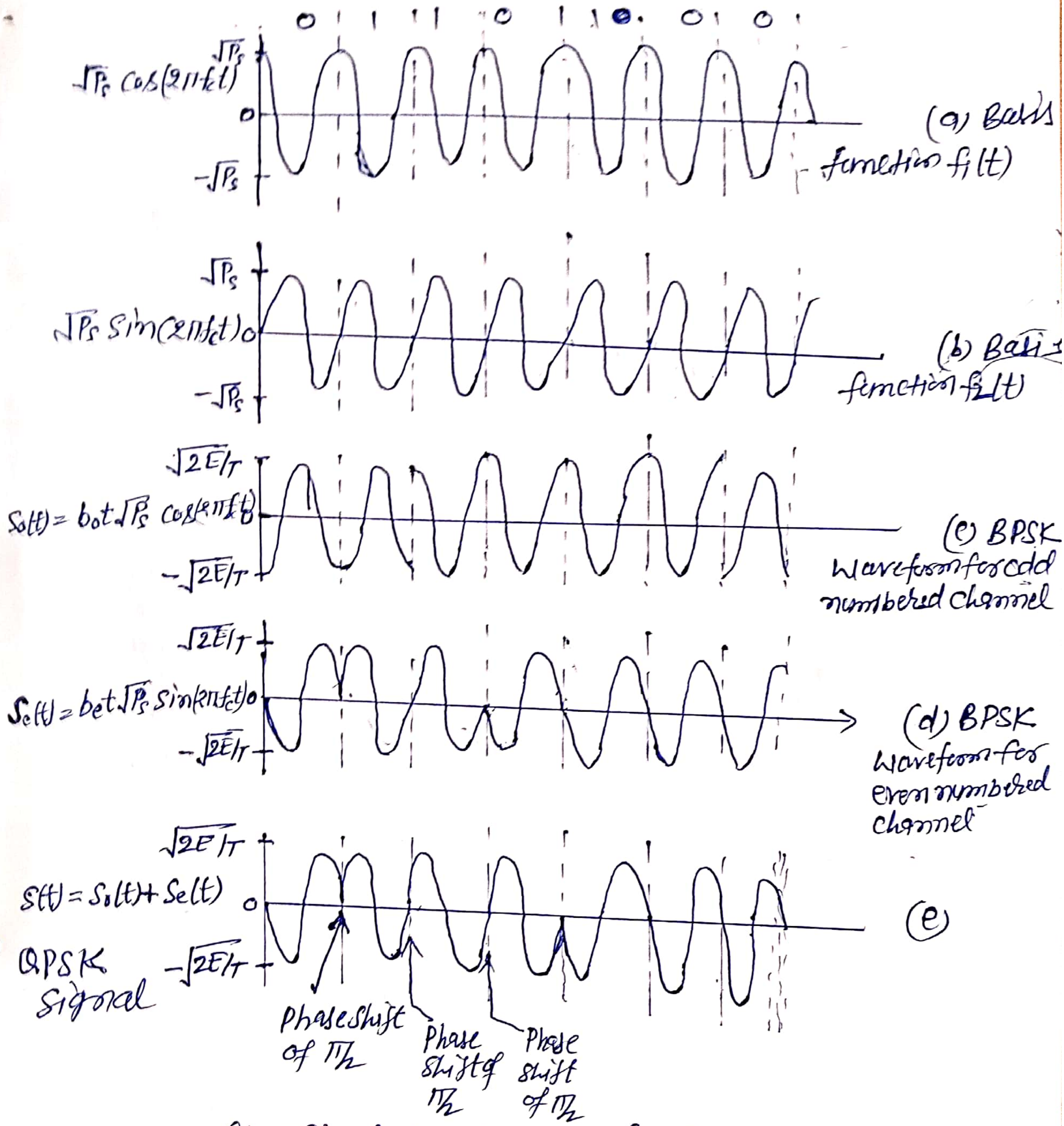


odd number =
0+1 = 1
1+0 = 1
1+0 = 1

Even numbered bit sequence $b_e(t)$:-



even number =
1+1 = 2



fig(e): final QPSK waveform

The o/p of the adder is QPSK signal as $S(t) = \underbrace{S_o(t)}_{\text{odd numbered sequence}} + \underbrace{S_e(t)}_{\text{even numbered sequence}}$

OR $S(t) = b_o(t)\sqrt{P_s}\cos(2\pi f_c t) + b_e(t)\sqrt{P_s}\sin(2\pi f_c t)$

Reception of QPSK (OR. Detection of QPSK) :→

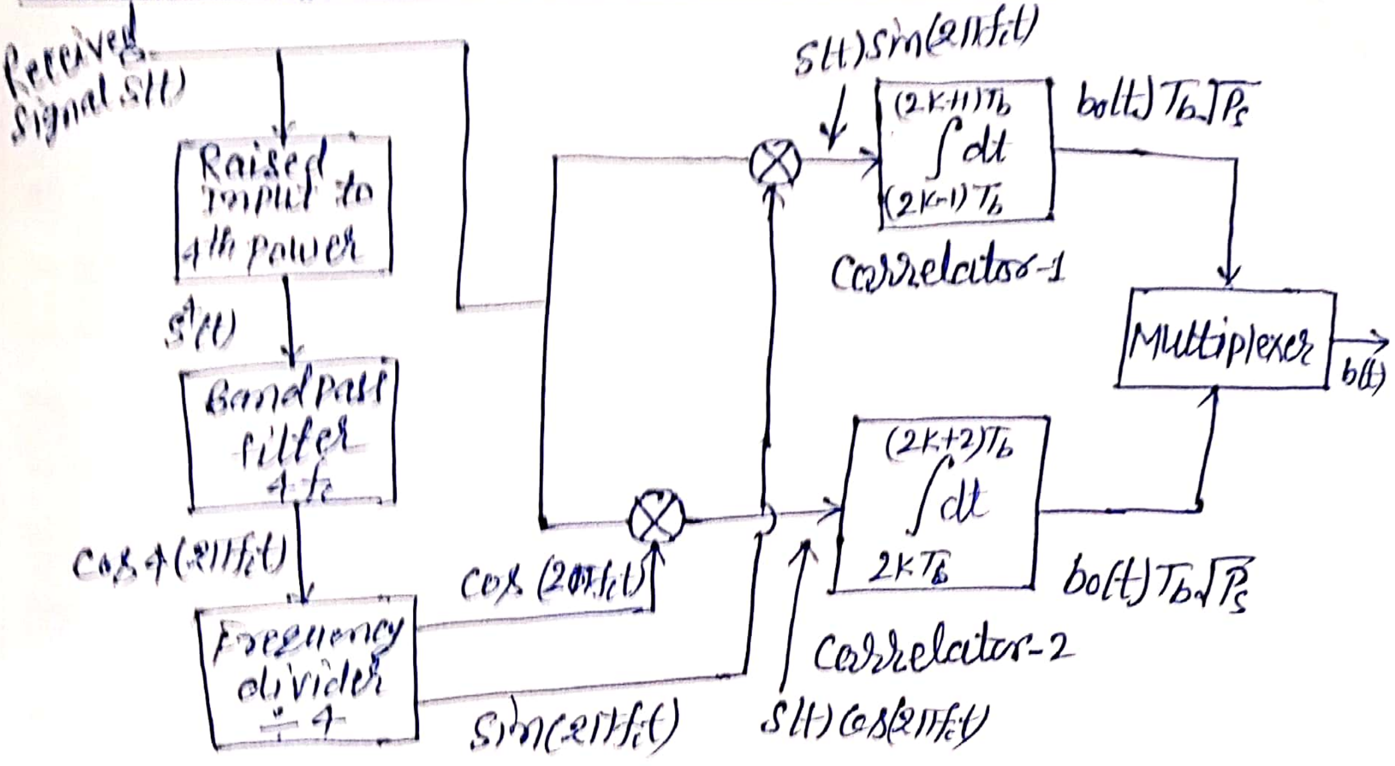


Fig. :- Reception of QPSK

BANDWidth of QPSK Signal :→

We have observed that the band width of BPSK signal is equal of $2f_b$. Here $T_b = \frac{1}{f_b}$ is the one bit period. In QPSK, the two waveforms $b_e(t)$ and $b_o(t)$ form the base band signals. one bit period for both of these signals is equal to $2T_b$. Therefore, band width of QPSK signal will be

$$BW = 2 \times \frac{1}{2T_b} = f_b$$

Hence, the bandwidth of QPSK signal is half of the bandwidth of BPSK signal.

Advantages of QPSK :→

1) For the same bit error rate, the bandwidth required by QPSK is reduced to half as compared to BPSK
2) Because of reduced bandwidth, the information transmission rate of QPSK is higher.

Minimimum Shift Keying (MSK) :→ Gen of MSK —

Transmitter —

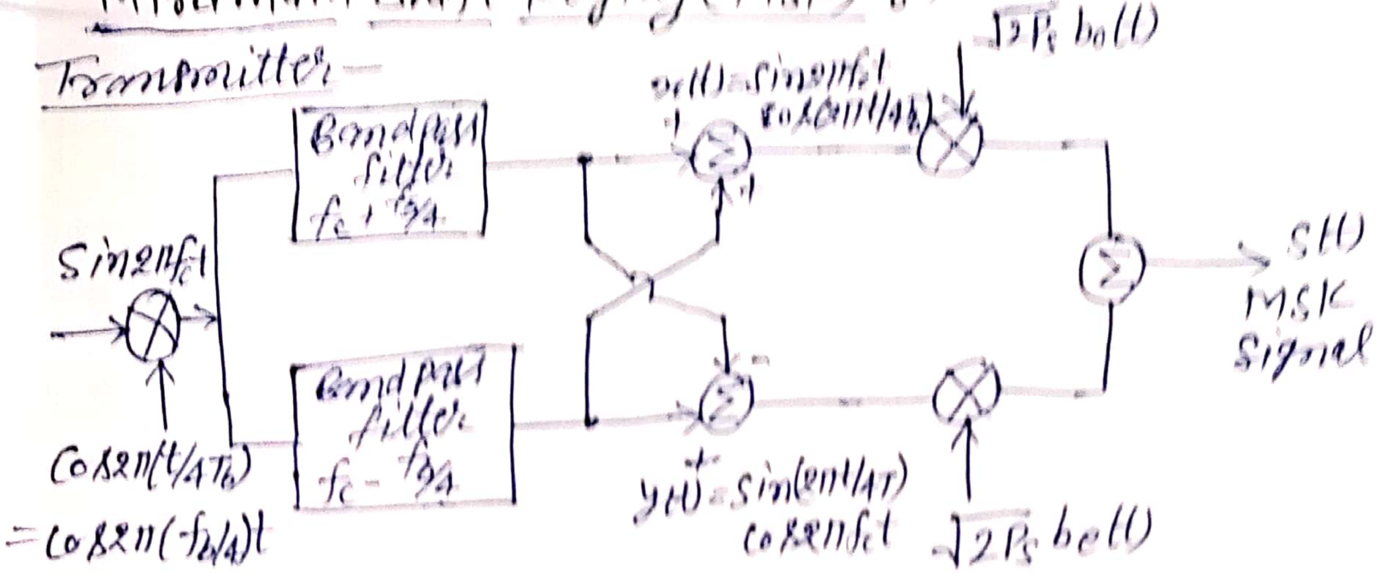


Fig.(a) BLOCK diagram of MSK Transmitter

MSK Receiver :→

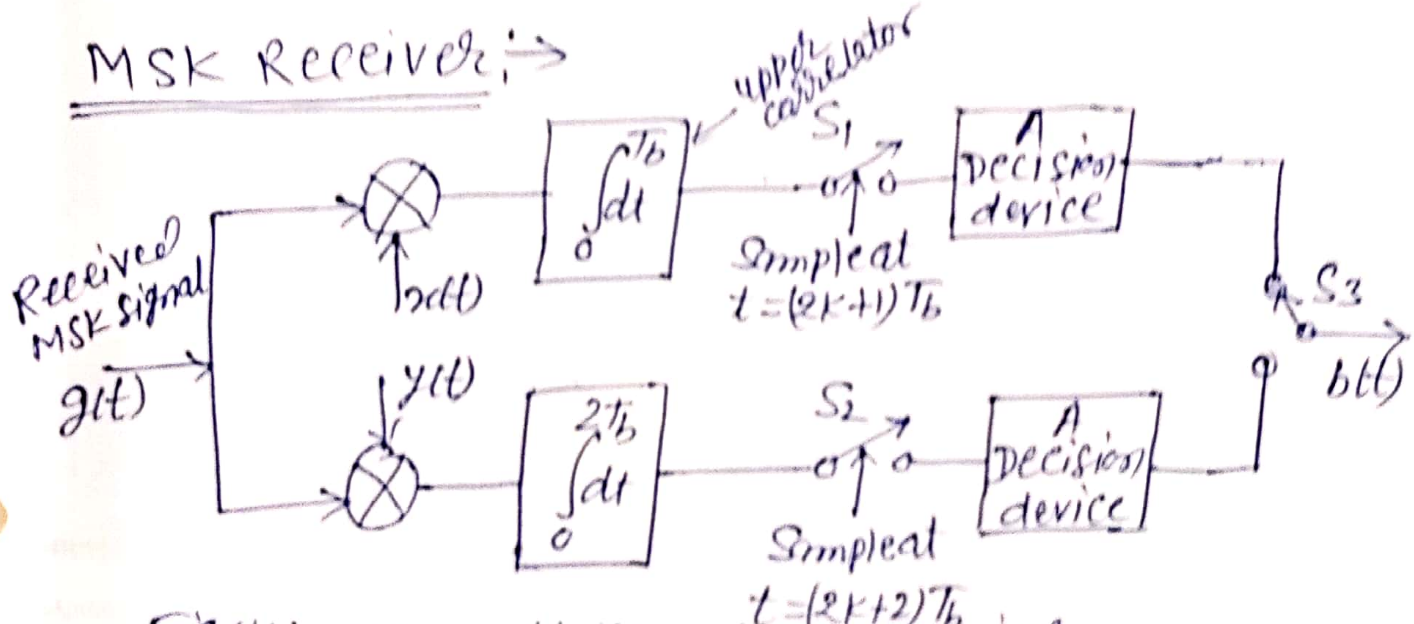


Fig.(b):- BLOCK diagram of MSK Receiver

Generation of MSK — The two sinusoidal signals $\sin(2\pi f_c t)$ and $\cos(2\pi t/4T_b)$ are mixed (i.e. multiplied). The Bandpass filters then pass only **sum** and difference components $f_c+\frac{f_b}{4}$ and $f_c-\frac{f_b}{4}$. The outputs of bandpass filters (BPFs) are then added and subtracted such that two signals $x(t)$ and $y(t)$ are generated. Signal $x(t)$ is multiplied by $\sqrt{2P_s}\,b_o(t)$ and $y(t)$ is multiplied by $\sqrt{2P_s}\,b_e(t)$. The outputs of the multipliers are then added to give final MSK signal

Advantages of MSK as compared to QPSK—

1) The MSK base band waveforms are smoother compared to QPSK

2) MSK signal have continuous phase in all the cases, whereas QPSK has abrupt phase shift of $\pi/2$ or π

3) MSK waveform does not have amplitude variations, whereas QPSK signals have abrupt amplitude variations.

4) The main lobes of MSK is wider than that of QPSK. Main lobe of MSK contains around 99% of signal energy whereas QPSK main lobe contains around 90% signal energy.

5) Side lobes of MSK are smaller compared to that of QPSK. Hence, inter channel interference because of side lobes is significantly large in QPSK.

6) To avoid inter channel interference due to side lobes, QPSK needs bandpass filtering, whereas it is not required in MSK.

8) Bandpass filtering changes the amplitude waveform of QPSK because of abrupt changes in phase. This problem does not exist in MSK.

Drawbacks of MSK as compared to QPSK—

1) The bandwidth requirement of MSK is $1.5 f_b$, whereas it is f_b in QPSK. Actually, this cannot be said serious drawback of MSK. Because power to bandwidth ratio of MSK is more. Infact, 99% of signal power can be transmitted within the bandwidth of $1.2 f_b$ in MSK. While QPSK need around $8 f_b$ to transmit the same power.

2) The generation and detection of MSK is slightly complex. Because of incorrect synchronization, phase shifter can be present in MSK. This degrades the performance of MSK

Reception of MSK OR Detection of MSK –

MSK uses synchronous detection.

The signals $x(t)$ and $y(t)$ are multiplied with the Received MSK Signal. Here $x(t)$ and $y(t)$ have same values as shown in tx. block dia. ∴ The output of the multipliers are $b_e(t)$ and $b_o(t)$. The integrators integrate over the period of $2T_b$. For the upper correlator, the sampling switch samples output of integrator at $t=(2k+1)T_b$. For the upper correlator, the sampling switch samples output of integrator at $t=(2k+1)T_b$. Then the decision device decides whether $b_o(t)$ is +1 or −1. Similarly, lower correlator output is $b_e(t)$. The output of two decision devices are staggered by T_b. The switch S_3 operates at $t = kT_b$ and simply multiplexes the two correlator outputs.

Wave form of MSK –

NRZ Data →

MSK wave form →