EC8395 COMMUNICATION ENGINEERING **L T P C**
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OBJECTIVES:

- To introduce the relevance of this course to the existing technology through demonstrations, case studies, simulations, contributions of scientist, national/international policies with a futuristic vision along with socio-economic impact and issues
- To study the various analog and digital modulation techniques
- To study the principles behind information theory and coding
- To study the various digital communication techniques

UNIT I ANALOG MODULATION 9

Amplitude Modulation – AM, DSBSC, SSBSC, VSB – PSD, modulators and demodulators – Angle modulation – PM and FM – PSD, modulators and demodulators – Superheterodyne receivers

UNITII PULSE MODULATION 9

Low pass sampling theorem – Quantization – PAM – Line coding – PCM, DPCM, DM, and ADPCM And ADM, Channel Vocoder - Time Division Multiplexing, Frequency Division Multiplexing

UNIT III DIGITAL MODULATION AND TRANSMISSION 9

Phase shift keying – BPSK, DPSK, QPSK – Principles of M-ary signaling M-ary PSK & QAM – Comparison, ISI – Pulse shaping – Duo binary encoding – Cosine filters – Eye pattern, equalizers

UNIT IV INFORMATION THEORY AND CODING 9

Measure of information – Entropy – Source coding theorem – Shannon–Fano coding, Huffman Coding, LZ Coding – Channel capacity – Shannon-Hartley law – Shannon's limit – Error control codes – Cyclic codes, Syndrome calculation – Convolution Coding, Sequential and Viterbi decoding

UNIT V SPREAD SPECTRUM AND MULTIPLE ACCESS 9

PN sequences – properties – m-sequence – DSSS – Processing gain, Jamming – FHSS – Synchronisation and tracking – Multiple Access – FDMA, TDMA, CDMA

TOTAL: 45 PERIODS

OUTCOMES:
At the end of the course, the student should be able to:

- Ability to comprehend and appreciate the significance and role of this course in the present contemporary world
- Apply analog and digital communication techniques.
- Use data and pulse communication techniques.
- Analyze Source and Error control coding.

TEXT BOOKS:

1. H Taub, D L Schilling, G Saha, "Principles of Communication Systems" 3/e, TMH 2007
2. S.Haykin "Digital Communications" John Wiley 2005

REFERENCES:

1. B.P.Lathi, "Modern Digital and Analog Communication Systems", 3rd edition, Oxford University Press, 2007
2. H P Hsu, Schaum Outline Series – "Analog and Digital Communications" TMH 2006
3. B.Sklar, Digital Communications Fundamentals and Applications" 2/e Pearson Education 2007.

EC 6392 - COMMUNICATION ENGINEERING.

UNIT-I. ANALOG MODULATION

Introduction:

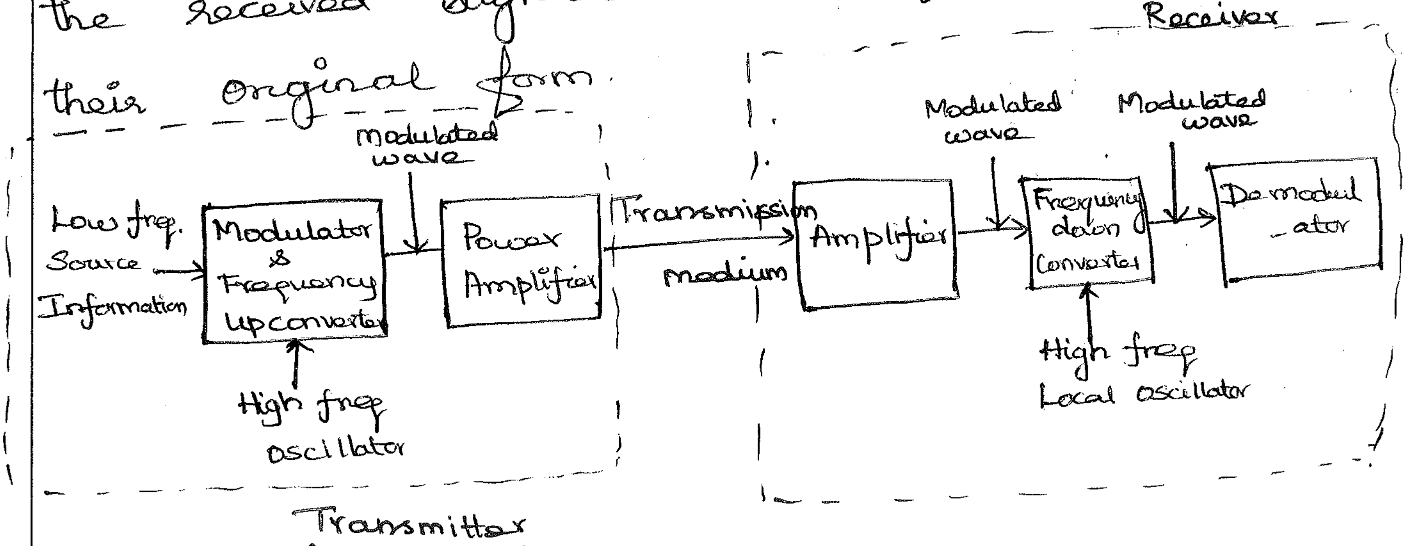
* Information signals are transported b/w a transmitter and a receiver over some form of Transmission medium
* Information signals must be transformed into a form that is more suitable for transmission.

Modulation:

The process of impressing low frequency information signals onto a high frequency carrier signal is called modulation.

Demodulation:

Demodulation is the reverse process where the received signals are transformed back to their original form.



Necessary for modulation

* It is difficult to radiate low frequency signals from an antenna in the form of em energy.
* Information signals often occupy the same frequency band, they interfer with each other.

audio freq = 5KHz.

1. Reduction in antenna height → $L = \lambda/2 = \frac{c}{2f} = \frac{3\times10^8}{2\times5\times10^3} = 30km$ (impossible)

2. Long distance communication.

3. Ease of radiation → easy to design amplifier ckts & Antennas at high freq.

4. Multiplexing → Diff signals can be transmitted over a same channel without interference.

5. Improve the quality of Reception → affect of Noise can be eliminated with the help of modulation Technique

6. Avoid mixing of signals → Each signal is modulated with different carrier frequency, then they will occupy diff. slots in the freq. domain.

Classification of Modulation:

- Modulation
  - Analog Modulation.
    - Continuous modulation
      - Amplitude Modulation
      - Angle modulation
        - Freq modulation
        - Phase modulation
    - Pulse Analog Modulation
      - PAM
      - PPM
      - PWM
  - Digital Modulation
    - Pulse digital modulation
      - PCM
      - DM
      - ADM
      - DPCM
    - Shift keying methods.
      - ASK
      - FSK
      - PSK
      - QAM

General Terms:

Analog Signals: The amplitude changes continuously w.r.t time. eg. sine wave.

Digital Signals: The amplitude constant level for the prescribed period of time and then it changes to another level. eg. binary signal.

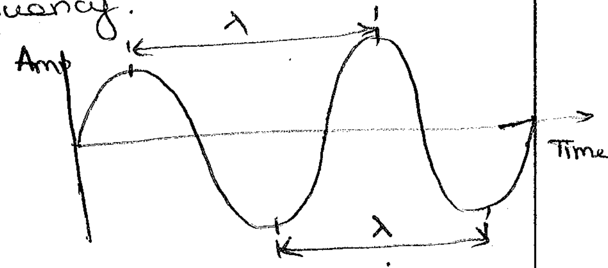
* All binary signals are digital. But all digital signals are not binary.

Frequency: Frequency is simply a number of times a periodic motion occurs in a given periodic time.
* Each complete alteration of wave is called Cycle.

Unit: (Hz) Hertz. $f = \frac{1}{T}$.

Wavelength: Wavelength is the length that one cycle of an electromagnetic wave occupies in space (ie distance b/w similar points in a repetitive wave).

$$\text{Wavelength} = \frac{\text{Velocity}}{\text{frequency}}$$



λ – wavelength
c – Speed of light $= (3 \times 10^8 \text{ m/s})$
f – frequency.

$$\lambda = \frac{c}{f}$$

Sinusoidal Signal:

Electrical signals are voltage or current time variations that can be represented by a series of sine or cosine waves.

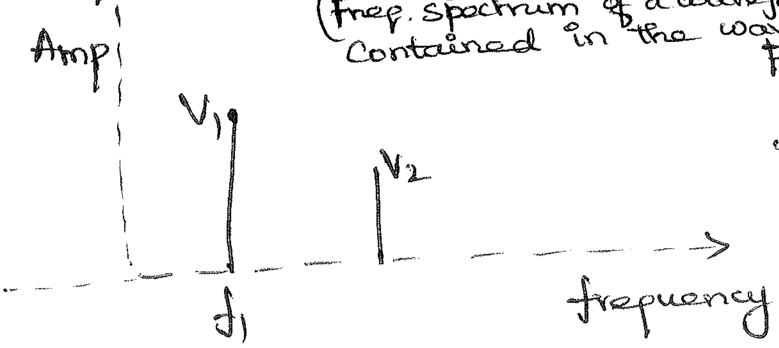
$$v(t) = V\sin(2\pi ft + \theta) \text{ (or) } v(t) = V\cos(2\pi ft + \theta)$$
$$i(t) = I\sin(2\pi ft + \theta) \text{ or } i(t) = I\cos(2\pi ft + \theta).$$

Where v(t)/i(t) – Time varying voltage / current sine wave.
V – Peak voltage (Volts).
f – frequency (hertz).
θ – phase shift (radians)
I – peak current (amperes).
2πf – angular velocity (radians per second).

Time domain Representation: (Signal waveform).
* A description of a signal w.r.t time is called a time domain representation.
* Instrument used is A standard oscilloscope, CRT.
* Signal waveform shows the shape and instantaneous magnitude of the signal w.r.t time.

Frequency Domain Representation:

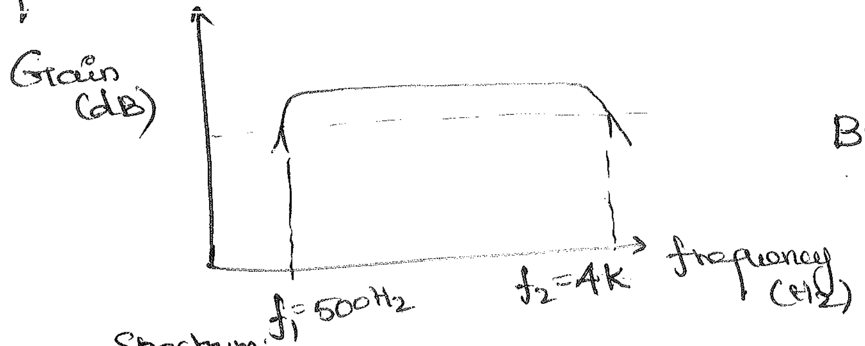
* A description of a signal w.r.t its frequency is called frequency domain representation.
* Instruments used : Spectrum analyzer.
* Amplitude versus frequency plot is shown in Spectrum analyzer. (known as frequency Spectrum).
(Freq. spectrum of a waveform consists of all the frequencies contained in the waveform and their respective amplitudes. Freq. Spectrum of plotted in freq domain)



2 Sine Signals.

Bandwidth:

* Bandwidth is defined as the range of frequency overwhich the information signal is transmitted.
* It is the difference between highest and lowest frequencies contained in the information signal.



$BW = f_2 - f_1$

$= 4K - 500Hz$

$= 3500Hz.$

Frequency Spectrum:

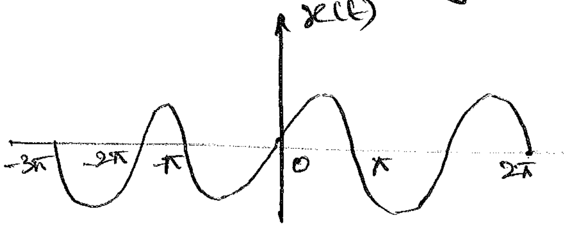
S.No	Frequency Range	Name	Application
1.	3K - 30K	VLF	Submarine Commn.
2.	30K - 300K	LF	Marine & aeronautical navigation
3	0.3M - 3M	MF	Commercial AM broadcasting
4.	3M - 30M	HF	Amateur Radio.
5.	30M - 300M	VHF	Mobile radio, Commercial FM.
6.	300M - 3G	UHF	Cellular Telephone, Microwave. Satellite radio systems.
7.	3G - 30G	SHF	Microwave, Satellite Radio Systems
8.	30G - 300G	EHF	Radio Communications.

Signal:

It is a physical quantity that varies with time, Space or any other independent Variable Eg. Radio signal, TV Signal, Computer signal etc.

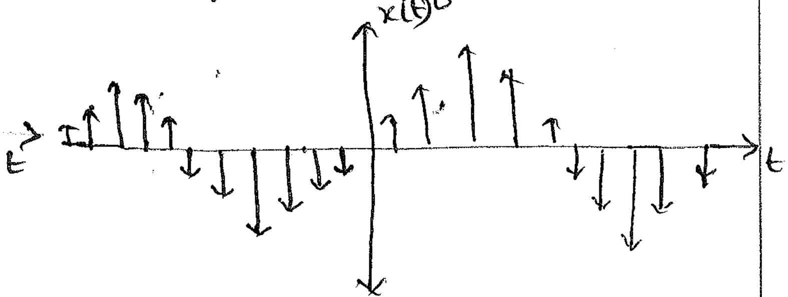
Continuous Signal.

A signal of continuous amplitude and time is known as a Continuous time signal.



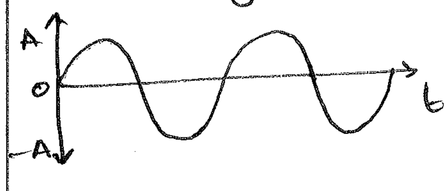
Discrete Signal.

A signal is said to be discrete if it exists at discrete interval of time.



Analog Signal.

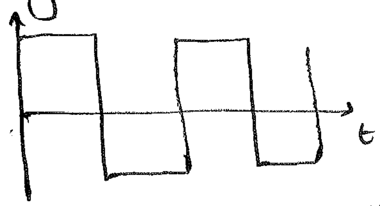
* Continuous signal in both amplitude and time is known as analog signal.



* The amplitude changes continuously w.r.t time
eg. Sine wave.

Digital Signal.

* Digital signal is a discrete time signal or continuous time waveform signal.

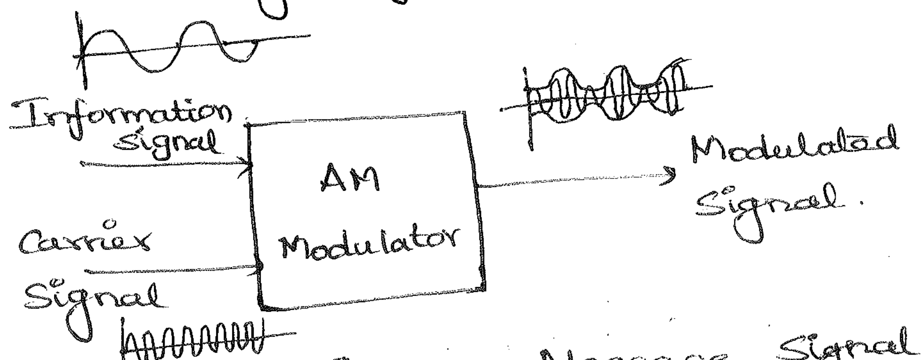


* The amplitude maintains constant level for the prescribed period of time and then it changes to another level.

eg. binary signal.

Amplitude Modulation:

Def: Amplitude modulation is the process by which the amplitude of the carrier signal is varied in accordance with the instantaneous value (amplitude) of the modulating signal. Here phase and frequency maintains constant.



* Information signal, Message signal or modulating signal: is comprised of low frequency information signal that may be single frequency or a complex waveform made up of many frequencies.
* Carrier input is a single high frequency signal having constant amplitude.
* The modulated output waveform from an AM modulator is called an AM envelope.

Mathematical represention of AM

Let un-modulated carrier can be described as $V_c(t) = E_c \sin(2\pi f_c t)$.

where $V_c(t)$ - time varying voltage waveform of carrier.

E_c - Peak carrier amplitude (Volts).

f_c - carrier frequency (Hz).

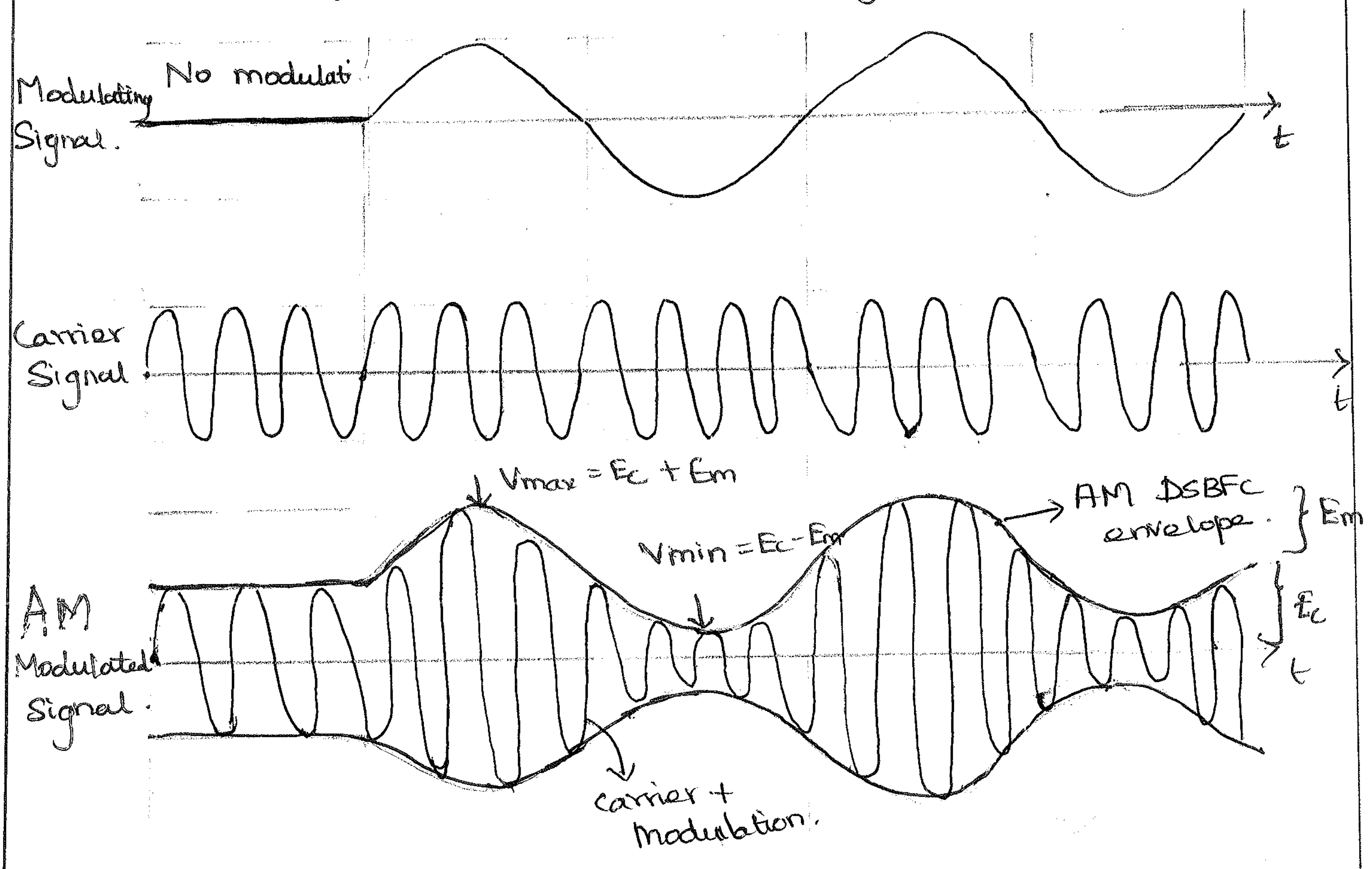
The modulating signal can be describe as

$$V_m(t) = E_m \sin(2\pi f_m t)$$

where $V_m(t)$ - time varying waveform of modulating signal.

E_m = Peak modulating signal.
f_m - modulating frequency (Hz).

* The modulated wave is $V_{am}(t)$
* An AM waveform is produced when a single frequency modulating signal acts on a high frequency carrier signal.
* The O/P waveform contains all the frequencies that make up the AM signal and is used to transport the information through the system.
* The shape of the modulated wave is called the AM envelope.
* **Note:** The repetition rate of the envelope is equal to the frequency of the modulating signal and the shape of the envelope is identical to the shape of the modulating signal.



The AM envelope used most commonly is known as DSBFC (double side band Full Carrier).

* The maximum amplitude of the modulated wave is equal to $E_c + E_m$.

* The instantaneous amplitude of the modulated wave can be expressed as.

$$V_{am}(t) = E_{am} \sin 2\pi f_c t.$$

$$E_{am} = E_c + E_m \sin 2\pi f_m t.$$

$$\therefore V_{am}(t) = (E_c + E_m \sin 2\pi f_m t) \sin 2\pi f_c t \quad \text{--- (1)}$$

Modulation Index and Percentage Modulation

or

Modulation Coefficient:

* It is used to describe the amount of amplitude change (modulation) present in an AM wave.

Def: Modulation coefficient is defined as the ratio of maximum amplitude of the modulating signal (E_m) to the maximum amplitude of the carrier signal (E_c)

$$m = \frac{E_m}{E_c}. \quad ; m \text{- modulation co-efficient (unitless).}$$

(1) $\Rightarrow$

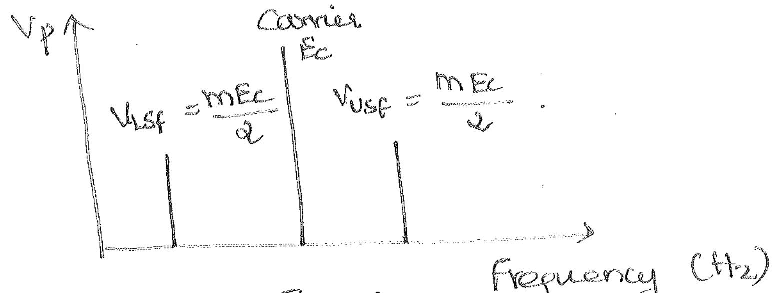
$$V_{am}(t) = E_c \left[1 + \frac{E_m}{E_c} \sin 2\pi f_m t\right) \sin 2\pi f_c t.$$

$$V_{am}(t) = \underbrace{(1 + m \sin 2\pi f_m t)}_{\text{Constant + modulating signal}} \underbrace{E_c \sin 2\pi f_c t}_{\text{unmodulated Carrier}}.$$

Rearranging,

$$V_{am}(t) = E_c \sin 2\pi f_c t + m E_c \sin 2\pi f_m t \sin 2\pi f_c t.$$

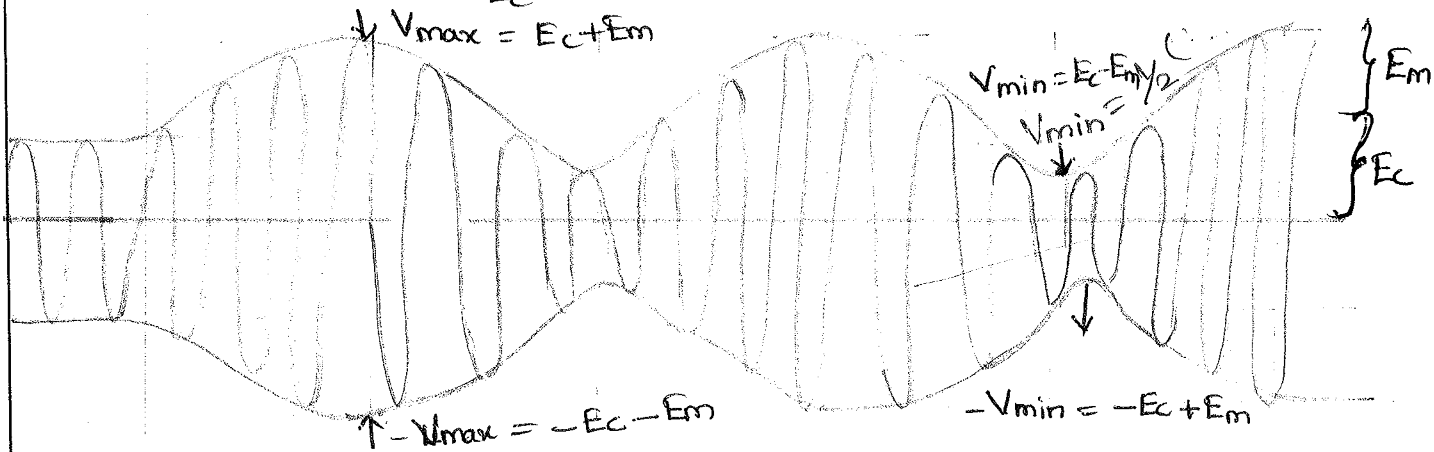
$$V_{am}(t) = \underbrace{E_c \sin 2\pi f_c t}_{\text{Carrier signal}} - \underbrace{\frac{mE_c}{2} \cos[2\pi(f_c + f_m)t]}_{\text{(upperside freq. signal)}} + \underbrace{\frac{mE_c}{2} \cos[2\pi(f_c - f_m)t]}_{\text{lower side freq. signal}} \quad \text{--- (2)}$$



Voltage Spectrum of AM

Percent Modulation: (M).

* $M = \frac{E_m}{E_c} \times 100$ or $m \times 100$.



$$E_{am} = E_c + E_m \sin 2\pi f_m t.$$

$$V_{max} = E_c + E_m \quad \text{——(1)} \quad \& \quad -V_{max} = -E_c - E_m$$

$$V_{min} = E_c - E_m \quad \text{——(2)} \quad \& \quad -V_{min} = -E_c + E_m$$

(1) - (2) ⇒ $V_{max} - V_{min} = 2E_m$.

$$E_m = \frac{1}{2}(V_{max} - V_{min}) \quad \text{——(3)}$$

Sub E_m in (1), $V_{max} = E_c + \left[\frac{V_{max} - V_{min}}{2}\right]$.

$$E_c = V_{max} - \frac{V_{max} + V_{min}}{2}$$

$$E_c = \frac{V_{max} + V_{min}}{2} \quad \text{——(4)}.$$

w.k, $m = \frac{E_m}{E_c} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$

$$M = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

For 100% modulation,

$$E_m = E_c$$

$\therefore$ ① $\Rightarrow$ $V_{max} = 2E_c$ or $V_{max} = 2E_m$.

② $\Rightarrow$ $V_{min} = 0$.

Degree of Modulation:

(i) Critical modulation or 100% Modulation.

When $E_m = E_c$, $V_{max} = 2E_m$, $V_{min} = 0$. $m = 1$

* The message or modulating signal is preserved and can be recovered from the envelope without any distortion.

$V_{max} = 2E_c$

$V_{min} = 0V$

E_m

E_c

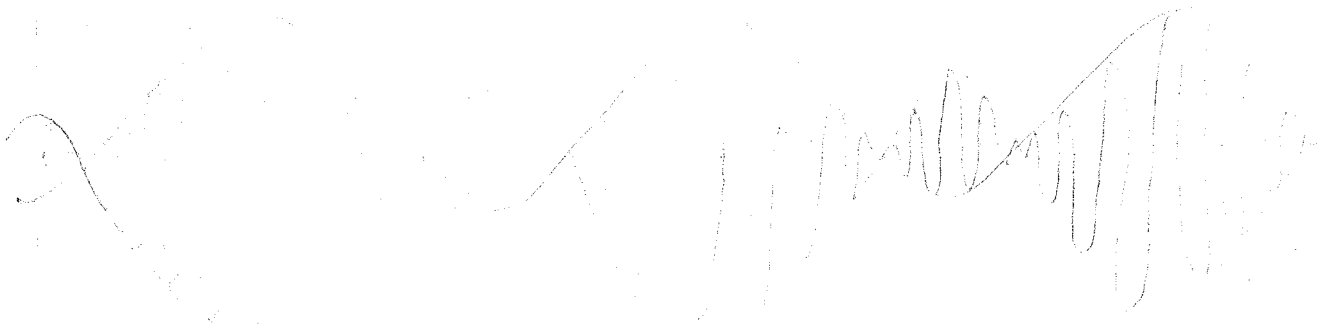
(ii) Under modulation:

When $E_m < E_c$, $m < 1$

* The message signal is fully preserved in the AM envelope.

— Refer Previous page.

iii) Over modulation:



* When $E_m > E_c$, $m > 1$

The message signal is not fully preserved in the AM Envelope.

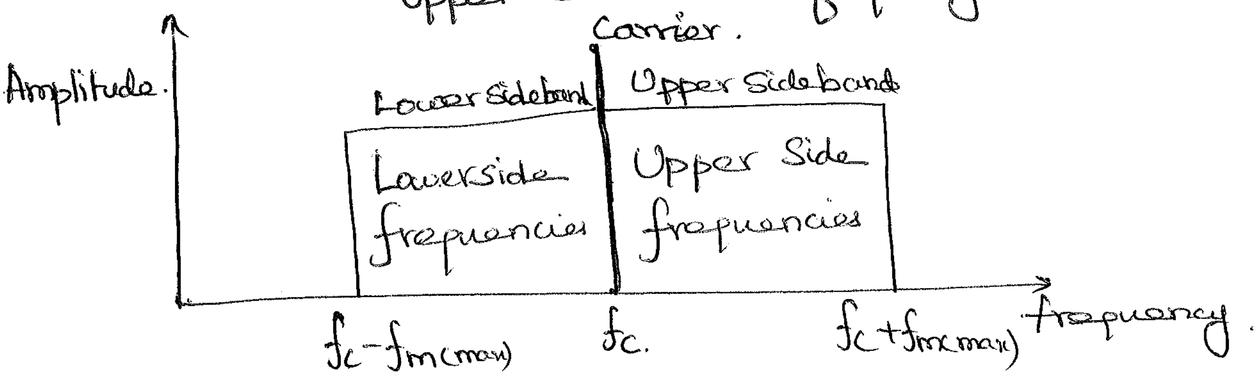
* Distorted signal occurs at the output of the envelope.

AM Frequency Spectrum and Bandwidth.

From eqn (2) Carrier frequency $= f_c$.

Lower Sideband frequency $= f_c - f_m$.

Upper sideband frequency $= f_c + f_m$.

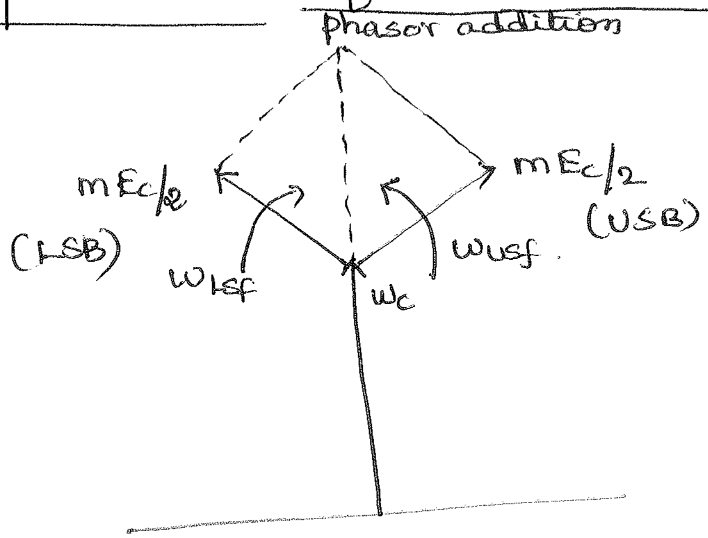


Bandwidth = USF − LSF.

$$B = \cancel{f_c} + f_{m(max)} - \cancel{f_c} + f_{m(max)}$$

$$\boxed{BW = 2 f_{m(max)}}$$

Phasor Representation of AM with Carrier



* $\omega_{usf} > \omega_c$.

* $\omega_{lsf} < \omega_c$.

AM Power Distribution:

We know, $P = \frac{V^2}{R}$. Average Power $= \frac{V_{rms}^2}{R}$.

Total power in an DSBFC is gn. by.

$$P_t = P_c + P_{usb} + P_{lsb}$$

$$= \frac{(E_c/\sqrt{2})^2}{R} + \frac{(mE_c/2\sqrt{2})^2}{R} + \frac{(mE_c/2\sqrt{2})^2}{R}.$$

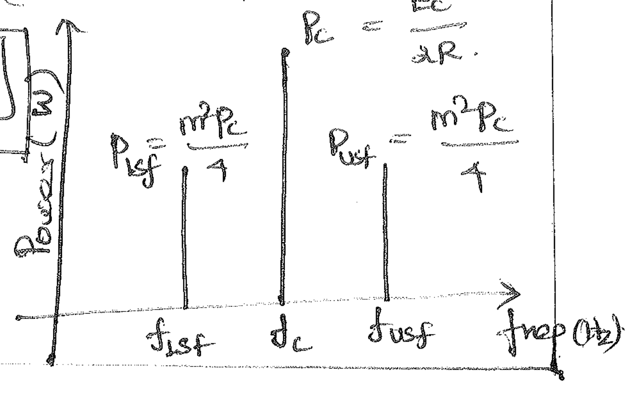
$$= \frac{E_c^2}{2R} + \frac{m^2E_c^2}{8R} + \frac{m^2E_c^2}{8R}.$$

$$= \frac{E_c^2}{2R} + \frac{m^2}{4}\frac{E_c^2}{2R} + \frac{m^2}{4}\frac{E_c^2}{2R}.$$

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

$$P_t = \underbrace{P_c}_{\text{(carrier power)}} + \frac{m^2}{2}P_c \text{ (sideband power)}.$$

$$\boxed{P_t = P_c\left[1 + \frac{m^2}{2}\right]} \quad (3)$$



Power Spectrum for an AM DSBFC

AM Current Calculations :

Let I_c be carrier current

I_t be total current of AM (rms)

From (3)

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

$$\frac{I_t^2/R}{I_c^2/R} = 1 + m^2/2.$$

$$\therefore \frac{I_t}{I_c} = \sqrt{1 + m^2/2}.$$

$$\therefore \boxed{I_t = I_c\sqrt{1 + m^2/2}.}$$

Transmission Efficiency (η)

$$\% \eta = \frac{\text{Total Power in sideband}}{\text{Total Power of AM}} \times 100$$

$$= \frac{\frac{m^2}{4}\frac{E_c^2}{2R} + \frac{m^2 E_c^2}{4 \times 2R}}{\frac{E_c^2}{2R}\left[1 + m^2/2\right]}.$$

$$= \frac{m^2/2.}{1 + m^2/2.}$$

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

when $m = 1$, $\% \eta = 33.3 \%$.

* 33.3% is used to transmit the signal, remaining power is wasted in the carrier and sideband signals.

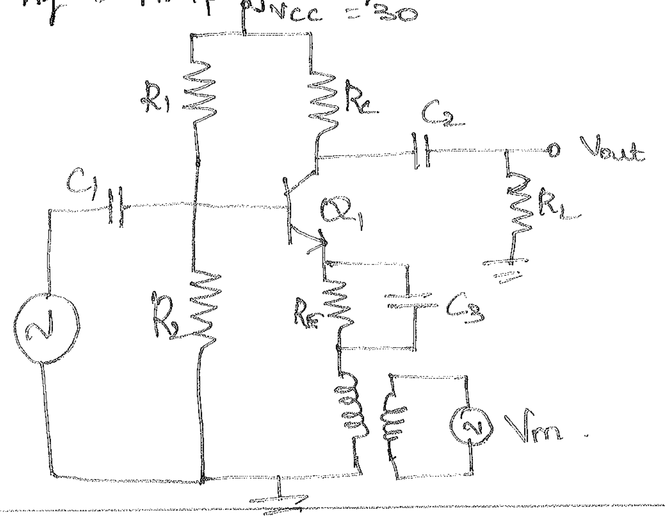
DSBFC MODULATORS.

AM modulating Circuits
- Square law or non linear modulators (Low level modulators) — Based on power out,
- Linear modulators (High Level Modulators).

Low level AM modulator

* A simple Class A amplifier is used for the generation of AM Signals.
* The carrier signal is applied at the base terminal. and modulating signal is applied at the emitter terminal. So it is called Emitter Modulation.
* When the modulating signal is applied to an emitter, the gain of the amplifier varies according to the Voltage of modulating signal.
* The carrier signal is amplied based on the Variation in the amplifier gain.
* The amplitude of the carrier signal is modulated by the modulating signal.
* The voltage gain is gn. by $A_v = A_q(1+m\sin 2\pi f_m t)$

where A_v - Amplifier voltage gain.
A_q - Amplifier quiescent (without modulation) Voltage gain.



Advantages of Low Level Modulation:

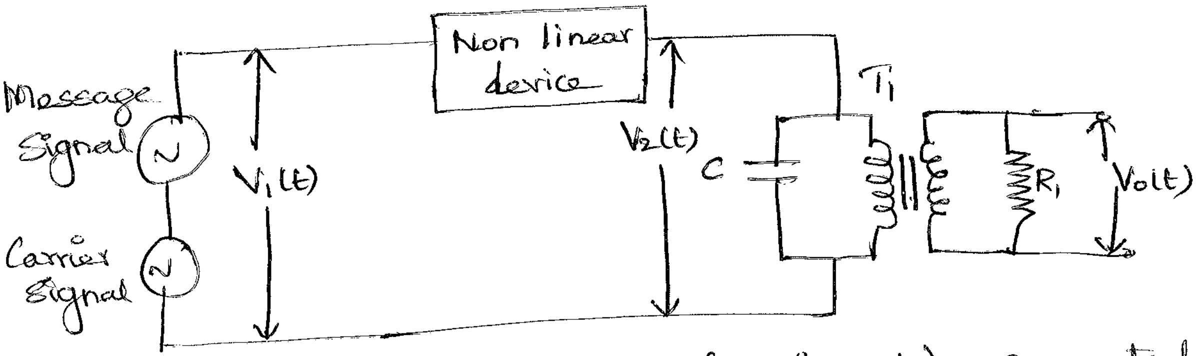
* less modulating power is required to obtain high percentage of modulation.
* Modulating Circuit is designed for low power.

Disadvantages of Low level Modulators:

* At high operating powers linear amplifier are very inefficient

Non-linear Modulators:

Square Law Modulator:



* Here, the carrier signal ($E_c \sin \omega_c t$) generated from the carrier source and message signal $E_m \sin \omega_m t$ generated from the modulating signal source are fed to the Non linear device such as transistor, triode or diode.

$$\therefore V_1(t) = E_m \sin \omega_m t + E_c \sin \omega_c t \quad \text{---} (1)$$

* The i/p o/p relation of any non linear device is gn. by

$$V_2(t) = a V_1(t) + b V_1^2(t) \quad \text{---} (2) \qquad a, \& \ b \text{ are constants}$$

$V_1(t)$ - i/p to non linear device ; $V_2(t)$:- o/p to non-linear device

Sub (1) in (2), $V_2(t) = a[E_m \sin \omega_m t + E_c \sin \omega_c t] + b[E_m \sin \omega_m t + E_c \sin \omega_c t]^2$

$$V_2(t) = \underbrace{a E_m \sin \omega_m t}_{\text{modulating Signal}} + \underbrace{a E_c \sin \omega_m t}_{\text{Carrier signal}} + \underbrace{b E_m^2 \sin^2 \omega_m t}_{\text{Squared mod. signal}} + \underbrace{b E_c^2 \sin^2 \omega_c t}_{\text{Squared car signal}} + b E_m E_c [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t].$$

* This signal $V_2(t)$ is passed through the BPF for the frequencies $(\omega_c + \omega_m)$ and $(\omega_c - \omega_m)$ AM signal with Side bands.

* The rest of the frequencies are filtered.

$\therefore \quad V_o(t) = aE_c \sin w_c t + bE_m E_c [\cos(w_c - w_m)t - \cos(w_c + w_m)t]$.

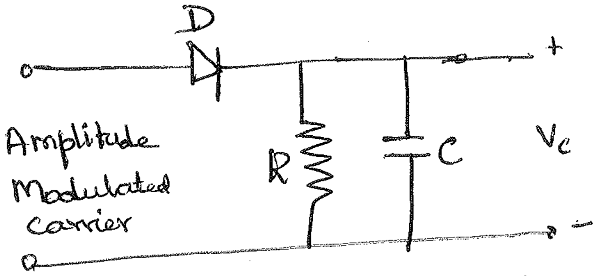
$V_o(t)$ - AM with carrier and 2SB frequencies.

DSB FC Demodulators:

Envelope Detector: (or) Diode Detector:

Def: It is the process in which the modulating signal is recovered back from the modulated signal.

* This is the reverse process of modulation.



* When the i/p AM signal is applied, the diode D Conducts the positive half cycle
* If R is not present, the Capacitor C charges to the peak positive voltage of the carrier (V_c).
* If carrier Voltage ↑, V_c ↑. If Carrier Voltage ↓, Capacitor Voltage should be decreased. In order to achaive that, the Resistor R is included. So that the capacitor C discharges through R with the time constant RC
* Since carrier cycle is small, time constant is also small.
∴ The V_c will follow the envelope more closely looking similar to the modulating signal.
* Hence the i/p modulating signal is recovered from the modulated signal.

Double Side Band Suppressed Carrier (DSB SC)

* The carrier frequency in the AM wave (DSB FC) spectrum is not carrying any information, the carrier frequency can be suppressed.
* The AM with suppressed carrier is called DSB SC

Generation of DSB SC (or) DSBSC Modulator:

Let $e_m(t) = E_m \sin 2\pi f_m t$ and $e_c(t) = E_c \sin 2\pi f_c t$

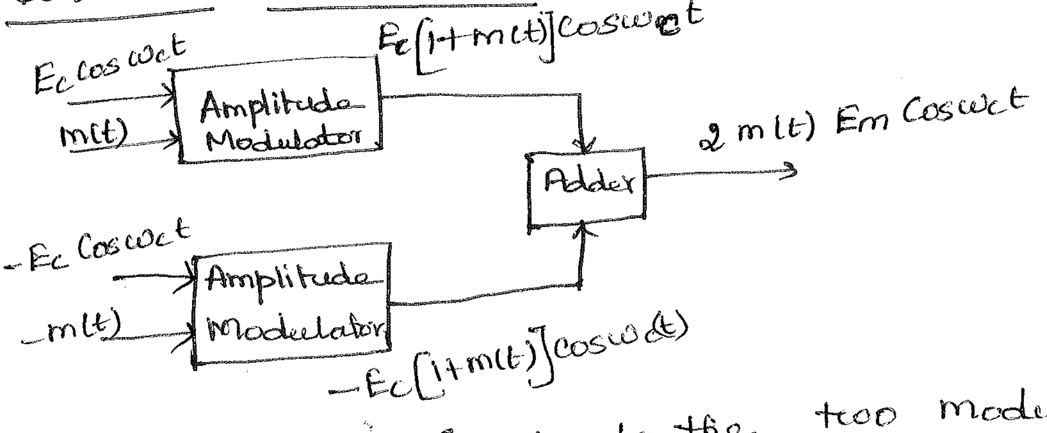
$$e_{DSBSC} = e_m(t)\, e_c(t)$$

$$= E_m \sin 2\pi f_m t \cdot E_c \sin 2\pi f_c t$$

$$E_{DSBSC} = \frac{E_m E_c}{2}\left[\cos 2\pi(f_c - f_m)t - \cos 2\pi(f_m - f_c)t\right]$$

* Only SB is present. No carrier is present in E_{DSBSC}.
* Product Modulator can be used to generate DSB SC

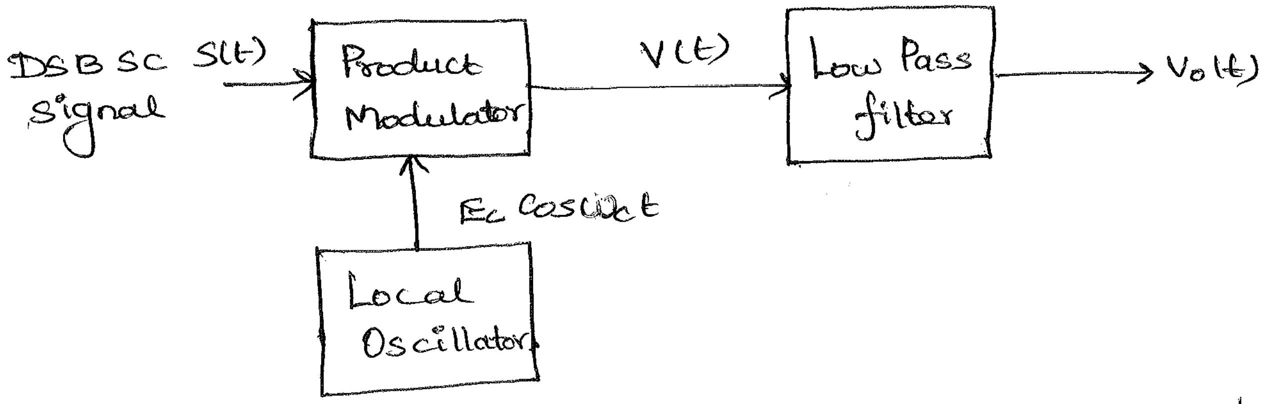
Balanced Modulator: (or) Product Modulator



* The carrier signal to the two modulators are reverse polarity and also the modulating signal.
* the modulated signals of both the modulators are added resulting DSB SC signal suppressing the carrier.

Demodulation of DSB Sc:

Coherent detection of DSBSC signal.

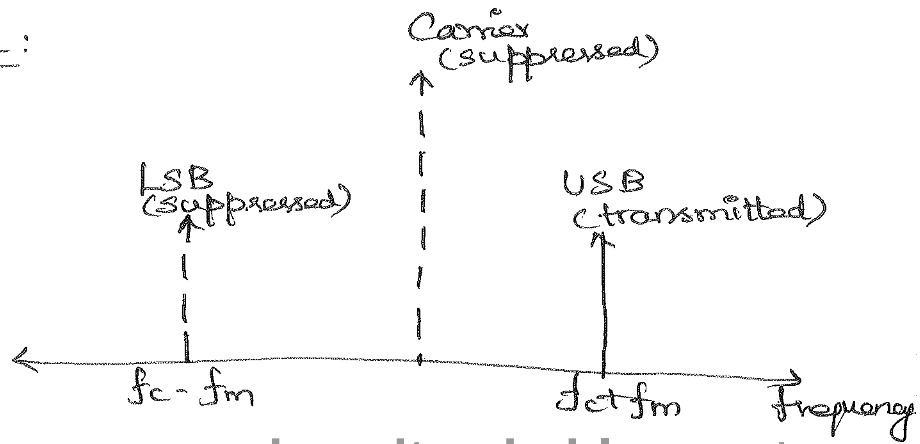


* The synchronous or coherent detector uses locacally generated carrier for detection.
* The DSBSC signal is multiplied with the carrier signal which is generated by a local oscillator at the receiver. (V(t)). The carrier signal should be sinchronized with modulating signal.
* The O/P V(t) is passed through the Low pass filter to get the message signal.

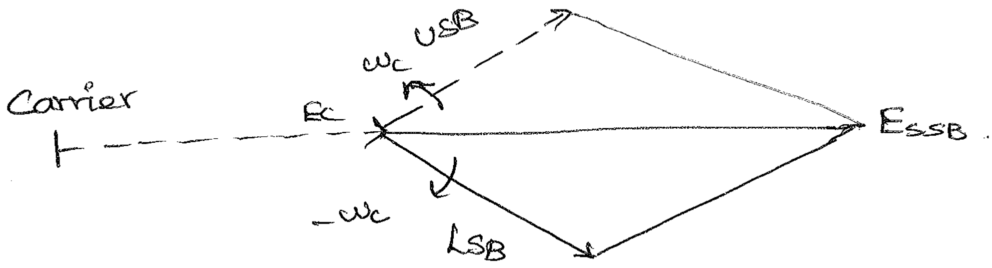
Single Side band Suppressed Carrier (SSBSC)

* Both the sidebands in the DSBSC are carrying the same information, it is sufficient to transmit only one sideband.
* Such a transmission is called single side Band Suppressed carrier (or) SSB system or SSB transmission.

Spectrum of SSB signal:



Phasor Representation of SSB_SC AM



Power Calculation in SSB SC AM

The total power in transmitted AM is

$$P_t = \frac{E_c^2}{2R}\left[1 + \frac{m^2}{2}\right].$$

$$P_t = P_c\left[1 + \frac{m^2}{2}\right].$$

If carrier and one side band is suppressed, then the total power is

$$P_t'' = P_{LSB} = P_{USB}.$$

$$= \frac{m^2 E_c^2}{8R}.$$

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

$$\therefore \text{Power Saving} = \frac{P_t - P_t''}{P_t}.$$

$$= \frac{P_c\left[1 + \frac{m^2}{2}\right] - \frac{m^2 P_c}{4}}{P_c\left[1 + \frac{m^2}{2}\right]}$$

$$= \frac{\cancel{P_c}\left[1+\frac{m^2}{2}-\frac{m^2}{4}\right]}{\cancel{P_c}\left[1+\frac{m^2}{2}\right]}$$

$$= \frac{1+\frac{2m^2-m^2}{4}}{1+\frac{m^2}{2}}$$

$$= \frac{1+m^2/4}{1+m^2/2}$$

$$= \frac{(4+m^2)/\cancel{4}}{(2+m^2)/\cancel{2}} =$$

$$= \frac{4+m^2}{2(2+m^2)}$$

$$= \frac{4+m^2}{4+2m^2}.$$

When $m=1$, the power saving in SSB SC is $5/6 = 83.3\%$.

When $m=0.5$, the power saving in SSB SC is 94.4%.

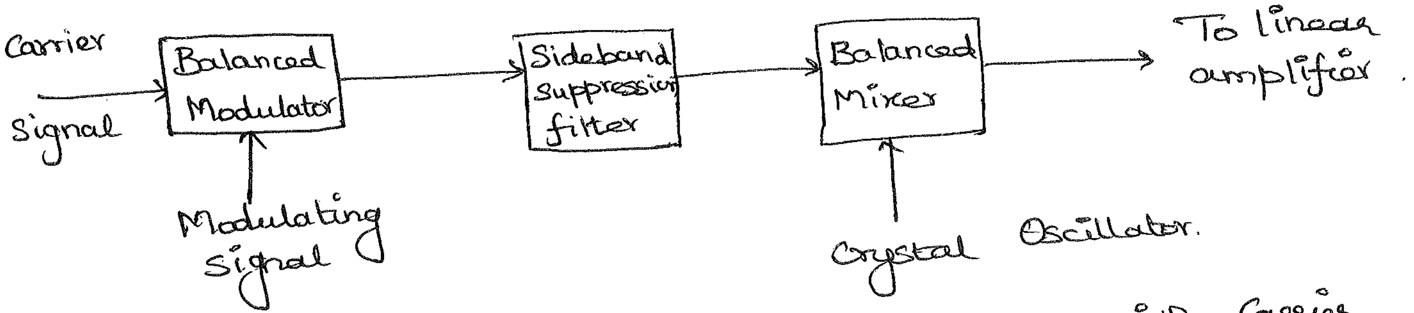
Advantage:

1. Bandwidth used is less (ie half of DSB SC)
* The power of the suppressed sideband is saved.
* The effect of noise is reduced at the receiver.
* Fading effect is removed in SSB.
(Fading effect arises because of the interference of carrier and 2 sidebands).

Disadvantage: Complex circuit for frequency stability.

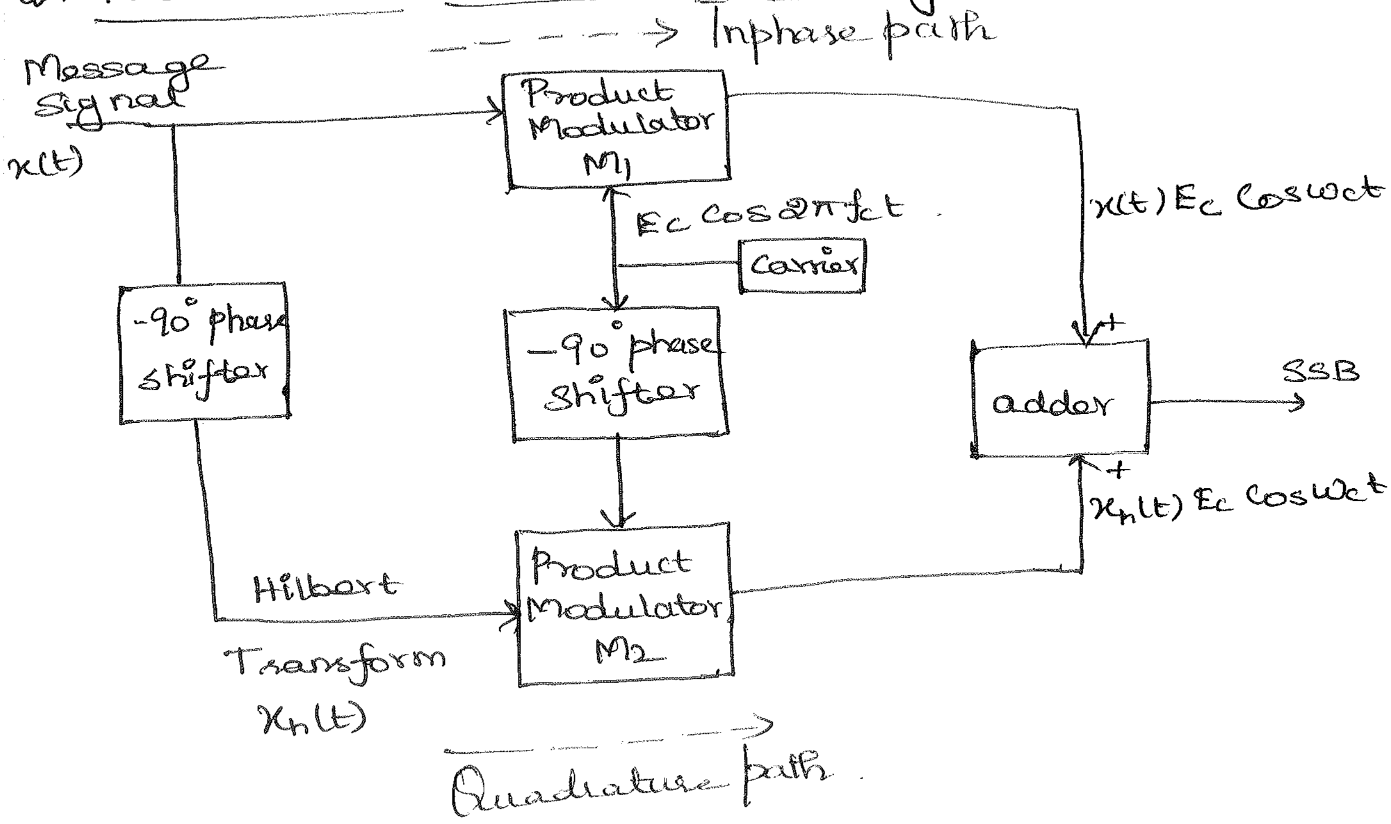
Generation of SSB:

1. Filter Method:



* The ~~modulating~~ balanced modulator is provided with carrier signal and modulating signal and the output is the sum and difference signal (Sideband Lower & Upper signal) (or) DSB SC signal.
* This LSB & USB signals are given to filter which suppresses one SB signal and select only one side band signals.
* The filter selects the high frequency component to get single SB of high frequency.
* ∵ modulation is performed before transmission it has to be converted to suitable high frequency for transmission. This process is done using mixer.

2. Phase shift Method or Hartley's Method :



* There are two seperate double side band modulators (balanced modulator 1 & 2).
* The modulating signal and Carrier are applied directly to M_1 then both are phase shifted $90°$ and applied to M_2.
* The O/Ps from the two modulators M_1 & M_2 are DSB SC with proper phase such that, when they are combined in a linear summer, USB is cancelled.

The o/p of M_1 is $E_c E_m \sin\omega_m t \sin\omega_c t$. ——(1)

The o/p of M_2 is $E_c E_m \cos\omega_m t \cos\omega_c t$. ——(2)

$$(1) \Rightarrow \frac{E_c E_m}{2}\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right] \quad \text{——(3)}$$

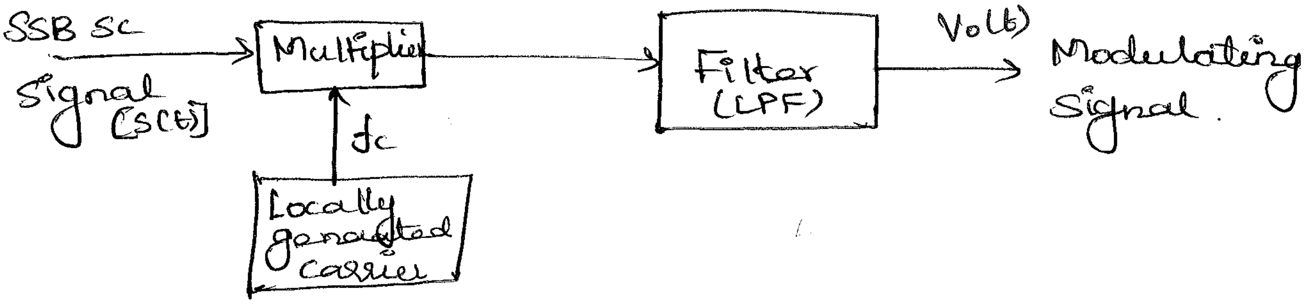
$$(2) \Rightarrow \frac{E_c E_m}{2}\left[\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t\right] \quad \text{——(4)}$$

$$(3) + (4) \Rightarrow E_c E_m \cos(\omega_c - \omega_m)t$$

(LSB signal).

Demodulation of SSB SC signals.

* The baseband signal can be retrieved from the SSB SC signal by using synchronous detection method.
* It involves multiplication of SSB SC signal with the locally generated carrier.



The o/p of Local oscillator is

$$C(t) = E_c \cos 2\pi f_c t.$$

The SSB SC signal having LSB is $\frac{E_m E_c}{2} \cos[2\pi(f_c - f_m)t]$

The o/p of product multiplier is $V(t) = S(t)\, C(t)$.

$$\therefore V(t) = \frac{E_m E_c}{2} \cos[2\pi(f_c - f_m)t]\, E_c \cos 2\pi f_c t.$$

$$= \frac{E_m E_c^2}{4} \{\cos 2\pi(2f_c - f_m)t + \cos 2\pi f_m t\}.$$

$$V(t) = \frac{E_m E_c^2}{4} \cos 2\pi f_m t + \frac{E_m E_c^2}{4} \cos 2\pi(2f_c - f_m)t$$

1^{st} term :- scaled version of message signal.

$\therefore$ From the V(t), the message signal can be retrieved by passing V(t) through a LPF.

$$\therefore \quad V_o(t) = \frac{E_m E_c^2}{4} \cos 2\pi f_m t.$$

where $\frac{E_m E_c^2}{4}$ is the scaling factor.

VSB - Vestigial Side Band.

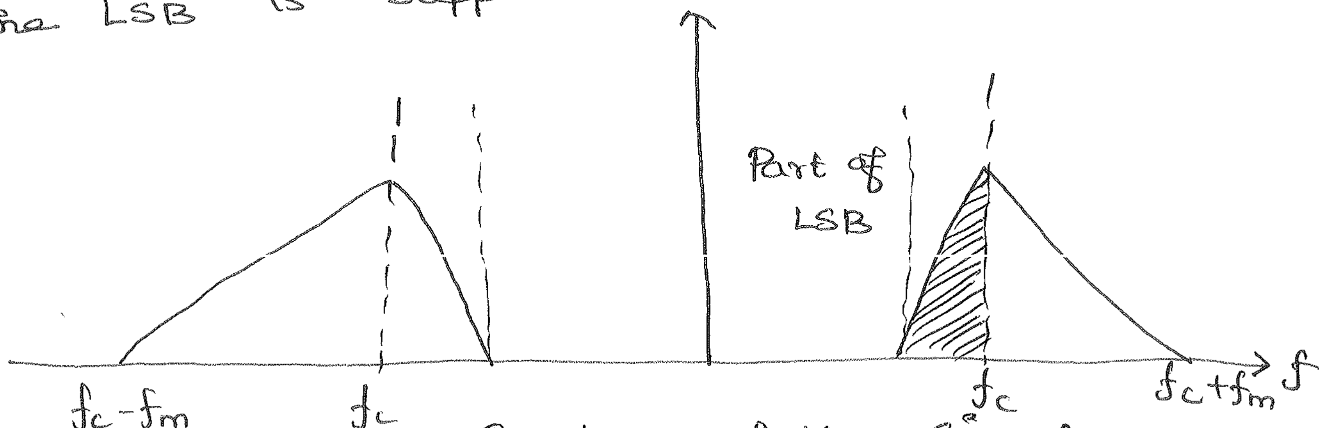
* Vestigial sideband modulator is a type of modulation in which one sideband is fully transmitted and part of the other side band is also transmitted suppressing the carrier.

* BW of VSB = $f_{SSBSC} < f_{VSB} < f_{DSB\,SC}$

* Need? Video signals, TV signals and High Speed data signals have important information at the lower frequency region.

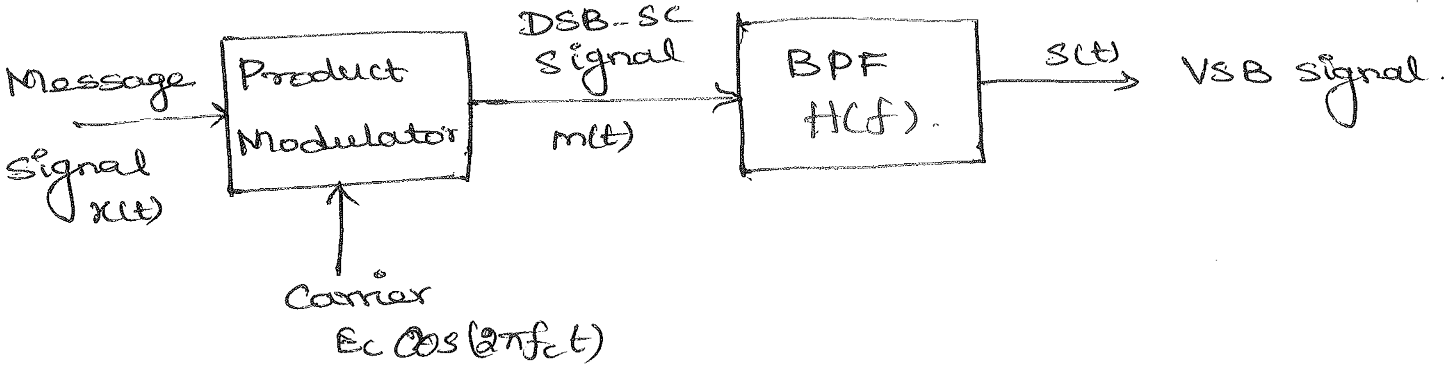
→ If SSB Modulation is used, the information will be lost completely,

→ To compromise this information loss, a part of the LSB is suppressed.



Spectrum of VSB signal.

Generation of VSB :



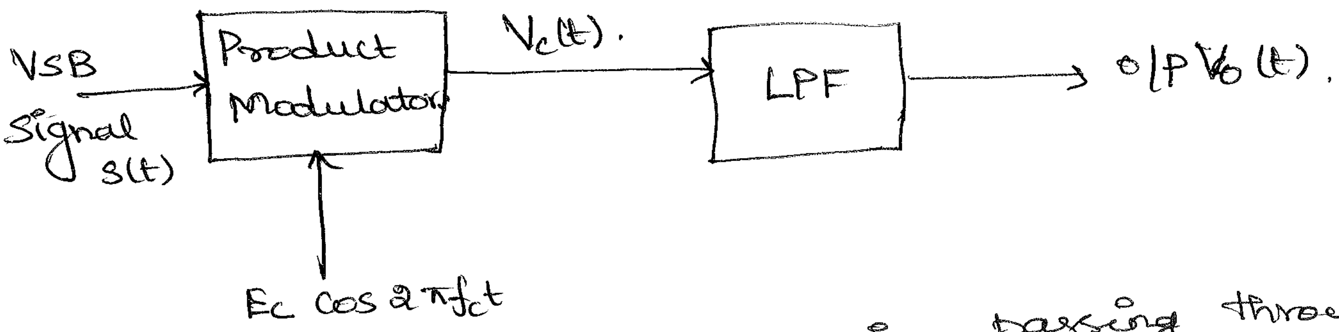
* The modulating signal $x(t)$ is applied to the product modulator along with the carrier signal $c(t)$
* The o/p of Product modulator is gn by

$$m(t) = x(t) \cdot c(t) = x(t) \cdot E_c \cos 2\pi f_c t .$$

* This represents the DSB SC signal. This can be passed through BPF to get the desired spectrum of Vestigial Side Band Signal.
* VSB signal is gn by $s(f) = \frac{E_c}{2}\left[X(f-f_c) + X(f+f_c)\right] H(f)$.

$$s(t) = \frac{E_c}{2} x(t) \cos 2\pi f t - x_2(t) \sin 2\pi f t$$

Demodulation of VSB:



* The VSB modulated wave is passing through a product modulator where it is multiplied by a locally generated carrier $E_c \cos 2\pi f_c t$.
* $\therefore V_c(t) = s(t)\, E_c \cos 2\pi f_c t$.
* The $V_c(t)$ is passed through the LPF to get the desired message signal.

ANGLE Modulation:

Def: When frequency or phase of the carrier is varied according to the amplitude of message signal, then it is called Angle Modulation.

* The Amplitude of the carrier is constant.

Adv * Better discrimination ~~compared~~ against to noise & Interference

DisAdv * Needs more BW.

Angle Modulation → Frequency Modulation
Angle Modulation → phase Modulation.

An Angle modulated signal is gn as

$$S(t) = A_c \cos(\cancel{\theta_c(t)}\ \omega_c(t) + \theta(t))$$ where

A_c - Peak Amplitude of the carrier.

ω_c - Carrier frequency.

$\theta(t)$ - Instantaneous phase deviation.

Phase Modulation

Def: PM is defined as the varying phase of a constant amplitude carrier directly proportional to the amplitude of the modulating signal at a rate equal to the frequency of the modulating signal.

Mathematically, $m(t) = V_c \cos[\omega_c t + K V_m \cos \omega_m t]$.

Frequency Modulation: Varying frequency of a constant amplitude carrier directly proportional to the amplitude of the modulating signal at a rate equal to the frequency of the modulating signal

$$m(t) = V_c \cos\left[\omega_c t + \frac{K_1 V_m}{\omega_m} \sin \omega_m t\right]$$

Note: Direct FM → Indirect PM

Direct PM → Indirect FM

FM and PM Waveforms:

(i) In FM, the maximum frequency deviation (change in the carrier frequency) occurs during the maximum positive and negative peaks of the modulating signal.

(ii) In PM, the maximum frequency deviation occurs during the zero crossings of the modulating signal.

(iii) In both FM & PM, the rate at which the frequency changes occur is equal to the modulating signal frequency.

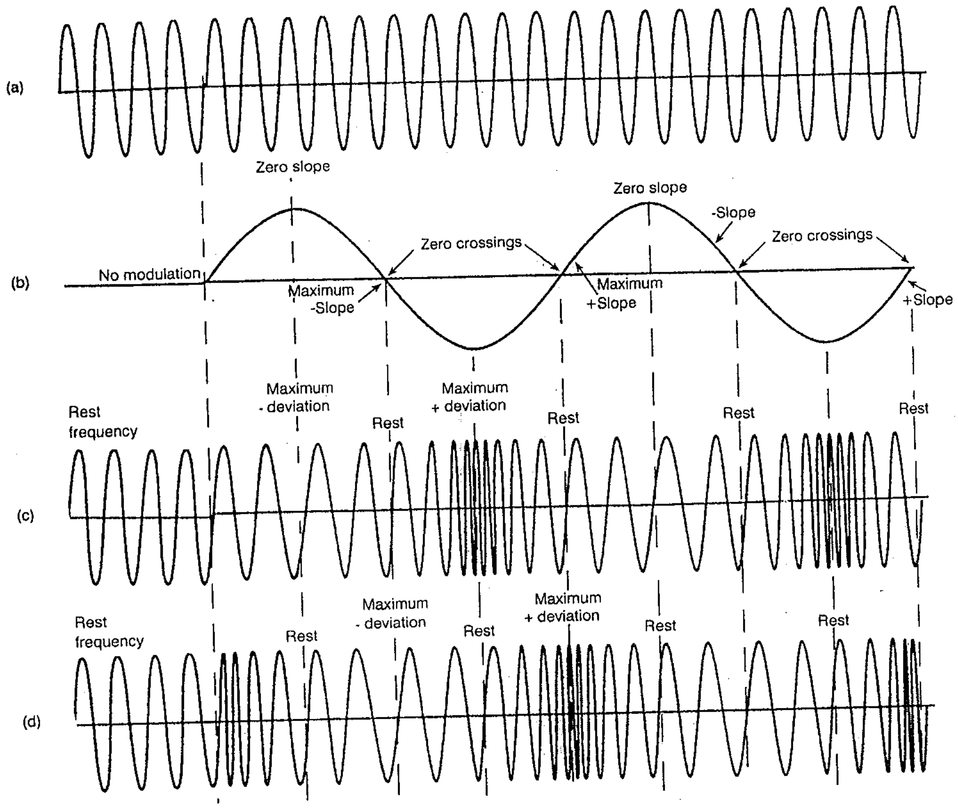


FIGURE 7-3 Phase and frequency modulation of a sine-wave carrier by a sine-wave signal: (a) unmodulated carrier; (b) modulating signal; (c) frequency-modulated wave; (d) phase-modulated wave

Frequency deviation:

It is the change in frequency that occurs in the carrier when it is acted on by a modulating signal frequency. Mathematically expressed as

$$\Delta f = K_1 V_m \ (Hz)$$

Where K_1 is deviation sensitivity, V_m - peak modulating signal amplitude.

Deviation Sensitivity $K_1 \left(\frac{rad/sec}{V}\right) = \frac{\text{Change in o/p frequency}}{\text{Change in the amplitude of i/p voltage}} \left(\frac{\Delta w}{\Delta V}\right)$.

<u>Modulation</u> <u>Index</u>. (Unitless)

$$m_f = \frac{K_1 V_m}{\omega_m} \quad \text{or} \quad \frac{K_1 V_m}{f_m}$$

$$m_f = \frac{\Delta f}{f_m} \quad \text{where } f_m = \text{cyclic frequency.}$$

<u>Percent Modulation</u> $\%\ \text{Modulation} = \frac{\Delta f(\text{actual})}{\Delta f(\max)} \times 100$.

<u>Phase Deviation</u>: or <u>Modulation</u> <u>Index</u> for PM.

$m_p = K V_m$ (radians).

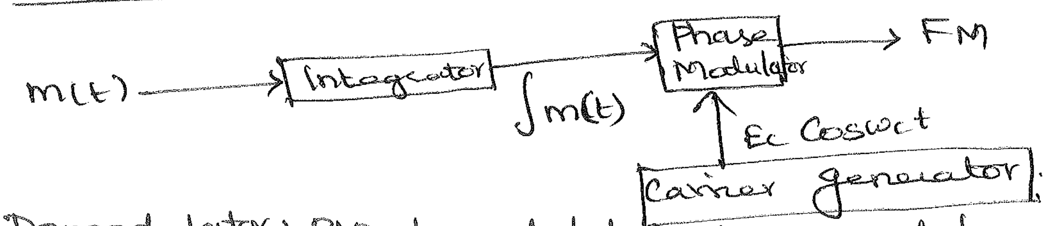
K - ~~Sens~~ deviation sensitivity

Vm - peak modulating signal Amplitude

Deviation Sensitivity $K = \frac{\text{Change in o/p phase } (\Delta\theta)}{\text{Change in the amplitude of i/p voltage } (\Delta V)} \left(\frac{rad}{V}\right)$

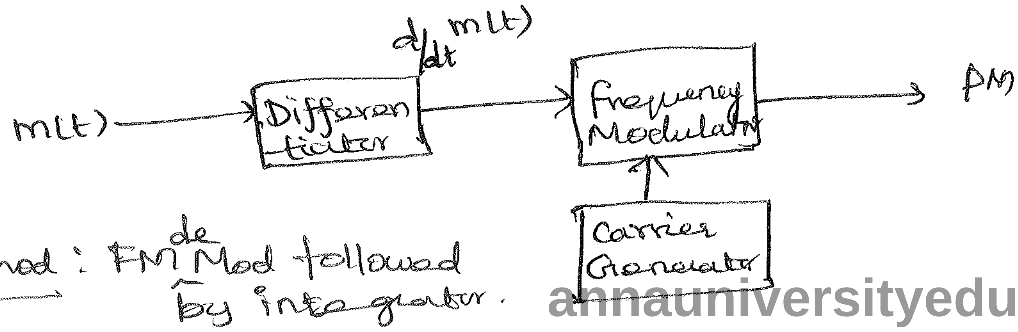
<u>Relation</u> <u>between</u> <u>FM & PM</u>:

<u>FM</u> <u>Generation</u> <u>using PM</u>:



<u>FM Demodulator</u>: PM demodulator followed by differentiator

<u>PM</u> <u>Generation</u> <u>using FM</u>:



<u>PM Demod</u>: FM deMod followed by integrator.

Expression for phase Modulation and Frequency Modulation.

Angle Modulated signal is expressed as

$$S(t) = V_c \cos[\omega_c t + \theta(t)]$$

In PM, instantaneous phase deviation is given by (1)

$$\theta(t) = k V_m(t) \text{ (rad)}$$ where, k - deviation Sensitivity.

$V_m(t)$ - Modulating signal.

In FM, instantaneous frequency deviation is given by (2)

$$\theta'(t) = k_1 V_m(t) \text{ rad/sec}$$ where,

k_1 – Deviation Sensitivity of FM

From (1) & (2)

$$\theta(t) = \int \theta'(t)\, dt.$$

$$= \int k_1 V_m(t)\, dt.$$

$$\theta(t) = k_1 \int V_m(t)\, dt.$$

Phase Modulation:

∴ The phase modulated signal is given by,

$$S(t) = V_c \cos[\omega_c(t) + k V_m(t)].$$

w.k.t $V_m(t) = V_m \cos\omega_m t.$

$$\therefore S(t) = V_c \cos[\omega_c t + k V_m \cos\omega_m t].$$

$$= V_c \cos[\omega_c t + m_p \cos\omega_m t].$$

where m_p - modulation index of PM.

Frequency Modulation:

The frequency modulated signal is given by,

$$s(t) = V_c \cos\left[\omega_c t + \theta(t)\right]$$

$$= V_c \cos\left[\omega_c t + k_1 \int V_m(t)\, dt\right].$$

$$= V_c \cos\left[\omega_c t + k_1 \int V_m \cos\omega_m t\, dt\right].$$

$$S(t) = V_c \cos\left[\omega_c t + \frac{k_1 V_m}{\omega_m} \sin\omega_m t\right].$$

$$= V_c \cos\left[\omega_c t + m_f \sin\omega_m t\right].$$

where m_f - modulation Index for FM.

Frequency Analysis of Angle Modulated wave.

* A Single frequency modulating signal produces an infinite number of pairs of side frequencies and thus has an infinite bandwidth.
* Each side frequency is displaced from the carrier signal by an integral multiple of the modulating signal frequency.

The angle modulated signal can be written as

$$S(t) = V_c \cos\left[\omega_c t + m_f \cos\omega_m t\right]. \quad (1)$$

To solve cosine function, bessel fn. identities are applied,

$$\therefore \cos(\alpha + m_f \cos\beta) = \sum_{n=-\infty}^{\infty} J_n(m_f) \cos\left[\alpha + n\beta + \frac{n\pi}{2}\right]. \quad (2)$$

where $J_n(m_f)$ is Bessel fn. of first kind of n^{th} order with argument 'm_f'.

Apply eqn ② in ①.

$$\therefore \; S(t) = V_c \sum_{n=-\infty}^{\infty} J_n(m_f) \cos\left[\omega_c t + n\omega_m t + n\pi/2\right] \quad \text{——— ③}$$

Expanding eqn ③,

$$S(t) = V_c \Big\{ J_0(m_f)\cos\omega_c t + J_1(m_f)\cos\left[(\omega_c+\omega_m)t + \pi/2\right] - $$
$$- J_1(m_f)\cos\left[(\omega_c-\omega_m)t - \pi/2\right] +$$
$$J_2(m_f)\cos\left[(\omega_c+2\omega_m)t\right] +$$
$$J_2(m_f)\cos\left[(\omega_c-2\omega_m)t\right] + \ldots$$

where $S(t)$ – angle modulated wave.

m_f – modulation index.

V_c – peak amplitude of the unmodulated carrier.

$J_0(m_f)$ – carrier component.

$J_1(m_f)$ – 1^{st} set of side frequencies displaced from the carrier by ω_m.

$J_n(m_f)$ – n^{th} set of side frequencies displaced from the carrier by $n\omega_m$.

→ Eqn (B) shows a single frequency modulating signal produces an infinite number of sets of side frequencies.

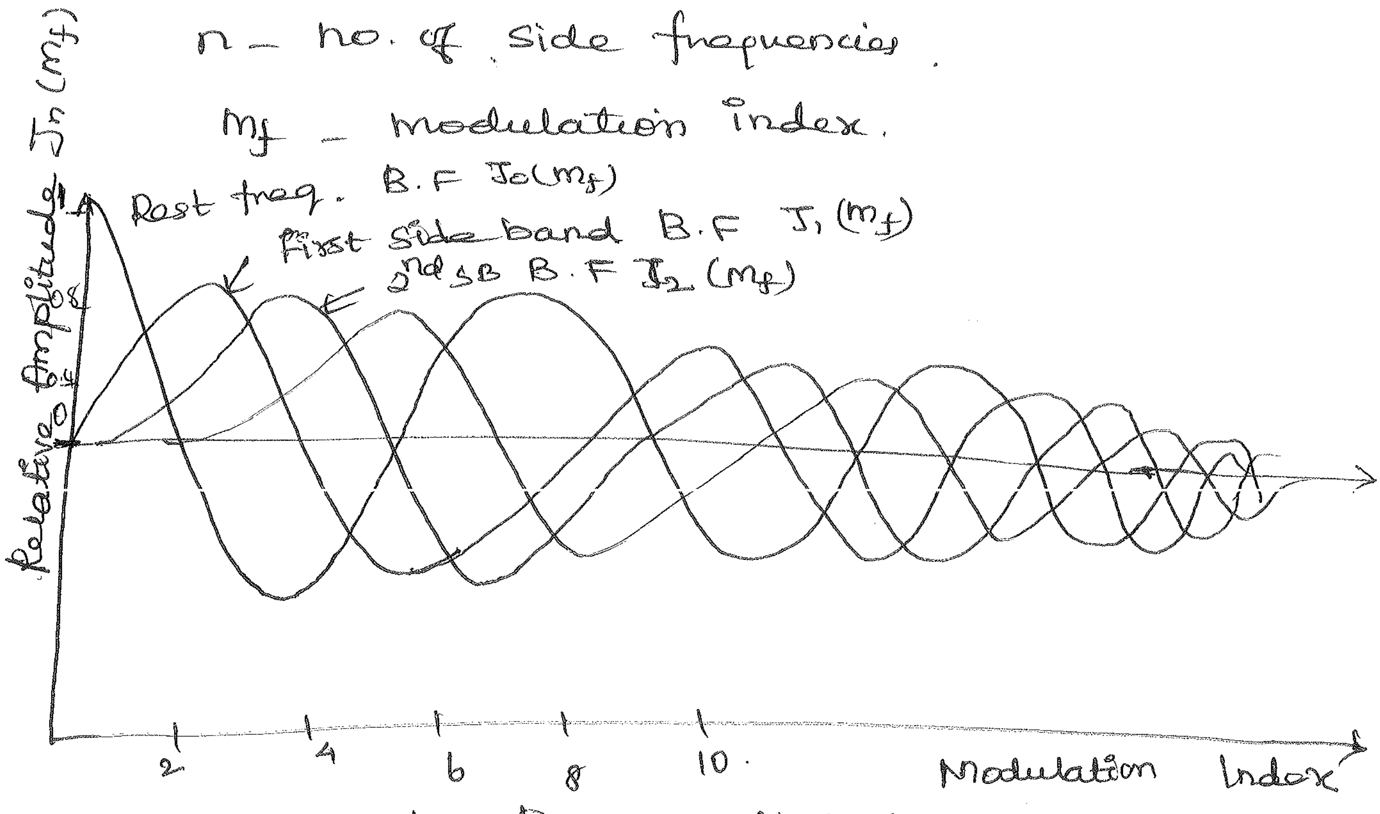
→ A sideband set contains an upper and lower side frequency ($f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm nf_m$, ...)

→ To solve the amplitude of the side frequencies J_n,

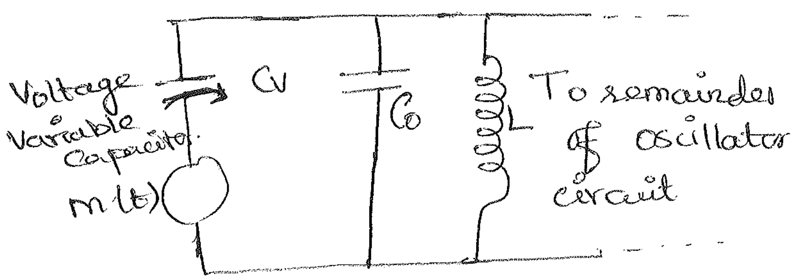
$$J_n(m_f) = \left(\frac{m_f}{2}\right)^n \left[\frac{1}{n} - \frac{(m_f/2)^2}{1!\,(n+1)!} + \frac{(m_f/2)^4}{2!\,(n+2)!} + \cdots\right].$$

n – no. of side frequencies.

m_f – modulation index.



FM Generation by Parameter Variation Method



* Here the modulating signal varies the voltage across C_v.
* As C_v changes, there is the corresponding change in oscillator freq.

$$f = \left[2\pi\sqrt{LC}\right]^{-1}$$

* The instantaneous frequency of the oscillator depends on the instantaneous value of modulating signal
* Frequency modulation can be achieved by the variation of any element or parameter on which freq. depends.

Bandwidth of FM

* BW of FM can be find using Carson's rule.
* Carson's rule states that the BW of FM is twice the sum of deviation and highest Modulation Frequency.

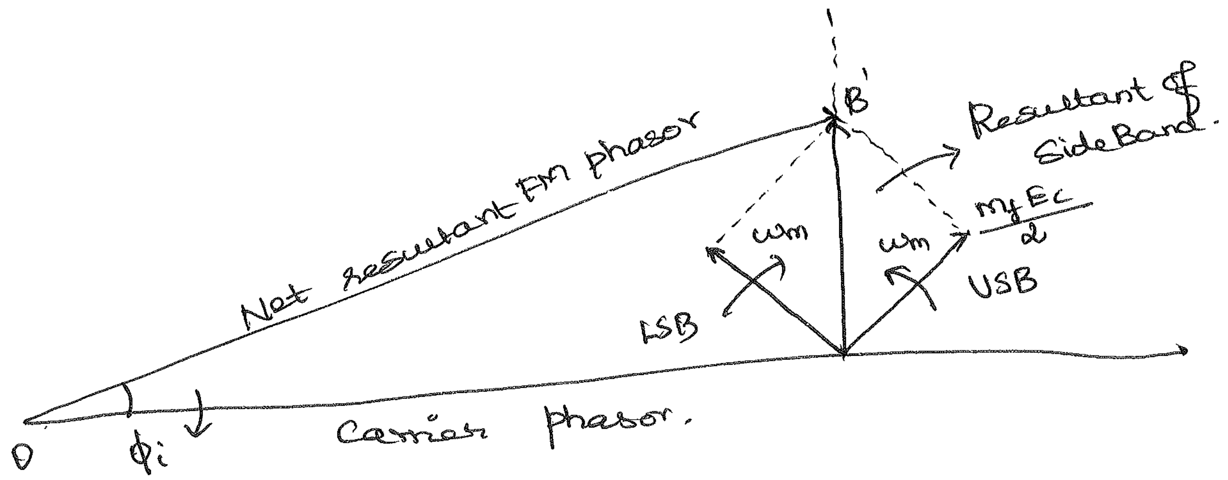
$$BW = 2(\Delta f + f_m) = 2\left(\Delta f + \frac{\Delta f}{m_f}\right)$$

$$\boxed{BW = 2\Delta f\left(1 + \frac{1}{m_f}\right)} \quad rad/sec$$

* FM can be classified as Narrowband and Wide Band

ie FM — NB FM $m_f << 1$
— WB FM. $m_f >> 1$.

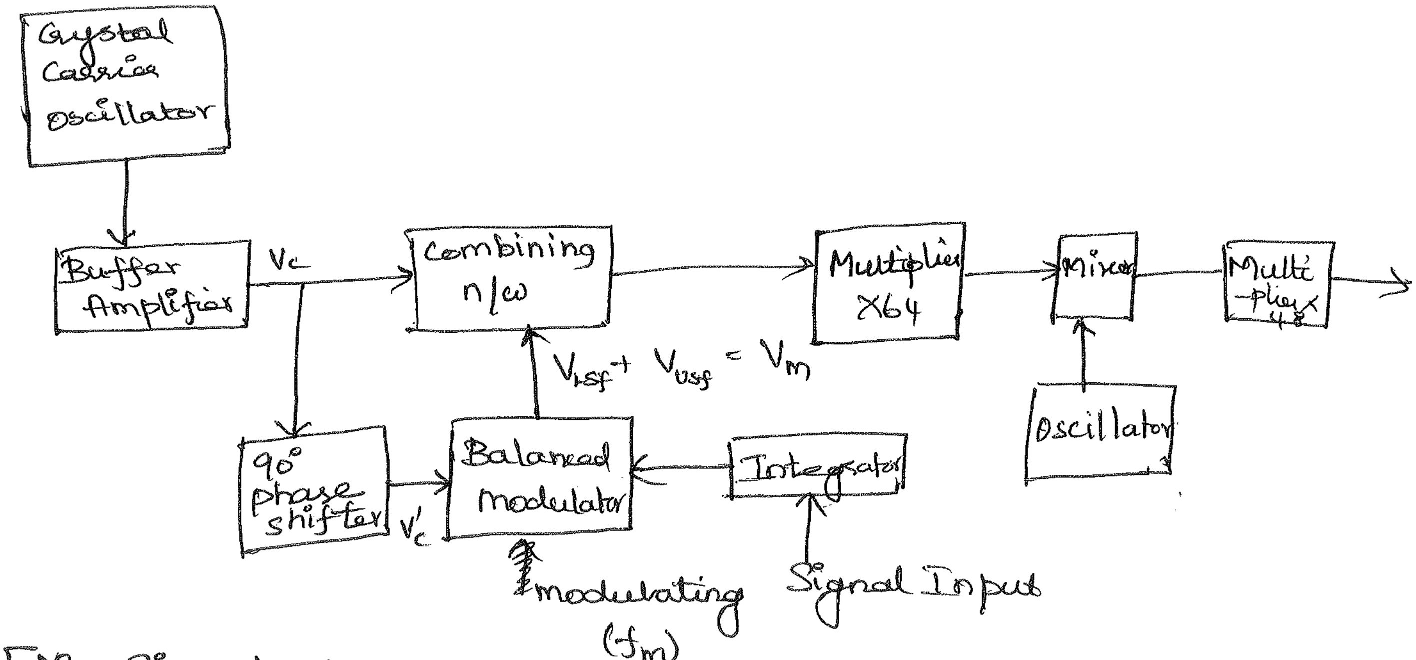
Phasor diagram of NBFM



FM Modulators:

Armstrong Indirect FM Transmitter:

* Indirect FM transmitters produce an o/p waveform in which the phase deviation is directly proportional to the modulating Signal.
* ∴ The carrier oscillator must be a Crystal Oscillator since carrier oscillator is not directly deviated.



* FM Signal can be obtained from phase modulator by integrating the modulating signal before giving to phase Modulator.
* In an Armstrong modulator, a relatively low frequency carrier is phase shifted and fed to the balanced modulator where it is mixed with integrated modulating signal.
* The output from the Balanced modulator is double Side band Suppressed Carrier wave. which is combined in a combining network with the original carrier signal from the Crystal Oscillator.
* The output of the combining n/w is a low index frequency modulated waveform. (NBFM)
* To convert NBFM to WBFM, mixers and multipliers are used.
* Mixing the NBFM signal with local oscillator frequency will produce sum (f_c+f_o) and difference (f_c-f_o) frequency component at the o/p of mixer.

* Thus mixer increases or decreases the centre carrier frequency, keeping frequency deviation constant.
* If FM signal with center carrier frequency f_c and deviation Δf are fed to multiplier, center carrier frequency and frequency deviation are multiplied.
* Thus resulting, increase in the modulation Index.

Mathematical Expression of generation of NBFM:

Let the angle modulated signal be $S(t) = A_c \cos[\omega_c t + \theta(t)]$ —— (1)

By applying $\cos(A+B) = \cos A \cos B - \sin A \sin B$,

$$(1) \Rightarrow \quad S(t) = A_c[\cos\omega_c t \cos\theta(t) - \sin\omega_c t \sin\theta(t)]$$

By small angle approximation, $\sin\theta = \theta$; $\cos\theta = 1$,

$$S(t) = A_c[\cos\omega_c t - \sin\omega_c t\ \theta(t)].$$

we know, $\theta(t) = m \sin\omega_m t$.

$$S(t) = A_c[\cos\omega_c t - \sin\omega_c t\ m \sin\omega_m t]$$

$$= A_c[\cos\omega_c t \quad \cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)] \text{ —— (3)}$$

From eqn (3)

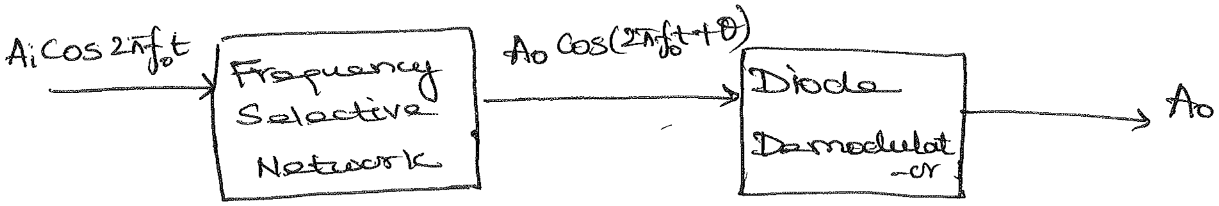
This is similar to AM wave. But phase change is there.

This indicates the generation of NBFM.

FM Demodulator:

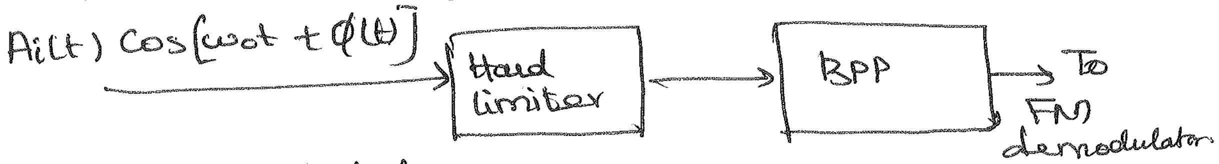
* The process of getting the modulating signal from the frequency modulated carrier is known as frequency demodulation or detection.
* To detect the FM signal, it is necessary to have a circuit whose output voltage varies linearly with a frequency of input signal.
* The circuit used is called frequency discriminator which converts FM signal into corresponding AM signal.

* Then, the modulating signal is obtained from AM signal by Envelope detection.

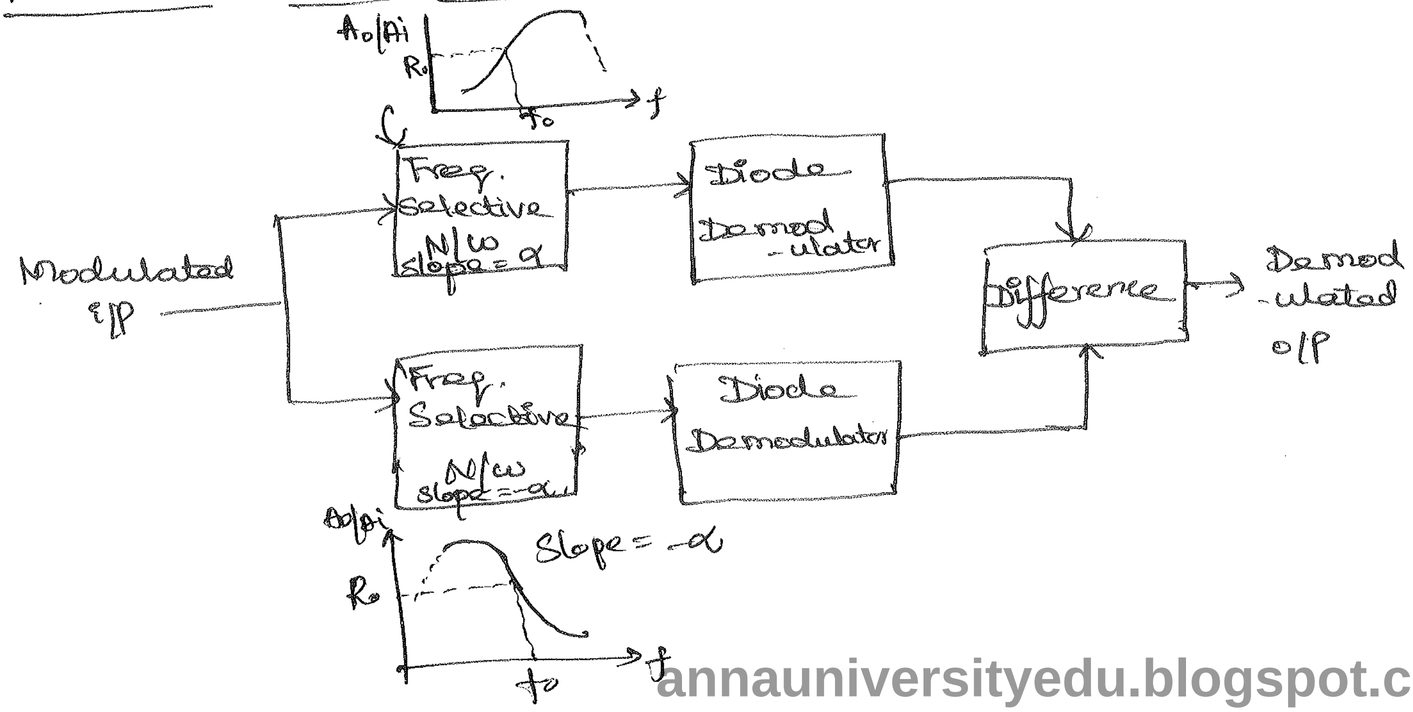


* Here, a waveform of frequency f_0 and input amplitude A_i is applied to a frequency selective network which yields an output of A_0
* The ratio of amplitudes A_0 / A_i is the absolute value of the transfer function of the network. ie $|H(j\omega)|$.
* The diode demodulator generates an output which is equal to A_0. The o/p $A_0 = R_0 A_i + \alpha A_i (f - f_0) + \beta A_i (f - f_0)^2 + \ldots$
* If the variation in the i/p ie $A_i(t) \cos(\omega_0 t + \phi(t))$ is present, then Hard limiter or Comparator is used before Frequency selective n/w.

ie



Balanced FM demodulator:



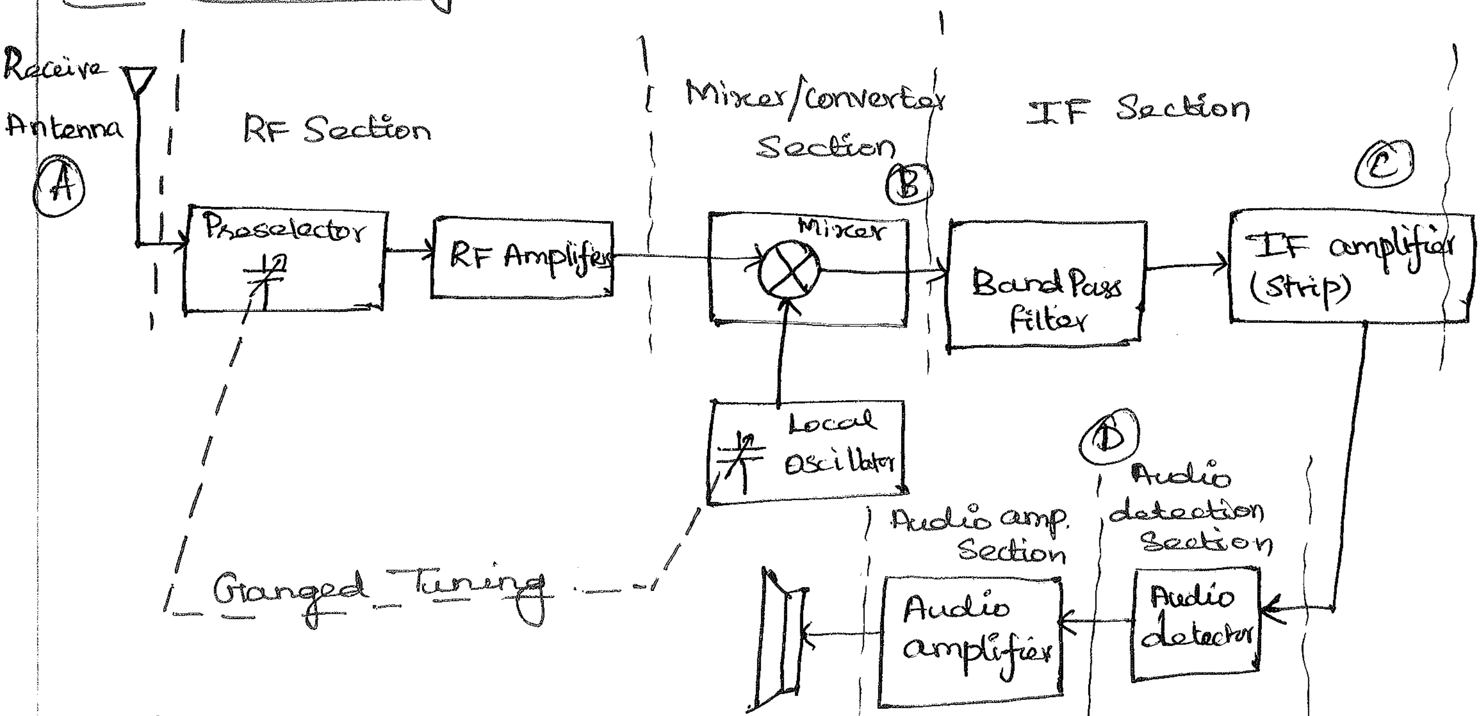
* Since a network cannot be constructed linearly constant R_oA_i and
* The balanced FM demodulator is used to remove all even harmonics, thereby reducing the distortion produced by the nonlinearity of the BPF.

* Here two demodulators are employed. One demodulator has slope α, other has $-\alpha$ slope.
* The output provided by this balanced modulator is the difference b/w the 2 individual demodulators.
$A_o' = R_oA_i + \alpha A_i(f - f_o) + \beta A_i (f-f_o)^2$; $A_o'' = A_iR_o - \alpha A_i (f-f_o) + \beta A_i (f-f_o)^2$
* The difference o/p is $A_o = 2\alpha A_i (f_o - f_o)$ and linearity is improved.
* In practical, frequency selective N/w is LC tuned circuits.
* ∴ Hardlimiter is not used in balanced FM demodulator.

Super Heterodyne Receivers:

* Here, all incoming radio frequencies are converted into a single Intermediate frequency (f_i) by the heterodyning process.
* The incoming carrier and a locally generated signal (f_l) are mixed in mixer, also referred as first detector.
* The mixer generates the sum and difference frequencies at the output.
* The difference frequency ($f_l - f_c$) is selected properly by a tuned circuit and the local oscillator frequency should be higher than (f_c) carrier. So it is called as super heterodyne receiver.
* In commercial appl'n the Intermediate frequency is fixed as 455 kHz.

AM Superheterodyne Receiver:



RF Section

* RF Section consists of preselector and an amplifier.
* Preselector is a broad band bandpass filter with an adjustable center frequency used to pass the desired RF frequency and to reject unwanted radio frequency (image frequency) and to reduce the noise ~~bandwidth~~ level. It is used to provide signal selectivity and prevent reradiation of local oscillator signal.
* RF amplifier determines the sensitivity of the receiver.

Mixer / Converter section:

→ It consists of local oscillator and Mixer.

→ The local oscillator is designed such that its frequency of oscillation is always above or below the desired RF carrier by an amount equal to the IF center frequency.

→ The adjustments for the center frequency of the preselector and the local oscillator frequency are gang tuned
(The two adjustments are tied together so that single adjustments will change the center frequency of the preselector and at the same time, change local oscillator). It is called Ganged Tuning.

→ Tracking is a process by which the local oscillator frequency follows or tracks the signal frequency to keep on correct frequency difference.

→ Super heterodyne action takes place when 2 different frequencies are mixed together.

→ Mixing involves adding and passing the result through a non linear device so that the o/p contains the product of 2 signals and 2 original signals.

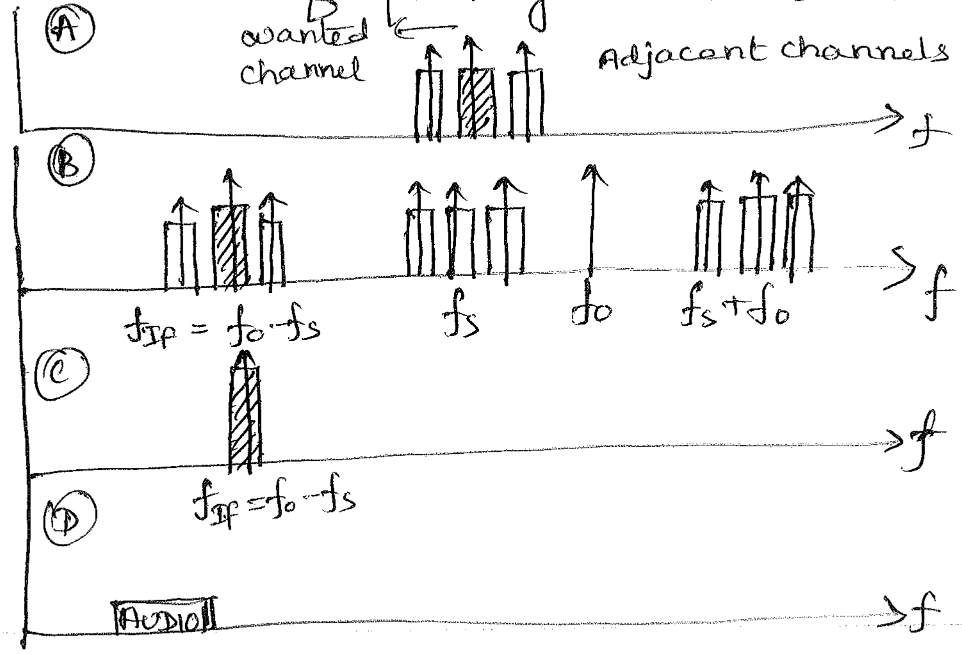
→ The product term can be separated as sum frequency and difference frequency.

→ In the frequency conversion process, the oscillator frequency may be placed above or below the signal frequency and either sum or difference frequency may be used as the output

→ For an up-conversion the sum frequency is used as output with the oscillator either above or below the signal frequency.

→ In super heterodyne receiver, usually down conversion is used where the received radio signal at frequency (f_s) is mixed with local oscillator signal (f_o) (usually above f_s) and the difference frequency produced is taken as intermediate frequency or IF as $IF = |f_o - f_s|$

Spectra



IF Section: The mixer output is fed to the IF section.

→ It consists of a series of IF amplifier and BPF to achieve most of the receiver gain and high selectivity to reject adjacent channel signals.

→ The IF is always lower than the RF because it is easier and less expensive to construct high gain, stable amplifiers for low frequency signals.

Detection Section:

→ It convert the IF signals back to the original source information.

→ It can be any demodulator circuit.

Audio Amplifier Section:

→ It consists of several cascaded amplifiers and speaker.

→ Depending on the output power, the amplifiers are cascaded.

Image Frequency:

→ An image frequency is any frequency other than the selected radio frequency carrier that will produce a cross product frequency that is equal to the intermediate frequency if allowed to enter a receiver and mix with the local oscillator.

2. For an AM DSBFC transmitter with an unmodulated power $P_c = 100W$ that is modulated simultaneously by three modulating signals with coefficients of modulation $m_1 = 0.2$, $m_2 = 0.4$ and $m_3 = 0.6$ determine.

(i) Total Co-efficient of Modulation

(ii) Upper and lower Sideband Power.

(iii) Total transmitted Power.

<u>Solution:</u> Given $P_c = 100W$, $m_1 = 0.2$, $m_2 = 0.4$, $m_3 = 0.6$

(i) Total Co-efficient of Modulation $m = \sqrt{m_1^2 + m_2^2 + m_3^2}$

$$= \sqrt{0.2^2 + 0.4^2 + 0.6^2}$$

$$\boxed{m = 0.748.}$$

172, 190

(ii) <u>Upper and Lower Sideband Power.</u>

$$P_{SB} = P_{LSB} + P_{USB} = \frac{m^2}{2} P_c = \frac{(0.75)^2}{2} 100.$$

$$\therefore \quad P_{SB} = \underline{28.12\ W}.$$

$$P_{USB} = P_{LSB} = \frac{m^2}{4} P_c = \frac{(0.75)^2}{4} \times 100 = \underline{14.06\ W}.$$

(iii). <u>Total Transmitted Power</u>

$$P_t = \left[1 + \frac{m^2}{2}\right] P_c = \left[1 + \frac{0.75^2}{2}\right] \times 100$$

$$\boxed{P_t = 128\ W}.$$

3). A 25MHz carrier is modulated by a 400Hz audio sine wave. If the carrier voltage is 4V and the maximum frequency deviation is 10kHz and the phase deviation is 25 radians. Write the equation of this modulated wave for (i) AM (ii) FM. If the modulating frequency is now changed to 2KHz, all else remaining constant. Write a new equation for FM and PM.

Solution:

Given, $f_c = 25MHz$, $f_m = 400Hz$, $V_c = 4V$.

freq. Deviation, Δf = 10kHz ; Phase deviation (m_p) = 25 Radians.

$$\therefore \quad m_f = \frac{\Delta f}{f_m} = \frac{10k}{400} = 25.$$

$\therefore$ FM signal $s(t) = A_c \cos[\omega_c t + m_f \sin\omega_m t]$

$$s_{FM}(t) = 4 \cos[157 \times 10^6 t + 25 \sin 2.5 \times 10^3 t]$$

PM signal $s_{PM}(t) = A_c \cos[\omega_c t + m_p \cos\omega_m t]$.

$$= \overset{4}{\cancel{A_c}} \cos[2\pi \times 25 \times 10^6 t + 25 \cos 2\pi \times 400 t].$$

If f_m is changed to 2kHz,

$$S_{FM}(t) = 4 \cos[157 \times 10^6 t + 25 \sin(4\pi \times 10^3 t)].$$

$$S_{PM}(t) = 4 \cos[157 \times 10^6 t + 25 \cos(4\pi \times 10^3 t)].$$

Pb:1

A modulating signal $15\sin(2\pi\times10^3t)$ is used to modulate a carrier signal $30\sin(2\pi\times10^4t)$. Find the modulation index, percentage modulation, frequencies of the side band components and their amplitudes. What is the bandwidth of the modulated signal? Draw the spectrum of the AM wave.

Solution:

Given, The modulating signal $V_m(t) = 15\sin(2\pi\times10^3t)$.

$\therefore\ E_m = 15\ ;\quad f_m = 1\times10^3\,Hz = 1\,KHz.$

The carrier signal $V_c(t) = 30\sin(2\pi\times10^4t)$.

$\therefore\ E_c = 30\ ;\quad f_c = 1\times10^4\,Hz = 10\,KHz.$

(i) Modulation Index: $m_a = \dfrac{E_m}{E_c} = \dfrac{15}{30} = 0.5$

(ii) Percentage Modulation: $M = m_a\times100 = 50\%.$

(iii) SideBand Frequencies: $f_{USB} = f_c + f_m = (10+1)\,KHz$

$f_{USB} = 11\,KHz.$

$f_{LSB} = f_c - f_m = (10-1)\,Hz$

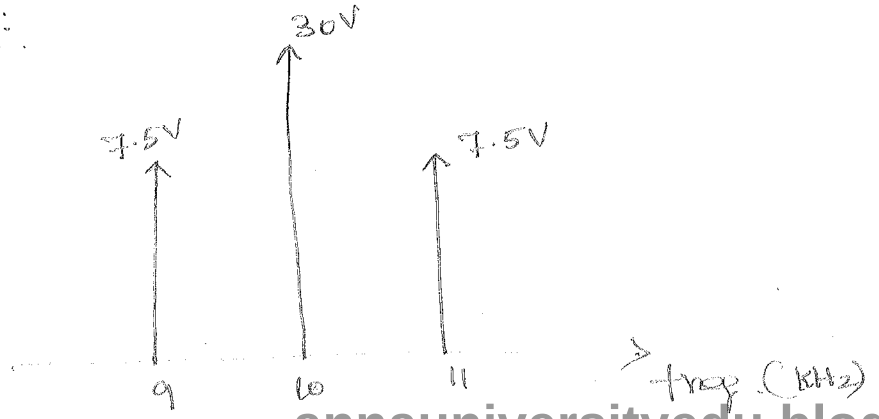
$f_{LSB} = 9\,KHz.$

Amplitude of SB: $V_{LSB} = \dfrac{m_a E_c}{2}$

$= \dfrac{0.5\times30}{2} = 7.5V.$

(iv). Bandwidth $BW = 2f_m = 2\times1\,KHz = 2\,KHz.$

(v). Spectrum:



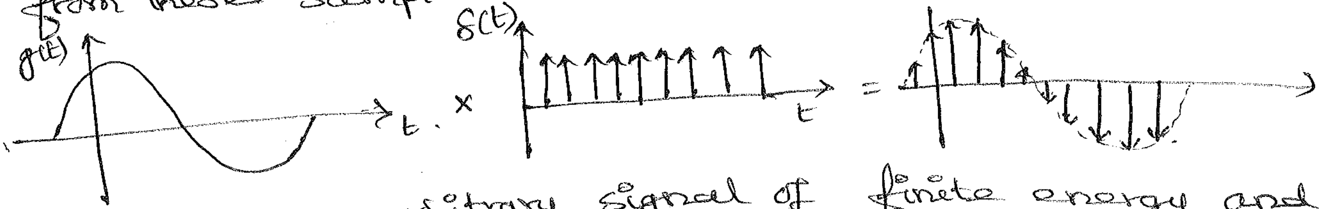
UNIT - II. PULSE MODULATION.

Introduction:

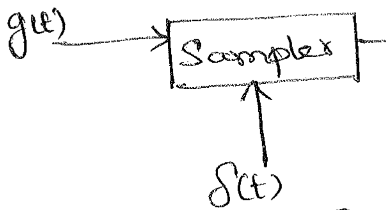
* In pulse modulation and digital modulation systems, the signal to be transmitted must be in discrete time form.
* The message signal is analog in nature (speech or video signal). So, the analog signal should be converted to discrete signal using sampling process
* Thus, sampling is the process of converting a continuous analog signal to a discrete signal.
* The sampled signal is the discrete representation of original analog signal.

Sampling Theorem for Low pass Signals (Shannon's Sampling Theorem).

Stmt: Let $g(t)$ be a signal which is bandlimited such that its highest frequency spectral component is f_M. Let the values of $g(t)$ be determined at regular intervals separated by times $T_s \le 1/2f_m$ that is the signal is periodically sampled every T_s seconds. Then these samples $g(nT_s)$, where n is an integer, uniquely determine the signal, and the signal may be reconstructed from these samples with no distortion.



* Let $g(t)$ be an arbitrary signal of finite energy and impulse train is represented by $\delta(t)$.
* $g(t)$ sampled at uniform rate T_s denoted as $g(nT_s)$
  T_s - Sampling period $f_s = \frac{1}{T_s} \Rightarrow$ Sampling rate.
* The minimum sampling rate is called as nyquist rate given as $f_s = 2f_M$. and nyquist interval is $T_s = 1/2f_M$



∴ The sampled signal is represented as $g_s(t) = g(t).\delta(t)$

We know, $\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$ is the delta function positioned at time $t = nT_s$.

$$\therefore \quad g_s(t) = \sum_{n=-\infty}^{\infty} g(t) \, . \, \delta(t-nT_s) = \sum_{n=-\infty}^{\infty} g(nT_s) . \delta(t-nT_s) \quad \text{——} (1)$$

where $g(nT_s)$ is the instantaneous amplitude of $g(t)$

Taking Fourier Transform to (1),

We get $\quad G_s(f) = F.T[g(t)] * F.T\left[\sum_{n=-\infty}^{\infty} \delta(t - nT_s)\right].$

$$= G(f) * \frac{1}{T_s}\sum_{n=-\infty}^{\infty} \delta(f - nf_s).$$

$$= f_s \sum_{n=-\infty}^{\infty} G(f) * \delta(f - nf_s).$$

Applying Shifting Property to impulse response,

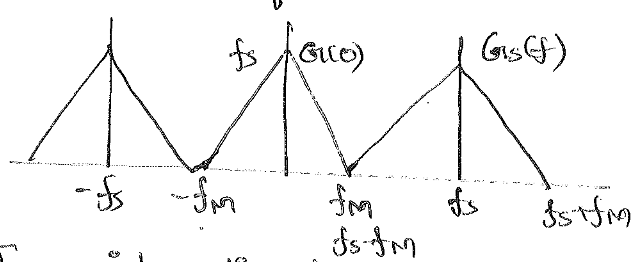
$$G_s(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s).$$

Expanding $\quad G_s(f) = f_s G(f) + f_s G(f \pm f_s) + f_s G(f \pm 2f_s) + \dots$

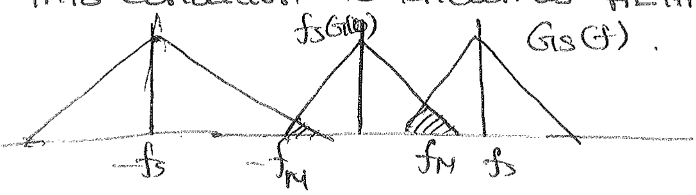
$$\therefore \quad G_s(f) = f_s G(f) + \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} f_s G(f - nf_s) \quad \text{———} (2)$$

1^{st} term of (2) indicates the spectrum without sampling

2^{nd} term of (2) indicates the spectrum repeating at multiple frequencies of sampling freq. f_s.



__Note:__ If $f_s < 2f_M$, the Samples overlap with each other. This condition is known as ALIASING.



__To avoid aliasing:__

* Low pass antialiasing filter is used prior to sampling to attenuate the high frequency components.
* When $f_s > 2f_M$, there is no aliasing. and when the signal is reconstructed it is made faithfully.

Quantization:

Def: Quantization is the process of Converting an infinite number of amplitude possibilities to a finite number of condition.

* Quantizing a signal $m(t)$ to $m_q(t)$ is an approximation to $m(t)$.

Operation of Quantization:

* Consider a signal $m(t)$ with the range from V_L to V_H.
* Divide the ~~total~~ range to M equal Intervals each of size S, called step size. (ie) $S = (V_H - V_L)/M$.
* For eg, take $M=8$. At the center of each level, locate quantization levels $m_0, m_1, \ldots m_7$.
* Whenever $m(t)$ is in the range of Δ_0, $m_q(t)$ maintains in m_0. whenever $m(t)$ is in the range of Δ_1, $m_q(t)$ maintains m_1. and so on.
* when $m_q(t)$ changes from m_0 to m_1, there is an abrupt change in the level when $m(t)$ passes L_{01} which is midway b/w m_0 and m_1.
* ~~There~~ At every instant of time, $m_q(t)$ has the value of quantization level to which $m(t)$ is closest.
* ∴ The quantized signal is an approximation to original signal.
* The quality of approximation is improved by reducing the step size.

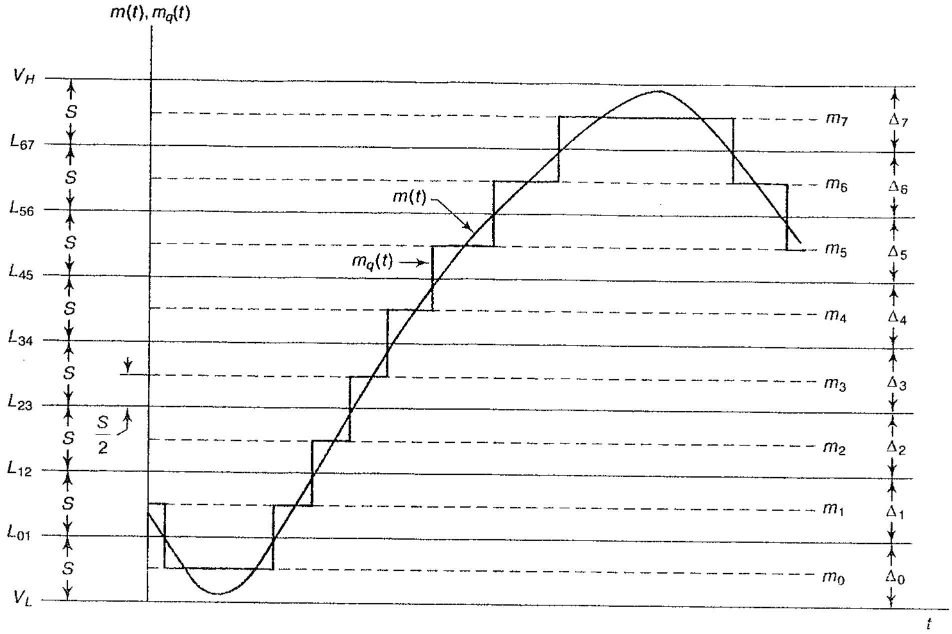


Fig. 5.21 The operation of quantization.

Quantization Error:

The difference or error between the quantized signal and the original signal viewed as a noise due to the quantization process is called quantization error.

(ie) $e = m(t) - m_k$, where $m(t)$ – Instantaneous value of original signal.
m_k – quantizer output.

Mean square quantization error $\overline{e^2} = \dfrac{S^2}{12}$. S – Step size.

Pulse Modulation:

* In pulse modulation System, the carrier is train of discrete pulses rather than continuous time signal (Sine sig).
* Pulse modulation consists of sampling the analog information signals and then converting those signals into discrete pulses and transmitting the pulses from source to a destination over a physical transmission medium or channel.

Types of Quantization:

* There are two types of Quantization.
  (i) Uniform Quantization
  (ii) Non Uniform Quantization.

Uniform Quantization:

* Here the quantization levels are uniformly spaced.
* There are two types of uniform Quantization.
  i) Mid Rise ii) Mid Thread type.

Mid Rise

* Here, origin lies in the middle of a raising part of the stair case like graph.
* The quantization levels are even in number.

Mid Tread

* The origin lies in the middle of the ~~stair~~ tread of the stair case like graph.
* The quantization levels are odd in number.

Note: Both Mid rise and Mid Tread are symmetric about origin

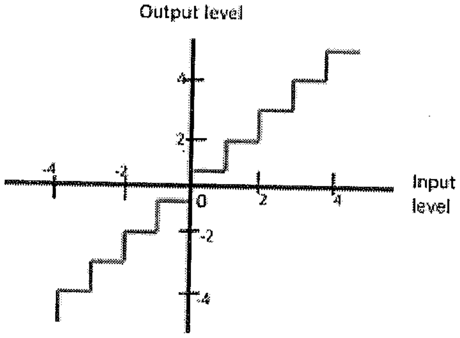


Fig 1 : Mid-Rise type Uniform Quantization

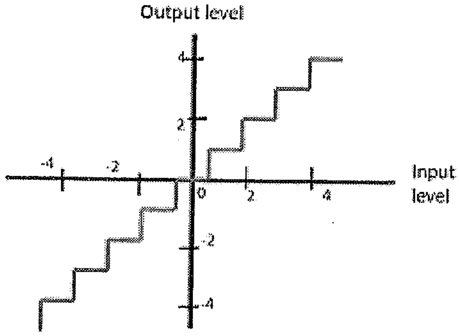


Fig 2 : Mid-Tread type Uniform Quantization

Non - Uniform Quantization:

* The quantization levels are unequal and mostly the relation between them is logarithmic.

Companding:

Compression + expansion ⇒ Companding.

* This is a non linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver.
* The effect of noise and crosstalk are reduced by this technique.
* There are two types of companding Technique.

μ Law Companding: (Used in North America, Japan).

$$|V| = \frac{\log_e(1+\mu|m|)}{\log_e(1+\mu)}$$

* Uniform quantization is achieved at $\mu = 0$, where the characteristic curve is linear and no compression is done.
* μ-Law has mid tread at the origin.
* μ-law Companding is used for speech and music signals.

A Law Companding:

* At $A=1$, uniform quantization is achieved.
* A law has mid-rise at the origin
* A law Companding is used for PCM telephone systems.

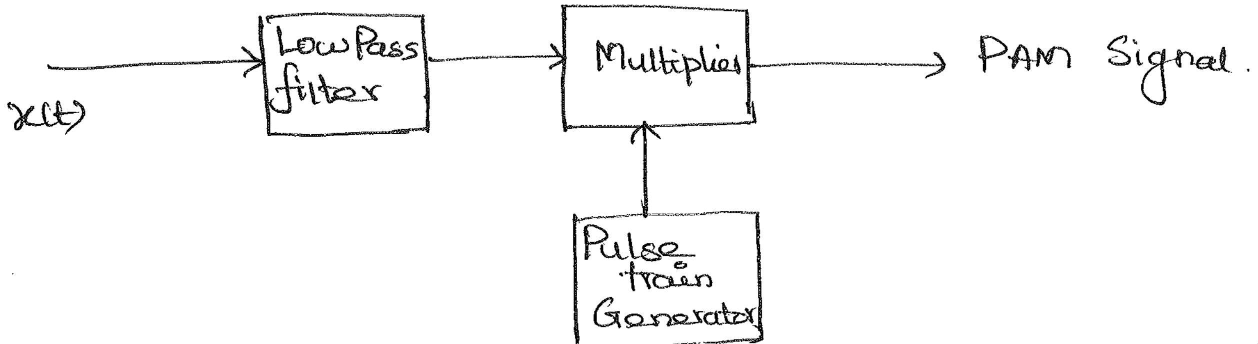
$$|V| = \begin{cases} \dfrac{A|m|}{1+\log_e A}, & 0 \le |m| \le \dfrac{1}{A} \\[2ex] \dfrac{1+\log_e(A|m|)}{1+\log_e A}, & \dfrac{1}{A} \le |m| \le 1. \end{cases}$$

Pulse Amplitude Modulation:

The amplitude of a constant width, constant position pulse is varied with respect to the amplitude of the sample of the modulating signal.

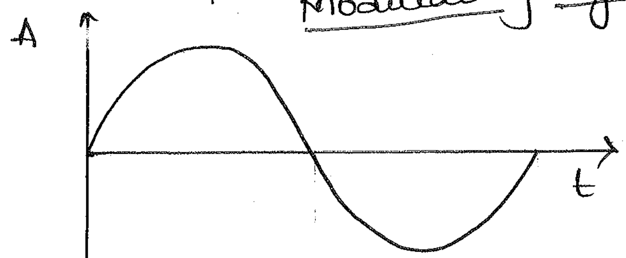
Generation of PAM:

(i) Natural PAM:

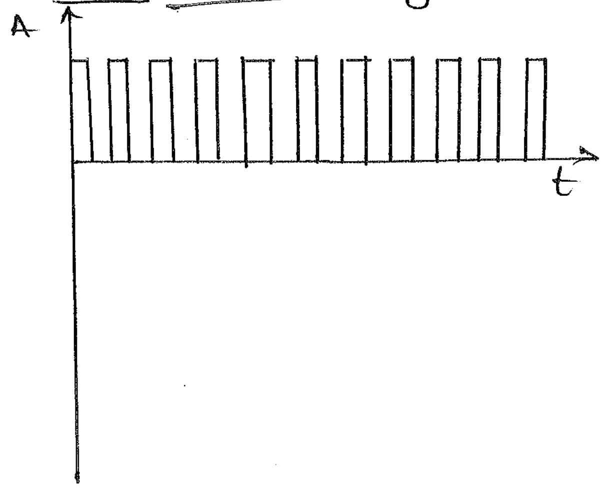


* The continuous analog signal x(t) is passed through a Lowpass filter. The LPF will bandlimit the signal to f_m.
* The pulse train generator generates a pulse train of frequency f_s.
* The uniform sampling takes place and producing the PAM waveform.
* The information is carried over the Amplitude variation of the pulsed carrier

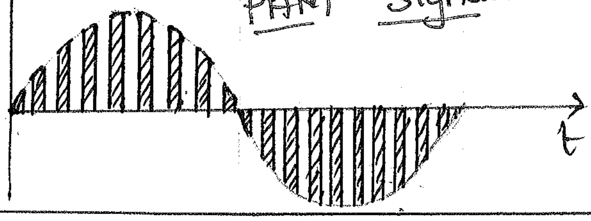
Modulating Signal



Pulsed Carrier Signal.

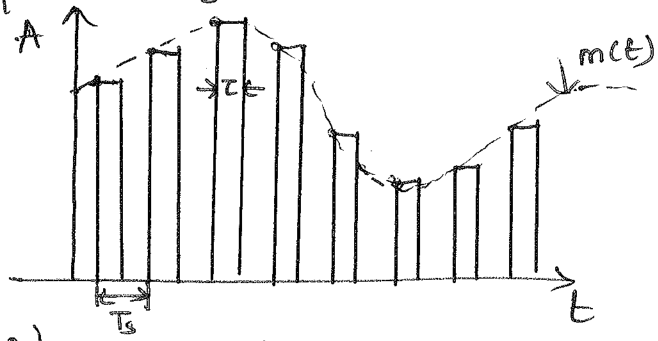


PAM Signal.

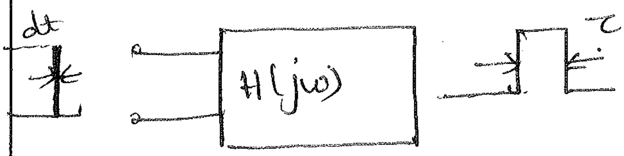


Flat Top PAM:

* A flat topped pulse has a constant amplitude ~~value~~ established by the sample value of the signal at some point within the pulse interval.
* It simplifies the circuit design.
* The flat top pulse can be generated by passing the instantaneously sampled signal through a n/w which broadens a pulse of duration (dt) into a pulse of duration τ.

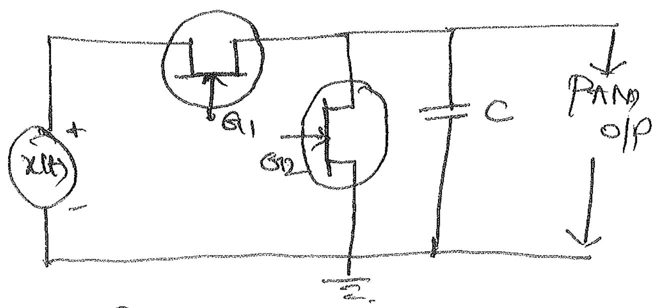


PAM Signal.



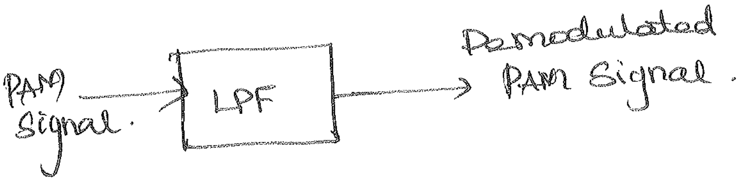
$$H(j\omega) = \frac{\tau}{dT}\,\frac{\sin(\omega\tau/2)}{\omega\tau/2}.$$

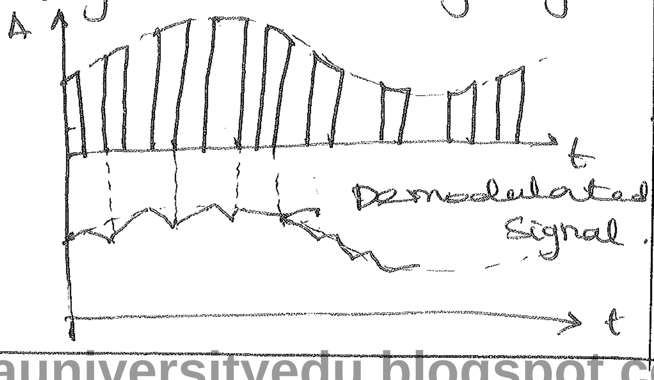
* Usually Flat Top signal is preferred over Natural PAM
* The output of Sample & Hold circuit contains sequence of flat top samples.



Detection of Natural PAM Signals:

* The PAM signal is passed through LPF whose cut-off frequency is adjusted to fm. Hence all the frequency ripple is removed and the original modulating signal is reconstructed back.





Line Coding:

* Analog data can be represented in binary ie 0 and 1.
* Line coding helps to assign electrical voltages to binary 0, 1. that suits communication over an electrical line.

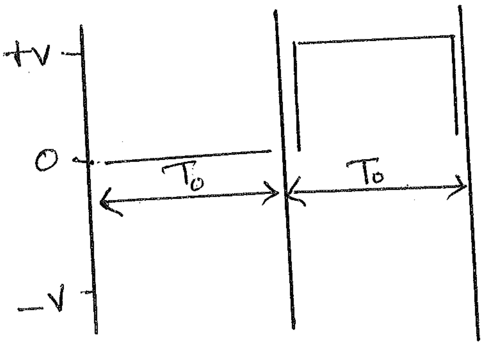
Factors of Line Codings:

(i) Power and bandwidth required for transmission.
(ii). Ability to extract timing information.
(iii) Presence of low frequency or dc component which is not suitable for ac coupled circuits.
(iv) Error monitoring Ability.

Line Code
- Linear Code : I/P data stream is continuously processed to generate o/p code.
- Block Code : I/P data stream is divided into blocks, each of which is coded to a symbol or another block of data.

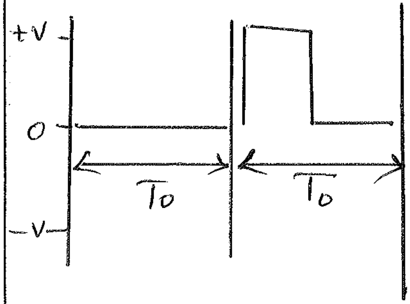
Unipolar NRZ:

* logic 0 : 0V ; logic 1 : +V - constant signal level during its entire bit interval T_0. bit rate is $1/T_0$.



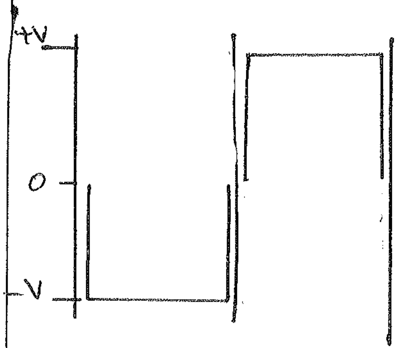
Unipolar RZ

* logic 0 - 0V ; logic 1 - Constant Voltage level (+V). But logic 1 pulse ~~is~~ returns to zero after a brief period (ie half bit period).



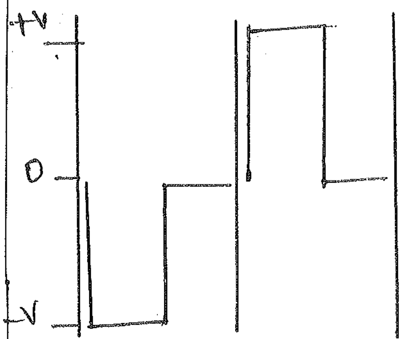
Bipolar NRZ :

* Since the Code relay on +V and -V, the code is known as bipolar.
* Here, pulses don't return to zero and stays at that level for entire bit duration.



Bipolar RZ :

* Here, both logic 1 and 0 are ~~p~~ represented by pulses that returns to zero within the bit interval.
* Instead of bipolar, polar is used.



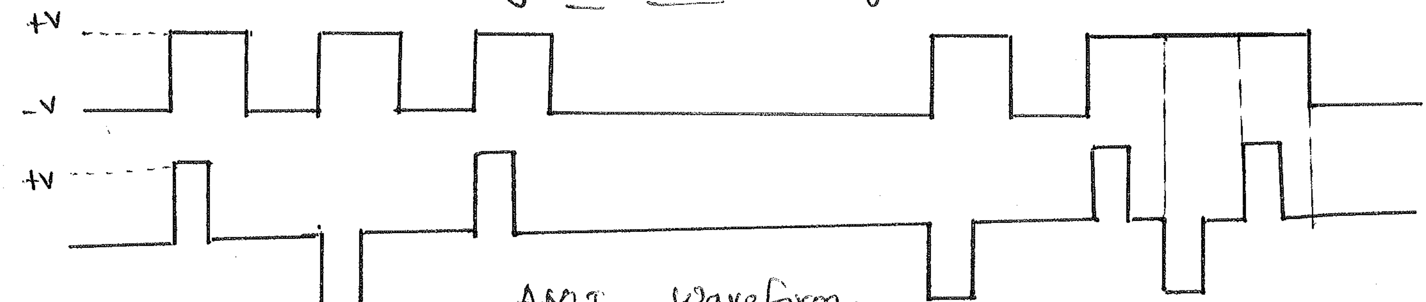
Alternate Mark Inversion : (AMI).

logic 0 : 0V for entire bit interval.

logic 1 : +V or −V for a fraction of bit interval.
(50% duty cycle).

* Here, Two Successive 1's whether occur in neighbouring bit or separated by 0's are represented by pulses of opposite polarity. Such a waveform is known as AMI.

AN NRZ waveform.



AMI Waveform.

High Density Bipolar Code : (HDB)

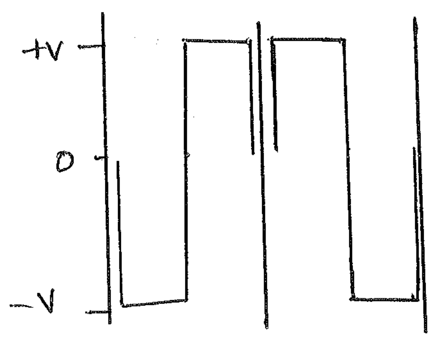
* When to strings of 8 zero appear in AMI Code, the 1's are inserted that they generate violations (ie) they do not alternate

eg: HDB − 3 code, More than 3 zeros in a string, 4th one is replaced by a violation pulse.

Split Phase (Manchester)

logic 0 : negative pulse followed by a positive pulse each of half bit duration.

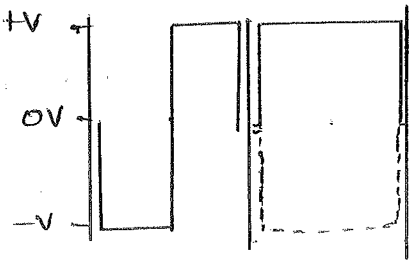
logic 1 : positive pulse followed by a negative pulse each of half bit duration.



Coded Mark Inversion Code : (CMI)

→ This is a two level NRZ code.

* logic 0 : negative pulse followed by a positive pulse, each of half bit duration.
* logic 1 : Represented alternatively by +V and -V for full bit interval.



2B1Q :- 2 Binary digit encoded into 1 Quaternary Symbol.

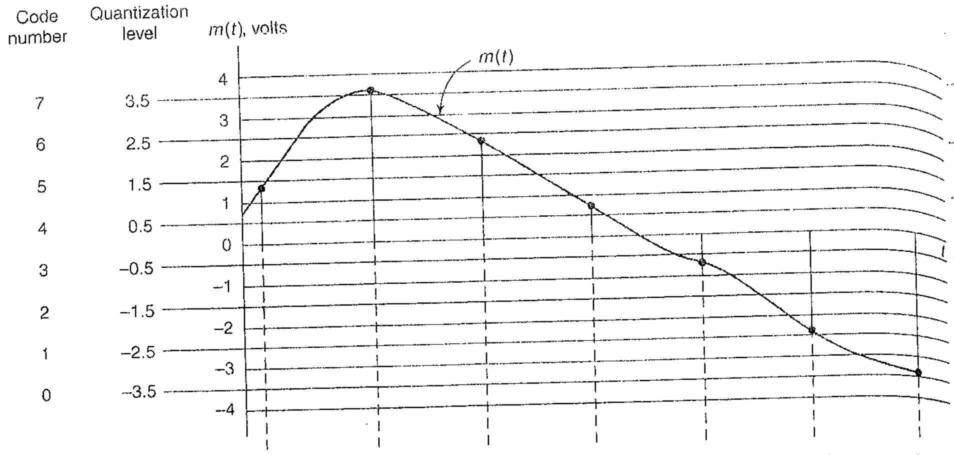
* This is a block code where block size is 2.

ie 00 → -3V ; 01 → -1V ; 10 → +1V ; 11 → +3V.

* This results to 4 level PAM signal having zero dc component.
* Adv : less bandwidth, better performance w.r.t ISI, cross talk compared to linear code.

PCM : Pulse Code Modulation :

* In PCM, the analog signal is sampled and then quantized. The quantized levels are represented by the code numbers.
* The code numbers are converted to binary arithmetic ie base 2 arith matic.
* The digits of the binary representation of the code number are transmitted as pulses. Thus the system of transmission is called (binary) Pulse Code Modulation.



Sample value	1.3	3.6	2.3	0.7	−0.7	−2.4	−3.4
Nearest quantization level	1.5	3.5	2.5	0.5	−0.5	−2.5	−3.5
Code number	5	7	6	4	3	1	0
Binary representation	101	111	110	100	011	001	000

Fig. 5.24 A message signal is regularly sampled. Quantization levels are indicated. For each sample the quantized value is given and its binary representation is indicated.

* The analog signal is represented as m(t) with the Voltage range −4 to 4
* the step size is 1V ∴ 8 quantization levels are presented −3.5, −2.5, −1.5 to 3.5.
* The level −3.5 is given with the code number 0, −2.5 as 1, −1.5 as 2 and so on.
* Each code number can be represented in binary ranging from 000 to 111. [∵ the level is 8].
* In Correspondance to each sample, the binary representation is specified and it is transmitted as pulses. (ie). 101 111 110 100 011 001 000.

The Encoder:

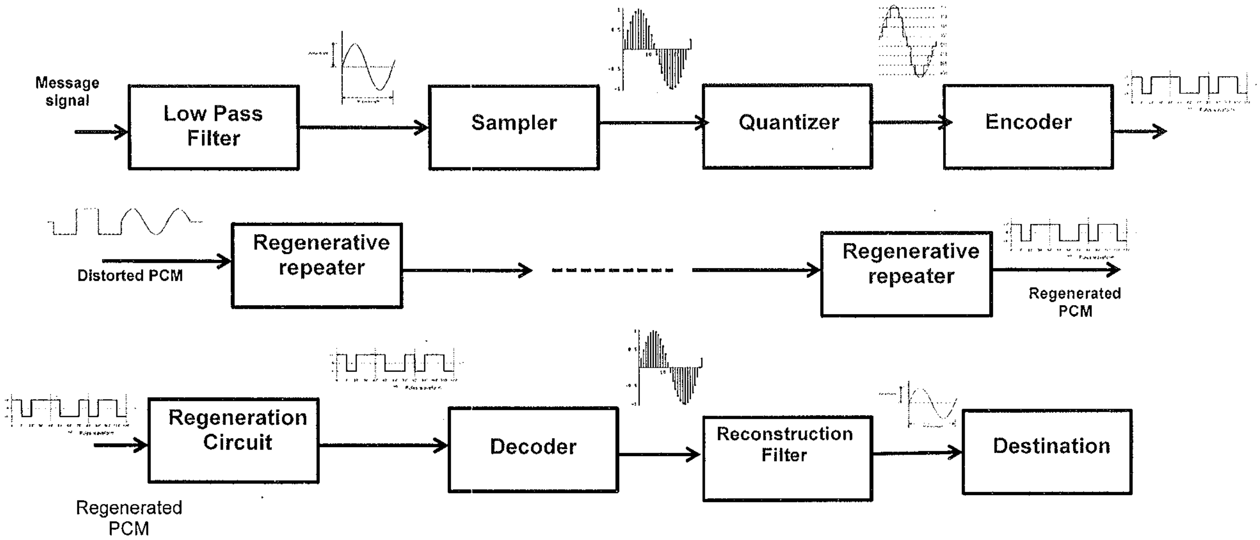
* The analog signal m(t) is sampled and these samples are subjected to the operation of quantization.
* The quantized samples are applied to an encoder
* The encoder responds to each sample by the generation of a unique and identifiable binary pulse pattern.
* The combination of the quantizer and encoder is ~~this~~ called an analog to digital converter.
* The A/D converter accepts an analog signal and replaces it with a succession of code symbols
* Each symbol consisting of a train of pulses in which each pulse may be interpreted as the digit in arithematic system.
* Thus, the signal transmitted over the communication channel in a PCM system is referred as digitally encoded signal.

The Decoder:

* When the digitally encoded signal arrives at the receiver, quantization has to be done to seperate the signal from the noise whis has been added during the transmission along the channel.
* The receiver quantizer decides whether a positive pulse or a negative pulse was received and transmits ~~the~~ its decisions, in the form of a reconstituted or regenerated pulse train, to the decoder.
* The decoder also called D/A converter performs the inverse operation of encoder.
*

(7)

PULSE CODE MODULATION (PCM System)



Low Pass Filter:

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing

Sampler:

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component $\mathbf{f_M}$ of the message signal, in accordance with the sampling theorem.

Quantizer:

Quantizing is a process of Converting analog sample of the signal into a discrete form. Resultant signal is discrete both in time and amplitude. These three sections (LPF, Sampler, and Quantizer) will act as an analog to digital converter.

Encoder:

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The electrical representation of a code is done by assigning a waveform (or a pulse) in the encoder. The output of the encoder is the PCM wave. Encoding minimizes the bandwidth used

Regenerative Repeater:

This section increases the signal strength. The output of the channel has regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

The Output of the regenerative repeater is the reconstructed signal

This regenerated PCM is given to the receiver. The most important feature of PCM system lies in the ability to control the effects of distortion and noise produced by transmitting PCM wave through channel.

Decoder:

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter:

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal

* The decoder o/p is the sequence of quantized multilevel sample pulses.
* The quantized PAM signal is now reconstituted.
* It is then filtered to reject any frequency components lying outside of the baseband.
* The final output signal m'(t) is identical with the input m(t).

Differential Pulse Code Modulation: (DPCM).

* In a PCM System, sampled signal exhibit a high correlation between adjacent samples since the signal does not change rapidly from one sample to the next.
* When theses correlated samples are encoded, the resulting encoded signal contains redundant information.
* The most efficient coded signal can be obtained by removing this redundancy before encoding.
* Here, we use linear prediction to remove the reduntant information.

→ Let x(t) be the baseband signal and it is sampled at the rate $f_s = 1/T_s$ and the sequence be denoted as $x(nT_s)$

→ It is possible to predict future values of the signal x(t) and provides hope for differential quantization.

→ The i/p to the quantizer is given by

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \quad \text{(1)}$$ where

$x(nT_s)$ — unquantized i/p signal.

$\hat{x}(nT_s)$ — prediction of $x(nT_s)$

$e(nT_s)$ — Prediction error

→ Prediction of $x(nT_s)$ is done by a predictor whose input consists of a quantized version of the the i/p signal $x(nT_s)$
Prediction error is the amount by which the predictor fails to predict the input exactly.
The input output characteristics of the quantizer is $Q[\cdot]$.
→ The quantizer output can be represented as

$$v(nT_s) = Q[e(nT_s)].$$

$$= e(nT_s) + q(nT_s) \quad \text{--- (2)}$$ where

$q(nT_s)$ – quantization error.

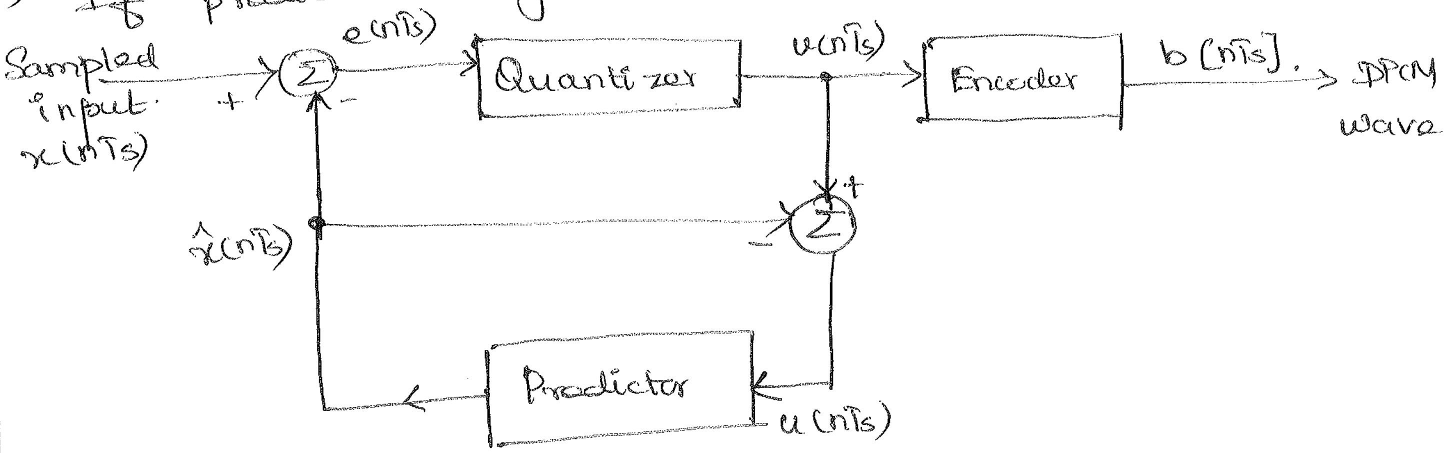
→ The predictor input $u(nT_s) = \hat{x}(nT_s) + v(nT_s)$. ——— (3).

Sub (2) in (3) ⇒ $u(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$

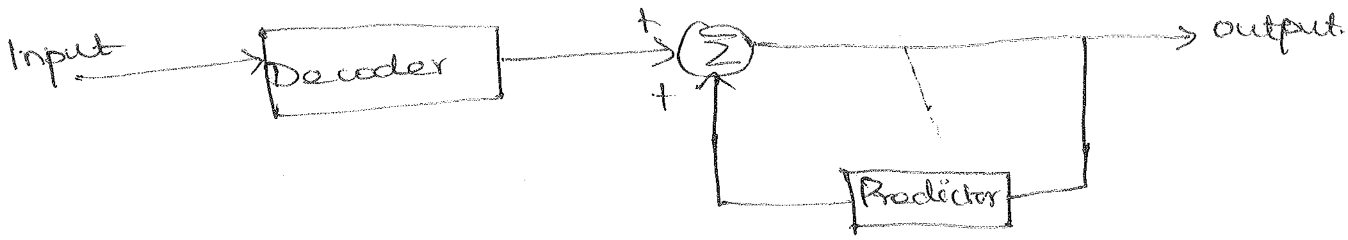
From (1) ⇒ $u(nT_s) = x(nT_s) + q(nT_s)$. ——— (4).

→ The equation (4) says that the quantized signal $u(nT_s)$ at the predictor input differs from the original i/p signal $x(nT_s)$ by the quantization error.

→ If prediction is good, $e(nT_s)$ is smaller.



DPCM Transmitter

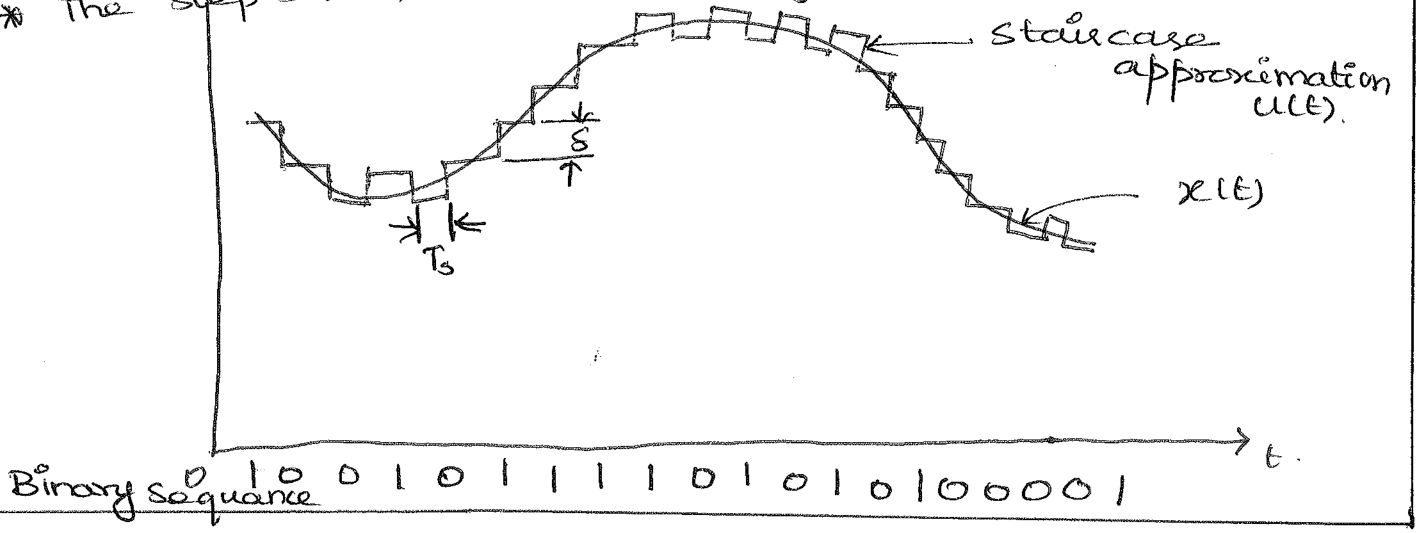


DPCM Receiver.

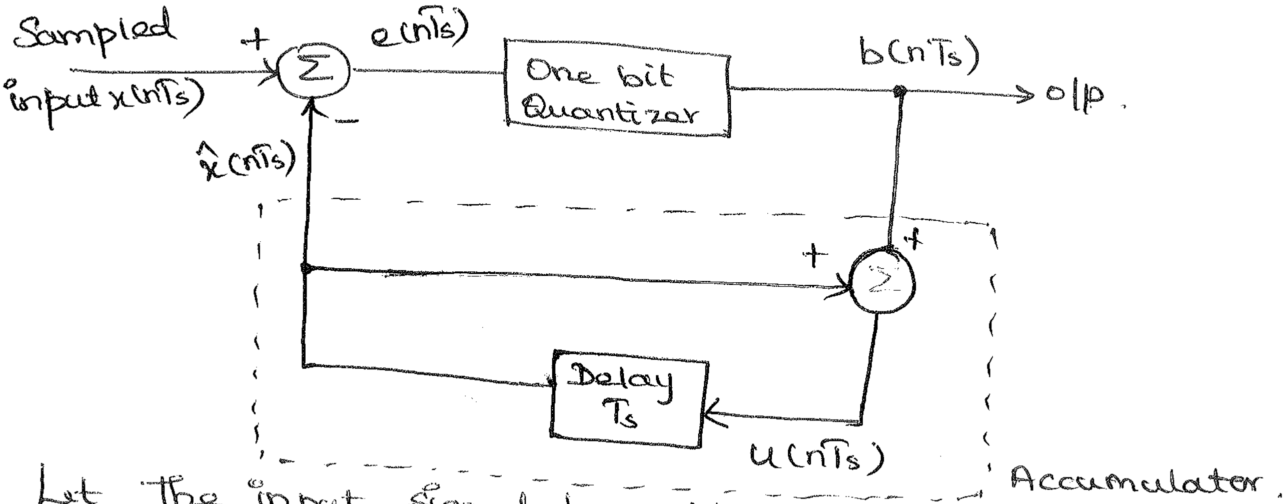
* The receiver for reconstructing the quantized version of the i/p consists of a decoder to reconstruct the quantized error signal.
* The original signal is reconstructed from the decoder using the same predictor used in the transmitter.
* The output of receiver is equal to $u(nT_s)$, which differs from the original i/p $x(nT_s)$ by $q(nT_s)$ as a result of quantizing the prediction error $e(nT_s)$.

Delta Modulation:

* Delta Modulation is a one bit version of DPCM.
* DM provides a staircase approximation to the over-sampled version of an input baseband signal.
* The difference between the input and the approximation is quantized into only two levels, namely $+\delta$ and $-\delta$, corresponding to positive and negative differences respectively.
* Thus, if the approximation falls below the signal at any sampling, it is increased by $+\delta$.
* If the approximation lies above the signal, it is decreased by δ.
* If the signal does not changes too rapidly from sample to sample, the staircase approximation remains within $\pm\delta$ of the input signal.
* The step size(Δ) is related δ by $\Delta = 2\delta$.



DM Transmitter:



Let the input signal be $x(t)$ and the staircase approximation as $u(t)$.

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

where $e(nT_s)$ is prediction error, T_s – Sampling Period.

$$= x(nT_s) - u(nT_s - T_s)$$

$$b(nT_s) = \delta \,\mathrm{Sgn}\,[e(nT_s)].$$

$$u(nT_s) = u(nT_s - T_s) + b(nT_s)$$

$$u(nT_s) = \delta \sum_{i=1}^{n} \mathrm{Sgn}[e(iT_s)]$$

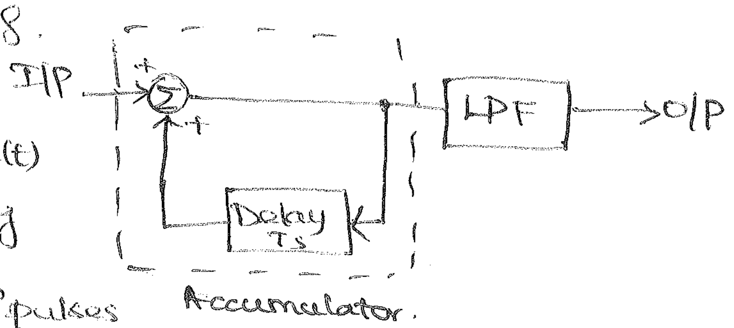
$$= \sum_{i=1}^{n} b(iT_s).$$

where $b(nT_s)$ is the algebric sign of the error $e(nT_s)$ except for the scaling factor δ.

$b(nT_s)$ is the one bit word transmitted by DM.

* DM Transmitter Consists of Summer, accumulator and a 2 level quantizer. Accumulator is initialized to zero.
* At each sampling instant, the accumulator approximates the input signal to $\pm\delta$.

DM Receiver:



* The staircase approximation $u(t)$ is reconstructed by passing the incoming sequence of positive and negative ~~sequence~~ pulses through an accumulator.
* The quantization noise is removed by passing it through Low Pass Filter whose BW equal to original message bandwidth.

Delta modulation subjects to two quantization error.

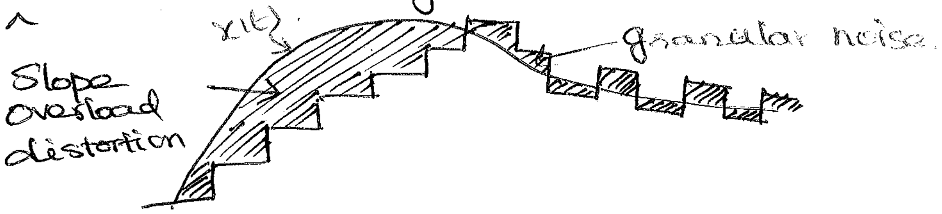
(i) <u>Slope Overload distortion</u>:

→ When the step size $\Delta = 2\delta$ is too small for the staircase approximation $u(t)$ to follow a steep segment of the input waveform $x(t)$, $u(t)$ falls behind $x(t)$.

→ This condition is called slope overload and the resulting quantization error is called slope overload distortion.

(ii) <u>Granular noise</u>:

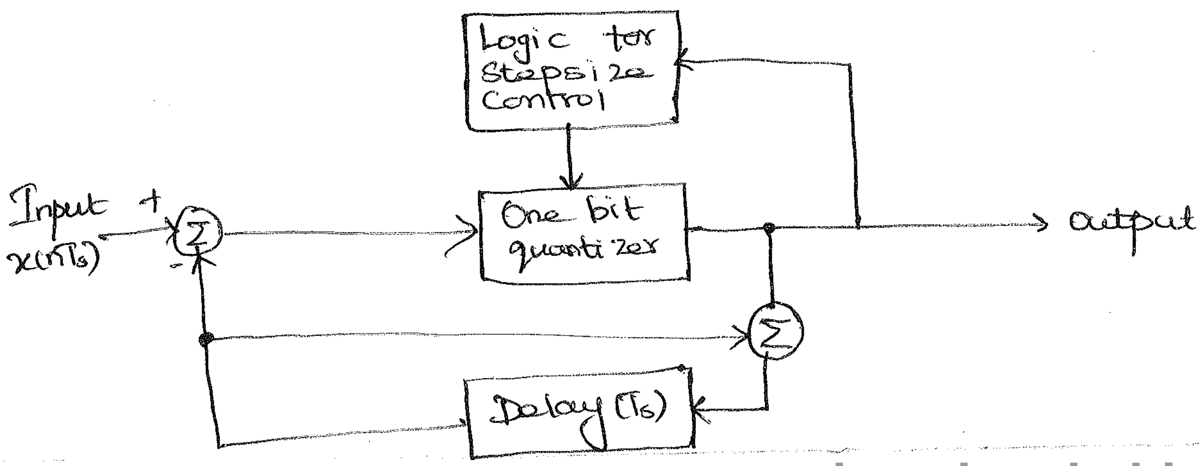
If the step size Δ is too large to the local input characteristics of the input waveform $x(t)$, $u(t)$ falls around the flat segment of the input waveform. This phenomenon is known as granular noise.



Adaptive Delta Modulation:

* In ADM, during a steep segment of the input signal, step size is increased. When the input signal is varying slowly, the stepsize is reduced.
* In short, the step size of the ADM is time varying (adjustable) one.
* Since the step size is adapted to the level of the input signal, the resulting method is called adaptive delta modulation.

∴ ADM Transmitter:



* Here, the stepsize $\Delta(nT_s)$ or $2\delta(nT_s)$ is constrained to lie between minimum and maximum values.

* We can write $\delta_{min} \le \delta(nT_s) \le \delta_{max}$

* δ_{max} controls the amount of slope overload distortion. δ_{min} controls the amount of idle channel noise.

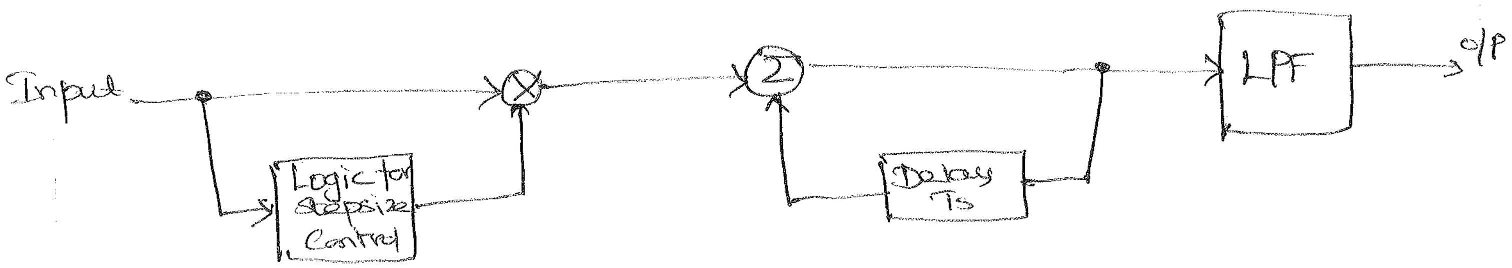
* The adaptation rule for $\delta(nT_s)$ is expressed in the general form $\delta(nT_s) = g(nT_s)\,\delta(nT_s - T_s)$, where $g(nT_s)$ depends on the present binary output $b(nT_s)$ of the delta modulator and M previous values $b(nT_s - T_s) \ldots b(nT_s - MT_s)$, and initiated with $\delta_{start} = \delta_{min}$.

ie $g(nT_s) = \begin{cases} K & \text{if } b(nT_s) = b(nT_s - T_s) \\ K^{-1} & \text{if } b(nT_s) \neq b(nT_s - T_s) \end{cases}$

* This adaptation algorithm is called constant factor ADM with one bit memory.

* One bit memory refers to explicit utilization of the single previous bit $b(nT_s - T_s)$.

ADM Receiver :



A digital coding scheme that uses both adaptive Quantization and adaptive prediction is called Adaptive Differential Pulse Code Modulation (ADPCM (11)

ADPCM

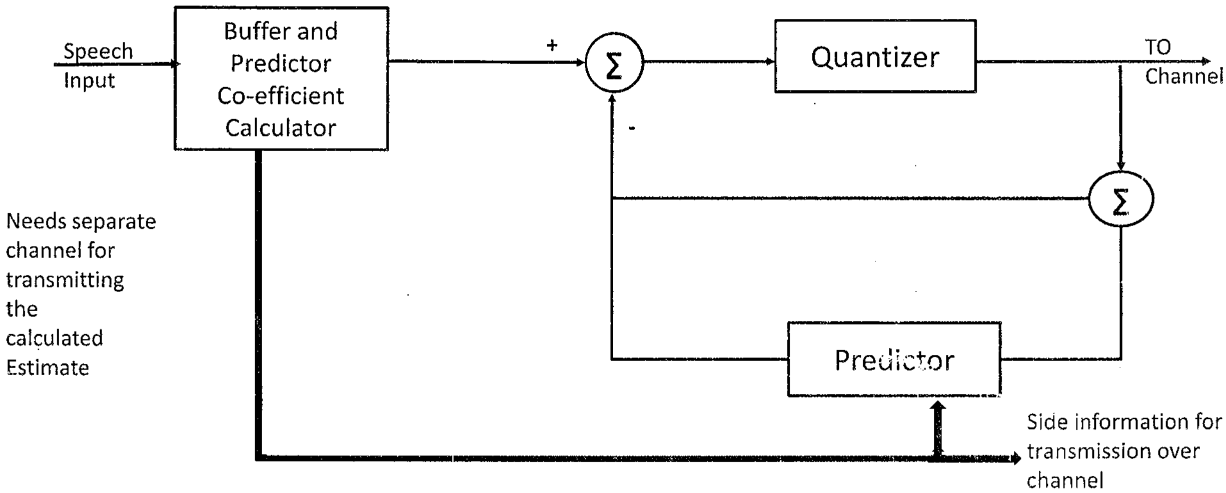
Adaptive Prediction should be used in ADPCM as speech signals are inherently non stationary.

There are two types of Adaptive Prediction

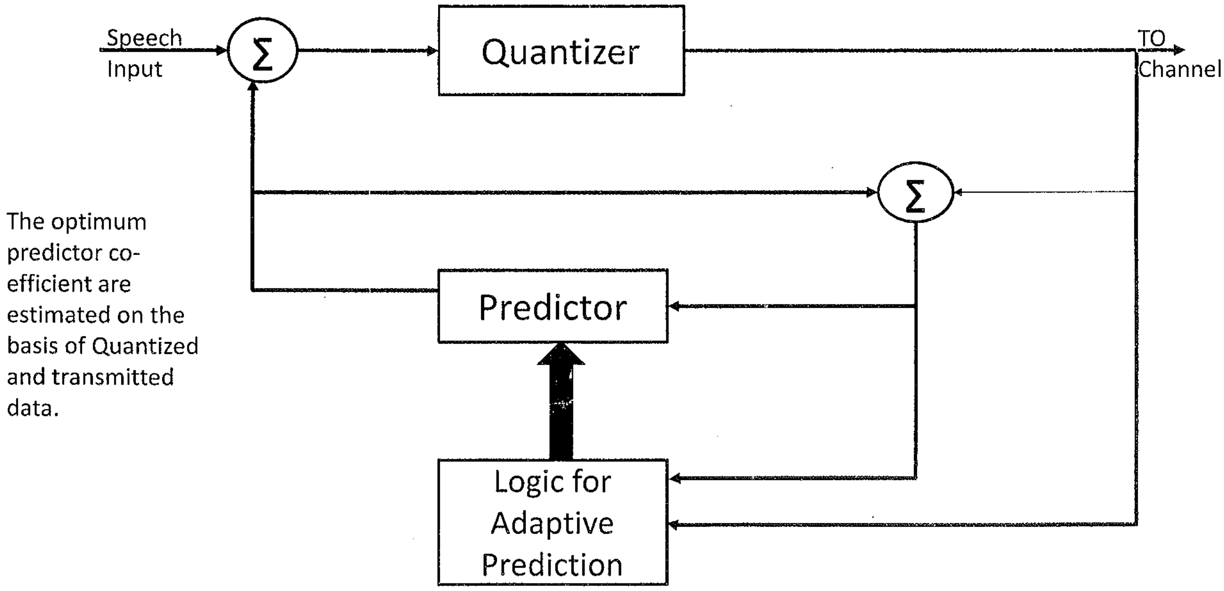
1. APF-Adaptive Prediction with Forward estimation
2. APB-Adaptive Prediction with Backward estimation

APF- Adaptive Prediction with Forward estimation

- Unquantized samples of the input speech signal are buffered and the released after the computation of M predictor co efficient.



APB- Adaptive Prediction with Backward estimation



* Adaptive ~~pros~~ Quantization refers to a quantizer that operates with a time varying step size $\Delta(nT_s)$.

* At any given time period identified by the index n, the adaptive quantizer is assumed to have a uniform transfer characteristic.

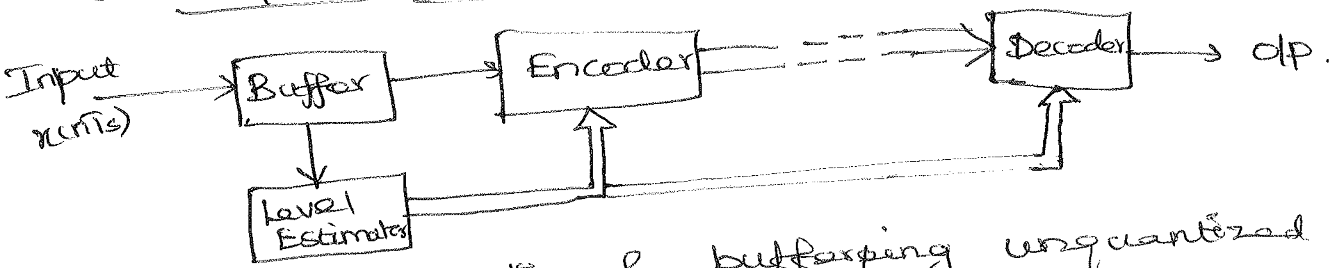
* The step size $\Delta(nT_s)$ is varied so as to match the variance σ_x^2 of the i/p signal $x(nT_s)$.

$$\therefore \Delta(nT_s) = \phi\, \hat{\sigma}_x(nT_s) \; ; \; \phi - \text{Constant}.$$

$\hat{\sigma}_x(nT_s)$ — an estimate of Std. Deviation $\sigma_x(nT_s)$

→ The adaptive quantization can be done by 2 methods.

(i) Adaptive Quantization with Forward estimation (AQF)



* First ~~It goes~~ through buffering unquantized samples of the i/p speech signal.

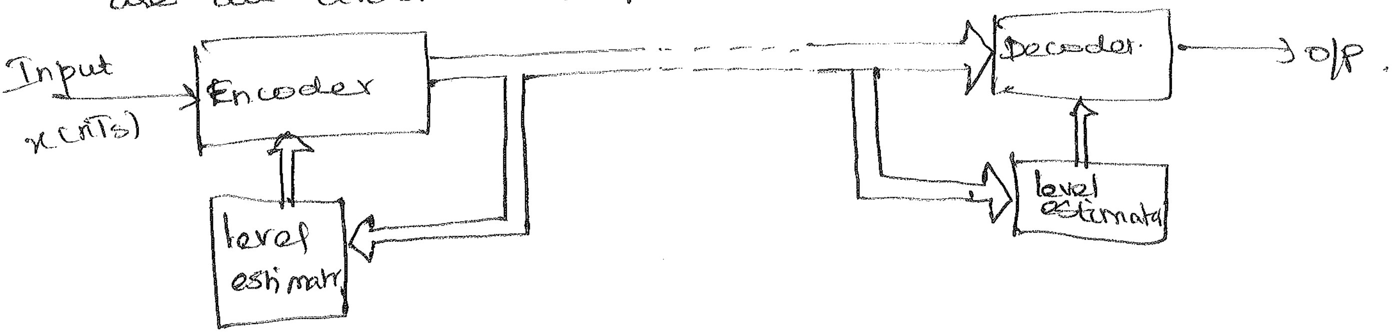
* The samples are released after the estimate $\hat{\sigma}_x(nT_s)$ has been released. (It is independent of quantizing noise).

* AQF requires the explicit transmission of level information.

(ii) Adaptive Quantization with Backward Estimation (AQB)

* Represents Non linear Feedback System.

* The problem of level transmission, buffering and delay intrinsic are all avoided in AQB.



Voice Coders: (Vocoders).

→ To transmit speech signal generated by the speaker, it is enough to transmit the information which can be reconstructed at the receiver.

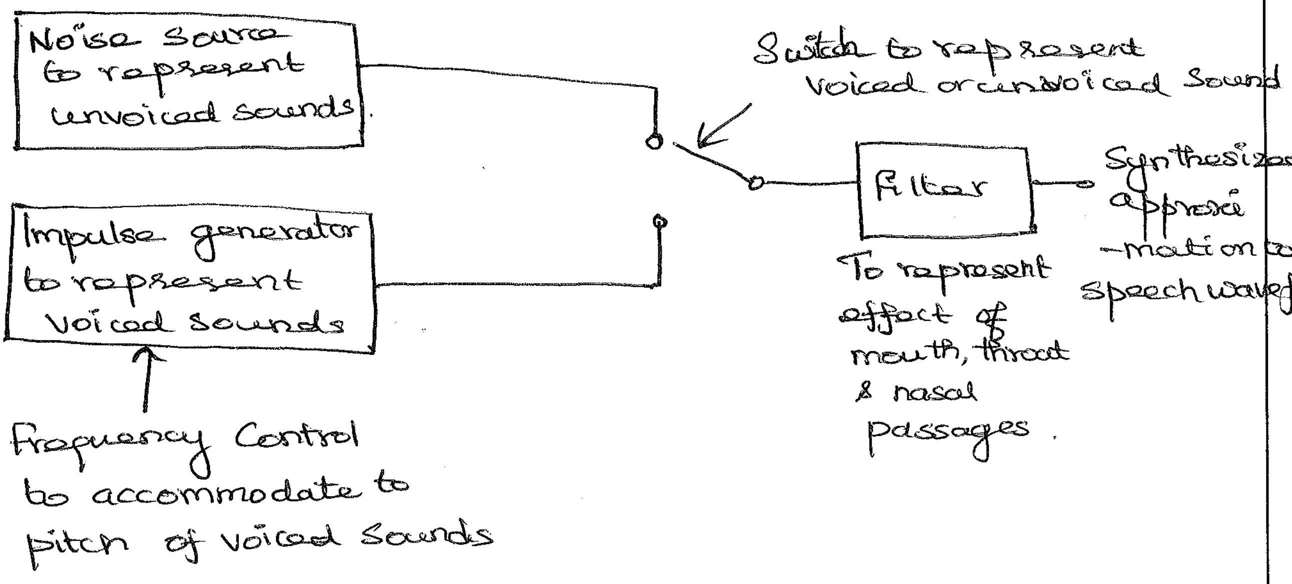
→ This can be achieved by a digital source coded transmission system at a lower bit rate.

→ The source coders employed are called vocoders (or) voice coders.

Voice Model:

* The speech consists of voiced and unvoiced sounds.
* The voice sounds are sounds generated by the vibrations of the vocal cords.
* The unvoiced sounds are generated when a speaker pronounces (s, f, p).

The generalized representation of a vocoder is shown as

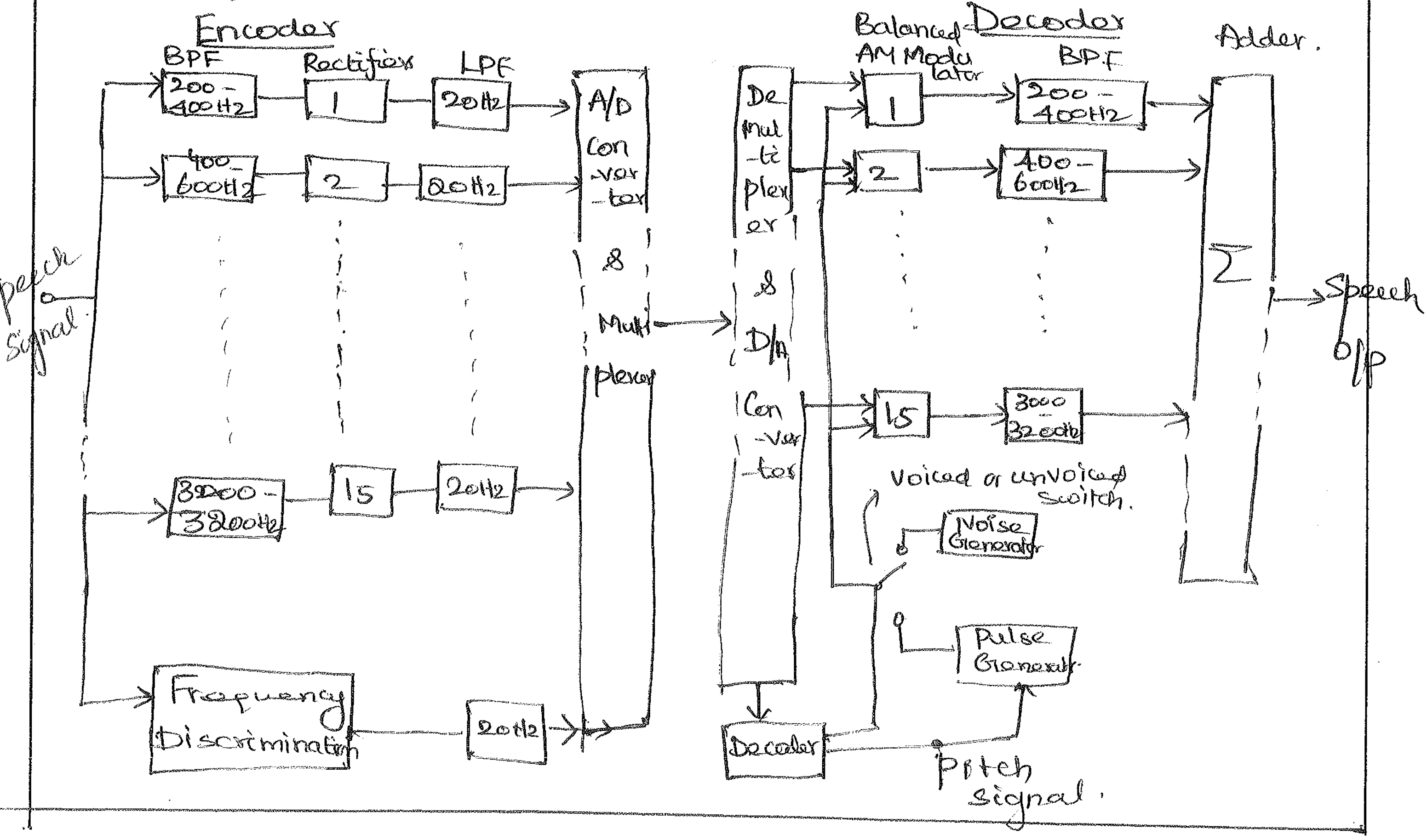


→ The filter represents the effect, on the generated sounds, of the mouth, throat and nasal passages of the speaker.

→ The voiced sounds are simulated by the impulse generator whose frequency is the fundamental frequency of vibration of the vocal cords.

Channel Vocoder:

* One of the vocoder systems is Channel Vocoder.
* In the encoding system, the spectrum of i/p speech is divided into 15 frequency ranges each of bandwidth 200Hz.
* Each 20Hz low pass filter will provide a voltage which is proportional to the amplitude at the output of its associated 200Hz bandpass filter.
* The i/p speech is applied to a frequency discriminator followed by a 20Hz LPF.
* When the signal is voiced, the o/p of the filter provides a voltage which is proportional to the voice frequency. This frequency is the pitch of the voice.
* When the speech is unvoiced, the output of the filter is a smaller voltage than the voiced speech.
* The output of 16, 20 Hz LPFs are sampled, multiplexed and A/D converted.



→ At the vocoder receiver, the signal is demultiplexed, and decoded (ie) converted back to analog form.

→ At the decoder, a balanced modulator and a BPF is provided to do the functions done by filter-rectifier combinations.

→ A balanced modulator is a modulator whose carrier input is the noise or pulse generator waveform and modulating i/p is the amplitude signal provided by the encoder.

→ At each sampling, the amplitude information is updated as is the information about whether the speech waveform is voiced or unvoiced, if voiced, the pitch is provided by the discriminator.

Multiplexing:

- In telecommunications and computer networks, multiplexing is a method by which multiple analog or digital signals are combined into one signal over a shared medium.
- The multiplexed signal is transmitted using a single communication channel such as a cable.

Multiplexing can be defined as a technique that allows simultaneous transmission of multiple signals across a single data link.

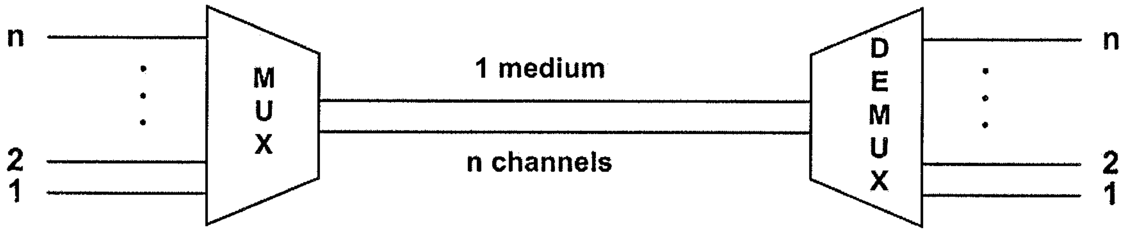


Figure 2.7.1 Basic concept of multiplexing

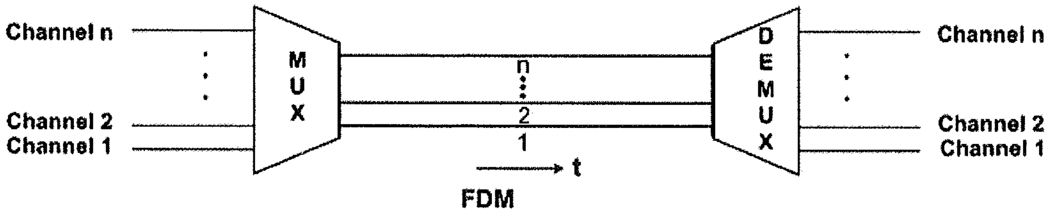
Multiplexing techniques can be categorized into the following types:

***Frequency-division multiplexing (FDM)*:** It is most popular and is used extensively in radio and TV transmission. Here the frequency spectrum is divided into several logical channels, giving each user exclusive possession of a particular frequencyband.

Time-division Multiplexing (TDM): It is also called synchronous TDM, which is commonly used for multiplexing digitized voice stream. The users take turns using the entire channel for short burst of time.

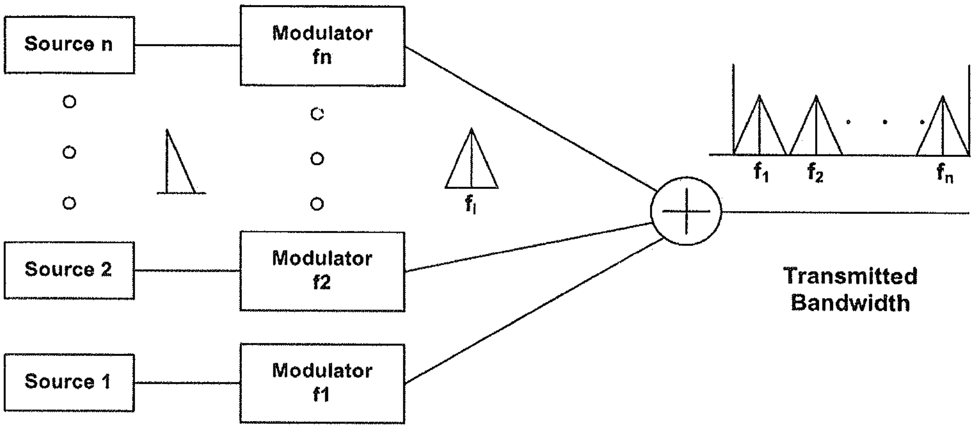
Frequency-Division Multiplexing (FDM)

- In frequency division multiplexing, the available bandwidth of a single physical medium is subdivided into several independent frequency channels.
- Independent message signals are translated into different frequency bands using modulation techniques, which are combined by a linear summing circuit in the multiplexer, to a composite signal.
- The resulting signal is then transmitted along the single channel by electromagnetic means as shown



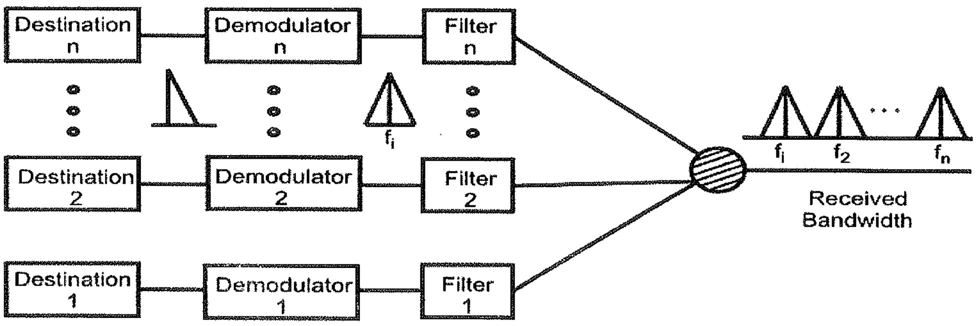
- Basic approach is to divide the available bandwidth of a single physical medium into a number of smaller, independent frequency channels.

- Using modulation, independent message signals are translated into different frequency bands.
- All the modulated signals are combined in a linear summing circuit to form a composite signal for transmission. The carriers used to modulate the individual message signals are called *sub-carriers*, shown as $f_1, f_2, \ldots, f_n$ in Fig.



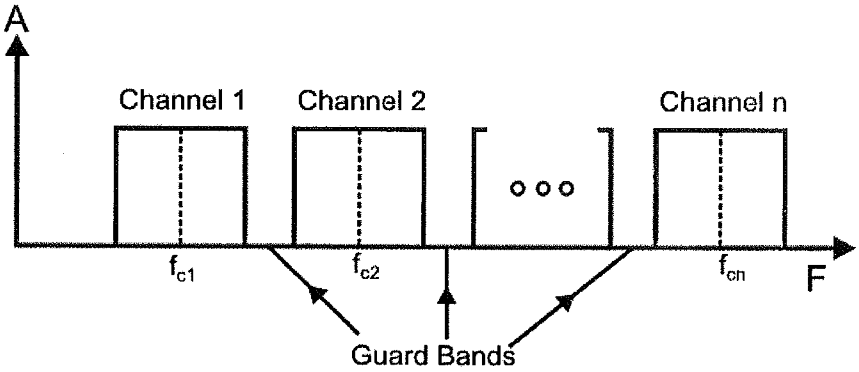
FDM Multiplexing Process

- At the receiving end the signal is applied to a bank of band-pass filters, which separates individual frequency channels.
- The band pass filter outputs are then demodulated and distributed to different output channels as shown in Fig



FDM Demultiplexing Process

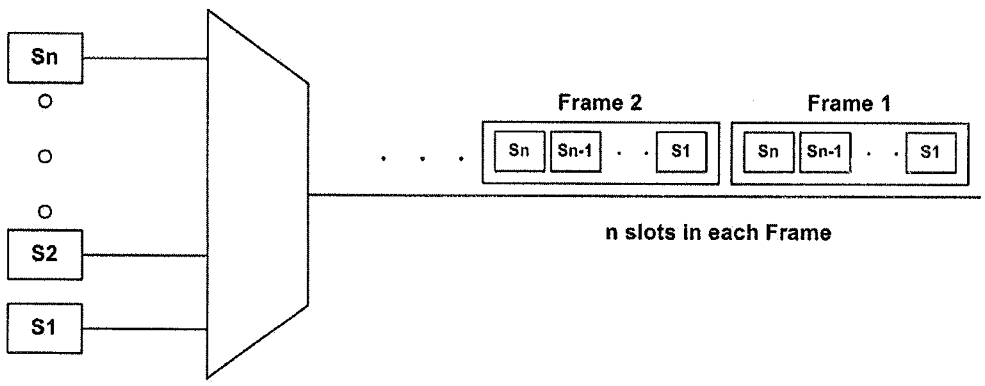
- If the channels are very close to one other, it leads to inter-channel cross talk.
- Channels must be separated by strips of unused bandwidth to prevent inter-channel cross talk.
- These unused channels between each successive channel are known as **guard bands** as shown in Fig
- FDM are commonly used in radio broadcasts and TV networks. For example, the AM radio uses 540 to 1600 KHz frequency bands while the FM radio uses 88 to 108 Mhz frequency bands



Use of guard bands in FDM

Time-Division Multiplexing (TDM)

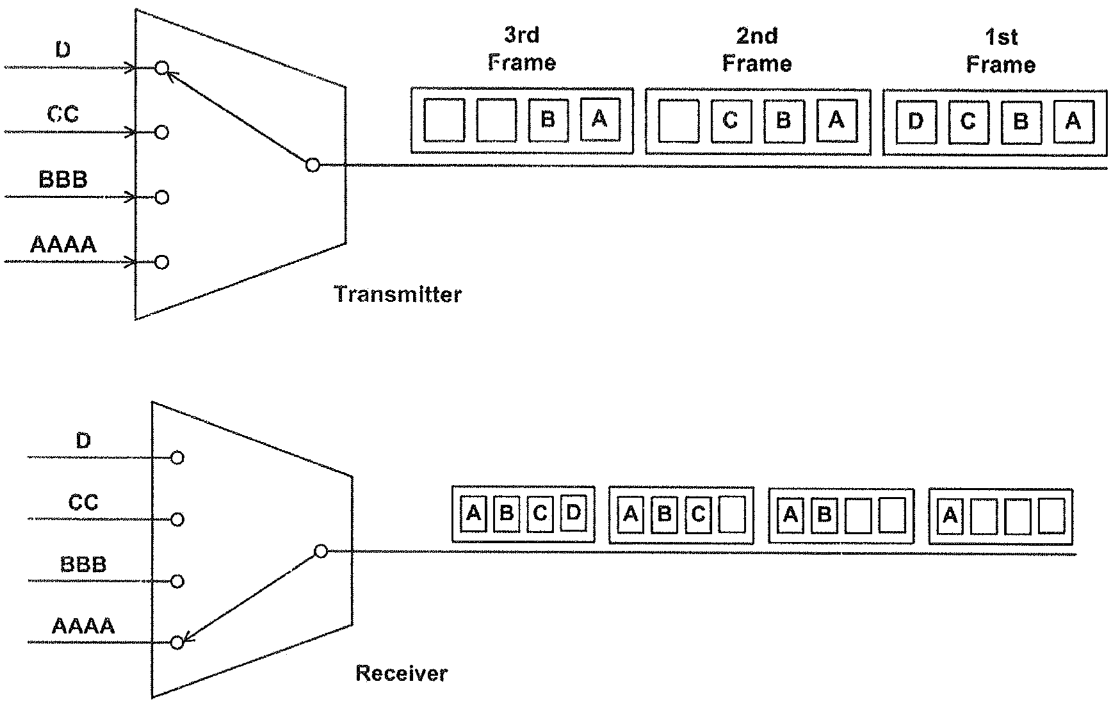
- In Time-division multiplexing, all signals operate with same frequency at different times.
- This is a base band transmission system, where an electronic commutator sequentially samples all data source and combines them to form a composite base band signal, which travels through the media and is being demultiplexed into appropriate independent message signals by the corresponding commutator at the receiving end.
- The incoming data from each source are briefly buffered. Each buffer is typically one bit or one character in length.
- The buffers are scanned sequentially to form a composite data stream. The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive.
- Composite data rate must be at least equal to the sum of the individual data rates.
- The composite signal can be transmitted directly or through a modem. The multiplexing operation is shown in Fig.



- As shown in the figure the composite signal has some *dead space* between the successive sampled pulses, which is essential to prevent interchannel cross talks.

- Along with the sampled pulses, one synchronizing pulse is sent in each cycle. These data pulses along with the control information form a *frame*.
- Each of these frames contain a cycle of time slots and in each frame, one or more slots are dedicated to each data source.
- The maximum bandwidth (data rate) of a TDM system should be at least equal to the same data rate of the sources.

Synchronous TDM is called synchronous mainly because each time slot is preassigned to a fixed source.



Multiplexing and demultiplexing in synchronous TDM

UNIT - III   J. GANAPATHI

DIGITAL MODULATION AND TRANSMISSION

different from Digital Transmission (Pulse Modulation)

* Digital modulation is the transmittal of digitally modulated analog signals (carriers) between two or more points in a communication system.

* It is some times called "digital radio" bcoz digitally modulated signals can be propagated through earth's atmosphere and used in wireless comm. systems. (unlike digital txn requires a physical medium)

* It has several advantages over AM, FM, &PM such as ease of processing, ease of multiplexing and noise immunity.

* In digital radio, the carrier facility could be a physical cable, or it could be free space.

* With digital modulation, the information signal is digital, which could be computer generated data or digitally encoded analog signals.

* We have various forms of digital modln (i) ASK (ii) FSK (iii) PSK (iv) QAM

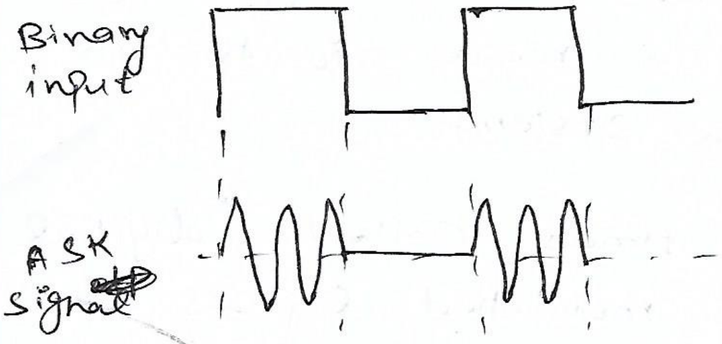
$$V(t) = V \sin(2\pi \cdot ft + \theta)$$

V → ASK, f → FSK, θ → PSK, ASK + PSK → QAM

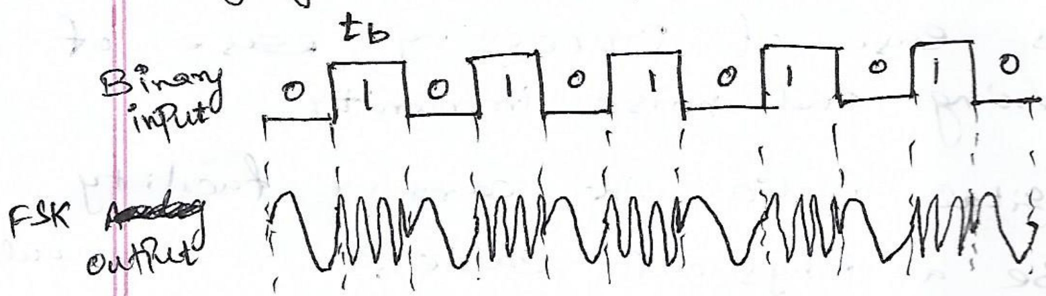
Referring to the above eqn,

If the information signal is digital and the amplitude (V) of the carrier is varied proportional to the information signal, a digitally modulated signal called amplitude shift Keying (ASK) is produced.

ASK



* If the frequency (f) is varied proportional to the information signal, frequency shift keying (FSK) is produced.



* If the phase of the carrier (θ) is varied proportional to the information signal, Phase shift keying (PSK) is produced.

* If both the amplitude and the phase are varied proportional to the information signal, Quadrature Amplitude Modulation (QAM) will be produced.

Information capacity:

It is a measure of how much information canbe propagated through a communications system and is a function of bandwidth and transmission time.

Bit rate:

It is the number of bits transmitted during one second and is expressed in bits per second (bps).

* Baud rate:

Baud refers to the rate of change of a signal on the transmission medium after encoding and modulation have occured.

A baud may represent more than one information bit.

M-ary encoding:

* M-ary is derived from the word binary.

* M simply represents a digit that corresponds to the no. of conditions, levels or combinations possible for a given number of binary variables.

* To express the no. of conditions possible with N bits, we can write

$$2^N = M.$$

* a digital s/g with four possible conditions (V, f & phase etc)
M = 4.
If eight possible cond,
M = 8.

Phase shift Keying:

* PSK is another form of angle modulated, constant amplitude digital modulation.

* PSK is an M-ary digital modulation scheme, similar to conventional P.M, except the i/p is a binary digital signal and there are a limited no. of o/p phases possible.

Binary Phase shift Keying: (BPSK):

* Simplest form of PSK, is BPSK where N=1 & M=2.

* ∴ with BPSK two phases are possible for the carrier.

* As the digital input signal changes state, the phase of the output carrier shifts b/w two angles that are separated by 180°.

BPSK Transmitter:

* Fig 1. shows the simplified block dgm of a BPSK Transmitter.

* The Balanced modulator acts as a phase reversing switch. Depending on the logic condition of the digital input, the carrier is transferred to the o/p either in phase

or 180° out of phase with the reference carrier oscillator.

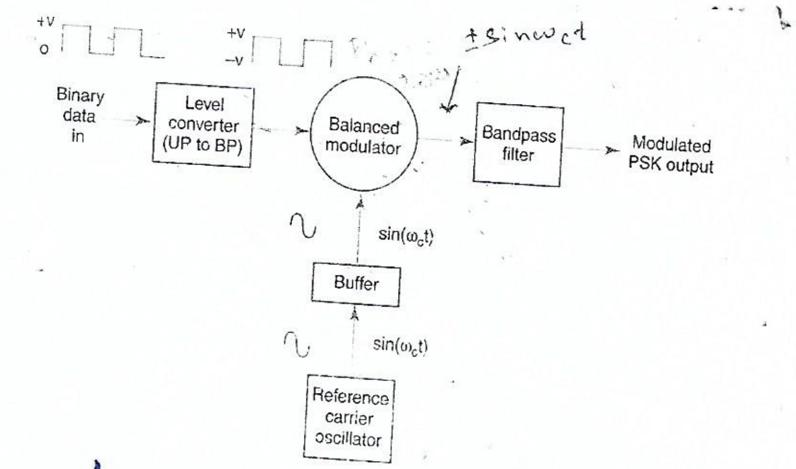


FIGURE 1 BPSK transmitter

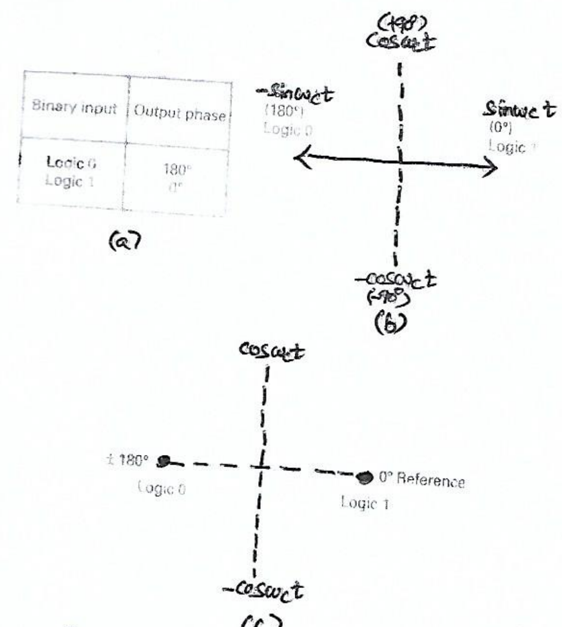


FIGURE 2 BPSK modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

* Fig 2 shows the truth table, Phasor diagram and constellation diagram for a BPSK modulator.
* A constellation diagram, which is sometimes called signal-state-space diagram is similar to a phasor diagram, where only the relative Position of the Peaks of the phasors are shown.

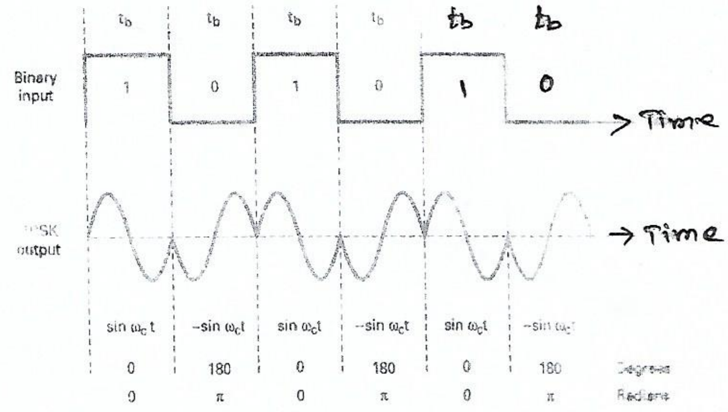


FIGURE 9-15 Output phase-versus-time relationship for a BPSK modulator

Bandwidth considerations of BPSK:

* In a BPSK modulator, the carrier s/g is multiplied by the binary data.

* If +1V is assigned logic 1 &
  −1V is " logic 0

  the o/p signal is either $+1 \cdot \sin\omega_c t$ → in phase
  (or)
  $-1 \sin\omega_c t$ → 180° out of phase

* The widest o/p BW occurs when the input binary data are an alternating 1/0 seq.

* The fundamental frequency (f_a) of an alternative 1/0 bit sequence is equal to one-half of the bit rate ($f_b/2$).

∴ Mathematically the o/p of BPSK modulator is

$$\text{BPSK output} = [\sin(2\pi f_a t)] \times [\sin(2\pi f_c t)]$$

whr,

f_a = max. fundamental freq. of binary i/p (Hz)

f_c = reference carrier freq. (Hz)

Solving the above eqn. using trig. identity,

$$\frac{1}{2}\cos[2\pi(f_c - f_a)t] - \frac{1}{2}\cos[2\pi(f_c + f_a)t]$$

Thus the minimum Nyquist BW (B) is,

$$B = \left\{ \begin{array}{r} f_c + f_a \\ -f_c + f_a \\ \hline 2f_a. \end{array} \right.$$

and because $f_a = f_b/2$

$$B = 2f_b/2 = \underline{\underline{f_b}}.$$

BPSK receiver:

* Fig shows the block diagram of a BPSK receiver.
* The i/p signal may be $+\sin\omega_c t$ or $-\sin\omega_c t$.
* The coherent carrier recovery circuit detects and regenerates a carrier signal that is both freq. and phase coherent with the original transmit carrier.

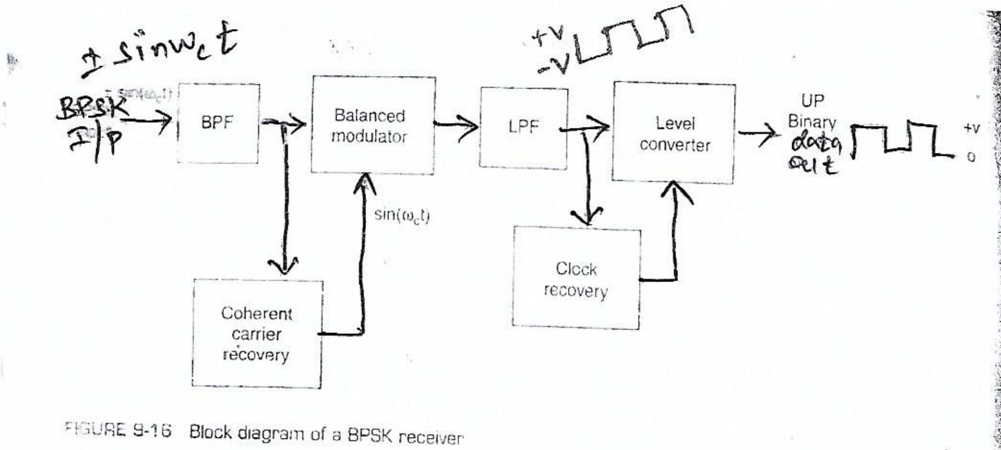


FIGURE 9-16 Block diagram of a BPSK receiver

* The LPF separates the recovered binary data from the complex demodulated signal.

Mathematically, the demodulation process is as follows.

For a BPSK i/p signal of $+\sin\omega_c t$ (logic 1) the o/p of Bal. Modulator is,

$$o/p = (\sin\omega_c t)(\sin\omega_c t) = \sin^2\omega_c t$$

$$= \frac{1}{2}(1 - \cos 2\omega_c t)$$

$$= \frac{1}{2} - \underbrace{\frac{1}{2}\cos 2\omega_c t}_{\text{filtered out.}}$$

leaving

$$o/p = +\frac{1}{2}V = \text{logic } 1.$$

The LPF has a cutoff frequency

much lower than $2\omega_c$, and thus blocks the second harmonic of the carrier and passes only the positive constant component. A +ve vltg repr. a demodulated logic 1.

For a BPSK input signal of $-\sin\omega_c t$ (logic 0) the o/p of bal. modulator is,

$$o/p = (-\sin\omega_c t)(\sin\omega_c t) = -\sin^2\omega_c t$$

$$= -\frac{1}{2}(1 - \cos 2\omega_c t)$$

$$= -\frac{1}{2} + \underbrace{\frac{1}{2}\cos 2\omega_c t}_{\text{filtered out}}$$

$$o/p = -\frac{1}{2}V = \text{logic } 0.$$

Again, the LPF blocks the second harmonic of the carrier and passes only the negative constant component. A -ve vltg repr. a demodulated logic 0.

Quaternary PSK (or) Quadrature PSK: (QPSK)

* It is other form of angle modulated, Constant amplitude digital modulation.

* QPSK is an M-ary encoding scheme, where $N=2$ & $M=4$. (hence quaternary).

* With QPSK four o/p phases are possible for a single carrier frequency.

* therefore, to produce four different i/p combinations, we need two bits, The four possible combinations are 00, 01, 10 & 11.

* Each dibit code generates one of the four possible o/p phases ($+45^\circ$, $+135^\circ$, -45°, -135°).

QPSK Transmitter:

* Block diagram of a QPSK modulator is shown in fig.

* Two bits are serially inputted, ~~to the~~ and parallely outputted using a bit splitter.

* One bit is directed to the I channel and another " " Q channel.

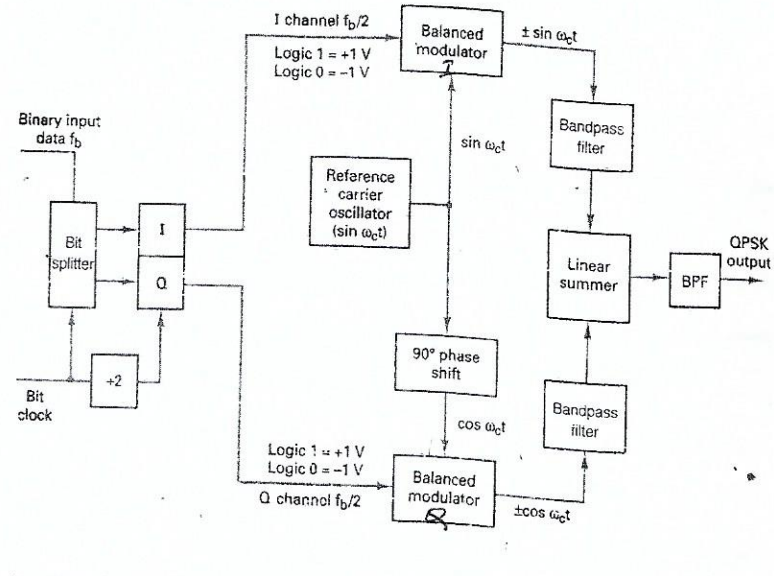


FIGURE 9-17 QPSK modulator

* The I-bit modulates a carrier that is inphase with the ref. oscillator (Hence I for inphase)
* The Q-bit modulates a carrier that is 90^0 out of phase (or) in quadrature with the reference oscl.. (Hence Q for quadrature channel).
* Once the dibit has been split into I and Q channels, the operation is same as in a BPSK modulator.
* Essentially a QPSK modulator is two BPSK modulators combined in parallel.
* For a logic 1 = +1V &

  a logic 0 = −1V,

  two phases are possible at the o/p of the I Balanced modulator (+ sin$\omega_c t$ & − sin$\omega_c t$)

and two phases are possible at the o/p of the Q Balanced modulator ($+\cos\omega_c t$ & $-\cos\omega_c t$)

* when the linear summer combines the two quadrature (90° out of phase) signals, there are four possible resultant phasors.

$+\sin\omega_c t + \cos\omega_c t$

$+\sin\omega_c t - \cos\omega_c t$

$-\sin\omega_c t + \cos\omega_c t$

$-\sin\omega_c t - \cos\omega_c t$.

Binary input Q	Binary input I	QPSK output phase
0	0	−135°
0	1	−45°
1	0	+135°
1	1	+45°

(a)

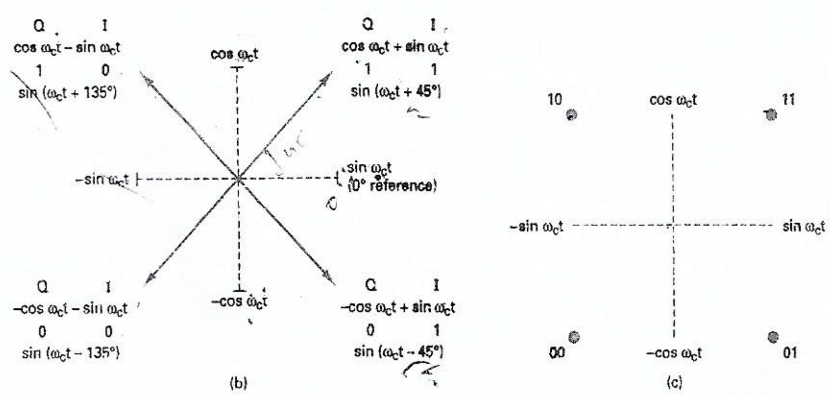


FIGURE 9-18 QPSK modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

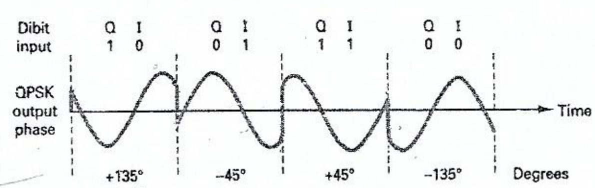


FIGURE 9-19 Output phase-versus-time relationship for a QPSK modulator

* From the above figures, it can be seen that each of the four o/p phasors has exactly the same amplitude.
* ∴ the binary info. must be encoded entirely in the phase of the o/p signal.

Bandwidth considerations of QPSK:

* with QPSK, because the i/p data are divided into two channels, the bit rate in either I or Q channel is equal to one half of the input data rate ($f_b/2$).
* Consequently the fundamental frequency present at the data i/p to the I or Q bal. modulator is one-fourth of the input data rate. (one half of $f_b/2 = f_b/4$).

Mathematically, o/p of Bal. modulators is,

$$o/p = (\sin \omega_a t)(\sin \omega_c t)$$

where, $\underbrace{\omega_a t = 2\pi \frac{f_b}{4} t}_{\text{modulating s/g}}$ and $\underbrace{\omega_c t = 2\pi f_c t}_{\text{carrier s/g.}}$

Thus,

$$o/p = \left(\sin 2\pi \frac{f_b}{4} t\right)(\sin 2\pi f_c t)$$

$$= \frac{1}{2}\cos 2\pi\left(f_c - \frac{f_b}{4}\right)t - \frac{1}{2}\cos 2\pi\left(f_c + \frac{f_b}{4}\right)t$$

The o/p freq. spectrum extends from $f_c + f_b/4$ to $f_c - f_b/4$, The minimum BW (f_N) is

$$= \left(f_c + \frac{f_b}{4}\right) - \left(f_c - \frac{f_b}{4}\right)$$

$$= \frac{2f_b}{4}$$

$$= \frac{f_b}{2}$$

QPSK receiver:

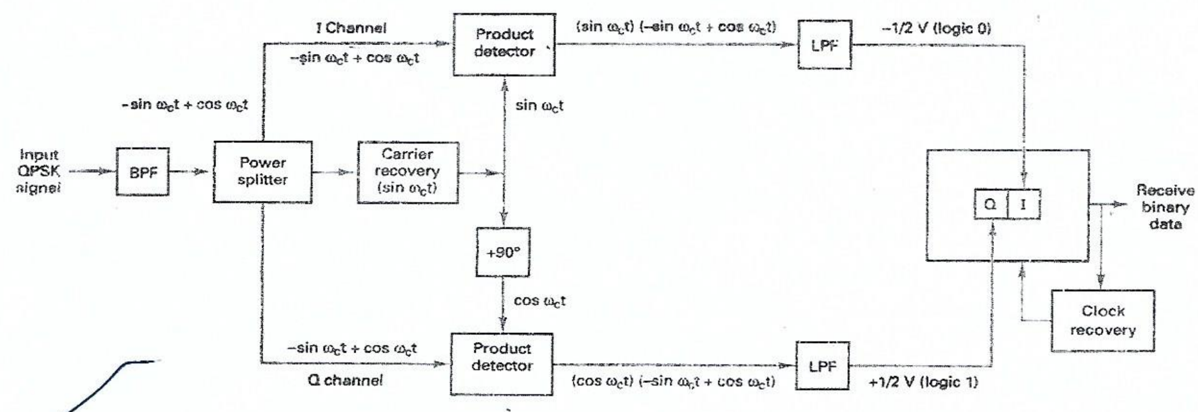


FIGURE 9-21 QPSK receiver

* The power splitter directs the input QPSK signal to the I and Q product detectors and the carrier recovery circuit.
* The carrier recovery circuit reproduces the original transmit carrier oscillator signal.

* The QPSK signal is demodulated in the I and Q product detectors, which generate the original I and Q data bits.

* These data bits are fed to the bit combining circuits, where they are converted from parallel I and Q data channels to a single binary output data stream.

* ~~From~~ out of the four possible o/p phases, Let the incoming QPSK signal be,

$\Rightarrow \underline{-\sin\omega_c t + \cos\omega_c t},$

* Mathematically the demodulation process is as follows.

$\Rightarrow$ The received QPSK signal $(-\sin\omega_c t + \cos\omega_c t)$ is one of the inputs to the I product detector. The other input is the recovered carrier $(\sin\omega_c t)$.

$\therefore$ The o/p of I product detector is

$$I = \underbrace{(-\sin\omega_c t + \cos\omega_c t)}_{\text{QPSK input signal}} \underbrace{(\sin\omega_c t)}_{\text{carrier}}$$

$$= (-\sin\omega_c t)(\sin\omega_c t) + (\cos\omega_c t)(\sin\omega_c t)$$

$$= -\sin^2\omega_c t + (\cos\omega_c t)(\sin\omega_c t)$$

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

$$I = -\frac{1}{2} + \underbrace{\frac{1}{2}\cos 2\omega_c t}_{\text{filtered out}} + \underbrace{\frac{1}{2}\sin 2\omega_c t + \frac{1}{2}\sin 0}_{\text{equals } 0}$$

$$I = -\frac{1}{2}V \text{ (logic 0)}$$

* Again, the received QPSK s/g $(-\sin\omega_c t + \cos\omega_c t)$ is one of the inputs of the Q product detector. The other input is the recovered carrier shifted 90° in phase $(\cos\omega_c t)$.

The o/p of Q product detector is

$$Q = \underbrace{(-\sin\omega_c t + \cos\omega_c t)}_{\text{QPSK i/p signal}}\underbrace{(\cos\omega_c t)}_{\text{carrier}}$$

$$= \cos^2\omega_c t - \sin\omega_c t \cos\omega_c t$$

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

$$= \frac{1}{2} + \underbrace{\frac{1}{2}\cos 2\omega_c t}_{\text{filtered out}} - \underbrace{\frac{1}{2}\sin 2\omega_c t - \frac{1}{2}\sin 0}_{\text{equals } 0}$$

$$Q = \frac{1}{2}V \text{ (logic 1)}$$

The demodulated I and Q bits (0 and 1 respectively) correspond to the constellation diagram and truth table for the QPSK modulator shown in fig.

Quadrature Amplitude Modulation: (QAM):

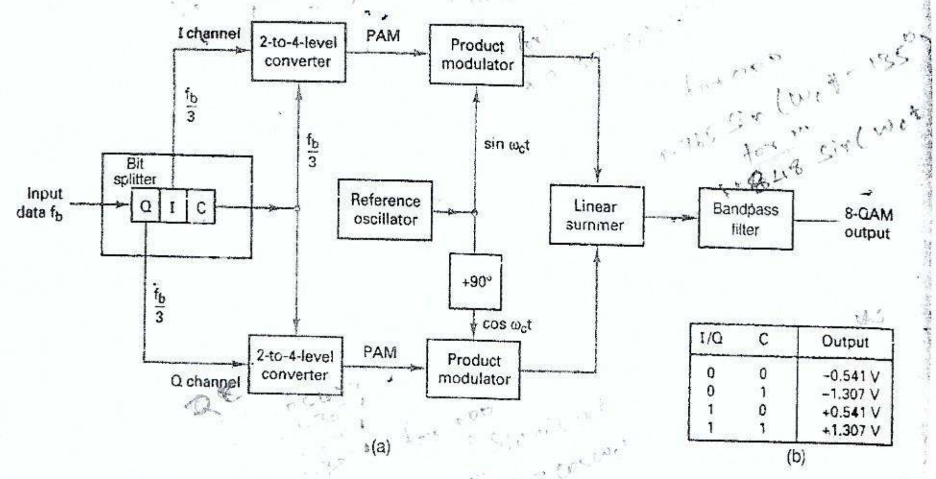
* Quadrature Amplitude Modulation is a form of digital modulation similar to PSK except the digital information is contained in both the amplitude and the phase of the transmitted carrier.

8-QAM:

* 8-QAM is an M-ary encoding technique where M = 8, and 8-QAM signal is not a constant-amplitude signal.

8-QAM Transmitter:

*



I/Q	C	Output
0	0	-0.541 V
0	1	-1.307 V
1	0	+0.541 V
1	1	+1.307 V

FIGURE 9-30 8-QAM transmitter: (a) block diagram; (b) truth table 2-4 level converters

* The incoming serial bit stream enters the bit splitter, where it is converted to a parallel, three channel output. & ~~the~~

I → Inphase channel

Q → In-Quadrature channel

C → control channel.

* The bit rate in each of the three channels is $f_b/3$.

* The bits in I & C channels enter the I-channel 2-to-4 level converter, and the bits in the Q and C channels enter the Q-channel 2-to-4 level converter.

* 2-to-4 level converters are Parallel input digital-to-analog converters (DAC's) with two input bits, four output voltages are possible.

Binary input			8-QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765 V	−135°
0	0	1	1.848 V	−135°
0	1	0	0.765 V	−45°
0	1	1	1.848 V	−45°
1	0	0	0.765 V	+135°
1	0	1	1.848 V	+135°
1	1	0	0.765 V	+45°
1	1	1	1.848 V	+45°

(a)

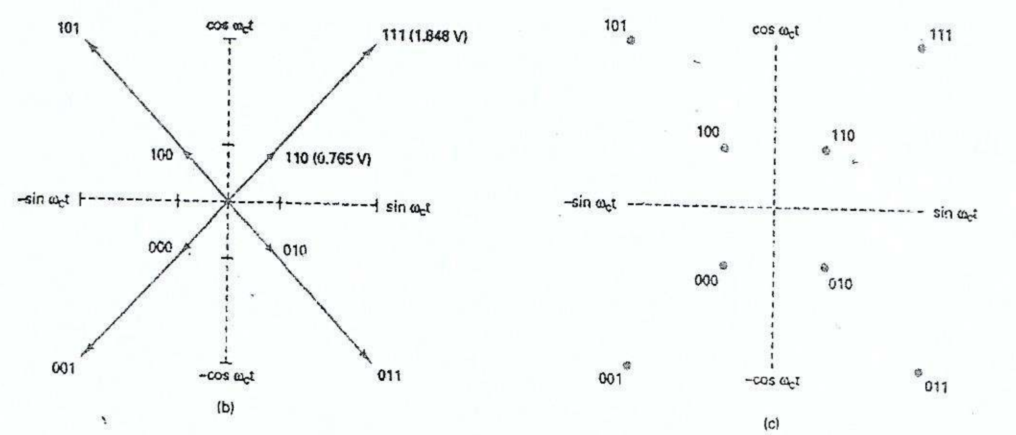


FIGURE 9-31 8-QAM modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

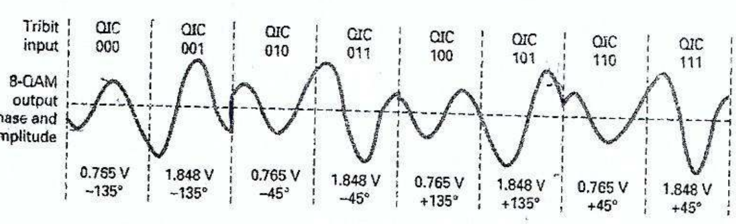


FIGURE 9-32 Output phase and amplitude-versus-time relationship for 8-QAM

* The I and Q bits determine the polarity of the PAM signal at the output of the 2-to-4 level converters and the C channel determines the magnitude.

* The magnitudes of I and Q PAM signals are always equal. Their polarities depend on the logic condition of the I and Q bits and therefore maybe different.

For eg: If the tribit is 000, then Q = 0, I = 0 and C = 0,

∴ the two i/p's of I channel product modulator are -0.541 & $\sin\omega_c t$

$$\therefore \quad I = (-0.541)(\sin\omega_c t) = -0.541 \sin\omega_c t$$

& The two i/p's of Q channel product modulator are -0.541 & $\cos\omega_c t$

$$\therefore Q = (-0.541)(\cos\omega_c t) = -0.541 \cos\omega_c t$$

The outputs from the I & Q channel product modulators are combined in the linear summer and produce a modulated output of.

$$\text{Summer output} = -0.541 \sin\omega_c t - 0.541 \cos\omega_c t$$

$$= 0.765 \sin(\omega_c t - 135^\circ)$$

Thus different phase angles with different amplitudes are produced.

Bandwidth considerations of 8-QAM:

* In 8-QAM, because the data are divided into 3 channels, the bit rate in I, Q, and C channel is equal to $f_b/3$.

* Also the highest fundamental frequency is equal to one-sixth of the binary input bit rate.

* Mathematically the output of balanced modulator is

$$= (\sin \omega_a t)(\sin \omega_c t)$$

$$= \left(\sin 2\pi \frac{f_b}{6} t\right)(\sin 2\pi f_c t)$$

$$= \frac{1}{2} \cos 2\pi \left(f_c - \frac{f_b}{6}\right)t - \frac{1}{2} \cos\left(f_c + \frac{f_b}{6}\right)t$$

* The output frequency spectrum extends from $f_c + f_b/6$ to $f_c - f_b/6$ and the minimum bandwidth f_N is

$$f_N = \begin{array}{l} f_c + f_b/6 \\ -f_c + f_b/6 \\ \hline 2f_b/6 \end{array}$$

$$\boxed{f_N = f_b/3}$$

8-QAM receiver:

J. GANAPATHI

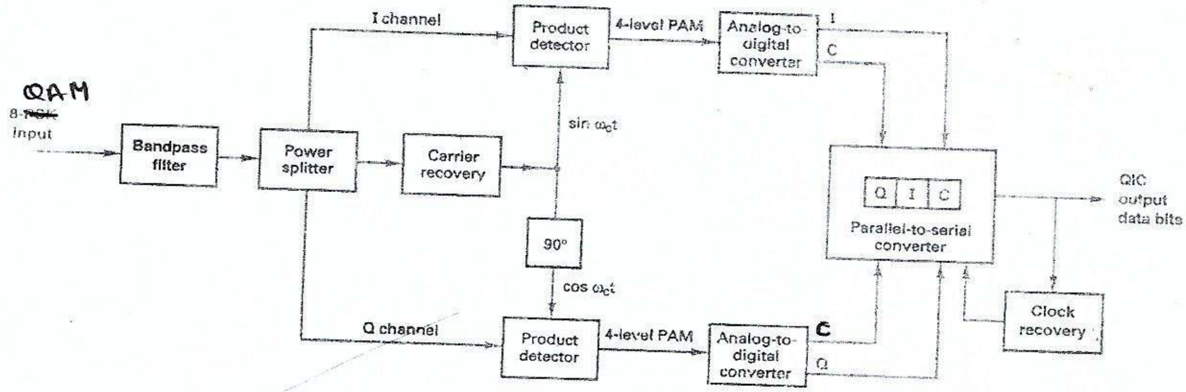


Fig: 8-QAM receiver

* The power splitter directs the input 8-QAM signal to the I and Q product detectors and the carrier recovery circuit.
* The o/p's of the product detectors are a 4-level PAM signals that are fed to the 2-to-4 level Analog to Digital converters (ADC's).
* The o/p's from the I-channel (I/c) and Q-channel (Q/c) ADC's are given to Parallel-to-serial converter circuit.
* This circuit converts the I/c and Q/c bit pairs to serial I, Q, and C output data streams.

Differential phase-shift Keying: (DPSK)

* DPSK is an alternative form of digital Modulation where the binary input information is contained in the difference between two successive signaling elements rather than the absolute phase.

* Differential BPSK Transmitter:

* An incoming information bit is X-NOR ed with the preceding bit prior to entering the BPSK modulator (Balanced modulator).

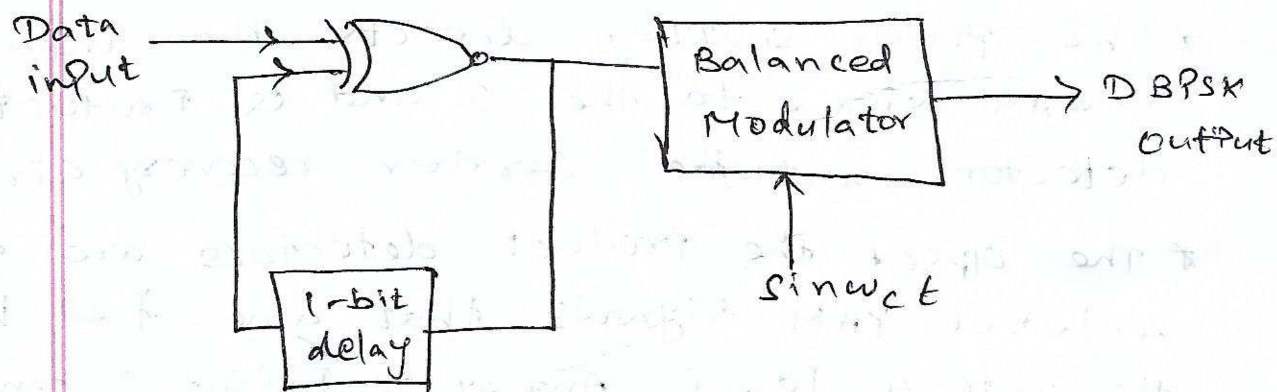


Fig (a) DBPSK Modulator Block diagram

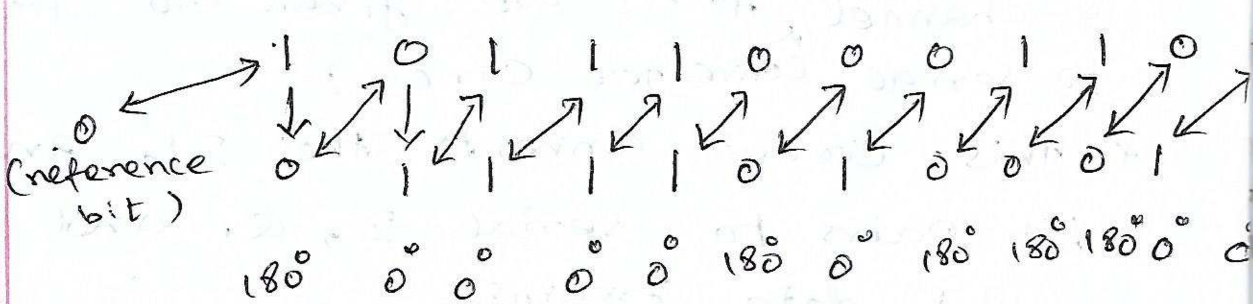


Fig (b) Timing diagram.

* For the first data bit, there is no preceding bit with which to compare it. ∴ an initial reference bit is assumed.

* The first data bit is XNORed with the reference bit. If they are the same, the XNOR output is a logic 1; if they are different, the XNOR output is a logic 0.

* The Balanced modulator operates the same as a conventional BPSK modulator; a logic 1 produces $+\sin\omega_c t$ at the output, and a logic 0 produces $-\sin\omega_c t$ at the output.

DBPSK receiver:

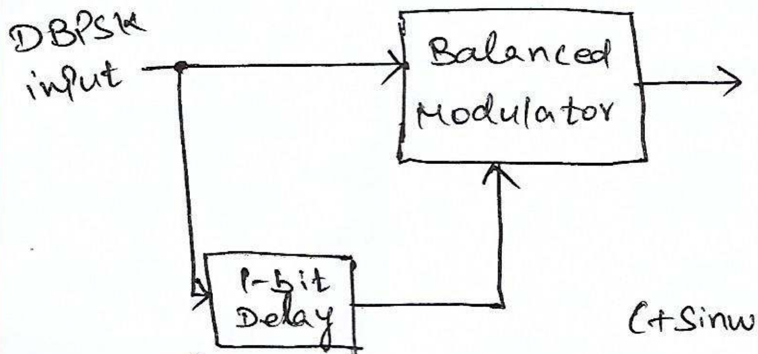


Fig (a) DBPSK DeModulator Block Diagram

Balanced Modulator output

$(+\sin w_c t)(+\sin w_c t) = +\frac{1}{2} - \frac{1}{2}\cos 2w_c t$

$(-\sin w_c t)(-\sin w_c t) = +\frac{1}{2} - \frac{1}{2}\cos 2w_c t$

$(-\sin w_c t)(+\sin w_c t) = -\frac{1}{2} + \frac{1}{2}\cos 2w_c t$

DBPSK i/p phase: 180° (reference phase) 180° 0° 0° 0° 0° 180° 0° 180° 180° 180° 0° 0°

1 0 1 1 1 0 0 0 1 1 0 1

Fig (b) Timing sequence:

* The received signal is delayed by one bit time, then compared with the next signaling element in the balanced Modulator.

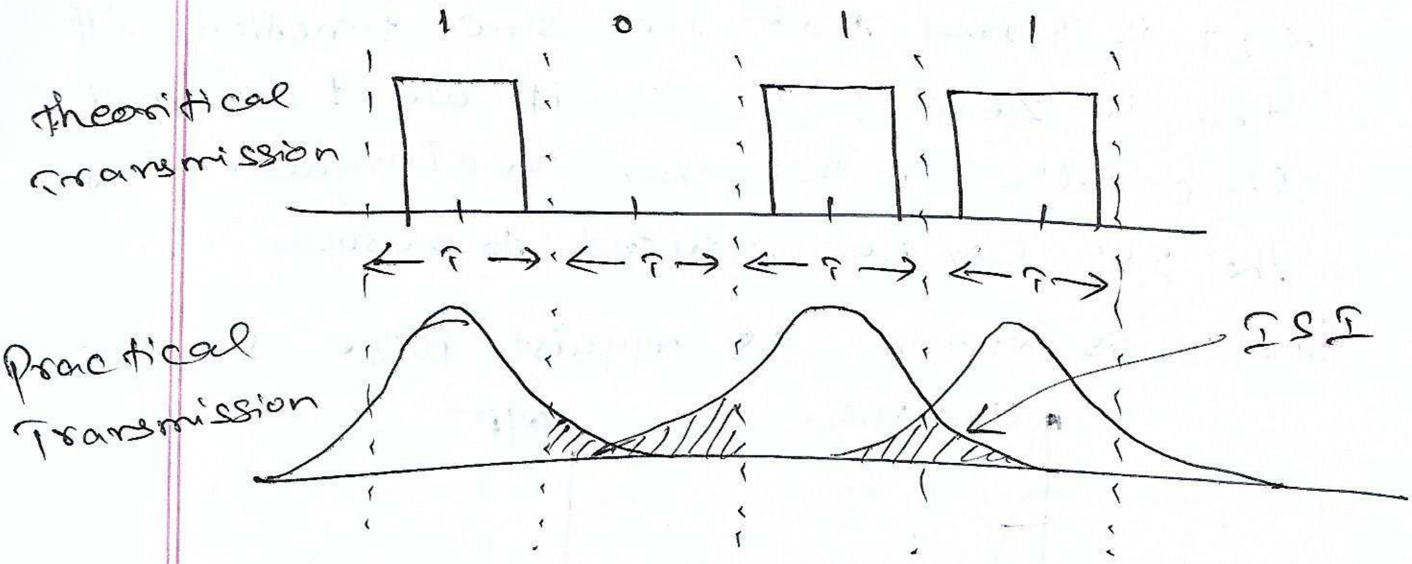
* If they are the same, a logic 1 (+ voltage) is generated. If they are different, a logic 0 (- voltage) is generated.

Advantages:

* The primary advantage of DBPSK is the simplicity with which it can be implemented.

* with DBPSK, no carrier recovery circuit is needed.

Inter symbol Interference : (ISI)



* In digital Signal transmission, the imperfection present in the Low pass filters, causes the digital pulses to be spread or disperse as shown in fig.

* This pulse spreading or dispersion causes overlap of pulses into adjacent time slots, as shown in the fig. This

* Phenomenon of pulse overlap is called Intersymbol Interference.

* ISI causes errors in the Digital Txn. Hence receiver can make an error in deciding whether it has received a logic 1 or a logic 0.

Remedy to reduce ISI:

* It is proved that the sinc function will provide zero ISI. ∴ if we transmit sinc pulse rather than rectangular pulse, the ISI can be reduced to zero.

* This is known as Nyquist pulse shaping.

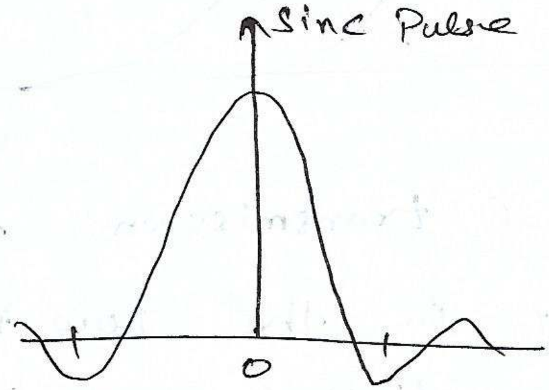


fig (a) Ideal sinc pulse for zero ISI

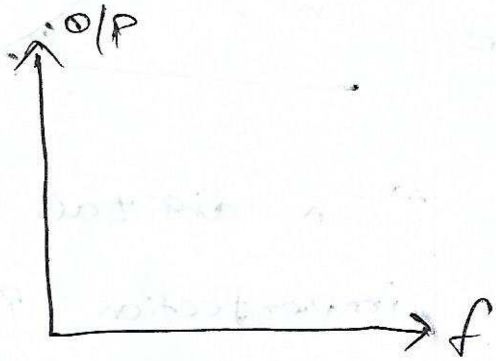


fig (b) Frequency response of filter.

* Fourier transform of sinc pulse is a rectangular function. Therefore to preserve all the frequency components, the frequency response of the filter must be exactly that in passband and zero in attenuation band as shown in fig (b).

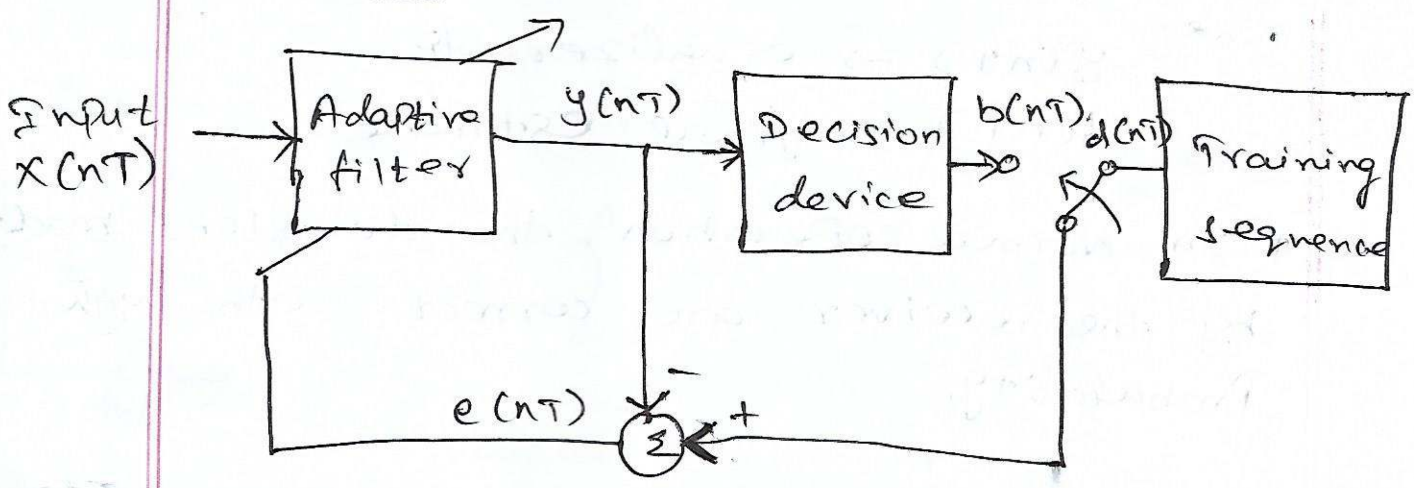
* Useful if the frequency response of the channel is known.

Equalizers:

* Equalizers are used to overcome the channel distortion & ISI, if the frequency response of the channel is either unknown or charge with time.

* Equalizers are filters whose parameters are varied on the basis of measurements of the channel characteristics during the transmission of data.

Adaptive Equalizer:



It consists of tapped delay line filter with set of delay elements, set of adjustable multipliers connected to the delay line taps and a summer.

* we have two modes of operation

1. Training Mode
2. Decision directed Mode.

Training mode:

* A known sequence $d(nT)$ is transmitted and synchronized version of it is generated in the receiver applied to adaptive equalizer.

* When the training period is completed, the adaptive equalizer is switched to its second mode of operation.

Decision Directed Mode:

In this mode, the error signal is

$$e(nT) = b(nT) - y(nT)$$

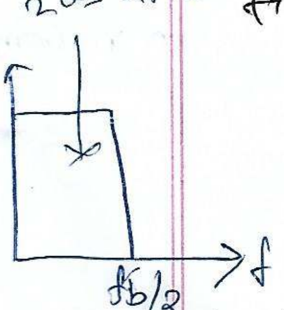
where, $y(nT) \rightarrow$ equalizer o/p

$b(nT) \rightarrow$ final estimate

* In normal operation the decisions made by the receiver are correct with high Probability.

Cosine filters:

* To minimize the intersymbol interference, the time domain shape of the symbol and so the spectral shape should be carefully chosen. This is known as pulse shaping.
* A convenient way of generating symbols with the time domain shape we require to generate an impulse of the appropriate strength for each symbol and then to shape this impulse by passing it through a shaping filter.
* Shaping of a signal is done by a Cosine filter.
* The transfer characteristics of cosine filter is given by,



$$H_c(f) = \begin{cases} 2\cos \pi f T_b & \text{for } f \le f_b/2 \\ 0 & \text{for } f > f_b/2 \end{cases}$$

* The raised cosine filter is defined as

$$H(\omega) = \begin{cases} 1 & 0 \le |\omega| \le (1-\alpha)W \\ 0.5 & (1-\alpha)W \le |\omega| \le (1+\alpha)W \\ 0 & |\omega| > (1+\alpha)W. \end{cases}$$

and its impulse response is given by,

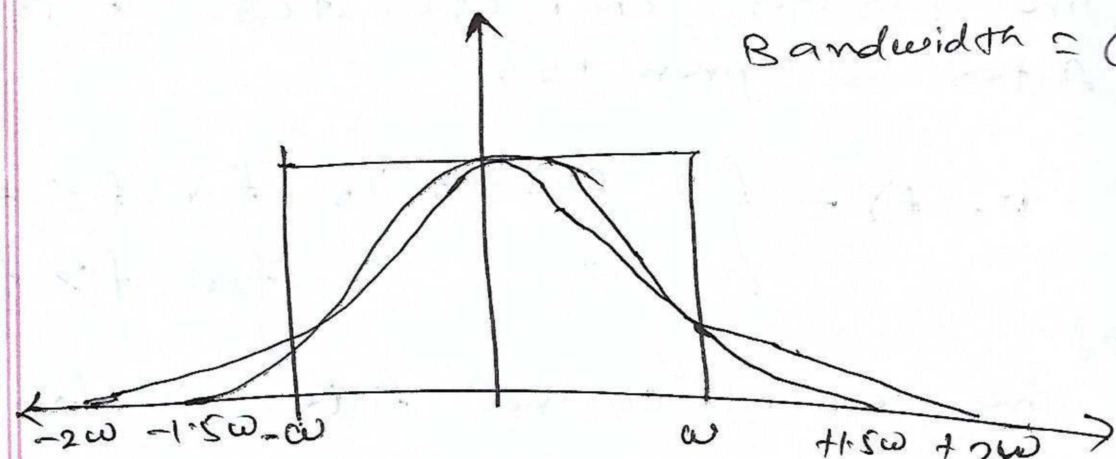
$$h(t) = \frac{1}{T_b}\left(\frac{\sin wt}{wt}\right)\cdot\left[\frac{\cos \alpha wt}{1-(2\alpha wt/\pi)^2}\right]$$

where,

$\omega = 2\pi f$, cutoff freq. in radians.

$\alpha \rightarrow$ roll off factor that decides sharpness or smoothness of the filter.

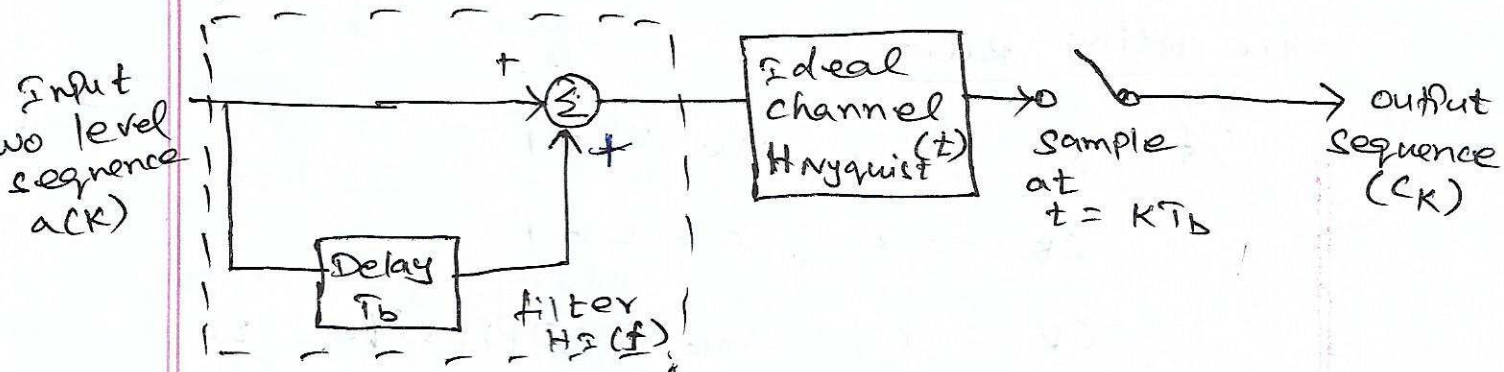
Bandwidth $= (1+\alpha)\omega$ in radians.



Duobinary Signaling:

* Duobinary signaling system doubles the transmission capacity of the binary system by reducing the maximum frequency of the baseband signal.

Encoder:



* Only from selected symbols ISI is introduced. Let the binary sequence be b_K. This is converted into binary waveform represented as a_K.

$$a_K = \begin{cases} +1 & \text{if symbol is } 1 \\ -1 & \text{if symbol is } 0 \end{cases}$$

* The encoder accepts two level sequence a_K and converts this into 3 levels +2, 0, −2.
* The output of the duobinary encoder is expressed as

$$C_K = a_K + a_{K-1}$$

For eg: binary sequence (b_k)

b_k	1	0	1	0	0
a_k	+1	−1	+1	−1	+1
a_{k-1}	+1	+1	−1	+1	−1
c_k	2	0	0	0	−2

Decoding Rule

If $c_k = 2$, $a_k = +1$

$c_k = -2$, $a_k = -1$

$c_k = 0$, a_k = opposite to previous estimate

Reconstruction:

Let $\hat{a}_k$ represent the estimate of a_k, we obtain

$$\boxed{\hat{a}_k = c_k - \hat{a}_{k-1}}$$

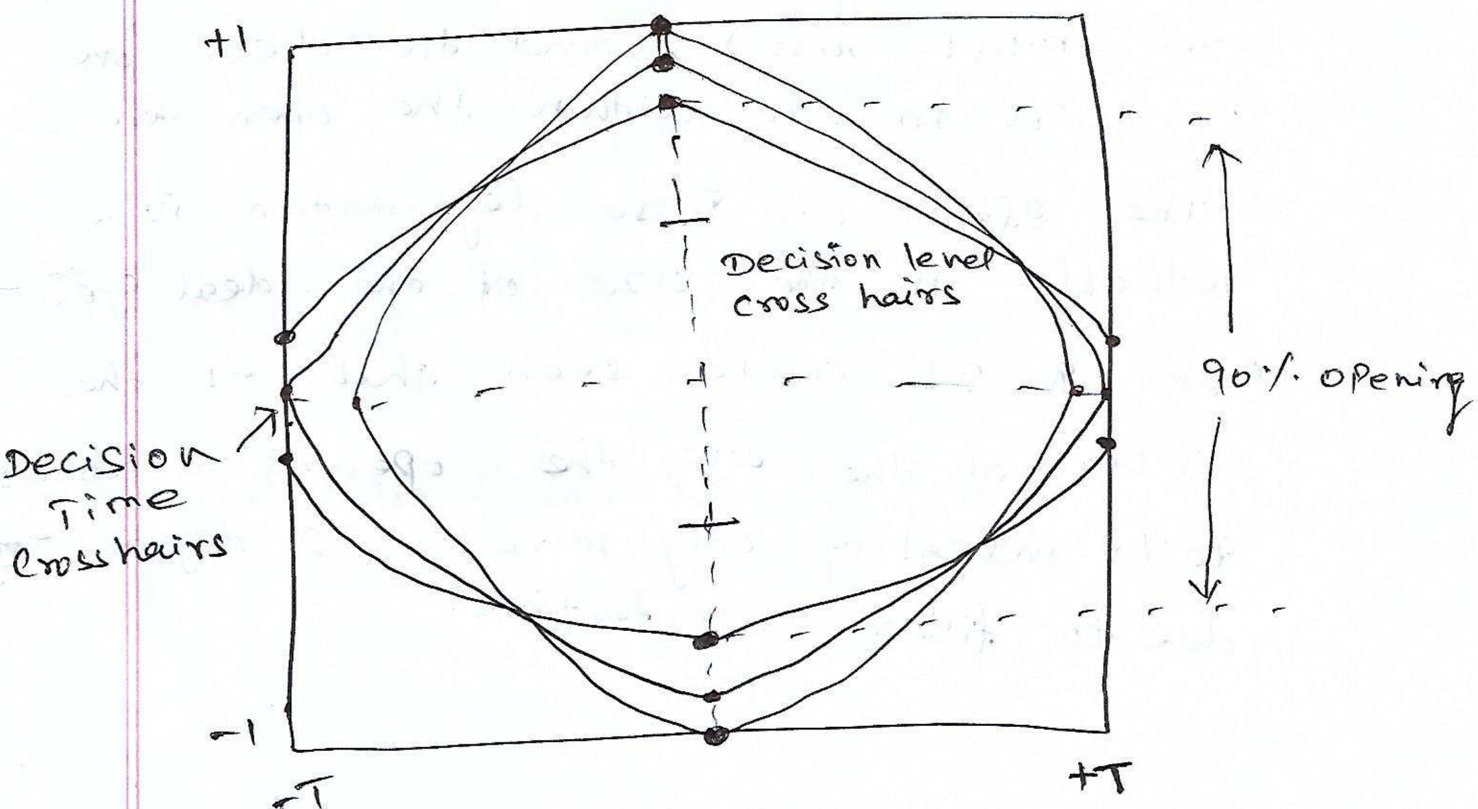
This shows that if c_k is received with error then $\hat{a}_k$ will also have error. This error will propagate in the output sequence.

Disadvantage:

Error propagation takes place in the decoder.

Eye Pattern:

* The performance of a digital txn, system depends, in part, on the ability of a repeater to regenerate the original pulses.
* ∴ The performance can be measured by displaying the received signal on an oscilloscope. Such a display is called an eye pattern or eye diagram.
* An eye pattern is a convenient technique for determining the effects of the degradations introduced into the pulses as they travel to the regenerator.



Eye diagram

* In a m-level system, there will be m-1 separate eyes.

* The vertical lines labeled +1, 0 and -1 correspond to the ideal received amplitudes ~~decision times~~. The horizontal lines, separated by the signaling interval T, correspond to the ideal decision times.

* The eye pattern shows the quality of shaping and timing and discloses any noise and errors that might be present in the line.

* To regenerate the pulse sequence without error, the eye must be open (ie, a decision area must exist), and the decision crosshairs must be within the open area.

* The effect of pulse degradation is a reduction in the size of the ideal eye.

* In fig. It can be seen that at the center of the eye, the opening is about 90% indicating only minor ISI degradation due to filter imperfections.

UNIT-5 J. GAJAPATHI

SPREAD SPECTRUM AND MULTIPLE ACCESS

* In a communication system, It may be required to provide a form of secure communication in a hostile environment such that the transmitted signal is not easily detected or recognized by unwanted listeners. This requirement is catered to by a class of signaling techniques known collectively as spread-spectrum modulation.

* The Primary advantage of a spread-spectrum communication system is its ability to reject interference

* Unintentional interference of another user simultaneously attempting to transmit through the channel.

* Intentional Inter-ference of a hostile transmitter attempting to jam the transmitter.

* Spread Spectrum modulation
  → Bandwidth & Power could be increased, ~~But~~ in order to provide secure communication.

We discuss two types of spread spectrum Modulation

(i) Direct sequence spread spectrum

(ii) Frequency-hop spread spectrum.

* In a direct sequence spread spectrum technique, two stages of modulation are used. First the incoming data sequence is used to modulate a wideband code. This code transforms the narrowband data sequence into a noise-like wideband signal. The resulting wideband signal undergoes a second modulation using a phase shift keying technique.

* In a frequency-hop spread spectrum technique, the spectrum of a data-modulated carrier is widened by changing the carrier frequency in a pseudo random manner.

* Both of these techniques rely on the availability of a noise-like spreading code called a pseudo-random or pseudo-noise sequence.

Pseudo-Noise Sequences:

* A pseudo-noise (PN) sequence is defined as a coded sequence of 1's and 0's with certain autocorrelation properties.

* It is usually periodic in that a sequence of 1's and 0's repeats itself exactly with a known period.

* The maximum-length sequence generator is a type of PN sequence generator, which uses linear feedback shift register.

* A shift-register of m consists of m flipflops (two-state memory stages).

* In order to prevent the shift register from emptying by the end of m clock pulses, we use a logical function, to compute a feedback term, and apply it to the input of the first flip-flop.

Example:

Consider a linear feedback shift register, where feedback is obtained using modulo-2 addition of the outputs of the various flipflops.

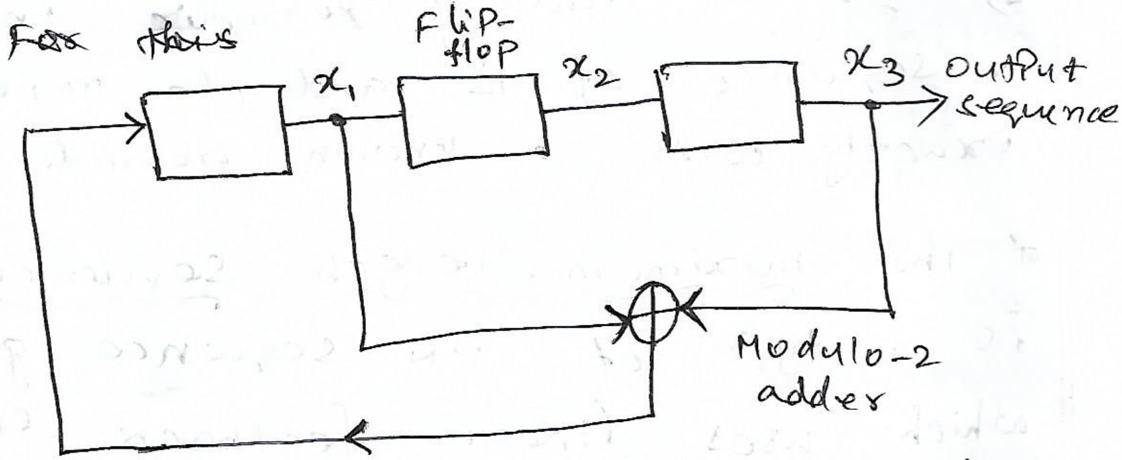


Fig.1: Maximum-length sequence generator.

* For this case $m=3$ (no. of flipflops) states of the flipflops are represented as x_1, x_2 and x_3. The feedback function is equal to the modulo-2 sum of x_1 and x_3.

* Assume the initial state of the shift register is 100 (reading the contents of the three flipflops from left to right),

* Then, the succession of states will be as follows:

100, 110, 111, 011, 101, 010, 001, 100....

The output sequence (the last position of each state of the shift register) is therefore

0 0 1 1 1 0 1

which repeat itself with period 7.

* 000 is not a state of the shift register sequence since this results in a "catastrophic cyclic code" (ie., once the state 000 is entered, the shift register cannot leave this state).

Properties of Maximum-length sequences:

Property 1: Balance property

In each period of a maximum-length sequence, the number of 1s is always one more than the number of 0's.

eg: Consider the maximum-length sequence generated by a feedback shift register.

$$\{c_n\} = \underbrace{0\,0\,1\,1\,1\,0\,1}_{N=7}\ldots\ldots$$

we see that there are three 0s and four 1s in one period of the sequence, which satisfies property 1.

Property 2 : Run Property:

Among the runs of 1s and 0s in each period of a maximum-length sequence, one-half the runs of each kind are of length one, one-fourth are of length two, one-eighth are of length three, and so on as long as these fractions represent meaningful numbers of runs.

eg: $\{c_n\} = \underbrace{0\,0\,1\,1\,1\,0\,1}_{N=7}\ldots$

with $N=7$, there are a total of four runs in one period of the sequence.

They are $\Rightarrow$ 00 , 111 , 0 and 1.

Two of the runs (a half of the total) are of length one, and one run (a quarter of the total) is of length two, which satisfies property 2.

Property 3: Correlation property:

The auto correlation function of a maximum-length sequence is periodic and binary valued.

* Let binary symbols 0 and 1 represented by −1 volt and +1 volt, respectively. By definition the auto correlation sequence of a binary sequence $\{c_n\}$, is

$$R_c(K) = \frac{1}{N} \sum_{n=1}^{N} c_n c_{n-k}$$

* For a maximum length sequence of length N, the auto correlation sequence is Periodic with Period N and two valued, as shown by,

$$R_c(K) = \begin{cases} \frac{1}{N} & k = lN \\ -\frac{1}{N} & k \neq lN \end{cases}$$

Where, $N \rightarrow$ length or period of the sequence,

$k \rightarrow$ lag of the auto correlation sequence

$l \rightarrow$ any integer

* when the length N is infinitely large, the auto correlation sequence $R_c(K)$ approaches that of a completely random binary sequence.

eg: fig.(a) shows two full periods of the maximum-length sequence. Fig.(b) shows the corresponding auto correlation function $R_c(\tau)$ plotted as a function of the Time lag (τ)

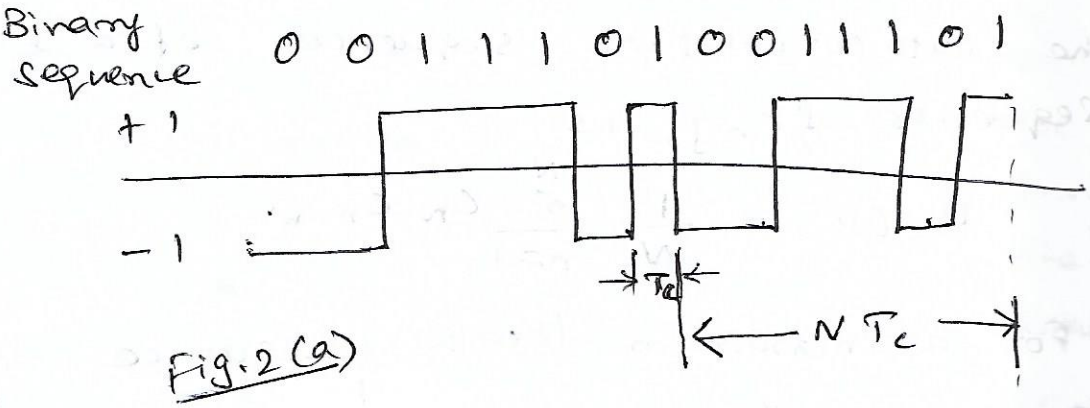


Fig.2(a)

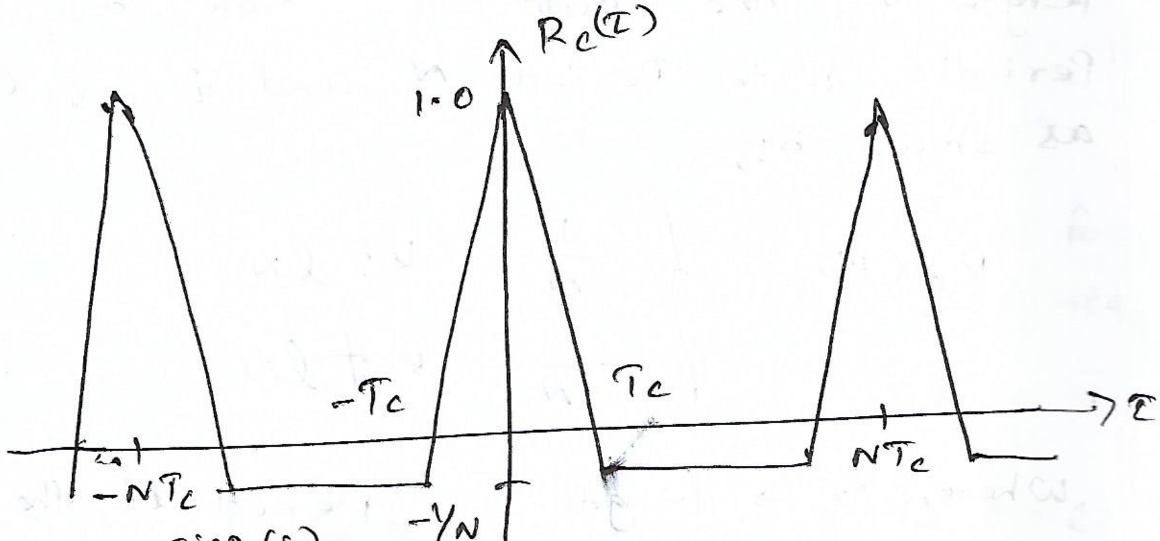


Fig2(b)

* The Parameter T_c denotes the duration of binary symbol 1 or 0 in the sequence,

Spread spectrum:

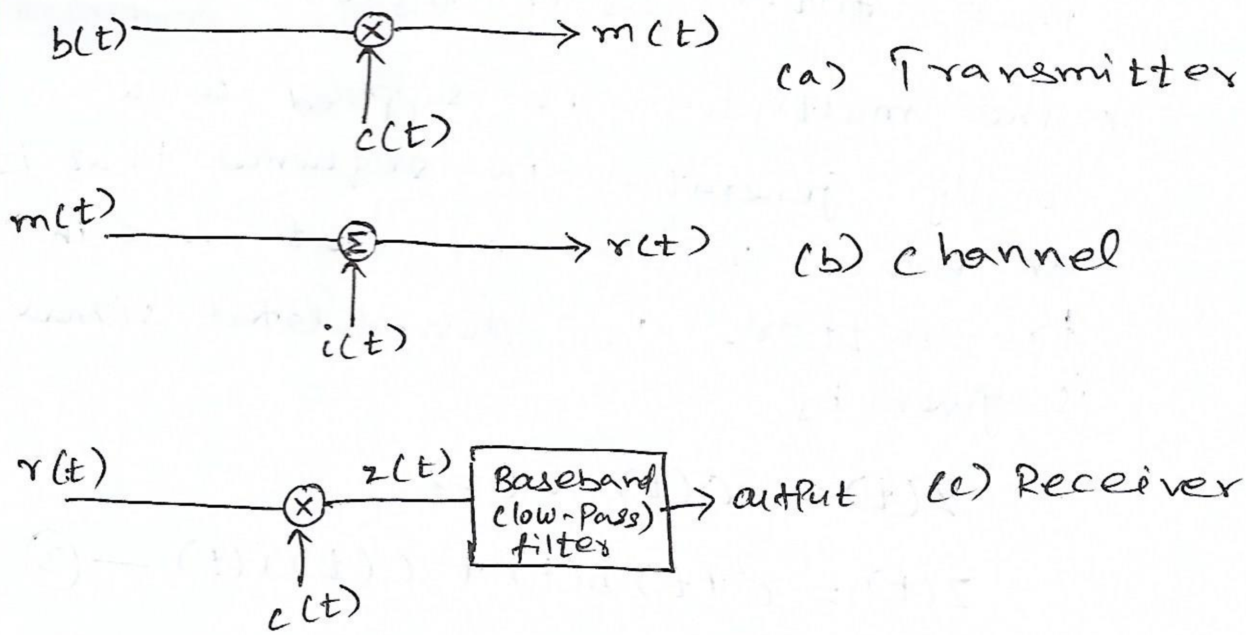
S. GAJAPATHI

* Spread spectrum modulation can provide protection against externally generated interfering (jamming) signals.

* Protection against jamming waveforms is provided by purposely making the information-bearing signal occupy a bandwidth far in excess of the minimum bandwidth necessary to transmit it.

* This has the effect of making the transmitted signal assume a noise-like appearance so as to blend into the background.

* One method of widening the bandwidth of an data sequence involves the use of modulation.



Fig(3) Idealized model of baseband spread spectrum system

* a data sequence b(t) is multiplied with the spreading code c(t), The transmitted signal m(t) is

$$m(t) = c(t)\, b(t) \quad \text{——— ①}$$

having wider Bandwidth

* The received signal r(t) consists of the transmitted signal m(t) plus an additive interference denoted by i(t), Hence

$$r(t) = m(t) + i(t)$$

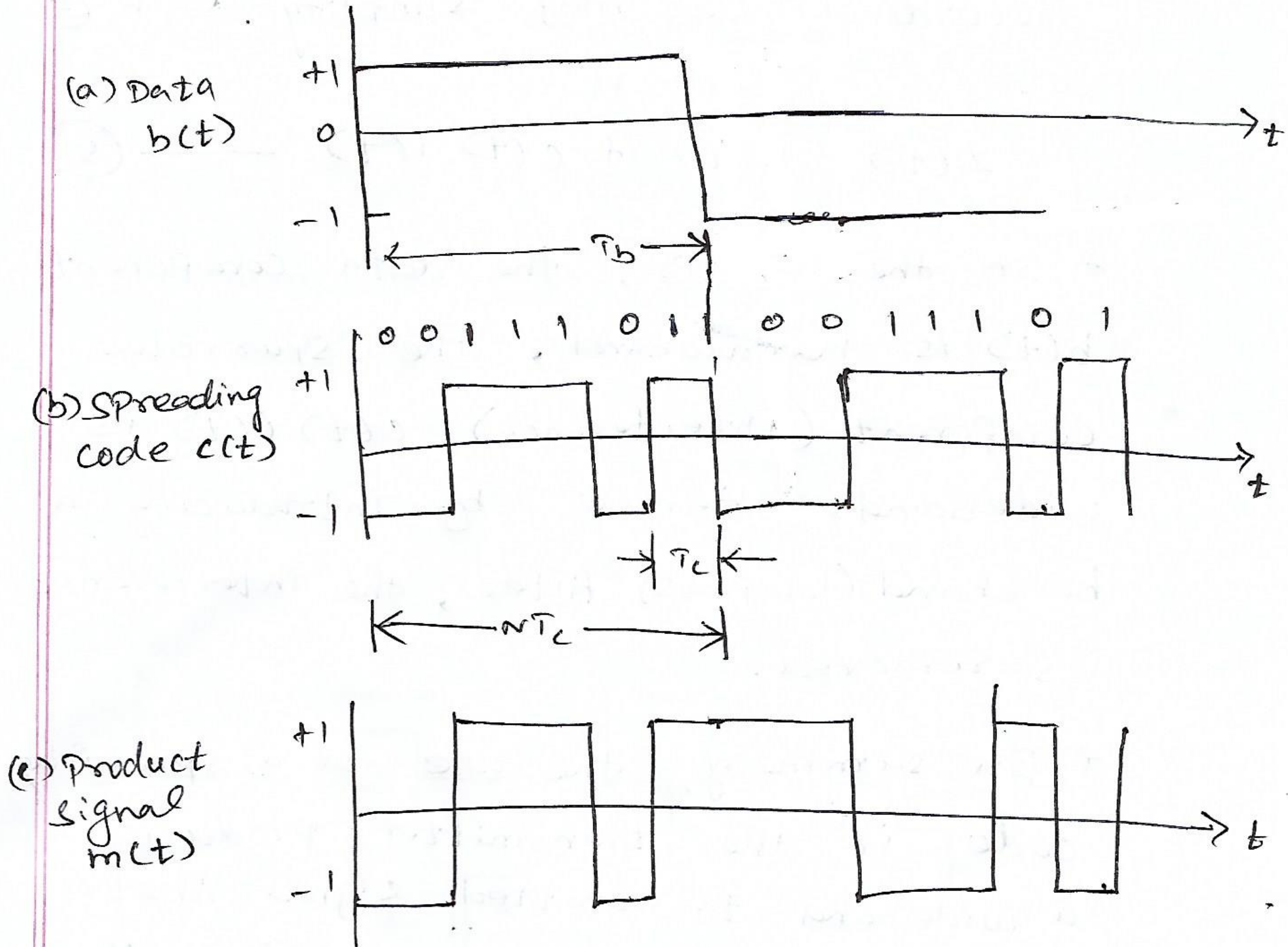
$$r(t) = c(t)\, b(t) + i(t) \quad \text{——— ②}$$

* To recover the original data sequence b(t), the received signal r(t) is applied to a demodulator, that consists of a multiplier followed by a low-pass filter as in fig (c).

* The multiplier is supplied with a locally generated PN sequence that is an exact replica of that used in the transmitter. The demodulated signal is given by,

$$z(t) = c(t)\, r(t)$$

$$z(t) = c^2(t)\, b(t) + c(t)\, i(t) \quad \text{— ③}$$



Fig(4): waveforms in the transmitter:

* Eqn. 3 shows that the desired signal b(t) is multiplied twice by the spreading code c(t), whereas the unwanted signal i(t) is multiplied only once. The spreading code c(t) alternates between -1 and +1. and the alternation is destroyed when it is squared; hence

$$c^2(t) = 1 \quad \text{for all } t \quad \text{——— (4)}$$

Accordingly we may simplify eqn ③ as

$$z(t) = b(t) + c(t)\, i(t) \quad \text{——— (5)}$$

* In the eqn ⑤, the data component b(t) is narrowband, the spurious component (interference) c(t) i(t) is wideband. Hence by introducing a baseband (lowpass) filter, the interference is removed.

* In summary, the use of a spreading code in the transmitter produces a wideband transmitted signal that appear noise-like to a receiver that has no knowledge of the spreading code.

* Therefore, To get improved protection against interference we have to tradeoff transmission bandwidth, system complexity and processing Delay.

* This spread-spectrum technique is referred as "Direct-sequence spread spectrum".

Direct-sequence spread coherent Binary Phase-shift Keying:

* TO use the Direct sequence spread spectrum technique in a band-pass channel (eg., satellite channel), we may incorporate coherent binary phase-shift keying (PSK) into the transmitter and receiver as shown in Fig (5).

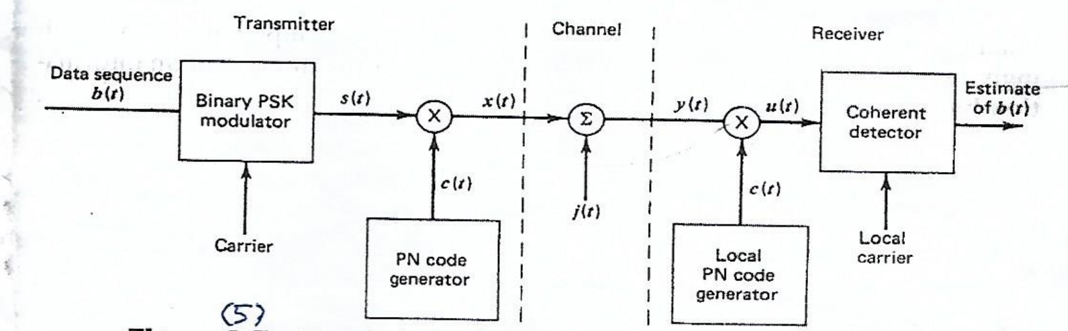


Figure (5) Model of direct-sequence spread binary PSK system.

* The transmitter section involves two stages of modulation. The second stage consists of a product modulator (or) multiplier with the data sequence and PN sequence as inputs. the first stage consists of a binary PSK modulator. The transmitted signal $x(t)$ is thus a direct-sequence spread binary phase-shift-keyed (DS/BPSK) signal.

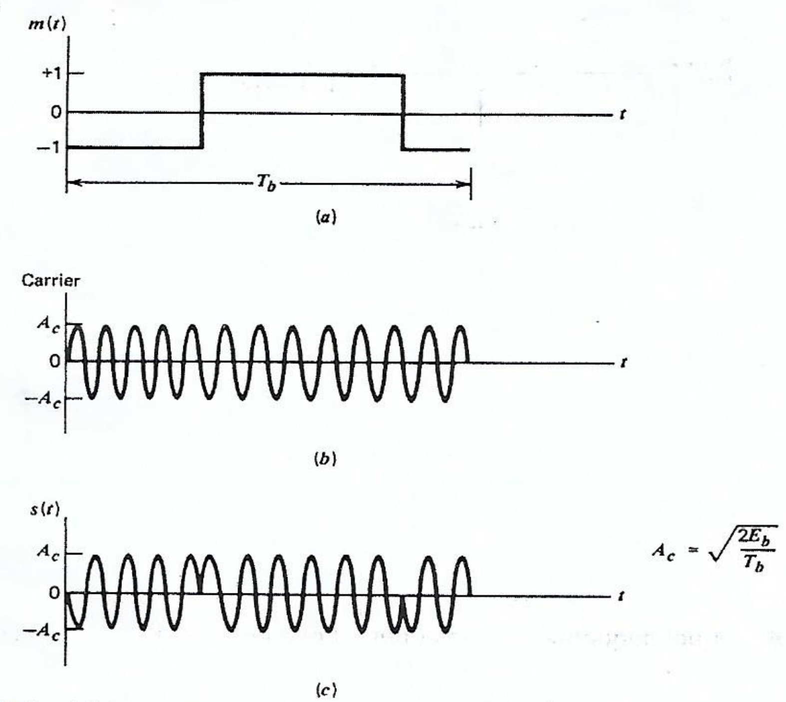


Figure 6 (a) Product signal $m(t) = c(t)b(t)$. (b) Sinusoidal carrier. (c) DS/BPSK signal.

* The phase modulation $\theta(t)$ of $x(t)$ has one of two values, 0 and π, depending on the polarities of the data sequence $b(t)$ and PN sequence $c(t)$ at time t in accordance with the truth table given by,

		Polarity of Data sequence b(t) at Time t	
		+	−
Polarity of PN sequence c(t) at time t	+	0	π
	−	π	0

* Similarly, the receiver consists of two stages of demodulation.

* In the first stage the received signal $y(t)$ and a locally generated replica of the PN sequence are applied to a multiplier.

* The second stage of demodulation consists of a coherent detector, the output of which provides an estimate of the original data sequence.

* The o/p of the multiplier is given by,

$$u(t) = c(t)\, y(t)$$

$$u(t) = c^2(t)s(t) + c(t)j(t)$$

$$u(t) = s(t) + c(t)j(t)$$

$\because y(t) = x(t) + j(t)$

$y(t) = c(t)s(t) + j(t)$

&

$c^2(t) = 1$ for all t

The above eqn. shows that the coherent detector input $u(t)$ consists of a binary PSK signal $s(t)$ imbedded in additive interference denoted by $c(t)\, j(t)$.

Performance parameters of DSSS system:

1. Processing gain (PG)
2. Probability of error (Pe)
3. Jamming Margin

1. Processing gain (PG)

↳ Achieved by processing a spread spectrum signal over an un spread signal.

* Processing gain is the ratio of the bandwidth of the spread spectrum signal to the bandwidth of the unspreaded signal.

$$\therefore \text{Processing gain} = \frac{\text{Bw of spread spectrum signal (after mixing)}}{\text{Bw of unspreaded signal (PN seq)}}$$

$$= \frac{1/T_c}{1/T_b}$$

reciprocal of one bit period.

$$= \frac{T_b}{T_c}$$

T_b → Time Period of PN seq.

T_c → Time period of spreaded sig.

Hence, for longer PN sequence, the processing gain will be larger.

2. Probability of Error:

The probability of error (P_e) of a coherent BPSK system is given by,

$$P_e = \frac{1}{2} \operatorname{erfc}\sqrt{\frac{E_b}{N_0}} \quad \text{———} \quad (1)$$

where,

E_b = Energy per bit

$N_0/2$ = Power spectral density of white noise.

* In DS-BPSK system, the interference maybe treated as a wideband noise signal with a power spectral density of $N_0/2$.

$$\therefore \frac{N_0}{2} = \frac{J T_c}{2}$$

$$N_0 = J T_c \quad \text{———} \quad (2)$$

where,

J = Average interference power

T_c = chip duration.

Sub eqn (2) in (1)

$$\boxed{P_e = \frac{1}{2} \operatorname{erfc}\sqrt{\frac{E_b}{J T_c}}}$$

3. Antijam characteristics (Jamming Margin)

Jamming Margin is the ratio of average interference power J and the signal power Ps.

$$\text{Jamming Margin} \Rightarrow \frac{J}{P_s} = \frac{PG \text{ (Processing gain)}}{E_b/N_0}$$

Jamming Margin canbe expressed in dB as,

$$\text{Jamming Margin (dB)} = \text{Processing Gain (dB)} - 10 \log_{10}\left[\frac{E_b}{N_0}\right]_{min.}$$

where,

$E_b = P_s T_b \rightarrow$ Energy per bit

$P \rightarrow$ average signal power

$T_b \rightarrow$ Bit duration

$N_0 = J T_c \rightarrow$ Power spectral density.

$T_c \rightarrow$ chip duration

$J \rightarrow$ Average interference power.

Frequency Hop Spread Spectrum: (FHSS)

* In DSSS, it may turn out that the processing gain so attained is still not large enough to overcome the effects of some jammers of concern, in which case we have to resort to other methods.

* One such alternative method is to force the jammer to cover a wider spectrum by randomly hopping the data-modulated carrier from one frequency to the next.

* The type of spread spectrum in which the carrier hops randomly from one frequency to another is called "frequency-hop (FH) spread spectrum.

* A common modulation format for FH systems is that of "M-ary frequency shift keying" (MFSK). The combination is referred to simply as FH/MFSK.

* Considering the rate at which the hops occuring, we may identify two basic characterization of frequency hopping:

1. Slow-frequency hopping, in which the symbol rate R_s of the MFSK signal is an integer multiple of the hop rate R_h. That is, several symbols are transmitted on each frequency hop.

2. Fast-frequency hopping, in which the hop rate R_h is an integer multiple of the MFSK symbol rate R_s. That is, the carrier frequency will change or hop several times during the transmission of one symbol.

Slow-Frequency Hoping:

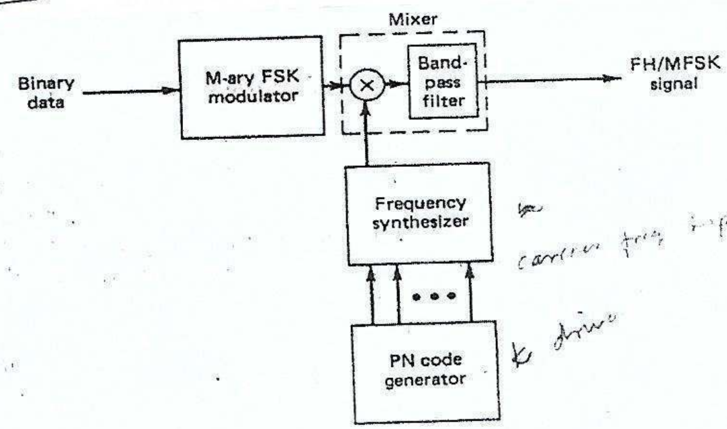


Fig (1)

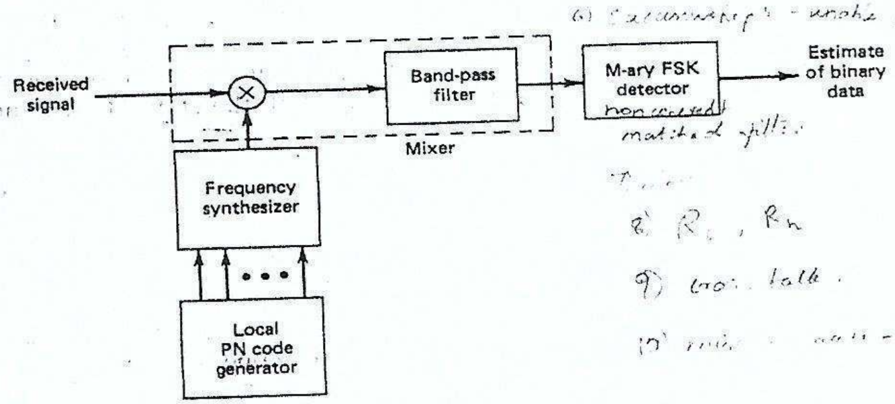


Fig. (2)

Figure 9.8 Frequency hop spread M-ary frequency-shift keying. (1) Transmitter. (2) Receiver.

Slow frequency hopping:

* Fig-1 shows the block diagram of an FH/MFSK transmitter. first the incoming binary data are applied to an M-ary FSK modulator.

* The resulting modulated wave and the output from a digital frequency synthesizer are then applied to a mixer that consists of a multiplier followed by a filter.

* Successive K-bit segments of a PN sequence drive the frequency synthesizer, which enables the carrier frequency hop over 2^k distinct values.

* On a single hop, the bandwidth of the transmitted signal is same as like a conventional MFSK format. However for a complete range of 2^k frequency hops, the transmitted FH/MFSK signal occupies a much larger bandwidth.

* An implication of these large FH bandwidths is that coherent detection is possible only within each ~~hop~~ hop, because frequency synthesizers are unable to maintain phase coherence over successive hops.

* In the receiver depicted in fig. 2 the frequency hopping is first removed by mixing the received signal with the output of a local frequency synthesizer, that is synchronously controlled in the same manner as that in the transmitter.

* The resulting output is then band-pass filtered, and subsequently processed by a non coherent M-ary FSK detector.

* An estimate of the original symbol transmitted is obtained by selecting the largest filter output.

* An individual FH/MFSK tone of shortest duration is referred as "a chip". The chip rate R_c, for an FH/MFSK system is defined by

$$R_c = \max(R_h, R_s)$$

where, $R_h \rightarrow$ hop rate,

$R_s \rightarrow$ Symbol rate.

* A slow FH/MFSK signal is characterized by having multiple symbols transmitted per hop. Hence each symbol of a slow FH/MFSK signal is a chip. Correspondingly, in a slow FH/MFSK system, the bit rate R_b of the incoming binary data, the symbol rate R_s of the MFSK symbol, the chip rate R_c and the hop rate R_h are related by,

$$R_c = R_s = \frac{R_b}{K} \geq R_h$$

where, $K = \log_2 M$.

Eg:

* Fig 3(a) illustrates the variation of the frequency of a slow FH/MFSK signal with time for one complete period of the PN

sequence. The Period of the PN sequence is $2^4 - 1 = 15$. The FH/MFSK signal has the following Parameters:

No. of bits per MFSK symbol $K = 2$

No. of MFSK tones $M = 2^K = 4$

Length of PN segment per hop $k = 3$

Total no. of frequency hops $2^k = 8$

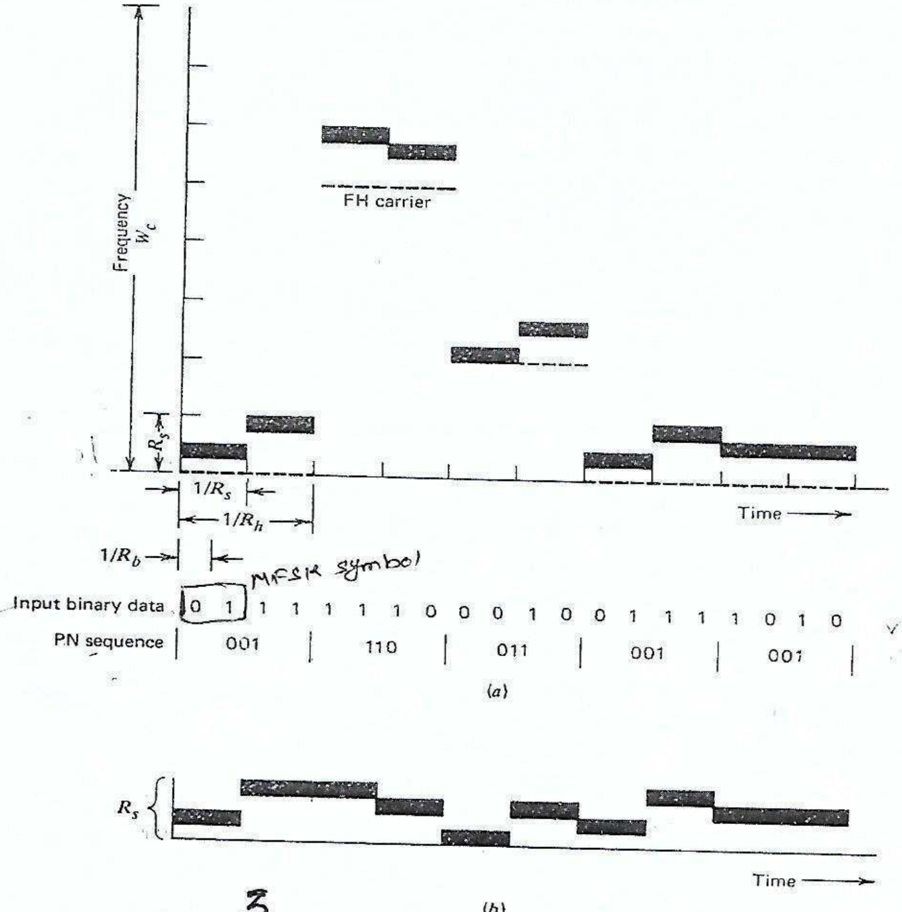


Figure 3 Illustrating slow-frequency hopping.

* In this example, the carrier is hopped to a new frequency after transmitting two symbols or equivalently, four information bits.

* fig 3.a also includes the input binary data and the PN sequence controlling the selection

of FH carrier frequency.

* Although there are eight distinct frequencies available for hopping, only three of them are utilized by the PN sequence.

* Fig-3b shows the variation of the dehopped frequency with time. This variation is recognized to be the same as that of a conventional MFSK signal produced by the given input data.

Fast-frequency Hopping:

* A fast FH/MFSK system differs from a slow FH/MFSK system in that there are multiple hops per M-ary symbol.

* In general, fast-frequency hopping is used to defeat a smart jammer's tactic that involves two functions: measurement of the spectral content of the transmitted signal, and retuning of the interfering signal to that portion of the frequency band.

* Clearly, to overcome the jammer, the transmitted signal must be hopped to a new carrier frequency before the jammer is able to complete the processing of these two functions.

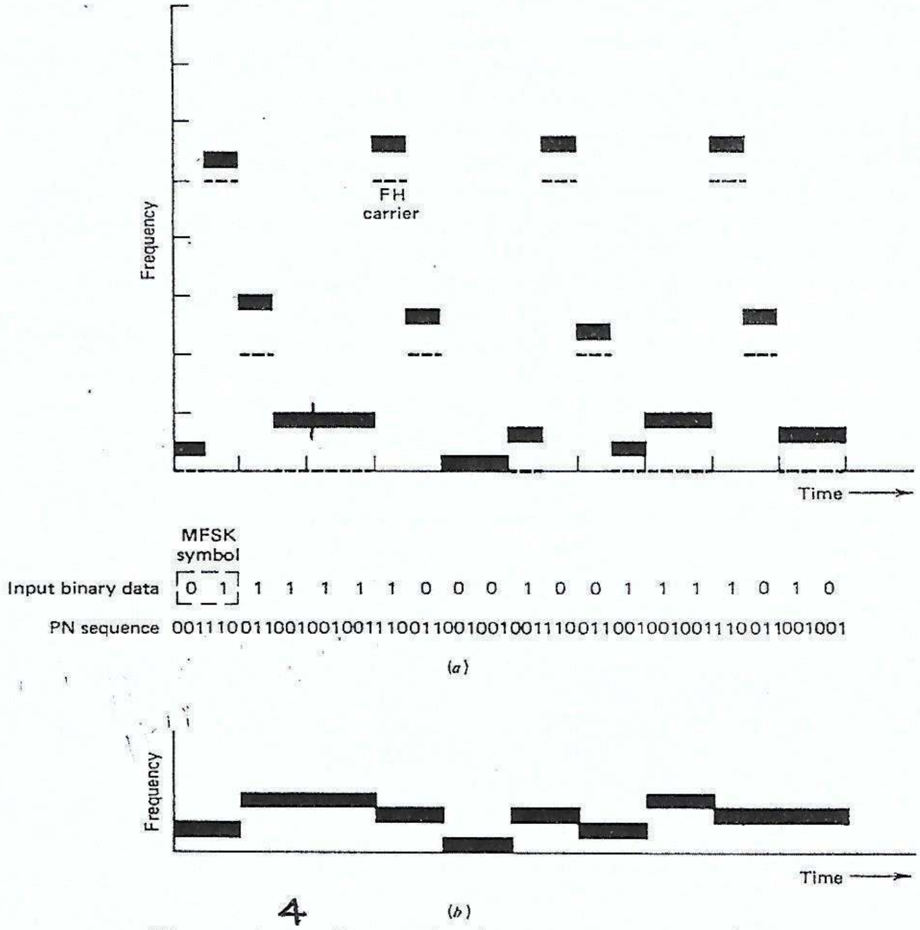


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Figure 9.[redacted] Illustrating fast-frequency hopping.

* Fig. 4 (a) illustrates the variation of the transmitted frequency of a fast FH/MFSK signal with time. The signal has the following parameters:

No. of bits per MFSK symbol $K = 2$

No. of MFSK tones $M = 2^K = 4$

Length of PN segment per hop $k = 3$

Total no. of frequency hops $2^k = 8$

In this example, each MFSK symbol has the same number of bits and chips; that is the chip rate R_c is the same as the bit rate R_b. After each chip, the carrier frequency of the transmitted MFSK signal is hopped to a different value, except for few occasions when the k-chip segment of the PN sequence repeats itself.

* Fig 4(b) depicts the time variation of the frequency of the dehopped MFSK signal.

* For data recovery at the receiver, non-coherent detection is used. For this, two procedures may be considered.

1. For each FH/MFSK symbol, separate decisions are made on the K frequency-hop chips received, and a simple rule based on majority vote is used to make an estimate of the dehopped MFSK symbol.

2. For each FH/MFSK symbol, likelihood functions are computed as functions of the total signal received over K chips, and the larger one is selected.

A receiver based on the second procedure is optimum.

Synchronization:

* For its proper operation, spread spectrum communication requires that the locally generated PN sequence used in the receiver to despread the received signal be synchronized to the PN sequence used to spread the transmitted signal in the transmitter.

* A solution to the synchronization problem consists of two parts:

⇒ acquisition

⇒ tracking

* In acquisition, or coarse synchronization the two PN codes are aligned to within a fraction of a chip in as short a time as possible.

* Once the incoming PN code has been acquired, tracking or fine synchronization takes place.

* Typically PN acquisition proceeds in two steps, first the received signal is multiplied by a locally generated PN code to produce a measure of correlation between it and the PN code used in the transmitter.

* Next, an appropriate decision-rule and search strategy is used to process the measure of correlation so obtained to determine whether the two codes are in synchronism and what to do if they are not.

* As for tracking, it is accomplished using phase-lock techniques very similar to those used for the local generation of coherent carrier references.

S.GASAPATHI

MULTIPLE ACCESS TECHNIQUES

* Multiple Accessing implies that more than one Satellite user has access to one or more radio channels within a satellite communications channel.

* Three commonly used multiple-accessing arrangements are:

(i) Frequency-division Multiple Access (FDMA)

(ii) Time-division Multiple Access (TDMA)

(iii) Code-division Multiple Access (CDMA).

(i) Frequency Division Multiple Access:

* With FDMA, each earth station's transmission are assigned specific uplink and downlink frequency bands within an allotted Satellite Bandwidth.

* FDMA transmissions are separated in the frequency domain and, therefore must share the total available bandwidth as well as Power.

* A given RF bandwidth is divided into smaller frequency bands called subdivisions. Each subdivision has its own IF carrier frequency.

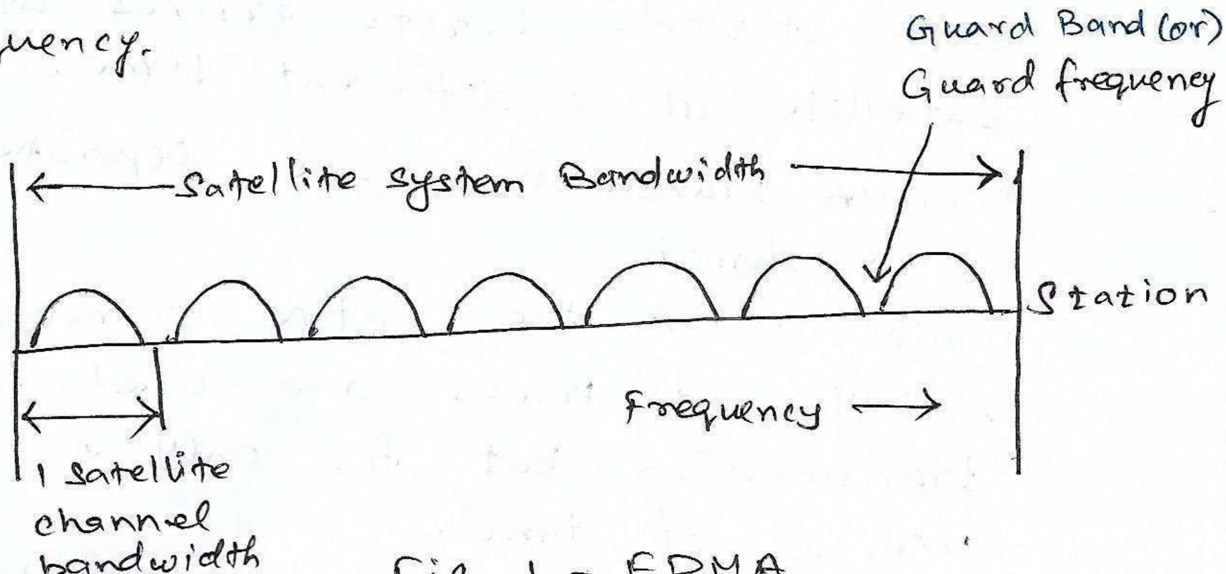

Fig-1 - FDMA

* A control mechanism is used to ensure that two or more earth stations do not transmit in the same subdivision at the same time.

* Essentially, the control mechanism designates a receive station for each of the subdivisions.

* Disadvantage:

Carriers from multiple earth stations may be present in a satellite transponder (radiochannel) at the same time. This results in Cross modulation. distortion, b/w them.

(ii) Time Division Multiple Access:

* With TDMA, each earth station transmits a short burst of information during a ~~[illegible]~~ specific time slot within a TDMA frame.

* The bursts must be synchronized so that each station's bursts arrives at the satellite at a different time.

* TDMA transmissions are separated in the time domain.

* With TDMA, the entire transponder bandwidth and power are used for each transmission but for only a prescribed interval of time.

* It is the predominant multiple-access method used today. It provides the most efficient method of transmitting digitally modulated carriers (PSK).

* Fig-2 shows a basic TDMA frame. Transmissions from all earth stations are synchronized to a reference burst.

* The reference burst contains a carrier recovery sequence (CRS) from which all receiving stations recover a frequency and phase coherent carrier for PSK demodulation.

* The Unique word (UW) sequence is used to establish a precise time reference that each of the earth stations uses to synchronize the transmission of its burst.

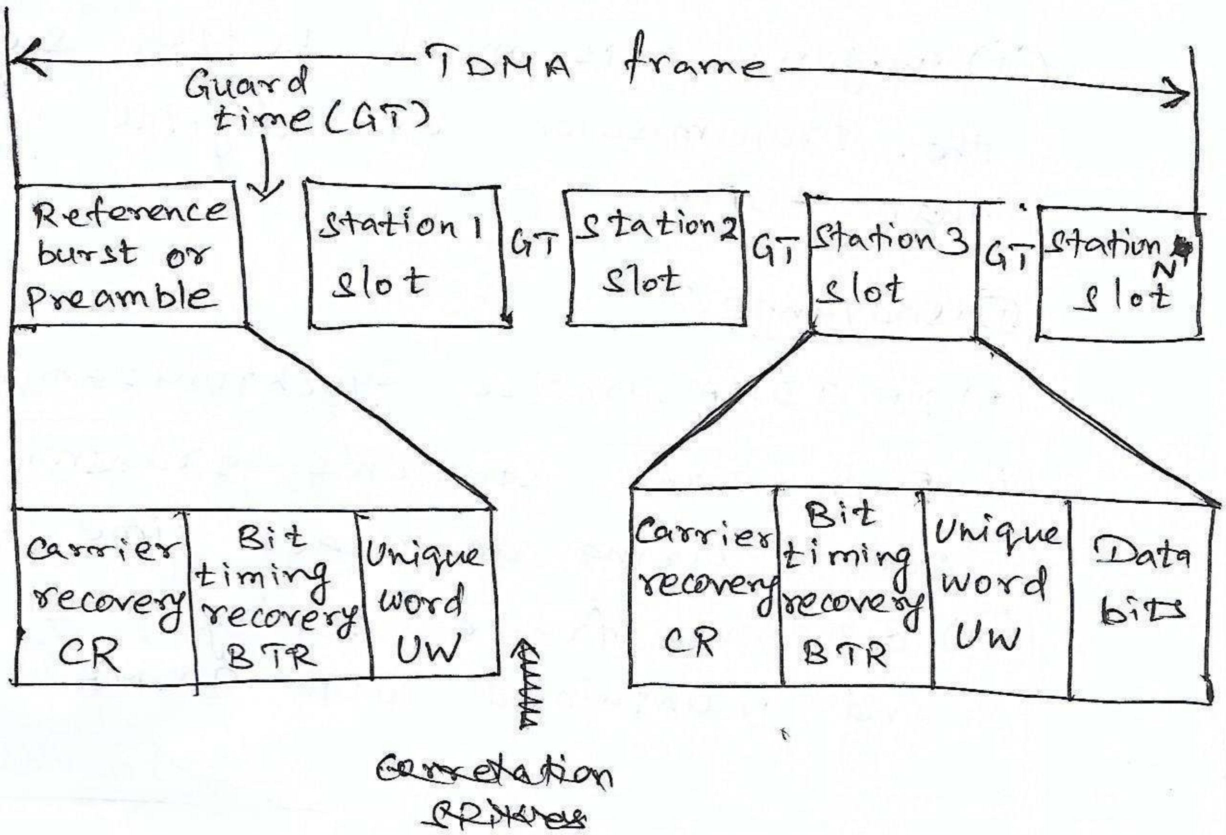

Fig-2: Basic TDMA Frame

* Guard time (GT) are introduced between successive stations to allow synchronization of the receivers.

* The preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other.

Advantages:

(i) With TDMA, only the carrier from one earth station is present in the satellite transponder at any given time, thus reducing intermodulation distortion.

(ii) ~~Digi~~ TDMA is much better suited to the transmission of digital information than FDMA.

Disadvantes:

(i) In TDMA precise synchronization is required

(ii) Each earth station's transmissions must occur during an exact time slot.

(iii) Bit and frame timing must be achieved and maintained with TDMA.

(iii) Code Division Multiple Access (CDMA)

* With CDMA there are no restrictions on time or bandwidth. Each earth station transmitter may transmit whenever it wishes and can use any or all the bandwidth allocated a particular satellite system or channel.

* Transmissions are separated through envelope encryption/decryption techniques.

* That is, each earth station's trans-missions are encoded with a unique binary word called a "chip code". Each station has a unique chip code. To receive a particular earth station's transmission, a receive station must know the chip code for that station.

* So, each user operates independently with no knowledge of other users.

Advantages:

(i) Less affected by noise.
(ii) It can accomodate more users.
(iii) Synchronization is not necessary.

Disadvantages:

(i) CDMA cannot offer international roaming as like GSM.
(ii) Data collision is possible. Near far effect and self jamming problem occurs.