Course Name : Analog and Digital Communications

Course Number : EC403PC

Course Designation : Core

Prerequisites : Signals and Systems, Probability Theory and Stochastic Processes

Prepared By

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Assistant Professor

SYLLABUS

Unit – I	**Amplitude Modulation:** Need for modulation, Amplitude Modulation - Time and frequency domain description, single tone modulation, power relations in AM waves, Generation of AM waves- Switching modulator, Detection of AM Waves - Envelope detector, DSBSC modulation - time and frequency domain description, Generation of DSBSC Waves - Balanced Modulators, Coherent detection of DSB- SC Modulated waves, COSTAS Loop, SSB modulation - time and frequency domain description, frequency discrimination and Phase discrimination methods for generating SSB, Demodulation of SSB Waves, principle of Vestigial side band modulation.
Unit – II	**Angle Modulation:** Basic concepts of Phase Modulation, Frequency Modulation: Single tone frequency modulation, Spectrum Analysis of Sinusoidal FM Wave using Bessel functions, Narrowband FM, Wide band FM, Constant Average Power, Transmission bandwidth of FM Wave - Generation of FM Signal- Armstrong Method, Detection of FM Signal: Balanced slope detector, Phase locked loop, Comparison of FM and AM., Concept of Pre-emphasis and de-emphasis.
Unit – III	**Transmitters:** Classification of Transmitters, AM Transmitters, FM Transmitters **Receivers:** Radio Receiver - Receiver Types - Tuned radio frequency receiver, Superhetrodyne receiver, RF section and Characteristics - Frequency changing and tracking, Intermediate frequency, Image frequency, AGC, Amplitude limiting, FM Receiver, Comparison of AM and FM Receivers.
Unit – IV	**Pulse Modulation:** Types of Pulse modulation- PAM, PWM and PPM. Comparison of FDM and TDM. **Pulse Code Modulation:** PCM Generation and Reconstruction, Quantization Noise, Non-Uniform QuantizationandCompanding,DPCM,AdaptiveDPCM,DMandAdaptiveDM,NoiseinPCMandDM.
Unit – V	**Digital Modulation Techniques:** ASK- Modulator, Coherent ASK Detector, FSK- Modulator, Non- Coherent FSK Detector, BPSK- Modulator, Coherent BPSK Detection. Principles of QPSK, Differential PSK and QAM. **Baseband Transmission and Optimal Reception of Digital Signal:** A Baseband Signal Receiver, Probability of Error, Optimum Receiver, Coherent Reception, ISI, Eye Diagrams.

TEXT BOOKS& OTHER REFERENCES BOOKS

Text Books	
1.	Analog and Digital Communications – Simon Haykin, John Wiley,2005.
2.	Electronics Communication Systems-Fundamentals through Advanced-Wayne Tomasi, 5th Edition, 2009, PHI.
3	Communications Systems by Simon Haykin, 2nd Edition
4	Analog and Digital Communications by Sanjay Sharma
Suggested / Reference Books	
1.	Principles of Communication Systems - Herbert Taub, Donald L Schilling, GoutamSaha, 3rd Edition, McGraw-Hill,2008.
2.	Electronic Communications – Dennis Roddy and John Coolean , 4th Edition , PEA,2004
3.	Electronics & Communication System – George Kennedy and Bernard Davis, TMH2004
4.	Analog and Digital Communication – K. Sam Shanmugam, Willey,2005

Websites References	
1.	https://en.wikipedia.org/wiki/Communications_system
2.	https://www.informit.com/articles/article.aspx?p=1157195&seqNum=6
3.	https://ict.iitk.ac.in/wp-content/uploads/EE320A-Principles-Of-Communication-CommunicationSystems-4ed-Haykin.pdf

Program Educational Objectives (PEO's)

PEO1 To develop the students knowledge in core and allied electronics and communication.

PEO2 To train the students in usage of modern tools which leads to realize a system in virtual environment?

PEO3 To provide enough training to ensure their higher education and employability in the reputed industry.

PEO4 To enhance Research and Development of the students and to appraise them in the latest trends of project management skills to work individually as an entrepreneur.

PEO5 To inculcate ethical practices, dynamic leadership qualities and effective communication skills.

PSO1 The graduates will be Equipped with knowledge of complete design flow from specification to silicon in areas of both digital and Analog VLSI Design and will be able to work in IC Design companies.

PSO2 The graduates will be Equipped with microprocessor and Microcontroller based system design skills and can work as design and verification engineers in the area of Embedded Systems Design.

PSO3 The graduates will be able to apply engineering knowledge for design and implementation of projects pertaining to signal processing and Communications.

Program Outcomes (PO's)

1. **Engineering knowledge:** Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization for the solution of complex engineering problems.
2. **Problem analysis:** Identify, formulate, research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
3. **Design/development of solutions:** Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for public health and safety, and cultural, societal, and environmental considerations.
4. **Conduct investigations of complex problems:** Use research-based knowledge and researchmethods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.
5. **Modern tool usage:** Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools, including prediction and modeling to complex engineering activities, with an understanding of the limitations.
6. **The engineer and society:** Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal, and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **Environment and sustainability:** Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **Individual and team work:** Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
10. **Communication:** Communicate effectively on complex engineering activities with the engineering community and with the society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.
11. **Project management and finance:** Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

Course Outcomes:

Upon successful completion of this course, students will be able to:

CO 1: Analyze and design of various continuous wave and angle modulation and demodulation Techniques

CO 2: Understand the effect of noise present in continuous wave and angle modulation Techniques.

CO 3: Attain the knowledge about AM, FM Transmitters and Receivers

CO 4: Analyze and design the various Pulse Modulation Techniques.

CO 5: Understand the concepts of Digital Modulation Techniques and Baseband transmission**.**

Course Schedule

Distribution of Hours Unit – Wise

Unit	Topic	Chapters		Total No. of Hours
		Book1	Book2	
I	Amplitude Modulation	TB 1		14
II	AngleModulation	TB 1		09
III	Transmitters and Receivers	RB 2		11
IV	Pulse Modulations, Pulse Code Modulation	TB 1		12
V	Digital Modulation Techniques, Baseband Transmission and Optimal Reception of Digital Signal	TB Ch9, 10		10
Contact classes for Syllabus coverage				**56**
Tutorial Classes : 05 ; Online Quiz : 1 Revision classes :1 per unit				

Number of Hours / lectures available in this Semester/

<u>The number of topic in every unit is not the same – because of the variation, all the units have an unequal distribution of hours</u>

ASSIGNMENT QUESTIONS

UNIT I:

1. Describe Amplitude modulation for single tone and draw the spectrum also?[CO 1, BL 2]
2. Describe the detection of AM wave using envelope Detector.[CO 1, BL 2]
3. Explain the generation of DSB-SC wave using ring Modulator.[CO 1, BL 2]
4. Explain the detection of DSB-SC wave using Costas loop. [CO 1, BL 2]
5. Describe the Phase discrimination method for generating SSBSC signal?[CO 1, BL 2]

UNIT II:

1. Discuss the generation of FM wave using direct method.[CO 1, BL 2]
2. What are the different demodulation techniques of FM? Explain the demodulation of F.M signal with the help of PLL.[CO 1, BL 2]
3. Compare the direct and indirect methods of generating FM signals. Explain Armstrong method of generating FM signals with a neat block schematic diagram.[CO 3, BL 4]
4. With a neat block diagram explain the generation of narrow band and wide band FM.[CO 1, BL 2]

UNIT III:

1. Draw and explain the block diagram of a high level AM transmitter [CO 3, BL 2]
2. Draw and explain the block diagram of a low level AM transmitter [CO 3, BL 2]
3. Draw the block diagram of Super heterodyne receiver and explain the function of each block. [CO 3, BL 2]
4. With neat block diagram explain the working principle of TRF receiver. [CO 3, BL 2]
5. Explain frequency changing and tracking. What is image frequency and the problems associated with image frequency? [CO 3, BL 2]

UNIT IV:

1. Explain themethodofgenerationanddetectionofPAM signalswithneatschematics.[CO 4, BL 2]
2. Whatis the different typeofPulseModulations?Explain?[CO 4, BL 2]
3. With a neat block diagram explain the Pulse code modulation system.[CO 4, BL 2]

4. What is meant by Quantization. Explain different types?[CO 4, BL 2]
5. Describe the Delta modulation indetail.[CO 4, BL 2]

UNIT V:

1. Explain about DPSK system. And also give the comparison between DPSK and PSK.[CO 5, BL 2]
2. Define eye diagram. Draw the eye diagram for FSK.[CO 5, BL 1]
3. Describe the BPSK modulation technique with the help of a neat diagram.[CO 5, BL 2]
4. Explain the concept of Inter Symbol Interference?[CO 5, BL 2]
5. What is Probability of Error.[CO 5, BL 1]

UNIT WISE SHORT ANSWER QUESTIONS

UNIT I:

1. Define communication. Explain with block diagram the basic communication system.[CO 1, BL 1]
2. Define modulation. Why is modulation required? [CO 1, BL 1]
3. Define is modulation index of AM wave? [CO 1, BL 1]
4. Describe the DSB-SC wave modulation with spectrum? [CO 1, BL 2]
5. Compare Square law detector with envelope detector? [CO 1, BL 4]

UNIT II:

1. Define modulation index and bandwidth of FM.[CO 1, BL 1]
2. Compare FM and AM.[CO 1, BL 1]
3. What is Carson's Rule?[CO 1, BL 1]
4. What is wideband FM & Narrowband FM?[CO 1, BL 1]
5. What are Advantages & Applications of FM?[CO 1, BL 1]

UNIT III:

1. What is a radio transmitter? Classify them.[CO 3, BL 1]
2. What is a radio receiver? Classify them.[CO 3, BL 1]
3. Define sensitivity of a radio receiver?[CO 3, BL 1]
4. Define selectivity of a radio receiver?[CO 3, BL 1]
5. What is fidelity?[CO 3, BL 1]

UNIT IV:

1. What are the types of pulse modulations?[CO 4, BL 1]
2. Compare PAM, PWM, PPM?[CO 4, BL 4]
3. Define sampling and sampling theorem.[CO 4, BL 1]
4. Define quantization and Quantization error?[CO 4, BL 1]
5. What is companding?[CO 4, BL 1]

UNIT V:

1. What is meant by Inter symbol interference?[CO 5, BL 1]
2. What are the uses of eye diagrams?[CO 5, BL 1]
3. Define ASK,PSK,FSK?[CO 5, BL 1]

TUTORIAL SHEETS

UNIT I:

1. What is the effect of frequency and phase over error in demodulation of DSB-SC wave using synchronous detector.[CO 1, BL 2]
2. a)Plotthe one cycle of AM wave and calculate the modulation index of it interms of Vmax and Vmin voltages[CO 1, BL 3]

   b) The rms antenna current of an AM transmitter is 10 A when un-modulated and 12 A when sinusoidally modulated. Calculate the modulation index.[CO 1, BL 3]

3. A modulating signal consists of a symmetrical triangular wave having zero dc component and peak to peak voltage of 12V. It is used to amplitude modulate a carrier of peak voltage 10V. Calculate the modulation index.[CO 1, BL 3]

UNIT II:

1. An FM radio link has a frequency deviation of 30 kHz. The modulatingFrequency is 3 kHz. Calculate the bandwidth needed for the link. What will bethe bandwidth if the deviation is reduced to 15 kHz?[CO 1, BL 3]
2. A 100 MHz carrier is frequency modulated by a sinusoidal signal of 10 KHz so that the maximum frequency deviation is 1 MHz. Determine the approximate bandwidth of the FM carrier. Now find the bandwidth of the FM carrier if the modulating signal amplitude is doubled. Determine the bandwidth of the FM carrier if the frequency of the modulating signal is also doubled.[CO 1, BL 3]
3. For an FM modulator with a modulating signal m(t)= $V_m \sin(300\times10^3 t)$, the carrier Signal $V_c(t)=8 \sin(6.5\times10^6 t)$ and the modulator index = 2. Find out the significant side band frequencies and their amplitudes.[CO 1, BL 3]

UNIT III:

1. Of all the frequencies that must be rejected by a superheterodyne receiver, why is the image frequency so important? What is the image frequency and how does it arise? If the image-frequency rejection of a receiver is insufficient, what steps could be taken toimprove it?[CO 4, BL 4]

2. In a broadcast super heterodyne receiver having no RF amplifier, the loadedQ of the antenna coupling circuit is 100. If the IF frequency is 455 kHz,determine the image frequency and its rejection ratio for tuning at 1.1. kHz astation.[CO 4, BL 3]

UNIT IV:

1. Write the advantages of digital communication. [CO 4, BL 1]
2. With neat sketch explain the TDM multiplexing and demultiplexing.[CO 4, BL 2]
3. Discuss the Delta modulation technique. Also discuss the noises in DM.[CO 4, BL 2]

UNIT V:

1. The bit stream 1011100011 is to be transmitted using DPSK. Determine the encoded sequence and transmitted phase sequence.[CO 5, BL 3]
2. Explain the working of non-coherent FSK detector.[CO 5, BL 2]
3. Draw and explain the working of optimum receiver with a neat diagram.[CO 5, BL 3]

TOPICS BEYOND SYLLABUS

ANALOG AND DIGITAL COMMUNICATIONS

- Fourier transform, Fourier Series. [CO 1]
- AM Square Law Modulator.[CO 1]
- AM square Law Demodulator.[CO 1]
- Hilbert Transform.[CO 1]

- Generation and Detection of Pulse Analog Modulations[CO 4]

INTRODUCTION TO COMMUNICATION SYSTEM:

→ Communication is the process whereby information is transferred from one point called the source to the other point called destination.

→ Information is basically the news one wishes to convey. Every news doesnot convery the same amount of information.

→ Message is the physical manifestation of the information as produced by the source.

→ Signal in communication is the electrical analog of the message at the source.

→ First telegraphic message invented by Samuel. F.B.Morse in 1838

Voice - 300Hz - 3.5K
Audio - 20Hz - 20kHz
Video - 0 - 4.5kHz

Elements of a Communication System:

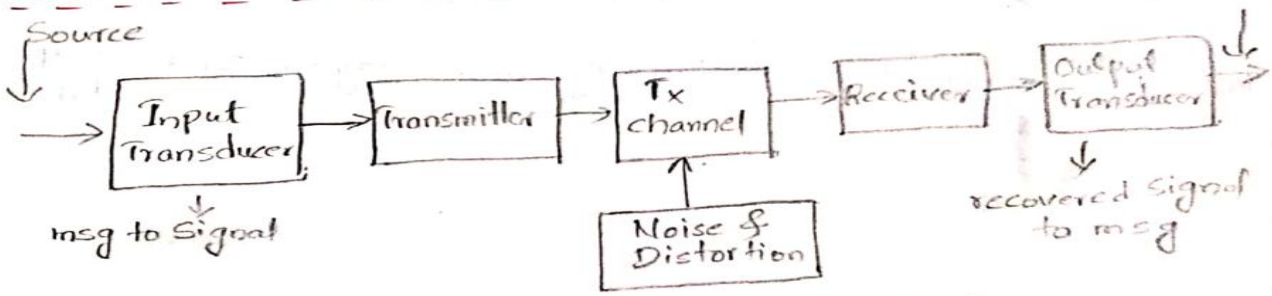


Fig1.1: Elements of Communication System.

Transmitter: The purpose of a transmitter is to modify the message signal to a suitable form for transmission over the communication channel. This can be achieved through a process called "Modulation."

Transmission channel: This is a medium which electrically connects the transmitter to the receiver. (pairs of wires, a coaxial cable, free space, optical fiber etc). The properties of the channel can strongly influence the performance of the communication system. Eg: Non-linearities, imperfection

Receiver: The main function of this unit is to reproduce the original message from the distorted signal available at the input of it. The reproduction of the signal is accomplished by the process known as demodulation (or) detection.

Ref: CS by Simon Haykin

Input Transducer: The input transducer converts the message to an electrical signal (Voltage or Current)

Eg: Microphone, Camera, Keyboard.

Output Transducer: The output transducer converts the output signal to the desired message form.

Eg: Speaker, Monitor

Ref: 1) Communication Systems by Simon Haykin, 2 Ed, Wiley Pub.
Page. No: 7, 8

* NEED FOR MODULATION:

Modulation:

- The purpose of modulation is to convert the signal to a suitable form to match the transmission medium, because the message signal is a low frequency signal & cannot be transmitted efficiently over the channel directly.
- The transmission channel is suited for high frequency signal transmission & the high frequency signals are called "Carriers".
- Modulation is a scheme which alters some characteristics of the high frequency carrier in accordance with the low frequency message signal called 'modulating signal'.

– Types of Modulation:-

i) Continuous Wave (CW) modulation (Continuous Process) — Carrier Signal → Sinusoidal

ii) Pulse modulation (discrete Process) — Carrier Signal is Pulse Train

- Continuous time varying signal can also be discretised by sampling.

– Need for Modulation:

- The process of modulation serves the following purposes.

i) Efficient radiation:

- In radio communication the information is transmitted in the form of electromagnetic waves from the transmitting antenna. For efficient radiation it is necessary that the size of the antenna (element) should be of the order of $\lambda/4$, $\lambda \rightarrow$ wavelength of the signal to be radiated.

Audio signals have low freq's

eg: 100 Hz $\lambda = \frac{c}{f} = \frac{3\times10^8}{100} = 300\times10^4$ Size $= \frac{300\times10^4}{4}$ $= 300$ km

· With the help of modulation the low frequency signal can be translated to higher frequency range & radiated efficiently from reduced size antenna

ii) Frequency Translation:

· Modulation enables one to translate the signals occupying similar frequency ranges to different regions in the frequency spectrum. This allows a user to tune his radio (or) television set to a particular broadcasting station.

iii) Multiplexing:

· Modulation scheme enables one to multiplex a no. of signals at the same time in a single channel without any interference among themselves. This scheme is utilised in long distance telephony, data telemetry etc.

iv) Reduction of noise:

· Noise & interference are two major limitations of any communication system. & cannot be eliminated totally. Certain modulation schemes can ~~Surpo~~ Supress the noise & interference to some extent.

→ AMPLITUDE MODULATION:

Def: Amplitude modulation (AM) is defined as a process in which the amplitude of the carrier wave $c(t)$ is varied about a mean value, linearly with the baseband signal $m(t)$

→ TIME DOMAIN AND FREQUENCY DOMAIN DESCRIPTION:

Consider a sinusoidal carrier wave $c(t)$ defined by.

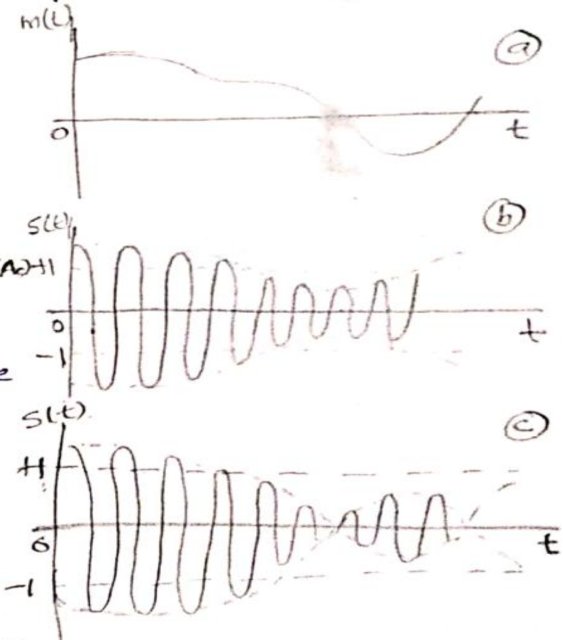
$$c(t) = A_c \cos(2\pi f_c t) \longrightarrow ①$$

Let $m(t)$ denote the baseband signal which carries the specification of the message. The carrier wave $c(t)$ is independent of $m(t)$.

An AM wave can be described as a function of time ~~in the~~ as.

$$s(t) = A_c\left[1 + K_a m(t)\right]\cos(2\pi f_c t) \rightarrow ②$$

$K_a \rightarrow$ a Const called the amplitude sensitivity of modulator



a → baseband signal $m(t)$

b → AM wave for $|K_a m(t)| < 1$

c → AM wave for $|K_a m(t)| > 1$

- From fig b), the envelope of $s(t)$ has the same shape as the baseband signal $m(t)$. Provided two requirements are satisfied.

1) The amplitude of $K_a m(t)$ is always less than unity, i.e., $|K_a m(t)| < 1$, for all 't'.
   The absolute maximum value of $K_a m(t)$ multiplied by 100 is referred to as % modulation

2) The carrier frequency f_c is much greater than the highest freq. component W of the message signal $m(t)$.
   $f_c \gg W$.

- Spectrum of AM Signal:

Consider the AM signal given by

$$s(t) = A_c[1 + K_a m(t)]\cos(2\pi f_c t)$$

Applying Fourier Transform to the AM signal $s(t)$

$$S(f) = \int_{-\infty}^{\infty} A_c[1+K_a m(t)]\cos(2\pi f_c t).\, e^{-j2\pi f t}\, dt$$

$$= \int_{-\infty}^{\infty}\left[A_c \cos(2\pi f_c t)\, e^{-j2\pi f t} + A_c K_a m(t)\cos(2\pi f_c t)\, e^{-j2\pi f t}\right] dt$$

$$\left\{\cos(\theta) = \frac{e^{j\theta} + e^{-j\theta}}{2}\right\}$$

$$= \frac{A_c}{2}\left\{\int_{-\infty}^{\infty}\left(e^{j2\pi f_c t} + e^{-j2\pi f_c t}\right)e^{-j2\pi f t} + K_a m(t)\left[e^{j2\pi f_c t} + e^{-j2\pi f_c t}\right] e^{-j2\pi f t}\right\} dt$$

$$= \frac{A_c}{2}\left\{\int_{-\infty}^{\infty}\left[e^{-j2\pi(f-f_c)t} + e^{-j2\pi(f+f_c)t} + K_a m(t)\left(e^{-j2\pi(f-f_c)t} + e^{-j2\pi(f+f_c)t}\right)\right] dt\right.$$

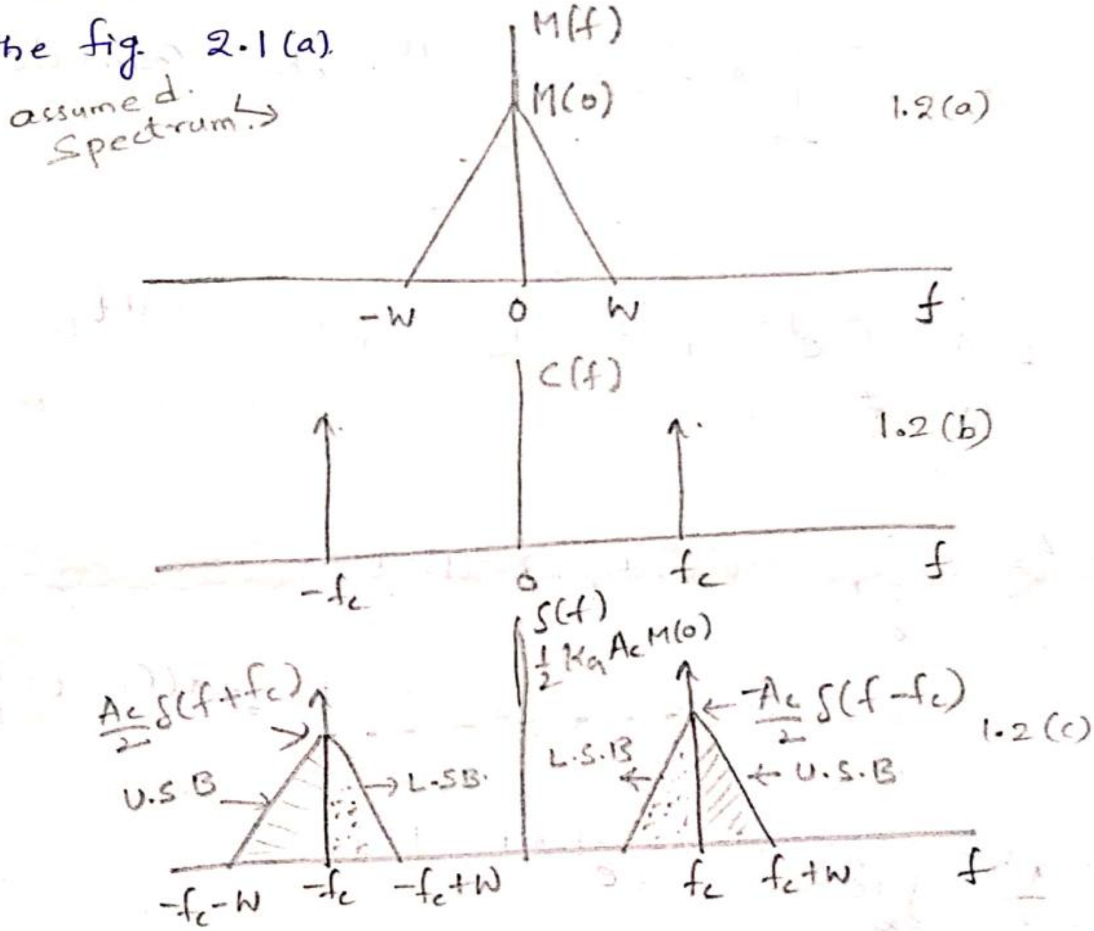
$$= \frac{A_c}{2}\left\{\int_{-\infty}^{\infty}\left(e^{-j2\pi(f-f_c)t} + e^{-j2\pi(f+f_c)t}\right) dt + K_a m(t)\int_{-\infty}^{\infty}\left(e^{-j2\pi(f-f_c)t} + e^{-j2\pi(f+f_c)t}\right) dt\right\}$$

$$= \frac{A_c}{2}\left\{\delta(f-f_c) + \delta(f+f_c) + K_a\left[M(f-f_c) + M(f+f_c)\right]\right\}$$

$$S(f) = \frac{A_c}{2}\left[\delta(f-f_c) + \delta(f+f_c)\right] + \frac{K_a A_c}{2}\left[M(f-f_c) + M(f+f_c)\right] \rightarrow (4)$$

- From equation (4) we observe that the spectrum of the AM wave consists of two impulse functions occuring at f_c & $-f_c$ and the original spectrum $M(f)$ shifted in the frequency domain by f_c & $-f_c$.

- Consider the baseband signal $m(t)$ is band-limited to the interval $-W \leq f \leq W$ as shown in the fig. 2.1 (a).



2.1(a) → spectrum of message signal.
2.1(b) → spectrum of Carrier signal
2.1(c) → Spectrum of AM wave.

- From the 2.1(c) spectrum of AM signal 2.1(c) consists of two delta functions weighted by the factor $A_c/2$ occuring at $\pm f_c$ and two versions of the baseband spectrum translated in frequency by $\pm f_c$ & scaled in amplitude by $K_a A_c/2$.

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- We have the following observations from the spectrum of AM wave.

i) For +ve frequencies, a portion of the spectrum of AM wave is lying above the carrier frequency & is referred to as the upper side band, whereas the symmetrical portion below f_c is referred to as Lower side band. Ily for -ve frequencies $a > -f_c$ L.S.B, $b < -f_c$ U.S.B.

ii) For +ve frequencies, the highest frequency component of the AM wave equals f_c+W & lowest frequency component equals f_c-W. The difference b/w these two frequencies is called transmission bandwidth B_T. For A.M wave $B_T = 2W$.

Eg: Single-tone modulation.

Consider a modulating wave m(t) that consists of a single tone or frequency component, i.e.,

$$m(t) = A_m \cos(2\pi f_m t) \rightarrow ①$$

$A_m \rightarrow$ amp. of modulating wave $f_m \rightarrow$ mod. wave freq

- Carrier wave as

$$c(t) = A_c \cos(2\pi f_c t) \rightarrow ②$$

- The corresponding AM wave is given by

$$s(t) = A_c\left[1+ \mu \cos(2\pi f_m t)\right] \cos(2\pi f_c t) \rightarrow ③$$

Where $\mu = k_a A_m \rightarrow ④$

$\mu \rightarrow$ modulation factor (dimensionless quantity)

- To avoid envelope distortion due to overmodulation, the modulation factor μ must be kept below unity.

- Let A_{max} & A_{min} denote the maximum & minimum values of the envelope of the modulated wave. Then from eq. ③ we have

$$\frac{A_{max}}{A_{min}} = \frac{A_c(1+\mu)}{A_c(1-\mu)}$$

$$\Rightarrow \qquad \mu = \frac{A_{max} - A_{min}}{A_{max} + A_{min}} \longrightarrow (5)$$

- Consider eq (3)

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

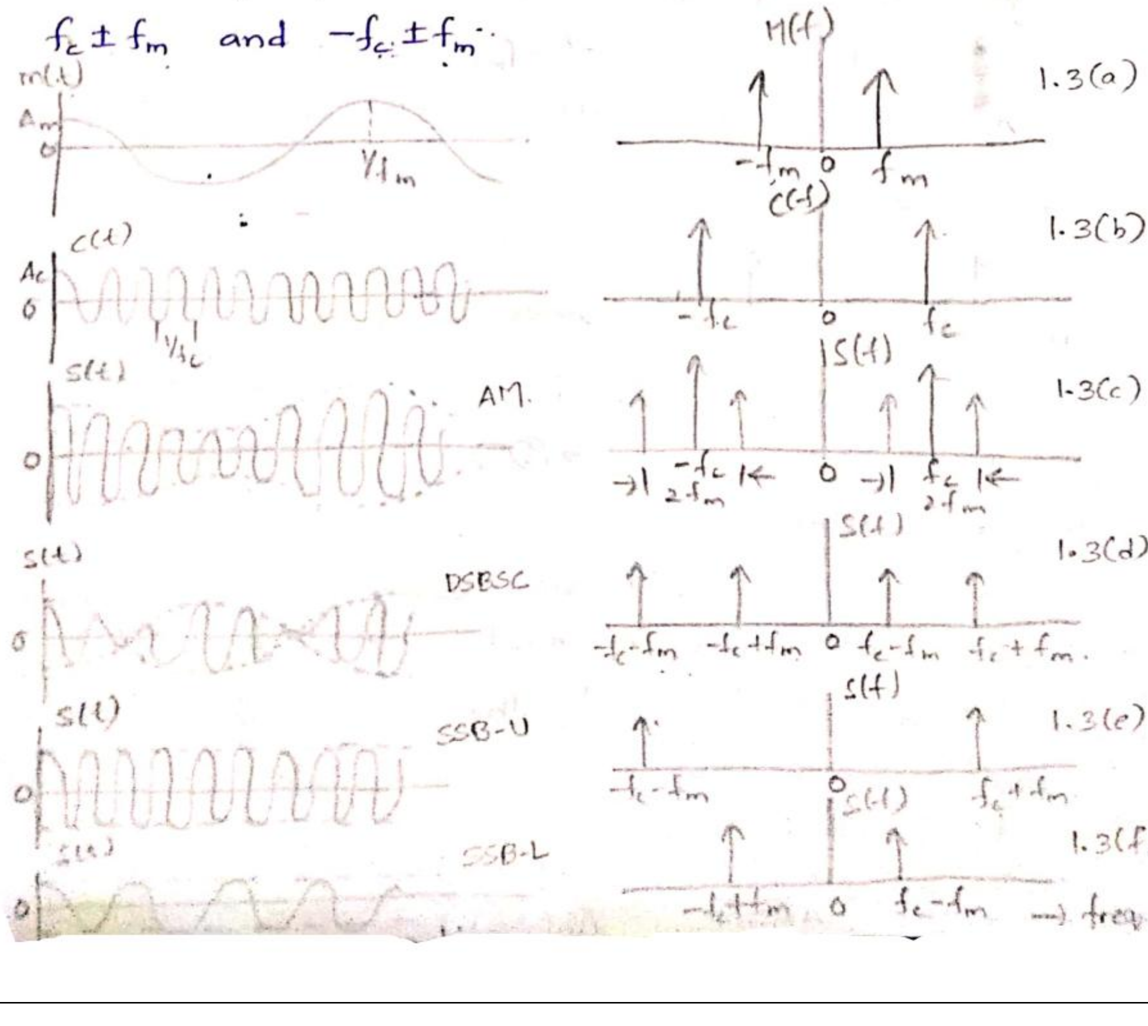
$$\because \cos A \cos B = \frac{1}{2}\left[\cos(A+B) + \cos(A-B)\right]$$

$$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + \frac{1}{2}\mu A_c \cos\left[2\pi(f_c+f_m)t\right] + \frac{1}{2}\mu A_c \cos\left[2\pi(f_c-f_m)t\right] \rightarrow (6)$$

Applying Fourier Transform

$$S(f) = \frac{1}{2}A_c\left[\delta(f-f_c) + \delta(f+f_c)\right] + \frac{1}{4}\mu A_c\left[\delta(f-f_c-f_m) + \delta(f+f_c+f_m)\right] + \frac{1}{4}\mu A_c\left[\delta(f-f_c+f_m) + \delta(f+f_c-f_m)\right] \longrightarrow (7)$$

- The spectrum of an AM wave for sinusoidal modulation, consists of delta functions at $\pm f_c$, $f_c \pm f_m$ and $-f_c \pm f_m$.



- **Power Calculations & Power Relations in AM Waves:**

Consider the AM wave

$$s(t) = A_c \cos(2\pi f_c t) + k_a A_c m(t) \cos(2\pi f_c t)$$

$$m(t) = A_m \cos(2\pi f_m t)$$

Now the resulting Spectrum contains a Carrier wave, U.S.B Spectrum, L.S.B spectrum.

In practice, the AM wave s(t) is a voltage or current wave. the average power delivered to a 1-ohm resistor by s(t) is comprised of three components

i) Carrier power ii) U.S. Freq. Power iii) L.S. Freq. power.

$$P = I^2 R \text{ or } V^2/R.$$

The carrier power is mean square value of $A_c \cos(2\pi f_c t)$.

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

$\because R=1$ $\quad P_{f_c} = \frac{A_c^2}{2}$

U.S. Freq. Power.

$$P_{f_c+f_m} = \left(\frac{A_c \mu}{2\sqrt{2} R}\right)^2 = \frac{A_c^2 \mu^2}{8R}.$$

L.S. Freq. Power.

$$P_{f_c-f_m} = \left(\frac{A_c \mu}{2\sqrt{2} R}\right)^2 = \frac{A_c^2 \mu^2}{8R}.$$

$$\text{Total Power} = \frac{A_c^2}{2R}\left[1 + \frac{\mu^2}{2}\right].$$

$$P_T = \underset{\text{Carrier}}{P_c} + \underset{\text{sideband}}{\frac{P_c \mu^2}{2}}. \qquad P_T = P_c\left[1 + \frac{\mu^2}{2}\right]$$

$$\text{Modulation Efficiency } (\eta) = \frac{\text{Total Sideband Power}}{\text{Total Power}}$$

$$\eta = \frac{\mu^2}{2+\mu^2}.$$

If $\mu = 1$, i.e., 100 percent modulation is used, the total power in the two side frequencies of the resulting AM wave is only one-third of the total power in the modulated wave.

Current calculations:

Let I_c be the unmodulated current & I_t the total or modulated, current of an AM transmitter, both being rms values. If 'R' is the resistance in which these currents flow, then

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

$$\frac{I_t}{I_c} = \sqrt{1 + \frac{\mu^2}{2}} \quad \text{(or)} \quad I_t = I_c \sqrt{1 + \frac{\mu^2}{2}}$$

Modulation by several sine waves:

We have, $P_t = P_c\left(1 + \frac{\mu^2}{2}\right) = P_c + P_{SB}$ $\left(P_{SB} = \frac{P_c \mu^2}{2}\right)$

If several sine waves simultaneously modulate the carrier, the carrier power will be unaffected, but the total sideband power ~~will~~ is the sum of the individual sideband powers.

$$P_{SB_T} = P_{SB_1} + P_{SB_2} + P_{SB_3} + \cdots$$

$$\frac{P_c \mu_t^2}{2} = \frac{P_c \mu_1^2}{2} + \frac{P_c \mu_2^2}{2} + \cdots$$

$$\mu_t^2 = \mu_1^2 + \mu_2^2 + \mu_3^2 + \cdots$$

PROBLEMS ON POWER RELATIONS

Q1) The amplitude of a carrier wave is 100V. Compute its RMS value when it has been amplitude modulated by a sinusoidal audio voltage to a depth of a) 20% b) 40%

Sol: Given, Carrier amplitude $A_c = 100V$

$$P_t = P_c\left[1+\frac{\mu^2}{2}\right] \quad (\text{or}) \quad \frac{V_{rms}^2}{2} = \frac{A_c^2}{2}\left[1+\frac{\mu^2}{2}\right]$$

$$\Rightarrow V_{rms} = A_c\sqrt{\left(1+\frac{\mu^2}{2}\right)}$$

a) $V_{rms} = 100\sqrt{1+\frac{(0.2)^2}{2}} = 101V$ b) $V_{rms} = 100\sqrt{1+\frac{(0.4)^2}{2}} = 103.9V$

Q2) Calculate the % power saving when the carrier & one sidebands are suppressed in an AM wave modulated to a depth of a) 100% b) 50%

Sol: a) For 100% modulation, mod. index $(\mu) = 1$.

$$\text{Total power } P_t = P_c\left[1+\frac{\mu^2}{2}\right] = P_c\left[1+\frac{1}{2}\right] = 1.5P_c.$$

$$\text{Power due to single sideband } P_{SB} = P_c\frac{\mu^2}{4} = \frac{P_c}{4} = 0.25P_c.$$

$$\text{Power saving due to single sideband} = \frac{P_t - P_{SB}}{P_t}\times 100\%.$$

$$= \frac{1.5-0.25}{1.5}\times 100$$

$$= 83.3\%$$

b) For 50% modulation, mod. index $\mu = 0.5$.

$$\text{Total Power } P_t = P_c\left[1+\frac{\mu^2}{2}\right] = P_c\left[1+\frac{(0.5)^2}{2}\right] = 1.125P_c.$$

$$P_{SB} = P_c\frac{\mu^2}{4} = P_c\left(\frac{0.5}{4}\right)^2 = 0.0625P_c$$

$$\text{Power saving due to single sideband} = \frac{P_t - P_{SB}}{P_t}\times 100.$$

$$= \frac{1.125-0.0625}{1.125}\times 100$$

$$= 94.4\%$$

Q3) An antenna current of an AM broadcast transmitter, modulated to a depth of 40% by an audio sine wave, is 11A. It increases to 12A as a result of simultaneous modulation by another audio sinewave. What is the modulation index due to this second wave?.

Sol: Current due to modulated carrier $I_t = 11A$.

We know, $\left(\frac{I_t}{I_c}\right)^2 = 1 + \frac{\mu^2}{2}$.

$$\Rightarrow I_c = \frac{I_t}{\sqrt{1+\frac{\mu^2}{2}}} = \frac{11}{\sqrt{1+\frac{(0.4)^2}{2}}} = 10.58A.$$

Total mod. index μ_t is.

$$\mu_t = \sqrt{2\left(\frac{I_t}{I_c}\right)^2 - 1} = \sqrt{2\left(\frac{12}{10.58}\right)^2 - 1}$$

$$= \sqrt{2(1.286-1)} = 0.757$$

$$\mu_t^2 = \mu_1^2 + \mu_2^2 \Rightarrow \mu_2^2 = \mu_t^2 - \mu_1^2$$

$$\Rightarrow \mu_2 = \sqrt{0.757^2 - 0.4^2} = 0.643$$

Q.4) Unmodulated RF carrier Power of 10KW sends a current of 10A RMS through an antenna. On amplitude modulation by another sinusoidal voltage, the antenna current increases to 11.6A. Calculate a) the mod. index. b) Carrier Power after modulation.

Sol: $I_c = I_{rms} = 10A$. , $I_t = 11.6A$ (after mod)

$$\mu_t = \sqrt{2\left(\frac{I_t}{I_c}\right)^2 - 1} = \sqrt{2\left(\frac{11.6}{10}\right)^2 - 1} = 0.8314.$$

Power of carrier after modulation.

$$P = P_c\left[1 + \frac{\mu_t^2}{2}\right] = 10\left[1 + \frac{0.8314^2}{2}\right]$$

$$= 13.456KW.$$

→ GENERATION OF AM WAVES

• The process of modulation translates the frequency spectrum. & the response of a modulator contains frequencies that are different from those present in the input signal. The device that generates an Amplitude Modulated wave is called Amplitude Modulator.

• The two methods used for generating AM waves are
i) Square Law Modulator ii) Switching Modulator

• These two methods require the use of a nonlinear element for their implementation and are well suited for low-power modulation purposes.

→ Switching Modulator:

• A typical switching modulator can be realised with the help of the arrangement shown in the figure.

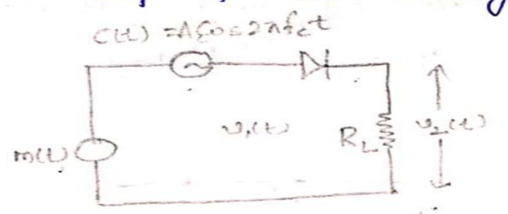


Fig: 1.5(a) Switching Modulator

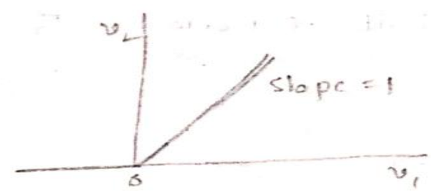


Fig: 1.5(b) Idealized i/p - o/p relation

• The carrier wave c(t) applied to the diode is large in amplitude & swings across the characteristic curve of the diode.

• The diode acts as an ideal switch, it presents zero impedance when F.B ($c(t) > 0$) & infinite impedance when R.B ($c(t) < 0$).

- We may approximate the transfer characteristic of the diode-load resistor combination by a piece-wise-linear characteristic, as shown in the graph.

• The input voltage applied to the diode is given by

$$v_1(t) = A_c \cos(2\pi f_c t) + m(t) \longrightarrow (1)$$

Where $|m(t)| << A_c$. & the resulting voltage $v_2(t)$ is

$$v_2(t) \simeq \begin{cases} v_1(t), & c(t) > 0 \\ 0 & c(t) < 0 \end{cases} \longrightarrow (2)$$

i.e, the load voltage $v_2(t)$ varies periodically b/w the values $v_1(t)$ & zero at a rate equal to the carrier freq. f_c.

• Mathematically the eq ② can be expressed as.

$$v_2(t) = \left[A_c \cos(2\pi f_c t) + m(t)\right] g_p(t) \longrightarrow ③$$

$g_p(t) \rightarrow$ a periodic pulse train of duty cycle equal to one-half of period $T_0 = 1/f_c$. as shown below.

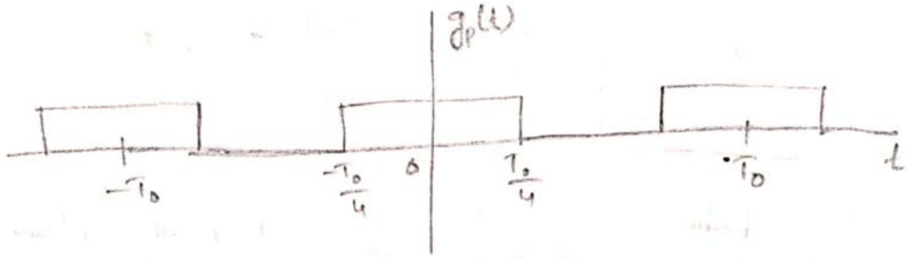


Fig.-1.5(c): Periodic pulse Train.

• Representing $g_p(t)$ by its Fourier Series, we have.

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos\left[2\pi f_c t(2n-1)\right] \longrightarrow ④$$

• Substituting eq ④ in eq ③

$$v_2(t) = \left[A_c \cos(2\pi f_c t) + m(t)\right]\left[\frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos\left[2\pi f_c t(2n-1)\right]\right].$$

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

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

$$= \frac{A_c}{2}\cos(2\pi f_c t) + \frac{2A_c}{\pi}\left[\cos^2(2\pi f_c t) - \frac{1}{3}\cos(3\pi f_c t)\cos(2\pi f_c t) + \frac{1}{5}\cos(10\pi f_c t)\cos(2\pi f_c t) \cdots\right]$$

$$+ \frac{m(t)}{2} + \frac{2}{\pi}m(t)\left[\cos(2\pi f_c t) - \frac{1}{3}\cos(3\pi f_c t) + \frac{1}{5}\cos(10\pi f_c t) + \cdots\right]$$

$$v_2(t) = \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c}m(t)\right]\cos(2\pi f_c t) + \frac{m(t)}{2} + \cdots$$

• The load voltage consists of two components.

1. The component $\frac{A_c}{2}\left[1 + \frac{4}{\pi A_c}m(t)\right]\cos(2\pi f_c t) \rightarrow$ desired AM wave

$k_a = 4/\pi A_c$.

2. An unwanted component, the spectrum of which contains delta fn's at $0, \pm 2f_c, \pm 4f_c$ & so on, & which occupies freq. intervals of width $2W$ centered at $0, \pm 3f_c, \pm 5f_c$ & so on ...

• The unwanted terms are removed by a B.P.F tuned to freq f_c & bandwidth $2W$, provided $f_c > 2W$.

↦ DEMODULATION OF AM WAVES:

• The process of demodulation provides a means to recover a signal that is proportional to the original modulating wave from the modulated wave.

• The two devices for demodulating the AM waves are
i) Square-law detector ii) Envelope detecter.

– Envelope Detector (Diode detector):

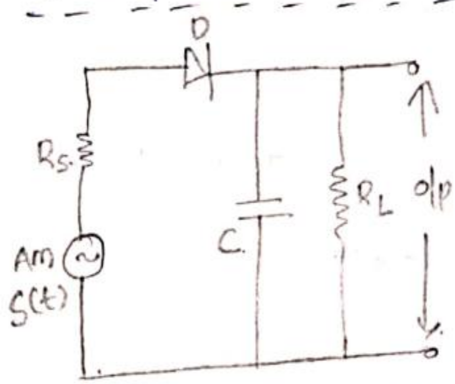
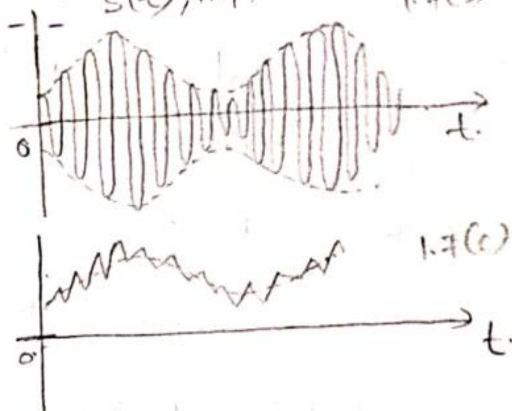


Fig: 1.7(a): Envelope detector



1.7(b): AM wave.
1.7(c): Envelope detector o/p

Assumptions:
$f_c \ggg W$, $\mu < 100\%$, diode is ideal

• An envelope detector produces an output signal that follows the envelope of the input signal waveform exactly.

Operation:

- During the positive half-cycle of the input signal, the diode is forward biased & the capacitor 'C' charges up rapidly to the peak value of the input signal. When the input signal falls below this value, the diode becomes reverse biased & the capacitor 'C' discharges slowly through the load resistor R_L.
- The discharging continues until the next positive half cycle. when the input signal becomes greater than the voltage across the capacitor, the diode conducts again & the process is repeated.
- $R_s \rightarrow$ internal impedance of the voltage source from where the AM wave is applied.
- The charging time constant R_sC must be short compared with the carrier period $1/f_c$ i.e., $R_sC << 1/f_c$, so that the capacitor charges rapidly.
- The discharging time constant R_LC must be long to ensure that the capacitor discharges slowly through the load R_L.

$$1/f_c << R_LC << 1/W \qquad W \rightarrow \text{message B.W.}$$

- The detector output (or) capacitor voltage is nearly the same as the envelope of the AM wave.

DOUBLE SIDE BAND SUPPRESSED CARRIER MODULATION (DSB-SC):

- In full AM (DSB-AM), the carrier wave $c(t)$ is completely independent of the message signal $m(t)$, i.e, the transmission of carrier wave represents a 'waste of power'.
- This is the disadvantage of AM. i.e, only a fraction of the total power is affected by $m(t)$.

$$\eta = \frac{\mu^2}{1+\mu^2}$$

$$\eta = 33.3\% \quad \text{when} \quad \mu = 1$$

$$\eta < 33.3\% \quad \text{when} \quad \mu < 1.$$

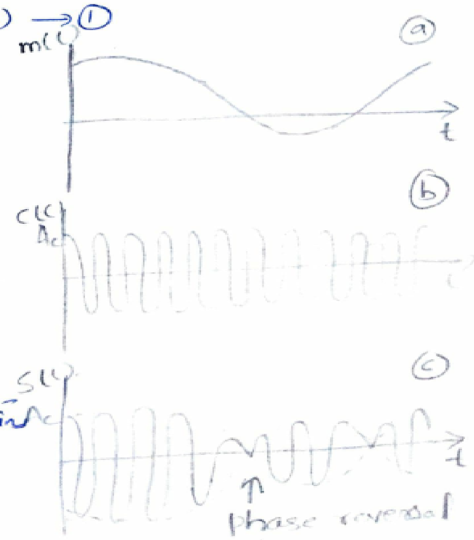
∴ To overcome this shortcoming we may suppress the carrier component from the modulated wave, resulting in double-side band suppressed carrier AModulation.

TIME DOMAIN AND FREQUENCY DOMAIN DESCRIPTION OF DSB-SC:

Time Domain DESCRIPTION:

- A DSB-SC signal is obtained by multiplying the message signal $m(t)$ with the carrier signal $c(t)$

$$s(t) = c(t).m(t)$$
$$= A_c \cos(2\pi f_c t) m(t) \rightarrow ①$$

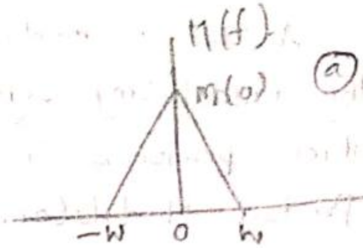


- This modulated wave undergoes a phase reversal whenever the modulating signal $m(t)$ crosses zero as shown in fig (c)
- Unlike AM, the envelope of a DSB-SC wave is different from the modulating signal.

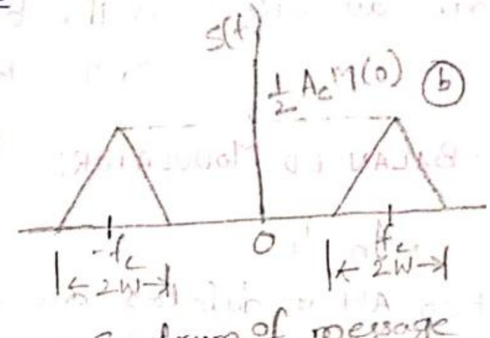
- Frequency Domain Description of DSB-SC:

- Consider the equation of DSB-SC signal.

$$s(t) = c(t)\, m(t)$$
$$= A_c \cos(2\pi f_c t)\, m(t) \rightarrow ①$$



Applying Fourier Transform to the above eq, we have



$$S(f) = \frac{1}{2} A_c \left[M(f-f_c) + M(f+f_c) \right] \rightarrow ②$$

- When the message signal $m(t)$ is band limited to the interval $-W \leq f \leq W$, the DSB-SC modulation process simply translates the spectrum of the message signal by $\pm f_c$.

a → Spectrum of message
b → Spectrum of DSB-SC wave

- The transmission bandwidth required by DSB-SC modulation is same as AM i.e, $B_T = 2W$.

- Power Content of the DSB-SC wave: (Sinusoidal Modulation)

$s(t)$ = Consider DSB-SC modulation process for a single tone

$$\therefore s(t) = c(t)\, m(t) = A_c \cos(2\pi f_c t)\, A_m \cos(2\pi f_m t) \rightarrow ①$$

$$\therefore s(t) = \frac{A_m A_c}{2} \cos 2\pi(f_c + f_m)t + \frac{A_m A_c}{2} \cos(2\pi(f_c - f_m)t)$$

Avg. power delivered to a 1Ω resistor is

$$\underset{U.S.B}{P_U} = \left(\frac{A_m A_c/2}{\sqrt{2}}\right)^2 \qquad \underset{L.S.B}{P_L} = \left(\frac{A_m A_c/2}{\sqrt{2}}\right)^2$$

$$P_U = \frac{A_m^2 A_c^2}{8} \qquad P_L = \frac{A_m^2 A_c^2}{8}$$

Total Power $P_T = P_U + P_L = \dfrac{A_m^2 A_c^2}{4}$

$$\therefore \frac{P_U}{P_T} = \frac{P_L}{P_T} = \frac{A_m^2 A_c^2/8}{A_m^2 A_c^2/4} \times 100 = 50\%.$$

- GENERATION OF DSB-SC WAVES:

. A DSB-SC modulated wave consists of the product of the modulating signal & the carrier signal. A device which performs the above requirement is called a 'Product Modulator'. Two forms of a product modulator are available. i) The Balanced Modulator
ii) The Ring Modulator

- BALANCED MODULATOR:

. In this method two AM modulators are arranged in a balanced configuration so as to suppress the carrier wave as shown in fig.

. The two AM modulators are identical, except for the sign reversal of the modulating wave applied to the input of one of the modulators.

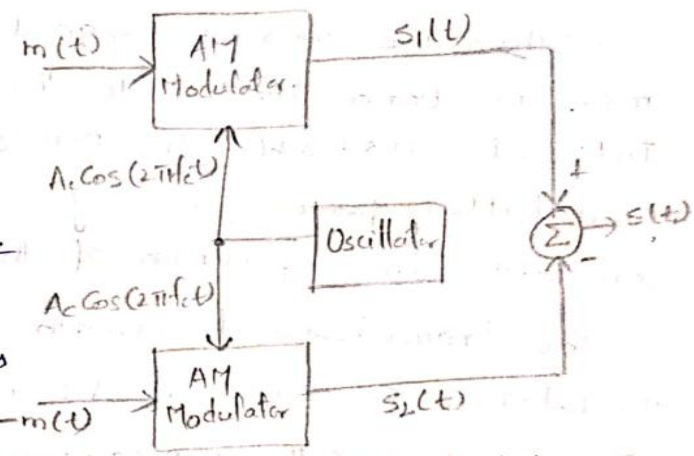


Fig: Balanced Modulator

. The output of two AM modulators are

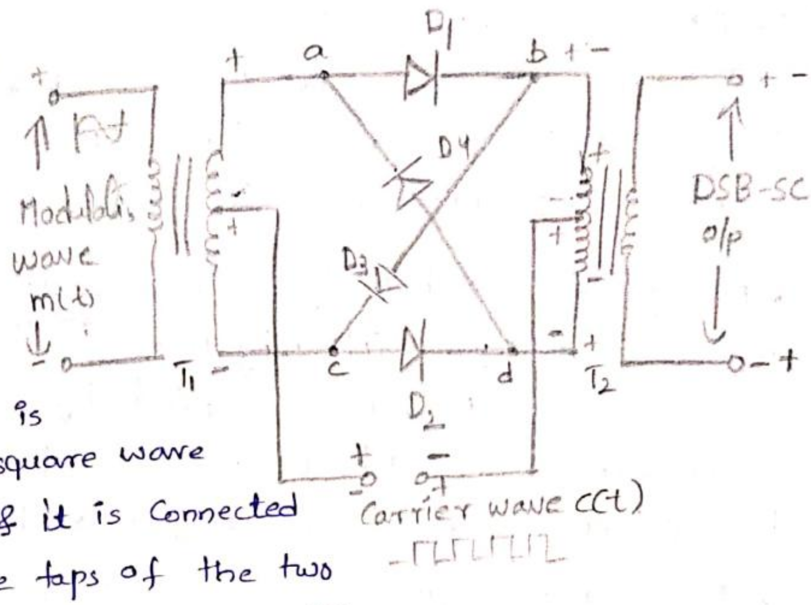
$$s_1(t) = A_c[1+K_a m(t)]\cos(2\pi f_c t) \rightarrow (1)$$
$$s_2(t) = A_c[1-K_a m(t)]\cos(2\pi f_c t) \rightarrow (2)$$

Substracting $s_2(t)$ from $s_1(t)$.

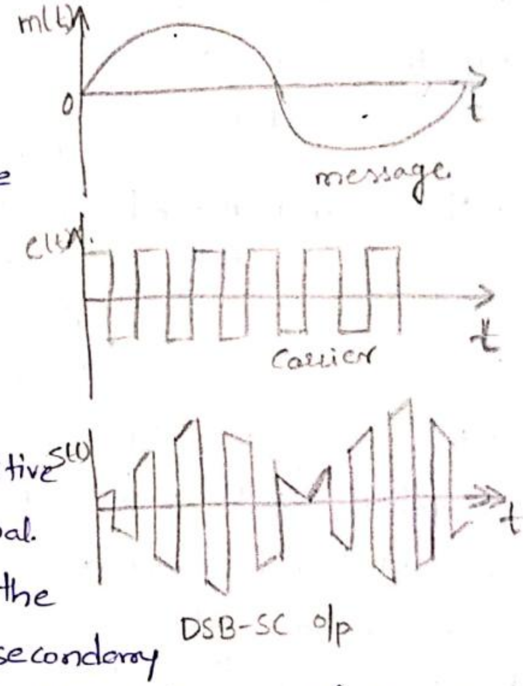
$$s(t) = s_1(t) - s_2(t)$$
$$= 2K_a A_c \cos(2\pi f_c t) m(t) \rightarrow (3)$$

. Except for a scaling factor $2k_a$, the balanced modulator output is equal to the product of the modulating wave and the carrier.

RING MODULATOR:



- A ring modulator consists of four diodes, an audio frequency transformer T_1 & an RF transformer T_2.
- The carrier signal is assumed to be a square wave with frequency f_c & it is connected between the centre taps of the two transformer.



Principle of Operation:

- During positive half cycle of the modulating signal.

i) The modulating signal m(t) is applied through the input of T_1 transformer. There are many cycles of the carrier signal, in the positive half cycle of the modulating signal.

ii) During positive half cycle of the carrier, D_1 & D_2 are 'ON' & the secondary of T_1 is applied as it is across the primary of T_2. Therefore, during positive half cycle of carrier, the output of T_2 is positive.

iii) In the negative half cycle of the carrier, D_3 & D_4 are turned 'ON' & the secondary of T_1 is applied in a reversed manner across the primary of T_2. The primary voltage of T_2 is negative & hence the output voltage also becomes negative.

• During negative half cycle of the modulating signal.
1) when modulating signal reverses the polarities, the operation of the circuit is same but the diode pair $D_3 D_4$ will produce a positive output voltage whereas $D_1 D_2$ will produce negative output voltage.

• During positive half cycle of the carrier, the message signal m(t) is multiplied by +1 & in the negative half cycle of the carrier m(t) is multiplied by -1. Thus, the ring modulator is an ideal form of product modulator & hence it produces the required DSB-SC output.

• the square wave carrier signal can be represented by the Fourier Series as.

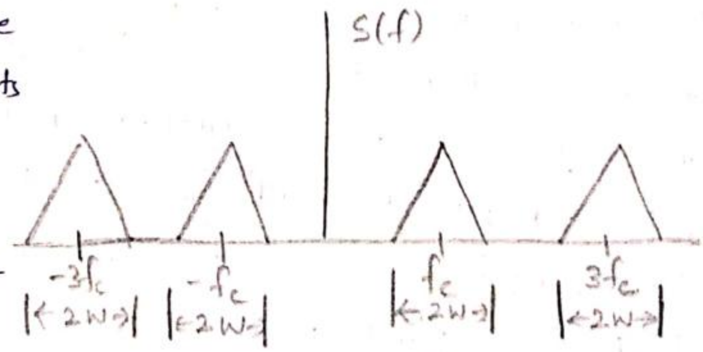
$$c(t) = \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{(2n-1)}\cos\left[2\pi f_c(2n-1)t\right]$$

• The ring modulator output is given by

$$s(t) = c(t)\, m(t).$$

$$= \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos\left[2\pi f_c(2n-1)t\right] m(t).$$

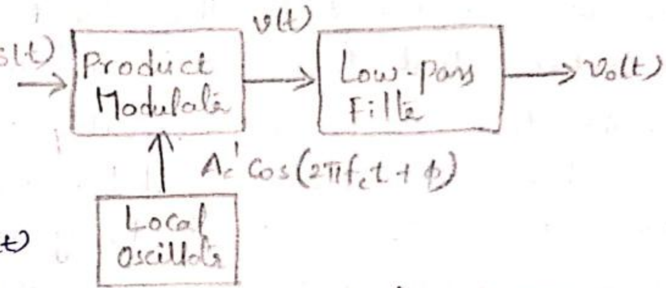
• The spectrum of the modulator output consists of sidebands around each of the odd harmonics of the square wave carrier.



• The desired sideband around carrier frequency f_c can be selected by using a band pass filter having. centre frequency f_c & bandwidth 2W..

COHERENT DETECTION OF DSB-SC WAVES:

- The baseband signal $m(t)$ can be uniquely recovered from a DSB-SC wave $s(t)$ by first multiplying $s(t)$ with a locally generated sine-wave & the low-pass filtering the product as shown



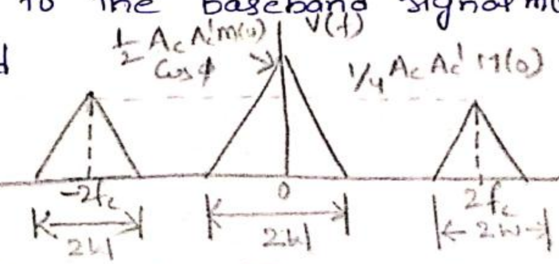
- The local oscillator signal should be exactly synchronized in both frequency & phase with the carrier wave $c(t)$ used in the product modulator to generate $s(t)$. Coherent detection of DSB-SC waves

This method of demodulation is known as Coherent detection (or) synchronous detection.

- The local oscillator signal is given by $A_c' \cos(2\pi f_c t + \phi)$. This signal has same frequency but arbitrary phase difference ϕ.
- The output of product modulator is given by

$$v(t) = A_c' \cos(2\pi f_c t + \phi)\, s(t)$$
$$= A_c A_c' \cos(2\pi f_c t) \cos(2\pi f_c t + \phi)\, m(t)$$
$$= \tfrac{1}{2} A_c A_c' \cos(4\pi f_c t + \phi)\, m(t) + \tfrac{1}{2} A_c A_c' \cos\phi\, m(t) \rightarrow (1)$$

- The first term in the above equation represents a DSB-SC wave with a carrier frequency $2f_c$, the second term is proportional to the baseband signal $m(t)$.
- The first term is removed by the low-pass filter with cut-off frequency greater than W but less than $2f_c - W$.



Spectrum of V(f)

- The filter output is given by

$$v_0(t) = \tfrac{1}{2} A_c A_c' \cos\phi\, m(t) \rightarrow (2)$$

- The demodulated signal $v_o(t)$ is proportional to $m(t)$ when the phase error ϕ is a constant. The amplitude of demodulated signal is max, when $\phi = 0$ & min when $\phi = \pm\pi/2$.
- The zero demodulated signal which occurs when $\phi = \pm\pi/2$ represents quadrature null effect of the coherent detector. The phase error ϕ in the local oscillator causes the detector output to be attenuated by a factor of $\cos\phi$.
- At the receiver, circuitry should be provided to maintain the local oscillator in perfect synchronism with the carrier used to generate the DSB-SC wave.
- The complexity in the receiver is due to suppressing the carrier wave to save transmitter power.

COASTAS RECEIVER:

• Coastas Receiver is a practical synchronous receiving system for demodulating DSB-SC waves.

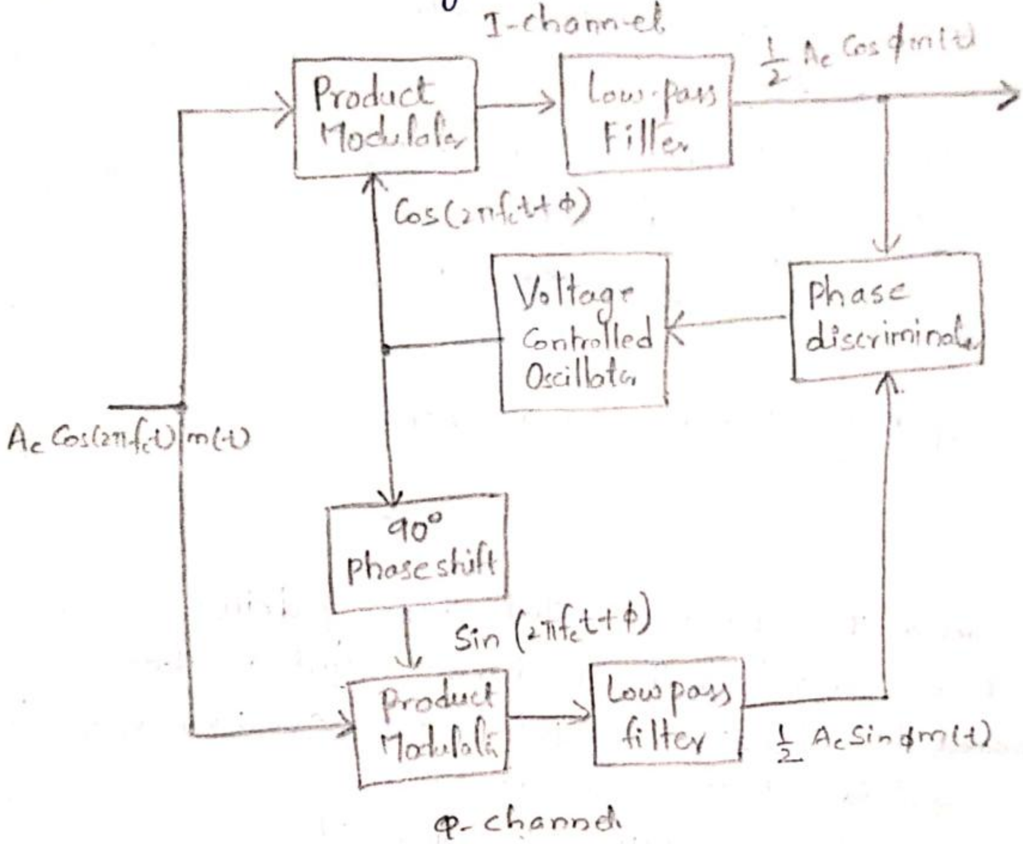


Fig: Costas Receiver

• This system consists of two coherent detectors with same input signal (DSB-SC wave), but with individual local oscillator signals that are in phase quadrature to each other. The frequency of the local oscillator is adjusted to be the same as the carrier frequency f_c present at the modulator. (transmitter).

• The detector in the upper path is referred to as the in-phase coherent detector (or) I-channel, & that in the lower path is referred to as the quadrature-phase coherent detector (or) Q-channel.

Operation:

. Assuming that the local carrier signal is synchronized with the transmitting carrier signal & $\phi = 0$, the output

eering

of I-channel is the desired modulating signal $m(t)$ ($\because \cos\phi = 1$) and the output of Q-channel is zero ($\because \sin\phi = 0$) due to quadrature null effect.

I-channel $v_1(t) = A_c \cos(2\pi f_c t) m(t) \cos(2\pi f_c t)$

$$= \frac{A_c m(t)}{2}\left[\cos(4\pi f_c t) + \cos(0)\right]$$

- The output of Low pass filter is $\frac{A_c m(t)}{2}$

Q-channel $v_2(t) = A_c \cos(2\pi f_c t) m(t) \sin(2\pi f_c t)$

$$= \frac{A_c m(t)}{2}\left[\sin(4\pi f_c t) - 0\right]$$

$$= \frac{A_c m(t)}{2} \sin(4\pi f_c t)$$

- The output of lowpass filter is zero

- Assuming that the local oscillator frequency drifts slightly i.e., ϕ (non-zero value), the I-channel output is almost unchanged, but Q-channel output is not zero, i.e., a signal proportional to $\sin\phi$ appears at the output

$$v_1(t) = A_c \cos(2\pi f_c t) m(t) \cos(2\pi f_c t + \phi)$$

$$= \frac{A_c m(t)}{2}\left[\cos(4\pi f_c t + \phi) + \cos\phi\right]$$

$$= \frac{A_c m(t)}{2} \cos(4\pi f_c t + \phi) + \frac{A_c m(t)}{2} \cos\phi$$

- The output of Low pass filter is $\frac{A_c m(t)}{2} \cos\phi \simeq \frac{A_c m(t)}{2}$

$$v_2(t) = A_c \cos(2\pi f_c t) m(t) \sin(2\pi f_c t + \phi)$$

$$= \frac{A_c m(t)}{2}\left[\sin(4\pi f_c t + \phi) - \sin\phi\right] = \overset{LPF}{\frac{A_c m(t)}{2} \sin\phi}$$

$\because \sin\phi \simeq \phi$, Output of LPF is $\frac{A_c m(t)}{2}\phi$.

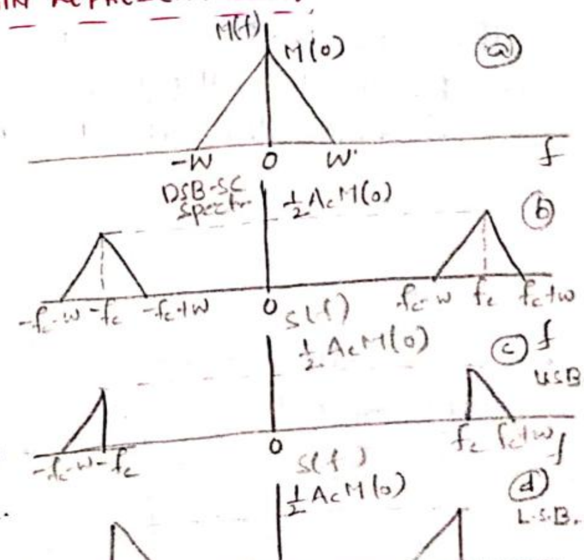
- The phase discriminator provides a dc control signal which may be used to correct local oscillator phase error. The local oscillator is a voltage controlled oscillator (VCO). Its frequency may be adjusted by an error control d.c. signal.

→ SINGLE SIDEBAND MODULATION (SSB-SC)

- Amplitude modulation and DSB-SC modulation are wasteful of bandwidth because both modulations require a transmission bandwidth equal to twice the message bandwidth.
- In either case, one-half of the transmission bandwidth is occupied by the upper sideband of the modulated wave & the other half by the lower sideband.
- The upper & lower sidebands are uniquely related to each other by the virtue of their symmetry about the carrier frequency f_c.
- If the amplitude and phase spectra of either sideband is given, the other sideband can be uniquely determined. When only one sideband is transmitted, the modulation system is referred to as a Single Sideband (SSB) system.

→ TIME DOMAIN AND FREQUENCY DOMAIN REPRESENTATION:

- The frequency domain description of an SSB wave depends on which sideband is transmitted.
- Consider a baseband signal m(t) with a spectrum M(f) band limited between $-W \le f \le W$ as shown in (a)



(b) → Spectrum of DSB-SC wave.
(c) → When U.S.B is transmitted
(d) → When L.S.B is transmitted.

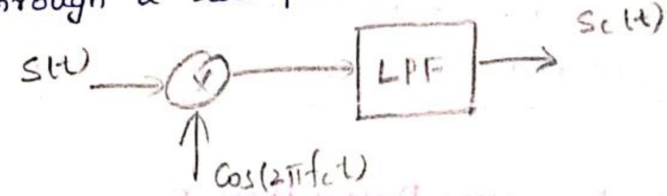
- An SSB modulation system translates the spectrum of the modulating wave, either with or without inversion, to a new location in the frequency domain & the transmission bandwidth requirement of the system is one-half that of an AM (or) DSB-SC system.

- According to the Canonical form representation of Band pass systems, the time-domain description of an SSB wave $s(t)$ is given by

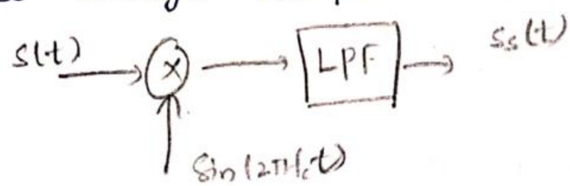
$$s(t) = s_c(t)\cos(2\pi f_c t) - s_s(t)\sin(2\pi f_c t) \rightarrow (1)$$

$s_c(t) \rightarrow$ in-phase Component of the SSB wave
$s_s(t) \rightarrow$ Quadrature Component "

- The in-phase component $s_c(t)$ can be derived from $s(t)$ by first multiplying $s(t)$ by $\cos(2\pi f_c t)$ & then passing the product through a low-pass filter.



- The quadrature component $s_s(t)$, can be derived from $s(t)$ by first multiplying $s(t)$ by $\sin(2\pi f_c t)$ & then passing the product through low-pass identical filter.



- The relation between the Fourier Transforms of $s_c(t)$ & $s_s(t)$ and the SSB wave $s(t)$ is given by

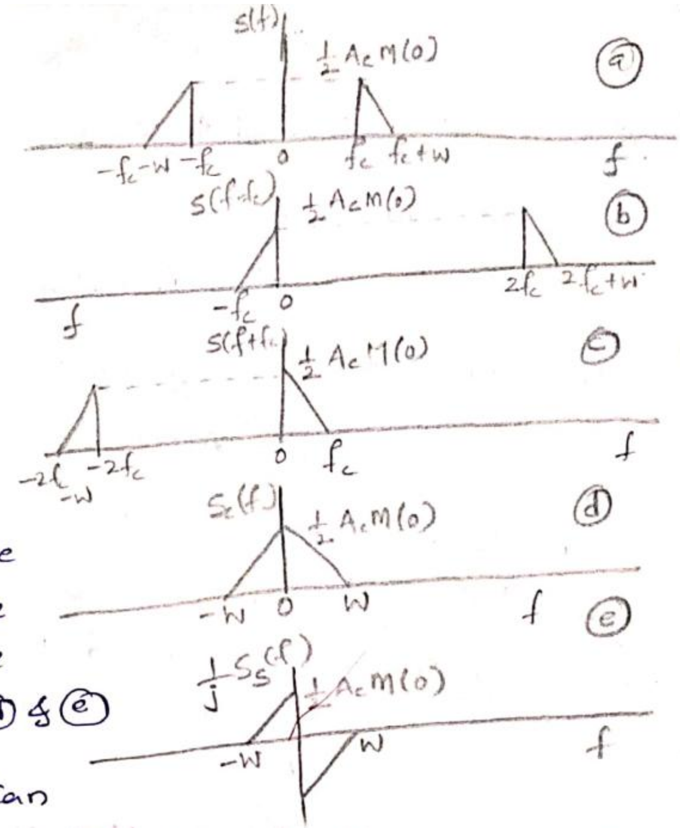
$$S_c(f) = \begin{cases} S(f-f_c) + S(f+f_c), & -W \leq f \leq W \\ 0 & \text{elsewhere} \end{cases} \rightarrow (2)$$

$$S_s(f) = \begin{cases} j[S(f-f_c) - S(f+f_c)], & -W \leq f \leq W \\ 0 & \text{elswhere} \end{cases} \rightarrow (3)$$

$-W \leq f \leq W \rightarrow$ freq. band occupied by message signal.

• Consider an SSB wave that is obtained by transmitting only the upper sideband. fig ⓐ ⓑ & ⓒ figs show the frequency spectra of $S(f-f_c)$, $S(f+f_c)$

• From eq's ② & ③ the spectra of in-phase component & quadrature phase component can be drawn as shown in ⓓ & ⓔ



• From figure ⓓ we can write,

$$S_c(f) = \frac{1}{2} A_c M(f) \rightarrow ④$$

$M(f) \rightarrow$ Fourier Transform of message signal $m(t)$.

• Therefore, the in-phase component $s_c(t)$ is defined by

$$s_c(t) = \frac{1}{2} A_c m(t) \rightarrow ⑤$$

• From figure ⓔ we have

$$S_s(f) = \begin{cases} -\frac{j}{2} A_c M(f) & ; f > 0 \\ 0 & , f = 0 \\ \frac{j}{2} A_c M(f) & , f < 0 \end{cases}$$

$$S_s(f) = -\frac{j}{2} A_c \operatorname{sgn}(f) M(f) \rightarrow ⑥$$

sgn(f) is the signum function equal to +1 for positive frequencies, zero for $f=0$, -1 for negative frequencies.

From eq. ⑥ we have

$$-j \operatorname{sgn}(f) M(f) = \hat{M}(f) \rightarrow ⑦$$

$\hat{M}(f)$ is the Fourier Transform of $\hat{m}(t)$, $g(t) - G(f)$

$- \hat{g}(t)$

$\hat{m}(t)$ is the Hilbert transform of $m(t)$.

$$\hat{g}(t) = \frac{1}{\pi}\int_{-\infty}^{\infty}\frac{g(\tau)}{t-\tau}d\tau$$

HT pairs.

$$g(t) = -\frac{1}{\pi}\int_{-\infty}^{\infty}\frac{\hat{g}(\tau)}{t-\tau}d\tau$$

Therefore we have.

$$S_s(f) = \frac{1}{2}A_c\hat{M}(f) \rightarrow (8)$$

- The quadrature component $s_s(t)$ is defined by

$$s_s(t) = \frac{1}{2}A_c\hat{m}(t) \rightarrow (9)$$

- Substituting eq (5) & (9) in eq (1) we have the canonical representation of an SSB wave $s(t)$ obtained by transmitting only the upper sideband is.

$$s(t) = \frac{1}{2}A_c m(t)\cos(2\pi f_c t) - \frac{1}{2}A_c\hat{m}(t)\sin(2\pi f_c t) \rightarrow (10)$$

→ GENERATION OF AM-SSB MODULATED WAVES:

• The two methods used for generating the SSB waves are i) frequency discrimination method
ii) phase discrimination method

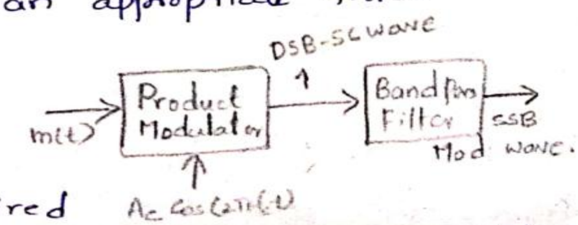
• These two methods are based on the frequency domain and time-domain descriptions of the SSB wave.

FREQUENCY DISCRIMINATION METHOD:

• This method is used to generate an SSB wave when the baseband is restricted & appropriately related to the carrier frequency.

• Under these conditions the desired sideband will appear in a non overlapping interval in the spectrum so that it can be selected by an appropriate filter.

• This method of SSB wave generation consists of a Product modulator (i.e. ring mod) & a filter that allows the desired side band of the DSB-SC wave & reject the other sideband



• In designing the band pass filter in the SSB modulation scheme, two requirements should be satisfied.

i) the passband of the filter should occupy the same frequency range as the spectrum of the desired SSB wave.

ii) the width of the transition band of the filter, separating the passband from the stopband should be twice the lowest frequency component of the modulating wave.

→ PHASE DISCRIMINATION METHOD:

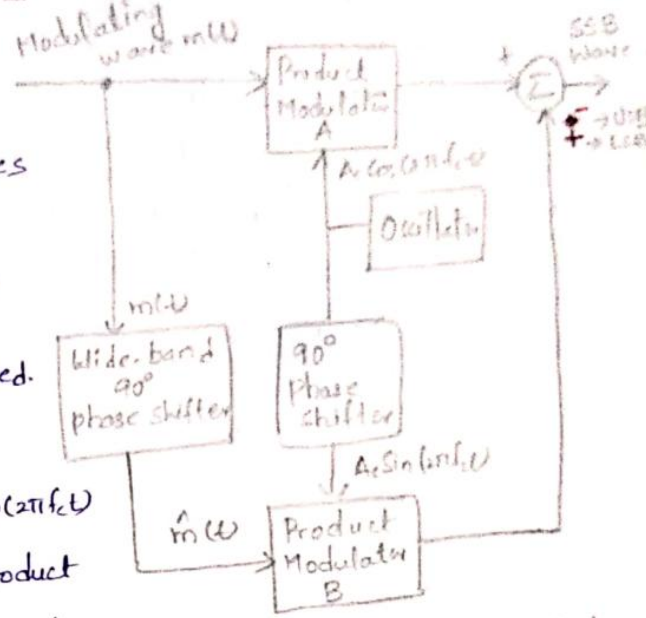


Fig: phase discrimination method for SSB wave generation.

- This method is based on the Canonical representation of SSB waves in the time domain.
- Consider the SSB wave equation for the case of upper sideband is transmitted.

$$s(t) = \frac{1}{2} A_c m(t) \cos(2\pi f_c t) - \frac{1}{2} A_c \hat{m}(t) \sin(2\pi f_c t)$$

- This system uses two product modulators A & B with carrier wave in phase quadrature to each other.
- The incoming baseband signal m(t) is applied to the product modulator A, producing a DSB-SC wave that contains reference phase sidebands symmetrically spaced about carrier frequency f_c.
- The Hilbert transform $\hat{m}(t)$ of m(t) is applied to product modulator B, producing a DSB-SC wave that contains sideband having identical amplitude spectra as that of modulator 'A'.
- The relative phase is such that the vector addition (or) subtraction of the two modulators outputs results in cancellation of one set of sidebands.
- The use of product modulator with plus sign outputs SSB wave with lower sideband, where as with minus sign it outputs SSB wave with upper sideband.
- The above modulator is also known as "Hartley modulator". The Hilbert transform of m(t) consists of a network that shifts the phase angle of every frequency component of m(t) by 90° but leaves the amplitude unchanged.

In practice, it is difficult to design a n/w which provides 90° phase shift over a wide freq range of the modulating wave m(t).

· Therefore, a phase shifting network is included in each modulation path, so that the required constant phase difference is maintained.

· The phase-shifts α & β are related by.

$$\beta - \alpha = \frac{\pi}{2}.$$

· This modulator does not require any sharp cut off filters.

· The degree to which the unwanted sideband is suppressed depends on.

i) balancing accuracy of the product modulator.

ii) Control accuracy of the quadrature phase relationship of the two carriers

iii) errors in the approximation of the constant 90° phase difference between m(t) & $\hat{m}(t)$.

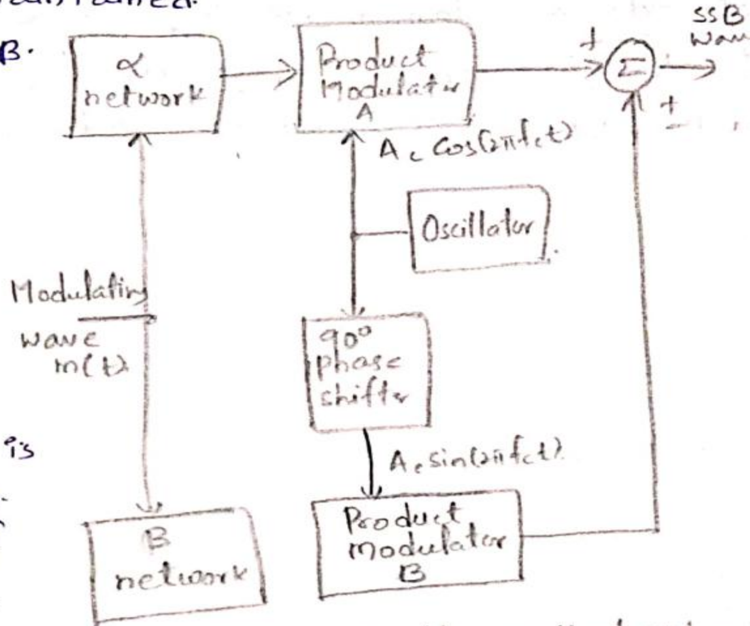
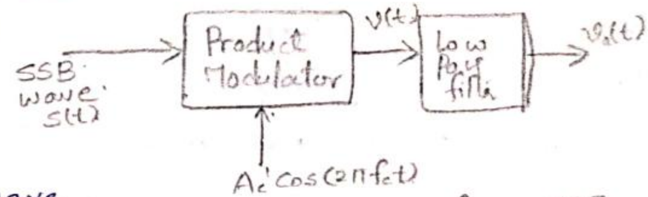


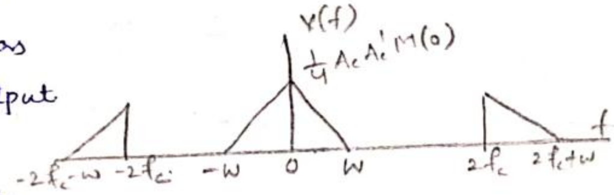
Fig: phase discrimination method using α & β phase shifting networks

DEMODULATION OF SSB WAVES:

- This method involves applying the SSB wave s(t), together with a locally generated sine wave $A_c' \cos(2\pi f_c t)$, to a product modulator & then low pass filtering the modulator output as shown in fig



Coherent detection of an SSB



Spectrum of P.M o/p v(t)

- Consider the time domain representation of SSB wave s(t) given by.

$$s(t) = \frac{1}{2}A_c m(t) \cos(2\pi f_c t) - \frac{1}{2}A_c \hat{m}(t) \sin(2\pi f_c t) \rightarrow (1)$$

USB transmitted

i) Consider the locally generated Carrier signal as $A_c' \cos(2\pi f_c t)$

- The output of product Modulator is given by

$$v(t) = \frac{1}{2}A_c A_c' \left[m(t) \cos(2\pi f_c t) \cos(2\pi f_c t) - \hat{m}(t) \sin(2\pi f_c t) \cos(2\pi f_c t) \right]$$

$$= \frac{1}{2}A_c A_c' \left[\frac{m(t)}{2}\{ \cos(4\pi f_c t) + \cos(0)\} - \frac{\hat{m}(t)}{2}\{ \sin(4\pi f_c t) + 0\} \right]$$

$$= \frac{1}{4}A_c A_c' m(t) + \frac{1}{4}A_c A_c' m(t) \cos(4\pi f_c t) - \frac{1}{4}A_c A_c' \hat{m}(t) \sin(4\pi f_c t)$$

$$= \frac{1}{4}A_c A_c' m(t) + \frac{1}{4}A_c A_c' \left[m(t) \cos(4\pi f_c t) - \hat{m}(t) \sin(4\pi f_c t) \right]$$

- The first term in the above equation is the desired demodulated signal, the second term represents an SSB wave corresponding to a carrier frequency $2f_c$.
- The high frequency components are removed by the low pass filter.

- Assuming that the local oscillator signal in the receiver, has a frequency error Δf, then the output of the product modulator is given by

$$v(t) = \frac{1}{2} A_c A_c' \Big[m(t) \cos(2\pi f_c t) \cos[2\pi(f_c+\Delta f)t] - \hat{m}(t) \sin(2\pi f_c t) \cos[2\pi(f_c+\Delta f)t] \Big]$$

$$= \frac{1}{4} A_c A_c' \Big[m(t) \{ \cos(4\pi f_c t + 2\pi \Delta f t) + \cos(2\pi \Delta f t) \} - \hat{m}(t) \{ \sin(4\pi f_c t + 2\pi \Delta f t) - \sin(2\pi \Delta f t) \} \Big]$$

$$= \frac{1}{4} A_c A_c' \Big[m(t) \cos(2\pi \Delta f t) \Big] + \frac{1}{4} A_c A_c' \Big[\hat{m}(t) \sin(2\pi \Delta f t) \Big] + \frac{1}{4} A_c A_c' m(t) \cos 2\pi(2f_c+\Delta f)t - \frac{1}{4} A_c A_c' \hat{m}(t) \sin 2\pi(2f_c+\Delta f)t$$

$$= \frac{1}{4} A_c A_c' \Big[m(t) \cos(2\pi \Delta f t) + \hat{m}(t) \sin(2\pi \Delta f t) \Big] + \frac{1}{4} A_c A_c' \Big[m(t) \cos\{2\pi(2f_c+\Delta f)t\} - \hat{m}(t) \sin\{2\pi(2f_c+\Delta f)t\} \Big]$$

- The first term in the above eq is a low frequency component which is allowed to pass through LPF & the other component is rejected by LPF.
- Assuming that the local oscillator signal in the receiver, has a phase error ϕ, then the output of the product modulator is given by

$$v(t) = \frac{1}{2} A_c A_c' \Big[m(t) \cos(2\pi f_c t) \cos(2\pi f_c t + \phi) - \hat{m}(t) \sin(2\pi f_c t) \cos(2\pi f_c t + \phi) \Big]$$

$$= \frac{1}{4} A_c A_c' \Big[m(t) \{ \cos(4\pi f_c t + \phi) + \cos\phi \} - \hat{m}(t) \{ \sin(4\pi f_c t + \phi) - \sin\phi \} \Big]$$

$$= \frac{1}{4} A_c A_c' \Big[m(t) \cos\phi + \hat{m}(t) \sin\phi \Big] + \frac{1}{4} A_c A_c' \Big[m(t) \cos(4\pi f_c t + \phi) - \hat{m}(t) \sin(4\pi f_c t + \phi) \Big]$$

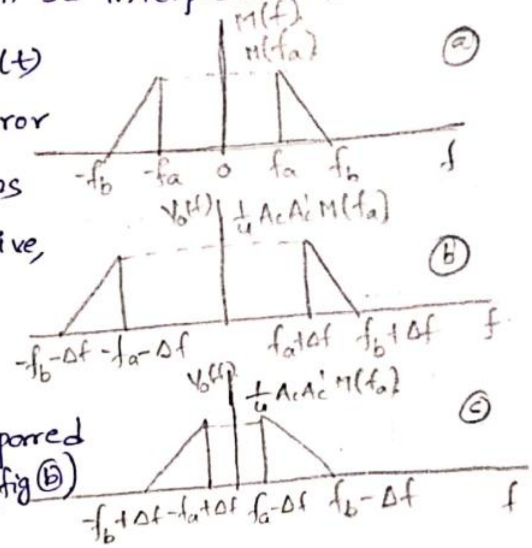
- The o/p of LPF is $\frac{1}{4} A_c A_c' \Big[m(t) \cos\phi + \hat{m}(t) \sin\phi \Big]$.

frequency error:

$$v_0(t) = \frac{1}{4} A_c A_c' \left[m(t) \cos(2\pi \Delta f t) + \hat{m}(t) \sin(2\pi \Delta f t) \right]$$

- The demodulated signal represents an SSB wave corresponding to a carrier frequency Δf. The effect of frequency error Δf in the local oscillator can be interpreted as.

i) If the incoming SSB wave $s(t)$ contains the LSB & the freq. error Δf is positive, (or) if $s(t)$ contains the upper sideband & Δf is negative, then the frequency components of the demodulated signal $v_0(t)$ are shifted outward by Δf, compared with the baseband signal $m(t)$ (fig (b))



ii) If the incoming SSB wave $s(t)$ contains the upper sideband & the freq. error Δf is positive, (or) if $s(t)$ contains the LSB & Δf is negative, then the frequency components of the demodulated signal $v_0(t)$ are shifted inward by Δf. (fig (c))

- In order to reduce the effect of frequency error distortion in telephone systems, the freq. error is limited to 2-5 Hz.

+

$$v_0(t) = \frac{1}{4} A_c A_c' \left[m(t) \cos\phi + \hat{m}(t) \sin\phi \right]$$

- the demodulated signal $v_0(t)$ contains an unwanted component proportional to $\hat{m}(t) \sin\phi$, which cannot be removed by filtering. This distortion appears as a phase distortion

Applying F.T to the above eq.

$$V_0(f) = \frac{1}{4} A_c A_c' \left[M(f) \cos\phi + \hat{M}(f) \sin\phi \right]$$

- From the definition of Hilbert transform $\hat{m}(t)$

we have

$$\hat{M}(f) = -j\,\text{sgn}(f)\,M(f)$$

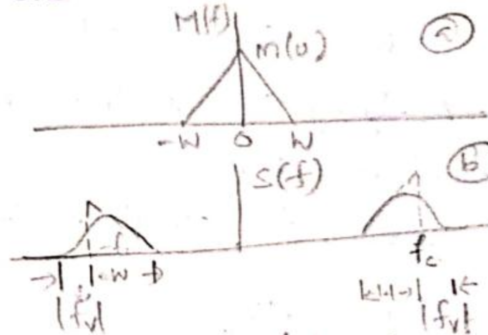
- Sub. $\hat{M}(f)$ in $V_0(f)$ we have

$$V_0(f) = \begin{cases} \frac{1}{4} A_c A_c' M(f) \exp(-j\phi), & f > 0 \\ \frac{1}{4} A_c A_c' M(f) \exp(+j\phi), & f < 0 \end{cases}$$

- The error in the phase of the local oscillator signal results in phase distortion, where each freq. component of m(t) undergoes a const. phase shift at the demodulator o/p.
- The human ear is relatively insensitive to phase distortion. The presence of phase distortion gives rise to a Donald Duck voice effect.

VESTIGIAL SIDEBAND MODULATION (VSB)

- Single sideband is suitable for the transmission of voice due to the energy gap that exists in the spectrum of voice signals between zero & few hundred hertz.
- The SSB modulation is inappropriate for the transmission of baseband signals ~~which~~ when the signal contains significant components at extremely low frequencies. This is due to the difficulty of isolating one sideband.
- In VSB modulation, one sideband is passed almost completely whereas just a trace (or) vestige of the other sideband is retained.
- The transmitted vestige of the unwanted sideband compensates for the amount removed from the desired sideband.

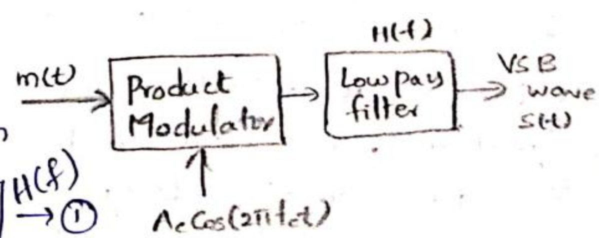


(a) Spectrum of baseband
(b) Spectrum of VSB

- The transmission bandwidth $B_T = W + f_v$. $f_v \rightarrow$ width of the vestigial sideband. VSB is a compromise between DSB-SC & SSB-SC.

Generation of DSB-SC VSB

- Vestigial sideband modulation can be generated by passing a DSB-SC wave through an appropriate filter of transfer function H(f) as shown.

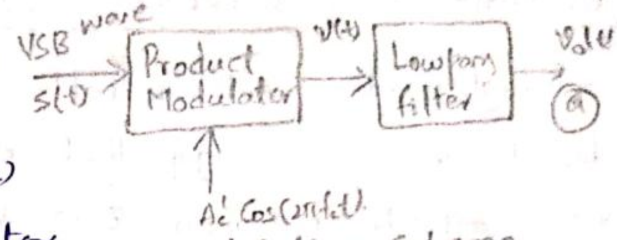


- The spectrum S(f) of the resulting VSB wave s(t) is given by

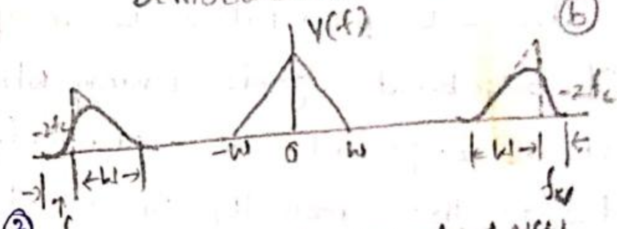
$$S(f) = \frac{A_c}{2}\left[M(f-f_c) + M(f+f_c)\right] H(f) \rightarrow (1)$$

$M(f) \rightarrow$ F.T of the baseband signal m(t)

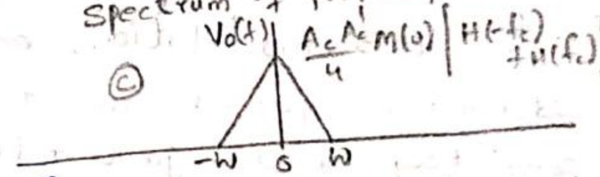
· The specification of the filter transfer function $H(f)$ can be determined by passing the VSB wave $s(t)$ through a coherent detector as shown in the fig. a



Demodulation Scheme.



Spectrum of P.M output v(t)



Spectrum of demodulated o/p $v_o(t)$

· The output of product modulator is given by

$$v(t) = A_c' \cos(2\pi f_c t)\, s(t) \rightarrow ②$$

· Applying F.T to the above eq. we have

$$V(f) = \frac{A_c'}{2}\left[S(f-f_c) + S(f+f_c)\right] \rightarrow ③$$

· Substituting eq ① in eq ③ & simplifying we have.

$$V(f) = \frac{A_c A_c'}{4} M(f)\left[H(f-f_c) + H(f+f_c)\right] + \frac{A_c A_c'}{4}\left[M(f-2f_c)H(f-f_c) + M(f+2f_c)H(f+f_c)\right] \rightarrow ④$$

· The spectrum $V(f)$ is shown in fig ⓑ The second term in the above eq. ④ represents a VSB wave corresponding to carrier freq. $2f_c$. This term is removed by the LPF to produce an output $v_o(t)$, the spectrum $V_o(f)$ is given by.

$$V_o(f) = \frac{A_c A_c'}{4} M(f)\left[H(f-f_c) + H(f+f_c)\right] \rightarrow ⑤$$

The spectrum of $V_o(f)$ is shown in fig ⓒ

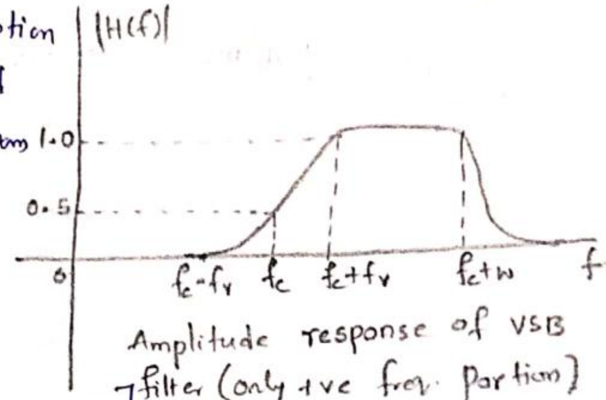
· For a distortionless reproduction of the original baseband signal $m(t)$, $V_o(f)$ should be a scaled version of $M(f)$. Therefore $H(f)$ must satisfy the condition

$$H(f-f_c) + H(f+f_c) = 2H(f_c) \rightarrow ⑥$$

$H(f_c) \rightarrow$ a const.

- The above requirement (eq. ⑥) is satisfied by using a filter with an amplitude response as shown in fig

- The time domain description of a VSB wave is based on canonical form for band-pass signals.



Amplitude response of VSB filter (only +ve freq. portion)

- According to this the in-phase component $S_c(f)$ is given by.

$$S_c(f) = \frac{1}{2}A_c M(f)\left[H(f-f_c) + H(f+f_c)\right] \rightarrow ⑦$$

Sub. eq ⑦

- Assuming the LPF transfer function H(f) of the VSB filter satisfies the condition of eq ⑥, the eq ⑦ is simplified as.

$$S_c(f) = \frac{1}{2}A_c M(f) \rightarrow ⑧$$

- The in-phase component in time-domain is given by

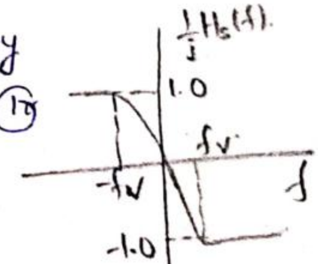
$$s_c(t) = \frac{1}{2}A_c\, m(t) \rightarrow ⑨$$

- Similarly the quadrature component $S_s(f)$ is given by

$$S_s(f) = \frac{j}{2}A_c M(f)\left[H(f-f_c) - H(f+f_c)\right] \rightarrow ⑩$$

- ~~The~~ From the above eq. we observe that $s_s(t)$ can be generated by passing the message signal m(t) through a filter with transfer function given by

$$H_s(f) = j\left[H(f-f_c) - H(f+f_c)\right] \rightarrow ⑫$$



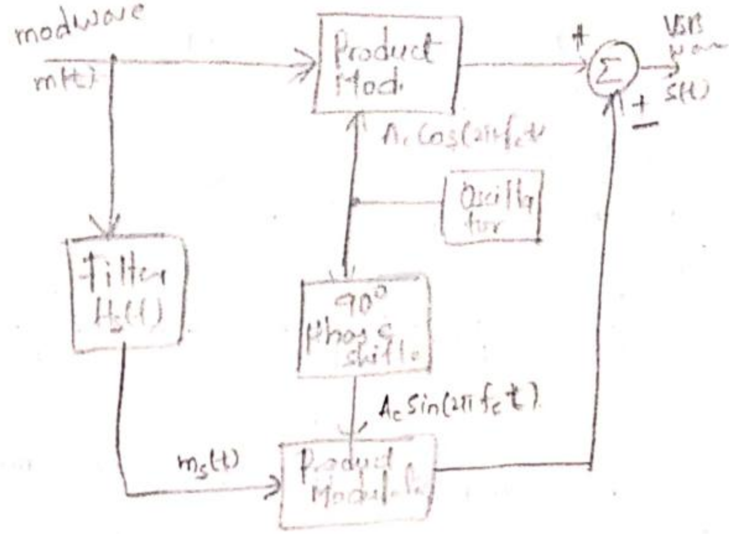
Freq res of filter producing the quadrature component of VSB wave

- Using $m_s(t)$ to denote the o/p of this filter for i/p m(t), we can express the quadrature component of VSB wave s(t) as

$$s_s(t) = \frac{1}{2}A_c m_s(t) \rightarrow ⑪$$

. Substituting eq (9) & (11) in the Canonical form representation we have

$$s(t) = \frac{1}{2} A_c m(t) \cos(2\pi f_c t) - \frac{1}{2} A_c m_s(t) \sin(2\pi f_c t) \rightarrow (12)$$



DEMODULATION OF VSB WAVE: (Envelope Detection):

• VSB modulation is used for the transmission of television & similar signals where good phase characteristics and transmission of low-frequency components are important.

• In commercial television broadcasting, a sizable carrier is transmitted together with the modulated wave. Therefore, envelope detector can be used for the demodulation of VSB waves.

• The time domain equation of VSB wave after adding the carrier component $A_c \cos(2\pi f_c t)$ scaled by a factor k_a is given by.

$$s(t) = A_c\left[1+\frac{1}{2}k_a m(t)\right]\cos(2\pi f_c t) - \frac{1}{2}k_a m_s(t)\sin(2\pi f_c t) \rightarrow (1)$$

$k_a \rightarrow$ Const, determines percentage modulation

• The envelope detector output, denoted by $a(t)$ is

$$a(t) = A_c\left\{\left[1+\frac{1}{2}k_a m(t)\right]^2 + \left[\frac{1}{2}k_a m_s(t)\right]^2\right\}^{1/2}$$

$$= A_c\left[1+\frac{1}{2}k_a m(t)\right]\left\{1+\left[\frac{\frac{1}{2}k_a m_s(t)}{1+\frac{1}{2}k_a m(t)}\right]^2\right\}^{1/2} \rightarrow (2)$$

• The above equation indicates that the distortion is contributed by the quadrature component $m_s(t)$ of the incoming VSB wave. This distortion can be reduced by

i) by reducing the percentage modulation to reduce k_a

ii) by increasing the width of the vestigial sideband to reduce $m_s(t)$.

• In commercial TV broadcasting, the VSB occupies a width of about 0.75 MHz. (one-sixth of a full sideband)

Summary:

In this Unit the students are introduced the concept of Modulation and its need. Then the concept related to Amplitude Modulation, Generation of AM, detection of AM was dealt. The others forms of AM modulation and their generation techniques, detection techniques are discussed. Some numerical problems on Power relations of AM wave is also discussed.

Assignment:

AMPLITUDE MODULATION

SHORT QUESTIONS

1. Define communication. Explain with block diagram the basic communication system.
2. Define modulation. Why is modulation required?
3. Define is modulation index of AM wave?
4. Describe the DSB-SC wave modulation with spectrum?
5. Compare Square law detector with envelope detector?
6. What are the advantages of ring modulator?
7. What is the difference between coherence detection and noncoherent detection?
8. Describe the principle of Vestigial side band modulation.
9. Compare AM, DSBSC, SSBSC and VSB?
10. Define under-modulation and over-modulation. Explain why over modulation is undesirable.

LONG ANSWER QUESTIONS

1. Describe Amplitude modulation for single tone and draw the spectrum also?
2. Explain the generation of AM using Switching Modulator?
3. Describe the detection of AM wave using envelope Detector.
4. Explain the generation of DSB-SC wave using ring Modulator.
5. Explain the detection of DSB-SC wave using Costas loop
6. Describe the Phase discrimination method for generating AM SSBSC signal?

References:

1. Communication Systems by Simon Haykin.
2. Electronic Communication systems by George F Kenedy.
3. Analog and Digital communications by Sanjay Sharma

BASIC CONCEPTS:

Angle Modulation:

- It is defined as the process in which the total phase angle of a carrier wave is varied in accordance with the instantaneous value of the modulating or message signal while keeping the amplitude of the carrier constant

Mathematical Representation:

- Let $\theta_i(t)$ denote the angle of a modulated sinusoidal carrier, which is a fn of msg. The resulting angle-modulated wave can be expressed as.

$$s(t) = A_c \cos[\theta_i(t)] \longrightarrow (1)$$

$A_c \rightarrow$ Carrier amplitude,

- A complete oscillation occurs whenever $\theta_i(t)$ changes by 2π radians. The average freq in hertz, over an interval from 't' to 't+Δt', is given by

$$f_{\Delta t}(t) = \frac{\theta_i(t-\Delta t) - \theta_i(t)}{2\pi \Delta t} \longrightarrow (2)$$

- The instantaneous freq. of the angle modulated wave s(t) is given by

$$f_i(t) = \lim_{\Delta t \to 0} f_{\Delta t}(t)$$

$$= \lim_{\Delta t \to 0} \left[\frac{\theta_i(t+\Delta t) - \theta_i(t)}{2\pi \Delta t} \right]$$

$$= \frac{1}{2\pi} \frac{d\theta_i(t)}{dt} \longrightarrow (3)$$

- The angular velocity is given by.

$$\omega_i(t) = \frac{d\theta_i(t)}{dt}.$$

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$$= \frac{1}{2\pi}\frac{d\theta_i(t)}{dt} \longrightarrow (3)$$

- The angular velocity is given by.

$$\omega_i(t) = \frac{d\theta_i(t)}{dt}.$$

- The total phase angle of an unmodulated carrier wave is given by $\theta_i(t) = 2\pi f_c t + \phi_c$. $\phi_c \rightarrow$ some phase angle.

ring

• If this angle $\theta_i(t)$ is varied according to the instantaneous value of the message or modulating signal, the carrier signal is then said to be angle modulated.

Types of Angle Modulation:

• The commonly used angle modulation techniques are i) Phase Modulation (PM) ii) Freq. Modulation (FM)

• Adv: Noise reduction, Improved system fidelity, Efficient use of Power

• Disadv: Increased B_T, Complex circuits.

Applications:

i) Radio Broadcasting
ii) Two way mobile radio
iii) Microwave Communication
iv) TV sound transmission
v) Cellular radio
vi) Satellite Communication

PHASE MODULATION:

Def: PM is that type of angle modulation in which the phase angle is varied linearly with a baseband (or) modulating signal m(t).

Math Rep:

Let, the unmodulated carrier signal is given by.

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

$$c(t) = A_c \cos[\theta_i(t)]$$

$$\theta_i(t) = \omega_c t + \theta_0$$

Neglecting θ_0 we have.

$$\theta_i(t) = \omega_c t \longrightarrow ①$$ {angle of unmodulated carrier}.

- This angle $\theta_i(t)$ is varied linearly with the base-band signal m(t). Therefore we have

$$\theta_i(t) = \omega_c t + K_p m(t)$$
$$= 2\pi f_c t + K_p m(t) \longrightarrow ②$$

- $2\pi f_c t \rightarrow$ angle of unmodulated carrier
  $K_p \rightarrow$ Const representing phase sensitivity of modulator in rad/volts.

- the phase modulated wave s(t) is given by.

$$s(t) = A_c \cos[\theta_i(t)]$$

$$s(t) = A_c \cos[2\pi f_c t + K_p m(t)] \longrightarrow ③$$

FREQUENCY MODULATION:

Def: FM is that type of angle modulation in which the instantaneous frequency $f_i(t)$ is varied linearly with a message signal m(t)

Math Rep:

The instantaneous freq. is given by.

$$f_i(t) = f_c + K_f m(t) \longrightarrow ①$$

$f_c \rightarrow$ freq. of the unmodulated carrier.

$k_f \rightarrow$ rep. frequency sensitivity of mod. hertz/volt.

From eq ①

$$2\pi f_i(t) = 2\pi f_c + 2\pi k_f m(t) \longrightarrow ②$$

Integrating the above eq. w.r.t 't'

$$\theta_i(t) = 2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt \rightarrow ③$$

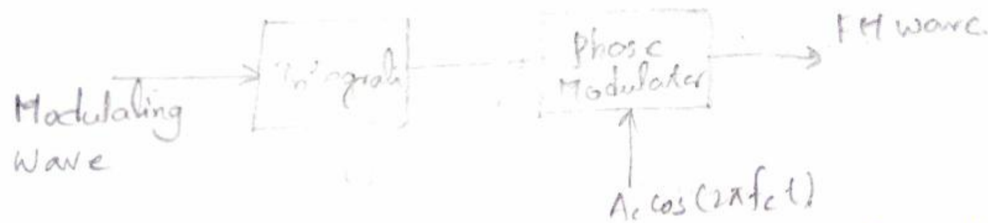
- The freq. modulated wave is given by.

$$S(t) = A_c \cos\left[\theta_i(t)\right]$$

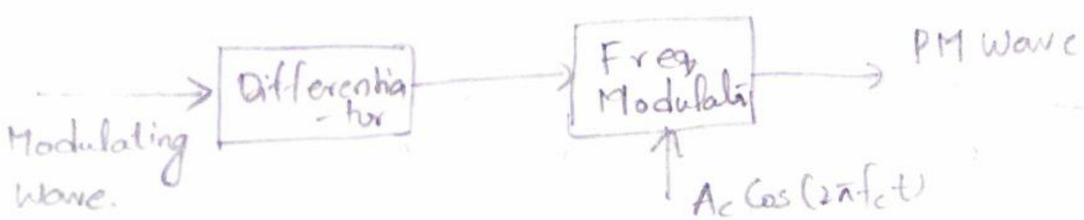
$$S(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right] \rightarrow ④$$

- Comparing PM and FM wave equations, FM wave may be regarded as a PM wave in which the modulating wave is $\int_0^t m(t)\,dt$ in place of $m(t)$.

- FM wave can be generated by first integrating $m(t)$ & then passing the result as i/p to Phase modulator, as shown



- Ily PM wave can be generated by first differentiating $m(t)$ & then passing the result as i/p to a freq. modulator as shown



SINGLE TONE FM & SPECTRAL ANALYSIS

- The equation of FM wave is given by

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right] \longrightarrow ①$$

- The above equation states that FM is a nonlinear modulation process. the B_T required by an FM wave is much greater than AM wave.

Spectral Analysis:

- Considering Single tone modulation, we have the message signal given by

$$m(t) = A_m \cos(2\pi f_m t) \longrightarrow ②$$

- The instantaneous freq. of the resulting FM wave is given by

$$f_i(t) = f_c + k_f A_m \cos(2\pi f_m t)$$

$$f_i(t) = f_c + \Delta f \cos(2\pi f_m t) \longrightarrow ③$$

Where $\Delta f = k_f A_m$ is called the 'frequency deviation' representing the max departure of the instantaneous freq. of the FM wave from the carrier freq. f_c.

- A fundamental property of FM wave is that the freq. deviation is proportional to the amplitude of the modulating signal & is independent of the modulation freq.

- The angle of the FM wave is given by

(from eq ③)

$$\theta_i(t) = 2\pi \int_0^t f_i(t)\,dt$$

$$= 2\pi f_c t + \frac{\Delta f}{f_m}\sin(2\pi f_m t) \longrightarrow ④$$

- The ratio of the freq. deviation Δf to the modulation freq. f_m is called as 'Modulation Index' of FM wave.

$$\therefore \beta = \frac{\Delta f}{f_m} \longrightarrow ⑤$$

$$\theta_i(t) = 2\pi f_c t + \beta \sin(2\pi f_m t) \longrightarrow (6)$$

- In the above eq, the parameter 'β' represents phase deviation of the FM wave, i.e., the max departure of the angle $\theta_i(t)$ from the angle $2\pi f_c t$ of the unmodulated carrier.

· The FM wave is given by

$$s(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \rightarrow (7)$$

· Depending on the value of the mod. index β, we have two types of FMs:

1) Narrowband FM for which β is small } compared to one radian.
2) Wideband FM for which β is large.

NARROW-BAND FM:

· Consider the equation of FM wave for single-tone modulation given by

$$s(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \rightarrow (1)$$

$\cos(A+B)$.

$$s(t) = A_c \cos(2\pi f_c t)\cos(\beta \sin(2\pi f_m t)) - A_c \sin(2\pi f_c t)\sin(\beta \sin(2\pi f_m t)) \rightarrow (2)$$

· For Narrow band FM, the mod. index β is small compared to one radian. From eq (1) we have the following approximations.

$$\cos\left[\beta \sin(2\pi f_m t)\right] \simeq 1 \rightarrow (3)$$

&

$$\sin\left[\beta \sin(2\pi f_m t)\right] \simeq \beta \sin(2\pi f_m t) \rightarrow (4)$$

Therefore, eq (2) simplifies to

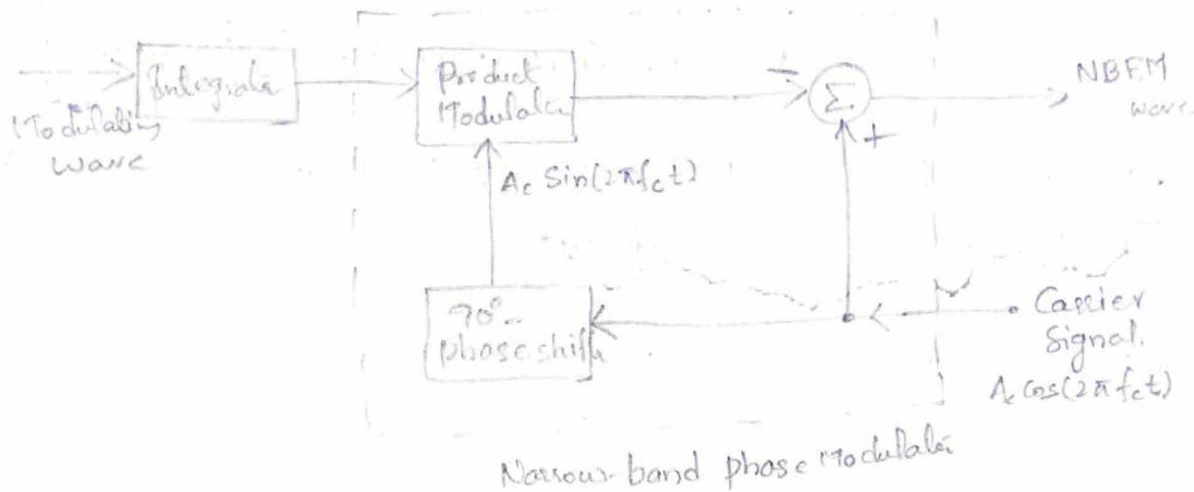
$$s(t) \simeq A_c \cos(2\pi f_c t) - \beta A_c \sin(2\pi f_c t)\sin(2\pi f_m t) \rightarrow (5)$$

· The above eq (5) defines the approx. form of a NBFM wave produced by single tone modulation.

* Generation of NBFM Wave:

· The eq. of NBFM wave is given by

$$s(t) \simeq A_c \cos(2\pi f_c t) - \beta A_c \sin(2\pi f_c t) \sin(2\pi f_m t)$$



Block diagram for generating a NBFM wave.

• The above method of generating NBFM wave is the direct implementation of NBFM wave equation given by eq (5)

• An FM wave ideally has a const. envelope, for a sinusoidal modulating wave of freq. f_m, the angle $\theta_i(t)$ is also sinusoidal with the same freq.

· The NBFM wave produced by the above method differs the ideal FM wave in two respects

i) The envelope contains a 'residual' amplitude modulation & therefore varies with time.

ii) For a sinusoidal modulating wave, the angle $\theta_i(t)$ contains 'harmonic distortion'

• By restricting the modulation index $\beta \leq 0.3$ radians, the effects of residual AM & harmonic distortions are limited to negligible levels.

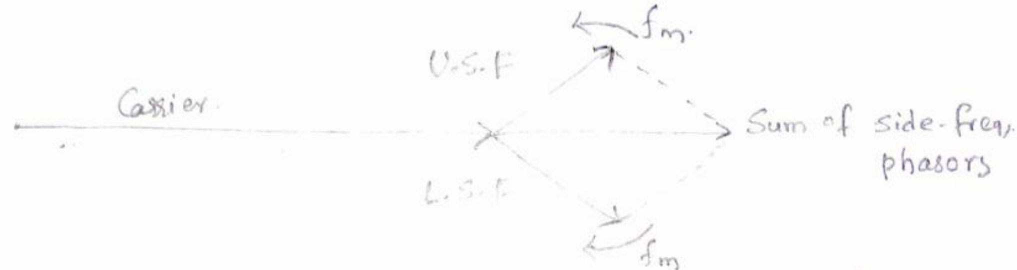
- Expanding the eq. of NBFM wave eq (5) we have

$$s(t) \simeq A_c \cos(2\pi f_c t) + \frac{1}{2}\beta A_c \left\{ \cos(2\pi(f_c+f_m)t) - \cos[2\pi(f_c-f_m)t] \right\} \quad \text{(6)}$$

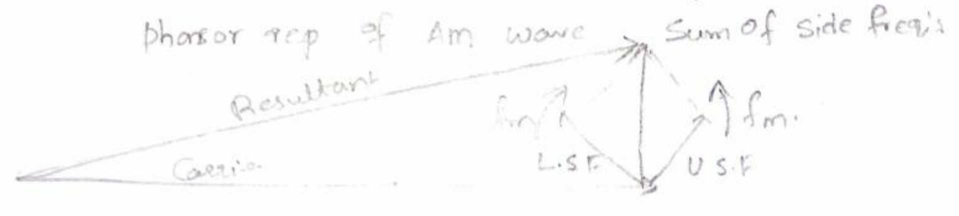
- The above eq. is similar to an AM eq. given by

$$S_{AM}(t) = A_c \cos(2\pi f_c t) + \frac{1}{2}\mu A_c \left\{ \cos[2\pi(f_c+f_m)t + \cos[2\pi(f_c-f_m)t] \right\}. \quad \text{(7)}$$

- The basic difference b/w an AM & NBFM wave is that the sign of the lower side-freq. in the NBFM wave is reversed.
- The NBFM wave req's the same transmission BW as the AM wave i.e., $B_T = 2f_m$.



phasor rep of Am wave



phasor rep. of NBFM wave.

- In the NBFM phasor, the resultant of the two side-frequency phasors is always at right angles to the carrier phasor. The effect of this is to produce a resultant phasor representing the NBFM wave which is approximately of the same amplitude as the carrier phasor, but out of phase with respect to it.
- In an AM phasor, the resultant phasor representing the AM wave has an amplitude different from that of the carrier phasor, but always in phase with it.

* WIDE BAND FM:

- When the value of modulation index 'β' is quite large, then in FM, a large number of sidebands are produced & hence the B.W of FM is sufficiently large. This type of FM system is known as wideband FM

- The expression for a single tone FM wave is given as

$$s(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \longrightarrow ①$$

- The above expression may be considered as a real part of the exponential phasor given by

$$s(t) = \mathrm{Re}\left[A_c \exp[j 2\pi f_c t + j\beta \sin(2\pi f_m t)]\right]$$

$$= \mathrm{Re}\left[\tilde{s}(t) \exp(j 2\pi f_c t)\right] \longrightarrow ②$$

Where $\tilde{s}(t) = A_c \exp[j\beta \sin(2\pi f_m t)] \longrightarrow ③$

↳ complex envelope of the FM wave $s(t)$.

- $\tilde{s}(t)$ is a periodic fn of time, $(1/f_m)$ & may be expanded in the form of a complex Fourier series as follows.

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j 2\pi n f_m t) \longrightarrow ④$$

Where

$$c_n = f_m \int_{-1/2f_m}^{1/2f_m} \tilde{s}(t) \exp(-j 2\pi n f_m t)\, dt$$

$$\left\{ \begin{aligned} &\underline{\underline{FS}} \quad g_p(t) = \sum_{n=-\infty}^{\infty} c_n \exp\left(\frac{j 2\pi n t}{T_0}\right) \\ &c_n = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} g_p(t) \exp\left(-\frac{j 2\pi n t}{T_0}\right) dt, \quad n = 0, \pm 1, \pm 2, \ldots \end{aligned} \right\}$$

$$c_n = f_m A_c \int_{-1/2f_m}^{1/2f_m} \exp[j\beta \sin(2\pi f_m t) - j 2\pi n f_m t]\, dt$$

Assuming. $x = 2\pi f_m t$, we have.

$$2\pi f_m t = x \qquad 2\pi f_m dt = dx \qquad dt = \frac{dx}{2\pi f_m}$$

$$c_n = \frac{A_c}{2\pi} \int_{-\pi}^{\pi} \exp[j(\beta \sin x - n x)]\, dx. \longrightarrow ⑤$$

• The integral on the right-hand side of eq ⑤ is known as the n^{th} order Bessel function of the first kind & argument β. This fn is denoted by $J_n(\beta)$.

$$J_n(\beta) = \frac{1}{2\pi}\int_{-\pi}^{\pi} \exp\left[j(\beta\sin x - nx)\right] dx \longrightarrow ⑥$$

∴ Eq ⑤ simplifies as

$$c_n = A_c J_n(\beta) \longrightarrow ⑦$$

• Substituting eq ⑦ in eq ④ we have

$$\tilde{s}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j2\pi n f_m t) \longrightarrow ⑧$$

• Substituting eq ⑧ in eq ② we have

$\to \cos x + j\sin x$

$$s(t) = A_c\, Re\left[\sum_{n=-\infty}^{\infty} J_n(\beta) \exp\left[j2\pi(f_c + nf_m)t\right]\right] \longrightarrow ⑨$$

• Evaluating the real part of RHS in the above eq. ⑨ we have

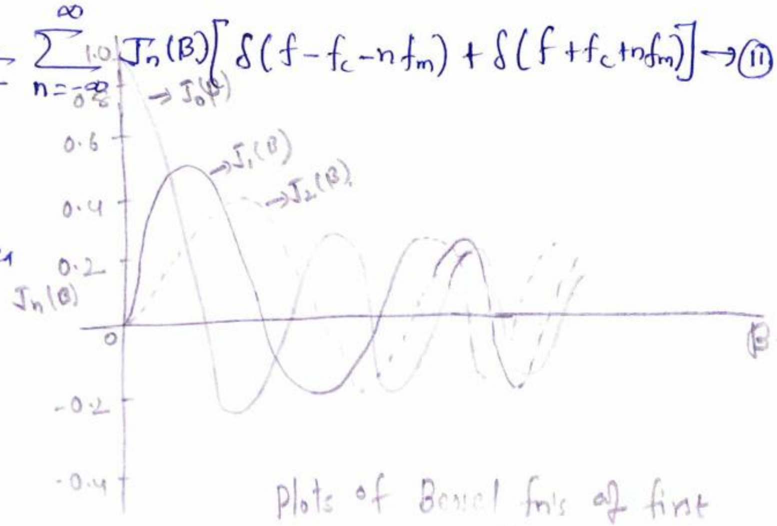
$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos\left[2\pi(f_c + nf_m)t\right] \longrightarrow ⑩$$

• The above eq. is the desired form for the F.S rep. of the single tone FM wave s(t) for an arbitrary value of β.

• The spectrum of s(t) is obtained by taking the F.T of both sides of eq ⑩.

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)\left[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)\right] \longrightarrow ⑪$$

• The graph shows Bessel fn $J_n(\beta)$ versus the modulation index β for different +ve integer values of 'n'.



Plots of Bessel fn's of first Kind

TRANSMISSION BANDWIDTH OF FM WAVES

IV-

- Theoritically, an FM wave contains an infinite number of side-frequencies & therefore the bandwidth required to transmit such a signal is similarly infinite in extent
- In practice, the FM wave is effectively limited to a finite number of significant side-frequencies compatible with a specified amount of distortion.
- Carson's rule provides a thumb formula to calculate the bandwidth of a single-tone WBFM.

Mathematically

$$B.W \simeq 2\Delta f + 2f_m = 2\Delta f\left[1+\frac{1}{\beta}\right].$$

Prob: Find the B.W of a commercial FM transmission if the freq. deviation $\Delta f = 75KHz$. & $f_m = 15KHz$.

$$B_T = 2(\Delta f + f_m) = 2(75+15) = 18KHz.$$

Prob: A single-tone FM is represented by the voltage eq. as $v(t) = 12\cos(6\times10^8 t + 5\sin 1250t)$.

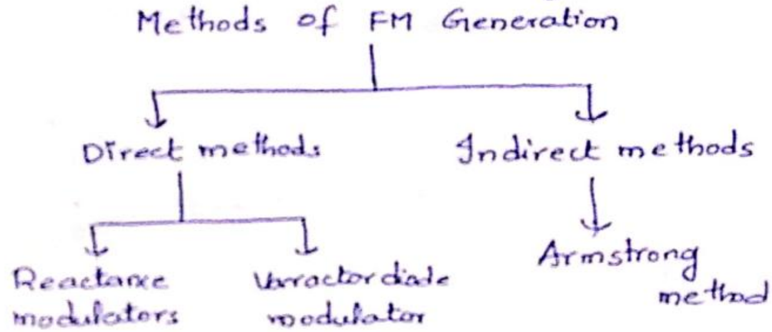
Determine i) Carrier freq. ii) Modulating freq.
iii) Mod. index iv) Max. deviation.
v) What power will this FM wave dissipate in 10Ω resistors.

GENERATION OF FM WAVES :

- The FM modulator circuits used for generating FM signals may be put into two categories as under.

i) The direct method or parameter variation method.

ii) The Indirect method or the Armstrong method.

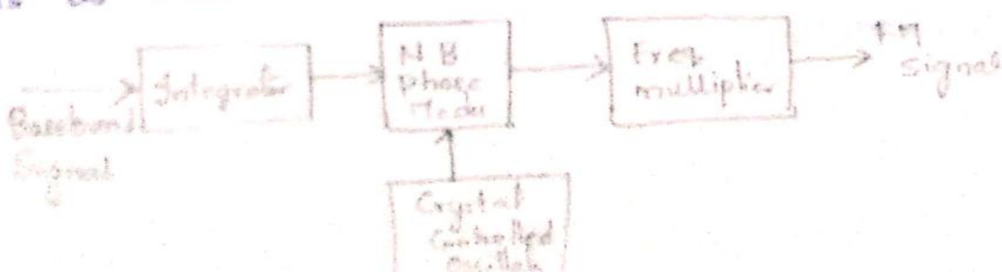


- In the indirect method the modulating wave is first used to produce a NBFM wave, & then frequency multiplication is used to increase the frequency deviation to the desired level.

- In the direct method of producing freq. modulation the carrier freq is directly varied in accordance with the i/p baseband signal.

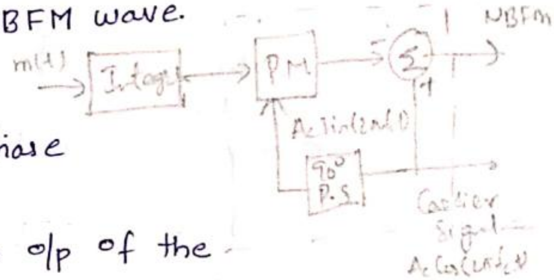
→ Indirect FM:

- The indirect method of generating a WBFM wave was first proposed by Armstrong & the arrangement is as shown below



Block diagram of Indirect method of generating a WBFM signal

- The baseband signal m(t) is first integrated & then used to phase modulate a crystal-controlled oscillator.
- The distortion inherent in the phase modulator is minimized by keeping the max. phase deviation (or) mod. index β very small. Therefore the o/p of phasemod is a NBFM (β << 1 rad).
- The o/p of the phase modulator is multiplied in frequency using a freq. multiplier circuit to produce the desired WBFM wave.



- The fig shows implementation of NBPhase modulator.
- Let $s_1(t)$ denote the o/p of the phase modulator, & is given by.

$$s_1(t) = A_1 \cos\left[2\pi f_1 t + 2\pi k_f \int_0^t m(t)\,dt\right] \rightarrow (1)$$

$f_1 \rightarrow$ freq. of the crystal-controlled oscillator,

$k_f \rightarrow$ Const. (freq. sensitivity).

- For a sinusoidal modulating wave, the o/p $s_1(t)$ is given by.

$$s_1(t) = A_1 \cos\left[2\pi f_1 t + \beta_1 \sin(2\pi f_m t)\right] \rightarrow (2)$$

$\beta_1 \rightarrow$ mod. index ($\beta_1 < 0.3$ rad)

- The phase modulator o/p is next multiplied 'n' times in freq. by the freq. multiplier, producing the desired WBFM.

$$s(t) = A_c \cos\left[2\pi f_c t + 2n\pi k_f \int_0^t m(t)\,dt\right] \rightarrow (3)$$

Where $f_c = n f_1$

- For sinusoidal modulating wave eq (3) is expressed as

$$s(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \rightarrow (4)$$

$\beta = n\beta_1$. By choosing 'n' properly, the final value of the mod. index 'β' is set to any desired value

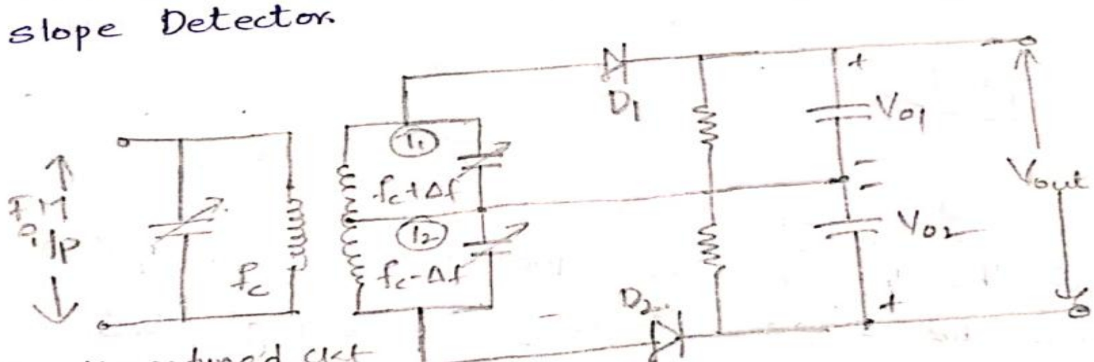
DETECTION OF FM WAVES:

• The detection of FM is different compared to AM. The FM detector should be able to produce the signal whose amplitude is proportional to the deviation in the frequency of FM signal. The purpose of FM detector is similar to freq to voltage converter.

• FM detectors ⟶ Slope detectors (Freq. discri).
⟶ phase discriminator
⟶ Ratio detector

→ **Frequency discriminator:**

– Also called as Round-Travis Detector (or) Blanced slope Detector.



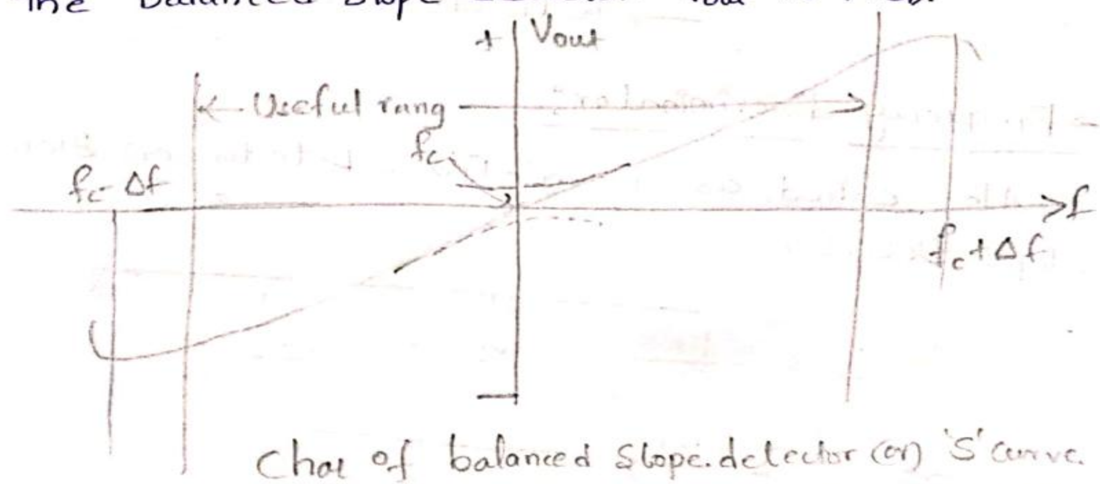
T1 — Upper tuned ckt
T2 - Lower tuned ckt

Balanced Slope Detector.

• The above figure shows the circuit of balanced slope detector. It consists of two identical circuits connected back to back. The FM signal is applied to the tuned LC circuit. Two tuned LC circuits are connected in series

• The inductance of the secondary tuned LC circuit is coupled with the inductance of the primary LC circuit; thus forming a tuned transformer.

- The input side LC circuit is tuned to the carrier freq f_c. T_1 is tuned to $f_c + \Delta f$ & T_2 is tuned to $f_c - \Delta f$. The input FM signal is coupled to T_1 & T_2 180° out of phase.
- The secondary side tuned circuits (T_1 & T_2) are connected to diodes D_1 & D_2 with RC loads. The total o/p V_{out} is equal to difference b/w V_{o1} & V_{o2}. The following fig shows the characteristic of the balanced slope detector. V_{out} Vs freq.



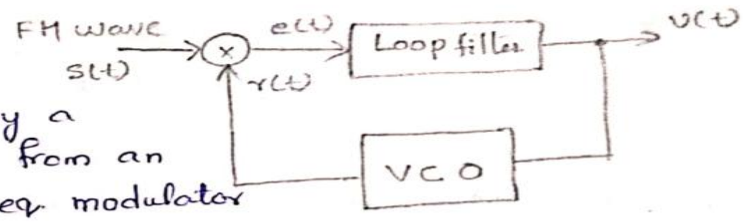
Char of balanced slope detector (or) 'S' curve.

- When the i/p freq. is equal to f_c, both T_1 & T_2 produce the same voltage & the voltages V_{o1} & V_{o2} are identical. Therefore V_{out} is zero. as shown in 's' curve
- When the i/p freq. is $f_c + \Delta f$, the upper tuned circuit T_1 produces max voltage whereas the lower tuned circuit T_2 produces min voltage. $\therefore V_{out} = V_{o1} - V_{o2}$ is max +ve for $f_c + \Delta f$.
- When the i/p freq. is $f_c - \Delta f$, the lower tuned ckt T_2 produces max voltage & the upper tuned circuit T_1 produces min voltage. $\therefore V_{out} = V_{o1} - V_{o2}$ is max -ve for $f_c - \Delta f$ as shown in 'S' curve.
- For other freq's of i/p, the o/p (V_{out}) is produced according to the characteristic shown in the fig. The linearity of the char. depends upon the alignment of tuning ckts & coupling char's of the tuned coils

→ PHASE LOCKED LOOP:

• The PLL is a -ve f/b system which consists of 3 major components. i) a multiplier, ii) a loop filter iii) a VCO. connected together in the form of a f/b loop shown below

• The VCO is a sine-wave generator whose freq. is determined by a voltage applied to it from an ext. source. Any freq. modulator may serve as a VCO.



• The i/p signal applied to the PLL is an FM wave defined by

$$s(t) = \underbrace{A_c}_{\text{carrier amp.}} \sin[2\pi f_c t + \phi_1(t)] \longrightarrow (1)$$

• With a modulating wave m(t) we have

$$\phi_1(t) = 2\pi k_f \int_0^t m(t)\,dt \longrightarrow (2)$$

k_f → freq. sensitivity of the Freq. Mod.

• Let the VCO o/p is defined as

$$r(t) = \underbrace{A_v}_{\text{Amp.}} \cos[2\pi f_c t + \phi_2(t)] \longrightarrow (3)$$

• With a control voltage v(t) applied to the VCO i/p, we have

$$\phi_2(t) = 2\pi k_v \int_0^t v(t)\,dt \longrightarrow (4)$$

k_v → freq. sensitivity of the VCO. (Hz/volt)

• The incoming FM wave s(t) & the VCO o/p r(t) are applied to the multiplier producing two components

i) a high freq. comp. given by.

$$k_m A_c A_v \sin[4\pi f_c t + \phi_1(t) + \phi_2(t)],$$

ii) a low-freq. comp. given by.

$$k_m A_c A_v \sin[\phi_1(t) - \phi_2(t)]$$

k_m → multiplier gain, measured in $volt^{-1}$.

• The high freq. component is eliminated by the combi. of filter & VCO. Therefore, the i/p to the loop filter is given by

$$e(t) = K_m A_c A_v \sin[\phi_e(t)] \rightarrow (5)$$

$\phi_e(t) \rightarrow$ phase error defined by

$$\phi_e(t) = \phi_1(t) - \phi_2(t).$$
$$= \phi_1(t) - 2\pi K_v \int_0^t v(t)\,dt \rightarrow (6)$$

• The loop filter operates on its i/p $e(t)$ to produce the o/p

$$v(t) = \int_{-\infty}^{\infty} e(\tau)\, h(t-\tau)\, d\tau \rightarrow (7)$$

$h(t) \rightarrow$ impulse response of the filter.

Using eq's (5), (6) & (7) we have.

$$\phi_e(t) = \phi_1(t) - 2\pi K_v \int_0^t \int_{-\infty}^{\infty} e(\tau)\, h(t-\tau)\, d\tau$$

$$\frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi K_v \int_{-\infty}^{\infty} K_m A_c A_v \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau$$

$$= \frac{d\phi_1(t)}{dt} - 2\pi K_0 \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau \rightarrow (8)$$

Where $K_0 = K_m K_v A_c A_v \rightarrow (9)$

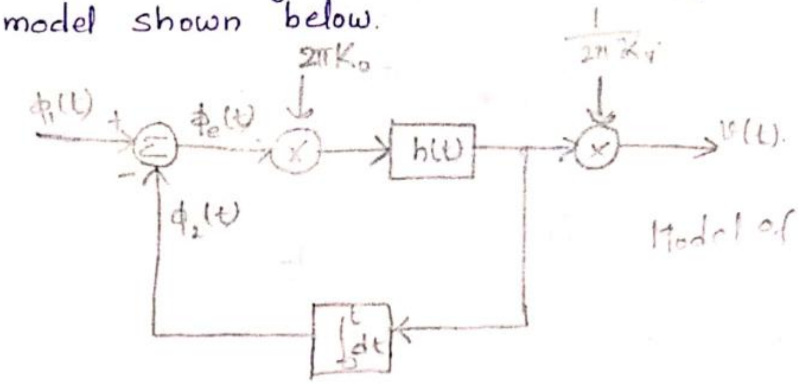
↳ & has the dimensions of freq.

• When the phase error $\phi_e(t)$ is zero, the PLL is said to be in phase-lock. When $\phi_e(t)$ is at all times small compared with one radian, we have.

$$\sin[\phi_e(t)] \simeq \phi_e(t) \rightarrow (9)$$

$$\therefore \frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi K_0 \int_{-\infty}^{\infty} \phi_e(\tau) h(t-\tau)\, d\tau.$$

. Thus we may represent the PLL by the linearized model shown below.



Model of PLL linearized

. According to this model, the phase error $\phi_e(t)$ is related to the i/p phase $\phi_1(t)$ by the integro-differential equation:

$$\frac{d\phi_e(t)}{dt} + 2\pi K_0 \int_{-\infty}^{\infty} \phi_e(\tau) h(t-\tau) d\tau = \frac{d\phi_1(t)}{dt} \rightarrow (10) \; (14)$$

. Transforming the above eq into freq. domain & solving for $\Phi_e(f)$ we have.

$$\Phi_e(f) = \frac{1}{1+L(f)} \Phi_1(f) \longrightarrow (11) \; (12)$$

$H(f) \rightarrow$ transfer fn of loop filter.

$L(f) \rightarrow$ open-loop transfer fn of the PLL defined as.

$$L(f) = K_0 \frac{H(f)}{jf} \longrightarrow (12) \; (13)$$

. If $L(f)$ is made very large compared to unity then from eq (11) 12 $\Phi_e(f)$ approaches zero. that is, the phase of the VCO becomes asymptotically equal to the phase of the incoming wave, & phase lock is thereby established.

. The F.T of the PLL o/p $v(t)$ is related to $\Phi_e(f)$ by.

$$V(f) = \frac{K_0}{K_v} H(f) \Phi_e(f) \longrightarrow (13) \; 14$$

(or)

$$V(f) = \frac{jf}{K_v} L(f) \Phi_e(f) \longrightarrow (14) \; 15$$

Substituting eq (11) in eq (14) we have

$$V(f) = \frac{(jf/K_v)\,L(f)}{1+L(f)}\,\Phi_1(f) \rightarrow (15)$$

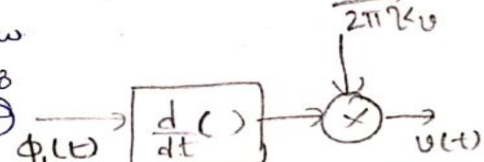
If $|L(f)| \gg 1$, the above eq is approx. as

$$V(f) \simeq \frac{jf}{K_v}\,\Phi_1(f) \rightarrow (16)$$

- The corresponding time-domain relation is

$$v(t) \simeq \frac{1}{2\pi K_v}\,\frac{d\Phi_1(t)}{dt} \rightarrow (17)$$

- From the above eq. the PLL may be modeled as a differentiator with its o/p scaled by the factor $1/2\pi K_v$. shown below



Simplified model. when loop gain is large compared to unity.

- Substituting eq (2) in eq (17) the resulting o/p signal of the PLL is

$$v(t) \simeq \frac{K_f}{K_v}\,m(t) \rightarrow (18)$$

$\therefore$ The o/p $v(t)$ of the PLL is approx. the same except for the scale factor K_f/K_v as the original baseband signal $m(t)$. & freq. demodulation is accomplished.

Summary:

Students are introduced the concept of Angle modulation and the types of Angle modulation namely Phase modulation and frequency modulation. The concept of Narrow band FM and Wide Band FM, Bandwidth requirements of FM are dealt. The various methods used for the generation and detection of FM are introduced.

Assignment:

SHORT QUESTIONS

1. Define modulation index and bandwidth of FM.
2. Compare FM and AM.
3. What is Carson's Rule?
4. What is wideband FM & Narrowband FM?
5. What are Advantages & Applications of FM?
6. Generate PM wave from FM and FM from PM?

LONG ANSWER QUESTIONS

1. Derive the expression for single tone FM wave and wide band FM wave.
2. Explain the detection of FM wave using balanced frequency discriminator.
3. Compare the direct and indirect methods of generating FM signals. Explain Armstrong method of generating FM signals with a neat block schematic diagram.
4. With a neat block diagram explain the generation of narrow band and wide band FM.
5. Explain phase locked loop.

References:

1. Communication Systems by Simon Haykin.
2. Electronic Communication systems by George F Kenedy.
3. Analog and Digital communications by Sanjay Sharma
4. Analog and Digital Communications by P. Chakrabarti.

Radio Transmitters:

A radio transmitter must generate a signal with the right type of modulation, with sufficient power at the right carrier frequency & reasonable efficienc

Types:

① Depending on service involved
- Radio Telegraph
- Television
- Radar
- Navigation

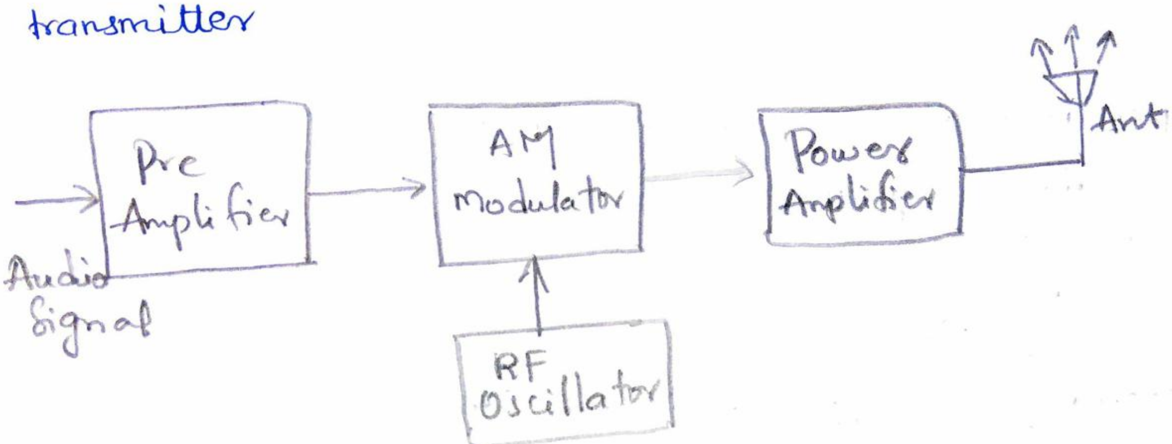
② Depending on Type of Modulation
- Amplitude Modulation
- Frequency modulation
- Pulse Modulation

③ Depending on the type of Carrier frequency
- Long Wave transmitters
- Medium wave transmitters
- Short wave transmitters
- Microwave transmitters
- V.H.F and U.H.F transmitters

④ Depending on the Power used.
- low level Modulated AM Transmitter
- High level Modulated AM Transmitter.

AM Transmitter:

AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted. The below figure shows an AM transmitter

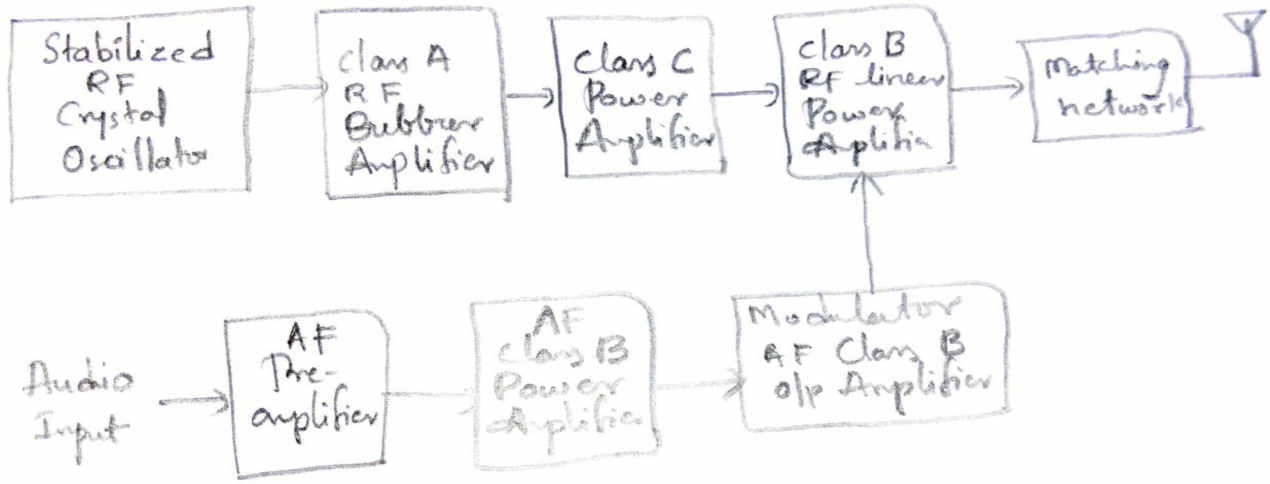


Working:

- The audio signal from the output of a microphone is sent to the preamplifier, which boosts the level of the modulating signal.
- The RF Oscillator generates the carrier signal.
- Both modulating & the carrier signal are applied to the Am modulator to generate an Amplitude modulated signal.
- Power amplifier is used to increase the power levels of Am wave. this wave is finally passed to the antenna to be transmitted.

→ low-level - AM transmitter

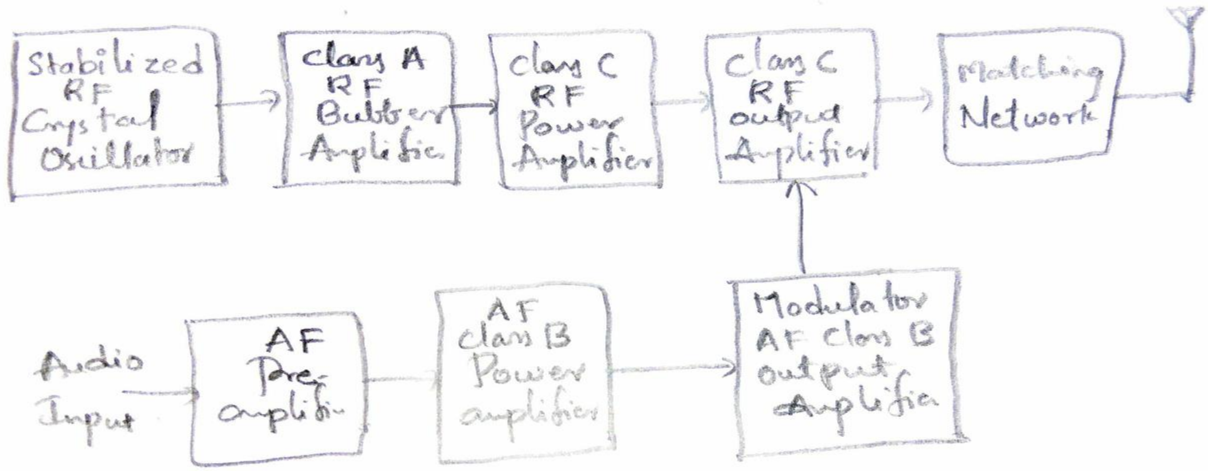
In low level modulation, the generation of AM wave takes place in the initial stage of amplification, i.e., at a low power level. The generated AM Signal then amplified using numbers of amplifier Stages



low level AM transmitter.

High level AM transmitter:

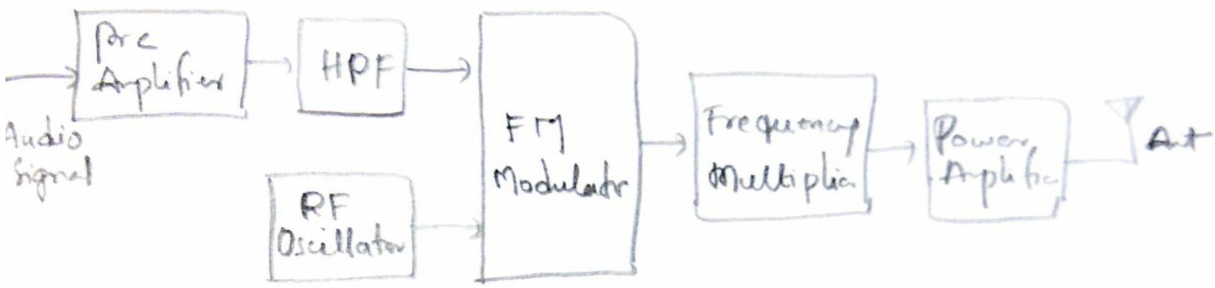
In high level modulation, modulation takes place in the final stage of amplification & therefore modulation circuitry has to handle high power



High level AM Transmitter

FM Transmitter:

FM transmitter takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted.



Working:

- The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
- this signal is then passed to the high pass filter, which acts as a preemphasis network to filter out the noise and improve the signal to noise ratio.
- This signal is further passed to the FM modulator
- the oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal
- Several stages of frequency multiplier are used to increase the operating frequency. RF power amplifier is used at the end to increase the power of the modulated signal. this FM output is finally passed to the antenna to be transmitted

RADIO RECEIVERS

• In a communication system, a radio transmitter radiates (or) transmits a modulated carrier signal which travels through a transmission medium & reaches the i/p of a "radio receiver."

• The functions of a radio receiver are
i) Intercept the incoming modulated signal by the Rx antenna.
ii) Select the desired signal & reject the unwanted signals.
iii) Amplify this selected R.F signal.
iv) Detect the modulated signal to get back the original modulating signal.
v) Amplify the modulating freq. signal.

→ Receiver Types:

A) Depending upon the applications.
i) Amplitude Modulation (A.M) Broadcast Receivers
ii) Frequency Modulation (F.M) Broadcast Receivers
iii) Communication Receivers
iv) Television Receivers
v) Radar Receivers

B) Depending upon the fundamental aspects,
i) Tuned Radio Freq (TRF) Receivers
ii) Superheterodyne Receiver.

→ Tuned Radio Frequency (TRF) Receivers:

• This is simplest radio receiver as shown below

- The first block of this receiver is an RF stage which contains two (or) three RF amplifiers. These RF amplifiers are tuned RF amplifiers i.e, they have variable tuned ckt at the i/p & o/p sides.
- The antenna at the i/p of the receiver selects the desired signal (i.e, station) from different sources (stations) with the help of variable tuned ckt of RF amplifiers.
- The selected signal of the order of μV is amplified by the RF amplifier (RF stage)
- The amplified incoming modulated signal is applied to the demodulator which produces the modulating (or) baseband signal (audio signal) at its o/p.
- This audio signal is amplified by audio amplifier. which is further amplified by ~~a~~ a power amplifier upto desired power level to drive the loudspeaker.
- The loudspeaker is the last stage of the receiver. which changes electrical signal into sound signal.

→ Drawbacks of TRF Receiver:

TRF receiver is cheaper & the simplest one but has certain drawbacks as follows.

i) The TRF receiver has the tendency to oscillate at higher frequencies from the multistage RF amplifiers with high gain & operating at same same frequency. This problem is also termed as instability of the receiver.

ii) The selectivity of TRF receiver is poor. The selectivity of the receiver is determined by the B.W which should be adequate to receive all components of the transmitted signal.

SUPERHETERODYNE RECEIVER:

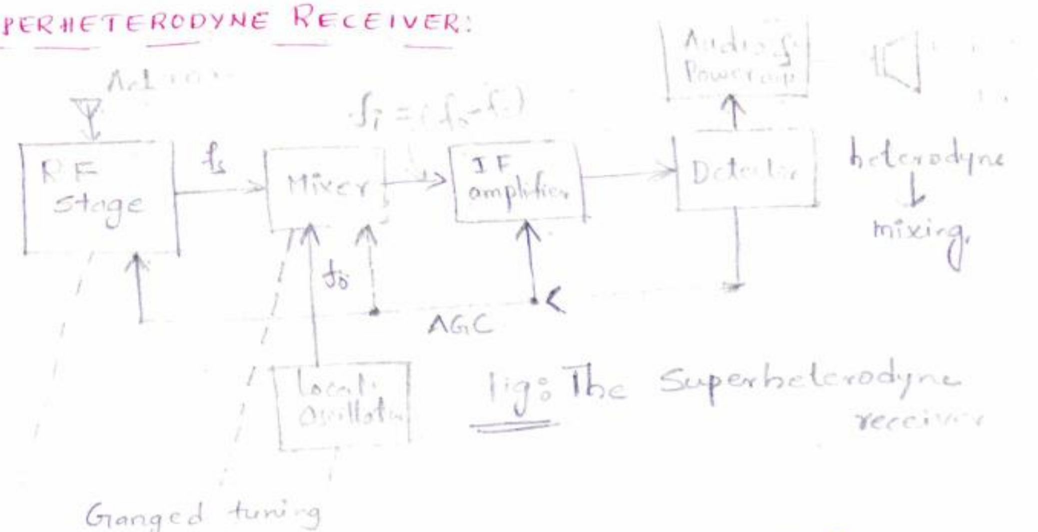


fig: The Superheterodyne receiver

- In this receiver, the incoming RF signal frequency is combined with the local oscillator signal frequency through a mixer & is converted into a signal of lower fixed frequency. This frequency is known as "intermediate frequency".
- This intermediate frequency signal is now amplified & demodulated to reproduce the original signal. Here the incoming signal frequency is mixed with local oscillator frequency. Therefore this receiver is known as Superheterodyne receiver.
- In this receiver, a constant frequency difference is maintained between the local oscillator signal freq. & incoming RF signals freq. through capacitance tuning in which the capacitances are ganged together & operated by a common control knob.
- The IF amplifier contains a number of tranformers each consisting of a pair of mutually coupled tuned circuits which are operated at a specially chosen frequency (I.F). The I.F amplifier provides most of the gain (sensitivity) & bandwidth requirement (selectivity)

• Due to the narrow bandwidth, the I.F amplifier rejects all other frequencies except intermediate freq. This rejection process reduces the risk of interference from other stations & sources (Main adv. of Superh. t)

• After the I.F amplifier, the signal is applied at the input of demodulator which extracts the original modulating signal. This signal is amplified by the audio amplifier & then Power amplifier to drive the loud speaker.

• This receiver is suitable for most of the radio receiver applications like AM, FM, Comm'n, SSB, television & radar receivers.

Advantages:

i) No variation is bandwidth.

ii) High sensitivity & selectivity.

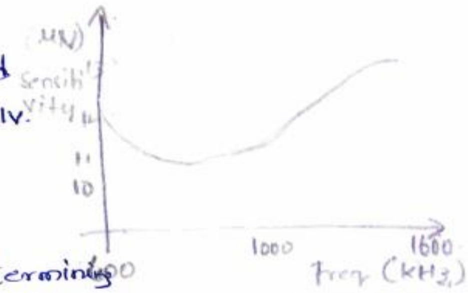
iii) High adjacent channel rejection.

Receiver Characteristics: (Superheterodyne AM Receiver)

Sensitivity:

The sensitivity of a radio receiver is its ability to amplify weak signals. It is defined in terms of the voltage that must be applied to the receiver i/p to give a standard output power, measured at the o/p terminals.

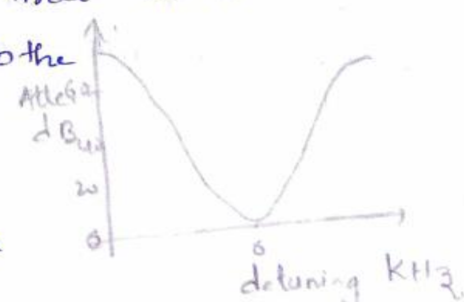
Sensitivity is often expressed in microvolts or in decibels below 1V & measured at three points along the tuning range.



· The important factors determining the sensitivity of a superheterodyne receiver are the gain of the IF amplifier(s) & that of the RF amplifier.

Selectivity:

The selectivity of a receiver is its ability to reject unwanted signals. It is expressed as a curve, which shows the attenuation that the receiver offers to signals at frequencies near to the one to which it is tuned.

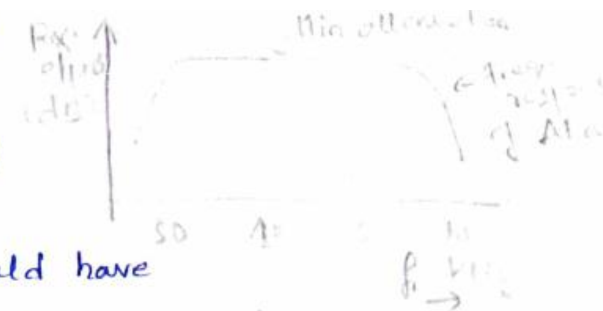


• Sensitivity determines the adjacent-channel rejection of a receiver.

Fidelity:

The fidelity is the ability of a receiver to reproduce all the modulating frequencies equally.

High fidelity is essential in order to reproduce the original signal without any distortion.



. So the AF amplifier should have a flat frequency response over a wide range of audio frequencies as shown.

Fidelity Curve

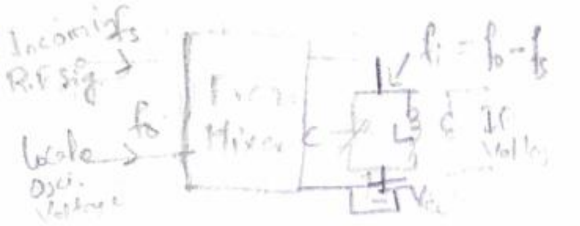
Double Spotting:

· When a receiver picks up the same shortwave station at two nearby points on the receiver dial, the double spotting phenomenon takes place. The main cause for double spotting is poor front end selectivity.

· If image-frequency rejection is improved, then there will be decrease in the double spotting occurrence.

Frequency Mixer

· A frequency Mixer is a non-linear device which produces a number of freq's when two different freq's are applied at the input of it.



Freq. Mixer

· A freq. mixer uses a device which has non-linear dynamic characteristics.

455 KHz for Am

· the mixer output contains the frequencies, $f_s, f_o, mf_o \pm nf_s$.

$m, n \rightarrow$ integers.

· $f_o - f_s$ is selected in the outside of the mixer by a tuned circuit which is tuned to this difference freq. term. This freq. is called Intermediate freq. (I.F)

→ Frequency Changing & Tracking:

• The frequency changer is the mixer. It is a non-linear device like multiplier with two inputs & one output terminal.

• The signal received by the antenna coupled to the RF stage is fed to one of the i/p terminals while the o/p of the L.O is fed to the other.

• This mixer has many frequencies present at the o/p including the difference b/w the two inputs. This difference freq is matched to the IF freq & the o/p ckt of the mixer is tuned to this freq.

• The most common type of mixers are made of BJTs, FETs & ICs.

• A superheterodyne receiver has several tuning stages which must be correctly tuned for proper reception of the signal of any particular station.

• The different tuning ckts are mechanically coupled & tuned by a single control dial. This is known as superheterodyne tracking.

• Irrespective of the freq. of the station to be received, the RF & the mixed i/p tuned ckt are tuned to this station freq.

• The local oscillator freq. is simultaneously changed with this tuning freq. so that the L.O. freq. is always greater than the station freq. by an amount equal to the freq. of the IF stage.

• If there is any error in selecting the exact L.O freq. so that the mixed L.O. freq. & the RF freq do not produce the exact IF freq, then it is called tracking error. This produces a wrong IF freq to be fed to the IF stage & is highly undesirable

→ Image Frequency & its Rejection, Intermediate freq:

• A superheterodyne receiver suffers from a major drawback known as Image frequency prob. This prob. arises because of the use of heterodyne principle.

• The freq. conversion process carried out by the local oscillator & the mixer often allows undesired freq. in addition to the desired incoming freq.

• In a standard broadcast receiver, the L.O. freq. is always made higher than the incoming signal freq. Mathematically.

$$f_o = f_s + f_i \rightarrow ① \text{ (or) } f_i = f_o - f_s.$$

$f_o \rightarrow$ L.O. freq, $f_s \rightarrow$ desired incoming freq,
$f_i \rightarrow$ intermediate freq.

$$\therefore f_i = f_o - f_s \rightarrow ②$$

• If a freq. f_{si} manages to reach the mixer, such that

$$f_{si} = f_o + f_i \rightarrow ③$$

then this freq. f_{si} would also produce f_i when it is mixed with f_o. This undesired IF signal will also be amplified by the I.F stage & thus would cause interference. This has the effect of two sources (or) stations being received simultaneously.

$f_{si} \rightarrow$ Image freq. & is defined as signal freq. plus twice the intermediate freq.

Sub eq ① in eq ③ we have.

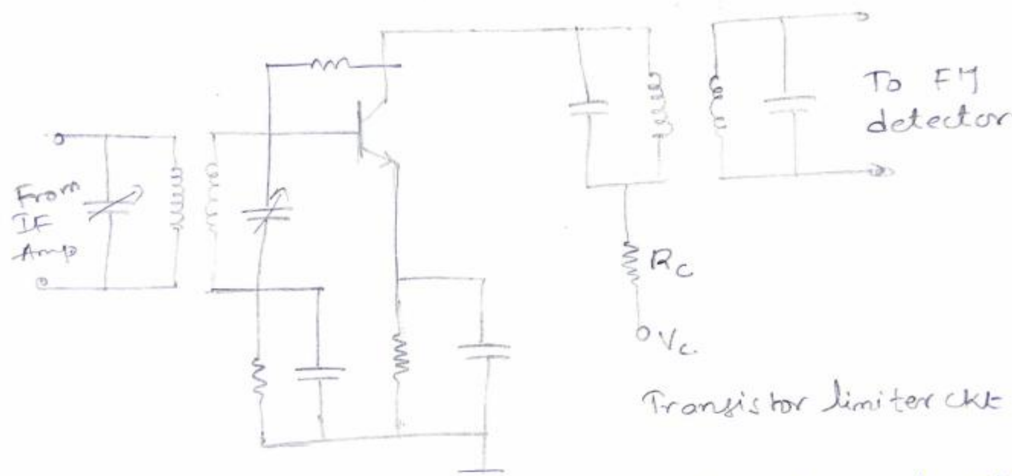
$$f_{si} = f_s + 2f_i \rightarrow ④$$

• The rejection of an image freq. signal by a single tuned ckt may be defined as the ratio of the gain at the signal freq. to the gain at the image freq. This is given by.

$$\alpha = \sqrt{1+Q^2\rho^2} \rightarrow ⑤, \quad \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} \rightarrow ⑥$$

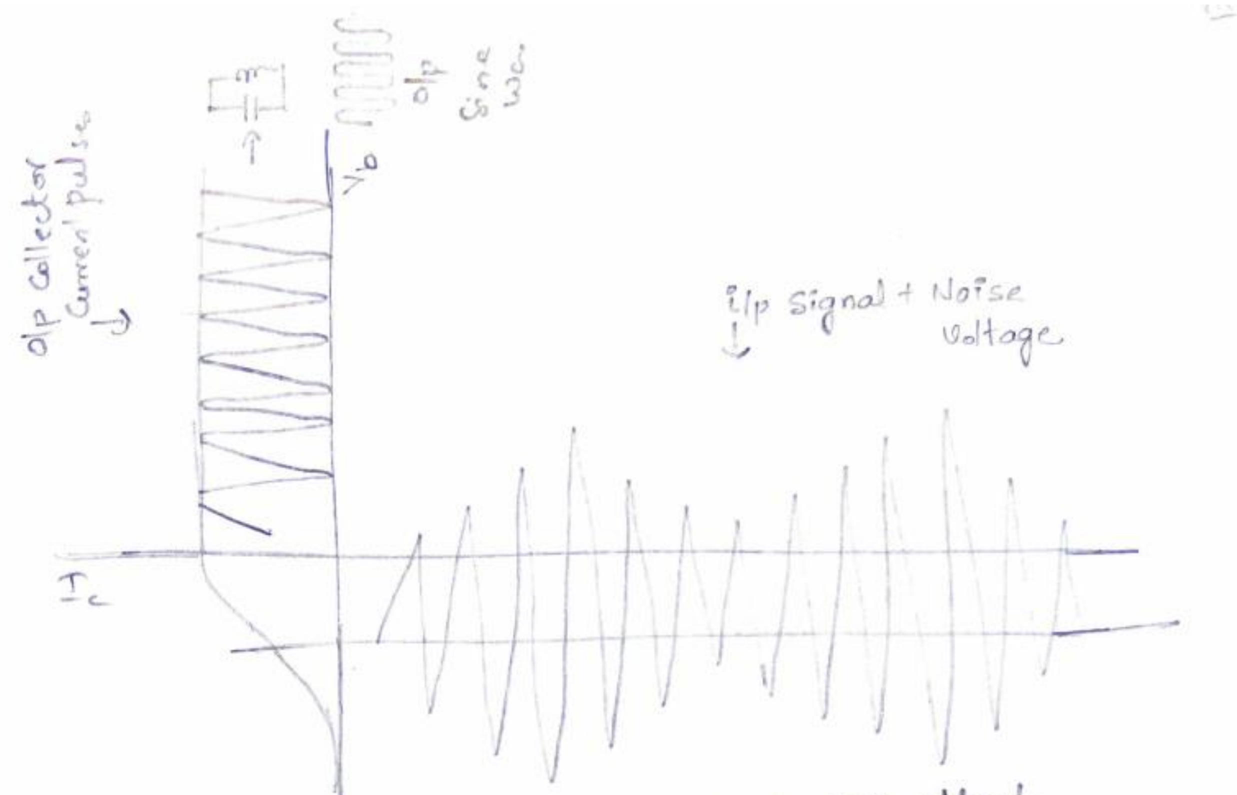
→ Amplitude Limiter:

- FM demodulators are always preceded by an 'amplitude limiter'. This is necessary because any amplitude changes in the signal fed to the FM demodulator is spurious. These spurious signals signals must be removed to avoid the distortion.
- A limiter is a form of clipping device, whose output tends to remain const. despite changes in the input signal. The limiter also provides AGC action. A transistor limiter circuit is shown below.



Transistor limiter ckt

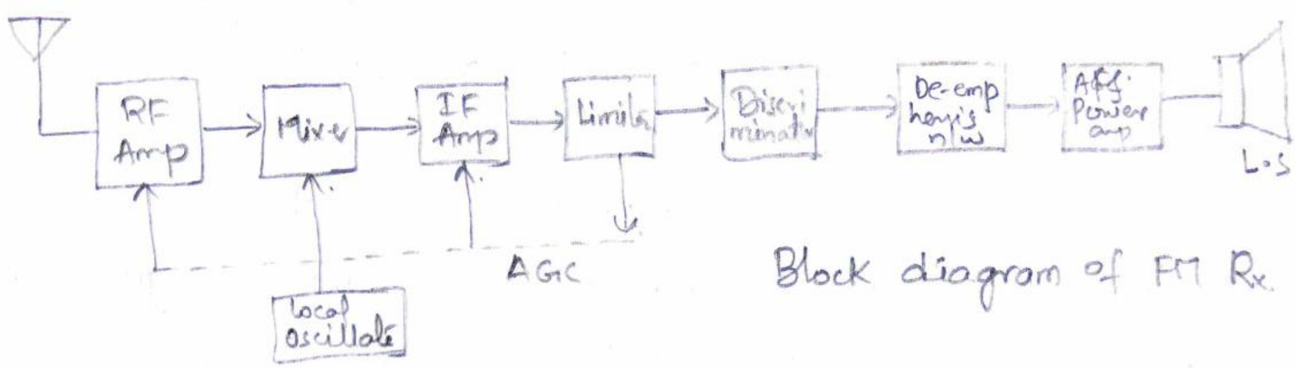
- In this ckt the resistor R_c limits the d.c supply voltage by providing a low d.c. collector voltage, which makes the stage easily overdriven.
- When the i/p is large enough to cause clipping at both extremes of collector current, the critical limiting voltage is attained & limiting action starts.
- The i/p, o/p characteristics of a limiter are shown in the graph. The fig also shows the desire clipping action & the effects of feeding the limited signal to LC tank ckt tuned to the centre freq. of the signal.
- The tank ckt converts the limited signal to a sinusoidal signal by removing all frequencies which are not near the centre freq.



i/p, o/p characteristics & LC tank ckt effect

→ FM RECEIVER:

• FM receivers are of superheterodyne type. The block diagram of a typical FM receiver is shown below.



Block diagram of FM Rx.

Summary:

The classification of Radio Transmitters and Radio Receivers is introduced. The functional blocks of different types of Radio transmitters and Receivers is discussed in detail.

Assignment:

SHORT ANSWER QUESTIONS

1. What is a radio transmitter? Classify them.
2. What is a radio receiver? Classify them.
3. Define Sensitivity, Selectivity and image frequency.

LONG ANSWER QUESTIONS

1. Draw and explain the block diagram of a low level AM transmitter.
2. Draw and explain the block diagram of a high level AM transmitter
3. Draw the block diagram of Super heterodyne receiver and explain the function of each block.

References:

1. Communication Systems by Simon Haykin.
2. Electronic Communication systems by George F Kenedy.
3. Analog and Digital communications by Sanjay Sharma
4. Analog and Digital Communications by P. Chakrabarti.

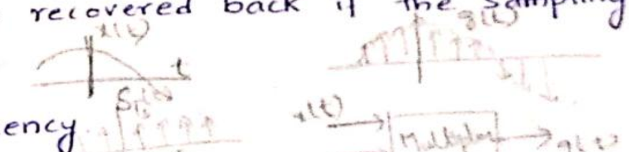
PULSE MODULATION

→ Sampling Theorem:

- There are two types of signals < Continuous Time Signal / Discrete Time Signal.
- Due to advancements in Digital technology the inexpensive, light weight, programmable & easily reproducible discrete time systems are available. Therefore, the processing of DTSs is more flexible & preferable.
- Fore this purpose the CTSs should be converted to DTSs. This problem is solved by a mathematical tool known as "Sampling Theorem".
- With the help of Sampling theorem, a continuous-time signal may be completely represented & recovered from the knowledge of samples taken uniformly.
- The concept of sampling is used to convert CTSs to DTSs. Samples must be taken fast enough in order for high-freq. components to be recognized & adequately represented.

Sampling Theorem Statement

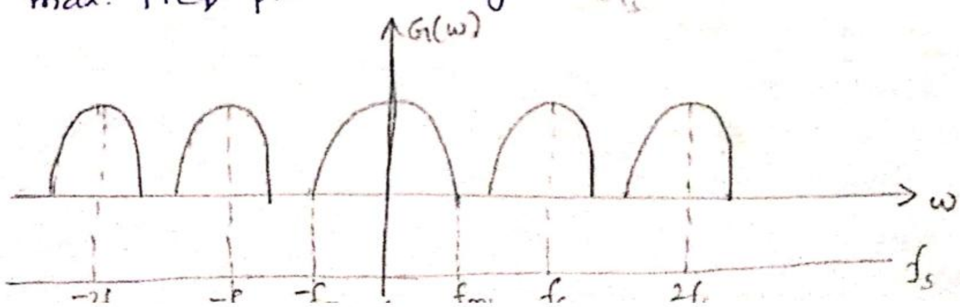
"A continuous-time signal may be completely represented in its samples & recovered back if the sampling frequency $f_s \geq 2f_m$."



$f_s \rightarrow$ Sampling frequency

$f_m \rightarrow$ max. freq. present in signal.

Note:



i) The spectrum of sampled signal extends upto infinity & the ideal bandwidth of sampled signal is infinite. The original (or) desired spectrum will be centered at $\omega=0$ & has BW equal to ω_m.

ii) For $f_s > 2f_m$, the successive cycles of $G(\omega)$ are not overlapping each other. Therefore for this case the original spectrum ~~of~~ can be recovered.

iii) For $f_s = 2f_m$, the successive cycles of $G(\omega)$ will not overlap but are touching each other. Therefore for this case the original spectrum can be recovered using a LPF with sharp cut-off freq. ω_m.

iv) For $f_s < 2f_m$, the successive cycles, of the sampled spectrum will overlap each other & hence in this case, the original spectrum cannot be extracted out of spectrum $G(\omega)$.

Hence, for reconstruction without distortion, we must have $f_s \geqslant 2f_m$.

Nyquist Rate & Nyquist Interval:

• When the sampling rate becomes exactly equal to $2f_m$ sam/sec, then it is called Nyquist rate. It is also called the minimum sampling rate.

$$f_s = 2f_m$$

lly, Maximum sampling interval is called Nyquist interval.

$$T_s = \frac{1}{2f_m} \text{ sec.}$$

Sampling Techniques:

→ There are three types of Sampling techniques

i) Instantaneous Sampling (or) Ideal Sampling (or) Impulse Sampling

ii) Natural Sampling

iii) Flat top Sampling (or) Rectangular Pulse Sampling

→ ANALOG PULSE MODULATION METHODS:

- In pulse modulation methods, some parameter of a carrier (pulse train) is vasied according to the instantaneous value of the modulating signal. There are two types of pulse modulation systems

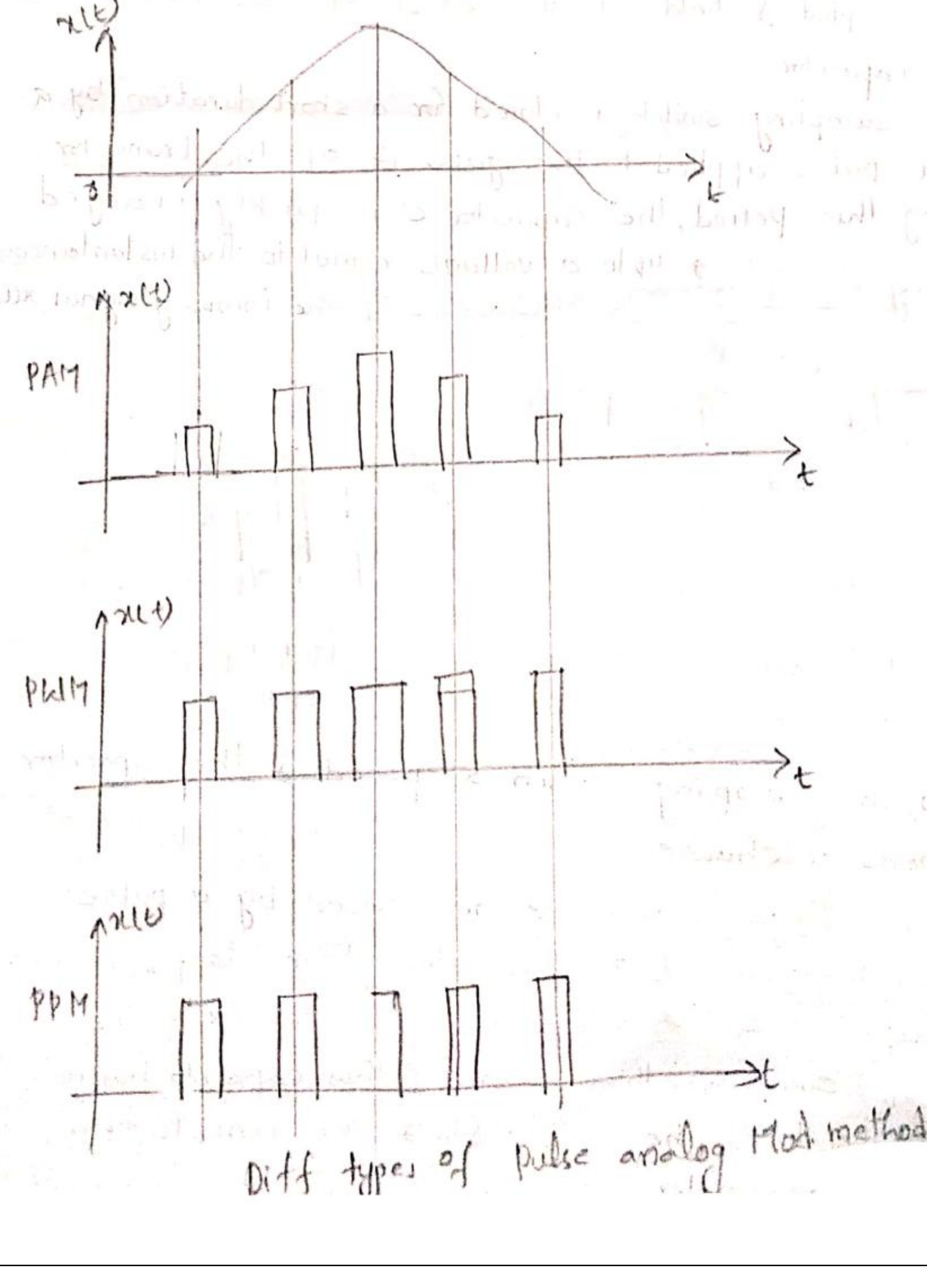
i) Pulse Amplitude Modulation (PAM)

ii) Pulse Time Modulation (PTM)

- There are two types of PTMs

i) Pulse Width modulation (PWM) {PDM}.

ii) Pulse Position modulation (PPM)



Diff types of Pulse analog Mod methods

→ PULSE AMPLITUDE MODULATION (PAM):

- PAM is defined as that type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to instantaneous value of the modulating (or) message signal.

  $m(nT_s) \rightarrow$ nth sample of the msg signal $m(t)$
  $T_s \rightarrow$ Sampling Period, $K_a \rightarrow$ ampli sensi, $g(t) \rightarrow$ Pulse train.

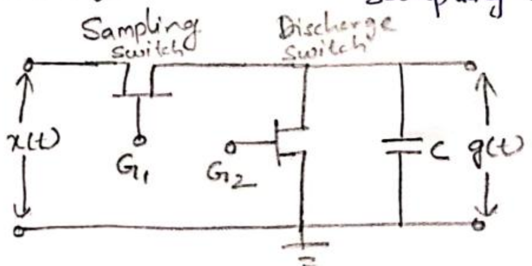
- The pulses in a PAM signal may be of flat top type (or) natural type (or) ideal type. The flat top PAM is most popular & is widely used. PAM wave is defined as

$$s(t) = \sum_{n=-\infty} [1 + K_a m(nT_s)] g(t - nT_s)$$

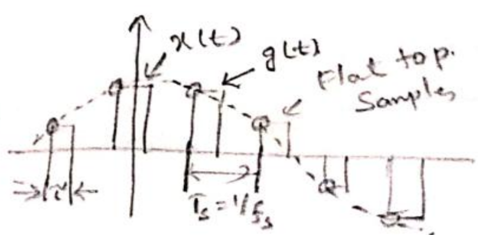
- A sample & hold circuit is used to produce Flat top sampled PAM. as shown below

Working Principle:

- The sampled & hold circuit consists of two FET switches & a capacitor.
- The sampling switch is closed for a short duration by a short pulse applied to the gate G_1 of the transistor. During this period, the capacitor 'C' is quickly charged up to a voltage equal to the instantaneous sampling value of the incoming signal $x(t)$.



Sample & hold ckt generating flat top Sampled PAM.

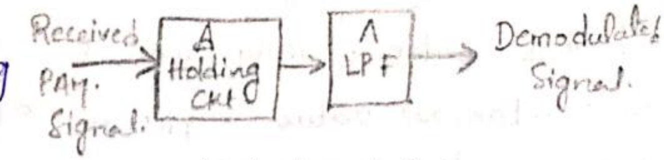


flat top Sampled PAM.

- Now, the sampling switch is opened & the capacitor 'C' holds the charge.
- The discharge switch is then closed by a pulse applied to gate G_2 of the other transistor. Due to this the capacitor 'C' is discharged to zero volts. The discharge switch is then opened & thus capacitor has no voltage.
- Hence, the o/p of the sample & hold ckt consists of a sequence of flat top samples.

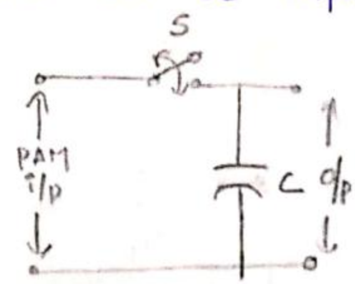
→ Demodulation of PAM Signals:

- PAM demodulation is performed using a holding circuit as shown in the figure.

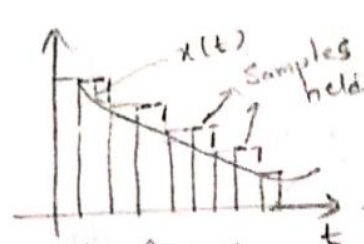


PAM demodulator

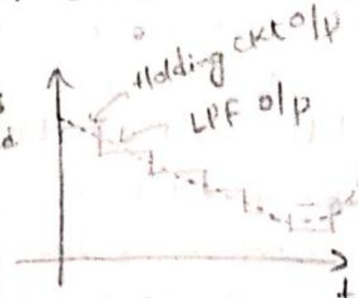
- In this method, the received PAM signal is allowed to pass through a Holding ckt & a LPF. A simple holding ckt can be implemented as shown below.



Zero order holding ckt



o/p of Holding ckt



o/p of LPF.

- The switch 'S' is closed after the arrival of the pulse & it is opened at the end of the pulse. In this way, the capacitor 'C' is charged to the pulse amplitude value & it holds this value during the interval b/w the two pulses.
- Hence the sampled value are held shown in fig 'b'. The holding ckt o/p is smothened by the LPF as shown in fig 'b'. Some kind of distortion is introduced due to the holding ckt.

→ Drawbacks of PAM Signal.

i) B.W required is very large compared to the max freq. of the mod signal.

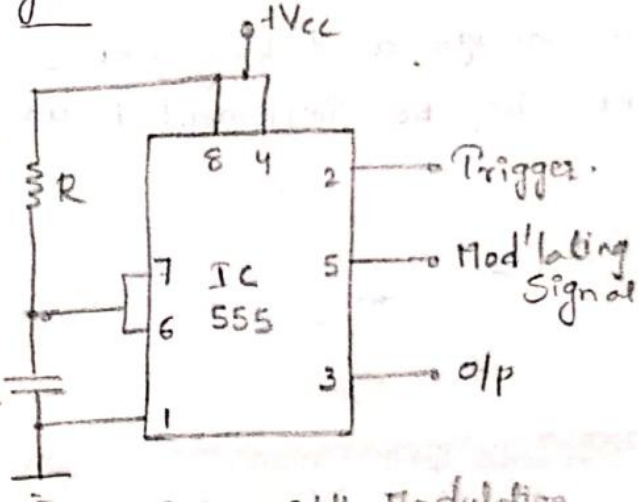
ii) Interference of noise is max. in PAM signal. (∵ pulses vary with mod. signal)

iii) The peak power varies with mod signal.

PULSE WIDTH MODULATION (PWM)

- PWM is that type of modulation in which the width of the pulses (carrier wave) is varied according to the instantaneous value (amplitude) of the message signal.
- The amplitude of the PWM wave is held constant.

→ **Generation of PWM Signal:**

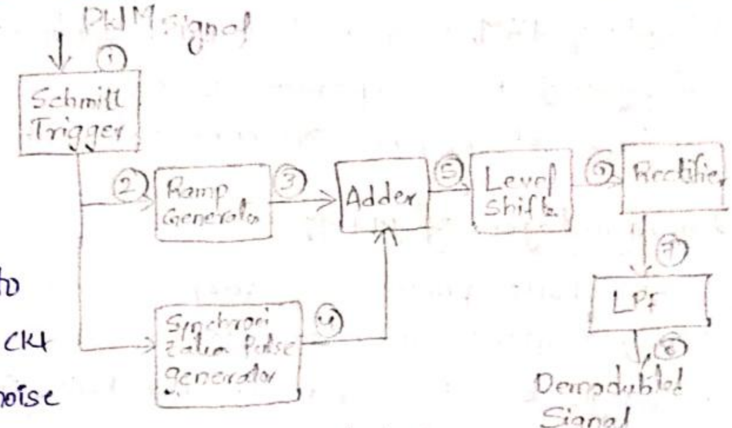


Pulse width Modulation

- The fig. shows a pulse width demodulator circuity based on mono-stable multivibrator.
- The modulating signal is applied at the control voltage input & internally the control voltage is adjusted to 2/3 Vcc.
- The width of the pulse is adjusted by the mod signal applied to the control pin.

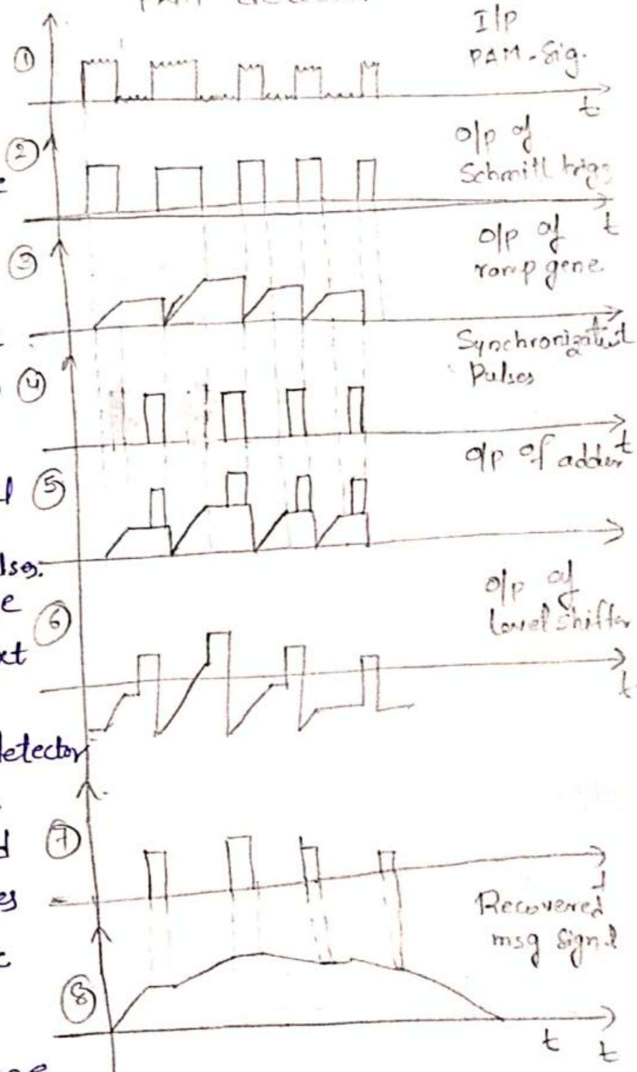
i) Apply a pulse of carrier signal at the trigger i/p
ii) Apply mod. signal at control pin
iii) Now the applied trigger pulse will decide the starting of pulse.
iv) RC combination & the applied msg signal will decide the end of pulse.

Demodulation of PWM signal:



PWM detector

- Fig shows block diagram of PWM detector.
- The received PWM signal is applied to the schmitt trigger ckt which removes the noise in PWM wave.
- The regenerated PWM is then applied to the ramp generator and the synchronization pulse detector.
- The ramp generator produces ramps for the duration of pulses such that the height of ramps are proportional to the widths of PWM pulses. the maximum ramp voltage is retained till the next pulse.
- The synchronous pulse detector produces reference pulses with const amplitude and pulse width. These pulses are delayed by specific amount of delay.



- The delayed reference pulses & the output of ramp generator are added & the resultant signal is given to the level shifter.
- Here the -ve offset shifts the waveform & then the -ve part of the waveform is clipped by rectifier. Finally, the o/p of rectifier is passed through LPF to recover the modulating signal.

Advantages of PWM:

i) Unlike, PAM, noise is less, since amplitude is const. in PWM
ii) Signal & noise separation is easy.
iii) Does not require synchronization b/w Tx & R_x

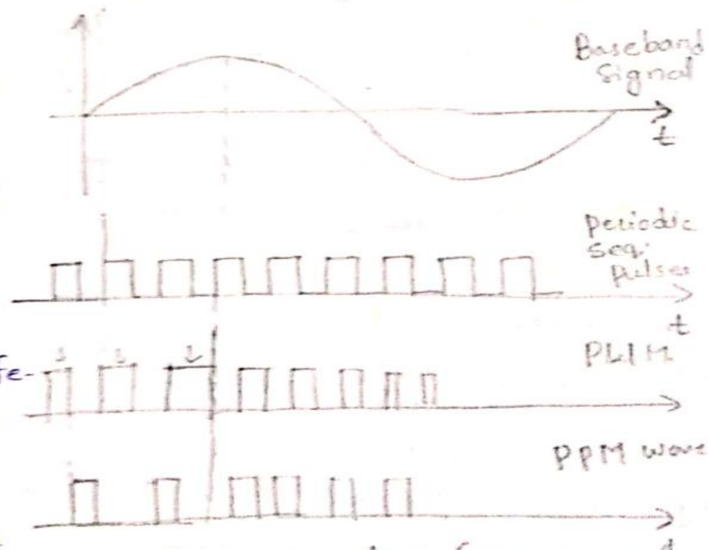
Disadvantages of PWM:

i) In PWM pulses are varying in width & therefore their power contents are variable.
ii) Large B.W is req'd for the PWM compared to PAM.

→ PULSE POSITION MODULATION:

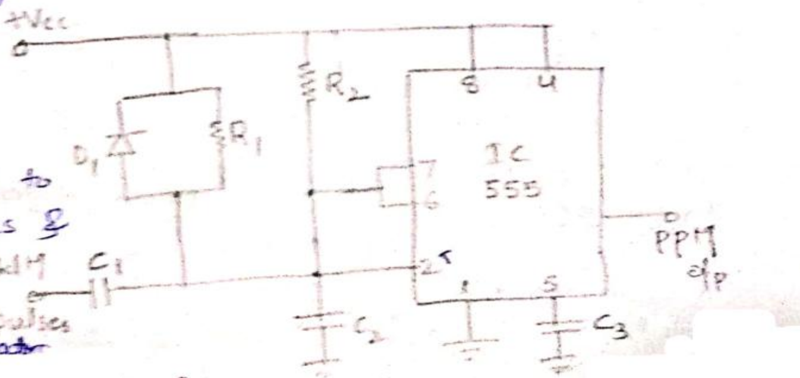
- In this system, the amplitude & width of the pulses are kept const, while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the mod. signal.
- The transmitter has to send synchronizing pulses to keep the transmitter & receiver in synchronism.
- Pulse position mod. is obtained from pulse width mod. as shown in the fig below. Each trailing edge of the PWM pulse is a starting point of the pulse in the PPM. Each pulse in PPM is proportional to the instantaneous amplitude of the sampled mod. signal

⇒ Generation of PPM:



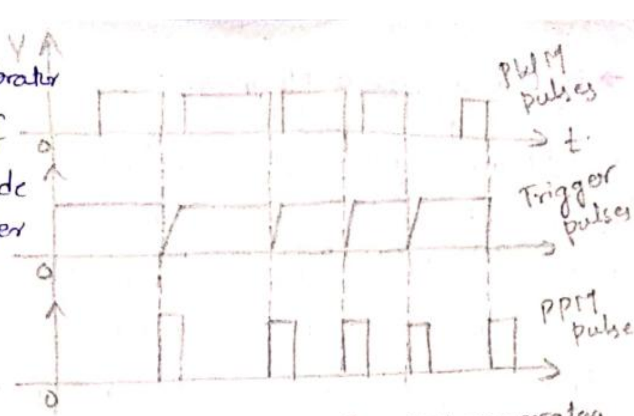
PPM generation from PWM.

- The figure shows a pulse position modulator consisting of a differentiator & a monostable multivibrator.
- The i/p to the differentiator is a PWM waveform.
- The differentiator generates +ve & −ve spikes corresponding to leading & trailing edge of the PWM wave form.
- Diode D_1 is used to bypass the +ve spikes & the −ve spikes are used to trigger monostable multivibrator



(3)

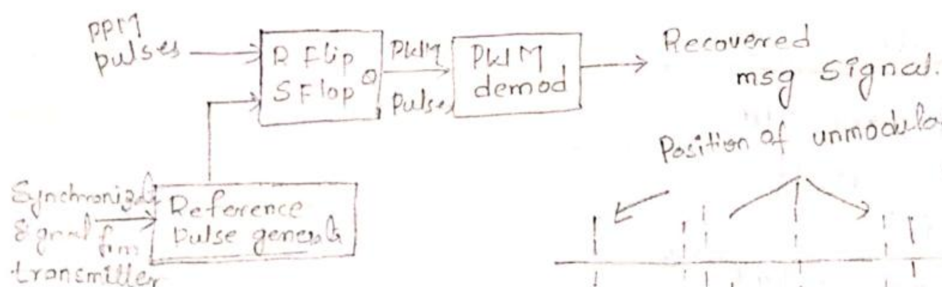
· The monostable multivibrator generates the pulses of same width & amplitude with reference to trigger i/p to produce PPM waveform as shown in fig(b)



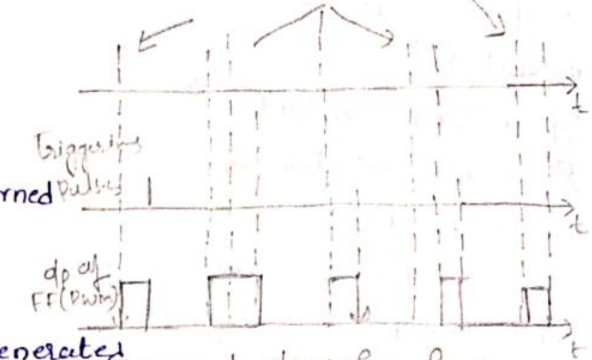
(b) Waveforms of PPM generator

→ Demodulation of PPM:

· For demodulating PPM waves, the received pulses that vary in position are converted to pulses that vary in length.
· One method to demodulate the PPM waves is as shown below.



PPM demodulator



Demod. waveform for PPM

· The flipflop ckt is set (or) turned 'ON' (giving high o/p) when the reference pulse arrives.

· The reference pulse is generated by Ref. pulse generator of the receiver with the synchronization signal from the transmitter.

· The flip-flop ckt is reset (or) turned 'OFF' (giving low o/p) at the leading edge of the position modulated pulse. This is repeated & PWM wave is obtained at the o/p of FF
· The PWM pulses are then demodulated by PWM demodulator to get original mod. signal.

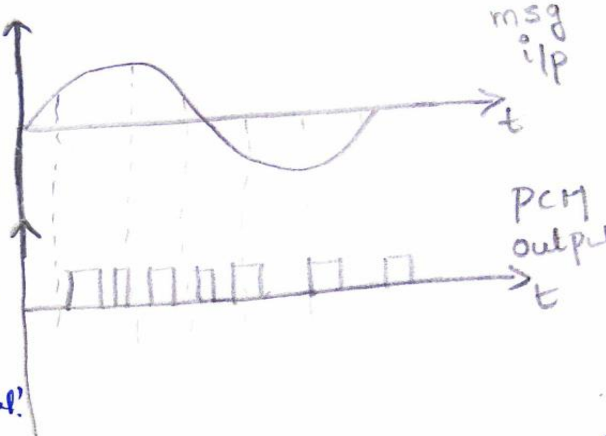
→ Advantages:
1) Less noise compared to PAM, PWM
2) Transmission power of each pulse is same because of same pulse width

Disadvantages:
1) Synchronization b/w transmitter & receiver is req'd.
2) large B.W is req'd compared to PAM.

Pulse Code Modulation (PCM):

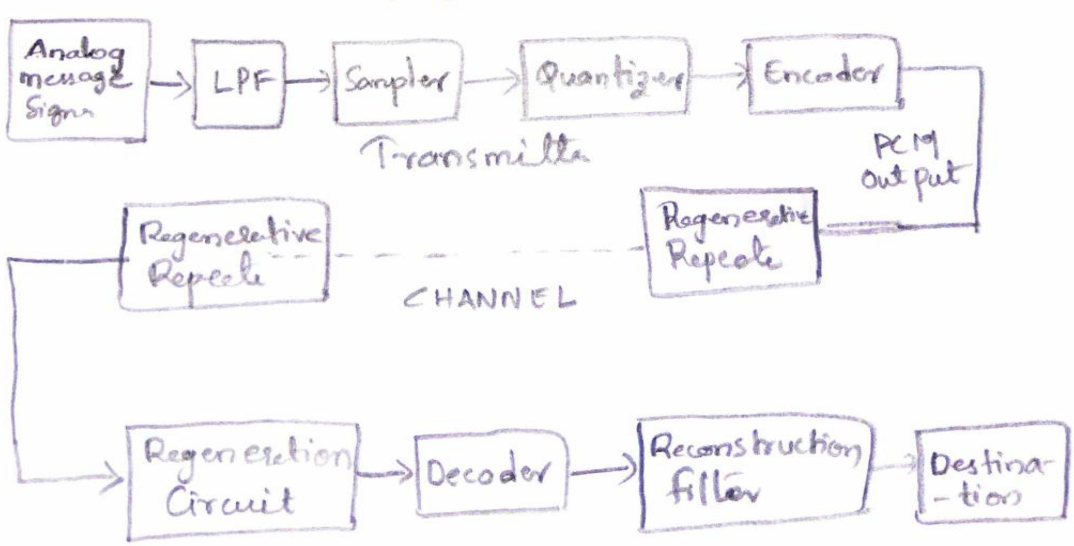
A signal is pulse coded modulated to convert its analog information into a binary sequence i.e., 1s & 0s. The output of PCM resembles a binary sequence as shown below

In PCM, the message signal is represented by a sequence of coded pulses. It produces a series of numbers (or) digits. & hence the process is called as 'Digital'.



Basic Elements of PCM:

- The transmitter section of pulse code modulator consists of Sampling, Quantizing and encoding. The LPF prior to Sampling prevents



Receiver

Fig: PCM System

Low Pass Filter:

This filter is used to eliminate the high frequency components present in the input analog signal which greater than highest frequency in the message signal to avoid aliasing effect

Sampler:

The sampler will perform the process of sampling to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal at the receiver side. The sampling rate must be greater than twice the highest frequency component of the message signal.

Quantizer:

The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is the process of representing the sampled values of the amplitude by a finite set of levels.

Encoder:

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The LPF, sampler & quantizer will act as an analog to digital converter. Encoding minimizes the bandwidth used.

ring

Regenerative Repeater:

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

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 & samples it, & then transmits it in an analog form. this whole process is repeated in a reverse pattern to obtain the original signal.

A q-level quantizer compares the discrete-time i/p $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion error. This error is called quantization error. Thus, the o/p of a quantizer is a digital level called $x_q(nT_s)$

Classification of Quantization Process:

Basically, quantization process may be classified as follows

Quantization
- Uniform Quantization
  - Midtread Type
  - Midrise type.
- Non-Uniform Quantization.

A uniform quantizer is that type of quantizer in which the 'step-size' varies according to the i/p range.

A Non-uniform quantizer is that type of quantizer in which the stepsize varies according to the i/p range.

Uniform Quantizers:

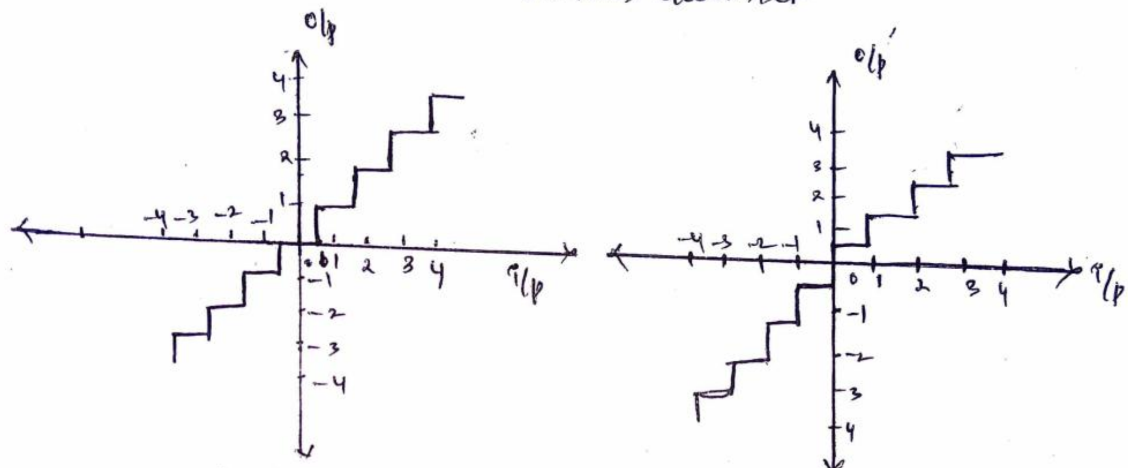
There are two types of uniform quantizer as under:

(i) Midtread quantizer
(ii) Midrise quantizer.

In a uniform quantizer, the representation levels are uniformly spaced: The quantizer characteristic can also be midtread (or) midrise type.

Below figure(a) shows the input - output characteristic of a uniform quantizer of the midtread type, which is so called because the origin lies in the middle of a tread of the staircase like graph. Below figure (b) shows the corresponding input-output characteristic of a uniform quantizer of the midrise type, in which the origin lies in the middle of a rising part of the staircase like graph.

Uniform Quantizer



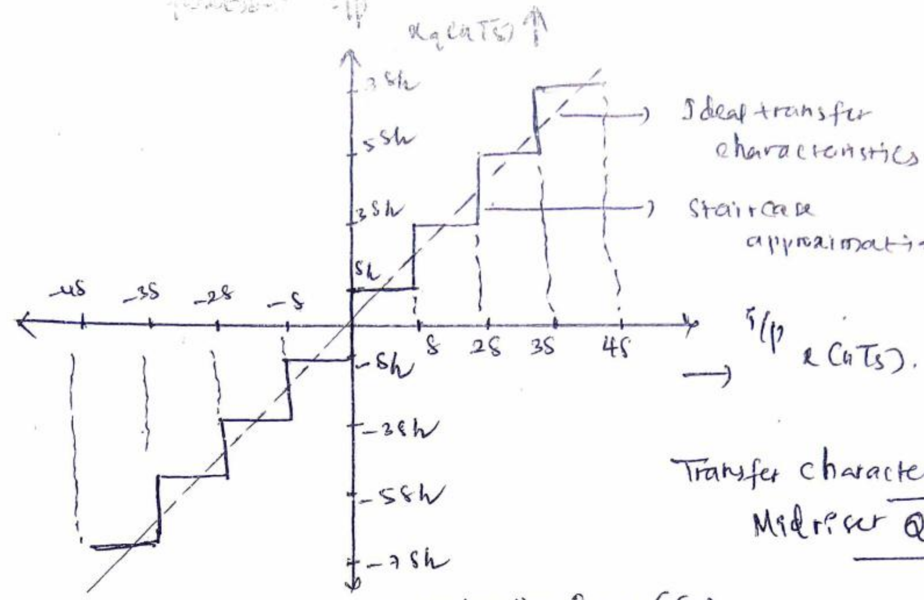
Midtread   Mid rise

Working Principle of a Quantizers

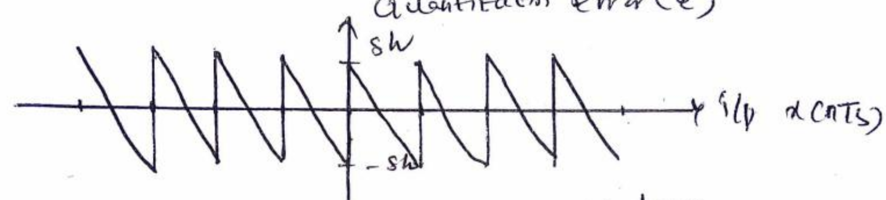
In this section, let us see how uniform quantization takes place. For this purpose, we shall consider uniform quantizer of midriser type.

Let us assume that the i/p to the quantizer $x(nTs)$ varies from 4Δ to $+4\Delta$. This means that the peak to peak value of $x(nTs)$ will be ~~taken~~ between -4Δ to $+4\Delta$. Here, 'Δ' is the step size.

Thus, i/p $x(nTs)$ can take any value between -4Δ to $+4\Delta$. Now, the fixed digital levels are available at $\pm\Delta/2$, $\pm 3/2\Delta$, $\pm 5/2\Delta$ and $\pm 7/2\Delta$. These levels are available at quantizer because of its characteristics.



Transfer characteristics of Midriser Quantizer



Here, according to figure, we have

If $x(nT_s) = 4\Delta$, then $x_q(nT_s) = \frac{7}{2}\Delta$

If $x(nT_s) = -4\Delta$, then $x_q(nT_s) = -\frac{7}{2}\Delta$.

Thus, it may be noted that, maximum quantization error would be $\pm \Delta/2$.

Here, the quantization error expressed as

$\epsilon = x_q(nT_s) - x(nT_s)$. $\epsilon \rightarrow$ quantization error.

At $x(nT_s) = 0$; then $x_q(nT_s)$ = either $-\Delta/2$ or $+\Delta/2$.

At $x(nT_s) = \Delta$; then $x_q(nT_s) = 3\delta/2$; $\epsilon = 3\delta/2 - \delta = \delta/$

At $x(nT_s) = -\Delta$; then $x_q(nT_s) = -3\delta/2$; $\epsilon = -3\delta/2 + \delta = -\delta$

Similarly,

$0 \le x(nT_s) < \delta$; $x_q(nT_s) = \delta/2$

$0 \le x(nT_s) < -\delta$; $x_q(nT_s) = -\delta/2$.

$\delta \le x(nT_s) \le 2\delta$; $x_q(nT_s) = 3/2\,\delta$

$-\delta \le x(nT_s) < -2\delta$; $x_q(nT_s) = -3/2\,\delta$.

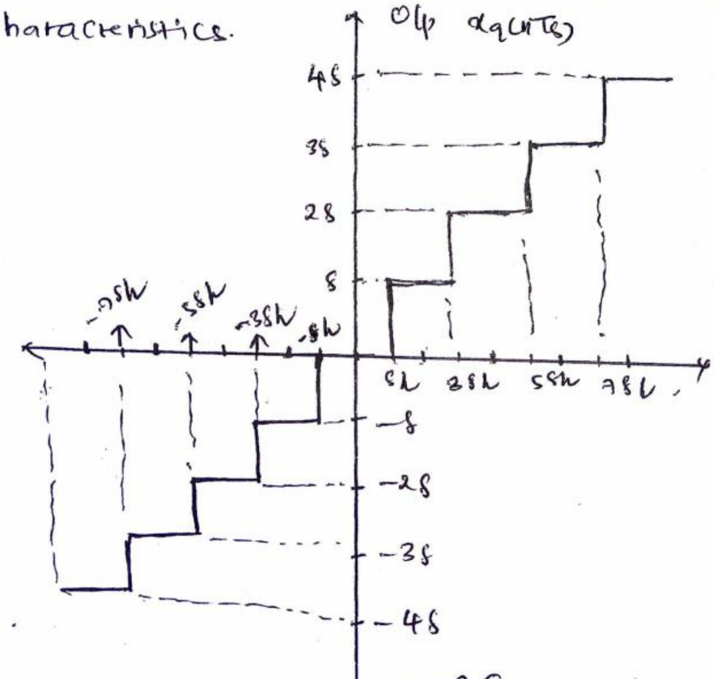
This means that the maximum quantization error will be $\pm \delta/2$.

In other words, the maximum quantization error is given by

$$\boxed{\varepsilon_{max} = |\delta/2|.}$$

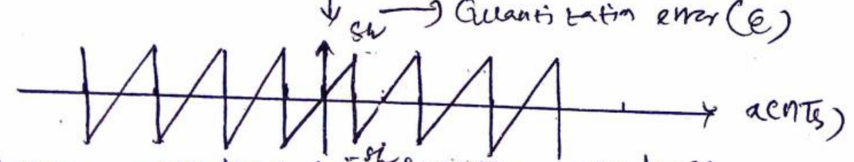
Midtread:

In this quantizer i/p signal varies from $-7\delta/2$ to $+7\delta/2$. This means that the peak to peak value of $x(nT_s)$ will be between ~~#1~~ $-7\delta/2$ to $+7\delta/2$. Here δ is the stepsize. Fixed digital levels are available at $\pm\Delta$, $\pm 2\Delta$, $+3\Delta$. These levels are available at quantizer because of its characteristics.



Midtread Quantizer Transfer characteristics.

$\varepsilon_{max} = |\delta/2|$.



Hence, according to the figure, we have

If $x(nT_s) = 7\delta/2$, then $x_q(nT_s) = 4\delta$

If $x(nT_s) = -7\delta/2$, then $x_q(nT_s) = -4\delta$

Maximum quantization error would be $\pm\delta/2$

Thats why it is called as midtread quantizer.

$$\varepsilon = x_q(nT_s) - x(nT_s).$$

At $x(nT_s) = 0$; $x_q(nT_s) = 0$. ; $\varepsilon = 0$.

At $x(nT_s) = \delta/2$; $x_q(nT_s) = \delta$; $\varepsilon = \delta - \delta/2 = \delta/2$

At $x(nT_s) = -\delta/2$; $x_q(nT_s) = -\delta$; $\varepsilon = -\delta + \delta/2 = -\delta/2$.

Similarly,

$-\delta/2 < x(nT_s) < +\delta/2$; $x_q(nT_s) = 0$.

$\delta/2 \leq x(nT_s) < 3\delta/2$; $x_q(nT_s) = \delta$.

$-\delta/2 \leq x(nT_s) < -3\delta/2$; $x_q(nT_s) = -\delta$.

This means that the maximum quantization error will be $\pm \delta/2$.

$$\therefore \underline{\varepsilon_{max} = |\delta/2|}.$$

Using midtread (or) midriser quantizer, we can conclude that maximum quantization error is

$$\varepsilon_{max} = |\delta/2| \quad \text{for uniform quantization}$$

Quantization Noise in PCM: *

In this section, we shall derive an expression for quantization noise (i.e; error). in a PCM system for uniform quantization. Because of quantization, inherent errors are introduced in the signal. This errors is called quantization error.

$$\varepsilon = x_q(nT_s) - x(nT_s).$$

Let us assume that the i/p $x(nT_s)$ to a linear or uniform quantizer has continuous amplitude in the range $-x_{max}$ to $+x_{max}$.

From the quantization [illegible]

observed that the total excursion of i/p x(nTs) is mapped into 'q' levels on vertical axis. This means that when input is 4δ, the output is $7/2\,\delta$ and when i/p is -4δ, output is $-7/2\,\delta$. Thus, $+x_{max}$ represents $7/2\,\delta$ and $-x_{max}$ represents $-7/2\,\delta$. Therefore, the total amplitude range becomes,

$$\text{Total Amplitude Range} = x_{max} - (-x_{max}) = 2x_{max}.$$

Now, if this total amplitude range is divided into 'q' levels of quantizer, then the step size 'δ' will be,

$$\text{'Step size' } \delta = \frac{x_{max} - (-x_{max})}{q} = \frac{2x_{max}}{q}.$$

Now, if signal $x(t)$ is normalized into minimum & maximum values equal to '1', then we have

$$x_{max} = 1 \quad , \quad -x_{max} = -1.$$

Therefore, step size would be

$$\delta = 2/q. \quad \text{(For normalized signal)}.$$

Now, if step size 'δ' is considered as sufficiently small, then it may be assumed that the quantization error 'ε' will be an uniformly distributed random variable.

We know that maximum quantization error is given as

$$\delta = \frac{2}{q}. \; ; \quad \varepsilon_{max} = |\delta/2|.$$

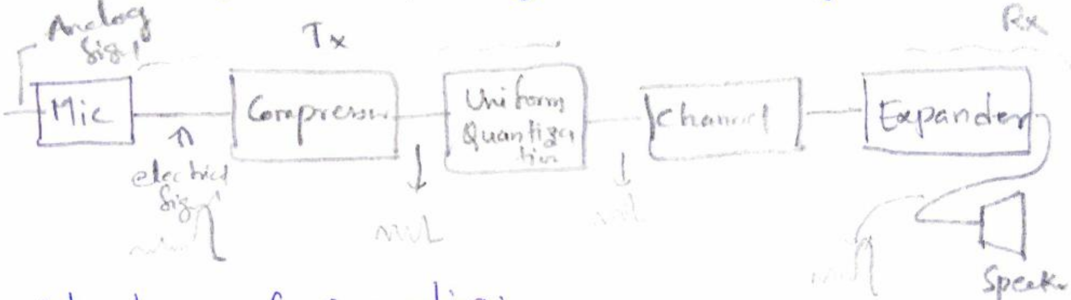
$$\text{i.e.; } -\delta/2 \le \varepsilon_{max} \le \delta/2.$$

Hence, over the interval $(-\delta/2, \delta/2)$ quantization error may be assumed as an uniformly distributed random variable.

Companding:

The word 'companding' is a combination of Compressing & expanding. This is a non-linear technique used in PCM which compresses the data at the transmitter & expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

Companding = Compressing + Expanding.



Advantages of Companding:

i) Dynamic range is reduced.

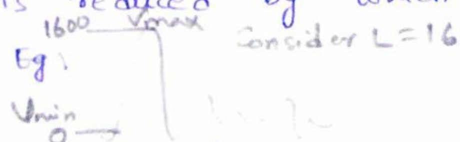
At the transmitter side the signal having maximum amplitudes will be attenuated and therefore reducing the dynamic rang.

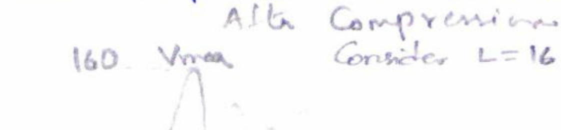
Eg

ii) Reduction in step size.

By compressing the signal the dynamic range is reduced by which the step size is also reduced

Eg:





$$\text{Step Size } (\Delta) = \frac{V_{max} - V_{min}}{L} = \frac{1600-0}{16} = 100$$

$$\text{Step Size } (\Delta) = \frac{V_{max} - V_{min}}{L} = \frac{160-0}{16} = 10.$$

Types of Companding:

There are two types of Companding techiques: They are:

A-law Companding Technique

- Uniform quantization is achieved at A = 1, where the characteristic curve is linear & no compression is done.

- A-law has mid-rise at the origin. Hence it contains a non-zero value.

- A law companding is used for PCM telephone systems.

- Used in Europe, Asia in PCM telephone system.

μ-law Companding technique.

- Uniform quantization is achieved at μ = 0, where the characteristic curve is linear & no compression is done.

- μ-law has mid-tread at the origin. Hence it contains a zero value.

- μ law companding is used for speech & music signals.

- Used in Japan, US in PCM telephone system.

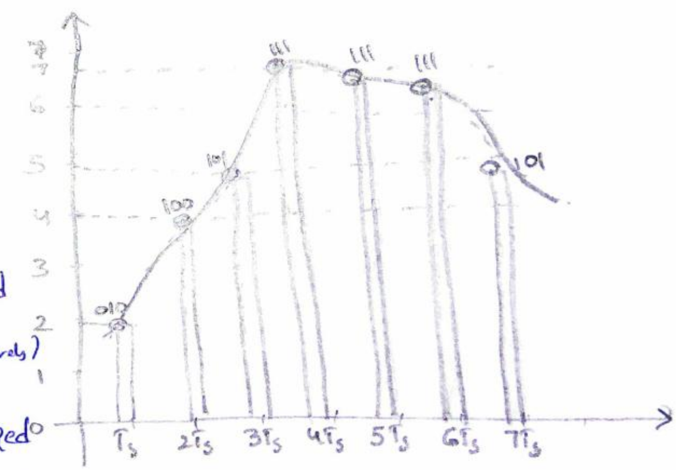
Differential Pulse Code Modulation (DPCM):

The PCM system had redundant information in it. The samples of a signal are highly connected with each other because the signal does not change fast. i.e, its present sample value & the next sample does not differ by a large amount.

Adjacent samples of the signal carry the same information with a little difference. When these samples are encoded by a standard PCM system the resulting encoded signal contains some redundant information.

Fig shows a continuous time signal $x(t)$.

This is sampled at intervals T_s, $2T_s$, $3T_s$ .... nT_s. The samples are encoded using 3-bit PCM ($2^3 = 8$ levels). The sample is quantized to the nearest digital level shown by small circles.



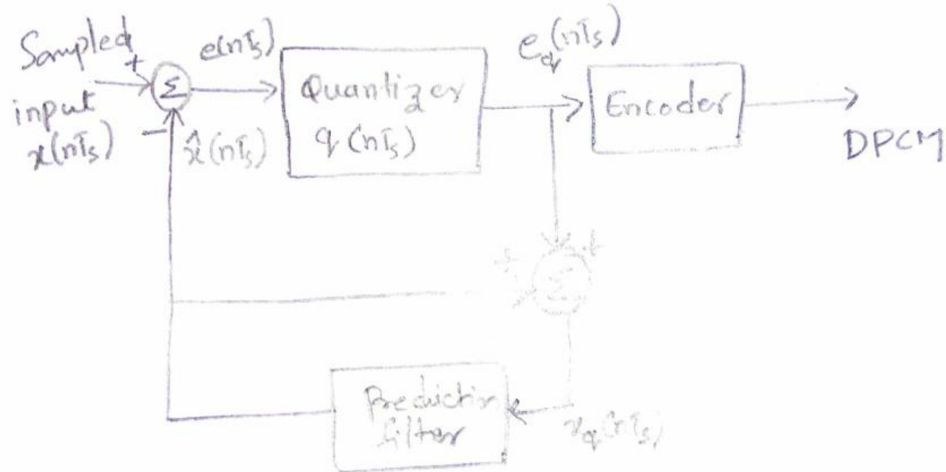
The samples at intervals $4T_s$, $5T_s$ & $6T_s$ are encoded to the same value 111. This information can be carried by only one sample, hence it is redundant. The samples at $2T_s$ & $3T_s$ differ only by one bit (100, 101), the first two bits are redundant.

Principle of DPCM:

If this redundancy is reduced, then the overall bit rate will decrease & the no. of bits required to transmit one sample will also be reduced.

- DPCM Transmitter:

The DPCM works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value.



DPCM Transmitter

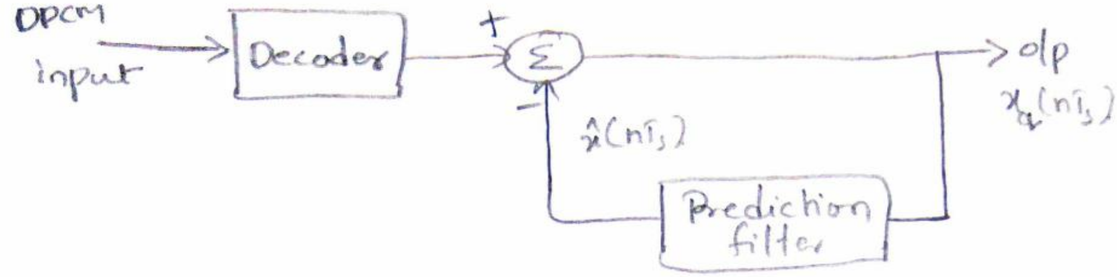
Above figure shows a DPCM transmitter system. The sampled signal is denoted by $x(nT_s)$ and the predicted sample signal is denoted by $\hat{x}(nT_s)$.

The Comparator finds the difference between the actual sample $x(nT_s)$ and the predicted signal $\hat{x}(nT_s)$ which is called error denoted as $e(nT_s)$

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

ng

DPCM Receiver;



DPCM Receiver

The decoder first reconstructs the quantized error signal from the incoming binary signal. The prediction filter o/p & the quantized error signals are summed up to give the quantized version of the original signal

Thus the signal at the receiver differs from the actual signal by quantization error $q(nT_s)$ which is introduced permanently in the reconstructed signal.

$$e_q(nT_s) + \hat{x}(nT_s) = x_q(nT_s)$$

Delta Modulation: DM:

This modulation transmits only one bit per sample, i.e., the present sample value is comparred with the previous sample. the input signal $x(t)$ is approximated to step signal by delta modulation, the step size is fixed

The difference between input $x(t)$ and straircase approximated signal is confined to two levels $+\delta$ & $-\delta$.

If the difference is positive, then the approximated signal is increased by one step i.e., 'δ'. If the difference is negative, the approximated signal is reduced by 'δ'

When the step is reduced, '0' is transmitted & if step size is increased, '1' is transmitted. i.e., for each sample, only one binary bit is transmitted

• The principle of delta modulation can be explained using following eqn.

The error $e(nT_s)$ is given by

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

$x(nT_s) \rightarrow$ Sampled Signal $x(t)$

$\hat{x}(nT_s) \Rightarrow$ last Sample approximation.

let us assume a quantity $b(nT_s)$ as

$$b(nT_s) = \delta \, \text{sgn}\left[e(nT_s)\right]$$ Sign of $e(nT_s)$.

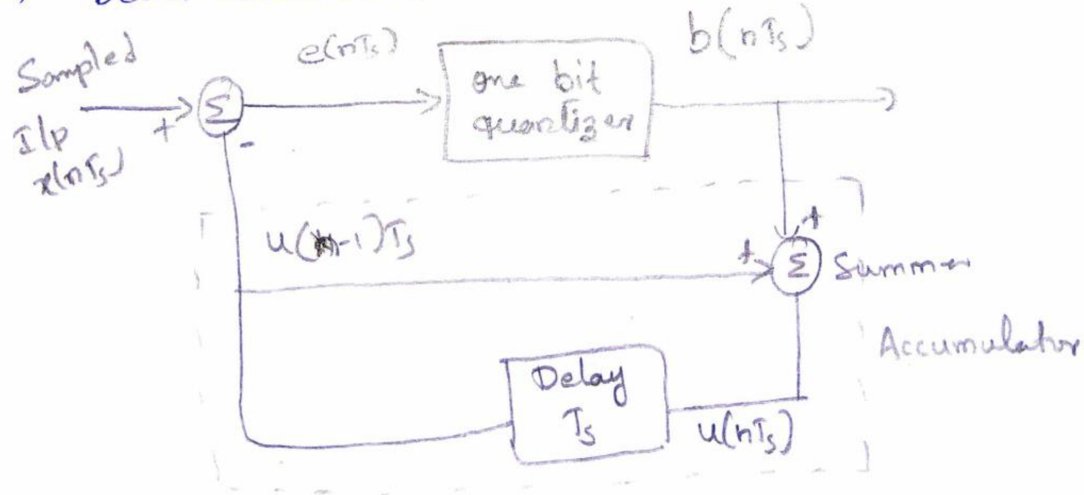
Depending on the Sign of error $e(nT_s)$ the Sign of Step Size 'δ' will be considered

$$b(nT_s) = +\delta \quad \text{if } x(nT_s) > \hat{x}(nT_s), \text{ '1' is transmitted}$$
$$= -\delta \quad \text{if } x(nT_s) < \hat{x}(nT_s) \text{ '0' is transmitted.}$$

$T_s \Rightarrow$ Sampling interval

Delta Modulation Transmitter:

The following figure shows transmitter part of Delta modulation



· The Summer in the accumulator adds quantizer output ($\pm\delta$) with the previous Sample approximation. This gives the present Sample approximation as

$$u(nT_s) = u(nT_s - T_s) + (\pm \delta)$$
$$= u[(n-1)T_s] + b(nT_s)$$

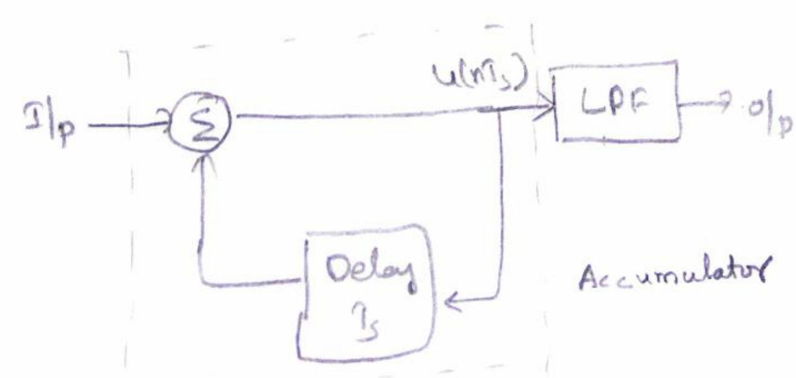
The previous Sample approx. $u[(n-1)T_s]$ is restored by delaying one Sample period T_s. The Sampled input Signal $x(nT_s)$ and straircase approx. Signal $\hat{x}(nT_s)$ are Subtracted to get error Signal $e(nT_s)$.

Thus, depending on the Sign of $e(nT_s)$, one bit quantizer generates an output of '$+\delta$' (or) '$-\delta$'. If the step Size is '$+\delta$', then binary '1' is transmitted and if it is '$-\delta$', then binary '0' is transmitted

The deta modulation transmits only one bit per Sample, indicating whether the Signal level is increasing (or) decreasing, but it needs a higher Sampling rate than PCM for equivalent results.

Delta Modulation Receiver:

The following figure shows the arrangement of receiver



At the receiver end, accumulator and Low pass filter are used.

The accumulator generates the staircase approximated signal output and is delayed by one sampling period 'Ts'. It is then added to the input signal.

If the input is binary '1' then one step 'δ' is added to the previous output. If the input is binary '0' then one step 'δ' is subtracted from the delayed signal.

The LPF has the cut off frequency equal to the highest frequency of x(t). The LPF smoothens the staircase signal to reconstruct original message signal x(t)

Advantages of Delta Modulation:

i) Since, the delta modulation transmits only one-bit for one sample, therefore the signaling rate and transmission channel bandwidth is quit less compared to PCM

ii) The tranmitter and receiver implementation is very much simple for demodulation.

Drawbacks of Delta Modulation:

① Slope overload distortion

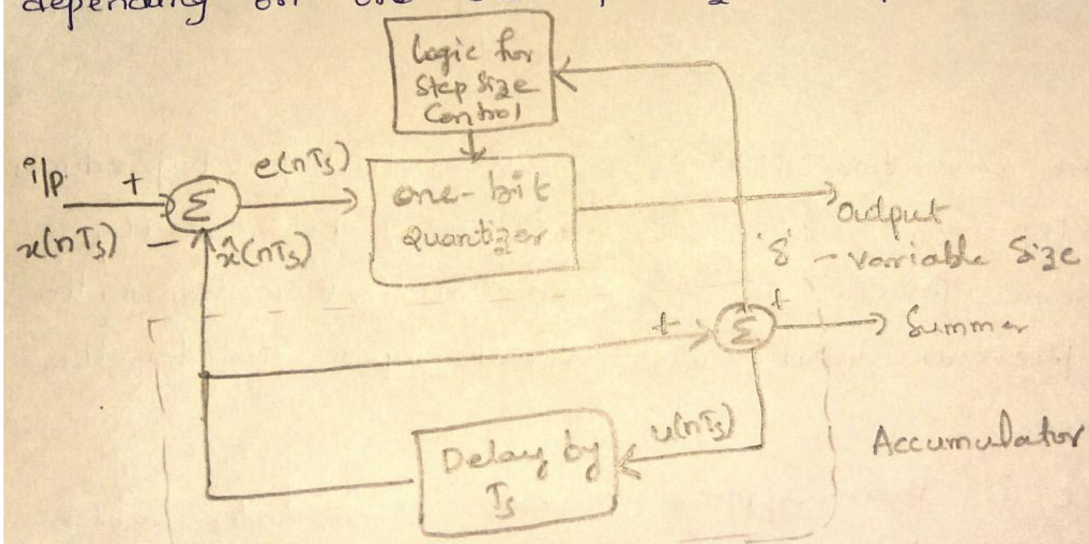
② Granular Noise.

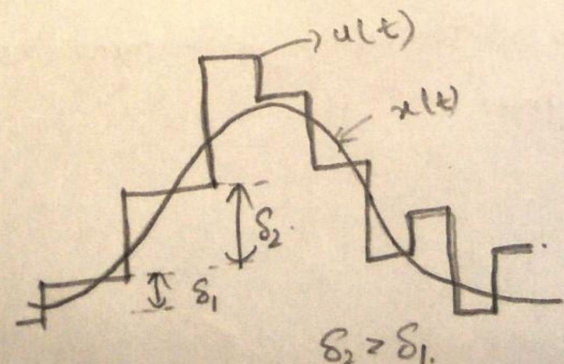
Adaptive Delta Modulation (ADM):

To overcome the quantization errors due to slope overload and granular noise, the step size is made adaptive to variations in the input signal x(t). This method is known as Adaptive Delta Modulation.

Generation of ADM:

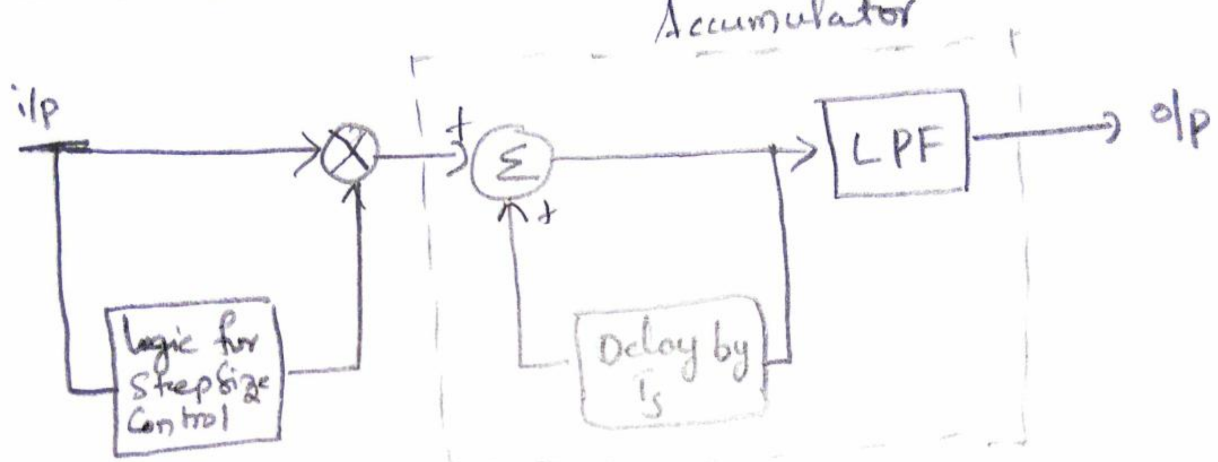
The generation process of ADM is same as DM except the inclusion of logic control for step size is added here. The step size here increases or decreases according to a specified rule depending on one-bit quantizer output.





As an example, if one bit quantizer output is high (i.e, 1) then the step size may be doubled for next sample. If one bit quantizer output is low (i.e., 0), then step size may be reduced by one step. Above fig show the staircase waveforms of ADM and sequence of bits to be transmitted.

Degeneration of ADM:



There are two parts here. The first part produces the step size from each incoming bit. exactly the same process is followed as that in transmitter. the previous input and present input decides the step size.

It is then applied to an accumulator which builds up staircase waveform. The LPF then smoothers out the staircase waveform to reconstruct the original signal

Summary:

Students are introduced the concept of Analog Pulse Modulations and Digital Pulse Modulations? The generation and detection methods of Analog Pulse modulations and Digital Pulse modulations are discussed in detail.

Assignment:

1. What are the types of Pulse modulations?
2. Explain the generation of PAM signal
3. Discuss the generation and detection of PWM signal.
4. With neat diagram explain the generation of PPM signal.
5. Define Quantization and types of Quantization.
6. Discuss about generation and degeneration of Delta Modulation.

References:

1. Communication Systems by Simon Haykin.
2. Electronic Communication systems by George F Kenedy.
3. Analog and Digital communications by Sanjay Sharma
4. Analog and Digital Communications by P. Chakrabarti.

Introduction:

In baseband pulse transmission, the input data is represented in the form of discrete PAM signals. These signals are transmitted over a low pass channel. The baseband signals have an adequately large power at low frequencies. So, they can be transmitted over a pair of wires or coaxial cables. But, it is not possible to transmit the baseband signal over radio links or satellites because impracticably large antennas would be required to be used. Hence, the spectrum of the message signal has to be shifted to higher frequencies. This is achieved by using the baseband digital signal to modulate a sinusoidal carrier. This is called as digital carrier modulation or digital passband communication. These signals are transmitted over a bandpass channel. The examples of bandpass channels are microwave radio links or a satellite channel.

There are three basic signaling schemes used in passband data transmission. These are as under.

(1) Amplitude shift keying (ASK)
(2) Phase shift keying (PSK)
(3) Frequency shift keying (FSK)

These are special cases of amplitude modulation (AM), phase modulation (PM), and frequency modulation (FM), respectively

Hierarchy of Digital Modulation Techniques:

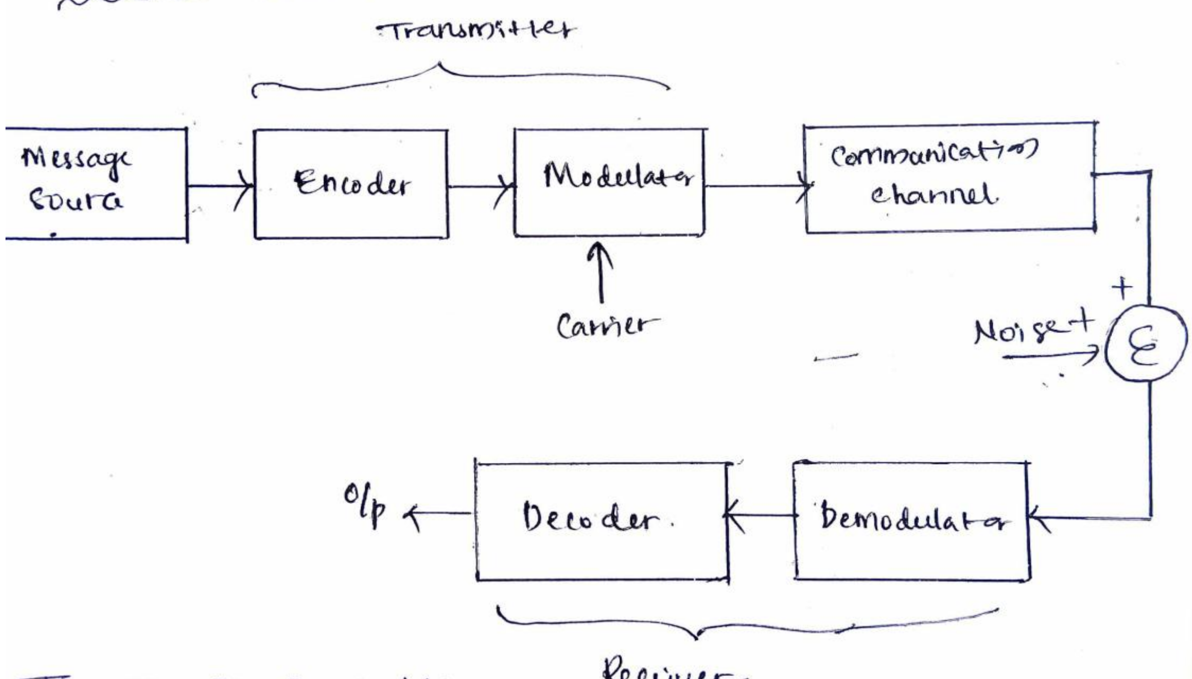
The digital modulation techniques may be classified into two categories as under:

(i) coherent techniques
(ii) Non-Coherent techniques.

In coherent digital modulation techniques, we have to use a phase synchronized carrier to be generated at the receiver to recover the information signal. The frequency and phase of this carrier produced at the receiver should be synchronized with that at the transmitter.

In the non-coherent techniques, no phase synchronized local carrier is needed at the receiver. These techniques are less complex. However, the performance is inferior to that of coherent techniques.

Pass Band Transmission Model:



The msg source emits the symbols at the rate of 'T' seconds. These symbols are encoded into binary signals by encoder. This encoded stream is modulated by using high frequency carrier wave. Here we have 3 different types of modulation techniques. (ASK, FSK, PSK). The modulated signal is ~~then~~ demodulated and decoded at the receiver. It is the basic block diagram of passband transmission model. This chapter will focus on various modulation techniques used in Digital modulation

Amplitude shift keying (or) ON-OFF keying:

(2)

ASK: It is the process of changing carrier amplitude in accordance with the i/p applied binary message signal. It is the simplest digital modulation technique. In this method, there is only one unit energy carrier and it is switched ON (or) OFF depending upon the binary input sequence.

Let carrier for ASK process is

$$c(t) = A\cos 2\pi f_c t$$

$A \rightarrow$ Amplitude of the carrier

$f_c \rightarrow$ frequency " "

Consider, power formula

$$P = \frac{V_{rms}^2}{R} = \frac{(A/\sqrt{2})^2}{1\Omega} = \frac{A^2}{2}. \quad (\because R = 1\Omega)$$

$$A = \sqrt{2P}.$$

$$\therefore \; c(t) = \sqrt{2P}\cos 2\pi f_c t.$$

As per the principle of ASK, it is mathematically defined as

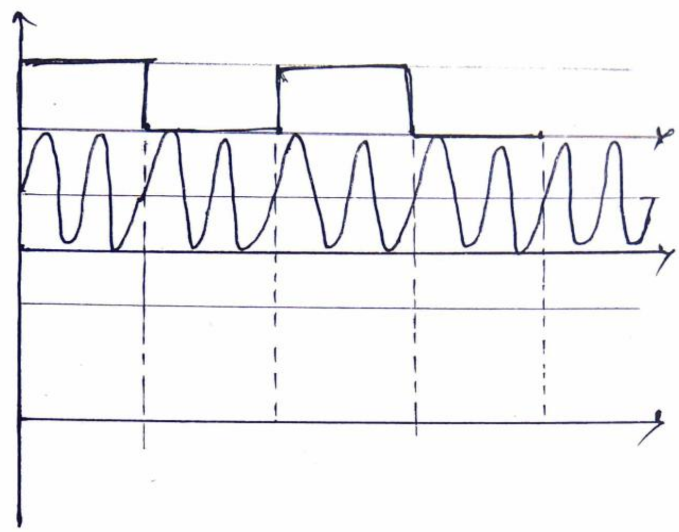
$$s(t) = \sqrt{2P}\cos 2\pi f_c t \quad (\text{To transmit '1'})$$

$$s(t) = 0 \quad (\text{To transmit '0'})$$

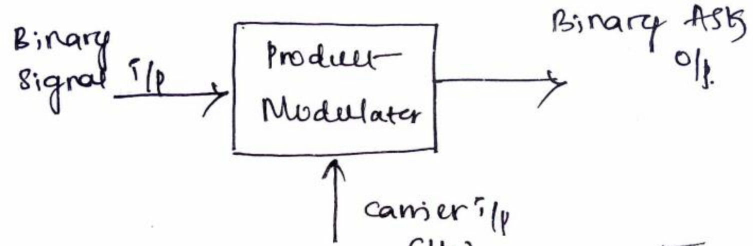
$$\therefore \; s(t) = \begin{cases} c(t) & \text{for symbol '1'} \\ 0 & \text{for symbol '0'} \end{cases}$$

To transmit symbol '0', the signal $s(t)=0$ i,e; no signal is transmitted. And, to transmit symbol '1', the carrier is transmitted as it is. Hence the ASK waveform looks like an ON-OFF of the signal. Therefore it is also known as ON-OFF keying.

Waveform of ASK:



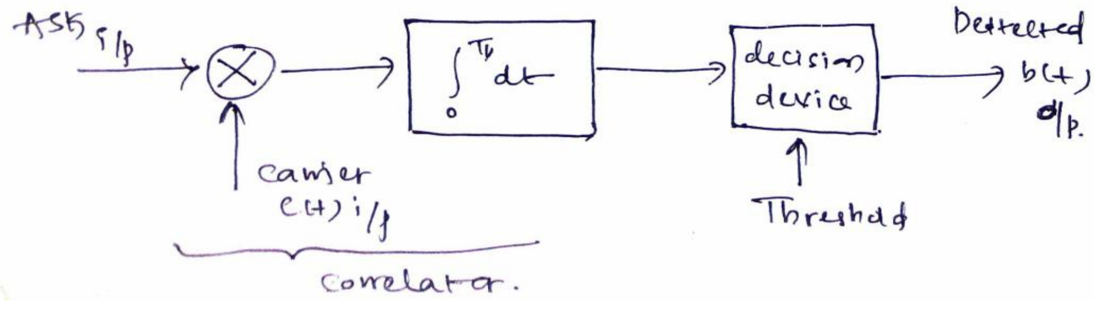
Generation of ASK:



ASK signal may be generated by simply applying the incoming binary data (unipolar form) and the sinusoidal carrier to the two inputs of a product modulator. The resulting output will be the ASK waveform.

Degeneration (or) Detection of ASK:

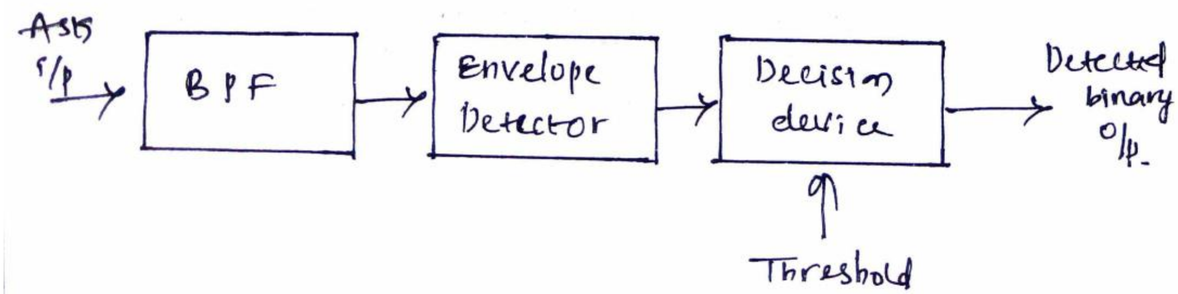
Coherent Detection:



The demodulation of binary ASK waveform can be achieved with the help of coherent detection. It consists of a product modulator which is followed by an integrator and a decision-making device. The ASK signal is applied to one input of the product modulator. The other input of the product modulator is supplied with a sinusoidal carrier which is generated with the help of a local oscillator. The output of the product modulator goes to input of the integrator. The integrator integrates over one bit duration of received i/p signal from multiplier. This output is fed to the input of decision device.

Now, the decision-making device compares the output of the integrator with a preset threshold. It makes a decision in favour of symbol '1' when the threshold is exceeded and in favour of symbol '0' otherwise. The coherent detection makes the use of linear operation. In this method we have assumed that the local carrier is in perfect synchronization with the carriers used in the transmitter. This means that the frequency and phase of the locally generated carrier is same as those of the carriers used in the transmitter.

Non-Coherent Detection of ASK:



Received ASK signal is applied to the input of BPF (Bandpass filter). This filter passes only carrier frequency, 'f_c'. The envelope detector generates high output voltage when carrier is present. When carrier (f_c) is absent, there is only noise at the i/p of the envelope detector. Hence it produces low output.

The decision device is basically a regenerator. It generates the the binary sequence $b(t)$. Threshold is provide to the decision device to overcome effects due to noise. when $y(t)$ is greater than threshold, $b(t) = 1$ and $y(t)$ is less than threshold, $b(t) = 0$. Non-coherent reception of ASK does not need any carrier synchronization.

Geometrical Representation (or) Signal Space diagram of ASK:

The ASK waveform for binary i/p symbols are represented as

$$S(t) = c(t) \quad \text{for symbol-'1'}$$
$$= 0 \quad \text{for symbol-'0'}$$

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

$$\therefore \quad s(t) = \begin{cases} \sqrt{2P} \cos 2\pi f_c t & \text{for symbol-'1'} \\ 0 & \text{for symbol-'0'} \end{cases}$$

for symbol-'1': It is represented as

$$s(t) = \sqrt{PT_b} \cdot \sqrt{2/T_b} \cos(2\pi f_c t) \quad (\because \text{Multiply \& divide by } \sqrt{T_b})$$

$$= \sqrt{PT_b}\, \phi_1(t) \quad \left(\because \text{If } \phi_1(t) = \sqrt{\tfrac{2}{T_b}} \cos 2\pi f_c t\right)$$

$$\therefore \quad s(t) = \begin{cases} \sqrt{PT_b}\, \phi_1(t) & \text{for symbol-'1'} \\ 0 & \text{for symbol-'0'} \end{cases}$$

Binary Frequency Shift Keying (BFSK).

In binary frequency shift keying, the frequency of the carrier is shifted according to the binary symbol. That is we have two different frequency signals according to the binary symbols.

Let two carriers; $c(t) = \sqrt{2P}\cos(2\pi f_c t)$

$c_1(t) = \sqrt{2P}\cos(2\pi f_c + \Omega)t$; $c_2(t) = \sqrt{2P}\cos(2\pi f_c - \Omega)t$.

Here 'Ω' is the frequency shift | $f_c + \Omega = f_H$, $f_c - \Omega = f_L$.

High frequency carrier $(f_H) = c_1(t) = \sqrt{2P}\cos(2\pi f_c + \Omega)t$

Low frequency carrier $(f_L) = c_2(t) = \sqrt{2P}\cos(2\pi f_c - \Omega)t$.

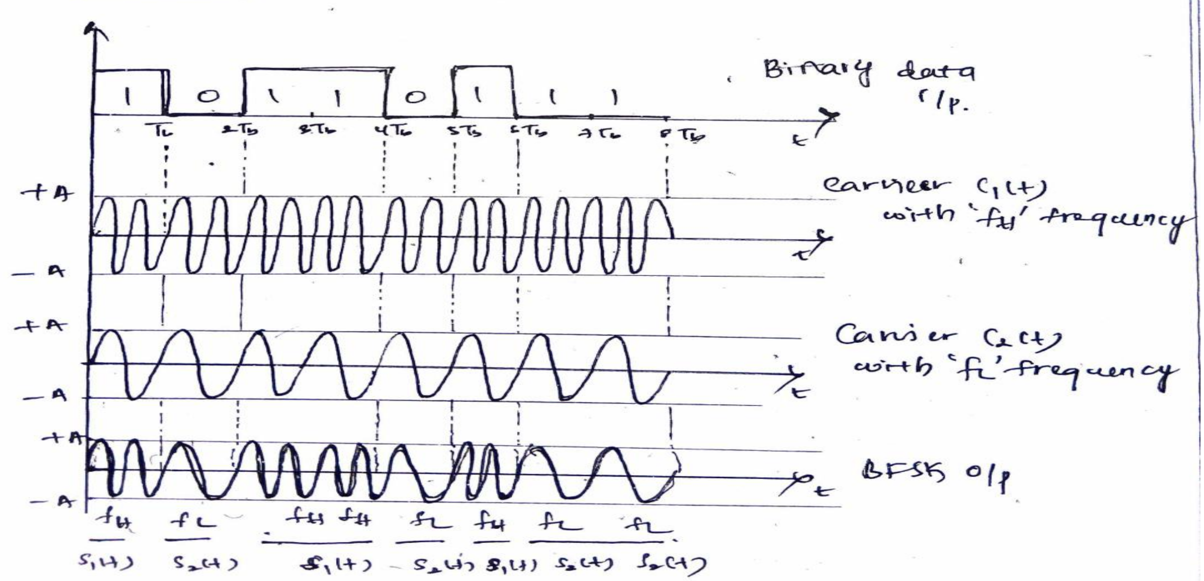
For binary symbol-'1', the modulator modulates with f_H carrier and for symbol-'0', it modulates with 'f_L' carrier. In this way we are modulating binary symbols with two different frequency carriers.

If $s(t)$ denotes o/p of FSK then BFSK o/p is

$$BFSK = \begin{cases} c_1(t) \text{ for symbol - '1'} \\ c_2(t) \text{ for symbol - '0'} \end{cases} \Rightarrow \begin{cases} \sqrt{2P}\cos 2\pi f_H t \\ \sqrt{2P}\cos 2\pi f_L t. \end{cases}$$

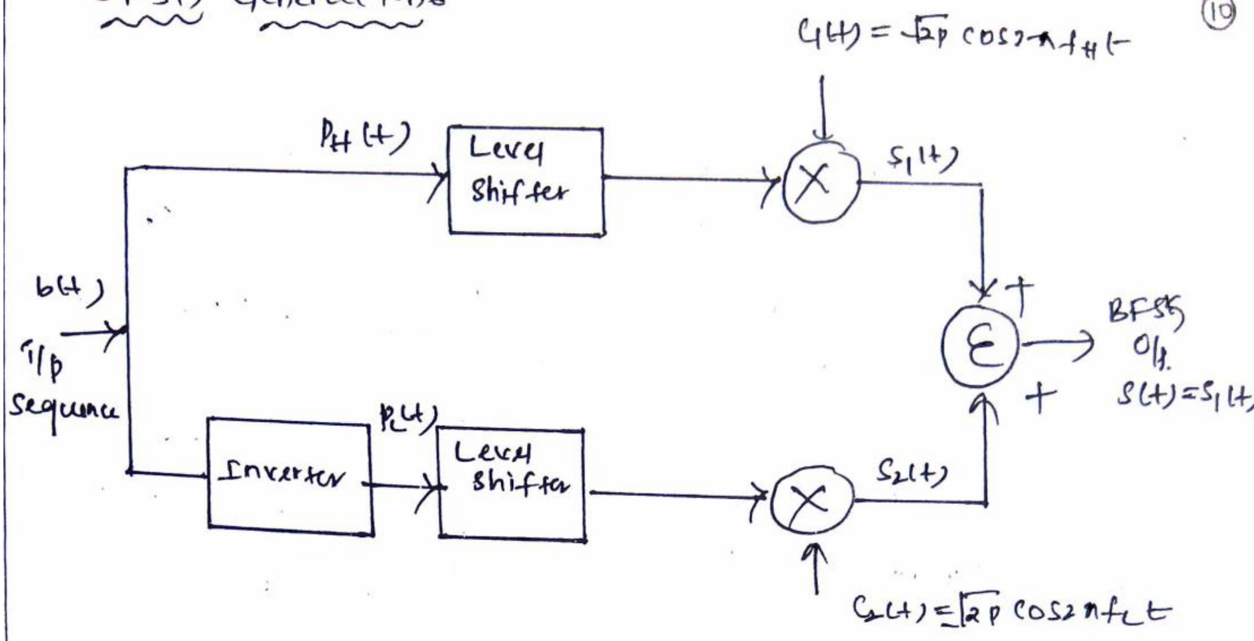
BFSK Generation:

BFSK Waveforms:



Here $\boxed{f_H > f_L}$ and $\boxed{f_H - f_L = 2f_b}$.

BFSK Generation



Above diagram shows the block diagram of FSK Generation. Input binary signal $b(t)$ is applied to the i/p of two branches. From above figure $P_H(t)$ is same as $b(t)$. And $P_L(t)$ is inverted version of $b(t)$. $P_H(t)$ and $P_L(t)$ are unipolar signals. The level shifter converts the symbol-'1' to +1V and symbol-0 to 0V. Further there are product modulators after level shifter. The two carrier signals f_H and f_L of $c_1(t)$ and $c_2(t)$ are used to modulate the i/p applied symbols. These two carriers are orthogonal to each other.

Here note that outputs from both the multipliers are not possible at a time. This is because $P_H(t)$ and $P_L(t)$ are complementary to each other. Therefore if $P_H(t) = 1$, then output will be only due to upper modulator and lower modulator output will be zero ($\because P_L(t) = 0$).

$$\therefore \text{BFSK signal} = S_1(t) + S_2(t)$$

$$= \begin{cases} \sqrt{2P}\cos 2\pi f_H t + 0 & ; \text{for symbol-'1'} \\ 0 + \sqrt{2P}\cos 2\pi f_L t & ; \text{for symbol-'0'} \end{cases}$$

Generally,

for digital modulation schemes, used carrier is

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

$f_c \rightarrow$ carrier frequency.

In FSK modulation, we should modulate two binary symbols: (i,e; '1' and '0'). Here amplitude and phase of the carrier is constant. In order to transmit these two symbols we need to consider two different carriers which are different in frequency only. So, 'f_c' is shifted by the factor of 'Ω' to define two different frequencies.

i,e; $(f_c + \Omega) = f_H$ (High frequency carrier).

$(f_c - \Omega) = f_L$ (low frequency carrier).

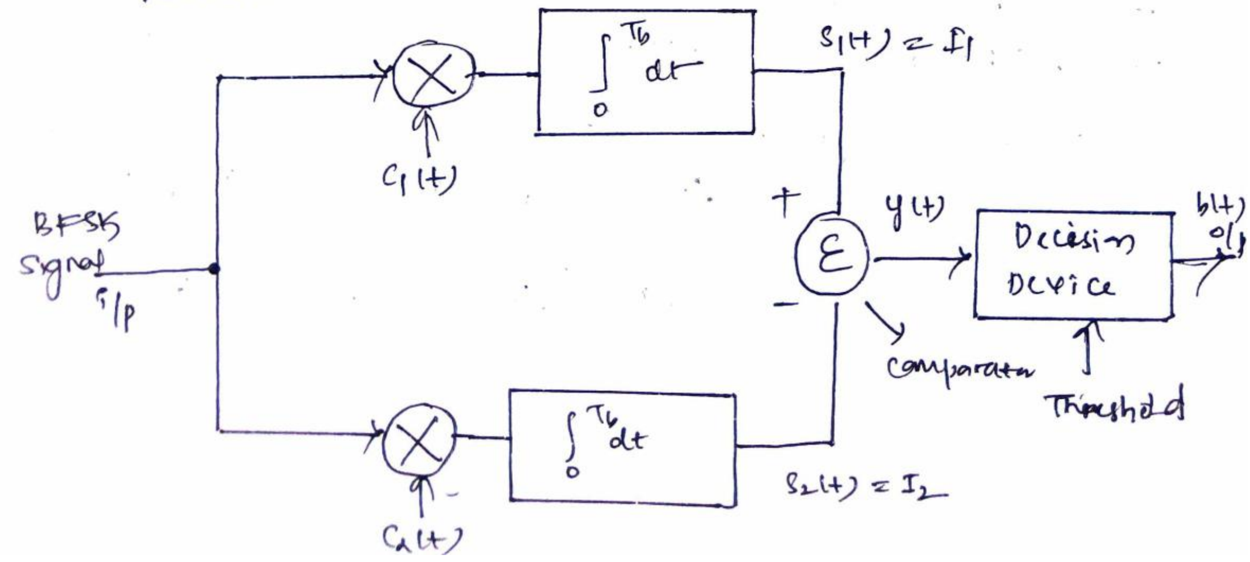
By observing above block diagram we can understand that for symbol-1, we will get 'f_H' component of carrier and for symbol-'0', we will get 'f_L' component of carrier. So, at a specific symbol the FSK o/p is either 'f_H' or 'f_L' component only.

Therefore, o/p of FSK is $s_1(t) + s_2(t)$.

$$\text{FSK o/p} = \begin{cases} s_1(t) + s_2(t) \\ \sqrt{2P} \cos 2\pi f_H t \text{ for symbol-1; } \sqrt{2P}\, P_H(t) \cos 2\pi f_H t \\ \sqrt{2P} \cos 2\pi f_L t \text{ for symbol-0; } \sqrt{2P}\, P_L(t) \cos 2\pi f_L t \end{cases}$$

BFSK Degeneration:

(i) Coherent:

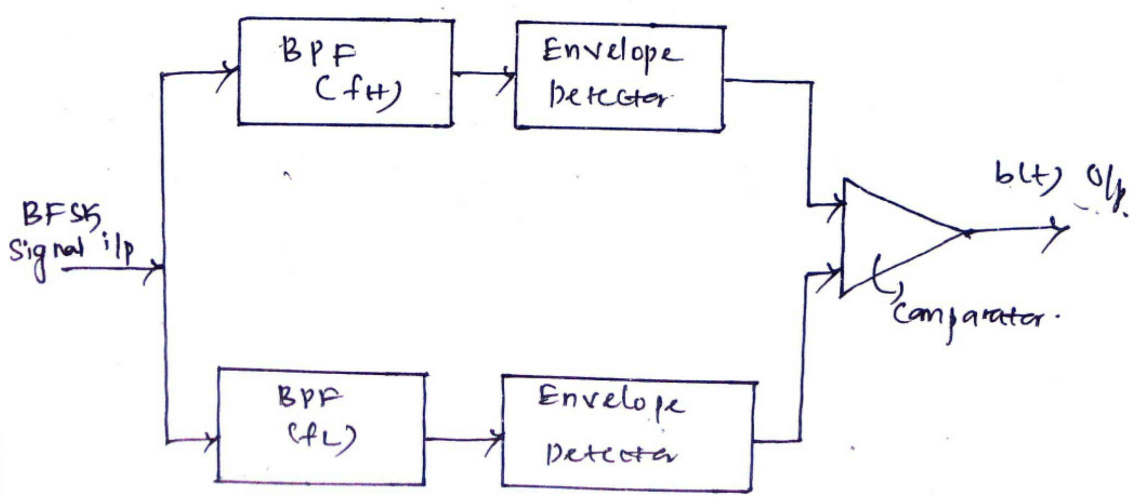


Blocks diagram of BFSK under coherent demodulation scheme is shown in above figure. It consists of two correlators that are individually tuned to two different carrier frequencies to represent symbols '1' and symbol '0'. A correlator consists of a multiplier followed by an integrator. Then, the received binary FSK signal is applied to the multipliers of both the correlators. To the other input of the ~~mod~~ multipliers, carriers with frequency f_H & f_L are applied. The multiplied output of each multiplier is sub frequently passed through integrators generating output 'I_1' and 'I_2' in the two paths.

The output of two integrators are then fed to the decision making device. The decision making device is essentially a comparator which compares the output I_1 path and output I_2 path. If I_1 is greater than I_2, the decision making device makes a decision in favour of symbol. If the output I_1 is less than I_2, then the decision making device decides in favour of symbol '0'. This way it will extract the information from received signal.

(ii) Non-Coherent:

Binary FSK signal may be demodulated non-coherently using enveloped detectors. It is shown in below blocks diagram.



The received FSK signal is applied to a bank of two Bandpass filters, one tuned to 'f_H' and the other tuned to 'f_L'. Each filter is followed by an envelope detector. The resulting outputs of the two envelope detectors are sampled and then compared with each other.

A decision is made in favour of symbol-'1' if the envelope detector output derived from the filter tuned to frequency f_H is larger than that derived from the second filter. otherwise a decision is made in favour of the symbol-'0'.

Binary Phase Shift Keying: (BPSK):

In BPSK technique, the phase of the sinusoidal carrier is changed according to the data bit to be transmitted. It is the most efficient of the three digital modulation techniques. (ASK, FSK, PSK). Hence BPSK is used for high bit rates.

Expression for BPSK:

In a binary phase shift keying (BPSK), the binary symbols '1' and '0' modulate the phase of the carrier.

Let carrier $c(t) = \sqrt{2P}\cos 2\pi f_c t$.

Now, when the symbol is changed, then the phase of the carrier will also be changed by an amount of 180 degrees (i.e., π radians).

Let us consider, for example,

for symbol-'1', we have

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

If next symbol-'0' then we have for symbol-'0':

$$s_2(t) = \sqrt{2P}\cos(2\pi f_c t + \pi)$$

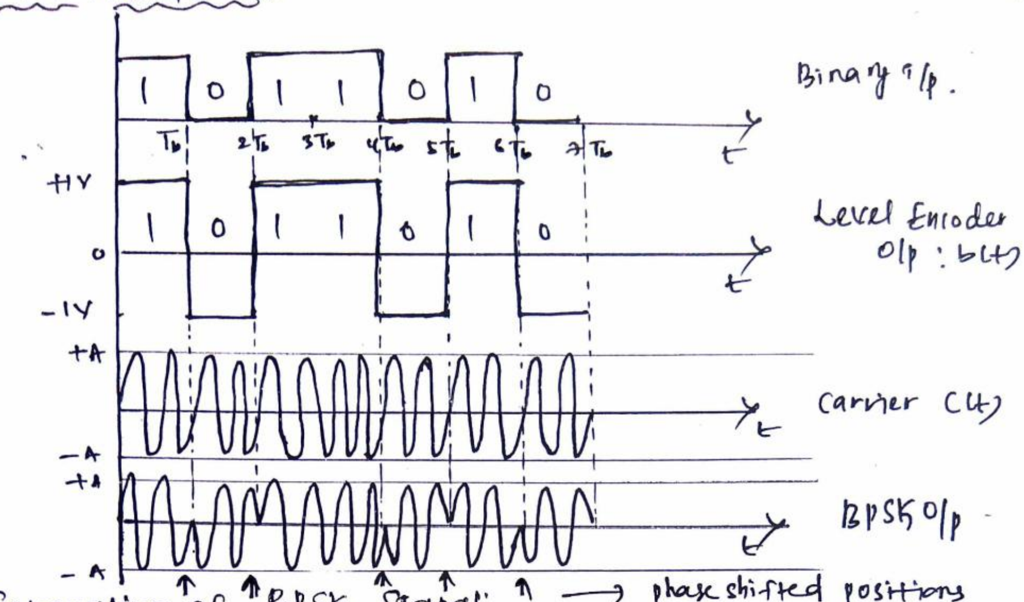
$$= -\sqrt{2P}\cos 2\pi f_c t \qquad \because \cos(\theta+\pi) = -\cos\theta.$$

With the above equation, we can define, BPSK signal combinely as

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

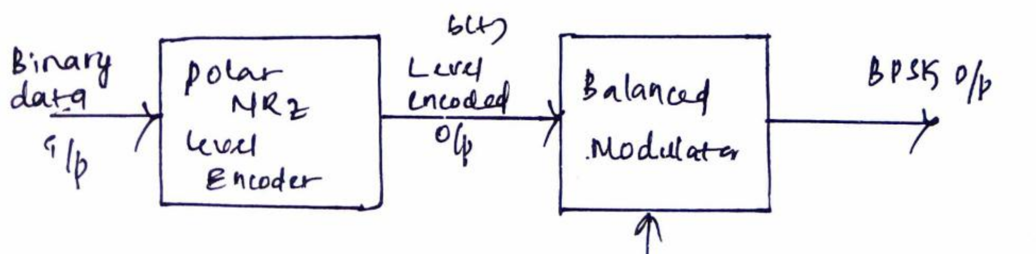
where $b(t) = +1$ when binary '1' is to be transmitted.
$= -1$ when binary '0' is to be transmitted.

BPSK Waveform:



Generation of BPSK Signal:

BPSK Generator consists of polar NRZ level encoder and balanced modulator. It is the process of simple multiplication. Usually, for any digital modulation process we should use line coding technique to convert the i/p binary symbols to the electrical equivalent signals. Because, modulator can understand only electrical signals. So, before going to modulation process we introduce line coding process. In this BPSK generation we mainly prefer polar NRZ (Non return to zero) technique, which converts (or) generates +1V for binary symbol-'1' and -1V for binary symbol-'0'.



i/p	b(t)	BPSK o/p
symbol-'0'	$b(t) = -1V$	$-\sqrt{2P}\cos 2\pi f_c t$
symbol-'1'	$b(t) = +1V$	$+\sqrt{2P}\cos 2\pi f_c t$

The binary data signal is converted to NRZ bipolar signal by an NRZ encoder. Here the bipolar signal b(t) is applied as a modulating signal to the balanced modulator. For this modulator, other input is fixed frequency carrier signal. It is high frequency and constant in amplitude, frequency and phase. Based on the input binary symbol this modulator changes its phase by 180°.

i/p digital signal	Bipolar NRZ signal b(t)	BPSK o/p signal.
Binary -'0'.	$b(t) = -1$	$-\sqrt{2P}\cos 2\pi f_c t$
Binary -'1'	$b(t) = +1$	$+\sqrt{2P}\cos 2\pi f_c t$

Above table shows the relation of i/p and o/p signals of balanced modulator.

Degeneration of BPSK Signal:

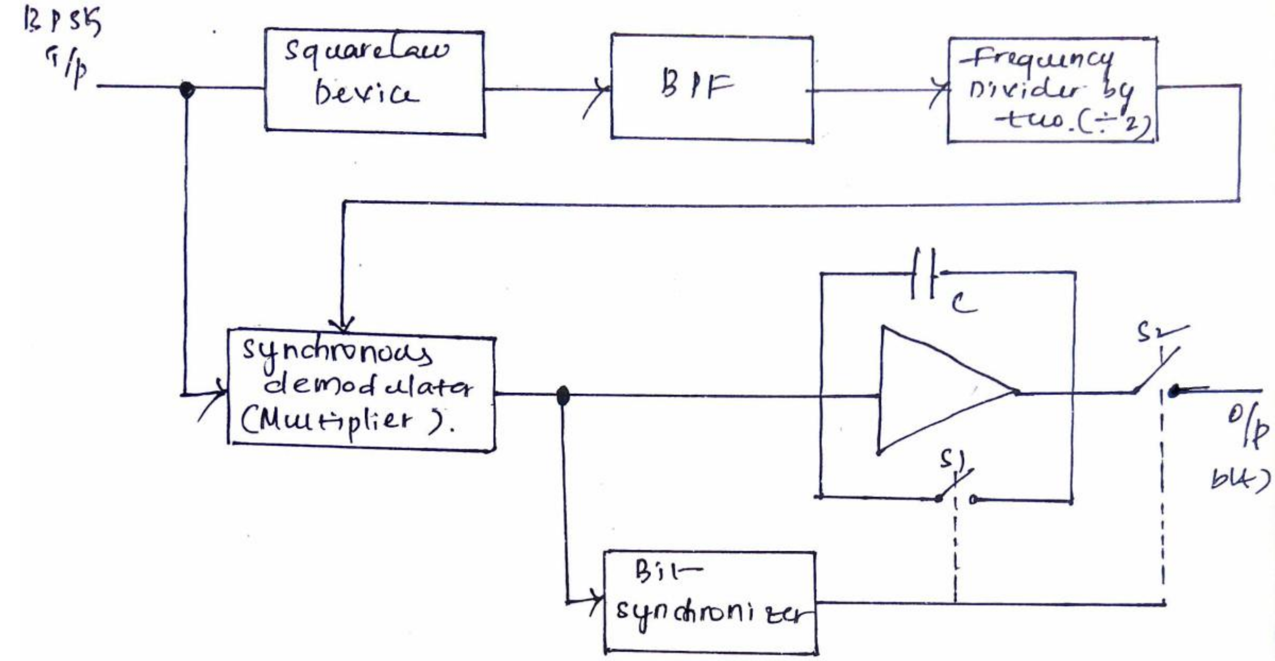
Under digital demodulation techniques we have two methods mainly (i) Coherent (ii) Non coherent. But in BPSK type of signals, frequency of the modulated waveform is constant in nature. So, Hence it will give equal outputs for both the binary symbols at envelope detector. So, it introduce some type of ambiguity to decide the exact transmitted symbol in received component. Therefore non-coherent technique is not applicable to PSK type of modulations.

Coherent Detection of PSK Signal:

Here, the received signal undergoes the phase change of depending upon the time delay from transmitter end to receiver end. This phase change is usually, a fixed phase shift in the transmitted signal. Let us consider that this phase shift is 'θ'. Because of this, the signal at the i/p of the receiver can be written as

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

Now, from this received signal, a carrier is separated because this is coherent detection.



The received signal is allowed to pass through a square law device. At the output of the squarelaw device, we get a signal which is given as

$$\cos^2(2\pi f_c t + \theta).$$

Here, it may be noted that we have neglected the amplitude, since we are only interested in the carrier of the signal.

Again, we know that

$$\cos^2\theta = \frac{1+\cos 2\theta}{2}$$

Therefore, we have

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

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

Here, $\frac{1}{2}$ represents DC level. This signal is then allowed to pass through a bandpass filter (BPF) whose passband is centred around '$2f_c$'. Bandpass filter removes the D.C level of $\frac{1}{2}$ and at the output, we obtain "$\cos 2(2\pi f_c t + \theta)$"

This signal is having frequency equal to "$2f_c$". Hence, it is passed through a frequency divider by-two. Thus, at the output of frequency divider, we get a carrier signal whose frequency is 'f_c' i.e., $\cos(2\pi f_c t+\theta)$

The synchronous demodulator multiplies the i/p signal and the recovered carrier. Here, at the output of multiplier, we get

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

$$= b(t)\sqrt{2P}\times\frac{1}{2}[1+\cos 2(2\pi f_c t+\theta)]$$

$$= b(t)\sqrt{\frac{P}{2}}[1+\cos 2(2\pi f_c t+\theta)]$$

This signal is then applied to the bit synchronizer and integrator. The integrator integrates the signal over one bit period. The bit synchronizer take care of starting and ending times of a bit. At the end of bit duration 'T_b', the bit synchronizer closes switch 'S_2' temporarily. This connects the o/p of an integrator to the decision device. In fact, it is equivalent to sampling the output of integrator. The synchronizer then opens switch 'S_2' and switch 'S_1' is closed temporarily. This resets the integrator voltage to zero. The integrator then integrates next bit. Let us assume that one bit period 'T_b' contains integral no. of cycles of the carrier. This means that the phase change occurs in the carrier only at zero crossing.

At k^{th} bit interval, we can write output signal as under:

$S_0(kT_b) \rightarrow$ o/p of integrator.

$$S_0(kT_b) = b(kT_b)\sqrt{\frac{P}{2}}\int_{(k-1)T_b}^{(kT_b)}[1+\cos 2(2\pi f_c t+\theta)]\,dt.$$

This equation gives the output of an interval for k^{th} bit. Hence, integration is performed from $(k-1)T_b$ to kT_b.

Here $T_b \rightarrow$ one bit period.

$$\therefore\ S_0(KT_b) = b(KT_b)\sqrt{\frac{P}{2}}\left[\int_{(K-1)T_b}^{KT_b} 1\, dt + \int_{(K-1)T_b}^{KT_b} \cos 2(2\pi f_c t + \theta)\, dt\right].$$

Here $\int_{(K-1)T_b}^{KT_b} \cos 2(2\pi f_c t+\theta)\, dt = 0$, since the average value of sinusoidal waveform is zero if integration is done over full cycles. Hence we may write

$$S_0(KT_b) = b(KT_b)\sqrt{\frac{P}{2}}\int_{(K-1)T_b}^{KT_b} 1\, dt = b(KT_b)\sqrt{\frac{P}{2}}\,[t]_{(K-1)T_b}^{KT_b}$$

$$= b(KT_b)\sqrt{\frac{P}{2}}\{KT_b - (K-1)T_b\}$$

$$= b(KT_b)\sqrt{\frac{P}{2}}\, T_b$$

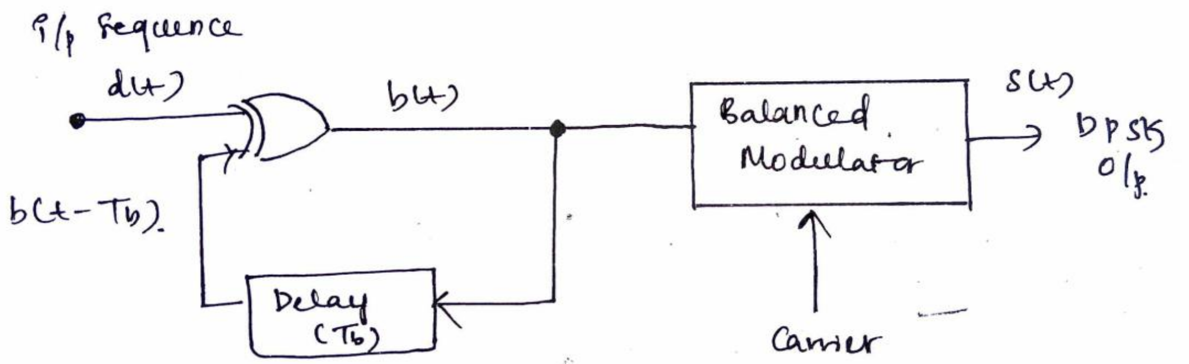
$$S_0(KT_b) \propto b(KT_b)$$

That means, the output of integrator depends upon i/p b(t). Depending upon the value of $b(KT_b)$, the output $S_0(KT_b)$ is generated in receiver. This signal is then applied to a decision device which decides whether transmitted symbol was zero (or) one.

Differential Phase shift Keying (DPSK):

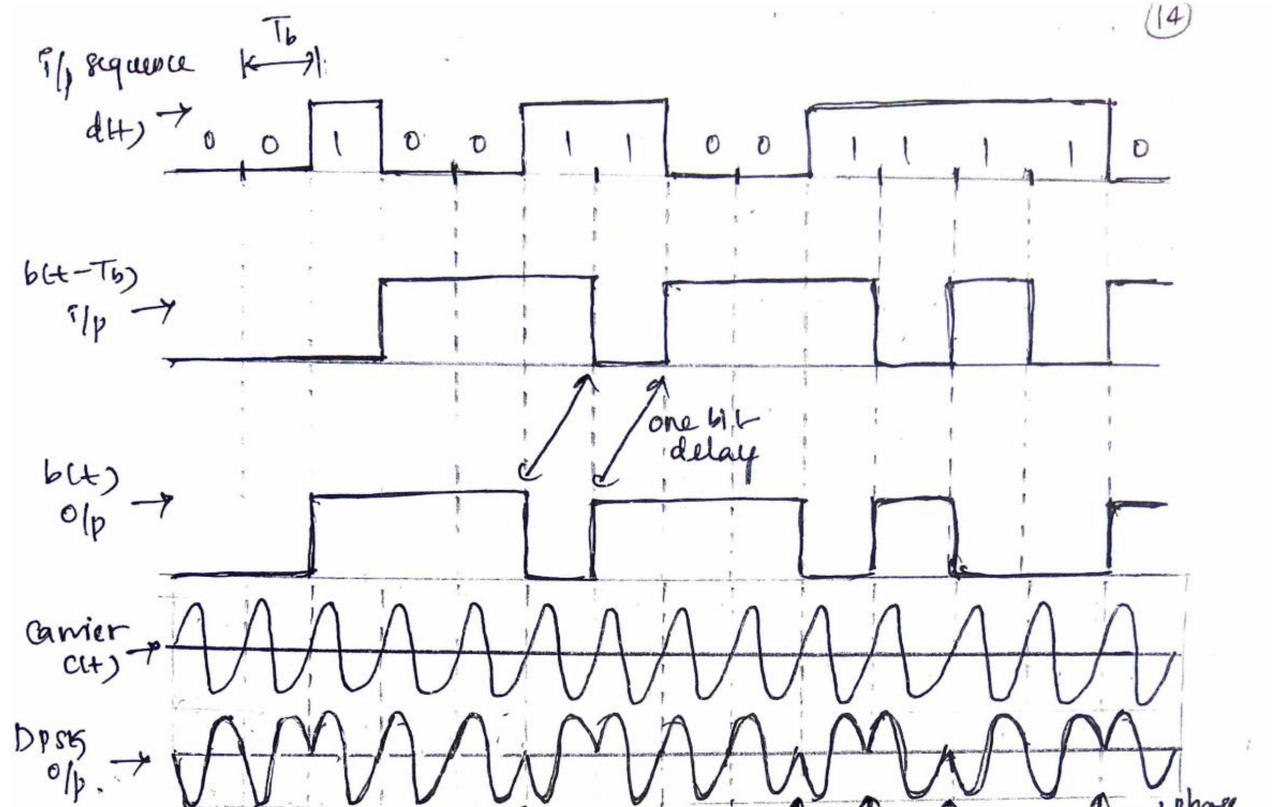
Differential phase shift keying does not need a synchronous carrier at the demodulator. The input sequence of binary bits is modified such that the next bit depends upon the previous bits. Therefore in the receiver the previous received bits are used to detect the present bit.

DPSK Transmitter



Above diagram shows the block diagram of DPSK generation. The input sequence d(t), Output sequence is b(t) and b(t-Tb) is the previous output delayed by one bit period. Depending upon values of d(t) and b(t-Tb), exclusive-OR Gate generates the output sequence b(t), based on below given truth table.

d(t)	b(t-Tb)	b(t)
0 (-1V)	0 (-1V)	0 (-1V)
0 (-1V)	1 (+1V)	1 (+1V)
1 (+1V)	0 (-1V)	1 (+1V)



From the waveforms it is clear that $b(t-T_b)$ is the delayed version of $b(t)$ by one bit period 'T_b'.

$$\therefore \quad b(t) = d(t) \oplus b(t-T_b).$$

By observing above figure (it is clear that $b(t-T_b)$) the output sequence $b(t)$ changes level at the begining of each interval in which $d(t)=1$ and it does not changes level when $d(t)=0$.

When $d(t)=0$; $\quad b(t) = b(t-T_b)$
when $d(t)=1$; $\quad b(t) = \overline{b(t-T_b)}$

The sequence $b(t)$ modulates sinusoidal carrier. When $b(t)$ changes the level, ~~the~~ phase of the carrier is changed. Since $b(t)$ changes its level only if $d(t)=1$; It shows that phase of the carrier is changed only if $d(t)=1$. It is noticed when we compare ~~with~~ o/p DPSK signal with i/p signal $d(t)$. But internally we are modulating the carrier wave with respect to phase only. Due to the Ex-OR operation of i/p applied signal it ~~ready~~ is converting like this.

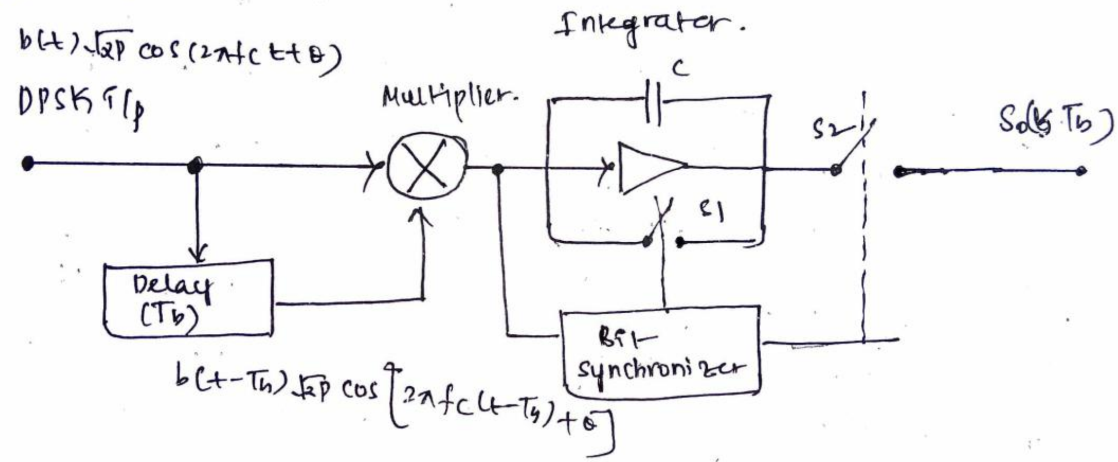
So, in BPSK phase of the carrier changes on both the symbols '1' and '0'. Where as in DPSK phase of the carrier changes only on symbol '1'. This is the main difference between BPSK and DPSK.

O/p of DPSK ~~signal is~~ operation is:

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

$$= \pm\sqrt{2P}\cos 2\pi f_c t.$$

DPSK Receiver: It does not need coherent detection to detect the original signal. Block diagram of DPSK receiver is shown below.



During the transmission, the DPSK signal undergoes some phase shift 'θ'. Therefore the signal received at the i/p of the receiver is $b(t)\sqrt{2P}\cos(2\pi f_c t+\theta)$.

Then the signal is multiplied with delayed version by one bit. Therefore the o/p of the multiplier is,

$$\Rightarrow b(t)\cdot b(t-T_b)(2P)\cos(2\pi f_c t+\theta)\cos[2\pi f_c(t-T_b)+\theta]$$

$$\because \cos A\cos B = \frac{1}{2}[\cos(A+B)+\cos(A-B)]$$

Here $A = 2\pi f_c t+\theta$, $B = 2\pi f_c(t-T_b)+\theta$.

$$\text{Multiplier o/p} = b(t)b(t-T_b)P\left\{\cos 2\pi f_c T_b + \cos\left[4\pi f_c\left(t-\frac{T_b}{2}\right)+2\theta\right]\right\}$$

$T_b \rightarrow$ one bit period

'T_b' contains integral no. of cycles of 'f_c'.

We know that

$$f_b = \frac{1}{T_b}.$$

If 'T_b' contains 'n' cycles of 'f_c' then we can write

$$f_c = n f_b$$

$$f_c = \frac{n}{T_b}$$

$$\therefore \boxed{f_c T_b = n}.$$

By substituting $f_c T_b = n$ in multiplier output we can write it as

$$\Rightarrow b(t).\, b(t-T_b) P\left\{ \cos 2\pi n + \cos\left[4\pi f_c\left(t - \frac{T_b}{2}\right) + 2\theta\right]\right\}$$

$$\because \cos 2\pi n = 1;$$

$$\Rightarrow b(t).\, b(t-T_b) P + b(t)\, b(t-T_b) P \cos\left[4\pi f_c\left(t - \frac{T_b}{2}\right) + 2\theta\right]$$

This signal is applied to the i/p of integrator. In the k^{th} bit interval the integrator output can be written as

$$s_0(kT_b) = b(kT_b)\, b[(k-1)T_b]\, P \int_{(k-1)T_b}^{kT_b} dt \; +$$

$$+ b(kT_b)\, b[(k-1)T_b]\, P \int_{(k-1)T_b}^{kT_b} \cos\left[4\pi f_c\left(t - \frac{T_b}{2}\right) + 2\theta\right] dt$$

The integration of the second term will be zero since it is integration of carrier over one bit duration. The carrier has integral no. of cycles over one bit period hence its integration is zero. Therefore we can write

$$s_0(kT_b) = b(kT_b)\, b[(k-1)T_b]\, P[kT_b - (k-1)T_b]$$

$$= b(kT_b)\, b[(k-1)T_b]\, P T_b.$$

We know that $PT_b = E_b$ (Energy of one bit).

The product $b(kT_b)\,b[(k-1)T_b]$ decides the sign of PT_b.

The transmitted data bit $d(t)$ can be verified easily from product $b(kT_b)\,b[(k-1)T_b]$.

From figure, we can observe that

when $b(t) = b(t-T_b)$, $d(t)=0$. That is if both are $+1V$ or $-1V$ then $b(t)\,b(t-T_b) = 1$. Alternatively we can write

if $\boxed{b(t) * b(t-T_b) = 1V \quad \text{then } d(t)=0.}$

We know that $b(t) = \overline{b(t-T_b)}$ then $d(t)=1$. That is $b(t) = -1V$, $b(t-T_b) = +1V$ and vice-versa. Therefore $b(t).b(t-T_b)$

Alternately we can write,

$\boxed{\text{if } b(t)\,b(t-T_b) = -1V, \text{ then } d(t)=1}$.

Using above notations, the decision device will decides the integrated data is relates to symbol-'1' or symbol-'0'.

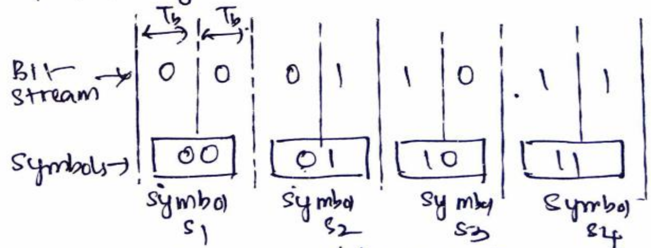
$$\therefore \quad S_0(kT_b) = \begin{cases} -PT_b, & \text{then } d(t)=1 \text{ and} \\ +PT_b, & \text{then } d(t)=0. \end{cases}$$

Quadrature Phase shift Keying (QPSK):

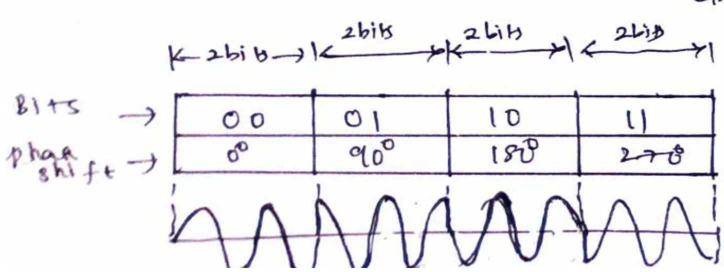
The modulation schemes discussed so far are all two level modulations (ASK and BPSK), because they can represent only two states of the digital data ('0' or '1'). Hence, the bit rate and baudrate are same for these systems.
The maximum bit rate which can be achieved using ASK, BFSK or BPSK systems does not meet the requirements of data communication systems. This happens due to the limited bandwidth of the telephone voice channel. We can keep the baudrate same and increase the bit rate by using multilevel modulation techniques. In this type of systems, the data groups are divided into groups of two or more bits and each group of bits is represented by a specific value of amplitude frequency (or) phase of the carrier. QPSK is an example of such multilevel phase of modulation.

In QPSK system, two successive bits in a bit stream are combined together to form a message and each message is represented by a distinct value of phase shift of the carrier. Every symbol contains two bits. Hence symbol duration is '$2T_b$'. These symbols are transmitted by transmitting the same carrier at four different phase shifts. Hence it is also called as 4-PSK system.

symbol	phase
00	0°
01	90°
10	180°
11	270°



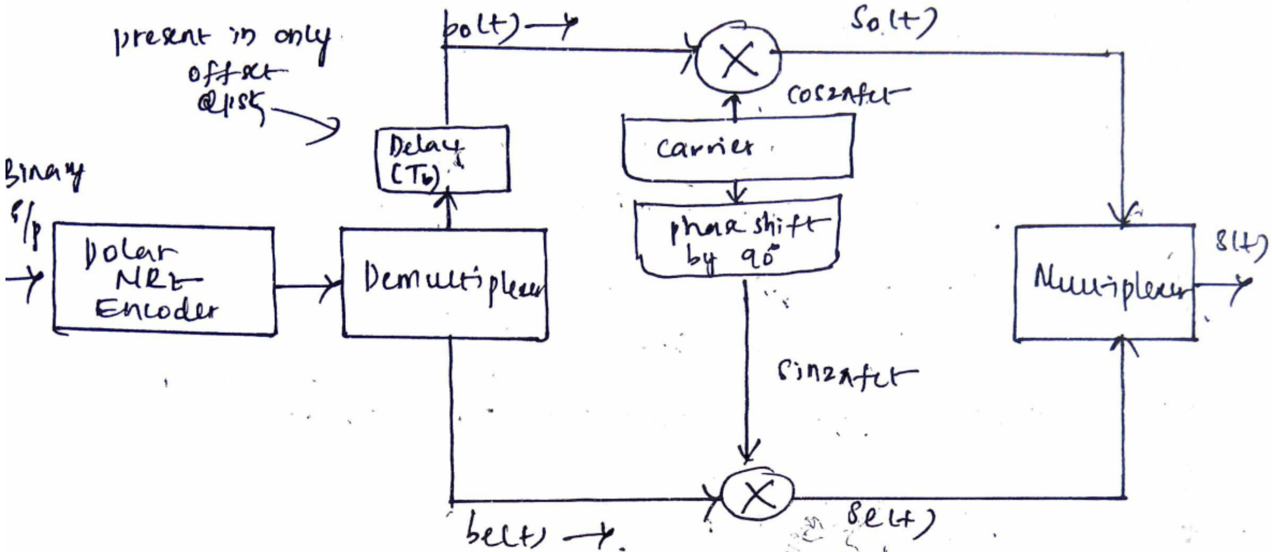
Grouping of bits in QPSK



waveforms of QPSK (4 PSK).

Generation: (offset QPSK).

The block diagram of QPSK is shown in below figure. This shows the mechanism by which a bit stream b(t) generates a QPSK signal for transmission.



The input binary sequence is first converted into bipolar NRZ signal, b(t). The value of b(t) = +1V for logic '1' i/p and b(t) = -1V when the binary input is equal to '0'. The demultiplexer will divide b(t) into two separate bit streams named $b_o(t)$ and $b_e(t)$. The bit stream $b_e(t)$ consists of only the even numbered bits 2, 4, 6 .... whereas the $b_o(t)$ bit stream consists of only the odd numbered bits i.e; 1, 3, 5 ....

Each bit in the even 'or' odd bit stream will be held for a period of '$2T_b$' seconds. This duration is known as 'symbol duration' T_s. Thus every symbol contains two bits. The first odd bit occur before the first even bit. Hence, the even bit stream $b_e(t)$ will start with a delay of one bit period after the first odd bit. This delay is equal to one bit period 'T_b'. It is called as an "offset". In order to eliminate this delay we need to introduce Delay by 'T_b' block at odd bit sequence. Hence both the sequences are start with changes its level at synchronising mode.

The bit streams $b_o(t) = \pm 1V$ is superimposed on a carrier $\sqrt{2P}\cos 2\pi f_c t$ and the bit stream $b_e(t) = \pm 1V$ is superimposed on $\sqrt{2P}\sin 2\pi f_c t$, by the use of two multipliers to generate $S_o(t)$ and $S_e(t)$ respectively. These two signals are basically BPSK signals. These signals are then added together to generate the QPSK output signal.

Here

$b_o(t) \rightarrow$ odd numbered sequence.

$b_e(t) \rightarrow$ even numbered sequence.

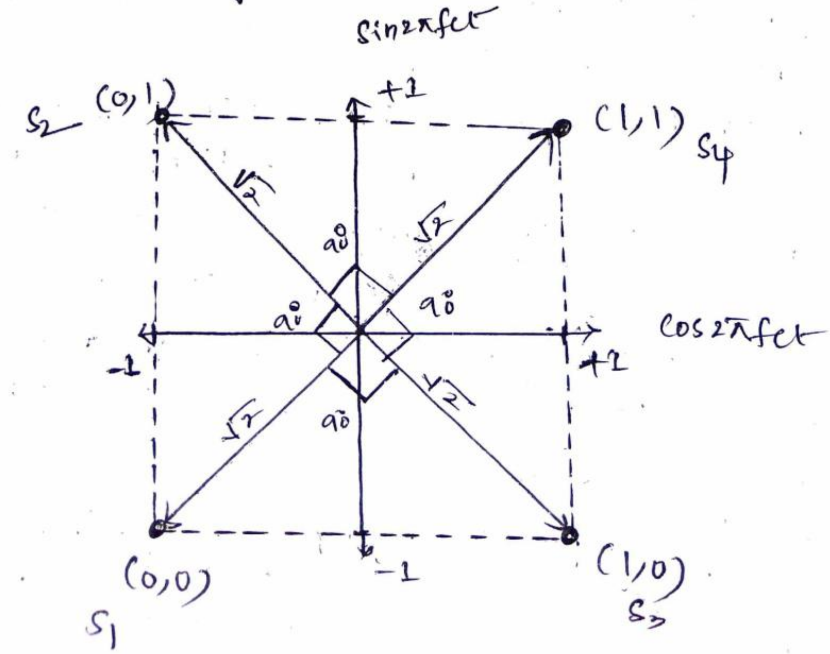
$$S_o(t) = b_o(t)\sqrt{2P}\cos 2\pi f_c t$$

$$S_e(t) = b_e(t)\sqrt{2P}\sin 2\pi f_c t$$

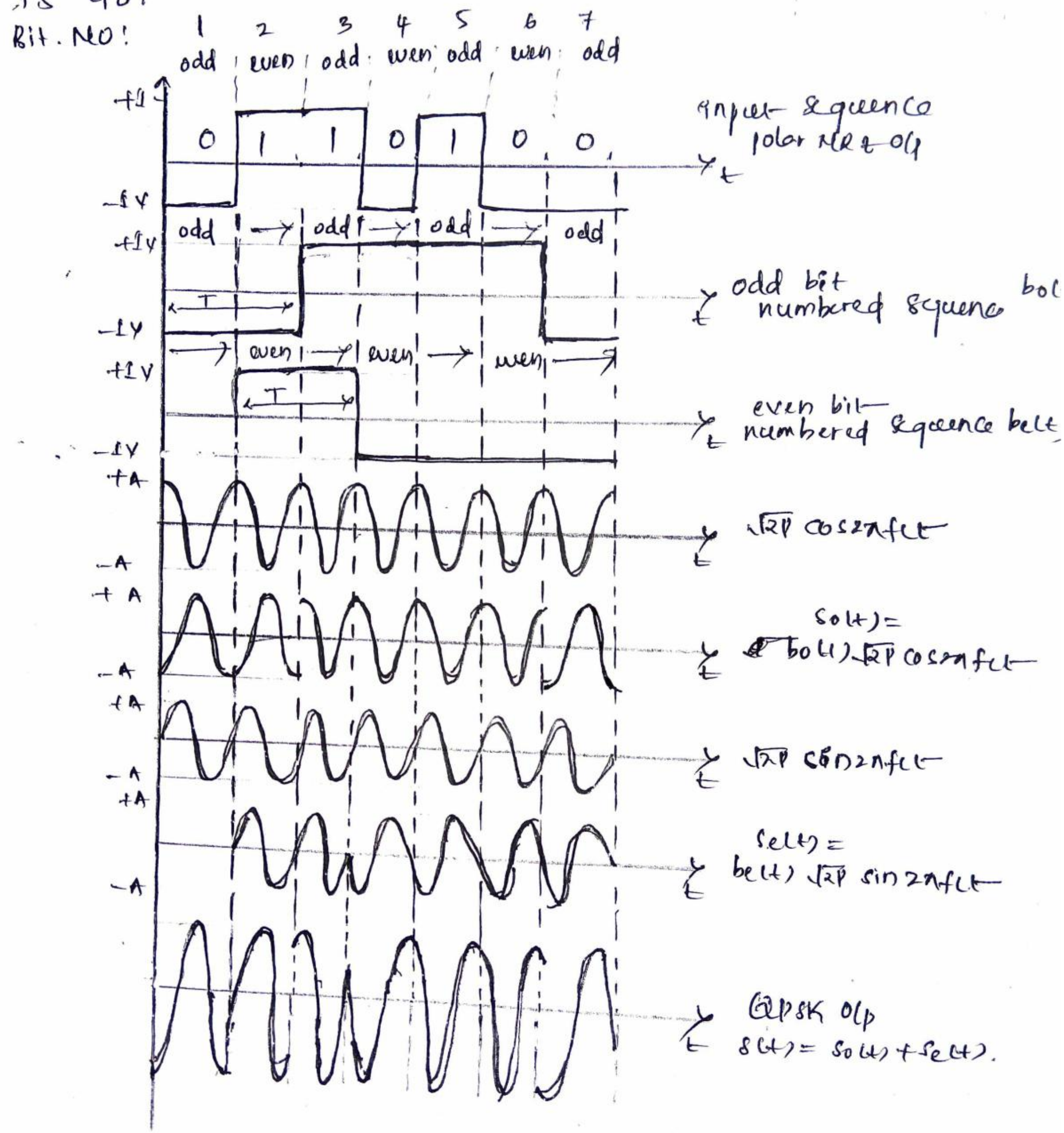
$$\therefore \; S(t) = b_o(t)\sqrt{2P}\cos 2\pi f_c t + b_e(t)\sqrt{2P}\sin 2\pi f_c t.$$

	$b_o(t)$	$b_e(t)$	
$S(t)$	0	0	$\rightarrow -\sqrt{2P_s}\cos 2\pi f_c t - \sqrt{2P_s}\sin 2\pi f_c t$
	0	1	$\rightarrow -\sqrt{2P_s}\cos 2\pi f_c t + \sqrt{2P_s}\sin 2\pi f_c t$
	1	0	$\rightarrow \sqrt{2P_s}\cos 2\pi f_c t - \sqrt{2P_s}\sin 2\pi f_c t$
	1	1	$\rightarrow \sqrt{2P_s}\cos 2\pi f_c t + \sqrt{2P_s}\sin 2\pi f_c t$.

Consider Phasor Diagram:



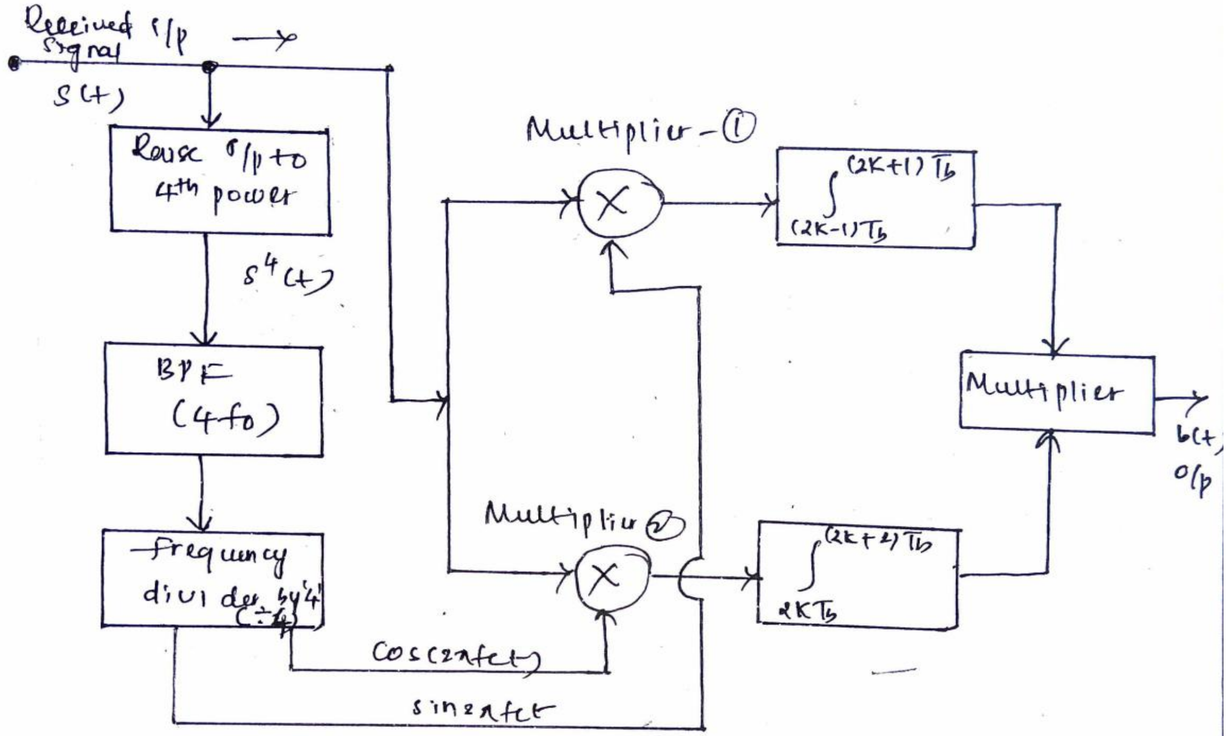
By using above phasor diagram we can observe that phase shift between ~~any~~ two symbols (or) symbol to symbol is "90°". So, in QPSK, maximum phase shift is 90°.



QPSK waveforms.

QPSK Receiver:

The block diagram of a QPSK receiver is shown in below figure. As shown, we use the synchronous detection technique. Therefore it is necessary to locally generate the carriers $\cos \omega_c t$ and $\sin 2\pi f_c t$.



The received QPSK signal s(t) raises to fourth power i,e; $s^4(t)$. This signal is then filtered by using a BPF with center frequency of $4\omega_c t$. The output of BPF is $\cos 4\omega_c t$. A frequency divider which divides the frequency at the filter output by 4 generates the two carriers signals $\sin 2\pi f_c t$ and $\cos 2\pi f_c t$. The incoming signal s(t) is applied to two synchronous demodulators consisting of a multiplier followed by an integrator. Each integrator integrates over a two-bit interval of duration $T_s = 2T_b$. One synchronous demodulator uses $\cos\omega_c t$ and the other one uses $\sin\omega_c t$ as a carrier signal.

The input to the upper integrator is given by

$$s(t) = b_o(t)\sqrt{2P}\cos(2\pi f_c t) + b_e(t)\sqrt{2P}\sin(2\pi f_c t)$$

At upper Multiplier - 1

$$s(t)\sin(2\pi f_c t) = b_o(t)\sqrt{2P}\cos(2\pi f_c t)\sin(2\pi f_c t) +$$
$$+ b_e(t)\sqrt{2P}\sin^2(2\pi f_c t)$$

At Integrator:

$$o/p = \int_{(2K-1)T_b}^{(2K+1)T_b} s(t)\sin(2\pi f_c t)\,dt$$

$$= \int_{(2K-1)T_b}^{(2K+1)T_b} \left[b_o(t)\sqrt{2P}\cos(2\pi f_c t)\sin(2\pi f_c t) + b_e(t)\sqrt{2P}\sin^2(2\pi f_c t)\right]$$

$$= b_o(t)\sqrt{2P}\int_{(2K-1)T_b}^{(2K+1)T_b}\cos(2\pi f_c t)\sin(2\pi f_c t)\,dt +$$
$$+ b_e(t)\sqrt{2P}\int_{(2K-1)T_b}^{(2K+1)T_b}\sin^2(2\pi f_c t)\,dt$$

$$\because \frac{1}{2}\sin(2x) = \sin x\cos x.$$
$$\& \sin^2(x) = \frac{1}{2}\left[1-\cos(2x)\right]$$

$$\Rightarrow \frac{b_o(t)\sqrt{2P}}{2}\int_{(2K-1)T_b}^{(2K+1)T_b}\sin(4\pi f_c t)\,dt + \frac{b_e(t)\sqrt{2P}}{2}\int_{(2K-1)T_b}^{(2K+1)T_b} 1\cdot dt -$$

$$- \frac{b_e(t)\sqrt{2P}}{2}\int_{(2K-1)T_b}^{(2K+1)T_b}\cos 4\pi f_c t\,dt$$

I - Integral and III - Integral denotes integration of sinusoidal signal over full cycle. Hence it tends to zero.

$$\therefore \int_{(2k-1)T_b}^{(2k+1)T_b} s(t)\sin(2\pi f_c t)\,dt = \frac{b_e(t)\sqrt{2P}}{2}\left[t\right]_{(2k-1)T_b}^{(2k+1)T_b}$$

$$\Rightarrow \frac{b_e(t)\sqrt{2P}}{2}\left[\cancel{2k T_b} + T_b - \cancel{2k T_b} + T_b\right]$$

$$= b_e(t)\sqrt{2P}\cdot T_b.$$

Similarly for lower multiplier & Integrator:

$$\Rightarrow b_o(t)\sqrt{2P}\,T_b.$$

Hence we can conclude that by using multiplier we get even component of signal and from lower multiplier we get odd component of signal. These two signals are multiplexed into one signal by multiplexer. Therefore, the final output indicates b(t).

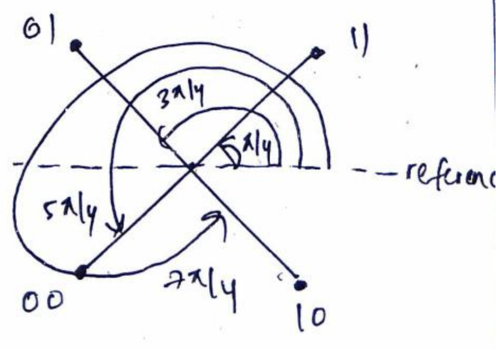
Signal space diagram of QPSK:

Consider alternate representation of QPSK:

Mathematically it is represented by

$$s(t) = \sqrt{2P}\cos\left[2\pi f_c t + (2m+1)\pi/4\right]$$

$m=0$; symbol - '11'.
$m=1$; symbol - '00'
$m=2$; symbol - '01'.
$m=3 \rightarrow$ symbol - '10'.



Let us rearrange the above equation:

$$s(t) = \sqrt{2P}\cos(2\pi f_c t)\cos\left[(2m+1)\pi/4\right] -$$

$$- \sqrt{2P}\sin(2\pi f_c t)\sin\left[(2m+1)\pi/4\right].$$

Intersymbol Interference:

This is a form of distortion of a signal in which one or more symbols interfere with subsequent signals, causing noise (or) delivering a poor output.

Causes of ISI:

- Multipath Propagation
- Non-linear frequency in channels

The ISI is unwanted & should be eliminated completely to get a clean output. The causes of ISI should also be resolved in order to reduce its effect.

The mathematical equations for the receiver output can be considered as

The receiving filter output $y(t)$ is sampled at time $t_i = iT_b$ given.

$$y(t_i) = \mu \sum_{K=-\infty}^{\infty} a_k p(iT_b - kT_b)$$

$$= \mu a_i + \mu \sum_{\substack{K=-\infty \\ k \neq i}}^{\infty} a_k p(iT_b - kT_b)$$

In the above equation, the first term μa_i is produced by the i^{th} transmitted bit.

The second term represents the residual effect of all other transmitted bits on the decoding of the i^{th} bit. This residual effect is Called as Inter Symbol Interference.

In the absence of ISI, the output will be

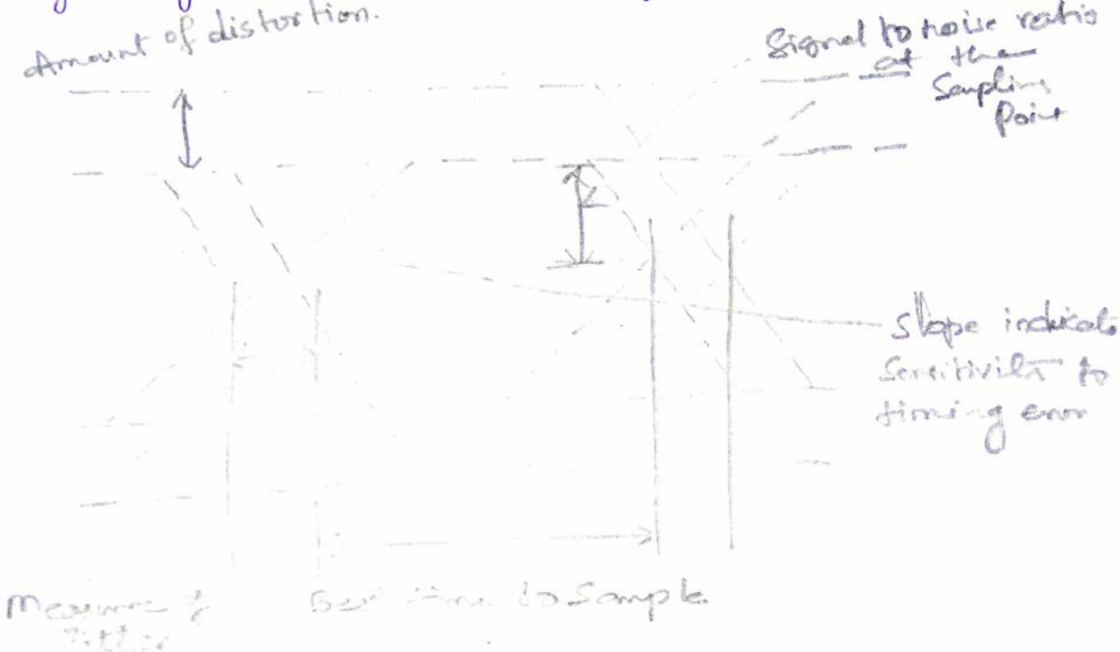
$$y(t_i) = \mu a_i$$

This equation shows that the i^{th} bit transmitted is correctly reproduced. However the presence of ISI introduces bit errors and distortions in the output.

While designing the transmitter or a receiver, it is important that we minimize the effects of ISI, so as to receive the output with least possible error rate.

Eye Pattern (or) Eye diagram

An effective way to study the effects of ISI is the Eye Pattern. The name Eye Pattern is given from its resemblance to the human eye for binary waves. The interior region of the eye pattern is called the 'eye opening'. The following figure shows the image of an eye-pattern.



Jitter is the short term variation of the instant of digital signal, from its ideal Position, which may lead to data errors

When the effect of ISI increases, traces from the upper portion to the lower portion of the eye opening increases and the eye gets completely closed, if ISI is very high.

An eye Pattern provides the following information about a particular system.

- Actual eye patterns are used to estimate the bit error and the signal to noise ratio.

- The width of the eye opening defines the time interval over which the received wave can be sampled ~~out~~ without error from ISI

- The time instant when the eye opening is wide, will be the preferred time for sampling.

- The rate of the closure of the eye, according to the sampling time, determines how sensitive the system is to the timing error.

- The height of the eye opening, at a specified sampling time, defines the margin over noise.

Summary:

Students are introduced the about the concepts of digital modulation techniques. The digital modulation techniques generation and degeneration is dealt. The concept of Baseband transmission and optimal reception is discussed along with intersymbol interference and eye diagrams.

Assignment:

1. With neat diagrams and equations, explain about PSK system.
2. Explain frequency shift keying. Describe coherent detection of FSK signals. What should be the relationship between bit-rate and frequency-shift for a better performance?
3. Explain the transmitter and receiver section of the DPSK techniques in detail.
4. What is the need of pulse shaping for optimum transmission in baseband transmission? Explain
5. What is Intersymbol interference. Discuss about eyediagram.

Reference:

1. Communication Systems by Simon Haykin.
2. Analog and Digital Communications by P. Chakrabarti.