NEHRU COLLEGE OF ENGINEERING AND RESEARCH CENTRE
(NAAC Accredited)
(Approved by AICTE, Affiliated to APJ Abdul Kalam Technological University, Kerala)

DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

COURSE MATERIAL

EE 401 ELECTRONIC COMMUNICATION

VISION OF THE INSTITUTION
To mould true citizens who are millennium leaders and catalysts of change through excellence in education.
MISSION OF THE INSTITUTION

NCERC is committed to transform itself into a center of excellence in Learning and Research in Engineering and Frontier Technology and to impart quality education to mould technically competent citizens with moral integrity, social commitment and ethical values.

We intend to facilitate our students to assimilate the latest technological know-how and to imbibe discipline, culture and spiritually, and to mould them in to technological giants, dedicated research scientists and intellectual leaders of the country who can spread the beams of light and happiness among the poor and the underprivileged.

ABOUT DEPARTMENT

- Established in: 2002
- Course offered: B.Tech Electrical and Electronics Engineering
- M.Tech (Energy Systems)
- Approved by AICTE New Delhi and Accredited by NAAC
- Affiliated to the University of Dr. A P J Abdul Kalam Technological University.

DEPARTMENT VISION

To excel in technical education and research in the field of Electrical & Electronics Engineering by imparting innovative engineering theories, concepts and practices to improve the production and utilization of power and energy for the betterment of the Nation

DEPARTMENT MISSION

1) To offer quality education in Electrical and Electronics Engineering and prepare the students for professional career and higher studies.

2) To create research collaboration with industries for gaining knowledge about real-time problems.

3) To prepare students with sound technical knowledge

4) To make students socially responsible

PROGRAMME EDUCATIONAL OBJECTIVES

1. Graduates shall have a good foundation in the fundamental and practical aspects of Mathematics and Engineering Sciences so as to build successful and enriching careers in the field of Electrical Engineering and allied areas

2. Graduates shall learn and adapt themselves to the latest technological developments in the field of Electrical & Electronics Engineering which will in turn motivate them to excel in their domains and shall pursue higher education and research

3. Graduates shall have professional ethics and good communication ability along with entrepreneurial skills and leadership skills, so that they can succeed in multidisciplinary and diverse fields.

PROGRAM OUTCOME (PO'S)

Engineering Graduates will be able to:

PO 1. Engineering knowledge: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.

PO 2. Problem analysis: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.

PO 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 the public health and safety, and the cultural, societal, and environmental considerations.

PO 4. Conduct investigations of complex problems: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.

PO 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.

PO 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.

PO 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.

PO 8. Ethics: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.

PO 9. Individual and team work: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.

PO 10. Communication: Communicate effectively on complex engineering activities with the engineering community and with 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.

PO 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.

PO 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.

PROGRAM SPECIFIC OUTCOME(PSO'S)

PSO 1: Apply Science, Engineering, Mathematics through differential and Integral Calculus, Complex Variables to solve Electrical Engineering Problems

PSO 2: Demonstrate proficiency in the use of software and hardware to be required to practice electrical engineering profession.

PSO 3. Apply the knowledge of Ethical and Management principles required to work in a team as well as to lead a team.

Course code	Course Name	L-T-P -Credits	Year of Introduction
EE401	Electronic Communication	3-0-0-3	2016

Prerequisite: Nil

Course Objectives

- To introduce the applications of communication technology.
- To understand the methods and techniques used in communication field.

Syllabus:

AM and FM fundamentals-AM and FM transmitters and receivers-Television and radar systems-Digital communication-Satellite communication-Cellular telephone.

Expected outcome

The students will

i. Understand the need of modulation in transferring a signal through either wireless or wired communication systems

ii. Be able to apply analog modulation techniques and receiver fundamentals in analog communication.

iii. Be to apply baseband digital encoding & decoding techniques in the storage / transmission of digital signal through wired channel

iv. Understand the performance of communication systems in the presence of noise and interference

Text Books:

1. Kennedy G., *Electronic Communication Systems*, McGraw-Hill, New York, 2008.
2. Roody and Coolen, *Electronic Communication*, Prentice Hall of India LTD., New Delhi, 2007.

References:

1. William Scheweber, *Electronic Communication Systems*, Prentice Hall of India LTD, New Delhi, 2004.
2. Wayne Tomasi, *Electronic Communication Systems*, Prentice Hall of India LTD, New Delhi, 2004.
3. Frank R. Dungan, *Electronic Communication Systems*, 3/e, Vikas Publishing House, 2002.
4. Simon Haykins, *Communication Systems*, John Wiley, USA, 2006.
5. Bruce Carlson. *Communication Systems*, Tata McGraw Hill, New Delhi, 2001.
6. Taub and Schilling, *Principles of Communication Systems*, McGraw-Hill, New York, 2008.
7. Anokh Singh, *Principles of Communication Engineering*, S. Chand and Company Ltd., Delhi.

Course Plan

Module	Contents	Hours	Sem. Exam Marks
I	**AM and FM fundamentals** AM – Frequency spectrum – vector representation – power relations – generation of AM – DSB, DSB/SC, SSB, VSB FM – frequency spectrum – power relations	6	15%
II	**AM and FM transmitters and receivers** Block diagrams of low power and high power AM transmission - AM receivers: straight receivers super hetrodyne receiver - choice of intermediate frequency - simple AVC circuit Block diagrams of direct FM transmitter and Armstrong transmitter - FM receivers (balanced - slope detector and Foster-Seely discriminator only).	8	15%
FIRST INTERNAL EXAMINATION			

III	**Television and radar systems** Principles of television engineering - Requirements and standards – need for scanning - types of camera tubes and picture tubes - B/W and colour systems - PAL - CCTV - Cable TV-high definition television. Radar and navigation: principle of radar and radar equation, block schematics of pulsed radar.	8	15%
IV	**Digital communication:** Principles of digital communication – - Sampling process-pulse modulation Techniques- sampling process-PAM, PWM and PPM concepts - PCM encoder and decoder Applications of data communication	6	15%
SECOND INTERNAL EXAMINATION			
V	**Satellite communication** Multiple access (MA) techniques-FDMA, TDMA, CDMA, SDMA - applications in satellite communication wire, MA techniques applications in wired communication. in satellite communication, earth station; Fibers – types: sources, detectors used, digital filters, optical link	8	20%
VI	**Cellular telephone -** Basic concepts, frequency reuse, interference cell splitting, sectoring, cell system layout, cell processing. Fibers – types: sources, detectors used, digital filters, optical link: Bluetooth, Zig-Bee, GPS, Wi-Fi, Wi-Max based communication	6	20%
END SEMESTER EXAM			

QUESTION PAPER PATTERN:

Maximum Marks: 100 Exam Duration: 3Hourrs.

Part A: 8 compulsory questions.
One question from each module of Modules I - IV; and two each from Module V & VI.
Student has to answer all questions. (8 x5)=40

Part B: 3 questions uniformly covering Modules I & II. Student has to answer any 2 from the 3 questions: (2 x 10) =20. Each question can have maximum of 4 sub questions (a,b,c,d), if needed.

Part C: 3 questions uniformly covering Modules III & IV. Student has to answer any 2 from the 3 questions: (2 x 10) =20. Each question can have maximum of 4 sub questions (a,b,c,d), if needed.

Part D: 3 questions uniformly covering Modules V & VI. Student has to answer any 2 from the 3 questions: (2 x 10) =20. Each question can have maximum of 4 sub questions (a,b,c,d), if needed.

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QUESTION BANK- MODULE – I

1) Define amplitude modulation.
2) Draw the frequency spectrum of SSB
3) Draw the phasor representation of AM
4) What is DSB SC AM?
5) Mention the advantages of VSB AM.
6) What is amplitude modulation? Draw its graphical and phasor representations and derive its power calculation.
7) What is double side band suppressed carrier AM? Compare it with AM with carrier with respective frequency spectrum, phasor representation and power calculation.
8) Mention the purpose of single side band suppressed carrier AM. Compare it with AM with carrier with respective frequency spectrum, phasor representation and power calculation.
9) What is vestigial side band modulation? Explain.
10) Give the comparison between AM with carrier, DSB SC and SSB SC modulation method in detail.

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QUESTION BANK- MODULE – II

1) Compare AM and FM transmitters
2) Explain low power AM transmission with block diagram
3) Briefly explain Armstrong transmitter with a neat block diagram.
4) Explain the choice of intermediate frequency in AM receivers.
5) Explain about super heterodyne receiver with a neat diagram.
6) Explain in detail about foster seely discriminator.
7) Explain in detail about balanced slope detector.
8) Explain the following 1) direct FM transmitter 2)Armstrong transmitter
9) With neat block diagram explain the types of AM transmitter.
10) Explain in detail about AVC circuit.

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QUESTION BANK- MODULE – III

1) Write a short note on principle of radar.
2) What is radar? Also write the radar equation.
3) What are the basic components of a radar system?
4) What types of camera tubes and picture tubes are used in television engineering?
5) Compare black and white and colour systems.
6) What are the principles of television engineering?
7) Explain in detail about CCTV.
8) What is meant by high definition television
9) What are the requirements and standards need for scanning? Also explain PAL
10) With a neat block diagram explain about pulsed radar.

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QUESTION BANK- MODULE – IV

1) What is sampling process?
2) What are the pulse modulation techniques?
3) What are the principles of digital communication?
4) Give the applications of data communication.
5) Distinguish between PWM and PPM
6) Explain PCM encoder and decoder.
7) Describe PAM concept
8) Explain PWM with necessary sketches.
9) Compare PAM and PWM
10) Explain the sampling process of PAM, PWM and PPM.

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QUESTION BANK- MODULE – V

1) What are the advantages of satellite communication?
2) Mention the applications of satellites
3) What is mean by multiple axis techniques?
4) Applications of MA techniques in wired communications
5) Briefly explain digital filters
6) Compare FDMA and TDMA
7) How MA techniques are applied in satellite communication and earth station?
8) Explain in detail about the sources and detectors in fiber communications.
9) Describe CDMA with a neat diagram
10) Explain in detail about SDMA

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QUESTION BANK- MODULE – VI

1) Explain the basic concepts of cellular telephone
2) Define cell splitting.
3) Write a short note on Bluetooth.
4) List the applications of ZigBee.
5) What is cell sectoring?
6) Explain the concept and principle of operation of mobile communication in detail.
7) What is ZigBee? Explain its functions and ZigBee stacks.
8) With neat sketches explain the concept of cellular architecture
9) Explain in detail about frequency reuse and interference cell splitting.
10) What is Wi-Fi? How does it work?

Module 1

AM and FM Fundamentals

AM AND FM FUNDAMENTALS

Ques 1) Define AM. Also write the advantages and disadvantages of AM.

Ans: AM (Amplitude Modulation)

Amplitude Modulation (AM) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to the waveform being transmitted. That waveform may, **for example**, correspond to the sounds to be reproduced by a loudspeaker, or the light intensity of television pixels. This technique contrasts with frequency modulation, in which the frequency of the carrier signal is varied, and phase modulation, in which its phase is varied.

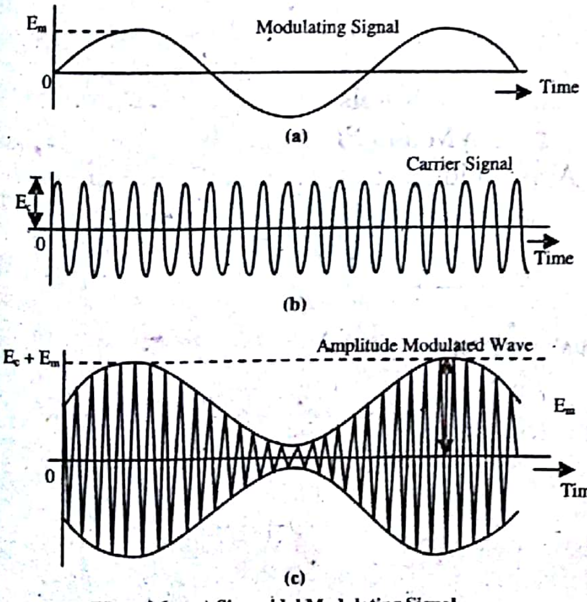


Figure1.1: a) Sinusoidal Modulating Signal
b) Sinusoidal High Frequency Carrier
c) Amplitude Modulated Signal

The modulating signal modulates amplitude, frequency or phase of the carrier according to its variations in amplitude. This results in amplitude, frequency or phase modulation. The frequency and phase modulation is also called angle modulation. In amplitude modulation, the amplitude of a carrier signal is varied according to variations in the amplitude of modulating signal. **Figure 1.1** shows the modulating signal in **figure 1.1 (a), figure 1.1 (b)** shows high frequency carrier and **figure 1.1 (c)** shows amplitude modulated signal. In **figure 1.1 (c),** observe that the carrier frequency remains same, but its amplitude varies according to amplitude variations of the modulating signal.

Advantages of AM

1) It is simple to implement.
2) It can be demodulated using a circuit consisting of very few components.
3) AM receivers are very cheap as no specialised components are needed.

Disadvantages of AM

1) It is not efficient in terms of its power usage
2) It is not efficient in terms of its use of bandwidth, requiring a bandwidth equal to twice that of the highest audio frequency
3) It is prone to high levels of noise because most noise is amplitude based and obviously AM detectors are sensitive to it.

Ques 2) Derive equation of AM signal. Also plot a frequency spectrum of an AM wave with the help of its mathematical analysis.

Or

Give a mathematical analysis of sinusoidal AM. Also define modulation index and percentage modulation?

Ans: Sinusoidal AM / Equation of AM Signal

The amplitude modulation technique, in which both carrier and base band signal are sinusoidal, is called sinusoidal AM. Thus, let the modulating signal be e_m and given by,

$$e_m = E_m \sin\omega_m t \quad .....(1)$$

And carrier signal can be represented by e_c as,

$$e_c = E_c \sin \omega_c t \quad .....(2)$$

Here,

E_m = Maximum amplitude of modulating signal
E_c = Maximum amplitude of carrier signal
ω_m = Frequency of modulating signal
ω_c = Frequency of carrier signal

Using the above mathematical expressions for modulating and carrier signals, we can create a new mathematical expression for the complete modulated wave. It is given as,

$$E_{AM} = E_c + e_m$$
$$= E_c + E_m \sin\omega_m t$$

By putting e_m from equation (1)

The instantaneous value of the amplitude modulated wave can be given as,

$$e_{AM} = E_{AM} \sin\theta = E_{AM} \sin\omega_c t$$

$$\therefore e_{AM} = (E_c + E_m \sin\omega_m t) \sin\omega_c t \quad \text{.....(3)}$$

This is an equation of AM wave.

Modulation Index and Percentage Modulation

The ratio of amplitudes of baseband to carrier signal is called **Modulation Index.** It is also called modulation factor, modulation coefficient or the degree of modulation. It is a number lying between 0 and 1 (maximum, when $E_m = E_c$), and it is very often expressed as a percentage and called the **percentage modulation.**

$$\text{Modulation Index, } m = \frac{E_m}{E_c} \quad \text{..... (4)}$$

$$\text{Percentage Modulation Index}, \%(m) = m \times 100$$
$$= \left(\frac{E_m}{E_c}\right) \times 100$$

Hence value of modulation index will be equal to 1.

Frequency Spectrum and Bandwidth

The modulated carrier has new signals at different frequencies, called side frequencies or sidebands. They occur above and below the carrier frequency.

That is, $f_{USB} = f_c + f_m$
$f_{LSB} = f_c - f_m$

Here,
f_c = Carrier frequency and
f_m = Modulating signal frequency
f_{LSB} = Lower sideband frequency
f_{USB} = Upper sideband frequency

Consider the expression of AM wave given by equation (3), i.e.,

$$e_{AM} = (E_c + E_m \sin\omega_m t) \sin\omega_c t \quad \text{.....(5)}$$

Since, $m = \frac{E_m}{E_c}$ from equation (4). Hence, $E_m = mE_c$.

Putting this value of E_m in above equation we get,

$$e_{AM} = (E_c + mE_c \sin\omega_m t) \sin\omega_c t$$
$$= E_c (1 + m \sin\omega_m t) \sin\omega_c t$$
$$= E_c \sin\omega_c t + m E_c \sin\omega_m t \sin\omega_c t \quad \text{.....(6)}$$

As,

$$\sin(A)\sin(B) = \frac{1}{2}\cos(A-B) - \frac{1}{2}\cos(A+B)$$

Applying this result to last term in above equation we get,

$$e_{AM} = E_c \sin\omega_c t + \frac{mE_c}{2}\cos(\omega_c - \omega_m)t - \frac{mE_c}{2}\cos(\omega_c + \omega_m)t \quad \text{..... (7)}$$

In the above equation, the first term represents unmodulated carrier, the second term represents lower side-band and last term represents upper sideband. Since, $\omega_c = 2\pi f_c$ and $\omega_m = 2\pi f_m$. Hence above equation can also be written as,

$$e_{AM} = E_c \sin 2\pi f_c t + \frac{mE_c}{2}\cos 2\pi(f_c - f_m)t - \frac{mE_c}{2}\cos 2\pi(f_c + f_m)t \quad \text{..... (8)}$$

$$e_{AM} = E_c \sin 2\pi f_c t + \frac{mE_c}{2}\cos 2\pi f_{LSB} t + \frac{mE_c}{2}\cos 2\pi f_{USB} t \quad \text{..... (9)}$$

From this equation the frequency spectrum of AM wave can be drawn as shown in **figure 1.2**:

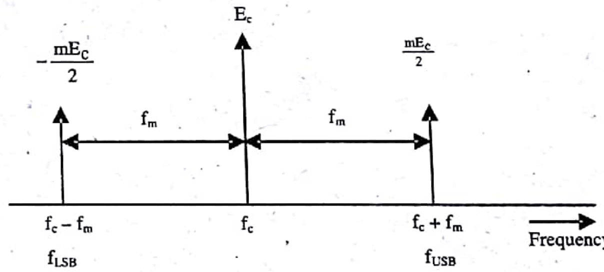


Figure 1.2: Frequency Domain Representation of AM Wave

This contains full carrier and both the sidebands; hence it is also called **Double Side-Band Full Carrier (DSBFC) system.**

Bandwidth of the signal can be obtained by taking the difference between highest and lowest frequencies. From **figure 1.2,** bandwidth of AM wave can be obtained as,

$$BW = f_{USB} - f_{LSB} = (f_c + f_m) - (f_c - f_m)$$

$$\therefore BW = 2f_m \quad \text{.....(10)}$$

Thus **bandwidth of AM signal is twice the maximum frequency of modulating signal.** For wave is an over-modulated **figure** shows the waveform for a sinusoidal baseband signal.

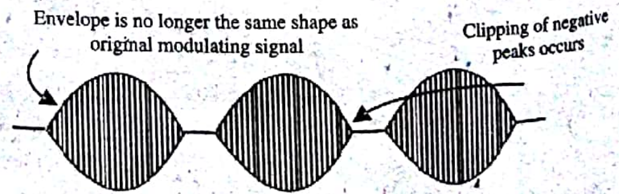


Figure 1.3

Ques 3) Discuss the vector representation of amplitude modulation.

Ans: Vector Representation of AM with Carrier

The vector representation of an AM with carrier is shown in **Figure 1.4**.

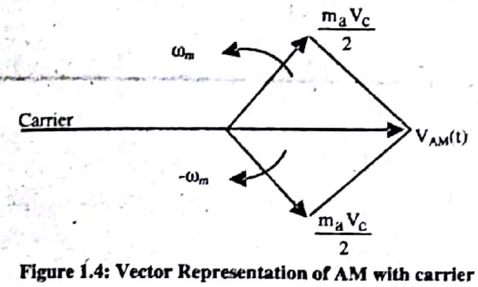


Figure 1.4: Vector Representation of AM with carrier

It is the easy way of representation of an AM wave, where V_c is the carrier wave phasor, taken as reference phasor. The two side band having a frequency of $(\omega_c + \omega_m)$ and $(\omega_c - \omega_m)$ are represented by two phasors rotating in opposite direction with angular frequency of ω_m. The resultant phasor is V(t). It depends on the position of the side-band phasor and carrier wave phasor.

Ques 4) Derive an expression for modulation index of sinusoidal AM.

Or

Given amplitude modulated wave at the detector end, give a mathematical analysis for the calculation of modulation index of given AM wave?

Ans: Calculation for Modulation Index of Sinusoidal AM

Consider the AM signal, the more descriptive form of the given waveform is shown in **figure 1.5**.

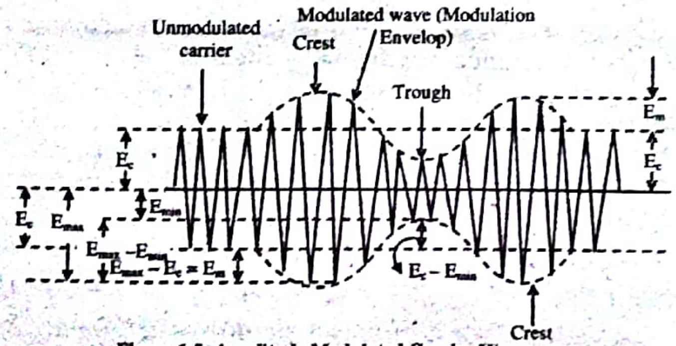


Figure 1.5: Amplitude Modulated Carrier Wave

We know that,

Modulation Index, $m = \frac{E_m}{E_c}$

Percentage Modulation,

$$\%m = m \times 100 = \left(\frac{E_m}{E_c}\right) \times 100$$

Another way of expressing the modulation index is in terms of the maximum and minimum values of the amplitude of the modulated carrier wave. This is shown in the **figure 1.6**.

From the figure we know that:

$$2E_m = E_{max} - E_{min}$$

$$E_m = \left(\frac{E_{max} - E_{min}}{2}\right) \quad \text{..... (1)}$$

And $E_c = E_{max} - E_m$ .....(2)

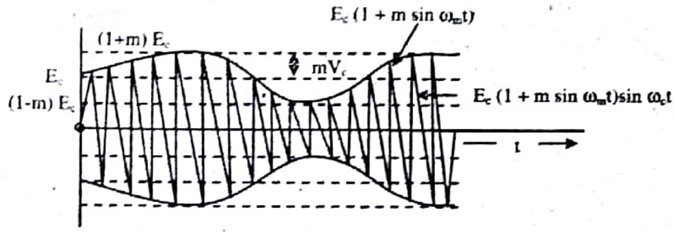


Figure 1.6: Amplitude Modulated Sine Wave with m < 1

By putting value of E_m from equation (1) to equation (2), we get:

$$E_c = E_{max} - \left(\frac{E_{max} - E_{min}}{2}\right)$$

$$E_c = \left(\frac{E_{max} + E_{min}}{2}\right) \quad \text{..... (3)}$$

Taking the ratio of equation (1) and (2),

$$m = \frac{E_m}{E_c} = \frac{\left(\frac{E_{max} - E_{min}}{2}\right)}{\left(\frac{E_{max} + E_{min}}{2}\right)}$$

$$\therefore \quad m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$$

This equation gives the technique of calculating modulation index from AM wave. For distortion less transmission m must be smaller than 1, practically preferred between 0.2 and 0.8.

Ques 5) Calculate the modulation index and percentage modulation if instantaneous voltages of modulating signal and carrier are 20 sin ω_mt and 80 sin ω_ct, respectively.

Ans: From the given instantaneous equation we have,

$E_m = 20$ and $E_c = 80$.

Hence modulation index will be,

$$m = \frac{E_m}{E_c} = \frac{20}{80} = 0.25$$

$\%$ modulation $= m \times 100 = 0.25 \times 100 = 25\%$

Ques 6) An amplitude modulated wave is represented by the expression
$e_m = 10(1 + 0.6 \cos 6280t) \sin 211 \times 10^4 t$ Volts. Calculate:

1) **The minimum and maximum amplitude of the A.M. wave.**
2) **The frequencies and amplitudes of the components contained in the modulated wave.**

Ans: The instantaneous voltage of A.M. wave is given by,

$$e_m = E_c(1 + m\cos\omega_m t)\cos\omega_c t$$

The given modulated wave is represented by the expression

$$e_m = 10(1 + 0.6 \cos 6280t) \sin 211 \times 10^4 t \text{ volts}$$

The Maximum and Minimum Amplitude of A.M. Wave

$E_m(max) = E_c(1 + m) = 10(1 + 0.6) = 16$ volts

$E_m(min) = E_c(1 - m) = 10(1 + 0.6) = 4$ volts

Frequency and Amplitudes of Compounds

Carrier wave,

$$f_c = \frac{\omega_c}{2\pi} = \frac{211 \times 10^4}{2\pi} = 33.43 \times 10^4 \text{ Hz}$$

Signal wave $f_m = \frac{\omega_m}{2\pi} = \frac{6280}{2\pi} = 1000\text{Hz}$

Carrier wave amplitude $E_c = 10$ volts

Signal wave amplitude E_m is given by $m = k\frac{E_m}{E_c}$

Where, k is the proportionality constant. Taking k = 1, we have

$$m = \frac{E_m}{E_c} \text{ or } E_m = mE_c = 0.6 \times 10 = 6 \text{ volts}$$

Ques 7) Write an expression for the effective voltage and current of a sinusoidal AM signal.

Ans: Effective Voltage and Current

In Amplitude Modulated systems, the modulated and unmodulated currents are necessary to calculate the modulation index of an AM wave. The effective or rms value of voltage E_t of the modulated wave can be find by the equation,

$$\frac{E_t^2}{R} = P_t$$

Similarly, the effective or root mean square voltage E_c of carrier can be find by the equation, $\frac{E_c^2}{R} = P_c$

Now, using the relation, $P_t = P_c\left(1 + \frac{m^2}{2}\right)$

We get, $E_t = E_c\left(\sqrt{1 + \frac{m^2}{2}}\right)$

A similar agreement applied to currents, yields

$$I_t = I_c\left(\sqrt{1 + \frac{m^2}{2}}\right)$$

Where, I_t is the rms current of modulated wave and I_c is the rms current of unmodulated carrier. The maximum power in the AM wave is $P_t = 1.5P_c$, when m=1. This is important, because it is the maximum power that relevant amplifiers must be capable of handling without distortion.

Ques 8) Discuss the power relations of sinusoidal AM.

Or

Calculate the average power of an AM system also derive the expression for:

1) **Carrier Power**
2) **Power in Side Band**
3) **Total Power of AM signal**
4) **Transmission Efficiency**
5) **Current Relation in AM**

Ans: Power Relations of Sinusoidal AM

AM wave has three components, unmodulated carrier, lower sideband and upper sideband. Therefore the total power of AM wave is the sum of the carrier power, P_C and powers in the two sidebands P_{USB}, and P_{LSB}. It is given as,

$$P_T = P_c + P_{USB} + P_{LSB} \quad \text{..... (1)}$$

Where all three voltages represent r.m.s. values and resistance R is a characteristics impedance of antenna in which the power is dissipated.

Carrier Power (P_c)

The carrier power is given as,

$$P_c = \frac{E_c^2}{R}$$

$\therefore$ The average carrier power $= \frac{(E_c/\sqrt{2})^2}{R}$

$$P_c = \frac{E_c^2}{2R} \quad \text{..... (2)}$$

Power in Side Band

The power in upper side band and lower side band is given as,

$$P_{LSB} = P_{USB} = \frac{E_{SB}^2}{R}$$

$$=\left(\frac{\frac{mE_c}{2}}{\sqrt{2}}\right)^2 \times \frac{1}{R}$$

$$\because\ E_{SB}=\frac{mE_c}{2}$$

$$P_{LSB}=P_{USB}=\frac{m^2E_c^{\ 2}}{8R} \qquad \text{..... (3)}$$

From the equation (3) it is realised that transmission efficiency increases as modulation index increase. In other words, increase in modulation index increase the information power (P_{LSB} + P_{USB}) in the total power. As a result strong signal is received at the receiving end for large modulation index. Hence, it is desirable to have higher modulation index. However, to avoid over modulation it should be less than 1.

$$\therefore P_{LSB}=P_{USB}=\frac{m^2E_c^{\ 2}}{8R}=\frac{m^2}{4}\times\frac{E_c^{\ 2}}{2R}$$

We know that, $\frac{E_c^{\ 2}}{2R}=P_c$

$$\therefore P_{LSB}=P_{USB}=\frac{m^2}{4}P_c \qquad \text{.....(4)}$$

Total Power or Average Power of AM signal
The average or total power of amplitude modulated signal can be obtained by adding carrier power and side band power as explained in equation (1) above:

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

$$=\frac{E_c^{\ 2}}{2R}\left(1+\frac{m^2}{4}+\frac{m^2}{4}\right)=\frac{E_c^{\ 2}}{2R}\left(1+\frac{m^2}{2}\right)$$

$$P_t=P_c\left(1+\frac{m^2}{2}\right) \qquad \text{..... (5)}$$

This equation (5) **relates the total power** of amplitude modulated wave with the power of unmodulated carrier. The maximum possible value of m in the amplitude modulated wave is 1, therefore from above equation it can be seen that the maximum total power of amplitude modulated wave is 1.5 P_c.

Transmission Efficiency
This transmission efficiency of the AM wave is defined as the ratio of the transmitted power which contains the information (i.e., sum of lower side band and upper sideband power) to the total transmitted powder (P_T).

$$\therefore \text{Transmission Efficiency}=\eta=\frac{P_{LSB}+P_{USB}}{P_{Total}}$$

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

$$\therefore \eta=\frac{m^2}{2+m^2} \qquad \text{.....(6)}$$

$\therefore$ The percentage transmission efficiency is given as,

$$\%n=\frac{m^2}{2+m^2}\times 100\%$$

If,
m = 1,

$$\%\eta=\frac{1}{3}\times 100=33.3\%$$

From this, it is calculated that only 33.3% of energy is used and the remaining power is wasted by the carrier transmission along with the side bands.

Current Relation in AM
From equation (5),

$$P_T=P_C\left[1+\frac{m^2}{2}\right]$$

We know that,

$$P_T=I_T^2.R$$

$$I_T^2.R=I_C^2.R$$

Hence,

$$I_T^2=I_C^2\left[1+\frac{m^2}{2}\right]$$

$$\therefore I_T=I_C\sqrt{\left[1+\frac{m^2}{2}\right]}$$

Where,
I_T is the total current, and I_C is the carrier current.

Ques 9) A 400 watt carrier is modulated to a depth of 80%. Calculate the total power in the modulated wave.

Ans:

$$P_T=P_c\left(1+\frac{m^2}{2}\right)=400\left(1+\frac{(0.8)^2}{2}\right)=528\text{ W}$$

Ques 10) A 400 watt carrier is modulated to a depth of 75%. Calculate total power in modulated wave.

Ans: Given, depth of modulation, m=75%
m=0.75
P_c=400 W

We know that,

$$P_m = P_c\left(1+\frac{m^2}{2}\right)$$

$$P_m = P_c\left(1+\frac{(0.75)^2}{2}\right) = 512.5\text{ W}$$

Ques 11) A 460 watt carrier is modulated to a depth of 65%. Calculate total power of modulated wave.

Ans: Given, depth of modulation m=65%

m=0.65

P_c=460 W

We know that,

$$P_m = P_c\left(1+\frac{m^2}{2}\right) = P_c\left(1+\frac{(0.65)^2}{2}\right)$$

$$P_m = 557.175\text{ W}$$

Ques 12) Explain in detail the generation of AM waves and also discuss the drawback of AM.

Ans: Generation of AM Waves

The circuits that are used to generate an amplitude modulated wave are called Amplitude Modulated circuits.

The most commonly used modulator circuits are:

1) **Square-Law Modulator Circuit:** Generation of AM Waves using the square law modulator circuit can be understood by observing the square law modulator circuit shown in **figure 1.7.** The generation of AM wave is based on the **principle** that, when a nonlinear element such as a diode is suitably biased and operated in a restricted portion of its characteristic curve, i.e., the signal applied to the diode is relatively weak, the transfer characteristic of diode load resistor combination can be represented closely by a square law i.e.,

$$V_o(t) = a_1V_i(t) + a_2 V_i^2(t)$$

Where, V_i is the input voltage and a_1 and a_2 are constants.

The **figure 1.7** below shows the general diagram of the square law modulator that consists of three main components:

i) **Non-Linear Device:** The non-linear devices are the devices that do not show a linear relationship between current flowing through it and voltage across it, the simple diode is an example of a non-linear device.

ii) **Band-Pass Filter:** A band-pass filter is a circuit that passes a range of frequencies and attenuates others.

iii) **Carrier Source and Modulating Signal:** The carrier source is the circuit that generates carrier signal. The base band signal or information signal is called modulating signal, x(t).

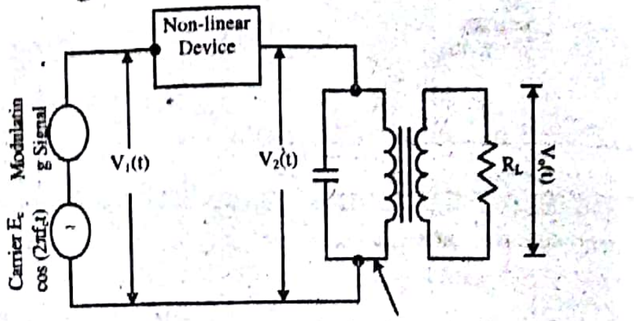


Figure 1.7: Square Law Modulator

The modulating signal and carrier are connected in series with each other and their sum $V_1(t)$ is applied at the input of the non-linear device.

Thus,

$$v_1(t) = x(t) + E_c\cos(2\pi f_c t) \quad .....(1)$$

The input output relation for non-linear device is as under:

$$v_2(t) = av_1(t) + bv_1^2(t) \quad ......(2)$$

Where, a and b are constants.

Substituting the expression in equation (1) in equation (2):

$$v_2(t) = a[x(t) + E_c\cos(2\pi f_c t)] + b[x(t) + E_c\cos(2\pi f_c t)]^2$$

Or,

$$v_2(t) = ax(t) + aE_c\cos(2\pi f_c t) + b\left[x^2(t) + 2x(t)\cos(2\pi f_c t) + E_c^2\cos^2(2\pi f_c t)\right]$$

$$v_2(t) = ax(t) + bx^2(t) + bE_c^2\cos^2(2\pi f_c t) + aE_c\cos(2\pi f_c t) + 2bx(t)E_c\cos(2\pi f_c t)$$

The LC tuned circuit shown above acts as a band pass filter. Its frequency response is tuned to frequency f_c and its bandwidth is equal to $2f_m$.

This bandpass filter eliminates the frequencies that are out of range thus equation $V_o(t)$ becomes,

$$V_o(t) = aE_c\cos(2\pi f_c t) + 2bx(t)E_c\cos(2\pi f_c t)$$

Or,

$$V_o(t) = [aE_c + 2bx(t)E_c]\cos(2\pi f_c t)$$

Therefore,

$$V_o(t) = aE_c\left[1+\frac{2b}{a}x(t)\right]\cos(2\pi f_c t) \quad ......(3)$$

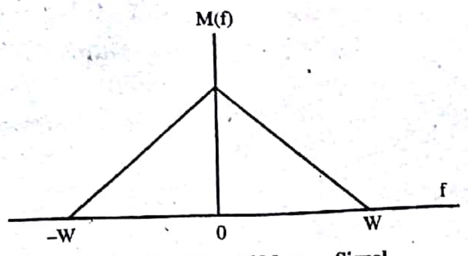


Figure 1.8: Spectrum of Message Signal

Comparing this with the expression for standard AM wave i.e,

$$s(t) = E_c[1 + mx(t)]\cos(2\pi f_c t)$$

The expression for $V_o(t)$ of equation (3) represents an AM wave with modulation index, m=(2b/a). Hence, the square law modulator produces an AM wave.

Assuming a message signal m (t), which is band limited to the interval

$W \le f \le W$

The Fourier transform of output voltage $V_o(t)$ is given by,

$V_o(f) = a_1 A_c/2[\delta(f-f_c) + \delta(f + f_c)] + a_2 A_c[M(f-f_c) + M(f+f_c)]$

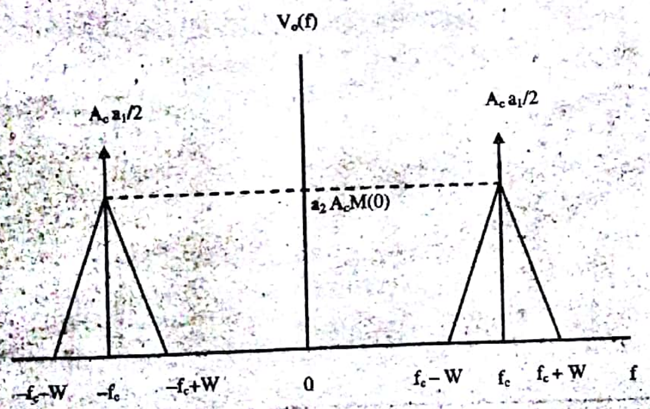


Figure 1.9: Spectrum of AM Signal

The AM spectrum consists of two impulse functions which are located at f_c and $-f_c$ and weighted by $A_c a_1/2$ and $a_2 A_c/2$, two USBs, band of frequencies from f_c to f_c +W and band of frequencies from $-f_c$ –W to f_c, and two LSBs, band of frequencies from f_c –W to f_c and $-f_c$ to $-f_c$ + W.

2) **Switching Modulator Circuit:** The switching modulator using a diode has been shown in **figure 1.10(a)**. This diode is assumed to operate as a switch. **Figure 1.10(b)** shows the transfer characteristics of diode. The modulating signal x(t) and the sinusoidal carrier signal c(t) are connected in series with each other.

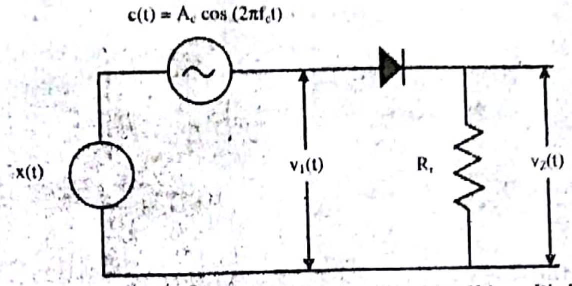


Figure 1.10(a): Switching Modulator Using a Diode

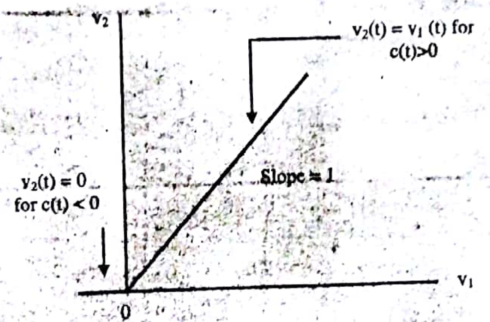


Figure 1.10(b): The Transfer Characteristic of Diode

Thus, the input voltage to the diode is given by,

$v_1(t) = c(t) + x(t) = E_c\cos(2\pi f_c t) + x(t)$

The amplitude of carrier is much larger than that of x(t) and c(t) decides the status of the diode (ON or OFF).

Considering an ideal diode, it acts as a closed switch when it is forward biased in the positive half cycle of the carrier and offers zero impedance. Whereas it acts as an open switch when it is reverse biased in the negative half cycle of the carrier and offers infinite impedance.

Therefore, the output voltage $v_2(t) = v_1(t)$ in the positive half cycle of c(t) and $v_2(t) = 0$ in the negative half cycle of c(t).

Hence, $v_2(t) = v_1(t)$ for $c(t) > 0$

$v_2(t) = 0$ for $c(t) < 0$

In other words, the load voltage $v_2(t)$ varies periodically between the values $v_1(t)$ and zero at the rate equal to carrier frequency f_c. Expressing $v_2(t)$ mathematically as under:

$$v_2(t) = v_1(t)n_p(t) = [x(t) + E_c\cos(2\pi f_c t)]n_p(t) \quad .....(4)$$

Where, $n_p(t)$ is a periodic pulse train of duty cycle equal to one half cycle period i.e. $T_0/2$ (where $T_0 = 1/f_c$).

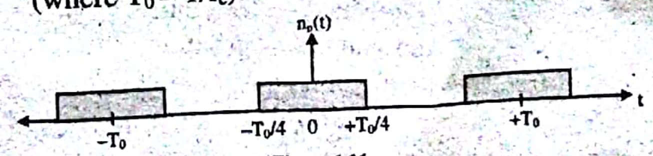


Figure 1.11

$n_p(t)$ can be expressed with the help of Fourier series as,

$$n_p(t) = \frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos[2\pi f_c t(2n-1)] \quad \text{..... (5)}$$

$$n_p(t) = \frac{1}{2} + \frac{2}{\pi}\cos(2\pi f_c t) + \text{odd harmonic components} \quad \text{..... (6)}$$

Substituting $n_p(t)$ into equation (4), we get

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

Therefore,

$$v_2(t) = [x(t) + E_c \cos(2\pi f_c t)] \left\{\frac{1}{2} + \frac{2}{\pi}\cos(2\pi f_c t) + \text{odd harmonics}\right\} \quad \text{..... (7)}$$

The odd harmonics in this expression are unwanted, and therefore, are assumed to be eliminated.

Hence,

$$v_2(t) = \underbrace{\frac{1}{2}x(t)}_{\text{Modulating Signal}} + \underbrace{\frac{1}{2}E_c\cos(2\pi f_c t) + \frac{2}{\pi}x(t)\cos(2\pi f_c t)}_{\text{AM Wave}} + \underbrace{\frac{2E_c}{\pi}\cos^2(2\pi f_c t)}_{\text{Second harmonic of carrier}}$$

In this expression, the first and the fourth terms are required terms whereas the second and third term together represents the AM wave. The second and third terms are taken together, we obtain

$$v_2(t) = \frac{E_c}{2}\left[1 + \frac{4}{\pi E_c}x(t)\right]\cos(2\pi f_c t) + \text{unwanted terms}$$

This is the required expression for the AM wave with $m = [4/\pi E_c]$. Using a band-pass filter (BPF), the unwanted terms can be eliminated.

Drawbacks of AM

In conventional AM double side-band system, the carrier signal does not carry information. The information is contained in the side bands. Due to the nature of this system, the drawbacks are as follows:

1) Carrier power constitutes two-thirds or more of the total transmitted power.
2) Both side bands contain the same information. Transmitting both side bands is redundant and thus causes it to utilise twice as much bandwidth as needed with single side-band system.
3) Conventional AM is both power and bandwidth inefficient.

Ques 13) Describe in brief about the various modulation technique of AM?

Ans: Modulation Technique of AM

The common amplitude modulation techniques are as follows:

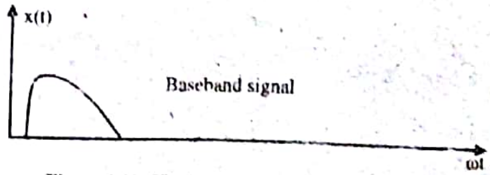


Figure 1.12: The Base Band or Message Signal

1) **Double-Sideband Modulation (DSB):** The simplest modulation method to implement is DSB, in which the translated spectrum of the message signal or base band signal is transmitted without further modification. Consider the base band signal be x(t) as shown aside:

The different types of DSB modulation signals are:

i) **Double-Sideband Modulation with Carrier (DSB-WC):** It is used on the AM radio broadcasting band. In this type of amplitude modulation technique the side-bands as well as the carrier signals are transmitted. The technique is also called **Full AM Modulation,** the modulated signal is shown in **figure 1.13.**

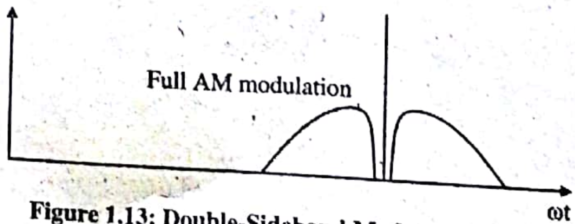


Figure 1.13: Double-Sideband Modulation with Carrier (DSB-WC)

ii) **Double-Sideband Suppressed-Carrier Transmission (DSB-SC):** In this type of amplitude modulation technique only the side-bands are transmitted. Double-sideband suppressed-carrier transmission (DSB-SC) is transmission in which frequencies produced by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed, the modulated signal is shown in **figure 1.14** below:

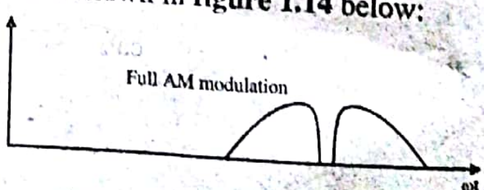


Figure 1.14: Double-Sideband Suppressed Carrier Modulation (DSB-SC)

In the DSB-SC modulation, unlike in AM, the wave carrier is not transmitted; thus, much of the power is distributed between the sidebands, which imply an increase of the cover in DSB-SC, compared to AM, for the same power used. It is used for radio data systems.

iii) **Double-Sideband Reduced Carrier Transmission (DSB-RC):** It is a technique of modulation in which, the frequencies produced by amplitude modulation are symmetrically spaced above and below the carrier and the carrier level is reduced for transmission at a fixed level below that, which is provided to the modulator.

2) **Single-Sideband Modulation (SSB, or SSB-AM):** Single sideband is widely used for HF (High Frequency) communications. It is formed by taking a signal that has the carrier and one sideband removed. In this way it becomes far more efficient in terms of both spectrum and power.

Types of SSB

The different types of SSB are as follows:

i) **SSB with Carrier (SSB-WC):** In this type of modulation only one of the side band is transmitted with the carrier signal. Usually never preferred due to increased power consumption. The **figure 1.15** shows the modulated signal with Lower Side Band (LSB) and Upper Side Band (USB). Either LSB with carrier or USB with carrier is transmitted.

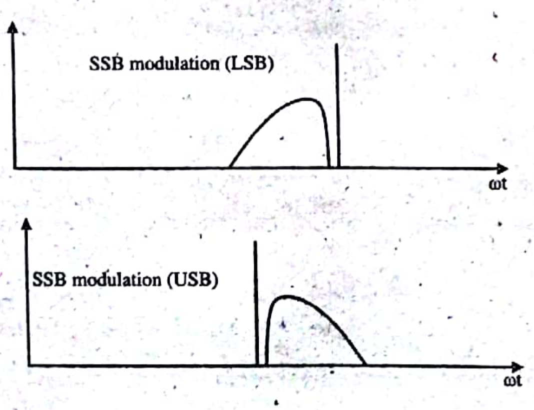


Figure 1.15: SSB with Carrier (SSB-WC)

ii) **SSB Suppressed Carrier Modulation (SSB-SC):** In this type of modulation only one of the side band (either LSB or USB) is transmitted without the carrier signal. Single-sideband suppressed carrier modulation reduces the bandwidth to half, and the power wasted on a carrier, at the cost of increased device complexity and more difficult tuning at the receiver. The **figure 1.16** shows the modulated signal.

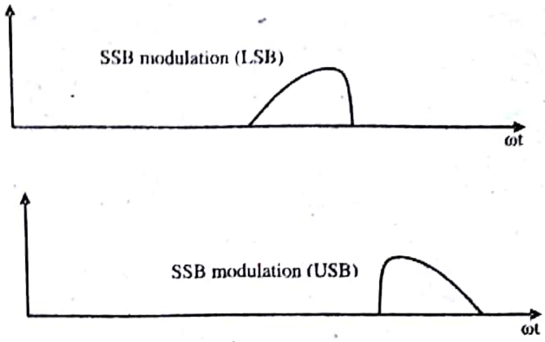


Figure 1.16: SSB Suppressed Carrier Modulation (SSB-SC)

3) **Vestigial Sideband Modulation (VSB, or VSB-AM):** The SSB-SC transmission requires a filter with sharp characteristics for signal demodulation which is hard and expensive to build, thus VSB technique is used. A vestigial sideband (in radio communication) is a sideband that has been only partly cut off or suppressed. Television broadcasts (in analog video formats) use this method if the video is transmitted in AM, due to the large bandwidth used.

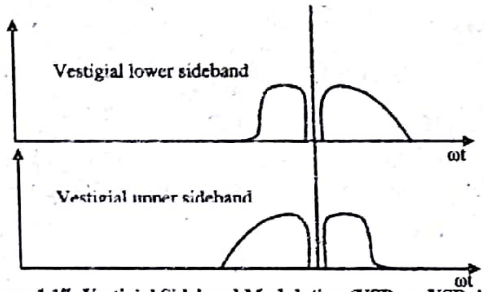


Figure 1.17: Vestigial Sideband Modulation (VSB, or VSB-AM)

4) **Quadrature Amplitude Modulation (QAM):** Quadrature amplitude modulation (QAM) is both an analog and a digital modulation scheme. It conveys two analog message signals, by changing (using modulation) the amplitudes of two carrier waves. The two carrier waves of the same frequency, usually sinusoids, are out of phase with each other by 90° and are thus called quadrature carriers or quadrature components, thus the name of scheme is quadrature amplitude modulation.

Ques 14) Discuss about the DSBSC modulator of AM wave.

Or

Explain balanced modulator and ring modulator of AM wave.

Ans: DSBSC Modulators

The following two modulators generate DSBSC wave.

1) **Balanced Modulator:** The block diagram of the balanced modulator is shown in **figure 1.18.**

Balanced modulator consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as Balanced modulator.

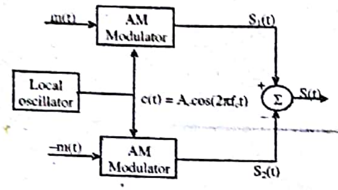


Figure: 1.18

The same carrier signal $c(t) = A_c \cos(2\pi f_c t)$ is applied as one of the inputs to these two AM modulators. The modulating signal m (t) is applied as another input to the upper AM modulator. Whereas, the modulating signal m (t) with opposite polarity, i.e., –m(t) is applied as another input to the lower AM modulator.

Output of the upper AM modulator is

$s_1 (t) = A_c [1 + k_a m (t)] \cos(2\pi f_c t)$

Output of the lower AM modulator is

$s_2 (t) = A_c [1 + k_a m (t)] \cos(2\pi f_c t)$

We get the DSBSC wave s (t) by subtracting s_2 (t) from s_1 (t). The summer block is used to perform this operation. s_1 (t) with positive sign and s_2 (t) with negative sign are applied as inputs to summer block. Thus, the summer block producer an output s (t) which is the difference of s_1 (t) and s_2 (t).

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

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

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

We know the standard **Equation** of DSBSC wave is

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

By comparing the output of summer block with the standard Equation of DSBSC wave, we will get the scaling factor as $2k_a$.

2) **Ring Modulator:** The block diagram of the ring modulator is shown in **figure 1.19**.

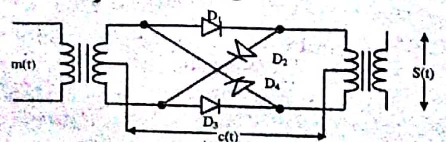


Figure: 1.19

In this diagram, the four diodes D_1, D_2, D_3 and D_4 are connected in the ring structure. Hence, this modulator is called as the ring modulator. Two center tapped transformers are used in this diagram. The message signal m (t) is applied to the input transformer. Whereas, the carrier signals c(t) is applied between the two center tapped transformers. For positive half cycle of the carrier signal, the diodes D_1 and D_3 are switched ON and the other two diodes D_2 and D_4 are switched OFF. In this case, the message signal is multiplied by + 1.

For negative half cycle of the carrier signal, the diodes D_2 and D_4 are switched ON and the other two diodes D_1 and D_3 are switched OFF. In this case, the message signal is multiplied by -1. This results in 180° phase shift in the resulting DSBSC wave. From the above analysis, we can say that the four diodes D_1, D_2, D_3, and D_4 are controlled by carrier signal. If carrier is a square wave, then the Fourier series representation of c(t) is represented as

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

We will get DSBSC wave s(t), which is just the product of carrier signal c(t) and the message signal m(t) i.e.,

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

The above Equation represents DSBSC wave, which is obtained at output transformer of the ring modulator. DSBCS modulators are also called as product modulators as they produce the output, which is product of two input signals.

Ques 15) Discuss about the DSBSC demodulators of AM wave.

Or

Give the brief description of Costas Loop and Coherent Detector demodulator of AM wave.

Ans: DSBSC Demodulators

The process of extracting an original message signal from DSBSC wave is known as detection or demodulation of DSBSC. Following demodulators (detectors) are used for demodulating DSBSC wave:

1) **Coherent Detector:** Here, the same carrier signal (which is used for generating DSBSC signal) is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. The block diagram of the coherent detector is shown in **figure 1.20**.

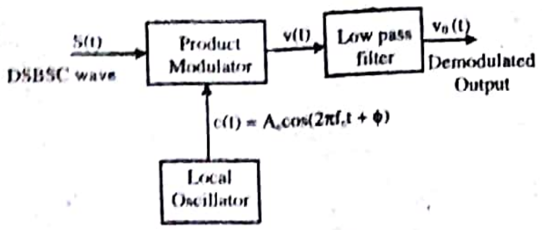


Figure: 1.20

In this process, the message signal can be extracted from DSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in DSBSC modulation. The resulting signal is then passed through a Low pass filter. Output of this filter is the desired message signal.

Let the DSBSC wave be,

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

The output of the local oscillator is,
$c(t) = A_c \cos(2\pi f_c t + \phi)$

Where, ϕ is the phase difference between the local oscillator signal and the carrier signal, which is used for DSBSC modulation. From the **figure 1.20**, we can write the output of product modulator as
$v(t) = s(t)\, c(t)$

Substitute, s(t) and c(t) values in the above Equation.

$$\Rightarrow v(t) = A_c \cos(2\pi f_c t)m(t)\, A_c \cos(2\pi f_c t + \phi)$$

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

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

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

In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.

Therefore, the output of low pass filter is,

$$v_0 t = \frac{A_c^2}{2}\cos\phi\, m(t)$$

The demodulated signal amplitude will be maximum, when $\phi = 0°$. That's why the local oscillator signal and the carrier Signal should be in phase, i.e., there should not be any phase difference between these two signals.

The demodulated signal amplitude will be zero, when $\phi = \pm 90°$. This effect is called as quadrature null effect.

2) **Costas Loop:** Costas loop is used to make both the carrier signal (used for DSBSC modulation) and the locally generated signal in phase. The block diagram of Costas loop is shown in **figure 1.21**.

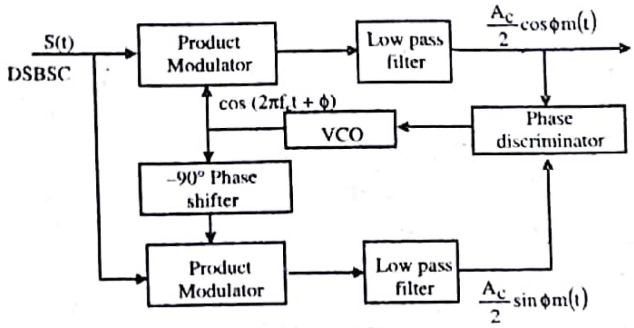


Figure: 1.21

Costas loop consists of two product modulators with common input s (t), which is DSBSC wave. The other input for both product modulators is taken from Voltage Controlled Oscillator (VCO) with –90° phase shift to one of the product modulator as shown in **figure 1.21**.

We know that the Equation of DSBSC wave is,

$s(t) = A_c \cos(2\pi f_c t)m(t)$
Let the output of VCO be

$c_1(t) = \cos(2\pi f_c t + \phi)$

This output of VCO is applied as the carrier input of the upper product modulator.

Hence, the output of the upper product modulator is,
$v_1(t) = s(t)\, c_1(t)$

Substitute, s (t) and c_1 (t) values in the above Equation.

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

After simplifying, we will get v_1 (t) as,

$$v_1(t) = \frac{A_c}{2}\cos\phi\, m(t) + \frac{A_c}{2}\cos(4\pi f_c t + \phi)m(t)$$

This signal is applied as an input of the upper low pass filter. Output of this low pass filter is

$$v_{01}(t) = \frac{A_c}{2}\cos\phi\, m(t)$$

Therefore, the output of this low pass filter is the scaled version of the modulating signal.

The output of –90° phase shifter is,
$c_2(t) = \cos(2\pi f_c t + \phi - 90°) = \sin(2\pi f_c t + \phi)$

This signal is applied as the carrier input of the lower product modulator.
The output of the lower product modulator is,
$v_2(t) = s(t)\, c_2(t)$

Substitute, s (t) and c_2 (t) values in the above Equation.

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

After simplifying, we will get v_2 (t) as,

$$v_2(t) = \frac{A_c}{2}\sin\phi m(t) + \frac{A_c}{2}\sin(4\pi f_c t + \phi) m(t)$$

This signal is applied as an input of the lower low pass filter. The output of this low pass filter is,

$$v_{02}(t) = \frac{A_c}{2}\sin\phi m(t)$$

The output of this low pass filter has –90° phase difference with the output of the upper low pass filter. The output of these two low pass filters is applied as inputs of the phase discriminator. Based on the phase difference between these two signals, the phase discriminator produces a DC control signal.

This signal is applied as an input of VCO to correct the phase error in VCO output. Therefore, the carrier signal (used for DSBSC modulation) and the locally generated signal (VCO output) are in phase.

Ques 16) Calculate the bandwidth and power of DSBSC wave

Ans: Bandwidth of DSBSC Wave

We know the formula for bandwidth (BW) is

$BW = f_{max} - f_{min}$

Consider the Equation of DSBSC modulated wave.

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

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

The DSBSC modulated wave has only two frequencies. So, the maximum and minimum frequencies are $f_c + f_m$ and $f_c - f_m$ respectively.

i.e.,

$f_{max} = f_c + f_m$ and $f_{min} = f_c - f_m$

Substitute, f_{max} and f_{min} values in the bandwidth formula.

$BW = f_c + f_m - (f_c - f_m)$

$\Rightarrow BW = 2f_m$

Thus the bandwidth of DSBC wave is same as that of AM wave and it is equal to twice the frequency of the modulating signal.

Power Calculation of DSBSC Wave

Consider the following Equation of DSBSC modulated wave.

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

Power of DSBSC wave is equal to the sum of powers of upper sideband and lower sideband frequency components.

$P_t = P_{USB} + P_{LSB}$

We know the standard formula for power of cos signal is

$$P = \frac{{v_{rms}}^2}{R} = \frac{\left(v_m\sqrt{2}\right)^2}{R}$$

First, let us find the power of upper sideband and lower sideband one by one. Upper sideband power

$$P_{USB} = \frac{\left(A_m A_c/2\sqrt{2}\right)^2}{R} = \frac{A_m^2 A_c^2}{8R}$$

Similarly, we will get the lower sideband power same as that of upper sideband power.

$$P_{USB} = \frac{A_m^2 A_c^2}{8R}$$

Now, let us add these two sideband powers in order to get the power of DSBSC wave.

$$P_t = \frac{A_m^2 A_c^2}{8R} + \frac{A_m^2 A_c^2}{8R}$$

$$\Rightarrow P_t = \frac{A_m^2 A_c^2}{4R}$$

Therefore, the power required for transmitting DSBSC wave is equal to the power of both the sidebands.

Ques 17) Give the brief description of SSBSC modulator and demodulator in AM wave.

Or

Explain about the frequency and phase discrimination method of AM wave.

Ans: SSBSC Modulators

SSBSC wave can be generated by using the following two methods:

1) **Frequency Discrimination Method:** The following **figure 1.18** shows the block diagram of SSBSC modulator using frequency discrimination method.

   In this method, first we will generate DSBSC wave with the help of the product modulator. Then, apply this DSBSC wave as an input of band pass filter. This band pass filter produces an output, which is SSBSC wave.

   Select the frequency range of band pass filter as the spectrum of the desired SSBSC wave. This means the band pass filter can be tuned to either upper sideband or lower sideband frequencies to get the respective SSBSC wave having upper sideband or lower sideband.

2) **Phase Discrimination Method:** The following **figure 1.22** shows the block diagram of SSBSC modulator using phase discrimination method.

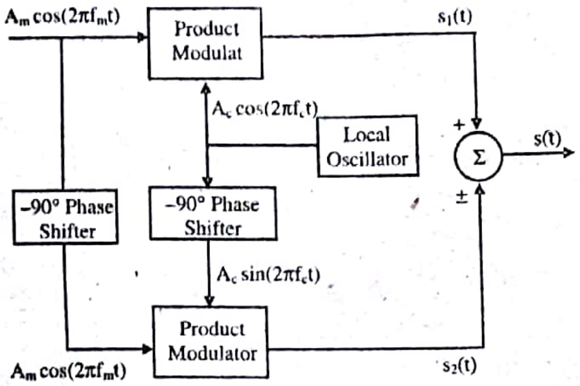


Figure: 1.22

This block diagram consists of two product modulators, two –90° phase shifters, one local oscillator and one summer block. The product modulator produces an output, which is the product of two inputs. The –90° phase shifter produces an output, which has a phase lag of –90° with respect to the input.

The local oscillator is used to generate the carrier signal. Summer block produces an output, which is either the sum of two inputs or the difference of two inputs based on the polarity of inputs.

The modulating signal $A_m\cos(2\pi f_m t)$ and the carrier signal $A_c\cos(2\pi f_c t)$ are directly applied as inputs to the upper product modulator. So, the upper product modulator produces an output, which is the product of these two inputs.

The output of upper product modulator is,

$$s_1(t) = A_m A_c\cos(2\pi f_m t)\cos(2\pi f_c t)$$

$$\Rightarrow s_1(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c+f_m)t] + \cos[2\pi(f_c-f_m)t]\}$$

The modulating signal $A_m\cos(2\pi f_m t)$ and the carrier signal $A_c\cos(2\pi f_c t)$ are phase shifted by –90° before applying as inputs to the lower product modulator. So, the lower product modulator produces an output, which is the product of these two inputs.

The output of lower product modulator is,

$$s_2(t) = A_m A_c \quad (2\pi f_m t - 90°) \quad (2\pi f_c t - 90°)$$

$$\Rightarrow s_2(t) = A_m A_c\sin(2\pi f_m t)\sin(2\pi f_c t)$$

$$\Rightarrow s_2(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c-f_m)t] - \cos[2\pi(f_c+f_m)t]\}$$

Add $s_1(t)$ and $s_2(t)$ in order to get the SSBSC modulated wave s(t) having a lower sideband.

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

$$\frac{A_m A_c}{2}\{\cos[2\pi(f_c-f_m)t] - \cos[2\pi(f_c+f_m)t]\}$$

$$\Rightarrow s(t) = A_m A_c\cos[2\pi(f_c-f_m)t]$$

Subtract $s_2(t)$ from $s_1(t)$ in order to get the SSBSC modulated wave s(t) having a upper sideband.

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

$$\frac{A_m A_c}{2}\{\cos[2\pi(f_c-f_m)t] - \cos[2\pi(f_c+f_m)t]\}$$

$$\Rightarrow s(t) = A_m A_c\cos[2\pi(f_c+f_m)t]$$

Hence, by properly choosing the polarities of inputs at summer block, we will get SSBSC wave having an upper sideband or a lower sideband.

Ques 18) Give the brief description of SSBSC demodulator in AM wave.

Or

Describe the coherent detector demodulator in AM wave.

Ans: SSBSC Demodulators (Coherent Detector)

The process of extracting an original message signal from SSBSC wave is known as detection or demodulation of SSBSC. Coherent detector is used for demodulating SSBSC wave.

Here, the same carrier signal (which is used for generating SSBSC wave) is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. The block diagram of coherent detector is shown in **figure 1.23**.

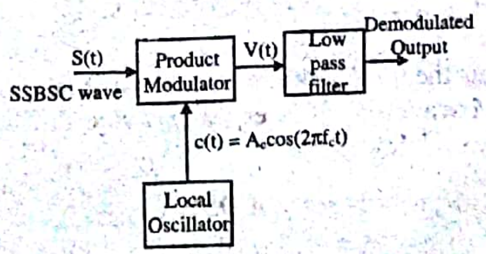


Figure: 1.23

In this process, the message signal can be extracted from SSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in SSBSC modulation. The resulting signal is then passed through a low pass Filter. The output of this filter is the desired message signal.

Consider the following SSBSC wave having a lower sideband,

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

The output of the local oscillator is,
$c(t) = A_c \cos(2\pi f_c t)$

From the **figure 1.23**, we can write the output of product modulator as,
$v(t) = s(t)\, c(t)$

Substitute s(t) and c(t) values in the above equation,

$$v(t) = \frac{A_m A_c}{2}\cos[2\pi(f_c - f_m)t]A_c\cos(2\pi f_c t)$$

$$= \frac{A_m A_c^2}{2}\cos[2\pi(f_c - f_m)t]\cos(2\pi f_c t)$$

$$= \frac{A_m A_c^2}{4}\cos[2\pi(2f_c - f_m)]\cos(2\pi f_m)t\}$$

$$v(t) = \frac{A_m A_c^2}{4}\cos(2\pi f_m t) + \frac{A_m A_c^2}{4}\cos[2\pi(2f_c - f_m)t]$$

In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.

Therefore, the output of low pass filter is,

$$v_0(t) = \frac{A_m A_c^2}{4}\cos(2\pi f_m t)$$

Here, the scaling factor is $\frac{A_c^2}{4}$

We can use the same block diagram for demodulating SSBSC wave having an upper sideband. Consider the following SSBSC wave having an upper sideband.

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

The output of the local oscillator is,

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

We can write the output of the product modulator as,

$v(t) = s(t)\, c(t)$

Substitute s(t) and c(t) values in the above equation.

$$\Rightarrow v(t) = \frac{A_m A_c}{2}\cos[2\pi(f_c + f_m)t]A_c\cos(2\pi f_c t)$$

$$= \frac{A_m A_c^2}{2}\cos[2\pi(f_c + f_m)t]\cos(2\pi f_c t)$$

$$= \frac{A_m A_c^2}{2}\{\cos[2\pi(2f_c + f_m)t] + \cos(2\pi f_m t)\}$$

$$v(t) = \frac{A_m A_c^2}{2}\cos(2\pi f_m t) + \frac{A_m A_c^2}{4}\cos[2\pi(2f_c + f_m)t]$$

In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.

Therefore, the output of the low pass filter is,

$$v_0(t) = \frac{A_m A_c^2}{4}\cos(2\pi f_m t)$$

Here too the scaling factor is $\frac{A_c^2}{4}$

Therefore, we get the same demodulated output in both the cases by using coherent detector.

Ques 19) Find the bandwidth and power of SSBSC wave.

Ans: Bandwidth of SSBSC Wave

We know that the DSBSC modulated wave contains two sidebands and its bandwidth is $2f_m$. Since the SSBSC modulated wave contains only one sideband, its bandwidth is half of the bandwidth of DSBSC modulated wave.

i.e., Bandwidth if SSBSC modulated wave $= \frac{2f_m}{2} = f_m$

Therefore, the bandwidth of SSBSC modulated wave is f_m and it is equal to the frequency of the modulating signal.

Power Calculation of SSBSC Wave

Consider the following equation of SSBSC modulated wave.

$$s(t) = \frac{A_m A_c}{2}\cos[2\pi(f_c + f_m)t] \text{ for the upper sideband}$$

Or

$$s(t) = \frac{A_m A_c}{2}\cos[2\pi(f_c + f_m)t] \text{ . for the lower sideband}$$

Power of SSBSC wave is equal to the power of any one sideband frequency components.

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

We know that the standard formula for power of cos signal is,

$$P = \frac{{v_{rms}}^2}{R} = \frac{\left(v_m/\sqrt{2}\right)^2}{R}$$

In this case, the power of the upper sideband is,

$$P_{USB} = \frac{\left(A_m A_c/2\sqrt{2}\right)^2}{R} = \frac{A_m^2 A_c^2}{8R}$$

Similarly, we will get the lower sideband power same as that of the upper sideband power.

$$P_{LSB} = \frac{A_m^2 A_c^2}{8R}$$

Therefore, the power of SSBSC wave is,

$$P_t = P_{USB} = P_{LSB} = \frac{A_m^2 A_c^2}{8R}$$

Ques 20) Discuss the advantages, disadvantages and applications of SSBSC wave.

Ans: Advantages of SSBSC Wave

1) Bandwidth or spectrum space occupied is lesser than AM and DSBSC waves.
2) Transmission of more number of signals is allowed.
3) Power is saved.
4) High power signal can be transmitter.
5) Less amount of noise is present.
6) Signal fading is less likely to occur.

Disadvantages of SSBSC Wave

1) The generation and detection of SSBSC wave is a complex process.
2) The quality of the signal gets affected unless the SSB transmitter and receiver have an excellent frequency stability.

Applications of SSBSC Wave

1) For the power saving requirements and low bandwidth requirements.
2) In land, air and maritime mobile communications.
3) In point-to-point communications.
4) In radio communications.
5) In television, telemetry, and radar communications.
6) In military communications, such as amateur radio, etc.

Ques 21) Discuss about the VSBSC modulator and demodulator of AM wave.

Ans: VSBSC Modulator

Generation of VSBSC wave is similar to the generation of SSBSC wave. The VSBSC modulator is shown in the following **figure 1.24**.

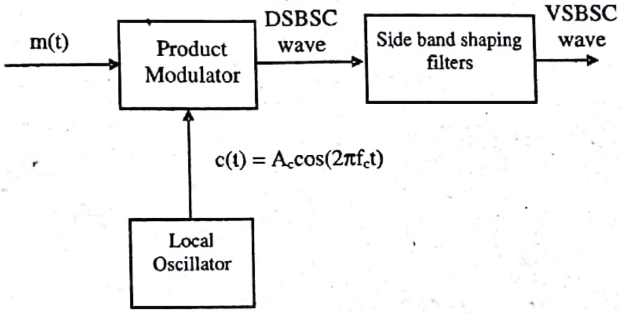


Figure: 1.24

In this method, first we will generate DSBSC wave with the help of the product modulator. Then, apply this DSBSC wave as an input of sideband shaping filter. This filter produces an output, which is VSBSC wave. The modulation signal m(t) and carrier signal $A_c \cos(2\pi f_c t)$ are applied as inputs to the product modulator. Hence, the product modulator produces an output, which is the product of the product modulator is

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

Apply Fourier transform on both sides

$$P(f) = \frac{A_c}{2}[M(f - f_c) + M(f + f_c)]$$

The above equation represents the equation of DSBSC frequency spectrum.

Let the transfer function of the sideband shaping filter be H(f). This filter has the input p(t) and the output is VSBSC modulated wave s(t). The Fourier transforms of p(t) and s(t) are P(t) and S(t) respectively.

Mathematically, we can write S(f) as,
S(t) = P(f) H(f)

Substitute P(f) value in the above equation.

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

The above equation represents the equation of VSBSC frequency spectrum.

VSBSC Demodulator

Demodulation of VSBSC wave is similar to the demodulation of SSBSC wave. Here, the same carrier signal (which is used for generation VSBSC wave) is used to detect the message signal. Hence, this process of detection is called as **coherent** or **synchronous detection.**

The VSBSC demodulator is shown in the following **figure 1.25.**

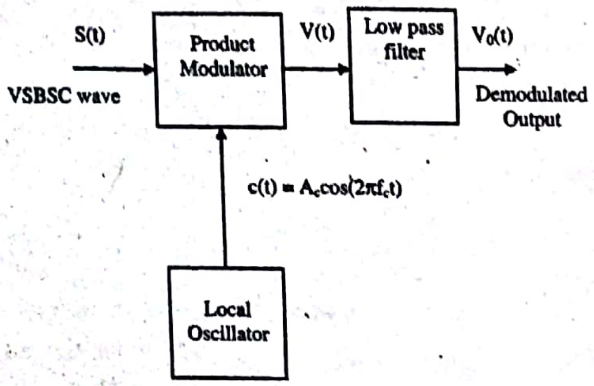


Figure: 1.25

In this process, the message signal can be extracted from VSBSC wave by multiplying it with a carrier, which is having the same frequency and the phase of the carrier used in VSBSC modulation. The resulting signal is then passed through a Low pass filter. The output of this filter is the desired message signal.

Let the VSBSC wave be s(t) and the carrier signal is $A_c \cos(2\pi f_c t)$.

From the **figure 1.25**, we can write the output of the product modulator as,

$$v(t) = A_c \cos(2\pi f_c t)s(t)$$

Apply Fourier transform on both sides,

$$V(f) = \frac{A_c}{2}[S(f-f_c)+S(f+f_c)$$

We know that,

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

From the above equation, let us find $S(f - f_c)$ and $S(f + f_c)$.

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

$$\Rightarrow S(f-f_c) = \frac{A_c}{2}[M(f-2f_c)+M(f)]\ H(f-f_c)$$

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

$$\Rightarrow S(f+f_c) = \frac{A_c}{2}[M(f)+M(f+2f_c)]\ H(f+f_c)$$

Substitute, $S(f-f_c)$ and $S(f + f_c)$ values in V (f).

$$V(f) = \frac{A_c}{2}\left[\begin{array}{l}\frac{A_c}{2}[M(f-2f_c)+M(f)]H(f-f_c)+ \\ \frac{A_c}{2}[M(f)+M(f+2f_c)H(f+f_c)]\end{array}\right]$$

$$\Rightarrow V(f) = \frac{A_c^2}{4}M(f)[H(f-f_c)+H(f+f_c)]$$

$$+\frac{A_c^2}{4}[M(f-2f_c)\ H(f-f_c)+M(f+2f_c)\ H(f+f_c)]$$

In the above equation, the first term represents the scaled version of the desired message signal frequency spectrum. It can be extracted by passing the above signal through a low pass filter,

$$V_0(f) = \frac{A_c^2}{4}M(f)[H(f-f_c)+H(f+f_c)]$$

Ques 22) Explain the bandwidth of VSBSC modulation. Also enlist the advantages, disadvantages and application of VSBSC wave.

Ans: Bandwidth of VSBSC Modulation

The bandwidth of SSBSC modulated wave is f_m. Since the VSBSC modulated wave contains the frequency components of one side band along with the vestige of other sideband, the bandwidth of it will be the sum of the bandwidth of SSBSC modulated wave and vestige frequency f_v.

i.e., Bandwidth of VSBSC Modulated Wave = $f_m + f_v$

Advantages of VSBSC Wave

Following are the advantages of VSBSC modulation:

1) Highly efficient.
2) Reduction in bandwidth when compared to AM and DSBSC waves.
3) Filter design is easy, since high accuracy is not needed.
4) The transmission of low frequency components is possible, without any difficulty.
5) Possesses good phase characteristics.

Disadvantages of VSBSC Wave

Following are the disadvantages of VSBSC modulation.

1) Bandwidth is more when compared to SSBSC wave.
2) Demodulation is complex.

Application of VSBSC Wave

The most prominent and standard application of VSBSC is for the transmission of television signals. Also, this is the most convenient and efficient technique when bandwidth usage is considered.

Now, let us discuss about the modulator which generates VSBSC wave and the demodulator which demodulates VSBSC wave one by one.

Ques 23) What do you mean by frequency modulation and sinusoidal frequency modulation? Differentiate between wide band FM and narrow band FM.

Or

Explain sinusoidal frequency modulation (FM) signal in detail. Discuss its frequency spectrum.

Ans: Frequency Modulation

Frequency modulation is a technique in which the amplitude of the modulated carrier is kept constant, whereas, its frequency is varied in accordance with the modulating signal. The angular frequency can be written as:

$$\omega_i(t) = \omega_c + k_f m(t)$$

If is the message signal is $m(t) = A_m\cos(2\pi f_m t)$, then the frequency modulated signal is given by,

$$2\pi f_i(t) = \omega_c + k_f A_m \cos(2\pi f_m t) \text{ and}$$

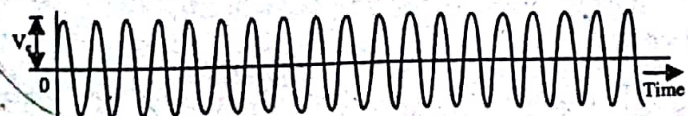
$$\theta_i(t) = \omega_c t + \frac{k_f A_m}{2\pi f_m}\sin(2\pi f_m t)$$

Here, $\frac{k_f A_m}{2\pi}$ is called frequency deviation (Δf) and

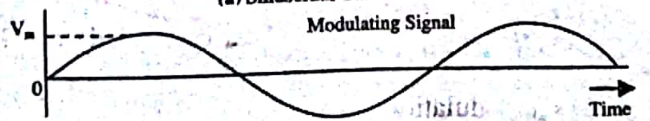
$\frac{\Delta f}{f_m}$ is called modulation index (β).

Sinusoidal Frequency Modulation (FM)

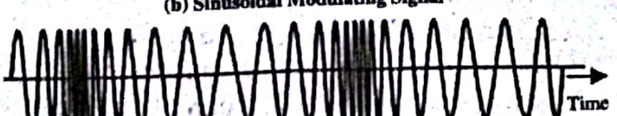
The frequency modulation in which both carrier and modulating signal are sinusoidal in nature is called sinusoidal frequency modulation (FM). The **figure 1.26(a)** below shows the carrier signal. The base band or modulating signal, as shown in **figure 1.26(b)** is superimposed by carrier signal and the resulting signal i.e., the frequency modulated signal, shown in **figure 1.26(c)**, is the signal whose instantaneous frequency varies with that of modulating signal.



(a) Sinusoidal Carrier Wave



(b) Sinusoidal Modulating Signal



(c) Sinusoidal Frequency Modulated Signal

Figure 1.26

The modulation index is denoted by β,

$$\beta = \frac{\Delta f}{f_m}$$

The angle of the modulating signal is given by,

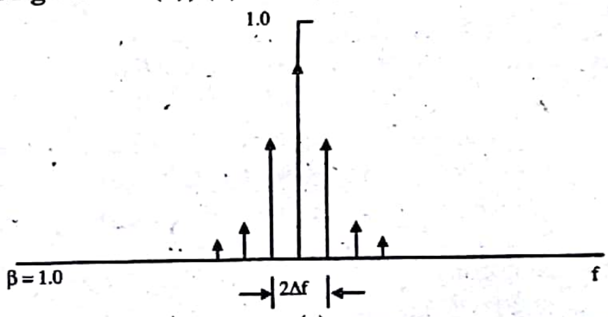
$$\theta_i(t) = 2\pi f_c t + \beta \sin(2\pi f_m t)$$

The modulating signal is given by,

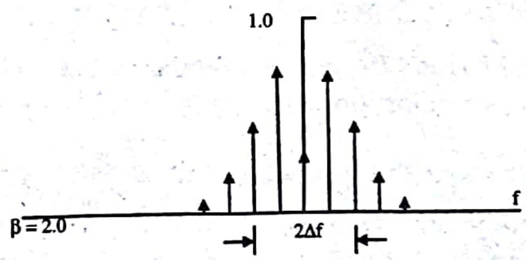
$$s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$$

Frequency Spectrum

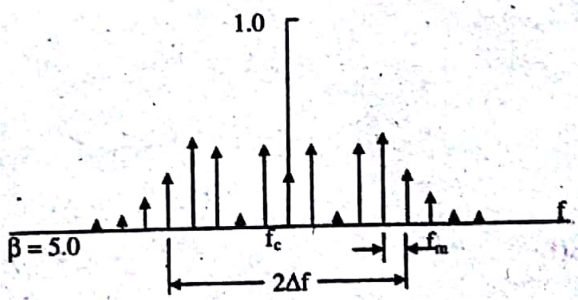
Discrete amplitude spectra of an FM signal, normalized with respect to the carrier amplitude, for the case of sinusoidal modulation of case of sinusoidal modulation of fixed frequency and varying amplitude. The FM spectra of the sinusoidal signal (only the spectra for positive frequencies) are shown in **figure 1.27(a), (b)** and **(c)** below.



(a)



(b)



(c)

Figure 1.27: Frequency Spectrum

From the **figure 1.27** above, it can be easily seen that with increase in the modulation index, β the number of side bands increases, where, $\beta = \frac{k_f A_m}{f_m}$. Thus, the spectrum of a signal with large β is difficult to trace.

Depending on the value of β, the frequency modulation index, the FM is classified as:

1) **Narrowband Frequency Modulated (NBFM) Signal:** Narrowband FM signal, is a FM signal in which the value of frequency modulation index is smaller than unity i.e., $\beta << 1$. It is used in mobile communication system to save bandwidth.

2) **Wideband Frequency Modulated (WBFM) Signal:** Wideband FM signifies the signal having theoretically infinite bandwidth. Wideband FM signal, is a FM signal in which the value of frequency modulation index approximately equal to unity, i.e., $\beta \approx 1$.

The **table 1.1** below shows the difference between wide band FM and narrow band FM.

Table 1.1: Difference between Wide Band FM and Narrow Band FM

S. No	Parameters/ Characteristics	Wideband FM	Narrowband FM
1)	Modulation Index	Greater than 1	Less than (or) slightly greater than 1
2)	Maximum deviation	75 kHz	5kHz
3)	Range of modulating frequency	30 Hz to 15 kHz	30 Hz to 3 kHz
4)	Bandwidth	$2(\Delta f + f_m)$	$2f_m$

Ques 24) Discuss the power relations in FM and also discuss its vector representations.

Or

Explain about the bandwidth calculation in FM.

Ans: Power Relations in FM

In frequency modulation, power of an angle-modulation wave is equal to the power of unmodulated carrier. The power from an unmodulated carrier signal is redistributed among carrier and sidebands.

The average power of angle modulation is defined as,

$$P_C = \frac{V_C^2}{2R} \quad \text{..... (1)}$$

Where,

V_C is the peak voltage of the unmodulated carrier, and R is the load resistance.

The instantaneous power for angle modulation as defined as,

$$P_C = \frac{m(t)^2}{R}$$

$$P_C = \frac{V_C^2}{R}\left[\frac{1}{2} + \frac{1}{2}\cos(2\omega_c t + 2\theta(t))\right] \quad \text{.... (2)}$$

Average power of the second term is zero. Thus:

$$P_c = \frac{V_C^2}{2R} \quad \text{.....(3)}$$

The total power for a modulated wave is defined as follows,

$$P_t = P_0 + P_1 + P_2 + \ldots.. + P_n \quad \text{..... (4)}$$

$$P_t = \frac{V_C^2}{2R} + \frac{2(V_1)^2}{2R} + \frac{2(V_2)^2}{2R} + \ldots.. + \frac{2(V_n)^2}{2R} \quad \text{..... (5)}$$

Where,

P_0 is the modulated carrier power,
P_1 is the power in the first set of sidebands,
P_2 is the power in the second set of sideband, and
P_n is the power in the nth set of sidebands.

Vector Representation of FM Wave

Any alternating wave having a frequency f_c and amplitude V may be represented as a rotating vector OA with OA representing magnitude and rotating as a constant angular velocity ω_c.

If its frequency is increased or decreased, the rotation of the phasor will be faster or slower than ω_c and the phasor would advance or retard to position OB or OC from the position OA. However, it will maintain a constant magnitude. Such a situation is depicted in **Figure 1.28**. The result is variation in phase angle ϕ of the wave.

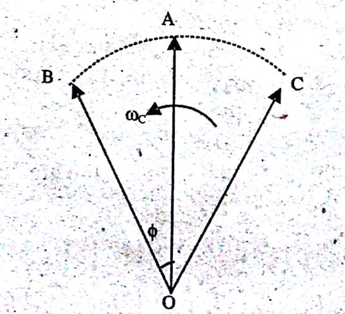


Figure 1.28: Vector Representation of an FM Wave

Bandwidth Calculation in FM

The bandwidth of frequency modulation can be determined by using Bessel table and is defined as,

$$B = 2(n \times f_m)\text{Hz}.$$

Where,

n is the number of significant sidebands, and
f_m is modulating signal frequency (Hz).

By using Carson's rule, the bandwidth of frequency modulation is defined as,

$$B = 2(\Delta f + f_m)\text{Hz}$$

and carrier swing is denoted as $2\Delta f$

Where,

Δf is the Peak frequency deviation (Hz), and

f_m is the Modulating signal frequency (Hz).

Such a system is termed narrowband FM. It finds its use in police, defence, fire, services, etc. with frequency deviation lying in the range of 15–25 kHz.

For values of modulation index $m_f < 0.6$, the side bands produce a wide frequency spectrum but their amplitudes decrease. In such cases, the number of significant side bands increases and the system is referred as wideband FM.

Wideband FM requires a considerably larger bandwidth a compared to the corresponding AM wave.

When $m_f > 5$, the Bessel coefficient $J_0(m_f)$ diminish rapidly and the system bandwidth, to an approximation may be given by,

$$B = 2.\ m_f.\ f_m\ \text{Hz}$$

Such a large bandwidth leads to improvement in the quality of reception, with very low noise as compared to AM. Narrowband FM does not possess this characteristic due to small bandwidth frequency deviation, bandwidth, etc.

Ques 25) Explain about the mathematical analysis of frequency modulation.

Or

Write a short note on the following in context of FM signal:

1) Modulation Index

2) Deviation Ratio

Ans: Mathematical Analysis of Frequency Modulation

In Frequency modulation technique since the amplitude of the modulated carrier is kept constant, whereas, its frequency is varied in accordance with the modulating signal. Phase modulation is similar system in which the phase of the carrier is varied instead of its frequency; as in frequency modulation, the amplitude of the carrier remains constant, the instantaneous angular frequency of frequency modulated wave can be written as,

$$\omega_i = \omega_c + K_f E_m \cos(\omega_m t) \quad .....(1)$$

Where, the proportionality factor K_f determines the maximum variation in frequency for a given signal strength E_m. If $\phi(t)$ is an instantaneous phase, then

$$\phi(t) = \int_0^t \omega_i dt = \int_0^t [\omega_c + K_f E_m \cos(\omega_m t) dt]$$

$$= \omega_c t + K_f \frac{E_m}{\omega_m} \sin(\omega_m t) + \theta_0 \quad ..... (2)$$

The initial phase (θ_0) is neglected as it plays no part in modulation process. Thus FM wave can be written as

$$e = E_c \sin\left[\omega_c t + K_f \frac{E_m}{\omega_m} \sin(\omega_m t)\right] \quad ..... (3)$$

The instantaneous frequency of FM wave is,

$$f = \frac{\omega}{2\pi} = f_c + K_f \frac{E_m}{2\pi} \cos(\omega_m t) \quad ..... (4)$$

Therefore,

$$f_{max} = f_c + K_f \frac{E_m}{2\pi} \text{ and } f_{min} = f_c - K_f \frac{E_m}{2\pi}$$

Hence Frequency Deviation

$$\Delta f = f_{max} - f_c = f_c - f_{min} = K_f \frac{E_m}{2\pi} \quad ..... (5)$$

The instantaneous frequency can then be expressed as,

$$f = f_c + \Delta f \cos(\omega_m t)$$

Modulation Index

The modulation index is given by,

$$m_i = \frac{\Delta f}{f_m} = K_f \frac{E_m}{\omega_m} \quad ..... (6)$$

and therefore the expression for FM wave becomes,

$$e = E_c \sin[\omega_c t + m_i \sin(\omega_m t)] \quad .....(7)$$

In FM broadcast system, Δf has a maximum permissible value of 75 kHz and f_m may vary from about 33Hz to 15 kHz, consequently, m_i can have both greater than unity or less than unity whereas in amplitude modulation m_a should not exceed unity for distortion less transmission.

Deviation Ratio

Minimum bandwidth is greatest when maximum frequency deviation is obtained with the maximum modulating signal frequency. Worst case modulation index and is equal to the maximum peak frequency deviation divided by the maximum modulating signal frequency. Worst case modulation index produces the widest output frequency spectrum

$$DR = \frac{\text{Max peak frequency deviation}}{\text{Max signal frequency}} = \frac{\Delta f_{max}}{f_{m(max)}}$$

Ques 26) Give the comparison between AM & FM.

Ans: Comparison between AM and FM

The comparison between AM and FM techniques are given below:

AM	FM
The equation for AM wave is $v = E_c[1 + m\sin\omega_m t] \sin \omega_c t$	The equation for FM wave is $v = A \sin[\omega_c t + \beta \sin \omega_m t]$
The value of modulation index is always between zero and one.	The modulation index can have value either less than one or more than one.
Transmitted power is dependent upon modulation index $P_T = P_c\left[1+\frac{m^2}{2}\right]$	Since in FM, amplitude of the carrier is constant, the transmitted power is constant, independent of the modulation index.
In an AM signal, only two sidebands are produced, for any value of modulation index.	The modulation index determines the number of significant pairs of sidebands in an FM signal.
The amplitudes of the sidebands is dependent on the modulation index, and is always less than the amplitude of carrier.	The amplitudes of the carrier and side bands vary with the modulation index and can be calculated with Bessel functions.
The sideband amplitude is never zero for any value of modulation index greater than zero.	The carrier or sideband amplitudes are zero at some modulation indices.
The bandwidth of an AM signal is twice the highest modulating frequency.	The bandwidth of an FM signal is proportional to the modulation index.
The AM system is more susceptible to noise and more affected by noise than FM.	The main advantage of FM over AM is its noise immunity, as limiter stage in FM receiver clips off noise signals.
When two AM signals occupy the same frequency, both signals will generally be heard regardless of their relative signal strength.	The capture effect in FM allows the strongest signal on a frequency to dominate without interference from the other signal.
The demodulation of AM signal is very easy practically by use of a single diode, which is very simple to operate and is cheaper.	The circuits to produce and demodulate FM are usually more complex and expensive than AM circuits.

Module 2

AM and Transmitters and Receivers

AM AND FM TRANSMITTERS AND RECEIVERS

Ques 1) What is AM transmitters/transmission? Also draw and describe the block diagram of low power and high power transmitters/transmission.

Or

Discuss the low power and high power transmitters/transmission. Also write there advantages and disadvantages.

Ans: AM Transmitters/Transmission

AM transmitter is a device that is responsible for transmission of amplitude modulated wave. There are two types of AM transmitters, these are, high power amplitude modulated (AM) transmitter and low power (AM) amplitude modulated Transmitter. **Figure 2.1** shows AM transmitter with low power modulation and **Figure 2.2** shows AM transmitter with high power modulation.

It must be noted that stable RF source, buffer amplifiers and subsequent RF power amplifiers are common for both, low power modulation transmitter and high power modulation transmitter.

1) **Low Power (AM) Amplitude Modulated Transmitter/Transmission:** Low level AM modulation is usually the best method for low power transmitters such as walkie-talkies (handhelds). **Figure 2.1** shows the block diagram of a low level amplitude modulated transmitter. The working of low level modulator is similar to that of high level modulator; with the exception that here Pre driver acts a modulator.

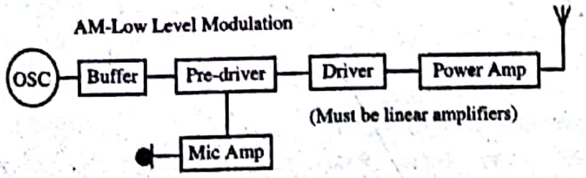


Figure 2.1: Low Power Amplitude Modulated Transmitter/Transmission

Advantage of Low Power Amplitude Modulated Transmitter/Transmission

The advantage of low power AM transmitter is that much audio power is not required, for modulation.

Disadvantage of Low Power Amplitude Modulated Transmitter/Transmission

All radio frequency amplifiers after the modulator (pre-driver in this case) must be linear. This is a disadvantage of low level transmitter as linear amplifiers run at lower efficiency.

2) **High Power Amplitude Modulated (AM) Transmitter/Transmission:** The stable RF source is provided by crystal oscillator with a carrier frequency or submultiple of it. The buffer amplifiers are usually class A amplifier whereas power amplifiers are class C amplifiers. In both, audio and power Audio Frequency (AF) amplifiers are present. In fact, the only difference is the point at which the modulation takes place. In case of low level modulation, modulation takes place at low power level, i.e., before the final output amplifier.

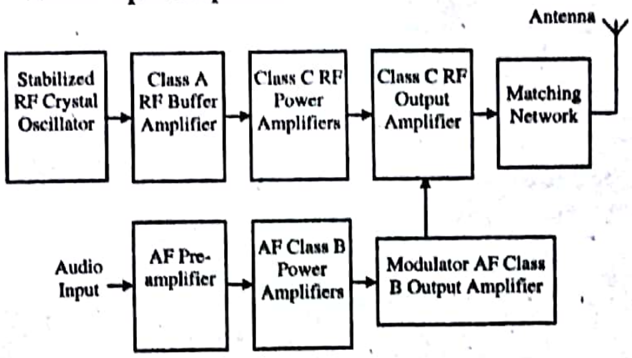


Figure 2.2: AM Transmitter/Transmission Block Diagram with High Power Modulation

Advantage of High Power Amplitude Modulated Transmitter/Transmission

The advantage of high power transmitter is that the carrier (the signal from oscillator) can be amplified using high efficiency Class C amplifiers.

Disadvantage of High Power Amplitude Modulated Transmitter/Transmission

The disadvantage of high power transmitter is that a large amount of audio amplification is required to modulate the high power carrier.

Ques 2) Give the comparison between low power and high power transmitter/transmission.

Ans: Comparison between Low Power and High Power Transmitter/Transmission

Table 2.1 below shows the comparison between low power and high power transmitter circuitry.

Table 2.1: Comparison Between Low Power and High Power Transmitter/transmission

S. No.	Parameter	Low Power Transmitter	High Power Transmitter
1)	Complexity	Modulation circuitry is simple as it has to handle low power.	Modulation circuitry is quite complex as it has to handle high power.
2)	Power level	Modulation circuitry has to handle low power.	Modulation circuitry has to handle high power.
3)	Point at which modulation takes place	Modulation takes place in the initial stages of amplification.	Modulation takes place in the final stage of amplification.
4)	Audio power	Low audio power is required to produce modulation.	High power is required to produce modulation.
5)	Prime factors in design	Simplicity is the prime requirement.	Prime requirement is high efficiency and low distortion.
6)	Amplifier used	Linear amplifier such as class A amplifier is used because all stages must be capable of handling amplitude variations caused by the modulation.	High efficient class C amplifiers are used.
7)	Design requirements of amplifier stages	Each amplifier stage following modulation must handle sideband power as well as carrier. All these subsequent amplifiers must have sufficient bandwidth for the sideband frequencies.	This is not the case with high level modulation because in this modulation takes place in the output stage.
8)	Efficiency	Lower than high level modulators.	Very high.
9)	Amplifiers used	Transistors and Op-amps.	Vacuum tubes and power transistors.

Ques 3) Discuss about the AM receiver with suitable block diagram. Also explain tuned radio frequency receiver in detail and list its advantages and disadvantages.

Ans: AM Receiver

The block diagram of general AM receiver is shown in **Figure 2.3**. The receiver consists of an RF section, the signal is first amplified to level that it can be detected by the Band pass filter, the mixer and converter section, then converts the frequency to a particular level,

IF section consists of various stages, these stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next.

The AM detector circuit simply consists of a diode and possibly a small capacitor to remove any remaining radio frequency. The signal is finally retrieved in the audio section.

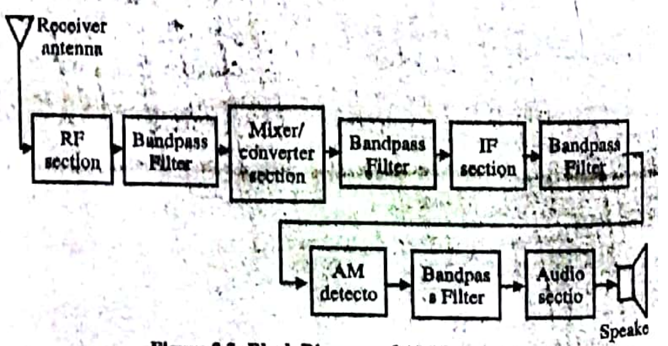


Figure 2.3: Block Diagram of AM Receiver

The tuned radio frequency receiver and Super Heterodyne Receiver are examples of some commonly used AM receivers.

Tuned Radio Frequency Receiver

The tuned radio frequency receiver is a receiver device in which the tuning or selectivity is provided at the radio frequency stages. The tuned radio frequency receiver was used in the early days of wireless technology but it is rarely used today as other techniques offering much better performance are available.

The TRF receivers consist of three main sections:

1) **Tuned Radio Frequency Stages:** This consisted of one of more amplifying and tuning stages. Early sets often had several stages, each proving some gain and selectivity.

2) **Signal Detector:** The detector enabled the audio from the amplitude modulation signal to be extracted. It used a form of detection called envelope detection and used a diode to rectify the signal.

3) **Audio Amplifier:** Audio stages are required to provide audio amplification. These amplifiers are not always required, but when the amplitude of the output of the detector is significantly low, the amplifier stages are required.

The tuned radio frequency receiver was popular in the early twenty century as it provided sufficient gain and selectivity for the receiving the data from broadcast stations. However, tuning took a little time as each stage in the early radios needed to be adjusted separately. Later ganged tuning capacitors were introduced, but by this time the super heterodyne receiver was becoming more popular.

Advantages

The advantages of TRF receiver are:

1) Simplest type of receiver since it does not involve mixing and IF operation.
2) Very much suitable to receive single frequency.
3) TRF receivers have good sensitivity.

Disadvantages

The TRF receiver has largely been disregarded in recent years. The disadvantages of a TRF receiver are:

1) Poor selectivity and low sensitivity in proportion to the number of tuned amplifiers used.

2) Other receiver topologies offer far better levels of performance, and with integrated circuit technology, the additional circuitry in advance receiver is not an issue.

3) Selectivity requires narrow bandwidth, and narrow bandwidth at a high radio frequency implies high Q or many filter sections.

   The problem with TRF receiver is the problem in tuning different frequencies.

4) Another problem is the narrow bandwidth tuning. Keeping several tuned circuits aligned is difficult.

5) The bandwidth of a tuned circuit doesn't remain constant and increases with the frequency increase.

Ques 4) Describe about the straight radio receiver and also write its disadvantages.

Ans: Straight Radio Receiver

The functional block diagram of a simple radio receiver is shown in **figure 2.4**. The receiving antenna receives the radiowaves from different broadcasting stations. The desire radiowave is selected by the radio frequency amplifier, which employs a tuned parallel circuit. The tuned RF amplifier amplifies this selected radiowave.

The amplified radiowave is fed to the detector circuit which consists of a PN diode. This circuit extracts the audio signal from the radiowave. The output of the detector is the audio signal, which is amplified by one or more stages of audio amplification. The amplified audio signal is given to the loud speaker for sound reproduction.

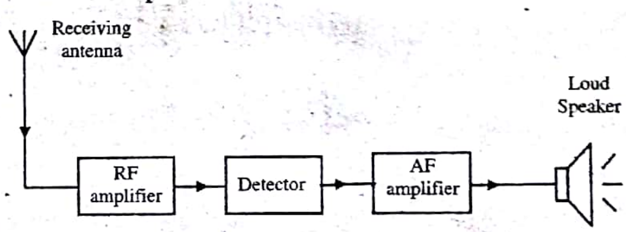


Figure 2.4: Simple Radio Receiver

Disadvantages

The simple radio receiver circuit has following disadvantages:

1) Poor sensitivity
2) Poor selectivity

Ques 5) Explain super heterodyne receiver in detail and also discuss its working with the help of block diagram.

Or

Write a short note on in context of a receiver circuit:

1) **Tuning range**
2) **IF amplifier**

Ans: Super Heterodyne Receiver

Super heterodyne receiver is one of the most popular forms of receiver used in a variety of applications from broadcast receivers to two way radio communications links as well as many mobile radio communications systems.

This form of receiver is based on the idea of mixing signals in a non-linear manner. This idea was first noticed when beats were detected between two signals. R.A Fessenden was the first person to notice this and he patented the idea in 1901.

The idea of the super heterodyne receiver revolves around the process of mixing. Here RF mixers are used to multiply two signals together. When two signals are multiplied together the output is the product of the instantaneous level of the signal at one input and the instantaneous level of the signal at the other.

It is found that the output contains signals at MHz frequencies other than the two input frequencies.

New signals are seen at frequencies that are the sum and difference of the two input signals, i.e., if the two input frequencies are f_1 and f_2, then new signals are seen at frequencies of (f_1+f_2) and (f_1-f_2).

Considering an example, if two signals, one at a frequency of 3 MHz and another at a frequency of 5 MHz are mixed together then new signals at frequencies of 11 MHz and 2MHz are generated.

Working of a Super Heterodyne Receiver

The received modulated signal enters at one input of the mixer. A locally generated signal (local oscillator signal) is fed into the other. The result is that new signals are generated.

These are applied to a fixed frequency intermediate frequency (IF) amplifier and filter. The signals that are converted down and fall within the pass-band of the IF amplifier will be amplified and passed on to the next stages. Those that fall outside the pass-band of the IF are rejected.

Tuning is accomplished very simply by varying the frequency of the local oscillator. The advantage of this process is that very selective fixed frequency filters can be used and these far out perform any variable frequency ones.

They are also normally at a lower frequency than the incoming signal and again this enables their performance to be better and less costly.

The block diagram of a simple super heterodyne receiver is shown in the figure 2.5.

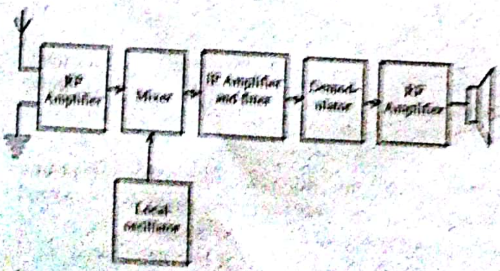


Figure 2.5: Block Diagram of a Basic Super Heterodyne Radio Receiver

The way in which the receiver works can be seen by following the signal as is passes through the receiver:

1) **Front End Amplifier and Tuning Block:** Signals enter the front end circuitry from the antenna. This circuit block performs two main functions:

   i) **Tuning:** Broadband tuning is applied to the RF stage. The purpose of tuning is to reject the signals on the image frequency and accept those of desired frequency. It must also be able to track the local oscillator so that as the receiver is tuned, so the RF tuning remains on the required frequency. Typically the selectivity provided at this stage is not high.

   The main purpose of tuning is to reject signals on the image frequency which is at a frequency equal to twice that of the IF away from the desired frequency. As the tuning within this block provides all the rejection for the image response, it must be sufficiently sharp to reduce the image to an acceptable level.

   However the RF tuning may also help in preventing strong off-channel signals from entering the receiver and overloading elements of the receiver, particularly the mixer or possibly even the RF amplifier.

   Tuning Range is the frequency range over which a receiver, transmitter or other piece of equipment (such as antennas) can be adjusted by means of a tuning control in consideration of required system performance.

   ii) **Amplification:** In terms of amplification, the level is carefully chosen such that the mixer is not overloaded when strong signals are present, but level enables the signals to be amplified sufficiently to ensure a good signal to noise ratio. The amplifier must also be designed at a low noise level as any noise introduced in this block will be amplified later in the receiver.

2) **Mixer / frequency Translator Block:** The tuned and amplified signal then enters to one port of the mixer. The local oscillator signal enters the other port. The performance of the mixer is crucial to many elements of the overall receiver performance. It should be as linear as possible to avoid unwanted signals.

3) **Local Oscillator:** The local oscillator may consist of a variable frequency oscillator that can be tuned by altering the setting on a variable capacitor, or it can be a frequency synthesizer that will enable greater levels of stability and setting accuracy.

4) **Intermediate Frequency Amplifier or IF Amplifier:** The signal now enters in the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters.

5) **Demodulator Stage:** The signals passed through IF stages of the super heterodyne receiver need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. Different demodulators used may include:
   i) AM diode detector
   ii) Synchronous AM detector
   iii) SSB product detector
   iv) Basic FM detector
   v) PLL FM detector
   vi) Quadrature FM detector

6) **Audio Amplifier:** The output from the demodulator is the recovered audio signal. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker.

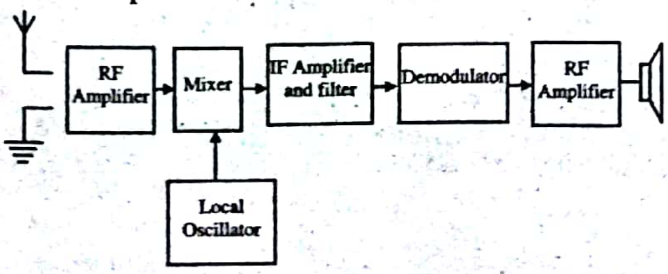


Figure 2.6: Block Diagram of a Basic Super heterodyne Radio Receiver

Ques 6) Explain the concept of Image frequency and its rejection. What are the main functions of RF stage in a receiver circuit?

Ans: Image Frequency and its Rejection

In a standard broadcast receiver (and, in fact, in the vast majority of all receivers made) the local oscillator frequency is made higher than the incoming signal frequency for reasons that will become apparent. It is made equal at all times to the signal frequency plus the intermediate frequency.

Thus $f_o = f_s + f_i$, or $f_s = f_o - f_i$, no matter what the signal frequency may be. When f_s and f_o are mixed, the difference frequency, which is one of the by-products, is equal to f_i. As such, it is the only one passed and amplified by the IF stage.

If a frequency f_{si} manages to reach the mixer, such that $f_{si} = f_o + f_i$, that is, $f_{si} = f_s + 2f_i$, then this frequency will also produce f_i when mixed with f_o.

Unfortunately, this spurious intermediate-frequency signal will also be amplified by the IF stage and will therefore provide interference.

This has the effect of two stations being received simultaneously and is naturally undesirable. The term f_{si} is called the image frequency and is defined as the signal frequency which is twice the intermediate frequency.

Reiterating, we have

$$f_{si} = f_s + 2f_i \qquad .....(1)$$

The rejection of an image frequency by a single-tuned circuit, i.e., the ratio of the gain at the signal frequency to the gain at the image frequency, is given by,

$$\alpha = \sqrt{1 + Q^2\rho^2} \qquad .....(2)$$

Where,

$$\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} \qquad .....(3)$$

Q = Loaded Q of tuned circuit

If the receiver has an RF stage, then there are two tuned circuits, both tuned to f_s. The rejection of each will be calculated by the same formula, and the total rejection will be the product of the two. Whatever applies to gain calculations applies also to those involving rejection.

Image rejection depends on the front-end selectivity of the receiver and must be achieved before the IF stage. Once the spurious frequency enters the first IF amplifier, it becomes impossible to remove it from the wanted signal. It can be seen that if f_{si}/f_s is large, as it is in the AM broadcast band, the use of an RF stage is not essential for good image-frequency rejection, but it does become necessary above about 3MHz.

Functions of RF Stage in Receiver Circuit
This is a class C tuned voltage amplifier. The main functions of this stage are:

1) Amplification of the received radio signal to provide better sensitivity and improved signal to noise ratio.
2) Rejection of the unwanted signals and improved adjacent channel selectivity.
3) Rejection of the image signal.

Ques 7) In a broadcast super heterodyne receiver having no RF amplifier, the loaded Q of the antenna coupling circuit (at the input to the mixer) is 80. If the intermediate frequency is 455 kHz, Calculate

1) The image frequency and its rejection ratio at 1000 kHz
2) The image frequency and its rejection ratio at 50 MHz

Ans:
1) The image frequency and its rejection ratio at 1000 kHz

$f_{si} = f_s + 2f_i$
$= 1000 \times 10^3 \times 2 \times 455 \times 10^3$
$= 1910$ kHz

$$\rho = \frac{1910\times10^3}{1000\times10^3} - \frac{1000\times10^3}{1910\times10^3}$$
$= 1.910 - 0.524$
$= 1.386$

Rejection ratio is given by,
$$\alpha = \sqrt{1+Q^2\rho^2} = \sqrt{1+(80)^2\times(1.386)^2}$$
$= 110.88$

2) The image frequency and its rejection ratio at 50 MHz

$f_{si} = f_s + 2f_i = 50 \times 10^6 + 2 \times 455 \times 10^3 = 50.91$MHz

$$\therefore \rho = \frac{50.91\times10^6}{50\times10^3} - \frac{50\times10^3}{50.91\times10^3}$$
$= 1.018 - 0.982 = 0.036$

Rejection ratio is given by,
$$\alpha = \sqrt{1+Q^2\rho^2} = \sqrt{1+(80)^2\times(0.036)^2}$$
$= 3.049$

Ques 8) What is the basic concept of using double-heterodyne receiver?

Or

Why double-heterodyne receiver is used? Explain the term double-conversion.

Or

What are the reasons behind using double super-heterodyne receiver?

Ans: Double-Heterodyne Receiver
Although the super heterodyne radio receiver works well, to ensure the optimum performance under a number of situations, an extension of the principle, known as the double super heterodyne or double super heterodyne radio receiver may be used.

The double heterodyne radio receiver improves the performance in a number of areas including stability (however, synthesizers work well to overcome this problem), image rejection and adjacent channel filter performance. It also meets the requirements of high selectivity at lower frequencies.

The double super heterodyne radio receiver is still widely used, especially at high frequencies where factors such as image rejection and filter performance are important.

Basic Double Super Heterodyne Receiver Concept
The basic concept behind the double super heterodyne radio receiver is the use of a high intermediate frequency to achieve the high levels of image rejection that are required, and a further low intermediate frequency to provide the levels of performance required for the adjacent channel selectivity.

Typically the receiver will convert the incoming signal down to a relatively high first intermediate frequency (IF) stage. This enables the high levels of image rejection to be achieved. As the image frequency lies at a frequency twice that of the IF away from the main or wanted signals, the higher the IF, the further away the image is and the easier it is to reject at the front end.

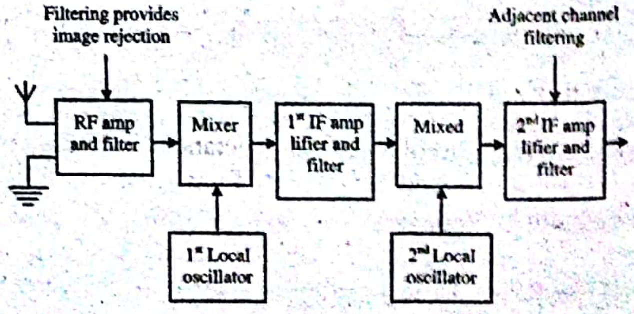


Figure 2.7: Basic Double Super Heterodyne Receiver

The signal once pass through the first IF at the higher frequency, is then passed through a second mixer to convert it down to a lower intermediate frequency where the narrow band filtering is accomplished so that the adjacent channel signals can be removed. As the lower frequency, filters are cheaper and the performance is often higher.

Although it must be said that filter technology now allows effective filters to be made at much higher frequencies than was previously possible. Since the receiver uses only two IF stages for conversion thus this process is called **double-conversion**. If there are three IF stages for conversion, it will be called as Triple-conversion and so on.

Ques 9) What are the various characteristics of amplitude modulated (AM) receiver?

Ans: Characteristics of Amplitude Modulated (AM) Receiver

The performance of the radio receiver can be measured in terms of following receiver characteristics:

1) **Adjacent-Channel Selectivity:** Adjacent-channel Selectivity is a measure of the performance of a radio receiver to respond only to the radio signal it is tuned to (such as a radio station) and reject other signals nearby in frequency, such as another broadcast on an adjacent channel.

   It is usually measured as a ratio in decibels (dBs), comparing the signal strength received against that of a similar signal on another frequency. If the signal is at the adjacent channel of the selected signal, this measurement is also known as **Adjacent-Channel Rejection Ratio (ACRR).**

   Selectivity in a receiver is obtained by using tuned circuits. These are LC circuits tuned to resonate at a desired signal frequency. The Q of these tuned circuits determines the selectivity. It shows attenuation that the receiver offers to signals at frequencies near to the one to which it is tuned. A good receiver isolates the desired signal in the RF spectrum and eliminates all other signals.

   Q is the ratio of inductive reactance to resistance ($Q = X_L/R$), and bandwidth of the tuned circuit is given by,

$$B_w = \frac{f_r}{Q}$$

   Where, f_r is the **resonant frequency**.

   The bandwidth of a tuned circuit is measure of the selectivity. Narrower the bandwidth better is the selectivity and to have narrower bandwidth and better selectivity the Q of the tuned circuit, B_w must be high.

   The **figure 2.8** shows the selectivity curve for the typical tuned circuit.

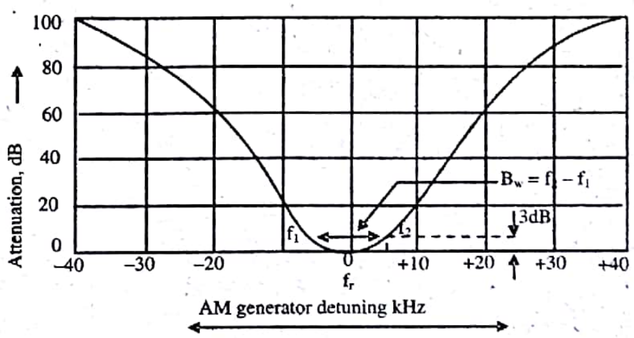


Figure 2.8: Selectivity Curve of a Tuned Circuit

2) **Fidelity:** Fidelity refers to the ability of the receiver to reproduce all the modulating frequencies equally. **Figure 2.9** shows the typical fidelity curve for radio receiver.

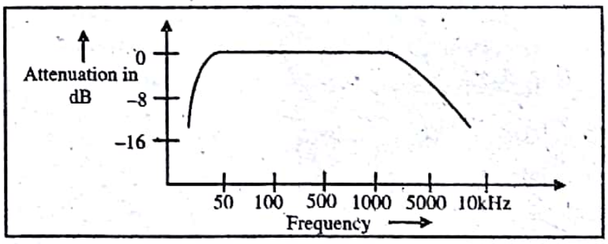


Figure 2.9: Typical Fidelity Curve

   The fidelity at the lower modulating frequencies is determined by the low frequency response of the IF amplifier and the fidelity at the higher modulating frequencies is determined by the high frequency response of the IF amplifier.

   It is also difficult to obtain in AM receiver because good fidelity requires more bandwidth of IF amplifier resulting in poor selectivity.

3) **Gain:** Gain is the factor by which an input signal is multiplied to produce the output signal. In general, higher the gain of a receiver, the better is its sensitivity. The more the gain of the receiver, the smaller the input signal, that is necessary to produce a desired level of output. High gain in communication receivers is obtained by using multiple stages of amplification.

4) **Sensitivity:** The sensitivity of a communication receiver refers to the ability of receiver to pick up weak signals, and amplify it. It is often defined in

terms of the voltage that must be applied to the receiver input terminals to give a standard output power, measured at the output terminals. The more the gain, a receiver has, the smaller the input signal necessary to produce desired output power. Therefore, sensitivity is a primary function of the overall receiver gain. It is often expressed in microvolts of in decibels.

Sensitivity in a receiver is normally taken as the minimum input signal (Smin) required to produce a specified output signal having a specified signal-to-noise (S/N) ratio and is defined as the minimum signal-to-noise ratio times the mean noise power.

The sensitivity of receiver mostly depends on the gain of the IF amplifiers. Good communication receiver has sensitivity of 0.2 to 1μV. Sensitivities of 5 to 10μV are typical for FM receivers, whereas the sensitivity of an AM receiver could be 100μA or more.

5) **Double Spotting:** The phenomenon of double spotting occurs at higher frequencies due to poor front end selectivity of the receivers. In this, receiver picks up same short-wave station at two nearby points on the receiver dial.

   When the receiver is tuned across the band, a strong signal appears to be at two different frequencies, once at the desired frequency and again when the receiver is tuned to 2 times IF below the desired frequency. In this second case, the signal becomes the image, reduced in strength by the image rejection, thus making it appear that the signal is located at two frequencies in the band.

6) **Image Frequency Rejection Ratio (IFRR):** Its value depends upon the value of the loaded Q of the tuned circuits of the RF stages, the value of the IF of the receiver (higher the better) and on whether f_c is close to the lower end or the higher end of the tuning range of the receiver. It should be at least 40dB, where IFFR is given by:

$$\text{IFRR} \triangleq 10\log_{10}\left|\frac{H_{RF}(f_c)}{H_{RF}(f')}\right|^2$$

Ques 10) Give the reason of choice of intermediate frequency.

Ans: Choice of Intermediate Frequency

Many factors are involved in the choice of a receiver IF. In most cases the choice is a compromise and may vary from one brand of receiver to another. The value of IF for a specific receiver is chosen by the designer, or manufacturer, from a wide range of values. In fact since the discovery of the heterodyne principle, intermediate frequencies ranging in value from 130 kHz to 485 kHz have been used in broadcast band "Superheat" receivers. Values at the low end of this range are used only occasionally.

The use of a low value IF results in slightly better gain and stability characteristics. A low IF also results in a 2 to 3 percent Improvement in selectivity. However, the advantages gained are overshadowed by the increased susceptibility to image- frequency reception. The majority of modern broadcast band receivers use one of three values for the IF; 455 kHz, 456 kHz, or 465 kHz. These values, through experience, have been found to give sufficient gain and stability characteristics while retaining acceptable image-frequency rejection.

As usual with making choices, the choice of an IF also constitutes a compromise between various requirements, some of which point to a low IF, while others demand a higher IF.

1) The higher the IF the poorer is the selectivity of the receiver.
2) Higher IF is more difficult to track and tune.
3) The lower the IF, the poorer the image-frequency rejection.
4) Very low IF can make the selectivity too sharp, and may even cut-off the side bands.
5) A lower IF requires a higher degree of stability from the local oscillator, as any "drift" is now of a greater proportion.Most standard AM broadcast receivers use an IF in the range 438 KHz to 465 KHz, with 455 KHz being the most commonly used IF. The intermediate frequency in a commercial AM receiver has a fixed value of 455 KHz. This value is chosen as a compromise between two conflicting factors.
   i) Adjacent channel selectivity and easy tracking for which IF should be low, and
   ii) Image signal rejection for which IF should be high.

For proper channel selectivity, the intermediate frequency f_i should be low. A lower f_i needs a lower Q, and the proper tank circuit can be easily designed, whereas if f_i is large, it will require a large Q and the tank circuit design will be complex. **For example,** if f_i = 455 kHz and the baseband is 10 kHz, the Q-value desired by the tank circuit is,

$$Q = \frac{455}{10} = 45.5$$

The tuned circuit for this Q can be easily designed. But, if f_i = 10MHz, the Q desired is,

$$Q = \frac{10\times10^6}{10\times10^3} = 1000$$

The design of a tuned circuit with such a high Q is impossible. Hence, for a better selectivity, f_i should be lower. Also, a low value of IF makes the difference between signal and local oscillator frequency small, and as a result tracking becomes easy.

The value of f_i should be large for image signal rejection. We have,

$$f_c = f_c + 2f_i$$

Or,

$$\frac{f_c'}{f_c} = 1+(2f_i / f_c)$$

Therefore, if f_i is kept large, the image signal f'_c will be far apart from desired signal f_c and can be easily rejected. In commercial AM receivers, the intermediate frequency is kept fixed at 455 kHz in order to achieve a compromise between the conflicting factors. This value of IF provides good image signal rejection at broadcast band even if no RF amplifier is used, as the ratio f'_c/f_c is large.

At short waves, however, the image signal rejection becomes poor due to a phenomenon known as double spotting. In this phenomenon, signals from the same short waves station is picked at two nearby points on the receiver tuning dial. Double spotting is harmful because it can mask a weak station by a nearby strong station at the spurious point on the dial.

The spurious point corresponds to an image signal. Double spotting can be avoided by a good front end selectivity to avoid image signal. The RF amplifier adds more tuning circuits and improves the front and selectivity of the receiver at short waves.

Ques 11) Describe the simple AVC circuit. Also draw the characteristics of AVC circuit.

Ans: Simple AVC (Automatic Volume Control) Circuit

Automatic volume control (AVC) bias is obtained to keep the receiver output substantially constant with time for any variations in receiver input voltage. The magnitude of the receiver input voltage varies with time due to fading, or when the receiver is tuned from one station to another having different signal strength. The AVC eliminates the effect of these variations.

Figure 2.10 provides a "simple AVC" bias if diode D in AVC circuit is removed. The AVC circuit samples a fraction of the detector output and converts it to AVC bias voltage. The AVC bias is applied to RF and IF stages to provide them a negative bias. As the input of the receiver signal increases, the AVC bias voltage also increases, and in turn, the negative bias to RF and IF amplifiers is increased, thereby reducing their gain. The output of the receiver is, thus, maintained constant.

The simple AVC circuit has a disadvantage is that it is operative even for low magnitude signals at the receiver input. This deteriorates the sensitivity of the receiver. Therefore, it is desirable that AVC should be operative only when the input signal is strong. This arrangement is made by a circuit called delayed Called delayed AVC, which is obtained by incorporating a forward biased diode D in the AVC filter circuit as shown in **Figure 2.10**.

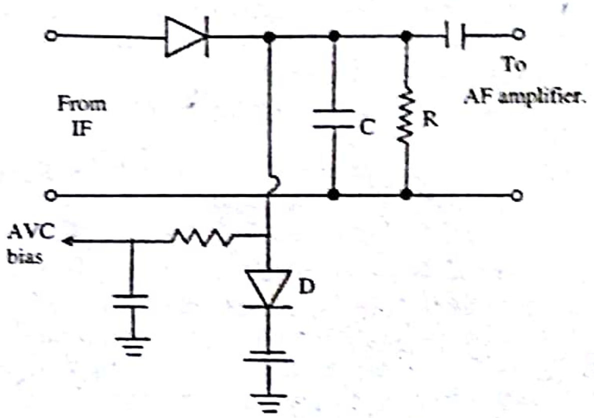


Figure 2.10: A Detector Circuit with AVC

The AVC is operative only if the signal is more than the diode bias voltage. The AVC characteristic can be further improved by amplifying the delayed AVC bias with the help of a DC amplifier. This is known as amplified and delayed AVC.

Characteristics of AVC circuit

The characteristics of the various types of AVC circuits are shown in **Figure 2.11** obviously, amplified and delayed AVC is very much near to the ideal AVC. An ideal AVC remains inoperative for signal strength between certain limits, and provides a constant output for signal strength exceeding this limit.

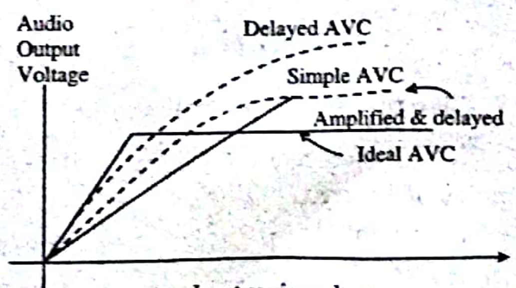


Figure 2.11: AVC Characteristics

Ques 12) What do you mean by FM transmitters? Also write the different types of FM transmitters.

Ans: FM Transmitters

The circuit that is used to transmit frequency modulated signal is called **FM transmitters**. The general block diagram of an FM transmitter is given in **figure 2.12**; the FM transmitter mainly consists of pre-amplifier, FM modulator, oscillator, frequency multiplier and power amplifier. Basically common FM transmitter contains following functional blocks.

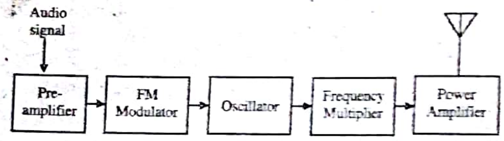


Figure 2.12: Block Diagram of Standard FM Transmitter

Types of FM transmitters

The different types of FM transmitters are as follows:
1) Direct FM Transmitter
2) Indirect FM transmitter

Ques 13) Explain the types of direct FM transmitter in brief.

Or

Write a short note on:
1) PLL direct FM transmitter
2) Varactor Diode Transmitter

Ans: Direct FM Transmitter

In a direct FM system the instantaneous frequency is directly varied with the information signal. The various methods of Direct FM transmission are:

1) **PLL Direct FM Transmitter:** PLL Direct FM transmitter is the most widely used FM transmitter. **Figure 2.13** shows the phase locked loop direct FM transmitter. This transmitter is used to generate high index wideband FM signal. When both the input frequencies of phase comparator are same, then they are locked to each other.

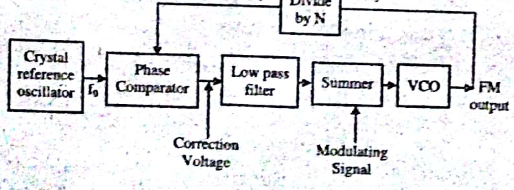


Figure 2.13: Block Diagram of PLL Direct FM Transmitter

Under this situation, phase comparator output is zero. It is passed through low pass filter to the summer. The summer has modulating signal as another input. This modulating signal is used to control the output frequency of VCO. Thus the output of VCO is FM signal whose frequency depends upon modulating signal. The output of VCO is divided by N and given to phase comparator.

If there is any shift in centre frequency (f_c) of VCO, then phase comparator generates the correction voltage, which is given to summer through low pass filter. This correction voltage adds to modulating signal voltage and corrects the VCO output. The lowpass filter is used to remove the rapid changes in correction voltage due to frequency variations in FM signal.

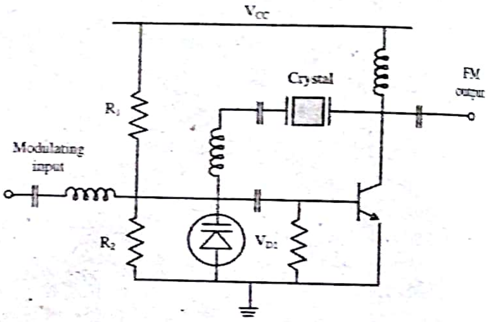


Figure 2.14: Varactor Diode Frequency Modulator

2) **Varactor Diode Transmitter:** The circuit below shows a varactor diode transmitter circuit. This circuit changes the frequency of the crystal oscillator using diode. R_1 and R_2 develop a DC voltage across the diode which reverse biases it. The voltage across the diode determines the frequency of the oscillations. Positive inputs increase the reverse bias, decrease the diode capacitance and thus increase the oscillation frequency.

Similarly, negative inputs decrease the oscillation frequency. The crystal oscillator ensures that the output waveform is very stable, but this is only the case if the frequency deviations are kept very small. Thus, the varactor diode modulator can only be used in limited applications. In direct FM the instantaneous frequency of the carrier is varied directly with the message signal by means of a device known as a voltage controlled oscillator (VCO).

A VCO is a sine wave generator (oscillator), that is capable of producing different frequency sinusoid depending on its input voltage.

Thus, a VCO is an FM modulator. If the input to the VCO is the message m(t)

$$v_{in}(t) = m(t)$$

$$f_{out} = f_c + k\, m(t)$$

$$\theta_{out}(t) =$$

$$2\pi\int_0^t f_{out}(t)\,dt = 2\pi f_c t + 2\pi k\int_0^t m(t)\,dt$$

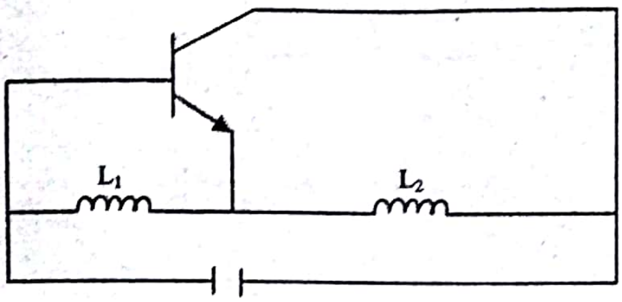


Figure 2.15: The Hartley Oscillator

Thus, the VCO output signal is

$$v_0 = \cos(\theta_{out}(t)) = \cos\left(2\pi f_c t + 2\pi k\int_0^t m(t)\,dt\right),$$

which is desired FM signal.

A typical implementation of the VCO is a Hartley Oscillator that uses a voltage variable capacitor to control the frequency of the oscillator.

Ques 14) Discuss the indirect FM transmitter. Also draw the block diagram of armstrong method and explain for FM generation.

Or

Give the brief description of indirect FM transmitter.

Ans: Indirect FM transmitter

The indirect transmitter uses a NBFM signal along with frequency multiplier and mixers to generate WBFM signal. A method to generate FM is given below, in this method a NBFM signal is passed through a frequency multiplier. The block diagram below shows various components of FM generation.

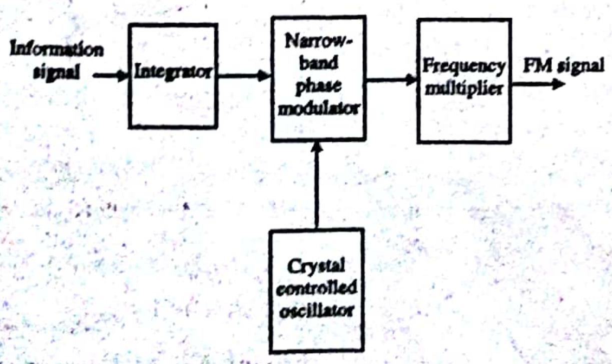


Figure 2.16: Block Diagram of the Indirect Method of generating a Wide-Band FM-Signal

Consider a signal x(t), where,

$$x(t) = A_c \cos(2\pi f_c t + \theta(t))$$

For a phase modulated system

$$\theta(t) = 2\pi k_p m(t)$$

For an angle modulated system

$$\theta(t) = 2\pi k_f \int_0^t m(\tau)\,d\tau$$

Where, τ is a dummy variable.

The information signal is passed through the integrator circuit, then is passed to narrow band phase Modulator, the signal is passed to the frequency multiplier circuit, the signal generated is a wide band modulated system, this increases the quality of signal reception at the receiver end, but in the expense of increased bandwidth.

The main disadvantage of direct FM is that, it is hard to build a stable high frequency Oscillator. Thus indirect method of FM is used.

Armstrong Method or RC Phase Shift Method

The stability of the circuit is a major problem in direct methods of FM generation. Particularly, even if crystal oscillators are used, stability problem still exists. Since, FM is one form of phase modulation. Hence it is possible to obtain FM from PM. This method is called indirect method to generate FM.

The phase modulated signal is represented as,

$$e_{pm} = E_c \sin(\omega_c t + \beta_p \cos \omega_m t) \quad .....(1)$$

The instantaneous angular frequency ω_p, of the above phase modulated signal can be obtained by,

$$\omega_p = \frac{d\theta(t)}{dt}$$

Here $\theta(t) = \omega_c t + m \cos \omega_m t$, hence above equation will be,

$$\omega_p = \frac{d}{dt}[\omega_c t + \beta_p \cos\omega_m t] = \omega_c - \beta_p \sin \omega_m t \times \omega_m \quad .....(2)$$

In terms of linear frequencies above equation can be written as,

$$f_p = f_c - \beta f_m \sin(2\pi f_m t) \quad .....(3)$$

The second term in the above equation represents the frequency shift with respect to centre frequency,

i.e., $f_p = f_c + \Delta f$ .....(4)

The above equation shows that frequency of the phase modulated signal varies around the carrier frequency f_c with the derivation of Δf, from equation (3), the frequency deviation can be written as,

$$\Delta f = \beta f_m \sin(2\pi f_m t)$$

The RC circuit is used to provide a phase-shift of 90° thus it is also known as the RC phase shift method of FM generation. Hence Maximum deviation will be,

$$\Delta f = f_m \beta \quad .....(5)$$

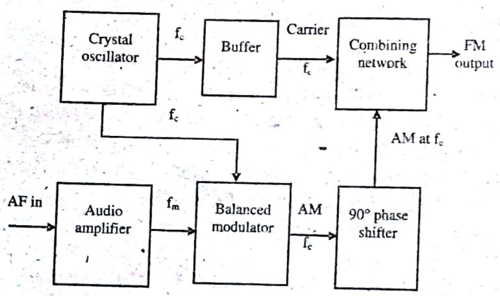


Figure 2.17: Block Diagram of Armstrong Method to Generate FM

The RC circuit is used to provide a phase-shift of 90° thus it is also known as The RC Phase Shift Method of FM generation.

Ques 15) Discuss FM receiver with the help of a block diagram.

Or

With the help of a block diagram of FM receiver, also discuss FM reception.

Ans: FM Receiver

The circuit that is used to receive the frequency modulated signal is called FM receiver. The process of receiving a modulated signal with the help of FM receiver is called FM reception. It is necessary to be able to successfully demodulate it and recover the original signal. The FM demodulator may be called a variety of names including FM demodulator, FM detector or an FM discriminator.

There are a number of different types of FM demodulator, but all of them enable the frequency variations of the incoming signal to be converted into amplitude variations on the output. These are typically fed into an audio amplifier, or possibly a digital interface if data is being passed over the system. The demodulator circuit is one of the most important part of FM receiver.The functional block diagram of an FM receiver is shown in **figure 2.18**

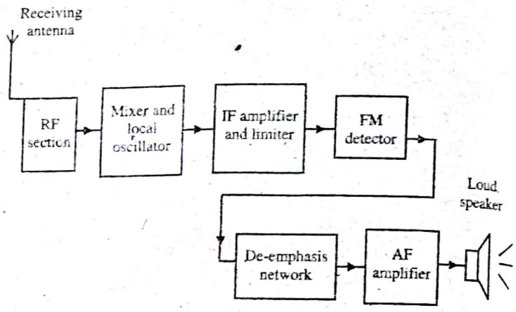


Figure 2.18: FM Receiver

The RF section selects the incoming modulated signals and is amplified. It is then fed into the mixer and local oscillator. Here the frequency of the modulated signal is changed to intermediate frequency. For FM receivers, this IF is 10.7 MHz. The intermediate frequency wave is amplified using IF amplifier and then its amplitude is maintained constant using a limiter.

The output of this section is applied to the FM detector which demodulates the modulated wave. The AF signal from the FM detector is then passed on through a de-emphasis network, where the various frequencies attain their original power distribution. Finally it is fed into the loud speaker after performing AF amplification.

Ques 16) Give the detail description of balanced slope detector.

Or

Discuss the working of balanced slop detector with the help of various cases.

Ans: Balanced Slope Detector

The circuit of balanced slope detector is shown in **Figure 2.19**. 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 this secondary tuned LC circuit is coupled with the inductance of the primary (or input side) LC circuit. Thus it forms a tuned transformer.

In **figure 2.19**, the upper tuned circuit is shown as T_1 and lower tuned circuit is shown as T_2. The input side LC circuit is tuned to f_c, carrier frequency. T_1 is tuned to $f_c + \delta f$, which represents highest frequency and lower LC circuit T_2 is tuned to $f_c - \delta f$, which represents the minimum frequency of FM signal. The input FM signal is Coupled to T_1 and T_2 180° out of phase.

AM and Transmitters and Receivers (Modul

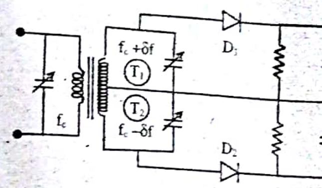


Figure 2.19: Balanced Slope Detect

The secondary side tuned circuits
connected to diodes D_1 and D_2 with
total output V_{out} is equal to differen
and V_{02}, since they subtract
figure 2.19. Figure 2.20 shows the
the balanced slope detector. It sh
respect to input frequency.

Working

The working of balanced slop detec
with the help of various cases:

1) **Case 1:** When the input frequenc
both T_1 and T_2 produce the same
V_{01} and V_{02} are identical and the
other. Therefore V_{out} is zero. Th
figure 2.20.

2) **Case 2:** When the input frequency
upper circuit T_1 produces maximum
is turned to this frequency i.e. f_c
lower circuit T_2 is tuned to $f_c - \delta f$,
away from $f_c + \delta f$. Hence T_2 proc
voltage. Hence the output V_{01} is m
V_{02} is minimum. Therefore
$V_{out} = V_{01} - V_{02}$ is maximum posit

3) **Case 3:** When input frequency is f_c
circuit T_2 produces maximum sig
tuned to it. But upper circuit
minimum signal. Hence a rectified
maximum and V_{01} is minimum. Th
$V_{out} = V_{01} - V_{02}$ is maximum nega
This is shown in **figure 2.20.**

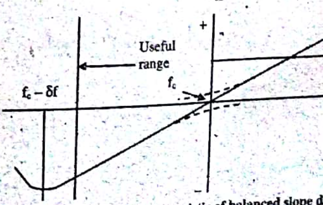


Figure 2.20: Characteristic of balanced slope d
'S'-curve

For the other frequencies of inpu
(V_{out}) is produced according to the
shown in **figure 2.20**. For exam

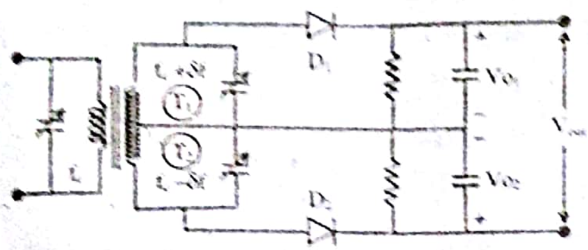


Figure 2.19: Balanced Slope Detector

The secondary side tuned circuits (T_1 and T_2) are connected to diodes D_1 and D_2 with RC loads. The total output V_{out} is equal to difference between V_{01} and V_{02}, since they subtract as shown in **figure 2.19. Figure 2.20** shows the characteristic of the balanced slope detector. It shows V_{out} with respect to input frequency.

Working

The working of balanced slop detector is described with the help of various cases:

1) **Case 1:** When the input frequency is equal to f_c, both T_1 and T_2 produce the same voltage. Hence V_{01} and V_{02} are identical and they subtract each other. Therefore V_{out} is zero. This is shown in **figure 2.20**.

2) **Case 2:** When the input frequency is $f_c + \delta f$ the upper circuit T_1 produces maximum voltage since it is turned to this frequency i.e. $f_c + \delta f$. Whereas lower circuit T_2 is tuned to $f_c - \delta f$, which is quite away from $f_c + \delta f$. Hence T_2 produces minimum voltage. Hence the output V_{01} is maximum where V_{02} is minimum. Therefore

   $V_{out} = V_{01} - V_{02}$ is maximum positive for $f_c + \delta f$.

3) **Case 3:** When input frequency is $f_c - \delta f$, the lower circuit T_2 produces maximum signal since it is tuned to it. But upper circuit T_1 produces minimum signal. Hence a rectified outputs V_{02} is maximum and V_{01} is minimum. Therefore output $V_{out} = V_{01} - V_{02}$ is maximum negative for $f_c - \delta f$. This is shown in **figure 2.20.**

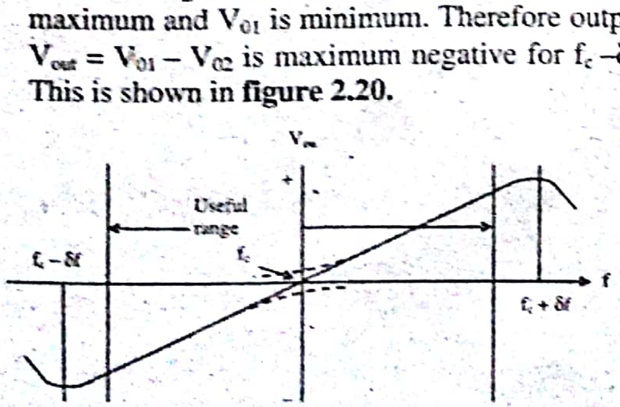


Figure 2.20: Characteristic of balanced slope detector, or 'S'-curve

For the other frequencies of input, the output (V_{out}) is produced according to the characteristic shown in **figure 2.20. For example,** if input frequency tries to increase above f_c then V_{01} will be greater than V_{02} and net output V_{out} will be positive. It is desirable that characteristic shown in **figure 2.20** should be linear between $f_c - \delta f$ and $f_c + \delta f$, then only proper detection will take place. The linearity of the characteristic depends upon alignment of tuning circuits and coupling characteristics of the tuned coils.

Ques 17) Explain foster seely discriminator and also discuss its phasor diagram.

Ans: Foster-Seely Discriminator (Phase Discriminator)

The phase shift between the primary and secondary voltages of the tuned transformer is a function of frequency. It can be shown that the secondary voltage lags primary voltage by 90°at the carrier center frequency. This carrier frequency (f_c) is the resonance (or tuned) frequency of the transformer. Foster-seeley discriminator utilizes this principle for FM detection.

The circuit diagram of basic foster-seeley discriminator is shown in **Figure 2.21**. In the **figure 2.21** observe that primary voltage is coupled though C_3 and RFC to the center tap on the secondary. The capacitor C_3 passes all the frequencies of FM. Thus the voltage V_1 is generated across RFC. RFC offers high impedance to frequencies of FM. The voltage V_1 thus appears across (RFC) center tap of secondary and ground also. The voltage of secondary is V_2 and equally divided across upper half and lower half of the secondary coil.

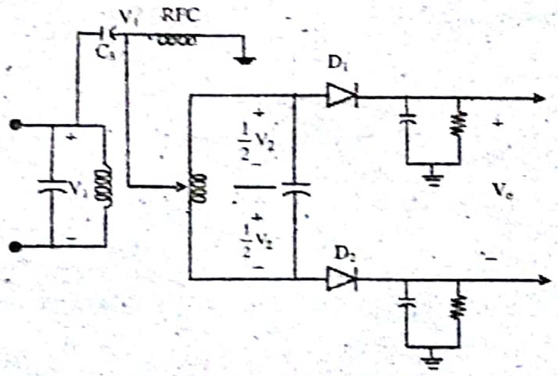


Figure 2.21: Basic Foster-Seeley Discriminator

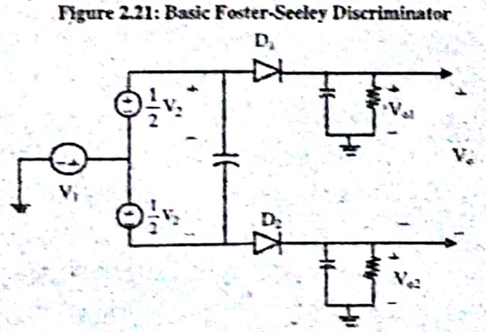


Figure 2.22: Voltage generator equivalent circuit

Figure 2.22 shows the generator equivalent circuit of Foster-Seeley discriminator. In this figure observe that the voltage across diode D_1 is $V_{D2} = V_1 + 0.5\ V_2$ and that across D_2 is $V_{D2} = V_1 - 0.5\ V_2$. The output of upper rectifier is V_{01} and lower rectifier is V_{02}. The net output $V_0 = V_{01} - V_{02}$ Since $V_{01} = |V_{D1}|$ and $V_{o2} = |V_{D2}|$ output, $V_0 = |V_{D1}| - |V_{D2}|$. Thus the net output depends upon the difference between magnitudes of V_{D1} and V_{D2}.

Phasor Diagram of Foster-Seeley Discriminator

At the center frequency, both V_{D1} and V_{D2} will be equal, since V_2 will have 90° phase shift with V_1. **Figure 2.23 (a)** shows how V_{D1} and V_{D2} are generated from V_1 and V_2. In the **figure 2.23 (b)** and (c) vector addition is shown and it shows that $|V_{D1}| = |V_{D2}|$.

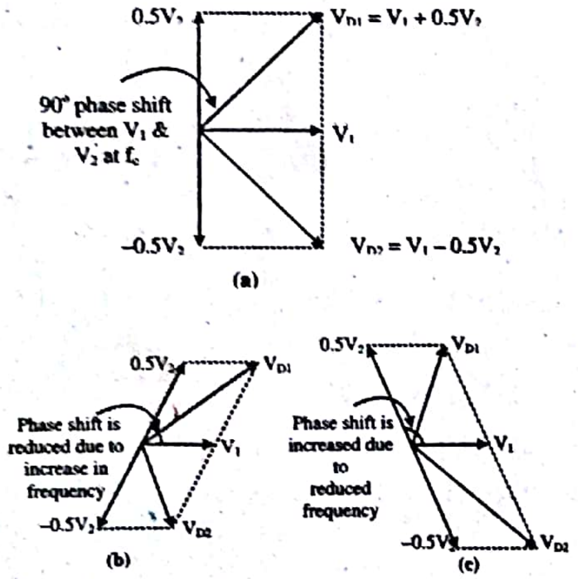


Figure 2.23

In **figure 2.23 (a)**, at center frequency, phase sh between V_1 and V_2 is 90°, hence $|V_{D1}| = |V_{D2}|$

In **figure 2.23 (b)**, for the frequencies above cente frequency, the phase shift between V_1 and V reduced. Hence $|V_{D1}| > |V_{D2}|$

In **figure 2.23 (c)**, for frequencies below cente frequency, the phase shift between V_1 and V_2 i increased. This makes $|V_{D1}| < |V_{D2}|$

Hence the net output of the discriminator will be zero. Now consider the situation when inpu frequency increases above f_c .Hence the phase shif reduces. Therefore $|V_{D1}|$ is greater than $|V_{D2}|$.

This is shown by vector addition in **figure 2.23 (b)**. Hence net output $V_o = |V_{D1}| - |V_{D2}|$ will be positive. Thus the increase in frequenc increases output voltage. Now consider the situation when frequency reduces below f_c. This makes $|V_{D1}|$ less than $|V_{D2}|$.

This is shown in the **figure 2.23 (c)**. Hence the output $V_o = |V_{D1}| - |V_{D2}|$ will be negative. Thus the Foster-Seeley discriminator produces output depending upon the phase shift. The linearity of the output depends upon the linearity between frequencies an induced phase shift.

Module 3

Television and Radar System

TELEVISION ENGINNERING

Ques 1) What do you mean by television broadcasting System? Also describe the principle of television engineering.

Ans: Television Broadcasting System

The word tele means far away. The word television then means viewing at a distance. In television systems, two-dimensional pictures and scenes are converted into one-dimensional electric signals and transmitted by means of electromagnetic radiations to distant places. These signals are received and demodulated into corresponding electric signals by means of TV receivers. The demodulated signals are then applied to respective TV screens to reproduce the picture televised.

In other words, television is used to extend the sense of sight beyond its natural limits and to transmit. Sound is associated with scene. Amplitude modulation (AM) is used for the picture signal and frequency modulation (FM) is used for sound signal.

Principle of Television Engineering

In television, light signals from the object being televised are converted into electrical signals by a television camera and transmitted to distant points by radio carrier waves. The television receiver separates the television signals from carrier waves and converts them into light signals which form a picture of the televised object on the screen of the picture tube.

However, in the television system sound has also to be transmitted along with the picture. Separate earner waves are used for the transmission of picture signals and sound signals but they are radiated by the same transmitting antenna.

At the receiving end, the same receiving antenna receives both carrier waves but the television receiver converts these signals separately into sound waves which drive a loudspeaker and light waves which produce a picture on the screen of the picture tube. For the proper display of the picture and the reproduction of accompanying sound, several controlling signals have also to be transmitted.

Ques 2) Discuss about the working operation of television system with the help of its block diagram.

Ans: Working Operation of Television System

A **block diagram** of a complete TV system for the transmission and reception of picture and sound signals is given in **figure 3.1.** At the TV studio, the TV camera focuses an optical image of the scene on a photosensitive plate in the camera and the picture elements of varying light intensity are converted into correspondingly varying electrical signals by a process of electronic scanning. The electrical signals so formed by scanning the picture image by an electric beam are called video signals.

At this stage, certain synchronising signals meant to keep the reassembly of the picture at the receiver in step with the scanning at the studios are also added to the video information. The composite video signal so formed is amplified by video-amplifiers and made sufficiently strong to amplitude-modulate a picture carrier wave which is transmitted by the transmitting antenna.

The sound picked up by the microphone is converted into electrical currents at audio frequencies (AF) and is strengthened by the audio amplifier which frequency-modulates a separate RF carrier whose frequency is generally 55MHz above the frequency of the video carrier. The frequency modulated (FM) sound carrier is radiated by the same transmitting antenna as used for the transmission of the video or picture carrier.

Thus, at the TV transmitting station, two separate RF carriers, one for the transmission of picture signals and the other for sound signals are radiated by a common transmitting antenna. The picture (video) carrier is amplitude-modulated (AM) and the sound carrier is frequency-modulated (FM).

At the receiving end, both the picture and sound carriers are intercepted by the same receiving antenna and passed onto a wideband circuit called the tuner. In the tuner two separate IFs for picture and sound signals are formed by heterodyning with a local oscillator as in a superheterodyne receiver. The picture and sound IF frequencies are amplified by a common IF amplifier and then detected by the video detector. At this stage, the sound IF of 5.5MHz (the difference between video and sound IF from the tuner) is separated and fed into the sound channel where it is detected by a method of FM detection and the AF is amplified and fed into the speaker to produce the sound as in a normal FM receiver.

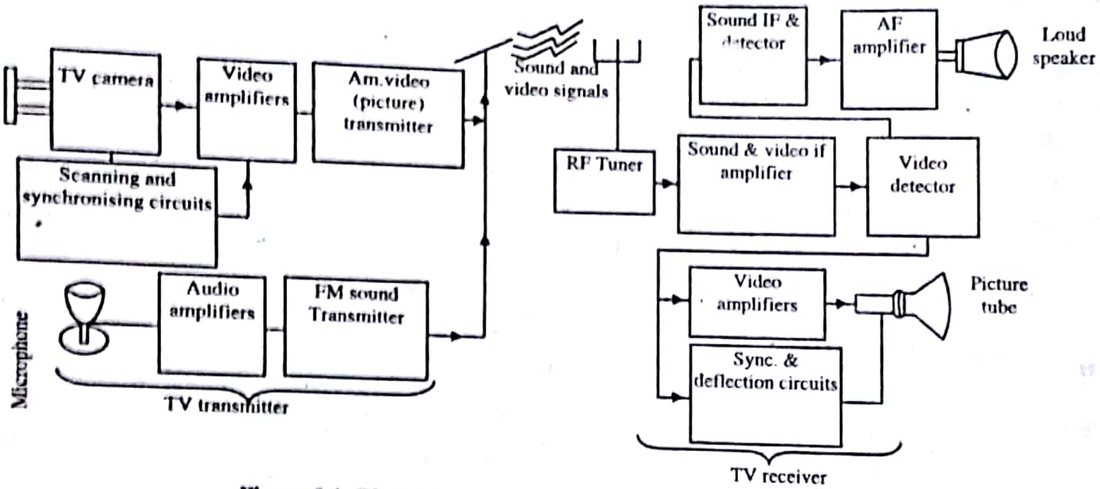


Figure 3.1: Block Diagram of a Complete Television System

The video signal from the video detector stage is amplified by a video amplifier and is used to modulate the electron beam in the picture tube to produce a picture of the television scene, A portion of the composite video signal is also fed to a synchronising separator where the synchronising signals are separated from the video signal and applied to the deflection circuits to keep the electronic scanning beam in the picture tube in step with the electronic beam at the transmitter.

The above method of obtaining the sound IF (5.5 MHz) by beating or heterodyning the video and sound carriers is known as the inter-carrier system and is the modem system used in TV receivers.

Ques 3) Explain about the requirements and standards of television engineering.

Ans: Requirements of Television Engineering

All television broadcast systems use a 4:3 aspect ratio, 2:1 interlace and a scanning sequence of left to right and top to bottom. The field scanning frequency is usually the same as the power supply frequency, except in some countries have a 50Hz supply frequency and yet use a scan rate of 60 fields per second. Modern low ripple receiver designs do not involve any great problem of hum-bar or rolling ripples in the picture. The 60Hz scanning systems provide a greater freedom from flicker at high brightness levels than the 50Hz systems.

High resolution systems, such as the French 819-line system, could not be fully exploited economically in the allocated UHF and VHF bands, because of large bandwidth requirements. There is a revived interest in high resolution high fidelity television service in HDTV, for cine film grade picture, if not better. All systems, except the old British 405-line system, employ horizontal polarisation because of reduction of interference from man-mode sources, such as cars, in the horizontal polarisation mode.

Standards of Television Engineering

In the international scenario, there are three main television systems. They are:

1) The CCIR PAL (Phase-Alternation by line) system
2) The US-based NTS (National Television systems Committee)
3) The French-based SECAM.

India follows the PAL-D system.

Table 3.1

Particulars	Western-Europe, Middle-East, India &most Asian Countries	North & South America including US, Canada, Mexico & Japan	England	USSR	France
Lines per frame	625	525	625	625	625
Frames per second	25	30	25	25	25
Field frequency (Hz)	50	60	50	50	50
Line frequency (Hz)	15,625	15,750	15,625	15,625	15,625
Video bandwidth (MHz)	5 or 6	4	5.5	6	6
Channel bandwidth (MHz)	7 or 8	6	8	8	8
Video modulation	Negative	Negative	Negative	Negative	Positive
Picture modulation	AM	AM	AM	AM	FM
Sound modulation	FM	FM	FM	FM	AM
Colour system	PAL	NTSC	PAL	SECAM	SECAM

Ques 4) What do you mean by scanning? Also discuss the need of scanning in television system.

Ans: Scanning

Scanning is the process by which a two-dimensional physical picture is converted into corresponding unidimensional electric signals. **Figure 3.2(a)** shows a full picture, which, in this case, is a circle. In **figure 3.2(b)**, shows a wide scanning line. Using such wide scanning wide scanning lines, we scan the circle from top to bottom, which results in the scanned circle, shown in **figure 3.2(c)**. We find from **figure 3.2(d)** that the scanned circle resembles the original circle only approximately. The smooth edge of the original circle is seen to become square edges in the recreated circle.

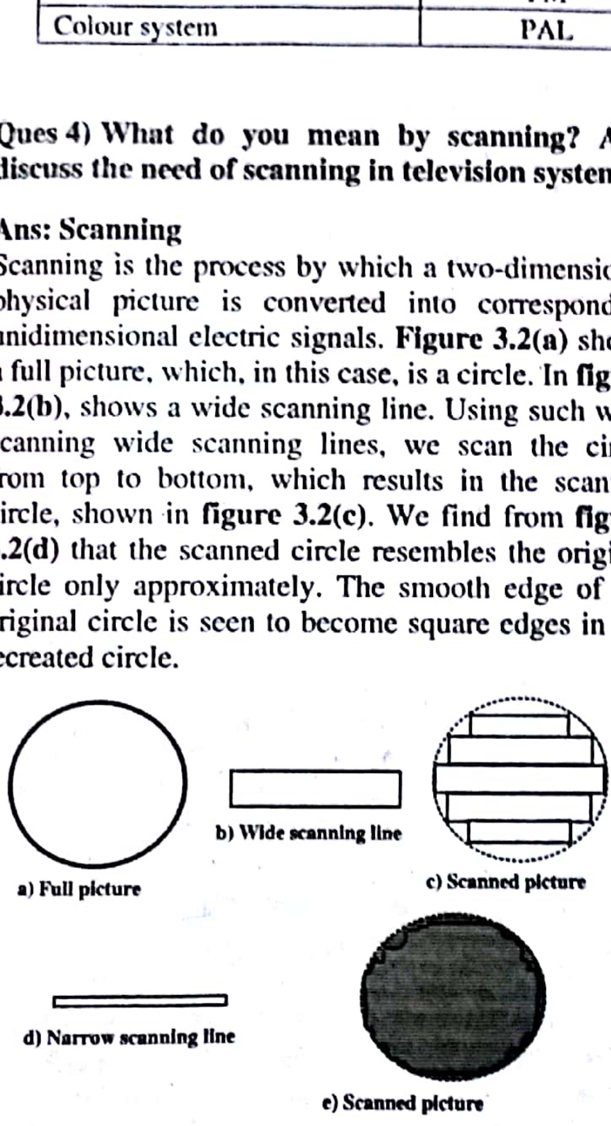


Figure 3.2: Scanning Details

Figure 3.2(d) shows a narrow scanning line and **figure 3.2(e)** shows the scanned picture using these narrow scanning lines. Now, comparing **figure 3.2(e)** and **figure 3.2(a)** we find that the circle recreated by using the narrow scanning lines resembles the original circles closer than that recreated by using the wider scanning lines. However, the circle recreated by using the narrow scanning lines requires more number of scanning lines. So, all scanning systems employ large number of scanning lines for scanning pictures.

Need for Scanning

To transmit all the variations of light present in the space of a picture at a single instant of time, thousands of channels would be needed. It is not feasible to use so many channels to transmit a single programme. Thus the first requirement for transmitting the variation of light in a picture is to convert them into signals varying with time so that there is only a single variation at any one instant. Hence, there is the **need of scanning.**

The whole picture is divided into a number of small elements called pixels. This infinite number of pieces of information are need to be picked up simultaneously for transmitting picture information but it is not practicable because, it is not feasible to provide separate signal (path channel) for each picture signal. This problem can be solved by a method known as **"Scanning"**.

Ques 5) Define camera tube and also discuss its principle.

Or

What do you mean by camera tube? Also write the types of camera tube.

Ans: Camera Tube

A camera tube is transducer which converts variations of light intensity into variations of electric current or voltage (video signal). It is the eye of a television system.

The essential requirements of a camera tube are as follows:

1) Photosensitive target plate capable of converting an optical image into an electric charge image. Greater the brightness of a pixel in the optical image, greater is the electrical charge produced at the corresponding point in the target plate.
2) Storage of the charge at the points in the target till the time it is not neutralized. The charge should not get dissipated during this time. Hence the target plate should not be a conductor.
3) Neutralization of the charge at every point of the target plate in quick succession by the scanning beam, causing current to flow through a load resistor.

Principle of Camera Tube

The basic principle employed in the construction of camera tubes is the photo electric effect which allows the conversion of light energy into electrical energy with the help of certain photosensitive materials like potassium, seetenium, lead and their oxides. Based on the principle of three **photo electric effects** used for converting variations of intensity into electrical variations they are:

1) Photo emission,
2) Photo conduction, and
3) Photo voltaic.

Beside the above two photoelectric effects, the solid-state image scanner based on the properties of charge coupled devices (CCD) are also coming into vogue.

Types of Camera Tube

There are various types of camera tube on the basis of target plate, which are as follows:

1) Vidicon camera tube
2) Plumbicon camera tube
3) Saticon camera tube
4) Silicon-vidicon camera tube
5) Pyroelectric vidicon camera tube
6) Image orthicon camera tube

Ques 6) Describe the vidicon camera tube.

Or

Discuss the vidicon camera tube with the help of its block diagram.

Ans: Vidicon-Camera Tube

Vidicon camera tube is of relatively simpler construction and makes use of the photoconductivity of certain semiconductor materials such as amorphous selenium, the resistances of which decreases on exposer to light. As shown in **Figure 3.3** the vidicon tube consists of target, a fine mesh screen (grid 4), beam focusing electrode (grid 3) and an electron gun.

The target plate consists of a very thin and transparent film of a conducting material coated directly on the inside of the glass face plate. This is the signal plate for the camera output signal. The gun side of the signal is coated with an extremely thin layer of a photoconductive material such as selenium or antimony compounds.

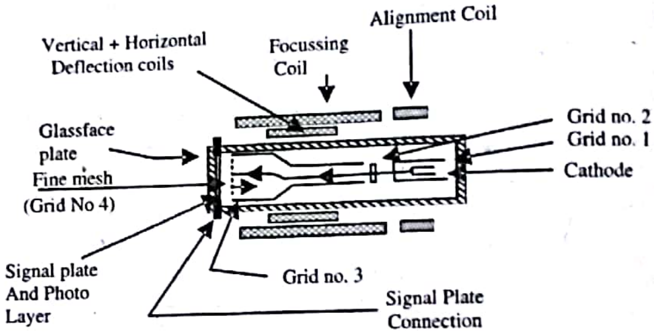


Figure 3.3: Vidicon

When illuminated with light from the optical image, the resistance of each element of the photoconductive layer decreases in proportion to the amount of light falling on it. The photoconductive layer is scanned by an electron beam from the electron gun.

This electron beam originates from the cathode that is at a potential of about –30 V with respect to the signal plate. The scanning electron beam deposits enough electrons on each spot that it scans to reduce the potential. Excess electrons not deposited on the target are turned back but not used in the vidicon.

Grid 4 is a wire mesh which provides a decelerating voltage for the electron beam so that the low velocity beam can deposit electrons on the charge image without producing secondary electrons from the photo layer. The potential difference across a particular spot on the photoconductive material is 30V before and after it has been scanned. This change in potential causes a capacitive signal current to flow in the signal plate circuit producing a voltage drop across the load resistance R_L. This is the video signal voltage.

The vidicon has the same sensitivity and resolution as the image orthicon but it is not able to reproduce rapid motion quite satisfactorily as the resistance of the photoconductive material does not vary instantaneously with changes in light intensity.

Ques 7) What do you mean by plumbicon camera tube? Also describe the working of plumbicon camera tube.

Ans: Plumbicon Camera Tube

Plumbicon is small camera tube like the vidicon. It has overcome most of the drawbacks of the standard vidicon tube. It has fast response and produces high quality pictures at low light levels. Because of its small size, light weight and low power operating characteristics, plumbicon is the most suitable camera tube for transistorised television cameras.

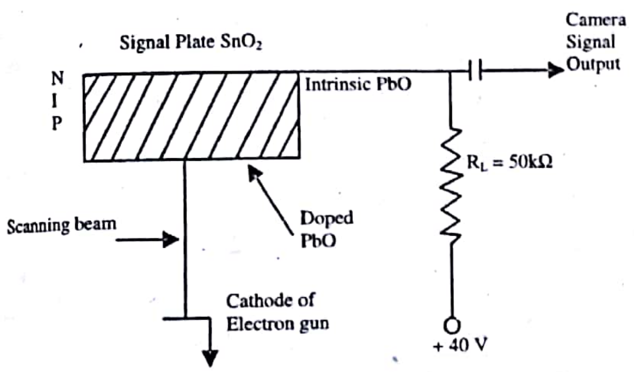


Figure 3.4: Plumbicon Camera Tube

Working

Except for the target, plumbicon is very much similar to standard vidicon. The target operates effectively as a PIN semiconductor diode coating on the inner side of the glass face plate. P-type semiconductor is doped to have an excess of positive charges, N-type has excess of electrons as negative charges. Intrinsic semiconductor, of I-type is pure PbO to be neutral without doping. In the manufacture, an intrinsic (I) layer of pure PbO is sandwitched between the N-type SnO_2 signal plate to form a PIN-type semiconductor diode.

The functioning of the photoconductive target in plumbicon is similar to the photo conductive material in vidicon except that in standard vidicon, each element acts as a leaky capacitor, with leakage resistance decreasing with increasing light intensity. In the plumbicon, however, each element serves as a capacitor in series with a reverse biased light controlled diode. The incidence of light on the target results in photo excitation of semiconductor junction between the pure PbO and doped layer.

The resultant is decrease in resistance causes signal current flow which is proportional to the intensity of light on each photo element. The average thickness of the target is 10 to 20 μm.

The higher sensitivity of plumbicon compared to vidicon is due to much reduced recombination of photo generated electrons in the intrinsic layer which contains very few discontinuities.

Ques 8) What is saticon camera tube? Also describe the construction and working of saticon camera tube.

Ans: Saticon Camera Tube

Saticon camera tube is suitable for broadcast. It consists of selenium, arsenic and tellurium, and hence the name saticon.

Construction and Working

Construction of face plate, signal plate (tin oxide), electron gun, focussing and deflecting devices is the same as in vidicon. In fact in all photoconduction type camera tubes, construction of the camera tubes is the same, except for the construction of the target plates.

Saticon's target plate consists mainly of selenium which is the first chemical element that was tried in TV camera tubes. At that time it was not successful because it suffered from crystallisation and instability. As a result of research, these problems have been successfully solved in saticon.

The target used in saticon is shown in **Figure 3.5.** Selenium was doped with arsenic which made it chemically stable and prevented crystallization.

To increase sensitivity for red light and to prevent halo, tellurium was added in traces close to the signal plate. Thus a thin layer at the signal plate consists of selenium, arsenic and tellurium.

Tellurium gives a black appearance to the target and prevents scattering. This layer gives n-type polarity due to conductor signal plate. After this layer, there is a layer of selenium and arsenic. Thereafter a thin layer of antimony trisulplide (p-type) is used on the gun side of the target to suppress emission of secondary electrons.

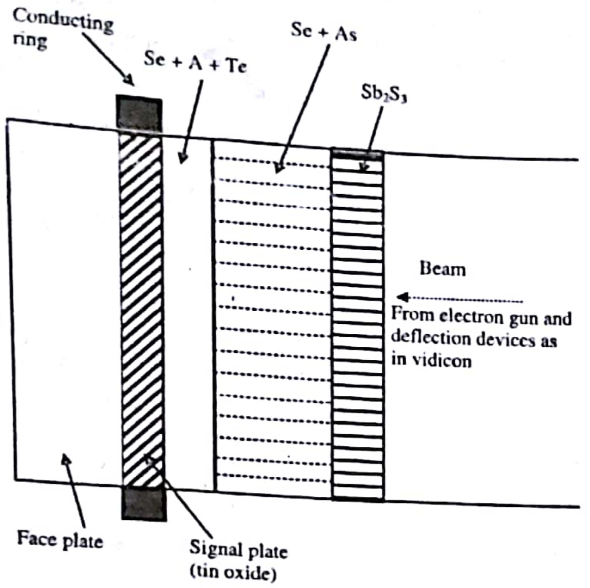


Figure 3.5: Target Plate of Saticon

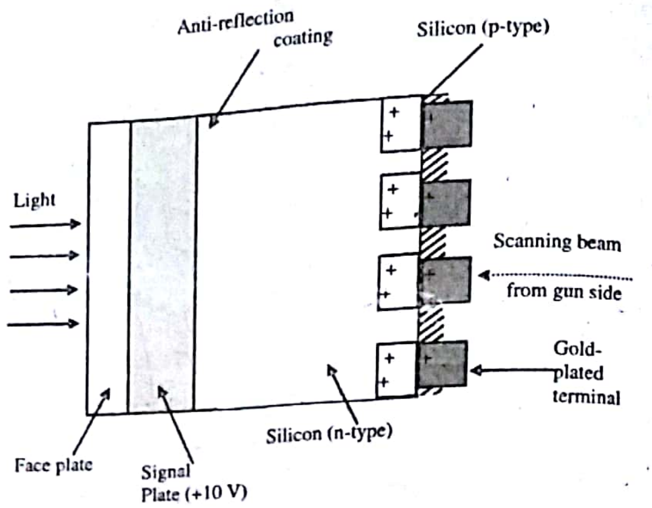


Figure 3.6: Silicon Vidicon Camera Tube

The transparent tin oxide layer allows light to fall on the target but being an electrical conductor, it connects the current path to the load resistor through the conducting.

Working of the target depends on the principle of photoconduction. Free electrons are created inside the semiconductor when light is incident on it. These electrons are removed by the positive voltage through the signal plate and hence, a charge image is formed on the gun side of the target. When a sharply focused electron beam from the electron gun scans the target, the charge at individual pixels is neutralised, resulting in signal current through the load resistor.

Ques 9) What is meant by silicon and Pyroelectric vidicon camera tube with the help of its construction.

Or

Write the short note on:

1) Silicon-vidicon camera tube

2) Pyroelectric vidicon camera tube

Ans: Silicon-Vidicon Camera Tube

It is also called silicon diode array. The target plate consists of a very thin n-type silicon wafer. A thin silicon dioxide layer is applied over the wafer. Then by photo masking and etching process p-type material (boron) is diffused into the wafer at various opening, resulting in an array of silicon photodiodes. A fine layer of gold is deposited for connection on the p-type layer.

Its construction is shown in **Figure 3.6**. This tube is useful in a high light environment which produces image lag in an ordinary vidicon.

Total number of diodes are 540 × 540 (about 300,000). The diodes are reverse biased by applying +10 V to the signal plate.

This reverse bias decrease due to production of electron-hole pairs when light falls on the target. When the electron beam scans the surface, it deposits electrons, neutralizing the p-type diode side to zero potential of cathode as in other photoconductive camera tubes. The diode current generated by this neutralisation causes signal voltage across the load resistor. Typical value of signal current is 7μA peak for white light.

Pyroelectric Vidicon Camera Tube

Pyroelectric vidicon camera tube is used for night vision. Pyroelectric effect means production of voltage by variation of temperature. Triglycine sulphate (TGS) crystal exhibited pyroelectric effect. When this type of detector is used in vidicon format, as shown in **figure 3.7**, it is called pyroelectric vidicon.

When the temperature is constant, no voltage is produced. Infrared light coming from the source modulates the constant ambient temperature and hence is detected. If the temperature changes by δT kelvin, there is change in charge by δQ and hence change in voltage by δV across the target as per equation (1).

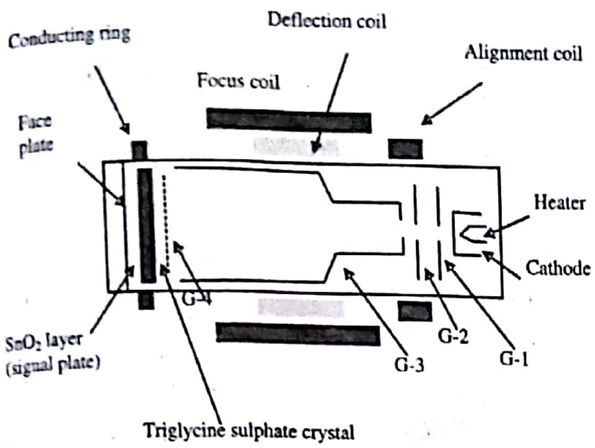
$$\delta Q = P \times A \times \delta T; \text{hence } \delta V = \frac{P \times A \times \delta T}{C} \quad \text{..... (1)}$$

Where,

P = Pyroelectric coefficient of the material

A = Area in m^2 in which incident energy is absorbed

C = Capacitance formed in the target.



Figre 3.7: Pyroelectric Vidicon

Ques 10) Give the detail description of image orthicon camera tube.

Or

Describe the image orthicon camera tube along with its construction.

Ans: Image Orthicon Camera Tube

The main transducer in image orthicon camera tube is a photocathode. It consists of a coating of bismuth-silver-cesium compound on the inside of the glass face plate of the tube. When light from a scene is focused on the photocathode, electrons are emitted from it. The energy of the electrons emitted from different points of the cathode depends on the intensity of light (energy of the photons).

Being a conductor, the photocathode cannot store charge. Hence the design of cathode is such that electrons come out in parallel beam. These electrons are accelerated towards the target, strike an n-type silicon plate placed parallel to the cathode. They cause emission of secondary electrons from the target plate.

The number of secondary electrons emitted from the target depends on the energy of the striking electrons. Loss of the secondary electrons from the semiconductor makes it positively charged.

The positive charge at any point on the semiconductor depends on the number of electrons lost from that point. Thus different points of the target possess different positive charges proportional to the intensity of light of the corresponding points in the scene. Hence a charge image of the picture is formed on the surface of the target.

A **constructional** detail of the image orthicon camera tube is given in **Figure 3.8**. Photocathode is at –400 V and G–6 at –320V. Thus G–6 is positive by 80V with respect to cathode and hence attracts the emitted electrons. These electrons pass through an extremely thin wire mesh screen, having about 300 holes per cm. It is at +400 V. The target plate (n-type silicon) is 50 microns away from the wire mesh.

The electrons from the photocathode varying in energy according to the intensity of light in the picture, strike the target plate at the corresponding points. It emits several secondary electrons for each electron striking it. The secondary electrons dislodged from the target are collected by the wire mesh screen and are returned to the photocathode, and so are lost to the target. The deficiency of electrons results in positive charge in the target, which at any point is proportional to the energy of the striking electrons from the corresponding point of the photocathode.

As the target plate is very thin, about 4 microns, this positive charge extend to the other side of the target plate. Thus charge image of the original picture is formed on the surface of the target plate. The charge at each point of the target remains stored at that point due to target being of glass, till it is not neutralised by the electron beam. Neutralization of charge is discussed below.

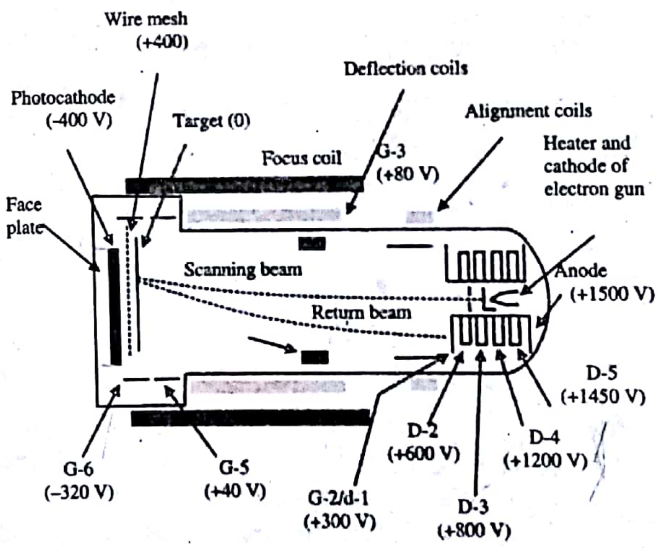


Figure 3.8: Construction Details of Image Orthicon Camera Tube

The charge from each point of the target surface is extracted by deflecting an electron beam produced by the electron-gun. Beam consists of electrons emitted by the thermionic cathode and controlled by control grid G-1 which is kept at –50 V with respect to the cathode. Generally cathode is kept at +50 V and grid at 0 V. The beam is deflected horizontally as well as vertically by sawtooth currents made to flow through respective deflection coils. Interlaced scanning of the target is thus achieved.

The electron beam reaches a point on the target perpendicularly with almost zero velocity due to decreasing voltages, 120 V, 40 V, 0 V, at G-4, G-5 and the target plate, respectively. The beam is rich in electrons and hence neutralises the positive charge by leaving electrons on the target. The beam leaves only as many electrons as are necessitated by the deficiency of electrons in the target, and then returns to the grid G-2 under the influence of positive voltages.

Ques 11) Discuss about the picture tube and also enlist its types.

Ans: Picture Tube

Picture tube is a transducer to convert electrical video signals varying with time into variations of light in space (through width and height of the screen) to reproduce the original picture. It uses the phenomenon of fluorescence and scanning.

A television picture tube can be regarded as a c.r.t. adapted for the special requirements of displaying a picture. Because a large rectangular picture is to be displayed, a rectangular rather than a circular screen is required let avoid unused screen area. The large display area requires a large deflection angle if the tube is not to be excessively long, This large deflection angle, together with the high beam current when white areas an being displayed on the screen, means that magnetic deflection has to be used instead or electrostatic deflection.

It is through considerations such as these that the present day black-and-white television picture tube has evolved.

Types of Picture Tubes

Mainly there are two types of picture tubes:

1) **Monochrome Picture Tube/ B/W System:** The monochrome picture tube is used to produce black and white picture.
2) **Colour Picture Tube/ Colour System:** The colour picture tube is to be used to reproduce original colour in the picture.

Ques 12) What do you mean by monochrome picture tube/black white system.

Or

Draw and discuss the block diagram of black white system.

Or

Define electron gun in black white system and also discuss its elements.

Ans Monochrome Picture Tube/Black White System

A monochrome picture tube consists of an evacuated glass envelope, housing an electron gun to produce a sharply focused electron beam and a fluorescent screen to produce light when the electron beam strikes the phosphor elements on the screen. Also, there are devices to deflect the electron beam horizontally and vertically to enable it to scan the screen through width and height to reproduce the original picture. Construction of the monochrome picture tube is shown in **figure 3.9.**

Electron Gun produces a welt focussed electron beam, using thermionic emission of electrons from a cathode, and a few grids for controlling, accelerating and focusing the electrons.

These elements of the electron gun are described as follows:

1) **Cathode:** It consists of a thoriated oxide of tungsten which is heated by a heater wire placed close to the cathode but without touching it electrically. The material between the heater and the cathode is a thermal conductor, but is an electric insulator. Such a cathode is known as indirectly heated cathode. In this type of cathode, the heater wire is heated by AC voltage. The insulator keeps the AC voltage isolated from the electron beam current and hence prevents production of ac hum in the picture. Heater requires 6.3V AC (at about 400mA).

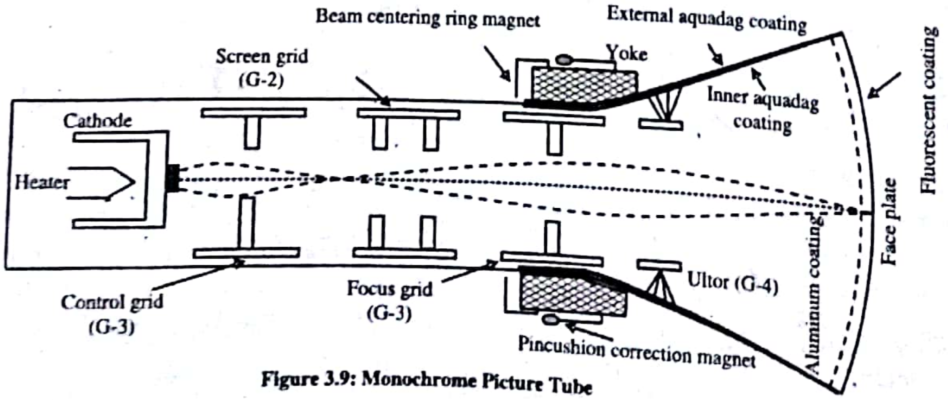


Figure 3.9: Monochrome Picture Tube

2) **Control Grid (G-1):** It is a metallic cylinder of nickel, and has a small aperture which allows the electrons to pass through. Control grid is kept negative with respect to cathode (either the cathode may be kept at positive voltage and grid grounded, or the grid may be kept at negative voltage and cathode grounded). The grid bias is -30 to -50V variable. Negative grid bias enables the control grid to control the space charge of the electrons coming out of the cathode.

3) **Screen Grid (G-2):** It is a metallic cylinder of nickel with internal baffles to restrict the beam to a narrow path. The screen grid has a positive voltage of about 400V and hence accelerates the electrons in the beam. It is therefore also called accelerating anode. Effect of voltages on the screen and control grids are such that the electrons converge in the apace between these two grids.

4) **Focus Grid (G-3):** This grid provides the required electrostatic field to prevent spreading of the electrons and help in producing a sharp spot on the screen with the help of a final anode described below.

5) **Fluorescent Screen:** It consists of a rectangular face plate made of an optically flat glass of high quality. The glass is 1.5 cm thick to withstand the outside air pressure against the internal vacuum. There is a very thin aluminium coating on the back of the phosphor surface (towards gun side). It improves brightness by reflecting the light back to the front.

Ques 13) Give the brief description of colour system with the help of diagram.

Ans: Colour System/Colour Picture Tube

Colour television transmission and reception can be described as a system which reproduces colour images (**Figure 3.10**). A colour scene is scanned and converted into a corresponding video signal by a colour television camera. In turn, this signal is processed by the colour-television system and finally supplied to a colour-television picture tube.

Thus, the original colour scene is reproduced as an identical colour image on the screen of the picture tube. There are various steps in the processing of the video signal. These steps do not necessarily have to be the same in different countries or not even in the same country, but the end result would always be the same. However, due to number of limitations, this system did not find practical application.

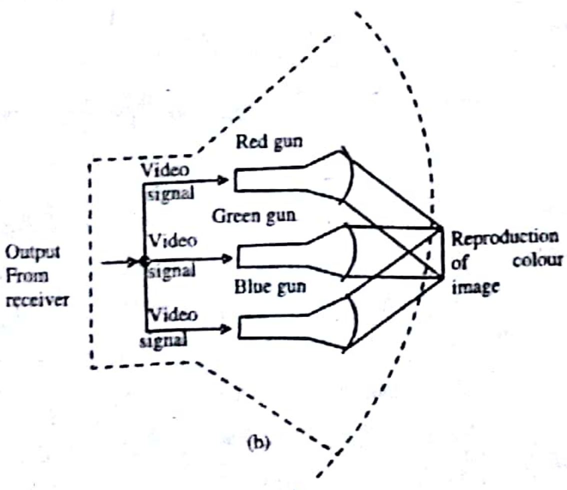


Figure 3.10: Colour Picture Tube Recombines the Primary Hues to Reproduce the Colour Image

Ques 14) Discuss the bandwidth in television system

Ans: Bandwidth in Television System

It is the highest video frequency related to the time taken in scanning two adjacent pixels. **Figure 3.11(a)** shows 4 successive pixels, and **Figure 3.11(b)** shows the corresponding 2 cycles of brightness. Time period T is the time taken in scanning one cycle or 2 adjacent pixels.

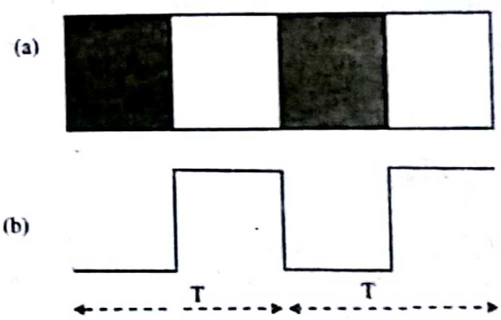


Figure 3.11: (a) Four Adjacent Pixels of Black and White Shades, and (b) Two Cycles of Brightness Signal Corresponding to Pixels of Figure (a)

If t is time in seconds for scanning one line (excluding retrace time), then R_H pixels are scanned in t seconds. Therefore 2 pixels shall be scanned in$2t/R_H$ second.

Thus time period T is given by equation (1).

$$t = \frac{2 \times t}{R_H}$$

Video bandwidth will be given by equation (2).

$$\text{Bandwidth} = \frac{1}{T} = \frac{R_H}{2t}$$

Ques 15) Describe the PAL TV along with description of colour schematics.

Ans: PAL (Phase Alternating Line) TV
PAL (Phase Alternating Line) is the colour encoding conversion standard. India has adopted the CCIR PAL-D colour TV.

This has the advantage over the NTSC system that phase changes that produce colour changes are automatically cancelled by the phase alternation circuit used in PAL system.

Also, it has more scanning lines (625 compared to the 525 in NTSC), which results in better picture clarity.

Basic of Colour Schematics
It has been found that any colour can be generated by employing three basic colour viz... red, green, and blue.

For example, white colour may be obtained by adding 30% of red, 59% of green and 11% of blue. That is, white (or luminance) signal.

$$Y = 0.3R + 0.59G + 0.11B \quad .....(1)$$

To get white colour, other similar ratios are also possible. However, each of these white colours are unique and different from one another. Equation (1) is the ratio of the white colour adopted for colour-television scheme.

Ques 16) Draw and discuss the block diagram of PAL TV transmitter.

Ans: PAL TV Transmitter
A PAL colour TV transmitter, called the PAL encoder. **Figure 9.3** shows the simplified block diagram of PAL coder.The TV transmitter consists of two transmitters: an audio transmitter and a video transmitter. The audio transmitter is an FM transmitter and produces frequency modulated audio signal, which is applied to a unit called the **Diplexer.**

The video transmitter is an AM transmitter. It consists of a colour TV camera that produces the luminance (light-intensity) signal Y, and the colour or chrominance signal U and V.

The Y signal is added with the respective synchronizing, blanking, and equalizing pulses, and TV colour cameras convert any given scene into a combination of R, G, and B, depending on their content in the scene televised.

For example, if as scene contains red only, the camera will output a current that is proportional to the intensity of the red colour only.

If the scene contains a combination of red and green (i.e., yellow) a certain proportion, then the output signal will contain the components of red and green in the same proportion.

To transmit a colour TV signal, and to receive it, we require 4 components of colours, viz, the red (R), the green (G), the blue (B), and the luminance (Y) components.

Even though four components are needed, all these four can be generated from the same set of the three primary colour R, G, and B, by using appropriate proportion. Thus, in PAL system, we use the signal Y, U and V.

$$U = R - Y - R\,(0.3R + 0.59G + 0.11B)$$
$$= 0.7R - 0.59G - 0.11B$$
$$V = B - Y = -0.3R - 0.59G + 0.89B$$

The U and V signals are added to the Y signal and transmitted. In the receiver we use these three signals to recover the original image.

It is applied to the adder, video amplifier, modulator and final power amplifier, as shown.

The chrominance signals U and V are modulated by a colour subcarrier frequency, which is produced by a colour subcarrier oscillator. In the PAL system, this frequency is fixed very accurately at 4.43361875 (≈4.43) MHz It is applied directly to balanced modulator BM 1, and applied through a ± 90° phase-shifter switch to balanced modulator BM 2.

The second input of BM 1 is the chrominance signal U, and its output is the term U cos $\omega_s t$, where, ω_s is the subcarrier frequency. In the same way, the output of BM 2 is ±V sin $\omega_s t$, as this is a ± 90° phase-shifted signal.Now, the ±V sin $\omega_s t$ term and U cos $\omega_s t$ term are added in adder 2, which produces the output,

$$C = U \cos \omega_s t \pm V \sin \omega_s t$$

This summation produces the desired side band terms, which can then be added on the Y signal as its side band using Adder1, as shown in below **figure 3.12.**

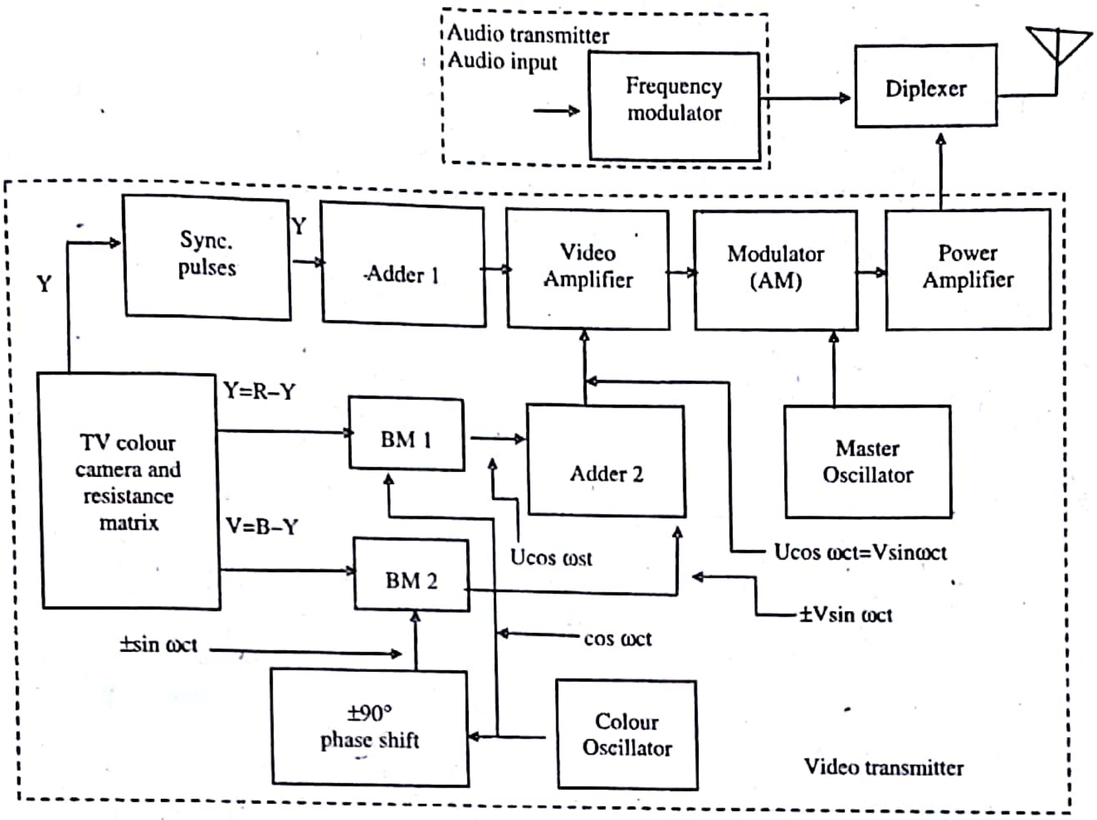


Figure 3.12: Block Diagram of PAL Colour TV Transmitter

The luminance Y signal acting as the carrier and the chrominance signal as its side-bands get amplified in the video amplifier, RF modulation in an AM modulator, power amplified and applied to the diplexer as its second input.

The diplexer is a circuit that isolates the audio and video signals from interfering with each other, and at the same connects them to the transmitting antenna.
It may be noted that the PAL transmitter makes use of the vestigial sideband transmission (VSB) scheme. This is a double sideband-full carrier (DSB-FC) AM scheme with the full upper sideband and a vestige (portion) of the lower sideband being used for transmission. For the Indian 625-line PAL system, the VSB scheme is shown in **figure 3.12.**

Ques 17) Give the brief description of PAL TV Receiver with its suitable block diagram.

Or

Discuss the block diagram of PAL TV Receiver.

Ans: PAL TV Receiver
The simplified block diagram of PAL receiver is shown in **figure 3.13**. The signal from a TV station is captured by the receiving antenna and applied to an RF tuner. The RF tuner is a single tuning block consisting of an RF amplifier, a local oscillator, and a mixer. The block tunes in the desired station and converts the signal frequency to an intermediate frequency (IF). For TV, the IF is 39.5 MHz, a very large value compared to the IF (455 KHz) in radio receiver. This large value is used because TV operates in the VHF or higher bands.

The output of the IF consists of audio, colour and video carrier frequencies and their sidebands. The IF signals is applied to the video detector, which rectifies this and outputs the three carrier frequencies, which are to be separated now into respective components.

Block Diagram of PAL TV Receiver

The block diagram of PAL TV receiver shown in the **figure 3.13**, the video detector output is applied as such to a video amplifier, which amplifies the signal as the luminance signal Y, and applies it to the Y-control grid of the TV picture tube. It may be noted that it is the Y component that gives the necessary light intensity to the scene reproduced on the TV screen.

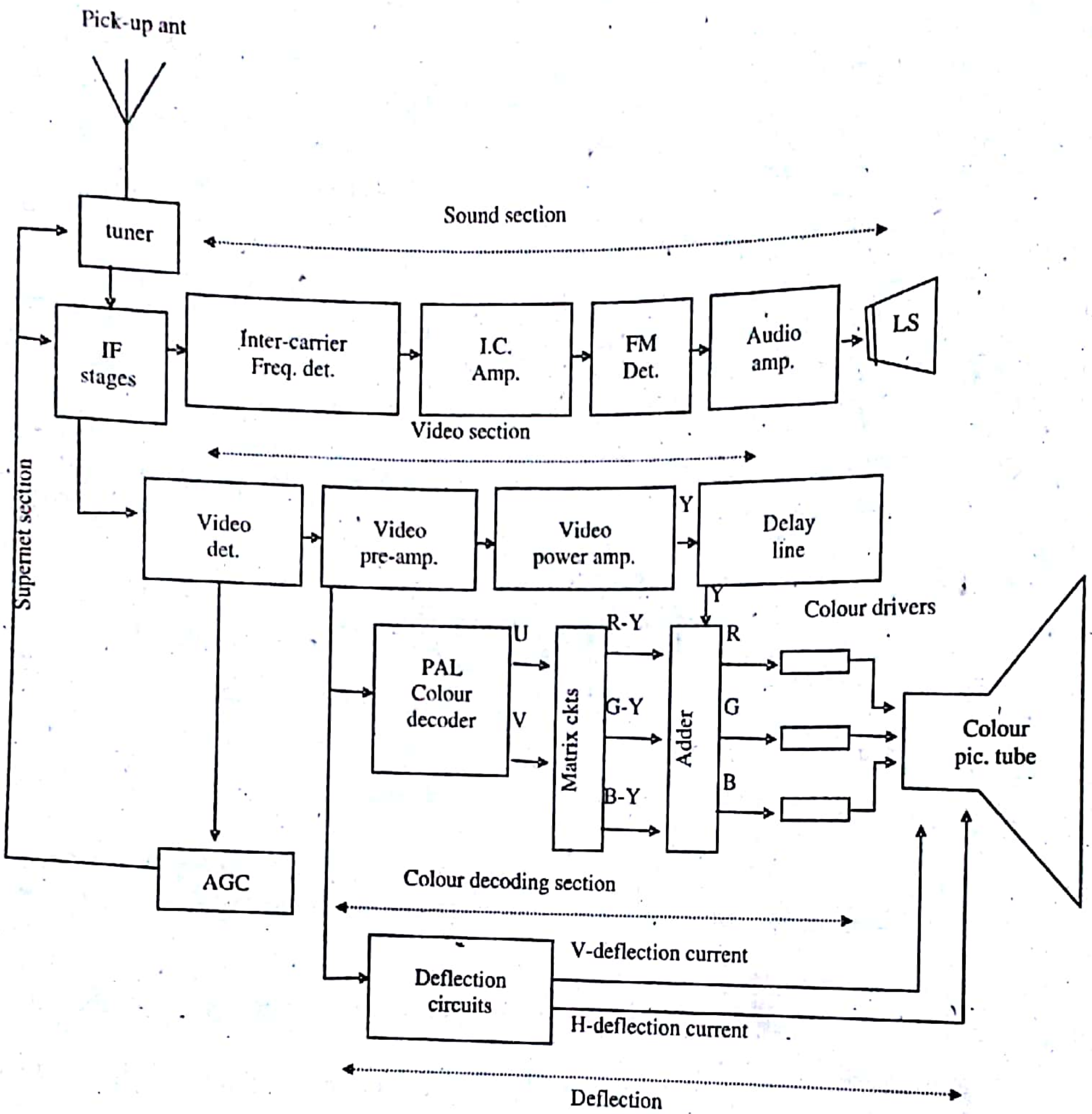


Figure 3.13: Block Diagram of PAL Receiver

The **block diagram of PAL TV receiver** is discussed below:

1) **Super Heterodyne Section:** It is similar to the super-heterodyne section in a monochrome receiver except that the inter-carrier frequency signal is taken out (for sound section) from the last IF stage instead of from the output of the detector or video pre-amplifier.

   The reason for this is to save the picture from the beat signal produced by mixing chroma signal (3.58 MHz) with inter-carrier frequency (45 MHz).Beat of the two signals would be 92KHz in NTSC system which falls within the video bandwidth and would cause interference known as sound in picture.

2) **Video Section:** Video detector recovers CCVS signal. Luminance signal (Y) is amplified and is delayed by a line so that it reaches the adder simultaneously with the chroma signal. Chroma signal propagated slowly in the colour decoder circuit due to lower bandwidth and hence Y signal has to be delayed.

3) **Colour Decoder:** CCVS signal is available at the video detector's output. It goes to the video pre-amplifier which amplifies the CCVS signal. The effective bandwidth of Y signal for colour receiver is about 3.13 MHz (1.3 MHz lower than 4.43 MHz).

Y signal is separated from C signal either by using a comb filter or preferably by a simple band pass filter to pass a bandwidth of 3.13 MHzY signal goes to the video power amplifier and C goes to the decoder.

There are two types of colour decoders:
i) Simple PAL.
ii) Delay line PAL.

Ques 18) Draw and describe the block diagram of basic CCTV system with its complete discription.

Or

What do you understand by CCTV System? Also enlist its application.

Ans: CCTV (Closed-Circuit Television)

CCTV (Closed-Circuit Television) is a television system that makes use of co-axial cables for transmission display and reception of TV signal.

CCTV in principle consists of a TV camera (black and white or colour), a cable, and a monitor. The camera shoots the scene to be televised, convert it into corresponding video signals, and transmits to a monitor through the cable connecting the transmitter and the monitor.

In this process no RF modulation such as AM or FM is used. The monitor receives the video signal and convert it back to the sent signal (transmitted by the transmitter) on its screen.

Figure 3.14 shows a simple CCTV schematic.

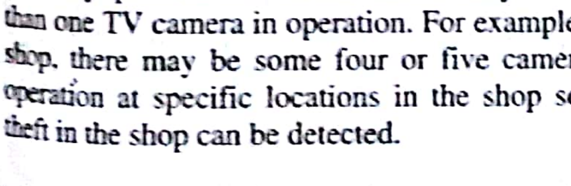


Figure 3.14 Basic CCTV System

In many pratical situations, threre may be more than one TV camera in operation. For example, in a shop, there may be some four or five cameras in operation at specific locations in the shop so that theft in the shop can be detected.

Block Diagram of CCTV System

When several cameras and monitors are used, video-switching scheme is to be used. Switching can be either manual or automatic. In manual switching, an operator can manually scan the cameras and supply the signals so obtained to preferred monitors. In automatic switching scheme, scanning and resulting display are done automatically. Such a scheme is shown in figure 3.15.

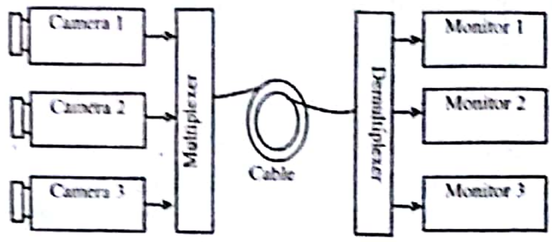


Figure 3.15: Block Diagram of CCTV System

As shown in the **figure 3.15**, there are three video cameras and three monitors in this system. Assume that these cameras are respectively placed above three counter of a shop to monitor the transactions in them. We use a multiplexer at the transmitting section, which will connect cameras in the sequence 1, 2, 3, 1, 2, 3….. To the cable. The cable, in turn, will take these respective signals to a de-multiplexer, which will connect monitor 1 to receive signal from camera 1, and monitor 2 to receive signal from camera 2 and so on.

Thus we find that the multiplexer and de-multiplexer in synchronized mode will transfer signals from respective cameras to respective receiver. Since the switching of the MUX and DEMUX are done at fast rates, the received signals will show no discontinuity and we get the feeling that each of the cameras is connected to its own receiver through individual cables.

CCTV Applications

The CCTV applications are as follows:
1) In security system in shops, offices, etc,
2) As outside display in meeting halls, wedding halls etc.
3) As displays in sports grounds.
4) For monitoring patients in hospitals.
5) For showing operation to persons standing outside the operation theatre.
6) For monitoring the conditions of various system that are not directly accessible and also dangerous.

Ques 19) Explain about the cable TV system. Also draw its block diagram.

Ans: Cable TV (CATV)

Cable television (CATV) system is a very popular form of television distribution system. For TV signal reception from individual stations individual antennas is requiring. Since each of these station operators on a different frequency, we require as many numbers of antennas as the number of stations received. This is a very cumbersome, expensive, and foolish method. Instead of several individual antennas, a single broadband antenna can be used. But such broadband antennas can receive only a limited number of stations.

For receiving very large number of stations, cable TV system is to be used, which in turn makes use of artificial satellites of the transmission and reception of TV signals. There are currently several communication satellites in operation in the outer space, each of which is capable of supporting several TV channels. Thus using one satellite antenna, and corresponding low-noise amplifier, a cable TV operator is able to receive several TV channels. If he installs several such antennas on a roof top, he can receive signal from a large number of TV stations.

After downloading signals from a satellite (or satellites), a cable operator converts them back into individual channel signals from several stations using hos down conversion equipment, multiplexes and distributes them to his subscribers using co-axial cables or optical fiber cables that capable of carrying several TV channels through them. The name cable TV network has been evolved from this type of distribution network.

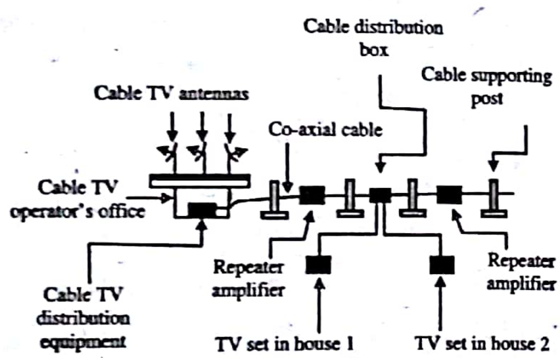


Figure 3.16: Cable TV Distribution

The block diagram of cable TV is shown in the **figure 3.16**, the distribution line consists of the co-axial/optical fiber cable, repeater amplifiers, distribution boxes, and cable-supporting posts. The signals transmitted through cables get attenuated due to various reasons. Therefore, we have to use repeaters.

Repeaters are receive-amplify-retransmit units. For co-axial microwave cables, we require repeaters at every 1-km distance. In the case of optical fibers, however, the repeater spacing varies from about 10 km to 100s of km depending on the type of cable used.

The distribution boxes shown in the **figure 3.16** are distribution points that contain tapping points from where cable-TV connections are given to individual houses. Distribution boxes also contain attenuators that are used to regulate signal strength to individual houses and impedance matching circuits that avoid reflection when tapping are made. Cable TV system has become a major area of business offering employment to thousands of people all around the world. It also has become a major source of entertainment to many houses.

Ques 20) What do you mean by high definition television system? And also explain HDTV formats.

Ans: High-Definition Television (HDTV) System

High-definition television (HDTV) is a TV system that gives more clarity (definition) in the picture reproduced on a TV screen. Consider a 625-line TV system displayed on a TV screen of 25 cm height. Then the spatial frequency of this screen is 625/25 = 25 lines/cm.

Now, assume if TV system has 1250 line. Then the spatial frequency of 25 cm high screen is 1250/25 = 50 lines/cm. Thus the same picture with a 25 lines/cm is now displayed with much higher (50 line/cm) spatial frequency, which means that the clarity of the second picture is much more than the first picture. This better clarity is also called better resolution.

The high-definition TV originally proposed contained doubling the number of existing scanning lines. Thus the 525-line NTSC (American) TV system had proposed a 1050-line HDTV system and the CCIR 625-lineTV system had proposed a 1250-line HDTV system.

However, due to several practical and technical problems, these could not be practically realized in their original formats.

HDTV Formats

The two new HDTV formats are in operation. They are:

1) **Progressive-Scan Mode (120p):** The 720p (720 progressive scan) system scans 720 lines per picture at 50 picture per second using the

progressive, sequential, or continuous-scanning mode. There are 1280 pixels (picture elements) on each line, resulting in a total resolution of $1280 \times 720 = 9.212 \times 10^5$ pixels per picture.

2) **Interlased-Scan Mode (1080i):** The 1080i (1080 interlaced scan) system employs 1080 scan lines, in interlaced mode. The 1080 lines are divided into even and odd fields, with each field containing 540 scanning lines. There are 1920 pixels on each line and the overall resolution is $1920 \times 1080 = 2.0736 \times 10^6$ pixels per picture.

The standard TV picture width to height aspect ratio is 4:3. This has been modified to 16:9 in HDTV system. This gives a wider screen display, and hence a more realistic TV picture reproduction.

Ques 21) Give the brief discirption of high definition television system. Also explain its advantages and disadvantages.

Or

Enlist the disadvantages of high definition television system.

Ans: High Definition Television System (HDTV) Transmitter

An HDTV transmitter is the same type of TV transmitter used for the standard low-resolution TV transmission. As stated above, if the number of scanning lines is doubled, then the channel width also is to be doubled.

However, channel width has to be preserved. So data compression teaching are adopted which will:

1) Provide higher picture resolution
2) At the same time, preserve bandwidth at the standard value of 5.5 MHz

This means that to increase our standard TV transmitter sweep frequency from 625 lines to 720 or 1080 lines per picture. This is not a difficult task since the clock generators that can produce clock frequency of over 4 GHz. However, in transmission over a standard existing channel, this increased frequency cannot be used as the channel bandwidth required will have to be doubled.

It is in this context that frequency-compression algorithms are used. Thus we conclude that an HDTV transmitter will be the regular type of TV transmitter that uses a higher sweep frequency and makes use of data-compression techniques.

High Definition Television System (HDTV) Receiver

To receive HDTV signal, an HDTV tuner is required. Conventional TV receivers require set-top boxes, which contains built-in HDTV tuner sections. However, specially constructed HDTV receivers have in-built HD tuners.

Further, in these receivers decompression algorithms are used to expand the compressed data. Except for this, the HDTV receiver will be very similar to the conventional TV receiver.

Advantages of HDTV

The advantages of HDTV over analog transmission are as follows:

1) Digital channels require shorter bandwidth, which means that more number of channels can be transmitted.
2) Can provide non-television services such as Multimedia and Interactive TV are possible.
3) Several programs can be multiplexed on the same channel.
4) Electronic program guides and additional languages, spoken or subtitled.
5) HDTV signals do not generate ghosts (reflections).
6) Noise is greatly reduced.

Disadvantages of HDTV

The disadvantage of HDTV is as follows:

1) HDTV images have some picture defects that are not present in regular TV transmission. This is due to the bandwidth limitations.
2) When a compressed image is decompressed to get the original data, some distortion may take place due to the inadequacy in the compression algorithm also.
3) HDTV signals have a significant delay when channels are manually changed.
4) Due to antenna mismatch, in digital TV, reception of a channel may become difficult leading to its no reception at all.
5) Multi-path fading is a serious problem for digital TV reception, which leads to intensity fluctuations in TV images.
6) Interlacing reduces the overall image quality and introduces image flickering and crawling scan lines.

RADAR AND NAVIGATION

Ques 22) Give the detail description of radar and also discuss its principle.

Or

Discuss the basic elements of radar system.

Ans: Radar

RADAR, stands for Radio Detection and Ranging, is an electromagnetic system for the detection of objects by radio waves or location of reflecting object such as aircraft, ship, spacecraft, vehicles, people and the natural environment.

Human eye though has great resolution and can identify colour, yet radar is considered to be more powerful as it can operate under conditions impervious to human vision/optical and infrared sensors such as darkness, haze, fog, rain and snow. Its ability to measure distance with high accuracy and in all weather conditions makes it a powerful electronic eye. Radars can be classified basically into two categories Pulsed Radars and Continuous-wave Rader.

Principle of Operation of Basic Radar

The radar operates by transmitting electromagnetic waves by directive antennas into a specified volume in space to search for targets. These targets reflect a fraction of incident energy back to the radar. The reflected electromagnetic energy is called radar returns or echoes. These echoes are processed by the radar receiver to obtain the information about the target.

The information includes range, angular position, velocity, size, direction of movement and identification of friendly and enemy targets, etc. In brief, the principle of operation depends on transmitted and reflected waves. The principle of operation is explained by the **figure 3.17**.

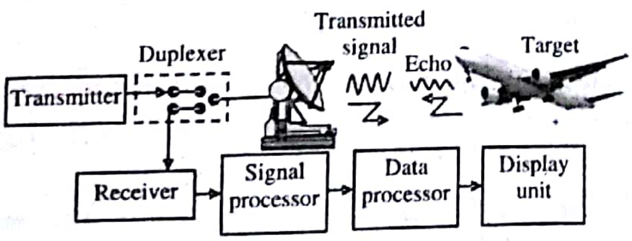


Figure 3.17: Basic Radar System

Basic Elements of Radar System

The basic radar consists of:

1) **Transmitter:** It provides a high power signal at a specified frequency to the transmitting antenna. The transmitter components include magnetron, klystron, Traveling-Wave Tube (TWT) and Cross-Field Amplifier (CFA), etc.

2) **Duplexer:** In most of the radars, the same antenna is used for transmitting and receiving. This is done by duplexer. It connects the transmitter and the receiver alternatively to the antenna.

   It is used to protect the receiver from the high transmitter power. Duplexer provides connection between Tx and antenna while transmission. It provides connection between antenna and Rx during reception. When the two antennas are used for Tx and Rx, then duplexer is not required.

3) **Antenna:** The antenna converts transmitter electrical signal into electromagnetic waves and sent these towards the target. It also receives echoes from the target and converts into electrical signal.

   This electrical signal is given to the receiver. Most often, the same antenna is used for transmitting and receiving purposes through a switch called duplexer.

4) **Receiver:** The electrical signal of echo from the antenna is given to the receiver. The receiver amplifier conditions the signal suitably. The output or the receiver is given to the signal processor unit. It includes ECCM circuits. It does the following tasks:
   i) Amplifies weak echoes
   ii) Separates noise and clutter.
   iii) Sends the amplified signal to a signal processor.

5) **Signal Processor:** It processes the received signals to increase the signal-to-noise ratio. It provides the information about the target.

6) **Data Processor:** It processes the information of the target and then stores it. It converts the entire data into easily understandable coordinates and then gives to the display unit.

7) **Display Unit:** In this unit the information of target is displayed in a form useful to radar operators and supervisors. The input to the display unit is from the data processor. The display unit usually consists of a cathode ray tube.

Ques 23) Enlist the advantages and disadvantages of basic radar system.

Ans: Advantages of Basic Radar

1) It acts as a powerful eye.
2) It can see through fog, rain, snow, darkness, haze, clouds and any insulators.
3) It can find out the range, angular position, location and velocity of targets.

Disadvantages of Basic Radar

1) It cannot recognise the colour of the targets.
2) It cannot resolve the targets at short distances like human eye.
3) It cannot see targets placed behind the conducting sheets.
4) It cannot see targets hidden in water at long ranges.
5) It is difficult to identify short range objects.

Ques 24) Derive the radar equation for the calculation of radar range. Also discuss about the factors affecting range of radar.

Ans: Radar Equation

Consider a pulse of very short duration (in the range of microseconds) is sent to a target, such as an enemy aircraft. Let the pulse reflected from the aircraft be collected by a receiver. This situation is shown in **figure 3.18.**

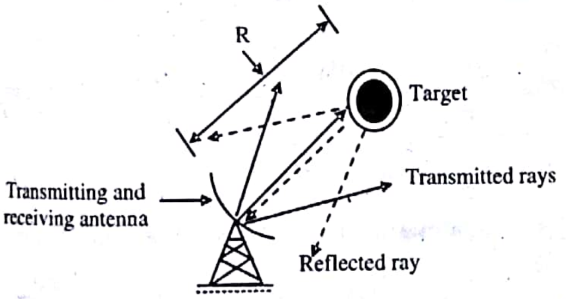


Figure 3.18: Diagram to Develop Radar-Range Equation

Let P_T = power of the transmitted radar waves. Assuming that the transmitter is an isotropic radiator (one that radiates equally in all directions), the power spreads in all direction around the transmitter uniformly. Therefore, the power transmitted per unit area around the transmitter is in a sphere of radius R:

$$P_T' = \frac{P_T}{4\pi R^2} w/m^2 \qquad ..... (1)$$

Now, if we assume that the transmitting antenna has a parabolic reflector, then the transmitted power gets concentrated along the direction of the reflector. Let G_T = gain of the transmitting antenna. Then the transmitted power per unit area will be,

$$P_T'' = \frac{P_T G_T}{4\pi R^2} w/m^2 \qquad ..... (2)$$

Let us assume that we have a target at distance R. Then the transmitted power absorbed by the target is given by,

$$P_T'' = \frac{P_T G_T A}{4\pi R^2} \text{watts} \qquad ..... (3)$$

Where, A= area of cross-section of the target. Assuming that the target reflects the entire power absorbed by it all around it, the reflected power at distance R from the target

$$P_R' = \frac{P_T''}{4\pi R^2} = \frac{P_T G_T A}{4\pi R^2} \times \frac{1}{4\pi R^2} = \frac{P_T G_T A}{(4\pi R^2)^2} W/m^2 \qquad ..... (4)$$

If we assume that the receiving antenna has a surface area of Sm^2, then the power received by it is given by,

$$P_R = \frac{P_T G_T A S}{(4\pi R^2)^2} \text{watts} \qquad ..... (5)$$

It can be proved that the target area,

$$S = \frac{\lambda^2 G_R}{4\rho} m^2 \qquad ..... (6)$$

Using the expression for S from equation (6) into equation (5), we get:

$$p_r = \frac{P_T G_T G_R \lambda^2 A}{(4\pi)^3 R^4} \text{watts} \qquad ..... (7)$$

It is quite common to use the same antenna for transmission and reception in radars. Therefore, we have $G_T = G_R = G$. Using this, we get the radar-range equation:

$$P_R = \frac{P_T G^2 \lambda^2 A}{(4\pi)^3 R^4} W \qquad ..... (8)$$

Rearranging **Equation (8)** yields the radar (range) Equation:

$$R = \sqrt[4]{\frac{P_T G^2 \lambda^2 A}{(4\pi)^3 P_R}} m \qquad ..... (9)$$

Equation (9) can be used to determine the distance to a target, since the values of P_T and G should be known. Target area A is a variable, and depending on this, that the range varies.

Factors Affecting the Range of Radar
Consider the radar-range equation,

$$r = \sqrt[4]{\frac{P_t G^2 \lambda^2 A}{(4\pi)^3 P_R}}$$

From this equation, the radar range is dependent on the following factors:

1) **Transmitter Power P_T:** To double the range, we have to increase the transmitter power by a factor of $2^4 = 16$.

2) **Wavelength of the Transmitted Electromagnetic Wave:** By using EM waves of longer wavelengths, range can be increased.

3) **Area of the Target:** In fact, we have no control over this. However, for the same target, various angles of its view with respect to the transmitter give various areas of cross-section.

   Choosing that cross-section offering maximum area of the target to the transmitting antenna optimizes the range.

4) **Received Power P_R:** This must be as minimum possible to increase the target range. This means that we must have a highly sensitive receiver that can detect even the weakest signal returned from a given target.

5) **Gain of the Antenna used:** It can be seen that gain of the antenna must be very large to increase the range.

Ques 25) Find the following:

1) **The power density at a target situated at a distance of 50km from a radar radiating a power of 100MW from a lossless isotropic antenna.**

2) **If this radar now employs a lossless isotropic antenna with a gain of 5000 and the target has a radar cross-section of 1.2 m², then what is the power density of the echo signal at the receiver?**

3) **If the minimum detectable signal of the radar is 10^{-8}MW and the wavelength of the transmitted energy is 0.02m, then what is the maximum range at which the radar can detect targets of the kind mentioned in (b)?**

4) **What is the effective area of the receiving antenna?**

Ans:

1) Power radiated by the radar $P_t = 100MW = 100 \times 10^6 W$.
   Distance of the target = R = 50Km = 50×10^3m

   Power density at the target,

$$\hat{P}_d = \frac{P_t}{4\pi R^2} = \frac{100 \times 10^6}{4\pi \times (50 \times 10^3)^2}$$

$$= 03183 \times 10^{-2} W/m^2$$

2) Antenna gain G = 5000.
   Radar cross-section of the target = σ = 1.2m²
   Power density at the target when a directive antenna is used

$$P_d = \hat{P}_d G = 0.3183 \times 10^{-2} \times 5000 = 15.915 W/m^2$$

   The amount of power intercepted by the target,

$$\hat{P} = P_d \sigma = 15.915 \times 1.2 = 19.098W$$

   This power is now reflected back to the receiving antenna. Hence, the power density of the echo signals at the receiver.

$$P_d^r = \frac{\hat{P}}{4\pi R^2} = \frac{19.098}{4\pi \times (50 \times 10^3)^2} = 6.079 \times 10^{-10} W/m^2$$

3) The wavelength of the transmitted energy, λ = 0.02m.
   The minimum detectable signal $S_{min} = 10^{-8}$ mW = 10^{-11}W.
   Then the maximum range R_{max} is given by (2.17) as

$$R_{max} = \left[\frac{P_t G^2 \lambda^2 \sigma}{(4\pi)^3 S_{min}}\right]^{1/4} = \left[\frac{100 \times 10^6 \times (5000)^2 \times (0.02)^2 \times 1.2}{(4\pi)^3 \times 10^{-11}}\right]^{1/4}$$

$$= 88183.6m = 88.1836Km.$$

4) The effective area of the receiving antenna

$$A_e = \frac{G\lambda^2}{4\pi} = \frac{5000 \times (0.02)^2}{4\pi} = 0.0159m^2$$

Ques 26) Define pulsed radar and also discuss the principle of operation of pulsed radar.

Ans: Pulsed Radar

Pulsed radar is radar, which transmits electromagnetic waves in the form of bursts or pulses. It measures the time interval between transmitted and received pulses. The pulse radar also measures the range of a target. It measures the slant range by determining the time taken by transmitted pulse to come back to the receiver.

The pulsed radar is used to determine direction and distance of the target. It also measures altitude of the target, if necessary.

Principle of Operation

Pulsed radar transmits a high power and high frequency electromagnetic waves in the form of pulses. The pulse interval is so designed that the echoes are conveniently received before next pulse is transmitted.

It is used to find distance and direction. The altitude of the target is also determined indirectly from the antenna position and propagation time of the pulse signal.

The pulsed radars transmit and receive a chain of modulated pulses. Range is extracted from the two-way time delay that takes place between a transmitted and received pulse. Doppler frequency is obtained from the range data.

This method is satisfactory if the range is not changed drastically over a time interval of Δt. If the range is changed drastically, pulsed radars use doppler filter bank. Its operation depends upon:
1) Carrier frequency,
2) Pulse width,
3) Modulation and
4) Pulse repetition frequency.

The carrier frequency is decided on the basis of radar mission. The pulse width decides bandwidth and range resolution. Different modulation techniques are used to enhance radar performance and capabilities. The pulse repetition frequency is designed to avoid Doppler and range ambiguities. It is also required to optimise average transmitted power.

Ques 27) Describe the block schematics of pulse radar.

Or

Explain the transmitter and receiver part of pulse radar.

Ans: Block Schematics of Pulse Radar

Figure 3.19 shows the block schematic of pulsed radar. It consists of a transmitter and a receiver part. The operation of the transmitter and receiver are synchronized by a synchronizing circuit.

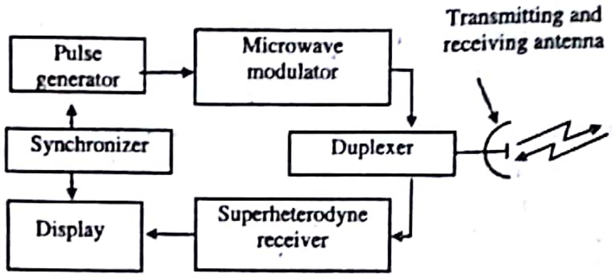


Figure 3.19 Block Schematic of Pulsed Radar

The radar transmitter, channel and receiver parts are discussed below:

1) **Radar Transmitter:** The radar transmitter consists of an RF pulse generator and microwave modular. These are discussed below:
   i) **RF Pulse Generator:** A typical hard-tube pulser is shown in **Figure 3.20**. The circuit consists of a transistor switch to which a capacitor, a diode, and a magnetron tube are connected.

   When the transistor T is off, the capacitor C charges through the diode D in the polarity shown. Now, the triggering pulses applied to the base of T will turn it on, and C discharges through T and the microwaves tube magnetron, as shown in **Figure 3.20**. This discharge of C through the magnetron produces sharp high-power pulses at microwave frequencies, which can be directed to the required target(s) by the antenna.

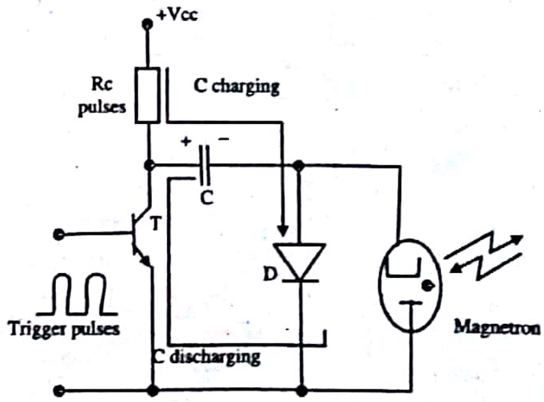


Figure 3.20: Hard-Tube Pulser

ii) **Microwave Modulation:** The output of the hard-tube pulsar is applied to for modulation of microwave, then the signal applied to antenna through duplexer.

2) **Radar Channel:** The duplexer is used to isolate the transmitter from the receiver. This kind of isolation between the transmitter and the receiver is required because the same antenna is used for both transmission and reception. It may be noted that, if the same antenna is used for transmission and reception, the transmitted high-power pulses will interfere with the operation of receiver.

3) **Radar Receiver:** The pulses reflected from various targets are received by the antenna and fed to a super-heterodyne radio receiver through the duplexer, as stated above.

i) **Superheterodyne Receiver:** The receiver demodulates the information regarding the targets contained in the reflected pulses, and applies this to display, unit. The display unit then shows the nature and range of the target(s) under consideration.

ii) **Plan-Position Indicator (PPI):** Plan position indicator is a major form of radar display. This scheme makes use of a cathode-ray tube with a rotating electron beam, as shown in **figure 3.21**. While the electromagnetic radiation from the transmitter scans the targets all around it at a finite scanning frequency, the rotating electron beam in the CRT display also rotates at the same frequency.

This results in the targets being displayed on the CRT screen at their respective positions in the space around the transmitting antenna, relative to the position of the antenna. The CRT screen can be calibrated to indicate the range of the target from the antenna.

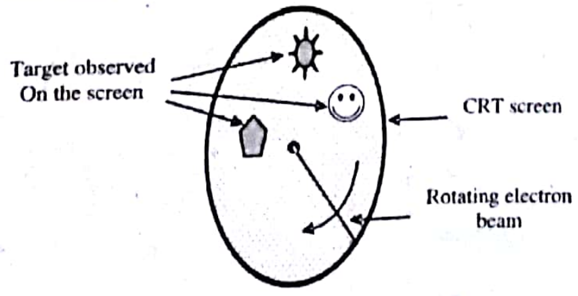


Figure 3.21: PPI Radar Display

Ques 28) Enlist the application of radar in measurement and navigation.

Or

Discuss about the application of radar in IFF System and LORAN.

Ans: Application of Radar in Measurment

Radars are used for accurate measurement of distance. Such an instrument is called a DME (Distance Measuring Equipment). **Figure 3.22** shows a long-jump pitch. Consider a jump being made. A radar fixed on the jumping base line, will send a pulse to a mirror fixed on the jump-falling point. The reflected pulse is captured by the receiver part of the radar. The time delay between the transmitted and received pulses is measured by the radar and using the computer in the equipment, this time delay is converted into its corresponding distance. Very accurate measurement is possible by this equipment.

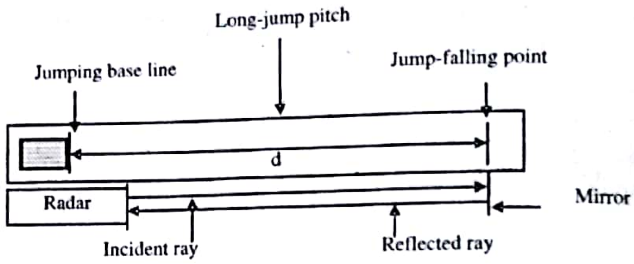


Figure 3.22: Distance Measuring Equipment

Application of Radar in IFF and LORAN (Long Range Navigation) System

The application of radar in IFF and LORAN is system discussed below:

1) **Radar Beacons in IFF System:** Radar beacons are established along air-plane routes and sea shores for the identification of the location of a moving aircraft or sailing ship, as the case may be. A radar beacon comprises of radar transmitter a receiver, and an antenna. **Figure 3.23** shows a radar beacon established to aid navigation in an aerodrome.

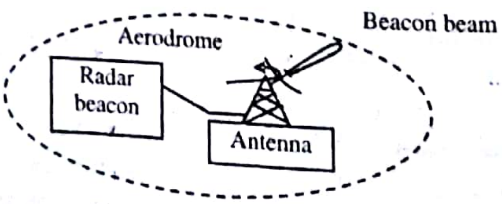


Figure 3.23: Radar Beacon

A beacon starts its action on the reception of a radar pulse from a target such as a moving aeroplane or ship. If this pulse is in a form of a particular frequency or code acceptable to the beacon, the beacon replies with its own identity and the moving target will know its position with respect to the location of the beacon. When this is a military application, the radar beacon is called an **IFF (identification of friend and foe)** system.

2) **Radar Beacons in LORAN:** A LORAN (Long-Range Navigation) is a navigation system using radar beacons to help moving ships to find their position with respect to and distance to a given shore. The operation of a LORAN system can be explained with reference to **figure 3.24.**

In **figure 3.24**, we notice that a typical LORAN system will have three radar beacons, located on the shore line of a given sea at distance of a few kilometers from each other. These beacons will be transmitting high-power radar pulses at regular intervals of time to the sea. They also maintain a constant time difference between themselves.

For example, assume that Beacon 1 is the master beacon and it emits radar pulses at time intervals t_1, t_4, t_7, t_{10},.... Let us also assume that Beacon 2 emits pulses at intervals t_2, t_5, t_8, t_{11},...., and that Beacon 3 emits pulses at intervals t_3, t_6, t_8, t_{12}, ... It must also be noted that $(t_1 - t_2) = (t_4 - t_5) =$ .... Similarly, $(t_1 - t_3) = (t_4 - t_6) =$ .... These equations shows that the time difference between power transmissions between two radar systems.

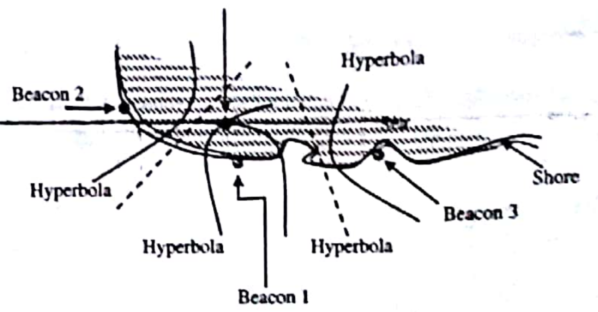


Figure 3.24 LORAN System.

Application of Radar in Navigation

The application of radar is as follows:

i) Radar is used to detect target such as enemy aircrafts and ship, and range them.
ii) Radar is used to detect and find the speed of fast moving objects.
iii) Radar beacons are used in navigation to help aircraft and ships to fix their location.
iv) Radar beacons are used to indicate standard time in maritime applications.

Module 4

Digital Communication

DIGITAL COMMUNICATION

Ques 1) Define digital communication and its principle.

Or

What do you mean by digital communication? Also list the advantages and disadvantages of digital communication.

Ans: Digital Communication

In digital communication, transmission of information takes place in digital form. The major reason of digital communication recently is the availability of wideband chann el such as optical fiber, satellite channel etc.

Principal of Digital Communication

Communication systems that use such a digital sequence as an interface between the source, and channel input (and similarly between the channel output and final destination) as shown in **figure 4.1**.

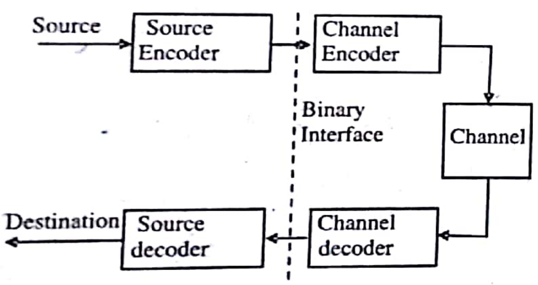


Figure 4.1

Figure 4.1 is placing a binary interface between source and channel. The source encoder converts the source output to a binary sequence and the channel encoder (often called a modulator) processes the binary sequence for transmission over the channels. The channel decoder (demodulator) recreates the incoming binary sequence (hopefully reliably), and the source decoder recreates the source output.

The idea of converting an analog source output to a binary sequence was quite revolutionary in 1948, and the notion that this should be done before channel processing was even more revolutionary.

By today, with digital cameras, digital video, digital voice, etc., the idea of digitizing any kind of source is common place even among the most technophobic. The notion of a binary interface before channel transmission is almost as common place.

For example, refer to the speed of our internet connection in bits per second.

Need of Digital Communication

The need of digital communication is discussed below:

1) Digital hardware has become so cheap, reliable, and miniaturized, that digital interfaces are eminently practical.
2) A standardized binary interface between source and channel simplifies implementation and understanding, since source coding/decoding can be done independently of the channel, and, similarly, channel coding/decoding can be done independently of the source.
3) A Standardized binary interface between source and channel simplifies networking, which now reduces to sending binary sequences through the network.
4) One of the most important of shannon's information theoretic results is that if a source can be transmitted over a channel in any way at all, it can be transmitted using a binary interface between source and channel. This is known as the source/channel separation theorem.

Advantages

1) Now a common format of digital can be used for different variety of signals. For example we can transmit voice, video, photo by using the binary digits.
2) By using error correcting and error detecting codes, we can improve the noise performance of system.

Disadvantage

1) Now we need the wideband channels due to increased bandwidth of signal.
2) In digital system, the systems are very complex.

Ques 2) Draw and describe the block diagram of digital communication system.

Ans: Block Diagram of Digital Communication

The basic **block diagram** of a digital communication system is shown in **Figure 4.2.** The source of information signal are also non-electrical quantity so cannot be processed directly.

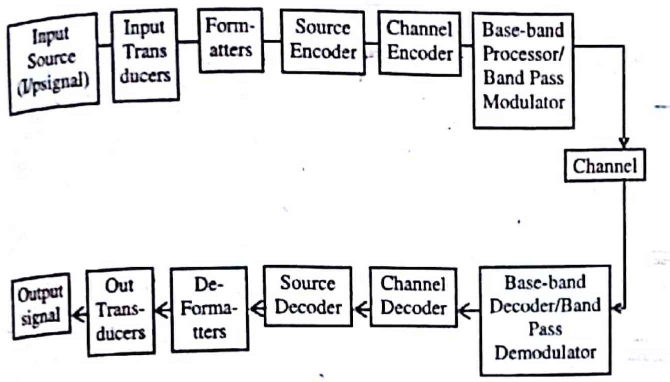


Figure 4.2: Element of Digital Communication System

The elements of block diagram of digital communication system are described below:

1) **Input Transducer:** Input transducers convert the applied non-electrical quantity into electrical quantity.

2) **Formatter:** Now the received signal should be converted into digital form that task is initiated by formatter; formatter converts an analog signal into a digital signal. If the source of information already provides the information in digital form such as computer there is no need of formatter.

3) **Source Encoder:** Source encoder simply provides a representation in the form of binary digits to the applied digital signal.

4) **Channel Encoder:** Channel encoder plays a very important role. According to the channel properties, channel encoder provides some reduency bits to the message so that at the receiver side, a error free signal can be reproduced.

5) **Baseband Modulator:** For very low-speed wireless transmission directly base-band signal can be transmitted. For high speed transmission as in case of analog transmission, digital data is modulated with the help of high frequency carrier. This task is performed by the baseband modulator.

6) **Channel**: The communication channel is the media via which the signal is transmitted from the transmitting end to the receiving end. Communication channel is the part of model at which the maximum noise is added to the signal.

7) **Baseband Demodulator**: At the receiver side we just reciprocal sequence of transmitting end. By baseband decoder by pulse shaped 'or' line coded data, original data is recovered back. If at the transmitter side we have used baseband modulator, at the receiver side; by demodulator, the data is demodulated to get original signal.

8) **Channel Decoder** At the receiver side, by the channel decoder, the redundancy bits are removed back to the signal.

9) **Source Decoder:** By using source decoder, we are able to decode the digital digits so that the original data is now available in digital form.

10) **Deformatter:** Deformatter converts data into analog and that data is applied to an appropriate output device.

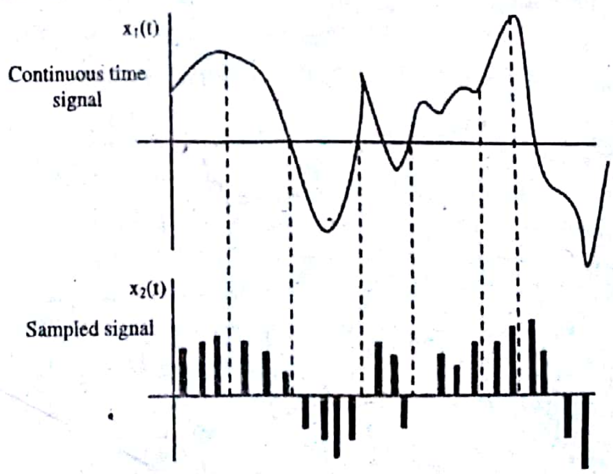
Ques 3) Give the detail description of sampling process and sampling theorem. Also give the mathematical description of sampling theorem.

Ans: Sampling Process

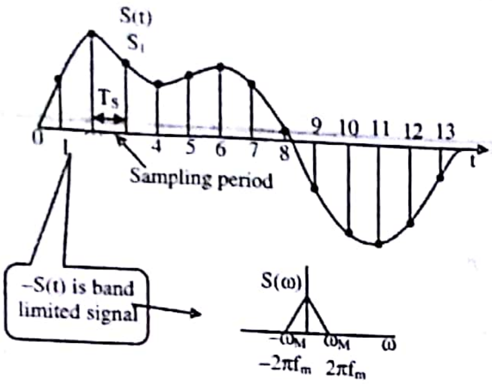
The process of converting continuous time signals into equivalent discrete time signals, can be termed as **Sampling**. A certain instant of data is continually sampled in the sampling process. The following **figure** indicates a continuous-time signal x(t) and a sampled signal xs(t). When x(t) is multiplied by a periodic impulse train, the sampled signal xs(t) is obtained.

A sampling signal is a periodic train of pulses, having unit amplitude, sampled at equal intervals of time Ts, which is called as the **Sampling time**. This data is transmitted at the time instants T_s and the carrier signal is transmitted at the remaining time.

Sampling Theorem



The sampling theorem shows that a continuos-time signal which is a band-limited to f_m Hz can be represented perfectly by a series of samples spaced T_s ($\leq \frac{1}{2} f_m$) seconds apart. T_s called **sampling period.**



The band-limited signal is a signal which its Fourier transform is non-zero for $-2\pi f_m < \omega < 2\pi f_m$. When m(t) is sampled uniformly at intervals of T_s seconds, the resultant sequence is denoted by $m(nT_s)$, for all integer values of n.

Let's define a sampling rate $f_s = 1/T_s$ which is the number of samples per second.
Thus, the sampling theorem states that a band-limited signal can be recovered completely from a set of samples taken at the rate of $f_s (\geq 2f_m)$ samples per second.

The preceding sampling theorem is often called the uniform sampling theorem for baseband or low-pass signal.
The minimum sampling rate, $2f_m$ samples per second, called the Nyquist rate; its reciprocal $1/2f_m$ is called **Nyquist interval.**

In each case the sampling rate must be at least twice the highest frequency contained in the analog signal, according to the Nyquist criterion. **Figure 4.3** shows the waveform for ideal sampling.

In ideal sampling, an arbitrary analogue signal is sampled by a train of impulses at uniform intervals, T_s. An impulse (having virtually no pulse width) is generated at each instant of sampling.

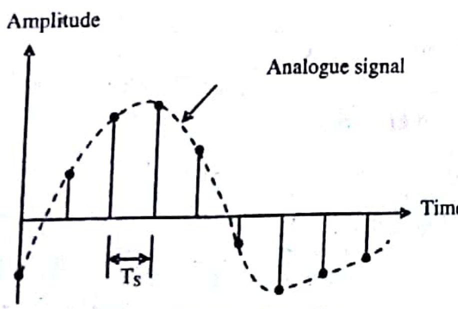
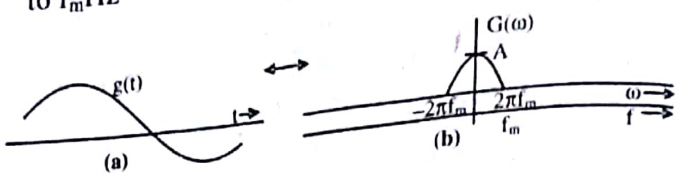
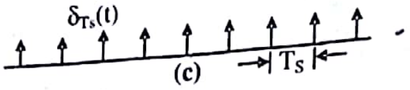


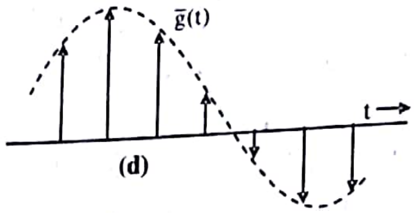
Figure 4.3: Ideal Sampling

Mathematical Description of Sampling Theorem
Let g(t) be a signal whose spectrum is band-limited to f_mHz



Sampling g(t) at a rate of f_s Hz (f_s samples per second) can be accomplished by multiplying g(t) by an impulse train $\delta_{T_s}(t)$, consisting of unit impulses repeating peridically every T_s seconds where $T_s = 1/f_s$





This results sampled signal g(t) consists of impulses spaced every T_s seconds.
The n^{th} impulse, located at $t = nT_s$ has a strength $g(nT_s)$ (the value of g(t),(at $t = nT_s$),

$$\bar{g}(t) = g(t)\delta_{T_s}(t) = \sum_n g(nT_s)\delta(t - nT_s)$$

Because the impulse train $\delta_{T_s}(t)$ is a periodic signal of period T_s, it can be expressed as a Fourier series. The trigonometric Fourier series is

$$\delta_{T_s}(t) = \frac{1}{T_s}[1 + 2\cos\omega_s t + 2\cos\omega_s t + 2\cos 3\omega_s t + ...] \quad ..... (1)$$

$$\left(\omega_s = \frac{2\pi}{T_s} = 2\pi f_s\right)$$

Therefore,

$$\bar{g}(t) = g(t)\delta_{T_s}(t) \quad ..... (2)$$

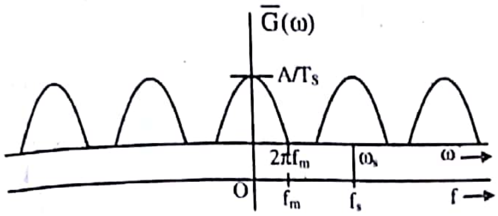
$$\bar{g}(t) = g(t)\delta_{T_s}(t)$$

$$= \frac{1}{T_s}\begin{bmatrix} g(t) + 2g(t)\cos\omega_s t + 2g(t)\cos 2\omega_s t \\ + 2g(t)\cos 3\omega_s t + ... \end{bmatrix}$$

To find $\bar{G}(\omega)$, the Fourier transform of $\bar{g}(t)$, we take the Fourier transform of the right-hand side of equation (2), term by term. The transform of the first term in the brackets is $G(\omega)$. The transform of the second term $2g(t)\cos\omega_s t$ is $G(\omega-\omega_s) + G(\omega+\omega_s)$. This represents spectrum $G(\omega)$ shifted to ω_s and $-\omega_s$.

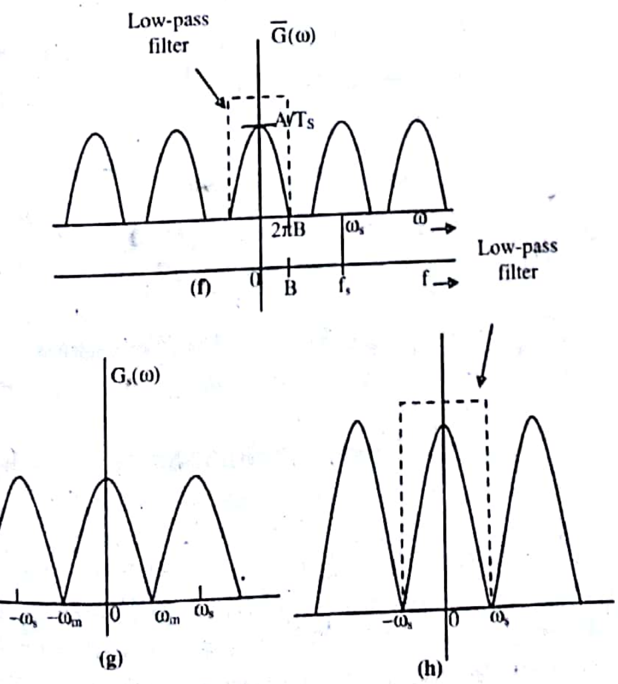
Similarly, the transform of the third term $2g(t)\cos 2\omega_s t$ is $G(\omega-2\omega_s)+G(\omega+2\omega_s)$, which represents the spectrum $G(\omega)$ shifted to $2\omega_s$ and $-2\omega_s$, and so on to infinity.

This mean that the spectrum $\overline{G}(\omega)$, consists of $G(\omega)$ repeating periodically with period ω_s



If we are to reconstruct $g(t)$ from $\overline{g}(t)$, we should be able to recover $G(\omega)$ from $\overline{G}(\omega)$. This is possible if there is no overlap between successive cycles of $\overline{G}(\omega)$.

Thus, as long as the sampling frequency f_s is greater than twice or equal the signal bandwidth B (in hertz), $\overline{G}(\omega)$ will consists of non-overlapping repetitions of $G(\omega)$. When this is true, $g(t)$ can be recovered from its samples $\overline{g}(t)$ by passing the sampled signal $\overline{g}(t)$ through an ideal low-pass filter of bandwidth B Hz. The minimum sampling rate $f_s=2f_m$ required to recover $g(t)$ from its samples $\overline{g}(t)$ is called the **Nyquist rate** from $g(t)$, and the corresponding sampling interval $T_s=1/2f_m$ called the **Nyquist interval** for $g(t)$.



Ques 4) Define sampling rate and also discuss the methods of sampling.

Or

Give the brief description of natural sampling and flat-top sampling. Also list its disadvantages.

Ans: Sampling Rate

To discretise the signals, the gap between the samples should be fixed. That gap can be termed as the sampling period T_s.

$$\text{Sampling Frequency} = \frac{1}{T_s} = f_s$$

Where, T_s = Sampling time

f_s = Sampling frequency or sampling rate

Methods of Sampling

There are two methods of sampling which is discussed as follows:

1) **Natural Sampling:** Natural sampling refers to PAM signals when tops of the sampled pulse retain their natural shape during the sample interval.

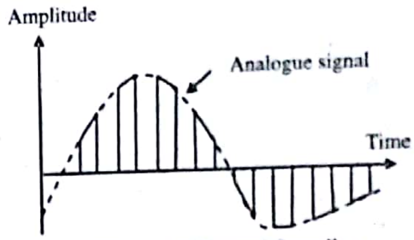


Figure 4.4: Natural Sampling

Figure 4.4 shows the waveform for natural sampling. In natural sampling, an arbitrary analog signal is sampled by a train of pulses having finite short pulse width occurring at uniform intervals. The amplitude of each rectangular pulse follows the value of the analogue information signal for duration of the pulse.

Disadvantages of Natural Sampling

The disadvantages of natural sampling are as follows:

i) It is difficult for an analogue-to-digital converter to convert the natural sample to a digital code.

ii) In fact, the output of analogue-to digital converter would continuously try to follow the changes in amplitude levels and may never stabilize on any code.

2) **Flat-Top Sampling:** Flat-top sampling refers to PAM signals when the tops of the sampled pulses remain constant during the sample integral.

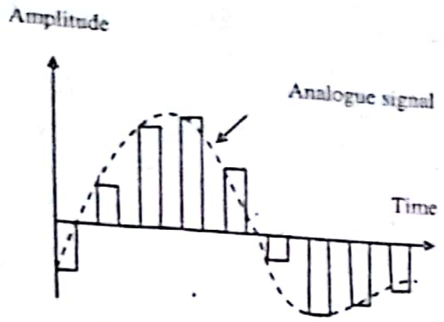


Figure 4.5: Flat-Top Sampling

Figure 4.5 shows the waveform for flat-top sampling.

In flat-top sampling, an arbitrary analogue signal is sampled by a train of pulses having finite short pulse width occurring at uniform intervals. The amplitude of each rectangular pulse is retained as the value of the analog information signal at the leading edge of the pulse.

Note: A sample-and-hold circuit is used to keep the **amplitude** constant during each pulse in flat-top sampling **process.**

Disadvantages of Flat-Top Sampling

The disadvantages of flat-top are as follows:

i) The use of flat-top PAM samples results into amplitude distortion.

ii) There is delay by $T_b/2$, where T_b is the width of the pulse that results into lengthening of the samples during transmission.

iii) At the receiver, amplitude distortion as well as delay causes errors in decoded data

Ques 5) Define pulse modulation technique and also enlist its types.

Ans: Pulse Modulation Technique

Pulse communication systems differ from continuous wave communication systems in the sense that the message signal or intelligence to be transmitted is not supplied continuously as in case of AM or FM. In turn, it is sampled at regular intervals and it is the sampled data that is transmitted. All pulse communication systems fall into either of two categories, namely analogue systems and digital systems. Analogue and digital communication systems differ in the mode of transmission of sampled information.

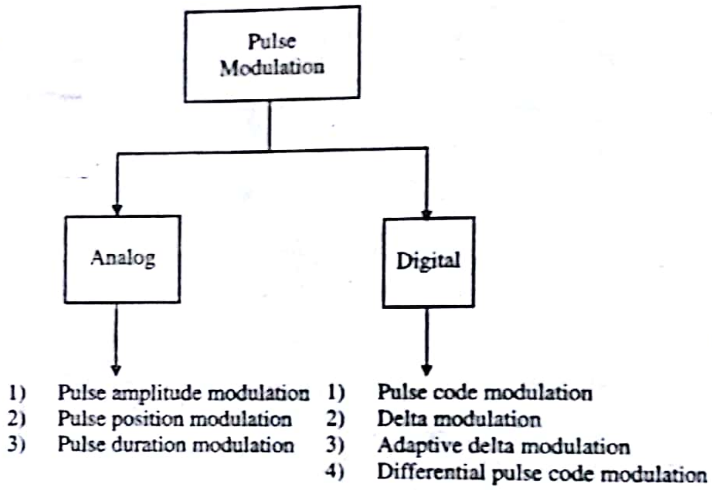
In the case of analogue communication systems, representation of the sampled amplitude may be infinitely variable whereas in the case of digital communication systems, a code representing the sampled amplitude to the nearest predetermined level is transmitted.

In pulse modulation, the carrier is no longer a continuous signal but consists of a pulse train. Now some parameter of pulse is varied according to the instantaneous value of the modulating signal. When the signal is in the form of pulses, the pulse modulation method is used.

In concern with a pulse, amplitude of the pulse, width of the pulse, location of the pulse are important parameters. Depending on them, we have various pulse modulation techniques such as Pulse Amplitude Modulation (PAM), Pulse Width Modulation (PWM), Pulse Frequency Modulation (PFM), Pulse Code Modulation (PCM) etc.

Types of Pulse Modulation Systems

These are two types of pulse modulation systems:



Ques 6) Discuss about the types of pulse modulation techniques with its proper waveform.

Or

Write a short note on:
1) **Pulse Amplitude Modulation (PAM)**
2) **Pulse Width Modulation (PWM)**
3) **Pulse Position Modulation (PPM)**

Ans: Types of Pulse Modulation Techniques

The types of pulse modulation techniques are as follows:

1) **Pulse Amplitude Modulation (PAM):** In the case of pulse amplitude modulation (PAM), the signal is sampled at regular intervals and the amplitude of each sample, which is a pulse, is proportional to the amplitude of the modulating signal at the time instant of sampling. The samples, shown in **figure 4.6**, can have either a positive or negative polarity.

In a single polarity PAM, a fixed D.C. level can be added to the signal, as shown in **figure 4.6(c)**

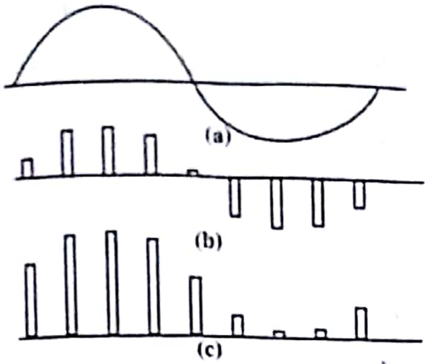
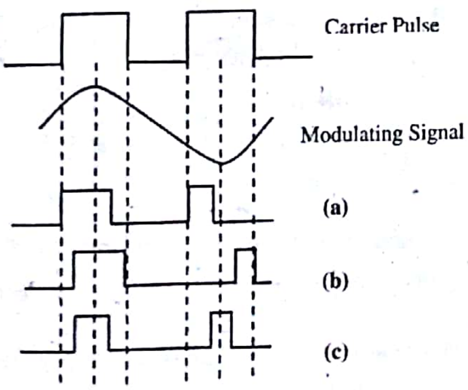


Figure 4.6: Pulse Amplitude Modulation

These samples can then be transmitted either by a cable or used to modulate a carrier for wireless transmission. Frequency modulation is usually employed for the purpose and the system is known as PAM-FM.

2) **Pulse Width Modulation (PWM):** In Pulse Width Modulation (PWM) or pulse duration modulation (PDM) or pulse time modulation (PTM) technique, the width or the duration or the time of the pulse carrier varies, which is proportional to the instantaneous amplitude of the message signal.

The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude to a desired level, and hence the noise is limited. The following **figure** explains the types of Pulse Width Modulations.



There are three types of PWM.

i) The leading edge of the pulse being constant, the trailing edge varies according to the message signal. The waveform for this type of PWM is denoted as in the above **figure (a).**

ii) The trailing edge of the pulse being constant, the leading edge varies according to the message signal. The waveform for the type of PWM is denoted as in the above **figure (b).**

iii) The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal. The waveform for this type of PWM is denoted as in the above **figure (c).**

3) **Pulse Position Modulation (PPM):** Pulse Position Modulation (PPM) is an analog modulation scheme in which, the amplitude and the width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.

The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and the receiver in sync. These sync pulses help to maintain the position of the pulses. The following figures explain the Pulse Position Modulation.

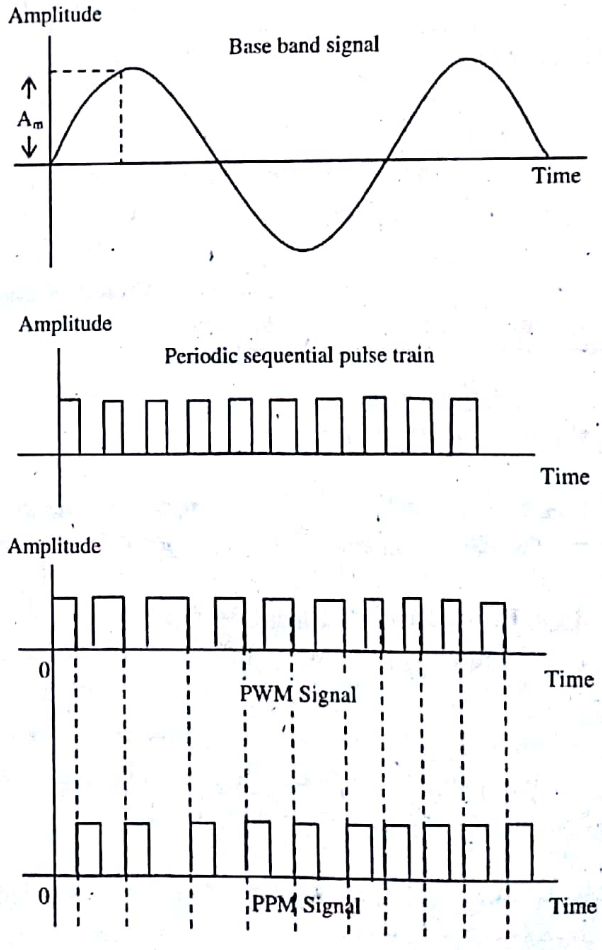


Figure 4.7

Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing edge of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses.

Ques 7) Give the comparison of various pulse modulation techniques.

Ans: Comparison between PAM, PWM, and PPM
The following table presents the comparison between three modulation techniques.

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and the width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

Ques 8) Describe the pulse code modulation with the help of its diagram.

Or

Draw the schematic diagram of pulse code modulation system with its proper description.

Ans: Pulse Code Modulation (PCM)
Two basic operations in the conversion of analog signal into the digital is time discretization and amplitude discretization in the context of PCM, the former is accomplished with the sampling operation and the latter by means of quantization.

In addition, PCM involves another step, namely, conversion of quantized amplitudes into a sequence of simpler pulse patterns (usually binary). Generally it is called as code words. (The word code in pulse code modulation refers to the fact that every quantized sample is converted to an R-bit code word.)

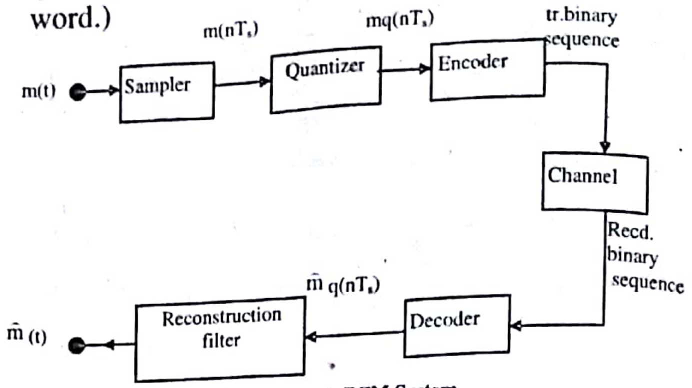


Figure 4.8: A PCM System

Figure: 4.8 show the PCM system. Here m (t) is the information bearing message signal that is to be transmitted digitally. m (t) is first sampled and then quantized. The output of the sampler is $m(nT_s) = m(t)\big|_{t=nT_s}$. T_s is the sampling period and n is the appropriate integer. $f_s = \frac{1}{T_s}$ is called the sampling rate or sampling frequency. The quantizer converts each sample to one of the values that is closest to it from among a pre-selected set of discrete amplitudes. The encoder represents each one of these quantized samples by an R-bit code word.

This bit stream travels on the channel and reaches the receiving end. With f_s as the sampling rate and R-bits per code word, the bit rate of the PCM system is $Rf_s = \frac{R}{T_s}$ bits/sec. The decoder converts the R-bit code words into the corresponding (discrete) amplitudes. Finally, the reconstruction filter, acting on these discrete amplitudes, produces the analog signal, denoted by $\hat{m}(t)$. If there are no channel errors then $\hat{m}(t)=m(t)$.

Ques 9) Describe briefly about the PCM encoder and decoder with its proper block diagram.

Or

Discuss about the types of pulse code modulation system.

Ans: Types of Pulse Code Modulation System
The pulse code modulation system divided into two parts which is described as follows:

1) **PCM Encoder:** The elements which come under PCM encoder are discussed as follows:
   i) **Low Pass Filter:** This filter eliminates the high frequency components present in the

input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

ii) **Sampler:** This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component W of the message signal, in accordance with the sampling theorum.

iii) **Quantiser:** Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer reduces the redundant bits and compresses the value.

iv) **Encoder:** The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections (LPF, Sampler, and Quantiser) will act as an analog to digital converter. Encoding minimizes the bandwidth used.

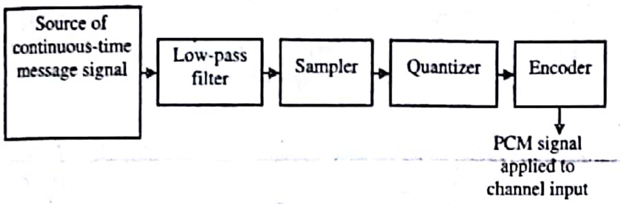


Figure 4.9 (a): PCM Encoder

2) **PCM Decoder:** The elements which come under PCM encoder are discussed as follows:

i) **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, and also to increase its strength.

ii) **Decoder:** The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

iii) **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.

iv) Hence, the pulse code modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

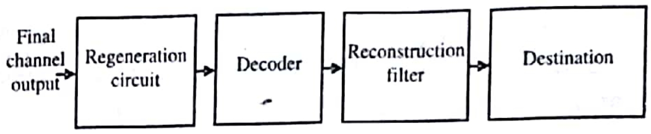


Figure 4.9 (b): PCM Decoder

Ques 10) Enlist the application of data communication.

Or

Write the various application of digital communication.

Ans: Application of Data Communication/ Digital Communication

The applications of data communication/ digital communication are as follows:

1) It is used in military application for secure communication and missile guidance.
2) It is used in image processing for pattern recognition, robotic vision and image enhancement.
3) It is used in digital signal processing.
4) The digital communication systems used in telephony for text messaging etc.
5) It is used in space communication where a spacecraft transmits signals to earth.
6) It is used in video compression and speech processing.
7) It is used in digital audio transmission and also in data compression.

Module 5

Satellite Communication

SATELLITE COMMUNICATION

Ques 1) What is mean by satellite communication? And also discuss the principle and features of satellite communication.

Or

Define satellite communication with the help of its principle of operation.

Ans: Satellite Communication

Satellite communications is the use of satellite technology in the field of communications. The services provided by satellite communications are voice and video calling, internet, fax, television and radio channels.

Satellite communications can provide communication capabilities spanning long distances and can operate under circumstances or conditions which are inoperable for other forms of communication. Communication satellite comprises of a transponder, antenna, communication payload, switching systems, command, and control system.

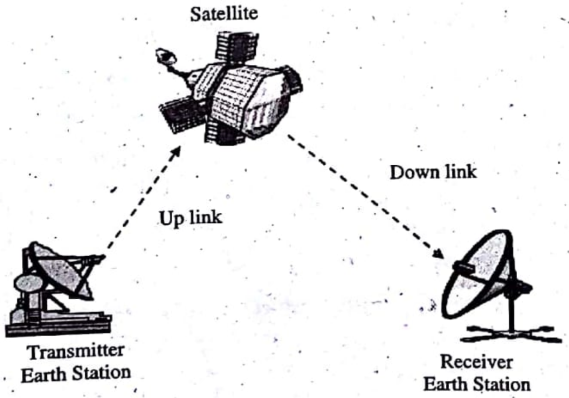


Figure 5.1: Satellite Communication

Principle of Operation of Satellite Communication

The transmission of signal from first earth station to satellite through a channel is called as uplink. A repeater is a circuit which is provided in a satellite, which increases the strength of the received signal and then transmits it. But, this repeater works as a transponder. Similarly, the transmission of signal from satellite to second earth station through a channel is called as downlink.

Uplink frequency is the frequency at which, the first earth station is communicating with satellite. The satellite transponder converts this signal into another frequency and sends it down to the second earth station. This frequency is called as downlink frequency. In similar way, second earth station can also communicate with the first one.

The process of satellite communication begins at an earth station. Here, an installation is designed to transmit and receive signals from a satellite in an orbit around the earth. Earth stations send the information to satellites in the form of high powered, high frequency (GHz range) signals.

The satellites receive and retransmit the signals back to earth where they are received by other earth stations in the coverage area of the satellite. Satellite's footprint is the area which receives a signal of useful strength from the satellite.

Features of Satellite Communication

1) Satellites used in satellite communications are usually in geostationary orbit. Some of them are placed in highly elliptical orbits.
2) Satellite communications can provide global availability. It can not only land masses but also maritime areas as well. Large distances can thus be covered quiet easily.
3) One of the main advantages provided by satellite communication is the superior reliability unlike other forms of communication. It does not need terrestrial infrastructure for operation.
4) Satellite communication could provide superior performance as uniformity and speed are much more pronounced than other forms of communication.
5) Scalability is higher in case of satellite communications.

6) Deployment cost is higher than most forms of communications in case of satellite communications.

7) As it is less vulnerable than other forms of communication, it is highly used in defense departments.

8) Satellite communications also provide weather information.

9) It can be helpful during times of disasters as the services rarely fail.

10) High amount of data can be transmitted with the help of satellites.

Ques 2) Discuss about the advantages, disadvantages and applications of satellite communication.

Ans: Advantages of Satellite Communication

1) Area of coverage is more than that of terrestrial systems
2) Each and every corner of the earth can be covered
3) Transmission cost is independent of coverage area
4) More bandwidth and broadcasting possibilities

Disadvantages of Satellite Communication

1) Launching of satellites into orbits is a costly process.
2) Propagation delay of satellite systems is more than that of conventional terrestrial systems.
3) Difficult to provide repairing activities if any problem occurs in a satellite system.
4) Free space loss is more
5) There can be congestion of frequencies.

Applications of Satellite Communication

1) Radio broadcasting and voice communications
2) TV broadcasting such as Direct To Home (DTH)
3) Internet applications such as providing Internet connection for data transfer, GPS applications, Internet surfing, etc.
4) Military applications and navigations
5) Remote sensing applications
6) Weather condition monitoring & forecasting

Ques 3) Define the multiple access (MA) technique and also write its types.

Ans: Multiple Access (MA) Techniques

Multiplexing deals with how multiple signals can utilize a single resource that is sampling, modulation etc. Whereas multiple access on the other hand, deals with which signal can utilize which particular resource frequency allocation/time slot allocation etc. When more than two nodes send at the same time, the transmitted frames collide. All collide frames are lost and the bandwidth of the broadcast channel will

be wasted. We need multiple –access protocol to coordinate access to multipoint or broadcast link (Nodes or stations are connected to or use a common link). Multiple access protocols are needed in wire and wireless LANs and satellite networks Channelization is a multiple-access method in which the available bandwidth of a link is shared in time, frequency, or through code, between different stations.

Types of Multiple Access Technique

The types of multiple access technique are as follows:

1) Frequency Division Multiple Access (FDMA)
2) Time Division Multiple Access (TDMA)
3) Code Division Multiple Access (CDMA)
4) Space Division Multiple Access (SDMA)

Ques 4) Explain FDMA (frequency division multiple access) techniques and also enlist its features.

Or

Draw the schematic of FDMA technique with complete description.

Ans: FDMA (Frequency Division Multiple Access)

This was the initial multiple-access technique for cellular systems in which each individual user is assigned a pair of frequencies while making or receiving a call as shown in **figure 5.2**. One frequency is used for downlink and one pair for uplink. This is called Frequency Division Duplexing (FDD).

That allocated frequency pair is not used in the same cell or adjacent cells during the call so as to reduce the co-channel interference. Even though the user may not be talking, the spectrum cannot be reassigned as long as a call is in place. Different users can use the same frequency in the same cell except that they must transmit at different times. In FDMA the total bandwidth is divided among M simultaneous users such that each user is allocated a channel with a bandwidth of,

$$W_{FDMA} = W/M \text{ Hz} \quad .....(1)$$

From equation the capacity of each channel is

$$C_{FDMA} = (W/M) \log_2(1+S/N) = C/M \quad .....(2)$$

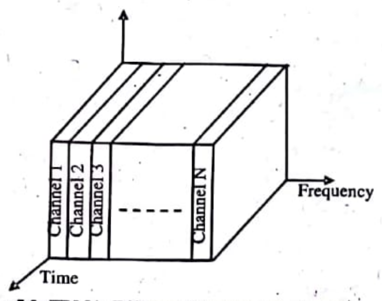


Figure 5.2: FDMA (Different Channels are Assigned Different Frequency Bands)

The schematic of FDMA technique is shown in **figure 5.2.** For a constant S/N, the capacity of the total bandwidth is the same as in equation (1), but is divided among the M users. In practice, each user occupies a bandwidth slightly narrower than W/M so that interference between channels will be acceptable in a system using realisable filters, which have a frequency response that does not go to zero abruptly outside the channel bandwidth. This approach is applicable to both digital and analogue modulation formats.The number of channels that can be simultaneously supported in a FDMA system is given by,

$$N = \frac{B_l - 2B_{guard}}{B_c} \quad \text{..... (3)}$$

Where,

B_l = Total spectrum

B_{gurad} = Guard band allocated at the edge of the allocated spectrum

B_C = Channel bandwidth

Features of FDMA

The features of FDMA are as follows:

1) The FDMA channel carries only one phone circuit at a time.
2) If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.
3) After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously.
4) The bandwidths of FDMA channels are relatively narrow (30 kHz) as each channel supports only one circuit per carrier that is; FDMA is usually implemented in narrowband systems.
5) The symbol time is large as compared to the average delay spread. This implies that the amount of inter-symbol interference is low and, thus, little or no equalization is required in FDMA narrowband systems.
6) The complexity of FDMA mobile systems is lower when compared to TDMA systems.
7) Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronisation and framing bits) as compared to TDMA.
8) FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design, and the need to use costly band pass filters to eliminate spurious radiation at the base station.
9) The FDMA mobile unit uses duplexers since both the transmitter and receiver operate at the same time. This results in an increase in the cost of FDMA subscriber units and base stations.
10) FDMA requires tight RF filtering to minimize adjacent channel interference.

Ques 5) Give the complete description of TDMA technique with its features.

Or

Write the short note on TDMA technique and also draw its diagram.

Ans: TDMA (Time Division Multiple Access)

Time Division Multiple Access (TDMA) shares the available bandwidth in the time domain. Each frequency band is divided into several time slots (channels). A set of such periodically repeating time slots is known as the TDMA frame.

Each node is assigned one or more time slots in each frame, and the node transmits only in those slots. For two-way communication, the uplink and downlink time slots, used for transmitting and receiving data, respectively, can be on the same frequency band (TDMA frame) or on different frequency bands.

The former is known as time division duplex – TDMA (TDD-TDMA), and the latter as frequency division duplex – TDMA (FDD-TDMA). TDMA is essentially a half-duplex mechanism, where only one of the two communicating nodes can transmit at a time.

The small duration of time slots creates the illusion of a two-way simultaneous communication. Perfect synchronization is required between the sender and the receiver.

To prevent synchronisation errors and inter-symbol interference due to signal propagation time differences, guard intervals are introduced between time slots. Since the sizes of slots are already small, the introduction of guard intervals results in a significant overhead for the system FDMA requires the device to have the capability of simultaneously receiving and transmitting signals, which leads to increased cost.

But when TDMA is used, the device can switch between slots and hence use the same transmitter for receiving also. Hence, the equipment cost in TDMA is less.

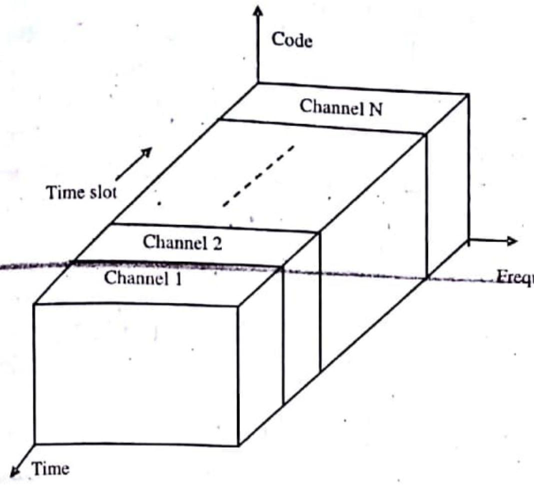


Figure 5.3: Basic Concept of TDMA

Features of TDMA

In the cases where continuous transmission is not required, there TDMA is used instead of FDMA. The features of TDMA include the following:

1) TDMA shares a single carrier frequency with several users where each user makes use of non-overlapping time slots.
2) Data transmission in TDMA is not continuous, but occurs in bursts. Hence hands-off process is simpler.
3) TDMA uses different time slots for transmission and reception thus duplexers are not required.
4) TDMA has an advantage that is possible to allocate different numbers of time slots per frame to different users.
5) Bandwidth can be supplied on demand to different users by concatenating or reassigning time slot based on priority.

Ques 6) Discuss about the CDMA (code division multiple access) technique. And also explain its features.

Ans: CDMA (Code Division Multiple Access)
Unlike other systems such as TDMA and FDMA, Code Division Multiple Access

(CDMA) does not assign a specific frequency to each user. Instead, every channel uses the entire spectrum. Individual conversations are encoded with a pseudorandom digital sequence. As the narrow-band transmission frequency is spread random digital sequenee.

As the narrow-band transmission frequency is spread over the entire wideband spectrum, the technique is also called as spread spectrum. The transmissions are differentiated through a unique code that is independent of the data being transmitted, assigned to each user. The orthogonality of the codes enables simultaneous data transmissions from multiple users using the entire frequency spectrum, thus overcoming the frequency reuse limitations seen in FDMA and TDMA.

CDMA was first used during World War II by the English Allies to foil attempts by the German army to jam the transmissions of the Allies. The Allies decided to transmit signals over several frequencies in a specific pattern, instead of just one frequency, thereby making it very difficult for the Germans to get hold of the complete signal.

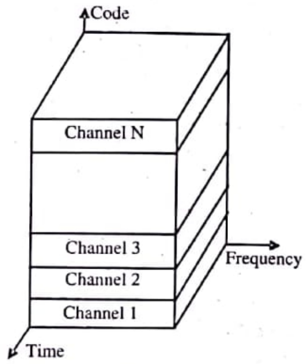


Figure 5.4: Basic Concept of CDMA

Features of CDMA

Code division multiple access technique is an example of multiple access where several transmitters use a single channel to send information simultaneously. Its features are as follows:

1) In CDMA every user uses the full available spectrum instead of getting allotted by separate frequency.
2) CDMA is much recommended for voice and data communications.
3) While multiple codes occupy the same channel in CDMA, the users having same code can communicate with each other.
4) CDMA offers more air-space capacity than TDMA.
5) The hands-off between base stations is very well handled by CDMA

Ques 7) What do you mean by SDMA (space division multiple access) technique?
Or
Discuss about the features of SDMA technique.

Ans: SDMA (Space Division Multiple Access)
The fourth dimension in which multiplexing can be performed is space. Instead of using omnidirectional

transmissions (as in FDMA, TDMA, and CDMA) that cover the entire circular region around the transmitter, Space Division Multiple Access (SDMA) uses directional transmitters/antennas to cover angular regions. Thus different areas/regions can be served using the same frequency channel.

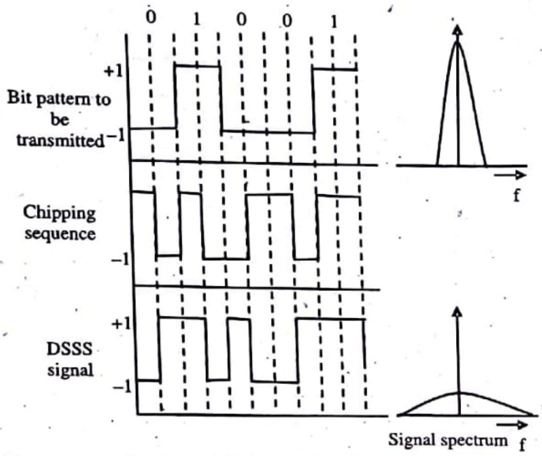


Figure 5.5: Illustration of DSSS

This method is best suited to satellite systems, which often used a narrowly focused beam to prevent the signal from spreading too widely and in the process becoming too weak. A satellite can thus reuse the same frequency to cover many different regions of the Earth's surface.

Feature of SDMA

Space division multiple access or spatial division multiple access is a technique which is MIMO (Multiple-Input Multiple-Output) architecture and used mostly in wireless and satellite communication. It has the following features:

1) All users can communicate at the same time using the same channel.
2) SDMA is completely free from interference.
3) A single satellite can communicate with more satellites receivers of the same frequency.
4) The directional spot-beam antennas are used and hence the base station in SDMA, can track a moving user.
5) Controls the radiated energy for each user in space.

Ques 8) Describe the application of SDMA in satellite communication.

Ans: Applications of SDMA in Satellite Communication

In the satellite business, SDMA is more commonly referred to as frequency reuse because it allows almost identical satellites to operate independently by separating them along the Clarke orbit and limiting their power. For a single satellite, SDMA, or frequency reuse, is accomplished by having distinct radio beams from the satellite pointed at different parts of the earth, these may be categorized by the regions they serve, such as hemispherical beams, regional beams, national beams, or spot beams. If the beams do not overlap, and then the same frequencies can be used in each beam, effectively multiplying the bandwidth available by the number of beams.

A satellite that is designed for mobile communication for users anywhere the satellite is visible must have a single global beam, 17.3° in diameter, that covers the entire region of the earth that the satellite can see. Since there is only one beam, there can be no frequency reuse except for cross-polarized beams. Previous systems have use frequency, time and code domains to achieve multiple access. Space Domain Multiple Access (SDMA) uses spatial separation.

A single satellite may achieve spatial separation by using beams with horizontal and vertical polarization or left-hand and right-hand circular polarization. This could allow two beams to cover the same earth surface area being separated by the polarization.

Additionally, the satellite could have multiple beams using separate antenna or using a single antenna with multiple feeds. For multiple satellites spatial separation can be achieved with orbit longitude or latitude and, for inter satellite links, using different planes.

The use of SDMA allows for frequency re-use and on-board switching which, in turn, enhances channel capacity. Additionally, the use of narrow beams from the satellite allows the earth station to operate with smaller antennae and so produces a higher power density per unit area for a given transmitter power.

SDMA is usually achieved in conjunction with other types of multiple access such as FDMA, TDMA and CDMA. Applications employ multiple-access systems to allow two or more Earth stations to simultaneously share the resources of the same transponder or frequency channel.

Ques 9) Discuss the application of MA (multiple access) technique in satellite communication.

Ans: Application of MA (Multiple Access) Technique in Satellite Communication

Applications employ multiple-access systems to allow two or more earth stations to simultaneously share the resources of the same transponder or frequency channel.

These include the three familiar methods: FDMA, TDMA, and CDMA, they are as follows:

1) **Application of FDMA Technique:** Practical application of satellite communication, FDMA has been developed in the form of the FDM/FDMA, where TV signals and bundles of frequency division multiplexed (FDM) telephone signals are transmitted as FM signals. FDMA is based on analog technology, and its development in the early days of satellite communication owned much to its efficient and reliable nature. Its use continues to this day, although primarily as a legacy from older network.

   Recently, due to advances in digital technology and the growth of networks with relatively small amounts of traffic in each earth station, such as business communication networks. FDMA is currently used by all the mobile satellite communication systems providing audio communication services in each country.

   This is for several reasons, including the ease with which new modulation techniques can be introduced into the system as they are developed, the ability to adapt to the specifications of various communication schemes, and the ability to expand them in terms of frequency allocation and the like to accommodate future growth in the number of mobile stations. The Mobile sat system also uses SCPC DAMA for communication between mobile stations and base stations.

2) **Application of TDMA Technique:** In TDMA, an entire network can be in principle configured by a single repeater, although depending on the network capacity it is possible to use multiple repeaters or just a part of the bandwidth of a single repeater. Satellite channels are divided by defining TDMA frames, which repeat at a constant period, and time slots of suitable length in each frame are assigned to each earth station. These time slots are called data bursts. TDMA can also be applied to continuous real-time signals such as audio and video.

   The bit rate of the baseband digital signals is converted to the transmission bit rate of the TDMA channel and then the signals are transmitted as data bursts after digital modulation such as PSK.SS-TDMA (satellite switched TDMA) is an advanced version of TDMA in which switching is performed on the satellite, and which can be combined with a multi beam antenna onboard the satellite to produce a highly efficient network.

3) **Application of CDMA Technique:** CDMA allows multiple earth stations to establish independent channels by sharing simultaneously the same frequency band of the satellite's onboard repeater, and is based on a different concept of FDMA and TDMA. It also makes it easy to design a system that is less susceptible to interference from existing terrestrial networks in the same frequency band. For such reasons, CDMA is often used in special circumstances.

   However, it has also recently attracted attention for use in mobile satellite communication systems using low earth-orbit satellites. Its communications quality degrades slightly as the number of stations communication simultaneously in the same bandwidth increases, but with fewer simultaneously communicating stations it allows the link margins to be increased. To deal with interference between beams when a multi-beam antenna is employed to re-use the frequencies in each beam.

   CDMA, also called spread spectrum communication, differs from FDMA and TDMA because it allows users to literally transmit on top of each other. This feature has allowed CDMA to gain attention in commercial satellite communication. It was originally developed for use in military satellite communication where its inherent antijam and security features are highly desirable. CDMA was adopted in cellular mobile telephone as an interference-tolerant communication technology that increases capacity above analog systems.

Ques 10) Write the MA techniques application in earth station.

Ans: MA Techniques Applications in Earth Station

Wireless communications is essentially a multi-user communication system. It also reuses frequencies in a certain predetermined pattern. The frequency reuse results in interference that places a performances limit that is related to the transmission rate. To increase the channel capacity, several interference mitigation techniques have been suggested. These include serial, parallel and hybrid interference cancellation techniques.

The other strategy could be to use certain multiple access technique which does not produces high levels of multiuser interference. The wideband spread spectrum multi-access could lead to higher spectrum efficiency.

By increasing the complexity of receivers, the MA efficiency can be increased nearly fourfold.

1) **Frequency Division Multiple Access (FDMA):** Where individual earth stations separate their transmissions from each other by up linking them on different frequencies. This is the simplest MA technique, since stations transmit to the satellite without coordination and with minimal interaction. Single channel per carrier (SCPC) is that form of FDMA where each individual signal (voice conversation, TV program channel, or data stream) gets its own carrier within the satellite repeater.

The alternative is to multiplex several channels into a carrier's baseband, which is called multiple channels per carrier (MCPC). In either case, loading of the transponder on a bent-pipe satellite repeater requires management of multiple carriers and the resulting RF intermodulation distortion (IMD). Due to the required amplifier output back off of 3 to 5 dB, the capacity of the transponder is reduced by the least 50%.

2) **Time Division Multiple Access (TDMA):** where separation is achieved by having earth stations transmit their data as bursts at different times, according to a preset time frame. Thus, the transmission must be synchronized in time to prevent collisions among the transmissions when received at the satellite. An alternative from of TDMA, called ALOHA, allows earth station transmissions to be uncoordinated and so introduces the possibility of collisions and a corresponding requirement for automatic retransmission.

In wideband TDMA, the burst transmissions at between 60 and 250 Mbps use the full bandwidth and power of the transponder, resulting in nearly 95% efficiency (allowing for the necessary synchronization overhead and guard time between bursts). More commonly, TDMA networks use lower data rates (between 256 Kbps and 20 Mbps) to share the capacity of a transponder in an FDMA mode and reduce the uplink power required from the earth station. The inherent digital feature of TDMA has made it the most popular multiple access technique for VSAT networks.

3) **Code Division Multiple Access (CDMA):** where earth station transmissions encoded using the direct sequence spread-spectrum waveform. This is another popular technique obtained by mixing the user data with a very high-speed stream of bits from a pseudo-random noise (PN) generator. Several carriers may be transmitted on the same frequency but are separated by virtue of the different spreading codes. The information on any particular CDMA channel is recovered at the receiver by multiplying the incoming PN-modulated data by the original PN stream. Prior to data recovery, the CDMA receiver must synchronize to the spreading sequence and lock onto its precise timing (a technique called autocorrelation).

CDMA hybrid, multicarrier CDMA, and orthogonal frequency division multiplexing (OFDMA), a form of CDMA multiple accesses that uses orthogonal carrier set to counter multitude of channel impairments. Another multiple access technique, though not legal at this time, is called impulse radio where information is modulated by the position of nanosecond pulses that results in infinitesimally small power spectral of the transmitted signal.

CDMA allows multiple earth stations to establish independent channels by sharing simultaneously the same frequency bond of the satellite's onboard repeater, and is based on a different concept to FDMA and TDMA. To separate one channel from the other channels, each earth station is allocated a specific spread spectrum code, for which a PN (Pseudo noise) code is normally used.

Using the PN code allocated to the station for which a signal is destined, a transmitting station spreads the outgoing carrier wave across a frequency band hundreds of times larger than that of the baseband signal either by direct sequence or by a method called frequency hopping. The receiving station uses its own spread code to detect the signals addressed to it.

Ques 11) Discuss the application of MA technique in wired communication.

Or

Explain application of MA technique in link between wireless sensors and wired network.

Ans: Application of MA Technique in Wired Communication

SDMA is considered in the form of the receiving Beam Forming (BF) in the wired end of the connection. As Compared to the other Multiple

Access (MA) techniques, SDMA does not require accurate scheduling as Time Division Multiple Access (TDMA) or as wide frequency band as Frequency Division Multiple Access (FDMA). It is applicable simultaneously with the other MA techniques for better communicational capacity.

Link between Wireless Sensors and Wired Network

SDMA is considered for the cellular systems in wireless sensor networks, which communicate ad-hoc, are obviously able to communicate with feedback to wired stations. In those cases, communication capacity between nodes and bases can be restricted, and hence the use of SDMA is worthwhile. If the wireless nodes are distant to the base of SINR is small, it is advisable to collect the information to a single node from a larger cluster of nodes and then send to the wired base station.

The upsides of connections without feedback are lighter hardware and software, leading to smaller battery usage. The lack of feedback enables wireless sensor node constructions without receivers. The advantage of the receiver less node is the smaller requirement for hardware and complex signal processing, and the small power consumption. After all, the power consumption of the analog/digital-conversion of receivers is high.

The downside of the feedback less connections is inability to use several multiple access and radio resource management techniques, e.g. TDMA, FDMA, Power Control (PC) and collision avoidance algorithms, and therefore the communication capacity is reduced. Also, SDMA with BF requires a control signal channel for efficient interference rejection in bursty traffic.

Module 6

Cellular Telephone and Optical Link

CELLULAR TELEPHONE

Ques 1) Explain the basic concepts of cellular telephone system. Also explain its basic structure.

Ans: Cellular Telephone System

A cellular telephone system provides a wireless connection to the PSTN (public switched telephone network) for any user location within the radio range of the system. Cellular systems accommodate a large number of users over a large geographic area, within a limited frequency spectrum. Cellular radio systems provide high quality service that is often comparable to that of the landline telephone systems.

High capacity is achieved by limiting the coverage of each base station transmitter to a small geographic area called a **cell** so that the same radio channels may be reused by another base station located some distance away.

A sophisticated switching technique called a **hand-off** enables a call to proceed uninterrupted when the user moves from one cell to another.

Basic Structure of Cellular System

Figure 6.1 shows a basic cellular system which consists of mobile stations, base stations and a mobile switching centre (MSC). The mobile switching centre is sometimes called a mobile telephone switching office (MTSO), since it is responsible for connecting all mobiles to the PSTN in a cellular system.

Each mobile communicates via radio with one of the base stations and may be handed-off to any number of base stations throughout the duration of a call. The mobile station contains a trans-receiver, an antenna, and control circuitry, and may be mounted in a vehicle or used as a portable hand-held unit.

The base stations consist of several transmitters and receivers which simultaneously handle full duplex communications and generally have towers which support several transmitting and receiving antennas. The base station serves as a bridge between all mobile users in the cell and connects the simultaneous mobile calls via telephone lines or microwave links to the MSC.

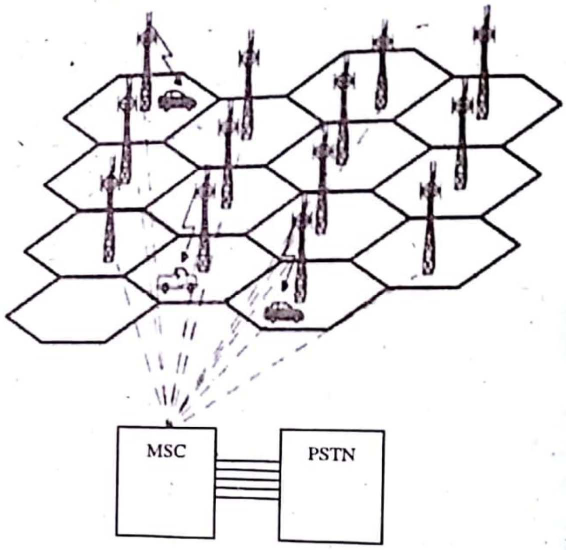


Figure 6.1

The MSC coordinates the activities of all of the base stations and connects the entire cellular system to the PSTN. A typical MSC handles 100,000 cellular subscribers and 5,000 simultaneous conversations at a time, and accommodates all billing and system maintenance functions, as well. In large cities, several MSCs are used by a single carrier.

Ques 2) Describe the working principle of cellular telephone system. Also write its features.

Ans: Working Principle of Cellular Telephone System

When a cellular phone is turned on, but is not yet engaged in a call, it first scans the group of forward control channels to determine the one with the strongest signal, and then monitors that control channel until the signal drops below a usable level. At this point, it again scans the control channels in search of the strongest base station signal.

For each cellular system the control channels are defined and standardized over the entire geographic area covered and typically make up about 5% of the total number of channels available in the system (the other 95% are dedicated to voice and data traffic for the end-users).

Since the control channels are standardized and are identical throughout different markets within the country or continent, every phone scans the same channels while idle. When a telephone call is placed to a mobile user, the MSC dispatches the request to all base stations in the cellular system. The mobile identification number (MIN), which is the subscriber's telephone number, is then broadcast as a paging message over all of the forward control channels throughout the cellular system.

The mobile receives the paging message sent by the base station which it monitors, and responds by identifying itself over the reverse control channel. The base station relays the acknowledgment sent by the mobile and informs the MSC of the handshake. Then, the MSC instructs the base station to move the call to an unused voice channel within the cell (typically, between ten to sixty voice channels and just one control channel are used in each cell's base station).

At this point, the base station signals the mobile to change frequencies to an unused forward and reverse voice channel pair, at which point another data message (called an alert) is transmitted over the forward voice channel to instruct the mobile telephone to ring, thereby instructing the mobile user to answer the phone.

Features of Cellular Systems

Wireless Cellular systems solve the problem of spectral congestion and increases user capacity. The features of cellular systems are as follows:

1) Offer very high capacity in a limited spectrum.
2) Reuse of radio channel in different cells.
3) Enable a fixed number of channels to serve an arbitrarily large number of users by reusing the channel throughout the coverage region.
4) Communication is always between mobile and base station (not directly between mobiles).
5) Each cellular base station is allocated a group of radio channels within a small geographic area called a cell.
6) Neighbouring cells are assigned different channel groups.
7) By limiting the coverage area to within the boundary of the cell, the channel groups may be reused to cover different cells.
8) Keep interference levels within tolerable limits.
9) Frequency reuse or frequency planning.
10) Organization of Wireless Cellular Network.
11) Cellular network is organized into multiple low power transmitters each 100w or less.

Ques 3) What is mean by frequency reuse in cellular telephone system?

Ans: Frequency Reuse in Cellular Telephone System

A given geographical area coverage can be sub-divided into hexagonal cells where each cell has its own set of channels (or band of frequencies) and a number of channels were required for cellular communications.

But this is not a practical approach to allow different sets of channels to each cell for entire geographical area due to limited range of frequency spectrum for wireless (cellular) communication. Therefore to overcome this drawback, a technique is adopted which uses same set of channels for different cells, the cells are separated from one another by a distance large enough to keep interference levels within tolerable limits.

In the cellular concept, frequencies allocated to the service are re-used in a regular pattern of areas, called 'cells', each covered by one base station. In mobile-telephone nets these cells are usually hexagonal. In radio broadcasting, a similar concept has been developed based on rhombic cells.

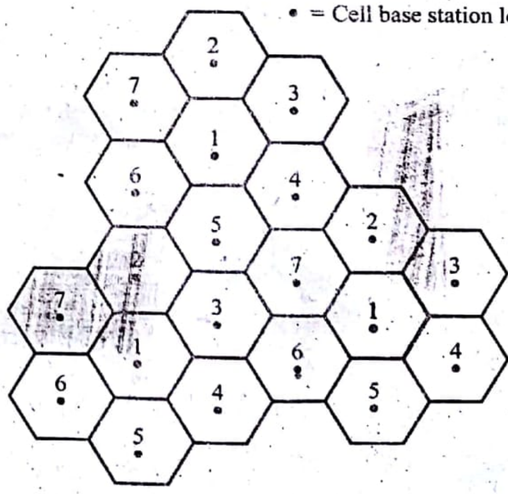


Figure 6.2: Frequency Reuse Format (N = 7)

Figure 6.2 shows frequency reuse concept (N = 7 reuse format). Frequency reuse plan can simply be defined as how technocrats and engineers sub-divide and assign the FCC allocated radio spectrum, throughout the coverage area.

Ques 4) Discuss about the frequency reuse distance in cellular telephone system.

Ans: Frequency Reuse Distance

To reuse the same set of radio channels in another cell, it must be separated by a distance called **frequency reuse distance**, which is generally represented by D.

Reusing the same frequency channel in different cells is restricted by co-channel interference between cells. So, it is necessary to find the minimum frequency reuse distance D in order to minimize the co-channel interference.

Figure 6.3 shows the separation of cells by frequency reuse distance in a cluster of 7 cells. In order to derive a formula to compute D, necessary properties of regular hexagon cell geometry.

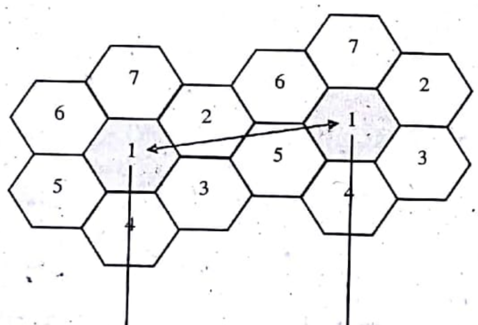


Figure 6.3: Frequency Reuse Distance

The frequency reuse distance (D), which allows the same radio channel to be reused in co-channel cells, depends on many factors:
1) The number of co-channel cells in the neighbourhood of the central cell
2) The type of geographical terrain
3) The antenna height
4) The transmitted signal strength by each cell-site

Suppose the size of all the cells in a cellular is approximately same, and it is usually calculated by the coverage area of the proper signal strength in every cell. The co-channel interference does not depend on transmitted power of each, if the cell size is fixed, i.e., the threshold level of received signal at the mobile unit is tuned to the size of the cell.

The co-channel interference depends upon the frequency reuse ratio, q, and is defined as,

$$q = D/R$$

Where, D is the distance between the two neighbouring co-channel cells, and R is the radius of the cells. The parameter q is also referred to as the frequency reuse ratio or co channel reuse ratio.

The following steps are used to find the relationship between frequency reuse ratio q and cluster size K. **Figure 6.4** shows an array of regular hexagonal cells. Where, R is the cell radius. Due to the hexagonal geometry each hexagon has exactly six equidistant neighbours.

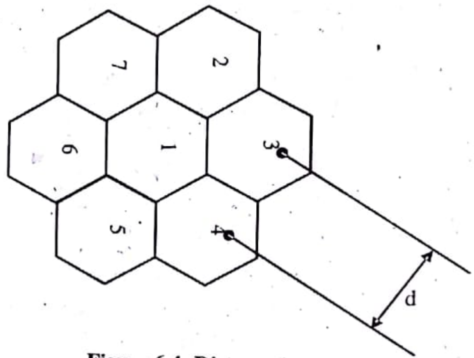


Figure 6.4: Distance between Two Adjacent Cells (d)

Let, d is the distance between two cell centers of neighbouring cells. Therefore,

$$d = \sqrt{3}R$$

The relationship between D, d, and shift parameters is,

$$D^2 = 3 \times R^2 \times (i^2 + j^2 + i \times j)$$

As, $K = i^2 + j^2 + i \times j$

$$D^2 = 3 \times R^2 \times K$$

$$\frac{D^2}{R^2} = 3 \times K$$

$$\frac{D}{R} = \sqrt{3K}$$

As, $q = D/R$

$$q = \sqrt{3K}$$

As the D/R measurement is a ratio, if the cell radius is decreased, then the distance between co-channels cells must also be decreased by the same amount, for keeping co-channel interference reduction factor same.

On the other hand, if a cell has a large radius, then the distance between frequency reusing cells must be increased proportionally in order to have the same D/R ratio.

Ques 5) Determine the frequency reuse ratio for a cell radius of 0.8km separated from the nearest co-channel cell by a distance of 6.4km.

Ans: Given,
Radius of a cell, R = 0.8 km
Distance between nearest co-channel cells, D = 6.4 km

To determine the frequency reuse ratio, q

We know that $q = D / R$
Or, $q = 6.4 / 0.8 = 8$

Hence, the frequency reuse ratio for given parameters, q = 8

Ques 6) Find the distance from the nearest co-channel cell for a cell having a radius of 0.64km and a co-channel reuse factor of 12.

Ans: Given,
Radius of a cell, R = 0.64km
Co-channel reuse factor, q = 12
To determine the distance from the nearest co-channel cell, D

We know that,
$q = D / R$
Or, $D = q \times R$

Therefore,
$D = 12 \times 0.64\text{km} = 7.68\text{km}$

Hence, the distance from the nearest co-channel cell, D = 7.68km.

Ques 7) What do you mean by interference in cellular system.And also discuss its sources and types.

Or

Explain co-channel and adjacent interference in cellular system.

Ans: Interference in Cellular System
Interference is the sum of all signal contribution that is neither noise not the wanted signal. In cellular mobile transmissions interference is a main limiting factor that degrades the system performance.

Source of Interference
There are many factors that act as source of interference such as:
1) A call progress in the neighbouring cell.
2) Some other mobile in same cell.
3) Foreign base stations that operate in same frequency band.
4) Leakage of energy due to any non-cellular systems.

The effect of interference due to any of the reasons given above may lead to occurrence of cross talk. Due to cross talk the user will hear interference in the background in his own set because of an undesired transmission.

Types of Interference in Cellular System
There are two types of interference in cellular system such as:
1) **Co-Channel Interference:** When frequency reuse is implemented, several cells within a given coverage area use the same set of frequencies. Two cells using the same set of frequencies are called co-channel cells, and the interference between them is called **co-channel interference**.

   Co-channel interference cannot be reduced by simply increasing transmit powers because increasing the transmit power in one cell increases the likelihood of that cell's transmissions interfering with another cell's transmission. To reduce co-channel interference, a certain minimum distance must separate co-channels.

   Figure 6.5 shows co-channel interference. The base station in cell A of cluster 1 is transmitting on frequency f_1, and at the same time, the base station in cell A of cluster 2 is transmitting on the same frequency. Although the two cells are in different clusters, they both use the group of frequencies.

   The mobile unit in cluster 2 is receiving the same frequency from two different base stations. Although the mobile unit is under the control of the base station in cluster 2, the signal from cluster 1 is received at a lower power level as co-channel interference.

   Interference between cells is proportional not to the distance between the two cells but rather to the ratio of the distance to the cell's radius. Since a cell's radius is proportional to transmit power, more radio channels can be added to a system by either:

i) Decreasing the transmit power per cell,
ii) Making cells smaller, or
iii) Filling vacated coverage areas with new cells.

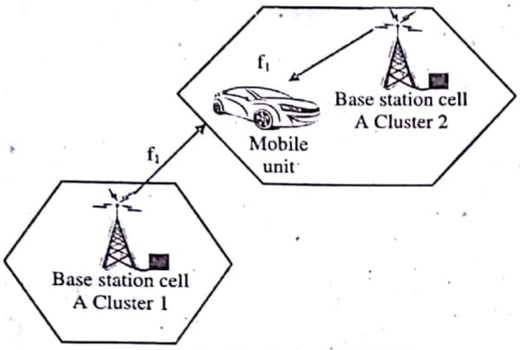


Figure 6.5: Co-channel Interference

In a cellular system where all cells are approximately the same size, co-channel interference is dependent on the radius (R) of the cells and the distance to the center of the nearest co-channel cell (D). Increasing the D/R ratio (sometimes called co-channel reuse ratio) increases the spatial separation between co-channel cells relative to the coverage distance. Therefore, increasing the co-channel reuse ratio (Q) can reduce co-channel interference. For a hexagonal geometry,

$$Q = \frac{D}{R}$$

Where,

Q = Co-channel reuse ratio (unitless)
D = Distance to center of the nearest co-channel cell (kilometers)
R = Cell radius (kilometers)

The smaller the value of Q, the larger the channel capacity since the cluster size is also smaller. However, a large value of Q improves the co-channel interference and, thus, the overall transmission quality. Obviously, in actual cellular system design, a trade-off must be made between the two conflicting objectives.

2) **Adjacent-Channel Interference:** Adjacent-channel interference occurs when transmissions from adjacent channel (channels next to another in the frequency domain) interfere with each other. Adjacent-channel interference results from imperfect filters in receivers that allow nearby frequencies to enter the receiver.

Adjacent-channel interference is most prevalent when an adjacent channel is transmitting very close to a mobile unit's receiver at the same time the mobile unit is trying to receive transmissions from the base station on an adjacent frequency. This is called the near-far effect and is most prevalent when a mobile unit is receiving a weak signal from the base station.

Adjacent-channel interference is depicted in **figure 6.6**. Mobile unit 1 is receiving frequency f_1 from base station A. At the same time, base station A is transmitting frequency f_2 to mobile unit 2. Because mobile unit 2 is much farther from the base station than mobile unit 1, f_2 is transmitted at a much power level than f_1. Mobile unit 1 is located very close to the base station, and f_2 is located next to f_1 in the frequency spectrum (i.e., the adjacent channel).

Therefore, mobile unit 1 is receiving f_2 at a much higher power level than f_1. Because of the high power level, the filters in mobile unit 1 cannot block all the energy from f_2, and signal intended for mobile unit 2 interferes with mobile unit 1's reception of f_1. f_1 does not interfere with mobile unit 2's reception because f_1 is received at a much lower power level than f_2.

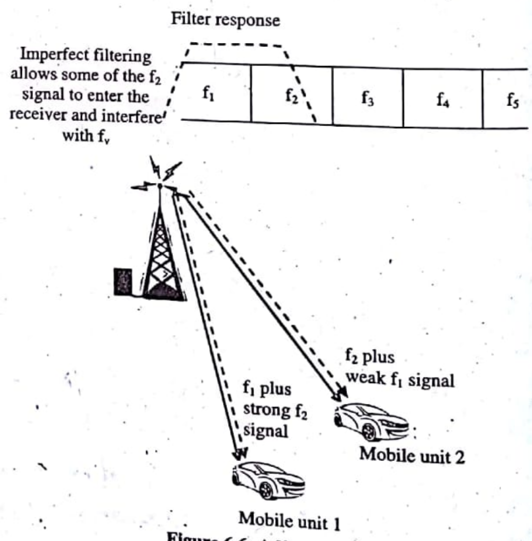


Figure 6.6: Adjacent-Channel Interference

Using precise filtering and making careful channel assignments can minimise adjacent channel interference in receivers. Maintaining a reasonable frequency separation between channels in a given cell can also reduce adjacent-channel interference. However, if the reuse factor is small, the separation between adjacent channels may not be sufficient to maintain an adequate adjacent-channel interference level.

Ques 8) Define cell splitting. And also explain the techniques for it.

Ans: Cell Splitting

Cell splitting is a method in which congested (heavy traffic) cell is subdivided into smaller cells, and each smaller cell is having its own base station with reduction in antenna height and transmitter power.

The original congested bigger cell is called macro cell and the smaller cells are called microcells. Capacity of cellular network can be increased by creating micro-cells within the original cells which are having smaller radius than macro-cells, therefore, the capacity of a system increases because more channels per unit area are now available in a network.

In principle, a cellular system can provide services for an unlimited number of users. However once a system is installed, it can only provide to a certain fixed number of users.

As soon as the number of users increases and approaches the maximum that can be served, some technique must be developed to accommodate the increasing number of users.

There are various techniques to enhance the capacity of a cellular system. One technique is cell splitting, a mechanism by which cells are split into smaller cells, each having the same number of channels as the original large cells, as shown in **Figure 6.7**.

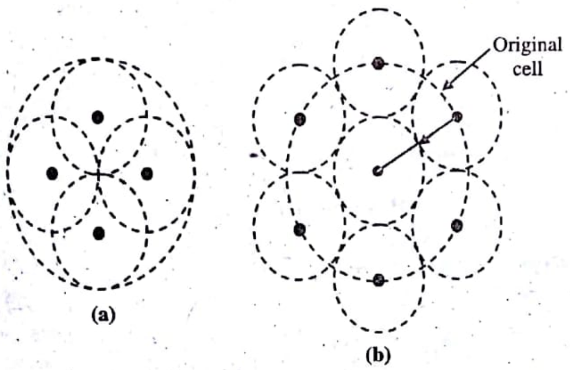


Figure 6.7: Concept of Cell Splitting

Cell Splitting Techniques

The two techniques of cell splitting are described below:

1) **Permanent Splitting:** The installation of every new split cell has to be planned a head of time; the number of channel, the transmitted power, the assigned frequencies, the choosing of cell site selection and the traffic load consideration should all be considered. When ready the actual service cut-over should be set at lowest traffic point usually at midnight or weekend. Hopefully, only a few cells will be dropped because of this cut-over assuming that the downtime of the system is within 2 hours.

2) **Dynamic Splitting:** This scheme is based on utilizing the allocated spectral efficiency in real time. The algorithm for dynamically splitting cell sites is a tedious job since we cannot afford to have one single cell unused during cell splitting at heavy traffic hours.

Ques 9) Discuss about the factor on which size of the splitting cell depends. And also write the advantages and disadvantages of cell splitting.

Ans: Factor on which Size of the Cell Splitting Depends

The size of splitting cells is dependent on a number of factors. These include radio aspect and capacity of switching processor:

1) **Radio Aspect:** The size of a small cell is dependent upon how well the coverage pattern can be controlled and how accurately vehicle locations will be known.

2) **Capacity of Switching Processor:** The smaller the cells, the more handoff will occur and the more cell splitting process is needed. This factor, the capacity of switching processor is a larger factor than handling of coverage area of small cells.

Advantages of Cell Splitting

The advantages of cell splitting are as follows:

1) It is normally possible to increase the system capacity through creating new cells, sectoring, cell splitting and or by simply scaling the network up progressively.
2) Lower transmission power.
3) Increasing the coverage area
4) Robustness

Disadvantages of Cell Splitting

The disadvantages of cell splitting are as follows:

1) The need to engineer necessary handover within (and between) the networks.
2) Cell splitting causes increased handoff
3) Inter cell interference occurs due to co-channel interference and adjacent channel interference.
4) Implementation of cell-splitting technique is costly.

Ques 10) Explain the cell sectroing function in cellular system.

Ans: Cell Sectoring

The co-channel interference in a cellular system may be decreased by replacing a single omnidirectional antenna at base station by several directional antennas for each sector within a cell. So "The technique of decreasing co-channel interference and thus increasing system capacity by using directional antennas is called **sectoring.**"

A cell is normally divided into three 120° sectors 'or' 60° sectors as shown in **figure 6.8 (a)** and **6.8 (b)** when sectoring is employed, the channels used in particular cell are broken down into sectored groups and are used only within particular sector as shown in **figure 6.8**.

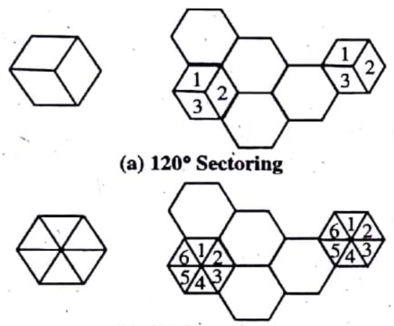


Figure 6.8: Concept of Cell Sectoring

But the only disadvantage of the cell sectoring is decrement in capacity of cellular system. Here radio channels have been fixed for a sector that's why; we have very less channels available in a sector for sharing by different users.

The numbers of the handovers are also increased; that also may be problem for service provider but now-a-days many base stations are available for sectored cell structure so handoff problem can be handled by base station without involving the main switching centre (MSC).

If cell splitting (D / R) ratio was constant and then improvement of capacity was achieved. In cell sectoring, the number of cells in a particular cluster is decreased. So the ratio (D / R) is decreased.

So following results have been achieved:

1) As cluster size decreases, capacity of cellular radio system is increased as repetition of radio channels also has been increased.
2) As the value of D / R is decreasing by relation, $D/R = \sqrt{3N}$

So the separation between co-channels has been decreased in sectoring technique. So in case of cell sectoring, the separations between co-channels have been decreased. For high capacity, the co-channel interference should be controlled.

Ques 11) Explain with diagram cell system layout in cellular system.

Ans: Cell System Layout in Cellular System

Cells are of arbitrary shape which is quite close to a circle, is the ideal radiation pattern of an omnidirectional antenna. Because of the randomness inherent nature of the mobile radio propagation and irregular geographical terrain, it is easier to obtain insight and plan the cellular network by visualizing all the cells as having the same shape.

By approximating a uniform cell size for all cells, it is easier to analyse and design a cellular topology mathematically. It is highly desirable to construct the cellular system such that the cells do not overlap, and are tightly packed without any dead signal spots. The cellular topology formed by using ideal circular shape results into overlaps or gaps between them which is not desirable in cellular communications which has to be essentially continuous.

This form of layout requires the use of regular topologies (such as hexagonal topology) instead of a circular shape, as shown in **figure 6.9**.

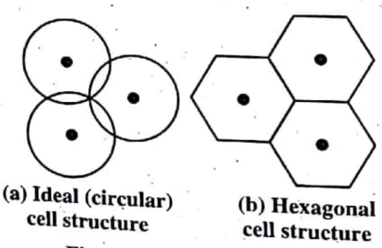


Figure 6.9: Ideal and Regular Hexagonal cell Structure

In **Figure 6.9**, the middle darks circles represent cell-sites. This is where the base-station radio equipment and their antennas mounted on tall towers are located. A cell-site gives radio signal coverage to a cell. In other words, the cell-site is a location or a point at the centre of the cell, whereas the cell is a wide geographical service area.

The design and performance of cellular systems using regular geometrical topologies may not correspond to real mobile environments, but these topologies do provide valuable information and guidelines for structuring practical cellular configuration layouts.

Cells of the same shape form a tessellation so that there are no ambiguous areas that belong to multiple cells or to no cell. The cell shape can be of only three types of regular polygons: equilateral triangle, square, or regular hexagon as shown in **Figure 6.10**.

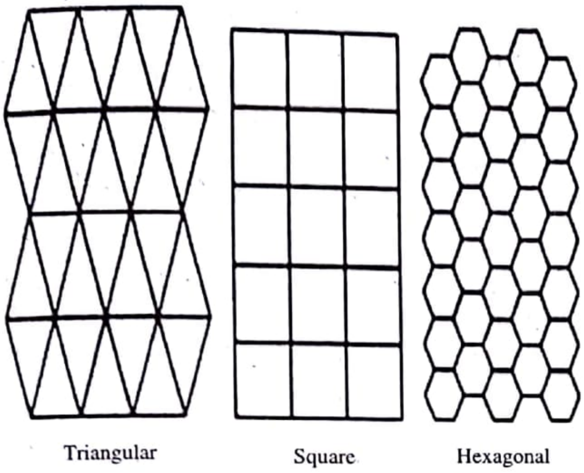


Figure 6.10: Possible Geometrical Cellular Structures

A cellular structure based on a regular hexagonal topology, though fictitious, offers best possible non-overlapped cell radio coverage. A regular hexagonal-shaped cell is the closest approximation to a circle out of these three geometrical shapes and has been used for cellular system design.

For a given radius (largest possible distance between the polygon centre and its edge), the hexagon has the largest area. It allows a larger region to be divided into non overlapping hexagonal sub regions of equal size, with each one representing a cell area.

Octagons and decagons geometrical patterns do represent shapes closer to a circular area as compared to hexagons, but they are not used to model a cell as it is not possible to divide a larger area into non-overlapping subareas of the same size.

Ques 12) Give the brief description of cellular cluster with the help of cluster patterns.

Ans: Cellular Cluster

A group of cells that uses a different set of frequencies in each cell is called a **cellular cluster.** Thus, a cluster is a group of cells with no reuse of channels within it. It is worth mentioning here that only a selected number of cells can form a cluster.

It follows certain rules before any cell can be repeated at a different location. Some common reuse cluster patterns are given in **Figure 6.11**.

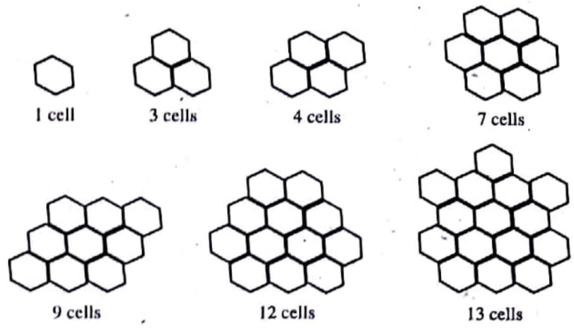


Figure 6.11: Common Reuse Patterns of Hexagonal Cell Clusters

Two or more different cells can use the same set of frequencies or channels if these cells are separated in space such that the interference between cells at any given frequency is at an acceptable level.

That means, the cluster can be repeated any number of times in a systematic manner in order to cover the designated large geographical service area. Let there be K number of cells having a different set of frequencies in a cluster. Then K is termed as the cluster size in terms of the number of cells within it.

Ques 13) Discuss about the cell processing or call processing of cellular system.

Or

Define cell processing and also enlist the components of a cellular mobile network.

Ans: Cell Processing

A mobile phone is a portable telephone used to receive or make calls through a cell site, or transmitting tower. Electromagnetic waves are used to transfer signals to and from the cell phone. Modem mobile phone networks use cells because radio frequencies are limited, shared resource. Cell-sites and handsets change frequency under computer control and use low-power transmitters so that a limited number of radio frequencies can be simultaneously used by many callers with less interference. **Figure 6.12** shows the cellular mobile network.

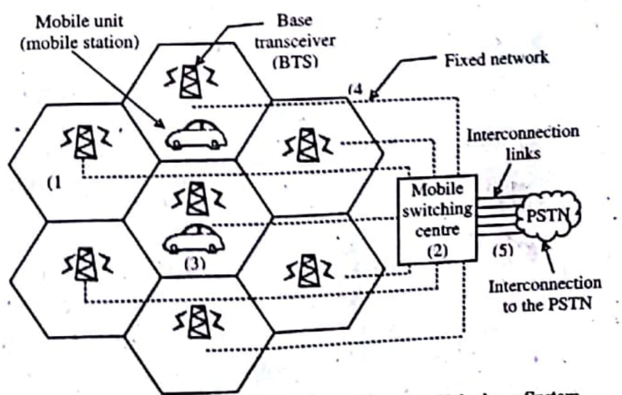


Figure 6.12: Five Main Components of a Cellular Telephone System

A cellular network is used by the mobile phone operator to achieve both coverage and capacity for their subscribers. Large geographical areas are split into smaller cells to avoid line-of-sight (LOS) signal loss and to support a large number of active phones in that area.

All cell sites are connected to telephone exchanges (or switches), which in turn connect to the public telephone network.

In cities, each cell site may have a range of up to approximately 1/2 mile, while in rural areas the range could be as much as 5 miles. It is possible that in clear open areas, a user may receive signals from a cell site 25 miles away.

Components of a Cellular Mobile Network

A cellular network is formed by connecting the following five components as shown in **Figure 6.12.**

1) Mobile station (MS)
2) Base station (BS)
3) Base station controller (BSC)
4) Mobile switching centre (MSC)
5) Public switched telephone network (PSTN)

Ques 14) Explain the function of each components of cellular mobile network.

Or

Write the short note on following:

1) **Mobile station**
2) **Base station**
3) **Base station controller**
4) **Mobile switching centre**
5) **Public Switched Telephone Network**

Ans: Function of Component of Cellular Mobile Network

The function of each network component is described in the following:

1) **Mobile Station (MS):** MSs are usually a mobile phone. Each mobile phone contains a transceiver (transmitter and receiver), an antenna, and control circuitry. Antenna converts the transmitted RF signal into an EM wave and the received EM waves into an RF signal. The same antenna is used for both transmission and reception, so there is a duplexer switch to multiplex the same antenna.

2) **Base Station (BS):** One of the important components in the cellular network is the BS. BS provides direct communication with mobile phones and it defines the cell. When cells are grouped together, a cluster is formed. Within a cluster, no channels are raised.

Two frequencies are required to establish communication between MS and BS: on from mobile phone (MS) to BS (uplink channel) and inverse (downlink channel). A group of BSs are in turn connected to a BSC. The BS is a transceiver station or system and consists of a number of different elements.

i) **BS or Celt Site Antenna:** Either omni-directional or directional antennas are used as BS antennas in the wireless industry. The typical directional antenna is shown in **figure 6.13**. The cell site mast generally has three "faces" each with several frequency agile, directional antennas. Each face covers approximately 120° of the cell and each face uses a different subset of the cell's assigned frequencies. Usually, the antenna lower is at the centre of the cell.

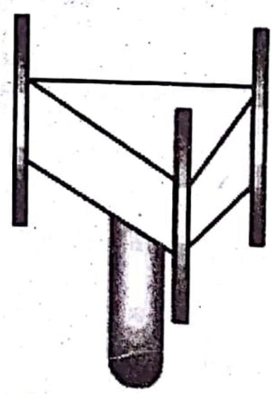


Figure 6.13 Face Directional Antenna as base Station Antenna

ii) **Omni-Directional Antenna:** The omni-directional antenna shown in **figure 6.14 (a)** at BSs exists only in rural areas for the most part. This is because of the lower subscriber densities in rural areas and the lack of requirement for the increased capacity that is afforded by using directional antennas and sectorised BSs. Omni-directional BSs are noted for their use of omni-directional antennas which are slender, long, and tubular. There are always two receive antennas at every BS, which are known as receive zero (Rx0) and receive one (Rx1).

The purpose of having two receive antennas at every BS is to provide for what is known as space diversity. It is also known as receive diversity, compensates for Rayleigh fading in the uplink to the BS. Space diversity is a tool used to optimize the signal received by a BS (transceiver); it counteracts the negative effects of Rayleigh fading. It ensures that the best possible receive signal is used to process all wireless calls. A typical antenna arrangement is shown in **figure 6.15**.

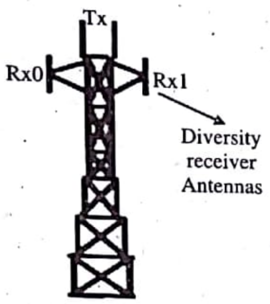


Figure 6.14: Horizontal View of Tower Mounting of Omni directional BS Antenna

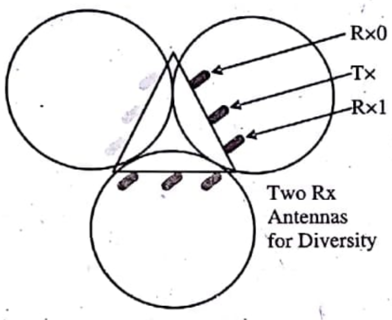


Figure 6.15: Typical Antenna Arrangement

3) **Base Station Controller (BSC):** A number of BSs are connected to a BSC as shown in **Figure 6.16**. An important function of BSC is that it manages the "handoff" from one BS to another as a subscriber moves from cell-to-cell. The BSC contains logic to control each of the BSs. Also, a group of BSCs are in turn connected to a MSC via microwave link or telephone lines.

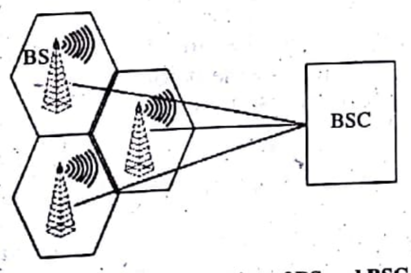


Figure 6.16: Interconnection of BS and BSC

4) **Mobile Switching Centre (MSC):** The MSC is the control centre for the cellular system. The MSC is also known as mobile telephone switching office (MTSO). It coordinates actions of the BSs providing overall control and acts as a switch and connects into the PSTN. Various functions performed by MSC areas follows:
   i) It communicates with the BSs, routing calls and controlling them as required.
   ii) It contains databases detailing the last known locations of the mobiles.
   iii) It also contains facilities for authentication centre allowing mobiles onto the network.
   iv) It contains facilities to generate billing information for individual accounts.

For this purpose, the MSC makes use of the three major components of the network subsystem (NSS) that is HLR, VLR, and AUC:

i) **Home Location Register (HLR):** The HLR contains the information related to each mobile subscriber, such as the type of subscription, services that the user can use the subscriber's current location, and the mobile equipment status. The database in the HLR remains intact and unchanged until the termination of the subscription.

ii) **Visitor Location Register (VLR):** The VLR comes into action once the subscriber enters the coverage region. Unlike the HLR, the VLR is dynamic in nature and interacts with the HLR when recording the data of a particular mobile subscriber. When the subscriber moves to another region, the database of the subscriber is also shifted to the VLR of the new region.

iii) **Authentication Centre (AUC):** The AUC (or AC) is responsible for policing actions in the network. This has all the data required to protect the network against false subscribers and to protect the calls of regular subscribers. There are two major keys in the GSM standards; the encryption of communications between mobile users and the authentication of the users. The encryption keys are held both in the mobile equipment and the AUC and the information is protected against unauthorized access.

5) **Public Switched Telephone Network (PSTN):** PSTN is a cellular network that can be viewed as an interface between mobile units and a telecommunication infrastructure as shown in **figure 6.17**, therefore, the PSTN network is nothing but the land-based section of the network. It is necessary that the BSs are to be connected to a switching network and that network is to be connected to other networks such as the PSTN, so that calls can be made to and from mobile subscribers.

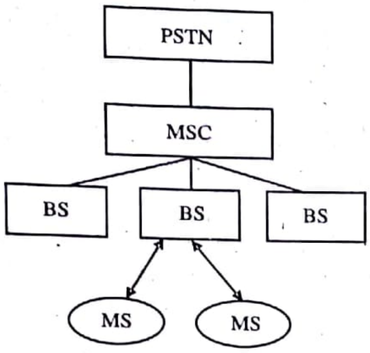


Figure 6.17: PSTN to Mobile Station Connectivity

OPTICAL FIBER

Ques 15) Discuss about the optical fibers.And explain the types of optical fibers.

Ans: Optical Fiber

An optical fiber is a very thin strand of plastic or glass that is used to transmit messages via light. These strands are bundled together in a protective sheath or cover and the whole assembly is often referred to as fiber optic cable or just fibe. It is a type of cabling technology that uses light to carry voice and data communications (telecommunications) over distances both great and small.

Types of Optical Fibers

Optical fiber can be manufactured to optimize performance for different communications applications. The major types are categorized by their diameter and refractive index profile, where the index profile is the refractive index as a function of fiber radius. The types of fiber are discussed below:

1) **Multimode Fiber:** Multimode (MM) fiber allows for the transmission of more than just a single mode. MM fiber is relatively inexpensive and easy to couple with LED sources and detectors. The addition of a large bandwidth (< 200 MHz-Km) makes MM fibers a popular choice for many shorter data link applications. Typical NA is about .20 and core/clad sizes (μm) are 50/125 or 62.5/125.

2) **Single-Mode Fiber:** Single-mode (SM) fiber allows for only a single mode to propagate. Since the core diameter must be less than about 9 μm (V < 2.4), this type of fiber is more difficult to handle and couple with devices, Single-mode fiber is more expensive and requires a laser source, but the extremely large bandwidth over long distances makes SM fiber ideal for high-speed.

3) **Graded-Index Fiber:** Graded-index fiber is manufactured by varying the refractive index between the central core and the cladding. Other variations such as triangular and parabolic refractive index profiles are also possible. While the process is more expensive than that for the previous fiber types, the improvement in dispersion and bandwidth over that of step-index multimode fiber is significant. The graded-index process works best for multimode fiber. The refractive index profile causes rays to refract continuously, tracing out a helical pattern as they propagate down the fiber.

Ques 16) Give the detail description of optical sources used in optical communication.

Ans: Optical Sources

Light emitting diode (LED) or a laser diode can be used as a source for lunching the signal into the fiber cable as their output can be modulated by varying the bias current. A laser source is a monochromatic (coherent) source and also has highly directional radiation characteristics whereas an LED source is not. Faster response time is obtained in laser based system than in LED based system.

But laser source is more complex to obtain and more expensive. Gallium Arsenide laser sources at 850 nm where used in the first generation systems. Later, Indium Gallium Arsenide laser sources were developed to work at 1300 nm operation enabling the optical system to work with lower attenuation.

While propagating through the fiber, the signal is progressively attenuated and distorted with increasing distance due to problems of scattering, absorption in the material and dispersion mechanisms in the fiber.

The principal light sources used for fiber optic communications applications are heterojunction-structured semiconductor laser diodes and light-emitting diodes (LEDs). A heterojunction consists of two adjoining semiconductor materials with different band-gap energies.

These devices are suitable for fiber transmission systems because then have adequate output power for a wide range applications, their optical power output can be directly modulated by varying the input current to the device, they have a high efficiency, and their dimensional characteristics are compatible with those of the optical fiber.

Ques 17) Explain briefly about the optical detectors and also discuss its principle and enlist the types of detector.

Ans: Optical Detectors

An optical detector is a device that converts light signals into electrical signals, which can then be amplified and processed. Such detectors are one of the most important components of an optical fiber communication system and dictate the performance of a fiber optic communication link.

There are many different types of photodetectors such as photomultiplier tubes, vacuum photodiodes, pyroelectric detectors, and semiconductor photodiodes. Semiconductor photodiodes are the most commonly used detectors in optical fiber systems since they provide good performance, are compatible with optical fibers (being small in size), and are of relatively low cost. These photodiodes are made generally from semiconductors such as silicon or germanium or from compound semiconductors such as GaAs, InGaAs, etc.

Principle of Optical Detection

The basic principle behind photodetection using semiconductors is optical absorption. When light is incident on a semiconductor, the light may or may not get absorbed depending on its wavelength. If the energy hν of a photo of the incident light beam is greater than the band gap of the semiconductor, then it can be absorbed, leading to generation of electron-hole pairs .When an electric field is applied across the semiconducting material, the photo generated electron-hole pairs are swept away, leading to a photo current in the external circuit.

If E_g is the bandgap of the semiconductor, then the maximum wavelength of absorption is given by,

$$\lambda_c = \frac{hc}{E_g} \qquad \text{..... (1)}$$

Substituting for $(h = 6.63 \times 10^{-34}\ \text{J.s})$ and c, the cutoff wavelength in micrometers is given by,

$$\lambda_c \approx \frac{1.24}{E_g(\text{ev})} \qquad \text{..... (2)}$$

The most common photo diode detectors are made of PIN diode (junctions made using p type-intrinsic type-n type materials). Avalanche photo diodes (APDs) are normally used where low power optical signal is received since it has better sensitivity due to avalanche effect. Important parameters of detectors are detector noise, detector efficiency and response time.

The principal source of noise in a photo detector is quantum noise, dark current noise generated in the bulk material of the photo diode and surface leakage current noise. The quantum efficiency is defined as number of electron-hole pairs generated per incident photon energy.

The response time is governed by the transit time of photon carriers in depletion region and the diffusion time of photon carriers generated outside the depletion region and RC time constant of the output circuit.

The figure of merit of a receiver is the minimum optical power necessary to detect data for a given signal to noise ratio (for analog signaling) or for a given probability of bit error (for digital signaling).

Types of Optical detectors

The optical detectors are classified into three categories:
1) Thermal detector
2) Photon detector
3) Coherent detector

Ques 18) Give the detail description of thermal detectors used in optical communication and also discuss its principle.

Ans: Thermal Detector

A thermal detector absorbs radiant energy, which causes a change of the detector's electrical characteristics. This electrical response to a change in the target temperature produces a signal that can be amplified and displayed, which respond to temperature changes of the material.

One of the most attractive characteristics of thermal detectors is the equal response to all wavelengths. This contributes to the stability of a system that must operate over a wide temperature range. Another significant factor is that thermal detectors do not require cooling. However, the response time of these detectors is in milliseconds and therefore relatively slows. The most common thermal detectors are thermocouple, thermopile, bolometer, and pyroelectric.

Principle of Operation of Thermal Detectors

The performance of a thermal detector will be calculated in two stages. First, by consideration of the thermal characteristics of the system, the temperature rise produced by the incident radiation is determined. Secondly, this temperature rise is used to determine the change in the property which is being used to indicate the signal. The first stage of the calculations is common to all thermal detectors, but the second stage will differ for the different types of thermal detectors.

Thermal detectors operate on a simple principle, that when heated by incoming IR radiation their temperature increases and the temperature changes are measured by any temperature-dependent mechanism, such as thermoelectric voltage resistance, pyroelectric voltage.

The simplest representation of the thermal detector is shown in **figure 6.18**. The detector is represented by a thermal capacitance C_{th} coupled via a thermal conductance G_{th} to a heat sink at a constant temperature T. In the absence of a radiation input the average temperature of the detector will also be T, although it

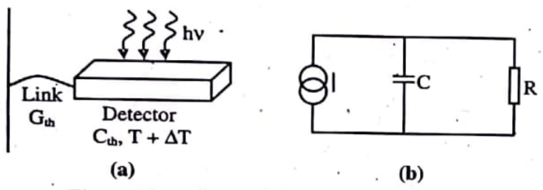


Figure 6.18: Thermal detector (a) and its electrical analogue (b)

Ques 19) What do you mean by photon detector? And also explain its working with the help of suitable schematic.

Or

Explain pin photo detector with the help of energy band diagram.

Ans: Photon (Quantum) Detector

Photon or quantum detectors operate on the quantum or photon effect (such as photoconductors and photodiodes) that forms an electrical signal directly from the interaction with individual photons. Photons are absorbed and produce free-charge carriers that change the electrical characteristic of the responsive element.

Photon detectors are much faster than thermal detectors; their response is in microseconds. Their detectivity is considerably higher. To obtain this high detectivity, the detector must be cooled.

For moderate temperature reductions, one or multistage thermoelectric coolers are employed. To provide cooling to very low temperatures of 77 K and even lower, cryogenic cooling methods must be applied.

Pin Photo detector is the most common semiconductor photon detector is the pin photodiode, shown schematically in **figure 6.19.**

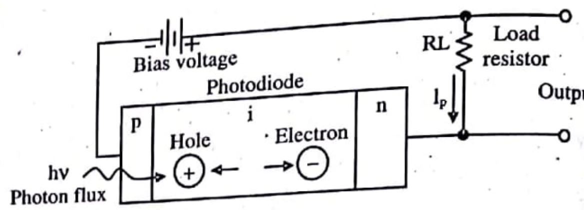


Figure 6.19: Pin Photodiode Circuit

Working of Photo Detector

The device structure consists of p and n regions separated by very lightly n-doped intrinsic region. In normal operation a sufficiently large reverse-bias voltage is applied across the device so that the intrinsic region is fully depleted of carriers. That is, the intrinsic n and p carrier concentrations are negligibly small in comparison with the impurity concentration in this region.

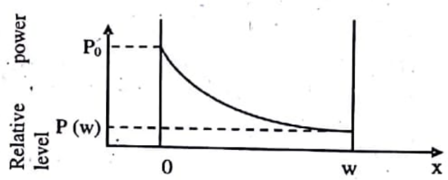


Figure 6.20: Incident Optical Level Decays Exponentially

When an incident photon has energy greater than or equal to the band-gap energy of the semiconductor material, the photon can give up its energy and excite an electron from the valance band to the conduction band. This process generates mobile electron-hole pairs, as shown in **Figure. 6.20**. These electrons and holes are known as photo-carriers, since they are photon-generated charge carriers that are available to produce a current flow when a bias voltage is applied across the device.

The number of charge carriers is controlled by the concentration level of impurity elements that are intentionally added to the material .The photo detector is normally designed so that these carriers are generated mainly in the depletion region where most of the incident light is absorbed.

The high electric field present in the depletion region causes the carriers to separate and be collected across the reverse-biased junction. This gives rise to a current flow in an external circuit, with one electron flowing for every carrier pair generated. This current flow is known as the **photocurrent**. As the charge carriers flow through the material, some electron-hole pairs will recombine and hence disappear. On the average, the charge carriers move a distance L_S or L_P for electrons and holes, respectively.

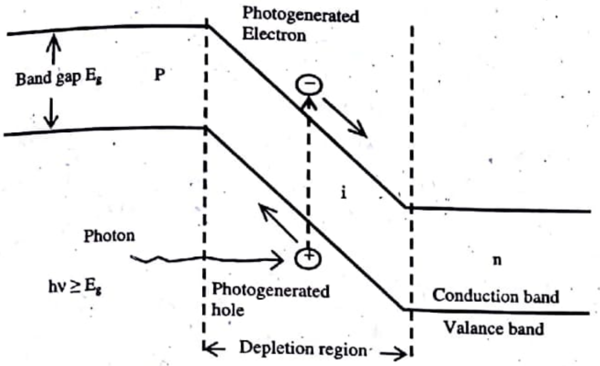


Figure 6.21: Energy-bond diagram

Photons with energies greater than or equal to the band-gap energy E_g can generate free electron hole pairs which act as photocurrent carriers. This distance is known as the diffusion length. The time it takes for an electron or hole to recombine is known as the carrier lifetime and represented by τ_n and τ_p respectively. The lifetimes and the diffusion lengths are related by the expressions,

$$l_n = (D_n \tau_n)^{1/2} \text{ and } l_p = (D_p \tau_p)^{1/2}$$

Where, D_n and D_p are the electron and hole diffusion coefficients (or constants), respectively, which are expressed in units of centimeters squared per second. As a photon flux ϕ penetrates into a semiconductor, it will be absorbed as it progress through the material.

Figure 6.21 gives an example of the power level as a function of the penetration depth into the intrinsic region, which has a width w. The width of the p region typically is very thin so that little radiation is absorbed there.

Ques 20) Describe about the coherent detectors with the help of suitable schematic.

Ans: Coherent Detectors

A "coherent" optical transmission system is characterized by its capability to do "coherent detection," which means that an optical receiver can track the phase of an optical transmitter (and hence "phase coherence") so as to extract any phase and frequency information carried by a transmitted signal.

Coherent detectors (heterodyne mixers) that operate on the interaction of the electric field of the incident radiation with a local oscillator.

For an optical coherent system, a narrow-line width tunnable laser, serving as an LO (local oscillator), tunes its frequency to "intradyne" with a received signal frequency through an optical coherent mixer, as shown in **Figure. 6.22**, and thereby recovers both the amplitude and phase information contained in a particular optical carrier.

Here, "intradyne" means that the frequency difference between an LO (local oscillator), and a received optical carrier is small and within the bandwidth of the receiver, but does not have to be zero. This implies that the frequency and phase of an LO do not have to be actively controlled to an extreme accuracy, therefore avoiding the use of a complicated optical phase locked loop.

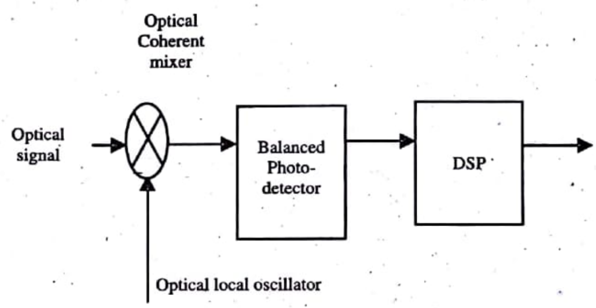


Figure 6.22: Closed-Loop System with Negative Feedback

Ques 21) Explain briefly about the digital filter. And also discuss its advantages.

Or

Enlist the advantage and disadvantages of digital filter.

Ans: Digital Filter

A **filter** is a function or device that passes only the desired signal components. A digital filter basically emulates an analog filter. A digital filter can be used to process an analog signal, but this requires that the signal be first sampled and quantized. An analog signal can be digitally filtered in real time if hardware with sufficient speed is available.

Advantages

A digital filter offers the following advantages:

1) Accurate signal values (within the available resolution);
2) High reproducibility (theoretically limited only by numerical rounding errors);
3) Processing flexibility (depending on the platform used for the realization);
4) Stable system operation (particularly if the effects of any operating system are excluded).

Disadvantages

The digital filters suffered from the following disadvantages:

1) Large size and physical complexity;
2) Difficulty of processing high-frequency signal in real time;
3) Significant arithmetic errors;
4) High power consumption (partly due to size and complexity).

OPTICAL LINKS

Ques 22) Discuss about the optical fiber link with the help its suitable schematic.

Or

Explain about the components of optical fiber link.

Ans: Optical Fiber Link

A fiber-optic link (or fiber channel) is a part of an optical fiber communications system which provides a data connection between two points (point-to-point connection). It essentially consists of a data transmitter, a transmission fiber (in some cases with built-in fiber amplifiers), and a receiver. Even for very long transmission distance, extremely high data rates of many (Gigabytes) Gbit/s or even several (Tetra bytes) Tbit/s can be achieved.

Components of Optical Fiber Link

The components of optical fibers link are discussed with the help of its schematic as shown in below **figure 6.23**.

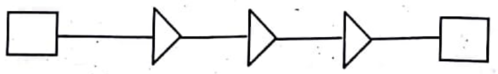


Figure 6.23: Schematic of fiber-optic link

The used components, which are mostly based on fiber optics are explained in the following:

1) **Transmitter:** The transmitter converts the electronic input signal into a modulated light beam. The information may be encoded e.g. via the optical power (intensity), optical phase or polarization; intensity modulation is most common.

   A typical transmitter is based on a single-mode laser diode, which may either be directly modulated via its drive current, or with an external optical modulator. Direct modulation is the simpler option, and can work at data rates of 10 Gbit/s or even higher. However, the varying carrier density in the laser diode then leads to a varying instantaneous frequency and thus to signal distortions in the form of a chirp.

   Therefore, external modulation is usually preferred for the combination of high data rates (10 or 40 Gbit/s) with long transmission distances. The laser can then operate in continuous-wave mode, and signal distortions are minimized. A light-emitting diode (LED) is used in the transmitter, but due to the poor spatial coherence this requires the use of multimode fibers. The transmission rate or distance is then restricted due to intermodal dispersion.

2) **Transmission Fiber:** The transmission fiber is usually a single-mode fiber in the case of medium or long-distance transmission, but can also be a multimode fiber for short distances. Long-range broadband fiber channels can contain fiber amplifiers at a certain points to prevent the power level from dropping to too low a level.

3) **Receiver:** The receiver contains photo detector, photodiode, and suitable high-speed electronics for amplifying the weak signal and extracting the digital or analog data. For high data rates, circuitry for electronic dispersion compensation may be included.

   Avalanche photodiodes can be used for particularly high sensitivity. The sensitivity of the receiver is limited by noise, normally of electronic origin; however, the optical signal itself is accompanied by optical noise. Such optical noise introduces limitations which cannot be removed with any receiver design.

4) **Bidirectional Transmission:** Full duplex links provide a data connection in both directions. These may simply be based on separate optical fibers, or work with a single fiber. The latter can be realized e.g. by using fiber-optic beam splitters at each end to connect a transmitter and a receiver. However, the need for bidirectional operation introduces various trade-offs, which in some cases (e.g. for very high data rates) make a system with two separate fibers preferable.

5) **Multiplexing:** A typical single-channel system for long transmission has transmission capacity of e.g. 2.5 or 10 Gbit/s; higher data rates of 40 Gbit/s or even 160 Gbit/s may be used in the future. For higher data rates, several data

channels can be multiplexed (combined), transmitted through the fiber, and separated again for detection.

The most common technique is wavelength division multiplexing (WDM). In different center wavelengths are assigned to different data channels. It is possible to combine even hundreds of channels in that way, but coarse WDM with a moderate number of channels is often preferred in order to keep the system simple. The main challenges are to suppress channel cross-talk via nonlinearities, to balance the channel power and to simplify the systems.

Another approach is time division multiplexing, where several input channels are combined by nesting in the time domain, and solitons are often used to ensure that the sent ultra-short pulses stay cleanly separated even at small pulse-to-pulse spacing.

6) **Active Optical Cables:** For short transmission distances, so-called active cables (AOC) can be used, where a transmitter and a receiver are rigidly attached to the ends of an optical fiber cable. Common electrical interfaces such as USB or HDMI ports are available, so the use of such an active optical cable is essentially the same as that of an electrical cable, while offering advantages like reduced diameter and weight and also a larger possible transmission distance.

Ques 23) Explain the bluetooth system. And also write its feature.

Or

Write a short note on bluetooth based communication system and also explain important terms related to bluetooth.

Ans: Bluetooth System

Bluetooth is a wireless LAN technology designed to connect devices of different functions such as telephones, notebooks, computers (desktop and laptop), cameras, printers, and so on. A bluetooth LAN is an ad-hoc network, which means that the network is formed spontaneously.

Bluetooth is an industrial specification for wireless personal area networks (PANs). Bluetooth provides a way to connect and exchange information between devices like personal digital assistants (PDAs), mobile phones, laptops, PCs, printers and digital cameras via a secure, low-cost, globally available short range radio frequency.

A bluetooth device has a built-in short-range radio transmitter. The current data rate is 1 Mbps with a 2.4-GHz bandwidth. This means that there is a possibility of interference between the IEEE 802.11b wireless LANs and Bluetooth LANs.

Feature of Bluetooth Based Comunication System

The usage of bluetooth has widely increased for its special features:

1) Bluetooth offers a uniform structure for a wide range of devices to connect and communicate with each other.
2) Bluetooth technology has achieved global acceptance such that any Bluetooth enabled device, almost everywhere in the world, can be connected with bluetooth enabled devices.
3) Low power consumption of bluetooth technology and an offered range of up to ten meters have paved the way for several usage models.
4) Bluetooth offers interactive conference by establishing an adhoc network of laptops.
5) Bluetooth usage model includes cordless computer, intercom, cordless phone and mobile phones.

Important Terms Related to Bluetooth

The important terms related to bluetooth are explained as follows:

1) **Spectrum:** Bluetooth technology operates in the unlicensed industrial, scientific and medical (ISM) band at 2.4 to 2.485 GHZ, using a spread spectrum hopping, full-duplex signal at a nominal rate of 1600 hops/sec. the 2.4 GHZ ISM band is available and unlicensed in most countries.

2) **Range:** Bluetooth operating range depends on the device Class 3 radios have a range of up to 1 meter or 3 feet Class 2 radios are most commonly found in mobile devices have a range of 10 meters or 30 feet Class 1 radios are used primarily in industrial use cases have a range of 100 meters or 300 feet.

3) **Data Rate:** Bluetooth supports 1Mbps data rate for version 1.2 and 3Mbps data rate for version 2.0 combined with error data rate.

Ques 24) What is the function of piconets and scatternets in bluetooth?

Ans: Function of Piconets in Bluetooth

A Bluetooth network is called a piconet, or a small net. A piconet can have up to eight stations, one of which is called the master; the rest are called slaves.

All the slave stations synchronize their clocks and hopping sequence with the master slave. A piconet can have only one master station. The communication between the master and the slaves can be one-to-one or one-to-many. **Figure 6.24** shows a piconet.

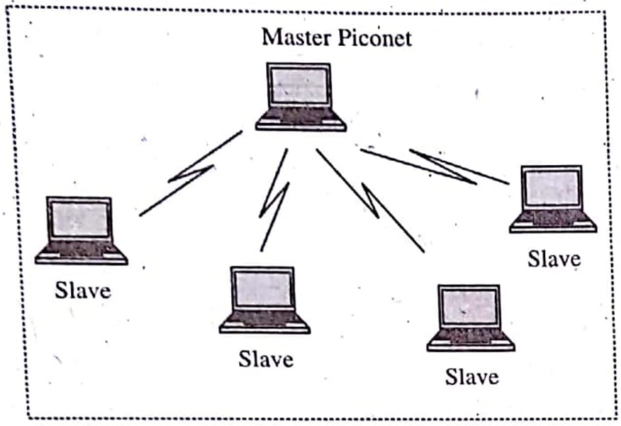


Figure 6.24: Piconet

Although a piconet can have a maximum of seven slaves, an additional eight slaves can be in the parked state. A slave in a parked state is synchronized with the master, but cannot take part in communication until it is moved from the parked state. Because only eight stations can be active in a piconet, activating a station from the parked state means that an active station must go to the parked state.

Function of Scatternet in Bluetooth

Piconets can be combined to form what is called a **scatternet**. A slave station in one piconet can become the master in another piconet. This station can receive messages from the master in the first piconet (as a slave) and, acting as a master, deliver it to slaves in the second piconet. A station can be a member of two piconets. **Figure 6.25** shows a scatternet.

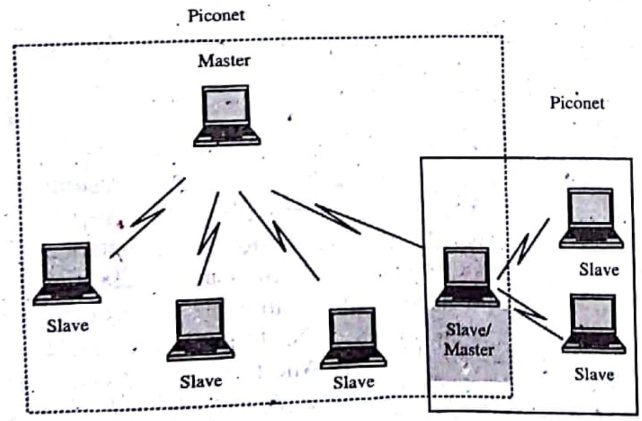


Figure 6.25: Scatternet

Ques 25) What is Zig-Bee technology? Also explain the function of Zig-Bee based communication system.

Ans: Zig-Bee Technology

Zig-bee is a wireless technology designed to address the unique needs such as affordability and power conservation. It is easy to implement and needs little power to operate. Zig-bee operates at the radio frequency of 2.4 GHz, which is used to deliver a reliable data and easy to use standards in the entire world.

Zig-bee is flexible in performance and is battery operated. Zig-bee operates in industrial, scientific and medical radio bands which operate at the frequency of 868 MHz in Europe and 915 MHz in Australia and USA. Zig-bee protocols are anticipated for embedded applications that require low power and low cost. Zig-bee can be used in different applications such as building automation, industrial control and medical data collection.

Zig-Bee Based Communication System

Now a day's confidential data transfer is a crucial task in many multinational companies, military departments, intelligence and surveillance departments, and so on. In such departments and companies lots of efforts are put forth for securing confidential data. Therefore, they need data encryption and decryption for their applications. Data encryption and decryption to secure data using zig-bee wireless communication technology for short distances.

A popular way to protect data is to encrypt the data while sending and decrypt it while receiving to regain the original message. Before transmitting, the data is converted into unreadable format, and then the data is encrypted and decrypted in the receiver end to get the original message.

Ques 26) Describe about the transmitter and receiver section of zig-bee communication system.

Ans: Transmitter and Receiver Section of Zig-Bee Communication System

The transmitter and receiver section of zig-bee communication system are as follows:

1) **Transmitter Section:** In a transmitter section, the data to be transferred to a remote location is entered using a keypad, and the data is sent to a microcontroller. The Microcontroller after receiving the data, based on its program

transfers the data to a MAX232 where in the TTL data is converted into a serial data. This serial encrypted data is then transmitted to the receiver section by a Zig-bee module and further gets displayed on the LCD.

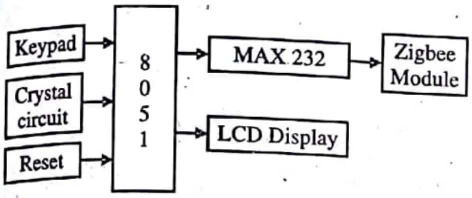


Figure 6.26: Wireless Data Encryption and Decryption Transmitter

The crystal circuit plays a key role in the microcontroller operation. This crystal circuit generates clock pulses so that the internal operation gets synchronized .When the reset pin is high, the microcontroller returns to a power on state, by leaving the currently executing program. RESET operation is performed by holding the RST pin high for at least two machine cycles.

2) **Receiver Section:** At the receiver end, the Zig-bee receiver module receives the data through air and the microcontroller decrypts the encrypted data. Finally the data is converted into the original data so that a user can read it and the decrypted data makes its way to the LCD display to get displayed there. Thus data can be protected at both the ends while transmitting and receiving.

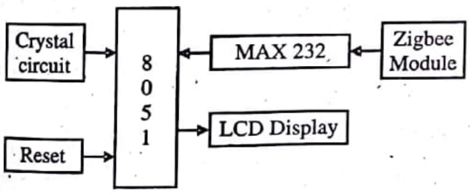


Figure 6.27: Wireless Data Encryption and Decryption

Ques 27) What is meant by GPS? And also discuss about the working of GPS system.

Or

Discuss about the GPS system with the help of its various segments.

Ans: GPS (Global Positioning System)

GPS (global positioning system) is a satellite navigation system that furnishes location and time information in all climate conditions to the user. GPS is used for navigation in planes, ships, cars and trucks also. The system gives critical abilities to military and civilian users around the globe. GPS provides continuous real time, 3-dimensional positioning, navigation and timing worldwide.

Working of GPS System

The GPS system consists of three segments:

1) **Space Segment:** The space segment is the number of satellites in the constellation. It comprises of 29 satellites circling the earth every 12 hours at 12,000 miles in altitude. The function of the space segment is utilized to route/navigation signals and to store and retransmit the route/navigation message sent by the control segment. These transmissions are controlled by highly stable atomic clocks on the satellites. The GPS space segment is formed by a satellite constellation with enough satellites to ensure that the users will have, at least, 4 simultaneous satellites in view from any point at the earth surface at any time.

2) **Control Segment:** The control segment comprises of a master control station and five monitor stations outfitted with atomic clocks that are spread around the globe. The five monitor stations monitor the GPS satellite signals and then send that qualified information to the master control station where abnormalities are revised and sent back to the GPS satellites through ground antennas. Control segment also referred as monitor station which is operated by the U.S. military.

3) **User Segment:** The user segment comprises of the GPS receiver, which receives the signals from the GPS satellites and determine how far away it is from each satellite. Mainly this segment is used for the U.S military, missile guidance systems, civilian applications for GPS in almost every field. Most of the civilian uses this from survey to transportation to natural resources and from there to agriculture purpose and mapping too.

Ques 28) Explain about the GPS based communication system. Also write its advantages, disadvantages and application.

Ans: GPS based Communication System

The working/operation of global positioning system is based on the 'trilateration' mathematical principle. The position is determined from the distance measurements to satellites. From the **figure 6.28**, the four satellites are used to determine the position of the receiver on the earth. The target location is confirmed by the 4th satellite. And three satellites are used to trace the location place. A fourth satellite is used to confirm the target location of each of those space vehicles.

Global positioning system consists of satellite, control station and monitor station and receiver. The GPS receiver takes the information from the satellite and uses the method of triangulation to determine a user's exact position.

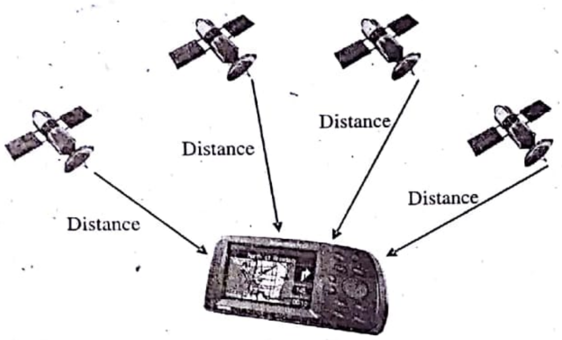


Figure 6.28

Advantages of GPS
The advantages of GPS communication system are as follows:

1) GPS satellite based navigation system is an important tool for military, civil and commercial users
2) Vehicle tracking systems GPS-based navigation systems can provide us with turn by turn directions
3) Very high speed

Disadvantages of GPS
The disadvantages are as follows:

1) GPS satellite signals are too weak when compared to phone signals, so it doesn't work as well indoors, underwater, under trees, etc.
2) The highest accuracy requires line-of-sight from the receiver to the satellite; this is why GPS doesn't work very well in an urban environment.

Application of GPS
The applications of GPS are as follows:

1) To determine position locations; **for example,** we need to radio a helicopter pilot the coordinates of our position location so the pilot can pick us up.
2) To navigate from one location to another; **for example**, we need to travel from a lookout to the fire perimeter.
3) To create digitized maps; **for example**, we are assigned to plot the fire perimeter and hot spots.
4) To determine distance between two different points.

Ques 29) What is the Wi-Fi? And also explain the working principle of Wi-Fi based communication system.

Or

Give the detail description of requirements for a Wi-Fi Network.

Ans: Wi-Fi
Wi-Fi is a popular wireless networking technology. Wi-Fi stands for "wireless fidelity". By using this Wi-Fi technology the information can exchange between two or more devices. Wi-Fi is a way for wireless devices to communicate. Wi-Fi is used for wireless communication.

Wi-Fi has been developed for mobile computing devices, such has laptops, but it is now extensively using for mobile applications and consumer electronics like televisions, DVD players and digital cameras. There should be two possibilities in communicating with the Wi-Fi connection that may be through access point to the client connection or client to client connection.

Wi-Fi is a one type of wireless technology. It is commonly called as wireless LAN (local area network). Wi-Fi allows local area networks to operate without cable and wiring. It is making popular choice for home and business networks. A computer's wireless adaptor transfers the data into a radio signal and transfers the data into antenna for users.

Working of Wi-Fi Network
A wireless network uses radio waves for transmission. In fact, communication across a wireless network is a lot like two-way radio communication. Following are the steps in wireless communication:

1) A computer's wireless adapter translates data into a radio signal and transmits it using an antenna.
2) A wireless router receives the signal and decodes it. The router sends the information to the Internet using a physical, wired Ethernet connection.

The process also works in reverse, with the router receiving information from the Internet, translating it into a radio signal and sending it to the computer's wireless adapter. They can transmit and receive radio waves, and they can convert 1s and 0s into radio waves and convert the radio waves back into 1s and 0s.

As long as they all have wireless adapters, several devices can use one router to connect to the Internet. This connection is convenient, virtually invisible and fairly reliable; however, if the router fails or if too many people try to use high-bandwidth applications at the same time, users can experience interference or lose their connections.

Requirements for a Wi-Fi Network

The requirements for a Wi-Fi network are as follows:

1) **Network Adapters**: A network adapter is the interface between a computer and a network. In a wireless network, the adapter contains a radio transmitter that sends data from the computer to the network and a receiver that detects incoming radio signals that contain data from the network and passes it along to the computer.

   A wireless adapter presents the same appearance to the computer's operating system as any other network interface.

2) **USB Adapters:** A USB port can be used to plug-in the wireless USB adapter; that is the best way for connecting the computer to the wireless network.

   Most USB adapters have captive antennas, often mounted on hinges or swivels that allow a user to make fine adjustments to their positions. Because the antennas on USB adapters are usually larger and easier to manipulate than the antennas in PC card adapters, hence somewhat better signal quality are expected through a USB device.

3) **Internal and External Antennas:** Many access points and most wireless network adapters come with captive antennas. For most users, in most situations, those built-in antennas will send and receive a strong, clean data stream between an access point and a nearby computer.

4) But if network adapters with built-in antennas are not able to provide a good enough signals, because of distance, obstructions, or interference from other radio signals, then an external antenna is best suited.

   There are two antennas in the link between a base station and a wireless network adapter – one at each end. A high-gain antenna at either end will have the same impact on the link, so it will be equally effective to replace the standard antenna on either the access point or the network interface. However, a directional antenna will focus most of the signal in one direction, so a directional access-point antenna can reduce the quality of links to other network nodes.

5) **Access Points:** A Wi-Fi access point broadcasts the wireless Internet connectivity to wireless devices. Most wireless network interface adapters perform just one function; they exchange data between a computer and a network.

6) Access points, in contrast, offer a wide variety of features and functions. They are available as simple access points and in combination with hubs, switches, and routers for wired connections to nearby computers and other devices. There is a whole category of wireless access points for home networks called residential gateways.

Ques 30) Discuss how to make the Security in Wi-Fi system? Also explain the various types of Wi-Fi technologies.

Ans: Security in Wi-Fi System

Security is impartment element in the Wi-Fi technology. Security is our personal decision but having a wireless connection so, attention should pay to protect our private details. We can connect easily to unsecured wireless routers.

The problem is any one is connected to your wireless router using the data like download games, download apps and planning terrorist activities, shirring illegal music and movie files etc. So it is necessary to provide security to the wireless technologies based devices.

All routers have a web page that can connect for configuring the Wi-Fi security. And turn on WEP (Wire Equivalence Privacy) and enter a password and remember this password. Next time when we will connect our laptop Wi-Fi router will ask you to enter the connection password and we enter that password.

Types of Wi-Fi Technologies

There are four major types of Wi-Fi technologies. These are explained as follows:

1) **Wi-Fi-802.11a:** 802.11a is the one of a series of wireless technology. That defines the format and structure of the radio signals sent out by Wi-Fi networking routers and antennas.

2) **Wi-Fi-802.11b:** 802.11b is the one of a series of wireless technology. 802. 11b support bandwidth 11mbps. Signal in unregulated frequency spectrum around 2.4 GHz. This is a low frequency compared with Wi-Fi-802.11a means it is working reasonable distance.

It is interference with micro owns cordless phones and other appliance. It is low-cost; signal range is good using home appliance.

3) **Wi-Fi-802.11g:** In 2002 and 2003, This Technology supporting a newer slandered products. It is best technology of 802.11a and 802.11b. The 802.11b support bandwidth up to 54mbps and it use a 2.4 GHz frequency for greater range. This cost is more than 802.11b. It is fast accessing and maximum speed.

4) **Wi-Fi-802.11n:** The 802.11n is the newest Wi-Fi technology. It was designed to improve on 802.11g .The amount of bandwidth supported by utilizing multiple wireless signals and antennas instead of one. It supports 100 mbps bandwidth and increased signal intensity.

Ques 31) Write application, advantages and disadvantages of Wi-Fi system.

Ans: Applications of Wi-Fi System

1) Mobile applications
2) Business applications
3) Home applications
4) Computerized application.
5) Automotive segment
6) Browsing internet
7) Video conference

Advantages of Wi-Fi System

1) Wireless laptop can be moved from one place to another place.
2) Wi-Fi network communication devices without wire can reduce the cost of wires.
3) Wi-Fi setup and configuration is easy than cabling process.
4) It is completely safe and it will not interfere with any network.
5) It can also connect internet via hot spots and wirelessly.

Disadvantages of Wi-Fi System

1) Wi-Fi generates radiations which can harm the human health.
2) There are some limits to transfer the data, it can't able to transfer the data for long distance.
3) Wi-Fi implementation is very expensive when compared to the wired connection.

Ques 32) What is Wi-Max? Explain the types of Wi-Max version in the communication system.

Ans: Wi-Max

Wi-Max (Worldwide Interoperability for Microwave Access) is a technology standard for long-range wireless networking, for both mobile and fixed connections. While Wi-Max was once envisioned to be a leading form of internet communication as an alternative to cable and DSL, its adoption has been limited.

Wi-Max can satisfy a variety of access needs. Potential applications include extending broadband capabilities to bring them closer to subscribers, filling gaps in cable, DSL, Wi-Fi, and cellular backhaul, providing last-100 meter access from fiber to the curb and giving service providers another cost-effective option for supporting broadband services.

Wi-Max can support very high bandwidth solutions where large spectrum deployments (i.e. >10 MHz) are desired using existing infrastructure keeping costs down while delivering the bandwidth needed to support a full range of high-value multimedia services Wi-Max can help service providers meet many of the challenges they face due to increasing customer demands without discarding their existing infrastructure investments because it has the ability to seamlessly interoperate across various network types.

Wi-Max can provide wide area coverage and quality of service capabilities for applications ranging from real-time delay-sensitive voice-over-IP (VoIP) to real-time streaming video and non-real-time downloads, ensuring that subscribers obtain the performance they expect for all types of communications.

Wi-Max, which is an IP-based wireless broadband technology, can be integrated into both wide-area third-generation (3G) mobile and wireless and wire line networks allowing it to become part of a seamless anytime, anywhere broadband access solution. Ultimately, Wi-Max is intended to serve as the next step in the evolution of 3G mobile phones, via a potential combination of Wi-Max and CDMA standards called 4G.

Types of Wi-Max Version

The two flavours of Wi-Max technology are used for different applications and although they are based on the same standard, the implementation of each has been optimized to suit its particular application:

1) **802.16d - DSL Replacement:** The 802.16d version is often referred to as 802.16-2004 and it is closer to what may be termed the original version of Wi-Max defined under 802.16a. It is aimed at fixed applications and providing a wireless equivalent of DSL broadband data.

   In fact the Wi-Max Forum describes the technology as "a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and DSL."

   802.16d is able to provide data rates of up to 75 Mbps and as a result it is ideal for fixed, DSL replacement applications. It may also be used for backhaul where the final data may be distributed further to individual users. Cell radii are typically up to 75 km.

2) **802.16e - Nomadic/Mobile:** While 802.16 / Wi-Max was originally envisaged as being a fixed only technology, with the need for people on the move requiring high speed data at a cost less than that provided by cellular services and opportunity for a mobile version was seen and 802.16e was developed.

   This standard is also widely known as 802.16-2005. It currently provides the ability for users to connect to a Wi-Max cell from a variety of locations, and there are future enhancements to provide cell handover. 802.16e is able to provide data rates up to 15 Mbps and the cell radius distances are typically between 2 and 4 km.