DRONACHARYA
College of Engineering

LABORATORY MANUAL

B.Tech. Semester- IV

COMMUNICATION SYSTEM LAB
Subject code: LC-ECE-204G

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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING
DRONACHARYA COLLEGE OF ENGINEERING
KHENTAWAS, FARRUKH NAGAR, GURUGRAM (HARYANA)

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Vision and Mission of the Institute

Vision:

To impart Quality Education, to give an enviable growth to seekers of learning, to groom them as World Class Engineers and managers competent to match the expending expectations of the Corporate World has been ever enlarging vision extending to new horizons of Dronacharya College of Engineering

Mission:

1. To prepare students for full and ethical participation in a diverse society and encourage lifelong learning by following the principle of 'Shiksha evam Sahayata' i.e. Education & Help.
2. To impart high-quality education, knowledge and technology through rigorous academic programs, cutting-edge research, & Industry collaborations, with a focus on producing engineers& managers who are socially responsible, globally aware, & equipped to address complex challenges.
3. Educate students in the best practices of the field as well as integrate the latest research into the academics.
4. Provide quality learning experiences through effective classroom practices, innovative teaching practices and opportunities for meaningful interactions between students and faculty.
5. To devise and implement programmes of education in technology that are relevant to the changing needs of society, in terms of breadth of diversity and depth of specialization.

Vision and Mission of the Department

VISION

To be a Centre of Excellence for producing high quality engineers and scientists capable of providing sustainable solutions to complex problems and promoting cost effective indigenous technology in the area of Electronics, Communication & Control Engineering for Industry, Research Organizations, Academia and all sections of society."

MISSION OF THE DEPARTMENT

M1 To frame a well-balanced curriculum with an emphasis on basic theoretical knowledge as well the requirements of the industry.

M2. To motivate students to develop innovative solutions to the existing problems for betterment of the society.

M3. Collaboration with the industry, research establishments and other academic institutions to bolster the research and development activities in department.

M4. To provide infrastructure and support for culmination of novel ideas into useful prototypes.

M5. To promote research in emerging and interdisciplinary areas and act as a facilitator for knowledge generation and dissemination through Research, Institute – Industry and Institute-Institute interaction**.**

Programme Educational Objectives (PEOs)

PEO1: To practice the profession of engineering using a systems perspective and analyze, design, develop, optimize & implement engineering solutions and work productively as engineers, including supportive and leadership roles on multidisciplinary teams

PEO2: To Continue their education in leading graduate programs in engineering & interdisciplinary areas to emerge as researchers, experts, educators & entrepreneurs and recognize the need for, an ability to engage in continuing professional development and life-long learning

PEO3: To Engineers, guided by the principles of sustainable development and global interconnectedness, will understand how engineering projects and affect society and the environment.

PEO4: To Promote Design, Research and implementation of products and services in the field of Engineering through strong Communication and Entrepreneurial skills.

PEO5: To Re-learn and innovate in ever-changing global economic and technological environments on the 21st century.

Programme Outcomes (POs)

PO1: Engineering knowledge: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.

PO2: 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.

PO3: 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.

PO4: 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.

PO5: Modern tool usage: Create, select, and apply appropriate techniques, resources, and modern engineering and software tools including prediction and modeling to complex engineering activities with an understanding of the limitations.

PO6: 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.

PO7: 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.

PO8: Ethics: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.

PO9: Individual and team work: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.

PO10: 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.

PO11: 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.

PO12: 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.

Programme Specific Outcomes (PSOs)

PSO1. Equip themselves to potentially rich & employable field of Engineering. Analyse and design electronic systems for signal processing, communications and other applications.

PSO2. Pursue higher studies in the contemporary Technologies and multidisciplinary fields with an inclination towards continuous learning. area of Electronics, Telecommunication ,VLSI or Instrumentation.

PSO3. Design, implement and evaluate processes, components and/or programs using modern techniques, skills and tools of core Information Technologies to effectively integrate effective communication-based solutions into using Electronic components.

PSO4. Develop impactful solutions by using research-based knowledge and methods in the fields of integration and implementation, alongside Meeting the requirements of the Industrial standard.

University Syllabus

LIST OF EXPERIMENTS:

1. To study and waveform analysis of amplitude modulation and determine the modulation index of amplitude modulation.
2. To study and waveform analysis of amplitude demodulation by any method.
3. To study and waveform analysis of frequency modulation and determine the modulation index of frequency modulation.
4. To study and waveform analysis of frequency demodulation by any method.
5. To study Amplitude Shift Keying (ASK) modulation.
6. To study Frequency Shift Keying (FSK) modulation.
7. To study Phase Shift Keying (PSK) modulation.
8. To study and waveform analysis of phase modulation.
9. To study Phase demodulation.
10. To study Pulse code modulation.
11. To study Pulse amplitude modulation and demodulation.
12. To study Pulse width modulation.
13. To study Pulse position modulation.
14. To study delta modulation.
15. To deliver a seminar by each student on ADVANCE COMMUNICATION SYSTEM.

Course Outcomes (COs)

After completion of this course, students will be able to :

CO1 : Students are able to analyze digital communication signals.

CO2 : Students understand the basics of PAM, QAM, PSK, FSK, and MSK.

CO3 : They can analyze noise and disturbance in modulated signals.

CO-PO Mapping

	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12
CO1	3	3	2	3	3						1	3
CO2	3	3	2	3	2					1	1	3
CO3	3	3	2	3	2						1	3

CO-PSO Mapping

	PSO1	PSO2	PSO3	PSO4
CO1	3	3		1
CO2	3	3		1
CO3	3	3		1

Course Overview

The Communication System Lab is a practical course that complements the theoretical concepts covered in the Communication Systems course. It provides students with hands-on experience in designing, implementing, and analyzing various communication systems and their components. The lab sessions focus on developing practical skills and enhancing understanding of communication system principles through experimentation and measurement. It provides an overview of basic communication system components, modulation techniques, and signal processing concepts. It will help in practical exercises involving analog modulation techniques such as Amplitude Modulation (AM), Frequency Modulation (FM), and Phase Modulation (PM). Students learn to generate and demodulate analog signals using modulation and demodulation circuits. It gives information about digital Communication Experiments: Hands-on experiments related to digital communication techniques, including Pulse Amplitude Modulation (PAM), Pulse Code Modulation (PCM), and digital modulation schemes like Phase Shift Keying (PSK) and Quadrature Amplitude Modulation (QAM). Laboratory exercises on signal processing techniques, such as filtering, noise reduction, and equalization. Students apply filters to different signals and evaluate their impact on the communication system performance. Error Detection and Correction: Practical sessions on error detection and correction codes, including Hamming codes, Reed-Solomon codes, and cyclic redundancy checks (CRC). Students implement these codes to detect and correct errors in digital communication systems. It provides knowledge about communication System Simulation: Introduction to software tools for simulating communication systems. Students use simulation software to design, analyze, and optimize various communication system parameters.

Throughout the course, students will learn how to use communication system hardware, measurement instruments, simulation software, and analysis tools. They will gain practical experience in troubleshooting, analyzing system performance, and interpreting measurement results. The Communication System Lab provides a valuable opportunity for students to bridge the gap between theory and practical implementation, fostering a deeper understanding of communication system concepts.

List of Experiments mapped with Cos

S. No	List of Experiments	Course Outcomes
1.	To study and waveform analysis of Amplitude Modulation and determine the modulation index of Amplitude Modulation.	CO2
2.	To study and waveform analysis of Amplitude Demodulation by any method.	CO2
3.	To study and waveform analysis of Frequency Modulation and determine the modulation index of Frequency Modulation.	CO2
4.	To study and waveform analysis of Frequency Demodulation by any method.	CO2
5.	To study Amplitude Shift Keying (ASK) Modulation.	CO1
6.	To study Frequency Shift Keying (FSK) Modulation.	CO1
7.	To study Phase Shift Keying (PSK) Modulation.	CO1
8.	To study and waveform analysis of Phase Modulation & Demodulation.	CO2,CO3
9.	To study Pulse Code Modulation.	CO2,CO3
10.	To study Pulse Amplitude Modulation and Demodulation.	CO2,CO3
11.	To study Pulse Width Modulation.	CO1,CO3
12.	To study Pulse Position Modulation.	CO1,CO3
13.	To study Delta Modulation.	CO1,CO3

DOs and DON'Ts

DOs

1. Login-on with your username and password.
2. Log off the Computer every time when you leave the Lab.
3. Arrange your chair properly when you are leaving the lab.
4. Put your bags in the designated area.
5. Ask permission to print output waveforms.

DON'Ts

1. Do not touch any components in operating conditions .
2. Do not remove or disconnect cables or hardware parts.
3. Do not personalize the computer setting.
4. Do not run programs that continue to execute after you log off.
5. Do not download or install any programs, games or music on computer in Lab.
6. Personal Internet use chat room for Instant Messaging (IM) and Sites is strictly prohibited.
7. No Internet gaming activities allowed.
8. Tea, Coffee, Water & Eatables are not allowed in the Computer Lab.

General Safety Precautions

Precautions (In case of Injury or Electric Shock)

1. To break the victim with live electric source, use an insulator such as fire wood or plastic to break the contact. Do not touch the victim with bare hands to avoid the risk of electrifying yourself.
2. Unplug the risk of faulty equipment. If main circuit breaker is accessible, turn the circuit off.
3. If the victim is unconscious, start resuscitation immediately, use your hands to press the chest in and out to continue breathing function. Use mouth-to-mouth resuscitation if necessary.
4. Immediately call medical emergency and security. Remember! Time is critical; be best.

Precautions (In case of Fire)

1. Turn the equipment off. If power switch is not immediately accessible, take plug off.

2. If fire continues, try to curb the fire, if possible, by using the fire extinguisher or by covering it with a heavy cloth if possible isolate the burning equipment from the other surrounding equipment.

3. Sound the fire alarm by activating the nearest alarm switch located in the hallway.

4. Call security and emergency department immediately:

Emergency : Reception

Security : Main Gate

Guidelines to students for report preparation

All students are required to maintain a record of the experiments conducted by them. Guidelines for its preparation are as follows:-

1) All files must contain a title page followed by an index page. ***The files will not be signed by the faculty without an entry in the index page.***
2) Student's Name, Roll number and date of conduction of experiment must be written on all pages.
3) For each experiment, the record must contain the following :
   (i) Aim/Objective of the experiment .
   (ii) Pre-experiment work (as given by the faculty) .
   (iii) Lab assignment questions and their solutions.
   (iv) Test Cases (if applicable to the course) .
   (v) Results/ output.

Note:

1. Students must bring their lab record along with them whenever they come for the lab.
2. Students must ensure that their lab record is regularly evaluated.

Lab Assessment Criteria

An estimated 10 lab classes are conducted in a semester for each lab course. These lab classes are assessed continuously. Each lab experiment is evaluated based on 5 assessment criteria as shown in following table. Assessed performance in each experiment is used to compute CO attainment as well as internal marks in the lab course.

Grading Criteria	Exemplary (4)	Competent (3)	Needs Improvement (2)	Poor (1)
AC1: Pre-Lab written work (this may be assessed through viva)	Complete procedure with underlined concept is properly written	Underlined concept is written but procedure is incomplete	Not able to write concept and procedure	Underlined concept is not clearly understood
AC2: Program Writing/ Modeling	Assigned problem is properly analyzed, correct solution designed, appropriate language constructs/ tools are applied, Program/solution written is readable	Assigned problem is properly analyzed, correct solution designed, appropriate language constructs/ tools are applied	Assigned problem is properly analyzed & correct solution designed	Assigned problem is properly analyzed
AC3: Identification & Removal of errors/ bugs	Able to identify errors/ bugs and remove them	Able to identify errors remove them with little bit of guidance	Is dependent totally on someone for identification of errors/ bugs and their removal	Unable to understand the reason for errors/ bugs even after they are explicitly pointed out
AC4:Execution & Demonstration	All variants of input /output are tested, Solution is well demonstrated and implemented concept is clearly explained	All variants of input /output are not tested, However, solution is well demonstrated and implemented concept is clearly explained	Only few variants of input /output are tested, Solution is well demonstrated but implemented concept is not clearly explained	Solution is not well demonstrated and implemented concept is not clearly explained
AC5:Lab Record Assessment	All assigned problems are well recorded with objective, design constructs and solution along with Performance analysis using all variants of input and output	More than 70 % of the assigned problems are well recorded with objective, design contracts and solution along with Performance analysis is done with all variants of input and output	Less than 70 % of the assigned problems are well recorded with objective, design contracts and solution along with Performance analysis is done with all variants of input and output	

LAB EXPERIMENTS

EXPERIMENT No. 1

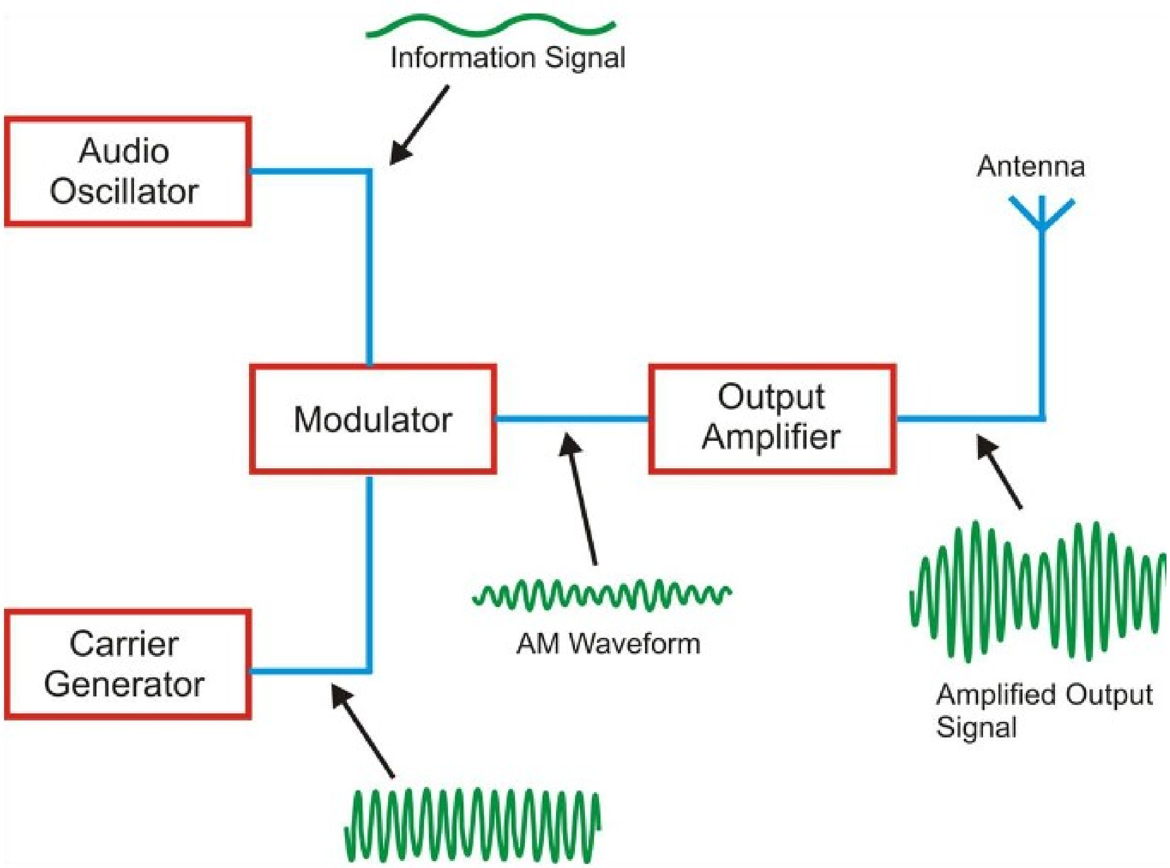
OBJECTIVE:- To study and waveform analysis of Amplitude Modulation and determine the modulation index of Amplitude Modulation.

APPARATUS REQUIRED:- AM transmitter 2201, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

Modulation is defined as the process by which some characteristics of a carrier signal is varied in accordance with a modulating signal. The base band signal is referred to as the modulating signal and the output of the modulation process is called as the modulated signal. Modulation is performed in a transmitter by a circuit called a modulator. Need for modulation is as follows:

- Avoid mixing of signals
- Reduction in antenna height
- long distance communication
- Multiplexing
- Improve the quality of reception
- Ease of radiation



AM Transmitter System

Amplitude Modulation is the process of changing the amplitude of a relatively high frequency carrier signal in proportion with the instantaneous value of the modulating signal. The general equation of Amplitude Modulated signal is given by

$$s(t) = A_c(1 + x(t))\cos\omega_c t$$

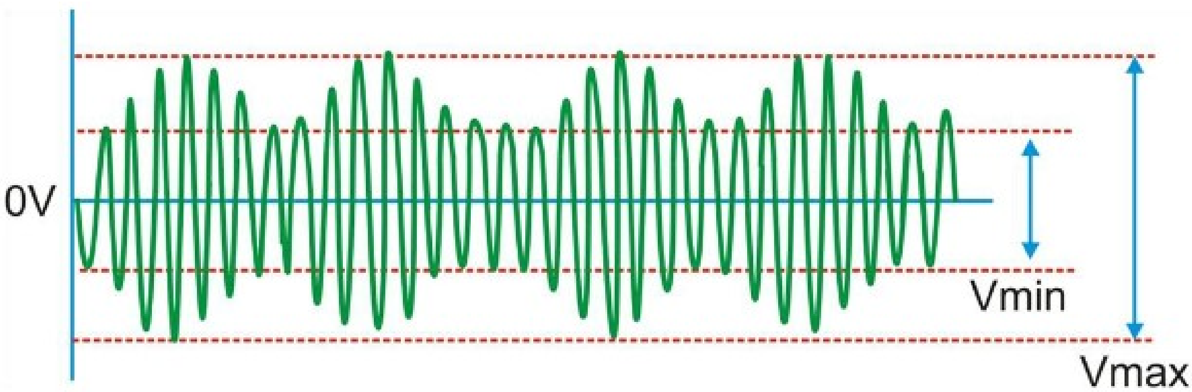
Where $A_c \cos\omega_c t$ is the carrier signal & x(t) is the message signal. If the type of Modulation is single-tone modulation then message signal is given by

$$x(t) = A_m \cos\omega_m t.$$

The shape of the modulated wave is called the AM envelope. With no modulating signal the output waveform is simply the carrier signal. Modulation index is a term used to describe the amount of amplitude change present in an AM waveform. There are three degrees of modulation available based on value of modulation index.

(i) Under modulation : $m_a < 1$, $A_m < A_c$

(ii) Critical modulation: $m_a = 1$, $A_m = A_c$



(iii) Over modulation: $m_a > 1$, $A_m > A_c$

The Modulation Index is calculated as,

$$m_a = \frac{A_m}{A_c} = \frac{(V_{\max} - V_{min})}{(V_{max} + V_{min})}$$

where V_{max} and V_{min} are the maximum and minimum amplitudes of the modulated wave.

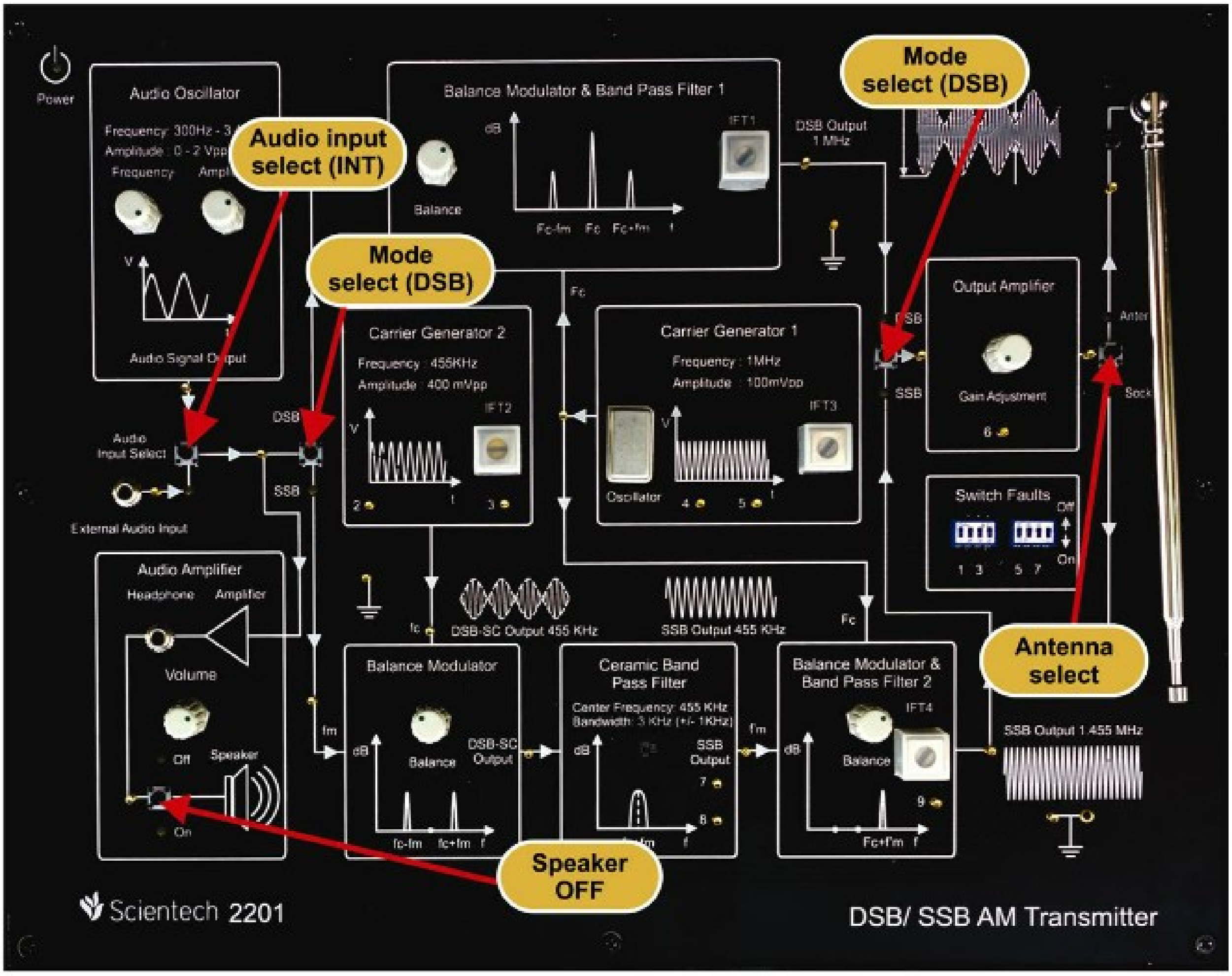
<u>PRE EXPERIMENT QUESTIONS</u>

Q.1. Define Amplitude Modulation.

Q.2. What is the need for Modulation?

<u>PROCEDURE</u>:-

Step 1 Ensure that the following initial conditions exist on the board.

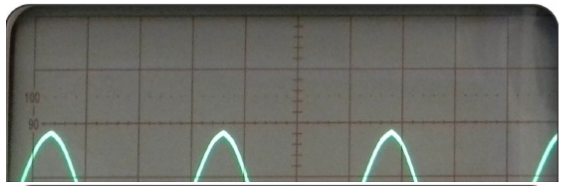


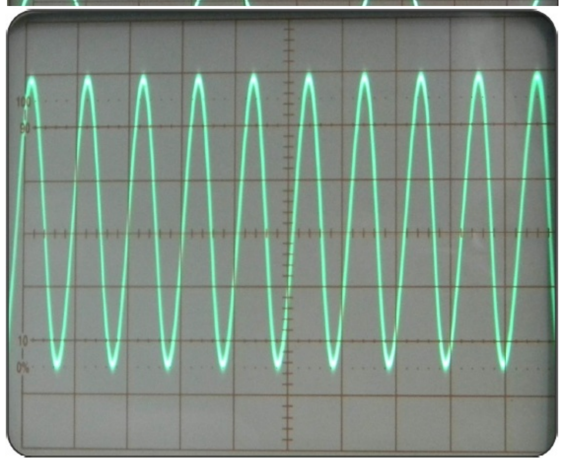
a. Audio input select switch should be in INT position to select onboard generated audio signal as a modulating signal.
b. Mode switch in DSB position to connect the DSB signal to Output Amplifier section.
c. Output amplifier's gain potentiometer in full clockwise position for maximum amplification.
d. Speakers switch in OFF position.

Step 2 Turn on power to the Scientech 2201 board.

Step 3 Observe the output of ‘Audio Oscillator’ block at ‘Audio Signal Output’ test point on Oscilloscope. Amplitude and Frequency of this audio signal can be varied using the respective Amplitude and Frequency control pots. The amplitude varies from 0 to 2vpp approx and frequency varies from 300 Hz to 3 KHz approx. This is the audio frequency sine wave which will use as modulating signal input to Balanced Modulator and Band Pass Circuit 1.

[CH1(Y) – 0.5V; Time base – 0.1 mS]

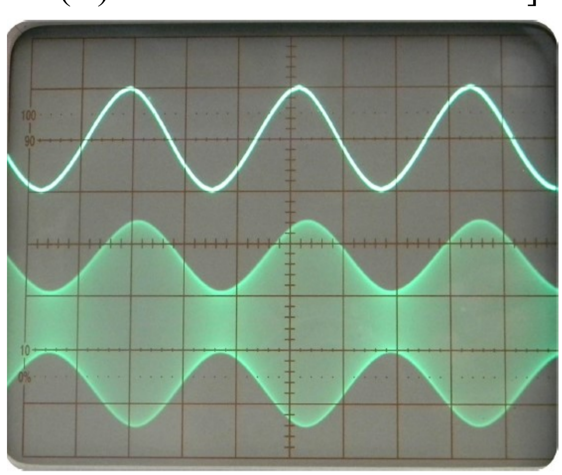




Step 4 1 MHz Crystal Oscillator Block generates a sine wave of 1 MHz frequency and 120mV pp amplitude approx, which is used as a carrier input to Balance Modulator and Band Pass Filter Circuit 1. Observe the carrier waveform at output test point on Oscilloscope. [CH1(Y) – 10mV; Time base – 1 uS]

Step 5 Balanced Modulator and Band Pass Filter Circuit 1 use to perform 'Double Side Band Amplitude Modulation'. Balance pot is used to vary the depth of modulation AM waveform. Initially turn the pot to its maximum position and observe the DSB AM output on Oscilloscope together with the modulating Audio Signal output l at Trigger the Oscilloscope on the Audio signal output. The output from the balanced modulator & band pass filter circuit 1 block is a double-sideband. AM waveform, which has been formed by amplitude modulating the 1MHz carrier sinewave with the audio-frequency sinewave from the audio oscillator.

[CH1(Y) – 1V; CH2(X) – 0.2V Time base – 0.1 mS]



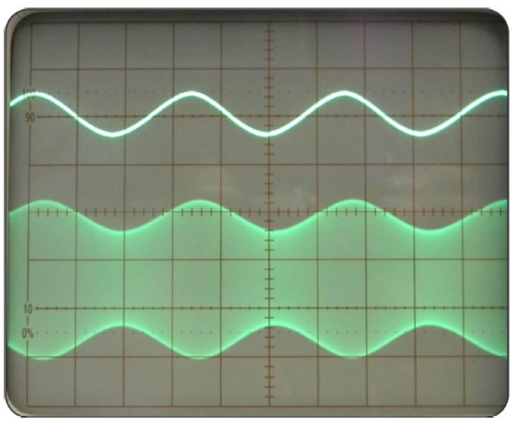
Step 6 To determine the depth of modulation, measure the maximum amplitude (V_{max}) and the minimum amplitude (V_{min}) of the AM waveform, and use the following formula:

$$\text{Percentage Modulation} = \frac{(V_{\max} - V_{min})}{(V_{max} + V_{min})} X\ 100\ \%$$

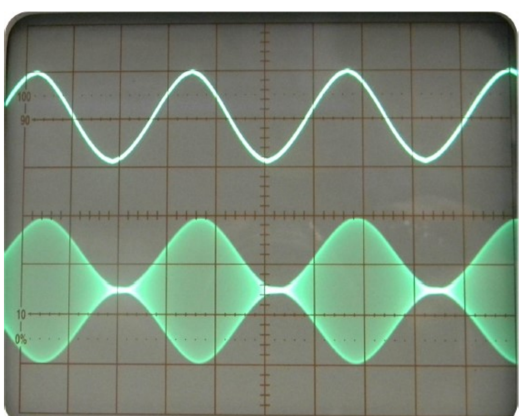
Step 7 Now vary the frequency of the modulating audio signal by varying the frequency pot of audio oscillator block and observe the effect on AM waveform. The frequency of envelop also varies with respect to the modulating audio signal frequency.

Step 8 Now vary the amplitude of the modulating audio signal by varying the amplitude pot in the audio oscillator block and observe the effect on AM waveform. The amplitude of two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero.

<u>WAVE FORMS OBSERVED</u>:-



50 % Modulation



100 % Modulation

<u>TABULATION:-</u>

INPUT SIGNALS:

Signals	Amplitude (V)	Time period (ms)	Frequency (kHz)
Modulating signal			

Carrier signal			

MODULATED SIGNAL:

V_{max} (V)	V_{min} (V)	$m_a = \frac{(V_{max}-V_{min})}{(V_{max}+V_{min})} X\ 100\ \%$	Type of Modulation

RESULT:-

Thus the characteristics of AM Transmitter is studied and the waveforms are observed and plotted.

1. As the amplitude of message signal increases, the Modulation index increases and vice versa.
2. When message signal and carrier signal are in-phase it represents V_{max}.
3. When message signal and carrier signal are out of phase it represents V_{min}.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

POST EXPERIMENT QUESTIONS:

Q.3. What are the different types of AM?

Q.4. What happens when the amplitude of the modulating signal is greater than the amplitude of the carrier?

Q.5. What is the reference line for the modulating signal?

Q.6. Explain under modulation, 100% modulation, over modulation.

Q.7. Write the formulae to calculate practical modulation index.

Q.8 What are advantages of AM?

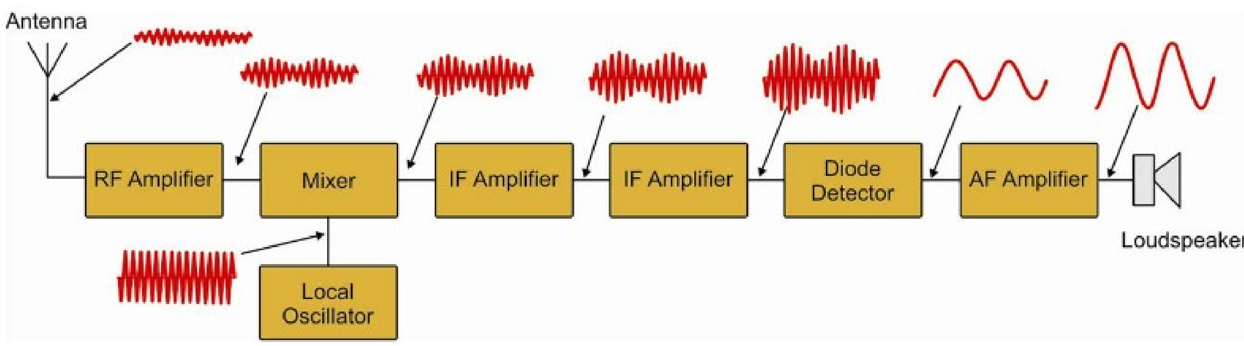
EXPERIMENT No. 2

OBJECTIVE:- To study and waveform analysis of Amplitude Demodulation by any method.

APPARATUS REQUIRED:- AM transmitter 2201, AM receiver 2202, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

The demodulation circuit is used to recover the message signal from the incoming AM wave at the receiver. An envelope detector is a simple and yet highly effective device that is well suited for the demodulation of AM wave, for which the percentage modulation is less than 100%. Ideally, an envelope detector produces an output signal that follows the envelope of the input signal wave form exactly; hence, the name. The depth of modulation at the detector output greater than unity and circuit impedance is less than circuit load ($R_L > Z_m$) results in clipping of



DSB Receiver

negative peaks of modulating signal. It is called “negative clipping.”

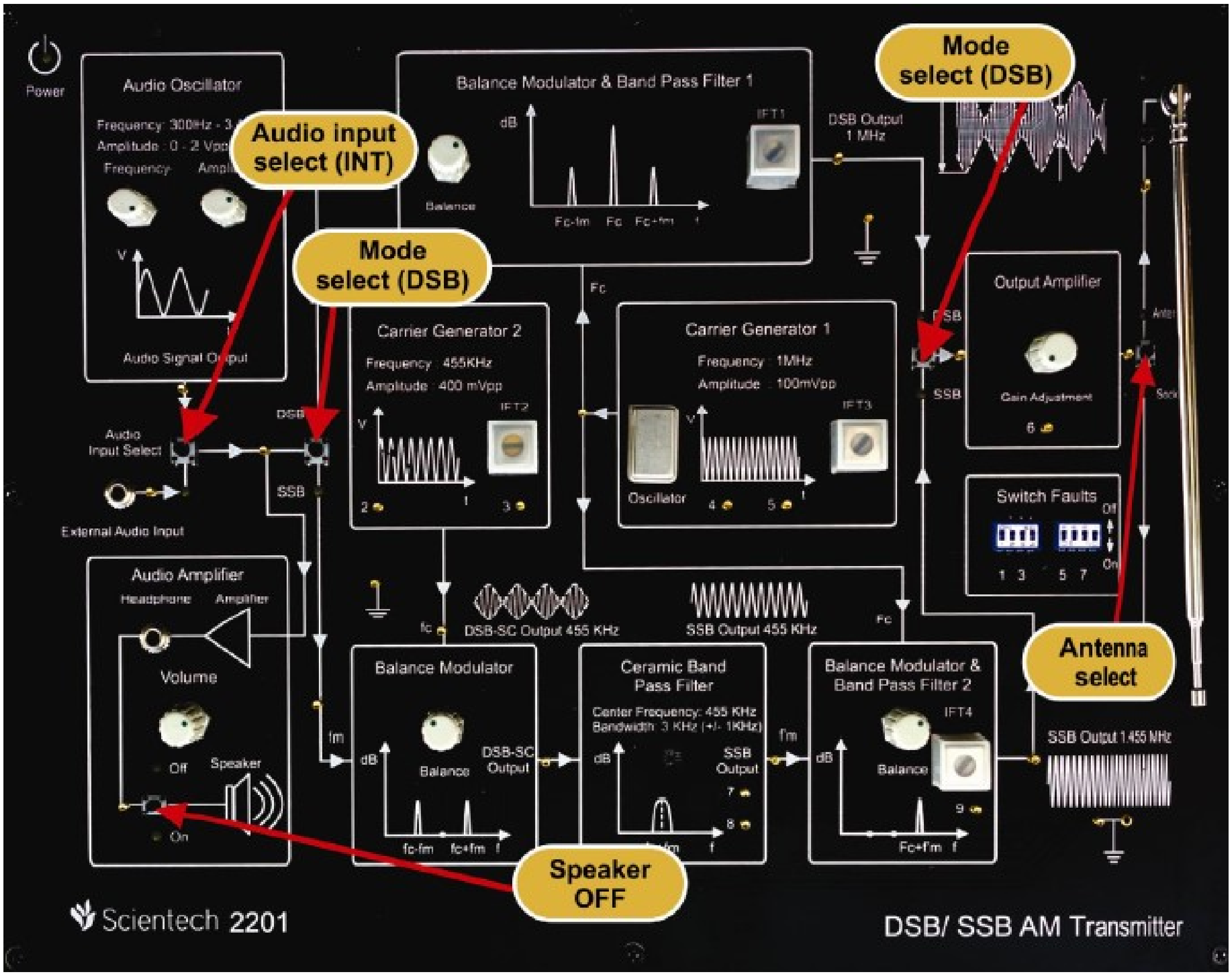
The envelope of the modulating wave has the same shape as the base band message provided the following two requirements are satisfied:

1. The carrier frequency f_c must be much greater than the highest frequency components f_m of the message signal m (t) i.e. $f_c >> f_m$.
2. The modulation index must be less than unity. If the modulation index is greater than unity, the carrier wave becomes over modulated.

PRE EXPERIMENT QUESTIONS:

Q.1. Define Amplitude Demodulation.

Q.2. When does over-modulation occur?



PROCEDURE:-

Step 9 Position the Scientech 2201 & Scientech 2202 modules, with the Scientech 2201 board on the left, and a gap of about three inches between them.

Step 10 Ensure that the following initial conditions exist on the Scientech 2201 board.

a. Audio oscillator's amplitude pot in fully clockwise position.
b. Audio input select switch in INT position.
c. Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position.
d. Mode switch in DSB position.
e. Output amplifier's gain pot in full clockwise position.
f. TX output select switch in ANT position.
g. Audio amplifier’s volume pot in fully counter-clockwise position.
h. ‘Speaker’ switch in ON position.
i. On-board antenna in vertical position, and fully extended.

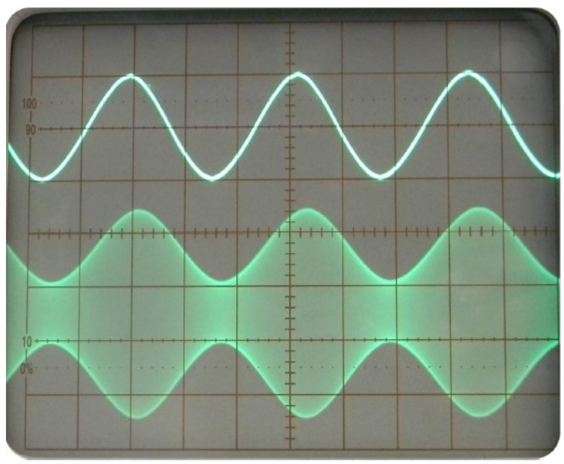
Step 11 Ensure that the following initial conditions exist on the Scientech 2202 board:

a. RX input select switch in ANT position.

b. R.F. amplifier's tuned circuit select switch in 'INT' position.
c. R.F. amplifier's gain pot in fully clock-wise position;
d. AGC switch in 'IN' position.
e. Detector switch in 'Diode' position.
f. Audio amplifier's volume pot in fully counter-clockwise position.
g. 'Speaker' switch in ON position.
h. Beat frequency oscillator switch in OFF position.
i. On-board antenna in vertical position, and fully extended.Turn on power to the Scientech 2201 board.

Step 12 Turn on power to the modules.

Step 13 On the Scientech 2201 module, examine the transmitter's output signal, together with the audio modulating Audio signal.
[CH1(Y) – 1V; CH2(X) – 0.2V Time base – 0.1 mS]



Since Scientech 2201 TX output select switch is in the ANT position, the AM signal at the output is fed to the transmitter's antenna. Prove this by touching Scientech 2201's antenna, and nothing that the loading caused by your hand reduces the amplitude of the AM waveform at the output.

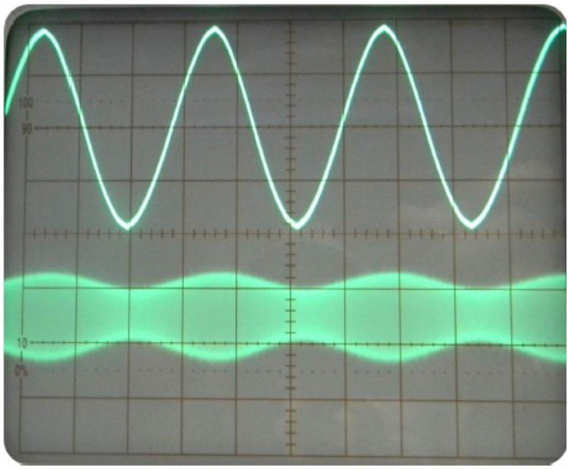
Step 14 On the Scientech 2201 module, turn the volume pot (in the audio amplifier block) clockwise, until you can hear the tone of the audio oscillator's output signal, from the loudspeaker on the board.

Step 15 On the Scientech 2202 receiver, adjust the volume pot so that the receiver's output can be clearly heard. Then adjust the receiver's tuning dial until the tone generated at the transmitter is also clearly audible at the receiver (this should be when the tuning dial is set to about 55-65 and adjust the receiver's volume pot until the tone is at a comfortable level.

Step 16 Check that you are tuned into the transmitter's output signal, by varying Scientech 2201's frequency pot in the audio oscillator block, and nothing that the tone generated by the receiver changes.
[CH1(Y) – 1V; CH2(X) – 2V Time base – 0.1 mS]

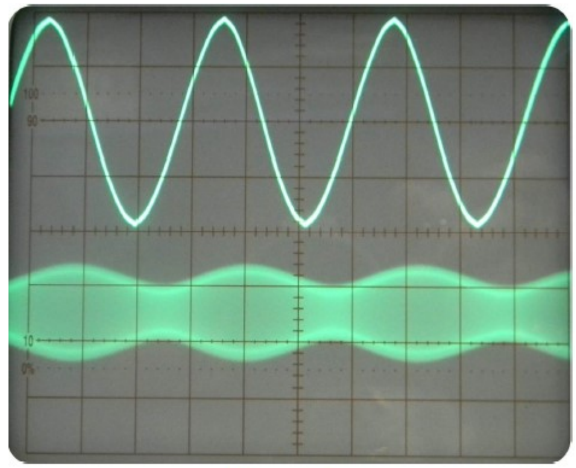
WAVE FORMS OBSERVED:-

[CH1(Y) – 0.5V; CH2(X) – 0.1V Time base – 0.1 mS]



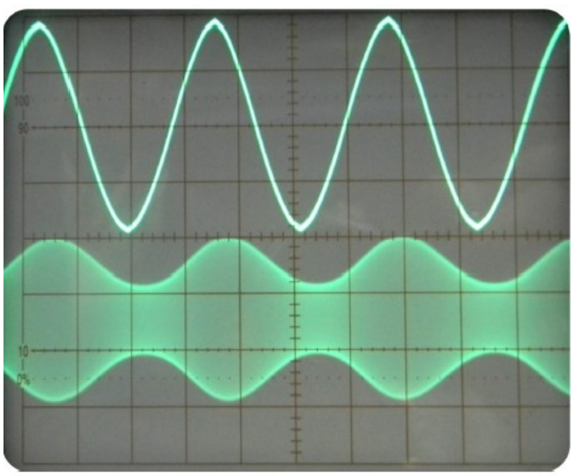
R. F. Amplifier output

[CH1(Y) – 0.5V; CH2(X) – 0.1V Time base – 0.1 mS]



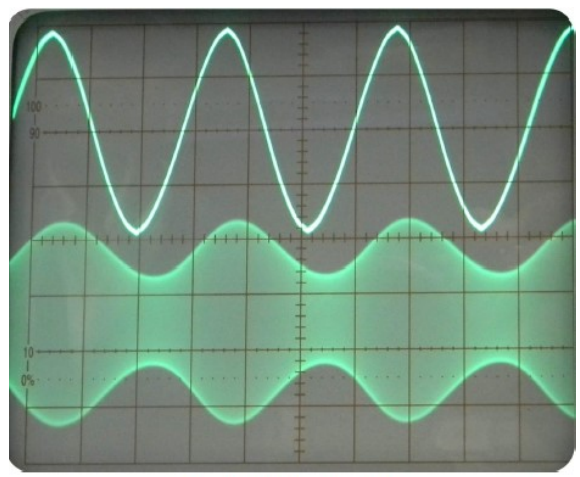
Mixer output

[CH1(Y) – 0.5V; CH2(X) – 0.5V Time base – 0.1 mS]

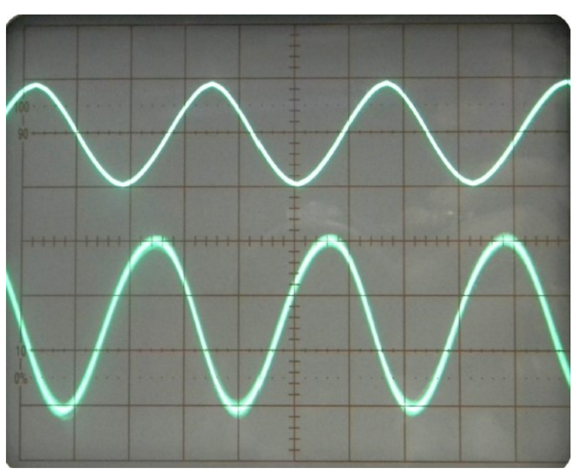


I.F. Amplifier 1 output

[CH1(Y) – 0.5V; CH2(X) – 0.5V Time base – 0.1 mS]



I.F. Amplifier 2 output



[CH1(Y) – 0.5V; CH2(X) – 0.2V Time base – 0.1 mS] [CH1(Y) – 1V; CH2(X) – 2V Time base – 0.1 mS]

TABULATION:-

DETECTED SIGNAL:

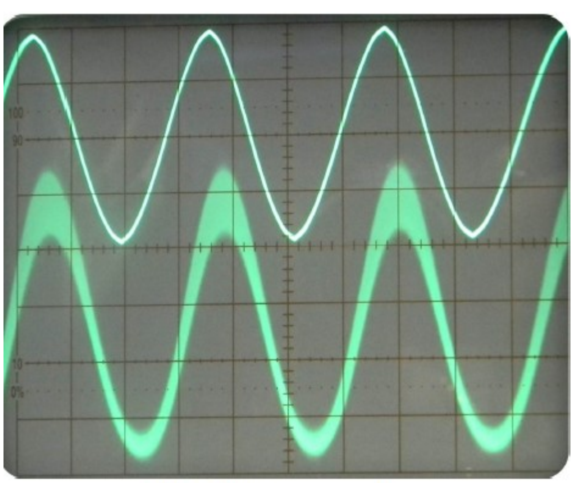
Amplitude (V)	Time period (ms)	Frequency (kHz)

RESULT:-

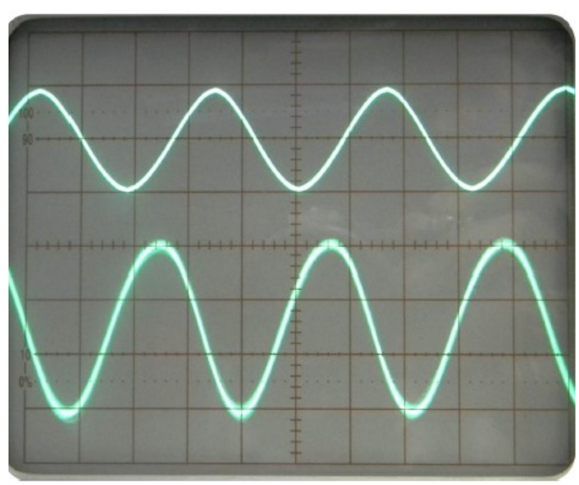
Thus the characteristics of AM Receiver is studied and the waveforms are observed and plotted. The phase difference between Message signal and demodulated signal are not same.

PRECAUTIONS:-

6. Do not use open ended wires for connecting to 230 V power supply.
7. Before connecting the power supply plug into socket, ensure power supply should be switched off.
8. Ensure all connections should be tight before switching on the power supply.
9. Take the reading carefully.
10. Power supply should be switched off after completion of experiment.



Diode Detector output



Audio Amplifier output

QUIZ/ANSWERS:-

.

Q.3. When does over-modulation occur?

Q.4. What is the condition for greatest output power at the transmitter without distortion?

Q.5. What is the main disadvantage of FM over AM?

EXPERIMENT No. 3

OBJECTIVE:- To study and waveform analysis of Frequency Modulation and determine the modulation index of Frequency Modulation.

APPARATUS REQUIRED:- FM Modulation and Demodulation Trainer (ST 2203), CRO 20 MHZ, CRO probes.

OBJECTIVE:-
Frequency modulation is a form of angle modulation in which the amplitude of the modulated carrier is kept constant while its frequency and its rate of change are varied by the modulating signal. In FM the instantaneous angular frequency W_i is varied linearly in accordance with the instantaneous magnitude of base band signal x(t), about an un-modulated carrier frequency (also called as resting frequency) W_c and the rate at which the carrier shifts from its resting point to its non-resting point is determined by the frequency of modulating signal while keeping the amplitude of the carrier wave constant.

$$\text{Carrier signal } C(t) = A \sin(W_c t + \theta_0) = A \sin \Phi \tag{1}$$

where W_c is the frequency of Carrier wave in radians/second and
Φ in radians = Total phase angle of the unmodulated carrier = $(W_c t + \theta_0)$ (2)
In FM while the amplitude A remains constant, instantaneous value of Φ changes.
If $W_i(t)$ = Instantaneous value of angular velocity,
and Φ_i = Instantaneous phase angle of FM wave,

$$\text{then } W_i(t) = d\Phi_i / dt \tag{3}$$

$$\text{and } \Phi_i = \int W_i(t)\, dt \tag{4}$$

Therefore FM wave can be represented as $S(t) = A \sin \Phi_i$ (5)
Modulating voltage Signal = X(t) volts (6)
Then instantaneous angular frequency of an FM signal is given by

$$d\Phi_i / dt = W_i(t) = W_c + K_f X(t) \tag{7}$$

where K_f = Constant of proportionality = frequency sensitivity of the modulator in Hertz per volt
Therefore Frequency variation = $|K_f X(t)|$ (8)
Since the value of W_c is assumed to be fixed,

$$\Phi_i = \int W_i(t)\,dt = \int [\,W_c + K_f X(t)]\,dt = W_c t + K_f \int X(t)\,dt \qquad (9)$$

Frequency Deviation- It is the amount by which carrier frequency is varied from its unmodulated value and it is same as frequency variation.

$$\text{Max Frequency deviation } \Delta W = \left| K_f X(t) \right| \max \qquad (10)$$

Very often we write $\Delta W = \delta$;

Maximum allowed deviation = 75 kHz

Frequency Modulation Index m_f - It is the ratio of frequency deviation ΔW in rad/sec to the angular frequency of modulating signal W_m or frequency deviation in Hertz/sec to the modulating frequency in Hertz/sec.

Thus mf = $\Delta W / W_m = \delta / W_m$ if δ is given in rad /Sec (11)

If δ is given in Hertz/Sec then $m_f = \delta / f_m$ (12)

Mathematical expression for FM wave

$$S(t) = A\text{Sin}\,\Phi_i = A\text{Sin}\,[W_c t + K_f \int X(t)\,dt] \qquad (13)$$

For Single tone FM

$$X(t) = Vm\text{Cos}\,W_c t \qquad (14)$$

Thus $\Phi_i = W_c t + K_f \int V_m \text{Cos}\,W_m t\,dt = W_c t + K_f V_m \text{Sin}\,W_m t$

$$W_m = W_c t + \Delta W\,\text{Sin}\,W_m t = W_c t + m_f\,\text{Sin}\,W_m t$$

Thus $S(t) = A\,\text{Sin}\,[W_c t + K_f \int X(t)\,dt]$

$$= A\,\text{Sin}\,[W_c t + m_f\,\text{Sin}\,W_m t] \qquad (15)$$

Deviation Ratio

It is the ratio of deviation in carrier frequency to the maximum modulating frequency.

In single tone FM, modulation index and the deviation ratio will be the same. If the modulating signal (AF) is 15 kHz at a certain amplitude and the carrier shift is 75kHZ, the transmitter will produce eight (8) significant sidebands as shown in the table above. The corresponding deviation ratio / modulation index is known as Maximum Deviation Ratio.

However in multi tone FM, the amplitude of highest frequency component may not necessarily be maximum. Modulation index will be different for each signal frequency component. The deviation ratio in this case will not be equal to any particular modulation index.

Frequency Spectrum

Analysis of equation (15) which is a sine function of another sine function shows:

$S(t) = A\{ J_0.(m_f) \sin W_ct + J_1(m_f)\{\sin(W_ct+ W_mt) + \sin(W_ct - W_mt)\}$
$+J_2(m_f)\{\sin(W_ct+2W_mt) + \sin(W_ct - 2W_mt)\}$
$+J3(m_f)\{\sin(W_ct+3W_mt) + \sin(W_ct - 3W_mt)\}$
$+J_4(m_f)\{\sin(W_ct+4W_mt) + \sin(W_ct - 4W_mt)]+\ldots\ldots\ldots\ ]$

The output consists of a carrier and an apparently infinite number of pairs of side bands having an amplitude coefficient J_n (m_f), which is a Bessel function of mf and of the order n denoted by the subscript. Values of these coefficients are available readily in table form as well as in graphic form as shown below.

Analysis of FM waveforms Wave forms of carrier, modulating signal, modulated signal as well as graphical form of plot of J_n (m_f) versus values of mf are shown below. It can be seen that:

1. Unlike AM, FM output contains carrier component of frequency fc as well as infinite number of side bands separated from the carrier frequency by f_m, $2f_c$, $3f_c$,…….and thus have a recurrence frequency of fm.

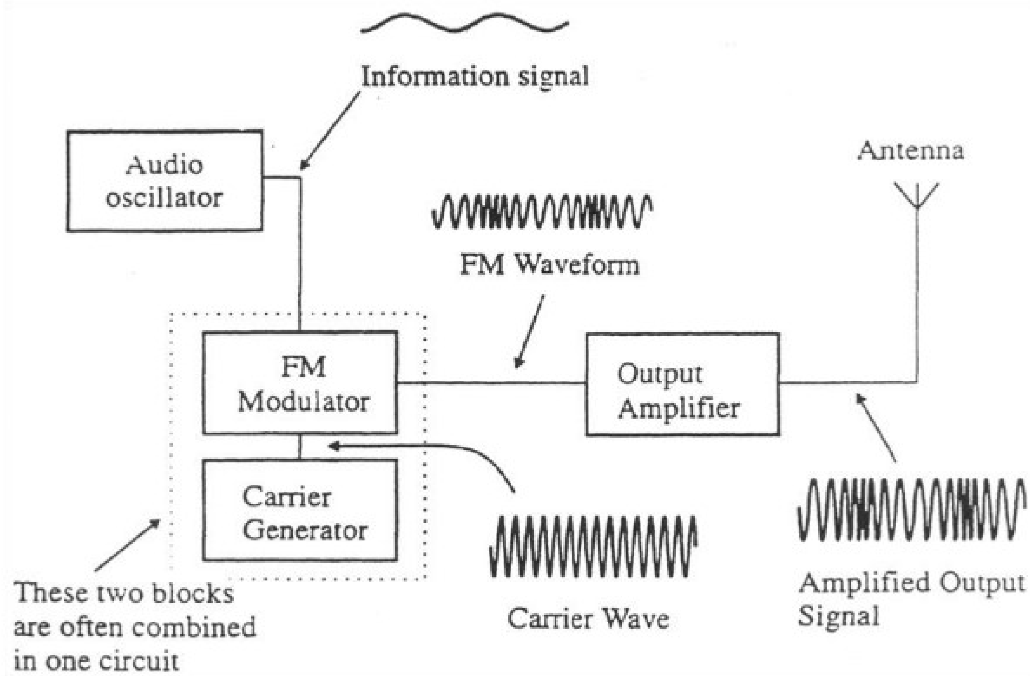
2. The values of each J_n coefficient, which represent the amplitude of a pair of side bands, fluctuates on either side of zero, gradually diminishes with increasing value of mf like damped oscillations. The values of J_n coefficients also eventually decrease, but only past increased value of n. As the value of mf increases, the value of J_0 decreases from 1 and the values of J_1 to J_n increases from 0 and fluctuate around mean value of 0.

3. The modulation index determines how many side band components have significant values.

4. Unlike AM, in FM, while the amplitude of modulated signal remains constant, the value of the carrier component decreases with increase in mf like a damped oscillation. It means that while the total transmitted power remains constant in FM, the number side bands of significant amplitude (and therefore the effective band width) increase with increase in mf .This increases the immunity to noise in FM unlike AM.

BLOCK DIAGRAM:-

The audio oscillator supplies the information signal and could, if we wish, be replaced by a microphone and AF amplifier to provide speech and music instead of the sine wave signals that we are using with ST2203. The FM modulator is used to combine the carrier wave and the information signal in much the same way as in the AM transmitter. The only difference in this case is that the generation of the carrier wave and the modulation process is carried out in the same block. It is not necessary to have the two processes in same block, but in our case, it is. The output amplifier increases the power in the signal before it is applied to the antenna for transmission just as it did in the corresponding block in the FM transmitter.



<u>Controlling the VCO:</u> To see how the VCO is actually controlled, let us assume that it is running at the same frequency as an un-modulated input signal. The input signal is converted into a square wave and, together with the VCO output, forms the two inputs to an Exclusive – OR gate. Remember that the Exclusive - OR gate provides an output whenever the two inputs are different in value and zero output whenever they are the same. The provided an output from the Exclusive -OR gate with an on-off ratio of unity and an average voltage at the output of half of the peak value. Now let us assume that the FM signal at the input decreases in frequency. The period of the 'squared up' FM signal increases and the mean voltage level from the Exclusive -OR gate decreases. The mean voltage level is both the demodulated output and the control voltage for the VCO. The VCO frequency will decrease until its frequency matches the incoming FM signal.

PROCEDURE:-

Step 1 S Ensure that the following initial conditions exist on the **ST2203** board.
(a) All switched faults off.
(b) Amplitude pot (in mixer amplifier block) in fully clockwise position.
(c) VCO switch in 'ON' position.

Step 2 Turn the audio oscillator block's amplitude pot to its fully clockwise position, and examine the block's output t.p.1 on an oscilloscope. This is the audio frequency sine wave, which will be used as our modulating signal. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency pot.

Step 3 Connect the output of audio oscillator to VCO section’s MOD In socket.

Step 4 Turn ON the power supply.

Step 5 Observe the modulating signal and modulated output at the VCO's MOD OUT socket by using CRO.
Step 6 Calculate mf = δ / fm.
Step 7 Vary the modulating frequency keeping carrier frequency constant and repeat steps 3 & 4.
Step 8 Vary the carrier frequency keeping modulator frequency constant and repeat steps 3 & 4.
Step 9 Tabulate the results.

OBSERVATION TABLE:-

1.				
2.				
3.				

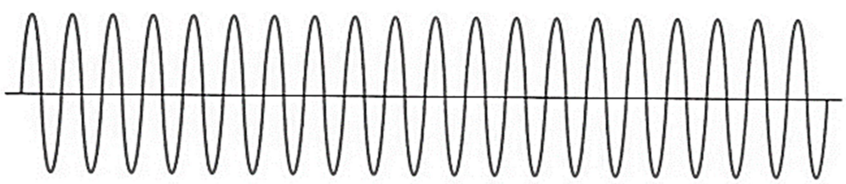
SAMPLE CALCULATION:-

$$m_f = \delta / f_m$$

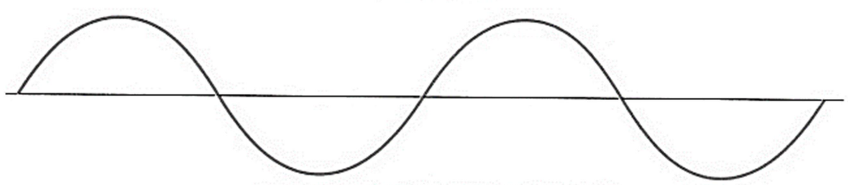
$$= 2 \times 8.3 \times 10^3 / 1000$$

$$= 16.6$$

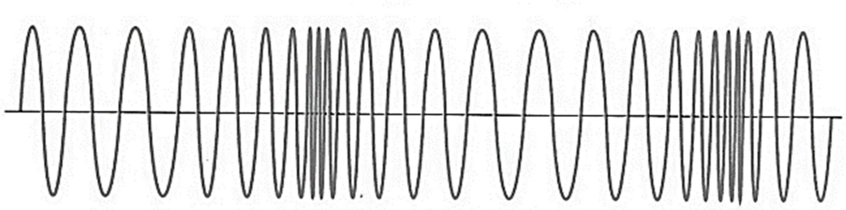
WAVE FORMS OBSERVED:-



Carrier Signal



Modulating Sin Wave Signal



Frequency Modulated Signal

RESULT:-

Frequency modulated wave using VCO is obtained on CRO and mf is calculated.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1. What is FM?

Ans. Frequency modulation (FM) is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave.

Q.2. How many types of FM are there? Write their names.

Ans. There two types of FM i.e. narrow band FM and wideband FM.

Q.3. What frequency deviation in FM?

Ans. The maximum change in instantaneous frequency from the average is known as frequency deviation.

Q.4. Which is the useful parameter for determination of bandwidth?

Ans. Frequency deviation is the useful parameter for determination of bandwidth.

Q.5. How many sidebands are there in FM?

Ans. Theoretically, number sidebands in FM are infinite.

Q.6. Which sidebands are ignored in FM?

Ans. The sidebands with small amplitude are ignored in FM.

Q.7. Which are significant sidebands?

Ans. The sidebands having considerable amplitudes i.e. more than or equal to 1% of the carrier amplitude are known as significant sidebands.

Q.8. What is the indirect method of FM generation?

Ans. Armstrong method.

Q.9. What is the direct method of FM generation?

Ans. The parameter variation method.

Q.10. What is VCO?

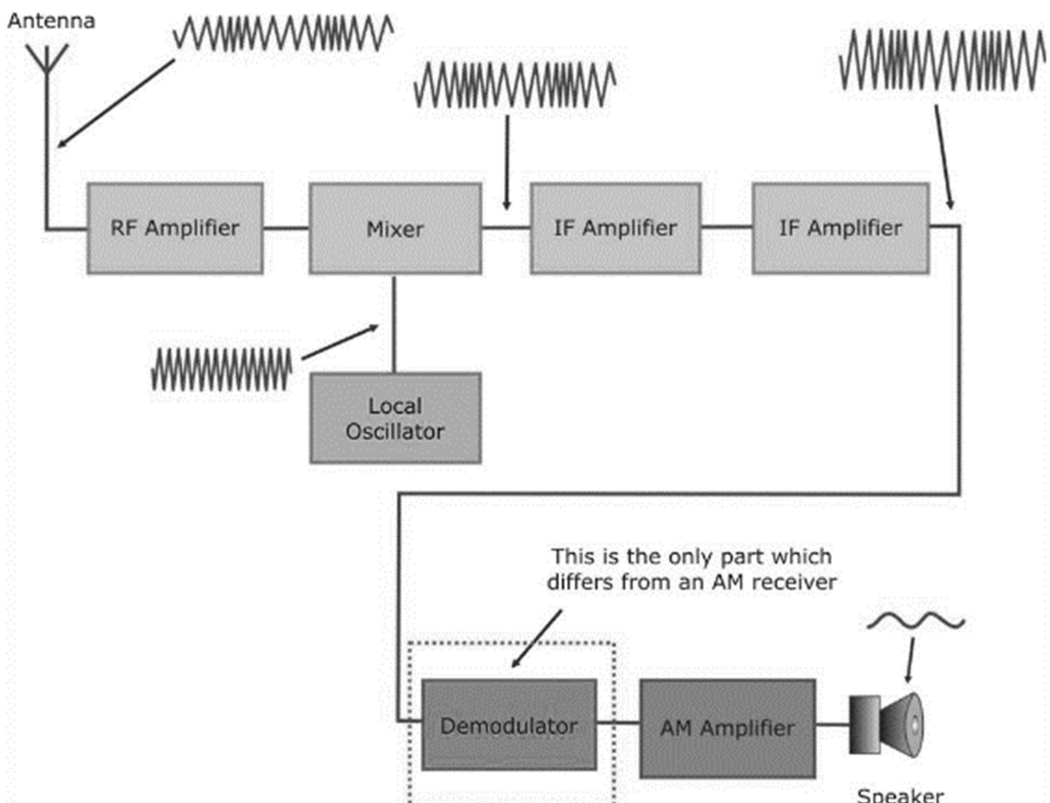
Ans. VCO stands for voltage controlled oscillator whose frequency is controlled by modulating voltage.

EXPERIMENT No. 4

OBJECTIVE:- To study and waveform analysis of Frequency Demodulation by any method.

APPARATUS REQUIRED:- FM Modulation and Demodulation Trainer (ST 2203), CRO 20 MHZ, CRO probes.

OBJECTIVE:-



FM Receiver

A FM receiver is very similar to an AM receiver. The most significant change is that the demodulator must now extract the information signal from a frequency rather than amplitude modulated wave. The basic requirement of any FM demodulator is therefore to convert frequency changes into changes in voltage, with the minimum amount of distortion. To achieve this, it should ideally have a linear voltage/frequency characteristic. A 'demodulator' can also be called a 'discriminator' or a 'detector'.

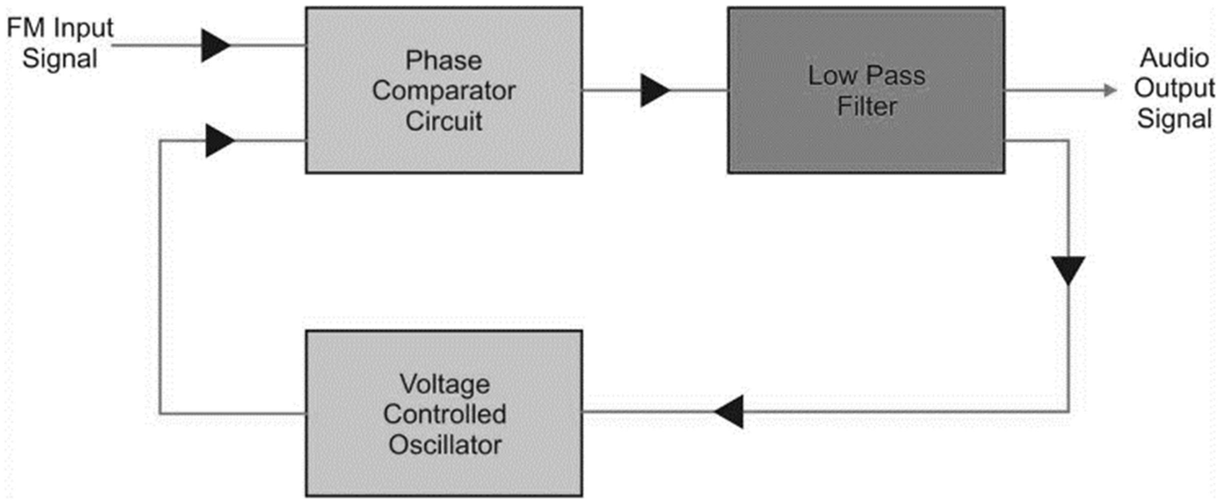


Figure 2 PLL FM demodulator

Phase Lock Loop Detector: This is a demodulator that employs a phase comparator circuit. It is a very good demodulator and has the advantage that it is available, as a self-contained integrated circuit so there is no setting up required. You plug it in and it works. For these reasons, it is often used in commercial broadcast receivers. It has very low levels of distortion and is almost immune from external noise signals and provides very low levels of distortion. Altogether it is a very nice circuit. The overall action of the circuit may, at first, seem rather pointless. As we can see in figure, there is a voltage-controlled oscillator (VCO). The DC output voltage from the output of the low pass filters controls the frequency of this oscillator. Now this DC voltage keeps the oscillator running at the same frequency as the original input signal and 90° out of phase. And if we did, then why not just add a phase shifting circuit at the input to give the 90° phase shift? The answer can be seen by imagining what happens when the input frequency changes - as it would with a FM signal. If the input frequency increases and decreases, the VCO frequency is made to follow it. To do this, the input control voltage must increase and decrease. These change of DC voltage level that forms the demodulated signal. The AM signal then passes through a signal buffer to prevent any loading effects from disturbing the VCO and then through an audio amplifier if necessary. The frequency response is highly linear.

Foster Seeley Detector: The foster Seeley circuit is shown in fig. 3. At first glance, it looks rather complicated but it becomes simpler if we consider it a bit at a time. When the input signal is un-modulated: We will start by building up the circuit a little at a time. To do this, we can ignore many of the companies we may recognize immediately that it consist of two envelope detectors like half wave rectifiers are fed from the center-tapped coil L2. With reference to the center-tap, the two voltages V1 and V2 are in anti-phase as shown by the arrows. The output voltage would be zero volts since the capacitor voltages are in anti-phase and are equal in

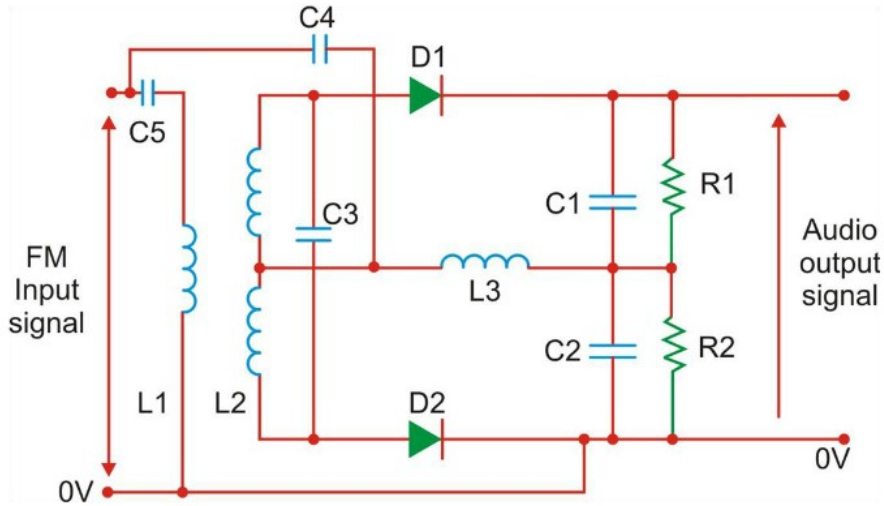


Figure 3 Foster –Seelay Detector

magnitude. After adding two capacitors: The next step is to add two capacitors and see their effect on the phase of the signals. L1 and L2 are magnetically tightly coupled and by adding C3 across the centre-tapped coil, they will form a parallel tuned circuit with a resonance frequency equal to the un-modulated carrier frequency. Capacitor C5 will shift the phase of the input signal by 90° with reference to the voltage across L1 and L2. The voltages are shown as V_a and V_b in the phasor diagram given in figure 39. Using the input signal V_{fm} as the reference, the phasor diagrams now look the way shown in figure 3. C4 is not important. It is only a DC blocking capacitor and has negligible impedance at the frequencies being used. But what it does do is to supply a copy of the incoming signal across L3. The entire incoming signal is dropped across L3 because C1 and C2 also have negligible impedance. If we return to the envelope detector section, we now have two voltages being applied to each diode. One is V1 or V2 and the other is the new voltage across L3, which is equal to Vfm. When the input Frequency changes: If the input frequency increased above its un-modulated value, the phasor of Va would fall below 90° due to the parallel tuned circuit becoming increasingly capacitive. This would result in a larger total voltage being applied across D1 and a reduced voltage across D2. Since the capacitor C1 would now charge to a higher voltage, the final output from the circuit would be a positive voltage. Conversely, if the frequency of the FM input signal decreased below the unmodulated value, the phase shift due to capacitor C5 increases above 90° as the parallel tuned circuit becomes slightly inductive. This causes the voltage across diode D2 to increase and the final output from the demodulator becomes negative. The effect of noise is to change the amplitude of the incoming FM signal resulting in a proportional increase and decrease in the amplitude of diode voltages VD1 and VD2 and the difference in voltage is the demodulated output, the circuit is susceptible to noise interference and should be preceded by a noise limiter circuit.

BLOCK DIAGRAM:-

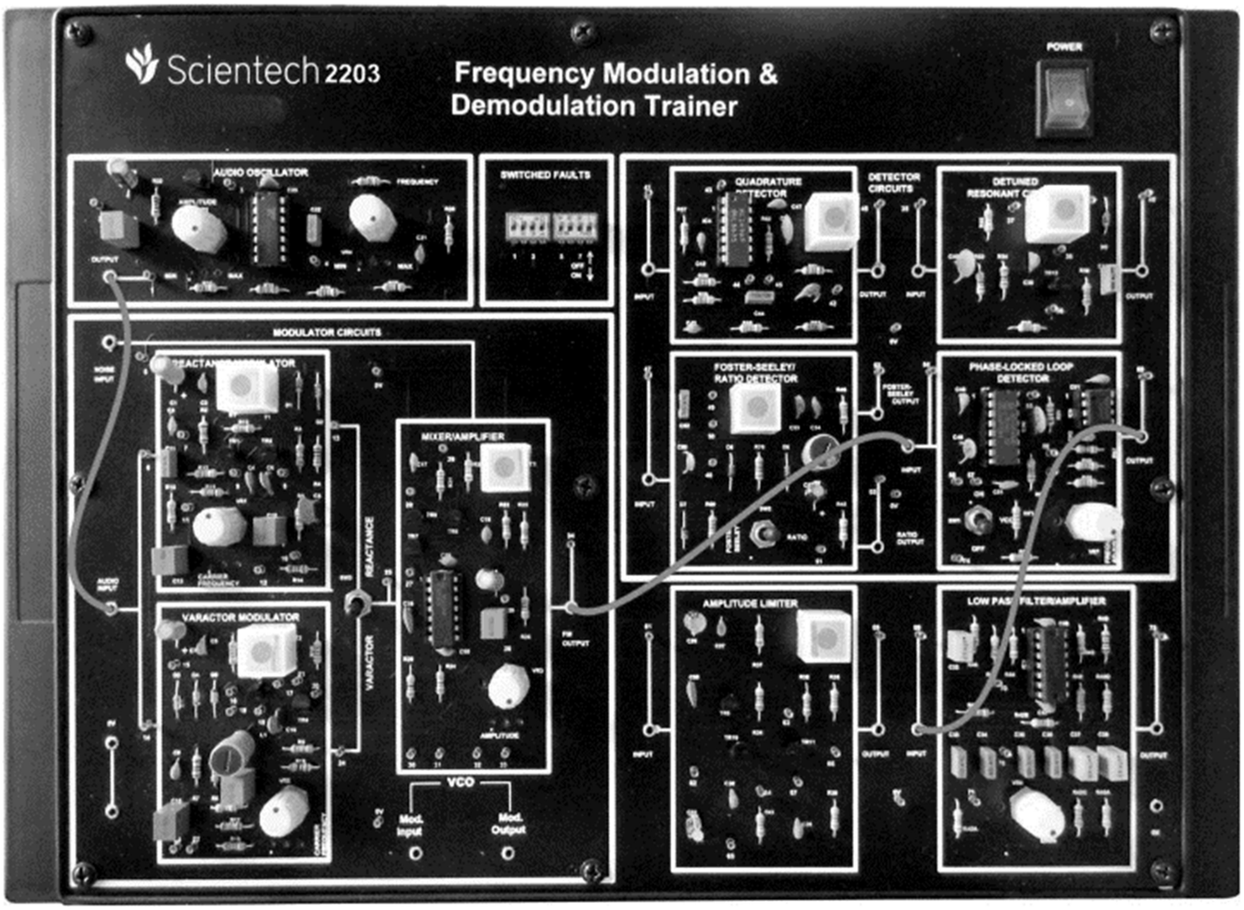


Figure 4 Connections for FM Demodulation using PLL

PROCEDURE:-

FM Detection using PLL:

Step 1 Ensure that the following initial conditions exist on the **ST2203** module:

(a) All switched faults off.

(b) Audio amplifier block's amplitude pot in fully clockwise (MAX) position.

(c) Audio amplifier block's frequency pot in fully counter clockwise. Ensure that the following initial conditions exist on the ST2203 clockwise (MIN) position.

(d) Amplitude pot (in the mixer/amplifier block) in fully clockwise position.

(e) VCO switch (in phase-locked loop detector block) in ON position.

Step 2 Make the connections shown in figure 4.

Step 3 Turn on power to the **ST2203** module.

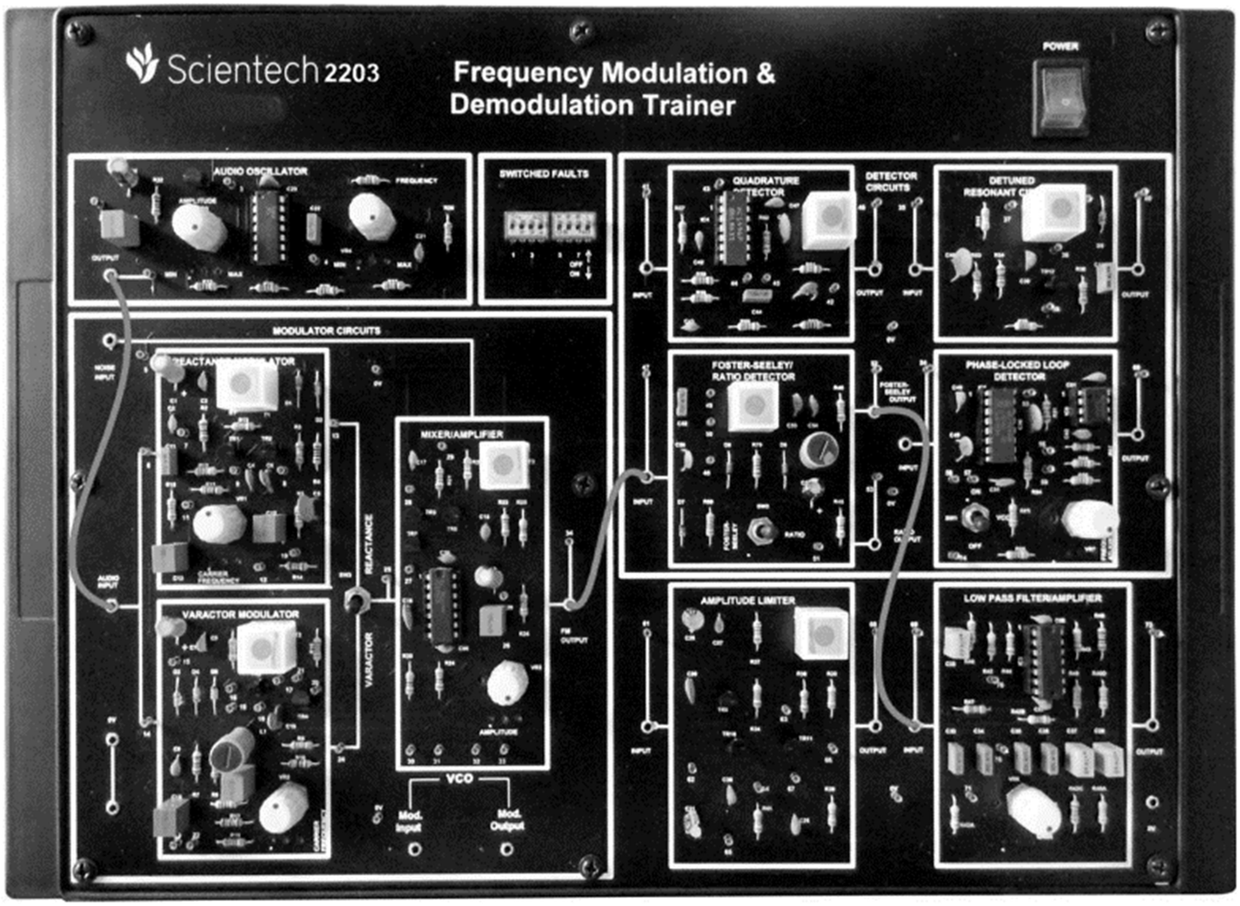


Figure 5 Connections for FM Demodulation using Foster-Seelay Detector

Step 4 Now monitor the audio input signal to the varactor modulator block (at t.p.14) together with the output from the phase-locked loop detector block (at t.p.60), triggering the oscilloscope in t.p.14. The signal at t.p.68 should contain three components:

- A positive D.C. offset voltage.
- A sine wave at the same frequency as the audio signal at t.p.14.
- A high - frequency ripple component.

Step 5 The low pass filter/amplifier block strongly attenuates the high-frequency ripple component at the detector's output and also blocks the D.C. offset voltage. Consequently the signal at the output of the low- pass filter/amplifier block (at t.p.73) should be very closely resemble the original audio making signal, if not then slowly adjust the freq. adjust pot of PLL block.

Step 6 Adjust the audio oscillator block's amplitude and frequency pots, and compare the original audio signal with the final demodulated signal.

FM Detection using Foster-Seelay Detector:

Step 1 Ensure that the following initial conditions exist on the **ST2203** module:

(a) All switched faults OFF;

(b) Audio amplifier block's amplitude pot in fully clockwise (MAX) position.

(c) Audio amplifier block's frequency pot in fully counter-clockwise (MIN) position.

(d) Amplitude pot (in the mixer/amplifier block) in fully clockwise position.

(e) VCO switch (in phase-locked loop detector block) in OFF position.

Step 2 Make connection as shown in figure 5.

Step 3 Turn on power to the ST2203 module.

Step 4 We will now investigate the operation of the foster-Seeley detector on the ST2203 module. In the Foster-Seeley / ratio detector block, select the Foster-Seeley detector by putting the switch in the Foster-Seeley position.

Step 5 Initially, we will use the varactor modulator to generate our FM signal, since this is the more linear of the two modulators, as fast as its frequency/voltage characteristic is concerned. To select the varactor modulator, put the reactance/ varactor switch in the varactor position. Ensure that the varactor modulator's carrier frequency pot is in the midway position.

Step 6 The audio oscillator's output signal (which appears at t.p.1) is now being used by the varactor modulator, to frequency-modulate a 455Khz carrier sine wave. As we saw earlier, this FM waveform appears at the FM output socket from the mixer/amplifier block. You will probably need to have an X-expansion control on your oscilloscope.

Step 7 Now monitor the audio input signal to the varactor modulator block (at t.p. 14) together with the foster-seeley output from the foster-seeley/ratio detector block (at t.p. 52), triggering the oscilloscope on t.p. 14. The signal at t.p. 52 should contain two components:

- A sine wave at the same frequency as the audio signal at t.p. 14.
- A High frequency ripple component of small amplitude.

Step 8 The low-pass filter/amplifier strongly attenuates this high-frequency ripple component, and blocks any small D.C. offset voltage that might exist at the detector's output. Consequently, the signal at the output of the low-pass filter/ amplifier block (at t.p. 73) should very closely resemble the original audio modulating signal.

Step 9 Monitor the audio input to the varactor modulator (at t.p. 14) and the output of the low pass filter / amplifier block (at t.p. 73) and adjust the gain pot (in the low pass filter/ amplifier block) until the amplitudes of the monitored audio waveforms are the same.

Step 10 Adjust the audio oscillator block's amplitude and frequency pots, and compare the original audio signal with the final demodulated signal.

OBSERVATION TABLE:-

1.				

2.				

RESULT:-
FM signal is being demodulated by using PLL and Foster-Seelay Method.

PRECAUTIONS:-
1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1. What is FM demodulation?

Ans. FM demodulator or detector circuit takes in frequency modulated RF signals and takes the modulation from the signal to output only the modulation that had been applied at the transmitter.

Q.2. What is frequency discriminator circuit?

Ans. A frequency discriminator is defined as a converter of frequency changes into amplitude changes. At low frequencies this converter can be implemented with the use of a differentiator or even the rising half of a tuned circuit

Q.3. What is the function of FM discriminator?

Ans. The foster-seeley discriminator is also known as the phase-shift discriminator. It uses a double-tuned RF transformer to convert frequency variations in the received FM signal to amplitude variations.

Q.4. What is the use of demodulator?

Ans. A demodulator is a circuit that is used in amplitude modulation and frequency modulation receivers in order to separate the information that was modulated onto the carrier from the carrier itself.

Q.5. Why do carriers have high frequency?

Ans. High frequency carrier waves increase the power that is radiated by the antenna to enhance the transmission range.

Q.6. Why do higher frequencies carry more data?

Ans. Higher-frequency transmissions have more bandwidth than lower-frequency transmissions, which means higher-frequency transmissions can send substantially more data between devices in less time.

Q.7. What is the advantage of high frequency?

Ans. The main advantage of high frequency signals is that the signal may be transmitted over very long distances and thus dissipates very less power. The antenna height required for transmission is reduced at higher frequencies.

Q.8. What is modulation factor on FM transmission?

Ans. The FM modulation index is equal to the ratio of the frequency deviation to the modulating frequency.

Q.9. What are the applications of FM?

Ans. Applications of frequency modulation include in FM radio broadcasting, radar, seismic prospecting, telemetry, & observing infants for seizure through EEG, music synthesis, two-way radio systems, magnetic tape recording systems, video broadcast systems, etc.

Q.10. What is the Carson's rule?

Ans. Carson's bandwidth rule defines the approximate bandwidth requirements of communications system components for a carrier signal that is frequency modulated by a continuous or broad spectrum of frequencies rather than a single frequency.

EXPERIMENT No. 5

OBJECTIVE:- To study Amplitude Shift Keying (ASK) Modulation.

APPARATUS REQUIRED:- ASK Modulation Kit, power supply, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

The binary ASK System was one of the earliest forms of digital modulation used in wireless telegraphy. This simplest form of digital modulation is no longer used widely in digital communication .Nevertheless it serves as a useful model which helps in understanding certain concepts. In an ASK system, binary symbol 1 is represented by transmitting a sinusoidal carrier wave of fixed amplitude Ac and fixed frequency f_c for the bit duration T_b seconds whereas binary symbol 0 is represented by switching off the carrier for T_b seconds. This signal can be generated by switching off the carrier of a sinusoidal oscillator on and off for the prescribed periods indicated by the modulating pulse train. For this reason the scheme is also known as on-off keying (OOK).

Let the sinusoidal carrier be represented by

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

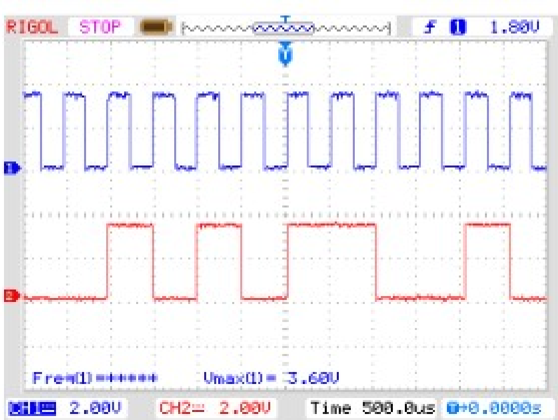
Then, the binary ASK signal can be represented by a wave s(t) given by

$$\begin{aligned} S(t) &= A_c \cos(2\pi f_c t) \quad \text{for symbol 1} \\ &= 0 \qquad\qquad\qquad \text{for symbol 0} \end{aligned}$$

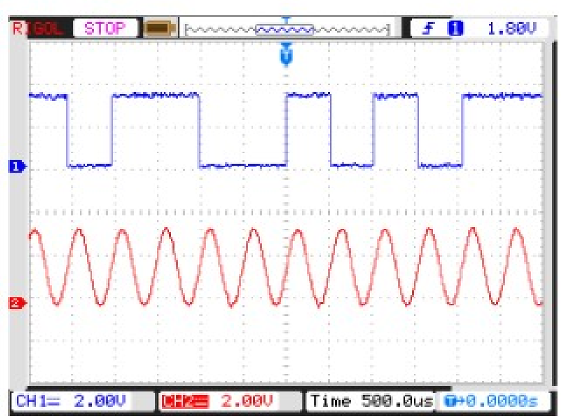
A typical ASK waveform is illustrated in figure for a binary data represented by 8-Bit: “10110010”.

Generation of ASK Signal:

ASK signal can be generated by applying the incoming binary data (represented in unipolar form) and the sinusoidal carrier to the two inputs of a product modulator



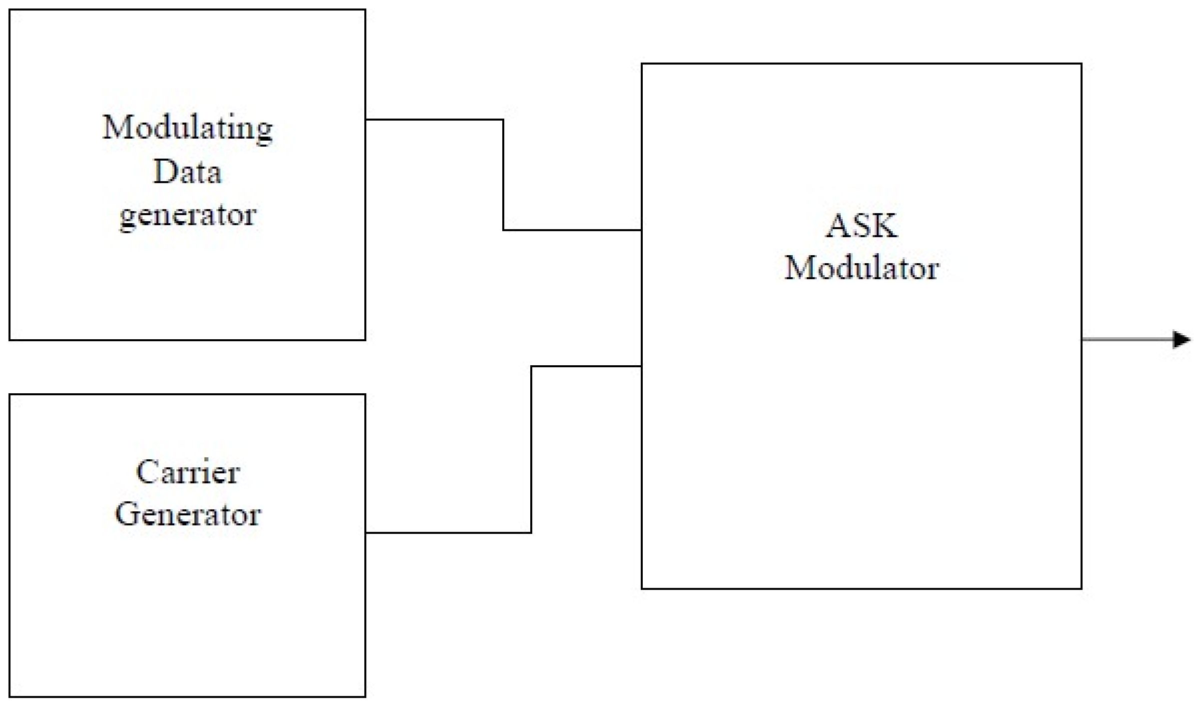
CH1: Input Data Clock (TP1), CH2: Input Data(TP2)



CH1: Input Data (TP3), CH2: Carrier Signal (TP4)

(balanced modulator) The resulting output is the ASK wave. This is illustrated in figure modulation causes a shift of the baseband signal spectrum. The ASK signal which is basically the product of the binary sequence and the carrier signal.

BLOCK DIAGRAM:-

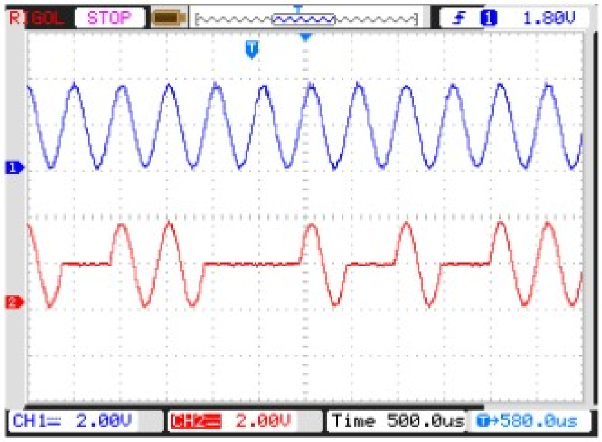


PROCEDURE:-

Step 1 Connect and switch on the Power Supply of Scientech 2807.
Step 2 Select input Data pattern using push button i.e. 8-Bit, 16-Bit, 32-Bit, 64-Bit. And respective LED will glow. Observe the input Data on test point (TP2).
Step 3 Select input data clock using push button i.e. 2 KHz, 4 KHz, 8 KHz, 16 KHz. Observe the change in frequency on test point (TP1).
Step 4 Observe the change in frequency of carrier signal at (TP4).
Step 5 ASK Modulator is by default selection when switch on the Power Supply of Scientech 2807 and LED of (TP3) will glow.
Step 6 Observe the ASK modulator output on (TP5).
Step 7 Observe the Input Data at TP2, Input data clock at TP1, encoded data input at TP3, Carrier signal at TP4, Modulated output at TP5 and data Patterns on TP2.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.



CH1: Carrier signal (TP4),
CH2: ASK Output (TP5)

RESULT:-

ASK output is obtained on CRO.

QUIZ/ANSWERS:-

Q.1. What is keying?

Ans. A method of encoding data by modulating the carrier either by phase or frequency.

Q.2. What is Digital Modulation?

Ans. In Digital Modulation analog carrier waveform is modulated by digital bitstream in this scheme analog signal is first converted into digital data by Analog to Digital conversion technique this digital data then transmitted by transmitter using modulation .

Q.3. List some digital modulation techniques.

Ans. ASK (Amplitude Shift Keying), FSK (frequency Shift Keying), BPSK (Binary Phase Shift Keying), QPSK (Quadrature Phase Shift Keying), MSK (Minimum Shift Keying) ,M-ary ASK etc.

Q.4. What is Carrier Signal?

Ans. A continuous waveform whose properties are modulated with an input signal for the purpose of conveying information upto far distances.

Q.5. What is Encoding and Decoding?

Ans. Encoding is a process of converting series of symbols, characters etc. into digital stream of l's and 0's (Binary form). Decoding is reverse process of encoding in which we decode our original transmitted data from digital stream of l's and 0's.

Q.6. What are the number of symbols available in binary?

Ans. 2.

Q.7. What is RZ and NRZ?

Ans. Return-to-Zero & Non Return-to-Zero.

Q.8. Why encoding is used?

Ans. To represent quantized samples in appropriate digital format.

Q.9. Why decoder is used?

Ans. To convert digital data into discrete sample values.

Q.10. Why channel encoder is used?

Ans. To avoid errors.

EXPERIMENT No. 6

<u>OBJECTIVE</u>:- To study Frequency Shift Keying (FSK) Modulation.

<u>APPARATUS REQUIRED</u>:- Data generator, FSK modulation kit, CRO 20 MHZ, CRO probes.

<u>OBJECTIVE</u>:-

FSK is one of the basic modulation techniques for the transmission of digital data .If the frequency of the sinusoidal carrier is switched depending upon the input digital signal, then it is known as frequency shift keying. As the amplitude remains constant in FSK, so the effect of non-linear ties, noise interference is minimum on digital detection. So FSK is preferred over ASK. Frequency shift keying consists of shifting of frequency of carrier from a mask frequency to a space frequency according to the base band digital signal. Frequency shift keying is identical to modulating an FM carrier with a binary digital signal. In an FSK system, two sinusoidal carrier waves of the same amplitude Ac but different frequencies fc1 and fc2 are used to represent binary symbols 1and 0 respectively. It can be easily verified that binary FSK waveform is a superposition of two binary ASK waveforms , one with a frequency fc1 and other with a frequency fc2. No discrete components appear in the signal spectrum of FSK signal. The main advantage of FSK lies in its easy hardware implementation.

Generation of FSK signal:-

The PSK signal can be generated by applying the incoming binary data to a frequency modulator. To the other input a sinusoidal carrier wave of constant amplitude Ac and frequency fc is applied. As the modulating voltages changes from one level to another, the frequency modulator output changes its frequency in the corresponding fashion.

Detection of FSK signal:-

FSK can be demodulated by using coherent and non-coherent detector. The detector
based on coherent detection requires phase and timing synchronization. Non coherent detection can be done by using envelop detector.

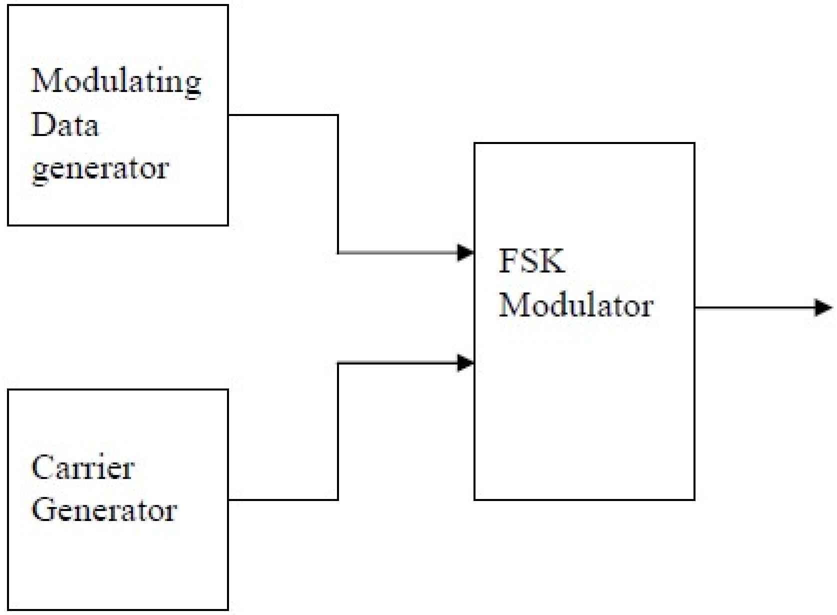
<u>PROCEDURE</u>:-

Step 1 Make the connection according to the circuit diagram.

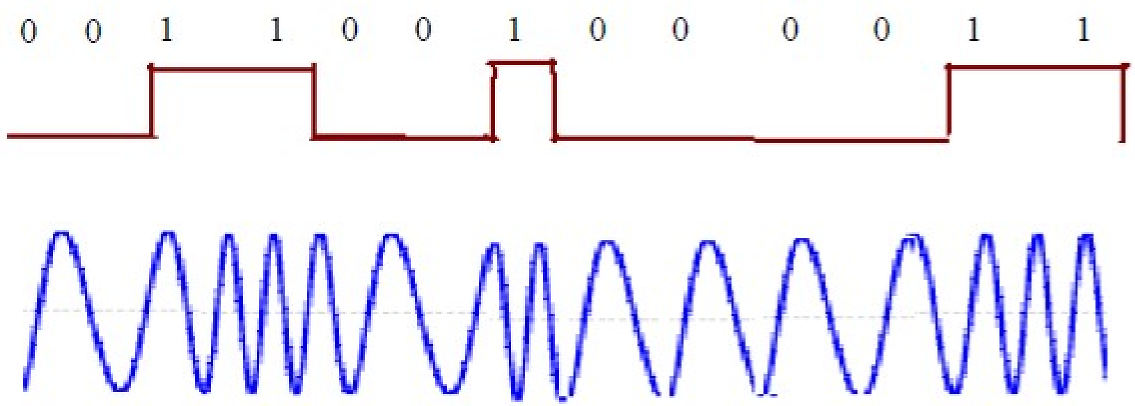
Step 2 Connect the modulator output to CRO.

Step 3 Observe output on CRO.

BLOCK DIAGRAM:-



WAVE FORMS:-



RESULT:-

FSK output is obtained on CRO.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1 What is FSK?

Ans. This is one of the basic modulation techniques for transmission of digital data. The frequency of carrier is switched on or off according to the input digital signal.

Q.2 Why FSK is preferred over ASK?

Ans. Because of constant amplitude of FSK the effect of non-linearity's and noise interference is minimum on signal detection.

Q.3 what are various components of FSK detector?

Ans. Two synchronous detector, differential amplifier, low-pass filter.

Q.4 What is BFSK?

Ans . In BFSK frequency of the carrier is sifted according to the binary symbol keeping the phase of the carrier unaffected.

Q.5 What is the difference between FM and FSK?

Ans. FM is a analog modulation technique where FSK is digital modulation technique.

Q6. How BFSK signal is generated?

Ans. An input signal is processed in two paths each existing of level shifter and product modulator. One path uses directly and other path uses inverter signal. Orthogonal carrier signal are used as the other input for the product modulator. The output of the product modulator are added which generates a BFSK.

Q.7 What is the bandwidth of BFSK?

Ans. 4fb where fb - bandwidth of the input signal.

Q.8 Compare bandwidth of BFSK and BPSK.

Ans. Bandwidth of BFSK= 2(bandwidth of BPSK)

Q.9 What is the disadvantage of BFSK?

Ans. The error rate of BFSK is more as compared to BPSK.

Q.10 How can you detect FSK by non-coherent method?

Ans. BFSK waves may be demodulated coherently using envelop detectors.

EXPERIMENT No. 7

OBJECTIVE:- To study Phase Shift Keying (PSK) Modulation.

APPARATUS REQUIRED:- Experimental kit, power supply, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

PSK:- PSK involves the phase change at the carrier sine wave between 0 to 180 in accordance with the data stream to be transmitted .

PSK modulator is similar to ASK modulator both used balanced modulator to multiply the carrier with balanced modulator signal. The digital signal with applied to modulation input for PSK generation is bipolar i.e. equal positive and negative voltage level.

When the modulating input is positive the out put at modulator is a line wave in phase with the carrier input whereas for positive voltage level, the output of modulator is a sine wave which is switched out of phase by 180 from the carrier input.

Quadrature Phase-shift Keying (QPSK)

QPSK:- in QPSK each pair at consecutive data bit is treated as a two bit code which is switch the phase of the carrier sine wave between one at four phase 90° apart. The four possible combinations at bib it code are 0°, 01, 10, and 11 each code represents either a phase of 45°, 185°, 225°, and 315° lagging, relative to the phase at the original un modulated carrier QPSK offers an advantage over PSK is a no carrier that how each phase represents a two bit code rather than a single bit. This means that either we can charge phase per sec. or the same amount of data can be transmitted with .

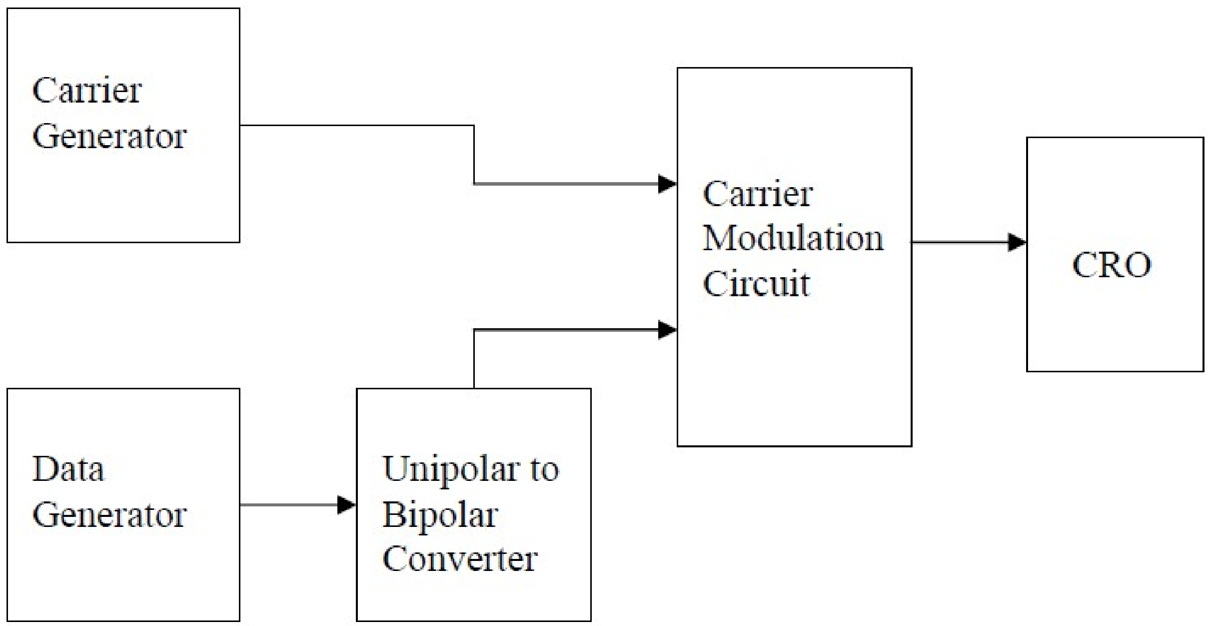
PROCEDURE:-

Step 1 Make the connection according to the circuit diagram.
Step 2 Connect the modulator output to CRO.
Step 3 Observe output on CRO.

PRECAUTIONS:-

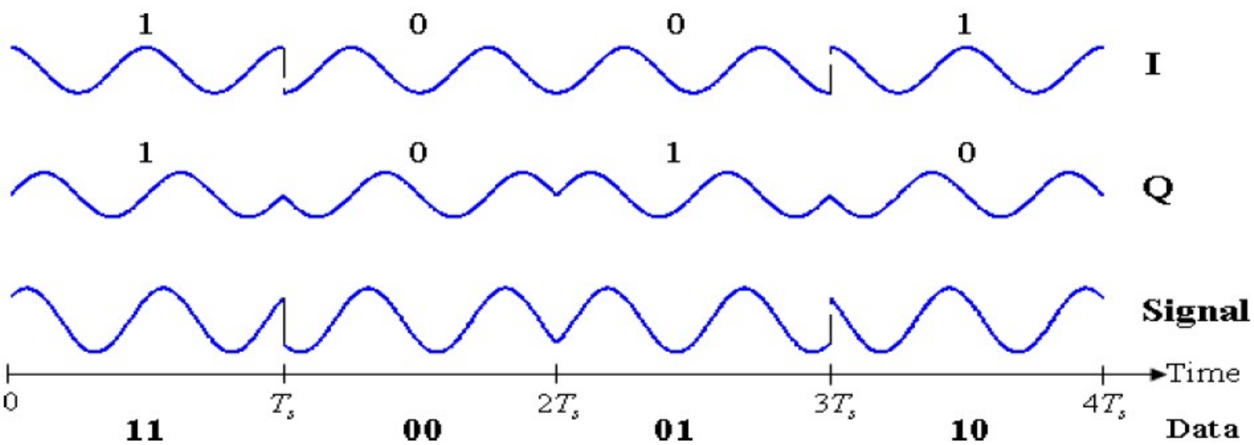
1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

BLOCK DIAGRAM:-



(Block diagram of PSK)

WAVE FORMS OBSERVED:-



RESULT:-

PSK output is obtained on CRO.

QUIZ/ANSWERS:-

Q.1. What are the number of symbols available in binary?

Ans. 2.

Q.2. What is RZ?

Ans. Return-to-Zero.

Q.3. What is NRZ?

Ans. Non Return-to-Zero.

Q.4. Why encoding is used?

Ans. To represent quantized samples in appropriate digital format.

Q.5. What is the unit of data rate?

Ans. Bits/s.

Q.6. Why decoder is used?

Ans. To convert digital data into discrete sample values.

Q.7. What is the advantage of PCM?

Ans. In PCM, S/N ratio is more than DM

Q.8. At which factor bandwidth of PCM depends?

Ans. It depends upon the bit duration i.e. sampling period/total no. of bits.

Q.9. What is no. of bits required to represent a sample of a system of 7 symbols?

Ans. 3

Q.10. Why channel encoder is used?

Ans. To avoid errors.

Q.11. How many types of FM are there? Write their names.

Ans. There two types of FM i.e. narrow band FM and wideband FM.

EXPERIMENT No. 8 & 9

OBJECTIVE:- To study and waveform analysis of Phase Modulation & Demodulation.

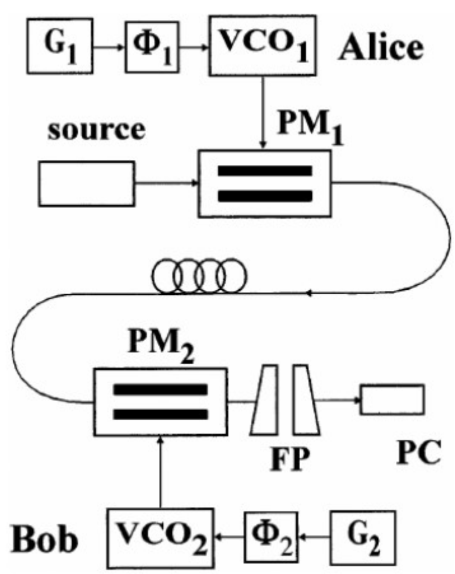
APPARATUS REQUIRED:- FM Modulation and Demodulation Trainer (ST 2203), CRO 20 MHZ, CRO probes.

OBJECTIVE:-
Angle modulation is a technique in which the angle of the carrier is varied with instantaneous values of message signal. Angle Modulation has been divided into two types.

i. Phase Modulation.   ii. Frequency Modulation.

In this phase modulator the carrier can be generated by a quartz oscillator, and so its frequency stabilization is easier. In the circuit used for the exercise, the frequency modulation is generated by a Hartley oscillator, which frequency is determined by a fixed inductance and by capacity (variable) supplied by varicap diodes. Phase modulation is calculated by adding the baseband signal to the argument of a sine or cosine function that represents the carrier. The modulation index makes the phase variations more or less sensitive to the behavior of the baseband signal. The frequency-domain effects of phase modulation are similar to those of frequency modulation.

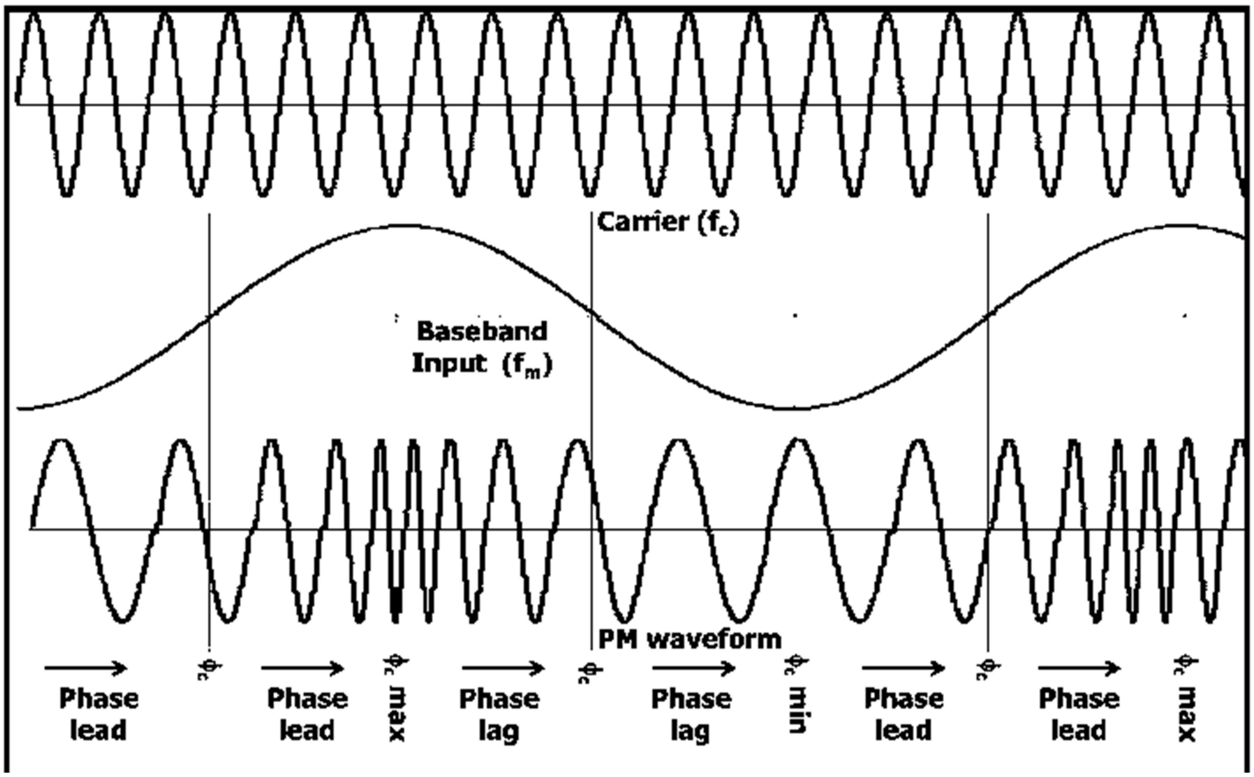
BLOCK DIAGRAM:-



PROCEDURE:-
Step 1 Make the connection according to the circuit diagram.
Step 2 Connect the modulator output to CRO.

Step 3 Observe output on CRO.

WAVE FORMS OBSERVED:-



RESULT:-
Phase modulation output is obtained on CRO.

PRECAUTIONS:-
1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-
Q.1. What are the number of symbols available in binary?

Ans. 2.

Q.2. What is RZ?

Ans. Return-to-Zero.

Q.3. What is NRZ?

Ans. Non Return-to-Zero.

Q.4. Why encoding is used?

Ans. To represent quantized samples in appropriate digital format.
Q.5. What is the unit of data rate?
Ans. Bits/s.
Q.6. Why decoder is used?
Ans. To convert digital data into discrete sample values.
Q.7. What is the advantage of PCM?
Ans. In PCM, S/N ratio is more than DM
Q.8. At which factor bandwidth of PCM depends?
Ans. It depends upon the bit duration i.e. sampling period/total no. of bits.
Q.9. What is no. of bits required to represent a sample of a system of 7 symbols?
Ans. 3
Q.10. Why channel encoder is used?
Ans. To avoid errors.
Q.11. How many types of FM are there? Write their names.
Ans. There two types of FM i.e. narrow band FM and wideband FM.

EXPERIMENT No. 10

OBJECTIVE:- To study Pulse Code Modulation.

APPARATUS REQUIRED:- TDM Pulse Code Modulation Transmitter Trainer (ST 2103) and TDM Pulse Code Modulation Receiver Trainer (ST 2104) CRO 20 MHZ, CRO probes.

OBJECTIVE:-

PCM is a digital process. In this instead of sending a pulse train capable of continuously varying one of the parameters, the PCM generator produces a series of numbers. Each one of these digits, almost always in binary code, represents the approximate amplitude of the signal sample at that instant.

In pulse code modulation, the massage signal is rounded to the nearest of a finite set of allowable values. So that both time and amplitude are discrete form. This allows the massage to be transmitted by means of coded electrical signals. There by distinguishing PCM from all other methods. Modulations with increasing ability of wide band communication channel coupled with the emergence of required device technology. The use of PCM has become a practical reality. The essential operation in the transmitter of PCM system are sampling, quantizing, and encoding. The quantizing and encoding operation are performed in the same circuit called A / D converter. The essential operations in the receiver are regeneration of unpaired signal, decoding and demodulation of train of quantized.

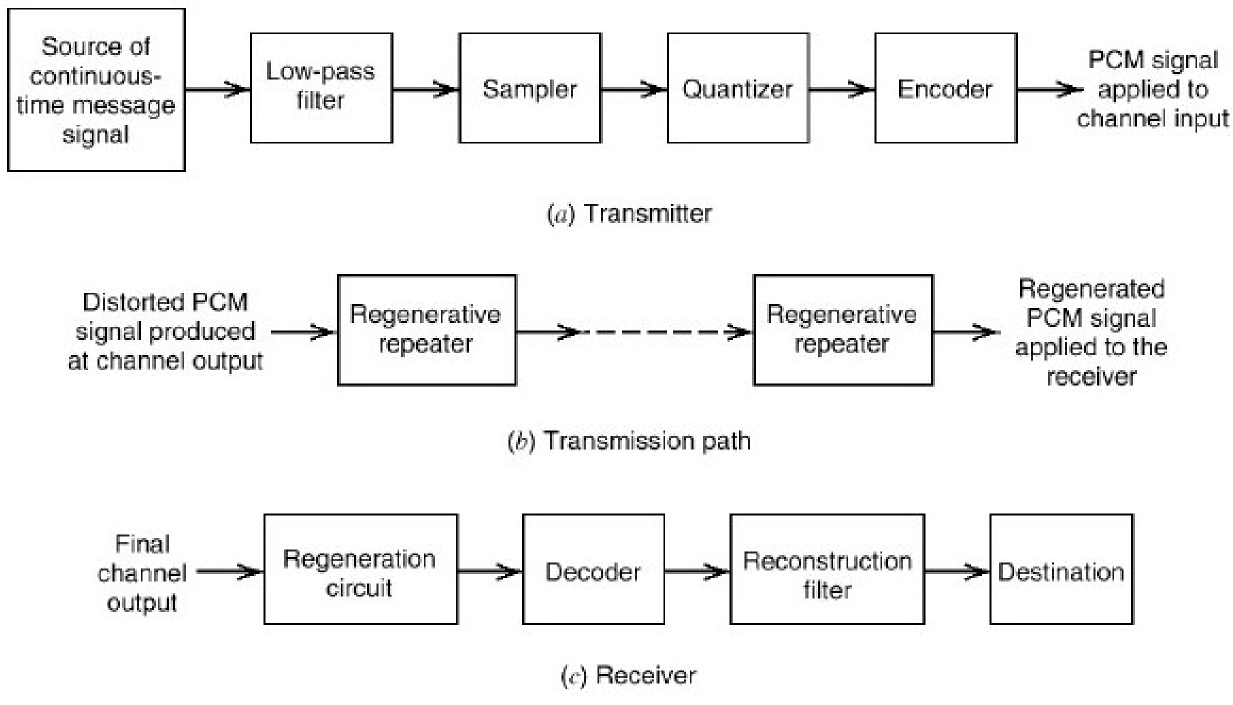
Steps in Pulse Code Modulation:

Sampling: The analog signal is sampled according to the Nyquist criteria. The nyquist criteria states that for faithful reproduction of the band limited signal, the sampling rate must be at least twice the highest frequency component present in the signal. For audio signals the highest frequency component is 3.4 KHz.

Practically, the sampling frequency is kept slightly more than the required rate. In telephony the standard sampling rate is 8 KHz. Sample quantifies the instantaneous value of the analog signal point at sampling point to obtain pulse amplitude output.

Allocation of Binary Codes: Each binary word defines a particular narrow range of amplitude level. The sampled value is then approximated to the nearest amplitude level. The sample is then assigned a code corresponding to the amplitude level, which is then transmitted. This process is called as *Quantization* & it is generally carried out by the A/D converter.

BLOCK DIAGRAM:-



Basic Elements of PCM System

PROCEDURE:-

Step 1 Make the connection according to the block diagram.
Step 2 Observe PCM output on CRO at the PCM OUT tp. on the ST 2103.
Step 3 Note the variations in the digital output as per variations in the value of DC1.
Step 4 Observe the operation of error check codes by putting switches A & B respectively in positions 00, 01, 10 &11.
Step 5 Change input from DC1 to 1kHz and 2 kHz sinusoidal signals and repeat from step 2 to 4.
Step 6 Observe the demodulated PCM output on ST 2104 output point.

RESULT:-

Pulse code modulation and demodulation is studied with error check codes.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1 In which category of PM is PCM?
Ans. Digital Modulation
Q.2 Which noise is occurs in PCM?
Ans. Quantization Noise
Q.3 What is Quantization?
Ans. In PCM, the total amplitude range which is signal may be divided into number of
standard level is called quantization.
Q.4 Which noise is occurs in PCM
Ans. Quantization noise.
Q.5 How analog signal can be encoded in to bits/
Ans. By delta modulation technique
Q.6 What is the advantage of DM over PCM?
Ans. DM needs a simple circuit as compared to PCM.
Q.7 What is the advantage of PCM?
Ans. In PCM, S/N ratio is more than DM
Q.8 At which factor bandwidth of PCM depends?
Ans. It depends upon the bit duration i.e. sampling period/total no. of bits.
Q.9 What is Elastic store?
Ans. A device which can store and reproduce data at different speed is Elastic store.
Q.10 Write down one example of Elastic store.
Ans. Tape-recorder.

EXPERIMENT No. 5

OBJECTIVE:- To study Pulse Amplitude Modulation and Demodulation.

APPARATUS REQUIRED:- Scientech 2110 with Power Supply cord, Oscilloscope with Connecting probe, Connecting cords.

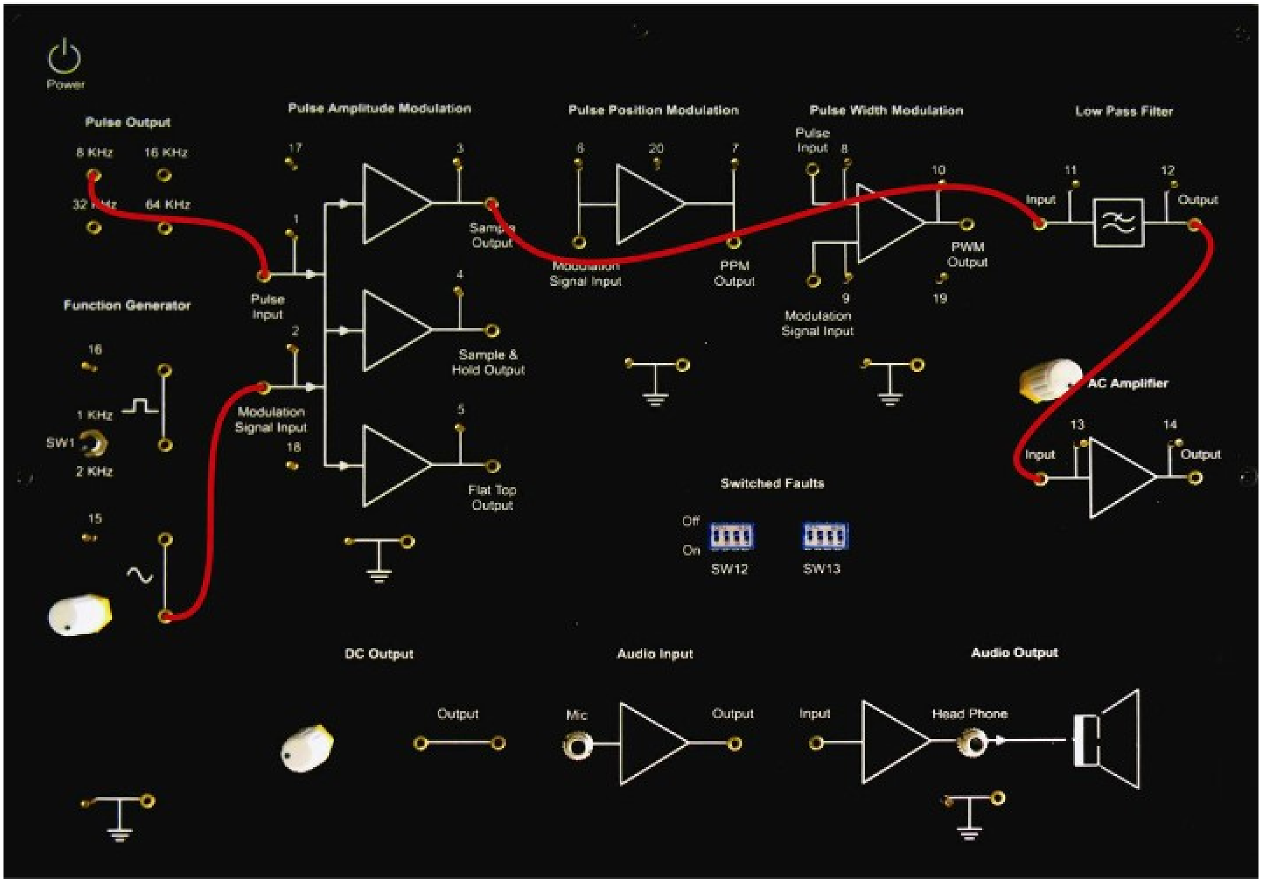
OBJECTIVE:-

Pulse Modulation is a system in which continuous waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times, together with any synchronizing pulses that may be required.

To process signals we use in the real world, such as our voice (Analog) for digital communication, we need to convert analog signals to "Digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert continuous time signal to discrete time signal, a process is used called as Sampling. There are three types of sampling techniques as under:

- Ideal Sampling or Instantaneous Sampling or Impulse Sampling
- Natural Sampling
- Flat top Sampling

The concept of 'Instantaneous' sampling is more of a mathematical abstraction as no practical sampling device can actually generate truly instantaneous samples. In the ADC process an analogue waveform is sampled to form a series of pulses whose amplitude is the amplitude of the sampled waveform at the time the sample was taken. In natural sampling the pulse amplitude takes the shape of the analogue waveform for the period of the sampling pulse. After an analogue waveform is sampled in the Analogue-to-Digital conversion process, the continuous analogue waveform is converted into a series of pulses whose amplitude is equal to the amplitude of the analogue signal at the start of the sampling process. Since the sampled pulses have uniform amplitude, the process is called flat top sampling. Note that due to the flat-top pulses, the spectrum of the sampled signal is distorted. The narrower the pulse width, the less distortion. In Pulse Amplitude Modulation system the amplitude of the pulse is varied in accordance with the instantaneous level of the modulating signal. The Pulse Amplitude Modulation is recovered by a low pass filter.



<u>CONNECTION DIAGRAM:-</u>

<u>PROCEDURE</u>:-

<u>Modulation</u>

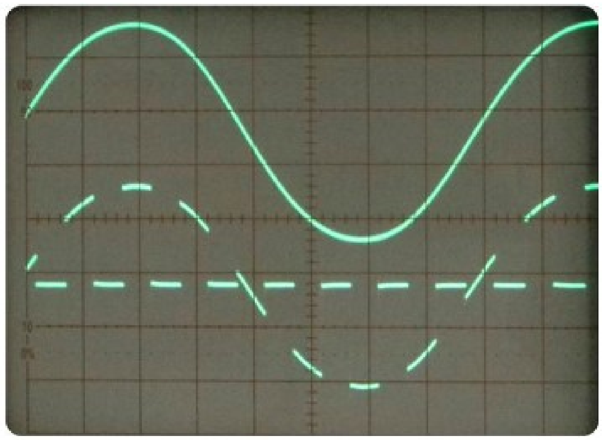
Step 1 Make Connect the circuit as shown in next figure

- Output of sine wave to modulation signal IN in PAM block keeping the switch in 1 KHz position. 8 KHz pulse output to pulse input.
- Output of low pass filter to input of AC Amplifier. Keep the gain pot in AC Amplifier block in anti-clock wise position.

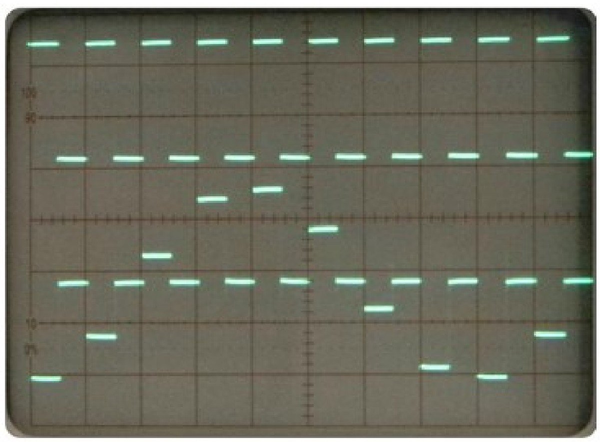
Step 2 Switch 'On' the Power Supply & Oscilloscope.

Step 3 Observe the Modulation signal input TP (2) and Pulse input TP (1).
[CH1(Y) - 2V; CH2(X) - 2V; Time base - 0.2 mS]

Step 4 Observe the outputs at TP (3) together with Modulation signal input TP (2). This is a **Natural Sampling output.**

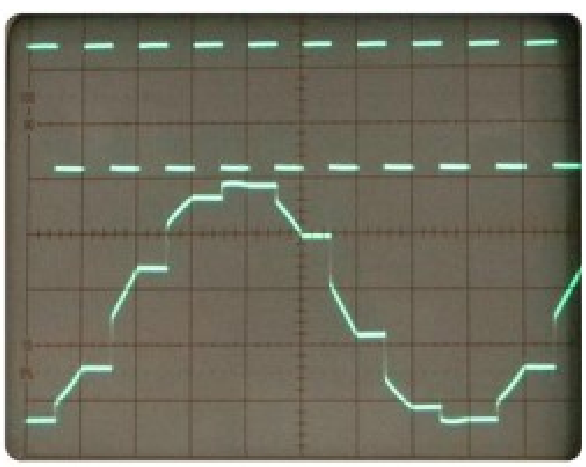


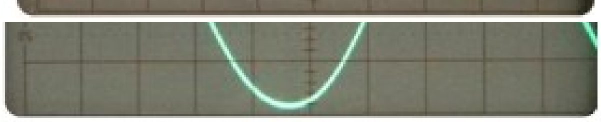
Step 5 Observe the Flat Top output at TP (5), together with Pulse input TP (1). This



is **Flat Top Sampling output.**

Step 6 Observe the output of Sample & Hold Circuit at TP (4), together with Pulse





input TP (1). This is **Sample & Hold output.**

Step 7 Vary the Amplitude Potentiometer and frequency change over switch & observe the effect on the output.

Step 8 Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.

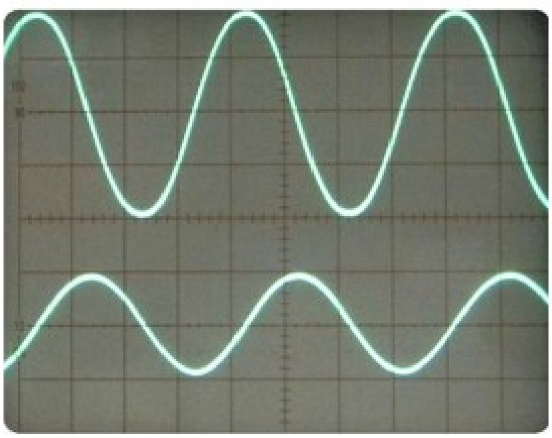
Step 9 Observe the difference between the three outputs.

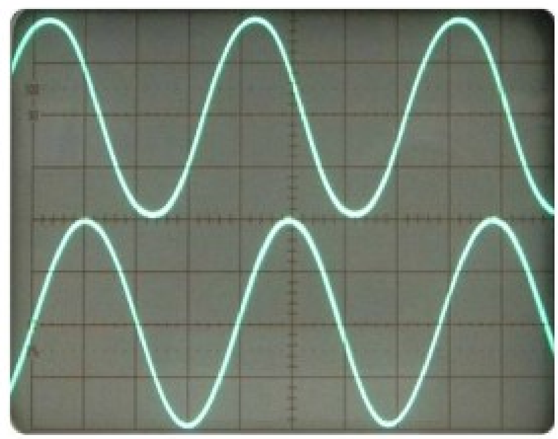
Step 10 Switch ‘On’ fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.

Demodulation

Step 11 Connect the **Sample output** to the input of low pass filter. Observe the output of the Low pass filter TP (12) together with Modulation signal input TP (2).

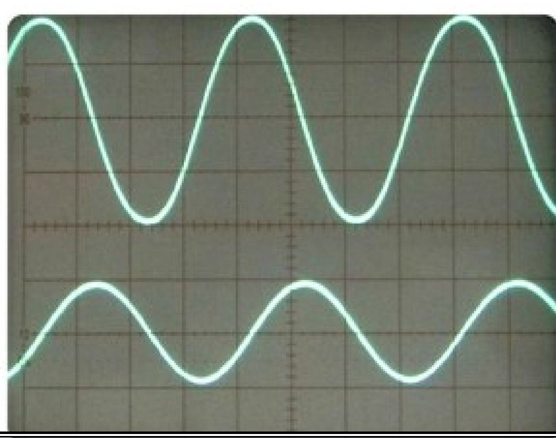
Step 12 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.

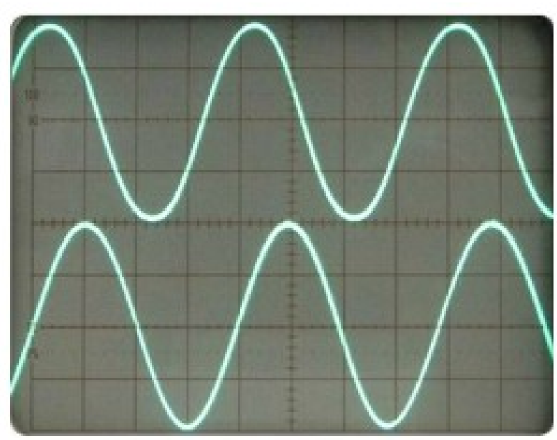




Step 13 Connect the **Flat top output** to the input of low pass filter Observe the output of the Low pass filter TP (12) together with Modulation signal input TP (2).

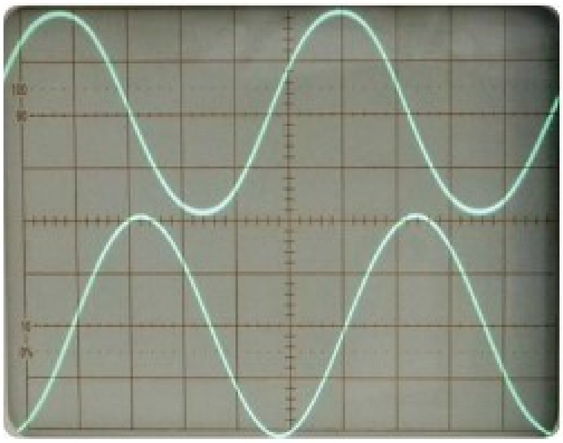
Step 14 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.

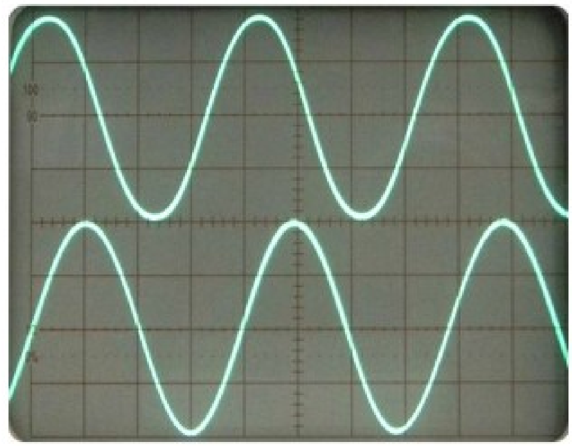




Step 15 Connect the **Sample & Hold output** to the input of low pass filter Observe the output of the Low pass filter TP (12) together with Modulation signal input TP (2).

Step 16 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.





Step 17 Vary the Amplitude Potentiometer and frequency change over switch & observe the effect on the output.

Step 18 Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.

Step 19 Switch 'On' fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.

Step 20 Switch 'Off" the Power Supply.

RESULT:- .

Pulse amplitude modulated waveform is obtained on CRO.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1. What is PAM?

Ans. Amplitude of the sampled pulse is varied according to the modulating signal.

Q.2. How many types of pulse modulation?

Ans. There are two types of PM i.e. PAM and PTM.

Q.3. How many types of pulse time modulation?

Ans. There are two types of PTM i.e. PWM and PPM.

Q.4. Give classification of Sampling.

Ans. There are three types of sampling techniques as under:

- Ideal Sampling or Instantaneous Sampling or Impulse Sampling
- Natural Sampling
- Flat top Sampling

Q.5. Why flat top sampling better than natural sampling?

Ans. The noise is interfered at top of the transmission pulse which can be easily removed if the PAM pulse in flat top.

Q.6. What is the significance of sampling?

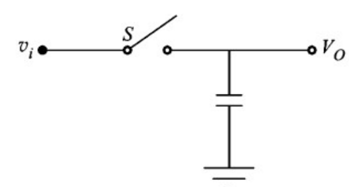
Ans. To convert a signal from continuous time to discrete time, a process called sampling is used. The value of the signal is measured at certain intervals in time.

Q.7. Which filter is used in PAM demodulator circuit?

Ans. Second order low pass filter.

Q.8. What do you understand by sample & hold circuit?

Ans. Sample and Hold Circuit takes samples from the analog input signal and hold them for particular period of time and then outputs the sampled part of input signal.



Basic Sample-and-Hold Circuit

Q.9. How PAM is recovered at the receiver end?

Ans. The Pulse Amplitude Modulation is recovered by a low pass filter.

Q.10. Mention application of PAM.

Ans. It is used in Ethernet communication, micro-controllers for generating the control signals, Photo-biology, as an electronic driver for LED lighting.

EXPERIMENT No. 12

OBJECTIVE:- To study Pulse Width Modulation.

APPARATUS REQUIRED:- Experimental kit, power supply, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

PWM is a part of PTM modulation. The PWM is also called PDM (pulse duration modulation) and sometimes it is also called PLM (pulse length modulation).

In PWM width of each pulse depends on the instantaneous value of the base band signal at the sampling instant. In pulse width modulation continuous waveform is sampled at regular intervals and the width of each pulse is kept proportional to the magnitude of signal at that instant in PWM. In pulse width modulation pulse is varied accordance with the modulating signal but the amplitude and starting time of each pulse is fixed .In PWM, the information about the base band signal lies in the trailing edge of the pulse

PWM has the disadvantage, when compared with PPM that its pulses are of varying width and therefore of varying power content .This means that transmitter must be powerful enough to handle the maximum- width pulses, although the average power transmitted is perhaps only half of the peak power. PWM still works if synchronization between transmitter and receiver fails.

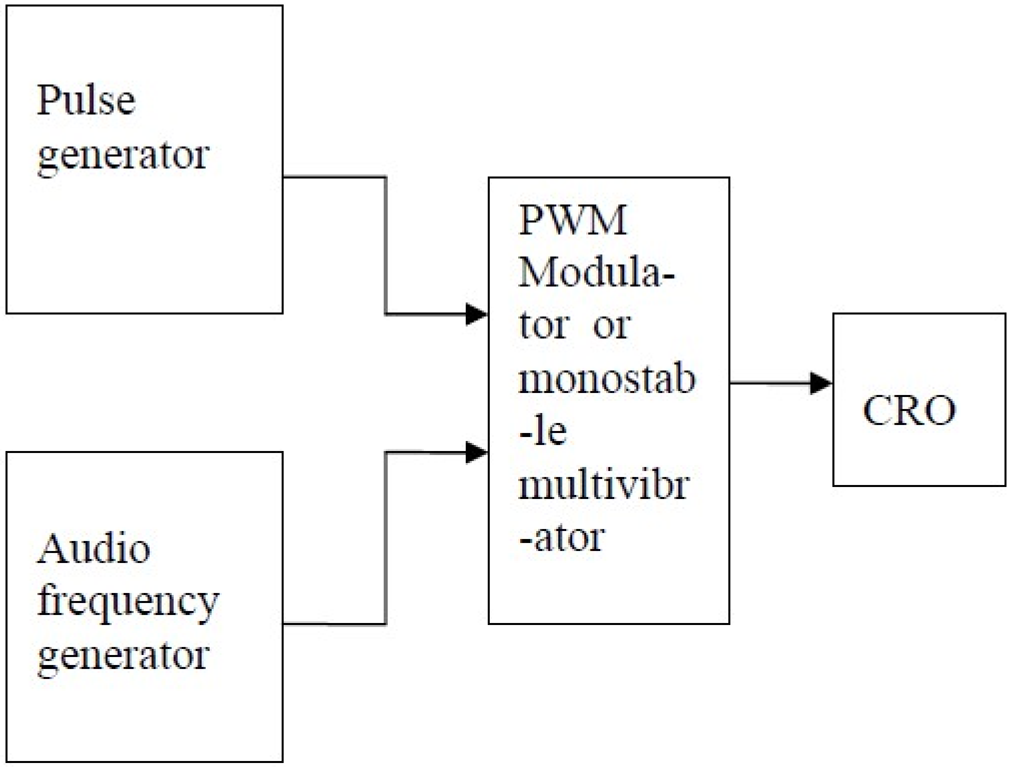
Generation and Demodulation of PWM

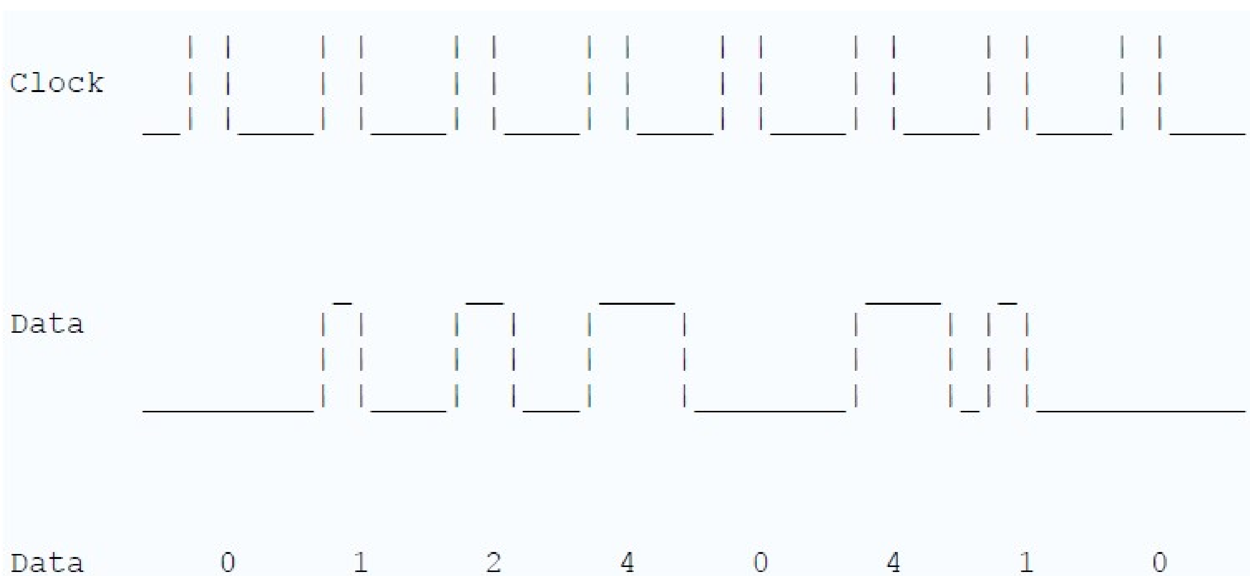
PWM may be generated by applying trigger pulses to control the duration of these pulses. The emitter coupled mono-stable multi-vibrator is used as voltage to time converter, since its gate width is dependent on the voltage to which the capacitor C is charged .If this voltage is varied in accordance with a signal voltage, a series of rectangular pulses will be obtained, with widths varying as required. The demodulation of pulse width modulation is a simple process. PWM is fed to an integrating circuit from which a signal emerges whose amplitude at any time is proportional to the pulse width at that time.

PROCEDURE:-

Step 1 Make the connection according to the block diagram.
Step 2 Connect the audio frequency of 2 KHz, 2V to modulator.
Step 3 Connect the modulator output to CRO.
Step 4 Switch ON the power supply.
Step 5 Observe output on CRO.

BLOCK DIAGRAM:-





WAVE FORMS OBSERVED:-

RESULT:-

Pulses width modulated wave is obtained on CRO.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1 Which modulation is PWM?
Ans. Analog Modulation
Q.2 What are two categories of Pulse Modulation.?
Ans. 1. Analog Modulation 2. Digital Modulation
Q.3 What is the unit of signaling speed?
Ans. Baud
Q.4 What is the disadvantage of PWM?
Ans. Due to varying of pulses width power contents of PWM also varying
Q.5 Which multivibrator is used for PWM?
Ans. Monostable Multivibrator
Q.6 Which circuit is used for PWM demodulator?
Ans. Integrating circuit.
Q.7 What is difference between PAM and PWM ?
Ans. In PAM, amplitude of pulse is varied according to modulating signal and in PWM, width is varied of pulses.
Q.8 How PWM may be generated?
Ans. PWM may be generated applying trigger pulses to contol the starting time of pulses from a monostable multivibrator.
Q.9 What is the use of sampling theorem?
Ans. Sampling Theorem is used to determine minimum sampling speed.
Q.10 What is the worldwide standard sampling rate?
Ans. Eight thousand samples per second.

EXPERIMENT No. 13

OBJECTIVE:- To study Pulse Position Modulation.

APPARATUS REQUIRED:- Experimental kit, power supply, CRO 20 MHZ, CRO probes.

OBJECTIVE:-

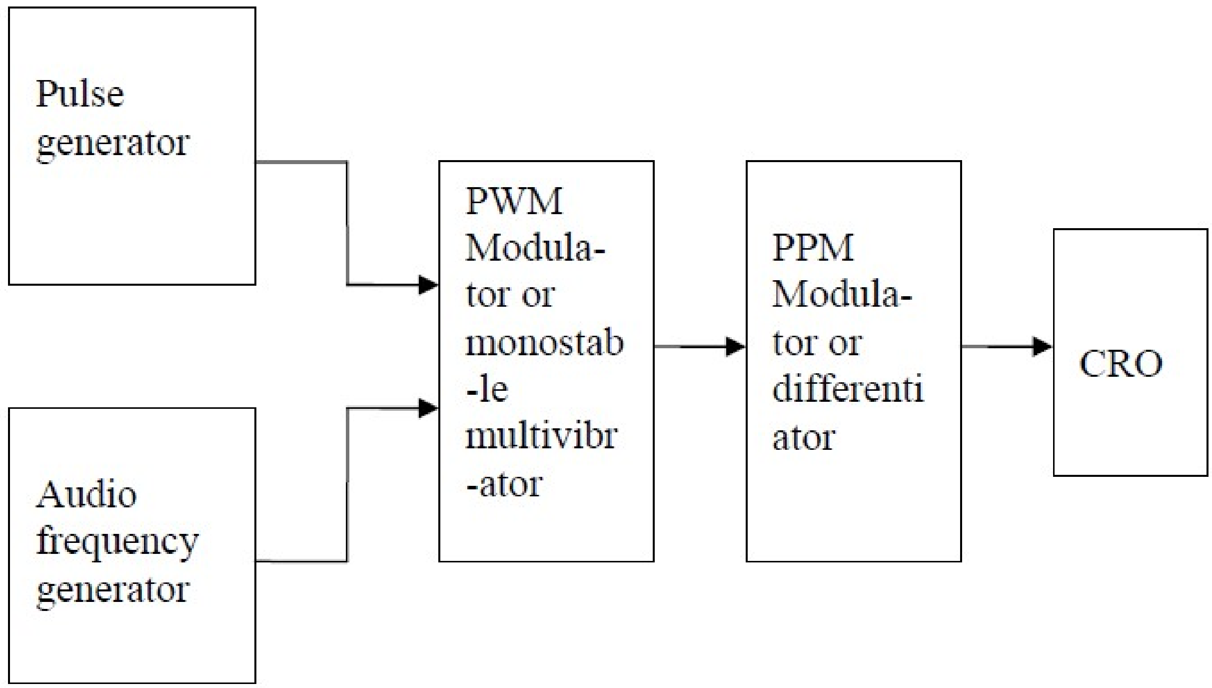
In pulse position modulating the amplitude of pulse is kept constant and position of the pulse in relation to the position of the reference pulse or synchronize pulse is varied by each sample value of modulating signal.

PPM may be obtained from PWM. In PWM each pulse has a leading edge and a trailing edge but the location of leading edge are fixed where trailing edge are depends on the pulse width. Thus PPM may be obtained from PWM by simply getting side of the leading edger and slots tops of PWM pulses. In pulse position modulating the amplitude of pulse is kept constant and position of the pulse in relation to the position of the reference pulse or synchronize pulse is varied by each sample value of modulating signal. In PWM each pulse has a leading edge and a trailing edge but the location of leading edge are fixed where trailing edge are depends on the pulse width. The trailing edges of PWM pulses are in fact position modulated. Thus PPM may be obtained from PWM by simply getting rid of the leading edge and slots tops of PWM pulses. In comparison with PWM, PPM has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter –receiver synchronization.

Generation and demodulation of PPM:

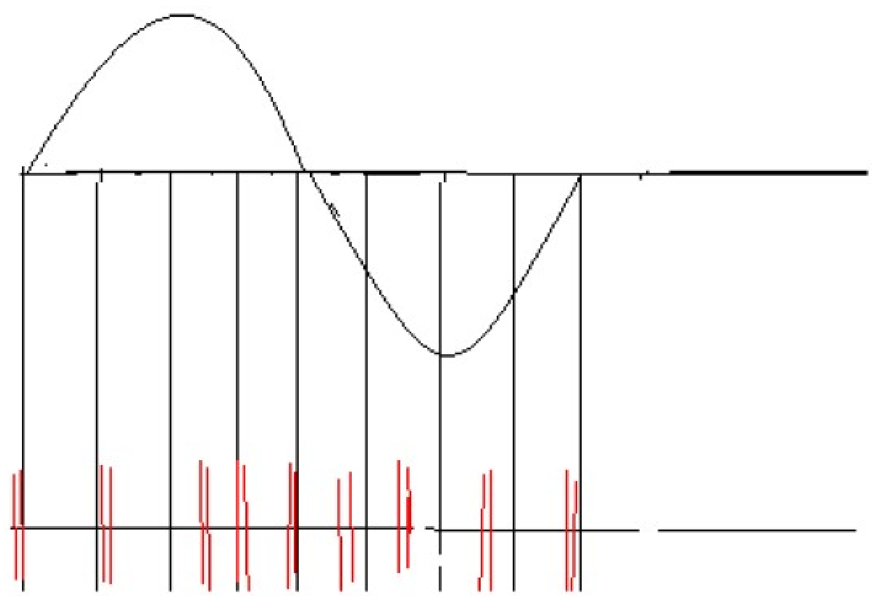
PPM may be generated from PWM easily. First of all, PWM pulses are generated and then they are differentiated. The result is another pulse train which has positive going narrow pulses corresponding to leading edges and negative going narrow pulses corresponding to trailing edges. If the position corresponding to the trailing edges of an un-modulated PWM pulse is counted as zero displacement, then the trailing edges of a modulated pulse will arrive earlier or later. An unmodulated PWM pulse is one that is obtained when the instantaneous signal value is zero. The differentiated pulses corresponding to the leading edges are removed with a diode clipper and the remaining pulses are nothing but position modulated output. When the PPM is demodulated in the receiver, it is again first converted into PWM by using flip-flop or bistable multivibrator. One input of the multivibrator receives trigger pulses from a local generator which is synchronized by trigger pulses received from the transmitter, and these triggers are used to switch off one of the stages of the flip-flop. The PPM pulses are fed to the other base of the flip-flop and switch that stage ON. The period of time during which this particular stage is OFF, depends on the time difference between the two triggers, so that the resulting pulse has a width that depends on the time displacement of each individual PPM pulse. The PWM pulse train thus obtained is a demodulated output.

BLOCK DIAGRAM:-



PROCEDURE:-

Step 1 Make the connection according to the block diagram.
Step 2 Connect the audio frequency of 2 KHz, 2V to modulator.
Step 3 Connect the PWM output to the PPM modulator.
Step 4 Connect the PPM modulator output to CRO.
Step 5 Switch ON the power supply.
Step 6 Observe output on CRO.



WAVE FORMS:-

RESULT:-
The Pulse Position Modulated wave is obtained on CRO.

PRECAUTIONS:-
1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-
Q.1 What is advantage of PPM?
Ans. It has no varying width of pulse so power content are not varying.
Q.2 What is PPM?
Ans. In PPM the position of pulses is varied and width and amplitude are constant.
Q.3 Which Multivibrator is used for PPM De-modulator?
Ans. Bi-stable Multivibrator.
Q.4 What is the difference between PPM & PWM?
Ans. In PWM, the width is varied and in PPM, the position is varied according to modulating signal.
Q5. Which filter is used in PPM demodulator?
Ans. Second order low pass filter.
Q6. In which category of PM is PPM?

Ans. Analog Modulation.
Q7. Which modulation is similar to PDM?
Ans. Phase modulation.
Q8. At which factor the band-width of PPM depends?
Ans. Bandwidth of transmission channel depends on rising time of the pulse.
Q.9 What is the use of sampling theorem?
Ans. Sampling Theorem is used to determine minimum sampling speed.
Q.10 What is the world wide standard sampling rate?
Ans. Eight thousand samples per second.

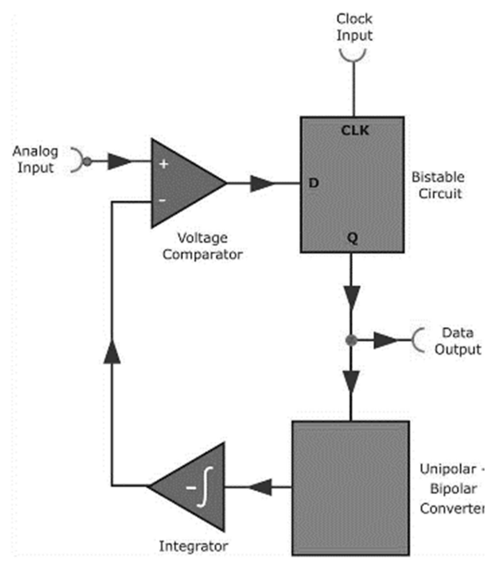
EXPERIMENT No. 14

OBJECTIVE:- To study Delta Modulation.

APPARATUS REQUIRED:- Delta Modulator trainer kit, CRO 20 MHZ, CRO probes.

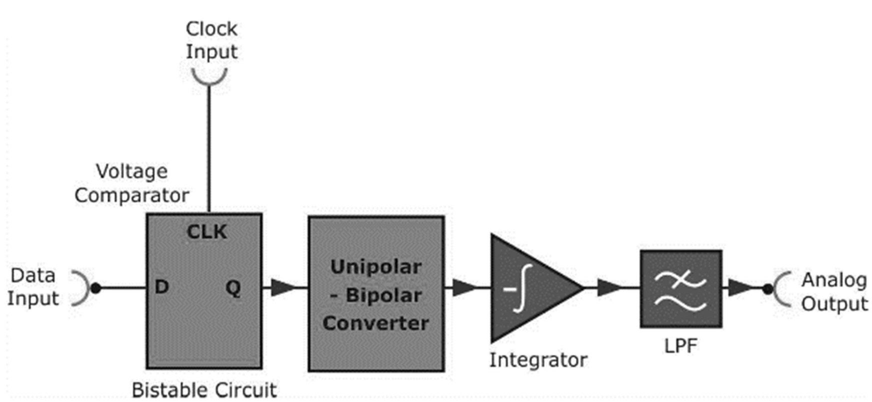
OBJECTIVE:-

PCM codes each quantized sample into a binary code that is sent and decoded at the receiver. Another form of coded modulation is called Delta Modulation.



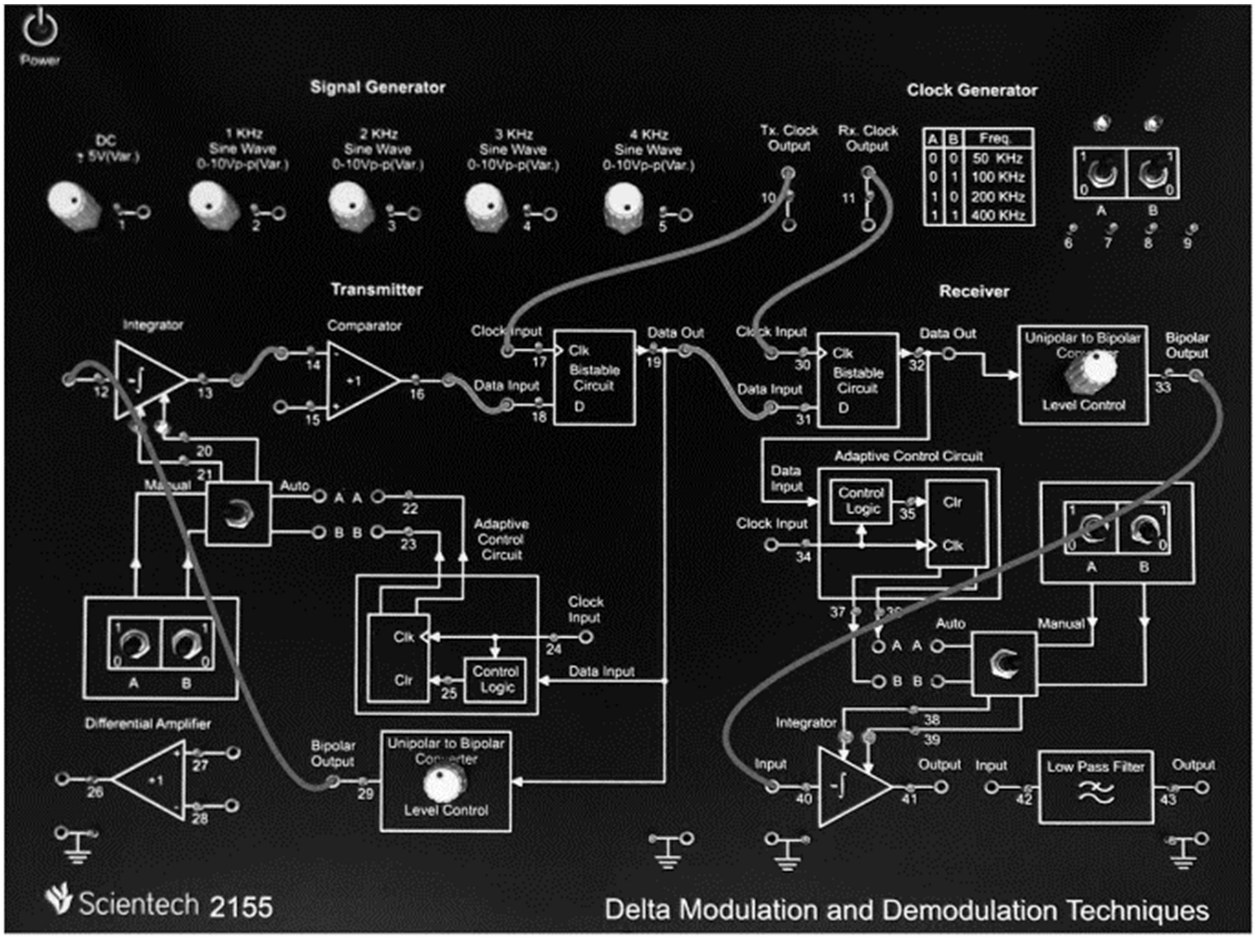
Delta modulator

The main principle behind the Delta Modulation is to purposely over sampling the base band signal purposely to increase the correlation between adjacent samples of the signal, so as to permit the use of a sample quantizing strategy for constructing the encoded signal. Delta Modulation is a process of converting Analog signal into one bit code. In Delta Modulation only one bit is sent per sample. This bit indicates whether the signal is larger or smaller than the previous samples. The advantage of Delta Modulation is that the Modulator and Demodulator circuits are much simpler than those used in traditional PCM systems. The purpose behind this form of Modulation is to minimize the effects of noise without increasing the number of bits being sent. This increases the Signal-to-Noise Ratio improving system performance. The idea behind delta modulation is to take samples close enough to each other so that each samples amplitude does not vary by more than signal step size. Then instead of sending a binary code representing the step size, a single bit is sent, signifying whether the sample size has increased or decreased by a single step. The Original Signal is first transmitted



Delta De Modulator

and quantized as with PCM. If the sample currently being coded is above the previous sample, then a binary bit is set to logic '1'.If the sample is lower than the



previous sample then the bit is set low.

PROCEDURE:-

Step 1 Make Connect PLA1 to PLAA.

Step 2 Connect Channel-1 of CRO to TPA1/TPAA. Adjust VR1 to minimum to get zero level signal.

Step 3 Connect Channel-1 to TP2 and Channel-2 to TPB1and adjust VR2 to obtain square wave half the frequency of the clock rate selected (Output at TP1).

Step 4 Connect Channel-1 to TP2 and set voltage/div of Channel 1 to mV range and observe a triangle waveform, which is output of integrator. It can be observed that as the clock rate is increased, amplitude of Triangle waveform decreases. This is called minimum step size (Clock rate can be changed by depressing SW1 switch).

Step 5 Connect Channel-1 to TPA1/TPAA; adjust VR1 in order to obtain a 1 KHz sine wave of 500 mV Pk-Pk approximately.

Step 6 Signal approximating 1 KHz is available at the integrator output (TP2). This signal is obtained by integrating the digital output resulting from Delta Modulation.

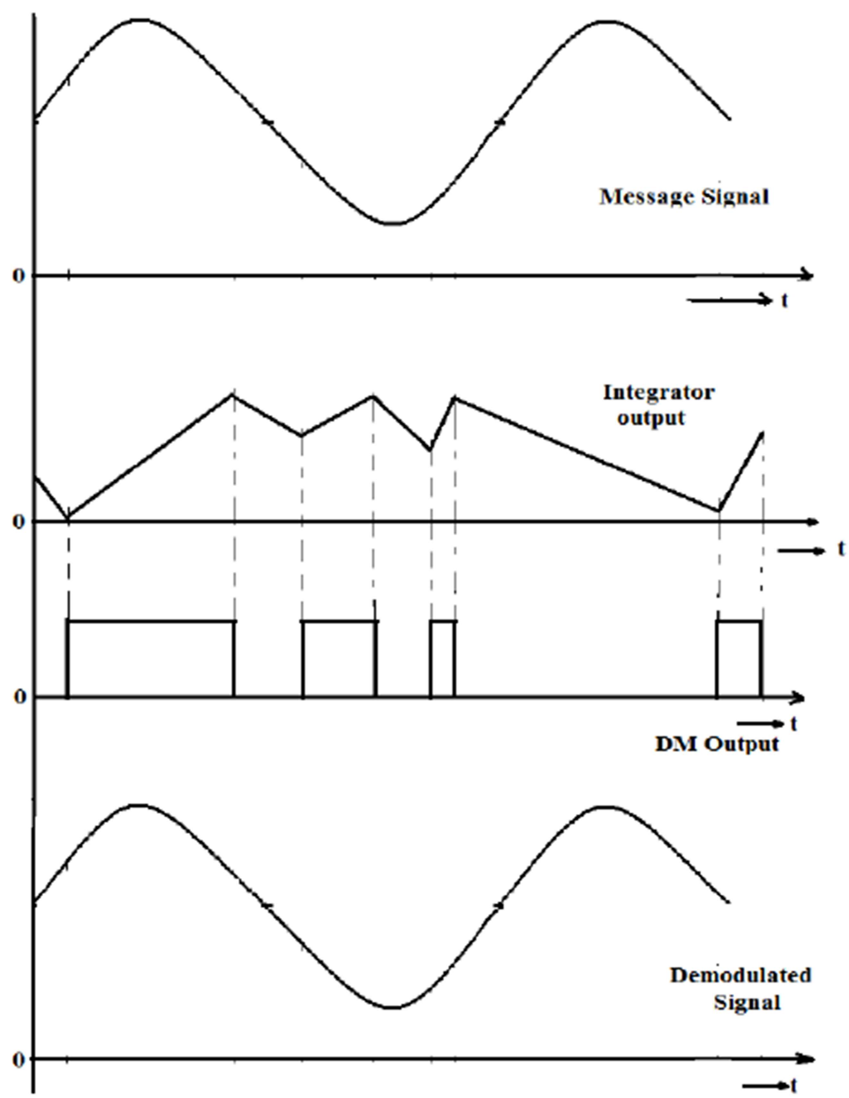
Step 7 Connect Channel-1 to TP2 and Channel-2 to TPB1.It can be observed that the digital high makes the integrator output to go upwards and digital low makes the integrator output to go downwards.

Step 8 With an Oscilloscope displaying three traces. It is possible to simultaneously observe the input signal of the modulation, the digital output of the modulator and the signal is obtained by the Integration from the modulator digital output.

Step 9 Notice that, when the output is lower than the analog input the digital output is high, whenever it is low when the analog input is lower than the integrated output.

Step 10 Increase the amplitude of 1 KHz sine wave by rotating VR1 1 V and high in the next case observe the changes in output signal. Repeat the same for different signal sources.

WAVE FO



Waveforms of Delta Modulation & Demodulation

RESULT:-

Thus the Delta modulation and demodulation is performed practically and the waveforms are plotted.

PRECAUTIONS:-

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3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ/ANSWERS:-

Q.1 What is DPCM?
Ans. Differential Pulse Code Modulation.
Q.2 What is the advantage of DPCM?
Ans. It require less BW as compared to PCM.
Q.3 What is quantizer?
Ans. It converts the sample values to some fixed finite levels.
Q.4 What is the use of predictor?
Ans. To estimate previous sample.
Q.5 Which one is better PCM or DPCM?
Ans. DPCM.
Q.6 Is DPCM analog modulation technique?
Ans. It belong to the class of pulse digital modulation.
Q.7 Which one has less BW requirement DPCM or Delta modulation?
Ans. Delta modulation.
Q.8 DPCM is suitable for which kind of input signals?
Ans. Where dynamic changes in signal are small, DPCM I very useful.
Q.9 Why DPCM is preferred over PCM?
Ans. Because of low BW.
Q.10 DPCM is preferably used for……….
Ans. Voice or picture Communication.

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