DRONACHARYA
Group of Institutions

COMMUNICATION ENGINEERING LAB

LABORATORYMANUAL

B.Tech. Semester -IV

Subject Code: BEC-451

Session: 2024-25, Even Semester

Name:	
Roll. No.:	
Group/Branch:	

DRONACHARYA GROUP OF INSTITUTIONS
DEPARTMENT OF ECE
#27 KNOWLEDGE PARK 3

GREATER NOIDA

AFFILATED TO Dr. ABDUL KALAM TECHNICAL UNIVERSITY, LUCKNOW

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Vision and Mission of the Institute

Vision:

Instilling core human values and facilitating competence to address global challenges by providing Quality Technical Education.

Mission:

- M1 - Enhancing technical expertise through innovative research and education, fostering creativity and excellence in problem-solving.
- M2 - Cultivating a culture of ethical innovation and user-focused design, ensuring technological progress enhances the well-being of society.
- M3 - Equipping individuals with the technical skills and ethical values to lead and innovate responsibly in an ever-evolving digital landscape.

Vision and Mission of the Department

Vision:

- Providing advanced education and engaging technocrats in cutting-edge research in field of Electronics and Communication Engineering and foster them to be globally competitive engineering professionals with a strong ethical foundation.

Mission:

- To deliver robust foundational knowledge and technical expertise through effective teaching learning Methodologies.
- To create an environment conducive to research by fostering partnerships between industry and academia.
- To integrate advancement of technology in Electronics and Communication Engineering with a focus on social responsibility and environmental consciousness.

Programme Educational Objectives (PEOs)

- To apply the scientific, mathematical and engineering fundamentals to provide solutions to the problems in Electronics and Communication Engineering and related fields.
- To exhibit creativity and innovation with ethical and professional behavior while addressing societal needs by engaging technocrats in independent learning.
- To empower technocrats by nurturing ethical values, creativity, and innovation, enabling them to pursue entrepreneurship or higher studies and establish their own startups, thereby impacting their lives positively.

Programme Outcomes (POs)

Engineering Graduates will be able to:

- **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 IT 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.

Program Specific Outcomes (PSOs)

ECE

- To apply fundamental knowledge of core Electronics and Communication Engineering subjects in the analysis, design, and development of various types of electronic systems.
- To adapt to modern software and hardware tools in Electronics and Communication Engineering for the design and analysis of complex electronic systems and their real-life applications.
- To prepare students to adjust in evolving work environments, effective interpersonal abilities, adherence to professional ethics and awareness of societal needs.

University Syllabus

1. To study DSB/ SSB amplitude modulation & determine its modulation factor & power inside bands.
2. To study amplitude demodulation by linear diode detector.
3. To study frequency modulation and determine its modulation factor.
4. To study sampling and reconstruction of pulse amplitude modulation system.
5. To study pulse amplitude modulation. a) Using switching method b) By sample and hold circuit
6. To demodulate the obtained PAM signal by 2^{nd} order LPF.
7. To study pulse width modulation and pulse position modulation.
8. To study pulse code modulation and demodulation technique.
9. To study delta modulation and demodulation technique.
10. To construct a square wave with the help of fundamental frequency and its harmonic component.
11. Study of amplitude shift keying modulator and demodulator.
12. Study of frequency shift keying modulator and demodulator.
13. Study of phase shift keying modulator and demodulator.
14. Study of single bit error detection and correction using hamming code.
15. Study of quadrature phase shift keying modulator and demodulator.
16. To simulate differential phase shift keying technique using MATLAB software.
17. To simulate M-ary Phase shift keying technique using MATLAB software (8PSK, 16PSK) and perform BER calculations.
18. Design a front end BPSK modulator and demodulator.

Course Outcomes (COs)

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

CO1	Analyze and compare different analog modulation schemes for their Modulation factor and power.
CO2	Study pulse amplitude modulation.
CO3	Analyze different digital modulation schemes and can compute the bit error Performance.
CO4	Study and simulate the Phase shift keying.
CO5	Design a frontend BPSK modulator and demodulator.

CO-PO Mapping

	PO1	**PO2**	**PO3**	**PO4**	**PO5**	**PO6**	**PO7**	**PO8**	**PO9**	**PO10**	**PO11**	**PO12**
CO1	3	3	2	2	1	2		2	2			1
CO2	2	2	1			1						
CO3	2	3	3	2	2	3		2	2			2
CO4	2	2						2	2			2
CO5	3	2	3	3	2	3		1	1			2
Course Correlation mapping	**2.4**	**2.4**	**1.8**	**1.4**	**1**	**1.8**		**1.4**	**1.4**			**1.4**

Correlation Levels: High-3, Medium-2, Low-1

CO-PSO Mapping

	PSO1	**PSO2**	**PSO3**
CO1	3	3	2
CO2	3	3	2
CO3	2	3	2
CO4	2	2	1
CO5	2	2	1

	2.4	**2.6**	**1.6**

Course Overview

Communication engineering laboratory focuses on training the students in both analog and digital transmission/reception of signal. The students here start in the analog communication area by constructing the circuits of amplitude modulation, frequency modulation and phase modulation. A mini AM transmitter is constructed by them in the MW frequency of electromagnetic spectrum and they do a live transmission with minimum power output. Learning the concepts, with small applications give them plenty of joy and real motivation towards their studies. The other important area of analog signal processing is the phase locked loop. Students are given NE 565 PLL IC and made to conduct an experiment to find the capture and lock ranges with their designed specifications.

Here the students make use of the function generators with frequency sweep facility. The students are enlightened yet in another area are audio systems, widely used to address the public.

The students perform the characteristics of microphone. So, they gain knowledge in Acoustical Engineering. They also test condenser, coil and piezo electric transducers used in the audio engineering. In the area of pulse techniques, faculty helps them in PAM and PWM experiments. In addition to all the above, the lab consists of the analog signal sampling kit, TDM trainer and microwave test benches.

List of Experiments mapped with COs

S. No.	Name of the Experiment	Course Outcome
1	To study DSB/SSB amplitude modulation & determine its modulation factor & power in side bands.	**CO1**
2	To study amplitude demodulation by linear diode detector.	**CO2**
3	To study frequency modulation and determine its modulation factor.	**CO1**
4	To study sampling and reconstruction of pulse amplitude modulation system.	**CO2**
5	To study pulse amplitude modulation. a)Using switching method b)By sample and hold circuit	**CO1**
6	To demodulate the obtained PAM signal by 2^{nd} order LPF.	**CO3**
7	To study pulse width modulation and pulse position modulation.	**CO3**
8	To study pulse code modulation and demodulation technique.	**CO2**
9	To study delta modulation and demodulation technique.	**CO2**
10	To construct a square wave with the help of fundamental frequency and its harmonic component.	**CO2**
11	Study of amplitude shift keying modulator and demodulator.	**CO2**
12	Study of frequency shift keying modulator and demodulator.	**CO3**
13	Study of phase shift keying modulator and demodulator.	**CO3**
14	Study of single bit error detection and correction using hamming code.	**CO4**
15	Study of quadrature phase shift keying modulator and demodulator.	**CO4**
16	To simulate differential phase shift keying technique using MATLAB software.	**CO4**
17	To simulate M-Ary Phase shift keying technique using MATLAB software (8PSK,16PSK) And perform BER calculations.	**CO5**
18	Design a front end BPSK modulator and demodulator.	**CO5**

Dos and DON'Ts

DOs

1. Observe type of sockets of equipment power to avoid mechanical damage.
2. Strictly observe the instructions given by the Teacher/ Lab Instructor.
3. It is mandatory to come with observation book and lab recording which previous experiment should be written in Record and the present labs experiment in Observation book.
4. Observation book of the present lab experiment should be get corrected on the same day and Record should be corrected on the next scheduled lab session.
5. Mobile Phones should be Switched OFF in the lab session.
6. Students have to come to lab in-time. Late comers are not allowed to enter the lab.
7. Prepare for the viva questions. At the end of the experiment, the lab faculty will ask the viva.
8. Bring all the required stationery like graph sheets, pencil & eraser, different color pens etc. for the lab class.

DON'Ts

1. Do not handle any equipment without reading the instructions/ Instruction manuals.
2. Do not insert connectors forcefully in the Sockets.

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 hall way.
4. Call security and emergency department immediately:

Emergency : 201 (Reception) Security : 231 (Gate No.1)

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 Course Outcomes attainment as well as internal marks in the lab course.

Grading Criteria	Exemplary (4)	Competent (3)	Needs Improvement(2)	Poor(1)
AC1: Designing experiments	The student chooses the problems to explore.	The student chooses the problems but does not set an appropriate goal for how to Explore them.	The student fails to define the problem adequately.	The student does not identify the problem.
AC2: Collecting data through observation and/ or experimentation	Develops a clear procedure for investigating the problem	Observations are completed with necessary theoretical calculations and proper identification of required components.	Observations are completed with necessary theoretical calculations but without proper understanding. Obtain the correct values for only a few components after calculations. Followed the given experimental procedures but obtained results with some errors.	Observations are incomplete. Lacks the appropriate knowledge of the lab procedures.
AC3: Interpreting data	Decides what data and observations are to be collected and verified	Can decide what data and observations are to be collected but lacks the knowledge to verify	Student decides what data to gather but not sufficient	Student has no knowledge of what data and observations are to be collected
AC4: Drawing conclusions	Interprets and analyses the data in order to propose viable conclusions and Solutions.	Incomplete analysis of data hence the quality of conclusions drawn is not upto The mark	Cannot analyze the data or observations for any kind of conclusions.	Lacks the required knowledge to propose viable conclusions and solutions
AC5: Labrecord assessment	Well-organized and confident presentation of record & ability to correlate the theoretical concepts with the concerned lab results with Appropriate reasons.	Presentation of record is acceptable	Presentation of record lacks clarity and organization	No efforts were exhibited

LAB EXPERIMENTS

EXPERIMENT No. 1

AIM:-To generate DSB-SC AM signal using balanced modulator.

APPARATUS REQUIRED:-(i)C.R.O.(ii) CRO Probe (ii) DSB/SSB Transmitter (ST 2201) and Receiver (ST2202) Trainer (iv) Connecting leads.

THEORY:-

A double sideband suppressed carrier signal, or DSBSC, is defined as the modulating signal and the carrier wave.

$$DSBSC=E.\ \cos\mu t.\ \cos\omega t \quad (1)$$

Generally, and in the context of this experiment, it is understood that: $\omega >> \mu$ (2)

Equation (3) can be expanded to give:

$$\cos\mu t\ .\ \cos\omega t = (E/2)\cos(\omega - \mu)t + (E/2)\cos(\omega + \mu)t \quad (3)$$

Equation (3) shows that the product is represented by two new signals, one on the sum frequency ($\omega + \mu$), and one on the difference frequency ($\omega - \mu$) - see Figure 1.

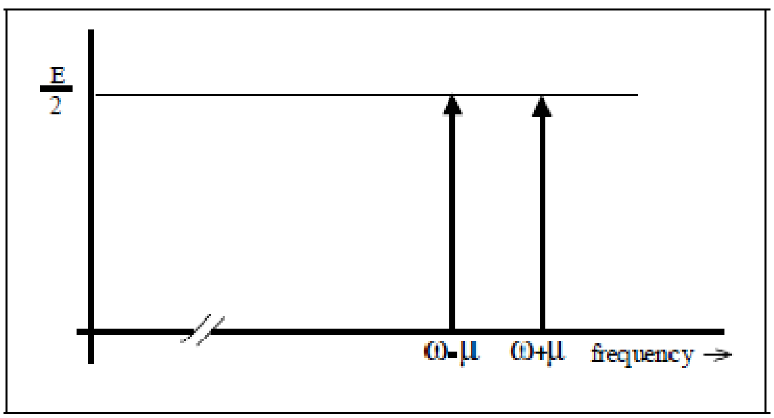


Figure1: Spectral components

Remembering the inequality of eqn. (2) the two new components are located close to the frequency ω rad/s, one just below, and the other just above it. These are referred To as the lower and upper side bands respectively.

These two components were derived from a _carrier‘term on ω rad/s, and a message on μrad/s. Because there is note r mat carrier frequency in the product signal it is described is a double sideband *suppressed* carrier (DSBSC) signal.

The term _carrier‘comes from the context of _double sideband amplitude modulation' (commonly abbreviated to just AM).

The time domain appearance of a DSBSC (eqn.1) in a text book is generally as shown in Figure 2.

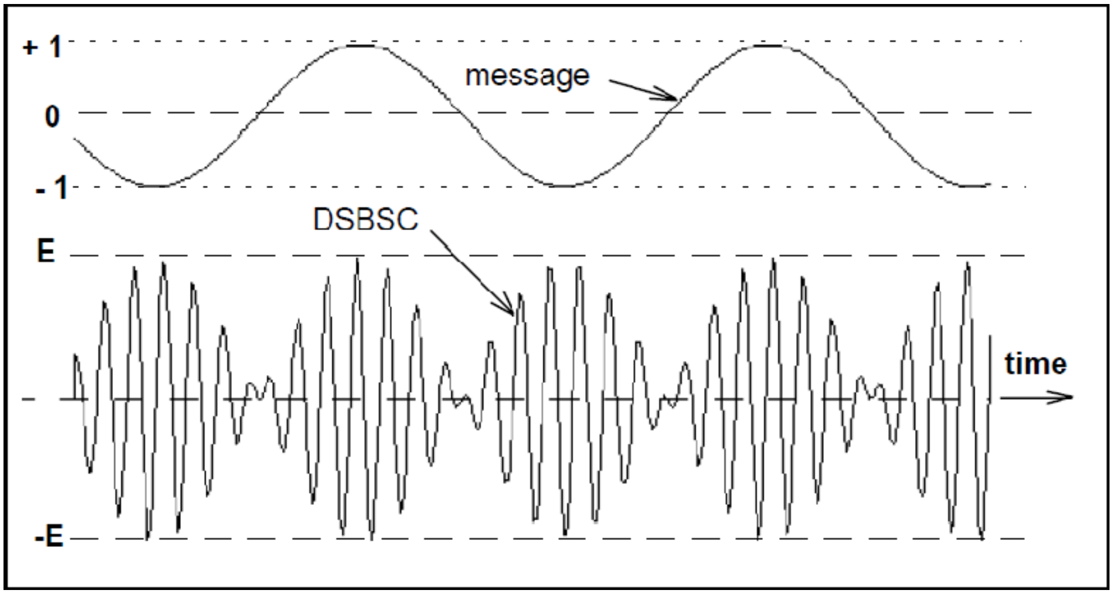


Figure2: DSBSC –seen in the time domain

Notice the waveform of the DSBSC in Figure 2, especially near the times when the message amplitude is zero. The fine detail differs from period to period of the message. This is because the ratio of the two frequencies μ and ω has been made non-integral. Although the message and the carrier are periodic waveforms (sinusoids), the DSBSC itself need not necessarily be periodic.

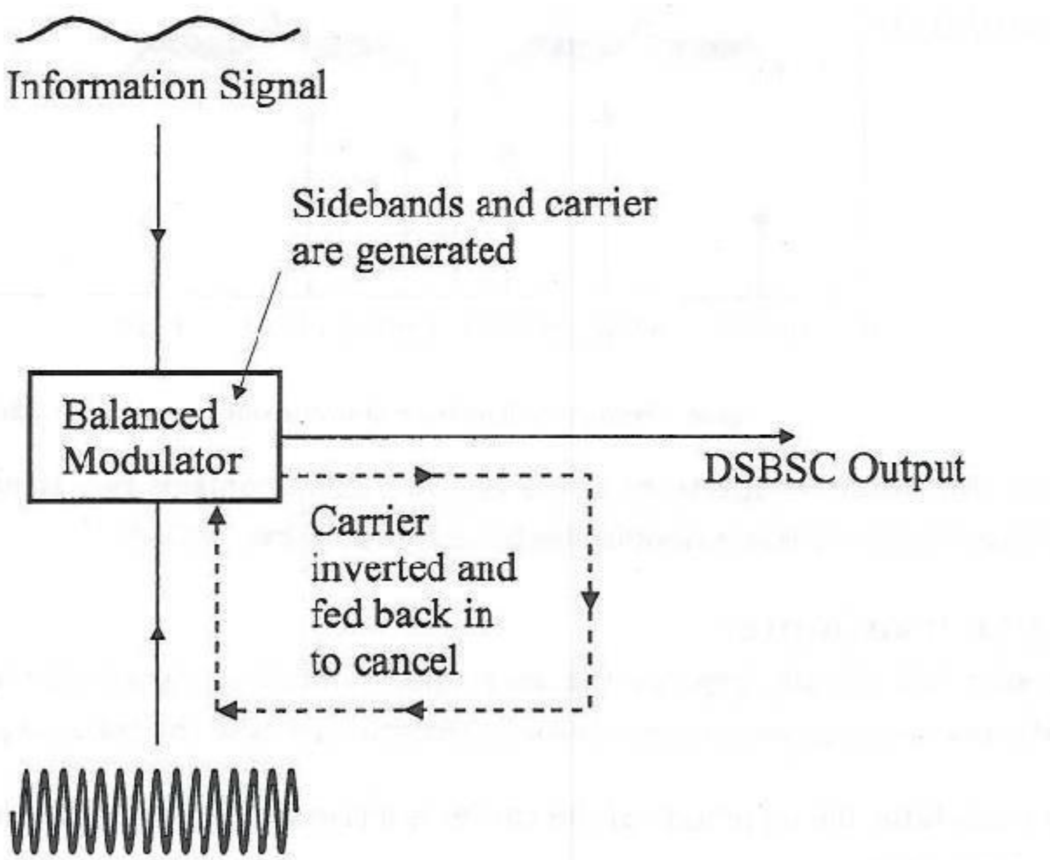


Figure3: DSBSC Generation using balanced modulator

By removing the carrier from an AM waveforms, the generation of double sideband suppressed carrier (DSBSC) AM is generated.

Properties of DSB-SC Modulation:

(a) There is a180 phase reversal at the point where m(t) goes negative. This is typical of DSB-SC modulation.

(b) The bandwidth of the DSB-SC signal is double that of the message signal, that is, $BW \quad =2B$ (Hz).

+

(c) The modulated signal is centered at the carrier frequency ω with two identical side bands (double-sideband) – the lower sideband (LSB) and the upper sideband (USB). Being identical, they both convey the same message component.

(d) The spectrum contains no isolated carrier. Thus the name suppressed carrier.

(e) The 180 phase reversal causes the positive (or negative) side of the envelope to have a shape different from that of the message signal, see Figure 2.

A balanced modulator has two inputs: a single-frequency carrier and the modulating signal. For the modulator to operate properly, the amplitude of the carrier must be sufficiently greater than the amplitude of the modulating signal (approximately six to seven times greater).

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

a. Audio input select switch in INT position:

b. Modes which in DSB position.

c. Output amplifier's gain pot in full clock wise position.

d. Speakers switch in OFF position.

2. Turn on powertotheST2201 board.

3. Turn the audio oscillator block's amplitude pot to its full clockwise (MAX) position, and examine the block's output (t.p.14) on an oscilloscope. This is the audio frequency sine wave which will be our modulating signal. Note that the sine wave‘s frequency can be adjusted from about 300 Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency pot meter. Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the Audio oscillator's amplitude pot meter to its fully counter clock wise(MIN) position. Return the amplitude represent to its max position.

4. Turn the balance pot, in the balanced modulator & band pass filter circuit 1 block, to its fully clockwise position. It is this blocks that we will use to perform *double-sideband amplitude modulation.*

5. Monitor, in turn, the two inputs to the balanced modulator & band pass filter circuits block, at t.p.1 and t.p.9. Note that:

a. The signal at t.p.1 is the audio-frequency sine wave from the audio oscillator block. This is the modulating input to our double-sideband modulator.

b. Test point 9 carries a sine wave of 1MHz frequency and amplitude 120mVpp approx. This is the carrier input to our double-sideband modulator.

6. Next, examine the output of the balanced modulator & band pass filter circuit 1 block (at t.p.3), together with the modulating signal at t.p.1 Trigger the oscilloscope on that .p. 1 signal. The output from the balanced modulator & band pass filter circuit 1 block (at t.p. 3) is a DSBFC AM waveform, which has been formed by amplitude-modulating the 1MHz carrier sine wave with the audio-frequency sine wave from the audio oscillator.

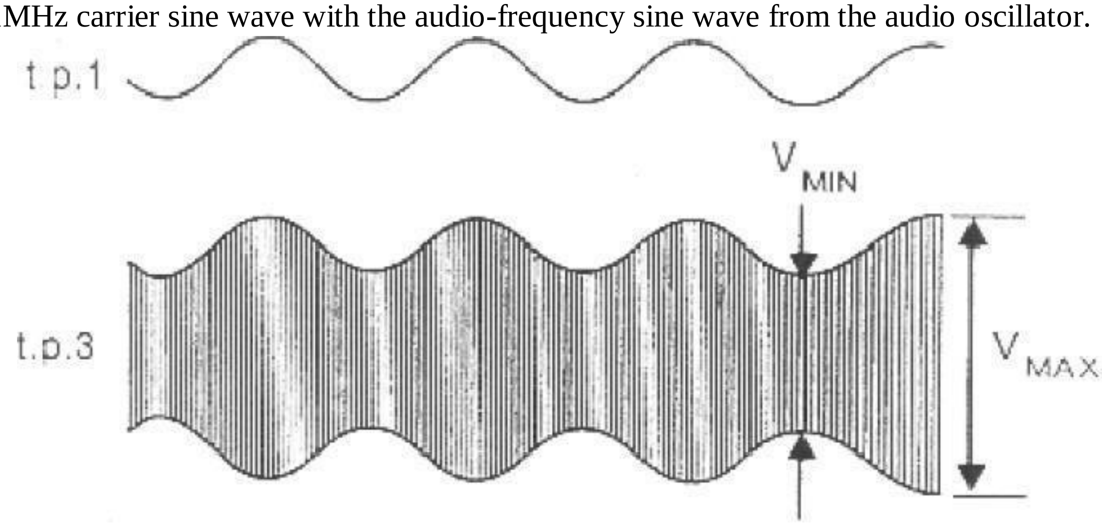


Figure4: DSBFC (AM) waveforms

7. Now vary the amplitude and frequency of the audio-frequency sine wave, by adjusting the amplitude and frequency present in the audio oscillator block. Note the effect that varying each pot has on the amplitude modulated waveform. The amplitude and frequency amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do this by turning the amplitude pot to its MIN position, and note that the signal at t.p3 becomes an un-modulated sine wave of frequency 1 MHz, indicating that only the carrier component now remains. Return the amplitude pot to its maximum position.

Now turn the balance pot in the balanced modulator & band pass filter circuit 1 block, until the signal at t.p. **3** is as shown in Fig. **5**

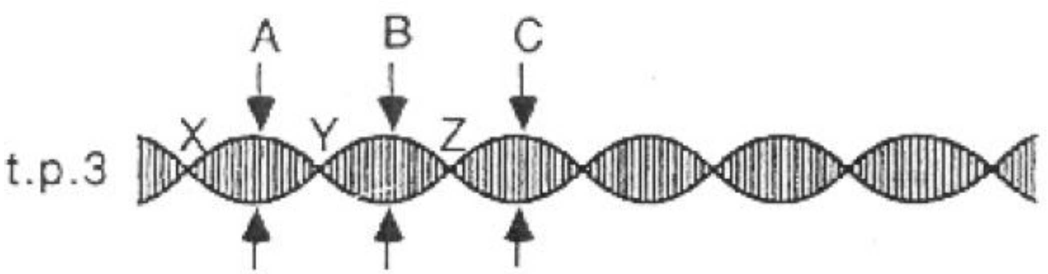


Figure**5**: Output of BPF

The balance pot varies the amount of the 1MHz carrier component, which is passed from the modulator's output. By adjusting the pot until the peaks of the waveform (A,B,C and so on) have the same amplitude, we are removing the carrier component altogether. We say that the carrier has been 'balanced out' (or 'suppressed') to leave only the two sidebands.

Note that once the carrier has been balanced out, the amplitude of t.p.3's waveform could be zero at minimum points X, Y & Z etc. If this is not the case, it is because one of the two sidebands is being amplified more than the other. To remove this problem, the band pass filter in the balanced modulator & band pass filter circuit 1 block must be adjusted so that it passes both sidebands equally. This is achieved by carefully trimming transformer T1, until the waveform's amplitude is as close to zero as possible at the minimum points. The waveform at t.p.3 is known as a double-side suppressed carrier (DSBSC) waveform, and its frequency spectrum is as shown in Fig.1.Notethatnowonly the two sidebands remain, the carrier component has been removed.

8. Change the amplitude and frequency of the modulating audio signal (by adjusting the audio oscillator block's amplitude and frequency pots), and note the effect that these changes on the DSBSC waveform. The amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do these by turning the amplitude present to its MIN position, and note that the monitored signal becomes a D C level, indicating that there .are now no frequency components present. Return the amplitude pot to its MAX position.

9. Examine the output from the output amplifier block (t.p.13), together with the audio modulating signal (at t.p.1), triggering the scope with the audio modulating signal. Note that the DSBSC waveform appears, amplified slightly at t.p.13, as we will see later, it is the output amplifier's output signal which will be transmitted to the receiver.

10. By using the microphone, the human voice can be used as the modulating signal, instead of using ST2201's audio oscillator block. Connect the module's output to the externalaudioinputontheST2201board, and put the audio input selects which in the ext position. The input signal to the audio input module may be taken from an external microphone or from a cassette recorder, by choosing the appropriate switch setting on the module.

RESULT:-

The DSBSC signal has been generated using balanced modulator.

WAVEFORMS OBSERVED:-

Draw waveforms as observed on CRO and label the different waveforms appropriately.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

Q.1.WhatisDSBSC?
Ans.DoubleSidebandSuppressedCarrier.
Q.2. Whicharethediscretefrequencies in DSBSC?
Ans.(1)Lowersidebandfrequency(2)Uppersidebandfrequency
Q.3InDSBSC,howmanysidebandsarethere?
Ans.TherearetwosidebandsinDSCBSC i.e.LSBand USB
Q.4.MentionadvantagesofDSBSCoverDSBFC. Ans.
Transmission efficiency is more.
Q.5.WhichtypeofcarrierisusedinRingmodulator? Ans.
Square wave carrier.
Q.6. WritethemethodsofDSBSC generation.
Ans.(1)BalancedModulator(2)RingModulator(3)SwitchingModulator
Q.7.WhatistheBWofDSBSCforasingletonemodulatingsignalwithfrequency*w*? Ans. 2*w*.
Q. 8.Wherethemodulationindexlies?
Ans.modulationindex always liesbetween0and 1.Morethan1isovermodulation.
Q.9.Whathappensincaseofovermodulation? Ans.
The wave will get distorted.
Q.10.Whatistherangeofaudiofrequencies? Ans. 20
Hz to 20 KHz.

EXPERIMENT No. 2

AIM:-To generate SSB-AM signal.

APPARATUSREQUIRED:-(i)C.R.O.(ii)CROProbe(ii)DSB/SSBTransmitter(ST 2201) and Receiver Trainer (ST 2202) (iv) Connecting leads

THEORY:-

Single Sideband Suppressed Carrier (SSB-SC) modulation was the basis for all long distance telephone communications up until the last decade. It was called "L carrier." It consisted of groups of telephone conversations modulated on upper and/or lower

sidebands of contiguous suppressed carriers. The groupings and sideband orientations (USB, LSB) supported hundreds and thousands of individual telephone conversations.

SSB Transmitter:

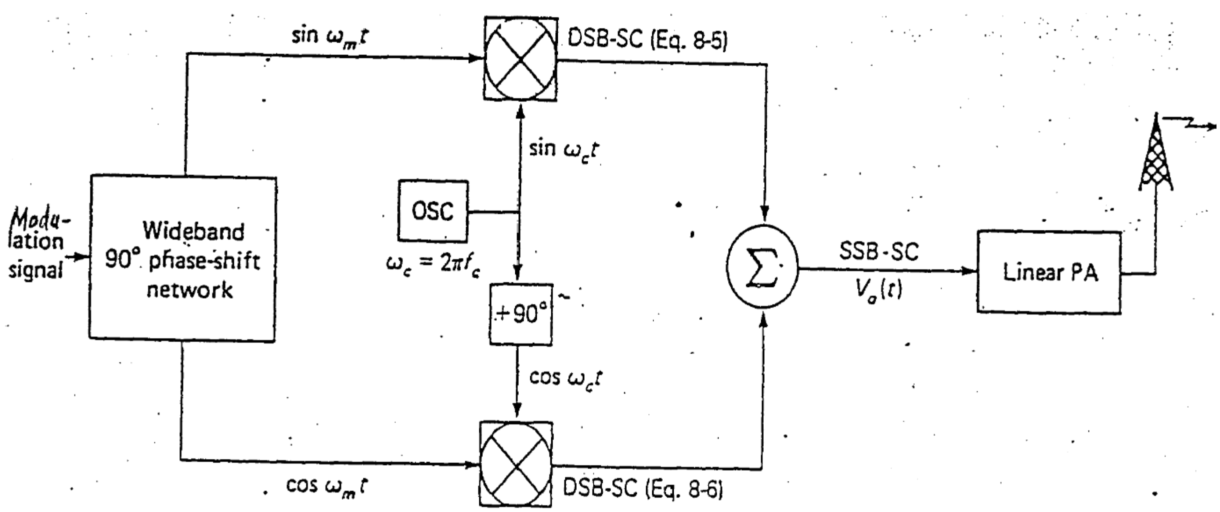


Figure1:SSB Transmitter

A double side band transmission was the first method of modulation developed and for broadcast stations, is still the most popular. Indeed, for medium and long range broadcast stations is still the most popular. In medium and long range broadcast stations is still the most popular .The reason for such wide spread use is that the receiver design can be simple and reliable. Radio is also used for communications in which the signal is addressed to a receiving station or a group of station .For this type of communication other system are used, one of which is investigated.

PROCEDURE:--

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

a) Audio input select switch in INT position.
b) Mode switch in SSB position.
c) Output amplifier's gain pot in fully clockwise position.
d) Speaker switches in OFF position.

2. Turn on power to the**ST2201** board.

3. Turn the audio oscillator block's amplitude pot to its fully clockwise (MAX) position, and examine the block's output (t.p.14) on an oscilloscope. This is the audio frequency sine wave which will be used as out 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.

Note: *That the amplitude of this audio modulating signal can be reduced to zero, by turning the audio oscillator's pot to its fully counter-clockwise (MIN) position. Leave the amplitude pot on its full clockwise position, and adjust the frequency pot for an audio frequency of 2 KHz, approx. (mid-way).*

4. To achieve signal-side band amplitude modulation , we will utilize the following three

Blocks on the **ST2201** module.

a) Balanced modulator.
b) Ceramic band pass filter
c) Balancedmodulator&bandpassfiltercircuit2.

We will now examine the operation of each of these blocks in detail.

5. Monitor the two inputs to the balanced modulator block, att.p.15 and t.p.6 noting that:

a) The signal t.p. 15 is the audio frequency sine wave from the audio oscillator block. This is the modulating input to the balanced modulator block.

b) The signal at t.p. 6 is a sine wave whose frequency is slightly less than 455 KHz. It is generated by the 455 KHz oscillator block, and is the carrier input to the balanced modulator block.

6. Next, examine the output of the balanced modulator block (att.p.17), together with the modulating signal at t.p.15 trigger the oscilloscope on the modulating signal. Check that the waveforms are as shown Fig. 2.

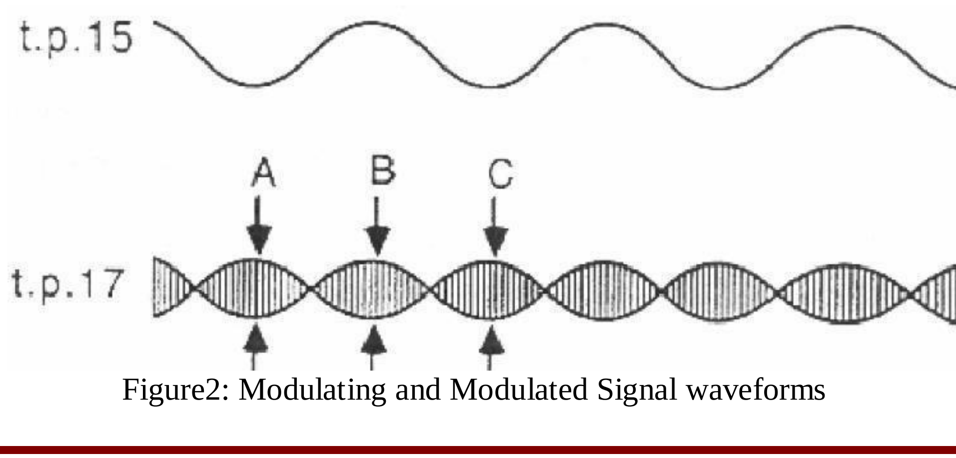


Figure2: Modulating and Modulated Signal waveforms

Note that it may be necessary to adjust the balanced modulator block's balance pot, in order to ensure that the peaks of t.p.17's waveform envelope (labeled A, B, C etc. in the above diagram) all have equal amplitude. You will recall that the waveform at t.p.17was encountered in the previous experiment this is a double-sideband suppressed carrier (DSBSC) AM waveform, and it has been obtained by amplitude-modulating the carrier sine wave at t.p. 6 of frequency fc with the audio-frequency modulating signal at t.p. 15 of frequency fm, and then removing the carrier component from the resulting AM signal, by adjusting the balance pot. The frequency spectrum of this DSBSC waveform is shown in Fig.3.

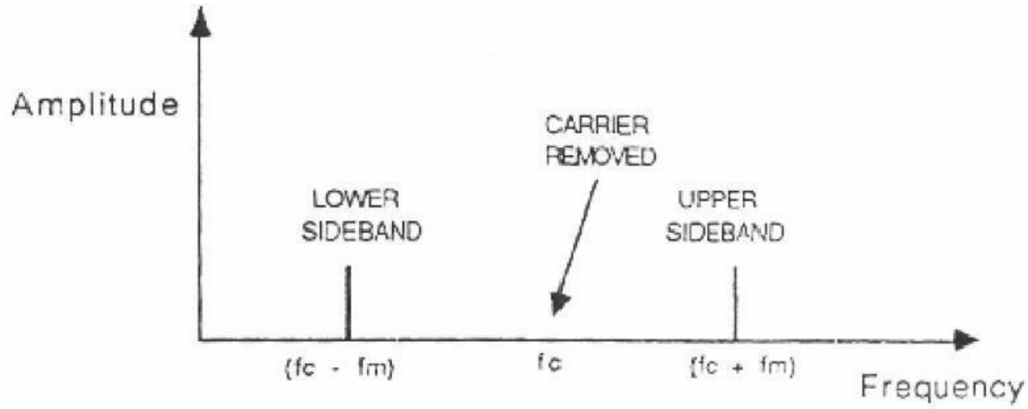


Figure3: DSBSC Side bands

7. TheDSBSCoutputfromthebalancedmodulatorblockisnextpassedontotheceramic filter block, whose purpose is to pass the upper sideband, but block the lower sideband. We will now investigate how this is achieved. First note that the ceramic band pass filter has a narrow pass band centered around 455 KHz. It was mentioned earlier that the frequency of the carrier input to the balanced modulator block has been arranged to be slightly less than 455 KHz. In fact, the carrier chosen so that, whatever the modulating frequency fm, the upper sideband (at fc+fm) will fall inside the filter's pass band, while the lower sideband (at fc-fm) always falls outside. Consequently, the upper sideband will suffer little attenuation, while the lower sideband will be heavily attenuated to such an extent that it can be ignored. This is shown in the frequency spectrum in fig 4.

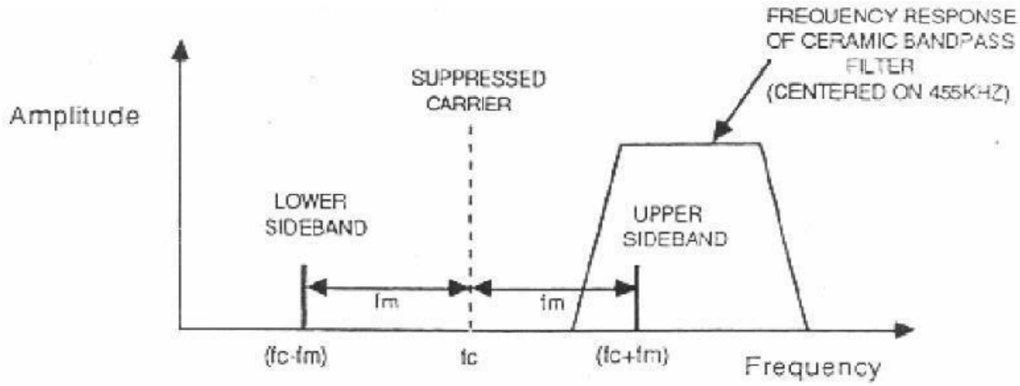


Figure4:Frequency Response of Ceramic BPF

8. Monitor the output of the ceramic band pass filter block (at t.p. 20) together with the audio modulating signal (at t.p.15) using the later signal to trigger theoscilloscope.Notethattheenvelopeofthesignalatt.p.20nowhasfairlyconstantamplitude,asshown in Fig.5.

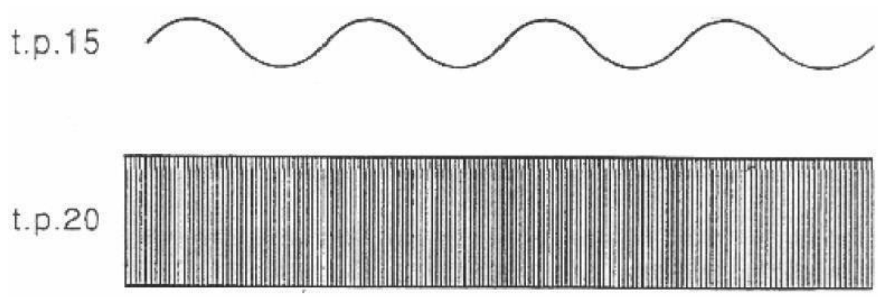


Figure5: Input Audio Signal and SSB output Signal

If the amplitude of the signal at t.p. 20 is not reasonably constant, adjust the balance pot in the balance modulator block to minimize variations in the signal's amplitude. If the constant-amplitude waveform still cannot be obtained, then the frequency of the 455 KHz oscillator needs to be trimmed.

9. Now, trigger the oscilloscope with the ceramic band pass filter's output signal (t.p.20) and note that the signal is a good, clean sine wave, indicating that the filter has passed the upper sideband only. Next, turn the audio oscillator block's frequency pot throughout its range. Note that for most audio frequencies, the waveform is a good, clean sine wave, indicating that the lower sideband has been totally rejected by the filter. For low audio frequencies, you may notice that the monitored signal is not such a pure sinusoid. This is because the upper and lower sidebands are now very close to each other, and the filter can no longer completely remove the lower sideband. Nevertheless, the lower sideband's amplitude is sufficiently small compared with the upper sideband, that its presence can be ignored. Since the upper sideband dominates for all audio modulating frequencies, we say that single sideband (SSB) amplitude modulation has taken place.

Note: *If the monitored waveform is not a good sine wave at higher modulating frequencies (i.e. when the frequency pot is near the MAX position), then it is likely that the frequency of the 455 KHz oscillator needs to be trimmed*

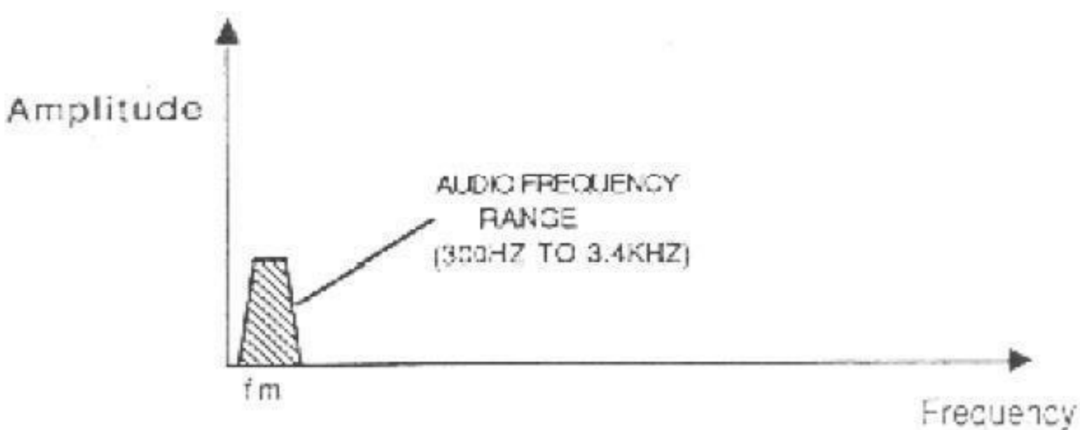
10. Note that there is some variation in the amplitude of the signal at the filter's output (t.p. 20) as the modulating frequency changes. This variation is due to the frequency response of the ceramic band pass filter, and is best explained by considering the spectrum of the filter's input signal at the MIN and MAX positions of the frequency pot, as shown in Fig. 4.

a. Modulating frequency fm=300Hz(pot in MIN position)

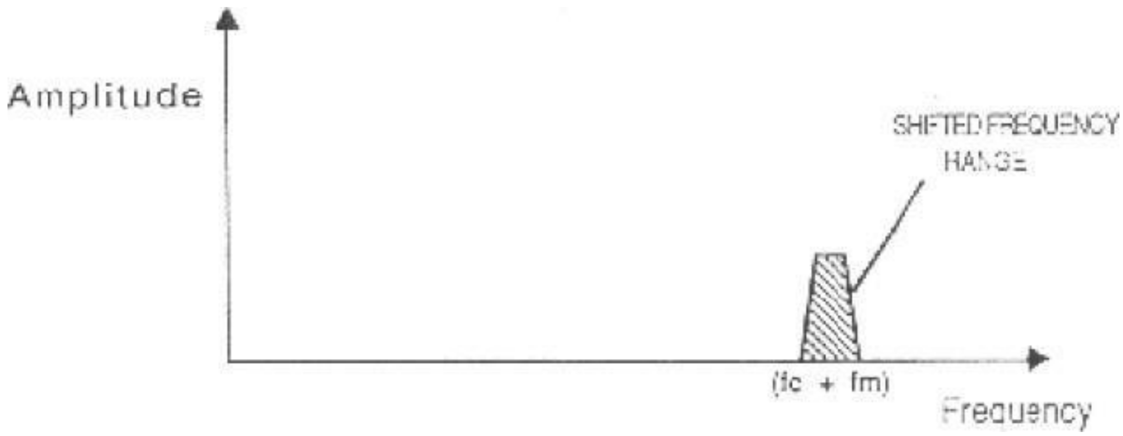
b. Modulating frequency fm=3.4KHz(pot in MAX position)

Notice that, since the upper sideband cuts rising edge of the filter's frequency response when fm=300Hz, there will be a certain amount of signal attenuation then the frequency pot is in its 'MIN' position.

11. Note that, by passing only the upper side band of frequency (fc+fm), all we have actually done is to shift out audio modulating signal of frequency fm up in frequency by an amount equal to the carrier frequency fc. This is shown in Fig.7.



(a). Range of frequencies available from audio oscillator



(b). Corresponding range of output frequencies from ceramic band pass filter block

Figure 7: Range of frequency output from audio oscillator and ceramic BPF

12. With the audio oscillator block's frequency pot roughly in its midway position (arrow head pointing towards the top), turn the block's amplitude pot to its MIN position, and note that the amplitude of the signal at the ceramic band pass filter's output (t.p. 20) drops to zero. This highlights one on the main advantages of SSB amplitude modulation if there is no modulating signal, then the amplitude of the SSB waveform drops to zero, so that no power is wasted. Return the amplitude pot to its MAX position.

13. The particular filter we are using has a pass band centered on 455 KHz, and this is why we have arranged for the wanted upper sideband to also be at about 455 KHz. However, there is a disadvantage of this type of filter is the range of frequencies that the filter will pass is fixed during the filter's manufacture, and cannot subsequently be altered. Note that since there is a large gap between the upper and lower side bands(a gap of about 910 KHz), a band pass filter with a very sharp response is not needed to reject the lower sideband, a simple tuned circuit band pass filter is quite sufficient.

14. Now examine the output of the balanced modulator & bandpass filter circuit 2 blocks (t.p.22), and check that the waveform is a good sine wave of frequency approximately 1.45MHz.Thisindicatesthatonlytheuppersidebandisbeingpassedbytheblock.Check thatthewaveformisreasonablygoodsinusoidforallaudio modulating frequencies (i.e. all positions of the audio oscillator‘s frequency pot). If this is not the case, it may be that the balance pot (in the balanced modulator & band pass filter circuit 2 blocks) needs adjusting, to remove any residual carrier component at 1 MHz. If a reasonably clean sine wave still cannot be obtained for all audio frequencies, thentheresponseofthetunedcircuitbandpassfilterneedsadjusting.Thisisachievedby adjusting transformer T4 in the balanced modulator & band-pass filter circuit 2 block When the modulating audio signal is swept over its entire range (a range of 3.4 KHz – 300 Hz = 3.1 KHz), the SSB waveform at t.p. 22 sweeps over thesamefrequency range. So single-sideband modulation has simply served to shift our range of audio frequencies up so they are centered on 1.455 MHz.

15. Monitor the 1.455 MHz SSB signal (at t.p. 22) together with the audio modulating signal (t.p. 15), triggering the scope with the later. Reduce the amplitude of the audio modulating signal to zero (by means of the audio oscillator block's amplitude pot), and notethattheamplitudeoftheSSBsignalalsodropstozero,asexpected.Returnthe amplitudepottoitsMAXposition.

16. Examine the final SSB output (at t.p. 22) together with the output from the output amplifier block (t.p. 13). Note that the final SSB waveform appears, amplified slightly, at t.p. 13. As we still see later, it is the output signal which will be transmitted to the receiver.

17. Byusingthemicrophonethehumanvoicecanbeusedastheaudiomodulatingsignal, insteadofusing**ST2201**'saudiooscillatorblock.Connectthemicrophonetotheexternal audio input on the **ST2201** board, and put the audio input select switch in the EXT position. The input signal to the audio input select may be taken from an external microphone (suppliedwiththemodule)offromacassette recorder,bychoosingtheappropriateswitch setting on the module.

RESULT:-

TheSSBsignalhasbeengeneratedusingbalancedmodulator.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshould beswitchedoffaftercompletionofexperiment

QUIZ/ ANSWERS:-

Whatisthemostcommonlyuseddemodulator?Ans.
Diode detector.

WhatisAGC?

Ans.AGCstandsfor automaticgain control.

Whatis theuseof AGC?

Ans.AGCcircuitisusedtopreventoverloadingreceiverandalsoreducetheeffectof fluctuations in the received signal strength.

WhatistherequiredoscillatorfrequencyinAMreceiver?

Ans.TherequiredoscillatorfrequencyinAMreceiverisalwayshigherthanthesignal frequency.

WhatistheuseofpilotcarrierinSSB?Ans. For
frequency stabilization.

Whatarethemethods ofSSB generation?

Ans.Frequencydiscriminationand(b)Phasediscrimination.

WhataretheadvantagesofSSBoverDSB?

Ans.1.Transmittercircuitismorestable.2.Increasedtransmissionefficiency3.Reduced BW.

Whichtypeof modulationis usedinIndiaforvideo transmission?

Ans Amplitudemodultion

WhichfilterisusedinSSBgeneration?Ans. Mechanical filters.
HowAMsignalswithlargecarrieraredetected?Ans. By using envelope detector.

EXPERIMENT No. 3

AIM:-To study envelope detector for AM signal and observe peak diagonal clipping effect.

APPARATUSREQUIRED:-(i)C.R.O.(ii)CROP robe(ii)DSB/SSB Transmitter (ST 2201) and Receiver Trainer (ST 2202) (iv) Connecting leads

THEORY:-

The AM Transmitter:
The transmitter circuits produce the amplitude modulated signals which are used to carryinformationoverthetransmissiontothereceiver.Themainpartsofthetransmitter areshowninFig.11. In Fig.11&12,wecanseethatthepeak-to-peakvoltageintheAM waveform increase and decrease in sympathy with the audio signal.

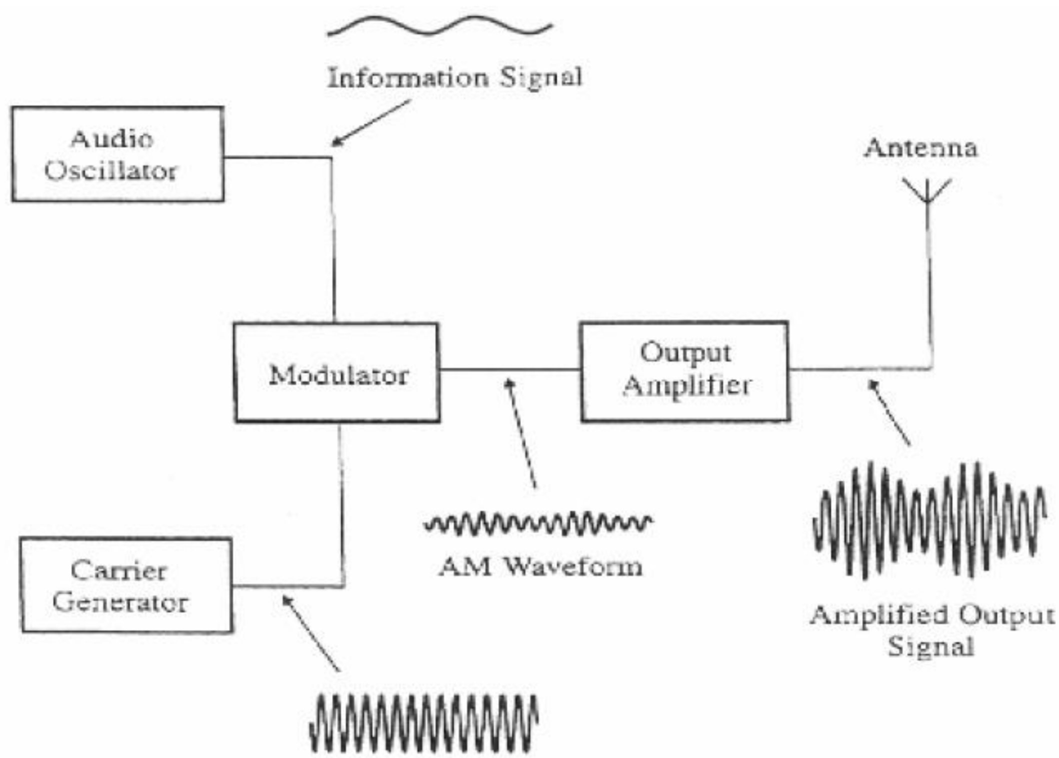


Fig.1:AMTransmitterSystem

Toemphasizetheconnectionbetweentheinformationandthefinalwaveform,aline is sometimes drawn to follow the peaks of the carrier wave as shown in Fig.12. This shape, enclosed by a dashed line in out diagram, is referred to as an 'envelope', or a 'modulation envelope'. It is important to appreciate that it is only a guide to emphasizeof the AM waveform.

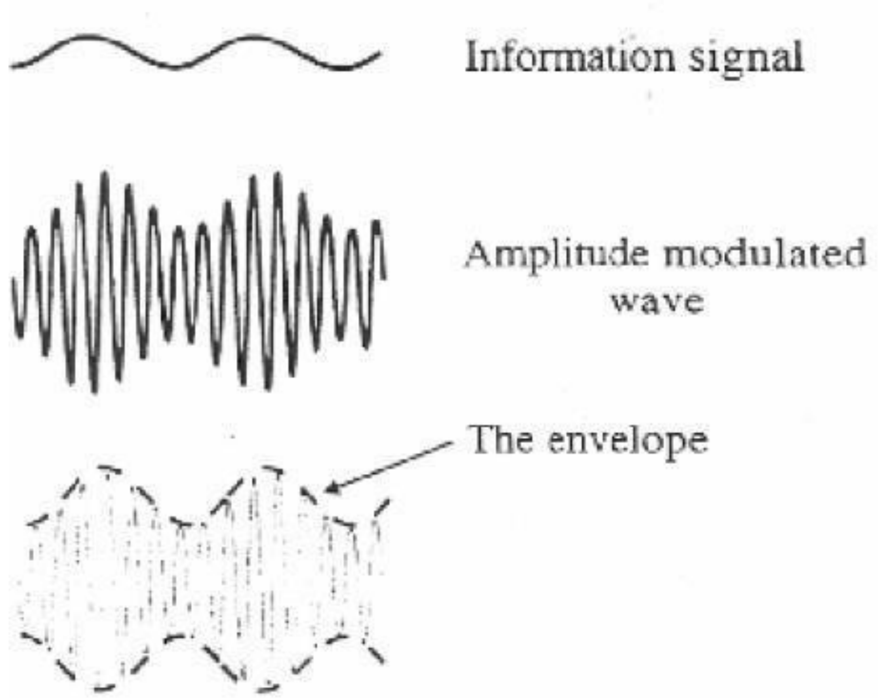


Figure2:WaveformsinAMtransmitter

*<u>AMReception:</u>*The'em'wavefromthetransmittingantennawilltraveltothereceiving antennacarryingthe informationwithit.Thestagesof AMreceptionareshown inFig. 3. :

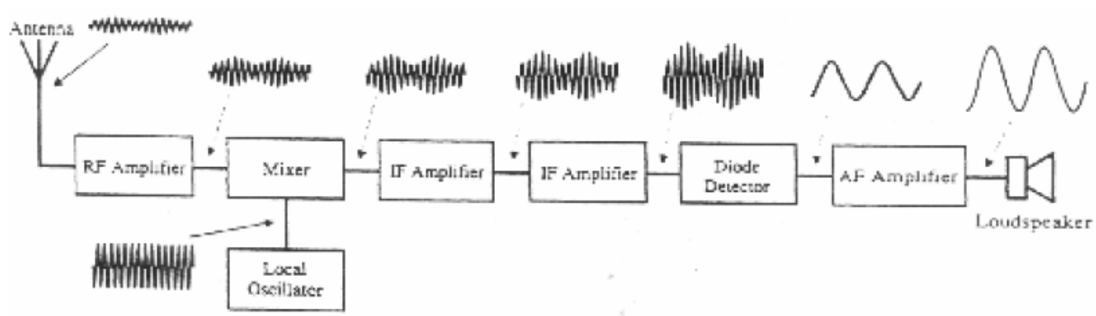


Figure3:AMReception

EnvelopeDetector:

The simplest form of envelope detector is diode detector. The function of the diode detectoristoextracttheaudiosignalfromthesignalattheoutputofthe IFamplifiers.It performs this task in a very similar way to a half wave rectifier converting an AC input to a DC output. Fig.4 shows a simple circuit diagram of the diode detector.

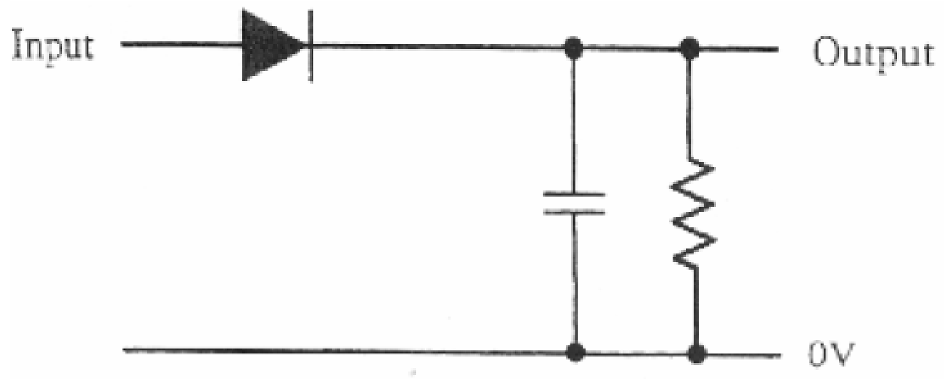
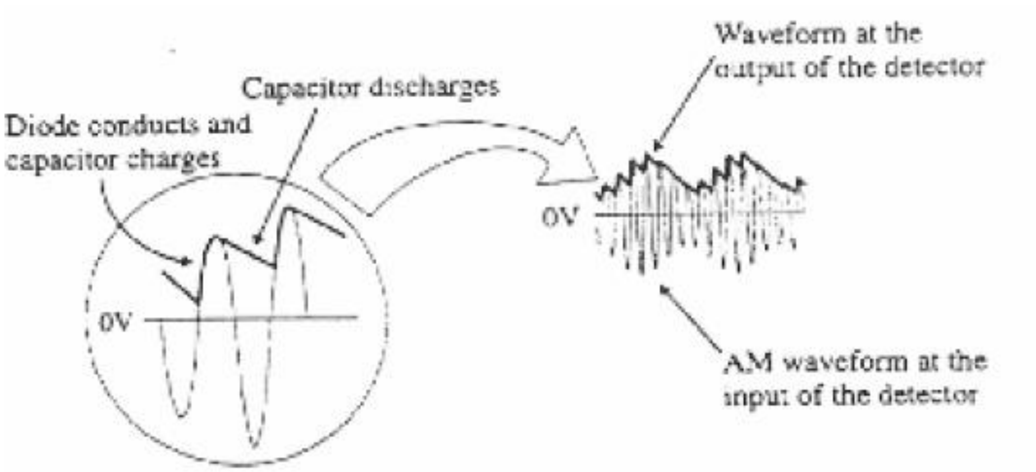


Figure4:DiodeDetector

InFig.4,thediodeconductseverytimetheinputsignalappliedtoitsanodeismore positive than the voltage on the top plate of the capacitor.

When the voltage falls below the capacitor voltage, the diode ceases to conduct and thevoltageacrossthecapacitorleaksawayuntilthenexttimetheinputsignalisable to switch it on again. See fig. 5



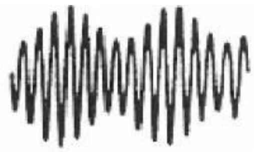
.

Fig.5ClippinginDiodeDetector

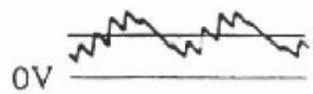
Theresultis anoutput whichcontainsthreecomponents:

1. Thewantedaudioinformation signal.
2. SomerippleattheIFfrequency.
3. ApositiveDCvoltagelevel.

Attheinputtotheaudio amplifier,alow passfilterisusedtoremovethe IFrippleanda capacitor blocks the DC voltage level. Fig.6 shows the result of the information signal passingthrough the diode detector and audio amplifier. The remaining audio signals are then amplified to provide the final output to the loudspeaker.



The input to the diode detector from the last IF amplifier



Output of diode detector includes : a DC level, the audio signal, ripple at IF frequency

Output after filtering

Figure6:OutputofDiodeDetectorandoutput Filter

<u>PROCEDURE:</u>-

1. Position the **ST2201** &**ST2202** modules, with the **ST2201** board on the left, and agap of about three inches between them.

2. Ensurethatthefollowing initialconditionsexistonthe**ST2201**board.

- **a.** Audiooscillator'samplitudepotin fullyclockwiseposition.
- **b.** Audioinputselect switchin INTposition.
- **c.** Balance pot in balanced modulator & band pass filter circuit 1 block, in fullclockwise position;
- **d.** ModeswitchinDSBposition.
- **e.** Outputamplifier'sgainpotinfullcounter-clockwise position.
- **f.** TXoutputselectswitchinANTposition:
- **g.** Audioamplifier‘svolumepot in fullycounter-clockwiseposition.
- **h.** SpeakerswitchinONposition.
- **i.** On-boardantennainverticalposition,andfullyextended.

3. Ensurethatthefollowing initialconditionsexistonthe**ST2102**board:

- **a.** RXinputselectswitchinANTposition.
- **b.** R.F.amplifier'stunedcircuitselectswitchinINT position.
- **c.** R.Eamplifier'sgainpotinfullyclock-wiseposition;
- **d.** AGCswitchinINTposition.
- **e.** Detectorswitchindiodeposition.
- **f.** Audioamplifier'svolumepot infullycounter-clockwiseposition.
- **g.** SpeakerswitchinONposition.
- **h.** BeatfrequencyoscillatorswitchinOFFposition.
- **i.** On-boardantennainverticalposition,andfullyextended.

4. Turnonpowertothemodules.

5. Onthe**ST2202**module, slowlyturn the audioamplifier's volumepot clockwise,until sounds can be heard from the on-board loudspeaker. Next, turn the vernier tuning dial until a broad cast station can be heard clearly, and adjust the volume control to a comfortable level.

Note: *If desired, headphones (supplied with the module) may be used instead of the on-boardloudspeaker.Tousetheheadphones,simplyplugtheheadphonejackintotheaudio amplifier block's headphones socket, and adjust controlled block's volume pot.*

6. Thefirststageor'frontend'ofthe**ST2202**AMreceiveristheR.Famplifierstage.This is a wide -bandwidth tuned amplifier stage, which is tuned into the wanted station by means of the tuning dial. Once it has been tuned into the wanted station, the R.F. amplifier, having little selectivity, will not only amplify, but also those frequencies that are close to the wanted frequency. As we will see later, these nearbyfrequencies will be removed by subsequent stages of the receiver, to leave only the wanted signal. Examine the envelope of the signal at the R.F. amplifier's output (at t.p. 12), with an a.c. -coupled oscilloscope channel. Note that:

a. The amplifier's output signal is very small in amplitude (a few tens of millivoltsatthemost).Thisisbecauseonestageofamplificationisnotsufficient to bring the signal's amplitude up to a reasonable level.

b. Only a very small amount of amplitude modulation can be detected, if any. This is because there are many unwanted frequencies getting through to the amplifieroutput,whichtendto'drownout'thewantedAMSignal.Youmaynoticethatthe waveformitself drifts upand down onthe scope display, indicating that thewaveform's averagelevel ischanging. Thisis dueto theoperation oftheAGCcircuit, which will be explained later.

7. The next stage of the receiver is the mixer stage, which mixes the R.F. amplifier's output with the output of a local oscillator. The Frequency of the local oscillator is also tunedbymeansofthetuningdial,andisarrangedsothatitsfrequencyisalways455KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency differenceisalwayspresent,irrespectiveofthepositionofthetuningdial,andisarranged sothatitsfrequencyisalways455KHzabovethesignalfrequencythatthe R.F.amplifieristunedto.Thisfixedfrequencydifferenceisalways present,irrespective of the position of the tuning dial, and is known as the intermediate frequency (IF for short). This frequency relationship is shown below, for some arbitrary position of the tuning dial.

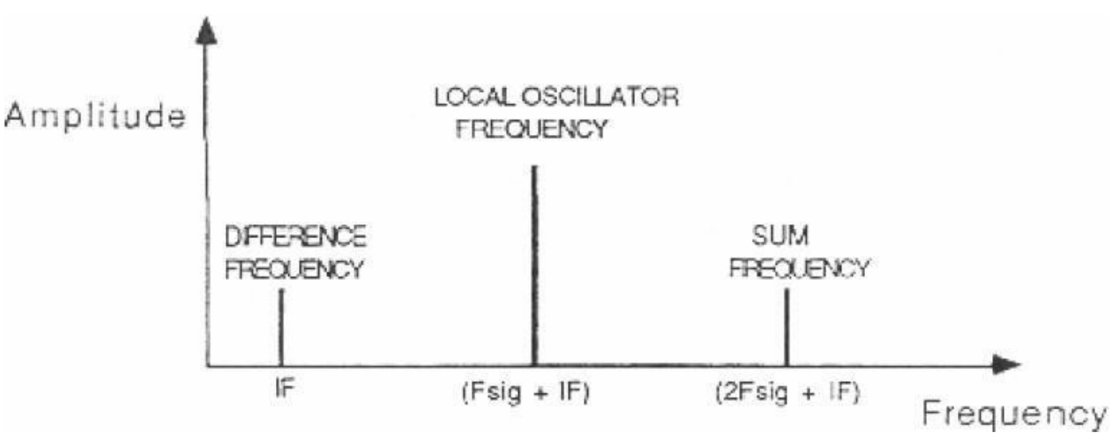


Figure7:FrequencyContentsinDSBAM

Examine the output of the local oscillator block, and check that its frequency varies as the tuning dial is turned. Re-time the receiver to a radio station.

8. Theoperationofthemixerstageisbasicallytoshiftthewantedsignaldownto theIF frequency, irrespective ofthe position ofthe tuning dial. Thisisachievedin twostages.

a. Bymixingthelocaloscillator'soutputsinewavewiththeoutputfromthe R.F.amplifierblock.Thisproducesthreefrequencycomponents:
The local oscillator frequency = (f sig + IF)
Thesumoftheoriginaltwofrequencies,fsum=(2fsig+IF) The difference between the original two frequencies,

b. By strongly attenuating all components. Except the difference frequency, IF this is done by putting a narrow-bandwidth band pass filter on themixer'soutput.

The end result of this process is that the carrier frequency of the selected AM station is shifteddownto455KHz(theIFFrequency),andthesidebandsoftheAMsignalarenow either side of 455 KHz.

9. Note that, since the mixer's band pass filter is not highly selective, it will not completelyremovethelocaloscillatorsandsumfrequencycomponentsfromthemixer's output. this is the case particularly with the local oscillator component, which is much larger in amplitude than the sum and difference components. Examine the output of the mixer block (t.p. 20) with an a.c. coupled oscilloscope channel, and note that the main frequency component present changes as the tuning dial is turned. This is the local oscillator component, which still dominates the mixer's output, in spite of being attenuated by the mixer's band pass filter.

10. Tune in to a strong broadcast station again and note that the monitored signal shows little, if any, sign of modulation. This is becausethe wanted component, whichis now at the IF frequencyof 455 KHz, is still verysmall in component, which is now at the IF frequencyof455KHz,isstillverysmallincomparisontothelocaloscillatorcomponent. Whatweneedtodonowistopreferentiallyamplifyfrequenciesaround 455 KHz, without amplifying the higher-frequency local oscillator and SUM components. This selective amplification is achieved by using two IF amplifier stages, IFamplifier1 and IF amplifier 2,which aredesigned toamplifystronglya narrow band of frequencies around 455 KHz, without amplifying frequencies on either side of thisnarrow band. These IF amplifiers are basically tuned amplifiers which have been pre-tuned to the IF frequency-they have a bandwidth just wide enough to amplify the 455 KHzcarrierandtheAM sidebandseithersideofit.Anyfrequencies outsidethisnarrow frequencyband will not be amplified. Examine the output of IF amplifier 1 (at. t.p. 24) with an a.c.-coupled oscilloscope channel, and note that:

a. Theoverallamplitudeof thesignalismuchlarger thanthesignalamplitude at the mixer's output, indicating that voltage amplification has occurred.

b. Thedominant component ofthe signal is now at 455 KHz, irrespective of any particular station you have tuned into. This implies that the wanted signal, at the IFfrequency,hasbeenamplifiedtoalevelwhereitdominatesovertheunwanted components.

c. The envelope of the signal is modulated in amplitude, according to the sound information being transmitted by the station you have tuned into.

11. Examine the output of IF amplifier 2 (t.p.28) with an a.c.-coupled oscilloscope channel, noting that the amplitude of the signal has been further amplified bythis second IF amplitude of the signal has been further amplified bythis second IF amplifier stage. IF amplifier 2 has once again preferentially amplified signals around the IF frequency (455 KHz), so that:

a. Theunwantedlocaloscillatorandsumcomponentsfromthemixerarenowso small in comparison, that they can be ignored totally,

b. Frequencies close to the I F frequency, which are due to stations close to the wanted station, are also strongly attenuated.

Theresultingsignal at theoutput of IF amplifier2 (t.p.28)is thereforecomposedalmost entirelyofa455KHzcarrier,andtheA.M.sidebandseithersideofitcarryingthewanted audio information.

12. The next step is extract this audio information from the amplitude variations of the signal at the output of IF amplifier 2. This operation is performed by the diode detector block,whoseoutputfollowsthechangesintheamplitudeofthesignalatitsinput.Tosee howthisworks,examinetheoutputofthediodedetectorblock(t.p.31),togetherwiththe output from. IF amplifier 2 (at t.p.28). Note that the signalatthediodedetector'soutput:

· Followstheamplitudevariationsoftheincomingsignalas required:

· Containssomeripple attheIFfrequencyof455KHz,and

· The signal has a positive DC offset, equal to half the average peak to peak amplitudeof the incoming signal. We will see how we make use of this offset later on, when we look at automatic gain control (AGC).

13. Thefinalstageofthereceiveristheaudioamplifierblockcontainsasimplelow-pass filter which passes only audio frequencies, and removes the high frequency ripple from the diode detector's output signal. This filtered audio signal is applied to the input of an audio power amplifier, which drives on board loudspeaker (and the headphones, ifthese areused). The final result is the sound you arelisteningto the audio signal which drives the loudspeaker can be monitored at t.p. 39 (providing that the audio amplifier block's volume pot is not in its minimum volume position). Compare this signal with that at the diode detector's output (t.p. 31), and note how the audio amplifier block's lowpass filter has'cleanedup'theaudiosignal.Youmaynoticethattheoutputfromtheaudioamplifier block(t.p.39)isinvertedwithrespecttothesignalattheoutputofthediodedetector(t.p. 31)thisinversionisperformedbytheaudiopoweramplifierIC,andinnowayaffectsthe sound produced by the receiver.

14. Nowthatwehaveexaminedthebasicprinciplesofoperationofthe**ST2202**receiver forthereceptionanddemodulationofAMbroadcastsignals,wewilltryreceivingtheAM signal from the **ST2201** transmitter. Presently, the gain of **ST2201'**s output amplifier block is zero, so that there is no output from the Transmitter. Now turn the gain pot in **ST2201**'s output amplifier block to its fullyclockwise (maximum gain) position, so that thetransmittergeneratesanAMsignal.Onthe**ST2201**module,examinethetransmitter's output signal (t.p.13), together with the audio modulating signal (t.p.1), triggering the 'scopewiththesignal'.Since**ST2201**TXoutputselectswitchisintheANTposition,the AM signal at t.p.13 is fed to the transmitter's antenna.Prove this by touching **ST2201**'s antenna, and nothing that the loading caused by your hand reduces the amplitude of the AM waveform. at t.p.13. The antenna will propagate this AM signal over a maximum distanceofabout1.4feet.WewillnowattempttoreceivethepropagatedAMwaveform with the **ST2201/ ST2202** board, by using the receiver's on board antenna.

Note: *If more than one* **ST2201** *transmitter/receiver system is in use at one time, it is possible that there may be interference between nearby transmitters if antenna propagation is used. To eliminate this problem, use a cable between each* *transmitter/ receiver pair, connecting it between* **ST2201**'*s TX output socket and* **ST2201/ST2202**'*s RX input socket. If you do this, make sure that the transmitter's TX output select switch, and the receiver's RX input select switch, are both in the SKT position, then follow the steps below as though antenna propagation were being used.*

15. OntheST2201module,turnthevolumepot(intheaudioamplifierblock)clockwise, until you can hear the tone of the audio oscillator's output signal, from the loudspeaker on the board.

Note: If desired, headphones may be used instead of the loudspeaker on the board. To use the headphones, simply plug the headphone jack into the audio amplifier block's headphones socket, and put the speaker switch in the OFF position. The volume from the headphones is still controlled by the block's volume pot. Turn the volume pot to the full counter-clockwise (minimum volume) position.

16. On the **ST2201/ST2202** receiver, adjust the volume pot so that the receiver's output canbeclearlyheard.Thenadjustthereceiver's tuningdial untilthetonegeneratedatthe 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. Check that you are tuned into the transmitter's output signal, byvarying **ST2201**'s frequency pot in the audio oscillator block, and nothing that the tone generated by the receiver changes.

The **ST2201/2202** receiver is now tuned into AM signal generated by the **ST2201** transmitter. Briefly check that the waveforms, at the outputs of the following receiver blocks, are as expected:

R.F.Amplifier(t.p.12)
Mixer (t.p.20)
I.F.Amplifier1(t.p.24)
I.F. Amplifier 2 (t.p.28)
Diode Detector (t.p.31)
AudioAmplifier(t.p.39)

17. By using the microphone, the human voice can be used as transmitter's audio modulating signal, instead of using **ST2201**'s audio oscillator block. Use DSB and not DSBSC. Connect the microphone's output to the external audio input on the **ST2201** board, and put the audio input select switch in the EXT position.

18. In theoutput ofdiodedetectorpeak diagonalclippingcan beobserved at lowvalues of time constant of tuning circuitry.

RESULT:-

AM signal has been demodulated using envelope detector and peak diagonal clippingeffect has been observed.

PRECAUTIONS:-

1. Donot useopen endedwiresforconnecting230 Vpowersupply.
2. Beforeconnectingthepowersupplyplugintosocket,ensurepowersupplyshouldbe switched off.
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.

5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

Whatisamplitudemodulation?
Ans. Amplitude Modulation is a process in which the amplitude of the carrier is madeproportional to the instantaneous amplitude of the modulating signal.
Whicharethethree discretefrequenciesinAM?
Ans.(1)Carrierfrequency(2)lowersidebandfrequency(3)uppersidebandfrequency
Q.3HowmanysidebandsinAM?
Ans.Therearetwosidebandsin AMi.e.LSBandUSB
Q.4.WhichcircuitisusedasLPF?
Ans. R-C circuit.
Q. 5.WhicharethetwomethodsofAMgeneration?
Ans. (1) single sideband (2) double sideband
Whatisdiagonalclipping?
Ans.Distortioncausedbecauseofsmallvalueoftimeconstantoftunedcircuitiscalled diagonal clipping.
WhatistheunitofmodulationindexinAM? Ans. It is unit less.
Wherethemodulationindexlies?
Ans.modulationindex always liesbetween0and 1.Morethan1isovermodulation.
Whathappensincaseofovermodulation? Ans. The wave will get distorted.
HowDSBSCcanbeconvertedintoconventionalAM? Ans. By carrier reinsertion.

EXPERIMENT No. 4

AIM:- To generate Frequency modulated signal using Voltage Control Oscillator.

APPARATUS REQUIRED:- (i)C.R.O. (ii) CRO Probe (ii) FM Modulation and Demodulation Trainer (ST 2203) (iv) Connecting leads

THEORY:-

Frequency modulation is a form of angle modulation in which the amplitude of the modulatedcarrieriskeptconstantwhileitsfrequencyanditsrateofchangearevariedby themodulatingsignal.InFMtheinstantaneousangularfrequency$W_i$isvariedlinearlyin accordance with the instantaneous magnitude of base band signal X(t) , aboutan un-modulated carrier frequency (also called as resting frequency) Wc and the rate at which the carrier shifts from its resting point to its non resting point is determined bythe frequencyofmodulatingsignalwhilekeepingtheamplitudeofthecarrierwaveconstant.

CarriersignalC(t)=ASin($W_c t + \theta_0$)=ASinΦ......................... (1)

whereWcisthefrequencyofCarrierwaveinradians/secondand

Φinradians=Totalphaseangleoftheunmodulatedcarrier=($W_c t+\theta_0$)….(2)

InFMwhiletheamplitudeAremainsconstant, instantaneousvalueofΦchanges.IfW_i (t)=Instantaneousvalueofangularvelocity,and and

Φ_i= Instantaneous phase angle of FM wave,

then$W_i(t)=d\,\Phi_i/dt$, ... (3)

and$\Phi_i=\int W_i(t)dt$................................ (4)

ThereforeFMwavecanberepresented asS(t)=ASin Φ_i .... (5)

ModulatingvoltageSignal =X(t)volts .... (6)

Theninstantaneous angularfrequencyofanFMsignalisgivenby

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

whereK_f=Constantofproportionality=frequencysensitivityofthemodulator inHertzper volt

ThereforeFrequencyvariation=$|K_f X(t)|$ .... (8)

SincethevalueofW$_c$isassumed to befixed,

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

<u>FrequencyDeviation</u>Itistheamountbywhichcarrierfrequencyisvariedfromits unmodulated value and it is same as frequency variation.

MaxFrequencydeviation$\Delta W=|K_f X(t)|_{max}$........(10)

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

Maxiumalloweddeviation=75khz

<u>Frequency Modulation Index m_f</u>- It is the ratio of frequency deviation ΔW in rad/sec to theangularfrequencyofmodulatingsignalW_m orfrequencydeviationinHertz/sectoto the modulating frequency in Hertz/sec.

Thus$m_f=\Delta W/W_m=\delta/W_m$ifδis givenin rad /Sec .... (11)

Ifδisgivenin Hertz/Secthen $m_f=\delta/f_m$........ (12)

Mathematicalexpressionfor FMwave

$$S(t)=ASin\Phi_i=ASin[W_ct+K_f\int X(t)dt]\ldots\ldots (13)$$

ForSingletoneFM

$$X(t)=V_mCosW_ct\ldots\ldots (14)$$

Thus $\Phi_i=W_ct+K_f\int V_mCosW_mtdt=W_ct+\frac{K_fV_m}{W_m}SinW_mt$

$=W_ct+\frac{\Delta W}{W_m}SinW_mt=W_ct+m_fSinW_mt$

Thus $S(t)=ASin[W_ct+K_f\int X(t)dt]$

$$=ASin\,[W_ct+m_fSin\,W_mt]=\ldots\ldots (15)$$

DeviationRatio

Itistheratioofdeviationincarrierfrequencytothemaximummodulatingfrequency.

In single tone FM, modulation index and the deviation ratio will be the same. If themodulatingsignal(AF)is15kHZatacertainamplitudeandthecarriershiftis75kHZ, 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 frequencycomponent. Thedeviation ratio in this case will not be equal to anyparticular modulation index.

FrequencySpectrum

Analysisofequation(15)whichisasinefunction ofanothersinefunctionshows:

$$\begin{aligned}S(t)=A\{&J_0.(m_f)sinW_ct+J_1(m_f)\{sin(W_ct+W_mt)+sin(W_ct-W_mt)\}\\&+J_2(m_f)\{sin(W_ct+2W_mt)+sin(W_ct-2W_mt)\}\\&+J_3(m_f)\{sin(W_ct+3W_mt)+sin(W_ct-3W_mt)\}\\&+J4(m_f)\{sin(W_ct+4W_mt)+sin(W_ct-4W_mt)]+\ldots\ldots]\end{aligned}$$

The output consists of a carrier and an apparently infinite number of pairs of side bands havinganamplitudecoefficient$J_n(m_f)$,whichisaBesselfunctionof$m_f$andoftheorder ndenotedbythesubscript.Valuesofthesecoefficientsareavailablereadilyintableform as well as in graphic form as shown below.

Analysis of FM waveforms Wave forms of carrier, modulating signal, modulated signal aswellasgraphicalformofplotof$J_n(m_f)$versusvaluesof$m_f$areshownbelow.Itcanbe seen that:

1. Unlike AM, FM output contains carrier component of frequency$f_c$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 off_m.
2. ThevaluesofeachJ$_n$coefficient,whichrepresenttheamplitudeofapairofsidebands, fluctuates on either side of zero, gradually diminishes with increasing value of m_f like damped oscillations. The values of $J_n$coefficients also eventually decrease, but

only pastincreasedvalueofn.Asthevalueofmfincreases,thevalueofJ$_0$decreases from1 and the values of J_1 to J_n increases from0 and fluctuate around mean value of 0.

3. Themodulationindexdetermineshowmanysidebandcomponentshavesignificant values.

4. Unlike AM, in FM, while the amplitude of modulated signal remains constant, the value of the carrier component decreases with increase in $m_f$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 m_f.This increases the immunity to noise in FM unlike AM.

5. As the value of m_f increases, The carrier component becomes zero for certain values ofmodulationindex,calledeigenvalueswhichareapproximately2.4,5.5.8.6.11.8andso on. These disappearances of carrier for specific values of $m_{f,}$ form a handy basis for measuring deviation.

BLOCK DIAGRAM:-

Theaudiooscillatorsuppliestheinformationsignalandcould,ifwewish,bereplacedby 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 combinethecarrier wave and the information signal in much the same wayas 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.

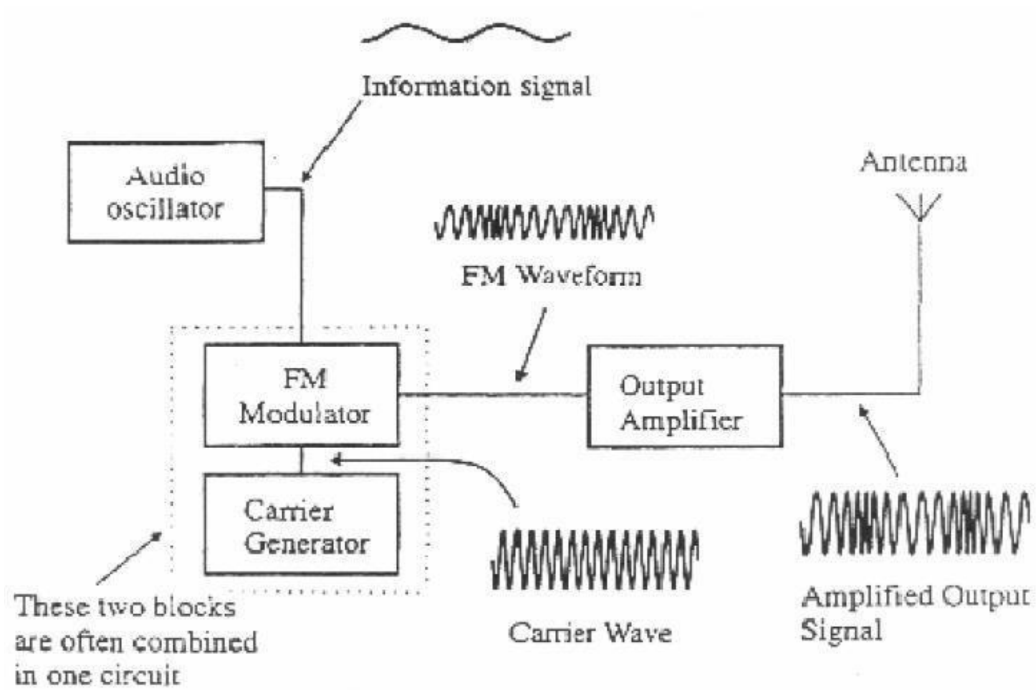


Figure1:FM Transmitter

ControllingtheVCO:

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 waveand,togetherwiththeVCOoutput,formsthetwoinputstoanExclusive-ORgate.
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 (see fig. 34). 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:--

1. Ensurethatthefollowing initialconditionsexistonthe**ST2202**board.
   a. Allswitchedfaultsoff.
   b. Amplitudepot(inmixeramplifierblock)infullyclockwiseposition.
   c. VCOswitch in'ON'position.
2. Turn the audio oscillator block's amplitude pot to its fully clockwise position, and examinetheblock'soutputt.p.1onanoscilloscope.Thisistheaudiofrequencysinewave, which will be used as our modulating signal. Note that the sine wave's frequencycan be adjustedfromabout300Hztoapproximately3.4KHz,byadjustingtheaudiooscillator's frequency pot.
3. ConnecttheoutputofaudiooscillatortoVCOsection‘sMODInsocket.
4. TurnONthepowersupply.
5. ObservethemodulatingsignalandmodulatedoutputattheVCO‘sMODOUTsocket by using CRO.
4. Calculate $m_f = \delta / f_m$.
5. Varythemodulatingfrequencykeepingcarrierfreqconstantandrepeatsteps3&4.
6. Varythecarrierfrequencykeepingmodulatorfreqconstantand repeatsteps3& 4.
7. Tabulatetheresults.

OBSERVATIONTABLE:-

1.				
2.				
3.				

SAMPLECALCULATION:-

$$m_f = \delta / f_m$$
$$= 2 \times 8.3 \times 10^3 / 1000$$
$$= 16.6$$

WAVEFORMS:-

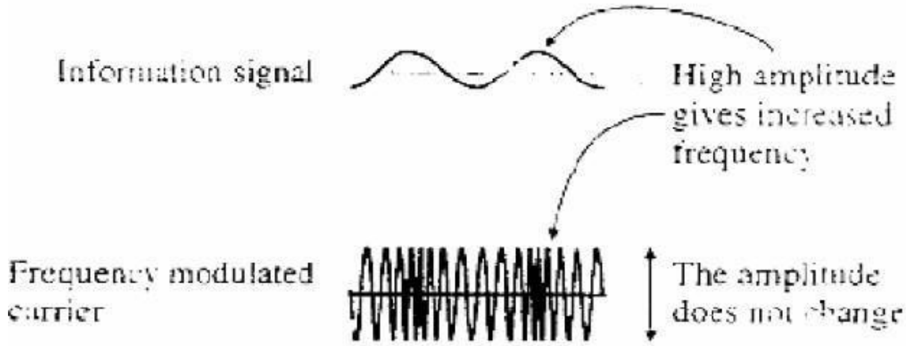


Figure2: Modulating and FM Modulated signal

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:-

HowmanytypesofFMarethere?Writetheir names.
Ans.TheretwotypesofFMi.e.narrowbandFMandwideband FM.
Whatfrequencydeviation inFM?
Ans. The maximum change in instantaneousfrequency from the average is known asfrequencydeviation.
Whichistheusefulparameterfordetermination of bandwidth?
Ans.Frequencydeviation istheusefulparameterfordetermination ofbandwidth.
Howmanysidebands aretherein FM?
Ans.Theoretically,numbersidebandsinFMareinfinite.
Whichsidebandsareignoredin FM?
Ans.Thesidebandswith smallamplitudeareignoredin FM.
Whicharesignificantsidebands?
Ans.Thesidebands havingconsiderable amplitudes i.e.morethanorequal to1%ofthe carrier amplitude are known as significant sidebands.
whatisCCIR?
Ans.CCIRstandsforConsultativeCommitteeforInternational Radio.
WhatistheindirectmethodofFMgeneration? Ans.
Armstrong method.

WhatisthedirectmethodofFMgeneration? Ans.
The parameter variation method.
What isVCO?
Ans.VCOstandsforvoltagecontrolledoscillatorwhosefrequencyiscontrolledby modulating voltage.

EXPERIMENT No. 5

AIM:-To generate FM signal using Varactor & reactance modulation.

APPARATUS REQUIRED:-(i)C.R.O. (ii) CRO Probe (ii) FM Modulation and Demodulation Trainer (ST 2203) (iv) Connecting leads

THEORY:

FM Using Varactor Modulator:
Thevariationsincapacitanceformpartofthetunedcircuitthatisusedtogeneratethe FM signal to be transmitted. Varactor modulator is shown in fig 1.

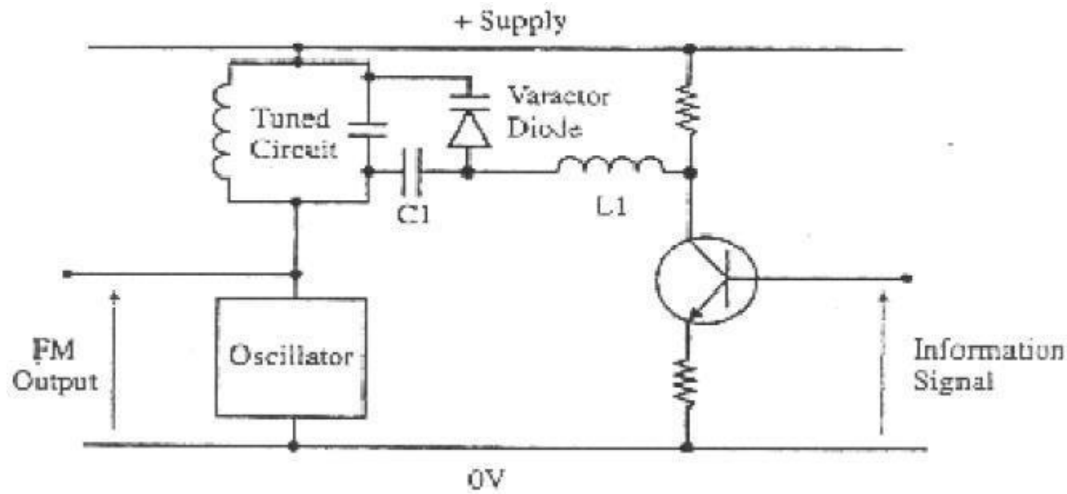


Figure1: FM generation usingVaractor Modulator

We can see the tuned circuit which sets the operating frequency of the oscillator and the varactor which is effectively in parallel with the tuned circuit. Two other components which may not beimmediatelyobviousareC1 and L1.C 1 is a DC blocking capacitor to provide DC isolation between the oscillator and the collector of the transmitter. L1 is an RFchokewhich allowstheinformation signal through to thevaractorbut blockstheRF signals.

The operation of the varactor modulator:

1. The information signal is applied to the base of the input transistor and appears amplified and inverted at the collector.

2. This low frequency signal passes through the RF choke and is applied across the varactor diode.

3. The varactor diode changes its capacitance in sympathy with the information signal and therefore changes the total value of the capacitance in the tuned circuit.

4. The changing value of capacitance causes the oscillator frequency to increase and decrease under the control of the information signal. The output is therefore a FMsignal.

Before we start the study of varactor/ reactance modulation techniques we shall study a simple VCO circuit.

Simply connect the audio output to the socket labeled VCO modulation in and observe the FM modulated waveform on the oscilloscope at the VCO modulation out terminal. Keep the amplitude of audio output to approx 4 V p-p and frequency 2 kHz approx. ObserveastableFMmodulatedwaveformonCRONowturnthetimebasespeedofCRO little higher and you will observe the same waveforms as under (like Bessel function).

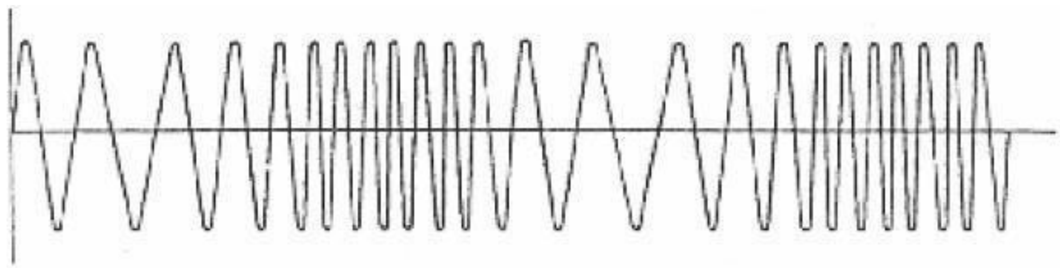


Figure2:FMmodulatedwave

Nowdisconnecttheaudioamplifier'soutputfrommodulationINandconnectittoaudio IN,keep thereactance/varactor switch in varactor position.Observetheoutput of mixer /amplifiercircuit.KeeptheoscilloscopeinX10positionnowobservethefullwaveform byshiftingtheXposition.Itisasshowninfig.Marktheresemblancebetweentheoutput of VCO and the Varactor modulator. They are same. The freq. modulation in VCO was morebecausetheFrequencydifferencebetweenthecarrierandthemodulatingsignalwas very less.

<u>FM Using Reactance Modulator:</u> In fig. 3, the left hand half is the previous varactor modulator simply an oscillator and a tuned circuit, which generates the un-modulated carrier. The capacitor C and the resistor R are the two components used for the phase shifting, and together with the transistor, form the voltage controlled capacitor. This voltage-controlled capacitorisactuallyinparallel withthetunedcircuit.Thisisnoteasy to see but figure 18 may be helpful. In the first part of the figure, the capacitor and associated components have been replaced by the variable capacitor, shown dotted.

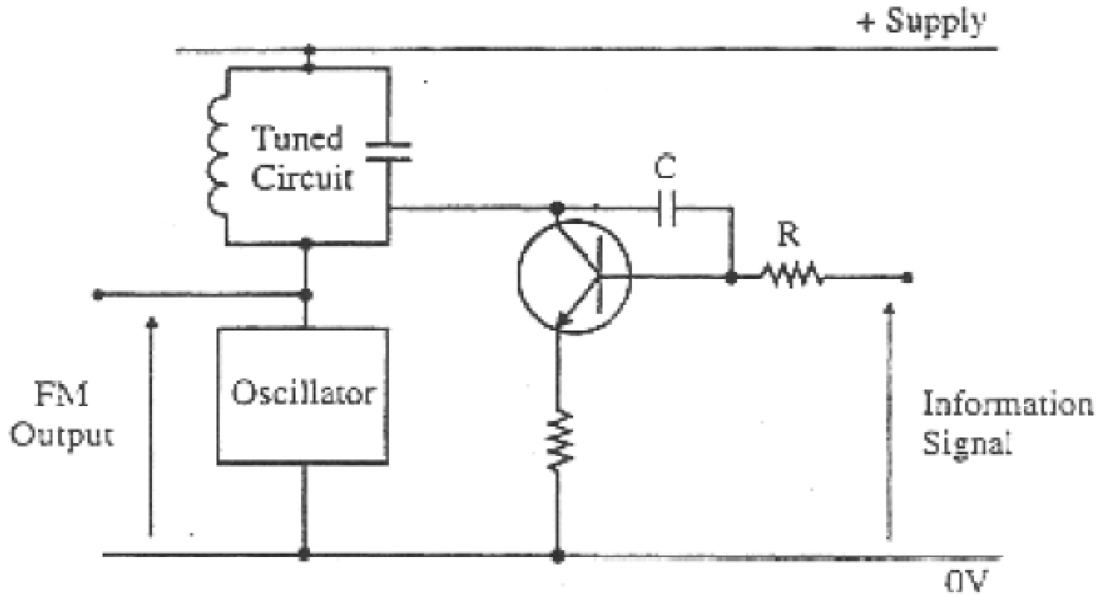


Figure3: FM using Reactance Modulation.

In the next part, the two supply lines are connected together. We can justify this by saying that the output of the DC power supply always includes a large smoothing capacitor to keep the DC voltages at a steady value. This large capacitor will have a very low reactance at the frequencies being used in the circuit less than a milli ohm. We can safely ignore this and so the two supply lines can be assumed to be joined together. Remember that this does not affect the DC potentials, which remain at the normal supply voltages. If the two supply voltages are at the same AC potential, the actual points of connectiondonotmatterandsowecanredrawthecircuitasshowninthethirdpart.

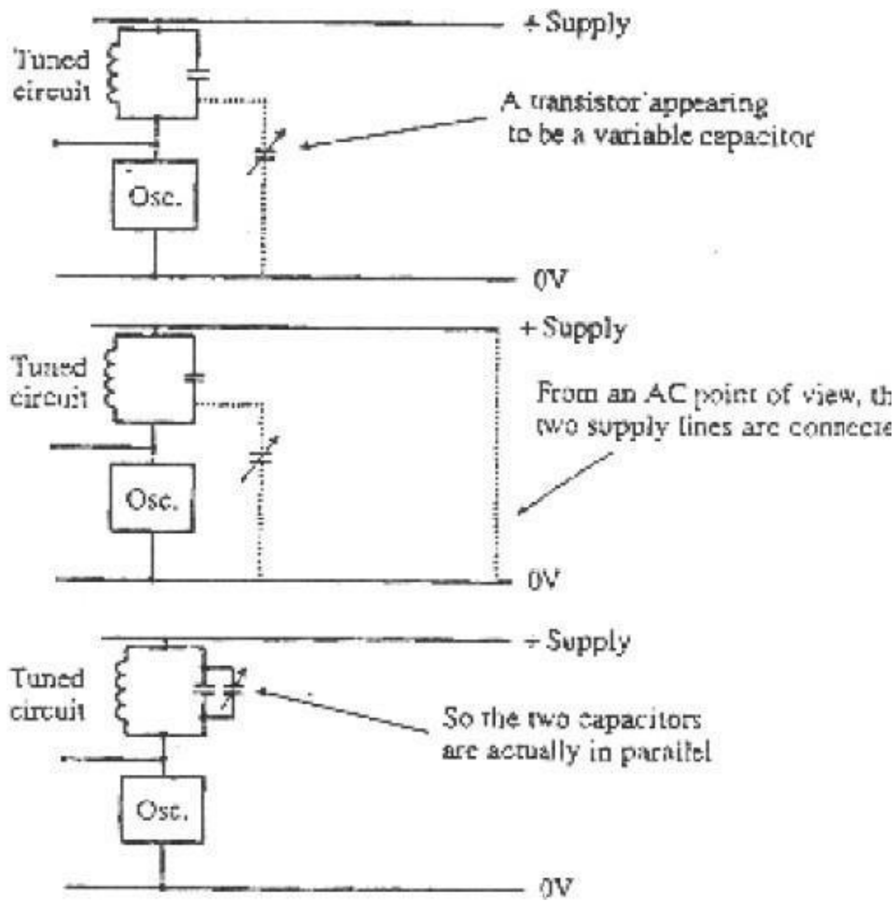


Figure4: VCO using capacitor

Operation of the Reactance Modulator :

1. The oscillator and tuned circuit provide the un-modulated carrier frequency and this frequency is present on the collector of the transistor.

2. Thecapacitorandtheresistorprovidethe90°phaseshiftbetweenthecollectorvoltage and current. This makes the circuit appear as a capacitor.

3. The changing information signal being applied to the base has the same effect as changing the bias voltage applied to the transistor and, this would have the effect of increasing and decreasing the value of this capacitance.

4. Asthecapacitanceiseffectivelyinparallelwiththetunedcircuitthevariationsinvalue will cause the frequency of resonance to change and hence the carrier frequency will be varied in sympathy with the information signal input.

BLOCKDIAGRAM:-

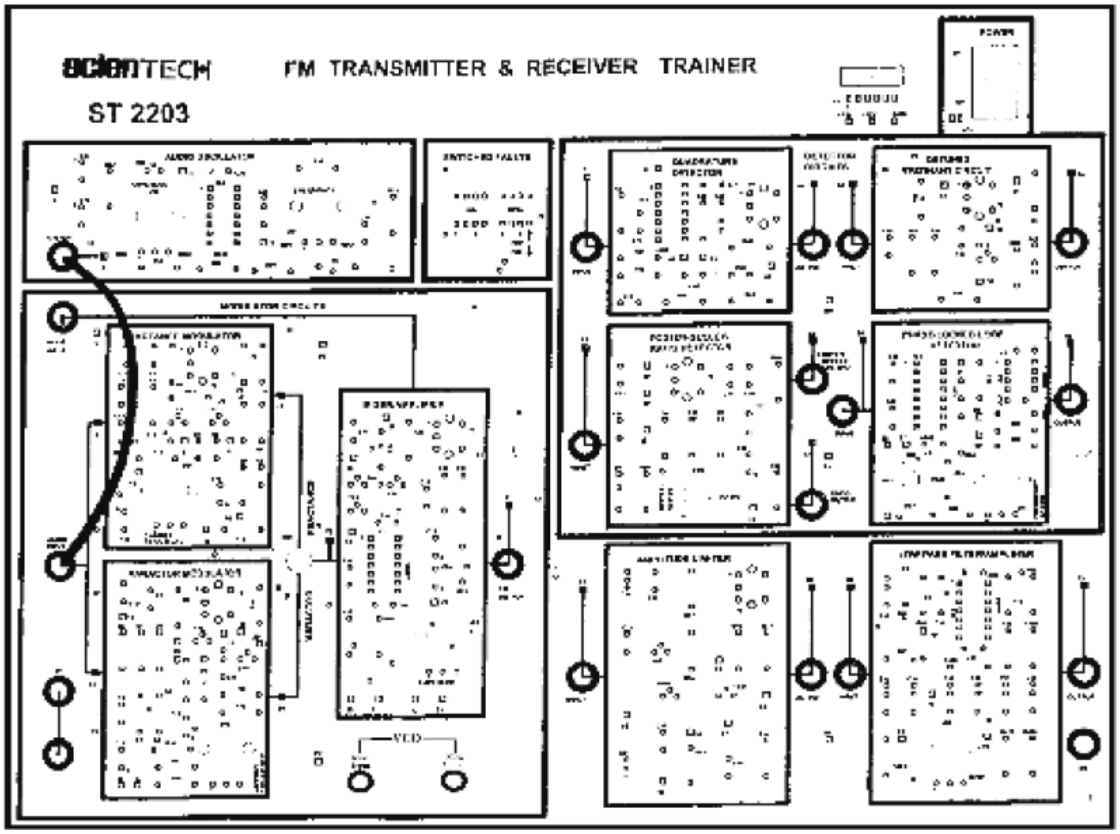


Figure3: Block Diagram of FM Trainer kit

PROCEDURE:-

1. Ensure that the following initialconditionsexistonthe**ST2202**board.

a. All switched faults off.

b. Amplitude pot (in mixer amplifier block) in fully clockwise position.

c. VCO switch (in phase locked loop detector block) in 'OFF' position.

2. Make the connections as shown in fig 3.

3. Switch ‘on 'the power.

4. Turn the audio oscillator block's amplitude pot to its fully clockwise position, and examinetheblock'soutputt.p.1onanoscilloscope.Thisistheaudiofrequencysinewave, which will be used as our modulating signal. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately3.4KHz, by adjusting the audio oscillator's frequencypot.Notealsothattheamplitudeofthismodulatingsignalisadjustedbyaudio oscillator amplitude pot. Leave the amplitude pot in min. position.

5. Connect the output socket of the audio oscillator block to the audio input socketof the modulator circuit‘s block.

For FM Varactor Modulator

6. Set the reactance / varactor switch to the varactor position. This switch selects the varactor modulator and also disables the reactance modulator to prevent any interference between the two circuits.

7. The output signal from the varactor modulator block appears at t.p. 24 before being buffered and amplified by the mixer / amplifier block, any capacitive loading (e.g. due to oscilloscope probe) may slightly affect the modulators output frequency. In order to avoid this problem we monitor the buffered FM output signal the mixer / amplifier block at t.p.34.

8. Put the varactor modulator's carrier frequency pot in its midway position, and then examine t.p. 34. Note that it is a sine wave of approximately 1.2 Vp-p, centered on 0V. This is our FM carrier, and it is un-modulated since the varactor modulators audio input signal has zero amplitude.

9. The amplitude of the FM carrier (at t.p.34) is adjustable by means of the mixer/amplifier block's amplitude pot, from zero to its pot level. Try turning this pot slowly anticlockwise, and note that the amplitude of the FM signal can be reduced to zero. Return the amplitude pot to its fully clockwise position.

10. Try varying the carrier frequency pot and observe the effects.

11. Also, see the effects of varying the amplitude and frequency pots in the audio oscillator block.

12. Turn the carrier frequency pot in the varactor modulator block slowly clockwise and note that in addition to the carrier frequency increasing there is a decrease in the amount of frequency deviation that is present.

13. Return the carrier frequency pot to its midway position, and monitor the audio input (att.p.6) and the FM output (atp.34) triggering the oscilloscope on the audio input signal. Turn the audio oscillator's amplitude pot throughout its range of adjustment, and note that the amplitude of the FM output signal does not change. This is because the audio information is contained entirely in the signals frequency and not in its amplitude.

14. By using the optional audio input module **ST2108** the human voice can be used as the audio modulating signal, instead of using **ST2203**'s audio oscillator block. If you have an audio input module, connect the module's output to the audio input socket in the modulator circuit‘s block. The input signal to the audio input module may be taken from an external microphone be (supplied with the module) or from a cassette recorder, by choosing the appropriate switch setting on the module.

For FM Reactance Modulator:

6. Put the reactance /varactor switch in the reactance position. This switches the output of the reactance modulator through to the input of the mixer/amplifier block ~ and also switches off the varactor modulator block to avoid interference between the two modulators.

7. The output signal from the reactance modulator block appears at tp.13, before being buffered and amplified by the mixer/amplifier block. Although the output from the reactance modulator block can be monitored directly at tp.13, any capacitive loading affect this point (e.g. due to an oscilloscope probe) may slightly affect the modulator's output frequency. In order to avoid this problem we will monitor the buffered FMoutput signal from the mixer/amplifier block at t.p. 34.

8. Put the reactance modulator's pot in its midway position (arrow pointing towards top of PCB) then examine t.p. 34. **Note** that the monitored signal is a sine wave of approximately1.2V peak/peak centered on 0 volts D.C. This is our FM carrier, and it is presently un-modulated since the reactance modulator's audio input signal has, zero amplitude.

9. The amplitude of the FM carrier (at t.p.34) is adjustable by means of the mixer/amplifierblock's amplitudepot, fromzero to itspresent level.Tryturningthispot slowlyanticlockwise,andnotethattheamplitudeoftheFMsignalcanbereducedtozero. Return the amplitude pot to its fully clockwise position.

10. The frequency of the FM carrier signal (at t.p.34) should be approximately 455Khz at the moment This carrier frequency can be varied from 453Khz to 460Khz (approx.) byadjustingthecarrierfrequencypotinthereactancemodulatorblock.Turnthispotover itsrangeofadjustmentandnotethatthefrequencyofthemonitoredsignalcanbeseento vary slightly. Note also that the carrier frequency is maximum when the pot is infully clockwise position.

11. Try varying the amplitude & frequency pot in audio oscillators block, and also sees the effect of varying the carrier frequency pot in the mixer/amplifiers block.

12. Monitor the audio input (at t.p.6) and the FM output (at t.p.34) triggering the oscilloscope on the audio input signal. Turn the audio oscillator's amplitude pot throughout its range of adjustment and note that the amplitude of the FM output signal doesnotchange.Thisisbecausetheaudioinformationiscontainedentirelyinthesignal's frequency, and not in its amplitude.

RESULT:-

Frequencymodulatedsignalis generatedbyusingvaractorandreactancemodulator.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

HowmanytypesofFMarethere?Writetheir names.

Ans.TheretwotypesofFMi.e.narrowbandFMandwideband FM.

Whatfrequencydeviation inFM?

Ans. The maximum change in instantaneousfrequency from the average is known asfrequencydeviation.

Whichis theusefulparameterfordetermination of bandwidth?

Ans.Frequencydeviation istheusefulparameterfordetermination ofbandwidth.

Howmanysidebands aretherein FM?

Ans.Theoretically,infinitenumbersofsidebandsaretherein FM.

Whichsidebandsareignoredin FM?

Ans.Thesidebandswith smallamplitudeareignoredin FM.

Whicharesignificantsidebands?

Ans.Thesidebands havingconsiderable amplitudes i.e.morethanorequal to1%ofthe carrier amplitude are known as significant sidebands.

WhatisCCIR?

Ans.CCIRstandsforConsultativeCommitteeforInternational Radio.

WhatistheindirectmethodofFMgeneration? Ans.
Armstrong method.
ClassifyFMonthebasisofbandwidth. Ans.
Narrowband and wideband FM.
WhichoneisbetterintermsofnoiseimmunityAMorFM? Ans. FM.

EXPERIMENT No. 6

AIM:-To Detect FM Signal using PLL & Foster-Seeley method.

APPARATUS REQUIRED:-(i)C.R.O. (ii) CRO Probe (ii) FM Modulation and Demodulation Trainer (ST 2203) (iv) Connecting leads

THEORY:-

AFM receiver is very similar to an AM receiver. Themost significant changeisthatthe demodulator must now extract the information signal from a frequency rather than amplitude modulated wave.

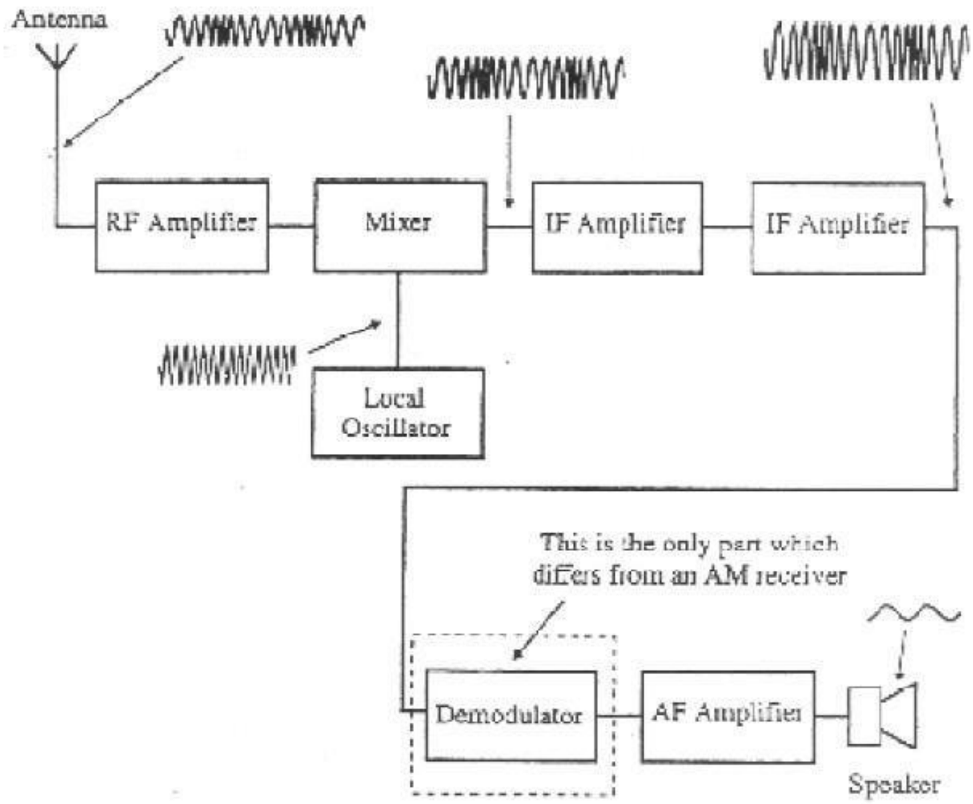


Figure1:FMReceiver

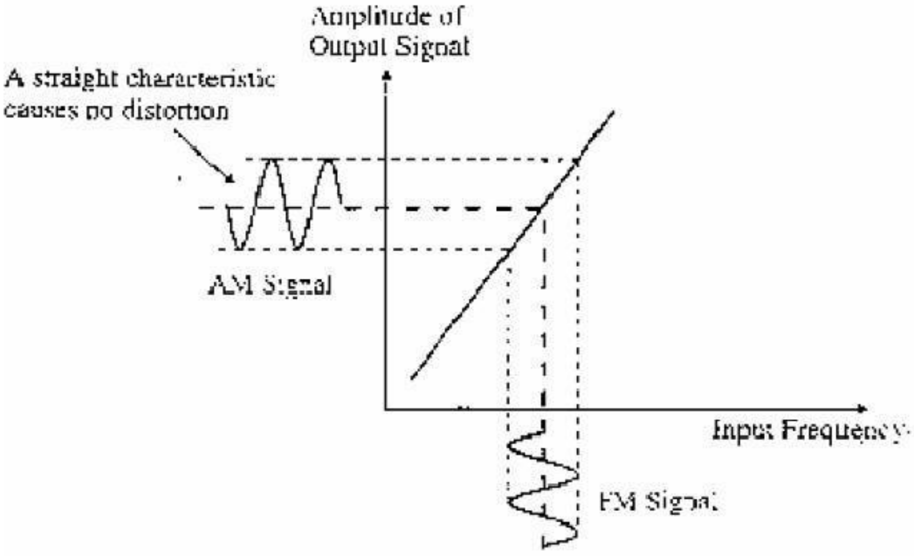


Figure2: Voltage/Frequency Characteristics.

ThebasicrequirementofanyFMdemodulatoristhereforetoconvertfrequencychanges intochangesinvoltage,withtheminimumamountofdistortion.Toachievethis,itshould ideallyhavealinearvoltage/frequencycharacteristic,similartothatshowninfigure2.A 'demodulator' can also be called a 'discriminator' or a 'detector'.

PHASELOCKLOOPDETECTOR

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,itisoftenusedincommercialbroadcastreceivers.Ithasverylowlevelsofdistortionand isalmostimmunefromexternalnoisesignalsandprovides verylowlevels ofdistortion. Altogether it is a very nice circuit.

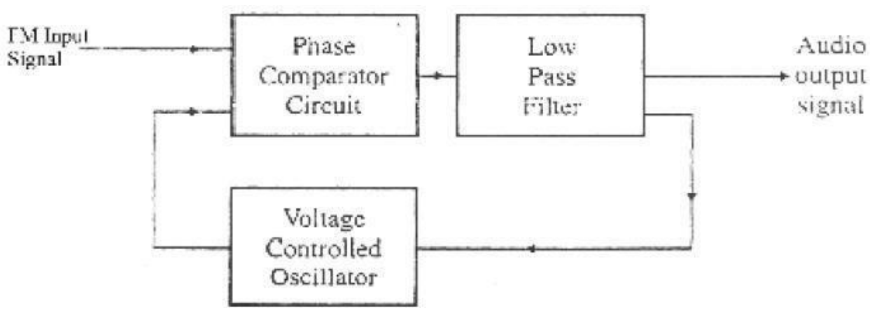


Fig.3PhaseLockLoop Detector

Theoverall action ofthe circuit may, at first, seem ratherpointless. As we can seein fig 3,thereisavoltage-controlledoscillator(VCO).TheDCoutputvoltagefromtheoutput ofthelowpassfilterscontrolsthefrequencyofthisoscillator.NowthisDCvoltagekeeps 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 byimagining 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 loadingeffectsfromdisturbingtheVCOandthenthroughanaudioamplifierifnecessary. The frequency response is highly linear as shown in figure 2.

FOSTERSEELEYDETECTOR

ThefosterSeeleycircuitisshowninfig.4.Atfirstglance,itlooksrathercomplicatedbut it becomes simpler if we consider it a bit at a time.

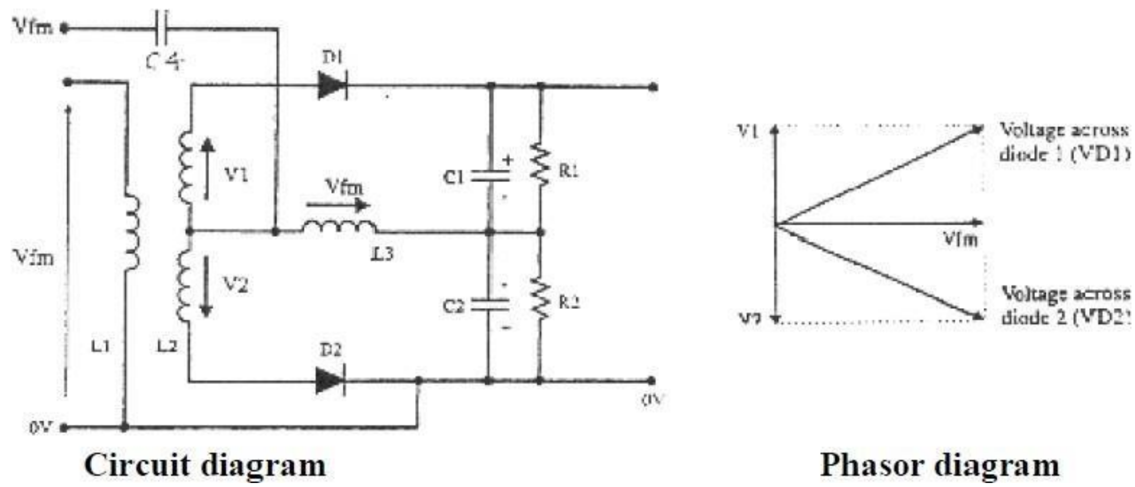


Figure4:Foster–SeelayDetector

When theinput signal isun-modulated: Wewill startbybuildingup thecircuit alittleat atime.Todothis,wecanignoremanyofthecompanieswemayrecognizeimmediately that it consist oftwo envelopedetectors likehalf waverectifiersarefed from thecenter- tapped coil L2. With reference to the center-tap, the two voltages V1and V2 are in anti- phase as shown by the arrows. The output voltage would be zerovolts since thecapacitor voltages are in anti-phase and are equal in magnitude. After adding two capacitors: Thenext stepis to add two capacitors and seetheir effect on thephaseofthe signals. L1 and L2 are magneticallytightlycoupled 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 thephase of the input signal by90°withreferencetothevoltageacrossL1andL2.ThevoltagesareshownasVaand Vbinthephasordiagramgiveninfigure39.UsingtheinputsignalVfmasthereference, the phasor diagrams now look the wayshown in figure 4. C4 is not important. It is only aDCblockingcapacitorandhasnegligibleimpedanceatthefrequenciesbeingused.But whatitdoesdoistosupplyacopyoftheincomingsignalacrossL3.Theentireincoming 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.OneisV1orV2andtheotheristhenewvoltageacrossL3,whichisequaltoVfm. 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 becomingincreasinglycapacitive.Thiswouldresultinalargertotalvoltagebeingapplied across D1 and a reduced voltage across D2. Since the capacitor C1wouldnow charge to a higher voltage, the finaloutput from the circuit would be a positive voltage. Conversely, if the frequency of the

FMinputsignaldecreasedbelowtheunmodulatedvalue,thephaseshiftduetocapacitor C5 increases above 90° as the parallel tuned circuit becomes slightly inductive. This causesthevoltageacrossdiodeD2toincreaseandthefinaloutputfromthedemodulator 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.

BLOCKDIAGRAM:-

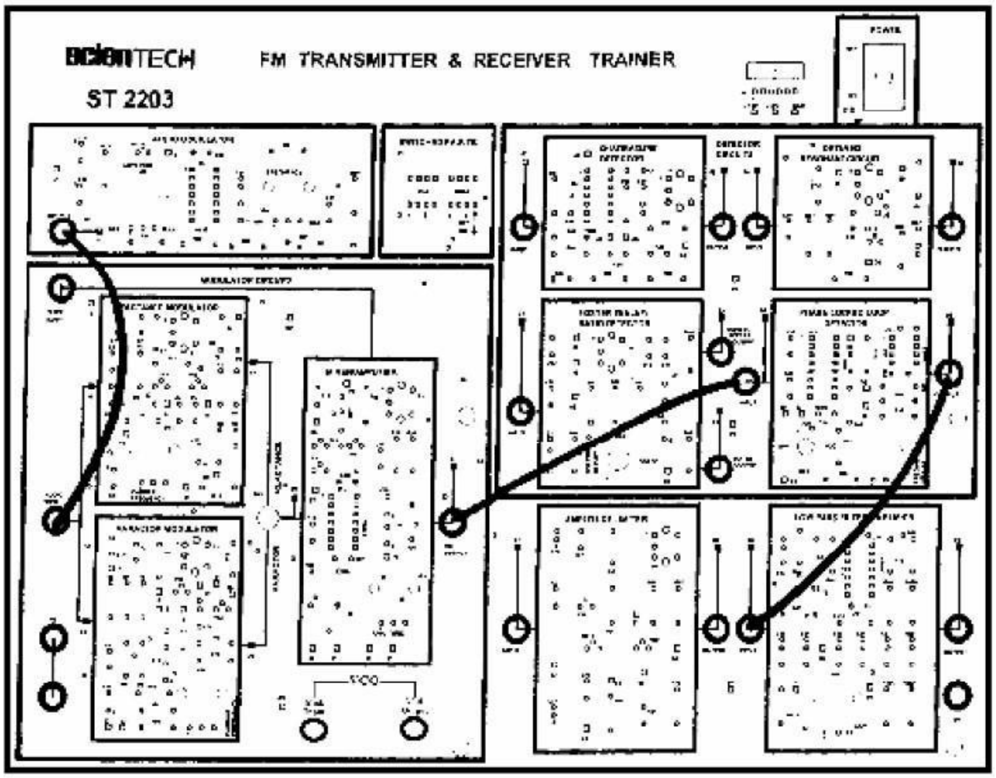


Figure5:Connections for FM Demodulation usingPLL

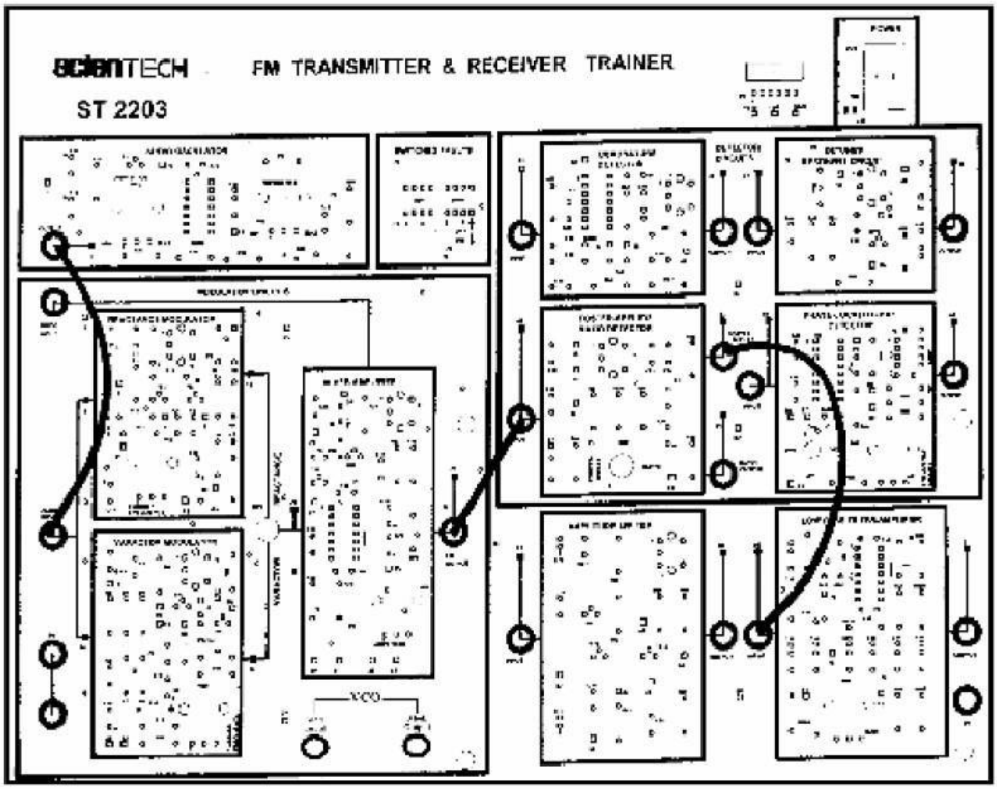


Figure6:ConnectionsforFMDemodulationusingFoster-SeelayDetector

PROCEDURE:-

FMDetectionusingPLL:

1. Ensurethatthefollowinginitialconditionsexiston the**ST2203** module:

a. AllswitchedfaultsOFF;

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

c. Audio amplifier block's frequency pot in fully counter clockwise. Ensurethatthefollowinginitialconditionsexistonthe**ST2203**clockwise (MIN) position.

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

e. VCOswitch(inphase-lockedloopdetectorblock)inONposition.

2. Maketheconnectionsshowninfigure5.

3. Turnonpowertothe**ST2203** module.

4. Nowmonitortheaudioinputsignaltothevaractormodulatorblock(att.p.14)together with the output from the phase-locked loop detector block (at t.p.60), triggeringthe oscilloscope in t.p.14. The signal at t.p.68 should contain three components:

- ApositiveD.C.offset voltage.
- Asinewaveat thesame frequencyastheaudiosignalat t.p.14.
- Ahigh -frequencyripplecomponent.

5. The low pass filter/amplifier block strongly attenuates the high-frequency ripple componentatthedetector'soutputandalsoblockstheD.C.offsetvoltage.Consequently the signal at the output of the low- pass filter/amplifier block (at t.p.73) should be very closelyresembletheoriginalaudiomakingsignal,ifnotthenslowlyadjustthefreq.adjust pot of PLL block.

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

<u>FMDetectionusingFoster-SeelayDetector:</u>

1. Ensurethatthefollowinginitialconditionsexiston the**ST2203** module:

- **a.** AllswitchedfaultsOFF;
- **b.** Audio amplifierblock'samplitudepotinfullyclockwise(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.** VCOswitch(inphase-lockedloopdetectorblock)inOFFposition.

2. Makeconnectionasshowninfigure42

3. Turnonpowertothe**ST2203** module.

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.

5. Initially,wewillusethevaractormodulatortogenerateourFMsignal,sincethisisthe 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.

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.

7. Nowmonitortheaudioinputsignaltothevaractormodulatorblock(att.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:

- · Asinewaveat thesamefrequencyastheaudio signalatt.p.14.
- · AHigh frequencyripplecomponentofsmall amplitude.

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.

9. Monitortheaudioinputtothevaractormodulator(att.p.14)andtheoutputofthelow 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.

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

RESULT:-

FMsignalisbeingdemodulatedbyusingPLLandFoster-SeelayMethod.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

HowmanytypesofFMarethere?Writetheir names.
Ans.TheretwotypesofFMi.e.narrowbandFMandwideband FM.
Whatfrequencydeviation inFM?
Ans. The maximum change in instantaneousfrequency from the average is known asfrequencydeviation.
Whichistheusefulparameterfordetermination of bandwidth?
Ans.Frequencydeviation istheusefulparameterfordetermination ofbandwidth.
Whataredifferentmethods ofFMdetection?
Ans.1.Slopedetection method2. Phasedetection method.
Whichsidebandsareignoredin FM?
Ans.Thesidebandswith smallamplitudeareignoredin FM.
Whicharesignificantsidebands?
Ans.Thesidebands havingconsiderable amplitudes i.e.morethanorequal to1%ofthe carrier amplitude are known as significant sidebands.
WhatisbasicprincipleofFMdetection?
Ans.Conversionoffrequencyvariations intoamplitudevariations.
WhatistheindirectmethodofFMgeneration? Ans.
Armstrong method.
WhatisthedirectmethodofFMgeneration? Ans.
The parameter variation method.
Whatis thefunctionofamplitudelimiter?
Ans.ItsuppressestheundesirableamplitudefluctuationingeneratedFMsignal.

EXPERIMENT No. 7

AIM:-ToStudySuperheterodyneAMreceiverandmeasurementofreceiver parameters viz. sensitivity, selectivity & fidelity.

APPARATUSREQUIRED:-(i)C.R.O.(ii) CRO Probe(ii)DSB/SSB Transmitter (ST 2201) and Receiver Trainer (ST 2202) (iv) Connecting leads

BLOCKDIAGRAM:-

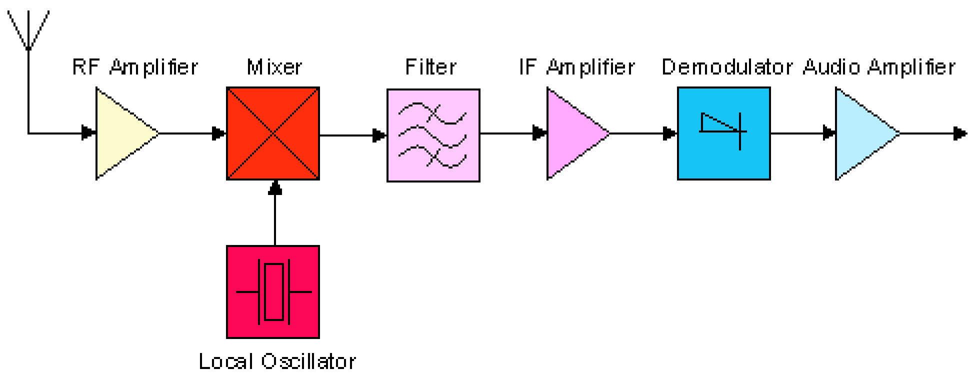


Figure1:SuperhetrodyneReceiver

THEORY:-

The principle of operation of the super heterodyne receiver depends on the use of heterodyning or frequency mixing. The signal from the antenna is filtered sufficiently at least to reject the *image frequency* (seebelow) and possibly amplified. A local oscillator inthereceiverproducesasinewavewhichmixeswiththatsignal,shiftingittoaspecific intermediatefrequency(IF),usuallyalowerfrequency.TheIFsignalisitselffilteredand amplifiedandpossiblyprocessedinadditionalways.Thedemodulatoruses the IF signal ratherthantheoriginalradiofrequencytorecreateacopyoftheoriginalmodulation(such as audio).

Fig.1. shows the minimum requirements for a single-conversion super heterodyne receiver design. The following essential elements are common to all superhet circuits: a receiving antenna, a tuned stage which may optionally contain amplification (RF amplifier), a variable frequency local oscillator, a frequency mixer, a band pass filter and intermediate frequency (IF) amplifier, and a demodulator plus additional circuitry to amplify or process the original audio signal (or other transmitted information). To receive a radio signal, a suitable antenna is required. This is often built into a receiver; especially in the case of AM broadcast band radios. The output of the antenna may be very small, often only a few micro volts. The signal from the antenna is tuned and may be amplified in a so-called radio frequency (RF) amplifier, although this stage is often omitted.Oneormoretunedcircuitsatthisstageblockfrequencieswhicharefarremoved fromtheintendedreceptionfrequency.Inordertotunethereceivertoaparticularstation, the frequency of the local oscillator is controlled by the tuning knob

(forinstance).TuningofthelocaloscillatorandtheRFstagemayuseavariablecapacitor, or varicap diode. The tuning of one (or more) tuned circuits in the RF stage must track the tuning of the local oscillator.

Mixer stage: The signal is then fed into a circuit where it is mixed with a sine wave from a variable frequency oscillator known as the local oscillator (LO). The mixer usesanon-linearcomponenttoproducebothsumanddifferencebeatfrequenciessignals, each one containing the modulation contained in the desired signal. The output of the mixer may include the original RF signal at f_d, the local oscillator signal at f_{LO}, and the two new frequencies f_d+f_{LO} and f_d-f_{LO}.The mixer may in advertently produce additional frequencies such as 3rd- and higher-order inter modulation products. The undesired signals are removed by the IF band pass filter, leaving only the desired offset IF signal atf_{IF} which contains the original modulation (transmitted information) as the received radio signal had at f_d.

Intermediate frequency stage: The stages of an intermediate frequency amplifier are tuned to a particular frequency not dependent on the receiving frequency; this greatly simplifies optimization of the circuit.[6] The IF amplifier (or *IF strip*) can be made highly selective around its center frequency f_{IF}, whereas achieving such a selectivity at a much higher RF frequency would be much more difficult. By tuning the frequency of the local oscillator f_{LO}, the resulting difference frequency f_{LO}-f_d (or f_d-f_{LO} when using so-called low- side injection) will be matched to the IF amplifier's frequency f_{IF} for the desired reception frequency f_d. One section of the tuning capacitor will thus adjust the local oscillator's frequency f_{LO} to f_d+f_{IF} (or. Less often, to f_d-f_{IF}) while the RF stage is tuned to f_d. Engineering the multi-section tuning capacitor (or varactors) and coils to fulfill this condition across the tuning range is known as *tracking*. Other signals produced by the mixer (such as due to stations at nearby frequencies) can be very well filtered out in the IF stage, giving the super heterodyne receiver its superior performance. However, if f_{LO} is set to f_d + f_{IF}, then an incoming radio signal at f_{LO} +f_{IF} will *also* produce heterodyne at f_{IF}; this is called the *image frequency* and must be rejected by the tuned circuits in the RF stage. The image frequency is 2f_{IF} higher (or lower) than f_d, so employing a higher IF frequency f_{IF} increases the receiver's *image rejection* without requiring additional selectivity in the RF stage. Usually the intermediate frequency is lower than the reception frequency f_d, but in some modern receivers (e.g. scanners and spectrum analyzers) it is more convenient to first convert an entire band to a much higher intermediate frequency; this eliminates the problem of *image rejection*. Then a tunable local oscillator and mixer convert that signal to a second much lower intermediate frequency where the selectivity of the receiver is accomplished. In order to avoid interference to receivers, licensing authorities will avoid assigning common IF frequencies to transmitting stations. Standard intermediate frequencies used are 455 kHz for medium-wave AM radio, 10.7 MHz for broadcast FM receivers, 38.9MHz(Europe)or45MHz(US)fortelevision,and70MHz for satellite and terrestrial microwave equipment.

Bandpass filter: The IF stage includes a filter and/or multiple tuned circuits in order to achievethedesiredselectivity.Thisfilteringmust thereforehave abandpassequaltoor less than the frequency spacing between adjacent broadcast channels. Ideally a filter would have a high attenuation to adjacent channels, but maintain a flat response across the desired signal spectrum inorder to retain the quality of the receivedsignal. This may be obtained using one or more dual tuned IF transformers or a multipole ceramic crystal filter.

Demodulation: The received signal is now processed by the demodulator stagewhere the audio signal (or other baseband signal) is recovered and then further amplified. AM demodulation requires the simple rectification of the RF signal (so-called envelope detection), and a simple RC low pass filter to remove remnants of the intermediate frequency. FM signals may be detected using a discriminator, ratio detector, or phase-lockedloop.Continuouswave(morsecode)andsinglesidebandsignalsrequireaproduct detector using a so-called beat frequency oscillator, and there are other techniques used for different types of modulation. The resulting audio signal (for instance) is then amplified and drives a loudspeaker. When so-called high-side injectionhas been used, where the local oscillator is at a *higher* frequency than the receivedsignal (as is common), then the frequency spectrum of the original signal will bereversed. This must be taken into account by the demodulator (and in the IF filtering) in the case of certain types of modulation such as single sideband.

RECEIVERCHARACTERISTICS:

Theimportantcharacteristicsofreceiversaresensitivity,selectivity,&fidelity described as follows:

Sensitivity:

The sensitivity of radio receiver is that characteristic which determines the minimum strength of signal input capable of causing a desired value of signal output. Therefore, expressing in terms of voltage or power, sensitivity can be defined as the minimum voltageorpoweratthereceiverinputforcausingastandardoutput.Incaseofamplitude-modulation broadcast receivers, the definition of sensitivity has been standardized as "amplitude of carrier voltage modulated 30% at 400 cycles, which whenapplied to the receiverinputterminalsthroughastandarddummyantennawilldevelopanoutputof0.5 watt in a resistance load of appropriate value substituted for the loud speaker" .

Selectivity:

The selectivity of a radio receiver is that characteristic which determines the extent to which it is capable of differentiating between the desired signal and signal of other frequencies.

Fidelity:

Thisisdefinedasthedegreewithwhichasystemaccuratelyreproducesatitsoutput the essential characteristics of signals which is impressed upon its input.

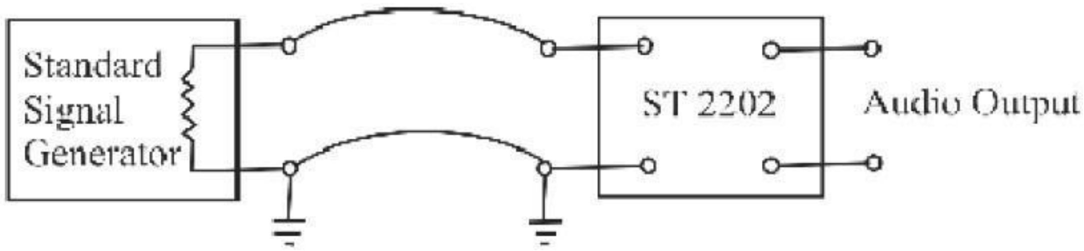


Figure2:SetupforDeterminingRecieverCharacteristics

Determination of receiver characteristics:
A laboratory method for the measurement of receiver characteristics is shown in Fig. 2. We use here an artificial signal to represent the voltage that is induced in the receiving antenna. This artificial signal is applied through 'dummy' antenna, which in association antenna with which the receiver is to be used. Substituting the resistance load of proper value for the loudspeaker and measuring the audio frequency power determine the receiver output.
Sensitivity:
Sensitivityis a determined byimpressing different RF voltages in series with a standard dummy antenna and adjusting the intensity of input voltage until standard outputs obtained at resonance for various carrier frequencies. Sensitivity is expressed in microvolt.
Selectivity: Selectivity is expressed in the form of a curve that give the carrier signal strength with standard modulation that is required to produce the standard test output plotted as a function off resonance of the test signal. The receiver is tuned to the desired frequencyandmanualvolumecontrolissetformaximumvalue.Atstandardmodulation, thesignal generatoris set at theresonant frequencyofthereceiver.Thecarrieroutput of thesignalgeneratorisvarieduntilthestandardtestoutputisobtained.Atthesametuning ofreceiver,thefrequencyofsignal generatorisvariedaboveandbelowthefrequencyto whichthereceiveristuned.Foreveryfrequency, thesignalgeneratorvoltage, appliedto the receiver input, is adjusted to give the standard test output from the receiver
Fidelity: Fidelityis the term expressing the behavior of receiver output with modulation frequencyof input voltage. To obtain a fidelitycurve, the carrier frequencyof the signal generator adjusted to resonance with the receiver, standard 400 cycles modulation is applied, the signal generator carrier level is set at a convenient arbitrary level and the manual volume control of the receiver is adjusted to give the standard test output. The modulationfrequencyisthenvariedovertheaudiorange,keepingdegree ofmodulation constant.

PROCEDURE:-

(a) ToPlotSelectivityofReceiver:
1. SettingonST2202
   a. Setthedetectorin diodemode.
   b. AGCon.
   c. Setthevolumecontrolfullclockwise.

2. Apply AM signal with 400 Hz modulating frequency and 30% modulationtaken from AM generator into Rx input socket.
3. Settheinputcarrierfrequencytosuitablevaluethatlieswithinthe AMband(525KHz -1600 KHz).Also setsignallevel to 100mV.
4. TunetheReceiverusingtuningcontrol.Alsoadjustgainpotentiometerprovidedin R.F. amplifiersection ofST2202 so as toget unclipped demodulated signal at detector's output (output of audio amplifier).
5. Note the voltage level at receiver's final output stage i.e. audio amplifier's output on CRO (voltage at resonance (Vr)).
6. Nowgraduallyoffsetthecarrierfrequencyinsuitablestepsof5KHzor10KHzbelow and above the frequency adjusted in step 2 without changing thetuningofreceiver while maintaining the input signal level.
7. Now record the signal level at output of audio amplifier for different input carrier frequency, on CRO (i.e. voltage off resonance (Vi)).
8. Tabulatethereadingsasunder:

Carrier Frequency	Output Voltage	Ratio = 20 log (Vi / Vr) dB

9. Plot the curve between ratio and carrier

frequency.<u>(b) To Plot Sensitivity of Receiver:</u>
1. SettingonST2202:
    a. Setthedetectorin diodemode.
    b. AGCon.
    c. Setthevolume controlfullyclockwise.
2. Apply AM signal, with 400 Hz modulating signal and 30% modulation, taken from AM generator into Rx input socket.
3. Settheinputcarrierfrequencysoastoliewithinthe AMBand(525KHz–1600KHz). Alsotunethedetectortothatcarrierfrequencyusingtuningcontrol.(Youwillhearatone)
4. SettheinputAMlevelto100mV.Alsoadjustthegainpotentiometerprovidedin R.F. amplifier section of ST2202 so as to get unclipped demodulated signal at detectors output.
5. Record input carrier frequency & signal level at the final output stage i.e. output ofaudio amplifier (observed on CRO).
6. Changetheinputcarrierfrequency&alsotunethereceivertothatfrequency&repeat step 4.
7. Tabulatethecollectedreadingsasunder:

Carrier frequency	Output (pp)

8. Plotthegraphbetweencarrierfrequency&output level.

(c) ToPlotFidelityofReceiver:

1. SettingonST220:
    a. Setthedetectorin diodemode.
    b. AGCon.
    c. Setthevolume controlfullyclockwise.
2. ApplyAMsignalof100mVwith400Hzmodulatingsignaland30%modulation,into Rx input.
3. Select a suitable carrier frequency that lies within AM Band (525 KHz - 1600 KHz). Tune the ST2202 receiver to that frequency using tuning control. Also adjust gain potentiometer provided in R.F. amplifier section so as to get unclipped demodulated signal at detector's output.
4. Note the demodulated signal level (Vr) at the final output stage i.e. output of audio amplifier (on CRO) for the applied AM signal with 400Hz modulating signal.
5. Now vary the modulating signal frequency over audio range (300 Hz-3 KHz) in suitable steps say 100Hz. Note the corresponding output level (Vi) at the output ofaudio amplifier (on CRO).
6. Tabulatereadings asunder:

Carrier frequency	Modulating frequency	Output Voltage

Relativeresponse=20 log(Vi /Vr)dB

7. Plotthegraphbetweenmodulatingfrequencyandrelativeresponse.

RESULT:-

Superhetrodynereceiverhas beenstudied andplot for receiver parametersviz. sensitivity, selectivity and fidelity has been drawn.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

Whatisamplitudemodulation?

Ans. Amplitude Modulation is a process in which the amplitude of the carrier is madeproportional to the instantaneous amplitude of the modulating signal.

Whicharethethree discretefrequenciesinAM?

Ans.(1)Carrierfrequency(2)lowersidebandfrequency(3)uppersidebandfrequency

Q.3Whatisreceiversensitivity?
Ans.Thesensitivityofradioreceiveristhatcharacteristicwhichdeterminestheminimum strength of signal input capable of causing a desired value of signal output.
Q. 4.Whatisreceiverselectivity?
Ans. Selectivity is expressed in the form of a curve that give the carrier signal strength withstandardmodulationthatisrequiredtoproducethestandardtestoutputplottedasa function off resonance of the test signal.
Whatisreceiverfidelity?
Ans.Thisisdefinedasthedegreewithwhichasystemaccuratelyreproducesatitsoutput the essential characteristics of signals which is impressed upon its input.
Whichtypeofreceiverisusedintheexperiment? Ans.
Superhetrodyne.
Wherethemodulation index lies?
Ans.modulationindex always liesbetween0and 1.Morethan1isovermodulation.
Whathappensincaseofovermodulation? Ans.
The wave will get distorted.
Mentionadisadvantageofsuperhetrodyning?
Ans.Generation of Image frequency.

EXPERIMENT No. 8(a)

AIM:-To study Pulse Amplitude Modulation.

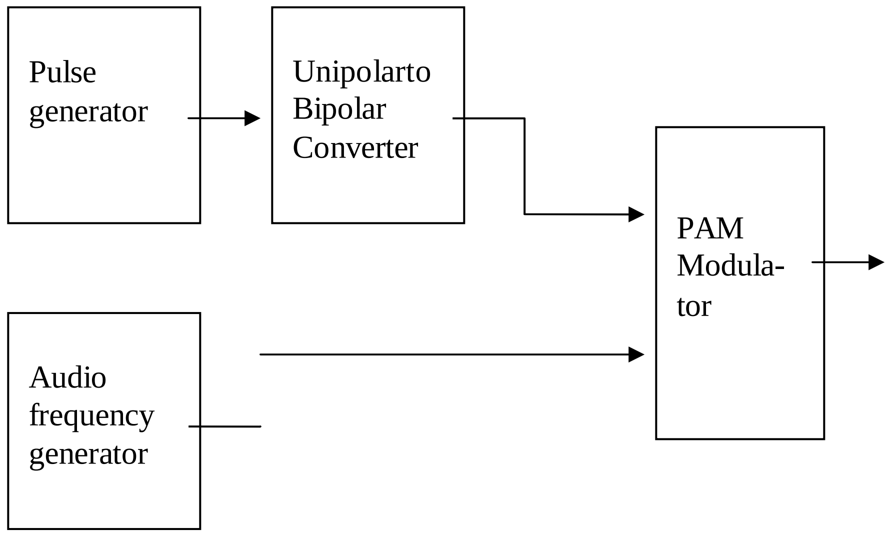
APPARATUSREQUIRED:-CRO, experimental kit, power supply, connecting leads.

THEORY:-

Pulse Modulation: We know that in Analog modulation systems, some parameter of a sinusoidal carrier (continuous in time domain) is varied according to the instantaneous value of the modulatingsignal. But in pulse modulation methods, the carrier is no longer a continuous signal but consists of a train of uniform pulses having a definedPRF (Pulse Repetition Frequency). The continuous modulating message signal waveforms are sampled at regular intervals. Information regarding the signal is transmitted onlyat the sampling times, together with anysynchronizing pulses that may be required. At the receiving end, the original wave form may be reconstituted with negligible distortion from the information regarding the samples, if these samples are taken with minimum sufficient frequency.

In Pulse Modulation some parameter of the pulsed carrier is varied according to the instantaneous value of the modulating signal. Pulse modulation may be broadly subdividedintotwocategories:Analog&Digital.Intheformer,theindicationofsample amplitudemaybeinfinitelyvariable,whileinthelatteracodewhichindicatesthesample amplitude to the nearest pre-determined level is sent. Pulse-Amplitude & Pulse-Time Modulation are both analog while the Pulse-code and Delta modulation are both digital. Allthemodulationsystemshavesamplingincommon,buttheydiffer fromeachotherin themannerofindicatingthesampleamplitude.InPAMthesignalissampledataregular intervals and each sample is made proportional to the instant of sampling. In single polarity PAM is fixed, AC level is acted to ensure that all the pulse are +Ve going. The frequencyspectrum is decayingbut with decaying amplitude. Therateofdecaydepends upon the width of the pulses. As the pulses are made wider, the spectrum decays faster.

BLOCKDIAGRAM:-



WAVEFORMFORMS:-

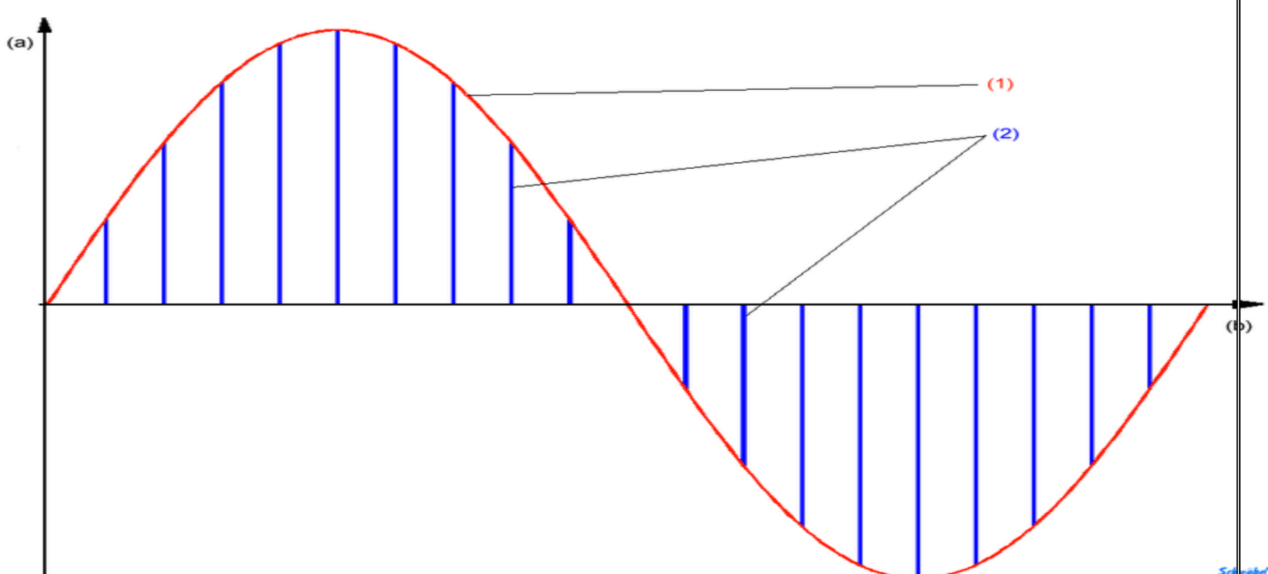


Fig1:PrincipleofPAM;(1)originalSignal,(2)PAM-Signal,(a)AmplitudeofSignal, (b) Time

PROCEDURE:-

1. Maketheconnection accordingtotheblockdiagram.
2. Connectpulsegenerator to theunipolartobipolarconverte
3. Connecttheaudiofrequencyof2 KHz,2Vto modulator.
4. ConnectthemodulatoroutputtoCRO.

5. ObserveoutputonCRO.

RESULT:-

Pulsemodulatedwaveformisobtainedon CRO.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Beforeconnectingthepowersupplyplugintosocket,ensurepowersupplyshouldbe switched off.
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

Whatispulseamplitudemodulation?
Ans.Amplitudeofthesampledpulseisvariedaccordingtothe modulating signal.
Howmanytypesofpulsemodulation?
Ans.TherearetwotypesofPMi.e.PAMand PTM.
How many types of pulse time modulation?
Ans.Therearetwotypes ofPTMi.e.PWMandPPM.
Whatisbaseband system?
Ans.Whenthebasicsignalistransmittedwithoutafrequencytranslationisknownas base band signal.
Howmanytypes ofpulse amplitude modulation?
Ans.TherearetwotypesofPAMi.e.singlepolarityanddoublepolarity.
WritethedemodulationmethodofPlatosamplesignal. Ans. (1) using an equalizer (2) using holding circuit.
WhichfilterisusedinPAMdemodulatorcircuit? Ans. Second order low pass filter.
Writethetwomethodofmultiplexing Ans.
(1) FDM (2) TDM
WhatisdirectmethodofPTM generation?
Ans. In the direct method of PTM generation wave form is generated withoutgeneratingthe PAM.
Whatisthemeritofflattopsampling?
Ans.Thetopsofpulsesareflatthusthepulseshaveconstantamplitudewithinthe pulse Interval.

EXPERIMENT No. 8(b)

AIM:-To study the pulse width modulation.

APPARATUSREQUIRED:-CRO, experimental kit, power supply, connecting leads

THEORY:-

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).

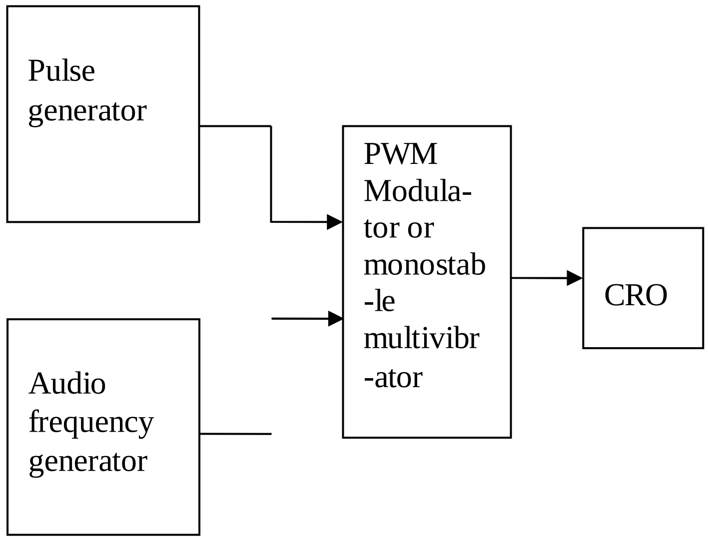
InPWMwidthofeachpulsedependsontheinstantaneousvalueofthebaseband signalatthesamplinginstant.Inpulsewidthmodulationcontinuouswaveformissampled at regular intervals and the width of each pulse is kept proportional to the magnitude of signalatthatinstantinPWM.Inpulsewidthmodulationpulseisvaried accordancewith themodulatingsignalbuttheamplitudeandstartingtimeofeachpulseisfixed.InPWM, 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.

GenerationandDemodulationofPWM

PWMmaybegeneratedbyapplyingtriggerpulsestocontrolthedurationofthesepulses. Theemittercoupledmono-stablemulti-vibratorisusedasvoltagetotimeconverter,since its gate width is dependent on the voltage to which the capacitor C is charged .If this voltageisvariedinaccordancewithasignalvoltage,aseriesofrectangularpulseswillbe obtained, with widths varying as required. The demodulation of pulse width modulation is a simple process. PWM is fed to an integrating circuitfrom which a signal emerges whose amplitude at any time is proportional to the pulse width at that time.

BLOCKDIAGRAM:-



PROCEDURE:-

1. Maketheconnection accordingtotheblockdiagram.
2. Connecttheaudiofrequencyof2 KHz,2Vto modulator.
3. ConnectthemodulatoroutputtoCRO.
4. SwitchONthepowersupply.
5. ObserveoutputonCRO.

OUTPUTWAVEFORMS:-

```
             _         _         _         _         _         _         _         _
            ||        | |       ||        | |       | |       | |       | |       ||
Clock       ||        ||        ||        ||        ||        ||        ||        ||
          ||________||______||______||______||______||______||______||_______



  Data                  ||        ||         |     |            |     |||
                       ||        ||        |      |           |      |||
                      ||        ||        |      |           |      |  ||



Data           0         1         2         4         0         4         1         0
```

RESULT:-

PulseswidthmodulatedwaveisobtainedonCRO.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.

2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Take the reading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

WhichmodulationisPWM? Ans.
Analog Modulation
WhataretwocategoriesofPulseModulation.? Ans.!
Analog Modulation 2. Digital Modulation
Whatistheunitofsignalingspeed? Ans.
Baud
WhatisthedisadvantageofPWM?
Ans.Dueto varyingof pulseswidthpowercontents ofPWM also varying
WhichmultivibratorisusedforPWM? Ans.
Monostable Multivibrator
WhichcircuitisusedforPWMdemodulator? Ans.
Integrating circuit.
Whatisdifference bet.PAMandPWM?
Ans.InPAM,amplitudeofpulseisvariedaccordingtomodulatingsignalandin PWM,width is varied of pulses.
HowPWMmaybegenerated?
Ans.PWMmaybegeneratedapplyingtriggerpulsestocontolthestartingtimeof pulses from a monostable multivibrator.
Whatistheuseofsamplingtheorem?
Ans.SamplingTheoremisusedtodetermineminimumsamplingspeed.
Whatistheworldwidestandardsamplingrate? Ans.
Eight thousand samples per second.

EXPERIMENT No. 8(c)

AIM:--To study the pulse position modulation.

APPARATUS REQUIRED:-:- CRO, experimental kit, power supply, connecting leads.

THEORY:-

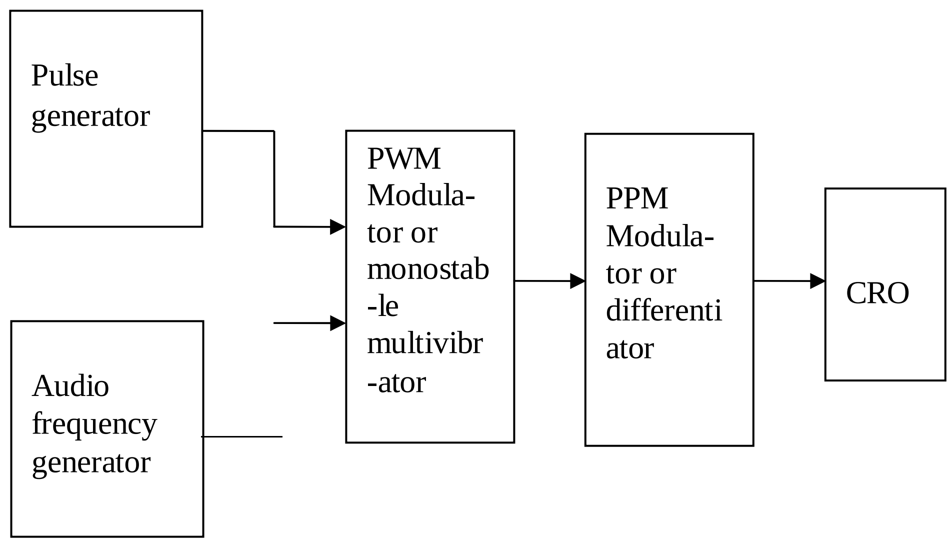
In pulse position modulating the amplitude of pulse is kept constant and position of the pulse in relation to the position of the reference pulse or synchronize pulse is varied by each sample value of modulating signal.

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 ofthe leading edgerand slots tops ofPWM pulses. In pulseposition modulating theamplitude ofpulseiskeptconstantandpositionofthepulseinrelationtothepositionofthereference pulseorsynchronizepulseisvariedbyeachsamplevalueofmodulatingsignal.InPWM eachpulsehasaleadingedgeandatrailingedgebutthelocationofleadingedgearefixed where trailing edge are depends on the pulse width. The trailing edges of PWM pulses areinfactpositionmodulated.ThusPPMmaybeobtainedfrom PWMbysimplygetting rid of the leading edge and slots tops of PWM pulses. In comparison with PWM, PPM hastheadvantageofrequiringconstanttransmitterpoweroutput,butthedisadvantageof 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. There sult is another pulse trainwhichhaspositivegoing narrow pulses corresponding to leading edges and negative going narrow pulses corresponding to trailing edges. If the position corresponding to the trailingedgesof an un-modulated PWM pulse is counted as zero displacement, then the trailing edges of a modulated pulse will arrive earlier or later. An un modulated 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,itisagainfirstconvertedintoPWMbyusingflip-floporbistablemultivibrator. One input of the multivibrator receives trigger pulses from alocal generator which is synchronized bytrigger pulses received from the transmitter, and these triggers areused to switch off one of the stages of the flip-flop. The PPM pulses are fed to the other base oftheflip-flopandswitchthatstageON.Theperiodoftimeduringwhichthisparticular stageisOFF,dependsonthetimedifferencebetweenthetwotriggers,sothattheresulting 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.

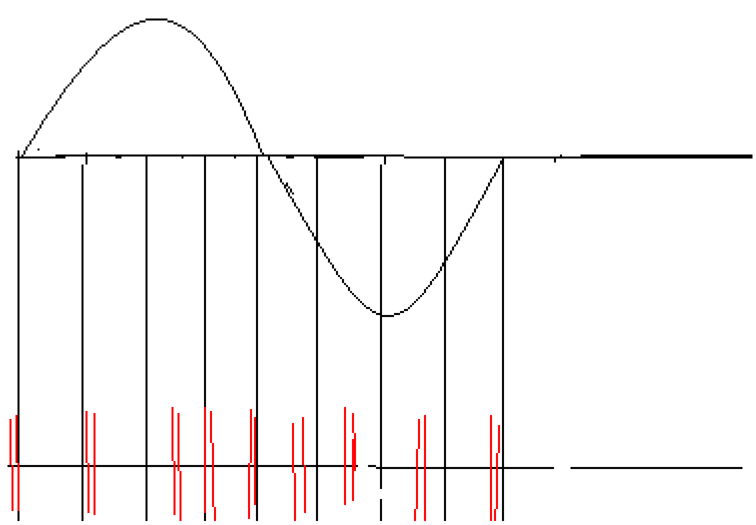
BLOCKDIAGRAM:-



PROCEDURE:--

1. Make the connection according to the block diagram.
2. Connecttheaudiofrequencyof2 KHz,2Vto modulator.
4.ConnectthePWMoutputto thePPMmodulator.
4. ConnectthePPMmodulator output toCRO.
5. SwitchONthepowersupply.
6. ObserveoutputonCRO

OUTPUTWAVEFORMS:-



RESULT:-

The Pulse Position Modulated wave is obtained on CRO.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Beforeconnectingthepowersupplyplugintosocket,ensurepowersupplyshouldbe switched off.
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment

QUIZ/ ANSWERS:-

WhatisadvantageofPPM?
Ans.Ithas novaryingwidthofpulsesopowercontentarenot varying.
What isPPM?
Ans.InPPMthepositionofpulsesisvariedandwidthandamplitudeareconstant.
WhichMultivibratorisusedforPPMDe-modulator? Ans.
Bi-stable Multivibrator.
Whatisthedifferencebetween PPM&PWM?
Ans.InPWM,thewidth isvariedandinPPM,thepositionisvariedaccordingto modulating signal.
Q5.WhichfilterisusedinPPMdemodulator? Ans.
Second order low pass filter.
Q6.InwhichcategoryofPMisPPM? Ans.
Analog Modulation.
Q7.WhichmodulationissimilartoPDM? Ans.
Phase modulation.
Q8.At which factortheband-width ofPPM depends?
Ans.Bandwidthoftransmission channel dependsonrisingtimeofthepulse.
Q.9Whatistheuseofsamplingtheorem?
Ans.SamplingTheoremisusedtodetermineminimumsamplingspeed.
Q.10 Whatistheworldwidestandardsamplingrate? Ans.
Eight thousand samples per second.

EXPERIMENT No. 9

AIM:--To study Time Division Multiplexing.

APPARATUSREQUIRED:- (i) C.R.O. (ii) CRO Probe (ii) TDM Pulse Code Modulation Transmitter Trainer (ST 2103) and TDM Pulse Code Modulation Receiver Trainer (ST 2104) (iv) Connecting leads.

THEORY:-

Time division multiplexing is a technique of transmitting more than one information on the same channel. As can be noticed from the fig. 11 below the samples consists of short pulses followed by another pulse after a long time intervals. This no-activity time intervals can be used to include samples from the other channels as well. This means that several information signals can be transmitted over a single channel by sending samplesfromdifferentinformationsourcesatdifferentmomentsintime.Thistechnique is known as time division multiplexing or TDM. TDM is widely used in digital communication systems to increase the efficiency of the transmitting medium. TDM can be achieved by electronically switching the samples such that they inter leave sequentially at correct instant in time without mutual interference. The basic 4 channel TDM is shown in fig. 2.

The switches S1 & S2 are rotating in the shown direction in a synchronized manner, whereS1issamplingchanneltothetransmissionmedia.Thetimingofthetwoswitches is very important to ensure that the samples of one channel are received only by the corresponding channel at the receiver. This synchronization between S1 & S2 must be established by some means for reliable communication. One such method is to send synchronization code (information) along itself to the transmitter all the time.

In practice, the switches S1&S2aresimulatedelectronically.

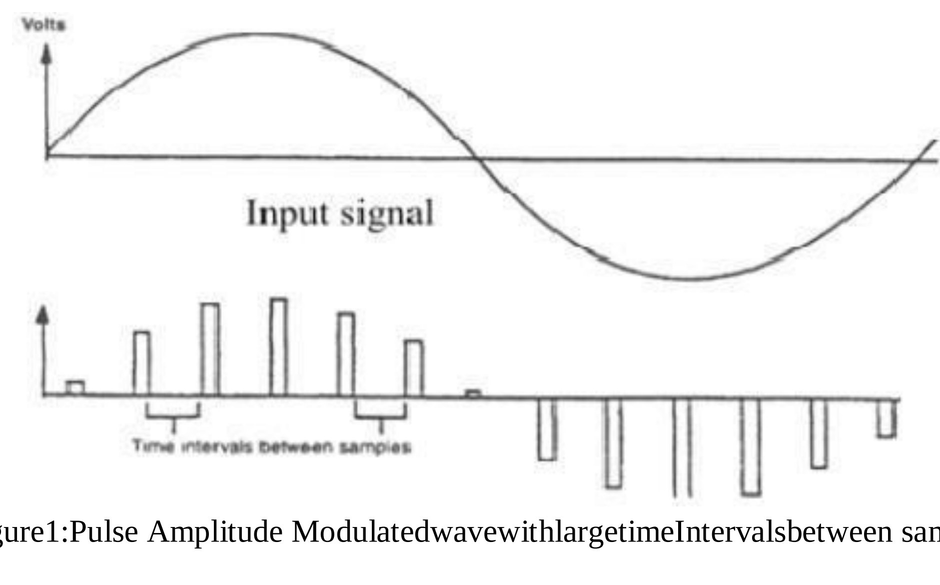


Figure1:Pulse Amplitude Modulatedwavewithlargetime Intervalsbetween samples

On ST2103, the sequence of operation is synchronized to the transmitter clock TX. Clock (t.p.3).Thetimeoccupiedbyeachclockpulseiscalleda*Bit.*Thesequenceof operation is repeated after every 15 bits. The complete cycle of 15 bits is called as*timing frame*. The start of the timing frame is denoted by the TX.TO signal (t.p.4)which goes high during the bit time 0. The various bits reserved for the data appearingin the middle of each transmitter clock cycle is shown in fig. The fig.12 shows the complete timing frame .

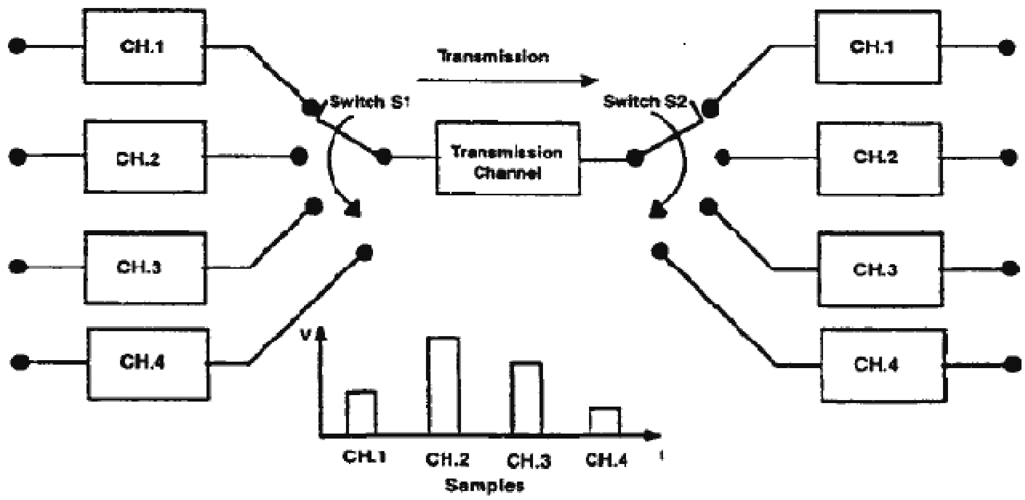


Figure2:Principleof4-ChannelTDMSystem

Bit 0: This bit is reserved for the synchronization information generated by the Pseudo randomsynccodegeneratorblockMoreaboutitsoperationinthelatersection.Whenthe Pseudo Random Sync Code is switched OFF a '0' is transmitted.
Bit 1 to 7: These carrya 7 bit data word correspondingto the last sample taken from the analog channel CH.0. Remember that the trainer transmits lowest significant bit (LSB) first. This time interval during which the coded information regarding the analog informationistransmittediscalledasthetimeslot.Sincethepresenttimeslotcorresponds to channel 0 it is known as timeslot 0.
Bit 8 to 14: This timeslot termed as timeslot 1 contains the 7 bit word corresponding to the last sample taken of analog channel1. As with channel 0 the least significant bit is transmitted first. The receiver requires two signals for its correct operation & reliable communication, namely.

a. Receiverclockoperatingat thesame frequencyasthat ofthe**ST2103**clock.
b. Synchronization signal, which allows the receiver to synchronies its clock/operationwiththetransmitter‘sclockoperation.Alltheserequirementscan be achieved by transmitting two essential information signals:

I. ATransmitclocksignal.
II. AFramesynchronizationsignal.

The simplest method is to transmit the synchronization information & the clock over a separate transmission link. This results in a simplest receiver. It is used in data communication LAN (Local Area Network) & in telemetry systems. However itis waste of media & is not economical for long distance communications. The ST2103 provides these two signals at TX. clock output (t.p.3) & TX.TO output (t.p.4). In this modethePseudorandomsynccodegenerator&detector(onST2104)areswitched

OFF.Thesecondtechniqueistotransmitthesynchronizationcodealongwithtransmitted data to be sufficiently different from the information samples. The ST2103 involves the use of a pseudo-random sync code generator. These codes are bit streams of '0's &'1's whose occurrence is detected by some rules. The Pseudo - Random Sync Code gets its namefromthefactthattheoccurrenceof'0's&'1'sinthestreamisrandom for a portion of sequence i.e. there is equal probability of occurrence of '0' and'1 '. This portion of sequence is 15 bit long on ST2103. On the receiver the pseudo- random sync code detectorrecognizesthePseudorandomcode&useittoidentify,whichincomingdatabit is associated, with which transmitter timeslot The advantage of this technique is that if the synchronization is temporarily lost, due to noise corruption, it can be re-established as the signal clears. Hence there is minimal loss of transmitted information. Also this technique reduces the separate link required for synchronization signal transmission.

Mode 1: Mode 1 is TDM system of three transmission links between transmitter & receiver. They are information, TX clock & TX.TO (synchronization) signal links. The Pseudo random sync code generator& Detector are switched OFF in this case.

Mode 2: Mode 2 is TDM system of two transmission links between transmitter & receiver. These are information & TX clock signal links. The synchronization is established by sync codes transmitted along with the data stream. No need to say thatthe pseudo random sync generator & detector are switched ON.

*Mode3:*Mode3 is TDM system of onelink between transmitter & receiver, namely the link carrying information. Synchronization is again established by the sync codes. The clocksignalisregeneratedbythephaselockedloop(PLL)circuitatthereceiverfromthe transition of the information data bits.

BLOCKDIAGRAM:-

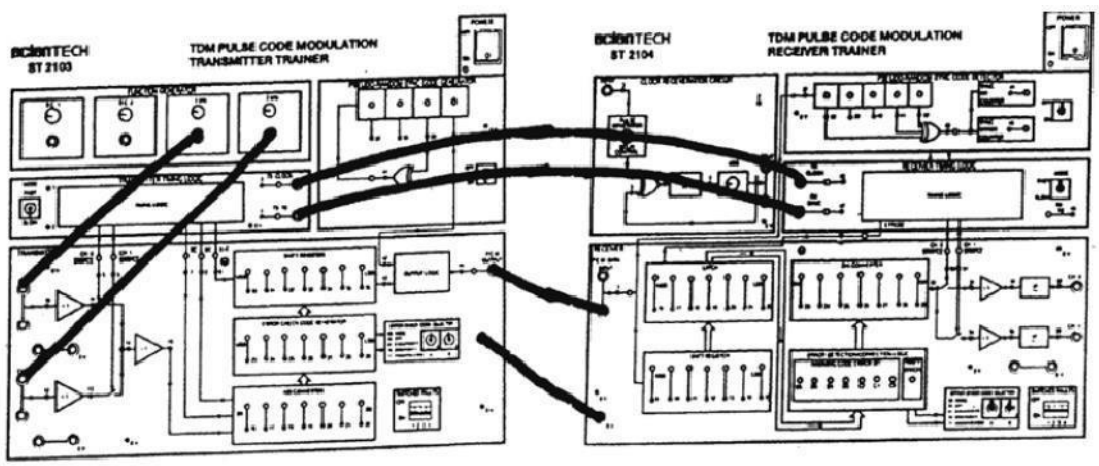


Figure3:Block diagram to study TDM

PROCEDURE:-

1. SetupthefollowinginitialconditionsonST2103:
   Mode Switch in fast position

DC1&DC2Controlsinfunctiongenerator blockfullyclockwise.
~1KHzand~2KHz controllevelssettogive10Vpp.
Pseudo-randomsynccodegeneratoron/offswitchinOFFPosition.
Errorcheckcodegeneratorswitch A&BinA=0&B=0position(OFFMode) All switched faults off.

2. First,connect onlythe1KHzoutput to CH0.
3. TurnONthepower.CheckthatthePAMoutputof1KHzsinewaveisavailableat t.p.15oftheST2103.
4. Connectchannel1of theoscilloscopeto t.p.10&channel2oftheoscilloscopeto t.p. 15.Observethetiming&phaserelationbetweenthesamplingsignalt.p.10&the sampled waveform at t.p.15.
5. TurnOFFthepower supply.Nowconnect also the2KHzsupplyto CH 1.
6. Connectchannel1oftheoscilloscopetot.p.12&channel2oftheoscilloscopeto t.p. 15.
7. Observe theindividualsignals,time division multiplexedandfinally demodulated and demultiplexed signal.

RESULT:-

TimeDivisionMultiplexinghasbeen studied.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Beforeconnectingthepowersupplyplugintosocket,ensurepowersupplyshouldbe switched off.
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

Whatismultiplexing?
Ans. Multiplexing is the technique used to send to information from a number of users through common channel.
Whatistheadvantageofmultiplexing?
Ans.BW utilization of channel is efficient.
Classifymultiplexingtechniques.
Ans. 1.Time Division Multiplexing 2.Frequency Division Multiplexing 3. Wavelength Division Multiplexing
ForwhatkindofsystemsTDMismoreappropriate? Ans.
Digital Systems.
ForwhatkindofsystemsFDMismoreappropriate? Ans.
Analog sytems.
WritedownoneexampleofElasticstore. Ans.
Tape-recorder.
WhatistheBWavailabletoeachuserin caseofTDM?

Ans.ItssameasthatofchannelBW.
ForwhatpurposecommutatorisusedinPAM-TDM? Ans. To allocate time slots to different users.
Why PCM-TDM is used?
Ans.Tousechannelefficiently.
WhatarestrategiesfortimeslotallocationinTDM? Ans. 1.Uniform 2.Non-uniform.

EXPERIMENT No. 10

AIM:--TostudythepulsecodemodulationanddemodulationwithparityandHamming codes.

APPARATUSREQUIRED:-(i)C.R.O.(ii)CROProbe(ii)TDMPulseCode Modulation Transmitter Trainer (ST 2103) and TDM Pulse Code Modulation Receiver Trainer (ST 2104) (iv) Connecting leads.

THEORY:-

PCMisadigitalprocess. Inthisinsteadof sendingapulsetraincapableof 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 communicationchannel coupledwiththeemergenceofrequireddevicetechnology.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 operationsinthereceiverareregenerationofunpairedsignal,decodinganddemodulation of train of quantized.

StepsinPulseCodeModulation :

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.

So,*SamplingFrequency*≥2 fm

≥ 2x3.4 KHz

≥ 6.8 KHz

Practically, the sampling frequency is kept slightly more than the required rate. In telephonythestandardsamplingrateis8KHz.Samplequantifiestheinstantaneousvalue ofthe analogsignal point at samplingpoint to obtain pulseamplitudeoutput. Allocation ofBinaryCodes:Eachbinarywordefines aparticularnarrowrangeofamplitudelevel. The sampled value is then approximated to the nearest amplitude level. The sample is thenassignedacodecorrespondingtotheamplitudelevel,whichisthentransmitted.This process is called as *Quantization* & it is generally carried out by the A/D converter.

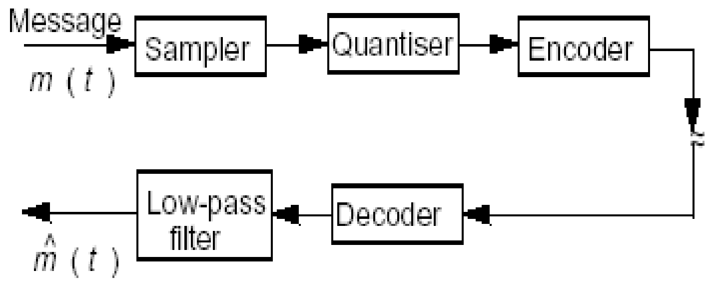


Figure 14.1 A single-channel PCM transmission system.

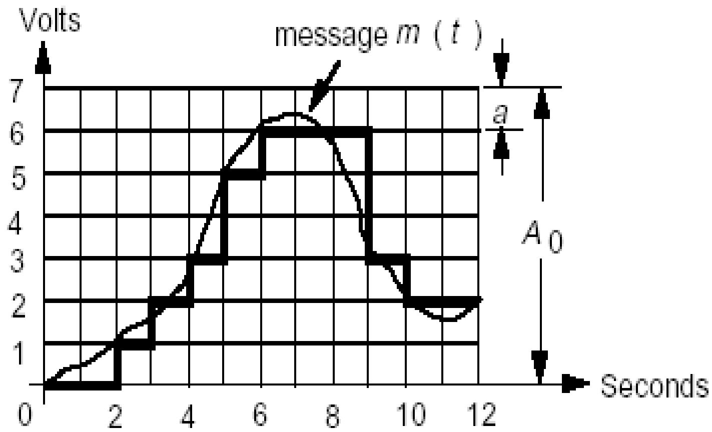


Figure 14.2 Message and quantised signal.

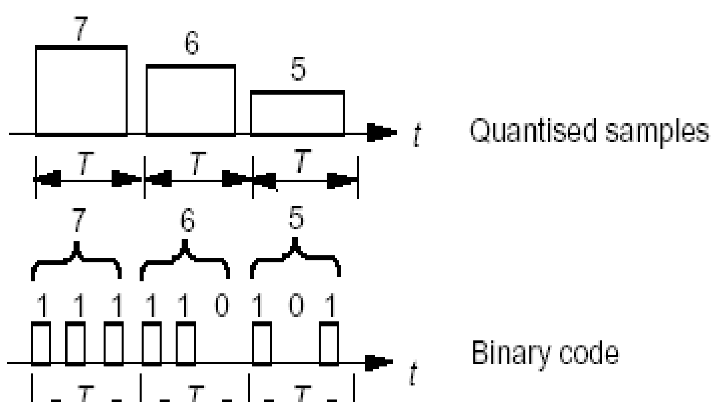


Figure1: Process of PCM

Manydifferenttypesofcodeshavebeendevelopedandareinusetoreduceerrorindigital communication. The commonly usedCodes employed in ST2103 & ST2104 are:

<u>ParityCoding</u>:

It is the simplest method of error coding. Parity is a method of encoding such that the numberof1'sinacodewordiseitherevenorodd Signalparityis establishedas follows. Each word is examined to determine whether it contains an odd or even number of '1' bits. If even parity is to be established (known as Even parity), a '1' bit is added to each wordcontainingodd'1'anda'0'bitisaddedtoeachwordcontainingeven'1'sotheresult isthatallthecode words containan evennumber of1bits after encoding. Similarly,the paritycodingcanensurethatthetotalnumberof'1'sintheencodedwordisodd. Insuch number of '1's in the encoded word is odd. In such cases it is called as *odd parity*. Continuing with the example of even parity, after transmission, each code word is examined to see if it contains an even numberof 1 bits. If it does not, the presence of an errorisindicated. Ifitdoes,theparitybitremainsandthedataispassedtotheuser.Note that single bit parity code can detect single errors only and it cannot provide error correction because there is no way of knowing which bit is in error. It is for this reason thatparitycodingisnormallyonlyusedontransmissionsystemswheretheprobabilityof error occurring is deemed to be low.

<u>HammingCoding</u>:

Hamming coding, decode each word at transmitter into a new code bystuffingthe word with extra redundant bits. As the name suggests, the redundant bits do not convey information but also provides a method of allowingthe receiver to decide when an error has occurred & which bit is in error since the system is binary, the bit in error is easily corrected.

Threebit hammingcodeprovidessinglebiterrordetectionandcorrection.

TheST2103& ST2104involves theuseof7 bit word. Thereforeonlyfourbits are used for transmitting data if hamming code is selected. The format becomes.

D6D5D4D3C2C1 C0

WhereC2, C1&C0are *HammingCodeBits*.

HammingcodewasinventedbyR.W.Hamming.Itusesthreeredundantbits,asopposed tothesingleredundantbitneededbysimpleparitychecking. Butitprovidesafacilityof single bit error detection & correction.

CodeGenerationonTrainer

Thecodeonthistrainerisgeneratedbyaddictingparitycheckbittoeachgroupasshown below :

Group1 D6,D5,D4ParityBit - C2
Group 2 D6, D5, D3 ParityBit - C1
Group3D6,D4,D3ParityBit–C0

The Groups & Parity bit forms an even parity check group. If an error occurs in any of thedigits,theparityislost&canbedetectedatreceivere.g.Letusencodebinaryvalue D6, D5, D4, D3 of '1101'

Group1D6D5D4C2 1
1 0 0
Group2D6D5D3C1 1
1 1 1
Group3D6D4D3C0 1
0 1 0

So,thedatawordaftercodingwillbe D6
D5 D4 D3 C2 C1 C0
1 1 0 1 0 1 0

At the receiver, the four digits representing a particular quantized value are taken in as threegroups.TheErrorDetection/CorrectionLogiccarriesoutevenparitychecksonthe three groups.

Group1D6D5D4C2. Group
2 D6 D5 D3 C1 Group 3 D6
D4 D3 C0

If none of them fails, then no error has occurred in transmission & all bit values are valid.

BLOCKDIAGRAM:-

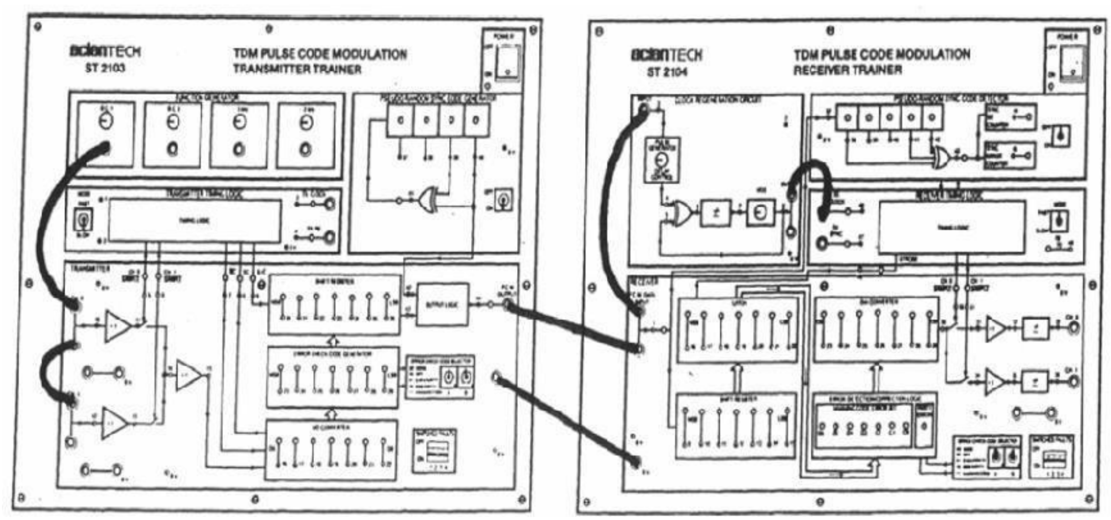


Figure2:PCMwithErrorcheckcodes

PROCEDURE:-

1. Maketheconnection accordingtotheblockdiagram.
2. ObservePCMoutputon CROatthePCMOUTtp.ontheST2103.
3. Notethevariationsinthedigitaloutputaspervariationsinthevalueof DC1.
4. Observetheoperationoferrorcheck codesbyputtingswitchesA&Brespectivelyin positions 00, 01, 10 &11.
5. Changeinput from DC1 to1kHzand2kHzsinusoidal signals and repeat fromstep2 to 4.
6. ObservethedemodulatedPCMoutputonST2104output point.

RESULT:-

Pulsecodemodulationand demodulationisstudiedwitherrorcheckcodes.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Beforeconnectingthepowersupplyplugintosocket,ensurepowersupplyshouldbe switched off.
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.

5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

QUIZ/ ANSWERS:-

InwhichcategoryofPMisPCM? Ans.
Digital Modulation
WhichnoiseisoccursinPCM? Ans.
Quantization Noise
WhatisQuantization?
Ans.InPCM,thetotalamplituderangewhichis signalmaybedividedintonumberof standard level is called quantization.
WhichnoiseisoccursinPCM Ans.
Quantization noise.
Howanalogsignalcanbeencodedintobits/ Ans. By
delta modulation technique
WhatistheadvantageofDMover PCM?
Ans.DMneedsasimplecircuitascomparedtoPCM.
What is the advantage of
PCM?Ans.InPCM,S/NratioismorethanD
M
Atwhichfactorbandwidth ofPCMdepends?
Ans.Itdependsuponthebitduration i.e.samplingperiod/totalno.ofbits.
WhatisElastic store?
Ans.Adevicewhichcanstoreandreproducedataatdifferentspeed isElasticstore.
WritedownoneexampleofElasticstore. Ans.
Tape-recorder.

EXPERIMENT No. 11

AIM:-To Studypulsedata codingand decoding formats.

APPARATUS:-Trainer Kit,Powersupply,ConnectingWires.

THEORY:-

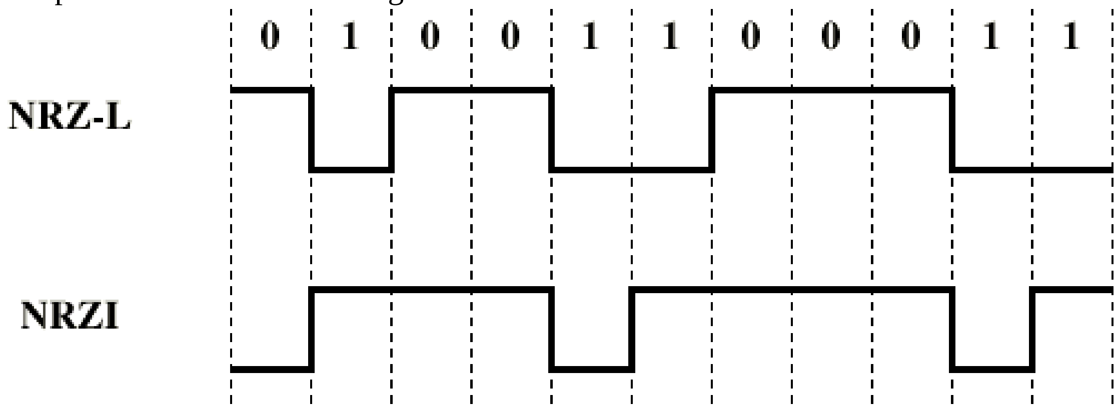
Encoding Schemes: Non-return to Zero-Level (NRZ-L), Nonreturn to Zero Inverted (NRZI), Manchester and Differential Manchester.

NonreturntoZero-Level(NRZ-L)

Twodifferentvoltages aretherefor0 and1bits.Voltageconstant duringbit interval no transition I.e. no return to zero voltage e.g. Absence of voltage for zero, constant positive voltage for one More often, negative voltage for one value and positive for the other, this is NRZ-L.

NonreturntoZero Inverted

Nonreturntozeroinvertedon onesConstantvoltagepulsefordurationofbit Data encoded as presence or absence of signal transition at beginning of bit time Transition(lowtohigh orhigh tolow)denotesabinary1 No transitiondenotesbinary0 An example of differential encoding



DifferentialEncoding

- Datarepresented bychangesratherthanlevels
- Morereliabledetectionoftransitionratherthanlevel
- Incomplextransmissionlayoutsitiseasytolosesenseofpolarity

TRANSISTORSERIESVOLTAGEREGULATOR

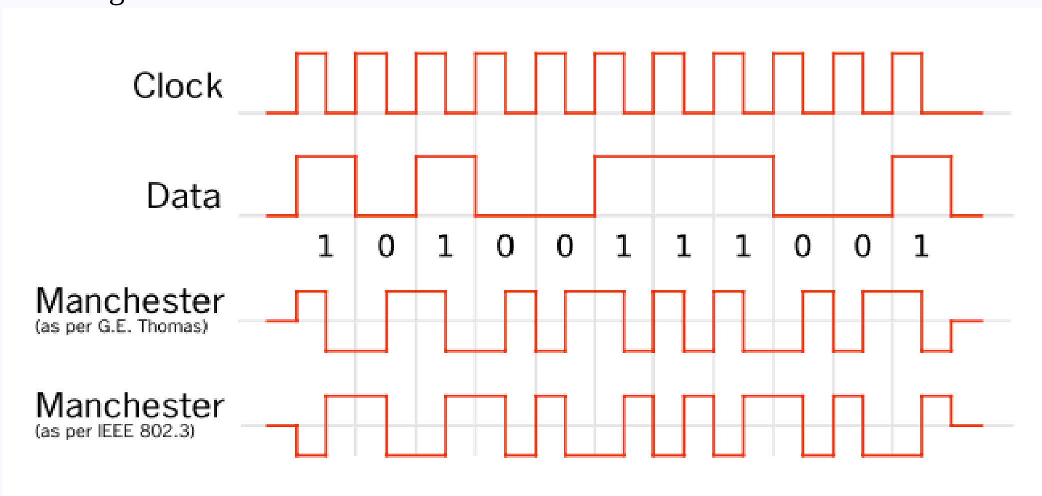
AvoltageRegulatorgenerallyemploysomeactivedevicessuchaszener,oratransistor orbothtoachieveitsobjective.Aseriesvoltageregularusingatransistorandzenerdiode is as shown,

The circuit is called a series voltage regulator because the load current passes through the series transistor Q1.
Themaindrawbackofseriesregulatoristhatthepasstransistorcanbedestroyedby excessive load current.

Machester code

Intelecommunication,Manchestercode(alsoknownasPhaseEncoding,orPE) isa for of data communications line code in which each bit of data is signifid by at least

onevoltagelevel transition.



Manchester encoding is therefore considered to be self-clocking, which means that

accuratesynchronisationofadatastreamispossible.Eachitistransittedovera predefinedtimeperiod.

Manchester coding provides a simple way to encode arbitrary binarysequences without everhavinglongperiodsithoutleveltransitions,thuspreventingthelossofclock

synchronisation,orbit errorsfromlow-frequencydriftonpoorly-equalizedanaloglinks (see ones-density). If transitted as an AC signal it ensures that the DC component of thencodedsigaliszero,againpreventingbaseline driftoftherepeatedsignal,making

iteasytoregenerateandpreventingwasteofenergy.However,therearetodaymany moresophisticated codes (8B/10Bencoding) which accomplish the same aims with less banwidthoverhead,andlesssynchronisationambiguityinpathologicalcases.

PROCEDURE:

1 Maketheconnectionaccordingtotheblockdiagram.Powersupplyshouldbe Switched off .
2 Connect frequency-modulated output to the AM De-Modulator input. connections should be tight.
3 ConnecttheDe-ModulatoroutputtoCRO.
4 ObserveoutputonCRO.Takeoutputcarefully.

RESULT: Differentpulsedatacodinganddecodingformats havebeenstudied.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshould beswitchedoffaftercompletion

QUIZ/ ANSWERS:-

Whatarethenumberofsymbolsavailableinbinary? Ans. 2.
What is RZ?
Ans.Return-to-Zero.
Q.2 What is NRZ?
Ans.NonReturn-to-Zero.
Whyencoding isused?
Ans.Torepresentquantizedsamplesinappropriatedigitalformat.
Whatistheunitofdatarate? Ans.
Bits/s.
Whydecoder isused?
Ans.Toconvertdigitaldataintodiscretesamplevalues.
What is the advantage of
PCM?Ans.InPCM,S/NratioismorethanD
M
Atwhichfactorbandwidth ofPCMdepends?
Ans.Itdependsuponthebitduration i.e.samplingperiod/totalno.ofbits.
Whatisno.ofbitsrequiredtorepresentasample ofasystemof7symbols? Ans. 3
Whychannelencoderisused? Ans.
To avoid errors.

EXPERIMENT No. 12(a)

AIM:--Study of Amplitude Shift Keying.

APPARATUSREQUIRED:-ASKmodulationkit,CROandconnecting leads.

THEORY:-

The binary ASK System was one of the earliest forms of digital modulation used in wireless telegraphy. Thissimplest 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-offkeying (OOK).

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

Then,thebinaryASKsignalcanberepresentedbyawaves(t)givenby $S(t) = A_c\cos(2\pi f_c t)$ symbol1

$=0,$ symbol0

AtypicalASKwaveformisillustratedinfigureforabinarydatarepresentedby {10110101}

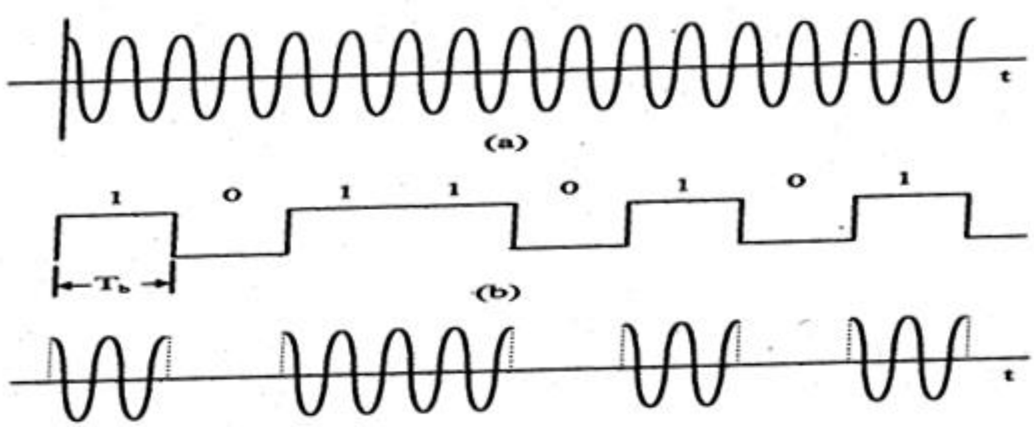


Figure1:ASKwaveforms:(a)Unmodulatedcarrier(b)Unipolarbitsequence(c)ASK wave.

GenerationOfASKSignal

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 (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 signalwhichisbasically the productof the binary sequence andthe carrier signal
.

BLOCKDIAGRAM:-

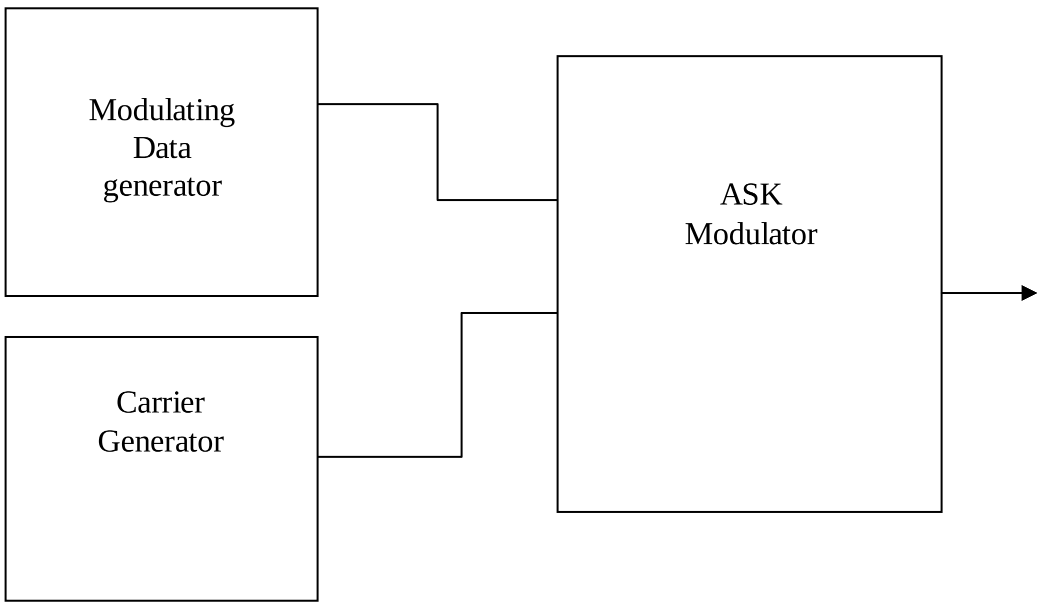


Figure2:BlockdiagramforASKGeneration

PROCEDURE:-

1. Maketheconnectionaccordingto thecircuitdiagram.
2. Connectthemodulator outputtoCRO.
3. Observeoutput on CRO.

RESULT:-:-ASKoutputisobtainedonCRO.

PRECAUTIONS:-

I.Donotuseopen endedwiresforconnectingto230Vpowersupply.
2. Before connecting thepower supplyplug into socket, ensurepower supplyshould be switched
Off.
3. Ensureallconnections shouldbetightbeforeswitchingonthepower supply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment.

EXPERIMENT No. 12(b)

AIM:--StudyofFrequencyShiftKeying.

APPARATUSREQUIRED:-Datagenerator, FSKmodulationkit,CROand connecting leads.

THEORY:-

FSKis oneofthe basic modulation techniquesforthetransmission ofdigital data.Ifthe frequency of the sinusoidal carrier is switched depending upon the input digital signal , thenit 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.
Frequencyshiftkeyingconsistsofshiftingoffrequencyofcarrierfromamaskfrequency toaspacefrequencyaccordingtothebasebanddigitalsignal.Frequencyshiftkeying is identical to modulating an FM carrier with a binary digital signal
In an FSK system, two sinusoidal carrier waves of the same amplitude A_c but different frequencies $f_{c1}$and $f_{c2}$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 f_{c1} and other with a frequency $f_{c2.}$ No discrete components appear in the signal spectrum of FSK signal. The main advantage of FSK lies in its easy hardware implementation.

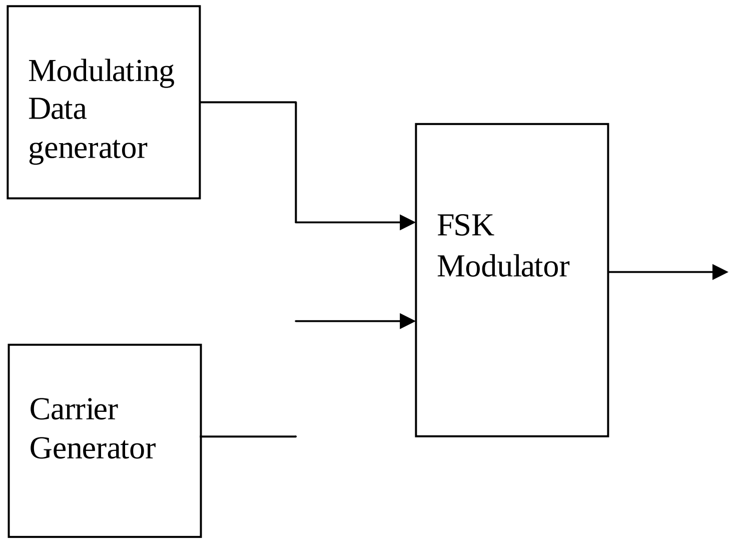
GenerationofFSKsignal:-

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 A_c and frequency f_c is applied. As the modulating voltages changes from one level to another, the frequency modulator output changes its frequency in the corresponding fashion.

DetectionofFSKsignal:-

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.

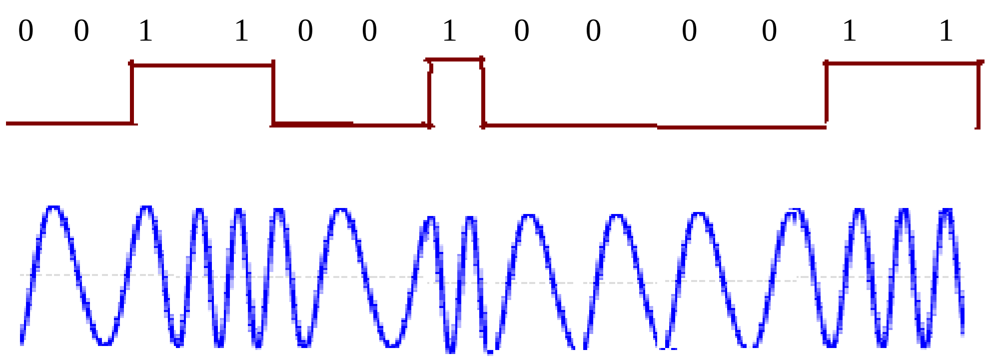
BLOCKDIAGRAM:-



PROCEDURE:-

1. Maketheconnection accordingtotheblockdiagram.
2. ConnectthemodulatoroutputtoCRO.
3. ObserveoutputonCRO.

WAVEFORMS:-



RESULT:-FSK output is obtained on CRO.

PRECAUTIONS:-

1. Do not use open ended wiresforconnectingto230Vpowersupply.

2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment

QUIZ/ ANSWERS:-

Whatis FSK?

Ans.Thisisoneofthebasicmodulationtechniquesfortransmissionofdigitaldata.The frequency of carrier is switched on or off according to the input digital signal.

WhyFSKispreferredoverASK?

Ans. BecauseofconstantamplitudeofFSKtheeffectofnon-linearity‘sandnoise interference is minimum on signal detection.

whatarevarious componentsofFSKdetector?

Ans. Twosynchronousdetector,differentialamplifier,low-passfilter.

Whatis BFSK?

Ans .InBFSKfrequencyofthecarrierissiftedaccordingtothebinarysymbol keeping the phase of the carrier unaffected.

WhatisthedifferencebetweenFMand FSK?

Ans. FM is aanalog modulationtechnique whereFSK isdigitalmodulation technique.

Q6. HowBFSKsignalisgenerated?

Ans.Aninput signal isprocessedintwo paths eachexistingoflevel shifter andproduct modulator. Onepath uses directlyand otherpath uses invertersignal.Orthogonal carrier signal areused as the other input for the productmodulator. The output of the product modulator are added which generates a BFSK.

Whatisthebandwidth ofBFSK?

Ans.4$f_b$wheref_b-bandwidth oftheinput signal.

Compare bandwidth of BFSK and BPSK. Ans.

BandwidthofBFSK=2(bandwidthofBPSK)

Whatisthedisadvantage ofBFSK?

Ans. Theerrorrateof BFSKismoreascomparedtoBPSK.

HowcanyoudetectFSKbynon-coherentmethod?

Ans. BFSKwaves maybedemodulatedcoherentlyusingenvelopdetectors.

EXPERIMENT No. 13

AIM:-TostudythePSKandQPSK.

APPARATUSREQUIRED:-CRO,experimentalkit,powersupply,connectingleads.

THEORY:-

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

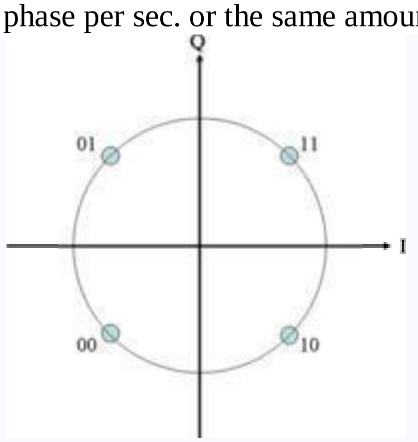
PSK modulator is similar to ASK modulator both used balanced modulator to multiply the carrier with balanced modulator signal. The digital signal with applied to modulation input for PSK generation is bipolar i.e. equal positive and negative voltage level.

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 modulatoris a sine wave which is switched out of phase by 180 from the carrier input.

QudraturePase-shiftKeying(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 90º apart. The four possible combinations at bib it code are 0º, 01, 10, and 11 each code represents either a phaseof45º,185º,225º,and315ºlagging,relativetothephaseattheoriginalun

modulatedcarrierQPSKoffersanadvantageoverPSKisanocarrierthathoweachphase 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 .

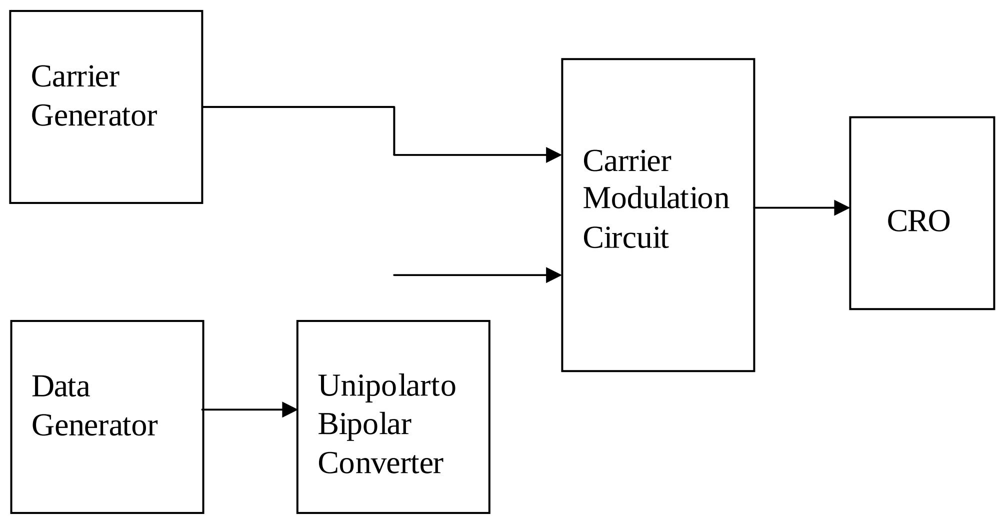


Constellation diagram for PSK with Gray coding. Each adjacent symbol only differs by one bit.

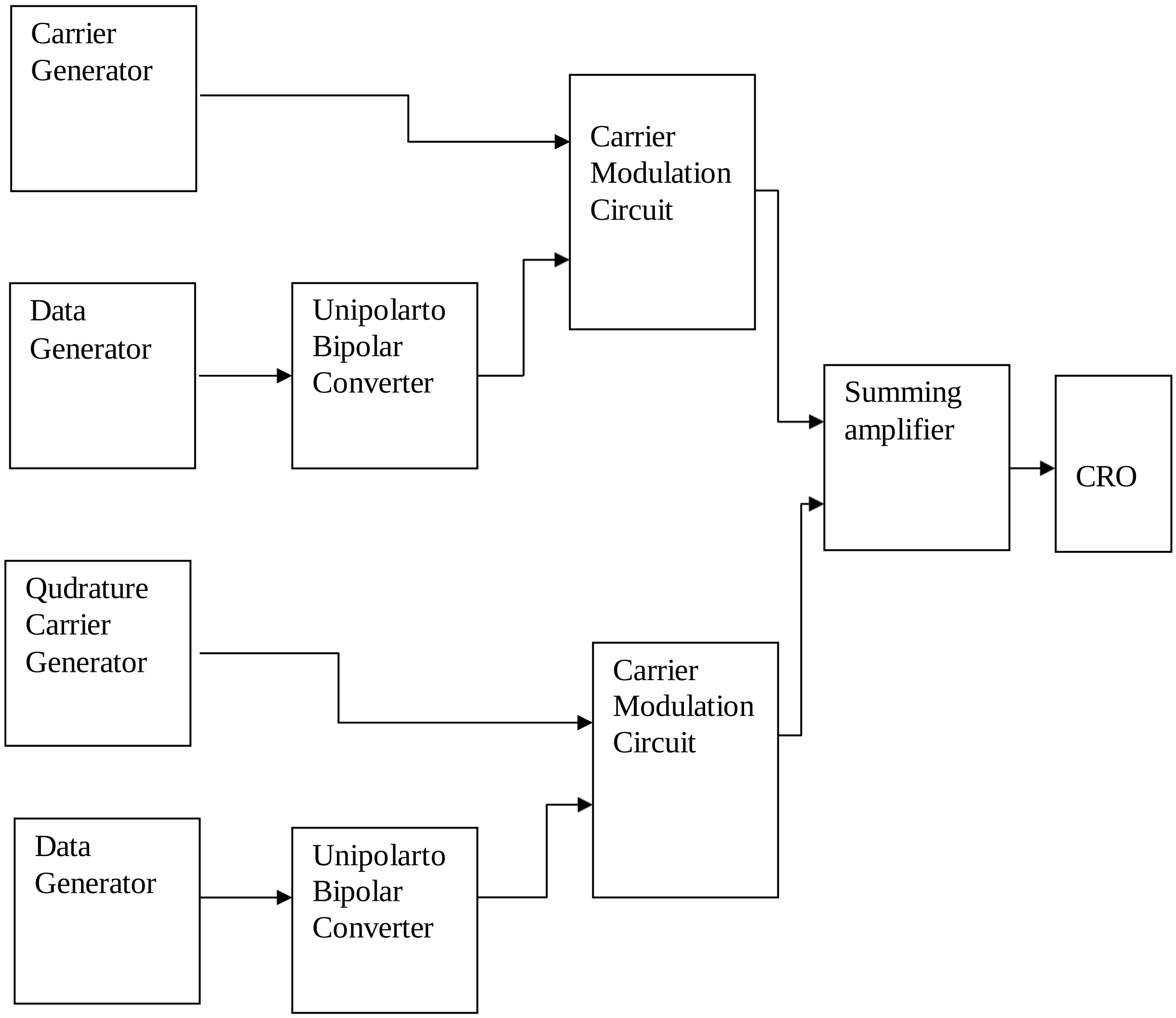
Sometimes known as quaternary or quadriphase PSK or 4-PSK, QPSK uses four points ontheconstellationdiagra,equispacedaroundacircle.Withfourphases,QPSKcanenc de two bits per symbol, shown in the diagra with Gray coding to minimize the BE — twice the rate of BPSK. Analysis shows that this may be used eiter to double thedata rate compared to a BPSKsystem whilemaintainingthe bandwidth ofthesignal or to maintain the data-rate of BPSK but halve the bandwidth needed.

Although QPSK can be viewed as a quaternary modulation, it is easier to see it as two independently modulated quadrature carriers. With this interpretation, the even (or odd) bits are used to modulate the in-phase component of the carrier, while the odd (or even) bits are used to modulate the quadrature-phase component of the carrier. BPSK is used on both carriers and they can be independently demodulated

BLOCKDIAGRAM:-



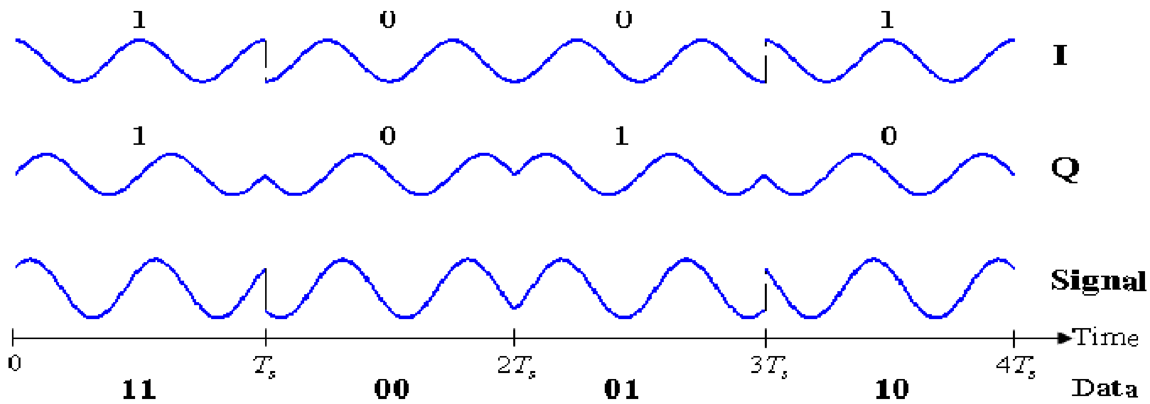
(BlockdiagramofPSK)



PROCEDURE:--

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

WAVEFORM:-



RESULT:-

PSK and QPSK output is obtained on CRO.

PRECAUTIONS:-

1. Donot useopenendedwiresforconnectingto230Vpowersupply.
2. Before connecting the power supplyplug into socket, ensurepower supply should be switched off
3. Ensureallconnections should betightbeforeswitchingonthepowersupply.
4. Takethereadingcarefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment

QUIZ/ ANSWERS:-

WhatisPSK?
Ans. It is one of the basic digital modulation technique. Here the phase of the carrier is switcheddependinguponthe inputdigitalsignal.ThisissimilartothePhase

ModulationandhasconstantAmplitudeenvelope.
WhatisthedisadvantageofPSK?
Ans.Itneedsacomplicatedsynchronizingcircuitofthereceiver.
WhatisBPSK?
Ans.BPSKisBinaryPhaseShiftKeying. Herebinarysymbol 1&0 modulatethephase of the carrier. The phase of carrier change by 180
HowBPSKisgenerated?
Ans. It can be generated by applying carrier signal and base-band signal as modulating signal to a balanced modulator.
WhatistheadvantageofPSK? Ans.
Error rate is less than DPSK.
WhatisthedifferencebetweenQPSKandBPSK?
Ans.InBPSKphaseshiftis 180whereasinPSKthephaseshift is45.
WhatisQPSK?

Ans.InQPSKtwosuccessivebitsarecombined.Thiscombinationoftwobitsformsfour distinguishingsymbols.Whenthesymbolischangedtonextsymbolthephaseofcarrier is changed by 45°°°.

HowQPSKisgenerated?

Ans. The input binarysequence is first converted to a bipolar NRZtype of signal called b(t) than it isdivided bydemultipluxer and added together after insertion of carrier. The generates QPSK signal.

HowQPSKisdetected?

Ans. BasicallyQPSK receiveruses asynchronous reception. A coherent carrierapplied tothetwosynchronousdemodulator,eachconsistsofamultiplierandanintegrator.The output is detected original signal.

WhatisDPSK?

Ans.DPSKisdifferentialphaseshiftkeyingandisa non-coherentversusofPSK.DPSK does not need acoherent carrierat thedemodulator.Theinput sequenceof binarybits is modified such that the next bit depends upon the previous bit.

EXPERIMENT No. 14

AIM:-To Study Differential pulse code modulation and Demodulation.

APPARATUS:-Trainer Kit, Power supply, Connecting Wires.

THEORY:-

Meaning of DPCM – ―Differential Pulse Code Modulation‖, is a modulation technique invented by the British Alec Reeves in 1937. It is a digital representation of an analog signal where the magnitude of the signal issampled regularly at uniform intervals. Every sample is quantized to a series of symbols in a digital code, which is usually a binary code. PCM is used in digital telephone systems. It is also the standard form for digital audio in computers and various compact discformats. Several PCM streams may be multiplexed into a larger aggregate data stream. This technique is called Time- Division Multiplexing. TDM was invented by the telephone industry, but today the technique is an integral part of many digital audio workstations such as Pro Tools. In conventional PCM, the analog signal may be processed (e.g. by amplitude compression) before being digitized. Once the signal is digitized, the PCM signal is not subjected to further processing (e.g. digital data compression). Some forms of PCM combine signal processing with coding. Older versions of these systems applied the processing in the analog domain as part of the A/D process, newer implementations do so in the digital domain. These simple techniques have been largely rendered obsolete by modern transform-based signal compression techniques.

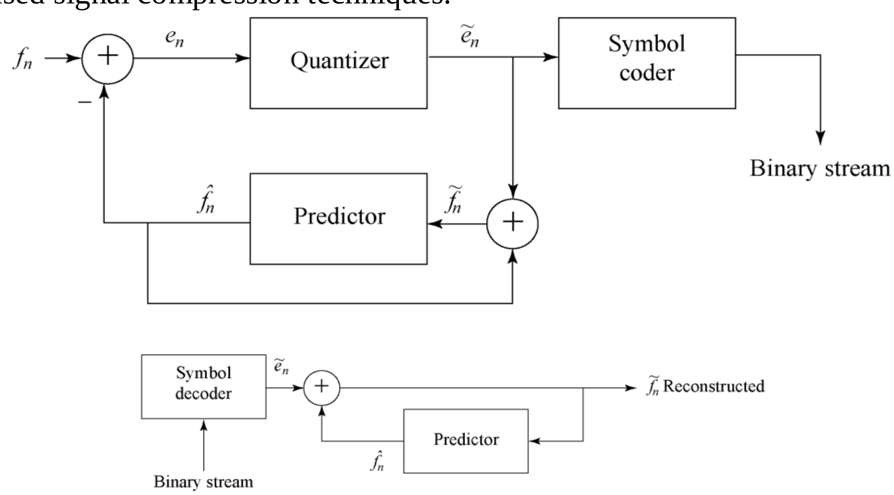


Figure1:DPSKSytem

In practical system bandwidth requirement for the transformation of information is very important aspect, since if bandwidth requirementis less more number of channels can be multiplexed on a single line and full utility of transmitting media is extracted out.

In asysteminwhichabasebandsignalm(t)istransmittedbysampling,thereisavailable aschemeoftransmissionwhichisanalternativetotransmittingthesamplevaluesateach samplingtime.Wecaninstead,ateachsamplingtime,saytheKth

samplingtime, transmit the differencebetween thesample value m(k) at sampling timeK and thesamplevaluem(K-1) at timek-1. Ifsuchchanges aretransmitted, thensimplybyaddingup these changes weshall generate at the receiver a waveformidentical in formto m (t).

PROCEDURE:-

1. Maketheconnectionaccordingto thecircuitdiagram.
2. Observeoutputon CRO.

RESULT:-DPCMmodulationanddemodulationhasbeenstudied.

PRECAUTIONS:-

1. Donotuseopenended wiresforconnectingto230Vpowersupply.
2. Before connecting the powersupplyplug into socket, ensurepower supplyshould be switched off
3. Ensureallconnectionsshouldbetightbeforeswitchingonthepowersupply.
4. Takethereading carefully.
5. Powersupplyshouldbeswitchedoffaftercompletionofexperiment

QUIZ/ ANSWERS:-

WhatisDPCM?
Ans.DifferentialPulseCodeModulation.
Whatis theadvantageofDPCM?
Ans.ItrequirelessBWascomparedtoPCM.
Whatisquantizer?
Ans.Itconvertsthesamplevaluestosomefixedfinitelevels.
Whatistheuseofpredictor?Ans. To
estimate previous sample.
WhichoneisbetterPCMorDPCM?Ans.
DPCM.
IsDPCManalogmodulation technique?
Ans.Itbelongtotheclassofpulsedigitalmodulation.
WhichonehaslessBWrequirementDPCMorDeltamodulation?Ans. Delta
modulation.
DPCMissuitableforwhichkindofinput signals?
Ans.Wheredynamicchangesinsignalaresmall, DPCMIveryusefull.
WhyDPCMispreferredoverPCM?Ans.
Because of low BW.
DPCMispreferablyusedfor………. Ans.
Voice or picture Communication.

This lab manual has been updated by

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Cross checked By
HOD ECE

Verified By
Director, DGI Greater Noida

Please spare some time to provide your valuable feedback.