QMP 7.1 D/F

Channabasaveshwara Institute of Technology

(Affiliated to VTU, Belagavi & Approved by AICTE,NewDelhi)
(**NAAC Accredited & ISO 9001:2015 Certified Institution)**
NH 206 (B.H. Road), Gubbi, Tumkur – 572 216.Karnataka.

Department of Electronics & Communication Engineering

COMMUNICATION LAB

BECL404

B.E - IV Semester Lab

Manual 2023-24

Name:____

USN:____

Batch:___________________________Section:____________________

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Department of Electronics and Communication Engineering

Communication Lab-BECL404
2023-24

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VISION OF THE INSTITUTE

"To create centres of excellence in education and to serve the society by enhancing the quality of life through value based professional leadership"

MISSION STATEMENT OF THE INSTITUTE

- **To provide high quality technical and professionally relevant education in a diverse learningenvironment.**
- **To provide the values that prepare students to lead their lives with personal integrity, professional ethics and civic responsibility in a globalsociety.**
- **To prepare the next generation of skilled professionals to successfully compete in the diverse globalmarket.**
- **To promote a campus environment that welcomes and honors women and men of all races, creeds and cultures, values and intellectual curiosity, pursuit of knowledge and academic integrity andfreedom.**
- **To offer a wide variety of off-campus education and training programmes to individuals andgroups.**
- **To stimulate collaborative efforts with industry, universities, government and professionalsocieties.**
- **Tofacilitatepublicunderstandingoftechnicalissuesandachieveexcellencein the operations of the institute.**

QUALITY POLICY OF THE INSTITUTE

Our organization delights customers (students, parents and society) by providing value added quality education to meet the national and international requirements. We also provide necessary steps to train the students for placement and continue to improve our methods of education to the students through effectivequality

management system, quality policy and quality objectives.

VISION OF THE DEPARTMENT

“To create globally competent Electronics and Communication Engineering professionals with ethical and moral values for the betterment of the society”

MISSION OF THE DEPARTMENT

- **To impart quality technical education in the field of electronics andcommunication engineering to meet over the current/future global industryrequirements.**
- **To create the centres of excellence in the field of electronics and communication in collaboration with industry anduniversities**
- **To nurture the technical/professional/engineering and entrepreneurial skillsfor overall self and societaluplifment**
- **To orient the student community towards the higher education, research and developmentactivities**
- **To provide a platform for equipping the students with necessary skills through co- curricular and extra-curricularevents.**
- **To have Industrial collaboration for strengthening the Teaching-Learning Process/Academics**
- **To associate with industries for training the faculty on the latest technologies through continuous educationprogrammes.**

PROGRAM EDUCATIONAL OBJECTIVES

PEO1 : Provide technical solutions to real world problems in the areas of electronics and communication by developing suitable systems.

PEO2 : Pursue engineering career in Industry and/or pursue higher education an Research.

PEO3 : Acquire and follow best professional and ethical practices in Industry and Society.

PEO4 : Communicate effectively and have the ability to work in team and to lead the Team

PROGRAM SPECIFIC OBJECTIVES

PSO1: Specify, design, build and test analog and digital systems for signal processing including multimedia applications, using suitable components or simulation tools.

PSO2: Understand and architect wired and wireless analog and digital communication systems as per specifications and determine their performance.

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

SYLLABUS

Communication LAB

Subject Code: BECL404 — IA Marks: 40

No. of Practical Hrs/Week: 02(Tutorial)+02(Laboratory) — Exam Hours: 03

Total no. of Practical Hrs.: 42 — Exam Marks: 60

1. Design and test a high-level collector Modulator circuit and Demodulation the signal using diode detector.

2.Test the Balanced Modulator / Lattice Modulator (Diode ring)

3.Design a Frequency modulator using VCO and FM demodulator using PLL (Use IC566 and IC565).

4.Design and plot the frequency response of Preemphasis and Deemphasis Circuits

5.Design and test BJT/FET Mixer

6 Design and test Pulse sampling, flat top sampling and reconstruction

7. Design and test Pulse amplitude modulation and demodulation.

8. Generation and Detection of Pulse position Modulation

9. Generation and Detection of Pulse Width Modulation

10. PLL Frequency Synthesizer

11. Data formatting and Line Code Generation

12.PCM Multiplexer and Demultiplexer

Communication Lab Course Objective's and CourseOutcome's

Course objectives:

This laboratory course enables students to

- **Understand the basic concepts of AM and FM modulation and demodulation.**
- **Design and analyse the electronic circuits used for AM and FM modulation and demodulation circuits.**
- **Understand the sampling theory and design circuits which enable sampling and reconstruction of analog signals.**
- **Design electronic circuits to perform pulse amplitude modulation, pulse position modulation and pulse width modulation..**

Course outcomes (Course Skill Set):

At the end of the course the student will be able to:

1. Illustrate the AM generation and detection using suitable electronic circuits.
2. Design of FM circuits for modulation, demodulation and noise suppression.
3. Design and test the sampling, Multiplexing and pulse modulation techniques using electronic hardware.
4. Design and Demonstrate the electronic circuits used for RF transmitters and receivers.

ProgramOutcomes(PO)

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

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

PO-3: Design/development of solutions: Design/development of solutions: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.

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

PO-5: Modern tool usage: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modelling to complex engineering activities with an understanding of the limitations.

PO-6: The engineering and society: Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal, and cultural issues and the consequent responsibilities relevant to the professional engineering practice.

PO-7: Environment and sustainability: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of need for sustainable development.

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

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

PO-10: Communication: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.

PO-11:Project management and finance: Demonstrate knowledge and understanding for the engineering and management principles and apply these to one‘s own work, as a member and leader ina team,to manage projects and in multidisciplinary environments.

PO-12: Life- long learning: Recognize the need for, and have the preparation and ability to engage inindependent and life-long learning in the broadest context of technological change.

Programme Specific Outcomes(PSO)

PSO1: Specify,design,build and test analog and digital systems for signal processing including multimedia applications,using suitable components or Simulation tools.

PSO2: Understand and architect wired and wireless analog and digital Communication systems as per specifications and determine their performance**.**

‘Instructions to the candidates’

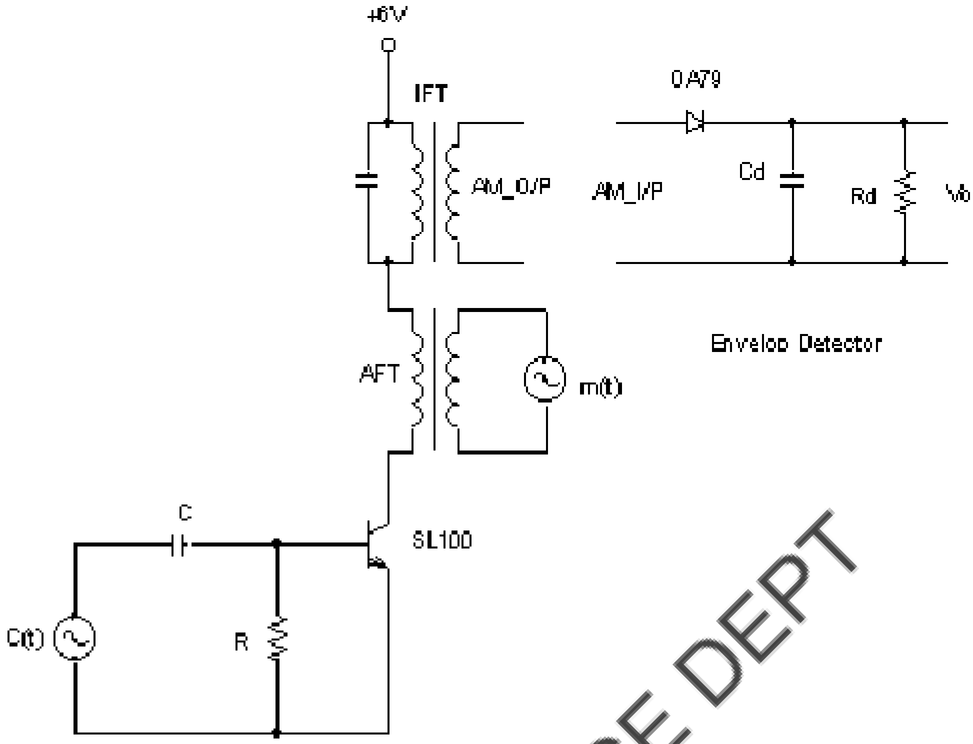
- Student should come with thorough preparation for the experiment to be conducted.
- Student should take prior permission from the concerned faculty before availing theleave.
- Student should come with proper dress code and to be present on time in thelaboratory.
- Student will not be permitted to attend the laboratory unless they bring the practical record fully completed in all respects pertaining to the experiment conducted in the previousclass.
- Student will not be permitted to attend the laboratory unless they bring the observation book fully completed in all respects pertaining to the experiment to be conducted in presentclass.
- Experiment should be started conducting only after the staff-in-charge has checked the circuitdiagram.
- All the calculations should be made in the observation book. Specimen calculations for one set of readings have to be shown in the practical record.
- Wherever graphs to be drawn, A-4 size graphs only should be used and the same should be firmly attached in the practicalrecord.
- Practical record and observation book should be neatlymaintained.
- Student should obtain the signature of the staff-in-charge in the observation book after completing eachexperiment.
- Theory related to each experiment should be written in the practical record before procedure in your own words with appropriatereferences.

CONTENTS

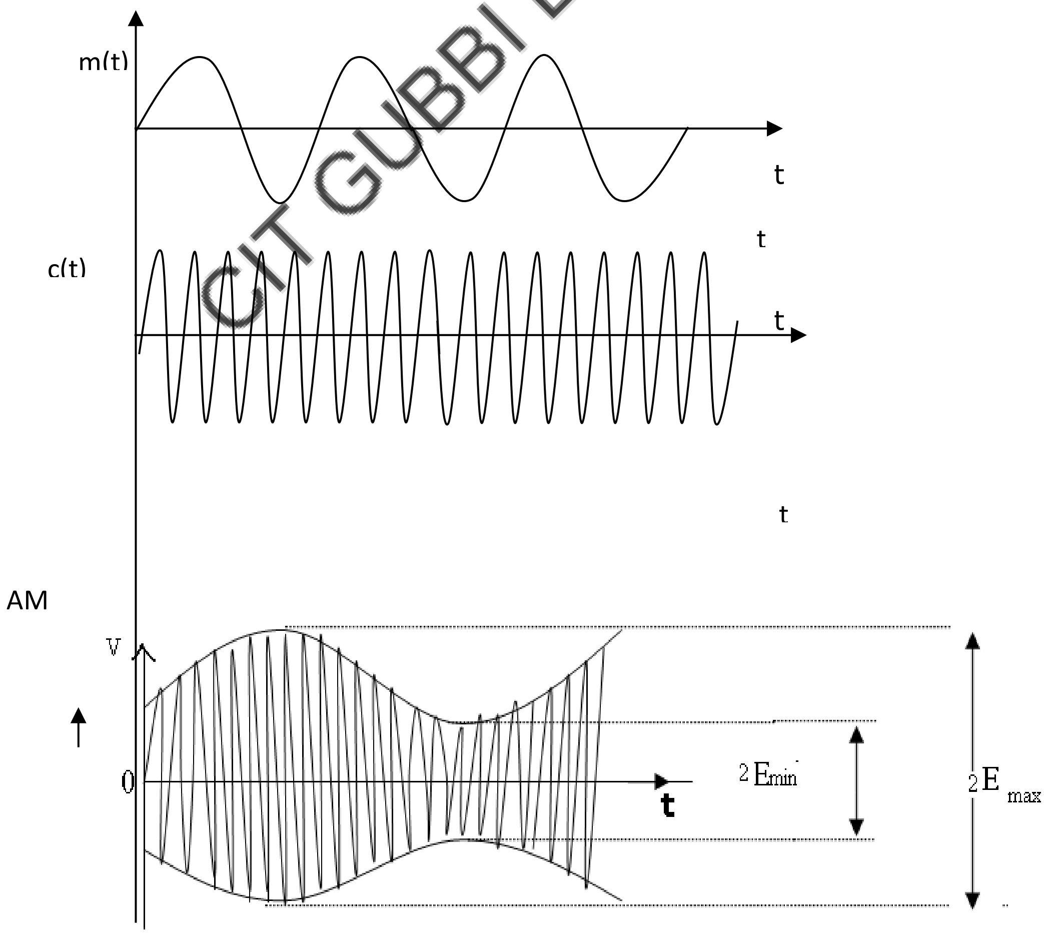
INDEX PAGE

Sl. No	Name of the Experiment	Date			Manual Marks (Max . 20)	Record Marks (Max. 10)	Signature (Student)	Signature (Faculty)
		Conduction	Repetition	Submission of Record				
1								
2								
3								
4								
5								
6								
7								
8								
9								
10								
11								
12								
Average								

Circuit Diagram :



Waveforms :



AM Signal

Experiment No. :01

Date:

COLLECTOR MODULATOR

Aim:

Design and test a high-level collector Modulator circuit and Demodulation of the signal using diode detector.

Apparatus:

Sl.No.	Particulars	Range	Quantity
1.	Transistor	SL100	01
2.	Resistors & Capacitors	As per design	-
3.	Diode	OA79	01
4.	IFT, AFT	-	01 each
5.	Probes	-	03 set

Theory:

Amplitude modulation (**AM**) is a form of modulation in which the amplitude of a carrier wave is varied in direct proportion to that of a modulating signal. AM is commonly used at radio frequencies and was the first method used to broadcast commercial radio. The term "AM" is sometimes used generically to refer to the AM broadcast (Medium wave) band. In its basic form, amplitude modulation produces a signal with power concentrated at the carrier frequency and in two adjacent sidebands. Each sideband is equal in bandwidth to that of the modulating signal and is a mirror image of the other. Thus, most of the power output by an AM transmitter is effectively wasted, half the power is concentrated at the carrier frequency, which carries no useful information (beyond the fact that a signal is present), the remaining power is split between two identical sidebands, only one of which is needed.

The modulator shown is a high power class C amplifier with high-level modulation. The modulator is a linear power amplifier that takes low level modulating signal and amplifies it to a high power level. The modulating signal is coupled through modulating transformer AFT to the class C amplifier. The secondary winding of the modulation transformer is connected in series with the collector supply voltage Vcc of the class C amplifier. This means that the modulating signal is applied in series with the collector power supply voltage of the class C amplifier applying collector modulation.

In the absence of modulating signal, there will be zero modulation voltage across the secondary of AFT. Therefore, the collector supply voltage will be applied directly to the class C amplifier generating current pulses of equal amplitudes and the output of the tuned circuit will be a steady sine wave.

When the modulating signal occurs, the ac voltage across the secondary of the modulating transformer will be added to and subtracted from the collector supply voltage. This varying supply voltage is then applied to the class C amplifier, resulting in variations in the amplitude of the carrier sine wave in accordance with the modulating signal. Due to this, amplitude of the current pulses also varies in accordance with the modulating signal. The tuned circuit then converts the current pulses into an amplitude modulated wave.

Modulation index

In AM, Modulation index is also called **Modulation depth**, indicates by how much the modulated variable varies around its 'original level'.

For AM,

$$\% m = [(E_{max} - E_{max}) / (E_{max} + E_{min})] * 100$$

If m = 0.5, the carrier amplitude varies by 50% above and below its unmodulated level, and for m = 1.0 it varies by 100%. Modulation depth greater than 100% is generally to be avoided - practical transmitter systems will usually incorporate some kind of limiter circuit. Modulation circuit designs can be broadly divided into low and high level.

Design :

f = 455 kHz ® T=2.2⌠sec

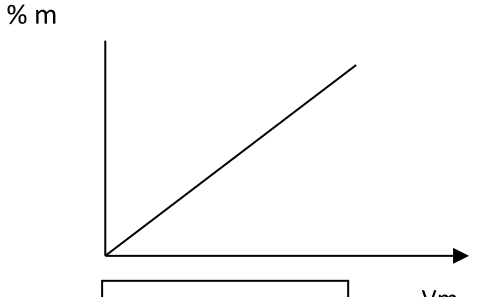
For clamping: R_bC_b >> T 4 Let R_bC_b =100T

With R_b = 10k∧, we get C_b = 0.022⌠F

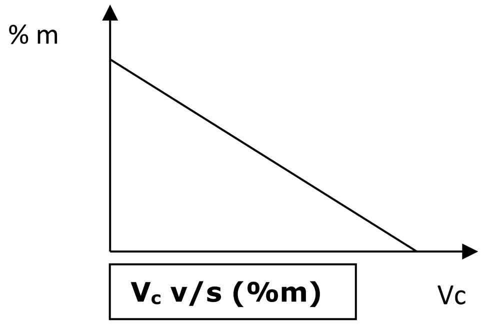
Procedure:

1. Connections are made as shown in the figure.
2. Without applying the message signal m(t), apply the carrier signal C(t) and adjust its frequency until we will get proper output and note down its voltage as carrier voltage Vc.
3. Calculate the carrier Power Pc.
4. Apply the Modulating Signal m(t) and note down E_{max} and E_{min} from modulated signal. Repeat the same for different voltages of m(t).
5. Calculate Modulation Index.

6. Connect the demodulator circuit and obtain demodulated message output.
7. The frequency of the demodulated signal is equal to message signal frequency.
8. Plot a Graph of(V_m v/s %m) and (V_c v/s %m)



V_m v/s (%m)



V_c v/s (%m)

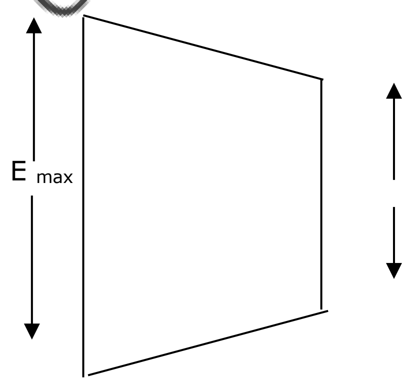
Formulae:

1. Power without modulation = $Pc = V_c^2 / 8R_L$
2. Modulation Index : $\%m = [(E_{max} - E_{max}) / (E_{max} + E_{min})] * 100$
3. Total power of AM Signal, $Pt = Pc[1+(m^2/2)]$

Tabular Column:

Sl.No.	V_m(p-p) volts	V_c (p-p)volts	E_{max}	E_{min}	Modulation index `m` (%)
1.					
2.					
3.					

Transfer Characteristic Curve :

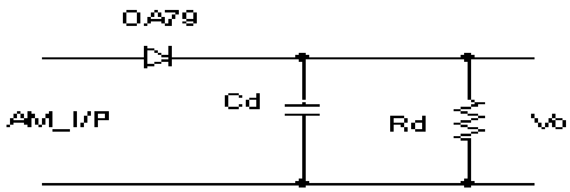


Diode Detector: (Design)

(1/fm) >> Rd Cd >> (1/fc) ; Let Rd Cd = 100/fc

Assume Cd = 0.001 μf, then Rd = 200 kΩ

Demodulation Circuit:



Diode/Envelope Detector

Sl. No.	Vo in Volt	fo in Hz	f_m in Hz

Note: To obtain the Trapezoidal waveform, feed the modulating signal to channel (i) and AM wave to channel (ii), press X-Y knob.

Result :

EXPERIMENT No.: 02 **Date:**

Balanced Modulator / Lattice Modulator (Diode ring)

Aim: Test the Balanced Modulator/Lattice Modulator (Diode ring) and observe DSBSC Waveform.

Components and equipments required:

Sl.No	Apparatus	Range
1	2 Audio Transformers	1:1 turns ratio
2	2 Audio Signal Generators	100Hz – 1 MHz
3	4 Identical Diodes	OA79
4	CRO	Digital

Theory:

DSBSC (Double Side band suppressed carrier) system is a continuous wave modulation scheme where the carrier is suppressed and only the two sideband (lower sideband LSB and upper sideband USB) frequencies are transmitted together. It is used for mixing of colour signals in colour TV transmission. Due to the absence of carrier in DSBSC, the power required to transmit a DSBSC signal is very less compared to that required to transmit an AM signal.

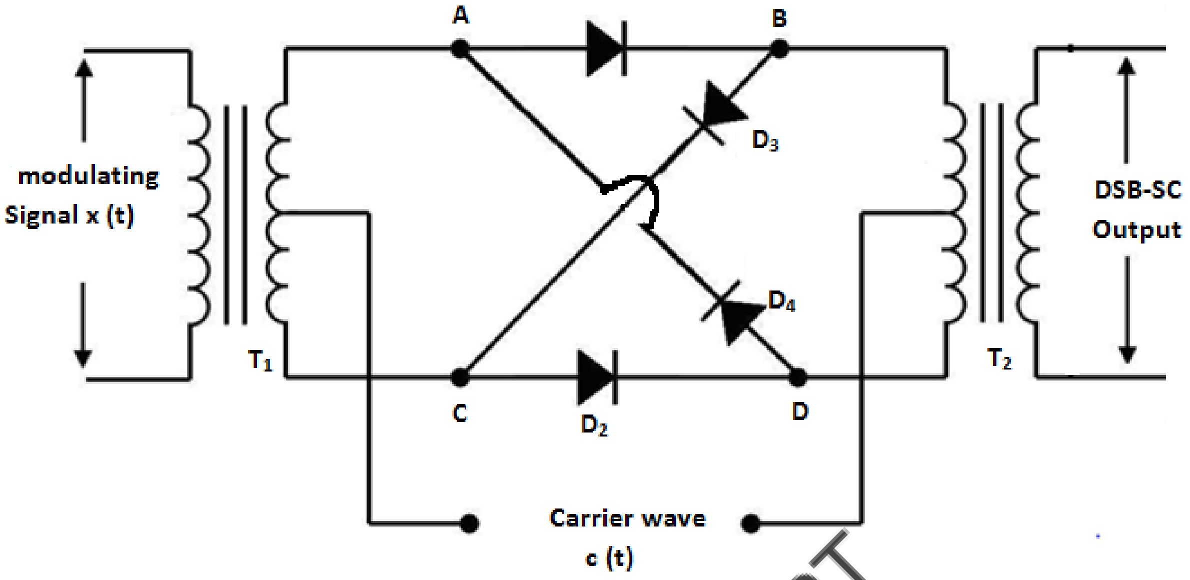
Principle of Working:

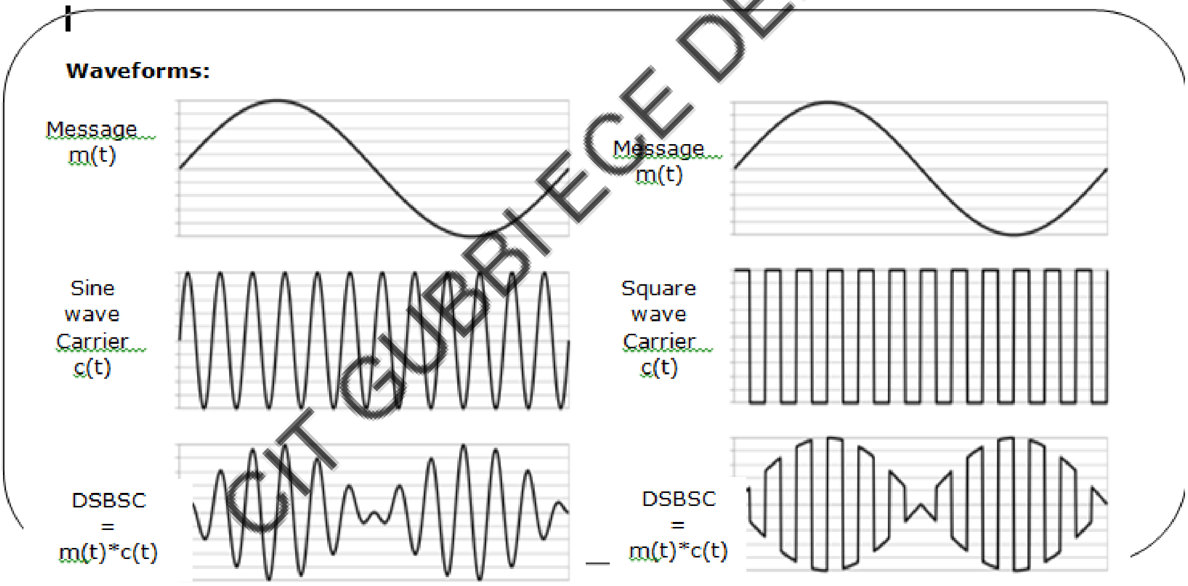
The Circuit used for generating a DSBSC signal is a ring modulator. Four Identical diodes are connected in the form of a ring as shown in the circuit diagram. The output contains only a pair of sidebands symmetrically placed on either side of carrier frequency position in the spectrum. The Input and Output transformers are audio frequency transformers of 1:1 ratio and must be identical.

Procedure:

1. Circuit connections are made as shown in the circuit diagram.
2. The Modulating and the carrier signal generators are switched ON.
3. The frequency of the modulating signal is kept at 500 Hz.
4. The frequency of the carrier signal is kept at 10 KHz.
5. With fine tuning of frequencies of modulating and carrier signals, an perfect DSBSC waveform is observed on CRO.
6. The variation in the DSBSC output is observed by varying the signal amplitude and frequencies of both modulating and carrier signals.

Circuit Diagram:





Result: Peak to Peak amplitude of DSBSC Signal: _________ Volts

EXPERIMENT No.: 3
Date:

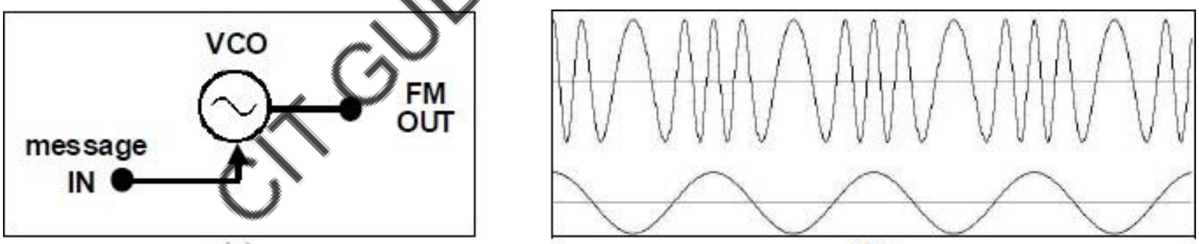
FREQUENCY MODULATION USING VCO AND PLL FM DEMODULATOR.

Aim: To Implement Frequency Modulation using VCO and Phase locked loop (PLL) for FM demodulation.

FM Generation:

A simple and direct method of generating an FM signal is by the use of a voltage controlled oscillator -VCO. The frequency of such an oscillator can be varied by the magnitude of an input (control) voltage. The block diagram of VCO-FM generator is shown in Figure.

For the VCO to work as a frequency modulator, it has to manifest a linear relation between the magnitude of the input signal and the output oscillation. Large signal amplitude may take the system out of its linear range of operation. Therefore a careful design of the deviation sensitivity of the VCO is required to ensure linear operation over the full range of input signal amplitudes.



(a) (b)

Figure.1: FM by VCO (a), and Resulting Output (b). Spectral analysis:

1. FM Bandwidth

In theory FM modulated signal will have an infinite number of sidebands and hence an **infinite bandwidth** but in practice all significant sideband energy (98% or more) is concentrated within the transmission bandwidth $\mathbf{B_T}$ defined by Carson's rule.

$$B_T = 2\Delta f + 2f_m$$
$$= 2f_m(1+\beta)$$
$$= 2\Delta f\ (1+(1/\beta))$$

β: is the FM modulation index and it is equal to Δf/f

There are a few interesting points of summary relative to frequency modulation bandwidth:

1. The bandwidth of a frequency modulated signal varies with both deviation and modulating frequency.
2. Increasing modulating frequency reduces modulation index - it reduces the number of sidebands with significant amplitude and hence the bandwidth.
3. Increasing modulating frequency increases the frequency separation between sidebands.
4. The frequency modulation bandwidth increases with modulation frequency but it is not directly proportional to it

2. FM signal types

There are two types of FM signal:

Narrowband signal (NBFM):

In this type Δf << f_m thus β<<1, and its bandwidth is approximately $2f_m$ based on Carson's rule.

Wideband signal (WBFM):

In this type Δf >> f_m thus β >>1, and its bandwidth is approximate $2\Delta f$ based on Carson's rule.

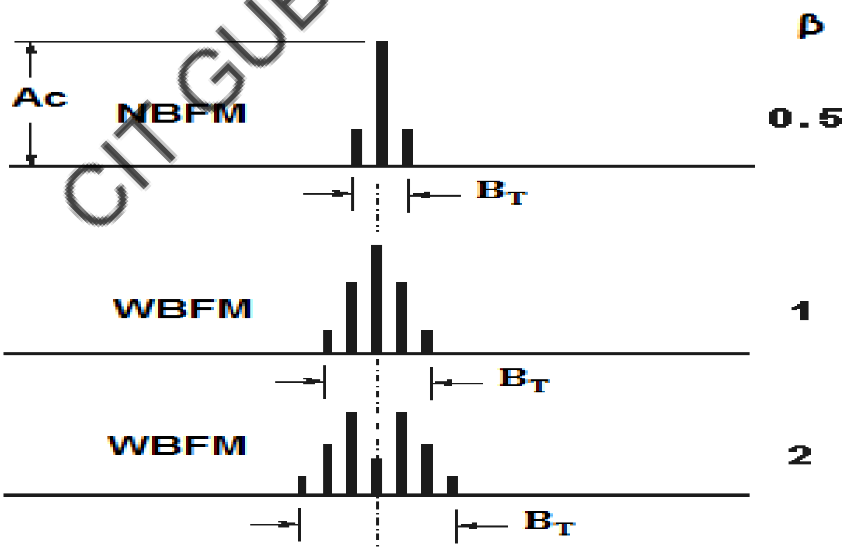


Figure.2 FM signal types

3. Phase Look Loop:

The block diagram of a phase locked loop (PLL) is shown in Figure 1. The principle of operation is simple. Suppose there is a non-modulated carrier at the input. If the VCO was tuned precisely to the frequency of the incoming carrier

(ω_0), then the instantaneous output would be a DC voltage of magnitude depending on the phase difference between the output of the VCO and the incoming carrier. Now suppose the incoming carrier started to drift slowly in frequency, then the output voltage will vary according to the frequency variation. If the incoming carrier is frequency modulated by a message, the output of the PLL will follow the message.

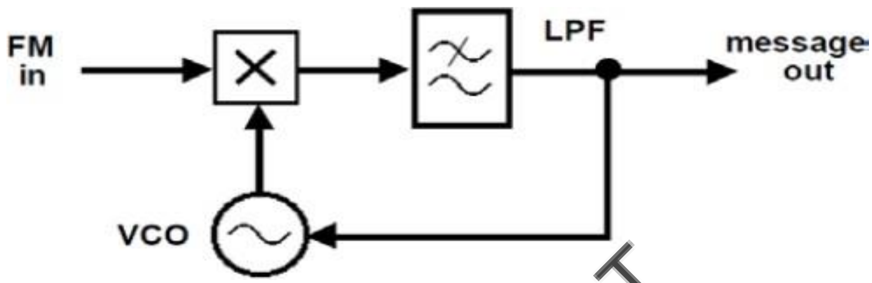


Figure.3 Phase Look Loop (PLL)

4. Frequency discriminator

FM can be demodulated as well by using a differentiator or a frequency discriminator. Frequency discrimination can be achieved by applying the FM signal to the linear part (transition region) of a BPF as depicted in Figure 4. The output of the discriminator is both FM and AM modulated. The message can be recovered by applying the discriminator output to an envelope detector as shown in Figure 5. The BPF of the 100 kHz channel filter module has close-to-linear pattern in the band 80- 90 kHz.

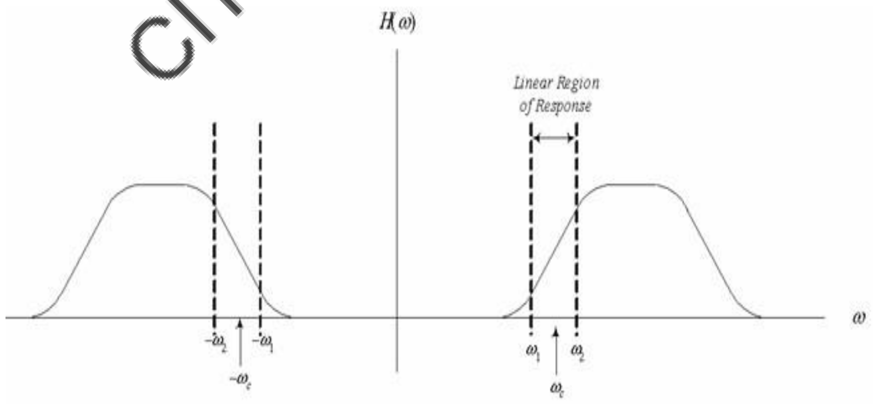


Figure.4: The BPF of the 100 kHz channel filter

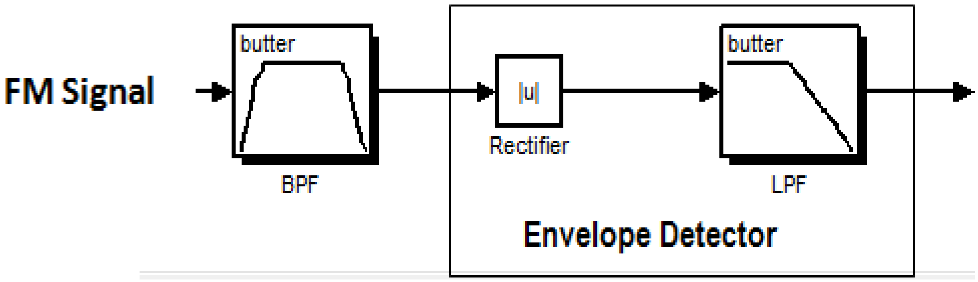


Figure.5: Frequency Discriminator

Modules Needed:

The following plug in modules are needed to complete the experiment:

Audio Oscillator, VCO, Multiplier, 100kHz channel filter, Utilities, Tunable LPF

Part I: Setting the Frequency Deviation

The frequency deviation is equal to the product of $V_{in(Max)}$ and Gain. Our objective is to design the Gain that yields frequency deviation of ±2 kHz.

1. Set a DC voltage of 2 V as input to VCO.
2. Set the Gain control fully anti-clockwise and the output frequency to 10 kHz.
3. Advance the Gain control until the frequency changes by 2 kHz.
4. Change to Variable DC to +2V and confirm that the deviation is about 2 kHz in the other direction. Record the measured frequency.

Part II: Time Domain and Frequency Domain Analysis

1. Fix the message frequency from the Audio Oscillator to 1 kHz.
2. Plot the message signal, carrier signal and the modulated signal in lab sheet.
3. Plot the spectrum of the modulated signal using Pico Scope, in lab sheet
4. Vary the message frequency and describe the impact on the spectrum of the FM signal.
5. Plot the spectrum of the FM signal at the minimum and maximum frequencies of the Audio oscillator.
6. Reset the frequency of the message to 1 kHz, and vary the deviation ratio (by varying the Gain in the VCO). Describe the effect on the spectrum of the FM signal (make sure you do not overload the VCO).
7. Plot the spectrum at the minimum value and maximum Gain setting (before

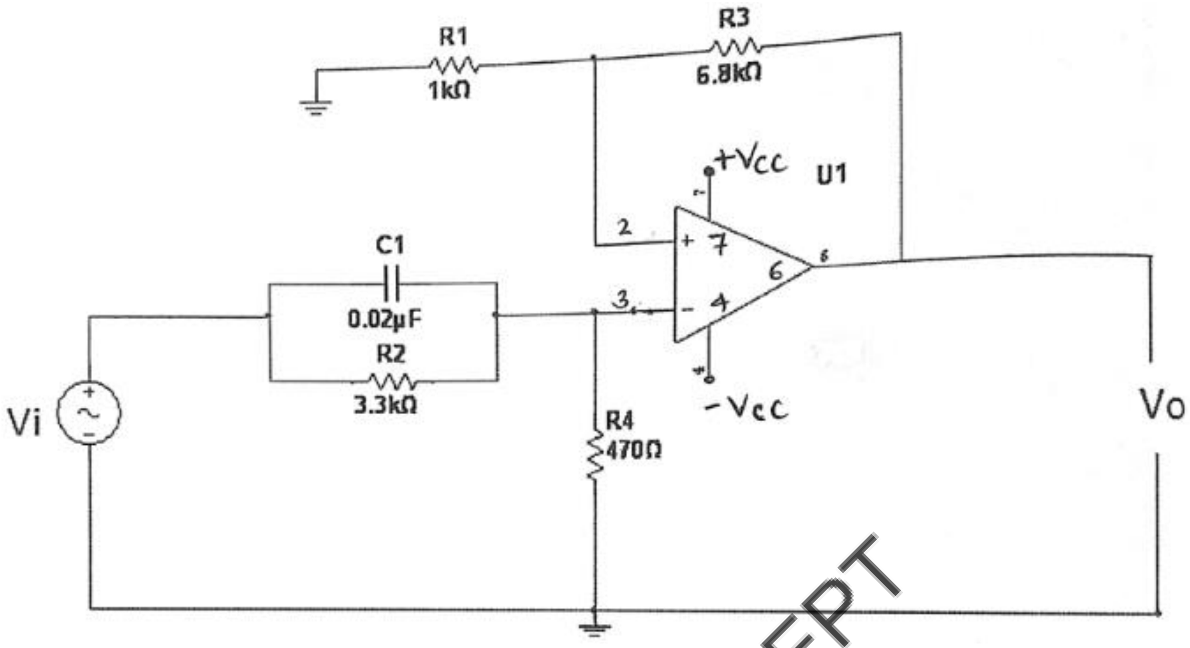
overload).

8. Explain the obtained spectra in light of Carson's Rule for bandwidth estimation.

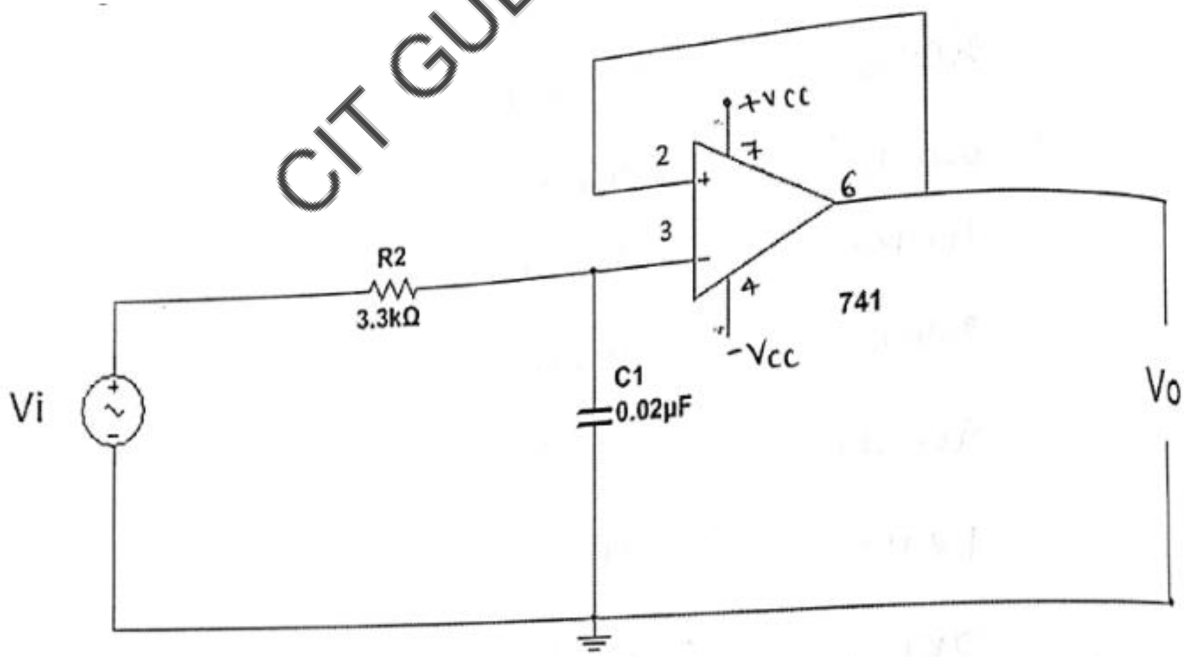
Part III: FM Demodulation Using PLL

1. Reconstruct the FM modulator as in the previous part.
   - Let the message frequency be 1 kHz from the Audio Oscillator,
   - Let the carrier frequency be 85 kHz from VCO,
   - Let the modulator VCO gain be around 20% of the maximum value.
2. Model the PLL demodulator illustrated in Figure 3.
   - For the filter use RC LPF provided in the Utilities Module.
   - In the Multiplier module set the toggle switch to AC.
   - Set the VCO in the demodulator to 85 kHz.
   - Set the Gain control to 20% of its maximum
3. Connect the output of the modulator to the input of the demodulator.
4. The PLL may or may not lock on to the incoming FM signal. Tune the Gain (and if necessary the center frequency) of the PLL-VCO until you obtain lock.
5. Examine the output of the PLL VCO and compare it with the original message.
6. Plot the message signal and recovered signal in lab sheet.

Pre-Emphasis



De-Emphasis:



EXPERIMENT No.: 04 **Date:**

Pre-Emphasis & De-Emphasis

Aim: Design and plot the frequency response of Pre-Emphasis and De-Emphasis Circuits.

Components required:

Sl.No	Apparatus	Range/Specifications
1	Op amp	µA 741
2	Audio Signal Generator	100Hz – 1 MHz
3	Resistors	3.3KΩ (02 No.s) 470Ω (02 No.s)
4	Capacitors	0.022µf
5	CRO	Digital

Theory: The Power Spectral Density of message signal falls off at higher frequencies as shown in graph. On the other hand the Power spectral Density of noise increases rapidly with frequency as shown in figure. Consequently, the signal-to-noise ration reduces drastically. To improve the signal-to-noise ration artificially, it is required to emphasize or boost up high frequency components of the message signal prior to modulation. This boosting of higher modulating frequencies in accordance with a pre-arranged curve is known as **"Pre-Emphasis"**. The reverse process is done at the receiver after discrimination (Demodulation). This reverse process is called as "De-Emphasis". Pre-Emphasis is achieved with a high pass circuit followed by a voltage follower using Op-amp, with RC time constant equal to the standard value of 75µ sec for FM. A low pass circuit followed by another voltage follower using Op-amp is used as De-Emphasis circuit.

Procedure:

Pre-Emphasis

1. Connections are made as shown in the circuit diagram
2. Apply a Sine wave of $0.5V_{p-p}$ amplitude, vary the frequency and note down the gain of the circuit
3. Plot a graph of Normalized gain V/s frequency

De-Emphasis

1. Connections are made as shown in the circuit diagram
2. Apply a Sine wave of $0.5V_{p-p}$ amplitude, vary the frequency and note down the gain of the circuit
3. Plot a graph of Normalized gain V/s frequency

Design:

Pre-Emphasis Circuit

RC = 75μsec for standard FM broadcasting.

Given f_1=2.1KHz, f_2=15KHz

$f_0 = \{(1)/(2\Pi RC)\} = 2.12KHz$

Let C = 0.022μf

Since, RC = 75 μsec

R = (75 μsec / 0.022 μf) = 3.4KΩ , Choose R = 3.3KΩ

Message Band width

$w= \{(1)/(2\Pi rc)\} = 15KHz$

$r = \{(1)/(2\Pi wc)\} = 482\Omega$

Choose r = 470Ω

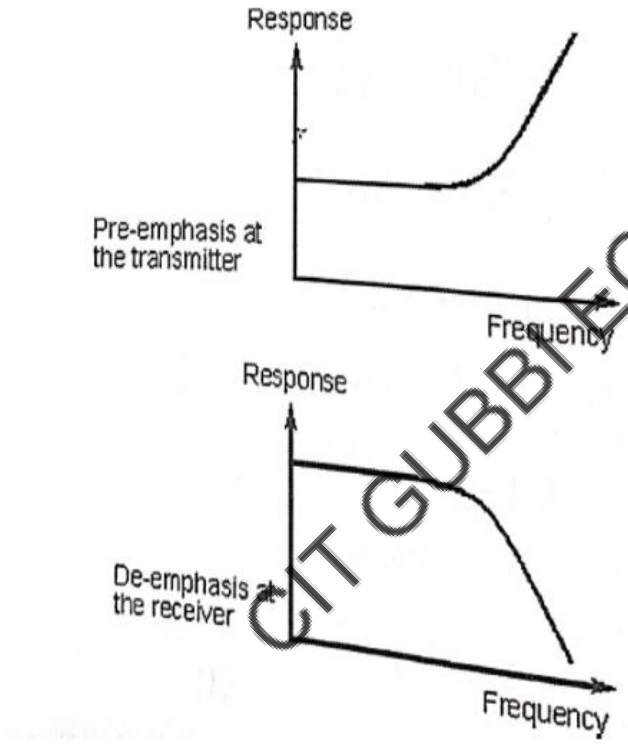
De-Emphasis Circuit

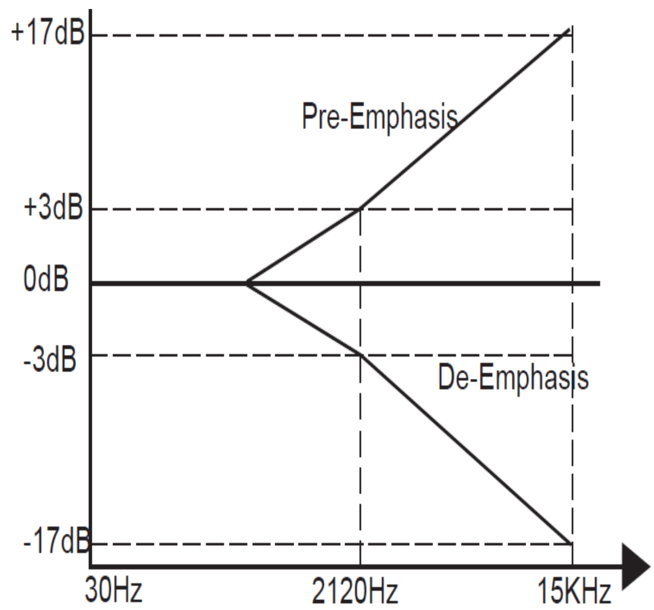
RC = 75 μsec

$f_c = \{(1)/(2\Pi RC)\} = 2.12$ kHz

R = 3.3KΩ and C = 0.022 μf

Graph :





75-µs Emphasis Curves

Tabular Column:

Pre-Emphasis : Input Voltage = 0.5 V_{p-p}

Sl.No	Frequency in Hz	Output Voltage V_{out}	Gain in dB = $20 \log_{10} (V_{out}/V_{in})$

De-Emphasis : Input Voltage = 0.5 V_{p-p}

Sl.No	Frequency in Hz	Output Voltage V_{out}	Gain in dB = $20 \log_{10} (V_{out}/V_{in})$

Result: Pre-Emphasis & De-Emphasis characteristics are verified by plotting response curves

EXPERIMENT No.: 05 **Date :**

BJT/FET Mixer

Aim: To design and test mixer circuit using transistor.

Tabular Column:

Sl. No.	Item & Specification	Quantity
1	Resistor- 22K, 33K, 3.3K	1No. each
2	Transistor BF194	1No.
3	Capacitor- 0.1 F	1No.
4	IFT	1No.
5	Signal Generator	2Nos.
6	CRO	1No.
7	Power Supply- =12V	1No.
8	Multimeter	1No.
9	Bread Board	1No.
10	Wires and probes	-

THEORY:

Mixer or frequency convertor is actually a non linear resistor having two sets of input terminals and one set of output terminal. The two inputs to the mixer are the input signal and the local oscillator signal. The output of the mixer contains many frequencies including the sum and difference frequencies between the two input signals. The mixer output is commonly tuned to the difference frequency. This frequency is called the intermediate frequency (IF).

The input to the mixer is the input signal voltage with magnitude V_S and frequency f_S. The output is usually a current component at IF frequency having a magnitude I_{IF} proportional to V_S. The proportionality constant is called transconductance and is given by

$g_C = I_{IF}/ V_S$

The conversion transconductance of a transistor mixer is of the order of 6ms.

Mixing takes placewhen the transconductance of the mixer is caused to vary with thelocal oscillator voltage.

PROCEDURE:

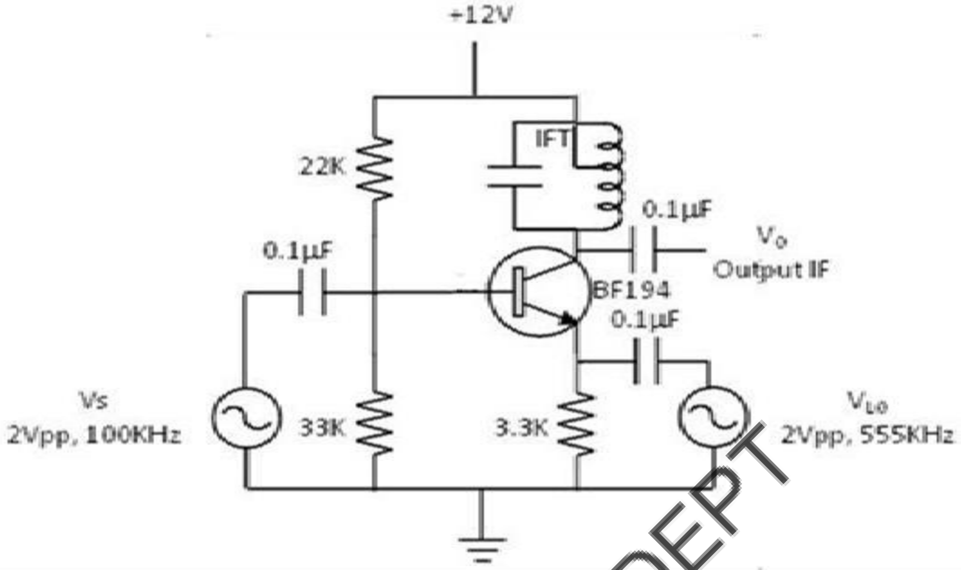
1. Test all the components and probes.
2. Set up the circuit as shown in figure on a breadboard.
3. Switch on the power supply.
4. Check the dc conditions of the transistor and make sure that it is working in the activeregion.
5. Feed a 2Vpp, 100KHz sine wave signal at the base of the transistor as shown in figure.
6. Feed a 2Vpp, 555KHz sine wave signal at the emitter of the transistor as shown in figure.
7. Observe the output waveform on a CRO and measure the frequency. Adjust the IFT toobtain 455KHz as the peak output frequency.
8. Plot the input/output waveforms.
9. Measure the output ac current (I_{IF}) and the input ac voltage(V_S) using a multimeter. Calculate the transconductance using the equation $g_C = I_{IF}/ V_S$
10. Check the output for 100KHz and 355KHz inputs.

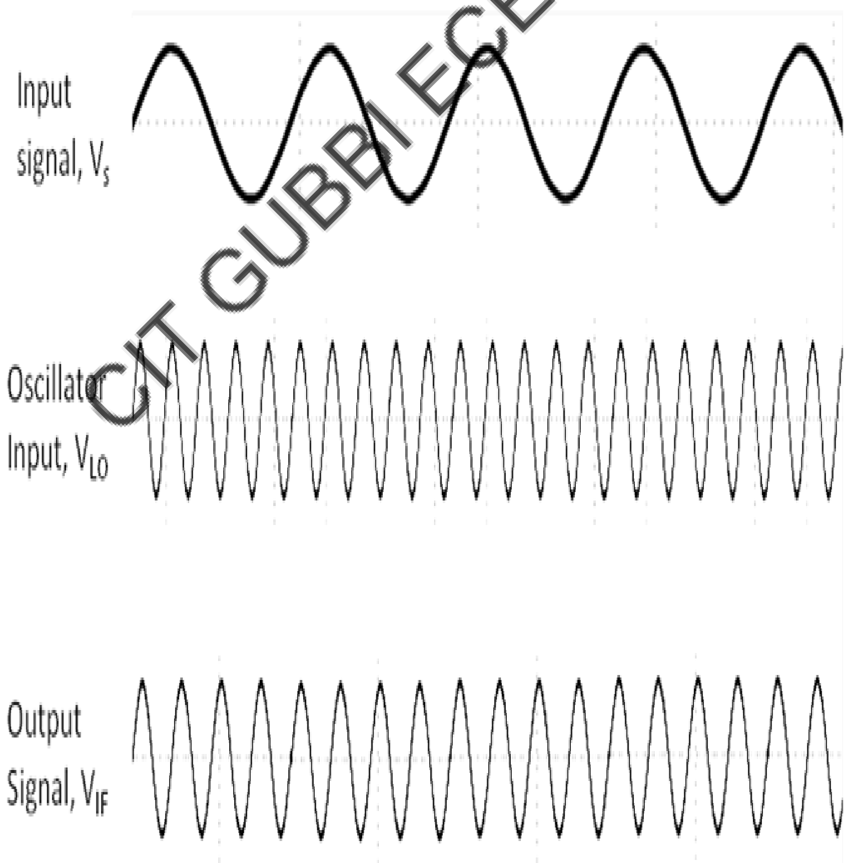
Note: The IFT centre tap point should be connected to Vcc. Connect one of the othertwo terminals of the IFT primary to the collector of the transistor. Try both terminalsand select the one that gives the better output.

Result: The transistor mixer circuit is designed for the difference frequency.

Circuit Diagram

Waveforms:





EXPERIMENT No.: 06 **Date:**

PULSE SAMPLING, FLAT TOP SAMPLING AND RECONSTRUCTION

Aim: To conduct an experiment to generate pulse sampling and to demodulate the same.

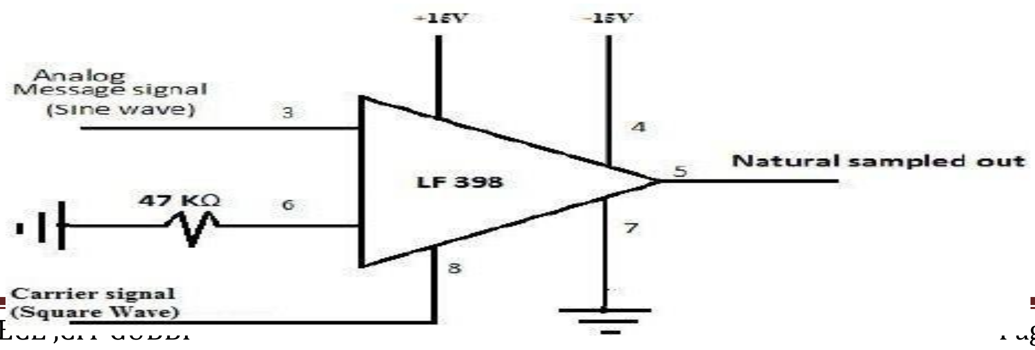
Apparatus Required:

Sl. No.	Apparatus	Range	Quantity
1	Sample & hold IC – LF398	-	1
2	Resistors & Capacitor	As Per the design	-
3	Springboard + connecting wires	-	1 Set

Procedure:

1. Check the components/Equipments for their working condition.
2. Connections are made as shown in the circuit diagram
3. Apply the square wave carrier signal of 15-20KHz of 20% duty cycle.
4. Apply sine wave modulating signal of frequency fm = 1kHz with 5V peak to peak amplitude.
5. Turn on the offset and vary the offset voltage until desired waveform is observed on CRO.
6. Observe the output waveform.
7. Connect the sampled output as a input to the low pass filter and reconstruct the original message signal and note fo and Vo.
8. Repeat the above steps for fc = 2fm and fc<2fm.

Circuit Diagram: (Natural sampling)



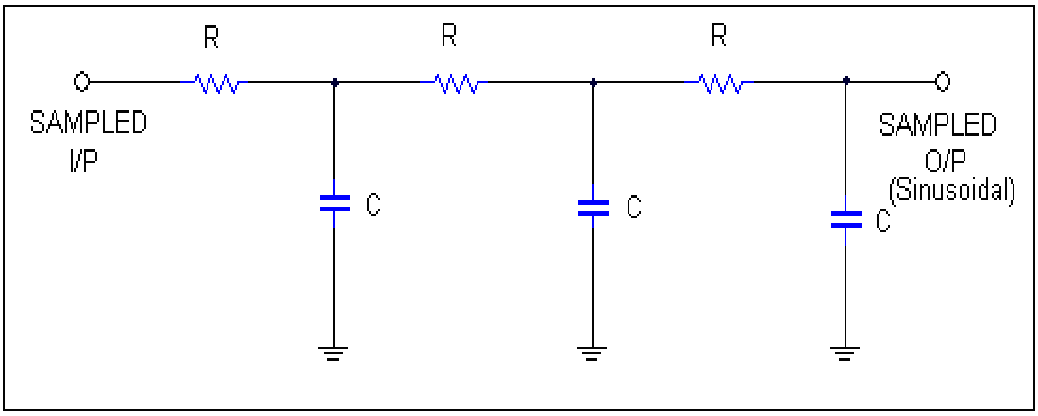


Fig 5.1: Natural Sampling using LF398

Filter design:

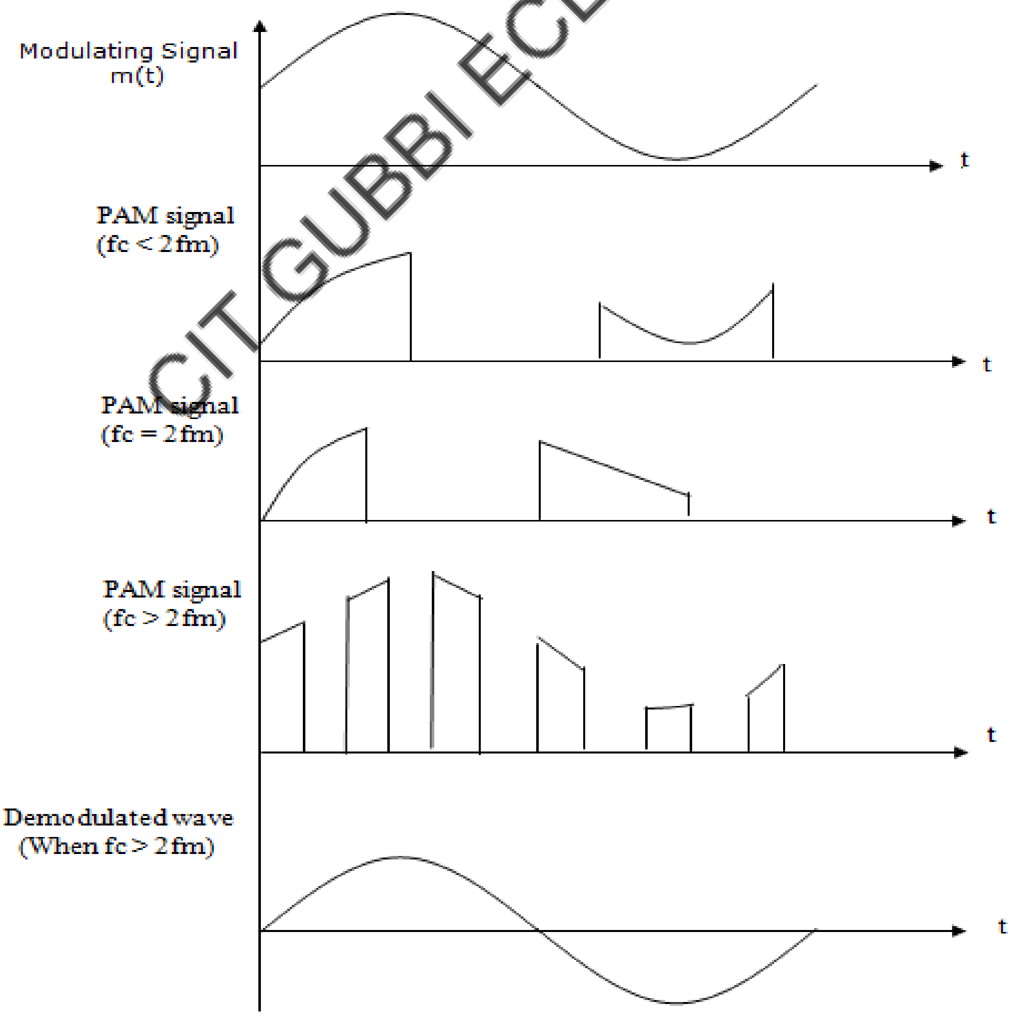
fs= 1/ Ts ; Ts=RC ; R=Ts/C

Cut off frequency of the filter

fo >> fm

Choose fo = 2kHz, fo = 1 / 2∏RC Assume C = 0.1 μf, then R = 500 Ω

Waveforms:



Sl. No.	Sampling methods	fc in Hz	fm in Hz	Vo of demodulated signal in Volt	fo of demodulated signal in Hz
1	Under Sampling (fc<2fm)				
2	Nyquist Rate fc=2fm				
3	Over Sampling fc>2fm				

Tabular Column: Vc (p-p)=_______V, Vm(p-p)=____V

FLAT TOP SAMPLING AND RECONSTRUCTION

Aim: To verify the Flat Top sampling theorem.

Apparatus Required:

Sl. No.	Apparatus	Range	Quantity
1	Function Generator	1MHz	1
2	Op-amps Transistor	μA741 SL 100	2 1
3	Resistors and capacitors	As per design	
4	Dual trace oscilloscope	20 MHz	1

Theory:

The analog signal can be converted to a discrete time signal by a process called sampling. The sampling theorem for a band limited signal of finite energy can be stated as: "**A band limited signal of finite energy, which has no frequency component higher than W Hz is completely described by specifying the values of the signal at instants of time separated by 1/2W seconds**"

It can be recovered from knowledge of samples taken at the rate of 2W per second.

Circuit diagram:

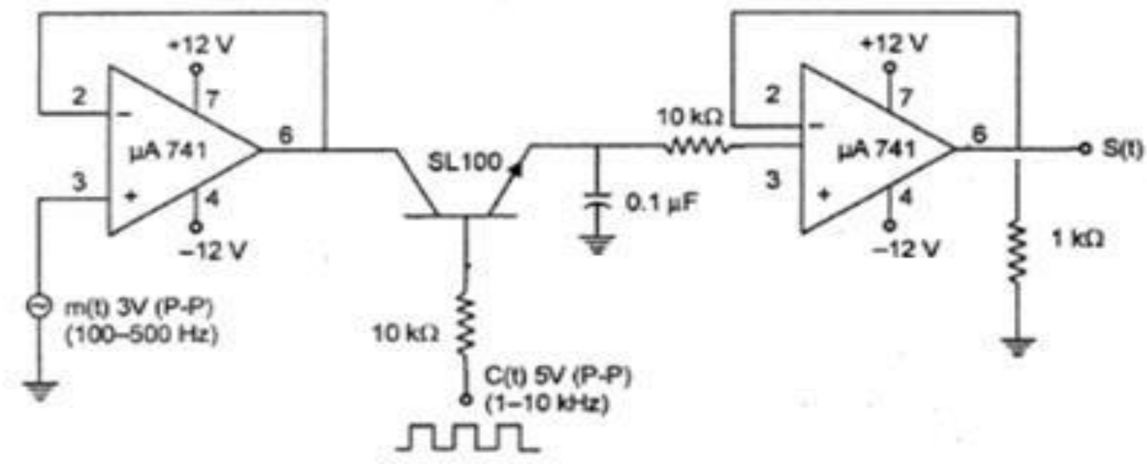


Fig 5.3: Sampling Circuit using transistor

Procedure:

1. The circuit is connected as per the circuit diagram shown in the fig 5.1 or 5.3
2. Switch on the power supply and set at +12V and -12V.
3. Apply the sinusoidal signal of approximately 3V (p-p) at 100-500 Hz frequency and pulse signal of 5V (p-p) with frequency between 100Hz and 10 KHz.
4. Connect the sampling circuit output and AF signal to the two inputs of oscilloscope
5. Initially set the sampling frequency to 200Hz and observe the output on the CRO. Now vary the amplitude of modulating signal and observe the output of sampling circuit. Note that the amplitude of the sampling pulses will be varying in accordance with the amplitude of the modulating signal.
6. Design the reconstructing circuit. Depending on sampling frequency, R & C values are calculated using the relations Fs = 1/Ts, Ts = RC. Choosing an appropriate value for C, R can be found using the relation R=Ts/C
7. Connect the sampling circuit output to the reconstructing circuit shown in Fig 5.4
8. Observe the output of the reconstructing circuit (AF signal) for different sampling frequencies. The original AF signal would appear only when the sampling frequency is 200Hz to 500Kz.

Design

(1) Flat top sampling

$$RC << T_m$$

where $T_m = 3.3$ ms

Assume $f_m = 300$ Hz

Let $RC = 1$ ms

Let $R = 10$ kΩ

$\therefore$ $C = 0.1$ μF

(2) Demodulation

$$f = \frac{1}{2\pi RC} = 500 \text{ Hz}$$

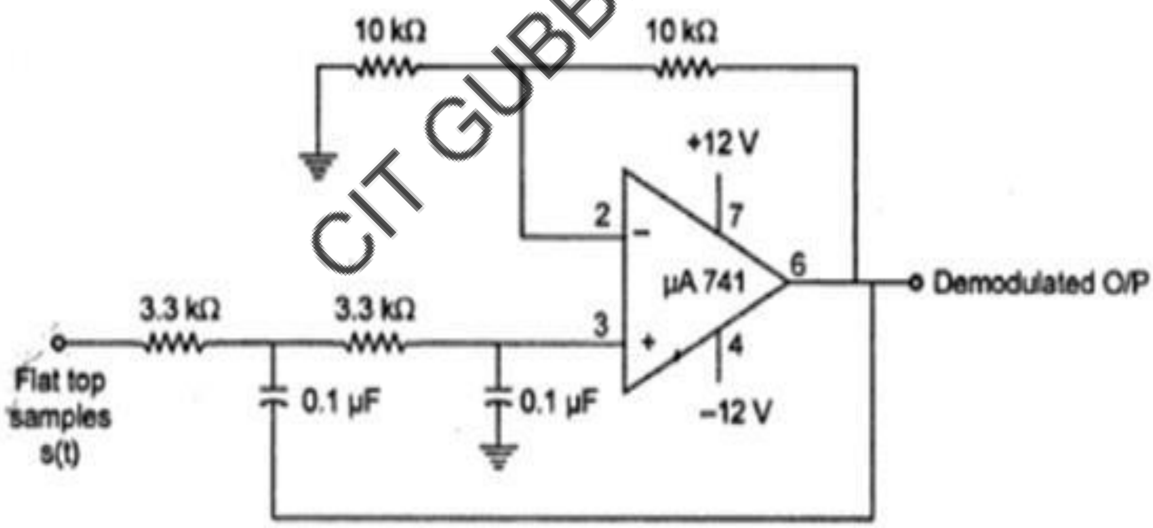
Let $C_1 = 0.1$ μF,

then $R_1 = 3.1$ kΩ

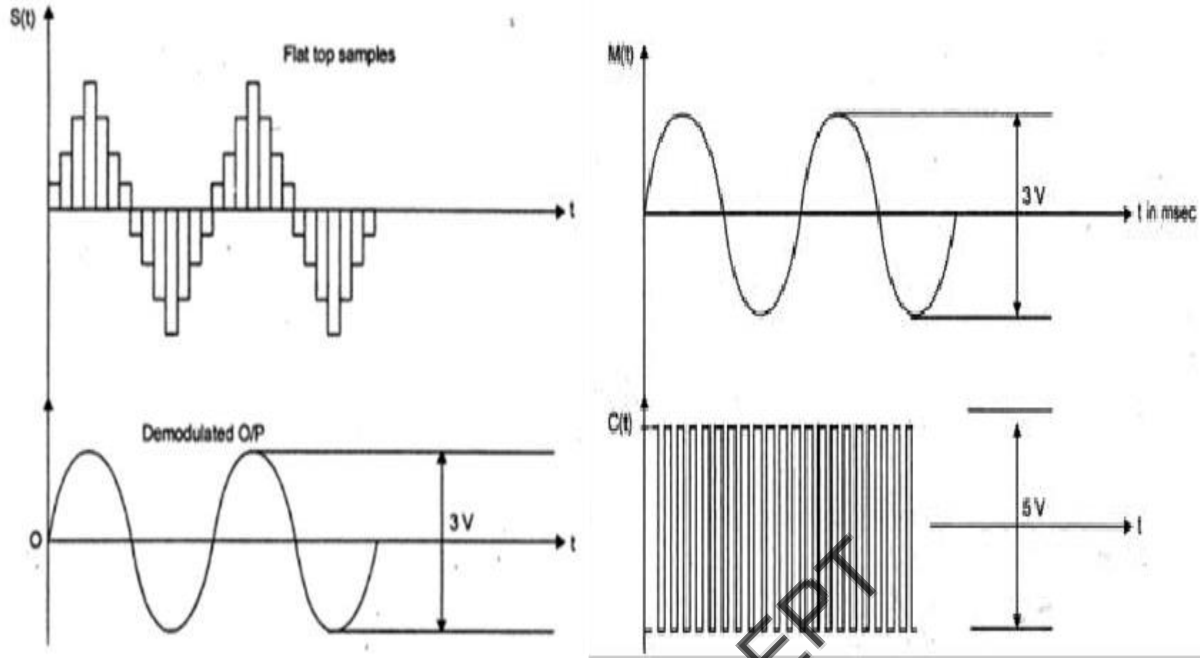
$\simeq 3.3$ kΩ

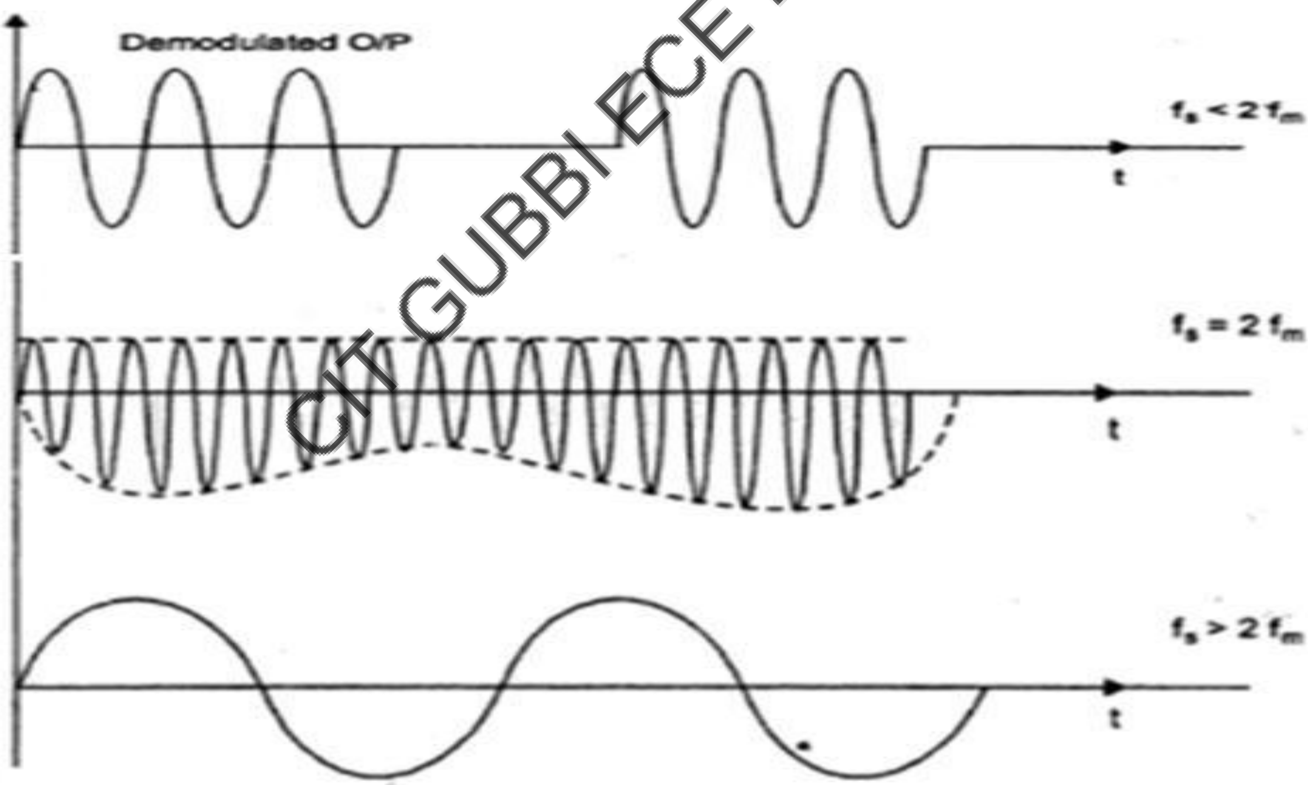
Circuit diagram:

Fig 5.4: Reconstruction circuit using Op-amp



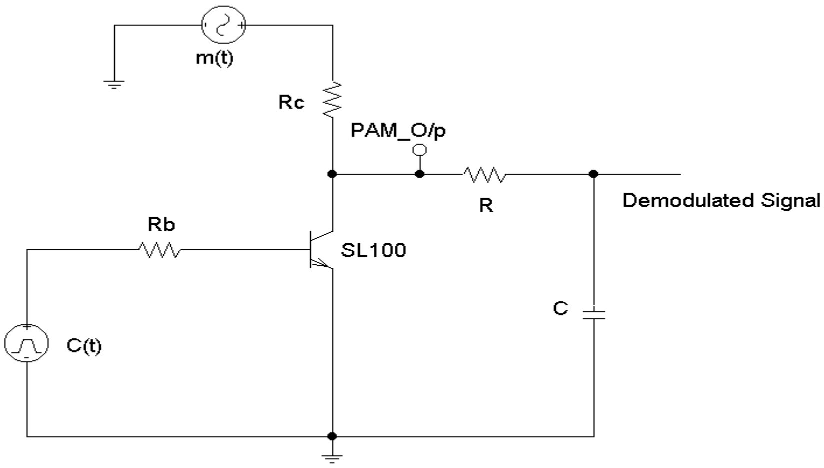
Waveforms:





Result: Sampling Theorem is verified for Pulse sampling & Flat top sampling circuits.

Circuit Diagram :



Design:

Specifications : Ic = 1mA, hfe = 100, $V_{CE(}sat)$ = 0.3 V, $V_{BE(}sat)$ = 0.7 V, fm = 100 Hz

Biasing:

$V_m(t) = Ic\ Rc + V_{CE(}sat)$

Let $V_m(t)$ = 2.5 V peak + 3 V DC shift = 5.5 V peak signal

Then Rc = 5.2 kΩ

$V_c(t) = I_B\ R_B + V_{BE(}sat)$

Let Vc(t) = 2 V peak – peak (1V peak) & I_B = IC / hfe = 10 μA

1 = R_B 10μA + 0.7

∴ R_B = 30KΩ

Filter:

Cut off frequency of the filter fo >> fm

Choose fo = 500 Hz, fo = 1 / 2πRC

Assume C = 0.1 μf, then R = 3.3 kΩ

Tabular Column :

Sl. No.	V_c (p-p) in Volt	fc in Hz	V_m (p-p) in Volt	fm in Hz	Vo of demodulated signal in Volt	fo of demodulated signal in Hz

EXPERIMENT No;7 **Date:**

PULSE AMPLITUDE MODULATION

Aim:

To conduct an experiment to generate PAM signal and to demodulate it.

Apparatus:

Sl.No.	Particulars	Range	Quantity
1.	Transistor	SL100	01
2.	Resistors & Capacitors	As per design	-
3.	Springboard + connecting wires	-	01 set

Theory:

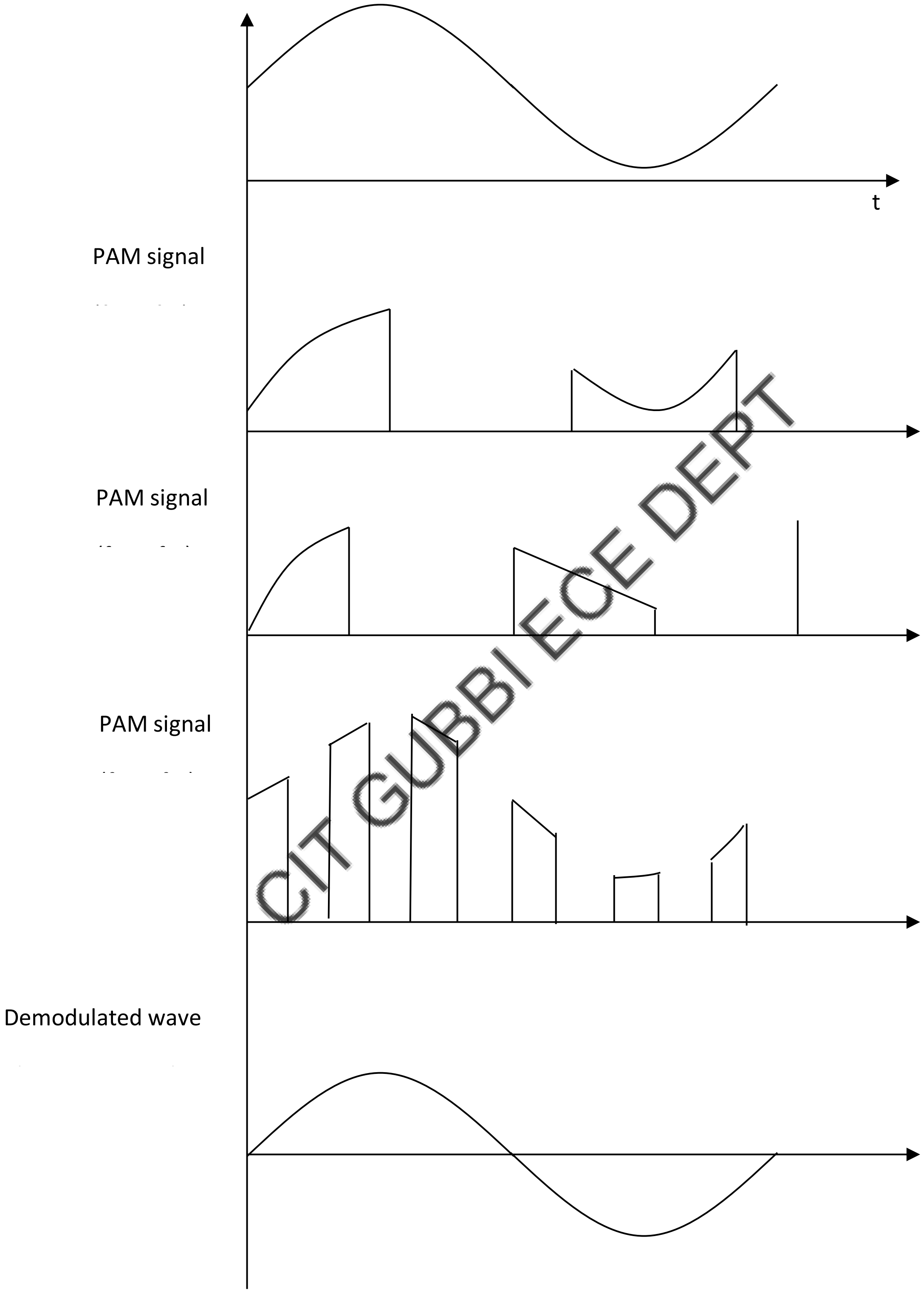
Pulse-amplitude modulation, acronym **PAM**, is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses. Demodulation is performed by detecting the amplitude level of the carrier at every symbol period. The samples are taken at regular interval of time. Each sample is a pulse whose amplitude of the variable at the instant of time at which the sample is taken. It is a simple process.

Pulse-amplitude modulation is now rarely used, having been largely superseded by pulse-code modulation, and more recently, by pulse-position modulation. The widely popular Ethernet communication standard is a good example of PAM usage.

Tabular Column:

Vc (p-p)=____V, Vm(p-p)=___V

Sl. No.	Sampling methods	fc in Hz	fm in Hz	Vo of demodulated signal in Volt	fo of demodulated signal in Hz
1	Under Sampling (fc<2fm)				
2	Nyquist Rate fc=2fm				
3	Over Sampling fc>2fm				


t
PAM signal
PAM signal
PAM signal
Demodulated wave

Experiment No.: 08 **Date:**

GENERATION AND DETECTION OF PULSE POSITION MODULATION

Aim: To conduct an experiment to generate PPM signal and to measure critical amplitude.

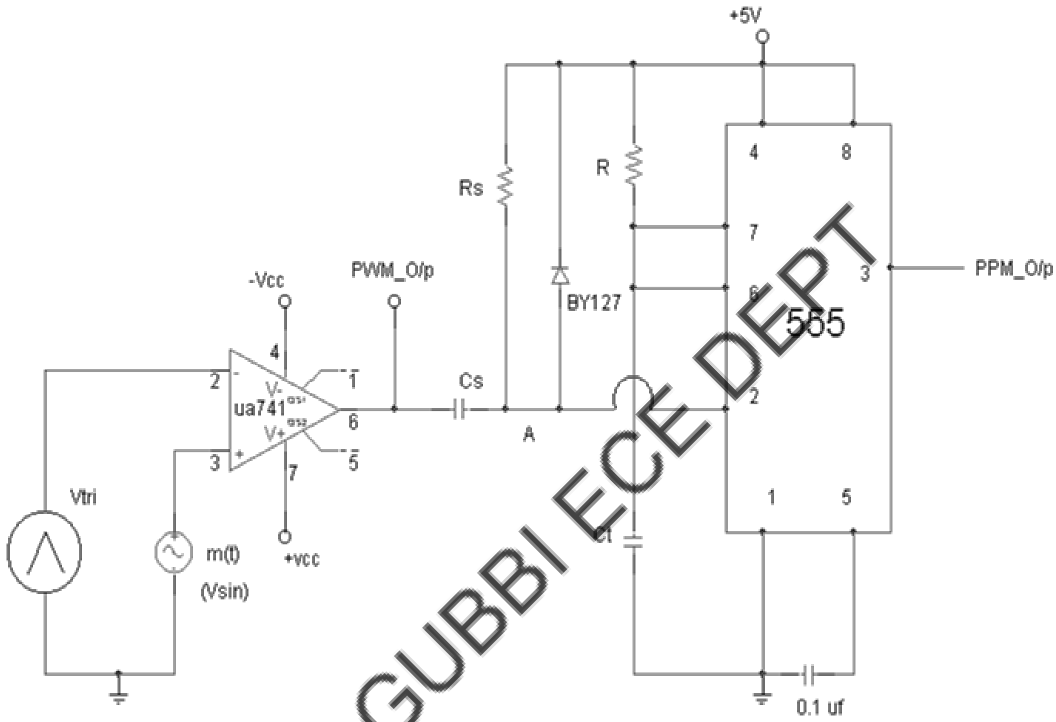
Apparatus:Sl.No.	Particulars	Range	Quantity
1.	ICs	555, uA741	01 each
2.	Resistors & Capacitors	As per design	-
3.	Diode BY127	-	01
4.	Spring board + connecting wires	-	01 set

Theory:Pulse Position Modulation: In this type the amplitude and width of the pulse are kept constant, with reference to the position of a reference pulse, It is changed according to the instantaneous sampled value of the modulating signal. Hence, transmitter has to send synchronizing pulses to keep the transmitter and receiver in Synchronism. It has advantage over that of PWM i.e it handles constant power output. But it require synchronization at transmitter as well as receiver which is the disadvantage. **Pulse-position modulation** is a form of signal modulation in which M message bits are encoded by transmitting a single pulse in one of 2^M possible time-shifts. This is repeated every T seconds, such that the transmitted bit rate is M/T bits per second. It is primarily useful for optical communications systems, where there tends to be little or no multipath interference.

Procedure:

1. Check the components/Equipments for their working condition.
2. Connections are made as shown in the circuit diagram.

3. Check the output of Monostable Multivibrator (unmodulated carrier) for the designed pulse width by giving unmodulated PWM signal (m(t) amplitude set to zero).
4. By increasing the amplitude of the modulating signal observe the PWM waveform and then observe the PPM waveform.



To find the critical amplitude and the dynamic range:

As the amplitude of the modulating signal is increased the position of the pulses will vary. Keep on increasing the amplitude of the modulating signal until the pulses will disappear. The corresponding amplitude of the modulating signal is the critical amplitude. The dynamic range is the difference between the critical amplitude and the amplitude of the modulating signal at which the PPM just begin.

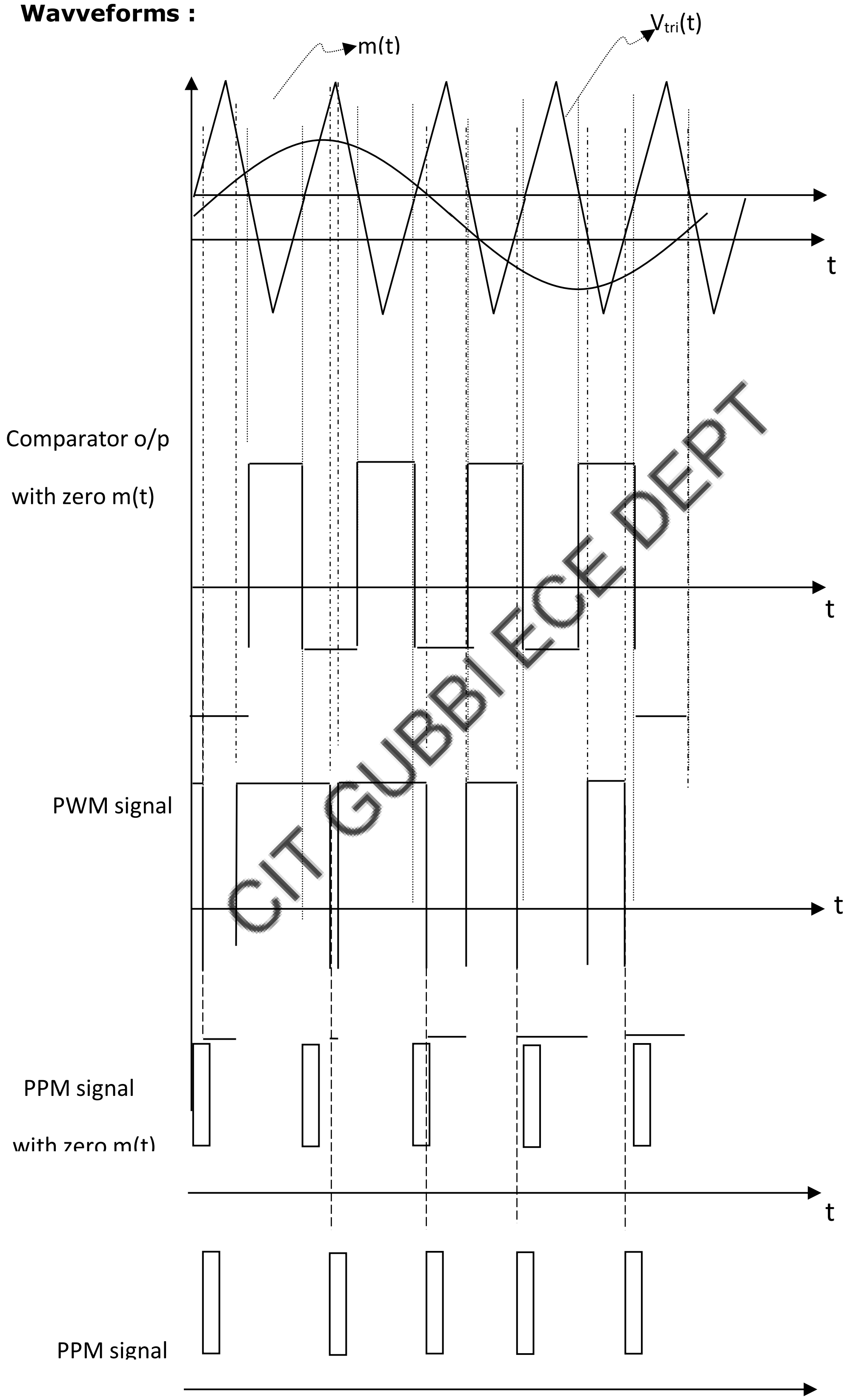
Result:

Pulse width = ________ ms

Dynamic range = ________ Volt

Critical Amplitude = ________ Volt

Wavveforms :


m(t)
$V_{tri}(t)$
t
Comparator o/p
with zero m(t)
t
PWM signal
t
PPM signal
with zero m(t)
t
PPM signal

EXPERIMENT No.: 09 t **Date:**

GENERATION AND DETECTION OF PULSE WIDTH MODULATION

Aim: To conduct an experiment to generate PWM signal and to measure critical amplitude.

Apparatus:

Sl.No.	Particulars	Range	Quantity
1.	IC	555 Timer	01
2.	Resistors & Capacitors	As per design	-
3.	Diode BY127	-	03
4.	Spring board + connecting wires	-	01 set

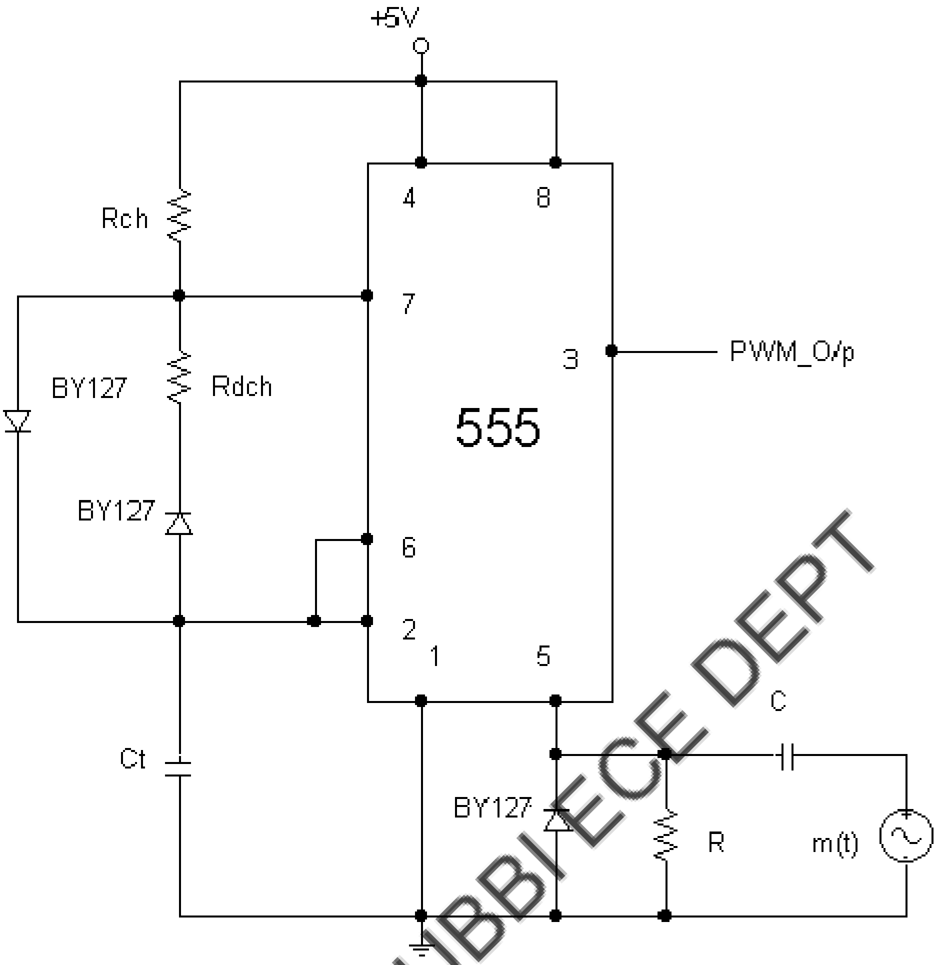
Theory:

Pulse width modulation is also known as pulse duration modulation. Three variation of pulse width modulation is possible.

1. The leading edge of the pulse is held constant and change in pulse width in accordance to the information signal.
2. The tail edge is held constant and with respect to it pulse width is measured.
3. The centre of the pulse is held constant and pulse width changes on either side of the centre of the pulse. **Pulse-width modulation** of a signal or power source involves the modulation of its duty cycle, to either convey information over a communications channel or control the amount of power sent to a load.

Procedure:

1. Check the components/Equipments for their working condition.
2. Connections are made as shown in the circuit diagram.
3. Keeping the modulating signal with minimum amplitude, observe the astable multivibrator output and verify the frequency and duty cycle.
4. Apply the modulating signal with frequency f_m at some convenient amplitude.
5. Observe the PWM waveform.
6. Observe the variation of pulse width with respect to clamped modulating signal (at Pin No.5).



Circuit Diagram:

Design :

Specifications : fc = 1 kHz, duty cycle = 50 %

Hence T = 1/fc = 1ms, Ton = 0.5 ms and Toff = 0.5 ms

Astable Multivibrator :

Ton = 0.693 (Rch + Rf) Ct, Toff = 0.693 (Rdch + rf) Ct

Since duty cycle = 50 %, Ton = Toff = 0.5 ms

Assume Ct = 0.1 μf and the forward resistance of diode Rf = 100 Ω

Then Rch = Rdch = 7.146 KΩ

Clamping Circuit :

Negative peak of the modulating signal should be clamped to zero volts.

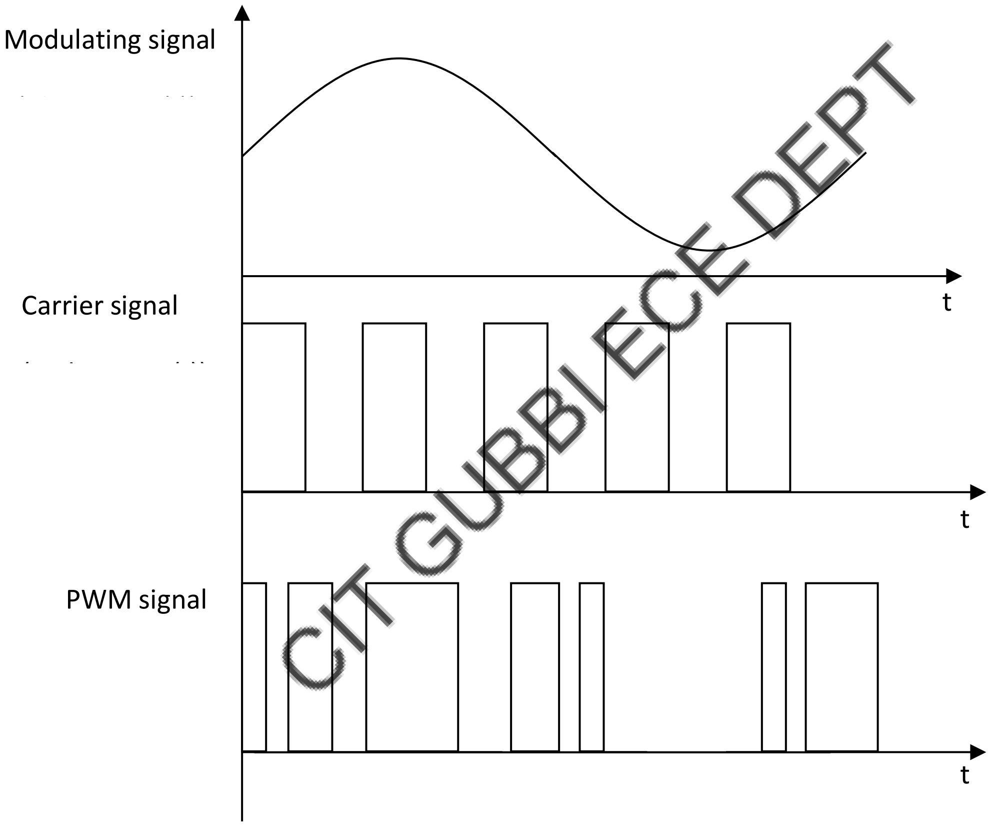
RC >> 1 / fm, therefore RC = 100/fm

Choose C = 10 μf, fm = 100Hz then R = 100 kΩ

Tabular Column: fm = ____________Hz

Unmodulated carrier signal			Modulated Signal		Dynamic range in volt
Ton in ms	**Toff in ms**	**fc in Hz**	**Max. width in ms**	**Min. width in ms**	

Waveforms :



Reference waveform

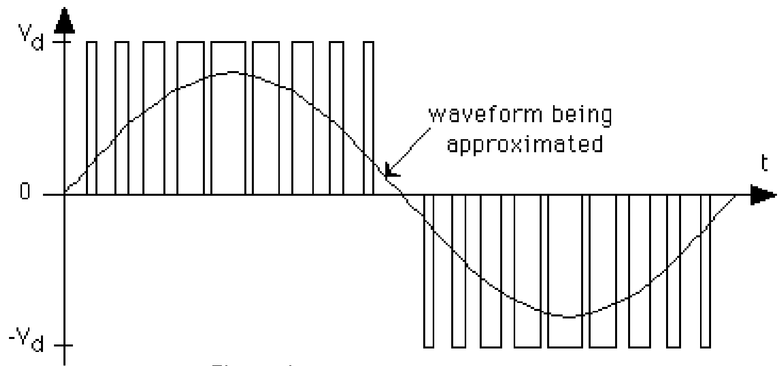


Figure 1

To find the critical amplitude and the dynamic range:

As the amplitude of the modulating signal is increased the width of the pulses during the negative half cycle of the modulating signal will reduce and during positive half cycle will increase. Keep on increasing the amplitude of the modulating signal until the pulses will disappear for the first time either because the width of the pulse may become zero during negative half cycle or the width of the pulse may become so large that it combines with the neighboring pulse during positive half cycle. The corresponding amplitude of the modulating signal is the critical amplitude.

The dynamic range is the difference between the critical amplitude and the amplitude of the modulating signal at which the PWM just begin.

Result:

Pulse width = __________ms

Dynamic range = _________ Volt

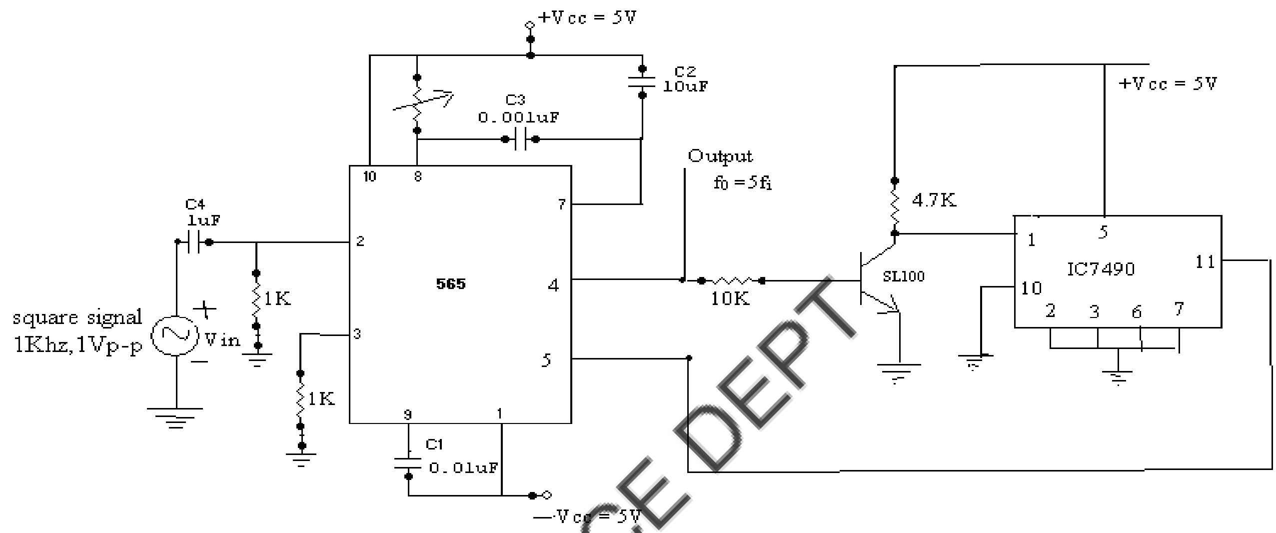
Critical Amplitude = _______Volt

EXPERIMENT No.: 10 **Date:**

FREQUENCY SYNTHESIS USING PLL

Aim: To study the frequency synthesis of PLL.

Circuit Diagram:



Design:

The center frequency of PLL is determined by free running frequency of VCO which is given by

$f_{out} = 1.2 / 4R_1C_1$ Hz

$f_L = \pm 8\ f_{out}\ / V$

Where V=(Vcc-(-Vcc))= 10V

$f_C = \pm [f_L /(2\pi x 3.6x10^3 x\ C_2)]^{1/2}$

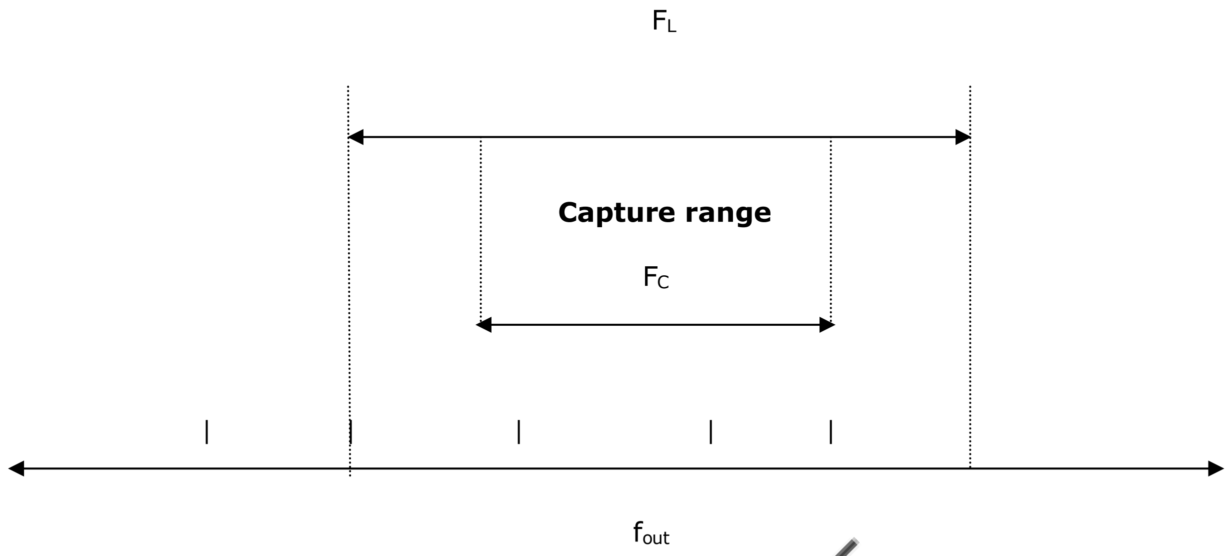
where 3.6K is the internal resistance of the capacitor C_2

$f_{out} = 1.2/ (4x\ 4.3x10^3\ x0.01x10^{-6}) = 70KHz$

$f_L = \pm (8x\ 70K) / 10 = \pm 56$ KHz

$f_C = \pm 49.75KHz$

Lock range



Frequency in Hertz

Procedure for Frequency synthesis:

1. Make the connections as shown in the circuit diadram.
2. Insert Mod-5 counter between pin 4 and 5.
3. Using function generator at pin 2 apply square wave of frequency 1 KHz to get 1V p_p input signal
4. By adjusting the DRB, set the VCO frequency till PLL is locked. Measure and note down the output frequency, it should be 5 times the input frequency.

Result:

Thus using IC NE 565 PLL the capture range and the locking range were determined.

NOTE:

Procedure for finding capture and locking range of PLL

1. Make the Connections as shown in the circuit diagram.
2. Measure and note down the frequency of 565 at pin 4 using CRO without input signal
   at pin 2.
3. Set the input signal at pin 2 to get 1Vp_p. Increase the input signal frequency slowly at
   the point of center frequency f_{out} . At some frequency VCO output will suddenly shifts
   from f_{out}
4. Measure and note down the frequency that is the lower edge of the Capture range f_{C1}
5. Increase the input signal frequency further till the output signal VCO is in phase with
   Input.
6. Measure and note down the frequency that is the upper edge of the Lock range f_{L2}
7. Now start decreasing at the point of center frequency f_{out} . At some frequency VCO
   output shifts from f_{out}.
8. Measure and note down the frequency that is the upper edge of the Capture range f_{C2}
9. Increase the input signal frequency further, till the output signal VCO is in phase with
   Input.
10. Decrease the input signal frequency further till the output signal VCO is out phase
    with Input.
11. Measure and note down the frequency that is the lower edge of the Lock range f_{L1}
12. Calculate Locking range as f_{L2} - f_{L1} and capture range as f_{C2} - f_{C1}

Result: Frequency Synthesis is designed and verified using PLL

EXPERIMENT No.: 11 **Date:**

DATA FORMATTING & LINE CODE GENERATION

Aim: To study Pulse data coding techniques for NRZ formats & generation of Line Codes.

Apparatus:

Sl.No.	Particulars	Range	Quantity
1.	Data Encoding kit	(Digital Trainer Kit)	01
2.	Data bit generator	--	01
3.	Patch cords	--	05
4.	CRO	20 MHz	01
5.	CRO Probes	--	02

Block Diagram:

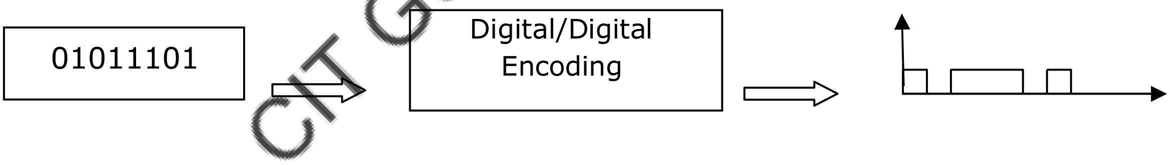
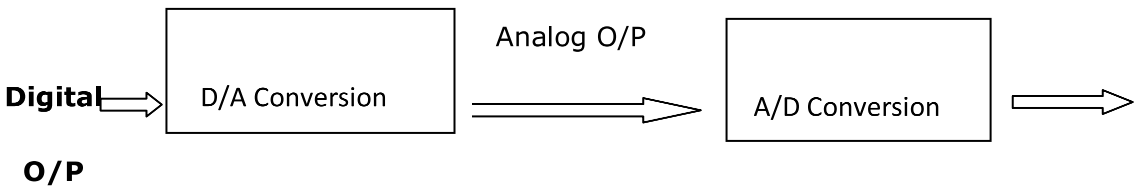


Fig 1: Digital-to-Digital Encoding



Theory:

Digital to Digital conversion is the representation of digital information by a digital signal. In this conversion, the binary 1's and 0's generated by a computer are translated into a sequence of voltage pulses that can be propagated over a wire. Figure 1 shows the relationship between the digital information, the digital-to-digital encoding hardware and the resultant digital signal. There are many mechanisms for digital-to-digital conversion, these are Uni-polar, polar and bipolar encoding/conversion. In our present experiment we are using polar conversion method.

Polar Encoding: It uses two-voltage levels, one positive and one negative, Of many existing variations of polar conversion we will examine only the three most popular: Non return to zero (NRZ), Return to zero (RZ), and Biphase. NRZ encoding includes two methods: Non return to zero, level (NRZ-L), and Non return to zero invert (NRZ-I). Biphase also refers to two methods. The first, Manchester, is the method used by Ethernet LANs. The second, Differential Manchester, is the method used by Token Ring LANs.

Non return to Zero(NRZ): In NRZ encoding, the level of the signal is always either positive or negative. The two most popular method of NRZ transmission are:

NRZ-L: In this encoding method, the level of the signal depends on the type of bit it represents. A positive voltage usually means the bit is a '0', and a negative voltage means the bit is a '1'(or vice-versa). Thus, the level of the signal is dependent upon the state of the bit.

It is the simplest form of data representation. The NRZ (L) waveform simply goes low for one bit time to represent a data '0' & high for one bit time to represent a data '1'. Thus the signal alternates only when there is a data change.

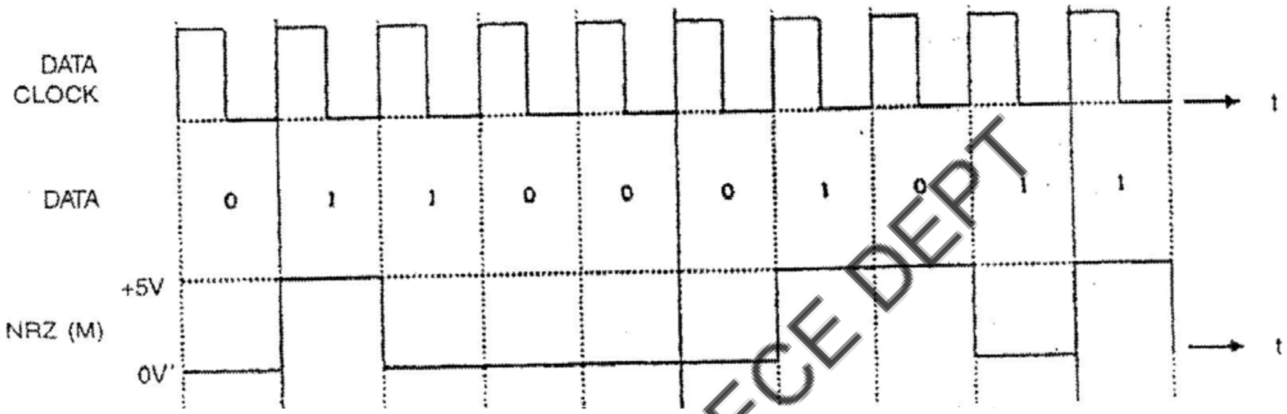
NRZ-I: In this method, an inversion of the voltage level represents a 1 bit. It is the transition between a positive and negative voltage, not the voltages themselves, that represents a 1 bit. A '0' bit is represented by no change.

Non - Return - To Zero (Mark) : [NRZ (M)] :

The NRZ (M) code is very much similar to the NRZ (L) code. Here if logic 1 is to be transmitted. The new level is inverse of the previous level i.e. change in level occurs. If a data '0' is to be transmitted the level remains unchanged. Thus in the case of NRZ (M) waveform the present level is related to the previous levels. See figure 2. Thus, no longer the absolute value of signal is necessary instead it is the change in the level for which we look now.

Remember: A change means a logic '1'

No change means logic '0'



Return to Zero: This method uses three values: positive, negative and zero. The signal changes not between bits but during each bit. A positive voltage means 1 and negative voltage means 0.

Biphase: In this the signal changes at the middle of the bit interval but does not return to zero. Instead it continues to the opposite pole.

Biphase (Manchester) Coding:

The encoding rules for biphase (Manchester) code are as follows: A data '0' is encoded as a low level during first half of the bit time and a high level during the second half. A data '1' is encoded as a high level during first half of the bit time and a low level during the second half. Thus string of 1's or 0's as well as any mixture of them will not pass any synchronization problem in receiver

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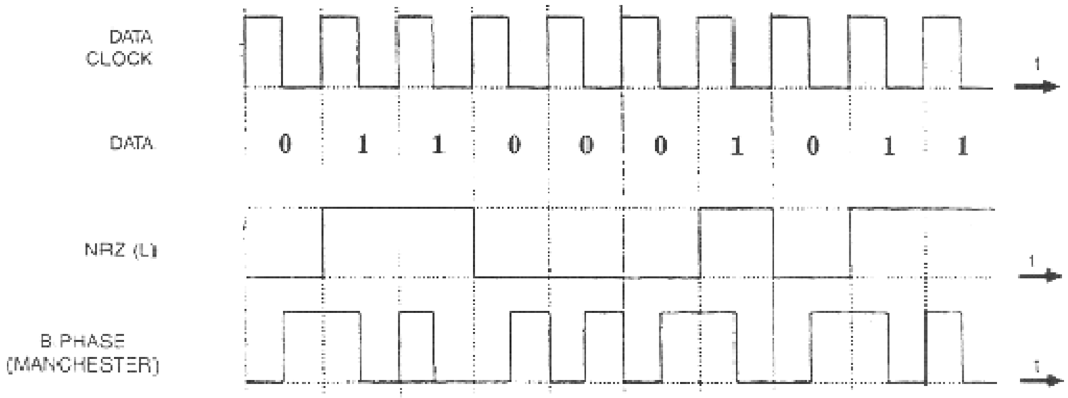


Fig. 3: Biphase (Manchester) waveform for a given data stream

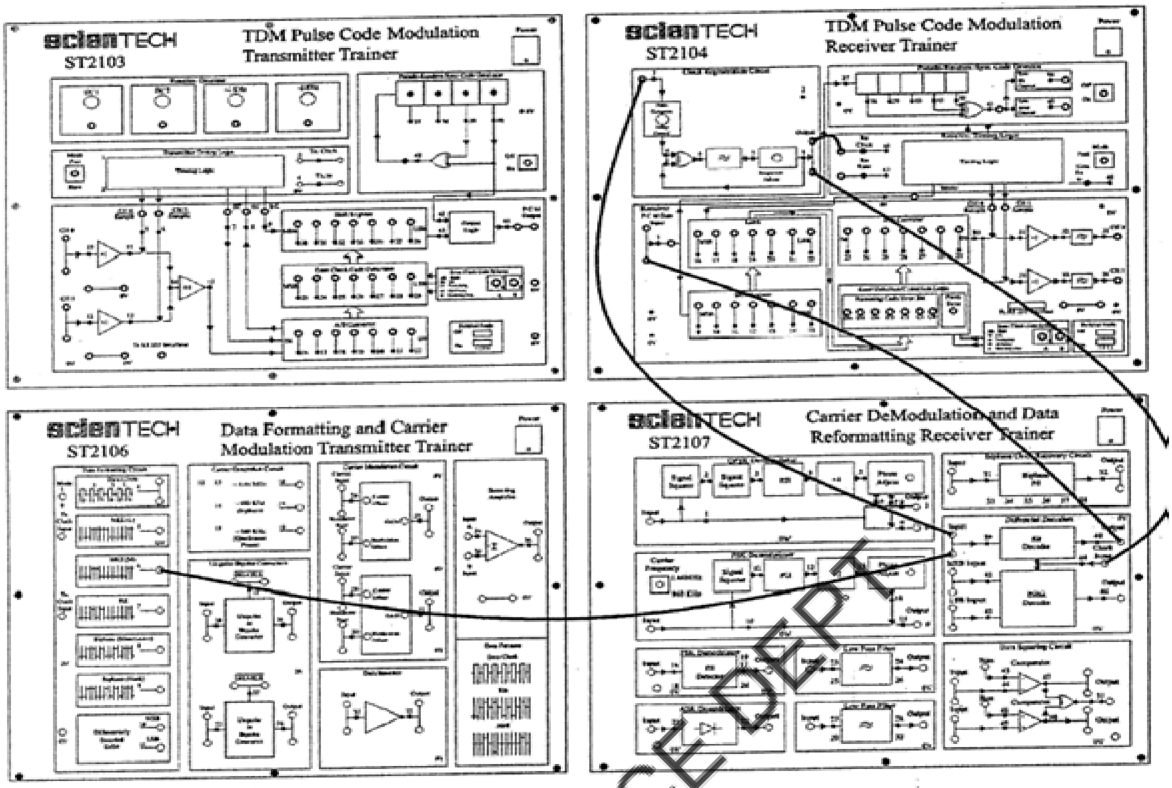
Procedure :

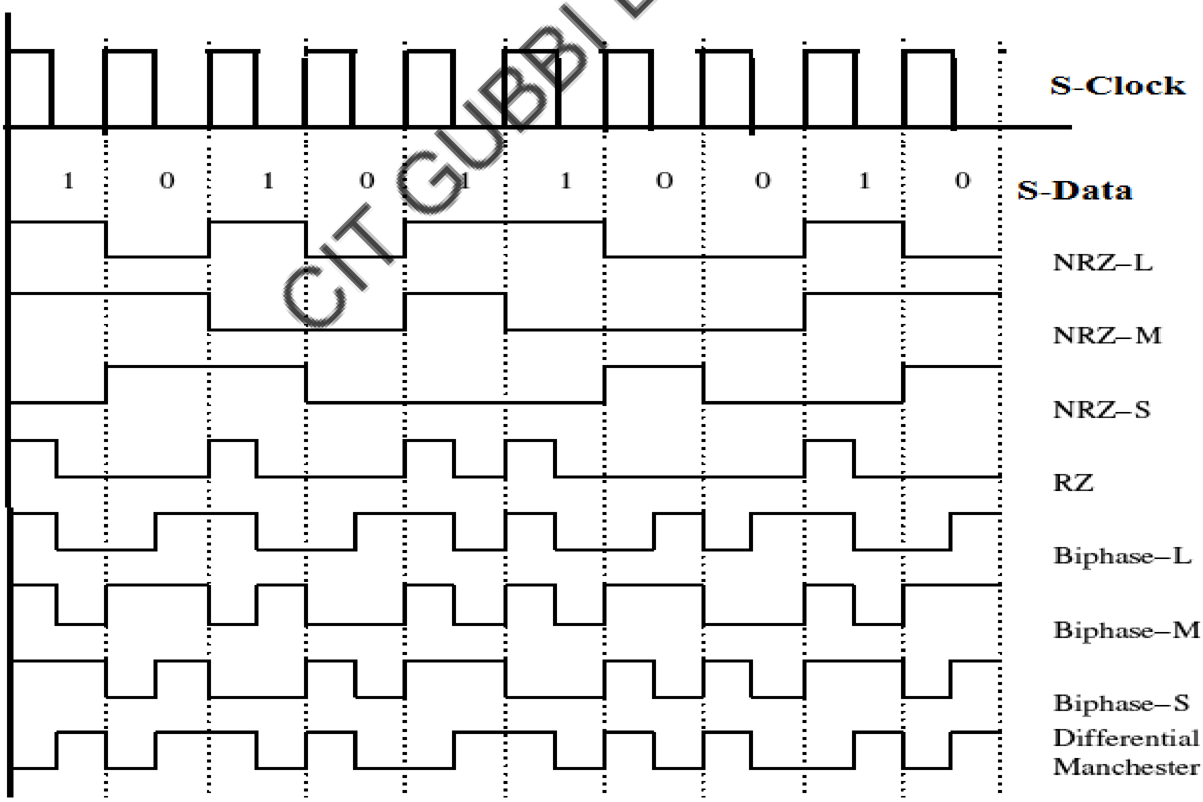
1. Data is generated with the help of a data bit generator.
2. Connect the data O/P of the data generator to the Tx data I/P of the trainer kit.
3. Now connect the clock of the generator to the Tx clock of the kit and ground with the ground terminal of the kit.
4. Select the data on the data generator and load it in the trainer kit by pressing load button.
5. Now observe the O/P of the NRZ-L, NRZ-M and Bi-phase.

Result:

Observed different data formatting methods & generation of line codes

Circuit Connections at digital trainer kit:





Waveforms of Data Formats

EXPERIMENT No.: 12
Date:

PCM MULTIPLEXER AND DEMULTIPLEXER

Aim: Study of Pulse code modulation (PCM) and its demodulation.

Apparatus Required:

Sl. No.	Component	Quantity
1.	PCM modulation / demodulation ST2103 trainer.	1
2.	CRO	1
3.	Connecting leads	2

Theory:

Pulse Code Modulation technique involves following steps:

(a) Sampling:

The analog signal is sampled according to the Nyquist criteria. The Nyquist criteria states that for faithful reproduction of a band limited signal, the sampling rate must be at least twice the highest frequency component present in the signal. So sampling frequency ≥ 2 fm, where fm is maximum frequency component present in the signal. Practically the sampling frequency is kept slightly more than the required rate.

(b) Allocation of Binary codes:

Each binary word defines a particular narrow range of amplitude level. The sampled value is then approximated to the nearest amplitude level. The sample is then assigned a code corresponding to the amplitude level, which is then transmitted. This process is called quantization and it is generally carried out by the A/D Converter as shown below in figure 01

Procedure:

1. Ensure that the MODE switch should be in FAST mode.
2. Connect CH 0 & CH 1 to DC1 AND DC2.
3. Ensure that the DC1 and DC2 controls, in Function Generator Block should be in fully clockwise direction and ~1KHz and 2 KHz signal controls set at 10Vpp.
4. Now turn ON the kit and see that the LED glows.
5. With the help of Digital Voltmeter, adjust the DC1 amplitude control until the DC1 output measures 0V.
6. Observe the output on the A/D Converter Block LED's (D0 to D6). The LED's represent the state of the binary PCM word allocated to the PAM sample being processed.
7. Adjust the D.C input from +5V to –5V in steps of 1V.
8. Observe the output of +5V is as follows:

D6	D5	D4	D3	D2	D1	D0
1	1	1	1	1	1	1

Where for the negative values it is less than 1000000. For –5V the output is as follows:

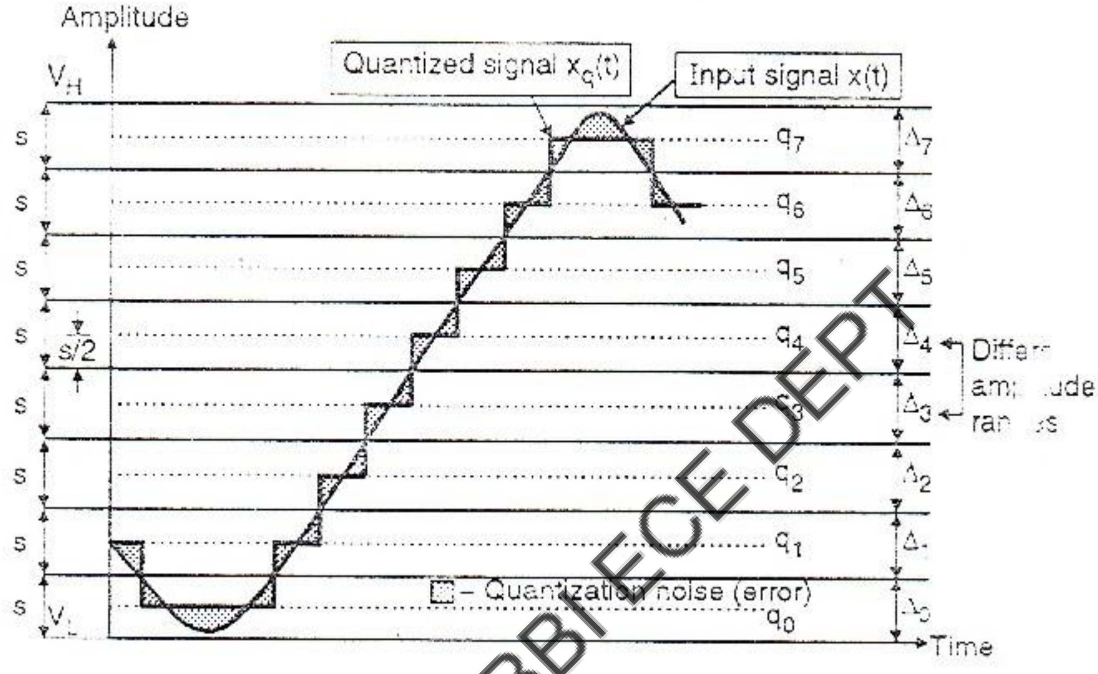
D6	D5	D4	D3	D2	D1	D0
0	0	0	0	0	0	0

This is obtained at the approximately full anti clockwise position of the DC Control.

9. Turn the DC1 control fully anticlockwise and repeat the above procedure by varying the DC2 control.
10. Trigger the dual trace oscilloscope externally by the CH.1 signal available at

t.p.12 and observe the signal at CH.0 and CH.1 at t.p.5 with reference to the signal at t.p.7.

Now connect the oscilloscope channel 1 to CH1 sample at t.p.6 and sketch the three waveforms

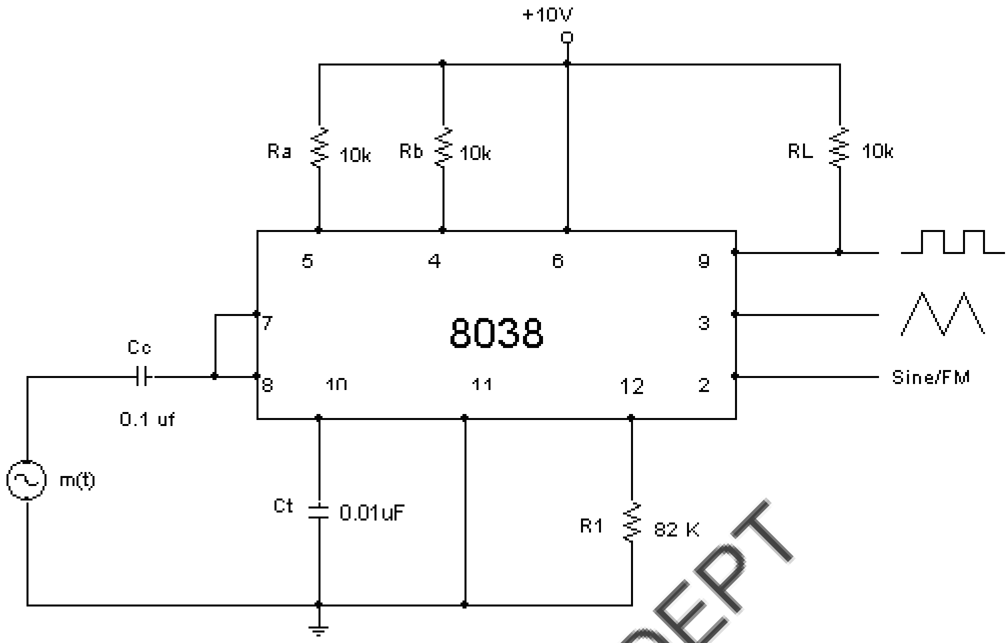


Process of quantization

Fig.1

Result: The PCM Modulation is performed and demodulated

Circuit Diagram :



Design :

Specifications : Carrier frequency fc = 3 kHz

fc = 0.3 / (R Ct) Where R = Ra = Rb

Assume R = Ra = Rb = 10 k∧ then Ct = 0.01 ⌈f

Choose RL = 10 k∧, R1 = 82 k∧, Cc = 0.1 ⌈f

Date:

ADDITIONAL EXPERIMENTS BEYOND SYLLABUS

FREQUENCY MODULATION

Aim: To conduct an experiment to generate Frequency Modulated wave and to measure frequency deviation and modulation index.

Apparatus:

Sl.No.	Particulars	Range	Quantity
1.	IC 8038	-	4
2.	Resistors & Capacitors	As per design	1 each
3.	CRO Probes	-	2 set

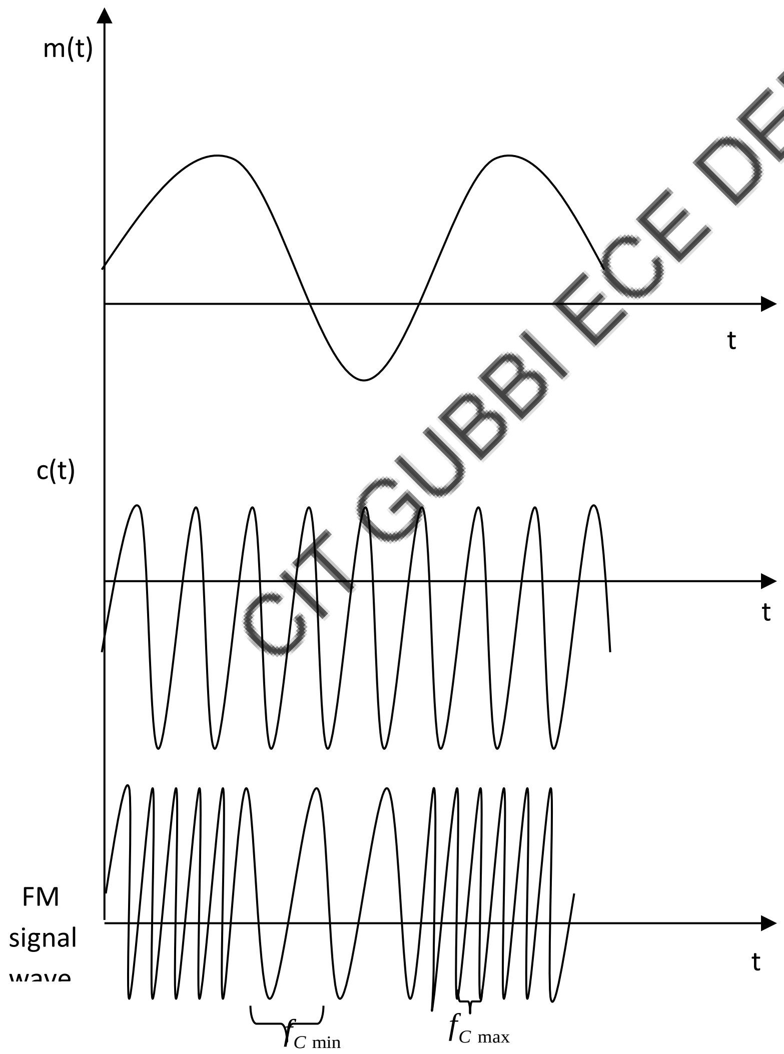
Theory:

Frequency modulation (**FM**) is a form of modulation which represents information as variations in the instantaneous frequency of a carrier wave. In analog applications, the carrier frequency is varied in direct proportion to changes in the amplitude of an input signal. Digital data can be represented by shifting the carrier frequency among a set of discrete values, a technique known as frequency-shift keying.

FM is commonly used at VHF radio frequencies for high-fidelity broadcasts of music and speech. Normal (analog) TV sound is also broadcasted using FM. A narrowband form is used for voice communications in commercial and amateur radio settings. The type of FM used in broadcast is generally called wide-FM, or W-FM. In two-way radio, narrowband narrow-fm (N-FM) is used to conserve bandwidth. In addition, it is used to send signals into space.

FM is also used at intermediate frequencies by most analog VCR systems, including VHS, to record the luminance (black and white) portion of the video signal. FM is the only feasible method of recording video to and retrieving video from magnetic tape without extreme distortion, as video signals have a very large range of frequency components — from a few hertz to several megahertz, too wide for equalizers to work with due to electronic noise below -60 dB. FM also keeps the tape at saturation level, and therefore acts as a form of noise reduction, and a simple limiter can mask.

aveforms :



Tabular Column :

fc = ____________ Hz, fm = ____________ Hz

Sl.No	Vm in V	fc $_{max}$ Hz	fc $_{min}$ Hz	™$_1$ Hz	™$_2$ Hz	™ Hz	® = ™/fm	B_T =2™+2fm In Hz

Where $\partial_1 = f_{C\max} - f_C$, $\partial_2 = f_C - f_{C\min}$

variations in the playback output, and the FM capture effect removes print-through and pre-echo. FM is also used at audio frequencies to synthesize sound. This technique, known as FM synthesis, was popularized by early digital synthesizers and became a standard feature for several generations of personal computer sound cards.

Modulation index

In FM modulation index indicates how much the modulated variable varies around its unmodulated level. For FM,

$$h = \frac{\Delta f}{f_m} = \frac{f_\Delta |x_m(t)|}{f_m}$$

With a tone-modulated FM wave, if the modulation frequency is held constant and the modulation index is increased, the (non-negligible) bandwidth of the FM signal increases, but the spacing between spectra stays the same. If the frequency deviation is held constant and the modulation index increased, the bandwidth stays roughly the same, but the spacing between spectra decreases.

Procedure:

1. Check the components/Equipments for their working condition.
2. Connections are made as shown in the circuit diagram.
3. By switching off the modulating signal m(t) note the frequency of the carrier wave at Pin No.2 of IC-8038.
4. Apply the modulating signal with suitable amplitude to get the FM signal.
5. Note the maximum (f_{cmax}) and minimum (f_{cmin}) frequency of the carrier wave in FM signal.
6. Calculate the frequency deviation, modulation index and bandwidth.

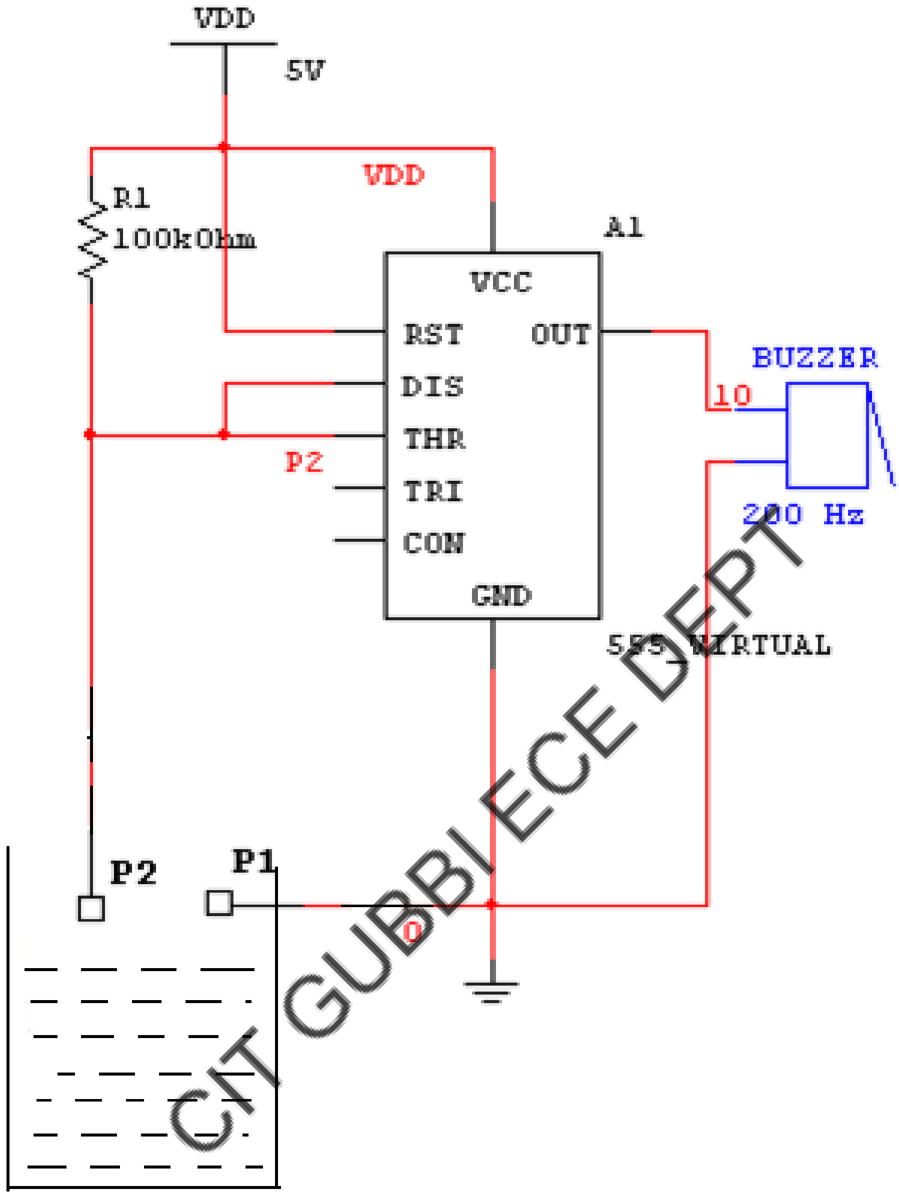
Result:

Modulation index = __________

Maximum Frequency Deviation = _____Hz.

Bandwidth of Operation = ___________ Hz.

Circuit Diagram



MINI PROJECT

MINI PROJECT-1:WATER LEVEL INDICATOR USING 555 TIMER

Aim: To make a water level detecting alarm using 555 Timer IC.

Apparatus:

Sl. No.	Particulars	Range	Quantity
1.	IC	555 Timer	1
2.	Resistor	100KΩ	1
3.	Buzzer	6-12V	1
4.	Connecting wires	-	1 set

Procedure:

1. Rig up the circuit as shown in the circuit diagram.
2. Apply +5V supply to VCC (pin no 8).
3. Keep the probes/wires inside the water container.
4. Observe the output.

Working:

The circuit is normally disabled and it gets enabled only when the probes touch the water, The distance between the probes should be less than a few centimeters to ensure that the conduction between the probes will take place when water is touched to these probes. When the water level rises to the height of the probes, then the 555 circuit will get enabled and the output is given to the mini loudspeaker/ Buzzer which then beeps.

Result: