COURSE LABORATORY MANUAL

A. LABORATORY OVERVIEW

Degree:	**B.E**	Programme:	EC
Semester:	IV	Academic Year:	2024-25
Laboratory Title:	Communication Laboratory-I	Laboratory Code:	BECL404
L-T-P-S:	0-0-2-0	Duration of SEE:	3 Hrs
Total Contact Hours:	24	SEE Marks:	50*
Credits:	1	CIE Marks:	50
Lab Manual Author:	**Mr. Nithin**		27/01/2025
Checked By:	**Mrs. Nisha G R**		30/01/2025

*The SEE will be conducted for 100 marks and proportionally reduced to 50 marks.

B. DESCRIPTION

1. PREREQUISITES:

- Knowledge of basic electronics.
- Electronic Principles and Circuits (BEC303)
- Digital System Design using Verilog (BEC302)
- Analog and Digital System design Lab (BECL305)

2. BASE COURSE:

- Principles of communication systems (BEC402)

3. COURSE OUTCOMES:

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.

4. RESOURSES REQUIRED:

- Signal generators, CROs, Dual regulated variable power supplies, IC trainer kit, Breadboards and Soldering Boards, Digital multimeters, Crocodile pins and connecting probes, Matlab Tool.

Prepared by: Nithin Checked by: Nisha G R HOD

5. RELEVANCE OF THE COURSE:

- Digital Communication
- Communication Engineering
- Optical and wireless communication

6. GENERAL INSTRUCTIONS:

- **Precautions for IC4052:** Applying a negative voltage to an input pin should be avoided. Make sure that the input pins have voltages within the specified values(0V-5V).
- Be regular to the lab.
- Maintain silence in the lab.

- Study the circuit, theory and procedures, expected output before doing the experiment.
- Get familiarize with the equipments in the lab.

7. CONTENTS:

Expt No.	Title of the Experiments	RBT	CO
1	Design and test a high-level collector Modulator circuit and Demodulation the signal using diode detector.	L3	CO1
2	Test the Balanced Modulator / Lattice Modulator (Diode ring)	L3	CO1
3	Design a Frequency modulator using VCO and FM demodulator using PLL.	L3	CO2
4	Design and plot the frequency response of Preemphesis and Deemphasis Circuits.	L3	CO2
5	Design and test BJT/FET Mixer.	L3	CO2
6	Design and test Pulse sampling, flat top sampling and reconstruction.	L3	CO3
7	Design and test Pulse amplitude modulation and demodulation.	L3	CO3
8	Generation and Detection of Pulse position Modulation.	L3	CO3
9	Generation and Detection of Pulse Width Modulation.	L3	CO3
10	PLL Frequency Synthesizer.	L3	CO4
11	Data formatting and Line Code Generation.	L3	CO4
12	PCM Multiplexer and Demultiplexer.	L3	CO3
13	Open ended experiment – 1	L3	
14	Open ended experiment – 2	L3	

8. REFERENCE:

1. Louis E Frenzel, Principles of Electronic Communication Systems, McGraw Hill Education (India) Private Limited, 2016.
2. B P Lathi, Zhi Ding, Modern Digital and Analog Communication Systems, Oxford University Press, 2015

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C. EVALUATION SCHEME

For CBCS 2021 scheme:

1. Laboratory Components: 50 Marks (Observation Writeup – 10 Marks + Lab Conduction – 10 Marks + Viva-Voce – 10 Marks + Record Writing – 20 Marks). 50 marks is scaled down to **30 marks**
2. Laboratory IA tests: **20** Marks
   (Minimum 1 IA test is mandatory. IA test shall be conducted for 50 Marks; for the final IA test marks shall be converted to maximum of 20).
3. Continuous Internal Evaluation (CIE) = 30 + 20 = 50 Marks.
4. *SEE* : 50* Marks
   (*The SEE will be conducted for 100 marks and proportionally reduced to 50 marks)

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D1. ARTICULATION MATRIX

Mapping of CO to PO

	POs											
COs	1	2	3	4	5	6	7	8	9	10	11	12
1.Illustrate the AM generation and	3	3	3	1	2	-	-	-	-	-	1	2

detection using suitable electronic circuits.												
2. Design of FM circuits for modulation, demodulation and noise suppression.	3	3	3	1	1	-	-	-	-	-	1	2
3. Design and test the sampling, Multiplexing and pulse modulation techniques using electronic hardware.	3	3	3	1	-	-	-	-	-	-	1	2
4. Design and Demonstrate the electronic circuits used for RF transmitters and receivers.	3	3	3	1	1	-	-	-	-	-	1	2
Note: Mappings in the Tables D1 (above) and D2 (below) are done by entering in the corresponding cell the Correllation Levels in terms of numbers. For Slight (Low): 1, Moderate (Medium): 2, Substantial (High): 3 and for no correllation: " - ".												

D2. ARTICULATION MATRIX CO v/s PSO

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

-

E. EXPERIMENTS

1. EXPERIMENT NO:1

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

3. LEARNING OBJECTIVES:
- To sketch and recognise the resulting waveforms for a sinusoidal carrier being amplitude modulated by a single frequency audio signal;
- To draw and analyse graphs to show the resulting waveform, and frequency spectrum for a sinusoidal carrier amplitude modulated by an audio signal, to a given depth of modulation, m;
- To select and use the formula: $m=\left(\frac{Emax-Emin}{Emax+Emin}\right)\times 100$

4. AIM:
- To design a circuit to generate AM wave using transistor.

5.MATERIAL / EQUIPMENT REQUIRED:

Transistor : BC 107/SL-100, Resistors: 1kΩ, 10kΩ, 22kΩ and 6.8kΩ, Capacitors: 1μF, 100nF, 4.7μF 0.1μF, Signal Generator, Power supply, Diode: 0A79, Decade Resistance Box, IFT, CRO.

6. THEORY :
- Amplitude Modulation is defined as a process in which the amplitude of the carrier wave c

(t) is varied linearly with the instantaneous amplitude of the message signal m(t). The standard form of amplitude modulated (AM) wave is defined by

Where K is a constant called the amplitude sensitivity of the modulator.
m<1------------ Under Modulation
m=1------------ 100% Modulation
m>1------------ Over Modulation .
The demodulation circuit is used to recover the message signal from the incoming AM wave at the receiver. An envelope detector is a simple and yet highly effective device that is well suited for the demodulation of AM wave, for which the percentage modulation is less than 100%. Ideally, an envelope detector produces an output signal that follows the envelop of the input signal wave form.

$$s(t) = A_c\big(1 + k_a m(t)\big)\cos(2\pi f_c t)$$

- The Modulation Index defined as, where Vmax and Vmin are the maximum and minimum amplitudes of the modulated wave.

$$m= (V_{max}- V_{min}) /(V_{max}+ V_{min})$$

7. FORMULA / CALCULATIONS:

$$m= (V_{max}- V_{min}) /(V_{max}+ V_{min})$$

where Vmax and Vmin are the maximum and minimum amplitudes of the modulated wave.

8. PROCEDURE :

<u>Modulation:</u>

- Rig up the connections as shown in the circuit diagram.
- Switch off the modulation signal source
- Set the input carrier frequency to 455kHz to 530kHz, 1Vp-p and tune the IFT to get maximum carrier amplitude
- Set the modulating signal to 500Hz, 2Vp-p and apply to the collector circuit through an audio driver transformer
- Observe the modulated carrier across the output terminals
- Adjust the amplitude of modulation and carrier signals so as to get undistorted modulation. By measuring $Emax$ and $Emin$ modulation index has to be calculated.

Percentage of modulation

$$m=\left(\frac{Emax-Emin}{Emax-Emin}\right)\times 100$$

<u>Demodulation:</u>

- Choose R and C values by using the condition

$$Tmodulation \gg T = R*C \gg T\ carrier$$

And $f\ modulation=\frac{1}{(2\pi RC\,m)}$ where m is the modulation index

Consider m= 0.5 and $f\ modulation = 3.3\text{kHz}$ for speech signal

Let C = 0.1μF

Then $R = \dfrac{1}{(2\pi * 3.3k * 0.1\mu * 0.5)}$ = 956Ω or choose R=1kΩ

- Connect decade resistance box and vary the value of R around 1kΩ so as to get demodulated AM having minimum carrier ripple.

9. CIRCUIT DIAGRAM :

MODULATOR:

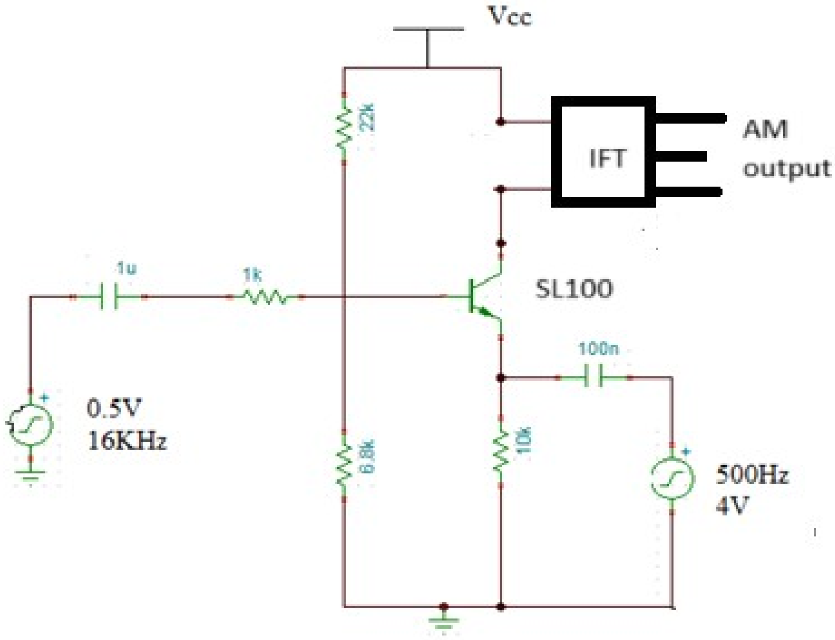


Fig-1a: Modulator Circuit

DEMODULATOR:

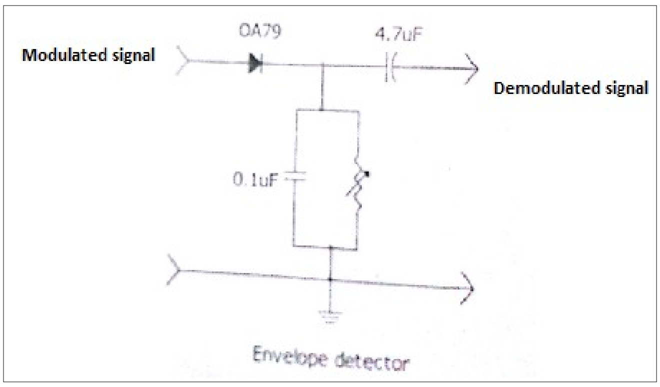


Fig-1b. Demodulator Circuit

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

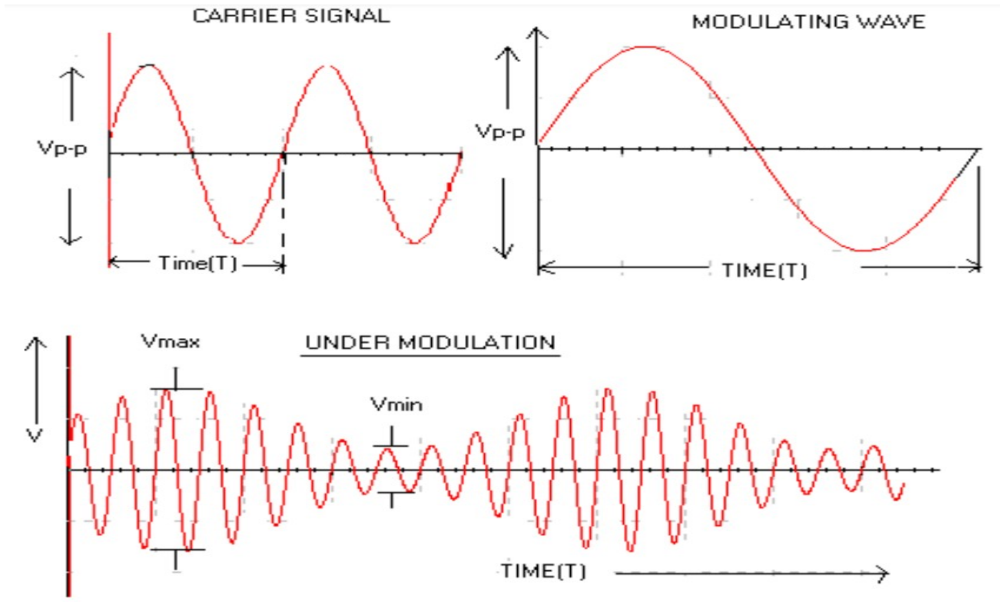


Fig 1c: AM graph

12. RESULTS & CONCLUSIONS:

- PERCENTAGE of modulation $m=\left(\frac{Emax-Emin}{Emax-Emin}\right)\times 100$.

13. LEARNING OUTCOMES :

- Amplitude modulated signal is observed and its envelope is compared with modulating signal. It is observed that the envelope of the modulated signal is same shape as the modulating signal.

14. APPLICATION AREAS:

- Broadcast transmissions
- Air band radio
- Cellular telecommunications

15. REMARKS:

- -

1. EXPERIMENT NO:2

2. TITLE: TEST THE BALANCED MODULATOR / LATTICE MODULATOR (DIODE RING).

3. LEARNING OBJECTIVES:

- To understand the concept of balanced modulator.
- To design a DSBSC generation circuit using IC 1496

4. AIM: To design a DSBSC generation circuit using Balanced Modulator IC 1496

5. MATERIAL / EQUIPMENT REQUIRED:

- DSBSC Kit, CRO, Power Supply

6. THEORY:

- Balanced modulator is used for generation of double side band suppress carrier signal. The output of balanced modulator is equal to the product of applied input signals .In order to generate this it uses non linear characteristics of semi conductor devices. Since the carrier does not convey any information, transmitting the carrier along with side band is only wasting of transmission power; therefore carrier is suppressed before transmission. By doing suppression 67% of transmission power can be saved. The method of transmission of modulated wave without carrier is DSBSC signal.

- Balanced modulator is also used in generation of SSB signals. The modulated signal undergoes a phase reversal whenever the base band signal crosses zero. Unlike AM, The envelope of DSBSC is different from base band signal .The ring modulator is another circuit for generation the DSBSC signal.

7. FORMULA / CALCULATIONS:

8. PROCEDURE / PROGRAMME / ACTIVITY:

MODULATION:

- Connect the circuit diagram as shown in above diagram.
- Carrier signal of 1Vp-p amplitude and frequency of 25 KHz is applied as carrier to S4.
- Message signal of 0.5Vp-p amplitude and frequency of 1.5 KHz is given as message signal to S1.
- Observe the DSB-SC waveform using CRO in Test point S2.
- Note down the Vmax and Vmin values and draw the waveform on the graph sheet.

DEMODULATION:

- Connections are made as shown above diagram.
- Connect the Balanced Modulator output (test point S2) to synchronous detector input s8.
- Vary the Pot VR4 to get clear Message signal at the output of Synchronous detector s7 using CRO.
- Sketch the waveforms for balanced modulated signal and demodulated signal.

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

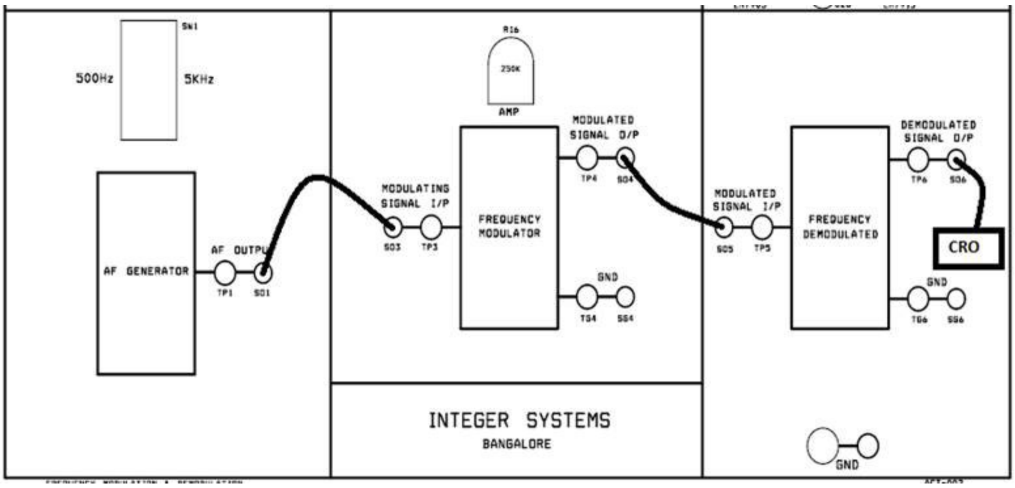


Fig: Modulation of DSBSC

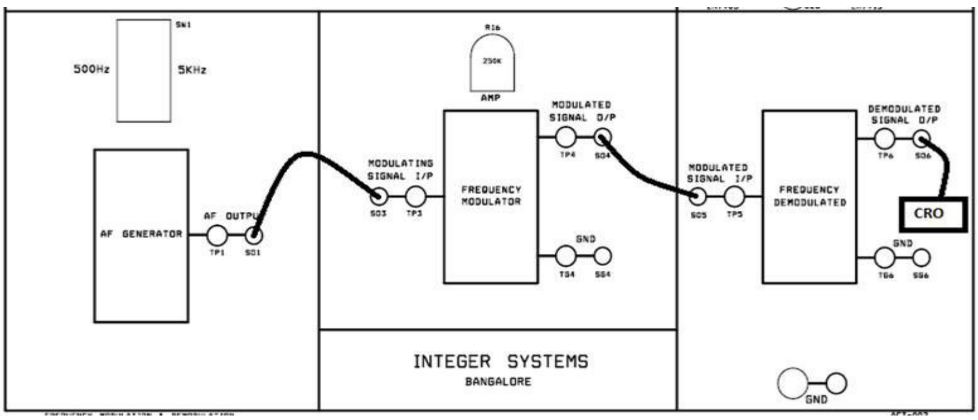


Fig : DSBSC demodulation connection

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11.GRAPHS / OUTPUTS:

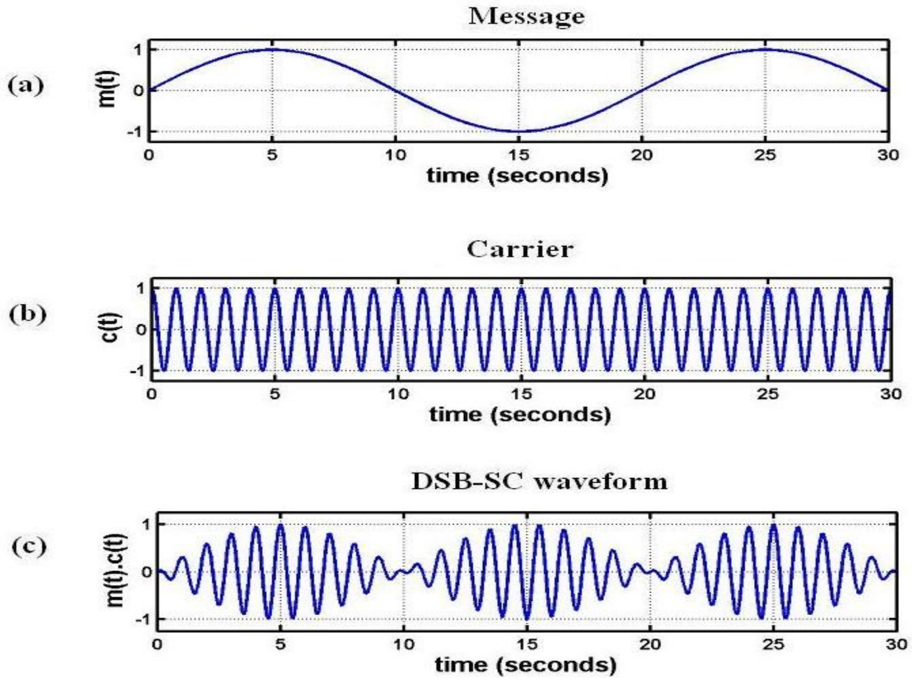


Fig : DSBSC Waveform

12. RESULTS & CONCLUSIONS:

- Modulated signal:

  Frequency =
  Time period =
  Vmax=
  Vmin=

13. LEARNING OUTCOMES :

- DSBSC generation circuit using Balanced Modulator IC 1496 is designed and the output is verified.

14. APPLICATION AREAS:

- Communication systems.
- Broadcast transmission.

15. REMARKS:

- -

1. EXPERIMENT NO: 3

2. TITLE: DESIGN A FREQUENCY MODULATOR USING VCO AND FM DEMODULATOR USING PLL.

3. LEARNING OBJECTIVES:

- To understand Angle modulation concept.
- To learn how to generate FM signal.
- To learn how to build FM demodulator.

4. AIM:

- To conduct a suitable experiment to generate an FM wave using IC 8038.

5. MATERIAL / EQUIPMENT REQUIRED:

- IC 8038 , IC 565, Resistors: 10KΩ-4, 81kΩ. Capacitors: 0.01uF, 0.001uF ,470uF, 10uF-2, 100uF, Signal generators, Regulated DC power supply, CRO,Patch chords and probes.

6. THEORY:

- Frequency modulation is that form of angle modulation in which the instantaneous frequency is varied linearly with the message signal. The IC 8038 waveform generator is a monolithic integrated circuit capable of producing high accuracy sine square , triangular, saw tooth and pulse waveforms with a minimum number of external components. The operation of IC 8038 is based on charging and discharging of a grounded capacitor C, whose charging and discharging rates are controlled by programmable current generators Ia and Ib. When switch is at position A, the capacitor charges at a rate determined by current source Ia.
- Once the capacitor voltage reaches Vout, the upper comparator (CMP 1) triggers and reset the flip-flop out put. This causes a switch position to change from position A to B. Now, capacitor charge discharging at the rate determined by the current sink Ib . Once the capacitor reaches lower threshold voltage, the lower comparator (CMP 2) triggers and set the flip-flop output. This causes the switch position to change from position B to A. And this cycle repeats. As a result, we get square wave at the output of Flip flop and triangular wave across capacitor. The triangular wave is then passed through the on chip wave shaper to generate sign wave. To allow automatic frequency controls, currents Ia and Ib are made programmable through an external control voltage Bi.
- For equal magnitudes of Ia and Ib, output waveforms are symmetrical conversely, when two currents are unequal, output waveforms are asymmetrical. By making, one of the currents much larger than other we can get saw tooth waveform across capacitor and rectangular waveform at the output of flip-flop. Frequency modulation, FM is widely used for a variety of radio communications applications, provides high quality audio, two way radio communications, mobile radio communications, taxis, etc.

7. FORMULA / CALCULATIONS:

- LET, $R_A = R_B = R$ THEN

  $f_{out} = (\frac{0.3}{RC})$ when 7 & 8 are shorted together.

  If a single timing resistor is used (common resistor for both 4 and 5 timing pins) than

  $f_{out} = (\frac{0.15}{RC})$ $Sensitivity = \frac{\Delta f}{Vin}$

8. PROCEDURE :

Modulation

- Ensure the proper working of IC 8038 by using the test circuit. The pin 7 provides FM bias of approximately 9V. Connect pin 7 to pin 8 and note down the output sinusoidal frequency and its Vp-p
- Disconnect pin 7 from pin 8. Apply a variable dc input to pin 8 in steps of 0.5V from 9V to 12V. This is called input sweep voltage. The sweep voltage should be within the range of (((2/3)Vcc)+2) < $V_{sweep} < V_{cc.}$ For 12Vcc it is 10V to 12V.
- Plot the same and obtain Vin v/s f_{out} characteristic of modulator.
- Obtain the sensitivity of the modulator , S = (Δfout/ΔVin)Hz/volt

- Demonstration part: Connect a voltage divider network comprising of 1kΩ and 5kΩ to pin
- The voltage drop of 10.5V across 5kΩ to pin 8. The voltage drop of 10.5V across 5kΩ will serve FM bias to pin 8. This voltage is the mid of (((2/3)Vcc)+2) and Vcc.

9. CIRCUIT DIAGRAM :

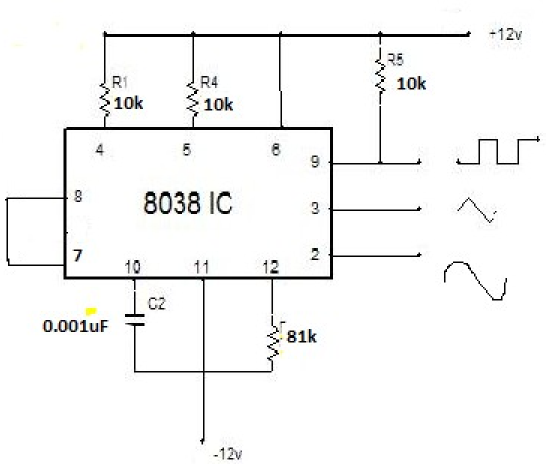


Fig 3a: 8038 IC Test Circuit

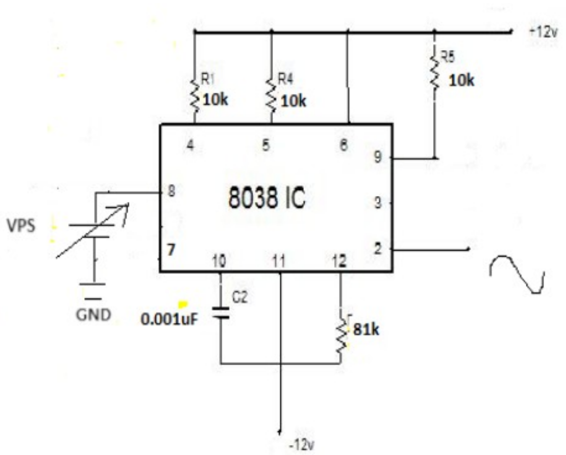


Fig 3b: 8038 Characteristics

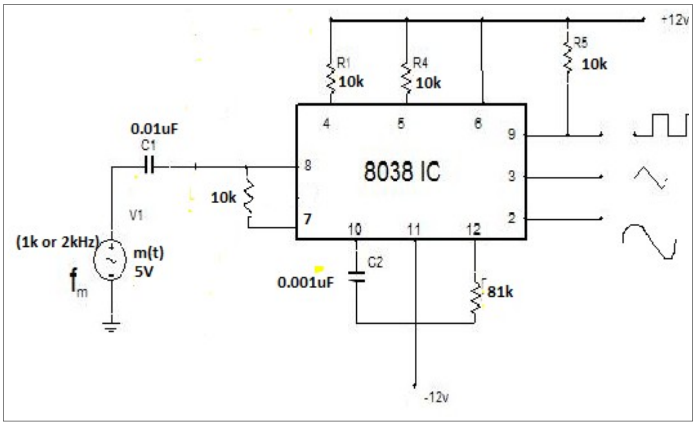


Fig 3c: FM modulator circuit diagram

10. OBSERVATION TABLE :

Vin v/s f_{out} characteristics

DC voltage (V)	Frequency (Hz)
11V	
:	
:	
:	
6V	

11. GRAPHS / OUTPUTS:

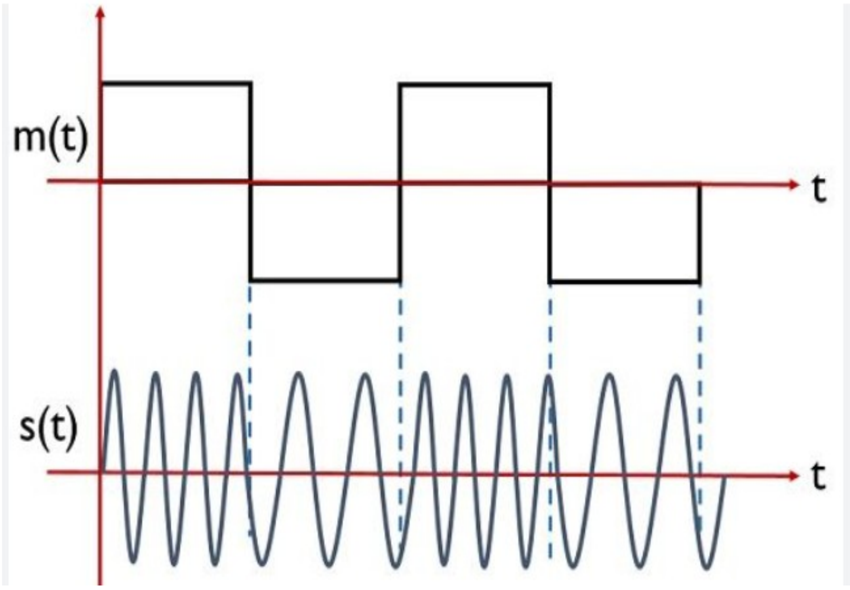


Fig-3c: FM waveform

12. RESULTS & CONCLUSIONS:

- MINIMUM FM WAVE/SIGNAL FREQUENCY = __________
- Maximum FM wave/signal frequency = __________

13. LEARNING OUTCOMES :

- Frequency modulation circuit using IC 8038 is designed and verified.

14. APPLICATION AREAS:

- FM radio
- Satellite TV
- TV broadcast

15. REMARKS:

1. EXPERIMENT NO: 4

2. TITLE: DESIGN AND PLOT THE FREQUENCY RESPONSE OF PREEMPHASIS AND DEEMPHASIS CIRCUITS.

3. LEARNING OBJECTIVES:

- To understand frequency response of preemphasis concept.
- To learn how to plot frequency response of preemphasis circuit.
- To learn how to plot frequency response of deemphasis circuit.

4. AIM:

- TO DESIGN AND PLOT THE FREQUENCY RESPONSE OF PREEMPHASIS AND DEEMPHASIS CIRCUITS.

5. MATERIAL / EQUIPMENT REQUIRED:

- Op-amp uA741, Capacitor 0.1µF, Resistors: 1KΩ, 820Ω, 560Ω, Connecting wire, CRO, Signal generator, DC supply

6. THEORY / HYPOTHESIS:

The pre-emphasis and de-emphasis help to improve the quality of any communication especially audio signals on the transmitter and receiver sides. The presence of noise is also an issue in FM and we know that noise usually has higher amplitude and higher frequency. When the amplitude of a high-frequency noise is higher than the current component in the modulation signal, it causes high-frequency interference. To deal with this issue, most FM circuits use a technique called pre-emphasis during transmission and de-emphasis during receiving. Pre-emphasis and de-emphasis circuits are commonly used in FM transmitters and receivers to improve the output signal-to-noise ratio. The pre-emphasis circuit is actually a high pass filter and de-emphasis circuit a low pass filter. The amount of pre-emphasis and de-emphasis used is defined by the time constant of a simple RC filter circuit. Simple pre-emphasis and de-emphasis circuits using op-amp are given in the diagram.

7. FORMULA / CALCULATIONS:

- Choose appropriate value for Time constant, T

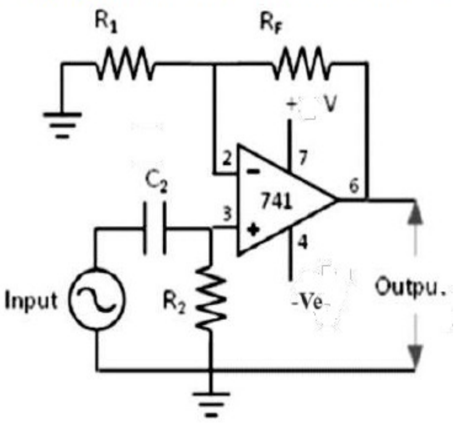
Sound transmission in TV have been standardized at 75 µsec. Therefore, the time constant T = 75µ = R_2C_2 With general assumptions of value of C_2 = 0.1µF then find R_2 = T / C_2=75µ /0.1µ = 750Ω (Assume 820Ω as standard Value). For Butterworth filters, Gain A = 1.586 Gain of non-inverting amplifier = 1+Rf/R1 → 1.586 = 1+Rf/R1 → Rf/R1 = 0.586 Take suitable value of R1=1KΩ, then find Rf =586 (Assume 560Ω as standard Value).

8. PROCEDURE / PROGRAMME / ACTIVITY:

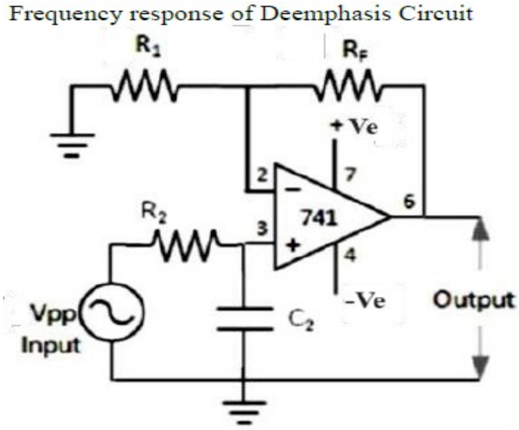
1. Test all the components and probes.
2. Set up the pre- emphasis circuit on a bread board as shown in figure.
3. Feed a sine wave as input =1V.
4. Vary the frequency from 100 Hz to 100KHz on step of 500 Hz and note down the values of the corresponding output voltage on a tabular column.
5. Plot frequency response on a graph sheet with log f on x-axis and gain in dB on yaxis.
6. Mark the cut-off frequencies corresponding to 3dB points.
7. Repeat the above steps for de-emphasis circuit.-

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

Frequency response of Pre-emphasis Circuit



Frequency response of Deemphasis Circuit

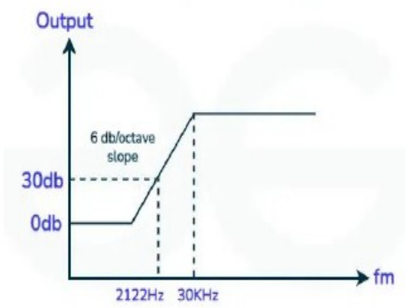


10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

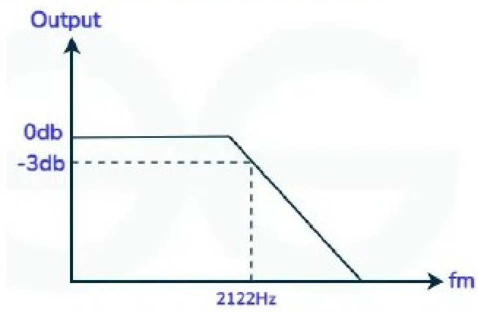
Frequency F in Hz	Output Voltage Vo in Volts	Output = 20log (Vo)
100 Hz		
...		
...		
....		

11. GRAPHS / OUTPUTS:

Frequency response of Pre-emphasis



Frequency response of Deemphasis



12. RESULTS & CONCLUSIONS:
- Designed and verified and ploted the frequency response of Pre-emphasis and Deemphasis Circuits

13. LEARNING OUTCOMES :
- The frequency response of preemphesis and deemphasis circuits are analyzed.

14. APPLICATION AREAS:
- FM broadcasting
- Audio recording
- Digital communication

15. REMARKS:
- -

1. EXPERIMENT NO: 5
2. TITLE: DESIGN AND TEST BJT/FET MIXER.
3. LEARNING OBJECTIVES:
   - To understand the working of mixer circuit.
   - To verify the output of mixer
4. AIM:
   - To design and verify the output of mixer.
5. MATERIAL / EQUIPMENT REQUIRED:
   - Transistor BC107/2N3094, Resistors: 100 kΩ-2, 10 kΩ-3, 22 kΩ, Capacitor 0.1μF-2, 0.001μF, 1μF, AFO, CRO, RPS.
6. THEORY :
   - An electronic mixer is a device that combines two or more electrical or electronic signals into one or more composite output signals.as shown in Fig.6a. Multiplying mixers multiply together two time-varying input signals instantaneously (instant-by-instant). If the two input signals are both sinusoids of specified frequencies fx and f y , then the output of the mixer will contain two new sinsoids that have the sum fx + f y frequency and the difference frequency absolute value | fx - f y |.

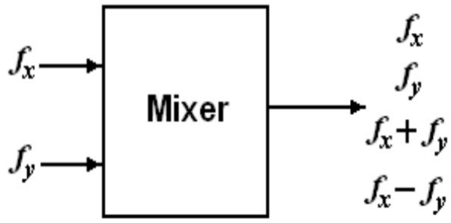


Fig6a: A mixer produces the original frequencies, the sum, and the difference

- In Fig.6a the input signals have frequencies of fx and f y . Because of non-linear distortion, the output signal contains the original input frequencies, sum and difference frequencies plus their harmonics. For instance, if the two input frequencies are 100 and 101 kHz, the output signal contains the frequencies, 100, 101, 200, 202, 300, 303 kHz, and so on. In addition to the harmonics that are produced, new frequencies appear in the output that are equal to the sum and difference of the two input frequencies.

- If fx and f y are the input frequencies, the new frequencies are
  Sum = fx+ fy
  Difference frequency absolute value = | fx - fy|

- Fig.6b is an example of a transistor mixer. One signal drives the base; the other drives the emitter. One of the input signals is large; this is necessary to ensure non-linear operation. The other input signal is usually small. One of the reasons this small is because it often is a weak signal coming from an antenna. At the output of collector, the two original frequencies, sum, difference and their harmonic frequencies are produced. The mixer output of Fig.6b is filtered by two low-pass RC circuits (2nd order LPF) that eliminates all the higher frequencies and passes only the difference frequency signal. The approximate cutoff frequency of each RC circuit is given by fc=1/2 πRC .

7. FORMULA / CALCULATIONS:

8. PROCEDURE :

- Connections are made as per the circuit diagram shown in Fig..
- Turn Vy down to 0. With the oscilloscope, adjust Vx to 0.1Vpp . Set the frequency 101 kHz.
- Now adjust Vy to 1Vpp and 100 kHz.
- Observe the final output signal with a vertical sensitivity of 0.1 V/cm (ac input) and a sweep time of 0.2 ms/cm. Vary the frequency of the Vx generator slowly in the vicinity of 101 kHz until you get a 1 kHz output signal. Tabulate the results.
- Observe at point B, the input to the final RC filter. Note the ripple on the 1kHz signal.
- Observe at point A, the input to the first RC filter. Use vertical sensitivity of 2V/cm. Note how large the ripple is here.
- Repeat the above steps for various in put frequencies.

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

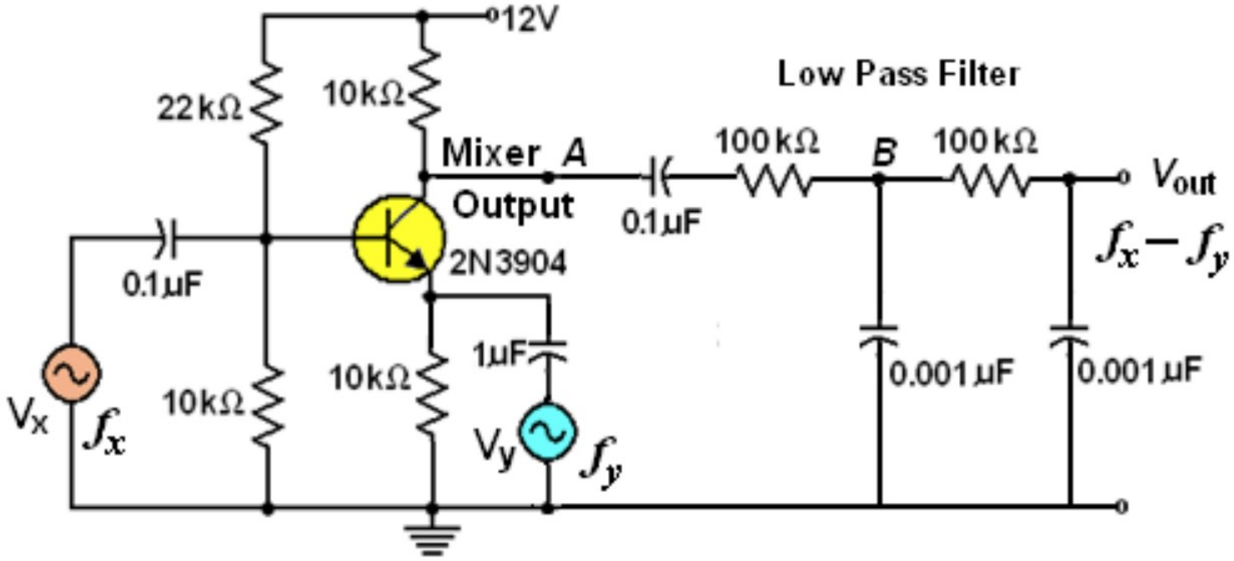


Fig 5: BJT Mixer circuit

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

Parameter	Input Signal 1	Input Signal 2	Output signal
Amplitude(V)			
Frequency(Hz)			

11. GRAPHS / OUTPUTS:

12. RESULTS & CONCLUSIONS:
- Frequency of the output signal=---------Hz

13. LEARNING OUTCOMES :
- BJT mixer circuit output has been verified

14. APPLICATION AREAS:
- Mixer circuits are used in superheterdyne receivers.

15. REMARKS:
- -

1. EXPERIMENT NO: 6

2. TITLE: DESIGN AND TEST PULSE SAMPLING, FLAT TOP SAMPLING AND RECONSTRUCTION.

3. LEARNING OBJECTIVES:
- To understand the concept of sampling.
- To design and demonstrate the working of flat top sampling circuit for a given message signal and to reconstruct the original message signal.
- To demonstrate the effect of Under sampling, Right Sampling and Over sampling.

4. AIM:
- To design and demonstrate the working of flat sampling circuit for a given message signal and to demonstrate the effect of Under sampling, Right Sampling and Over sampling.

5. MATERIAL / EQUIPMENT REQUIRED:
- IC 4052
- IC 741
- Resistors:8.2KΩ, Capacitors: 1µF, 0.1µF, 4.7µF, Diode:0A79

- Dual Power supply
- Function generator, CRO

6. THEORY:

- In order to perform any form of processing by digital computers, the signals must be reduced to discrete samples of a discrete time domain. The operation that transforms a signal from the continuous time to the discrete time is called sampling and it is performed by picking up the values of the continuous time signal at time instants that are multiple of a quantity T, called sampling interval. The quantity Fs=1/T is called sampling rate.
- After an analog signal is sampled, it is converted into a series of pulses whose amplitude is equal to the amplitude of the analog signal at the start of the sampling process. Since the sampled pulses have uniform amplitude the process is called flat top sampling. Alternatively , in a process called natural sampling, the amplitude of the sampled pulse is allowed to vary with the amplitude of the analog waveforms as it changes during the sampling period.

7.FORMULA / CALCULATIONS:

8. PROCEDURE :

- Rig up the circuit as shown in the figure.
- Apply a sinusoidal message signal of 200Hz, 5Vp-p to the pin No 5 of IC4052
- Apply keying input square wave of say 3KHz and 5Vp-p to the pin No.10
- Observe the flat top sampled output across the holding capacitor C at pin 3.
- Vary the sampling frequency and verify the effect of over sampling, under sampling and Right sampling.
- Apply the flat top sampled signal to the reconstruction filter f= $\frac{1}{2\pi RC}$ demodulate it and compare with the message signal and verify the Nyquist sampling rate.

9. CIRCUIT DIAGRAM:

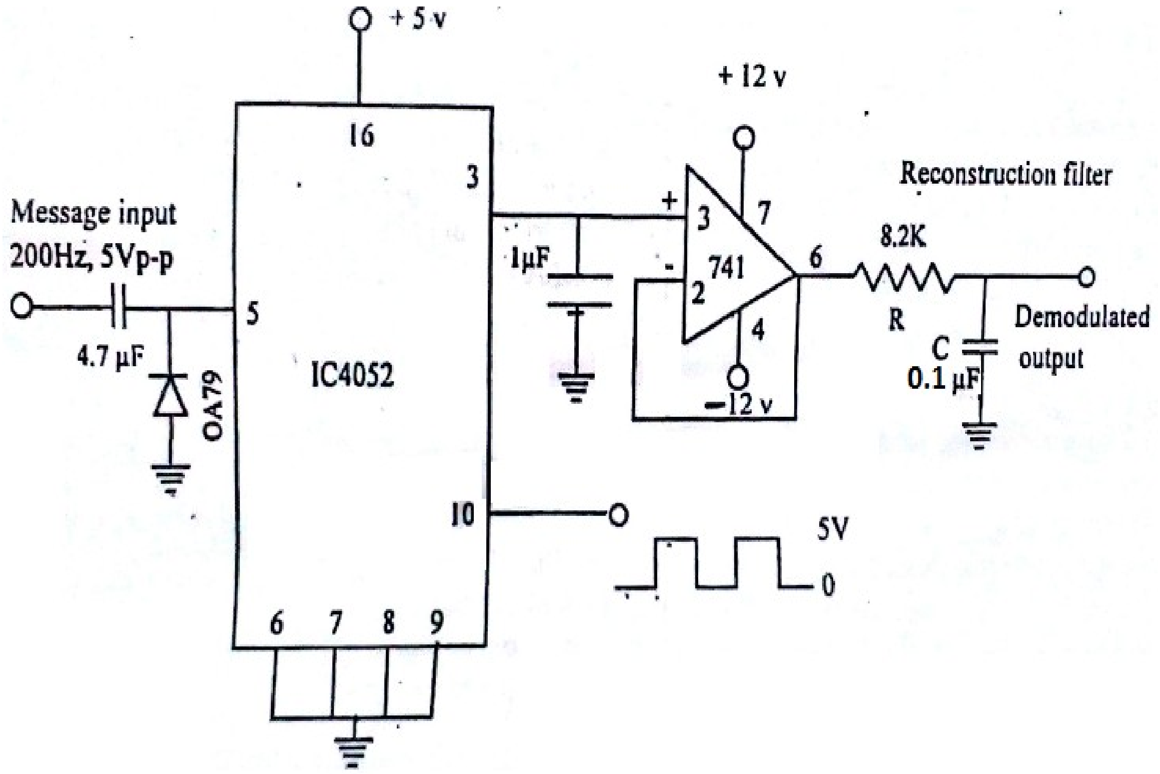


Fig-6a: Pulse flat top sampling and reconstruction circuit diagram

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

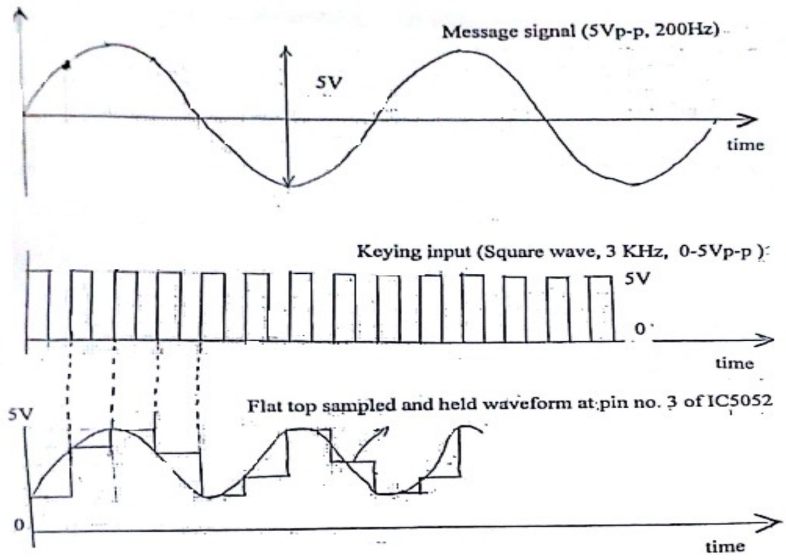


Fig 6b: Flat top sampling output waveform

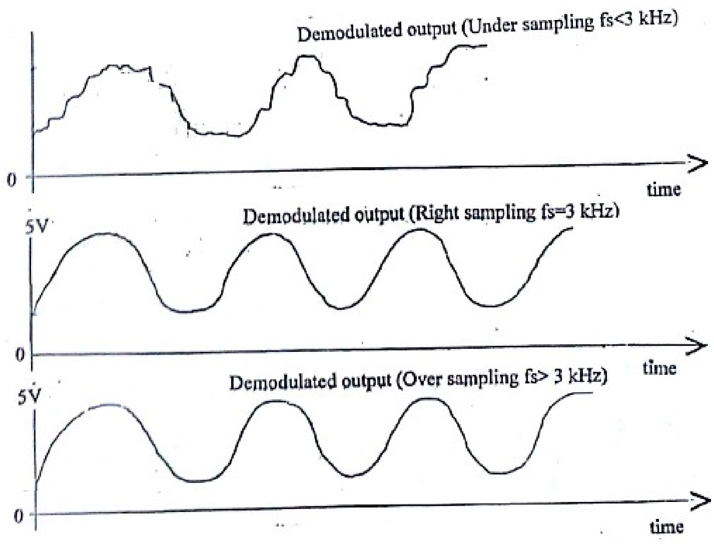


Fig 6c: Reconstruction waveform

12. RESULTS & CONCLUSIONS:

- Flat sampling circuit for a given message signal is designed and the reconstructed signal is verified.

13. LEARNING OUTCOMES:

- Flat sampling circuit for a given message signal is designed and the reconstructed signal is verified.

14. APPLICATION AREAS:

- Digital-to-analog Converters.
- Digital Communication.

15. REMARKS:

- -

1. EXPERIMENT NO: 7

2. TITLE: DESIGN AND TEST PULSE AMPLITUDE MODULATION AND DEMODULATION.

3. LEARNING OBJECTIVES:
- To understand the concept of pulse amplitude modulation and demodulation.
- To design and test pulse amplitude modulation and demodulation.

4. AIM: To generate the Pulse Amplitude modulated and demodulated signals.

5. MATERIAL / EQUIPMENT REQUIRED: Resistors 1KΩ, 10KΩ, 22KΩ, 560Ω, 5KΩ 2.2KΩ, Transistor BC107, Capacitor 4.7µF, CRO 30MHz, Function Generator 1MHz, Regulated Power Supply 0-30V, 1A and CRO Probes

6. THEORY :
- Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals. At the receiving end, the original waveforms may be reconstituted from the information regarding the samples. The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system is which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cables are used to modulated carrier.
- The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short. Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform. Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave.
- PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which can be easily distorted during transmission by noise, crosstalk, other forms of distortion. They are used frequently as an intermediate step in other pulse modulating methods, especially where time-division multiplexing is used.

7. FORMULA / CALCULATIONS:

8. PROCEDURE :
- Connect the circuit as per the circuit diagram shown in the fig: 7a.
- Set the modulating frequency to 3KHz and apply 0.5V message signal
- Set the sampling frequency to 15KHz and apply 1.5V sampling signal
- Observe the o/p on CRO i.e. PAM wave.
- Measure the levels of Vmax & Vmin
- Feed the modulated wave to the demodulated circuit as in fig 7b.
- The output observed on CRO will be the demodulated wave.
- Note down the amplitude (p-p) and time period of the demodulated wave.
- Vary the amplitude and frequency of modulating signal.
- Observe and note down the changes in output.
- Plot the wave forms on graph sheet.

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

Modulation

22KΩ

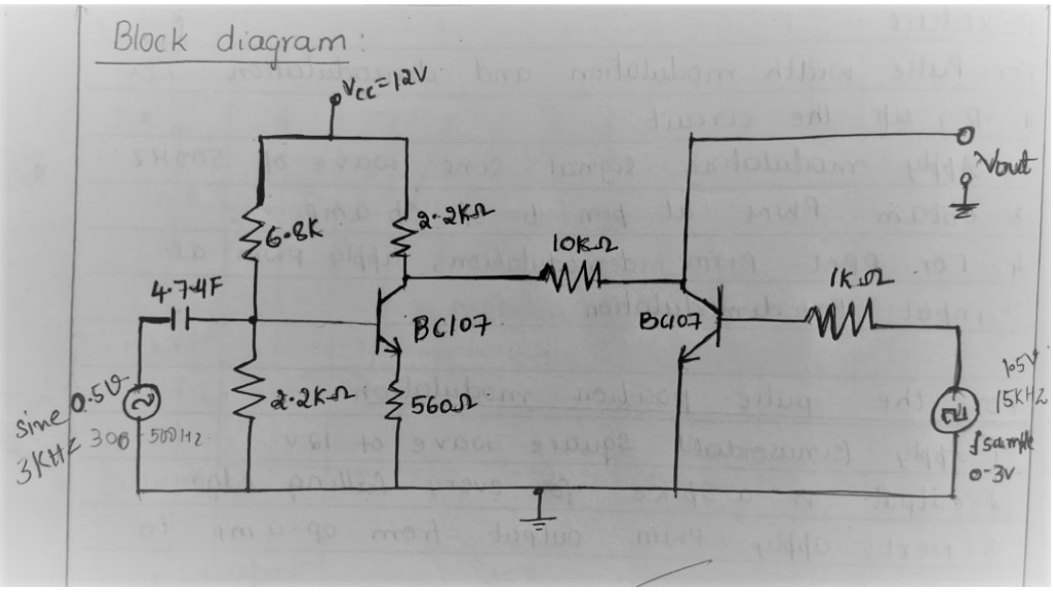


Fig 7a: PAM circuit diagram

Demodulation

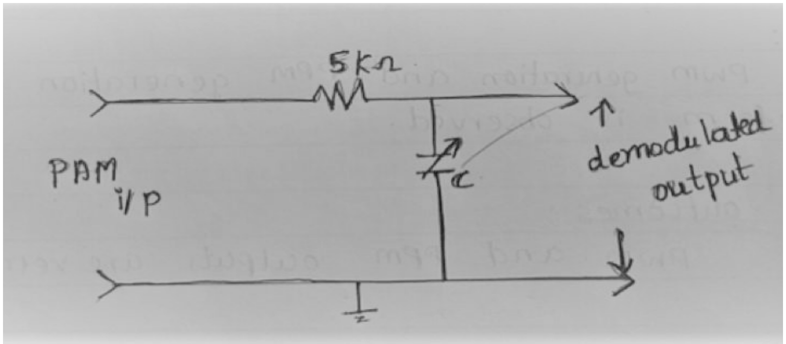
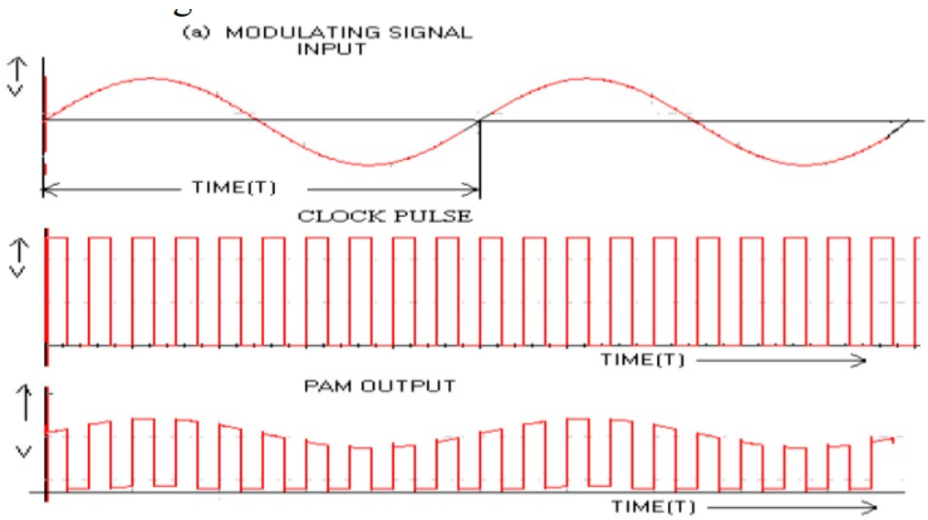


Fig 7b: Demodulation circuit diagram

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS :



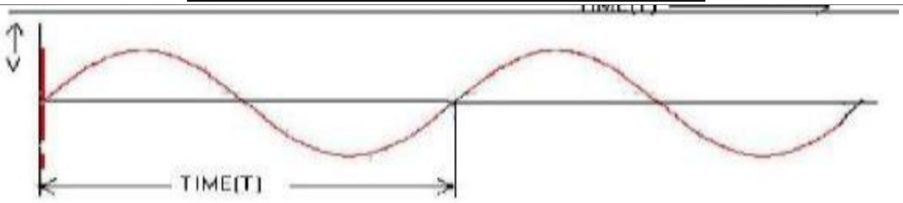


Fig 7c: PAM Modulation and Demodulation Waveform

12. RESULTS & CONCLUSIONS:

- Pulse Amplitude modulation, Vmax=---------V & Vmin=------------V.
- Frequency of demodulated signal=---------Hz

13. LEARNING OUTCOMES :

- Pulse amplitude modulation and demodulation waveforms observed.

14. APPLICATION AREAS:

- Ethernet Communications
- LED Drivers

15. REMARKS:

- -

1. EXPERIMENT NO: 8 & 9

2. TITLE: PULSE WIDTH MODULATION AND PULSE POSITION MODULATION

3. LEARNING OBJECTIVES:

- To understand and analyse the working of pulse modulation and demodulation.

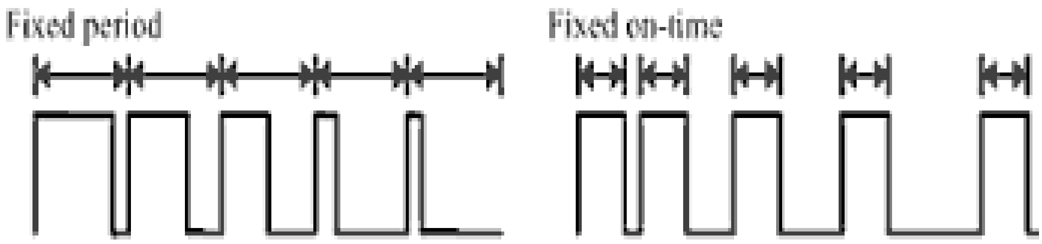
4. AIM:

- TO DESIGN CIRCUIT FOR PULSE MODULATION AND DEMODULATION AND ALSO TO GENERATE PULSE POSITION MODULATION.

5. MATERIAL / EQUIPMENT REQUIRED:

Sl. No	Components	Value	Qty
1	Op-amp	µA741	1
2	Resistors	10kΩ 1kΩ	2 3
3	Capacitors	0.1uF 0.01uF	1 2
4	Diode	OA79 1N4001	1 1
5	Decade capacitance box		1
6	IC555		

6. THEORY :



(a) Pulse-width modulation keeps constant frequency (period) and varies the duty cycle.

(b) Pulse-period modulation keeps constant on- or off-time and varies the period (frequency).

Pulse width modulation is also known as pulse duration modulation. PWN signal is binary in amplitude . The width or duration of the pulses varies according to the amplitude of the analog signal: at low analog voltages, the pulses are narrow; at the higher amplitudes, the pulses get wider as shown in fig.a. The instantaneous power of the transmitter varies with width of pulses. Finds application in DC motor control where speed of the motor proportional to the pulse width. In PPM, the pulses change position according to the amplitude of the analog signal. The pulses are very narrow as shown in fig.b. The instantaneous power of transmitter remains constant with width of pulses. Its disadvantages is the need for transmitter=receiver synchronization for proper operation.

THERE ARE TWO TYPES OF PULSE TIME MODULATION

(i)Pulse width modulation

(ii)Pulse position modulation

The pulse width modulation also called as pulse duration modulation. In pulse width modulation, the samples of the message signal are used to vary the duration of the individual pulses. The pulse width may be varied by varying the time of occurrence of the leading edge, the trailing edge or both edges of the pulse in accordance with the sampled value of the modulating wave. A typical PWM circuit in which the trailing edge of each pulse is varied in accordance with message signal is shown In pulse position modulation the position of a pulse relative to its unmodulated time occurrence is varied in accordance with a message signal. PPM generator consists of differentiator and a monostable multivibrator. The input to the differentiator is a PWM waveform. The differentiators generates positive and negative spikes corresponding to leading and trailing of the PWM waveform. Diode is used to bypass the positive spikes. The negative spikes are used to trigger monostable multivibrator. Pulse width Modulation (PWM) is also known as Pulse duration Modulation (PDM). Three variations of PWM are possible. In One variation, the leading edge of the pulse is held constant and change in the pulse width with signal is measured with respect to the leading edge. In other Variable, the tail edge is held in constant and w.r.t to it, the pulse width is measured in the third variation, the centre of the pulse is held constant and pulse width changes on either side of the centre of the pulse. The PWM has the disadvantage when compared to 'PDM' that its pulses are of varying width and therefore of varying power content, this means the transmitter must be powerful enough to handle the max width pulses.

7. PROCEDURE:

FOR PULSE WIDTH MODULATION AND DEMODULATION:

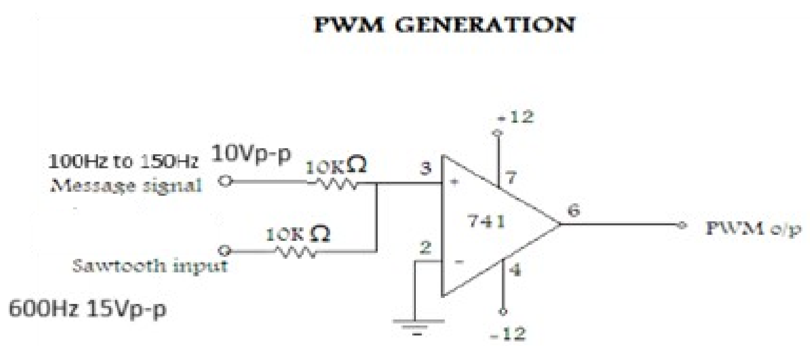
- Rig up the circuit as shown in PWM
- Apply modulating signal sine wave of 500Hz and sampling pulse –sawtooth waveform of 2KHz to pin 3 of the op amp shown
- Obtain PWM at pin 6 of op-amp. Observe the waveform at pin 3 and 6
- For PWM demodulation, apply PWM as input to the demodulation circuit shown
- Vary the capacitance to obtain undisputed message signal

FOR THE PULSE POSITION MODULATION:

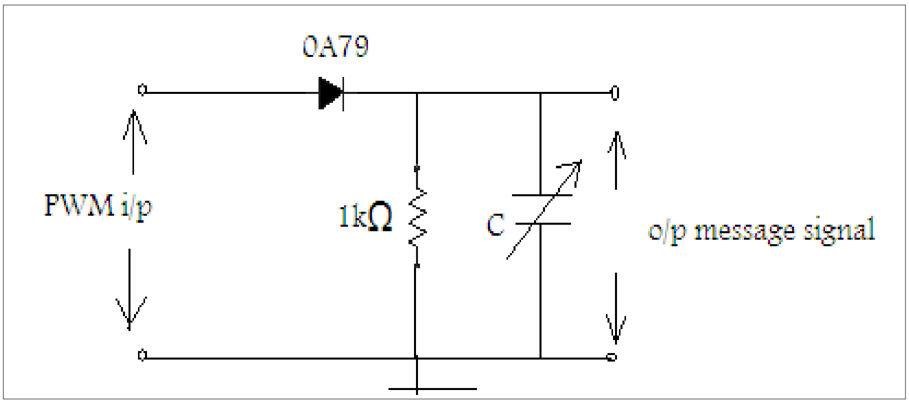
- Apply the square wave of =12V at pin2
- Output is a spike for every falling edge
- Spike should cross atleast 1/3rd of Vcc.
- Next, apply PWM output from op-amp as input to pin 2 of 555 timer
- Observe input waveform at pin 2 and PPM output at pin 3

8. CIRCUIT DIAGRAM :

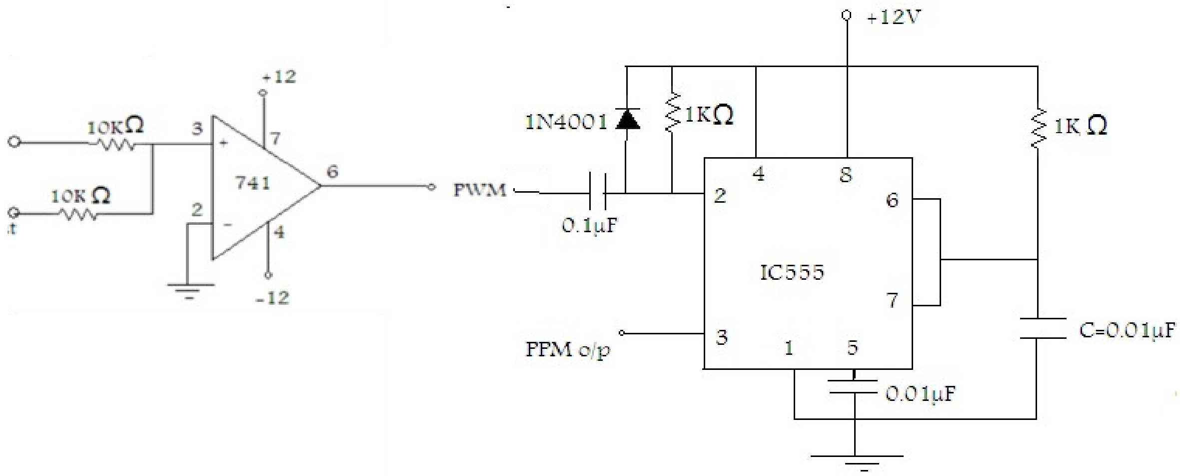
PWM GENERATION:



PWM DEMODULATION:



PPM GENERATION:



9. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

10. FORMULA / CALCULATIONS:

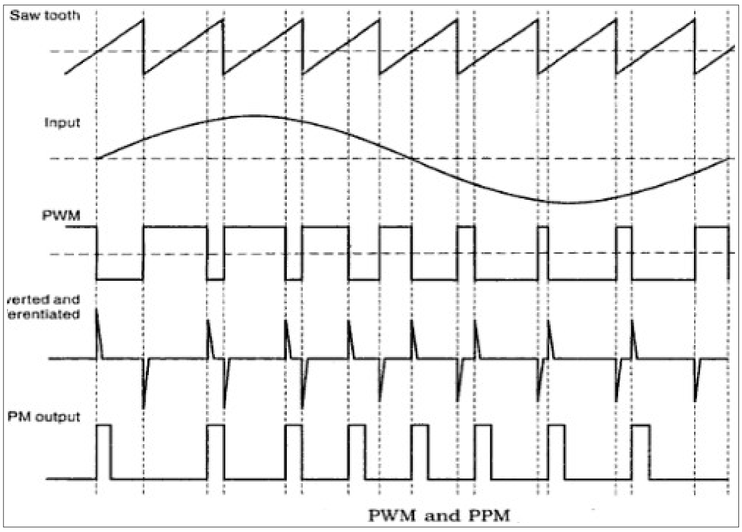
Choose $Ton = 0.01\text{msec}$

$T = 1.1\text{RC}$

Let C = 0.01µF

Hence , R= 1KΩ

11. GRAPHS / OUTPUTS:



12. RESULTS & CONCLUSIONS:
-

13. LEARNING OUTCOMES :
- PWM and PPM output waveforms are verified.

14. APPLICATION AREAS:
- Used in efficient voltage regulators
- Used in sound (music) synthesis
- Telecommunication

15. REMARKS:
-

1. EXPERIMENT NO: 10
2. TITLE: PLL Frequency Synthesizer
3. LEARNING OBJECTIVES:
   - To synchronize the output oscillator signal with a reference signal.
4. AIM:To study the operation of Frequency Synthesizer using PLL LM565.
5. MATERIAL / EQUIPMENT REQUIRED:
   - Frequency Synthesizer Kit, CRO, Connecting wires.
6. THEORY :
   - Most oscillators can be made variable in frequency, but they do not have good frequency stability. A crystal oscillator has exceptional frequency stability but the frequency cannot be varied significantly. Various techniques can be used to create other frequencies from crystal oscillators and maintain the same frequency stability as the crystal. Among these are frequency multiplication, mixing (or heterodyning) and the use of phase locked loops in combination with frequency dividers.

Operation of Frequency Synthesizer kit.

In this experiment an integrated circuit phase locked loop and frequency dividers will be used to demonstrate the idea of frequency synthesis. Although the function generator
Will be used as the source rather than a stable crystal oscillator, the principle of frequency
Synthesis will remain the same.
1.Shows a diagram for a basic frequency synthesizer using a phase locked loop. The 565 phase locked loop (PLL) will be used for the phase comparator, low pass filter and voltage controlled

oscillator (VCO) in the figure and the 74HC161CMOS BCD counter will be used as the divide by N programmable divider. The function generator will serve as the crystal oscillator. Connect the circuit shown on the following page and connect a jumper from pin 4 of the 565 PLL to pin 15 of the first 4029 counter shown on the left side of the drawing. The other 4029 counter (shown on the right of the drawing) will not be used in Part I of the experiment. Connect another jumper from pin 2 (Q3) of the 4029 counter to pin 5 of the 565 PLL

2.Set the function generator to a 1 V p-p, 1 kHz square wave and connect it to the Reference input of the PLL.

3.input on the 74161 determines whether the counter is binary or BCD. With pin 9 at +12V, the counter is binary and at ground it is BCD. Set the counter for binary operation and determine the expected divide ratio. Observe the output signal and determine its frequency

4.Vary the generator frequency over a narrow range while observing the output.

The output signal should follow the input frequency variations multiplied by the divide ratio of the 4029. What is the divide ratio?

Set the counter to BCD and observe that the output frequency changes to the new ratio times the input frequency.

7. FORMULA / CALCULATIONS:

- The free running frequency of the PLL is given as fr = (1.2)/(4R1C1) Hertz

The lock range of the PLL is given as fLock = (+/-){(8fr)/V} Hertz

The capture range of PLL is given as fc = (fLock/[2* 10^3*C2])^1/2

R1=10K ohm Variable

C1=220pF

V= -8Volts

C2= 1nF.

8.PROCEDURE :

- Connect the signal generator output to phase detector.
- Connect the phase detector output to the Frequency divider output F/16 or F/8 or F/4 or F/2.
- Connect the CRO to Frequency divider output as shown above diagram.
- Vary the phase detector pot to get n*F output at the frequency divider output
- And repeat the steps for other frequency of frequency divider output.

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

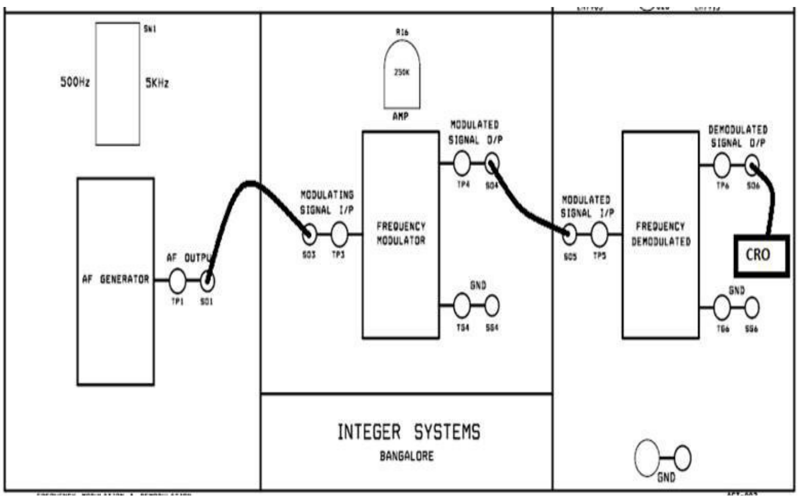


Fig 10a: PLL circuit

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:
12. RESULTS & CONCLUSIONS: • Free running frequency, f_0 = ____________Hz.
13. LEARNING OUTCOMES : • PLL to synchronize the output oscillator signal with a reference signal
14. APPLICATION AREAS: • To generate local oscillator signals to up and down convert.
15. REMARKS: • -

1. EXPERIMENT NO: 12

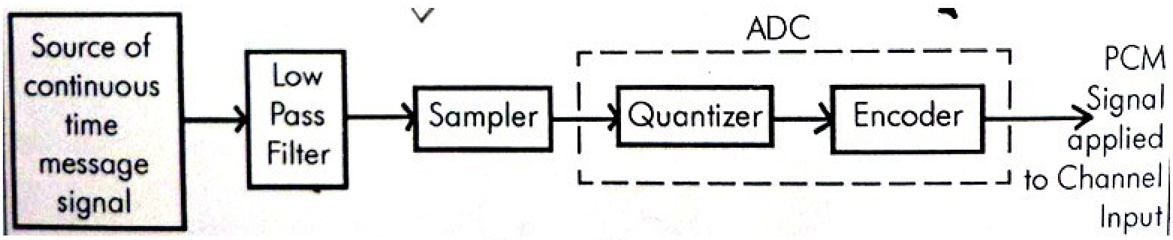
2. TITLE: PCM Multiplexer and Demultiplexer

3. LEARNING OBJECTIVES:
- To study about pulse code modulation and demodulation

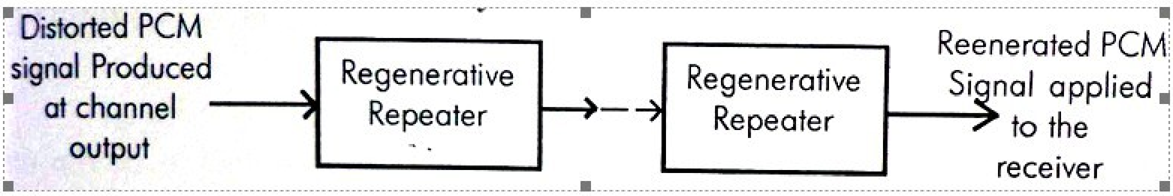
4. AIM:
- To study the operation of Frequency Synthesizer using PLL LM565.

5. MATERIAL / EQUIPMENT REQUIRED:
- Pulse code modulation and demodulation trainer
- CRO
- Connecting Wires

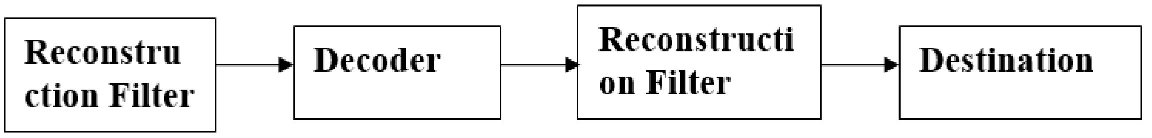
6. THEORY : In pulse code modulation (PCM) a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude. The basic elements of a PCM system are shown below.



Transmitter



(b) Transmission Path

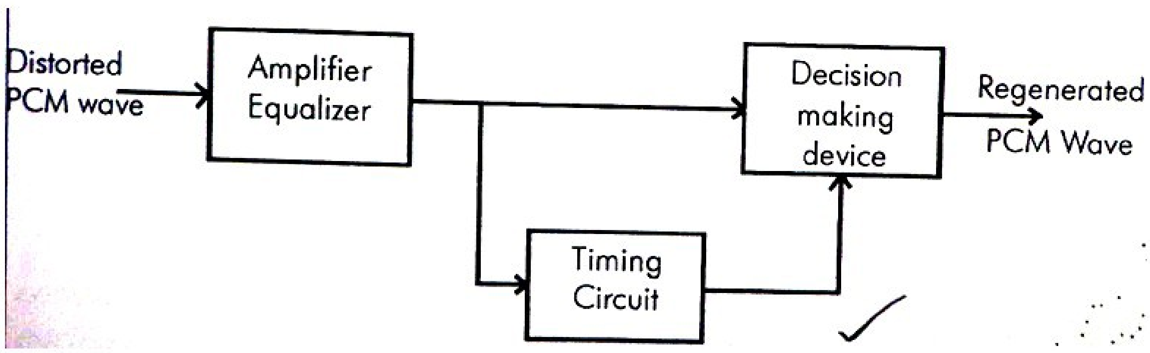


c) Receiver

The basic operations performed in the transmitter of a PCM system are sampling, quantizing and encoding. The low pass filter prior to sampling is included to prevent aliasing of the message signal. The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process. To ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than twice the highest frequency component W of the message signal in accordance with the sampling theorem.
The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter. The same circuit, which is called and analog-to-digital converter. The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude.
In combining the process of sampling and quantization, the specification of a continuous message (baseband) signal becomes limited to a discrete set of values, but not in the form best suited to transmission. To exploit the advantages of sampling and quantizing for the purpose of making the transmitted signal more robust to noise, interference and other channel impairments, we require the use of an encoding process to translate the discrete set of sample values to a more appropriate form of signal.

Regeneration:

The most important feature of PCM system lies in the ability to control the effects of distortion and noise produced by transmitting a PCM signal through a channel. This capability is accomplished by reconstructing the PCM signal by means of a chain of regenerative repeaters located at sufficiently closed spacing along the transmission route. As illustrated in figure below three basic functions are performed by a regenerative repeater: equalization timing and decision making.



7. FORMULA / CALCULATIONS:

8. PROCEDURE :

- Switch ON Pulse Code Modulation and Demodulation Trainer Kit.
- Connect the variable DC O/P to the Analog I/P of modulation section.
- Vary the VR-7 DC source and observe the PCM waveform on CRO. And also observe the LED representation of A/D output.
- Connect the AF output to Analog I/P of modulation section. Here you can vary the amplitude

and frequency of a sine wave.

- Connect the CRO probe to TP2 S/H to see the sampled signal and also TP4 to see the quantized output.
- Connect the PCM O/P to PCM I/P of demodulation section.
- Observe the DAC O/P at TP23 and also demodulated signal output at TP25 either it is an sine wave or DC source.

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

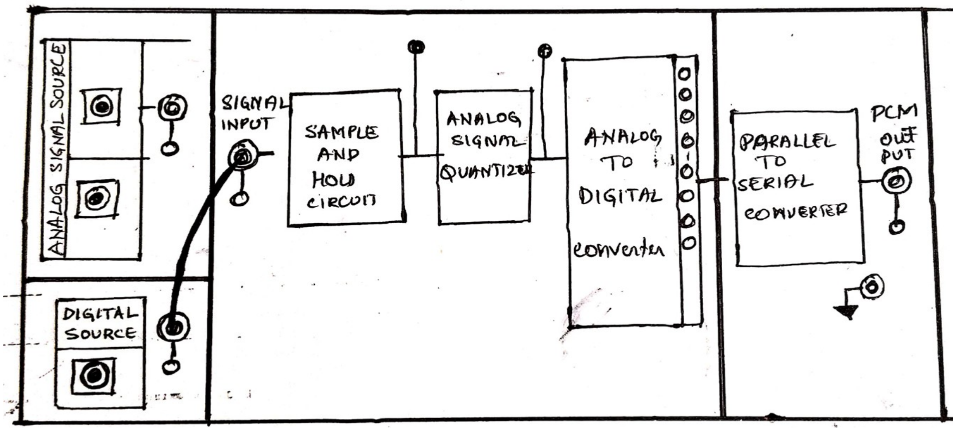


Fig 7a: Connection diagram for PCM

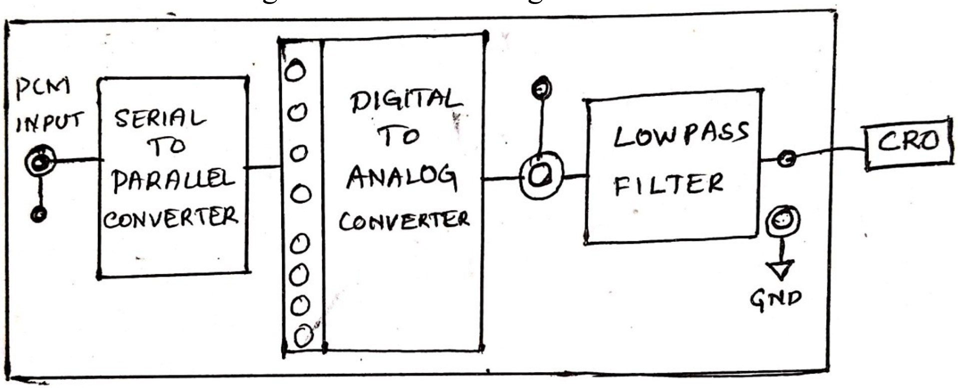
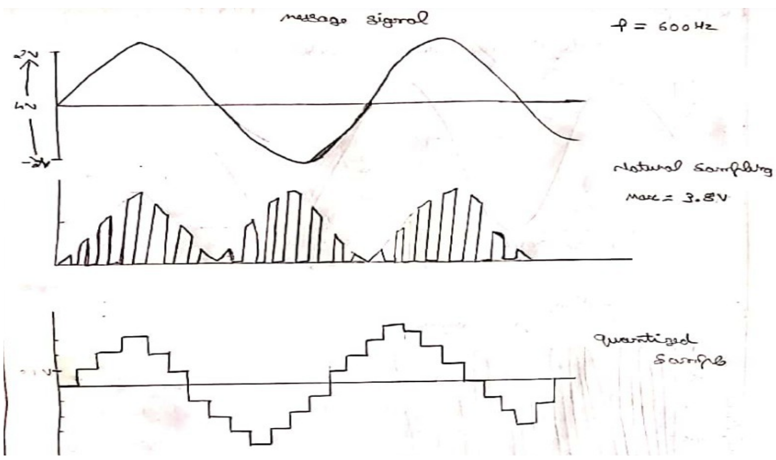


Fig 7b: Connection diagram for Demodulation

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:



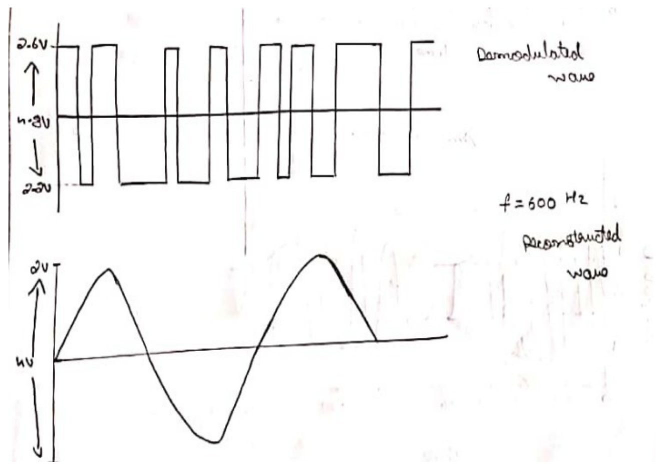


Fig 7b: PCM waveforms

12. RESULTS & CONCLUSIONS:
The Pulse Code modulation and demodulation output waveforms are observed.

13. LEARNING OUTCOMES :
- Pulse code modulation and demodulation studied.

14. APPLICATION AREAS:
- Digital telephony
- Digital audio recording
- In Compact Discs

15. REMARKS:
- -

1. EXPERIMENT NO:Open ended experiment-I

2. TITLE:**DESIGN OF ACTIVE SECOND ORDER BUTTERWORTH LOW PASS AND HIGH PASS FILTERS.**

3. LEARNING OBJECTIVES:
- To Know the concept of active Low Pass and High Pass Butterworth Filter
- To obtain the roll-off factor and cutoff frequency of the filter designed
- To compare the designed cut-off frequency with the desired cut-off frequency
- To understand the working of µA741 IC (Op Amp)

4. AIM:
- To design a Second order active low pass and High pass filter for a given cutoff frequency f c=1kHz

5. MATERIAL / EQUIPMENT REQUIRED:
- OPAMP µA741, Capacitors 0.01µF, Resistors 1kΩ , 16kΩ , 27KΩ, 560Ω , Potentiometer 22KΩ, CRO, Dual Power Supply, Function Generator

6. THEORY:
- A filter is frequency selective circuit that passes a specified band of frequencies and blocks signals of frequencies outside this band. Filters may be classified as follows:Analog & Digital, Passive & Active, Audio frequency & Radio frequency filters. Depending on the elements used in construction, filters may be classified as passive and active. Elements used in passive filters are: Resistors, Capacitors and Inductors, and in active filters Op-amps or Transistors are used along with resistors and capacitors.

- There are many advantages of using active filters:
  1. Since op-amp is used in the construction of active filter, there is flexibility in gain and frequency adjustment.
  2. Since Op-amp input impedance is infinite, no loading problem on filter characteristics.
  3. This is applicable over a wide range of frequency.
  4. This is cheaper in cost wise since big inductors are not used.
- A low pass filters is the filter that passes low frequency signals but attenuates signals with higher frequencies than the cut-off frequency. It is a Second order low pass filter which means that noise above a certain preset cut-off frequency is weakened by 40db/decade. Op-amp stage is unity gain amplifier. They are often rated for general audio/video, automotive, avionics, commercial, computers and many industrial, medical and military applications. Circuit for the LPF uses two stage passive RC filters connect to the input of Non-inverting opamp. The frequency response of op-amp will be same as that of passive RC filter, except that the amplitude of output signal is increased by pass band voltage gain of the amplifier and non-inverting amplifier as this is given as 1+Rf /R1 .For non-inverting amplifier circuit the voltage gain of the filter is generally expressed in decibels and is a function of feedback (Rf) divided by corresponding input resistor (R1) value and is given by
  Voltage gain = 20 log(Vout/ Vin)= Af 1+(f /f c) where Af = pass band gain of a filter f = Frequency of input signal in Hertz High pass filters are used in applications requiring the rejection of low-frequency signals. One such application is in high-fidelity loudspeaker systems. Music contains significant energy in the frequency range from around 100Hz to 2KHz, but high-frequency drivers can be damaged if low frequency audio signals of sufficient energy appear at their input terminals. A high-pass filter between the broadband audio signal and the tweeter in conjunction with a LPF for the low frequency driver. Active HPF allows to pass frequencies above the lower cutoff frequency fc. High pass filter are often formed simply by interchanging frequency determining resistors and capacitors in low pass filters. The range of frequency f > f c constitute a pass band and the frequencies f < f c constitute a stop band. Since high pass filter is also second order,the theoretical value of the roll off factor is 40db/decade

7. FORMULA / CALCULATIONS:
- For a second order Butterworth filter, the gain is given by,
  Af =1+Rf /R1 And the value of Af is 1.586
- Let R1=1k , then Rf=586Ω
- Also for second order filter, the cut of frequency is given by, f H = f c = 1 /2√(R3 R2C1C2)
  Let R2=R3=R and C2=C1=C then f H = f c = 1 2π RC for f H=1kHz we get R=16kΩ for C= 0.01μF

8. PROCEDURE:
- Before wiring the circuit check all the components using multimeter and IC tester.
- Rig up the circuit as shown in the figure
- Set the signal generator amplitude (input voltage) say 1Vp-p and observe the input Vin and output Vout signals of the circuit simultaneously on CRO screen.
- By varying the frequency of the input from 100Hz to higher kHz range note the frequency of signal and corresponding output voltage across pin 6 of IC with respect to the ground.
- For Low Pass Filter:The output voltage remains constant at lower frequency range and drops its amplitude by 40db/decade after designed cutoff frequency.
- For High Pass Filter:The output voltage increases its amplitude by 40 db/decade after designed cut off frequency and remains constant at higher frequency range.
- Tabulate the reading in tabular column and plot the graph with frequency along X-axis and gain in db along Y-axis.

9. CIRCUIT DIAGRAM :

- LOW PASS FILTE

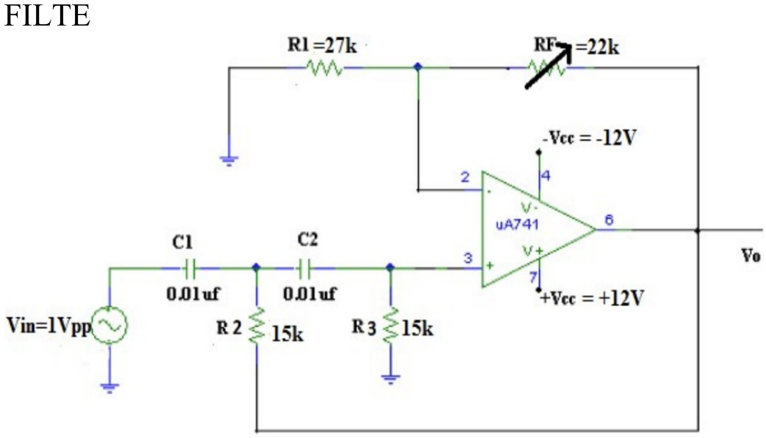


Fig1a. : Low Pass Filter

- HIGH PASS FILTER

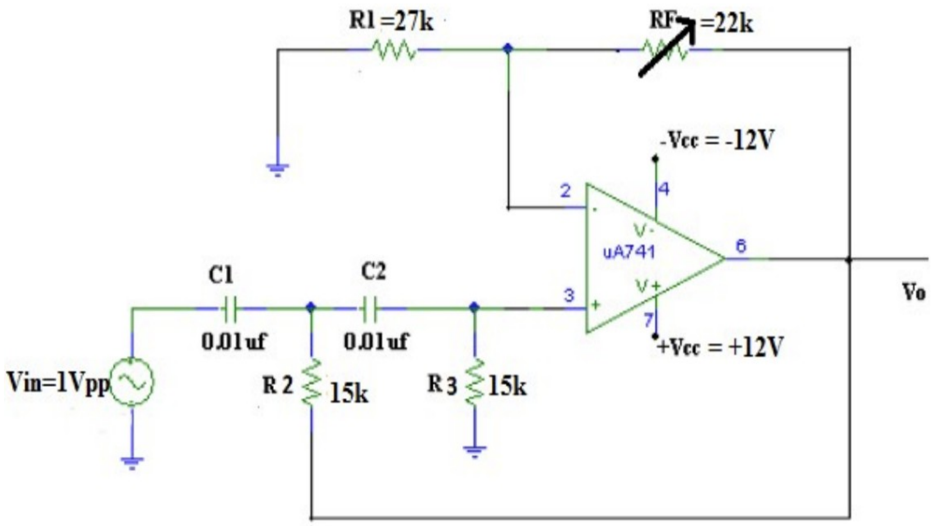


Fig1b. : High Pass Filter

10. OBSERVATION TABLE :

Vin=1Vp-p

Frequency in Hertz	Vout in Volts	G=Vout/Vin	Gain in dB=20log(G)

11. GRAPHS / OUTPUTS:

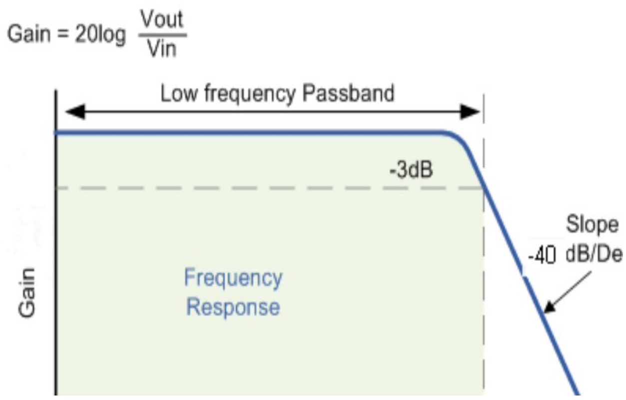


Fig1c.: Low Pass Filter graph

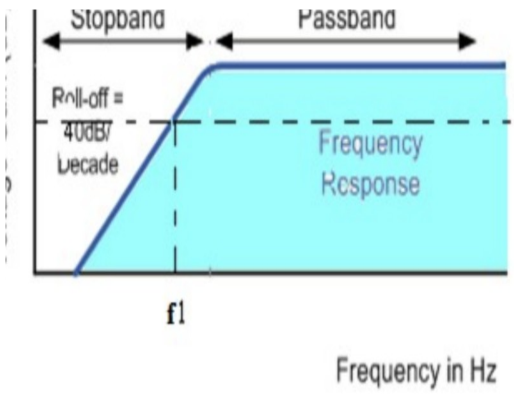


Fig1d.: High Pass Filter graph

12. RESULTS & CONCLUSIONS:

- Higher cut-off frequency , f H =____________
- Lower cut-off frequency , f L =____________

13. LEARNING OUTCOMES :

- The second order Low Pass and High pass filter circuit is designed , tested and the experimental result is obtained

14. APPLICATION AREAS:

- Used in Low frequency circuits.
- These filters are used as hiss filters in audio speakers to reduce the high frequency hiss produced in the system and these are used as inputs for sub woofers.
- Also used in equalizers and audio amplifiers.
- High pass filters are used in high frequency applications

15. REMARKS:

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COURSE LABORATORY MANUAL

1. EXPERIMENT NO: Open end experiment-II

2. TITLE: **DESIGN AND TEST TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING OF TWO BANDLIMITED SIGNALS.**

3. LEARNING OBJECTIVES:
- To demonstrate the working of TDM signals.
- To recover the two band limited signals.

4. AIM: To design and demonstrate the working of TDM and recovery of two band limited signals of TDM signals.

5.MATERIAL / EQUIPMENT REQUIRED:
- IC 4052, IC 7493 ,CRO,Signal generators,IC trainer kit,Patch chords and probes.

6. THEORY :
- TDM is a technique used for transmitting several message signals over a communication channel by dividing the time frame into slots, one slot for each message signal. This is a digital technique in which the circuit is highly modular in nature and provides reliable and efficient operation. There is no cross talk in TDM due to circuit non-linearities since the pulses are completely isolated. But it also has its disadvantages, which include timing jitter and synchronization is required.
- In pulse-amplitude modulation, the amplitude of a periodic train of pulses is varied in proportion to a message signal. TDM provides an effective method for sharing a communication channel.

7. FORMULA / CALCULATIONS:

8. PROCEDURE :

Multiplexing

- Rig up the circuit as shown in the fig 1
- Apply the square wave of frequency f_1 Hz to pin 5 and sine wave of f_2 Hz to pin 2 of 4052.
- Connect IC 7493 in trainer kit and apply logic HIGH and logic LOW to pin no 2 and 3.
- Apply 1Khz square wave in pin no 1 of IC 7493.
- In IC 7493 check counter output at pin no 8 and 11.
- Observe the output at pin 3.

Demultiplexing

- Rig up the circuit as shown in the fig 2.
- Short pin 3 and pin13 and observe the demultiplexed output at pin 14 and pin 15

9. CIRCUIT DIAGRAM :

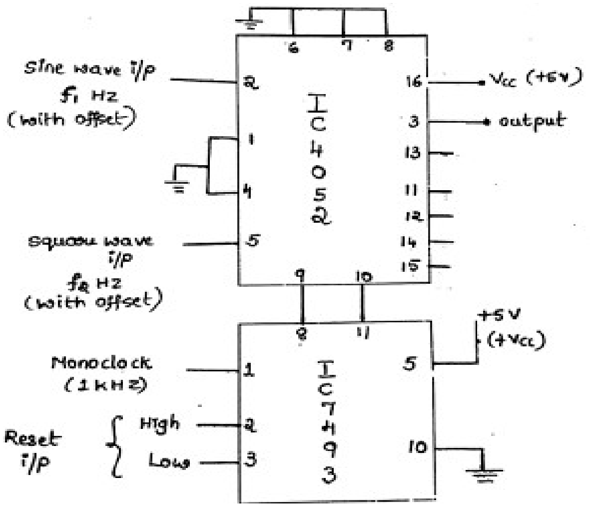

Fig.1: TDM multiplexing

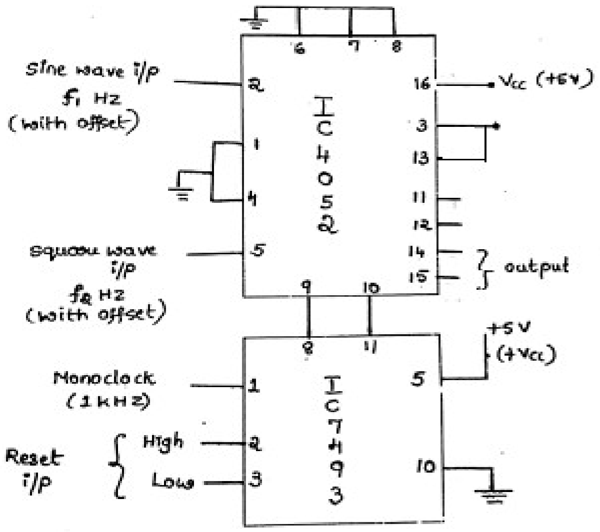

Fig. 2: TDM demultiplexing

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

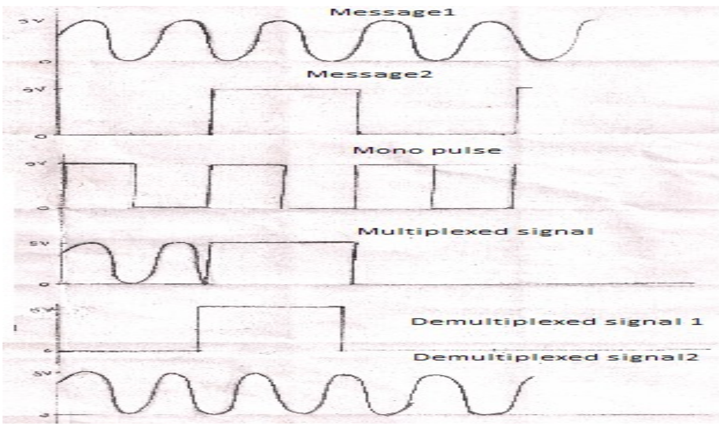

Fig 3: TDM Waveform

12. RESULTS & CONCLUSIONS:

The circuit to demonstrate the working of TDM for PAM signals was designed and the output waveforms were verified.

Multiplexing

Frequency of sine wave _______Hz
Amplitude of sine wave _______Volt
Frequency of suare wave _______Hz
Amplitude of sqare wave _______Volt

Demultiplexing

Frequency of sine wave _______Hz
Amplitude of sine wave _______Volt
Frequency of suare wave _______Hz
Amplitude of sqare wave _______Volt

13. LEARNING OUTCOMES :

The students will be able to demonstrate the working of TDM signals and to recover the two band limited signals.

14. APPLICATION AREAS:

- Communication channels.
- Telephone based application for the efficient use of channel band width.
- To transmit message of different users in same channel.
- TV cable line.

15. REMARKS:

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