

AC & DC LABORATORY MANUAL (ECE-319)

III/IV ECE SEM-I



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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING



ANIL NEERUKONDA INSTITUTE OF TECHNOLOGY & SCIENCES
(Autonomous)
Sangivalasa-531162, Bheemunipatnam Mandal, Visakhapatnam Dt.

VISION OF THE INSTITUTE

ANITS envisions to emerge as a world-class technical institution whose products represent a good blend of technological excellence and the best of human values.

MISSION OF THE INSTITUTE

To train young men and women into competent and confident engineers with excellent communication skills, to face the challenges of future technology changes, by imparting holistic technical education using the best of infrastructure, outstanding technical and teaching expertise and an exemplary work culture, besides molding them into good citizens

VISION OF THE DEPARTMENT

To become a centre of excellence in Education, research and produce high quality engineers in the field of Electronics and Communication Engineering to face the challenges of future technological changes.

MISSION OF THE DEPARTMENT

To achieve vision department will

Transform students into valuable resources for industry and society by imparting contemporary technical education.

Develop interpersonal skills and leadership qualities among students by creating an ambience of academic integrity to participate in various professional activities

Create a suitable academic environment to promote research attitude among students.

PROGRAM EDUCATIONAL OBJECTIVES (PEOS):

PEO1 : Graduates excel in their career in the domains of Electronics, Communication and Information Technology.

PEO2 : Graduates will practice professional ethics and excel in professional career through interpersonal skills and leadership qualities.

PEO3 : Graduates demonstrate passion for competence in higher education, research and participate in various professional activities.

PROGRAM OUTCOMES (POS):

Engineering Graduates will be able to:

1. **Engineering knowledge:** Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.
2. **Problem analysis:** Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
3. **Design/development of solutions:** Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.
4. **Conduct investigations of complex problems:** Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.
5. **Modern tool usage:** Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.
6. **The engineer and society:** Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **Environment and sustainability:** Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **Individual and team work:** Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
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.
11. **Project management and finance:** Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

PROGRAM SPECIFIC OUTCOMES (PSOS):

- PSO1** : Implement Signal & Image Processing techniques using modern tools.
- PSO2** : Design and analyze Communication systems using emerging techniques.
- PSO3** : Solve real time problems with expertise in Embedded Systems.

AC & DC LABORATORY [ECE 319]

COURSE OBJECTIVES:

The course objectives are to enable the students to:

1. Implement different analog and digital modulation schemes like AM, FM, ASK, PSK, and FSK.
2. Analyze Pulse analog and Pulse Digital modulation /demodulation schemes.
3. Demonstrate Super Heterodyne Radio Receiver characteristics.
4. Understand different source/channel coding techniques.
5. Simulate different blocks of analog and digital communication system.

COURSE OUTCOMES:

At the end of the course, the students will be able to:

CO1: Evaluate the performance of analog and digital modulation - demodulation techniques.

CO2: Implement analog and digital pulse modulation and demodulation methods.

CO3: Analyze the performance of Super Heterodyne Radio Receiver.

CO4: Analyze source/channel encoding & decoding methods.

CO5: Simulate functional blocks of analog and digital communication system using MATLAB.

CO-PO-PSO MAPPING:

		PO												PSO		
		1	2	3	4	5	6	7	8	9	10	11	12	1	2	3
CO	1	2		2	2					2			2		3	
	2	3		3	2					2			2		3	
	3	2		3	2					2			2		3	
	4	2		2	2					2			2		3	
	5	2		2	2	3				2			2		3	

LIST OF EXPERIMENTS		
S.No.	Name of the Experiment	CO
TRAINER KIT BASED EXPERIMENTS		
1.	Amplitude Modulation & Demodulation	CO1
2.	Frequency Modulation & Demodulation	CO1
3.	Balanced Modulator & synchronous Detector	CO1
4.	Generation and Detection of FSK.	CO1
5.	Generation and Detection of DPSK.	CO1
6.	Pulse Amplitude - Pulse Position Modulation & Demodulation	CO2
7.	Generation and Detection of Pulse Code Modulation for both A.C and D.C signals	CO2
8.	Generation and Detection of Delta Modulation	CO2
9.	Super Heterodyne Radio Receiver	CO3
10.	Linear Block code-Encoder and Decoder	CO4
11.	Convolution code-Encoder and Decoder	CO4
SIMULATION BASED EXPERIMENTS(Open source/Matlab/Multisim)		
12.	Amplitude Modulation & Demodulation	CO5
13.	Frequency Modulation & Demodulation	CO5
14.	Amplitude Shift Keying	CO5
15.	Phase Shift keying	CO5

LIST OF MAJOR EQUIPMENT

SL.NO	NAME OF THE EQUIPMENT	MAKE	QUANTITY
1.	50 MHZ DIGITAL STORAGE OSCILLOSCOPE	TEKTRONICS	12
2.	0-30V REGULATED POWER SUPPLY	ITL /FALCON/APLAB	14
3.	1MHZ FUNCTION GENERATOR	APLAB	07
4.	20 MHZ DUAL TRACE OSCILLOSCOPE	CADO	04
5.	3KVA ONLINE UPS	MEGAPOWER	01
6.	BLOCK CODE ENCODER&DECODER	SCIEN TECH	02
7.	CONVOLUTION CODE ENCODER&DECODER	SCIEN TECH	02
8.	ACL-AMPLITUDE MODULATION & DEMODULATION	AKADEMIKA LAB SOLUTIONS	02
9.	PL-DSP TRAINER KIT	AKADEMIKA LAB SOLUTIONS	02
10.	PERSONAL COMPUTER SYSTEM	HCL	06
11.	SPECTRUM ANALYZER	AGILENT TECHNOLOGIES PVT.LIMITED	01

TOTAL EXPENDITURE OF THE LABORATORY (including consumables) : Rs 27,79,409.37/-

SCHEME OF EVALUATION
[AC & DC LABORATORY]

Total marks for each student to evaluate in lab: 100 marks

Out of 100 marks:

- 1. External exam Evaluation : 50 marks**
- 2. Internal Evaluation : 50 marks**
 - i. Internal exam : 25 marks**
 - ii. Evaluation in lab : 25 marks**

S.NO .	Student Continuous evaluation Max.Marks:25M	Internal exam Max Marks: 25M	External Exam Max Marks:50M
1.	Student performance during every lab session / Observation book: 10M 1. Observation & Performance :5M 2. Viva-voce :5M	Internal exam Max Marks: 25M 1. Procedure :5M 2. Circuit/Block diagrams / MATLAB program :5M 3. Graphs/plots :5M 4. Result&Observations:5M 5. Viva-voce :5M	End Exam Marks : 50M 1. Procedure :10M 2. Circuit/Block diagrams/ MATLAB program :10M 3. Observations / Tabular forms :10M 4. Results : 5M 5. Graphs/plots :5M 6. Viva-voce :10M
2.	Record marks (Avg. of all experiments) : 10M 1. Aim, Apparatus, Theory and Procedure : 3M 2. Circuit/Block diagrams/ MATLAB program, Model Graph, Observations : 3M 3. Result :2M 4. Graphs/plots: 2M		
3.	Attendance: 5Marks Depends on percentage of LAB attendance 1. >90= 5M 2. >80-90=4M 3. >75-80=3M 4. >=66-75=2M 5. <66% =0M		

RUBRICS
(CSE LABORATORY, DC LABORATORY)

S.No	Performance Indicator	Excellent (A) 100%	Good(B) 80%	Need improvement (C) 60%	Fail (D) <40%
1.	Laboratory preparation & ability to construct the circuit diagram on a bread board and use meters/ instruments to record the measured data according to the range selected (Based on physical observation) (5M)	Read and understand the lab manual before coming to lab. Observations are completed with necessary theoretical calculations including the use of units and significant figures & Obtain the correct values of the components after calculations. Follow the given experimental procedures, to obtain the desired output.	Observations are completed with necessary theoretical Calculations but With-out proper understanding & Obtain the correct values for only few components after calculations. Follow the given experimental procedures, but obtained results with some errors.	Observations are Incomplete Obtain the incorrect values for components. Lacks the appropriate knowledge of the lab procedures. Has no idea what to do	No effort exhibited
2.	Stating clearly the aim of the experiment, its scope and importance for purpose of doing experiment & Oral Presentation (Based on viva)(5M)	Clearly describes the purpose of doing experiment and its scope. Responds confidently, and precisely in giving answers to questions correctly	Clearly describes the purpose of doing experiment. Responds in giving answers to questions but some answers are wrong.	Some idea of doing experiment but not very clear & responds in giving answers to questions but all answers are wrong.	No effort exhibited
3.	Presentation of record & Conclusions of the lab experiment performed. (Based on Lab record)(10M)	Well-organized, interesting, confident presentation of record & able to correlate the theoretical concepts with the concerned lab results with appropriate reasons.	Presentation of record acceptable	Presentation of record lacks clarity and organized	No effort exhibited

Do's

1. Be punctual and regular to the laboratory.
2. Maintain Discipline all the time and obey the instructions.
3. Read and understand how to carry out an experiment thoroughly before coming to the laboratory.
4. Check the connections properly before turning ON the circuit.
5. Turn OFF the circuit immediately if you see any component heating.
6. Dismount all the components and wires before returning the kit.
7. Report any broken plugs/apparatus or exposed electrical wires to the faculty member/laboratory technician immediately.
8. Shut down the systems properly

Dont's

1. Don't touch live electric wires.
2. Don't turn ON the circuit unless it is completed.
3. Avoid making loose connections.
4. Do not remove anything from the kits/experimental set up without permission.
5. Do not handle any equipment without reading the instructions/Instruction manuals

1. AMPLITUDE MODULATION & DEMODULATION

AIM:

To obtain the amplitude modulated signal and to calculate its Percentage modulation and to detect the AM signal.

APPARATUS:

1. AM HiQ-4101 Trainer Kit
2. Bread Board and connecting wires
3. Cathode Ray Oscilloscope and CRO Probes.
4. Function Generator
5. Regulated Power Supply

THEORY:

In AM, the amplitude of the carrier signal is varied in accordance with the amplitude of the modulating signal, whose frequency is invariably lower than the carrier. In practice the carrier may be high frequency (HF) while the modulating signal is of frequency. Formally, amplitude modulation is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating voltage.

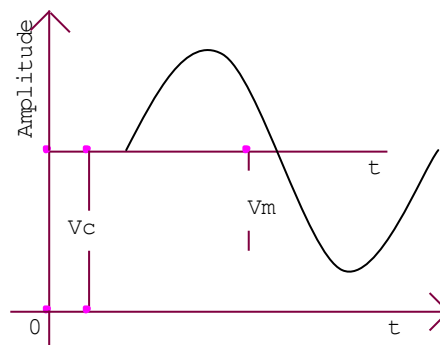
Let the carrier voltage V_c and modulating voltage V_m be represented by

$$V_c = V_c \sin \omega_c t$$

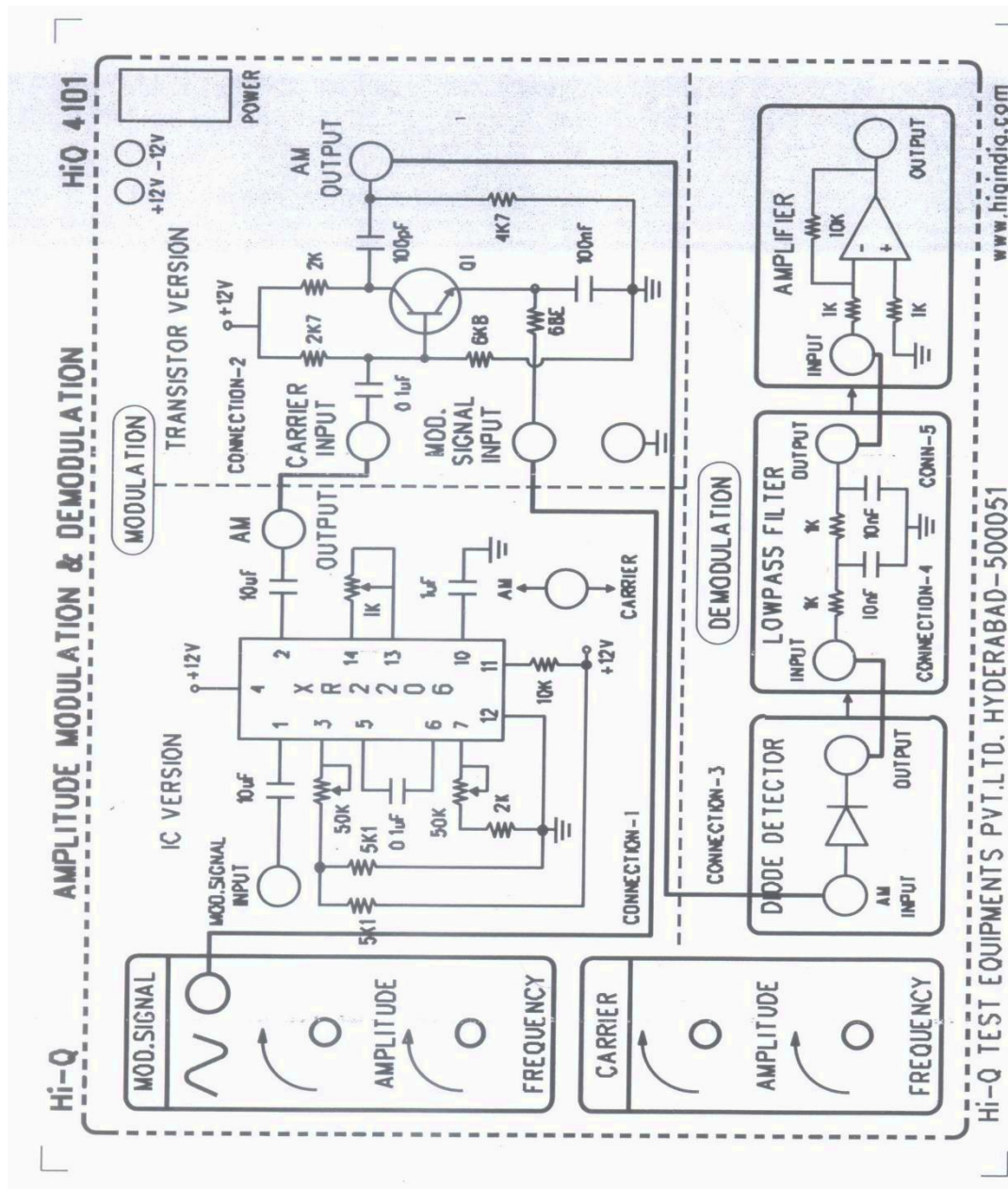
$$V_m = V_m \cos \omega_m t$$

Note that the phase angle has been ignored in both expressions since it is unchanged by the AM process. The amplitude (A) of the amplitude-modulated voltage is given by

$$\begin{aligned} A &= V_c + V_m \sin \omega_m t \\ &= V_c + V_c (V_m \sin \omega_m t) / V_c \\ &= V_c + V_c m \sin \omega_m t \\ &= V_c (1 + m \sin \omega_m t) \end{aligned}$$



CIRCUIT DIAGRAM:



Here, the ratio V_m / V_c leads to the definition of modulating index m .

$$m = V_m / V_c$$

It is a number lying between 0 and 1.

AM is shown simply as consisting of three discrete frequency of these, the central frequency i.e., the carrier, has the highest amplitude, and the other two are disposed symmetrically about it, having amplitudes which are equal to each other but which can never exceed half the carrier amplitude. The top envelop of the AM wave is given

$$A = V_c + V_m \sin \omega_m t.$$

Similarly, the maximum negative amplitude or bottom envelop is given By

$$A = - (V_c + V_m \sin \omega_m t).$$

The modulated wave extends between these two limiting envelops and has a repetition rate

equal to the unmodulated carrier frequency.

From the AM wave waveform,

$$V_m = (V_{\max} - V_{\min}) / 2$$

$$\text{And } V_c = V_{\max} - V_m$$

$$= V_{\max} - (V_{\max} - V_{\min}) / 2$$

$$= (V_{\max} + V_{\min}) / 2$$

$$\text{Since } m = V_m / V_c = (V_{\max} - V_{\min}) / 2 / (V_{\max} + V_{\min}) / 2$$

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

If $m > 1$, then it is termed 'over modulation' which is undesired.

Detection or demodulation is the process of retranslation of the spectrum to its original position. The method of recovering the original base band signal from the modulated signal is called synchronous (or) coherent (or) homodyne detection.

In an envelop detector, the o/p of the detector follows the envelop of the modulated signal. It is essentially a rectifier circuit with a capacitor across the output terminals. In the positive half cycle of input signal, the capacitor 'C' charges input to the peak voltage of the input signal. As the input signal falls below the peak value, diode is reverse biased because the capacitor voltage (peak voltage) is greater than the input signal voltage, thus causing the diode to open. The capacitor discharges through the resistor 'R' at a slow rate. This discharging continues until the next positive cycle, at its peak, the input signal becomes greater than the capacitor voltage at the peak value and diode conducts and the process is repeated. The charging time of the capacitor is very small compared to the carrier period i.e.

$$RC \ll 1 / f_c.$$

The discharging time of the capacitor i.e., RLC should be large enough to ensure that capacitor discharges slowly through R_L between the positive peaks of the carrier.

PROCEDURE:

1. Modulation

1. Connect the modulating signal to modulating input terminal of XR2206 circuit
2. Set the carrier switch to 'carrier' or 'AM' to observe carrier or modulated output.
3. Observe the output waveform at the output terminals on the CRO.
4. Note down the different frequency components and observe different conditions of modulation index.

2. De Modulation

1. Connect the modulated output to the diode detector input..
2. Observe the output of LPF(demodulated signal) on the CRO (Sinusoidal waveform)
3. Measure its frequency.

TABULAR FORM:

1. Modulation:

S.NO	Carrier Signal V_C (V)	Modulating Signal V_m (V)	Carrier Frequency f_c (Hz)	Modulating Signal V_{\max} (V)	Modulating Signal V_{\min} (V)	% Modulation Index

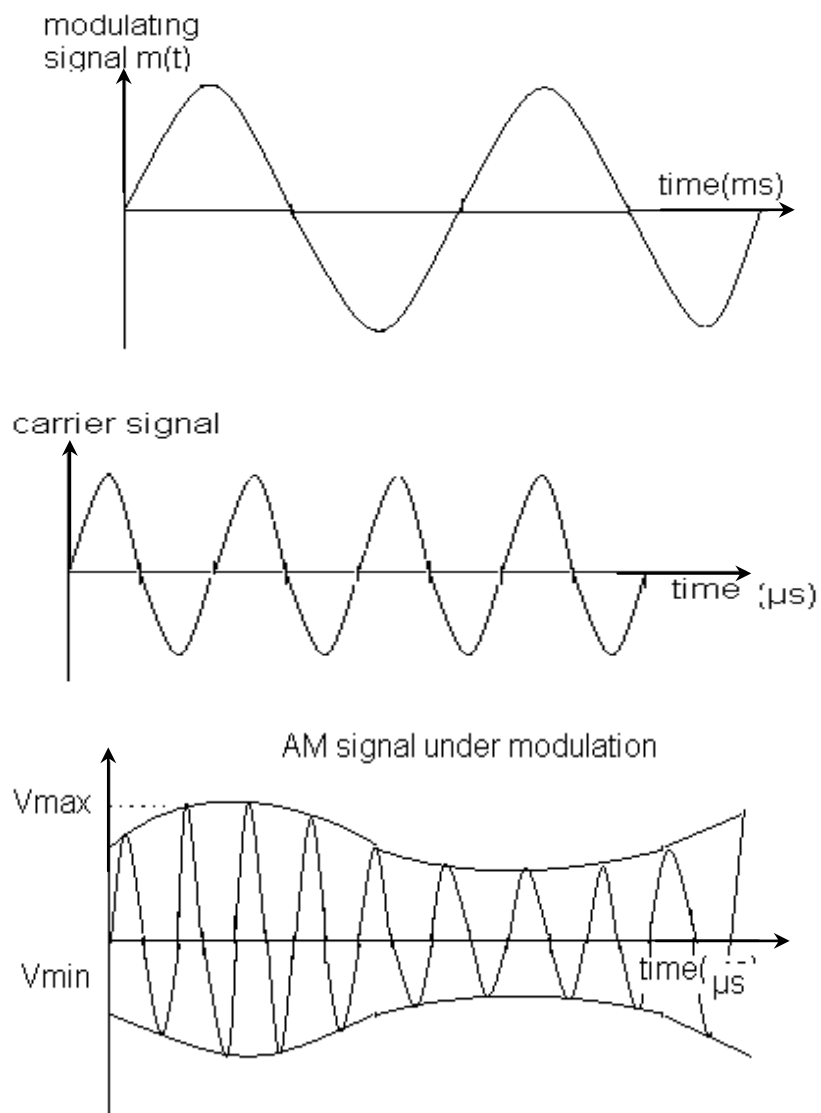
2. De Modulation:

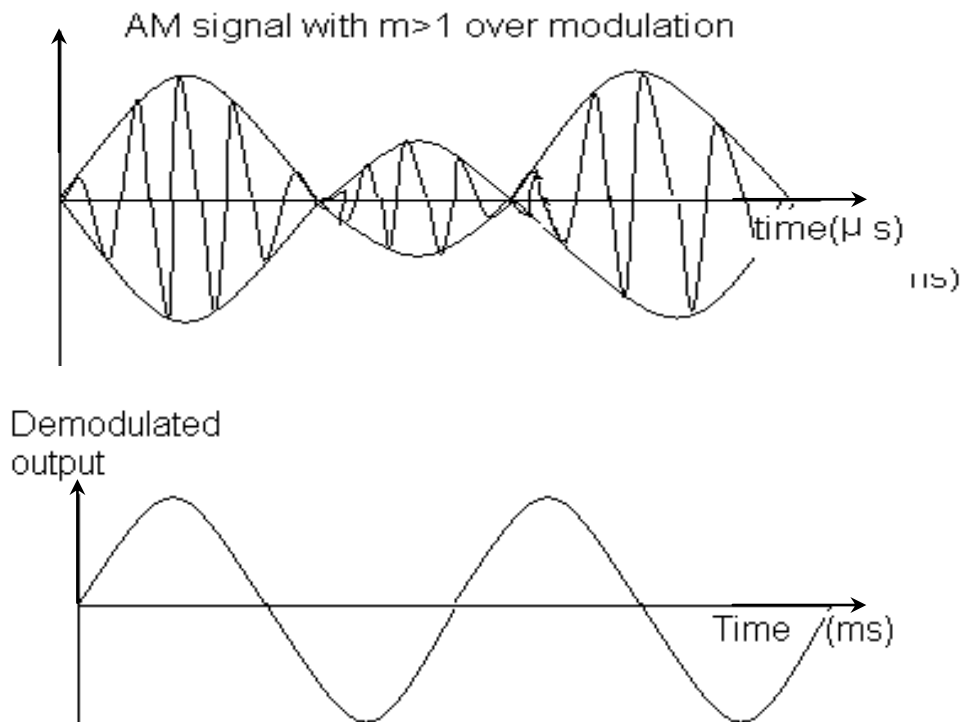
S.NO	Carrier Signal V_c (V)	Modulating Signal V_m (V)	Modulating Frequency f_m (Hz)	Modulated Signal f_d (Hz)

GRAPH:

1. Plot amplitude modulated waveform on a graph sheet.
2. Plot the % modulation versus modulating voltage.
3. The graph is a linear variation.

Model Graph:





PRECAUTIONS:

1. Note down the readings without any parallax error.
2. Avoid loose and wrong connections.
3. Always the carrier frequency must be higher than the modulating signal i.e., $f_c > f_m$
4. The carrier frequency f_c is a fixed value

RESULT:

The amplitude-modulated signal is obtained and its corresponding percentage modulation for over modulated, under modulated and perfect modulated signals are calculated.

APPLICATIONS:

AM is used in TV, Radio & RADAR Broadcasting.

VIVA VOCE:

- 1) Define modulation and need for modulation.
- 2) Define modulation index.
- 3) Define amplitude modulation and give the spectrum of AM.
- 4) What is the transmitted power in AM?
- 5) What is the transmission bandwidth of AM?
- 6) What are the requirements of envelope detector for detecting AM signal?
- 7) What are the advantages of AM?
- 8) Give the applications of AM
- 9) What is over modulation? Why it is not transmitted?

2. FREQUENCY MODULATION AND DEMODULATION

AIM:

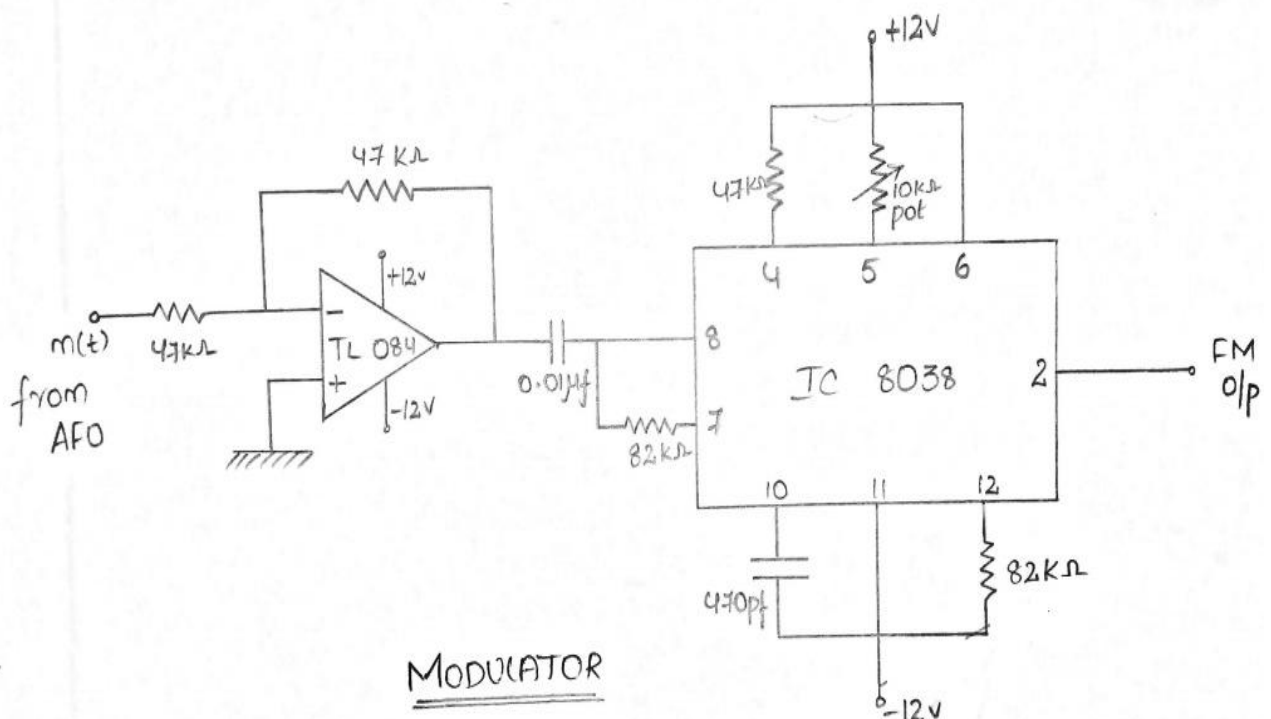
To generate a frequency modulated waveform and to calculate the frequency deviation and to detect the FM signal.

APPARATUS:

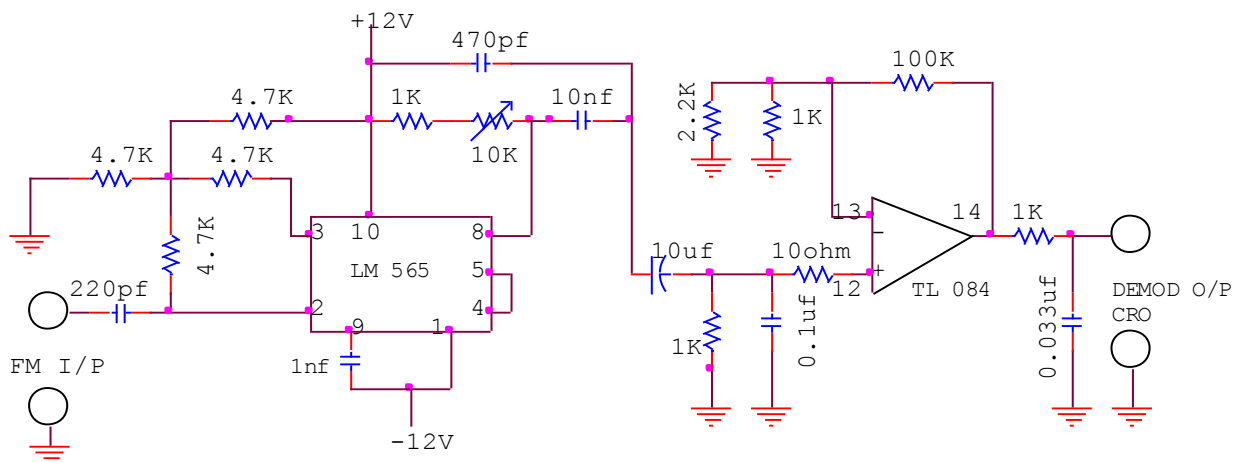
1. FM Modulation & Demodulation Kit .
2. Digital Storage Oscilloscope.
3. CRO & CRO Probes.Patch Cords.

CIRCUIT DIAGRAMS:

1.MODULATOR:



2.DEMODULATOR:



THEORY:

Frequency Modulation is a process in which the frequency of the carrier signal is varied in accordance with the amplitude of the base band signal provided that the amplitude of the carrier signal is kept constant.

$$\text{FM signal} = S_{FM}(t) = A_c \cos \left[2\pi f_c t + 2\pi k_f \int m(t).dt \right]$$

Instantaneous frequency of FM

$$f_i = f_c + k_f m(t) = f_c + k_f A_m \cos \omega_m t = f_c + \Delta f \cdot \cos \omega_m t$$

Maximum value of $f_i = f_c + \Delta f$

The maximum departure of the instantaneous frequency from the carrier frequency is called the frequency deviation (δ).

$$\Delta f = k_f \cdot A_m$$

Deviation varies with

1. Amplitude of the base band signal and
2. Also the carrier frequency.

Let modulation index, $\beta = \frac{\Delta f}{f_m}$

In AM, as the modulation index increases, the transmitted power increases while the transmitted power is independent of the modulating index in FM. Detection or demodulation is the process of retranslation of the spectrum to its original position. The method of recovering the original base band signal is called detection or demodulation of an FM signal.

PROCEDURE:

1. Turn on experimental kit and check the power supply.
2. Connect a DSO at the output.
3. With the variation of the amplitude of the input signal, the frequency deviation varies.
4. Find out f_{\max} & f_{\min} from the output waveform of DSO, then calculate Δf from the formula $\Delta f = (f_{\max} - f_{\min}) / 2$.
5. Calculate the BW of an FM signal.

For $\beta < 6$

$$\text{BW} = 2(\beta + 1) = 2(\Delta f + f_m).$$

DETECTION:

1. This FM output is given as input to the FM demodulator kit, and then the output observed is the original modulating signal.

SPECTRUM OF FM:

1. It contains a carrier and an infinite set of side bands located on either side of the carrier by frequency separations of $f_m, 2f_m, 3f_m$ & so on.
2. The side bands are symmetrical and are of equal amplitude.
3. The modulation index β determines the no. of side bands in a FM system.
4. The Bessel coefficients tend to zero as β increases and it will not directly decay to zero but the

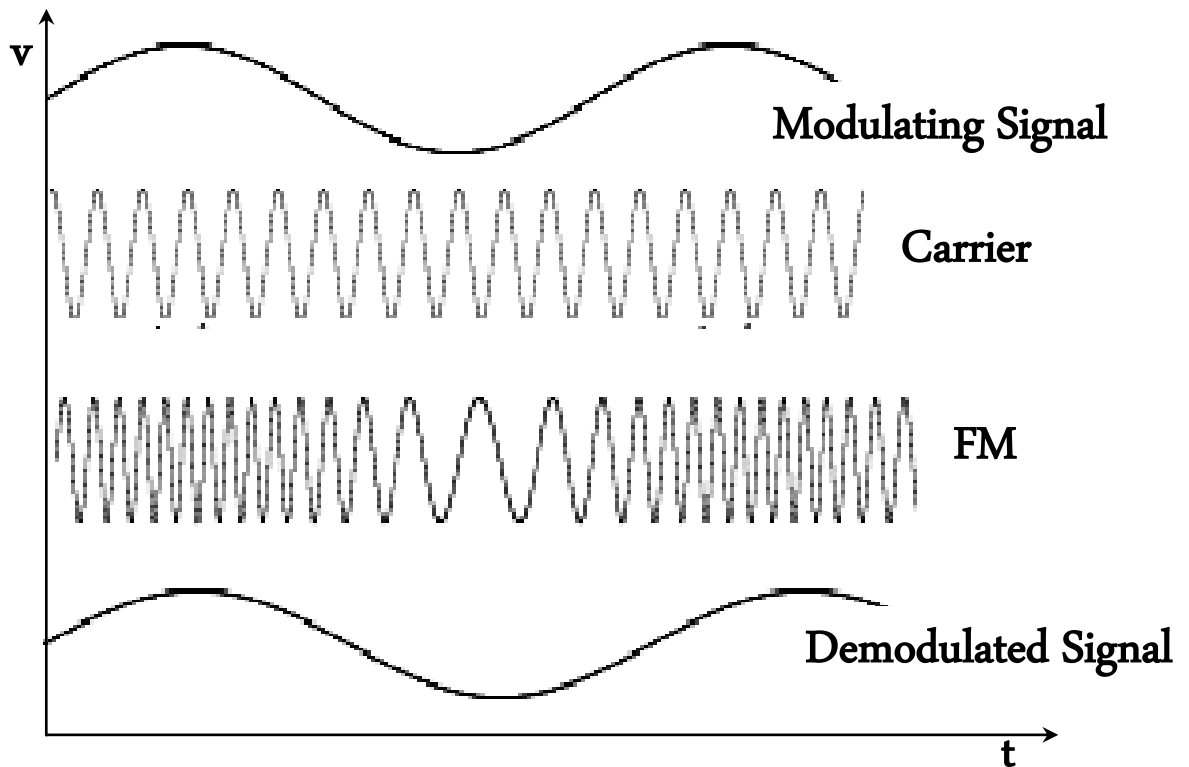
value fluctuates on either side of zero and gradually becomes zero.

TABULAR FORM:

MODULATION

S. No	T_{\min} (sec)	T_{\max} (sec)	$F_{\max} = 1/T_{\min}$ (Hz)	$F_{\min} = 1/T_{\max}$ (Hz)	Deviation $\Delta = (f_{\max} - f_{\min}) / 2$

MODEL WAVEFORMS:



DEMODULATION:

S. No.	Frequency of the base band signal f_m (Hz)	Frequency of the demodulated signal f_d (Hz)

PRECAUTIONS:

1. Wrong and loose connections should be avoided.
2. The potentiometer is varied carefully to obtain accurate readings.

RESULT:

The frequency modulated waveforms are generated and the corresponding frequency deviations and bandwidth are calculated. Also, the frequency modulated signal is detected (demodulated).

VIVA VOCE:

- 1) Define modulation and need for modulation.
- 2) Define FM.
- 3) What is frequency deviation?
- 4) Define modulation index.
- 5) How can you increase the bandwidth of FM signal
- 6) Draw the frequency spectrum of FM.
- 7) Give the advantages of FM over AM.
- 8) How the FM signal is deviated
- 9) What are the power requirements of FM.?

3. BALANCED MODULATOR & SYNCHRONOUS DETECTOR

AIM:

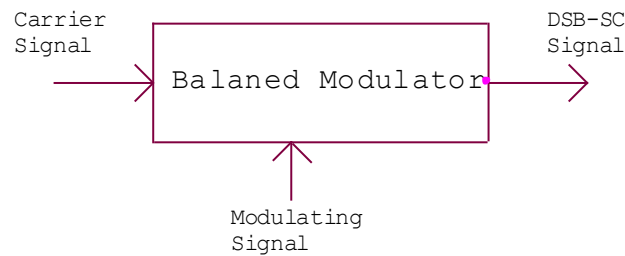
To obtain the DSB-SC (Double Side Band Suppressed Carrier) signal using a Balanced Modulator and its detection.

APPARATUS:

1. Cathode Ray Oscilloscope
2. Trainer Kit
3. Patch chords
4. CRO Probes

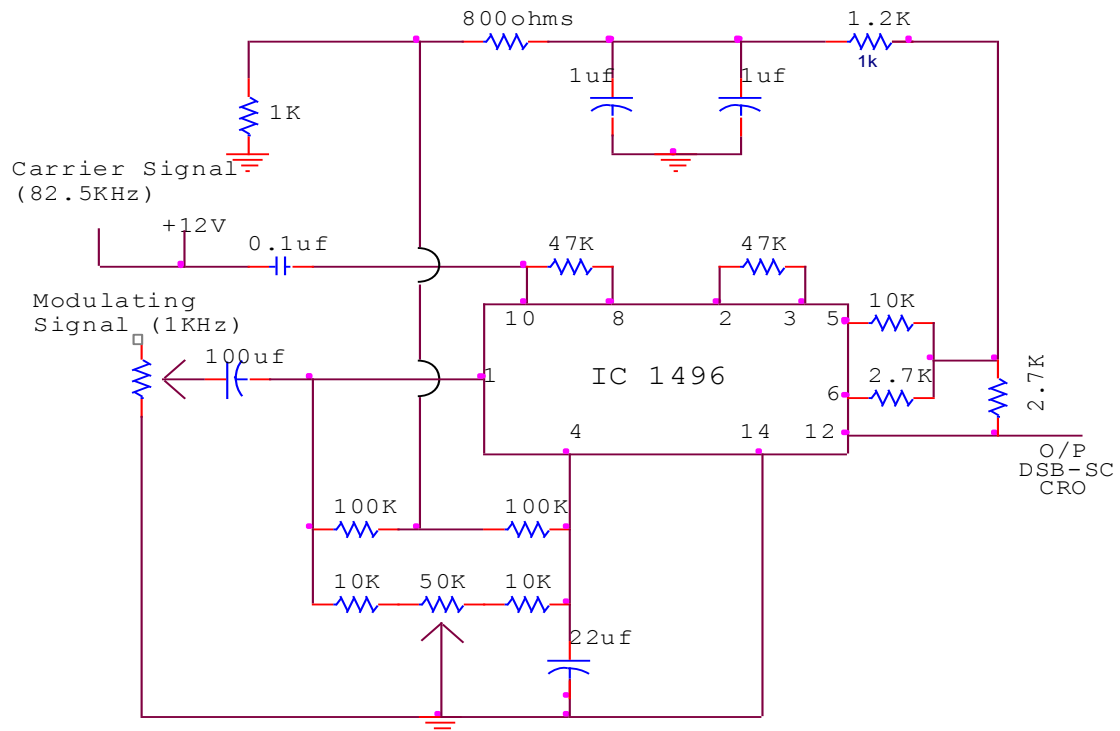
THEORY:

The balanced modulator circuit is used to suppress the carrier from the AM signal. The inputs to a balanced modulator are the carrier and a modulating signal. The outputs of a balanced modulator are the upper and lower side bands.



CIRCUIT DIAGRAM:

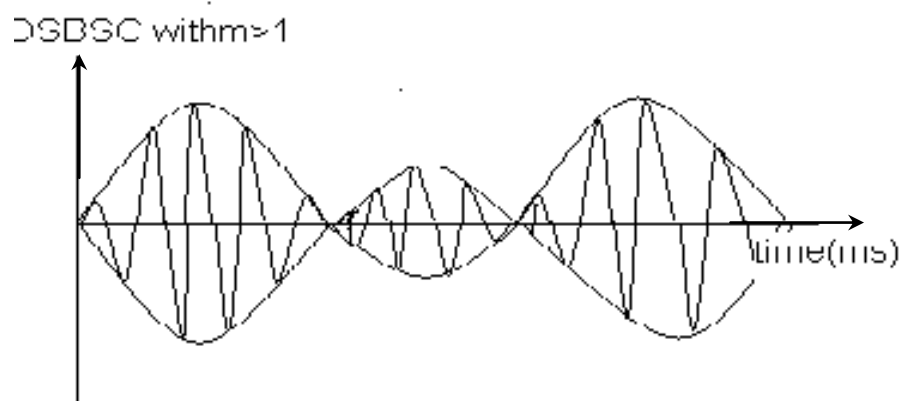
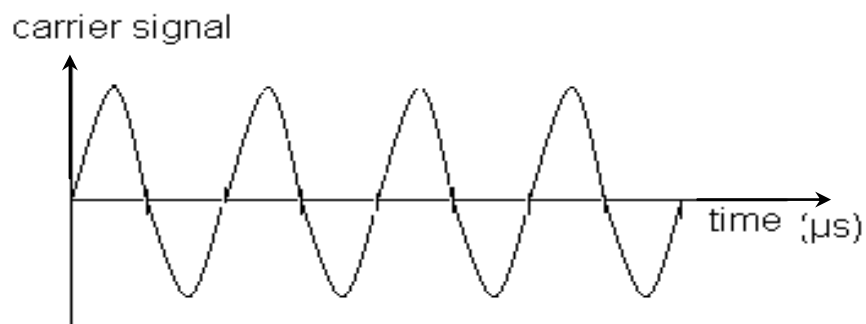
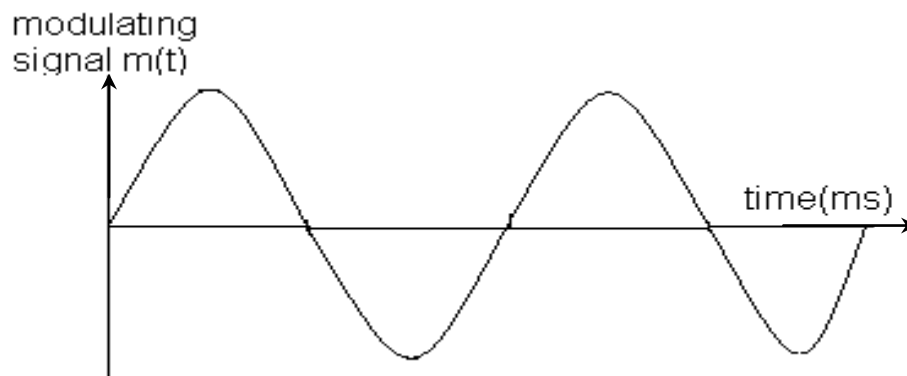
1. MODULATOR:

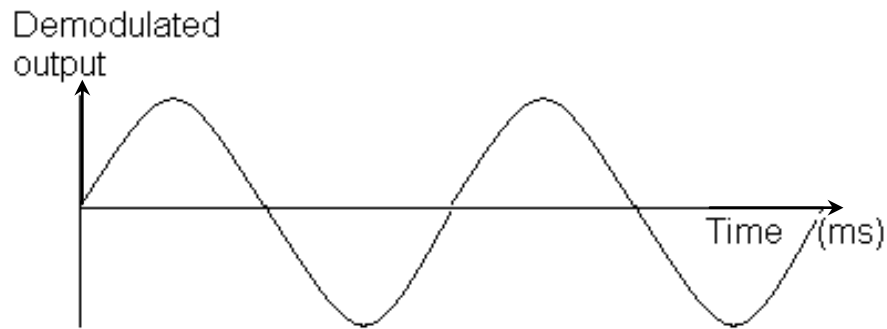


Demodulation:

F_m in hz	F_d in hz	V_d in volts

GRAPHS:





RESULT:

Thus, the DSB-SC (Double side band suppressed carrier) signal is obtained by using a balanced modulator. The original signal is demodulated from DSB-SC signal by detecting it using a synchronous (or) coherent detector.

VIVA VOCE:

- 1) What is balanced modulator?
- 2) Draw the spectrum of DSBSC.
- 3) What is the advantage of DSBSC over amplitude modulation?
- 4) What is the power saving in DSB?
- 5) How a balanced modulator generates a DSB signal?
- 6) How the DSB signal is demodulated?
- 7) What is the difference between coherent detection and non-coherent detection?

4. DIFFERENTIAL PHASE SHIFT KEYING MODULATION & DEMODULATION

AIM

To study the various steps involved in generating the differential binary signal and differential phase shift keyed signal at the modulator end and recovering the binary signal from the received DPSK signal.

INTRODUCTION

Digital communications became important with the expansion of the use of computers and data processing, and have continued to develop into a major industry providing the interconnection of computer peripherals and transmission of data between distant sites.

Phase shift keying (PSK) is a relatively a new system, in which the carrier is phase shifted by + 90 degrees for a mark, and by -90 degrees for a space.

PSK has a number of similarities to FSK in many aspects, as in FSK, frequency of the carrier is shifted according to the modulating data level.

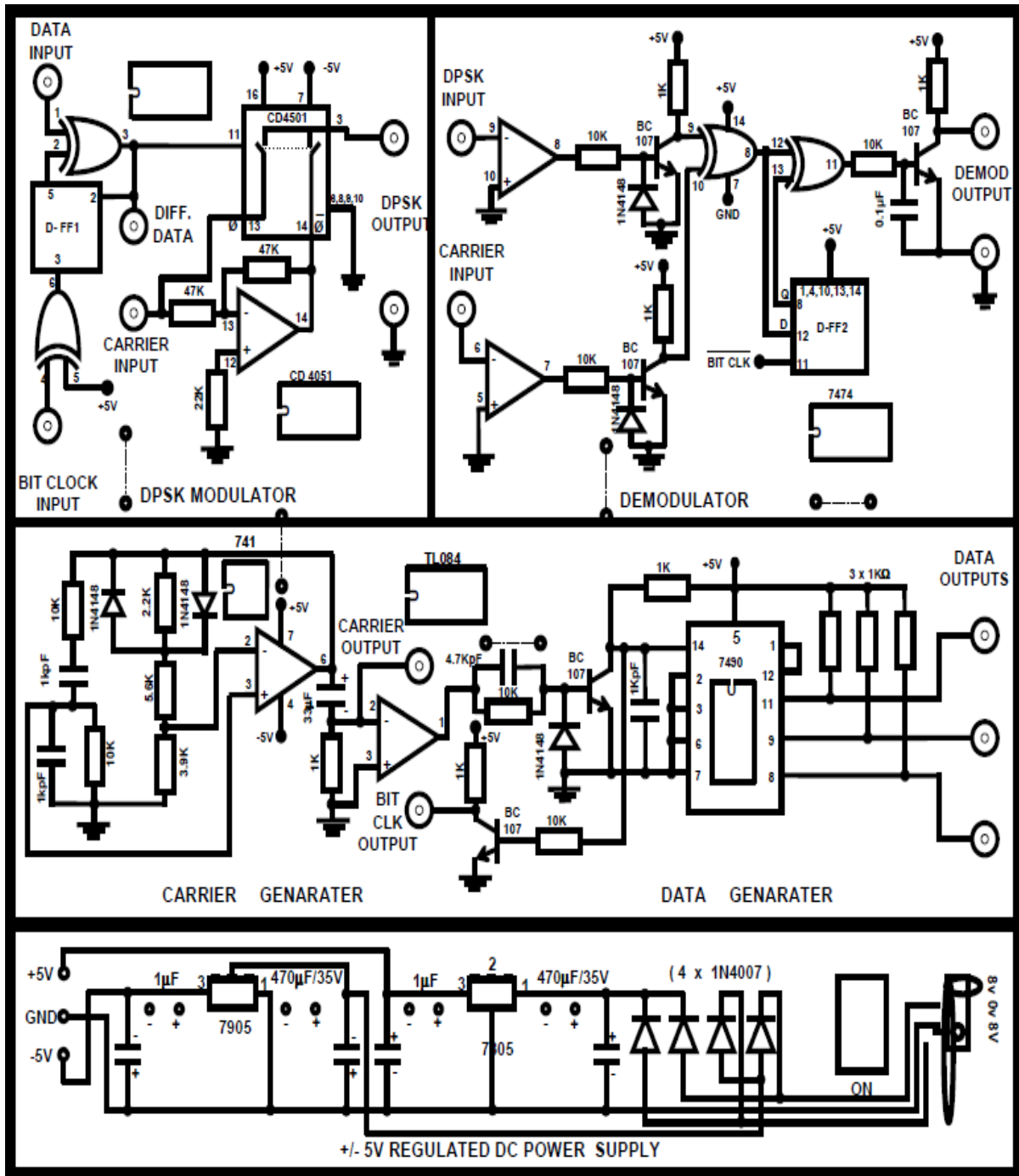
THEORY

The carrier wave signal is generated by a wein bridge oscillator around *** KHz at $\pm 5V$ P-P sine wave using 741 the sine wave is convert into square wave using TL084 in comparator mode. The Transistor BC 107 converts the square signal to TTL levels. This is used as a basic bit clock or 180° for a mark and 0° for space. This Square wave is used as a clock input to a decade counter (IC 7490) which generates the modulating data outputs.

MODULATION

The Differential signal to the modulating signal is generated using an Exclusive -OR gate (7486) and a 1-bit delay circuit using D Flip Flop 7474 (It is shown in fig-1). CD 4051 is an analog multiplexer to which carrier is applied with and without 180° degrees Phase shift (created by using an operational amplifier connected in inverting amplifier mode) to the input of the TL084. Differential signal generated by Ex-OR gate (IC7486) is given to the multiplexer's control signal input. Depending upon the level of the control signal, carrier signal applied with or without phase shift is steered to the output. 1-bit delay generation of differential signal to the input is created by using a D-flip-flop(IC7474).

Circuit diagram:



DEMODULATION

During the demodulation, the data and carrier are recovered through a TL084 op amp in comparator mode. This level is brought to TTL level using a transistor and is applied to one input of an EX-OR gate. To the second input of the gate, carrier signal is applied after conversion into a +5V signal. So the EX-OR gate output is equivalent to the differential signal of the modulating data. This differential data is applied to one input of an Exclusive -OR gate and to the second input, after 1-bit delay the same signal is given. So the output of this Ex-OR gate is the recovered Modulating signal.

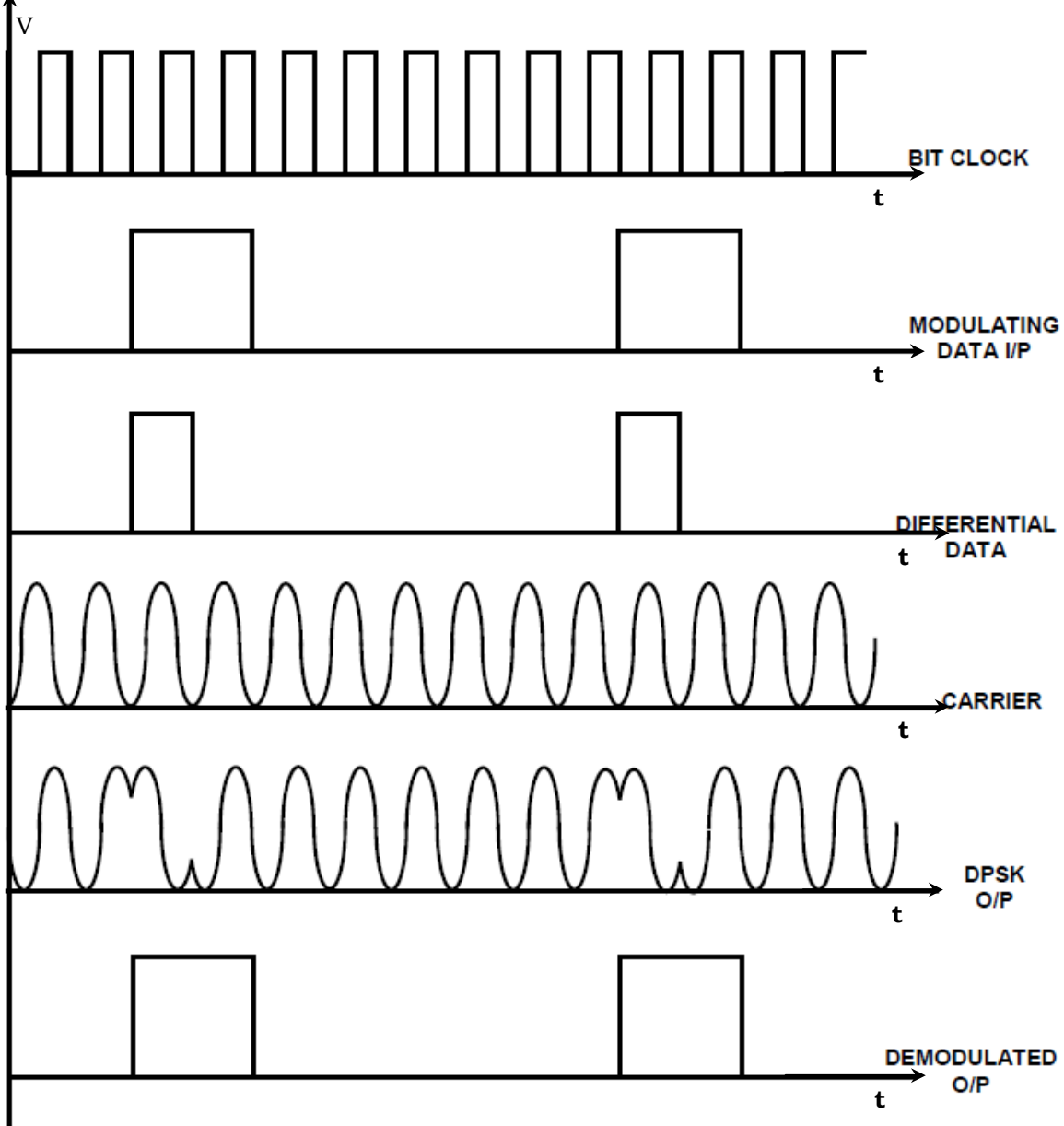
HARD WARE SPECIFICATIONS

1. Panel lay out diagram with components mounted on PCB /Panel
2. Built in variable DC power supplies $\pm 5V$, $\pm 350mA$, the figures given.
3. ICs 741, 7490, 7474, CD4051, 7486 are provided on the trainer and the hardware details are given.
4. User Manual
5. Set of patch chords

6.EXPERIMENTAL PROCEDURE

1. 'Switch ON' the experimental board.
2. Check the carrier signal and the data generator signals initially.
3. Apply the carrier signal to the carrier input of the DPSK modulator and give the data generator to the data input of DPSK modulator and bit clock output to the input of DPSK modulator and Bit Clk O/P to Bit Clk input of modulator.
4. Observe the DPSK modulating output with respect to the input data generator signal of dual trace oscilloscope (observe the DPSK modulating signal on channel 1 and the data generator signal on channel 2), and observe the DPSK Signal with respect to Differential data also.
5. Give the output of the DPSK modulator signal to the input of demodulator, give the bit clock output to the bit clock input to the demodulator and also give the carrier output to the carrier input of demodulator.
6. Observe the demodulator output with respect to data generator signal (modulating signal)

Model waveforms:



DPSK Signal Waveforms

PRECAUTIONS:

1. Avoid loose and wrong connections.
2. Readings should be noted without parallax error.

Result:

Thus the DPSK modulation and demodulation is performed and graphs were plotted.

5. FREQUENCY SHIFT KEYING

AIM

To study the generation of the Frequency Shift Keyed output and also to demodulate the FSK output with help of FUTURE TECH demonstration board.

INTRODUCTION

It would be quite possible to transmit teletype by the ordinary ON-OFF keying of the transmitter. In other words, we could use Amplitude modulation with pulses, ON corresponding to mark & OFF to space. However such a system has the inherent disadvantages that there is no real indication for the space. In addition, a system such as this would suffer from all the usual elements of Amplitude Modulation, as a result of which it is never used for automatic telegraphy (it is, of course, widely used for manual morse code CW operation). A system known as Frequency Shift Keying is generally used instead.

FSK is a system of frequency modulation, in it the nominal unmodulated carrier frequency corresponds to the mark condition, and a space is represented by a downward frequency shift. The amount was 850Hz in the original wideband FSK system designed for HF radio. For transmission by line or broad band systems, the current shift is 50Hz. This is known as narrow band FSK, or frequency modulated voice frequency telegraph (FM VFT). FSK is still often used for HF radio transmissions, with a frequency shift that is commonly 170Hz.

As with other forms of FM, the main advantage of the wide band system is greater noise immunity, while the narrow band systems are used to conserve the allocated frequency spectrum. Note that FSK may be thought of as an FM system in which the carrier frequency is midway between the mark and space frequencies, and modulation is by a square wave. In practice of course, only the fundamental frequency of the square wave is transmitted, and regeneration takes place in the receiver.

In the FSK generator, the frequency shift may be obtained by applying the varying DC output of the telegraph machine to a varactor diode in a crystal oscillator. At the receiving end, the signal is demultiplexed (if, as in common, FDM was used to send a number of telegraph or telex transmissions together) and applied to a standard phase discriminator.

From the discriminator, signals of either polarity will be available. After some pulse shaping, they are applied to the receiving teletype writer. If the telegraph transmission is by HF radio. The phase discriminator works at a (fairly low) intermediate frequency, although other methods are also possible for demodulation. An amplitude limiter is always used in the receiver, to take full advantage of the noise immunity of FSK.

THEORY

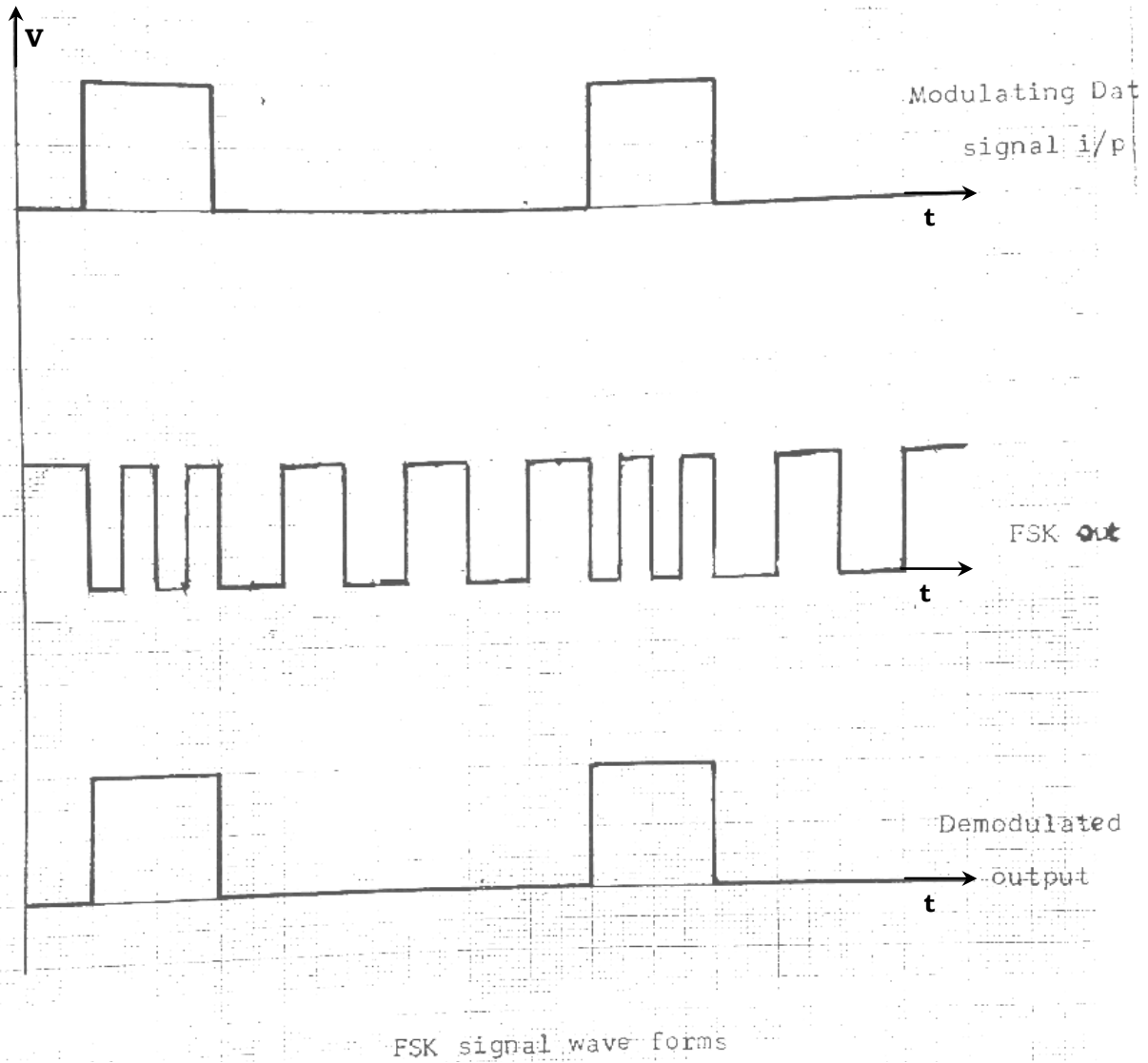
Panel diagram shows the FSK modulator and demodulator system. Basically a 555 IC is connected in Astable Multivibrator mode, generates a clock pulse of frequency determined by the values of RT and CT. This clock signal is given to a divided by 16 counter (74163 IC) which generates divided by 2,4,8 & 16 outputs of the input clock signal. In this system, divided by 2 & 8 outputs are taken as two carrier frequencies. So these are given to a FSK modulator constructed by using NAND gates. Divided by 16 output is given to a decade counter (IC 7490) which generates the modulating data signals. So depending on the level of the modulating data signal given to the FSK modulator, either divided by 2 or divided by 8 frequency outputs of the IC 74163 are transmitted to the output of the FSK modulator.

In the demodulator section, the FSK output is given to a high - Q tuned filter which is tuned to any frequency either divided by 2 or 8 of the outputs of 74163 counter. So the filter passes one frequency and stops the other frequency. This filter is constructed with operational amplifier TL084IC. The passed frequency is given to a full wave rectifier using operational amplifier TL084 IC, which rectifies and its output is given to an envelop detector which acts as a peak detector during pulse period. Then the output of the envelop detector is given to a comparator, its output is equivalent to the modulating data given at the input of the FSK modulator.

EXPERIMENTAL PROCEDURE

1. Connect the AC Adaptor to the mains and the other side to the Experimental Trainer.
2. Apply any one data output of the decade counter (7490IC) to the data input point of the FSK modulator and observe the same signal in one channel of a dual trace oscilloscope.
3. Observe the output of the FSK modulator on the second channel of the CRO.
4. During the demodulation, connect the FSK output to the input of the demodulator.
5. Adjust the potentiometers P1 & P2 until we get the demodulated output equivalent to the modulating data signal.

Model Waveforms:



PRECAUTIONS:

1. Avoid loose and wrong connections.
2. Readings should be noted without parallax error.

Result:

Thus the FSK modulation and demodulation is performed and graphs are plotted

6 (A). PULSE AMPLITUDE MODULATION & DEMODULATION

AIM:

To study the Pulse Amplitude Modulation and Demodulation circuit.

APPARATUS:

1. Pulse Amplitude Modulation Trainer kit
2. Cathode Ray Oscilloscope (CRO)
3. Patch Chords

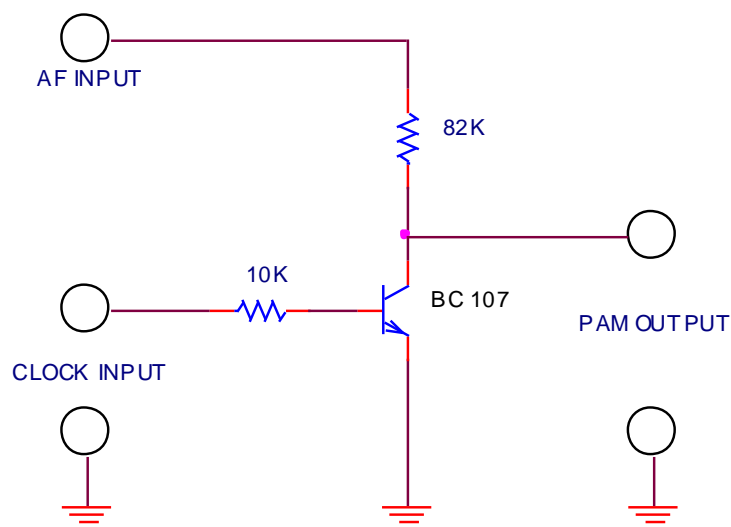
THEORY:

Pulse modulation may be used to transmit analog information, such as continuous speech or data. It is a system in which continuous waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times, together with any synchronizing pulses that may be required. At the receiving end, the original waveforms may be reconstituted from the information regarding the samples, if these are taken frequently enough.

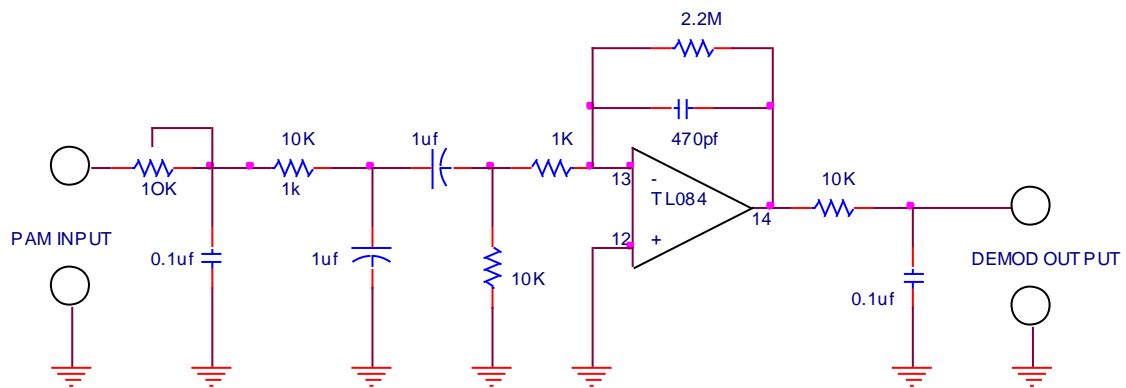
Pulse amplitude modulation, the simplest form of pulses of pulse modulation. PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier. It is very easy to generate and demodulate PAM. In a generator, the signal to be converted to PAM is fed to one input of an AND gate. Pulses at the sampling frequency are applied to the other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse-shaping network, which gives them flat tops. In the receiver, the pulses are first recovered with a standard FM demodulator. They are then fed to an ordinary diode detector, which is followed by a low pass filter. If the highest signal frequency, but low enough to remove the sampling frequency ripple, and undistorted replica of original signal is reproduced.

CIRCUIT DIAGRAM:

MODULATOR:



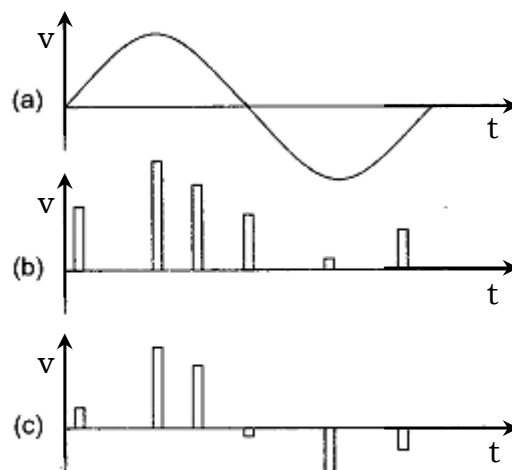
DEMODULATOR:



PROCEDURE:

1. Switch on the experimental board by connecting the power card to the AC mains.
2. Observe the modulating signal and carrier signal (Synchronous clock generator) outputs.
3. Adjust the modulating signal generator O/P to 1Vp-p amplitude.
4. Apply the modulating signal generator output and synchronous clock generator output to the PAM modulator.
5. By varying the amplitude of the modulating signal, depth of modulation varies.
6. During demodulation, connect PAM output to the input of the PAM demodulator and observe the output of PAM demodulator.

MODEL WAVEFORMS:



Pulse Amplitude Modulation (a) Signal
(b) Single Polarity PAM (c) Double Polarity PAM

RESULT:

Thus a pulse amplitude modulated signal is generated and demodulated.

6(B) PULSE POSITION MODULATION AND DEMODULATION

AIM:

To generate Pulse Position Modulation signal and its Demodulation signal.

APPARATUS:

Pulse Position Modulation and Demodulation Trainer Kit
Cathode Ray Oscilloscope
Patch Cords

THEORY:

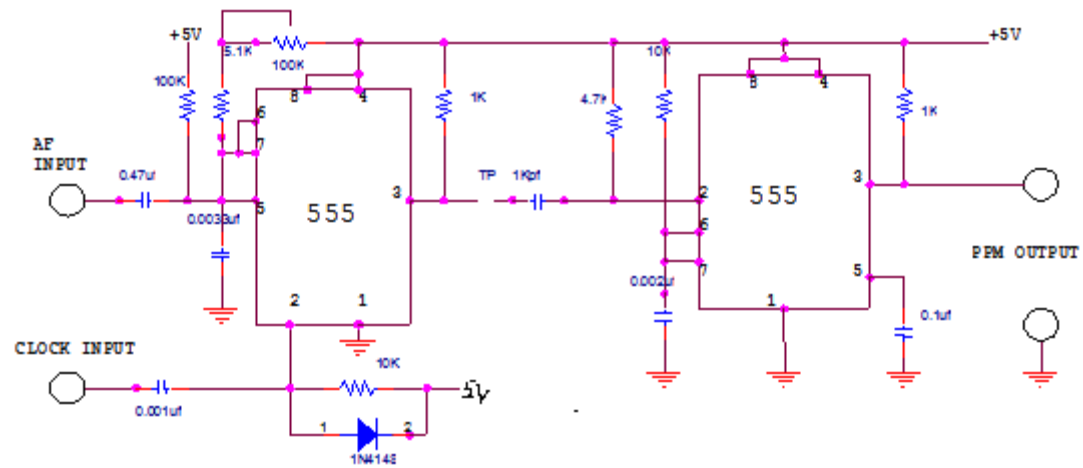
Pulse position modulation may be obtained very simply from PWM. Considering PWM and its generation again, it is seen that each such pulse has a leading edge and trailing edge (like any other pulse, of course). However, in PWM the locations of the leading edges are fixed, whereas those of the trailing edges are not. Their position depends on pulse width, which determined by the signal amplitude at that instant. Thus, it may be said that the trailing edges of PWM is pulses are, in fact, position modulated. The method of obtaining PPM from PWM is thus accomplished by “getting rid of “the leading edges and bodies of the PWM pulses.

Once again, PWM corresponding to a given signal, If the train of pulses thus obtained is differentiated, then, another pulse train results. This has positive going narrow pulses corresponding to leading edges and negative going pulses corresponding to trailing edges. If the position corresponding to the trailing edge of an unmodulated pulse is counted as zero displacement, then the other trailing edges will arrive earlier or later. An unmodulated PWM pulse is one that is obtained when the instantaneous signal value is zero. They will therefore have a time displacement other than zero; this time displacement is proportional to the instantaneous value of signal voltage. The differentiated pulses corresponding to the leading edges are removed with a diode clipper or rectifier, and the remaining pulses are position modulated.

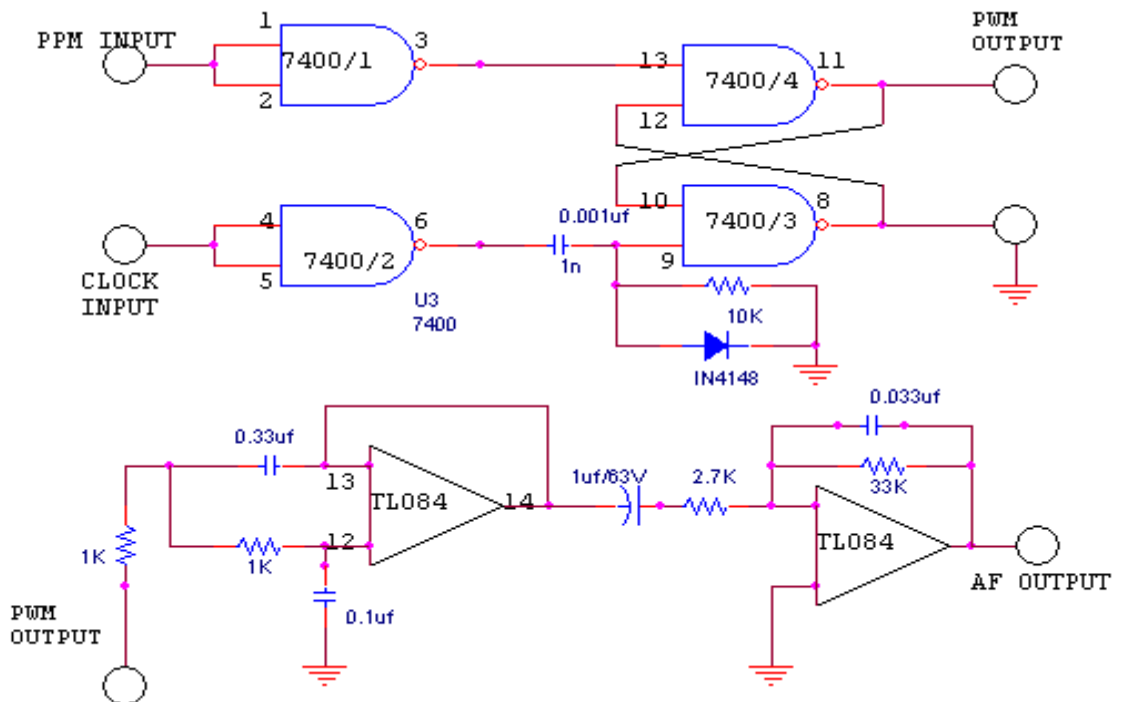
When PPM is demodulated in the receiver, it is again first converted into PWM. This is done with a flip-flop, or bistable multivibrator. One input of the multivibrator receives trigger pulses from a local generator, which is synchronized by trigger the stages of the flip-flop. The PPM pulses are fed to the other base of the flip-flop and switch that stage ON (actually by switching the other one OFF). The period of time during which this particular stage is OFF depends on the time difference between the two triggers, so that the resulting pulse has a width that depends on the time displacement of each individual PPM pulse. The resulting PWM pulse train is then demodulated.

CIRCUIT DIAGRAM:

MODULATOR:



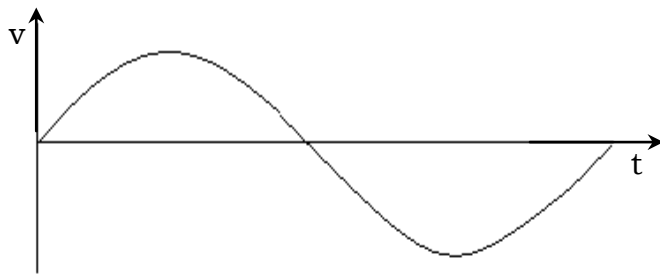
DEMODULATOR:



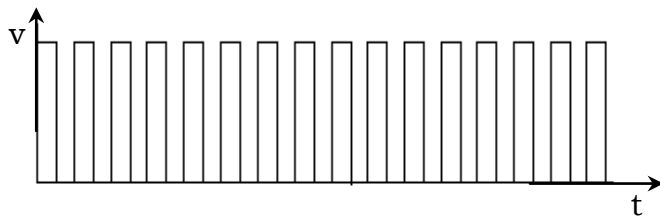
PROCEDURE:

1. Connect the trainer kit to supply and make sure the LED should glow indicating proper operation of the kit.
2. Check the output of sine wave generator provided on the board and make sure it is of 1 KHz frequency and vary in amplitude from 0V to +5V.
3. Now check the pulse generator output and make sure it is of 8 KHz.
4. The modulating signal and carrier signal are observed on the CRO and the waveforms are noted.
5. Now connect modulating signal generator and pulse generator outputs to the PPM Modulator, observe the output on CRO and noted.
6. The PPM signal is given as input to the demodulation circuit and the demodulated output is observed on the CRO and noted.

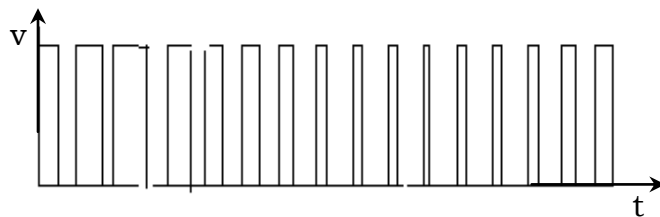
MODEL WAVEFORMS:



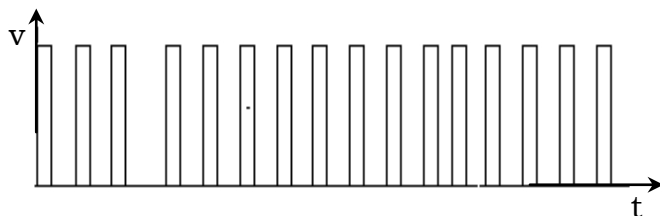
AF Signal



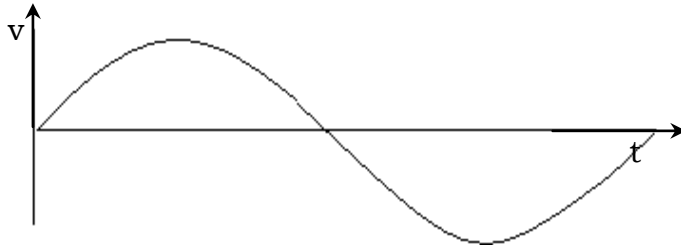
Synchronous Clock



PWM Output



PPM Output



Demodulated Output

PRECAUTIONS:

Avoid loose and wrong connections.
Readings should be noted without parallax error.

RESULT:

Thus a pulse position modulated signal is generated and it is demodulated and the original signal is obtained.

VIVA-QUESTIONS:

1. What is modulation? Explain the need for modulation.
2. What are the different types of modulation?
3. What is pulse amplitude modulation?
4. Advantages and disadvantages of PPM
5. Comparison of PPM with PPM, PWM
6. Explain in detail about all the IC's present in the Train

7.PULSE CODE MODULATION

AIM

1. To Study the Pulse Code Modulation and Demodulation.
2. To Study the effect on the variation of the Amplitude of the Modulating Signal.
3. To Study the variation of Sampling frequency on the output Demodulation signal and to prove the Nyquist Sampling Theorem.

INTRODUCTION & THEORY

In analog communication systems, the limitation is that once noise is introduced at any place along the channel, we are 'stuck up' with it. To overcome such a situation, a process known as 'Quantisation' is introduced in which the analog signal is approximated to the nearest whole number of small steps of each size S . This process of quantisation is during every sampling interval. Sampling of the Analog signal however is done at a rate keeping in view of the Nyquist criterion.

In the process of quantisation approximation is made to make the number of steps a whole number resulting in an approximation error of $\pm S/2$. This approximation noise or quantisation noise is reproduced in the demodulation and is inseparable from the original signal.

The quantised steps represented as a binary number (of Pulses) is transmitted to the Receiver. This whole process of sampling, Quantisation and conversion into a Binary code is known as Pulse Code Modulation (PCM). Thus in PCM the code represented at a particular time slot (sampling period) is representative of the instantaneous amplitude of the Analog signal.

At the receiving end, the pulses which are accompanied by noise or rounded off due to channel bandwidth limitation are reconstructed and clock frequency is derived from it using PLL techniques. The reconstructed signal retimed with the regenerated clock.

The resulting digital signal is converted into Analog form using D/A Converter and Low Pass Filter.

The PCM signal being in digital form transmission through low quality channels (with low SNR) is possible without any loss of quality.

However the band width required for PCM signals increases largely on the basis of Sampling frequency and number of clock pulses per sample.

The Digital Telephony, Digital Video etc. are a result of PCM techniques being used extensively for better quality in combating noise in transmission.

HARDWARE DESCRIPTION :

SIGNAL SOURCES

1. AF Signal Generator : 200Hz - 2KHz
2. Clock Generator
 - Conversion Clock : 68KHz - 300 KHz
 - Data Clock : 32KHz - 150 KHz
 - Byte Clock : Data Clock
 - Sampling Clock : 8.75KHz - 19KHz
3. Variable DC Source : +2.5V - -2.5V

PCM MODULATOR

4. Analog Sample and Hold.
5. 8 Bit Analog to Digital converter.
6. Parallel to Serial converter.

PCM DEMODULATOR

7. Serial to Parallel converter.
8. 8 Bit Latch.
9. 8 Bit Digital to Analog converter with Current output.
10. Current to Voltage converter
11. Low Pass Filter(3.4KHz cut off).

CIRCUIT DESCRIPTION :

1. AF SIGNAL GENERATOR

The 8038 IC is a versatile Function Generator IC which generates Sine ,Square and Triangle Waveforms. The frequency is 200Hz to 2KHz which is generated by an independent capacitor 0.01 μ F. The Sine wave Symmetry is adjustable by means of a preset 10K Ω connected between pins 4 and 5 of the IC 8038. The 100K Ω potentiometer adjusts the frequency of oscillations . The Sine wave output of 8038 at pin 2 is buffered by two Op Amps of TL084. The first Op Amp gives a unity non inverting gain while the latter gives an inverting gain of unity. The final output is taken through a 1K Ω potentiometer to adjust to required amplitude.

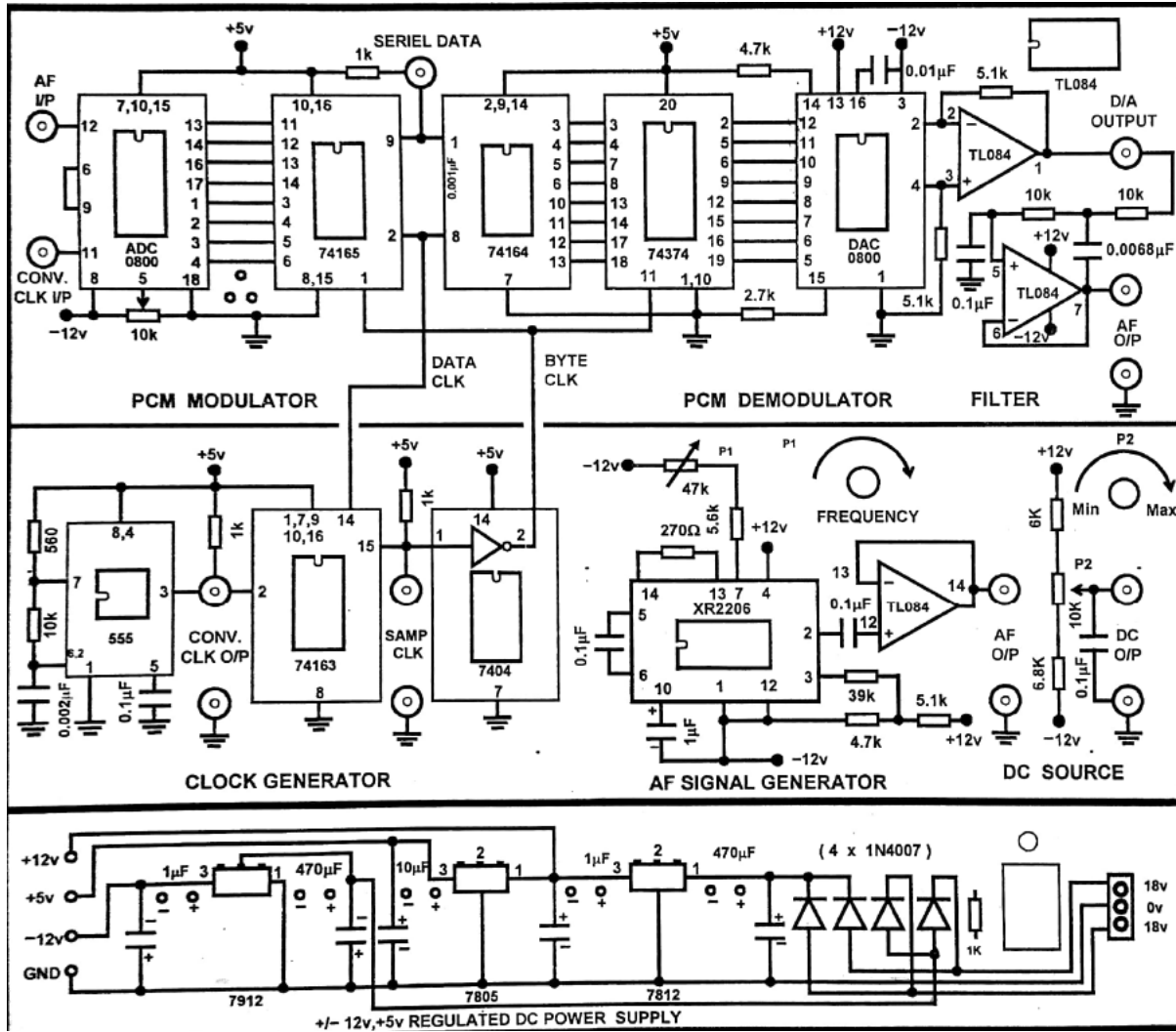
Additionally a DC signal variable from 0V to ± 2.5 V DC is provided through a potentiometer P₃ for studying steady-state conditions.

2. CLOCK GENERATOR

The circuit is designed around the popular timer IC 555 operated in Astable mode.Using a timing capacitor of 470pF and a potentiometer P₄ which is 22K Ω the required variation of 68KHz to 300KHz Square wave output is generated.

This output is divided by Flip - Flop in 74163 IC to give frequencies from 34Hz to 150KHz. This is used as a Bit clock in PCM.

Circuit Diagram:



3. SAMPLE PULSE GENERATOR

The output of Bit clock generator i.e 34KHz to 150KHz is given to a series of 3 Flip-Flops(divided by 8) to give frequencies from 8.75KHz to 19KHz.

4. ANALOG SAMPLE AND HOLD

The Analog Signal from the AF Signal Generator or Variable DC Source is buffered by a unity gain non inverting amplifier connected to the Sampling gate. The Buffer Amplifier is used to give a low output impedance so that the Hold capacitor gets charged instantaneously. CD4052 Analog Switch is used as a Sampling gate.

The gate is closed during the 'ON' time (+5V) of the Sampling Pulse and open (0V) during its 'OFF' time. During the 'OFF' time the capacitor holds the output level of the signal.

5. ANALOG TO DIGITAL CONVERTER

ADC 0800 is an 8 Bit Analog to Digital converter using successive approximation method. A high frequency Conversion Clock converts the Analog Signal into a Parallel Digital Word. For the Input range varies from -2.5V to +2.5V, the output varies monotonically from 00 to FF i.e -126 to +126 levels(i.e total 256 levels).

6. PARALLEL TO SERIAL CONVERTER

The Parallel Digital Word from the ADC is transferred by Parallel Load Clock in a Parallel to Serial Converter 74165. A Serial Shift Clock which is 8 times higher than Parallel Load Clock shifts the Bits in the 8 Bit Word serially. This is the Serial PCM Data.

7. SERIAL TO PARALLEL CONVERTER

At the Demodulating end the received Serial data is converted back into a Parallel Word using Serial to Parallel converter 74164. The same Serial Shift Clock helps in shifting the bits.

8. 8 BIT LATCH

The Parallel Word available at the Serial to Parallel Converter cited above is transferred into 8 Bit Word in a 8 Bit Latch (74374) using Parallel Load Clock. The 8Bit Word is held till the next word arrives after the next Sampling Pulse.

9. DIGITAL TO ANALOG CONVERTER

The DAC 0800 is a 8 Bit Digital to Analog Converter. The output is proportional (current form) to the Bit Code present at the 8 data input. An Op Amp is used to convert the output current to voltage. This output voltage is in the form of a stepped waveform similar to the S & H waveform except for the quantisation error.

10. LOW PASS FILTER

The stepped Analog waveform from the DAC is filtered and smoothed by a Low Pass Filter with 3.4KHz cut off frequency. This is a replica / reproduced / demodulated Analog Signal.xx

EQUIPMENT REQUIRED

1. FUTURE TECH Pulse Code Modulation and Demodulation Trainer.
2. Oscilloscope - 20 MHz , Dual Trace.
3. Set of Patch Chords (1set).
4. User Manual.

EXPERIMENTAL PROCEDURE

I . STUDY OF THE PULSE CODE MODULATION AND DEMODULATION

- 1a Connect the AC Adaptor to the mains and the other side to the Experimental Trainer. Switch 'ON' the power.
- 1b Observe the 'Analog output' signal on the Oscilloscope .Familiarize its variation of amplitude and frequency by means of potentiometers P_1 and P_2 .
2. Set the frequency about 1KHz and its amplitude to $\pm 2.5V$ (i.e 5V p-p).
3. Connect the 'Analog output' to the ' Analog input 'of the Sample and Hold circuit using a 2mm patch chord.
4. Observe the output of 555 IC timer 'Conversion Clock output ' . Observe the variation of frequency by means of Potentiometer P_4
5. Simultaneously observe the ' Parallel Load Clock output ' (I.e divided by 16) on the other channel of the Oscilloscope
6. Adjust the frequency of 'Sampling Clock' so that its is about 8KHz and connect it to Sample and Hold circuit using a 2mm patch chord.
7. Observe the Analog output Signal and Sample and Hold output on both the channels of the Oscilloscope.
8. During the 'ON' period of Sampling Pulse the 'Analog Signal' is sampled and during its 'OFF' period the Analog Signal' value is held.Vary the Analog frequency on either side of the set value and observe the S&H output.
9. Connect S&H output to the ADC 'S&H input' and the 555 IC 'Conv Clock output' to ADC 'Conv Clock input'.
The ADC 0800 successive approximation 8 Bit Analog to Digital Converter. The 8 successive pulses convert the Analog input into 8 Bits which appear in Parallel format. This can be observed on the LEDs provided after the ADC.
10. Observe the PCM output (Serial Shift Data) on CH1 and Sample Clock on CH2 of the Oscilloscope.
11. The PCM Serial output is already connected internally to PCM Demodulator input.
12. The Serial Shift Clock and Parallel Load Clock to the Demodulator are also connected internally.
13. Observe the output of the DAC of the amplifier (used as I to V converter) at pin 1 of TL 084(D/A output) . The DAC output resembles the S&H signal except that it is now in quantised form.
- 14.Observe the output of the Low Pass Filter (3.4 KHz cut off frequency) which is replica of the input Analog Signal.

2. STUDY OF THE EFFECT OF AMPLITUDE VARIATION OF MODULATING SIGNAL

1. Observe the Analog input to PCM modulator and the output of PCM Demodulator on 2 channels of the Oscilloscope.
2. Increase the Amplitude by means of Potentiometer P_2 . The Amplitude is limited to $\pm 2.5V$ p-p only. The output will be reproduced undistorted.
3. Any further increase will saturate the ADC levels and the output peaks are flattened.
4. Decrease the Amplitude. The output signal amplitude will be reduced accordingly. Since lesser levels are there in the Sine wave at lower levels, some distortion is observed even after filtering.

3. STUDY OF SAMPLING FREQUENCY ON THE OUTPUT

The Analog Signal frequency (1KHz) and Sampling frequency (8KHz) are restored using Oscilloscope before starting this part of the experiment.

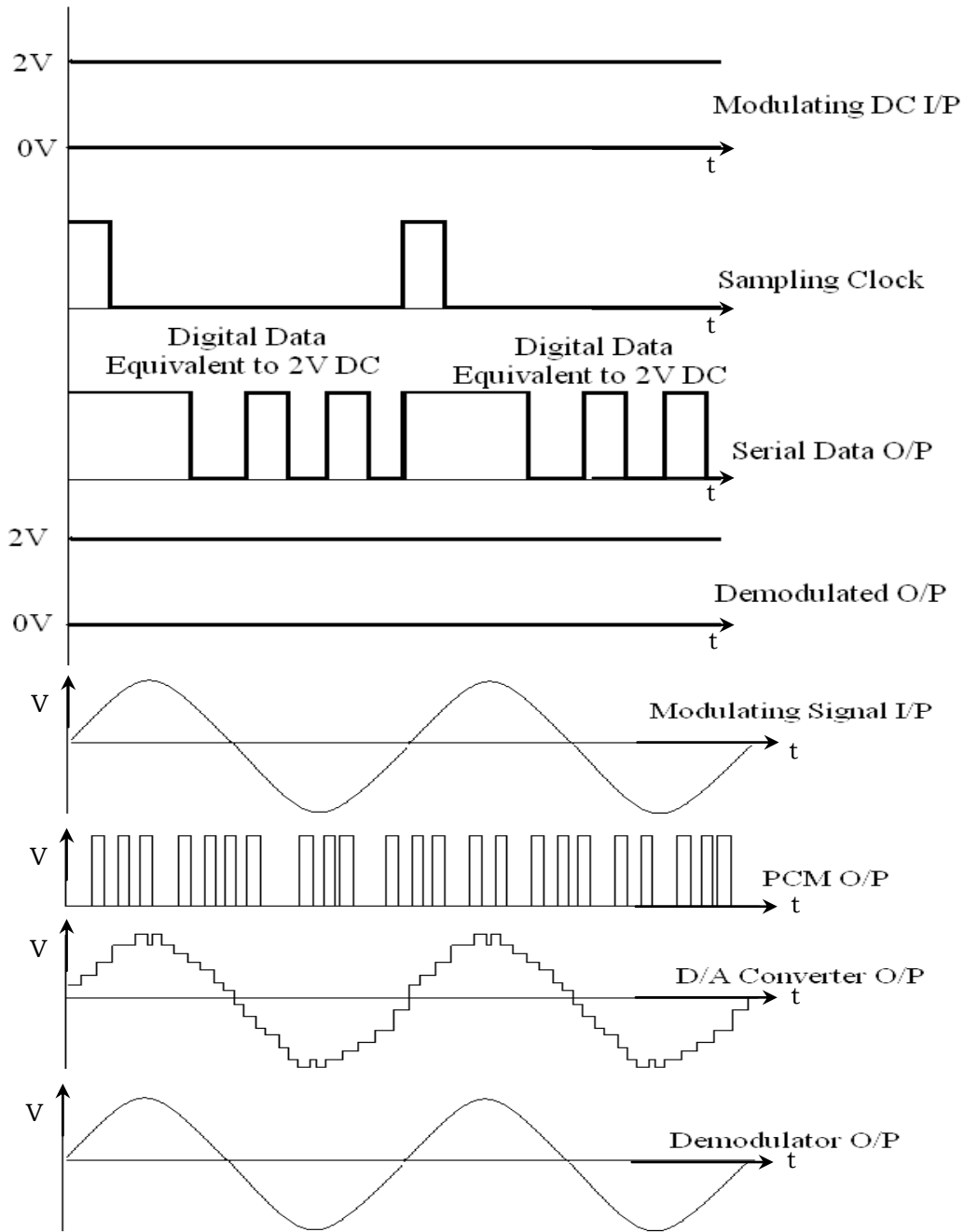
1. In each cycle of the Analog input we observe that there are about 8 Sampling Pulses. Thus the PCM Demodulated output is a true replica of input as observed on both the channels of the Oscilloscope.
2. Gradually reduce the Clock frequency using potentiometer P_4 which reduces the Sampling frequency. As the frequency decreases, the Sine wave output at one point gets so distorted that it does not resemble a Sine wave. Measure the Sampling frequency at this point. It is found that the frequency is 2KHz or even less as indicated by the Nyquist Theorem.

Usually for better reproduction, the Sampling frequency should be much more than the Nyquist Rate.

HARDWARE SPECIFICATIONS

1. Pulse Code Modulation and Demodulation Trainer Kit.
2. Built in DC power supply $\pm 12V$, $\pm 5V$ / 350mA.
3. Provided with 2mm Sockets.
4. The ICs provided on the board are TL084, 7404, 74163, 555, 8038, 74164, 74165, 74374, DAC 0800, ADC 0800, CD 4053.
5. Set of Patch chords Stackable 2mm - 10 Nos.
6. User Manual.

MODEL WAVEFORMS:



PRECAUTIONS:

1. Avoid loose and wrong connections.
2. Readings should be noted without parallax error.

RESULT:

Thus Pulse Code Modulation signal is generated for both AC and DC input signals and it is Demodulated.

8. DELTA MODULATION

AIM

To study the Delta Modulation process by comparing the present signal with the previous signal of the given Modulating signal.

INTRODUCTION

In radio transmission, it is necessary to send audio signal (e.g. Music, speech etc.) from a broadcast station over great distances to a receiver. This communication of audio signal which does not employ any wire and is sometimes called wireless. The audio signal cannot be sent directly over the air for appreciable distance. Even if the audio signal is converted into electrical signal, the latter cannot be sent very far without employing large amount of power. The energy of a wave is directly proportional to its frequency. At audio frequencies (20Hz to 20KHz), the signal power is quite small and radiation is not practicable.

The radiation of electrical energy is practicable only at high frequencies e.g. Above 20KHz. The high frequency signals can be sent thousands of miles even with comparatively small power. Therefore, if audio signal is to be transmitted properly, some means must be devised which will permit transmission to occur at high frequencies while it simultaneously allows the carrying of audio signal. This is achieved by imposing electrical audio signal on high frequency carrier/ The resultant waves are known as modulated waves or radio waves and the process is called modulation. At the radio receiver, the audio signal is extracted from the modulated wave by the process called demodulation. The signal is then amplified and reproduced into sound by the loudspeaker.

MODULATION

A high frequency carrier wave is used to carry the audio signal which is done by changing some characteristic of carrier wave in accordance with the signal. Under such conditions, the audio signal will be contained in the resultant wave. The process is called modulation and defined as "The process of changing some characteristic (e.g. Amplitude, Frequency or Phase) of a carrier wave in accordance with the intensity of the signal is known as Modulation".

Modulation means to "change". In modulation, some characteristic of a carrier wave is changed in accordance with the intensity (i.e. Amplitude) of the signal. The resultant wave is called modulated wave or radio wave and contains the audio signal. Therefore, modulation permits the transmission to occur at high frequency while it simultaneously allows the carrying of the audio signal.

NEED FOR MODULATION

Modulation is extremely necessary in communication system due to the following reasons.

1. PRACTICAL ANTENNA LENGTH : In order to transmit a wave effectively, the length of the transmitting antenna should be approximately equal to the wavelength of the wave.

$$\text{Now Wavelength} = \frac{\text{Velocity}}{\text{frequency}} = \frac{3 \times 10^8}{\text{frequency (Hz)}} \text{ meters}$$

As the audio frequencies range from 20Hz to 20KHz, therefore, if they are transmitted directly into space, the length of the transmitting antenna required would be extremely large. For instance, to radiate a frequency of 20KHz directly into space, we would need an antenna length of $3 \times 10^8 \div 20 \times 10^3 = 15,000$ meters. This is too long antenna to be constructed practically. For this reason, it is impracticable to radiate audio signal directly into space. On the other hand, if a carrier wave say of 1000KHz is used to carry the signal, we need an antenna length of 300 meters only and this size can be easily constructed.

2. OPERATING RANGE

The energy of a wave depends upon its frequency. The greater the frequency of the wave, the greater the energy possessed by it. As the audio signal frequencies are small, therefore these cannot be transmitted over large distances if radiated directly into space. The only practical solution is to modulate a high frequency carrier wave with audio signal and permit the transmission to occur at this high frequency (i.e carrier frequency).

3. WIRELESS COMMUNICATION

One desirable feature of radio transmission is that it should be carried without wires i.e. Radiated into space. At audio frequencies radiation is not practicable because the efficiency of radiation is poor. However, efficient radiation of electrical energy is possible at high frequencies. For this reason, modulation is always employed in communication systems.

INTRODUCTION

Delta Modulation is a Differential Pulse Code Modulation Technique, in which the difference signal between two successive samples is encoded into a single bit code.

Circuit Diagram:

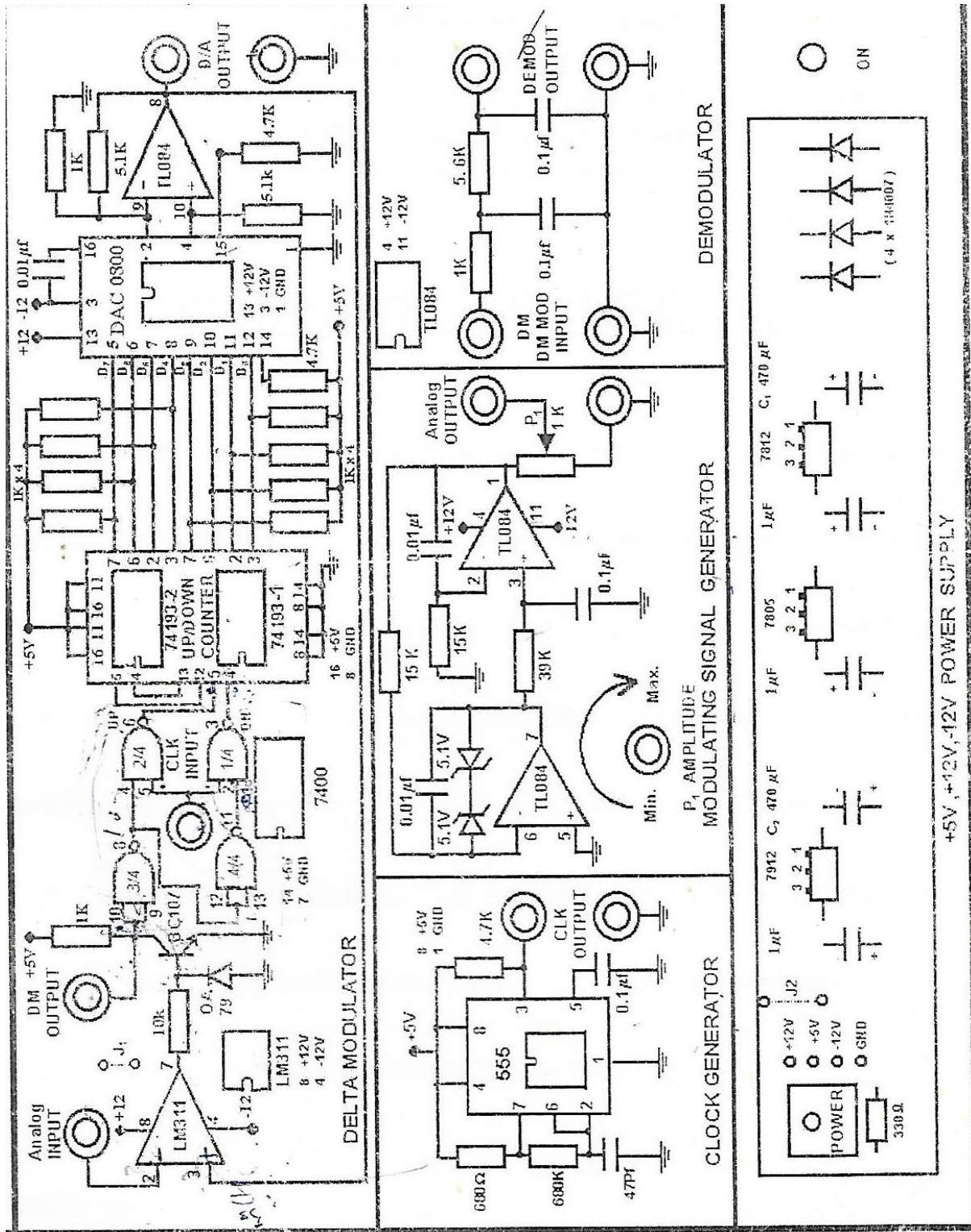


Fig-1 shows the block diagram of the Delta Modulation. This is also known as Linear Delta Modulator. The signal $m(t)$ is the analog input signal. While $r(t)$ is a reconstructed signal which is same as the quantised input signal with 1 bit delay. The signal $r(t)$ tries to follow the input signal $m(t)$ with one bit period delay.

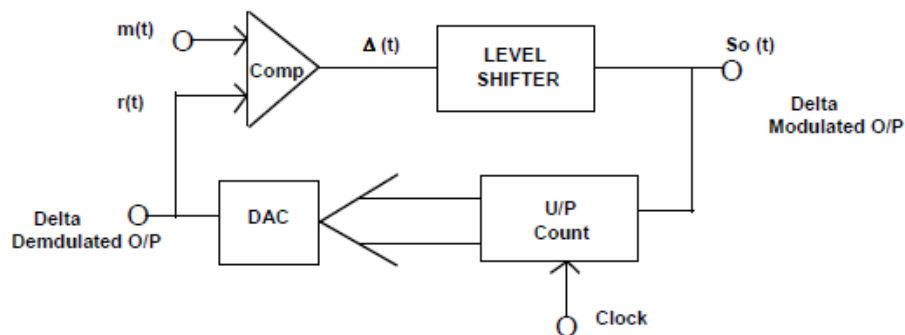


FIG - 1 BLOCK DIAGRAM OF DELTA MODULATION

THEORY

The process of encoding is as follows. The comparator compares the input signal $m(t)$ and $r(t)$. If $m(t) > r(t)$ a logic 1 is generated at the output of the comparator, otherwise a logic 0 is generated. The value of logic 1 or logic 0 turned as $\Delta(t)$ is held for the bit duration by the sample and hold current to generate $S_o(t)$, the Delta Modulated Output.

This output $S_o(t)$ is fed to the 8 bit binary up/down counter to control its count direction. A logic 1 at the mode control input increases the count value by one and a logic '0' decrements the count value by one. All the 8 outputs of the counter are given to DAC to reconstruct the original signal. In essence the counter & Decoder forms the Delta Modulator in the feedback loop of the comparator. Thus, if the input signal is higher than the reconstructed signal the counter increments at each step so as to enable the DAC output to reach to the input signal values. Similarly if the input signal $m(t)$ is lower than the reconstructed signal $r(t)$, the counter decrements at each step, and the DAC output gets reduced to reach a value to that of $m(t)$.

The block diagram of Delta demodulator is shown below. It works in the same way as it was in the feedback loop of the Delta modulator. The received Delta modulated signal $S_o(t)$ is given to the mode control input (U/ D) of the up/ down counter. The counter is 8 bit wide and counts in binary fashion. All the 8 outputs are connected to an 8 bit DAC which gives a quantised analog signal (stepped waveform). A low pass filter is used to smooth out the steps. A buffer amplifier provides the necessary drive capability to the output signal. Thus the digital Delta modulated data is demodulated and reconstructed into an analog signal.

Although this process of Delta Modulation and Demodulation is a simple and cost effective method of coding, there will be poor approximation at starting buildup and 'hunting' at flat top signals.

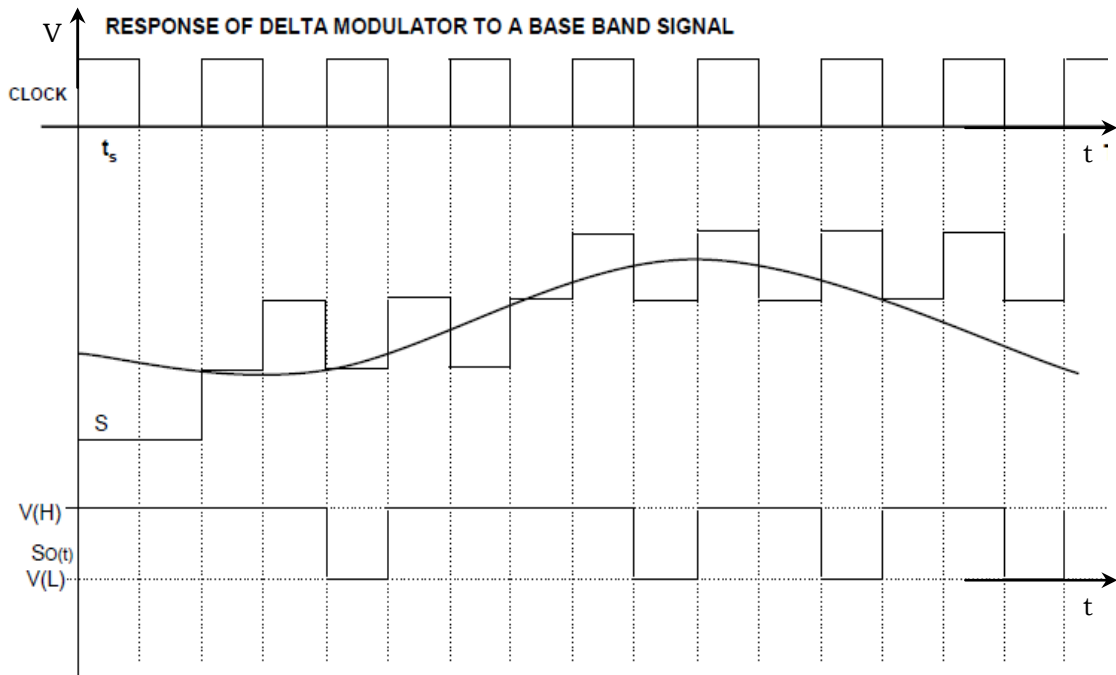


Fig - 2

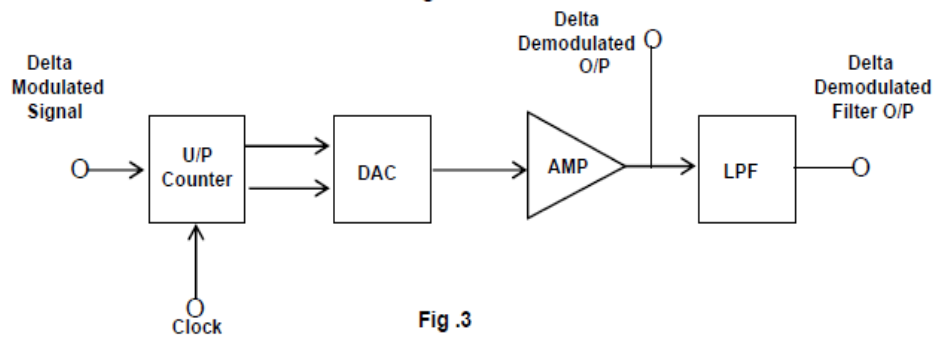


Fig .3

RESPONSE LIMITATIONS OF DELTA MODULATION

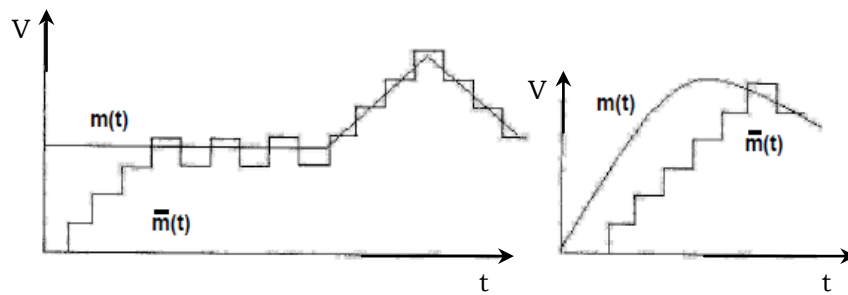


Fig .4

Another limitation in Delta Modulation is ' slope overloading;. Basically the DAC can produce a Max. signal variation of (256×5) volts in 256 clock pulses where is the quantisation step size T_c is the clock period the max. Slope that the DAC can produce in S / T_s volts / second which is the limiting factor where S is the step size and t_s is clock period. If the input signal slope is higher than this or in other words, if the input signal frequency is greater than the limiting value, slope overloading occurs. In such a case true reproduction of the analog signal is not possible. A sinusoidal waveform of amplitude A & frequency f has a maximum slope of $2\Delta fA$ which occurs at zero crossing of the sine wave. If the overloading is to be avoided then the following condition should be satisfied.

$$S_f > 2\pi fA$$

When S = quantisation step size,
 f_s = sampling frequency (Bit frequency), f = Signal frequency , A = Signal Amplitude

$$f_s = \frac{2\pi fA}{S} = \frac{\pi f 2A}{S}$$

When $2A$ = Peak to Peak Amplitude
 $= 256 \times S = \text{DAC Max. Amplitude}$
 $f_s = \pi f \times 256 = 256 \pi f$

For a signal frequency of 200Hz, the sampling frequency should be
 $f_s = 256 \times \pi \times 200 = 160\text{KHz}$

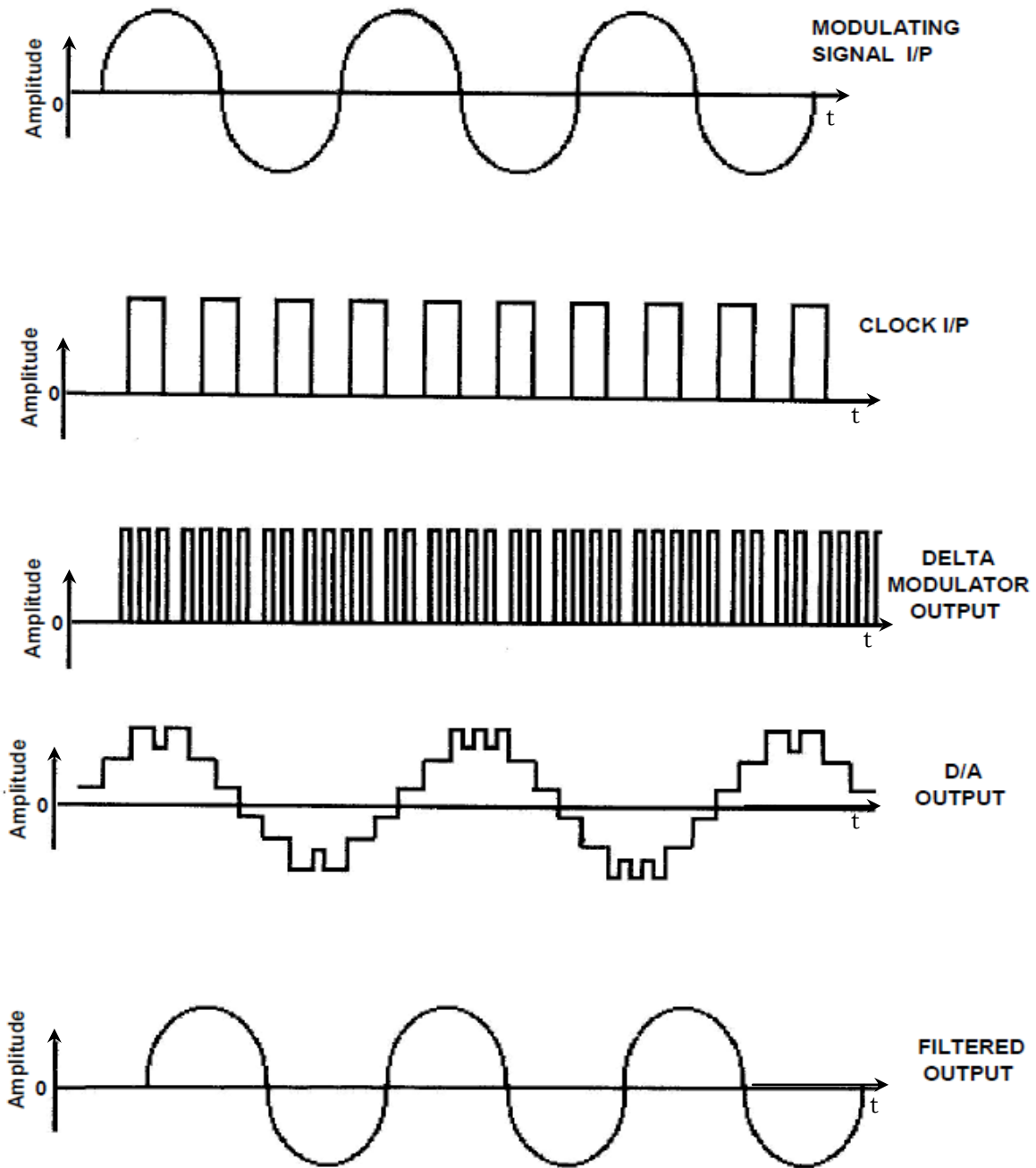
HARDWARE SPECIFICATIONS

1. Delta Modulation and Demodulation Trainer.
2. Built in DC power supply + 5V / 350mA , +/- 12V / 350mA.
3. Provided with 2mm Sockets.
4. The ICs provided on the board are TL084(1No), 555 (1No), (74193 (2 Nos), DAC0800(1 No) , 7400 (1 No) , LM 311 (1 No).
5. Set of Patch chords Stackable 2mm - 6 Nos.
6. User Manual.

EXPERIMENTAL PROCEDURE

1. Connect the AC Adaptor to the mains and the other side to the Experimental Trainer.
2. Switch ON the experimental board.
3. Connect Clock Signal to the Delta Modulator circuit.
4. Connect Modulating Signal to the Modulating signal input of the Delta Modulator and observe the same on channel 1 of a Dual Trace Oscilloscope.
5. Observe the Delta Modulator output on channel II.
6. Connect this Delta Modulator output to the Demodulator.
7. Also connect the clock signal to the demodulator.
8. Observe the Demodulator output with and without RC filter on CRO.

MODEL WAVEFORMS:



PRECAUTIONS:

1. Avoid loose and wrong connections.
2. Readings should be noted without parallax error.

RESULT:

Thus Delta Modulated signal is generated for given input signal and it is Demodulated

9. Super heterodyne Receiver

AIM:

- A. To study the frequency relations with the Mixer.
- B. To study the effect of Image Frequency.

Apparatus:

- A. Superhetrodeyne TX and RX module by AKADEMIA
- B. Connecting Wires

THEORY:

IMAGE FREQUENCIES:

Investigation of Image Frequencies:

This experiment investigates the concept of image frequencies, using the ACL-AD kit. In order to explain what these are, here is a short review of how AM receiver operates. You will recall that the frequency of the LOCAL OSCILLATOR in the receiver section is set higher than the selected signal frequency by a constant amount, irrespective of the frequency of the selected station. The amount of that constant difference frequency is chosen to be 455KHz, the Intermediate Frequency. The MIXER in the receiver section mixes the output of the R.F.AMPLIFIER with the output of the LOCAL OSCILLATOR, to extract the 455 KHz difference frequency. This frequency is then passed on to the I.F. Amplifiers, which preferentially amplify signals of around 455KHz.

The output from I.F. AMPLIFIER 2 is then passed on to the detector, in order to recover the original audio signal. The incoming signal, whose frequency is exactly 455KHz above the frequency of the LOCAL OSCILLATOR, is known as the image frequency. For every wanted frequency there is a corresponding image frequency. If there is a strong station at or near the image frequency, it will result in irritating whistling noises at the output of the receiver, which will spoil the reception of the wanted station.

In the last section, we saw we could receive a station being broadcast on 550KHz by tuning the local oscillator to a frequency of 1 MHz thus giving the difference (IF) frequency of the required 450KHz.

What would happen if we were to receive another station broadcasting on a Frequency of 1.45MHz?

This would also mix with the local oscillator frequency of 1.45MHz to produce the required IF frequency of 450KHz. This would mean that this station would also be received at the same time as the desired one at 550KHz.

STATION 1:

Frequency 550KHz, Local oscillator 1.0MHz, IF = 450KHz.

STATION 2:

Frequency 1.450MHz, Local oscillator 1.0 MHz, IF = 450KHz.

An 'image frequency' is an unwanted frequency that can also combine with the Local Oscillator output to create the IF frequency. Notice how the difference in frequencies of the

wanted and unwanted stations is twice the IF frequency. In the ACL-AD, it means that the image frequency is always 910KHz above the wanted station. This is a large frequency difference and even the poor selectivity of the RF amplifier is able to remove the image frequency unless it is very strong indeed. In this case it will pass through the receiver and will be heard at the same time as the wanted station. Frequency interactions between the two stations tend to cause irritating whistles from the loudspeaker.

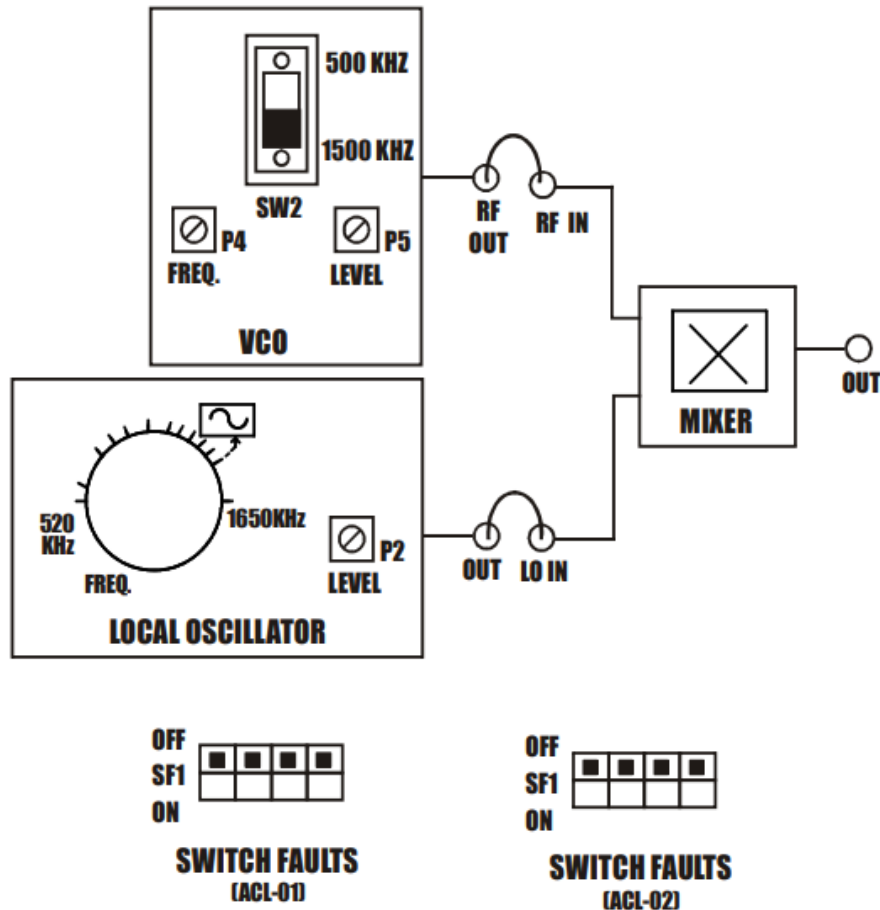


FIG. 8.1 BLOCK DIAGRAM FOR STUDY OF IMAGE FREQUENCY

A. TO STUDY THE FREQUENCY RELATIONS WITH THE MIXER.

Apparatus:

- Modules ACL-AM & ACL-AD.
- Power supply
- Oscilloscope.
- Frequency meter.
- Connecting Links.

NOTE : KEEP ALL THE SWITCH FAULTS IN OFF POSITION.

PROCEDURE:

PROCEDURE:

1. Refer to the FIG. 8.1 & Carry out the following connections; make jumper connections as per diagram.
2. Connect the o/p of **VCO (ACL-AM) OUT** post to the i/p of **MIXER (ACL-AD) RF IN** post.
3. Connect local oscillator **OUT** post to **LO IN** of the mixer section of **ACL-AD**.
4. Connect the power supply with proper polarity to the kit **ACL-AM & ACL-AD** while connecting this; ensure that the power supply is OFF.
5. Switch on the power supply and Carry out the following presetting:
 - VCO (ACL-AM): LEVEL about 100mVpp; FREQ. 550 KHz, Switch on 1500 KHz.
 - LOCAL OSCILLATOR (ACL-AD): LEVEL about 1Vpp; FREQ. 1000 KHz.
6. Connect the oscilloscope to the output of the mixer OUT, and accurately vary the frequency of LO until you detect a sine wave signal.
7. Measure the frequencies of the two input signals and the output signal, and check that also the last is the difference of the first two:

$$f_{IF} = f_{LO} - f_{RF}.$$

SWITCH FAULTS:

Note: Keep the connections as per the procedure. Now put the corresponding switch fault button in ON position and observe the effect on the output. The faults are normally used one at a time.

1. Put switch 2 of SF1 in ACL-AM switch fault section to ON position. This will short the 1K resistor (R32) in VCO. So when the frequency selector switch is in 1500 KHz position, we will get the constant frequency.

B. TO STUDY EFFECT OF IMAGE FREQUENCY

Apparatus:

- Modules ACL-AM & ACL-AD.
- Power supply
- Oscilloscope.
- Frequency meter.
- Connecting Links.

NOTE: KEEP ALL THE SWITCH FAULTS IN OFF POSITION.

PROCEDURE:

1. Refer to the FIG. 8.1 & Carry out the following connections; make jumper connections as per diagram.
2. Repeat the Procedure as explained in A.
3. Switch on the power supply and Carry out the following presetting:
 - VCO (ACL-AM): LEVEL about 100mV p-p; FREQ. 1450 KHz, Switch on 1500 KHz.
 - LOCAL OSCILLATOR: LEVEL about 1V; FREQ. 1000 KHz.
4. Accurately adjust the above frequency until the output is crossed again by a sine waveform.
5. Assure the frequencies and check that now:

$$f_{IF} = f_{LO} - f_{RF}$$

6. The last frequency relations indicate that the two signals with different frequency RF are converted to the same frequency IF. If they were contemporarily present, there would be interference between the two and this would make a proper reception of the signal impossible. The unwished frequency is called image frequency. To prevent this inconvenience it is necessary to prevent that the image signal reaches the input of the mixer and this is carried out interposing selective filters between the input of the signal RF and the mixer.

RESULT:

10. LINEAR BLOCK CODE - ENCODER & DECODER

AIM:

To Study the Hamming Code 7-bit Generation.

APPARATUS:

1. Linear Block Code- Encoder & Decoder Trainer Kit (Sciencetech 2121A & 2121B)
2. 2 mm Banana Cable
3. Regulated Power Supply

THEORY:

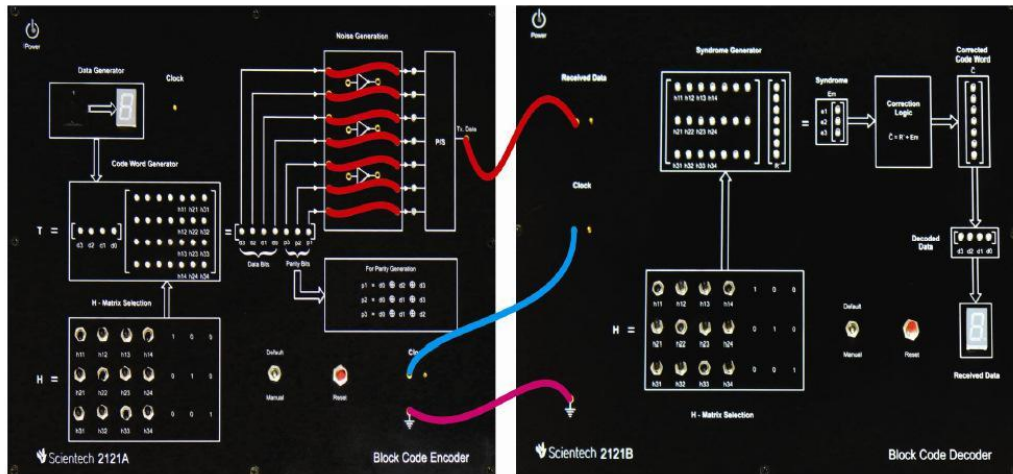
Error Detection and Correction:

Error detection is the ability to detect the presence of errors caused by noise or other impairments during transmission from the transmitter to the receiver. Error correction is the additional ability to reconstruct the original, error-free data. There are two basic ways to design the channel code and protocol for an error correcting system.

Linear block codes:

Linear block codes are conceptually simple codes that are basically an extension of single-bit parity check codes for error detection. A single-bit parity check code is one of the most common forms of detecting transmission errors. This code uses one extra bit in a block of n data bits to indicate whether the number of 1s in a block is odd or even. Thus, if a single error occurs, either the parity bit is corrupted or the number of detected 1s in the information bit sequence will be different from the number used to compute the parity bit: in either case the parity bit will not correspond to the number of detected 1s in the information bit sequence, so the single error is detected. Linear block codes extend this notion by using a larger number of parity bits to either detect more than one error or correct for one or more errors. Unfortunately linear block codes, along with convolutional codes, trade their error detection or correction capability for either bandwidth expansion or a lower data rate, as will be discussed in more detail below. We will restrict our attention to binary codes, where both the original information and the corresponding code consist of bits taking a value of either 0 or 1.

BLOCK DIAGRAM:



PROCEDURE:

1. Connect the power supply mains cord to the Scientech 2121A and Scientech 2121B but do not turn ON the power supply until connections are made for this experiment.

2. Keep default/manual switch in Manual mode.

3. There are some conditions regarding H-Matrix selection manually which are: Any row should not be identically selected like there should not all 1's or all 0's.

Each row selection should be different from other row.

The matrix should be so chosen that all the rows are distinct and consist of at least three 1's in them.

4. Switch 'On' the power supply and press reset button.

5. Check the clock pulse of 2 KHz on Oscilloscope at given test point.

6. At Scientech 2121A Block Code Encoder unit now select the data at seven Segment display with the help of BCD (binary coded decimal) switch.

7. Check the data at seven segment display and its binary equivalent (d3, d2, d1, d0), in the Code Word Generator block T where bit pattern is selected in the form of 8, 4, 2, 1 format.

8. Now set the H matrix as per the condition given in step 3. In Observation Table 3.1, some example sets are given (Set 1, Set2, Set3 and Set4). You can set your own matrix or you can choose any set from example sets and select the H Matrix as per the table.

6. After that check the H matrix in the form of $H = [Ik] [P]$; Identity matrix and Parity matrix corresponding to the selected set as given in the Observation Table 3.1.
7. Check the message signal in the form of $(d_3, d_2, d_1, d_0, p_3, p_2, p_1)$ and verify the status of 'Parity Bits' (p_3, p_2, p_1) as per the equations given for parity generation (see Observation Table 3.1).
8. Connect 2mm patch cords between horizontal bit stream and p/s block as per the connections diagram.
9. Observe the bit pattern output of codeword Generator at vertical 7-bit stream.
10. Now connect the Data output to the Data In of 2121B which is block code decoder.
11. Now connect the clock and ground of 2121A to 2121B via a 2mm patch cord.
12. Now set the H-Matrix section of 2121B Block code Decoder unit as per the same set what you have chose for 2121A Encoder unit. Refer the Observation Table 3.1.
13. Now first set Data '0' at Encoder unit and press reset switch until you get same decoded data on LED display and as well as at the seven segment display in numeric form. Once you get the same data 0 at decoder unit you can vary BCD switch to get the sequential data from 0-9.
14. For any selected data from 0-9, check the H matrix in the form of $H = [P] [Ik]$; Parity matrix and Identity matrix as given in the Observation Table 3.1
15. Also check the message signal in vertical matrix 'R' in the form of $(d_3, d_2, d_1, d_0, p_3, p_2, p_1)$ and check the status of Syndrome Em. As there is no error in the bits it will show (0 0 0).
16. Check the corrected code word and match it with the code word of Encoder unit.
17. Also check the Decode Bits (d_3, d_2, d_1, d_0) and match with the data at Encoder unit.

Equations for Parity Generation:

$$p1 = d0 \oplus d2 \oplus d3$$

$$p2 = d0 \oplus d2 \oplus d3$$

$$p2 = d0 \oplus d1 \oplus d2$$

Code Word Generator:

Identity Matrix				Parity Matrix		

Observation Table:

Decimal	Binary
7 Segment	d3 d2 d1 d0
0	
1	
2	
3	
4	
5	
6	
7	
8	
9	

Code Word						
d3	d2	d1	d0	p3	p2	p1

RESULT:

Linear Block Code encoding and decoding are verified.

11. CONVOLUTION CODE- ENCODER AND DECODER

Aim:

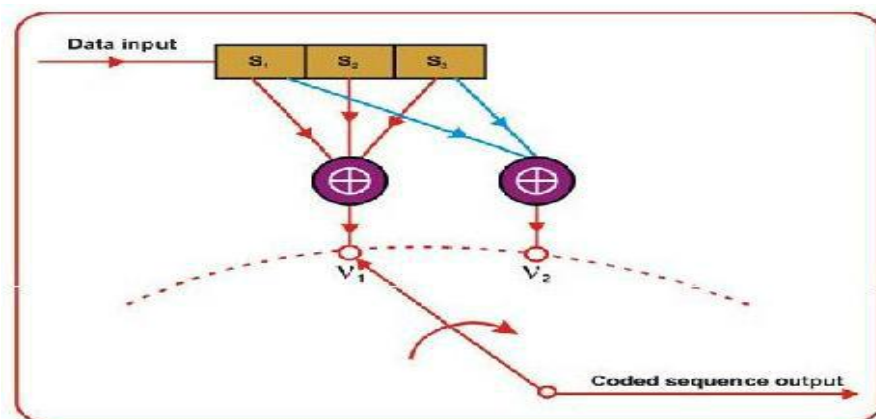
To study the Convolution Encoding and Code tree of Encoder for $K=3$ With Rate $= \frac{1}{2}$ & Study of Decoder with State and Trellis Diagram for $K = 3$ With Rate $= \frac{1}{2}$

Apparatus Required:

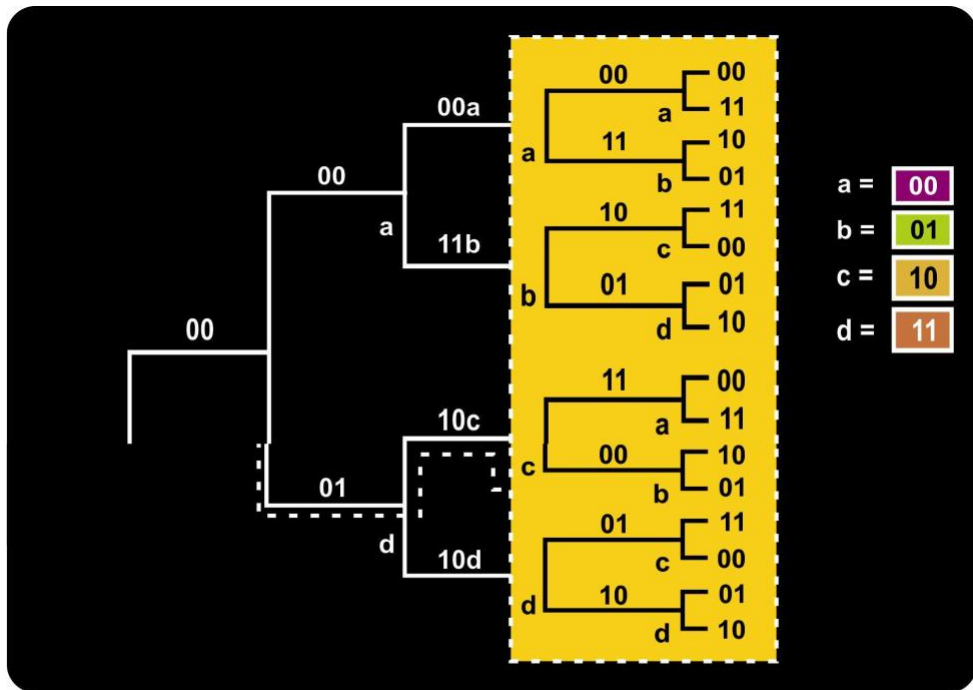
1. ST2122A & ST2122B Convolution Encoding and Decoding Kit
2. 2 mm Banana cable

Theory:

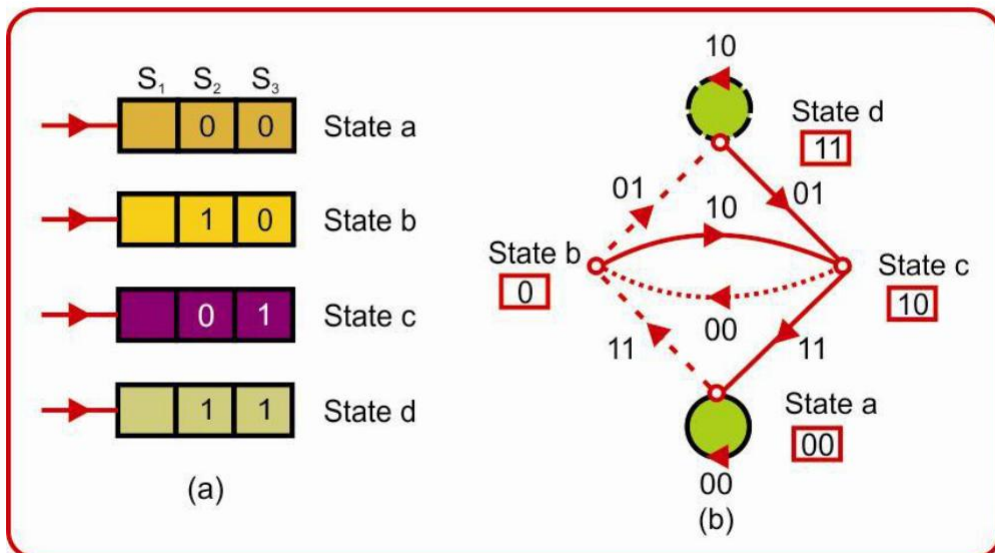
Coding Techniques are used for several reasons: Reduction of dc wandering, Suppression of inter-symbol interference, and reduce error rate. Noise within a transmission channel inevitably causes discrepancies or error between the channel input and channel output. Convolution codes were first introduced by Elias in 1955 as an alternative to block codes. These problems are of engineering interest in the transmission of digital data the coding schemes are used for error detection and correction. In the block coding techniques, the encoder splits up incoming data stream into blocks of finite-number digits and processes each block by adding extra bits (called redundancy) according to a pre defined algorithm. The output of the encoder is a codeword with another finite number of digits. In the convolution coding techniques, the encoder processes the incoming data stream continuously while its decoder employs the Viterbi algorithm among others



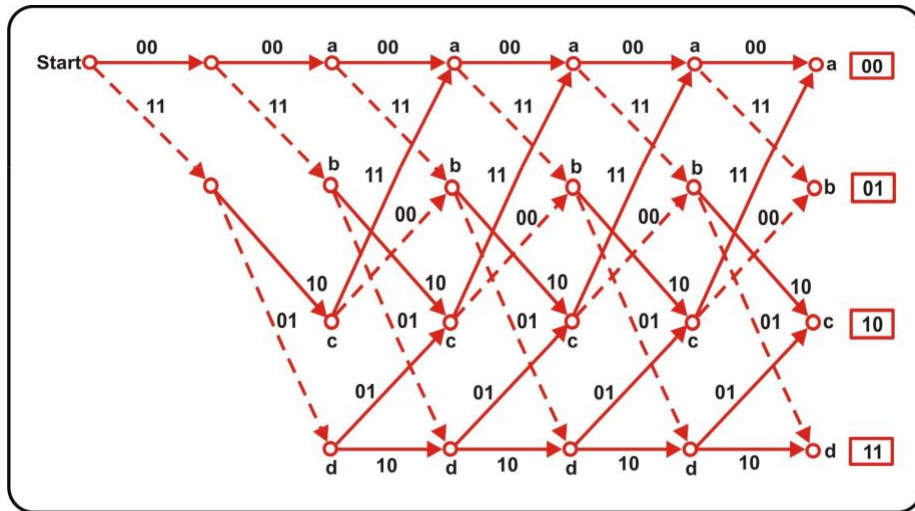
Convolution Coder



Code Tree for Coder

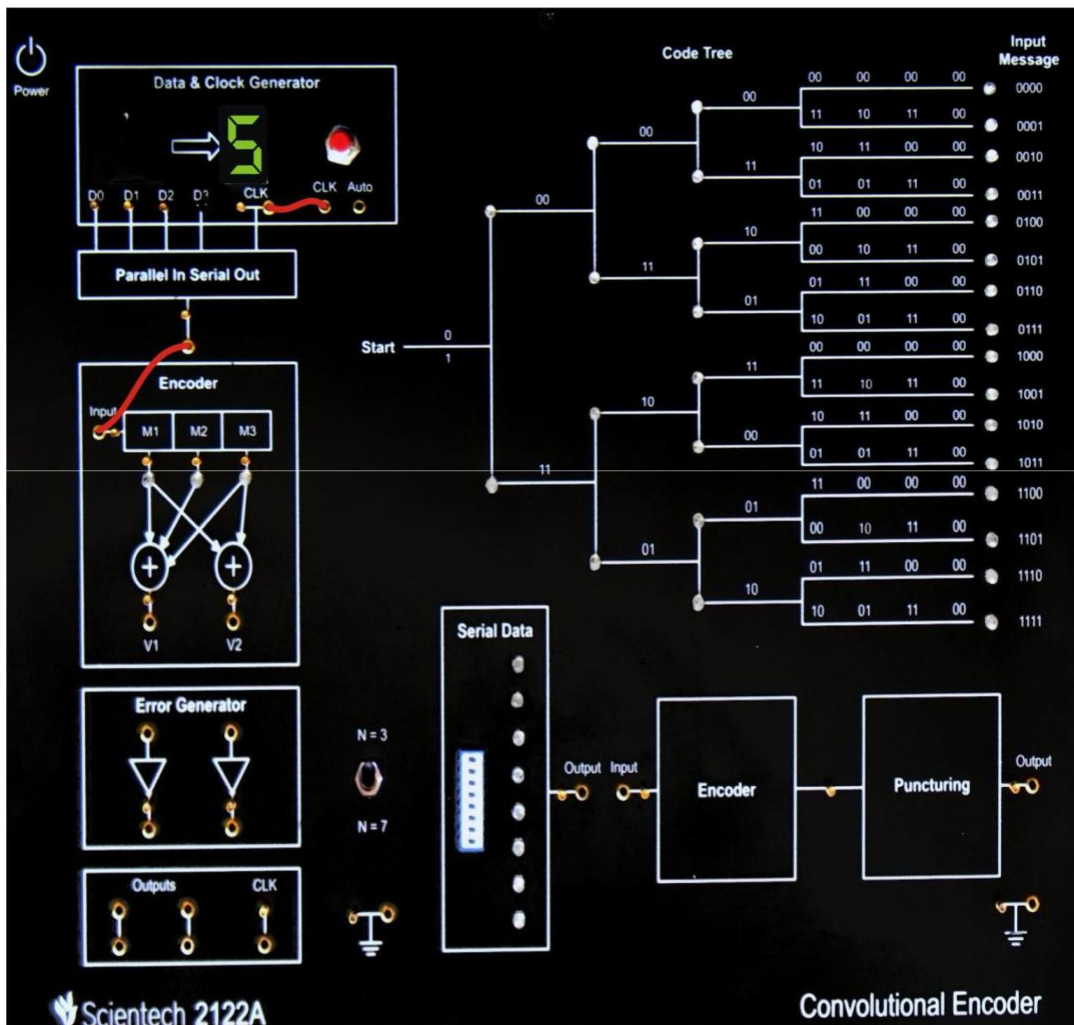


State Diagram for Coder



Trellis diagram for the coder

Connection diagram:



Connection diagram for convolution encoder

Procedure for Encoder:

1. Connect all the patch cord as per above image.
2. Connect the power supply mains cord to the Scientech 2122A, but do not turn ON the power supply until connections are made for this experiment.
3. Keep switch at "N = 3" mode in Scientech 2122A.
4. Switch 'On' the power supply (Using Touch Switch) of Scientech 2122A and press reset button.
5. Initially all the LEDs of Scientech 2122A should be off.
6. Now, select the data at seven segment display with the help of BCD (binary coded decimal) switch. (Note: To select data from 9 to 15 then remove "D3" and refer table 1.0).
7. Check the data at seven segment display and its binary equivalent (d3, d2, d1, d0), in the "Data and Clock Generator block" where bit pattern is selected in the form of 8, 4, 2, 1 format and output is shown in Table 1.1.

Data Table1.1

Data	8	4	2	1
0	0	0	0	0
1	0	0	0	1
2	0	0	1	0
3	0	0	1	1
4	0	1	0	0
5	0	1	0	1
6	0	1	1	0
7	0	1	1	1
8	1	0	0	0
9	1	0	0	1
10	1	0	1	0
11	1	0	1	1
12	1	1	0	0
13	1	1	0	1
14	1	1	1	0
15	1	1	1	1

8. For experiment we have select data at seven segments display is "5". So for this data the corresponding two outputs (V1 and V2) and memory state (M1, M2 and M3) are shown in Table1.2

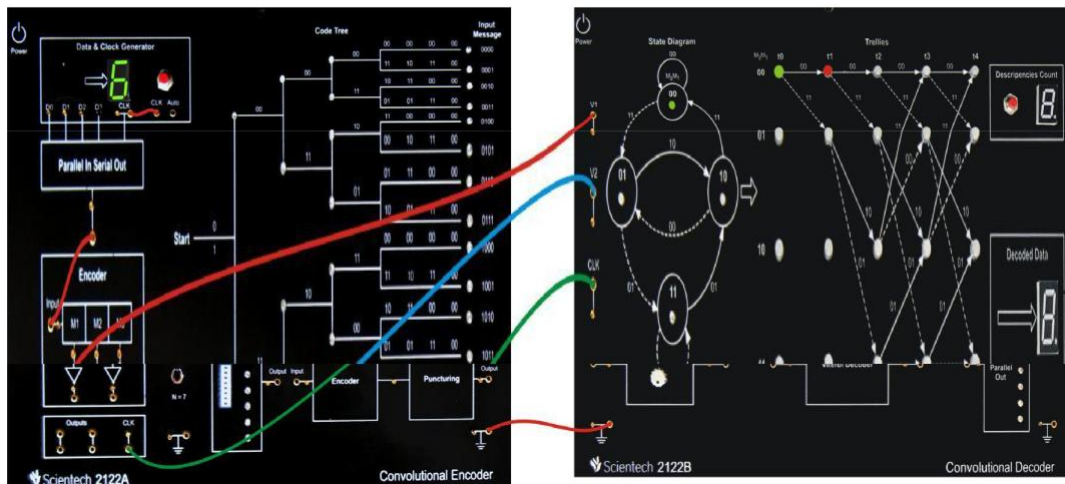
Equations for output: Table for Data = "5" (0101)

Output Table1.2

Clock	Memory M1	Memory M2	Memory M3	Output V1	Output V2
Initial	0	0	0	0	0
1	0	0	0	0	0
2	1	0	0	1	1
3	0	1	0	1	0
4	1	0	1	0	0
5	0	1	0	1	0
6	0	0	1	1	1
7	0	0	0	0	0

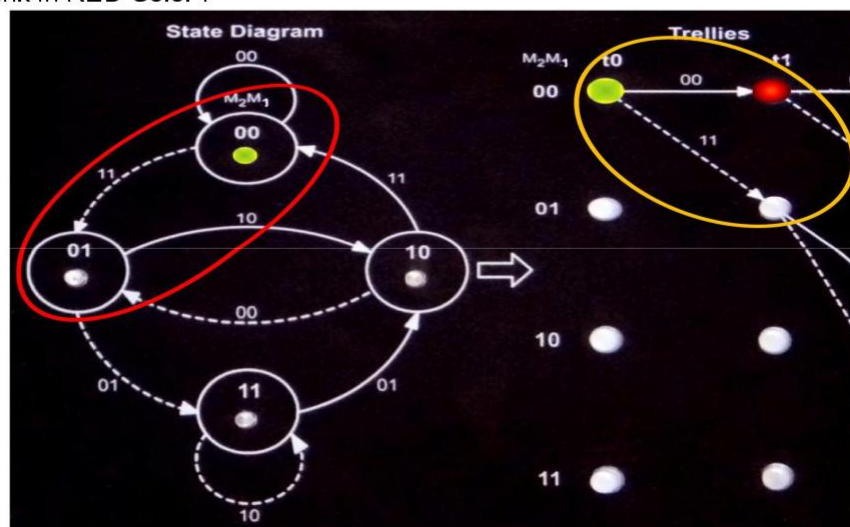
9. Now ,Press clock switch for nest and check the above output on Scientech 2122 A Encoder and code tree Section.
10. Repeat Step 9 up to 7 clocks
11. Also, if user wants to check data in auto clock mode then connect patch card one to auto mode.
12. For code Data = 5 the LED Statues on code tree.

Connection Diagram for Convolution Decoder:



Procedure for Decoder:

1. Connect all the patch cord as per above image.
2. Connect the power supply mains cord to the Scientech 2122A, but do not turn ON the power Supply until connections are made for this experiment.
3. Keep Switch at "N=3" mode in both trainer.
4. Switch 'ON' the power supply (Using touch Switch) of Scientech 2122A and press reset button.
5. Initially all the LEDs of Scientech 2122A should be off
6. Now, select the data at seven segment display with the help of BCD (binary coded Decimal) Switch. (Note: To select data from 9 to 15 then remove "D3" and refer table1.0)
7. Check the data at seven display and its binary equivalent (d3,d2,d1,d0), in the Data clock generator block. Where bit pattern is selected in the form of 8,4,2,1 format and output is shown in table1.0
8. Switch ON the power supply of Scientech 2122B and press reset button.
9. Initially if there is no error, LED "00" and "01" will blink in state diagram and LED to ("00") on is green colour and its corresponding two LED will blink in RED Color .



10. Now , as we have selected the data at seven segment display is “6” . So for this data output table will be as shown below.

Table for data =6 (0110)

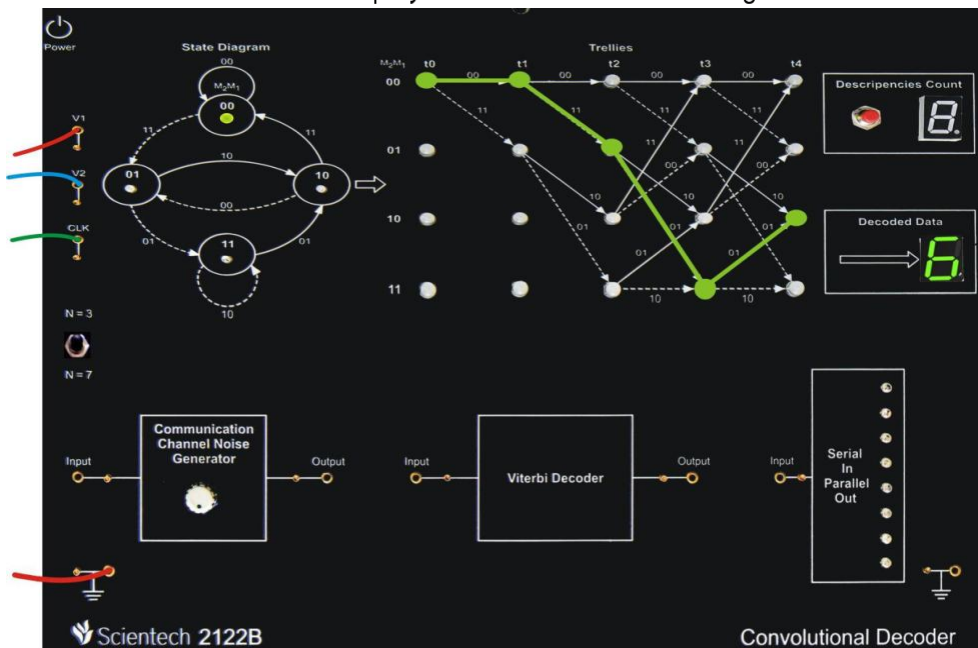
Output Table1.3

Clock	Memory M1	Memory M2	Memory M3	Output V1	Output V2
Initial	0	0	0	0	0
1	0	0	0	0	0
2	1	0	0	1	1
3	1	1	0	0	1
4	0	1	1	0	1
5	0	0	1	1	1
6	0	0	0	0	0
7	0	0	0	0	0

11. Now, press clock switch for next clock and check the above output on scientech 2122A encoder and code tree section and scientech 2122B state diagram and Trellis Section.

12. In Scientech 2122B the state diagram and trellis will indicate current state and next two possible states.

13. Up to four clock LED status will be shown below and after next three clock data will be Display on decoded data seven segment.



Decoded Data

Results:

The study of convolution encoding and decoding is verified.

SIMULATION BASED EXPERIMENTS

12. AMPLITUDE MODULATION AND DEMODULATION

Aim:

- A. To generate the amplitude modulated signal (AM wave) by using given message signal and carrier signals in MATLAB software
- B. To demodulate the AM wave using envelope detector principle

Apparatus:

Personal computer (PC)

MATLAB Software

Theory:

In amplitude modulation, the amplitude of the carrier voltage varies in accordance with the instantaneous value of modulating voltage. Let the modulating voltage be given by expression,

$$V_m = V_m \cos \omega_m t$$

Where ω_m is angular frequency of the signal & V_m is the amplitude. Let the carrier voltage be given by expression,

$$V_c = V_c \cos \omega_c t$$

On Amplitude Modulation, The instantaneous value of modulated carrier voltage is given by,

$$V = V(t) \cos \omega_c t$$

$$V(t) = V_c + k_a V_m \cos \omega_m t$$

$$V = V_c [1 + m_a \cos \omega_m t] \cos \omega_c t$$

Where m_a is modulation index and the modulation index is defined as the ratio of maximum amplitude of modulating signal to maximum amplitude of carrier signal.

$$m_a = V_m / V_c$$

The demodulation circuit is used to recover the message signal from the incoming AM wave at the receiver. An envelope detector is a simple and yet highly effective device that is well suited for the demodulation of AM wave, for which the percentage modulation is less than 100%. Ideally, an envelope detector produces an output signal that follows the envelope of the input signal wave form

PROGRAM:

AM without functions:

```
clc
```

```
clearall
```

```
closeall
```

```
t=linspace(0,0.02,10000); %defining time range for the signal
```

```
fc=5000; %frequency of carrier signal
```

```
fm=200; %frequency of message signal
```

```
fs=40000; %sampling frequency-----fs>=2(fc+BW)
```

```
Am=5; %amplitude of the message signal
```

```
Ac=10; %amplitude of the carrier signal
```

```
m=Am/Ac %modulation index for the AM wave
```

```
wc=2*pi*fc*t; %carrier frequency in radians
```

```
wm=2*pi*fm*t; %message frequency in radians
```

```
ec=Ac*sin(wc); %carrier signal
```

```
em=Am*sin(wm); %messagesignal
```

```
y=Ac*(1+m*sin(wm)).*sin(wc); %amplitude modulated signal
```

```
z=y.*ec; %in synchronous detection the AM signal is
```

```
multiplied with carrier signal and passed through LPF
```

```
z1=conv(z,exp(-t/0.000795));% the LPF filter response in time domain is given
```

```
by  $\exp(-t/RC)$ , the cut off frequency for filter should be  $f_m=200$ 
```

```
% $F=1/(2*\pi*R*C)$ , replacing  $F=200$ , and
```

```
%assuming  $R=1k$  ohm then  $C=0.795$  MICROFARAD
```

```
%so  $RC=0.000795$ 
```

```
%we will get the demodulated signal by convolving the AM signal with LPF response
```

```
l=10000;
```

```
subplot(4,1,1),plot(t(1:l),em(1:l))
```

```
xlabel('time(sec)');
```

```
ylabel('amplitude in volts(V)');
```

```
title('MODULATING SIGNAL');
```

```
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2))
```

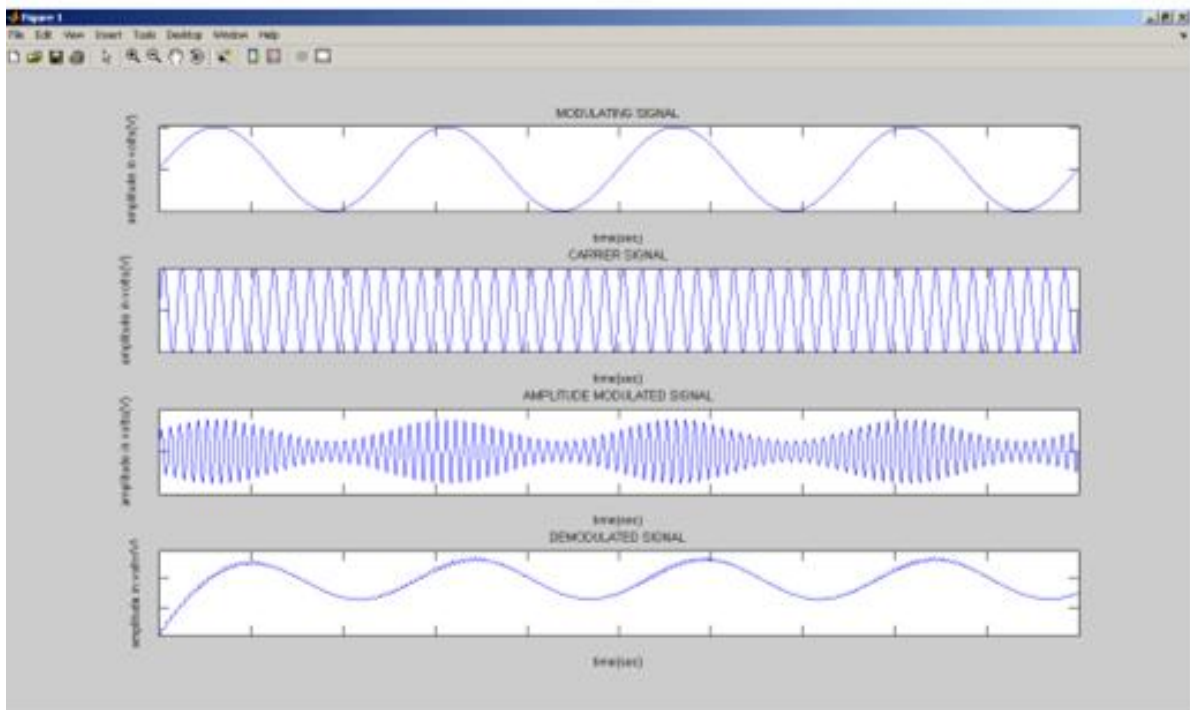
```
xlabel('time(sec)');
```

```

ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(4,1,3),plot(t(1:l),y(1:l))
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('AMPLITUDE MODULATED SIGNAL');
subplot(4,1,4),plot(t(1:l),z1(1:l))
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');

```

Model Waveforms:

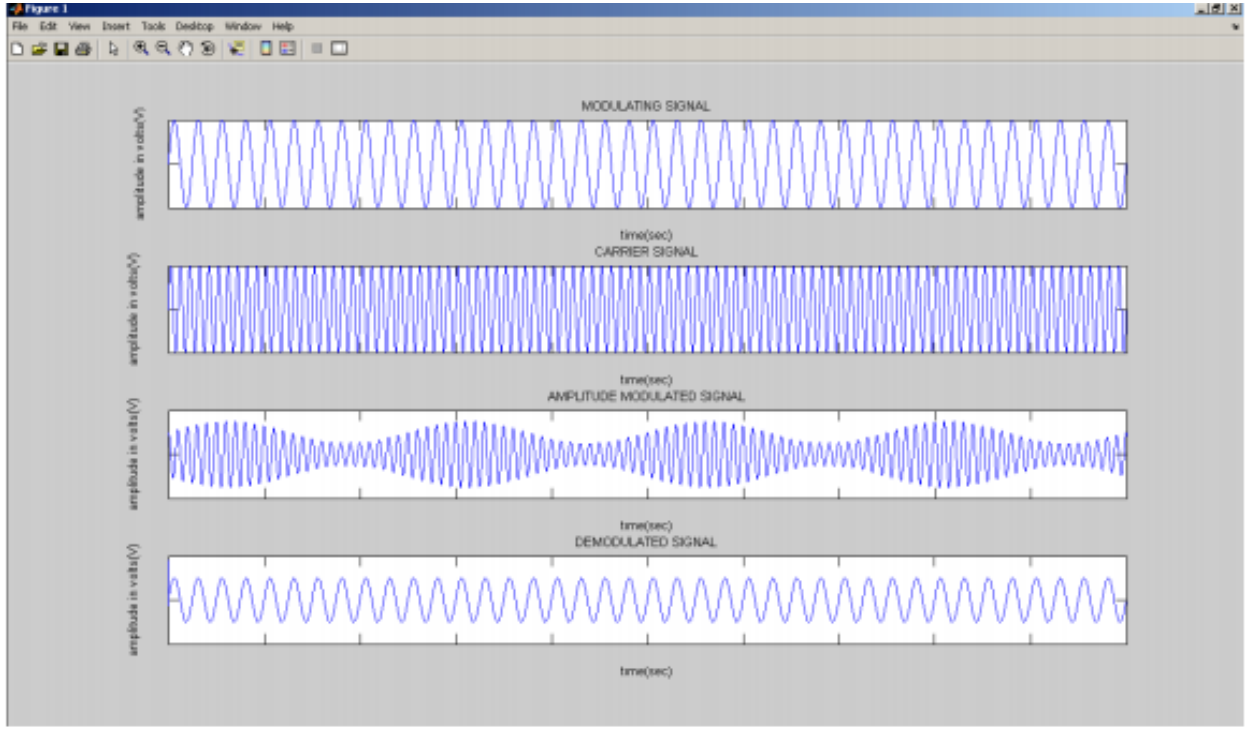


AM with functions:

```
clc
clearall
closeall
t=linspace(0,0.2,100000);%defining time range for the signal
fc=1000;%frequency of carrier signal
fm=200;%frequency of message signal
fs=100000;%sampling frequency-----fs>=2(fc+BW)
Am=5;%amplitude of the message signal
Ac=10;%amplitude of the carrier signal
m=Am/Ac%modulation index for the AM wave
wc=2*pi*fc*t;%carrier frequency in radians
wm=2*pi*fm*t;%message frequency in radians
ec=Ac*sin(wc);%carrier signal
em=Am*sin(wm);%messagesignal
y=ammod(em,fc,fs,0,Ac);%amplitude modulated signal
z=amdemod(y,fc,fs,0,Ac);%demodulated AM signal
l=100000;
subplot(4,1,1),plot(t(1:l),em(1:l))
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('MODULATING SIGNAL');
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2))
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(4,1,3),plot(t(1:l),y(1:l))
axis([0 0.02 -20 20])%setting axis dimensions
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('AMPLITUDE MODULATED SIGNAL');
subplot(4,1,4),plot(t(1:l),z(1:l))
```

```
xlabel('time(sec)');  
ylabel('amplitude in volts(V)');  
title('DEMODULATED SIGNAL');
```

Model Waveforms:



Result:

13. FREQUENCY MODULATION AND DEMODULATION

Aim:

- A. To generate frequency modulated signal and observe the characteristics of FM wave using MATLAB software.
- B. To demodulate a Frequency Modulated signal using MATLAB software

Apparatus:

- A. Personal computer(PC)
- B. MATLAB Software

Theory:

Frequency modulation consists in varying the frequency of the carrier voltage in accordance with the instantaneous value of the modulating voltage. Thus the amplitude of the carrier does not change due to frequency modulation. Let the modulating voltage be given by expression:

$$V_m = V_m \cos \omega_m t$$

Where ω_m is angular frequency of the signal & V_m is the amplitude. Let the carrier voltage be given by expression,

$$V_c = V_c \sin(\omega_c t + \theta)$$

On frequency modulation, the instantaneous value of modulated carrier voltage is given by,

$$V_c = V_c \sin \phi, \text{ Where } \phi = \omega_c t + \theta;$$

$$\phi = \omega_c t + k_f V_m / \omega_m \sin \omega_m t + \theta;$$

Hence the frequency modulated carrier voltage is given by,

$$V = V_c \sin[\omega_c t + k_f V_m / \omega_m \sin \omega_m t]$$

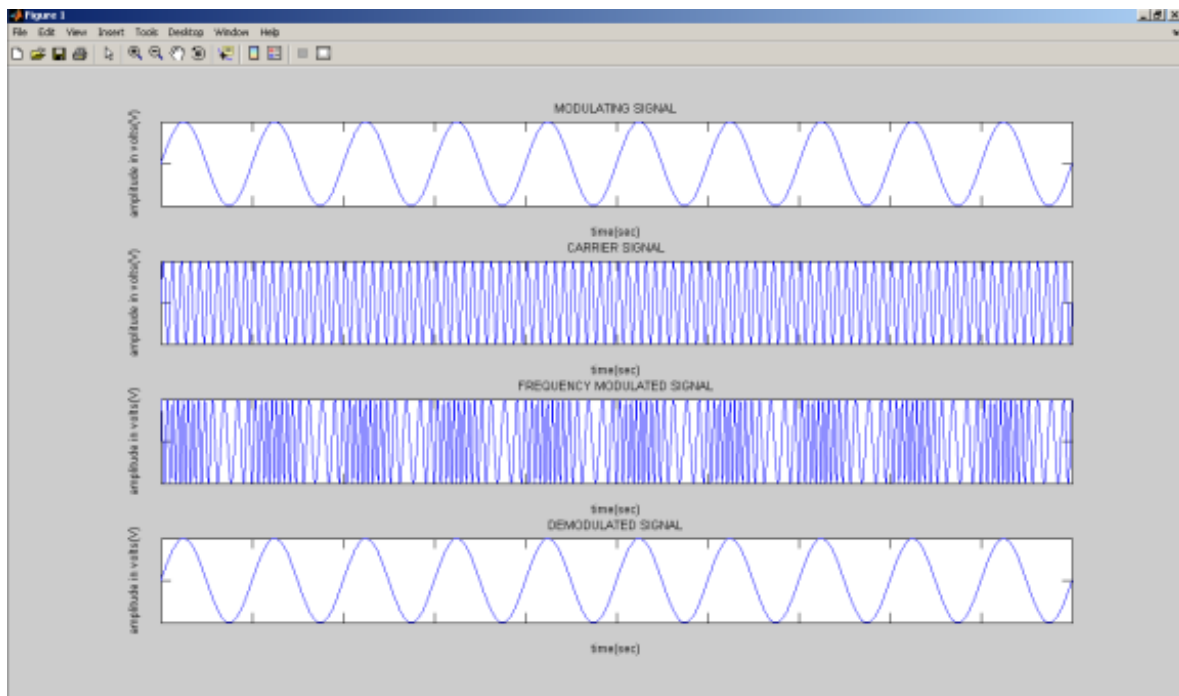
The modulation index is defined as the ratio of frequency deviation to frequency of modulating signal $m_f = d/f_m$ where deviation $d = (f_{max} - f_{min})/2$.

Program:

FM with functions:

```
clc
clearall
closeall
Fs = 8000; % Sampling rate of signal
Fc = 100; % Carrier frequency
t = linspace(0,1,10000); % Sampling times
x = sin(2*pi*10*t) % Channel 1
dev = 50; % Frequency deviation in modulated signal
y = fmod(x,Fc,Fs,dev); % Modulate both channels.
z = fmdemod(y,Fc,Fs,dev); % Demodulate both channels.
subplot(411),plot(t,x)
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('MODULATING SIGNAL');
subplot(412),plot(t,sin(2*pi*Fc*t))
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(413),plot(t,y)
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('FREQUENCY MODULATED SIGNAL');
subplot(414),plot(t,z)
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');
```


Model waveforms:



Result:

SOFTWARE EXPERIMENTS:

14. AMPLITUDE SHIFT KEYING

AIM:-

To plot the wave form for Binary Amplitude Shift Keying (BASK) signal using MATLAB for a stream of bits.

THEORY:-

Amplitude Shift Keying (ASK) is the digital modulation technique. In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes. In ASK, the amplitude of the carrier assumes one of the two amplitudes dependent on the logic states of the input bit stream. This modulated signal can be expressed as:

$$x_e(t) = \begin{cases} 0 & \text{symbol "0"} \\ A \cos \omega_c t & \text{symbol "1"} \end{cases}$$

Amplitude shift keying (ASK) in the context of digital signal communications is a modulation process, which imparts to a sinusoid two or more discrete amplitude levels. These are related to the number of levels adopted by the digital message. For a binary message sequence there are two levels, one of which is typically zero. Thus the modulated waveform consists of bursts of a sinusoid. Figure 1 illustrates a binary ASK signal (lower), together with the binary sequence which initiated it (upper). Neither signal has been band limited.

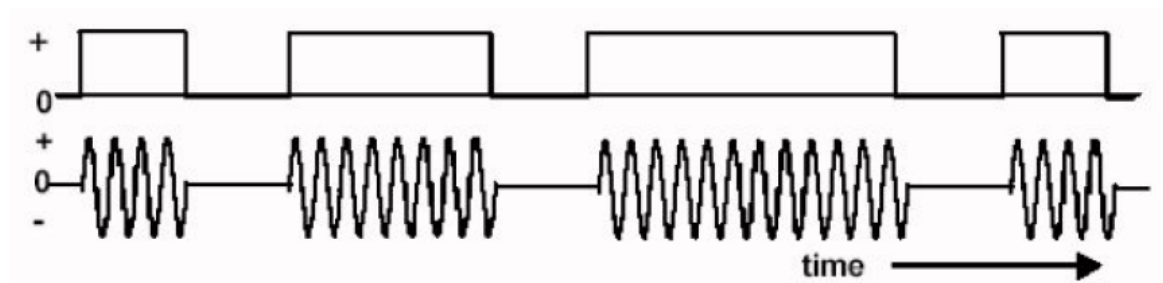


Fig: an ASK signal (below) and the message (above)

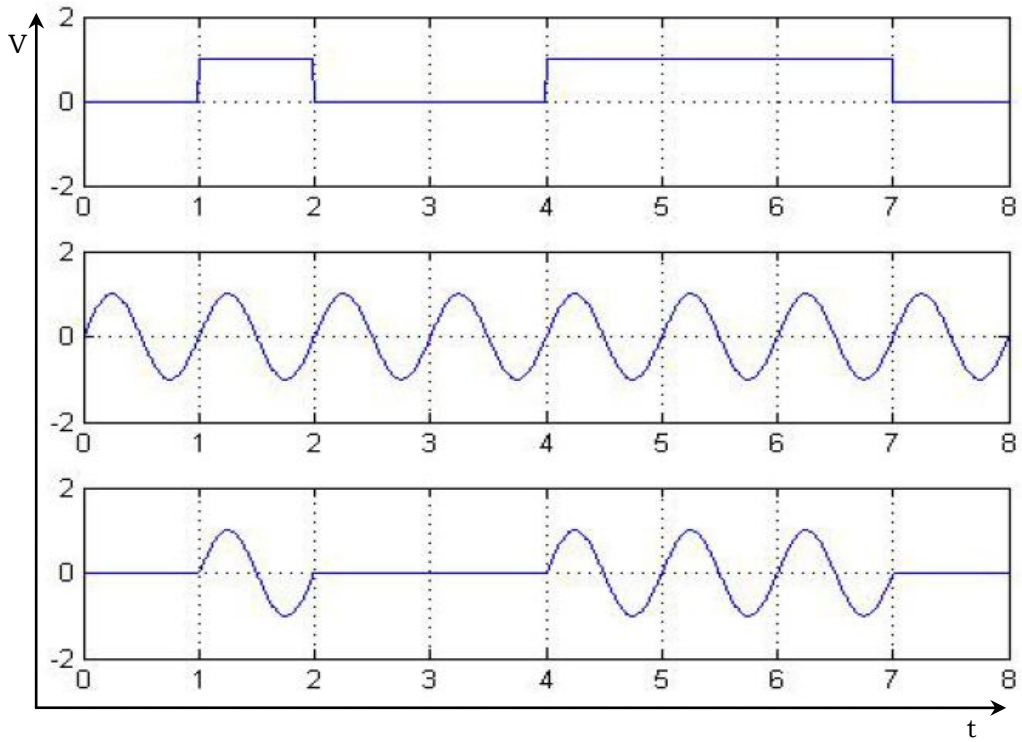
There are sharp discontinuities shown at the transition points. These result in the signal having an unnecessarily wide bandwidth. Band limiting is generally introduced before transmission, in which case these discontinuities would be 'rounded off'. The band limiting may be applied to the digital message, or the modulated signal itself. The data rate is often made a sub-multiple of the carrier frequency. This has been done in the waveform of Fig.

MATLAB PROGRAM:-

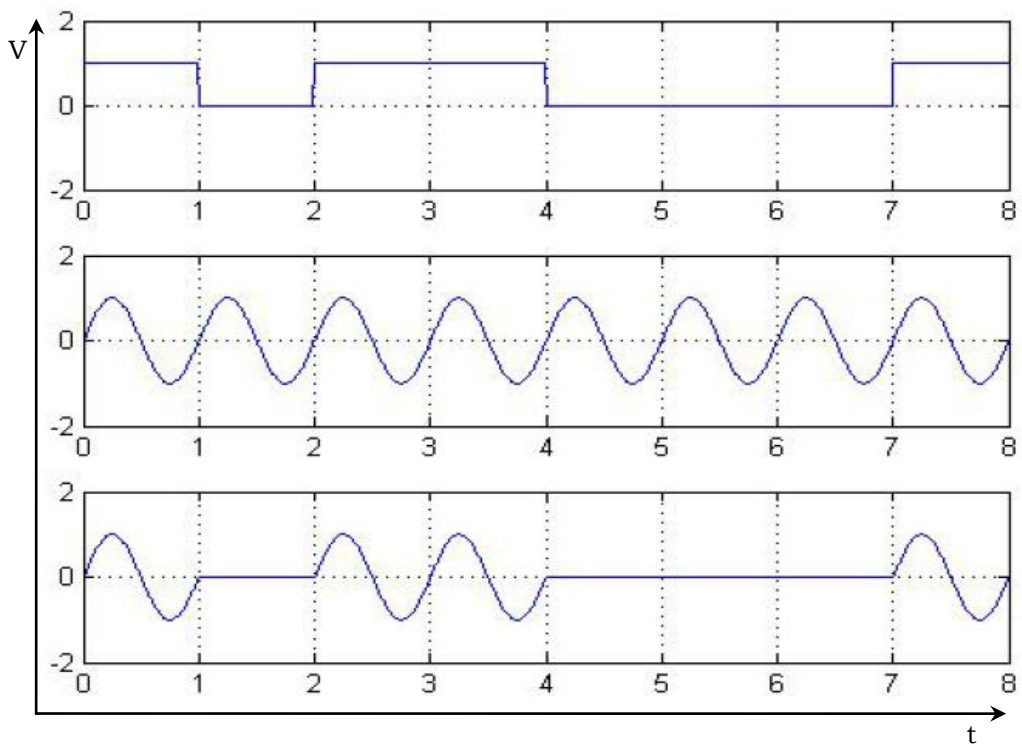
```
clear;
clc;
b = input('Enter the Bit stream \n '); %b = [0 1 0 1 1 1 0];
n = length(b);
t = 0:.01:n;
x = 1:1:(n+1)*100;
for i = 1:n
    for j = i:1:i+1
        bw(x(i*100:(i+1)*100)) = b(i);
    end
end
bw = bw(100:end);
sint = sin(2*pi*t);
st = bw.*sint;
subplot(3,1,1)
plot(t,bw)
grid on ; axis([0 n -2 +2])
subplot(3,1,2)
plot(t,sint)
grid on ; axis([0 n -2 +2])
subplot(3,1,3)
plot(t,st)
grid on ; axis([0 n -2 +2])
```

OBSERVATION:-

Output waveform for the bit stream [0 1 0 0 1 1 1 0]



Output waveform for the bit stream [1 0 1 1 0 0 0 1]



Result:

The program for ASK modulation and demodulation has been simulated in MATLAB and necessary graphs are plotted

15. PHASE SHIFT KEYING

AIM:-

To plot the wave form for Binary Phase Shift Keying signal (BPSK) using MATLAB for a stream of bits.

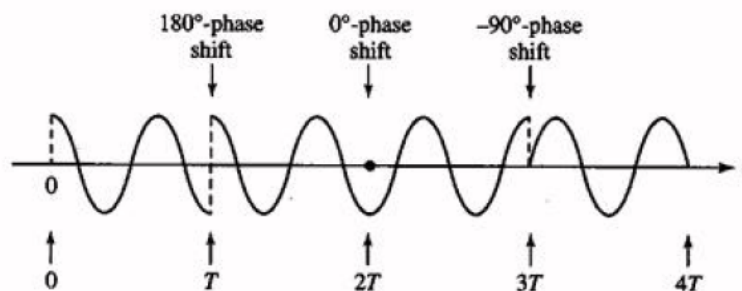
THEORY:-

In carrier-phase modulation, the information that is transmitted over a communication channel is impressed on the phase of the carrier. Since the range of the carrier phase is $0 \leq \theta \leq 2\pi$, the carrier phases used to transmit digital information via digital-phase modulation are $\theta_m = 2\pi m/M$, for $m=0,1,2,\dots,M-1$. Thus for binary phase modulation ($M=2$), the two carrier phases are $\theta_0 = 0$ and $\theta_1 = \pi$ radian. For M -array phase modulation $= 2^k$ where k is the number of information bits per transmitted symbol.

The general representation of a set of M carrier-phase-modulated signal waveforms is

$$u_m(t) = AgT(t) \cos(2\pi fct + 2\pi m/M), m=0,1,\dots,M-1$$

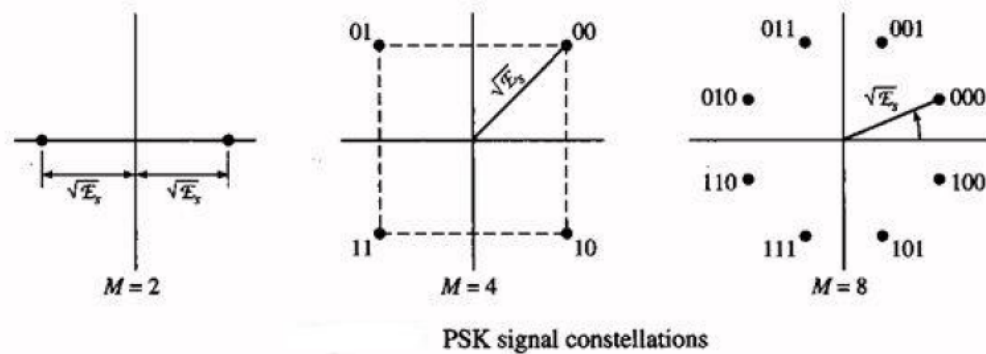
Where, $gT(t)$ is the transmitting filter pulse shape, which determines the spectral characteristics of the transmitted signal, and A is the signal amplitude. This type of digital phase modulation is called phase-shift-keying.



Example of a four-phase PSK signal

Signal point constellations for $M=2, 4$ and 8 are illustrated in figure. We observe that binary phase modulation is identical to binary PAM (binary antipodal signals). The mapping or assignment, of k information bits into the $M=2^k$ possible phases may be done in a number of ways. The preferred assignment is to Gray in coding, in which adjacent phases differ by one binary digit, as illustrated

below in the figure. Consequently, only a single bit error occurs in the k-bit sequence with Gray encoding when noise causes the incorrect selection of an adjacent phase to the transmitted phase



In figure shows that, block diagram of M=4 PSK system. The uniform random number generator fed to the 4-PSK mapper and also fed to the compare. The 4-PSK mapper split up into two phases. On the other hand, Gaussian RNG adds to the modulator. The two phases are fed to the detector. The output goes to the compare. The Uniform random number generator and detector also fed to the detector and finally fed to the bit-error counter and symbol-error counter.

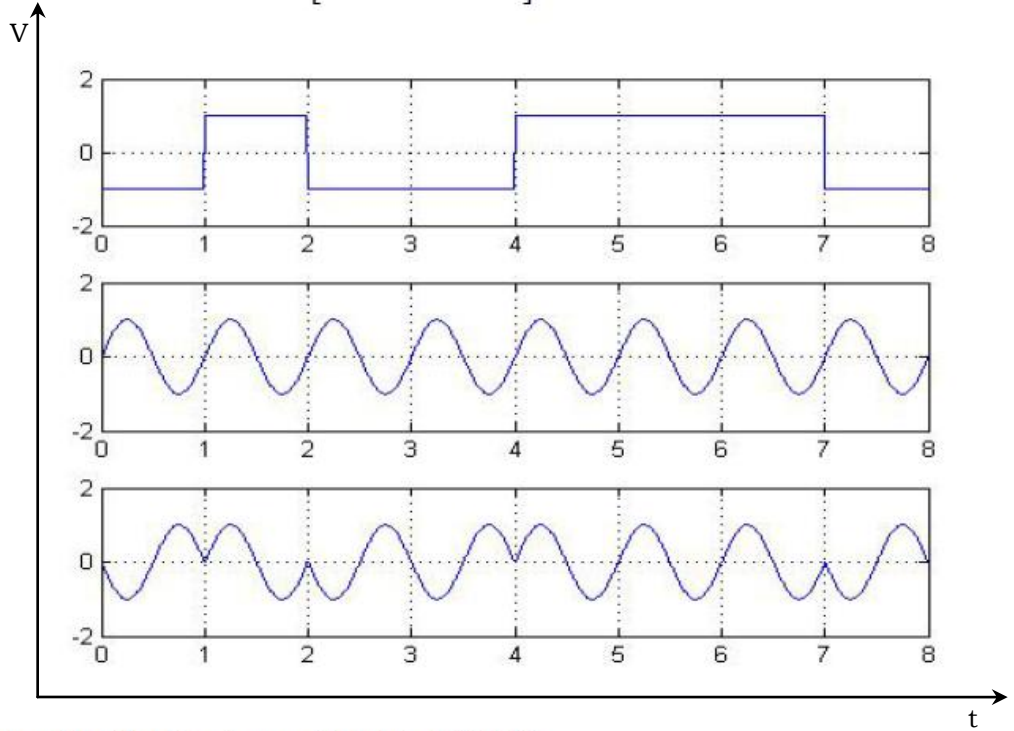
MATLAB PROGRAM:-

```
clear;
clc;
b = input('Enter the Bit stream \n ');      %b = [0 1 0 1 1 1 0];
n = length(b);
t = 0:.01:n;
x = 1:1:(n+1)*100;
for i = 1:n
    if (b(i) == 0)
        b_p(i) = -1;
    else
        b_p(i) = 1;
    end
    for j = i:.1:i+1
        bw(x(i*100:(i+1)*100)) = b_p(i);
    end
end
end
```

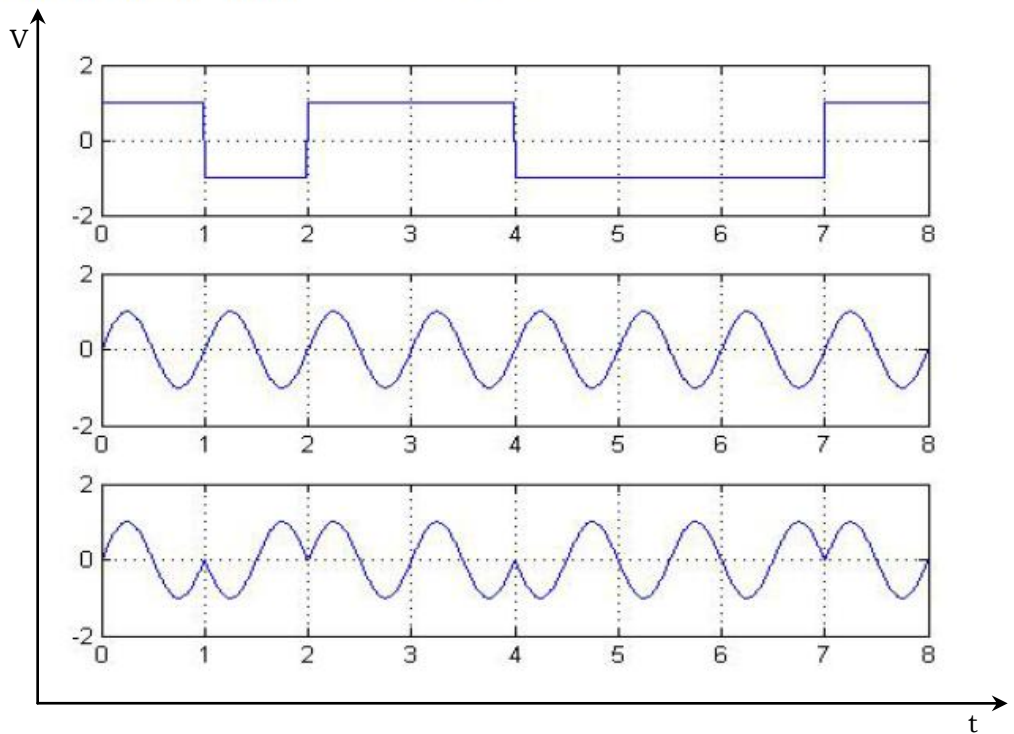
```
bw = bw(100:end);
sint = sin(2*pi*t);
st = bw.*sint;
subplot(3,1,1)
plot(t,bw)
grid on ; axis([0 n -2 +2])
subplot(3,1,2)
plot(t,sint)
grid on ; axis([0 n -2 +2])
subplot(3,1,3)
plot(t,st)
grid on ; axis([0 n -2 +2])
```

OBSERVATION:-

Output waveform for the bit stream [0 1 0 0 1 1 1 0]



Output waveform for the bit stream [1 0 1 1 0 0 0 1]



Result

The program for PSK modulation and demodulation has been simulated in MATLAB and necessary graphs are plotted.

ABOUT AC & DC LABORATORY

AC & DC LAB CONSIST OF ANALOG AND DIGITAL MODULATION AND DEMODULATION TRAINER KITS, SPECTRUM ANALYZER, SUPER HETRODYNE RECEIVER MODULE TRAINER KIT, 16 DIGITAL STORAGE OSCILLOSCOPES, UNINTERUPTED 0-30V REGULATED POWER SUPPLY WITH 3KVA ONLINE UPS AND SPACIOUS LAB WITH GOOD LIGHTING WHICH IS COMFORTABLE FOR BLENDED LEARNING AND FLIPPED LEARNING.



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