

## EXPERIMENT NO:1

### SIGNAL SAMPLING AND RECONSTRUCTION

#### AIM:-

#### Study of Sampling Process and Signal Reconstruction

#### APPARATUS REQUIRED:-

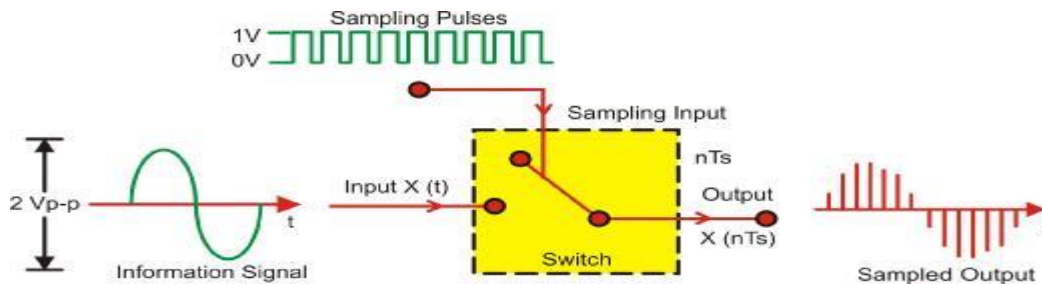
1. ST2101 with power supply cord.
2. Oscilloscope with connecting probe.
3. Connecting cords.

#### THEORY:-

The signals we use in the real world, such as our voice, are called "analog" signals. To process these signals for digital communication, we need to convert analog signals to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert continuous time signal to discrete time signal, a process is used called as sampling. The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample.

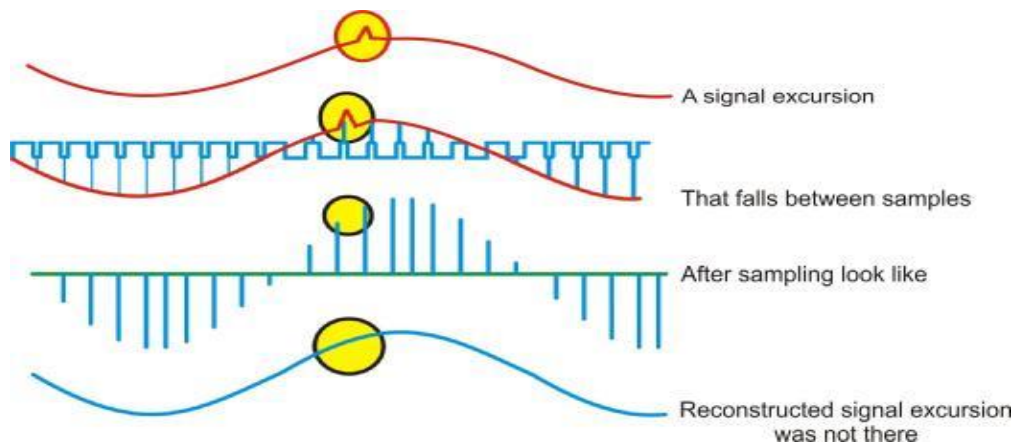
#### Principle of sampling:-

Consider an analogue signal  $x(t)$  that can be viewed as a continuous function of time, as shown in figure. We can represent this signal as a discrete time signal by using values of  $x(t)$  at intervals of  $nT_s$  to form  $x(nT_s)$  as shown in figure . We are "grabbing" points from the function  $x(t)$  at regular intervals of time,  $T_s$ , called the sampling period.



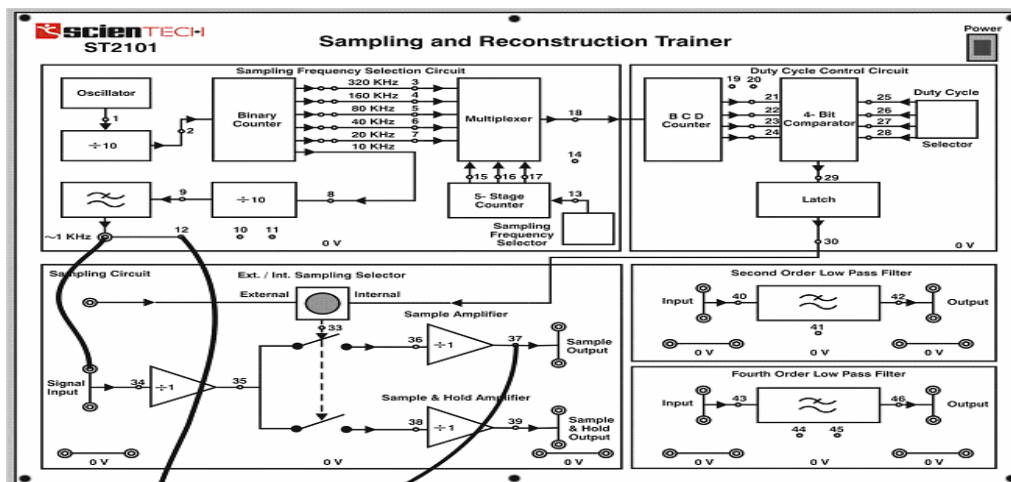
#### Aliasing:

A precondition of the sampling theorem is that the signal to be band limited. However, in practice, no time-limited signal can be band limited. Since signals of interest are almost always time-limited (e.g., at most spanning the lifetime of the sampling device in question), it follows that they are not band limited. However, by designing a sampler with an appropriate guard band, it is possible to obtain output that is as accurate as necessary. Aliasing is the presence of unwanted components in the reconstructed signal. These components were not present when the original signal was sampled. In addition, some of the frequencies in the original signal may be lost in the reconstructed signal. Aliasing occurs because signal frequencies can overlap if the sampling frequency is too low. As a result, the higher frequency components roll into the reconstructed signal and cause distortion of the signal. Frequencies "fold" around half the sampling frequency. This type of signal distortion is called aliasing. We only sample the signal at intervals. We don't know what happened between the samples. A crude example is to consider a 'glitch' that happened to fall between adjacent samples. Since we don't measure it, we have no way of knowing the glitch was there at all.

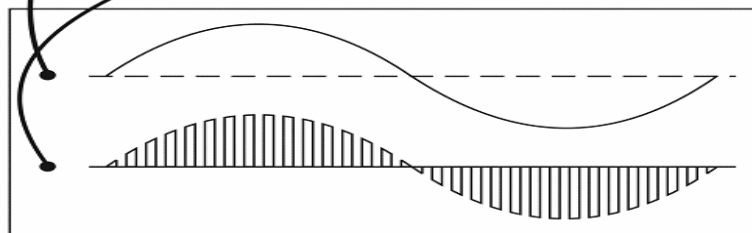


In a less obvious case, we might have signal components that are varying rapidly in between samples. Again, we could not track these rapid inter-sample variations. We must sample fast enough to see the most rapid changes in the signal. Sometimes we may have some a priori knowledge of the signal, or be able to make some assumptions about how the signal behaves in between samples.

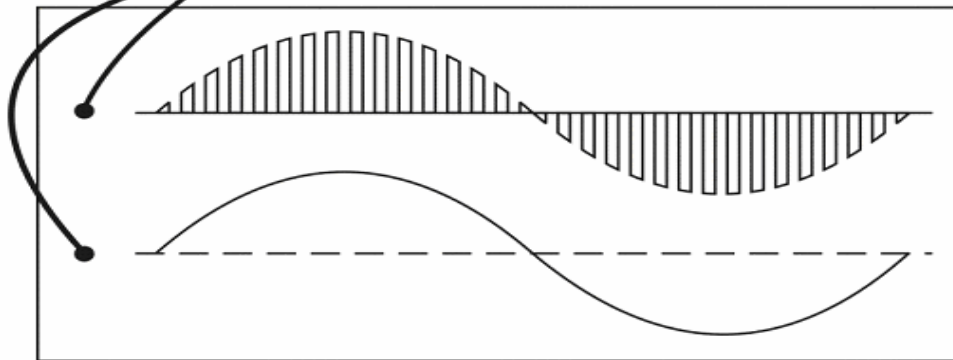
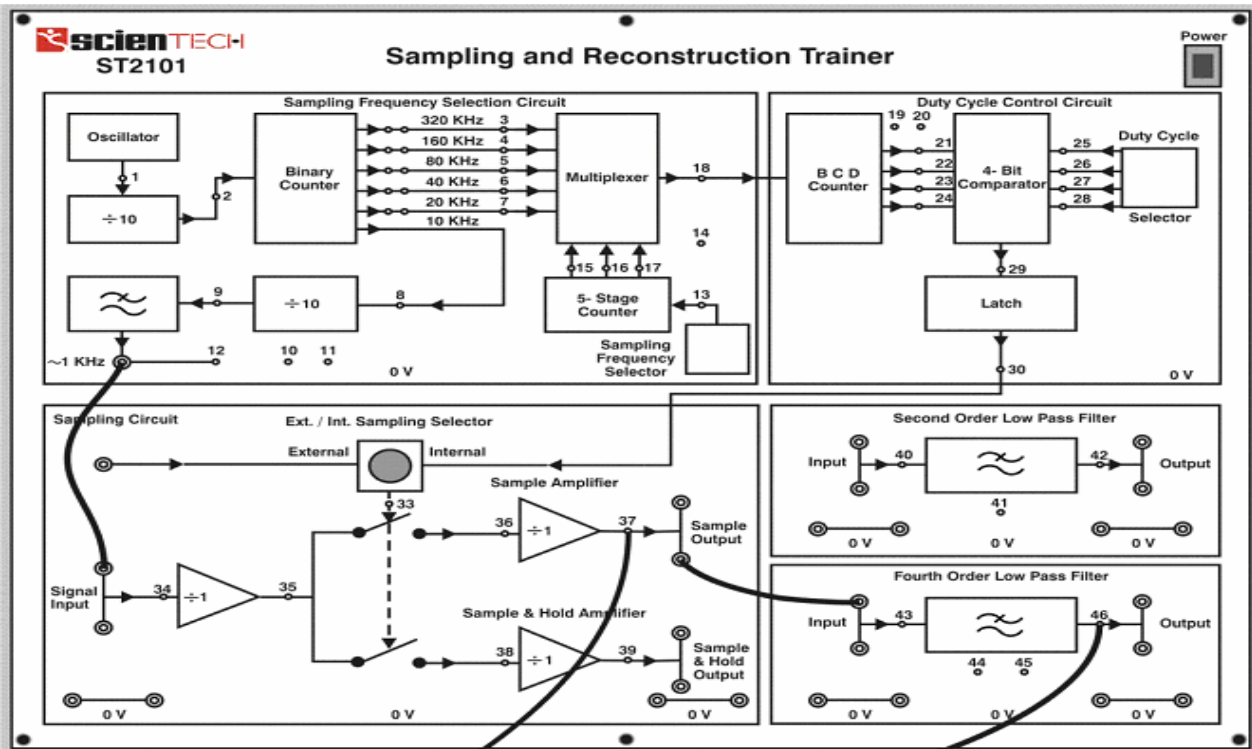
**CIRCUIT DIAGRAM:-**



**Signal Sampling**



## SIGNAL RECONSTRUCTION:-



**Signal Reconstruction**

### PROCEDURE:-

A. Set up for Sampling and reconstruction of signal. Initial set up of trainer: Duty cycle selectors switch position: Position 5.

Sampling selector switch: Internal position.

1. Connect the power cord to the trainer. Keep the power switch in 'Off' position.
2. Connect 1 KHz Sine wave to signal Input.
3. Switch 'On' the trainer's power supply & Oscilloscope.
4. Connect BNC connector to the CRO and to the trainer's output port.
5. Select 320 KHz (Sampling frequency is 1/10th of the frequency indicated by the illuminated LED) sampling rate with the help of sampling frequency selector switch.
6. Observe 1 KHz sine wave (TP12) and Sample Output (TP37) on Oscilloscope. The display shows 1 KHz Sine wave being sampled at 32 KHz, so there are 32 samples for every cycle of the sine wave.

7. Connect the Sample output to Input of Fourth Order low pass Filter & observe reconstructed output on (TP46) with help of oscilloscope. The display shows the reconstructed original 1 KHz sine wave.
8. By successive presses of sampling Frequency Selector switch, change the sampling frequency to 2KHz, 4KHz, 8KHz, 16KHz and back to 32KHz (Sampling frequency is 1/10th of the frequency indicated by the illuminated LED). Observe how SAMPLE output changes in each cases and how the lower sampling frequencies introduce distortion into the filter's output waveform. This is due to the fact that the filter does not attenuate the unwanted frequency component significantly. Use of higher order filter would improve the output waveform.
9. So far, we have used sampling frequencies greater than twice the maximum input frequency.

### **CONCLUSION:-**

As the sampling frequency increases the output of sample port has more number of samples of applied input signal.

### **B. Setup of Nyquist criteria and aliasing:-**

Initial set up of trainer: Duty cycle selector switch position: Position 5

Sampling selector switch: Internal position.

1. Keep the power switch in 'Off' position.
2. Connect 2 volts peak, 2 KHz sine wave from 600 ohms output of the Function Generator to the signal Input of the trainer.
3. Switch 'On' the trainer's power supply & Oscilloscope.
4. Connect BNC connector to the CRO and to the trainer's output port.
5. Select 320 KHz (Sampling frequency is 1/10th of the frequency indicated by the Illuminated LED) sampling rate with the help of sampling frequency selector Switch.
6. Connect the sample output to fourth order low pass filter & observe the output (TP46) on oscilloscope. Observe the two waveforms (applied input signal & filter output) which are similar but the second waveform (filter output) is lagging in phase. This is as expected from filters phase/ frequency response.
7. Decrease the sampling rate from 32 KHz to 2 KHz. Observe the distorted waveform at filter's output (TP46). This is due to the fact that we under-sampled the input waveform overlooking the Nyquist criteria and thus the output was distorted even though the signal lies below the cut-off frequency (3.4 KHz) of the filter. This explains the phenomena of Aliasing.

### **RESULT:-**

As the input sampling frequency is smaller than the applied input signal then the output is distorted means the original signal cannot be reconstructed.

## **EXPERIMENT.NO:2**

### **GENERATION OF AMPLITUDE MODULATION AND DEMODULATION**

**AIM:** To study the function of Amplitude Modulation & Demodulation (under modulation, perfect modulation & over modulation) and also to calculate the modulation index, efficiency,

#### **APPARATUS REQUIRED:-**

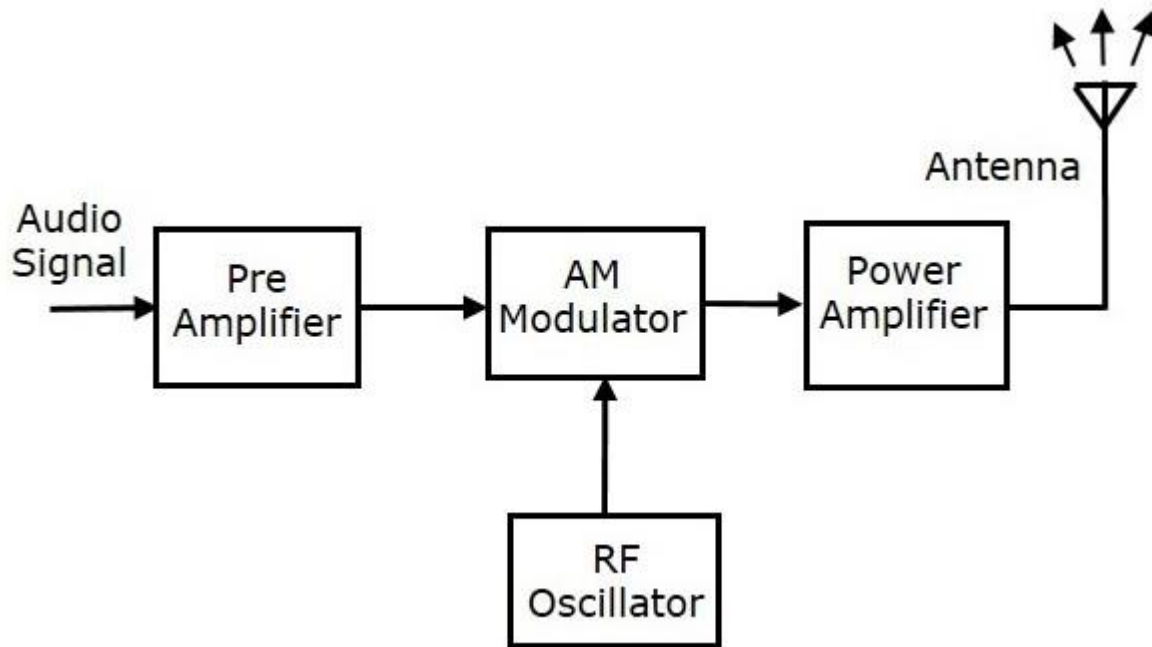
1. ACL-01 Amplitude & ACL-02 Amplitude with power supply cord.
2. CRO with connecting probe.
3. Connecting cords.

#### **THEORY:-**

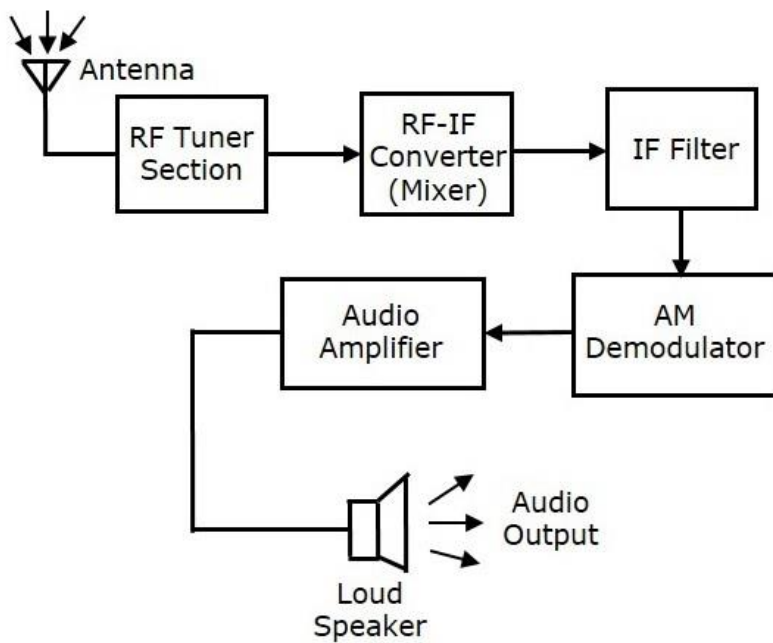
Most digital modulation systems are based on pulse modulation. It involves variation of a pulse parameter in accordance with the instantaneous value of the information signal. This parameter can be amplitude, width, repetitive frequency etc. Depending upon the nature of parameter varied, various modulation systems are used. Pulse amplitude modulation, pulse width modulation, pulse code modulation are few modulation systems cropping up from the pulse modulation technique. In pulse amplitude modulation (PAM) the amplitude of the pulses are varied in accordance with the modulating signal. In true sense, pulse amplitude modulation is analog in nature but it forms the basis of most digital communication and modulation systems. The pulse modulation systems require analog information to be sampled at predetermined intervals of time. Sampling is a process of taking the instantaneous value of the analog information at a predetermined time interval. A sampled signal consists of a train of pulses, where each pulse corresponds to the amplitude of the signal at the corresponding sampling time. The signal sent to line is modulated in amplitude and hence the name Pulse Amplitude Modulation (PAM).

**Natural sampling:-** In the analogue-to-digital conversion process an analogue waveform is sampled to form a series of pulses whose amplitude is the amplitude of the sampled waveform at the time the sample was taken. In natural sampling the pulse amplitude takes the shape of the analogue waveform for the period of the sampling pulse as shown in figure.

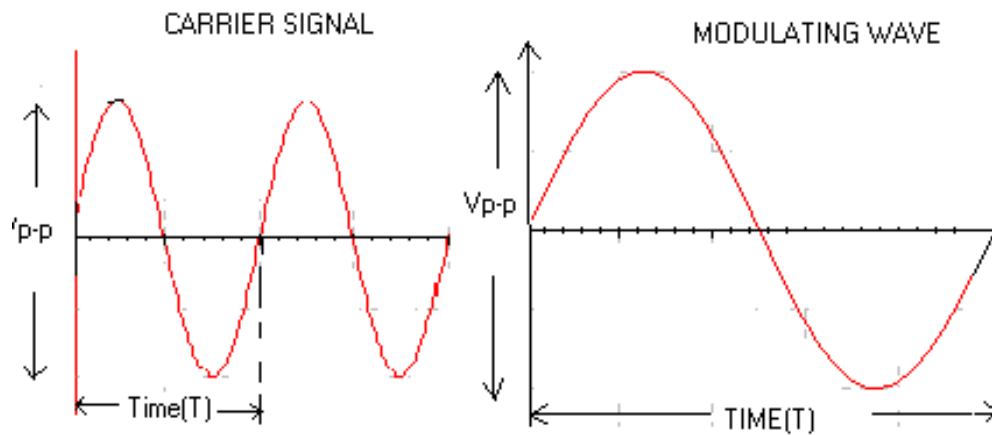
#### **BLOCK DIAGRAM OF AM TRANSMITTER:**



**BLOCK DIAGRAM OF AM RECEIVER:**

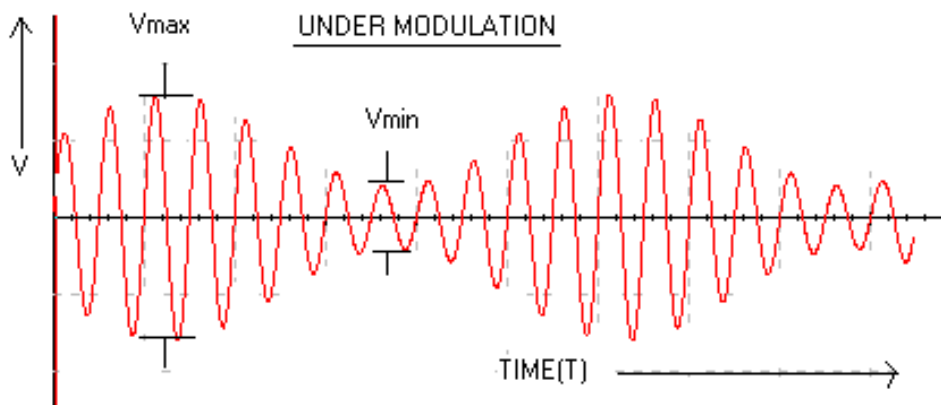


**EXPECTED WAVE FORMS**

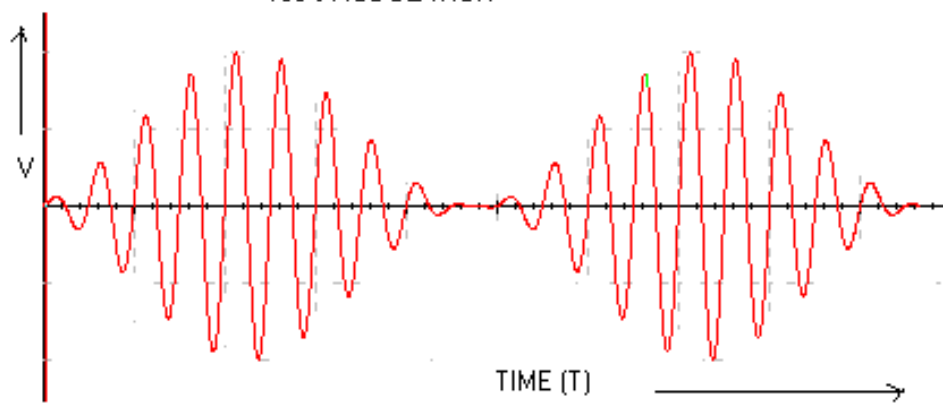


**PROCEDURE:-**

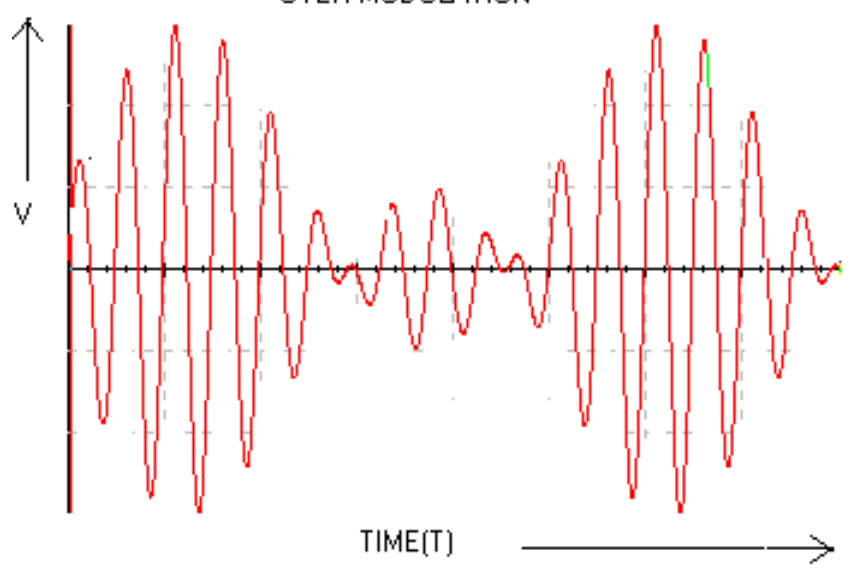
1. Connect the circuit as shown in Figure.
  - a. Output of sine wave to modulation signal input in PAM block keeping the switch in 1 KHz position.
  - b. 8 KHz pulse output to pulse input.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the outputs at TP(3 & 5) these are natural & flat top outputs respectively.
4. Observe the difference between the two outputs.
5. Vary the amplitude potentiometer and frequency change over switch & observe the effect on the two outputs.
6. Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.



PERFECT MODULATION/  
100% MODULATION

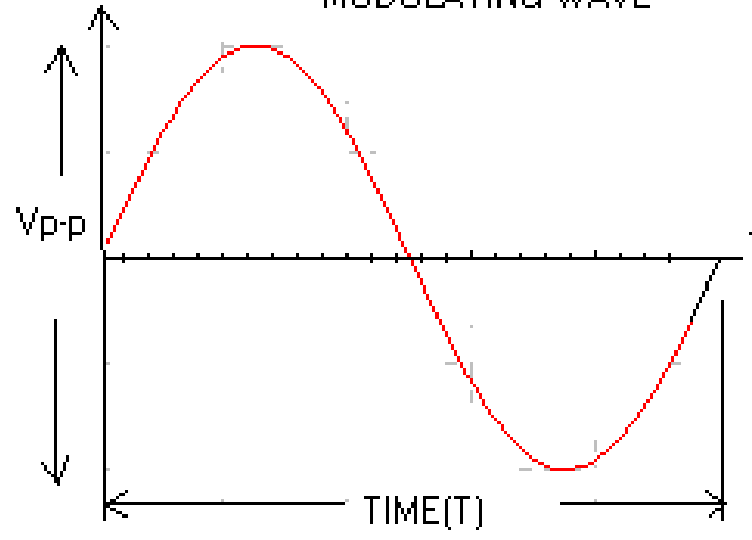


OVER MODULATION



**Demodulated signal**

MODULATING WAVE





**OBSERVATIONS:**

**MODULATION:**

MODULATIONS	$V_c$	$V_m$	$V_{max}$	$V_{min}$	$\mu = \frac{(V_{max} - V_{min})}{(V_{max} + V_{min})}$	$\mu = \frac{V_m}{V_c}$
Under modulation						
Perfect modulation						
Over modulation						

**DEMODULATION:**

Modulating signal frequency	Demodulated signal frequency

**RESULT:**

The amplitude modulated wave is observed for different modulation indexes. In amplitude modulation by increasing the message amplitude we observed different modulation indexes such as under modulation ( $\mu < 1$ ), over modulation ( $\mu > 1$ ) and exact modulation ( $\mu = 1$ ).

**EXPERIMENT NO: 3**  
**FREQUENCY MODULATION & DEMODULATION**

**AIM:**

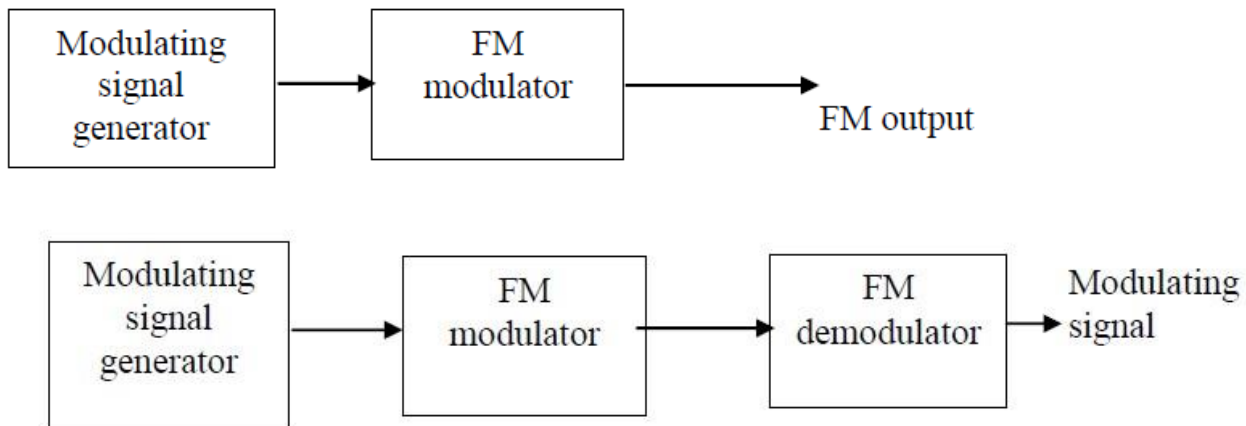
To study the functioning of frequency modulation & demodulation and to calculate the modulation index.

**APPARATUS:**

1. ACL-03 Frequency modulation & ACL-04 demodulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting wires

**BLOCK DIAGRAM:**

**Frequency modulator & De modulator**



**THEORY:**

FM is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal.

By the definition of FM, the amount by which the carrier frequency is varied from its unmodulated value, called the deviation, is made proportional to the instantaneous amplitude of the modulating voltage. The rate at which this frequency variation changes or takes place is equal to the modulating frequency.

FM is that form of angle modulation in which the instantaneous frequency  $f_i(t)$  is varied linearly with the message signal  $m(t)$ , as

$$f_i(t) = f_c + k_f m(t)$$

The term  $f_c$  represents the frequency of the unmodulated carrier, and the constant  $K_f$  represents the frequency sensitivity of the modulator expressed in Hertz per volt.

Unlike AM, the spectrum of an FM signal is not related in a simple manner to that of modulating signal, rather its analysis is much more difficult than that of an AM signal

**PROCEDURE:**

1. Switch on the experimental board.
2. Observe the FM modulator output without any modulator input which is the carrier signal and note down its frequency and amplitude.
3. Connect modulating signal to FM modulator input and observe modulating signal and FM output on two channels of the CRO simultaneously.
4. Adjust the amplitude of the modulating signal until we get less distorted FM output.

**OBSERVATIONS:      MODULATION**

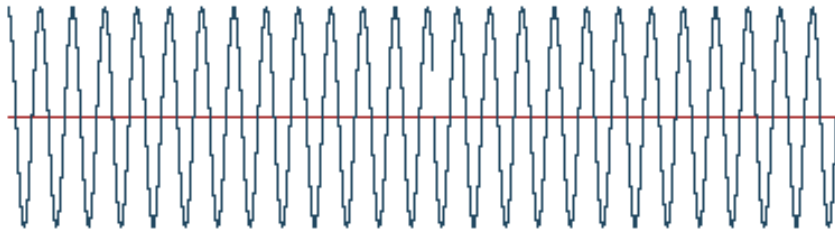
$V_m$	$F_1$	$F_2$	Frequency deviation $\Delta f = F_1 - F_2$	Modulating index $\beta = \frac{\Delta f}{F_m}$	Bandwidth $2(\beta+1)f_m$

**DEMODULATION**

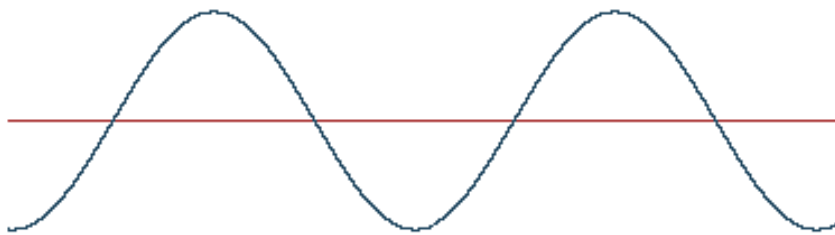
Modulating Signal frequency	Demodulating signal frequency

**EXPECTED WAVE FORM:**

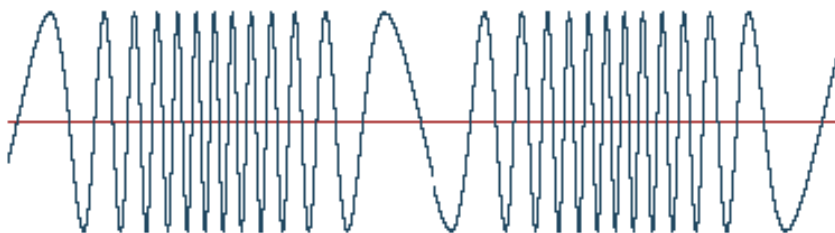
Carrier



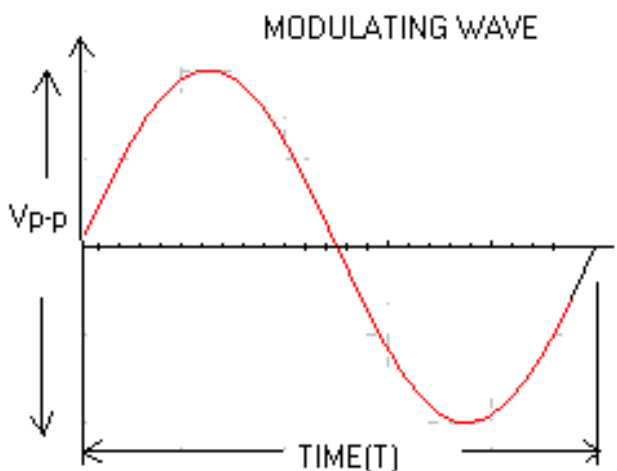
Modulating Wave



Modulated Wave



**Demodulated signal**



**RESULT :**

The Frequency modulated wave is observed for different modulation indexes. In frequency modulation by increasing the message amplitude we observed different modulation indexes such as under modulation ( $\mu < 1$ ), over modulation ( $\mu > 1$ ) and exact modulation ( $\mu = 1$ ).

## EXPERIMENT NO:4 PRE-EMPHASIS AND DE-EMPHASIS

**AIM:**

**To Construct and Verify Pre-emphasis and De-emphasis and Plot the Waveforms.**

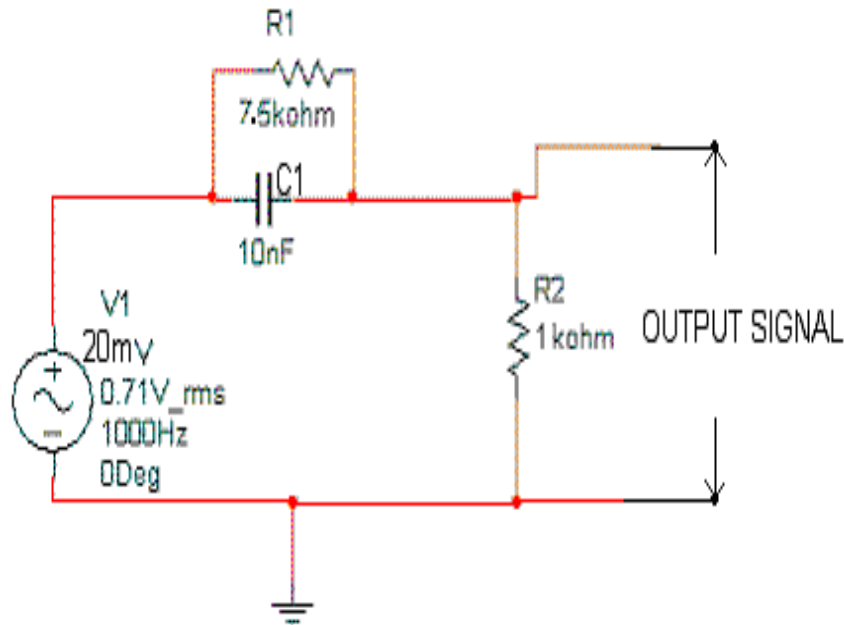
**APPARATUS REQUIRED:**

1. Resistors (10 K-2, 47K, 75K, 1K)
2. Capacitors (22 $\mu$ F, 0.1 $\mu$ -2,)
3. Transistor BC107
- 4 Function generators
5. CRO
6. Connecting Wires
7. RPS (15V)

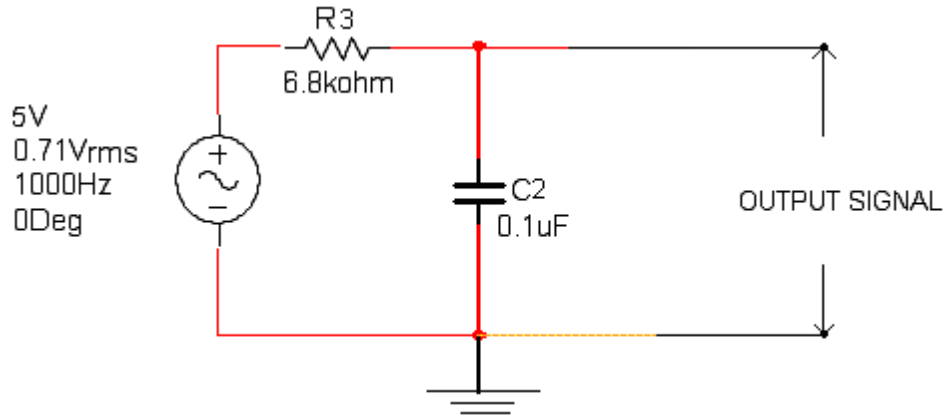
**THEORY:**

The noise has a effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the SNR ratio. This boosting of the higher modulating frequencies at the transmitter is known as pre-emphasis and the compensation at the receiver is called de-emphasis

**CIRCUIT DIAGRAM:**



**Pre-Emphasis:**



**De-Emphasis:**

**PROCEDURE:**

1. Connect the circuit as per circuit diagram as shown in Fig.
2. Apply the sinusoidal signal of amplitude 20mV as input signal to pre emphasis circuit.
3. Then by increasing the input signal frequency from 500Hz to 20 KHz, observe the output voltage (VO) and calculate gain  $20 \log (v_o/v_i)$ .
4. Plot the graph between gain Vs frequency.
5. Repeat above steps 2 to 4 for de-emphasis circuit (shown in Fig.2). by applying the sinusoidal signal of 5V as input signal.

**TABULAR FORM:**

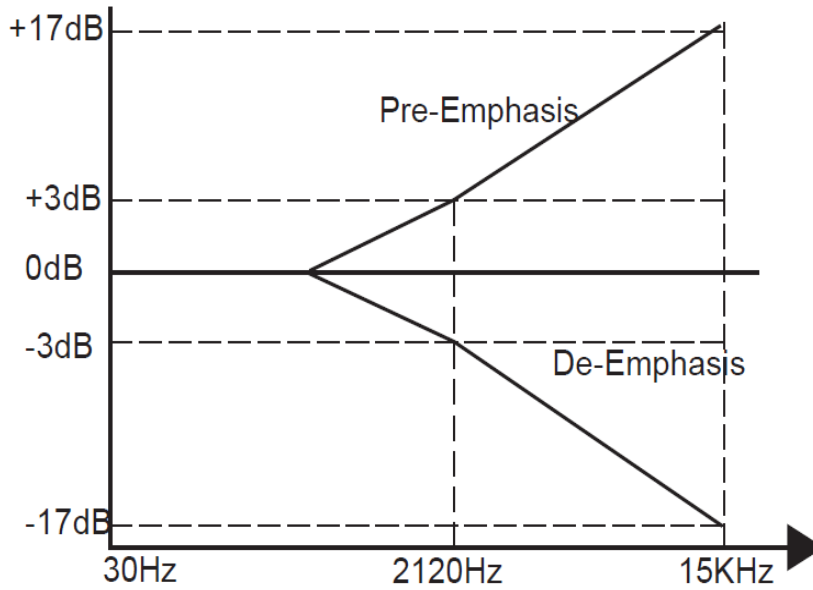
**PRE-EMPHASIS:**

S.NO.	Frequency (Hz)	I/P Voltage $V_i$	O/P Voltage $V_o$	Gain in dB $20\log(V_o / V_i)$

**DE-EMPHASIS:**

S.NO.	Frequency (Hz)	I/P Voltage $V_i$	O/P Voltage $V_o$	Gain in dB $20\log(V_o / V_i)$

**MODEL GRAPH**



**75-μs Emphasis Curves**

**RESULT:-**

Thus the pre-emphasis and de-emphasis characteristics are studied

## EXPERIMENT NO: 5

### PULSE AMPLITUDE MODULATION (PAM) & PULSE POSITION MODULATION

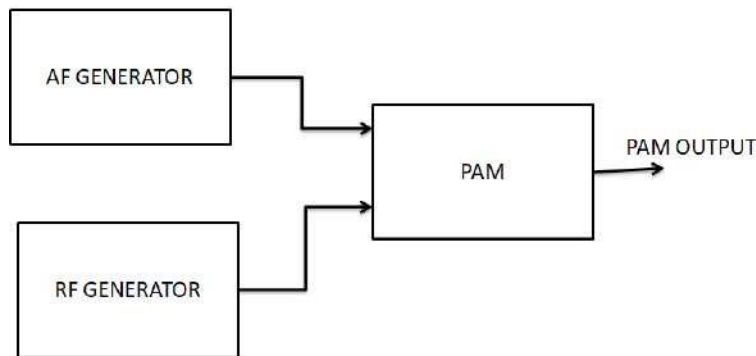
#### AIM:

To study Pulse Amplitude modulation and demodulation process with Relevant waveforms.

#### APPARATUS:-

1. Pulse amplitude modulation & demodulation Trainer Kit.
2. Dual trace CRO.
3. Patch chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

#### BLOCK DIAGRAM: PULSE AMPLITUDE MODULATION



#### THEORY:

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system 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 cables are used to modulated carrier.

The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM,

in which a fixed dc level is added to the signal to ensure that the pulses are always



positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short.

Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform.

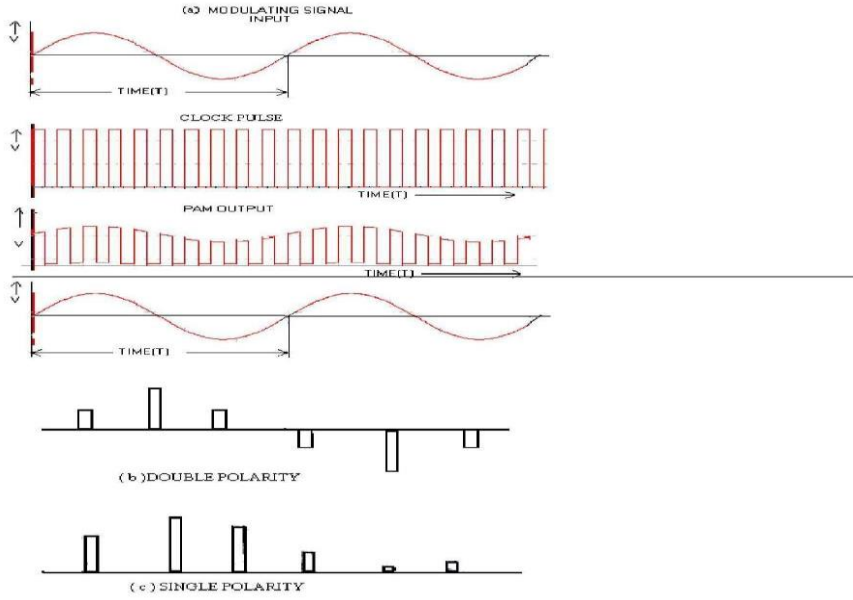
Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave.

PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which is used frequently as an intermediate step in other pulsemodulating methods, especially where time-division multiplexing is used.

### **PROCEDURE:**

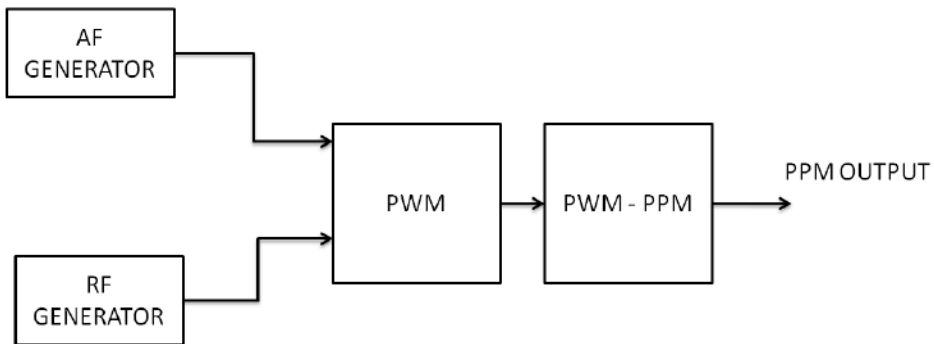
1. Switch "ON" the experimental kit.
2. Observe the clock generator output & modulation signal outputs.
3. Connect clock generator output to the clock input point of PWM modulator And observe the same clock on channel of a dual trace CRO.
4. Trigger the CRO with respect to CH 1.
5. Apply a variable DC voltage of 8 to 12 volts from any external regulated Power supply.
6. Observe the PWM output on CH 2.
7. If we observe the PWM output, it's width varies according to the Modulating voltage.
8. A variable amplitude modulating signal is given to observe how the PWM are varying for AC modulating voltages.
9. In this case we have to trigger the CRO with respect to modulating voltage.

EXPECTED WAVE FORMS:

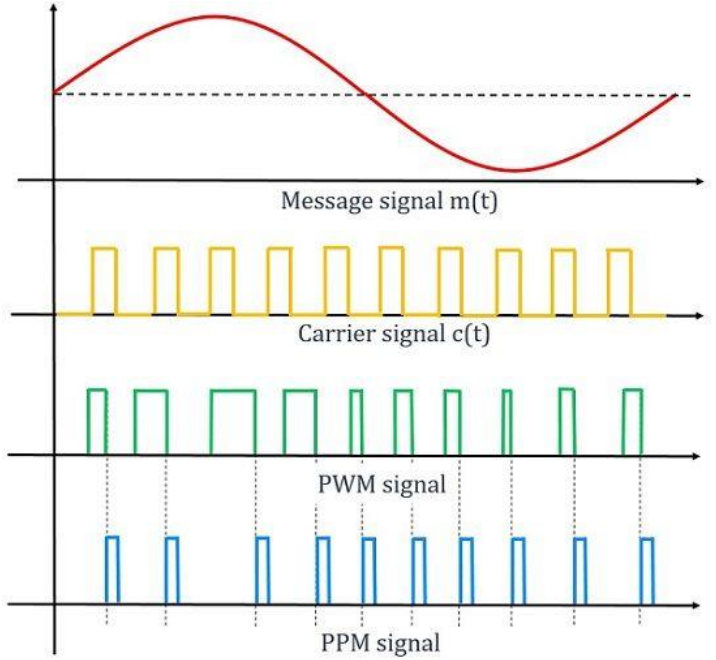


**BLOCK DIAGRAM:**

**PPM MODULATION**



**EXPECTED WAVE FORMS**



**Waveform representation of PPM signal generation**

Electronics Coach

**RESULT:**

The PAM AND PPM wave forms are generated and plotted.

## EXPERIMENT NO:6 GENERATION OF ASK AND FSK

### AIM:

To Generate ASK and FSK

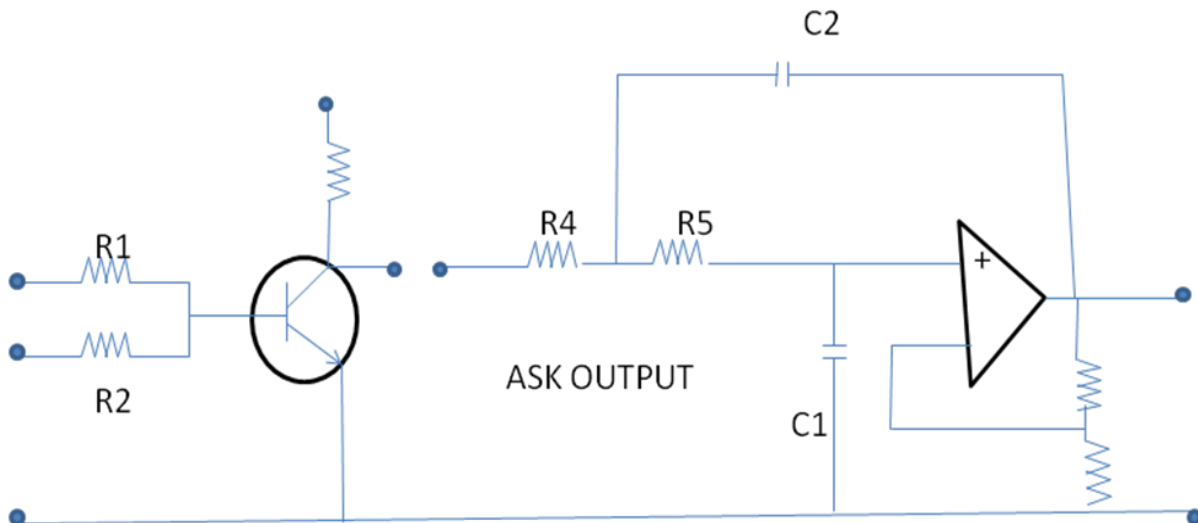
### APPARATUS REQUIRED:

ASK kit, connecting probes, CRO.

### THEORY:

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

### CIRCUIT DIAGRAM:-



### PROCEDURE:

#### FOR ASK MODULATION:

1. Connect the output of carrier wave generator (carrier output) to carrier input of ASK modulator & connect the modulating output to modulating input through patchcords.
2. Connect the CRO across output of ASK modulator.
3. Switch on the CRO as well as instrument.
4. Observe the output wave shape on CRO.

5. Change the amplitude & frequency of modulating signal & observe corresponding effect on ASK modulated wave. set the output shape (Ask modulated) through offset potentiometer provided on the front panel.

## PROCEDURE

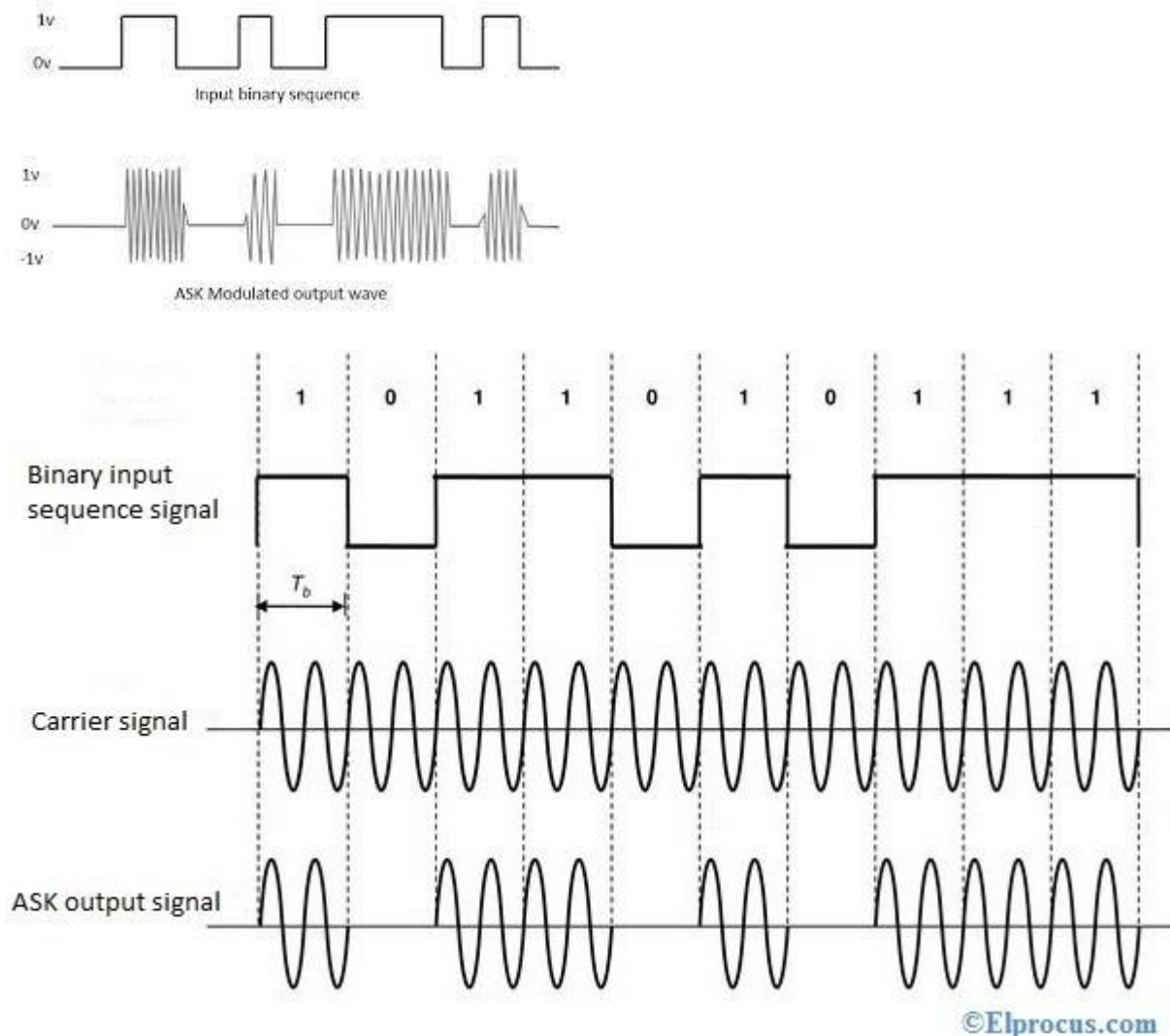
### FOR ASK DEMODULATION:

1. Connect the output of ASK modulator to the input of demodulator through patchcord.
2. Connect CRO probe across output of demodulator.
3. Observe the demodulated (square wave) shape on CRO. We will observe that the demodulated output is of same frequency as the modulated signal with little distortion.

### PRECAUTIONS:

1. Check the continuity of the connecting probes.
2. Handle the CRO properly.

### EXPECTED WAVE FORMS:



### (a) Study of Frequency Shift Keying.

#### APPARATUS REQUIRED:-

Data generator, FSK modulation kit, DSO and connecting leads.

#### THEORY:-

FSK is one of the basic modulation techniques for the transmission of digital data .If the frequency of the sinusoidal carrier is switched depending upon the input digital signal , then it is known as frequency shift keying. As the amplitude remains constant in FSK, so the effect of non-linearities, noise interference is minimum on digital detection. So FSK is preferred over ASK.

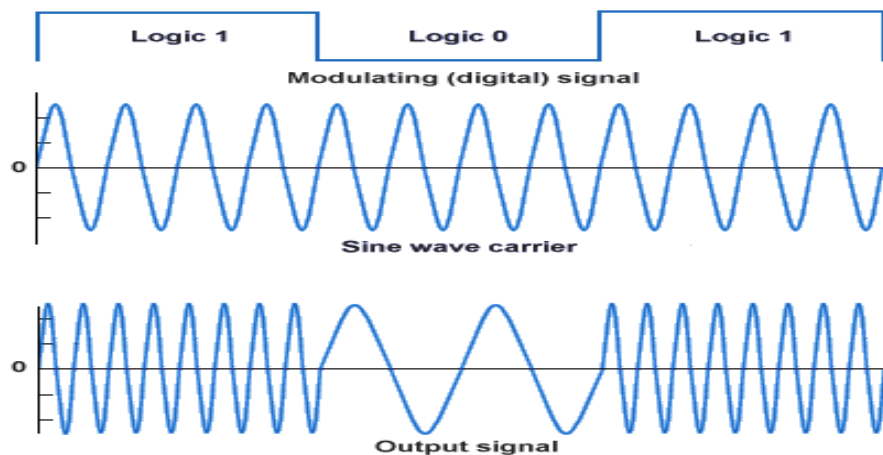
Frequency shift keying consists of shifting of frequency of carrier from a mark frequency to a space frequency according to the base band digital signal Frequency shift keying is identical to modulating an FM carrier with a binary digital signal In an FSK system, two sinusoidal carrier waves of the same amplitude  $A_c$  but different frequencies  $f_{c1}$  and  $f_{c2}$  are used to represent binary symbols 1 and 0 respectively. It can be easily verified that binary FSK waveform is a superposition of two binary ASK waveforms, one with a frequency  $f_{c1}$  and other with a frequency  $f_{c2}$ . No discrete components appear in the signal spectrum of FSK signal. The main advantage of FSK lies in its easy hardware implementation.

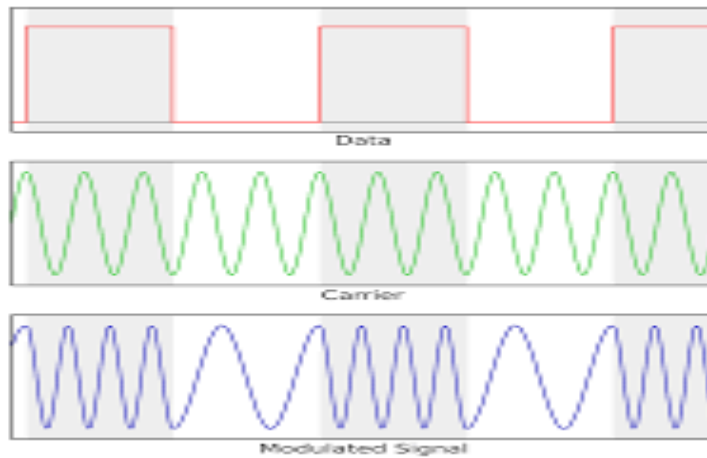
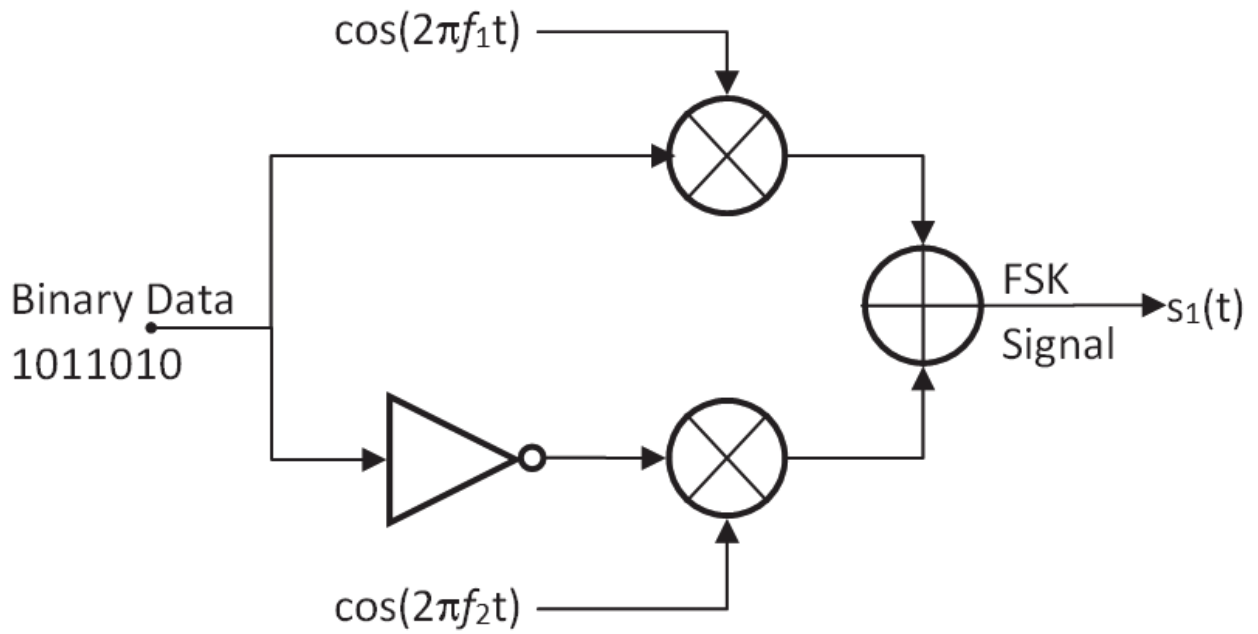
#### Generation of FSK signal:-

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

#### Detection of FSK signal:-

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





**RESULT:-**

The ASK and FSK output is obtained on DSO.

## EXPERIMENT NO:7

### GENERATION OF PSK AND QPSK

**AIM :**

**TO GENERATE AND PLOT OF PSK AND QPSK**

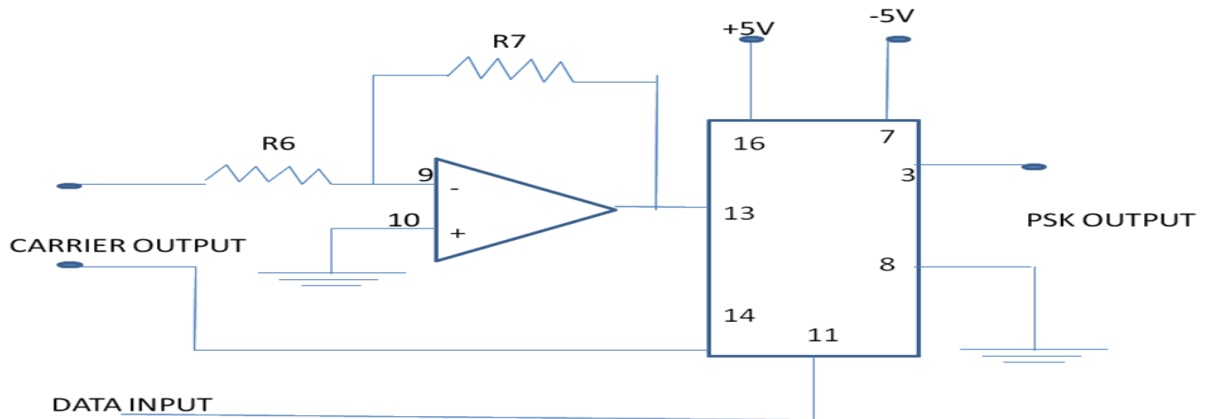
**APPARATUS REQUIRED:-**

PSK Kit , connecting probes, CRO.

**THEORY:-**

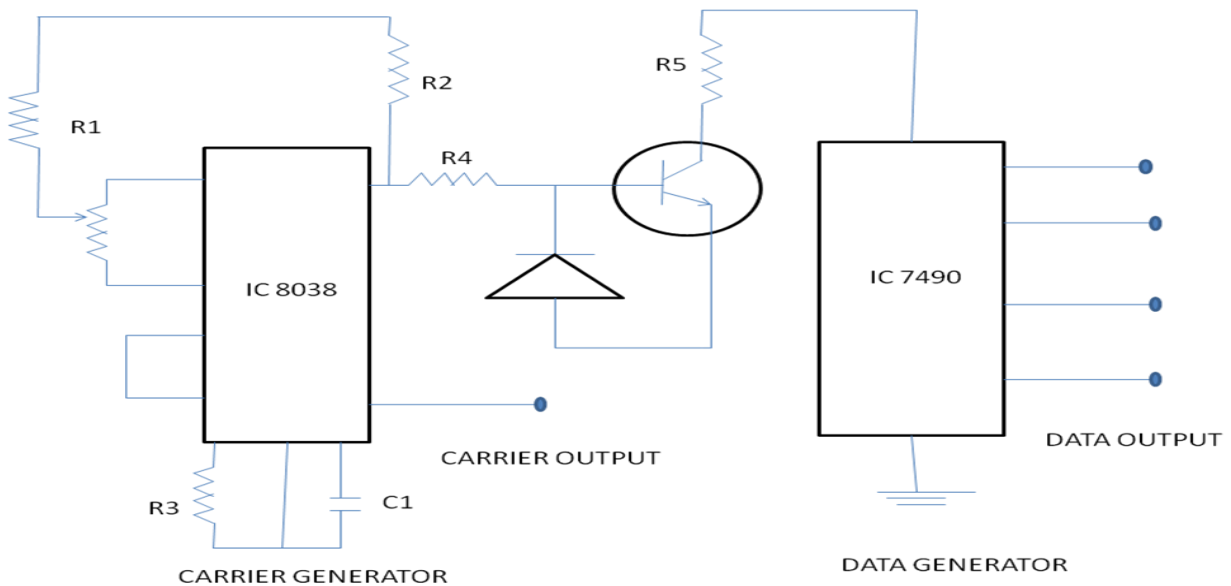
Digital communication became important with the expansion of the use of the computers and data processing and have continue to develop in to a major industry providing the interconnection of computer peripherals and transmission of data between distance sides. PSK is a relatively new system in which carrier may be phase shifted by  $90^\circ$  for a mark and by minus  $90^\circ$  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 modulating square wave.

**CIRCUIT DIAGRAM:-**



PSK MODULATOR





### PROCEDURE:-

#### FOR PSK MODULATION:-

1. Connect the carrier Output of the carrier wave generator (IC 8038) to carrier input of PSK modulator through patch cords. Also connect any data output from the data outputs of the data generator to the data input of the PSK modulator.
2. Connect the channel 1 of CRO across output of PSK. And channel 2 across data input on a dual trace oscilloscope.
3. Switch on the instrument using ON/OFF toggle switch provided on the front panel.
4. Observe the output wave shape on CRO.
5. Change the data inputs and observe the PSK output on CRO.

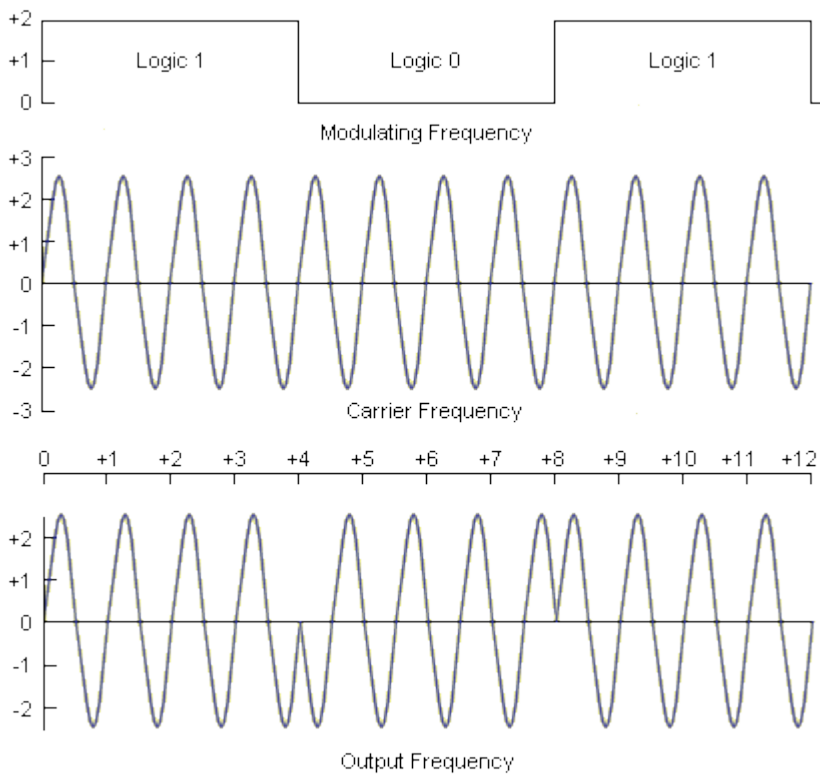
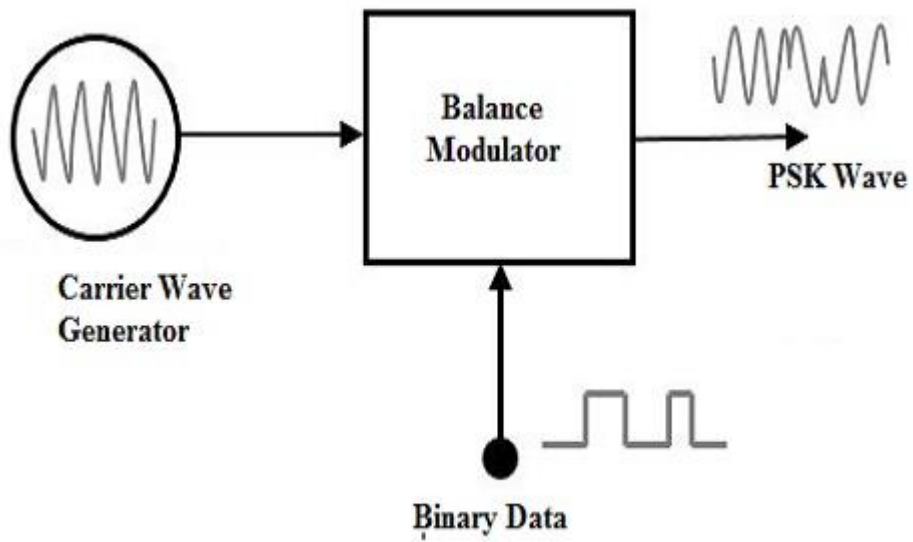
#### FOR PSK DEMODULATION:-

1. Connect the output of PSK to the demodulator input through patch cord.
2. Connect the carrier to the carrier input of the PSK demodulator.
3. Connect the channel 1 of CRO across output of demodulator.
4. Observe the demodulated output on CRO (it will be same as data input applied in the modulated in the modulator section).

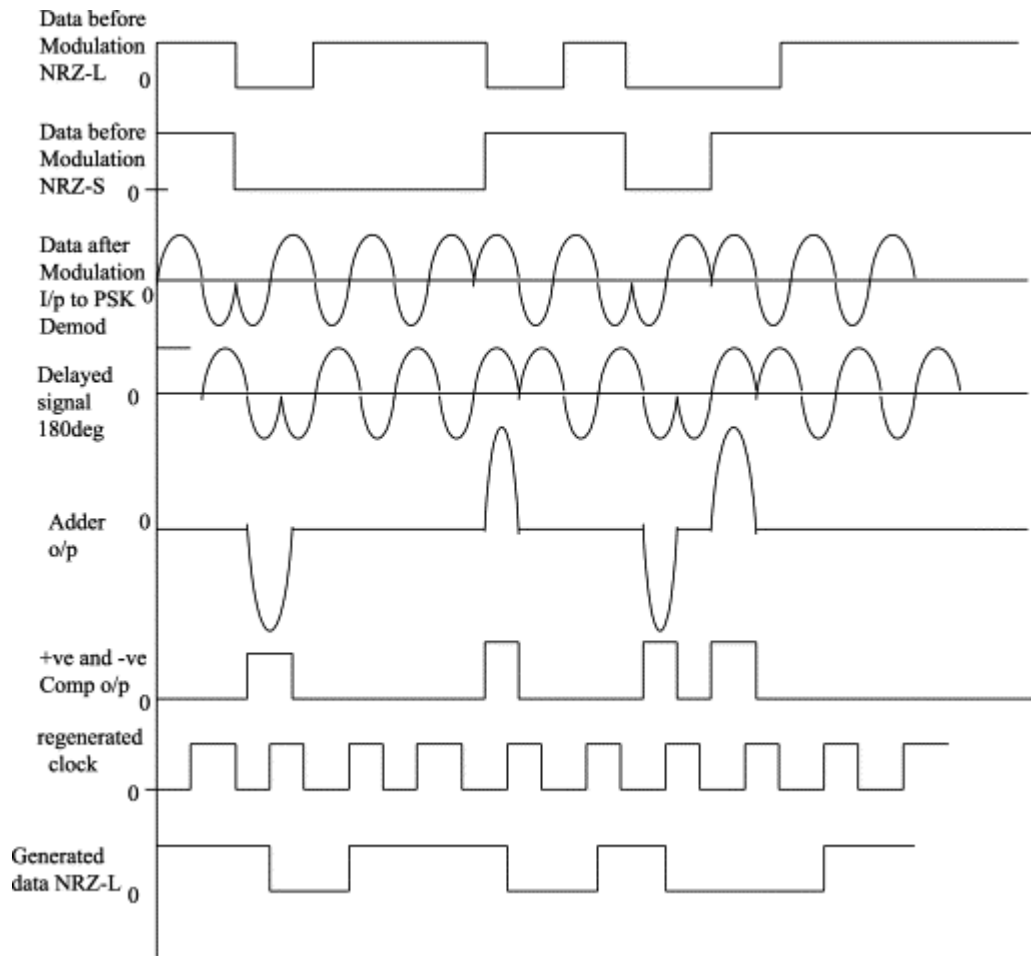
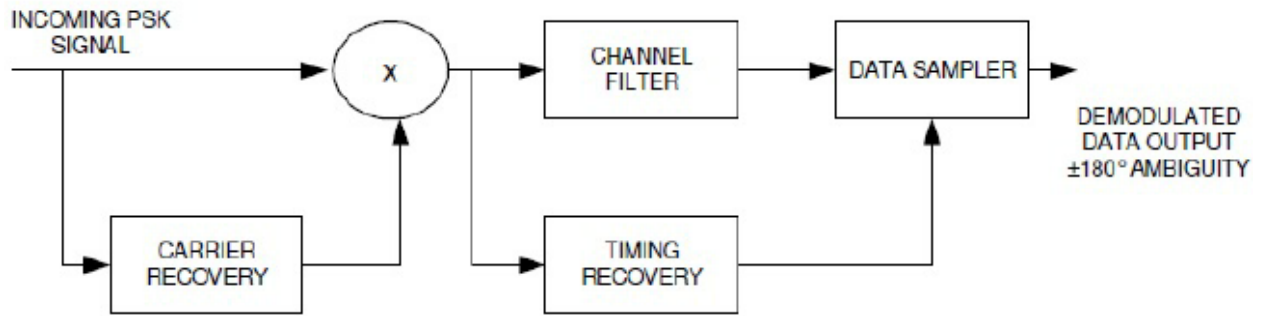
### PRECAUTIONS:-

1. Check the continuity of the connecting probes.
2. Handle the CRO properly.

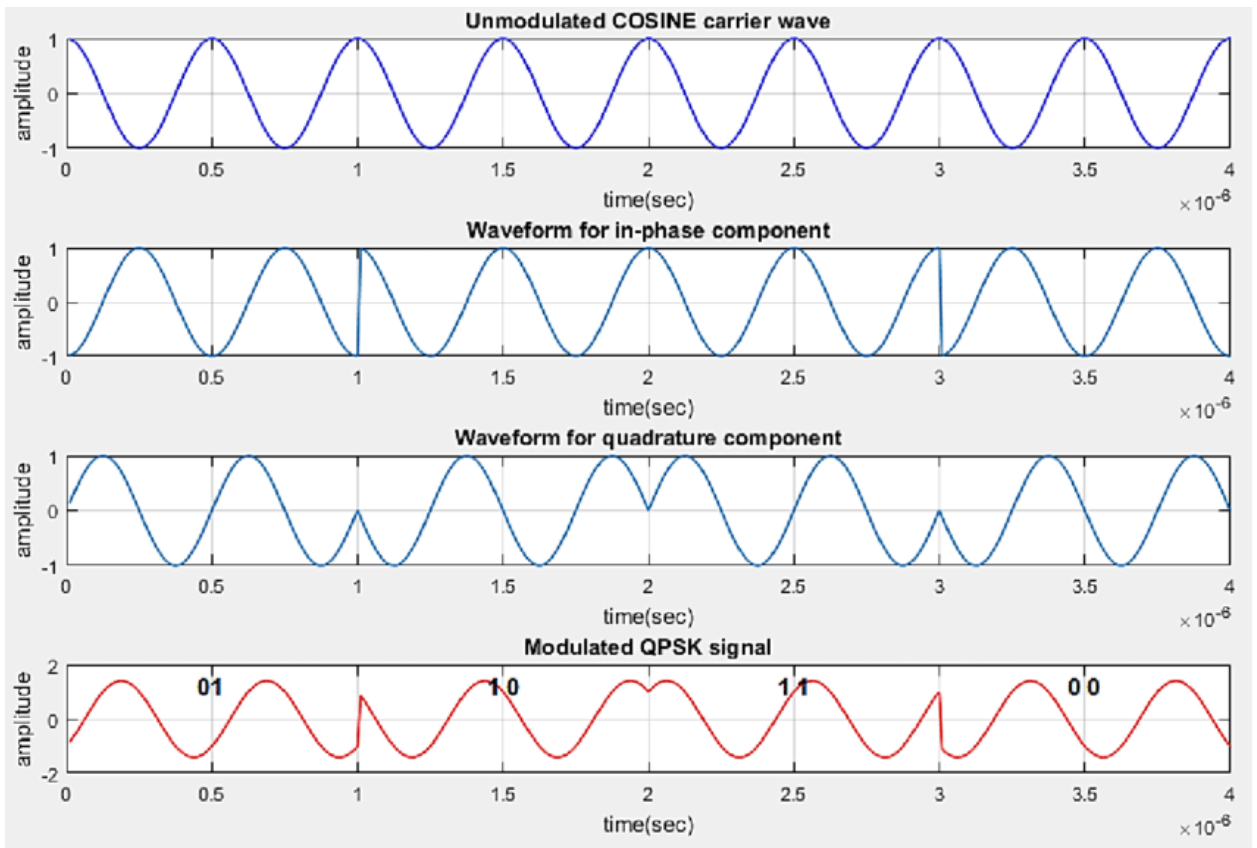
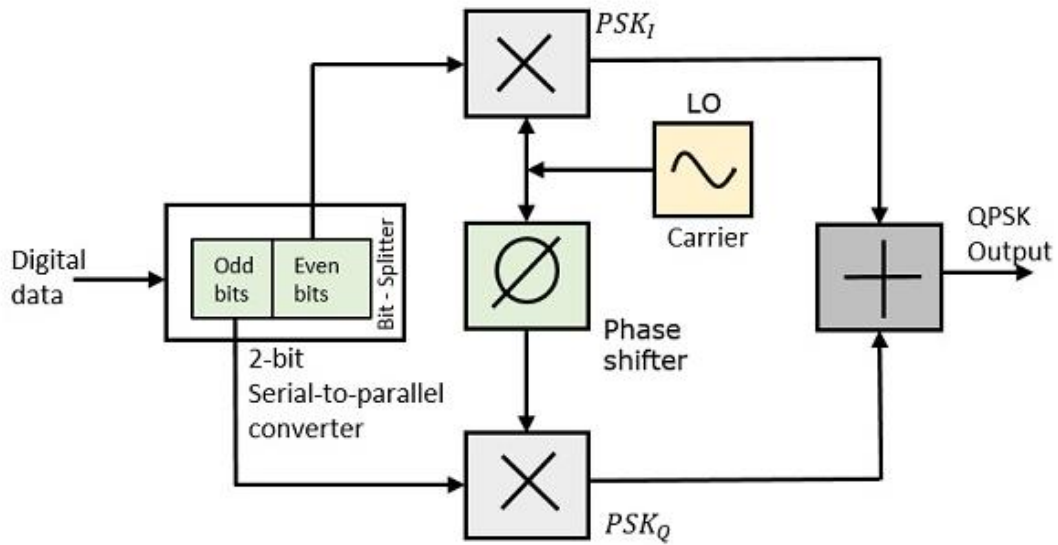
## PSK MODULATION:



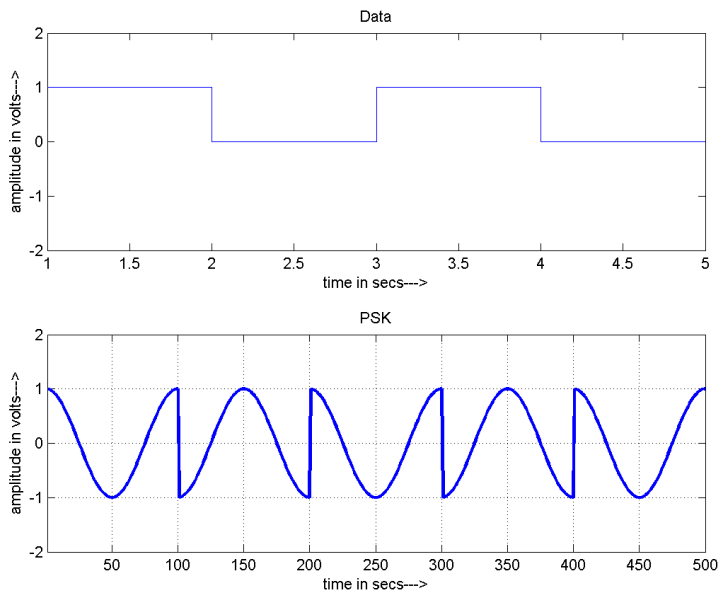
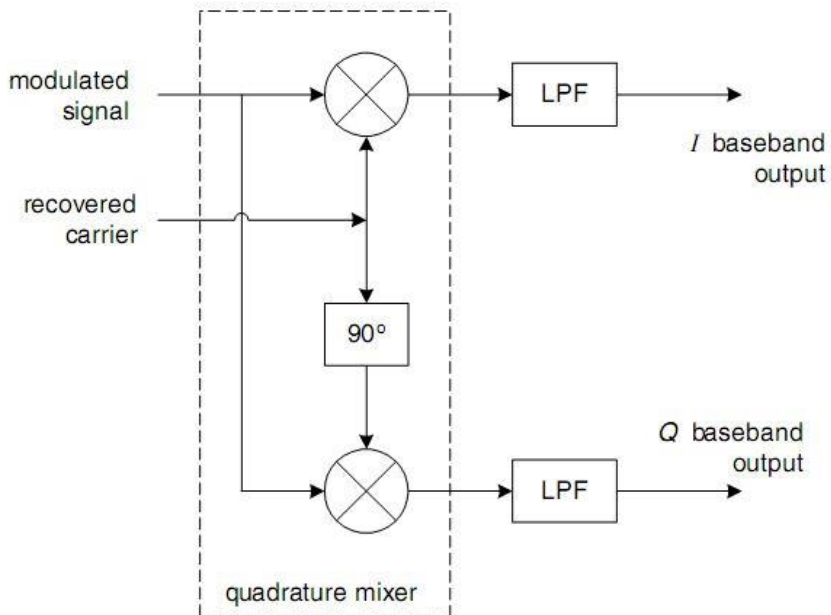
## PSK DEMODULATION:



## QPSK MODULATOR:



## QPSK DEMODULATOR:



## RESULT:

PSK AND QPSK are successfully generated.

**EXPERIMENT NO:8**  
**GENERATION OF PN SEQUENCES AND DIRECT SEQUENCE SPREAD SPECTRUM**  
**SPECTRUM**

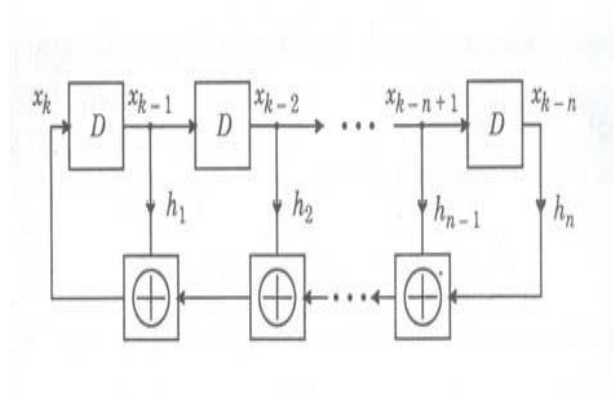
**AIM:**

**To Generate PN SEQUENCES AND DIRECT SEQUENCE SPREAD SPECTRUM.**

**PN Sequence Generation**

**Theory:**

Pseudo-Noise (PN) sequences are commonly used to generate noise that is approximately "white". It has applications in scrambling, cryptography, and spread-spectrum communications. It is also commonly referred to as the Pseudo-Random Binary Sequence (PRBS). These are very widely used in communication standards these days. The qualifier "pseudo" implies that the sequence is not truly random. Actually, it is periodic with a (possibly large) period, and exhibits some characteristics of a random white sequence within that period. PN sequences are generated by Linear Feedback Shift Registers (LFSR), as shown in the following figure:



In the figure, the output  $x_k$  is binary (0 or 1), and so are the constants  $h_j, j=0,1,\dots,n$ , and  $\oplus$  denotes the XOR operation. This means that  $x_k$  is given by:

$$x_k = h_1 x_{k-1} \oplus \dots \oplus$$

$h_n x_{k-n}$  Since  $x_k \oplus x_k = 0$ , it follows from the above that:

$$x_k \oplus h_1 x_{k-1} \oplus \dots \oplus h_n x_{k-n} = 0$$

or

$$h(D)x(D) = 0$$

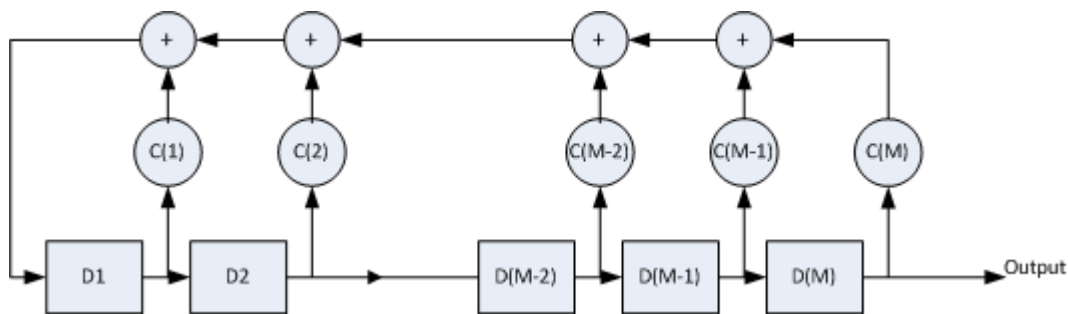
where  $h(D) = 1 \oplus h_1 D \oplus \dots \oplus h_n D^n$  and  $D$  denotes a unit delay ( $x_k D^j = x_{k-j}$  for any  $j$ ). Note

that the “1” in the polynomial does not correspond to a tap.

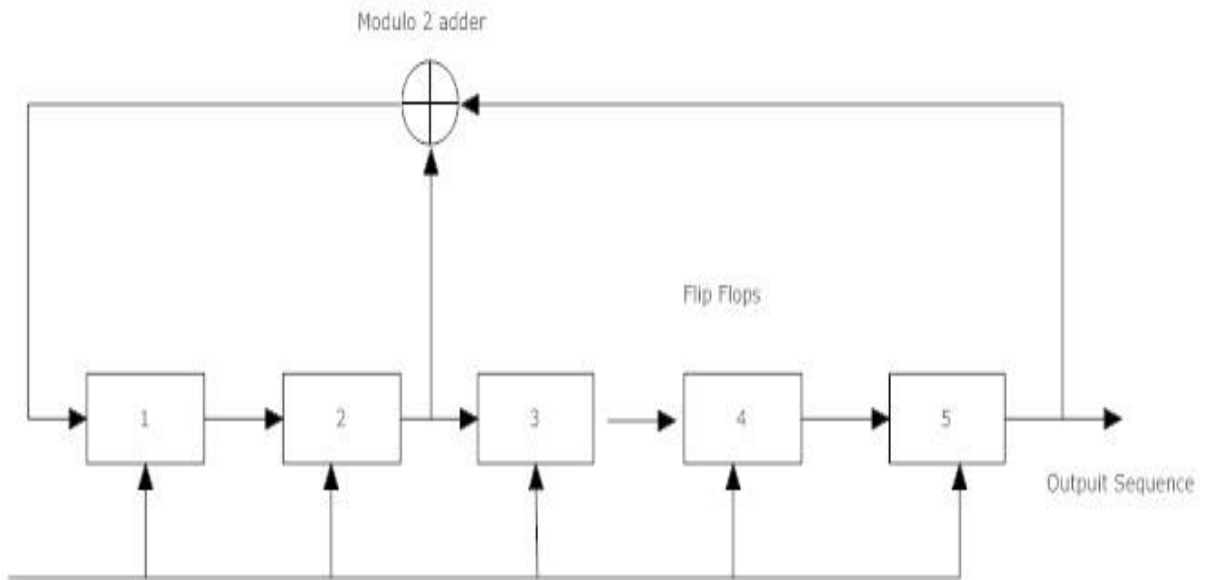
It is not difficult to see that the output  $x_k$  will be periodic. However, the dependence of the length of the period (which we would like to be as large as possible) on the constants  $h_j$ ,  $j=1,2,\dots,n$  is not obvious. We can see that the "state"  $(x_{k-1} \dots x_{k-n})$  can assume at most  $2^n$  values. We note the following:

- If the state of the shift register is all zero at any time, it remains so for all time. We need to ensure that this never happens (we start with a non-zero value).
- If the state ever remains unchanged from one clock cycle to the next, it remains the same forever.
- The sequence must be periodic (since there are at most  $2^n - 1$  states).
- Since all the all zero state is not allowed, the period of the output sequence can be at most  $2^n - 1$ . A feedback shift register that generates a sequence of this period is said to be of *maximal length*.

How do we design the feed-back shift register (i.e.,  $h_j$ ) so that it is maximal length, keeping the hardware (XORs) minimum? The answer to this question involves the divisibility of  $1 \oplus D^m$  by  $h(D)$  for  $m < 2^{n-1}$ , and need not concern us here. The following table lists the tap-points (points where  $h_j$  is 1) for registers of various sizes. The first column lists the order  $n$  of the register, and the second column lists the tap points or  $h(D)$  in octal notation. For example, for a register of size 12, the table lists 10123 in octal or 10000010



Clock Pulse	D3	D2	D1
0	0	0	1
1	1	0	0
2	0	1	0
3	1	0	1
4	1	1	0
5	1	1	1
6	0	1	1
7	0	0	1



Clock

Generator Polynomial is  $x^5 + x^2 + 1$   
 Initial State state ff1(0) ff2(1) ff3(1) ff4(0) ff5(1)

arguments to pass to function (a,b,c)  
 a, no of flip flops = 5  
 b, tap function or generator polynomial = [1 0 0 1 0]  
 c, initial state = [0 1 1 0 1]



n=2	7	14	42103	26	400000107
3	13	15	100003	27	1000000047
4	23	16	210013	28	2000000011
5	45	17	400011	29	4000000005
6	103	18	1000201	30	10040000007
7	211	19	2000047	31	20000000011
8	435	20	4000011	32	40020000007
9	1021	21	10000005	33	-
10	2011	22	20000003	34	-
11	4005	23	40000041	35	-
12	10123	24	100000207	36	-
13	20033	25	200000011		

Minimal weight polynomials of order 2 to 32 are listed in the above table. Each entry in the table is an octal number for each  $n$  in the first column, which when converted to binary specifies the coefficients of  $h(D)$ . The most significant (leftmost) bit  $h_n$  is 1, and so is the least-significant (right-most) bit  $h_0$ . . the tap points for larger length shift registers in various sites on the internet (use this [hyperlink](#) for example)

If you look at any  $n$ -length segment of the output, you will find all possible sequences with the exception of the all zero sequence. However, when you look at a smaller segment, you will come across all possible sequences. Intuitively therefore, when the register length is large, the sequence is approximately "white". Consider for example the case when  $n=2$  and  $h_1=h_2=1$  implying that both points are tapped so that  $x_k=x_{k-1}\oplus x_{k-2}$ . Starting with  $x_{k-1}=0$  and  $x_{k-2}=1$  gives the following sequence of shift register contents: (0,1),(1,0),(1,1),(0,1),... Notice that periodicity is three since  $2^2-1=3$  and this happens to be the tapping to get a maximal length sequence. Note that the shift-register contents are shifted versions of each other and it makes no difference which register output is considered the output.

The output of the PN sequence generator is purely deterministic – given the state of the generator, the output is uniquely determined for all time. With the zero level mapped to a “-1” to make it an antipodal sequences, the autocorrelation of the maximal-length PN sequence is periodic, and its value in one period is

$-1/M$  except at one location where it is 1 ( $M$  is  $2^n-1$ ). This sequence can be filtered to generate bandlimited Gaussian-like noise.

## SPREAD SPECTRUM – DSSS MODULATION & DEMODULATION

### INTRODUCTION:

Recall that when a sinusoidal carrier is DSBSC modulated by a message, the two signals are multiplied together. Recall also that the resulting DSBSC signal consists of two sets of sidebands but no carrier.

When the DSBSC signal is demodulated using product detection, both sidebands are multiplied with a local carrier that must be synchronized to the transmitter's carrier that is, it has the same frequency and phase. Doing so produces two messages that are in phase with each other and so add to form a single bigger message.

Direct sequence spread spectrum is a variation of DSBSC modulation scheme with a pulse train for the carrier instead of a simple sine wave. This may sound radical until you remember that pulse trains are actually made up of a theoretically infinite number of sine waves. That being the case, spread spectrum is really the DSBSC modulation of a theoretically infinite number of sinusoidal carrier signals. The result is a theoretically infinite number of pairs of tiny sidebands about a suppressed carrier.

In practice, not all of these sidebands have any energy of significance. However, the fact that the message information is distributed across so many of them makes spread spectrum signals difficult to deliberately interfere with "jam". To do so, you have to upset a significant number of the sidebands which is difficult considering their number.

Spread spectrum signals are demodulated in the same way as DSBSC signals using a product detector. Importantly, the product detector's local carrier signal must contain all the sine waves that make up the transmitter's pulse train at the same frequency and phase. If this is not done, the tiny demodulated signals will be at the wrong frequency and phase and so they won't add up to reproduce the original message. Instead, they'll produce a garbage signal that looks like noise.

The only way to obtain the right number of sine waves at the right frequency and phase at the receiver is to use a pulse train with an identical sequence to that used by the transmitter. Moreover, it must be synchronized. This issue gives spread spectrum another of its advantages over other modulation schemes. The transmitted signal is effectively encrypted.

Of course, with trial and error it's possible for an unauthorized person to guess the correct PN sequence to use for their receiver. However, this can be made difficult by making the sequence longer before it repeats itself. Longer sequences can produce more combinations of unique codes which would take longer to guess using a trial and error approach. To illustrate this point, an 8 bit code has 256 combinations while a 20 bit code has 1,048,576 combinations. A 256 bit code has  $1.1579 \times 10^{77}$  combinations.

Increasing the sequence's chip-length has another advantage. To explain, the total energy in a spread spectrum signal is distributed between all of the tiny DSBSC that make it up. A mathematical technique called Fourier Analysis shows that the greater the number of chips in a sequence before repeating, the greater the number of sinewaves of significance needed to make it.

That being the case, using more chips in the transmitter's PN sequence produces more DSBSC signals and so the signal's total energy is distributed more thinly between them. This in turn means that the individual signals are many and extremely small. In fact, if the PN sequence is long enough, all of these DSBSC signals are smaller than the background electrical noise that's always present in free space. This fact gives spread spectrum yet another important advantage. The signal is difficult to detect.

Spread spectrum finds use in several digital applications including: CDMA mobile phone technology, cordless phones. The global positioning system and two of the 802.11 Wi-fi standards

Generate a DSSS signal by implementing its mathematical model. You'll then use a product detector (with a stolen carrier) to reproduce the message. Once done, you'll examine the importance of using the correct PN sequence for the local carrier and difficulty of jamming DSSS signals.

#### Equipment

1. Emona Telecoms Trainer 101 (plus power pack)
2. Dual channel 20 MHz oscilloscope
3. Two Emona Telecoms Trainer 101 oscilloscope leads
4. Assorted Emona Telecoms Trainer 101 patch leads.

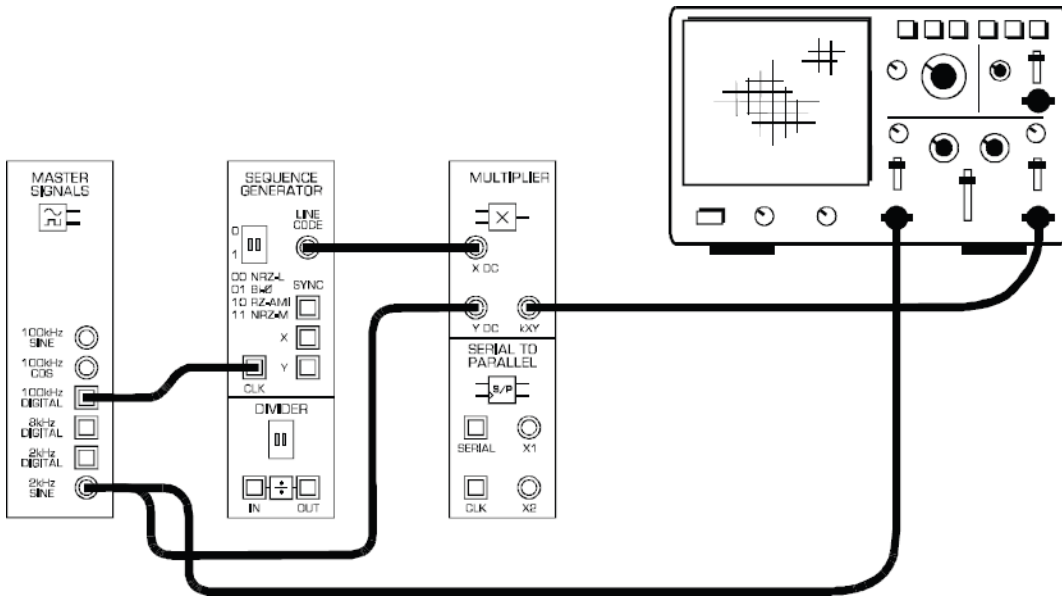
## Procedure:

### Part A

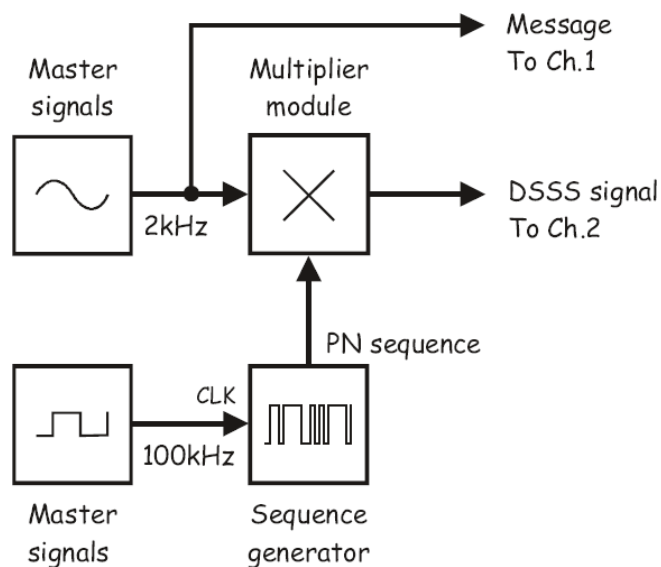
As DSSS is basically just DSBSC with a pulse train for the carrier instead of a simple sinusoid, it can be generated by implementing the mathematical model for DSBSC.

1. Gather a set of the equipment listed on the previous page.
2. Set up the scope per the instructions in experiment 1.
  - a. The Trigger source control is set to the CHI position
  - b. The Moe control is set to the CHI position.
3. Set the scope's Trigger source coupling control to the HF REJ position.
4. Locate the sequence Generator module and set its dip-switches to 00

- a. To do this, push both switches up.
5. Connect the set-up shown in Figure 1 below.
  - a. Note : Insert the black plugs of the oscilloscope leads into a ground (GND) socket.



The set up in FIG 1 can be represented by the block diagram in Fig 2 below. It multiplies the 2 kHz sinewave message with a PN sequence modeled by the sequence Generator's 32 bit pulse train output.

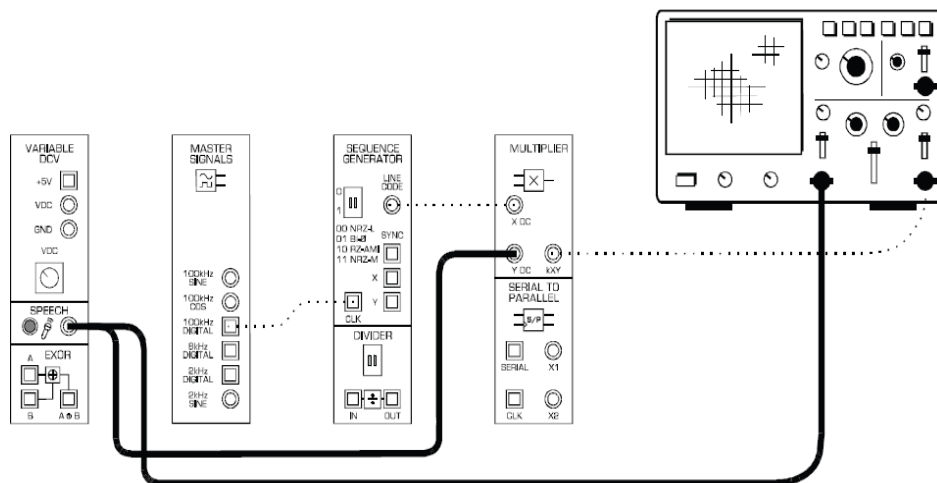


6. Adjust the scope's Time base control to view two or more cycles of the Master signals modules 2 kHz sine output.
7. Set the scope's Mode control to the DUAL position to view the DSSS signal out of the Multiplier Module as well as the message signal.
8. Adjust the scope's Vertical Attenuations controls to the appropriate settings for the signals.
9. Draw the two waveforms to scale in the space provided on the next page leaving room to draw a third waveform.
  - a. Tip: Draw the message signal in the upper third of the graph and DSSS signal in the middle third.
10. Use the scope's channel 1 Vertical position control to overlay the message with the DSSS signal's envelope's and compare them.

## Part B – Generating a DSSS signal using speech.

So far , this experiment has generated a DSSS signal using a sinewave for the message. The next part of the experiment lets you see what a DSSS signal looks like when modulated by speech.

11. Disconnect the plugs to the Master signals module's 2kHz sine output.
12. Connect them to the speech module's output as shown in fig 3 below.
  - a. Remember Dotted lines show leads already in place.

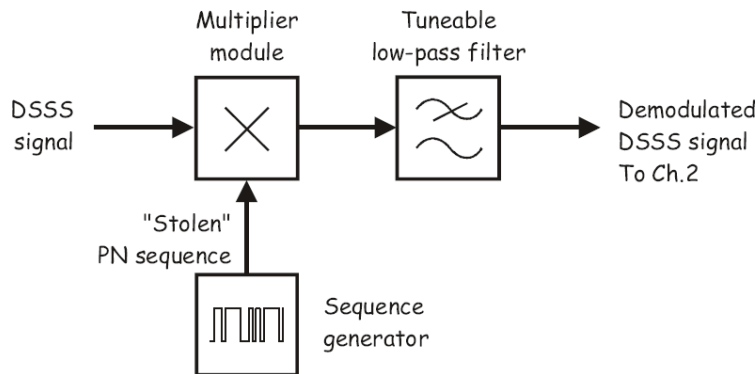
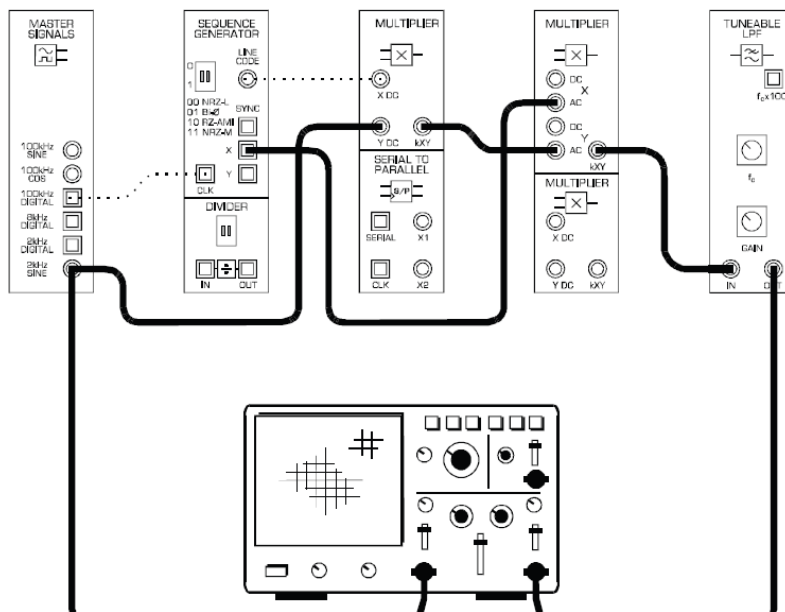


13. Set the scope's time base control to the 2ms/div position.
14. Talk, sing or hum while watching the scope's display.

## Part C

Using the product detector to recover the message.

15. Return the scope's Time base control to its original position.
16. Locate the Tunable low pass filter module and set its Gain control to about the middle of its travel.
17. Turn the tunable low pass filter modules cut-off frequency adjust control fully anti-clockwise.
18. Disconnect the plugs to the speech module's output and modify the set-up as shown in Fig 4 below

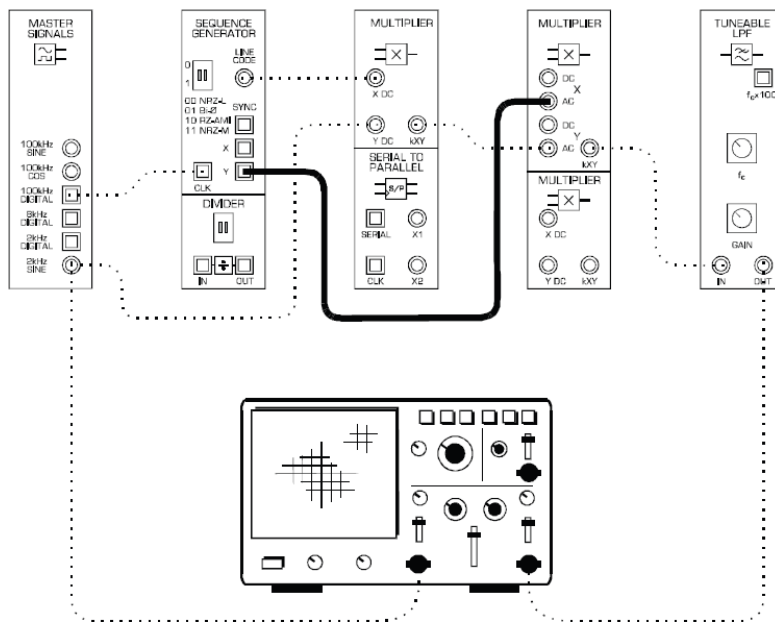


19. Slowly turn the Tunable low-pass Filter module's cut – off Frequency control clockwise while watching the scope's display and stop when it's at about half its travel.

20. Draw the demodulated DSSS signal to scale in the space that you left on the graph paper.

Recall that the message can only be recovered by the product detector if an identical PN sequence to the DSSS modulator's carrier is used. The next part of the experiment demonstrates. This.

21. Modify the setup as shown in Fig 7 below to make the demodulator's local carrier a different PN sequence to the transmitter's carrier.



22. Compare the message with the product detector's new output.

## Part D

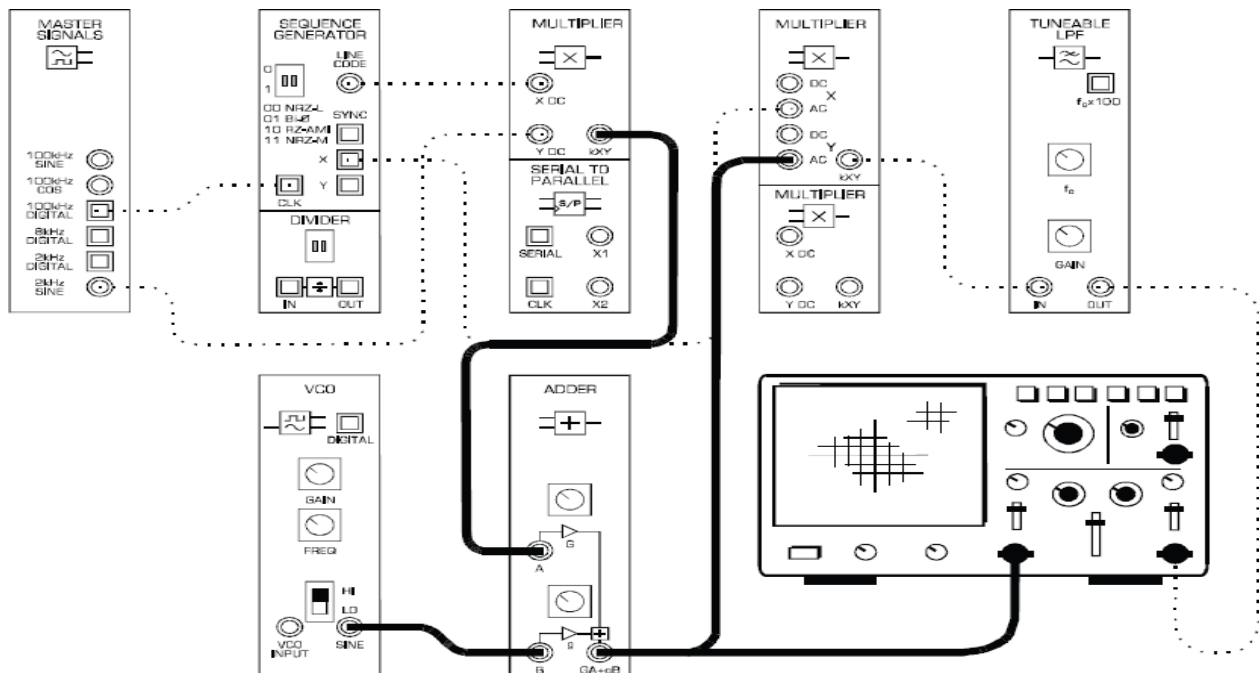
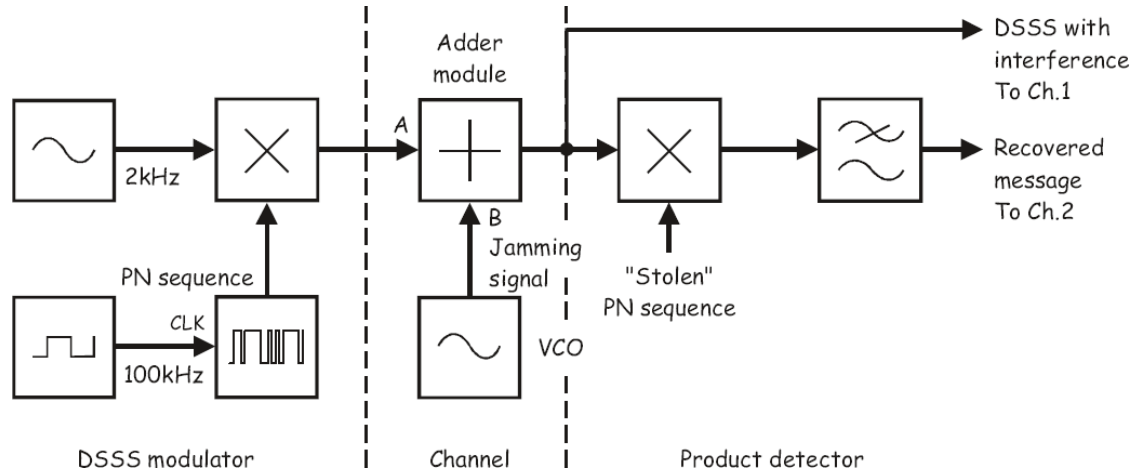
### DSSS and deliberate interference (Jamming)

Interferences occurs when an unwanted electrical signal gets added to the transmitted signal and changes it enough to change the recovered message. Electrical noise is a significant source of unintentional interference.

However, sometimes noise is deliberately added to the transmitted signal for the purpose of interfering or Jamming it. The next part of the experiment models deliberate interference to show how spread spectrum signals are highly resistant to it.

23. Move the patch lead from the sequence Generator's Y output back to its X output.
  - a. Note : The product detector should now be recovering the message again.
24. Locate the VCO module and set its Range control the HI position.
25. Set the VCO modules Frequency Adjust control to about the middle of its travel.
26. Locate the Adder Module and turn its g control fully anti clockwise.

27. Set the Adder module's G control to about the middle of its travel.
28. Modify the setup as shown in Fig 8 below.



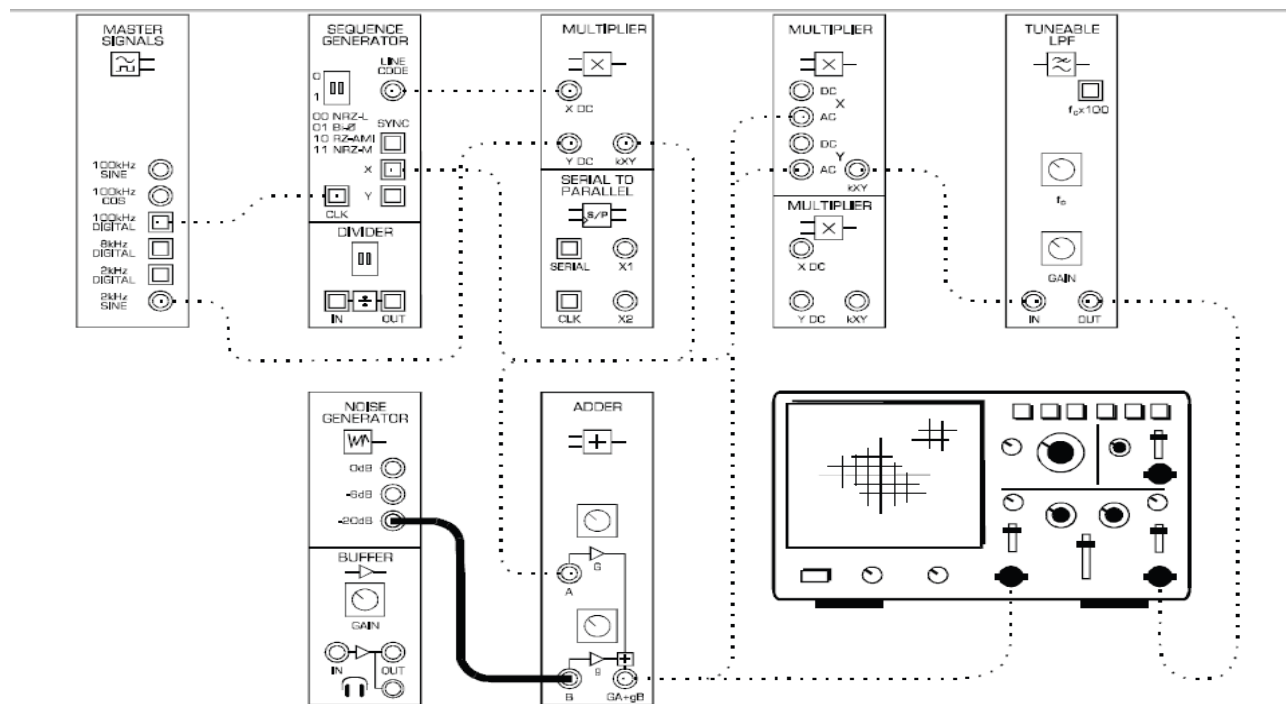
This modification forces the VCO module's output to sweep continuously through a wide range of Frequencies.

29. compare the two signals. Notice that the DSSS signal with interference is very distorted but the recovered message is only mildly affected.



An even more sophisticated approach to jamming involves using jamming signals at once to increase the chances of upsetting the transmitted signal. The next part of the experiment let's you see how spread spectrum handles this.

30. Modify the setup shown in Fig 11 below. This modification uses the Noise generator module to model a jamming signal that consists of thousands of frequencies.
31. Compare the two signal. Notice that the DSSS signal with interference is distorted but the recovered message is only mildly affected.
32. Increase the strength of the broadband jamming signal by connecting the Adder.
33. Compare the DSSS signal and the recovered message.
34. Increase the strength of the broadband jamming signal even more by connecting the Adder module's B input to the Noise Generator module's 0dB output.
35. Compare the two signals. Notice how distorted DSSS signal is but how little therecovered message is affected.
36. Modify the Setup as shown.



Result:

Thus the Experiment was performed successfully.

## EXPERIMENT NO:9

### SIMULATION OF ANALOG MODULATION SCHEMES IN MATLAB

#### AIM:

To simulate ANALOG MODULATION SCHEMES IN MATLAB

#### APPARATUS REQUIRED:

1. Amplitude Modulation and Demodulation Trainer
2. Function Generator
3. Oscilloscope
4. Connecting Wires

#### THEORY:

Modulation is defined as the process by which some characteristics of a carrier signal is varied in accordance with a modulating signal. The base band signal is referred to as the modulating signal and the output of the modulation process is called as the modulation signal.

Amplitude modulation is defined as the process in which is the amplitude of the carrier wave is varied about a means values linearly with the base band signal. The envelope of the modulating wave has the same shape as the base band signal provided the following two requirements are satisfied

- (1) The carrier frequency  $f_c$  must be much greater then the highest frequency components  $f_m$  of the message signal  $m(t)$

$$\text{I.e. } f_c \gg f_m$$

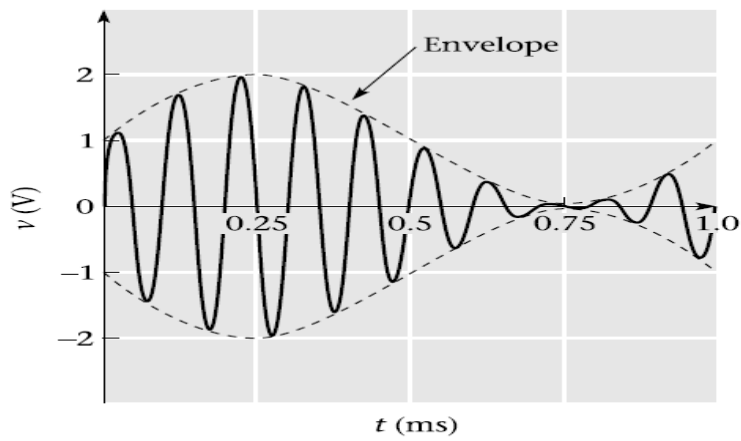
- (2) The modulation index must be less than unity. If the modulation index is greater than unity, the carrier wave becomes over modulated

#### PROCEDURE:

- 1 Switch on the trainer and check the O/P of carrier generator on oscilloscope.
2. Connect 1 KHz with 2 Volts A.F signal at AF I/P to the modulator circuit.
3. Connect the carrier signal at carrier I/P of modulator circuit.
4. Observe the modulator output signal at AM O/P by making necessary changes in A.F. Signal.
5. Vary the modulating frequency and amplitude and observe the effects on the modulated waveform.
6. The depth of modulation can be varied using the variable knob (potentiometer) provided at A.F. input.
7. The percentage of modulation or modulation factor can be calculated using the following formulas.

$$\% \text{ of Modulation (ma) } = \frac{V_{\max} - V_{\min}}{V_{\max} + V_{\min}} \times 100$$

$$\text{or Modulation factor(ma) } = \frac{V_{\max} - V_{\min}}{V_{\max} + V_{\min}}$$



(c) 100% Amplitude modulation

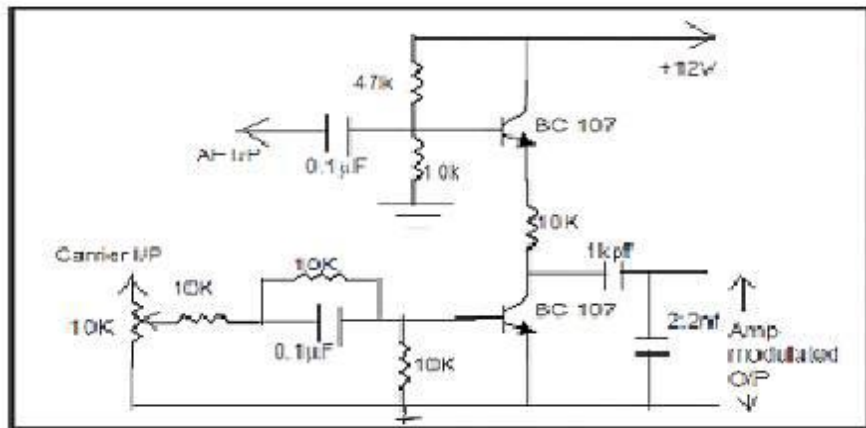
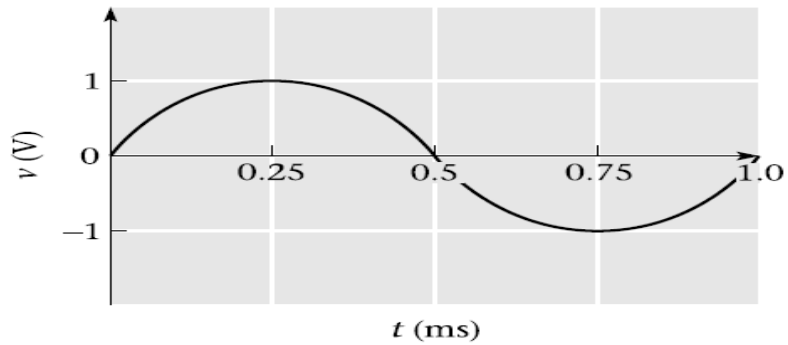
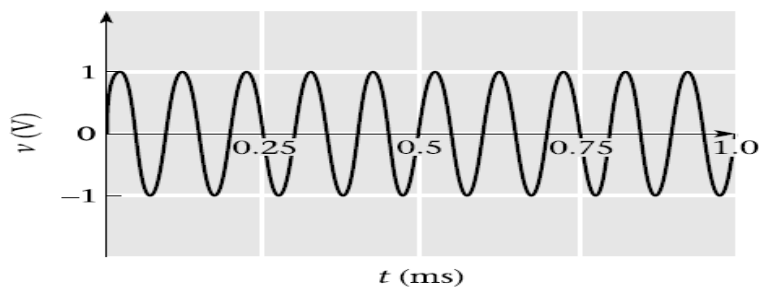


Fig:- (1) Circuit Diagram For modulation



(a)  $e_m$



(b)  $e_c$

```

t = [0:2*Fs+1]/Fs;
Fc = 10; % Carrier frequency
x = sin(2*pi*2*t); % message signal
Ac=1;
% compute spectra of am
xam=ammod(x,Fc,Fs,0,Ac);
zam = fft(xam);
zam = abs(zam(1:length(zam)/2+1));
frqam = [0:length(zam)-1]*Fs/length(zam)/2;
% compute spectra of dsbdc
ydouble = ammod(x,Fc,Fs, 3.14,0);
zdouble = fft(ydouble);
zdouble = abs(zdouble(1:length(zdouble)/2+1));
frqdouble = [0:length(zdouble)-1]*Fs/length(zdouble)/2;
% compute spectra of ssb
ysingle = ssbmod(x,Fc,Fs,0,'upper');
zsingle = fft(ysingle);

zsingle = abs(zsingle(1:length(zsingle)/2+1));
frqsingle = [0:length(zsingle)-1]*Fs/length(zsingle)/2;
% Plot spectrums of am dsbdc and ssb
figure;

subplot(3,1,1); plot(frqam,zam); title('Spectrum of am signal'); subplot(3,1,2); plot(frqdouble,zdouble);
title('Spectrum of double-sideband signal');subplot(3,1,3); plot(frqsingle,zsingle); title('Spectrum of
single-sideband

```

**RESULT:**

The simulation of ANALOG MODULATION SCHEMES IN MATLAB ARE GENERATED

## EXPERIMENT NO:10

### SIMULATION OF ANALOG DEMODULATION SCHEMES IN MATLAB

#### AIM:

To simulate analog demodulation schemes in matlab

#### THEORY :

Amplitude modulation (AM) is defined as a process in which the amplitude of the carrier wave  $c(t)$  is varied about a mean value, linearly with the base band signal  $m(t)$ .

An AM wave may thus be described, in its most general form, as a function of time as follows.

$$S(t)=A [1+K_a m(t)] \text{Cos} (2\pi f_c t)$$

The amplitude of  $K_a m(t)$  is always less than unity, that is  $|K_a m(t)| < 1$  for all  $t$ . It ensures that the function  $1+K_a m(t)$  is always positive. When the amplitude sensitivity  $K_a$  of the modulator is large enough to make  $|K_a m(t)| > 1$  for any  $t$ , the carrier wave becomes over modulated, resulting in carrier phase reversals. whenever the factor  $1+K_a m(t)$  crosses zero.

The absolute maximum value of  $K_a m(t)$  multiplied by 100 is referred to as the percentage modulation.

#### PROCEDURE:

1. Connect the AC Adapter to the mains and the other side to the Experimental Trainer.

Switch „ON“ the power.

2. Observe the carrier and modulating waveforms and note their frequencies.

(Carrier frequency is around 100 KHz and amplitude is variable from 0 -8Vp-p, modulating signal is 1KHz).

3. Connect the carrier and modulating signals to the modulator circuit.

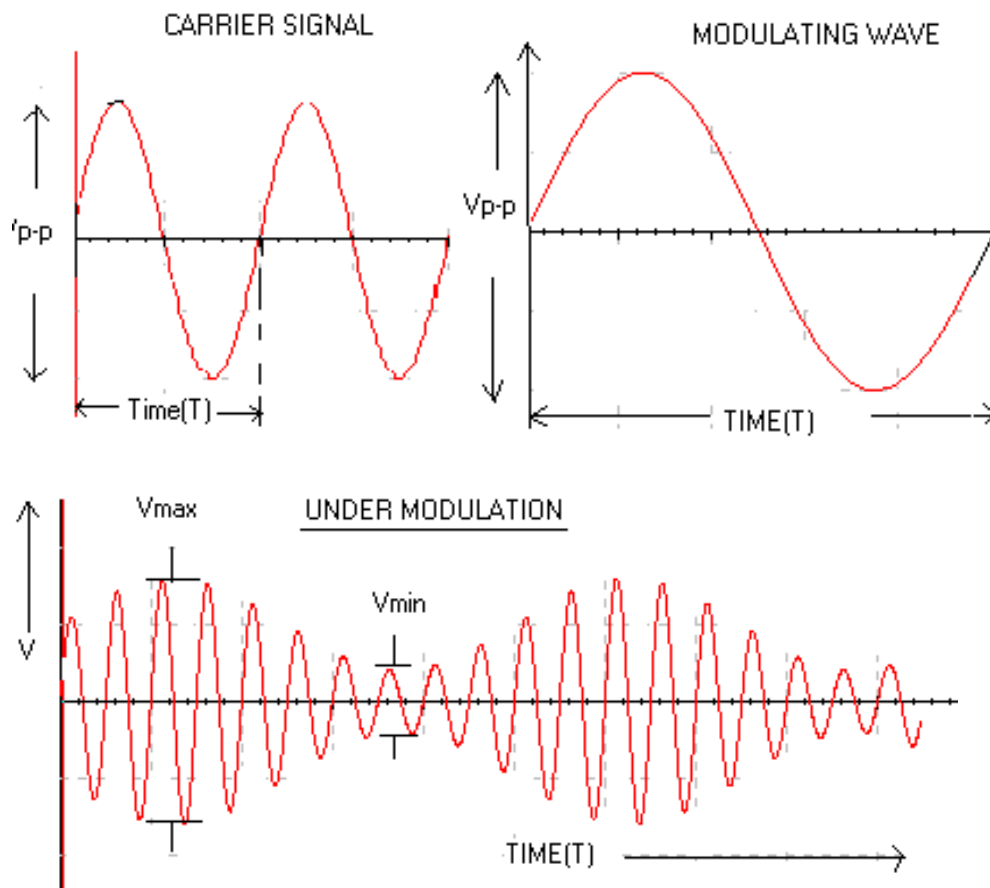
4. Observe the amplitude modulated wave.

5. Connect Carrier I/P to ground and apply a 2V peak to peak AF Signal input to (modulating I/P) and adjust P1 in anti-clock wise position to get minimum A.C output.

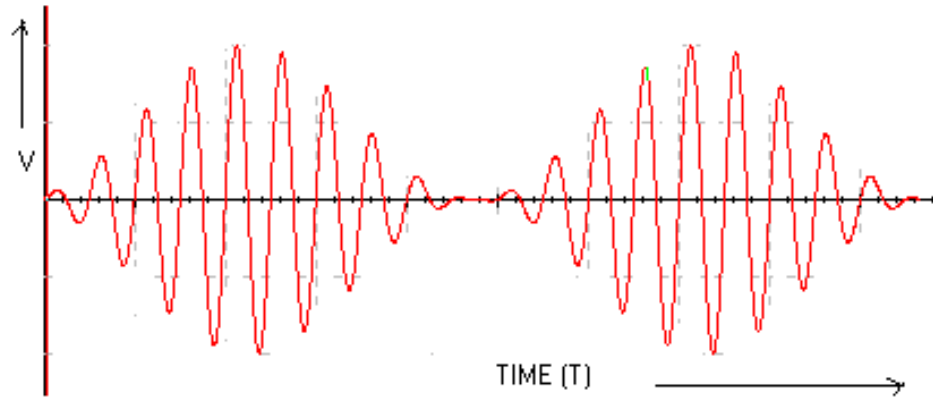
6. Connect modulating I/P to ground and apply a 3V peak to peak carrier signal to carrier I/P and adjust P2 in clock wise direction to get minimum A.C ouyput..

7. Connect modulating input & carrier input to ground and adjust P3 for zero D.C output.
8. Make modulating i/p 2 Vpp and carrier i/p 3 Vpp peak to peak and adjust potentiometer P4 for maximum output.
9. Calculate maximum and minimum points on the modulated envelope on a CRO and calculate the depth of modulation.
10. Observe that by varying the modulating voltage, the depth of modulation varies.

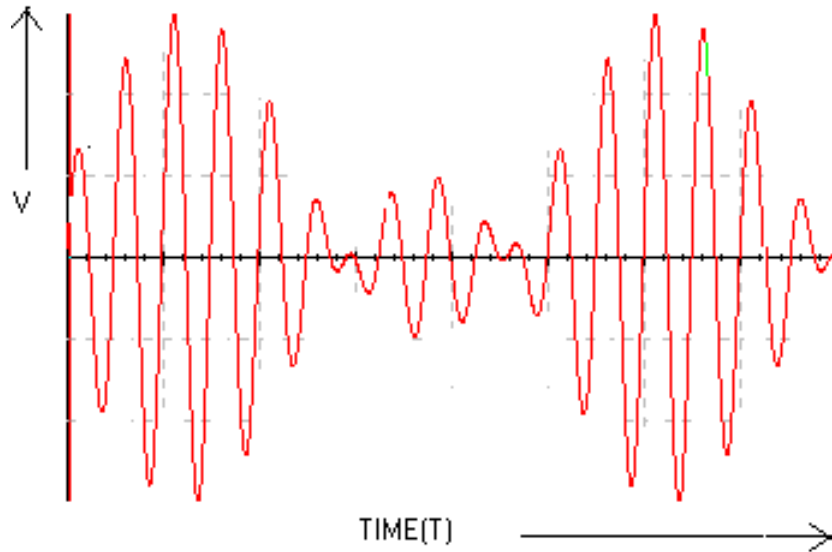
EXPECTED WAVE FORMS:



PERFECT MODULATION/  
100% MODULATION

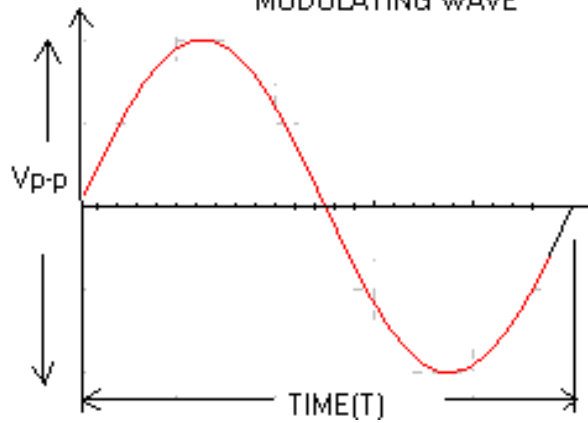


OVER MODULATION



### Demodulated signal

MODULATING WAVE



**Matlab code:**

```
fs=8000;
f
m
=
2
0
;
f
c
=
5
0
0
;
A
m
=
1
;
A
c
=
1
;
t=[0:.1*fs]/fs;
m=Am*cos(
2*pi*fm*t);
c=Ac*cos(2
*pi*fc*t);
ka=0.5;
u=ka*Am;
s1=Ac*(1+u*cos(2*pi*fm*t)).*cos(2
*pi*fc*t);subplot(4,3,1:3);
plot(t,m);
title('Modulating or Message
signal(fm=20Hz)');subplot(4,3,4:6);
plot(t,c);
title('Carrier
signal(fc=500Hz)');
subplot(4,3,7);
plot(t,s1);
title('Under Modulated
signal(ka.Am=0.5)');Am=2;
ka=0.5;
u=ka*Am;
s2=Ac*(1+u*cos(2*pi*fm*t)).*cos(2
*pi*fc*t);subplot(4,3,8);
plot(t,s2);
title('Exact Modulated
signal(ka.Am=1)');Am=5;
ka=0.5;
u=ka*Am;
```

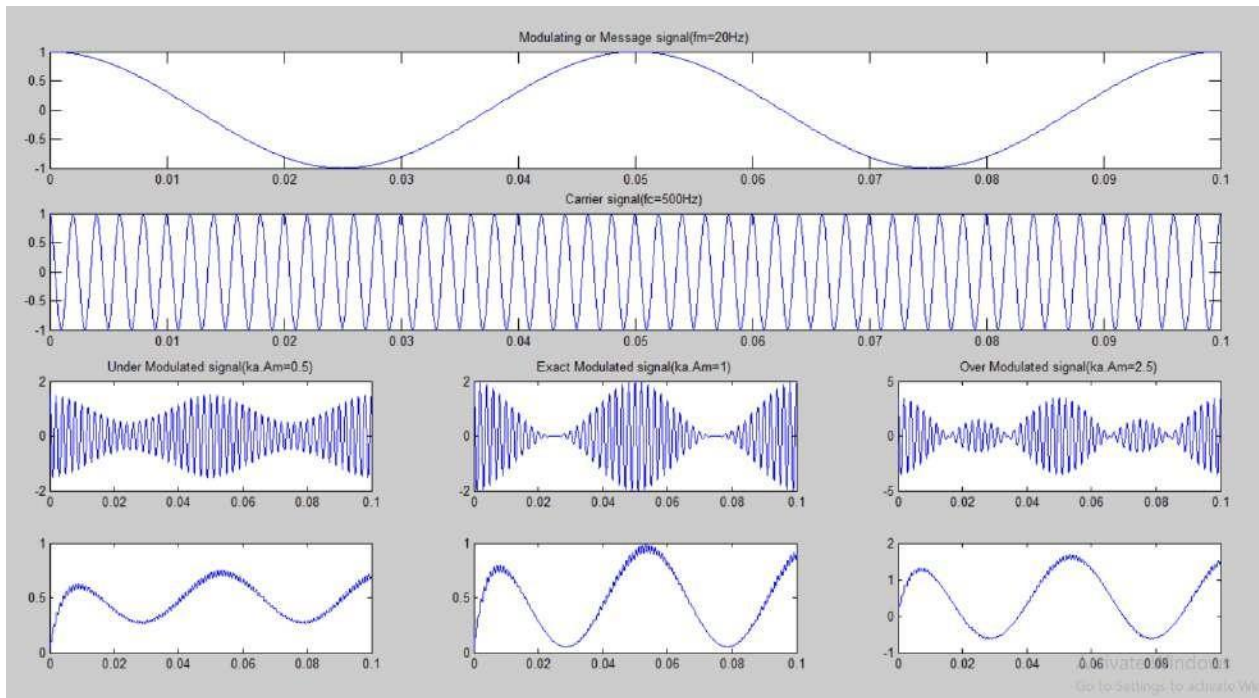


```

s3=Ac*(1+u*cos(2*pi*fm*t)).*cos(2
*pi*fc*t);subplot(4,3,9);
plot(t,s3);
title('Over Modulated
signal(ka.Am=2.5)');r1= s1.*c;
[b a] = butter(1,0.01);
mr1=
filter(b
,a,r1);
subplo
t(4,3,1
0);
p
l
o
t
(
t
,
m
r
1
)
;
r
2
=
s
2
.
*
c
;
[b a] = butter(1,0.01);
mr2=
filter(b
,a,r2);
subplo
t(4,3,1
1);
plot(t,mr2);

r3= s3.*c;

```



Go to settings to adjust Wb