EXPERIMENT:-1

AIM:-To simulate amplitude modulation and demodulation with carrier

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

Amplitude modulation (AM) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to the waveform being transmitted.

Consider a carrier wave (sine wave) of frequency f_c and amplitude A given by:

$$c(t) = A.\sin(2\pi f_c t)$$

Let m (t) represents the message signal. For this example we shall take the message to be simply a sine wave of a frequency f_m, a much lower frequency (such as an audio frequency) than f_c:

$$m(t) = M.\cos(2\pi f_m t + \emptyset)$$

Where M is the amplitude of the message signal. We shall insist that M<1 so that *(1+m(t))* is always positive. If M>1 then over modulation occurs and reconstruction of message signal from the transmitted signal would lead in loss of original signal. Amplitude modulation results when the carrier *c(t)* is multiplied by the positive quantity*(1+m(t))*:

$$y(t) = [1 + m(t)].c(t)$$
$$=[1+ M.\cos(2\pi f_m t + \emptyset)].A.\sin(2\pi f_c t)$$

In this simple case M is identical to the modulation index, when M>1 it is said to be over modulation, when M<1 it is said to be under modulation, when M=1 it is 100% modulation. Using prosthaphaeresis identities, $y(t)$ can be shown to be the sum of three sine waves:

$$y(t) = A.\sin(2\pi f_c t) + \frac{AM}{2}[\sin(2\pi(f_c + f_m)t + \emptyset) + \sin(2\pi(f_c - f_m)t - \emptyset)]$$

Therefore, the modulated signal has three components: the carrier wave *c(t)* which is unchanged and two pure sine waves (known as sidebands)$f_c - f_m$ *or* $f_c + f_m$ with frequencies slightly above and below the carrier frequency f_c.

DEMODULATION

Demodulation is the process of extracting the original message signal from the modulated signal. There are

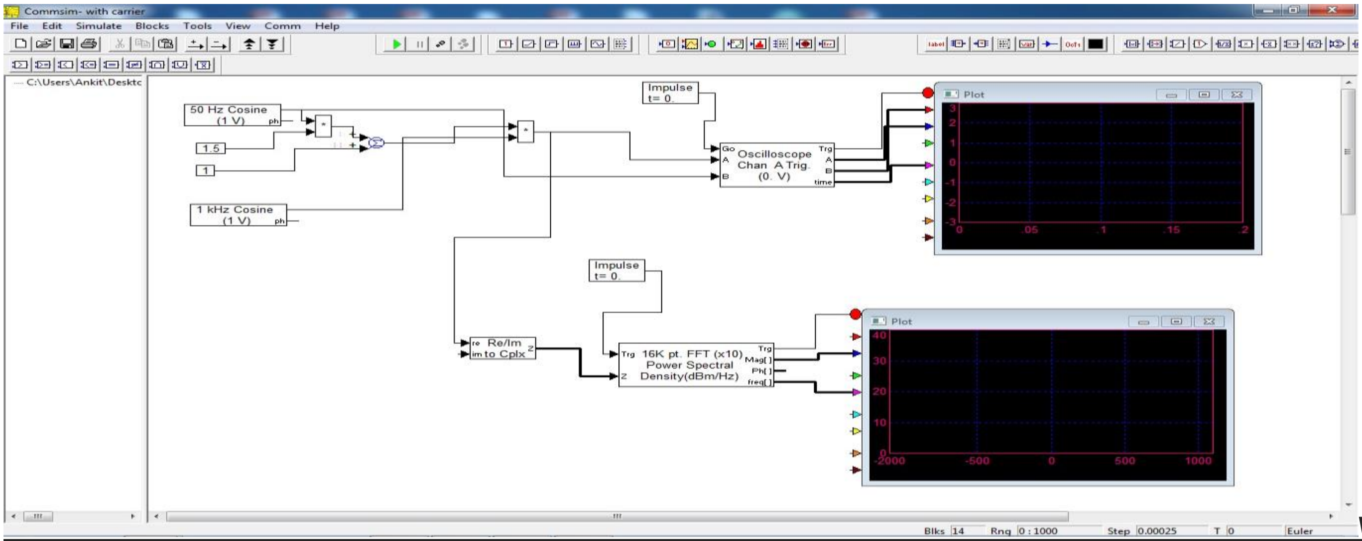
Various methods of demodulation but in this experiment demodulation are done by multiplying the DSB-SC signal with the carrier signal just like the modulation process. This resultant signal is then passed through a low pass filter to produce a scaled version of original message signal. DSB-SC can be demodulated by a simple envelope detector, if the modulation index is less then unity. Full depth modulation requires carrier re-insertion

$$\frac{V_c V_m}{2}\left[\cos\left((\omega_m + \omega_c)\right)t + \cos\left((\omega_m - \omega_c)\right)t\right] \times V_c\cos(\omega_c t)$$

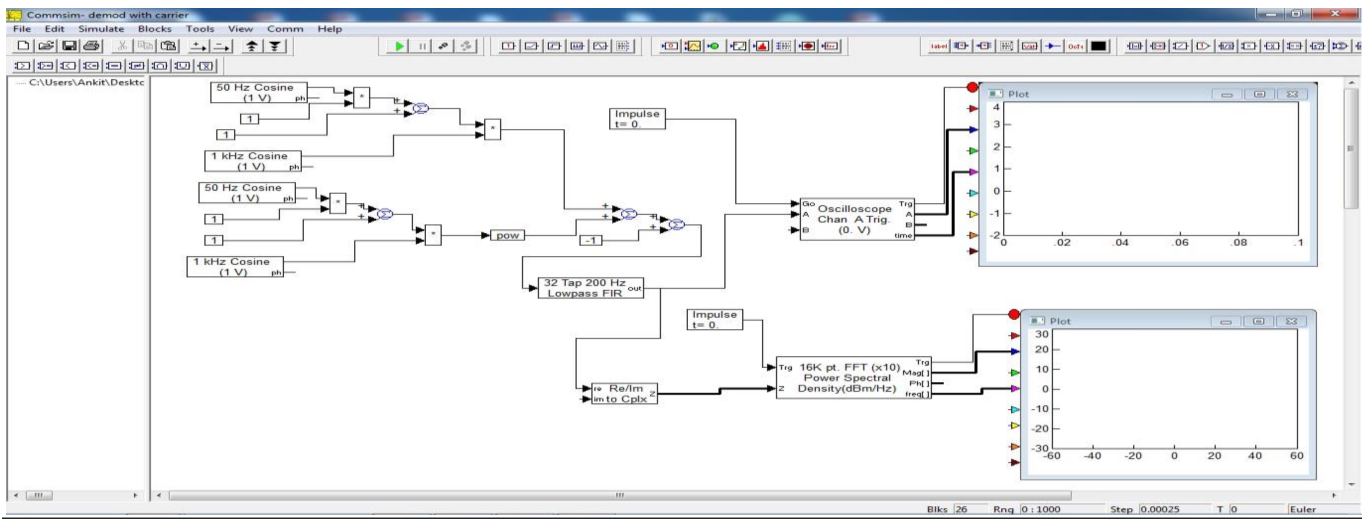
The equation above shows that by multiplying the modulated signal by the carrier signal, the result is a scaled version of the original message signal and also a second term. That second term is much higher in frequency than the original message. Once this signal pass through a low pass filter, the higher frequency component is removed, leaving the original message signal.

BLOCK DIAGRAM:-

MODULATION

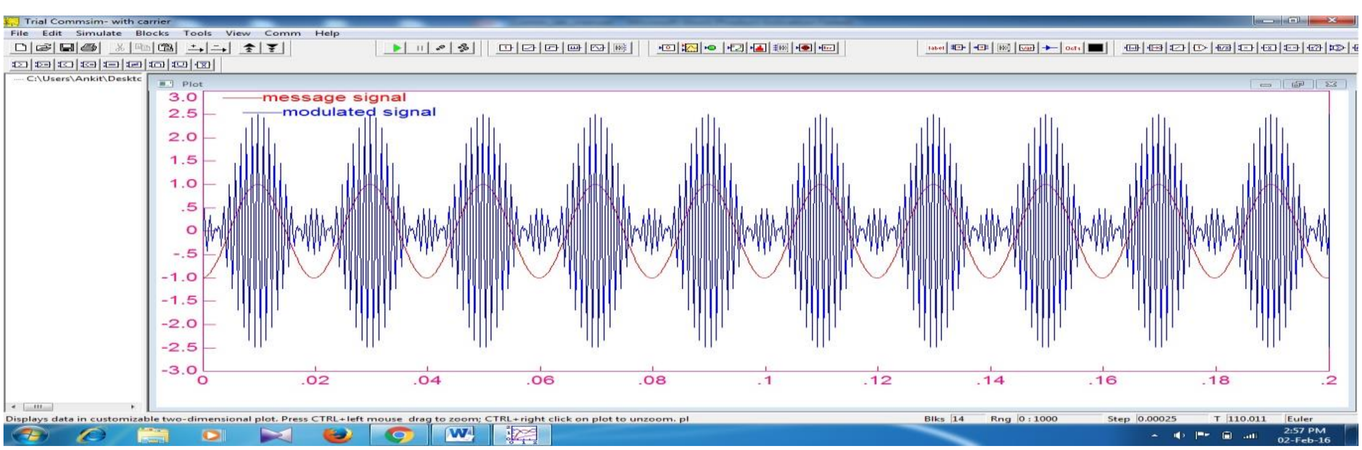


DEMODULATION

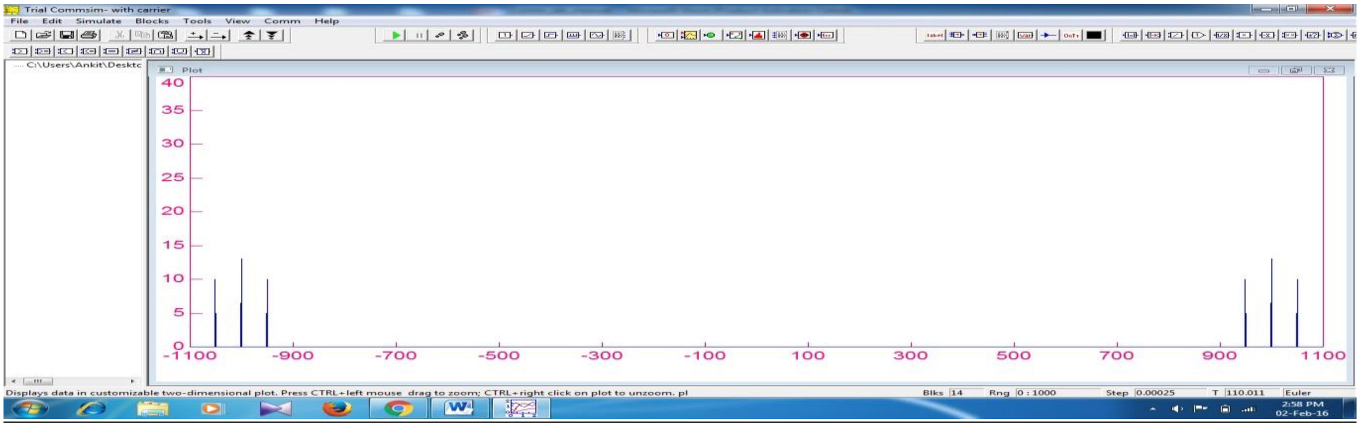


OUTPUT WAVEFORM:-

MODULATION(M=1.5)

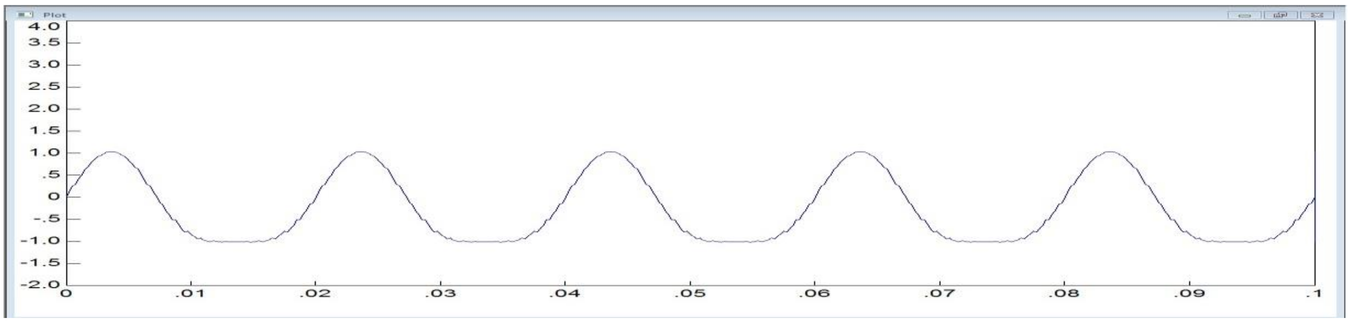


Modulation in Time domain

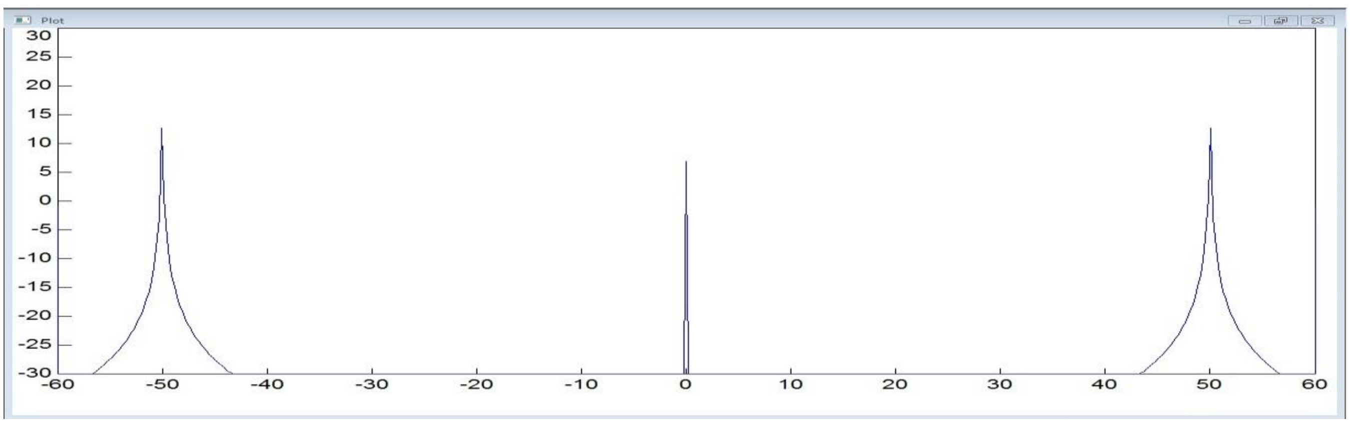


Spectrum analysis of modulated signal

DEMODULATION



Recovered Message Signal



Spectrum analysis ofRecovered Message Signal

CONCLUSION:-

We are taking a message signal having frequency 50 Hz and a carrier signal having a frequency 1000 Hz and modulation index 1.5. So after modulation we are getting an over modulated signal and we are getting frequency at 1000±50 and at – (1000±50). But after demodulation we are getting the message signal exactly at 50 Hz.

EXPERIMENT:-2

AIM:-To simulate amplitude modulation and demodulation without carrier

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

DSB-SC is basically an amplitude modulation wave without the carrier, therefore reducing power waste. It gives 50% efficiency. This is an increase compared to normal AM transmission(DSB), which has maximum efficiency of 33.33%, since 2/3 of the power is in the carriers no intelligence, and each sideband carries the same information. Single side band (SSB) suppressed carrier is 100% efficient.

MATHEMATICAL EXPRESSION:-

$$S(t) = V_m \cos(\omega_m t) \times V_c \cos(\omega_c t)$$

DEMODULATION

Demodulation can be done through synchronous detector. In a synchronous or coherent detector, the incoming AM signal is mixed with original carrier frequency.

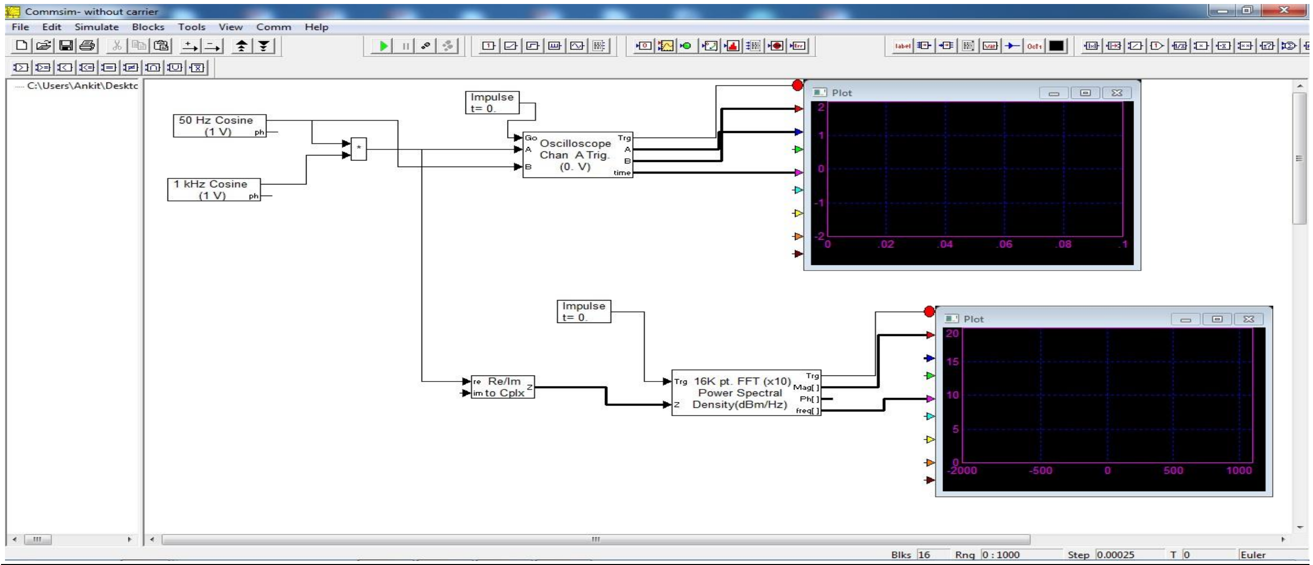
A synchronous detector is one where the difference between the two input frequency is zero Hz. In other words the two input frequency are the same.

At the multiplier output, we obtain:

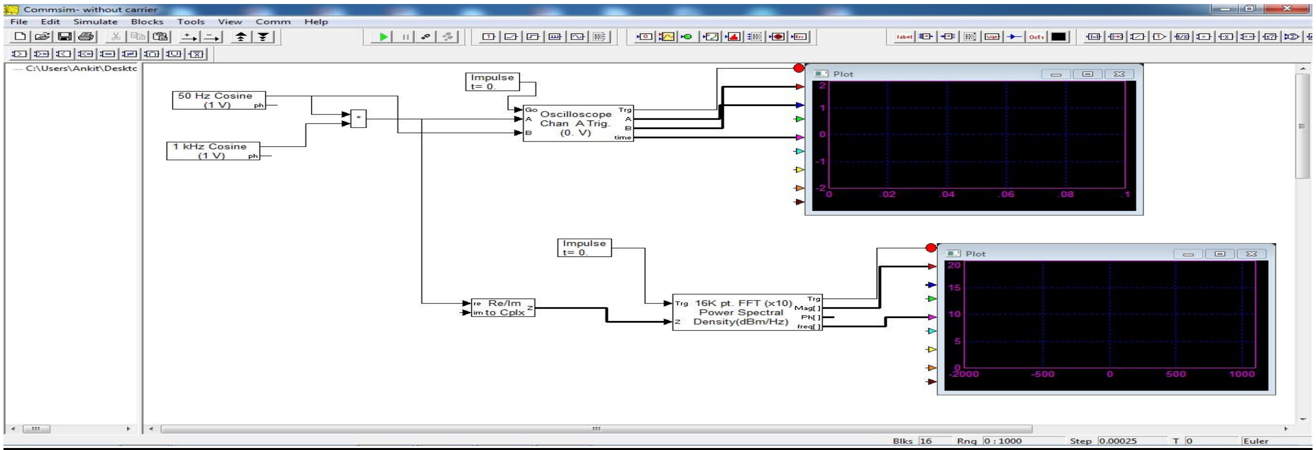
MIXER OUTPUT= original modulation signal+ AM signal centered at 2 times the carrier frequency

BLOCK DIAGRAM:-

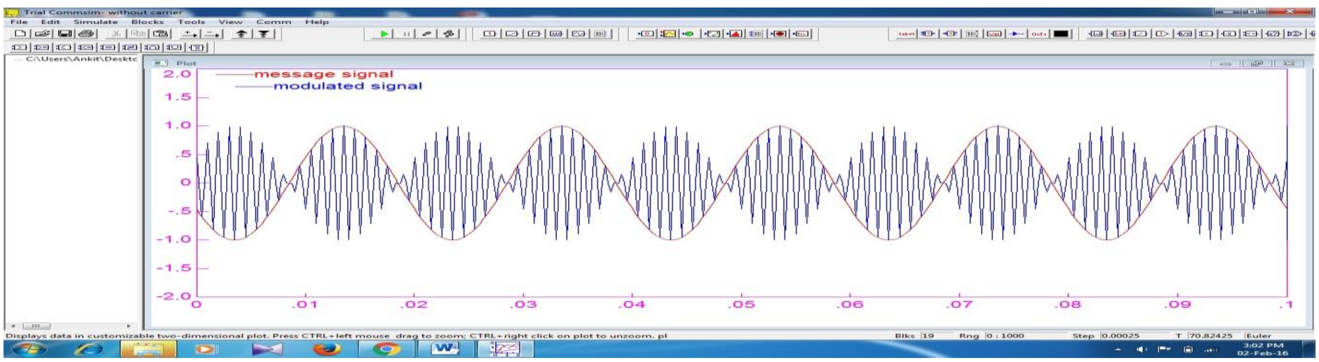
MODULATION



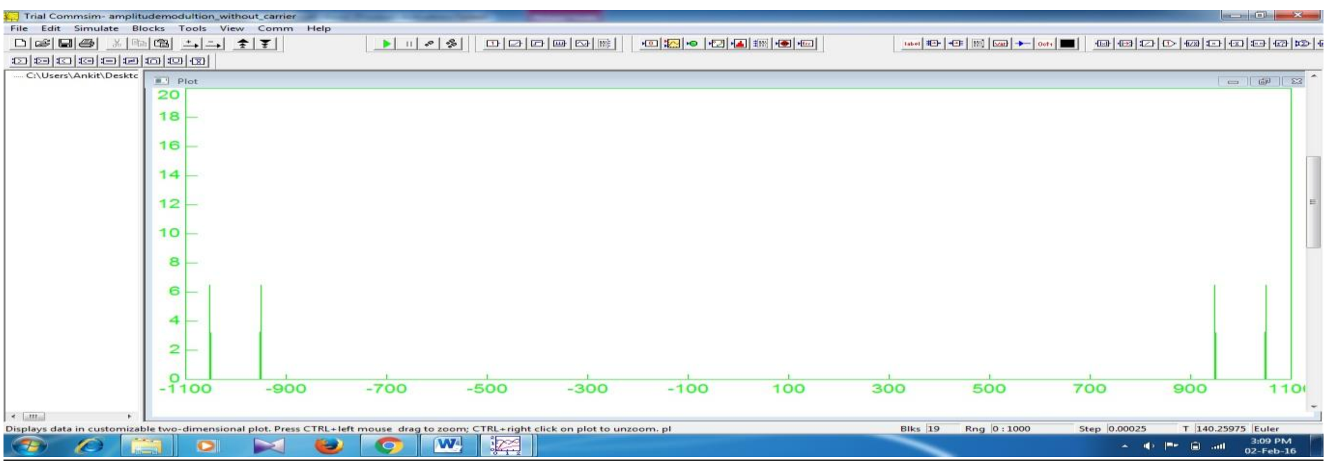
DEMODULATION



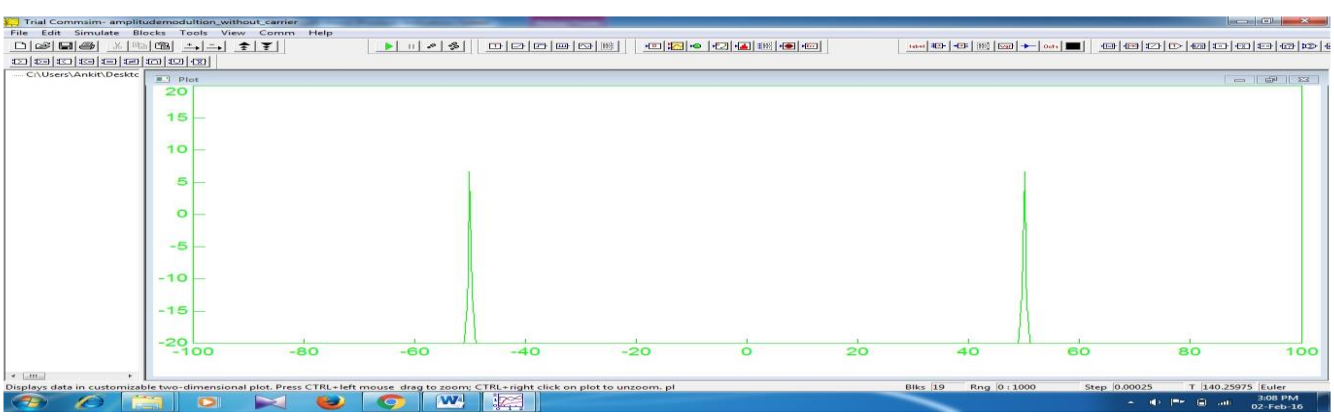
OUTPUT WAVEFORM:-



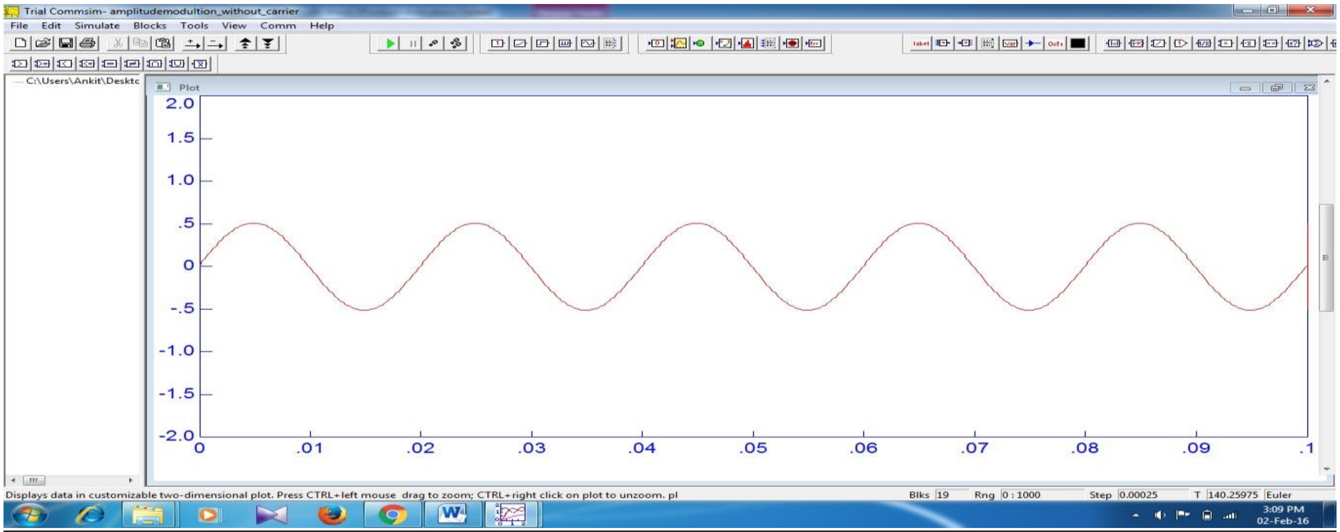
Modulation in Time domain



Spectrum Analysisin Frequency Domain



Spectrum analysis ofRecovered Message Signal



Recovered Message Signal

CONCLUSION:-

We are taking a message signal having frequency 50 Hz and a carrier signal having a frequency 1000 Hz. So after modulation we are getting a modulated signal and we are getting frequency at 1000±50 and at –(1000±50). And after demodulation we are getting the message signal exactly at 50 Hz.

EXPERIMENT:-3

AIM:-To simulate SSB modulation and demodulation

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

In radio communications, **Single-SideBand modulation** (**SSB**) or **Single-SideBand Suppressed-Carrier** (**SSB-SC**) is a refinement of amplitude modulation which uses transmitter power and bandwidth more efficiently. Amplitude modulation produces an output signal that has twice the bandwidth of the original baseband signal. Single-sideband modulation avoids this bandwidth doubling, and the power wasted on a carrier, at the cost of increased device complexity and more difficult tuning at the receiver.

MATHEMATICAL EXPRESSION

Single-sideband has the mathematical form of quadrature amplitude modulation (QAM) in the special case where one of the baseband waveforms is derived from the other, instead of being independent messages:

$$S_{ssb}(t) = s(t).\cos(2\pi f_0 t) - \hat{s}(t).\sin(2\pi f_0 t)$$

Where s(t) is the message, $\hat{s}(t)$ is its Hilbert transform, and f_0 is the radio carrier frequency.

To understand this formula, we may express s(t) as the sum of two complex-valued functions:

$$s(t) = \underbrace{\frac{1}{2}\left(s(t) + j.\hat{s}(t)\right)}_{s_a(t)} + \underbrace{\frac{1}{2}\left(s(t) - j.\hat{s}(t)\right)}_{s_a^*(t)}$$

Where j represents the imaginary unit, $s_a(t)$ is the analytic representation of s(t) and $s_a^*(t)$ is itscomplex conjugate. This representation divides s(t) into its non-negative frequency components and its non-positive frequency components. In other words:

$$\frac{1}{2}s_a(f) = \begin{cases} s(f), & For\ f > 0, \\ 0, & For\ f < 0 \end{cases}$$

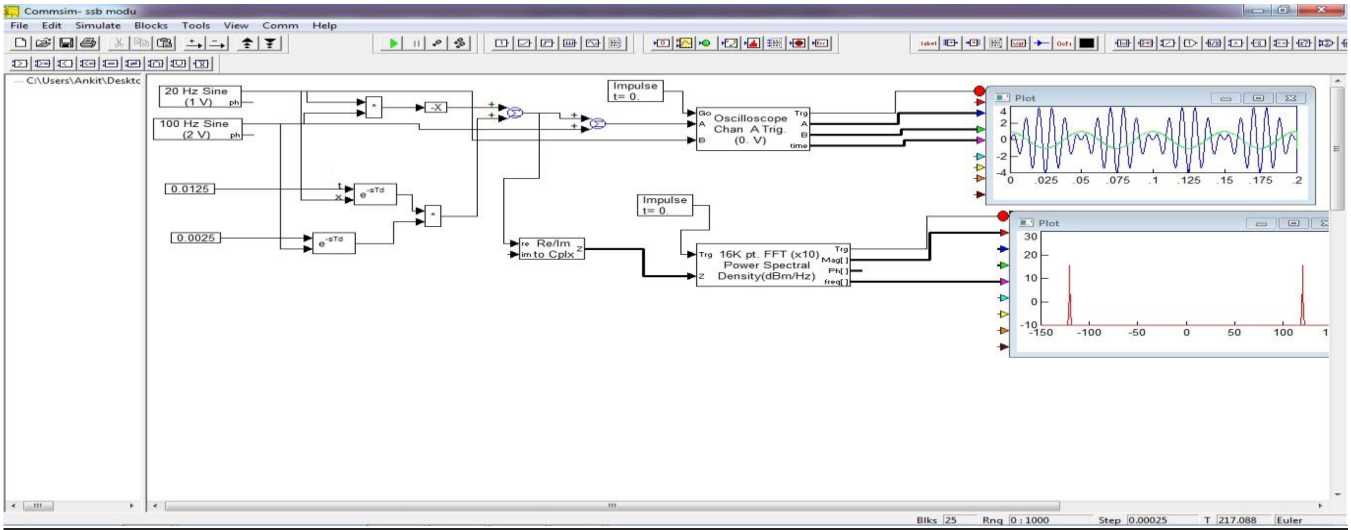
Where $s_a(f)$ and $s(f)$ are the respective Fourier transforms of $s_a(t)$ and s(t).

DEMODULATION

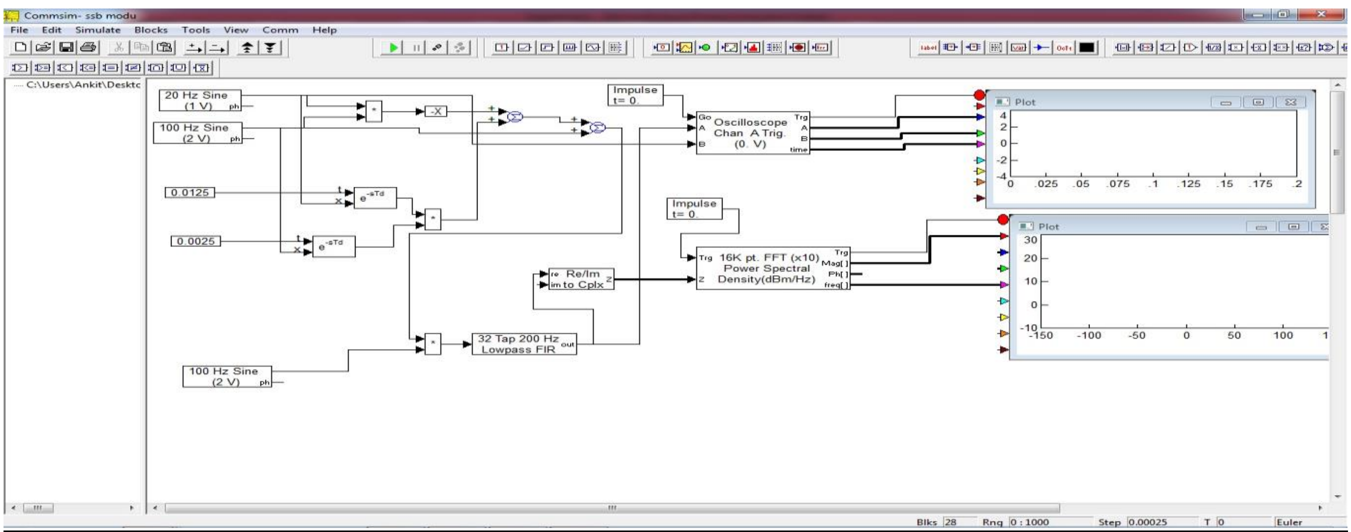
Single-sideband modulation (SSB) or Single-sideband suppressed-carrier (SSB-SC) is a refinement of amplitude modulation that more efficiently uses electrical power and bandwidth. Amplitude modulation produces a modulated output signal that has twice the bandwidth of the original baseband signal. Single-sideband modulation avoids this bandwidth doubling, and the power wasted on a carrier, at the cost of somewhat increased device complexity and more difficult tuning at the receiver.
The baseband or modulating signal can be recovered from the SSB-SC signal by using the synchronous detection. Coherent Demodulation of SSB signals ΦSSB (t) is multiplied with cos(ωc t) and passed through low pass filter to get back the original signal.

BLOCK DIAGRAM:-

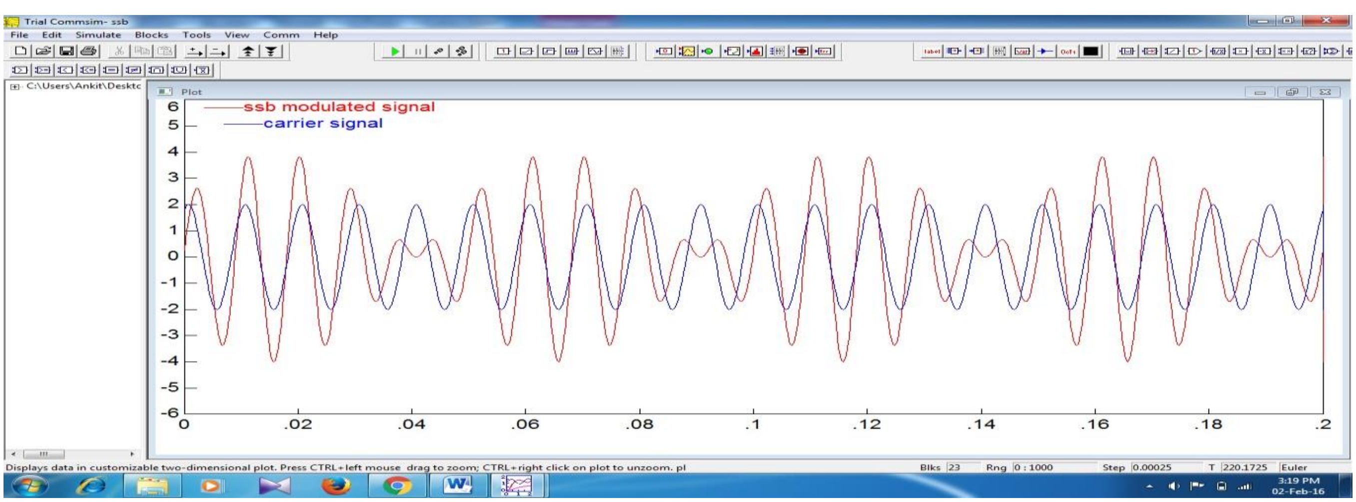
MODULATION



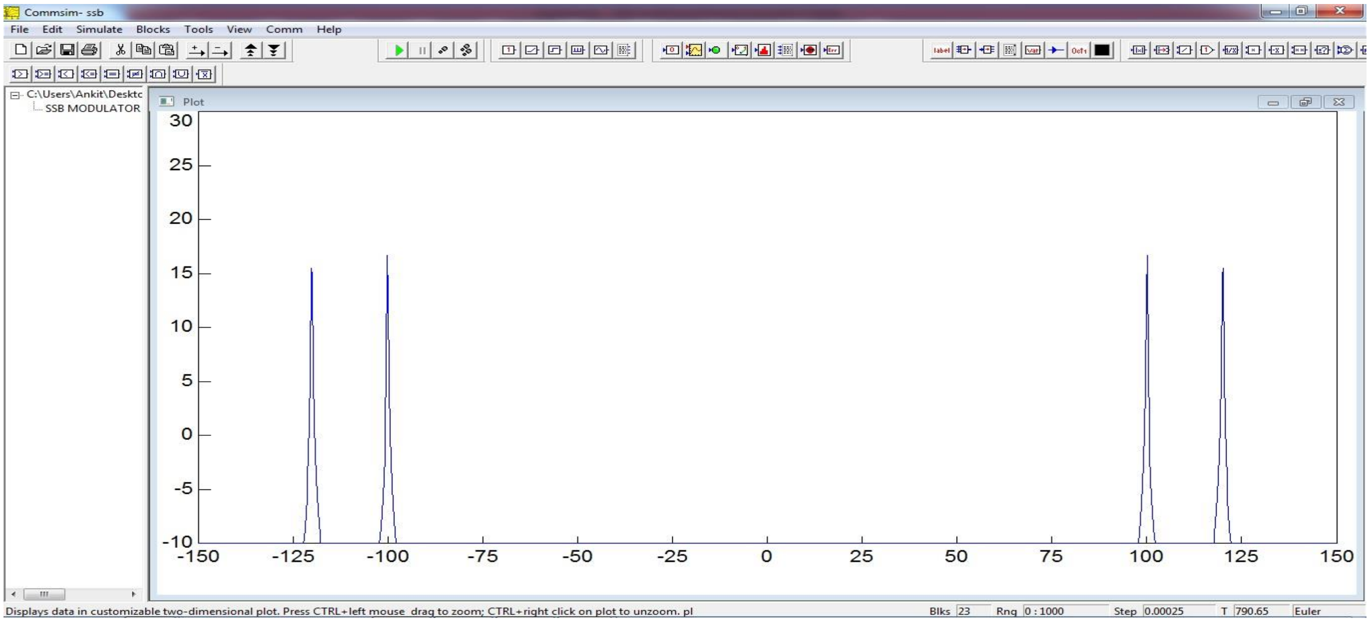
DEMODULATION



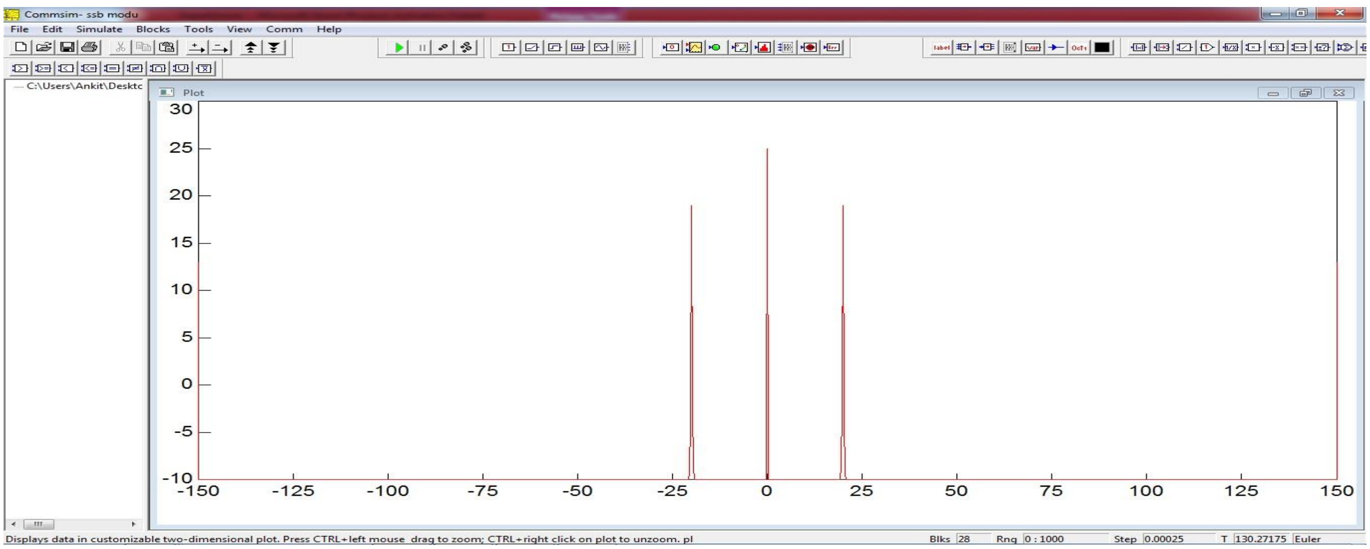
OUTPUT WAVEFORM:-



Modulation in Time Domain



Spectrum analysis of Modulated Signal



Spectrum analysis of Recovered Message Signal

CONCLUSION:-

We are taking a message signal having frequency 50 Hz and a carrier signal having a frequency 100 Hz. So after modulation we see that modulated signal changes their phase and we are getting frequency at 100±50 and at -(100±50). But after demodulation we are getting the message signal exactly at 50 Hz.

EXPERIMENT:-4

AIM:-**To simulate FM modulation and demodulation**

SOFTWARE USED:-**Commsim 6.0**

THEORY:-

MODULATION

In frequency modulation, the frequency of the carrier wave is shifted proportionally to the amplitude of the modulating signal. Therefore the frequency of the modulated wave is shifted continuously, and since a different spectrum from the modulating signal occurs centered on the carrier frequency, modulation is nonlinear.In frequency modulation, the modulated wave Sf_m can be expressed with the following equation.

$$S_{fm}(t) = A_c \cos\left[2\pi f_c t + \emptyset_c + K_{fm} \int_{-\infty}^{t} A_m \cos(2\pi f_m t)\, dt\right]$$

K_{fm}: Constant, With frequency modulation, the carrier wave phase Φc is shifted proportionally to the integral of the modulating signal m(t). The formula shows a form similar to phase modulation, while frequency modulation integrates the modulating signalm(t) before phase modulation. The maximum frequency shift Δf of the instantaneous frequency is as follows.Δf is also called deviation.

$$\Delta f = \frac{K_{fm}}{2\pi} |m(t)|_{max}$$

When the modulating signal m(t) is a single sine wave

$$m(t) = A_m \cos(2\pi f_m t)$$

Therefore

$$\Delta f = \frac{K_{fm} A_m}{2\pi} \; and \; K_{fm} = \frac{2\pi\Delta f}{A_m}$$

When the initial phase Φc of the carrier wave is Φc = 0, the frequency modulated wave is

$$S_{fm}(t) = A_c \cos\left[2\pi f_c t + \frac{2\pi\Delta f}{A_m} \int_{-\infty}^{t} A_m \cos(2\pi f_m t)\, dt\right] = A_c \cos\left[2\pi f_c t + \frac{\Delta f}{f_m} \sin(2\pi f_m t)\right]$$

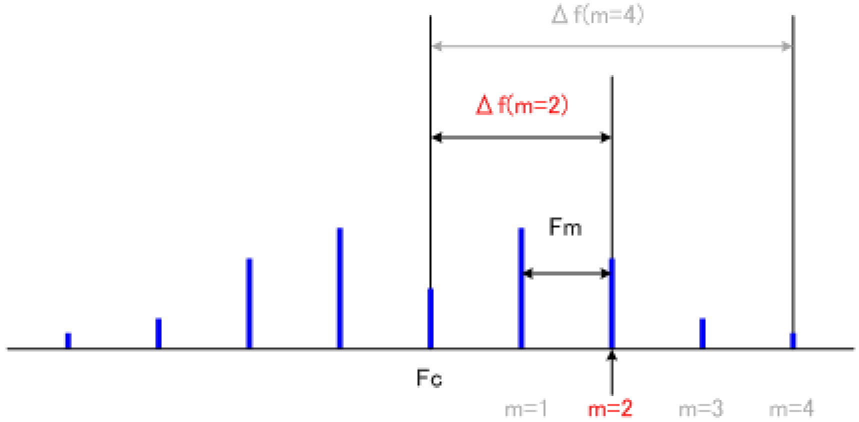
Specifying the frequency modulation index m gives the following.

$$m = \frac{\Delta f}{f_m}$$

When the frequency modulation index m is less than 1, it's known as narrowband FM, and if it's greater than 1, it's known as wideband FM. Wideband FM has good signal to noise ratio when it's demodulated at the receiver, so it's used widely in FM radios and so on. The higher the modulation index, the wider the bandwidth must be for transmission. Higher deviation during demodulation improves signal to noise, but this increases the bandwidth required for transmission. The priority given here depends on the system requirements. Improving the signal to noise by increasing the deviation is called FM gain.

◆FM modulated wave spectrum

The spectrum of the frequency modulation modulated wave occurs above and below the carrier wave Fc, and the frequency is the integral multiple of the modulating signal Fm. In the spectrum, the modulation signal frequency Fm, deviation Δf, and frequency modulation index m are related as follows. (When the modulating signal is a single sine wave) The spectrum interval is the modulation signal frequency Fm, and it spreads in an infinite frequency band. Deviation Δf is the difference in the center frequency Fc of the carrier wave and the frequency of the modulation index number from Fc.



FM modulated wave spectrum
Modulation index: m=2

◆The required bandwidth for frequency modulated waves

the envelope amplitude of frequency modulated waves is said to be constant, but this is only when all spectrum components are collected, and actually, amplitude fluctuation appears due to frequency band restrictions. The bandwidth B required so that amplitude fluctuation doesn't become a problem for demodulation is as follows. Δf is the maximum frequency shift, m is the modulation index, and Fm is the modulation signal frequency.

$$B \cong 2(\Delta f + f_m) = 2(m+1)f_m$$

When the maximum frequency of the modulating signal is fixed, a wide transmission band is required if the modulation index m is high. When the maximum frequency shift Δf is fixed, the spectral interval becomes narrow if the modulation index m is high.

DEMODULATION

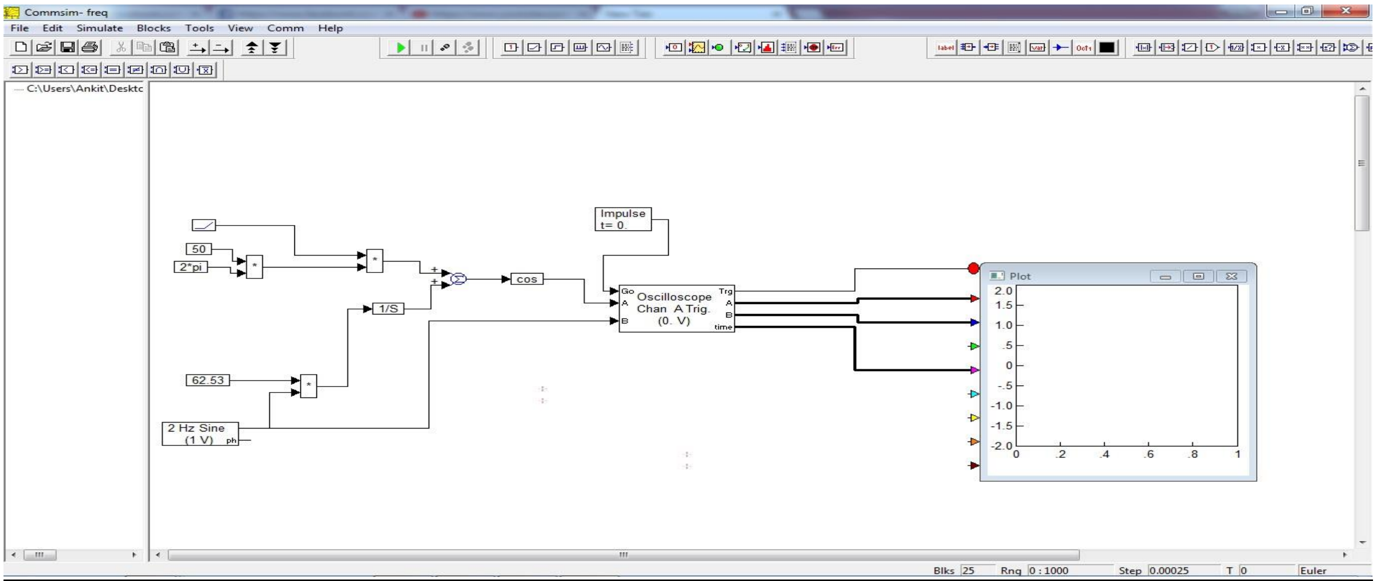
There are a number of circuits that can be used to demodulate FM. Each type has its own advantages and disadvantages, some being used when receivers used discrete components and others now that ICs are widely used.Below is a list of some of the main types of FM demodulator or FM detector. In view of the widespread use of FM, even with the competition from digital modes that are widely used today, FM demodulators are needed in many new designs of electronics equipment.

- ***Slope FM detector:*** This form of detector uses the slope of a tuned circuit to convert the frequency variations into amplitude variations. As the frequency of the FM signal varies, it changes its position on the slope of the tuned circuit, so the amplitude will vary. This signal can then be converted into a baseband signal by using an AM diode detector circuit. Read more about the ***Slope Detector***
- ***Ratio detector:*** This FM demodulator circuit was widely used with discrete components, providing a good level of performance. It was characterized by the transformer with three windings that was required. Read more about the ***Ratio Detector***
- ***Foster-Seeley FM detector:*** Like the Ratio detector the Foster Seeley detector or discriminator was used with discrete components, providing excellent performance for the day in many FM radios. Read more about the ***Foster-Seeley Detector***
- ***PLL, Phase locked loop FM demodulator:*** FM demodulators using phase locked loops, PLLs cna provide high levels of performance. They do not require a costly transformer and can easily be incorporated within FM radio ICs. Read more about the ***PLL FM Detector***
- ***Quadrature FM demodulator:*** This form of FM demodulator is very convenient for use within integrated circuits. It provides high levels of linearity, while not requiring many external components. Read more about the ***Quadrature FM Detector***

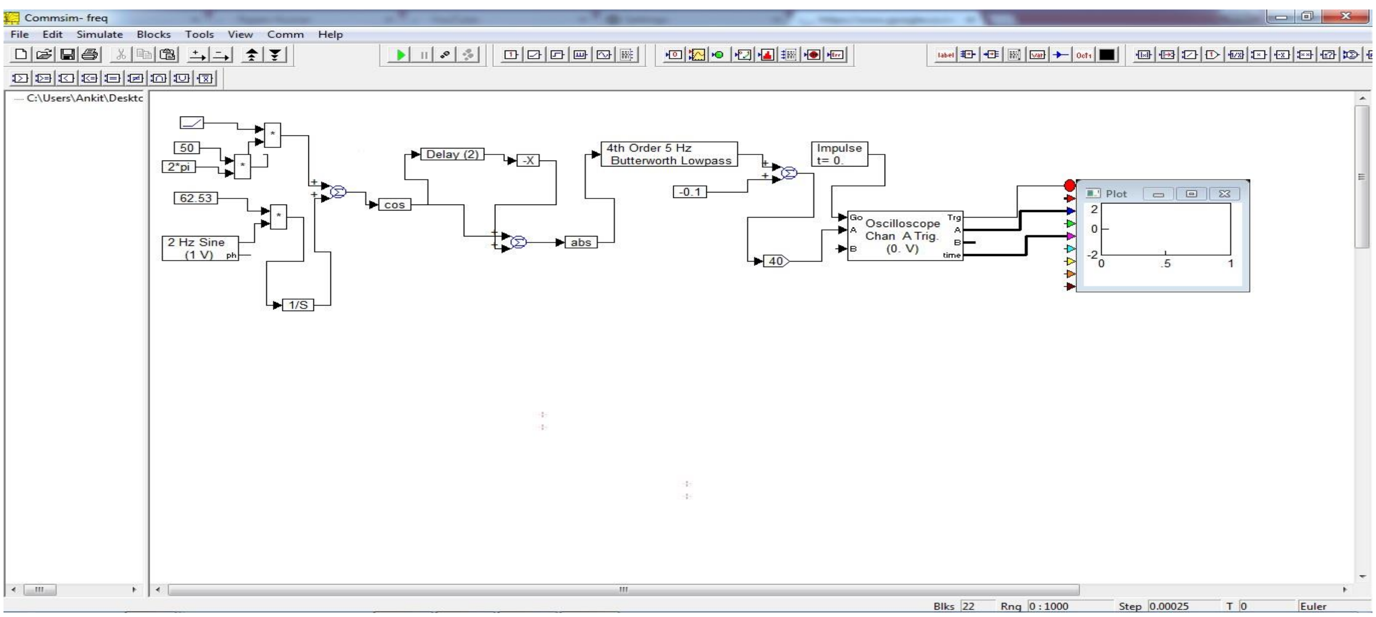
- ***Coincidence FM demodulator:*** This form of demodulator has many similarities to the quadrature detector. It uses digital technology and replaces a mixer with a logic NAND gate.

BLOCK DIAGRAM:-

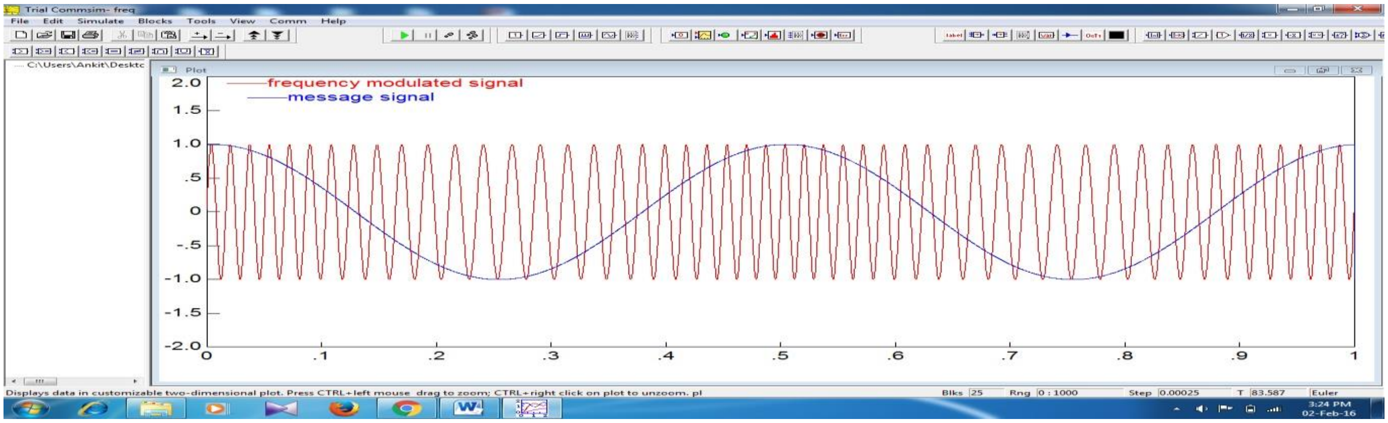
MODULATION



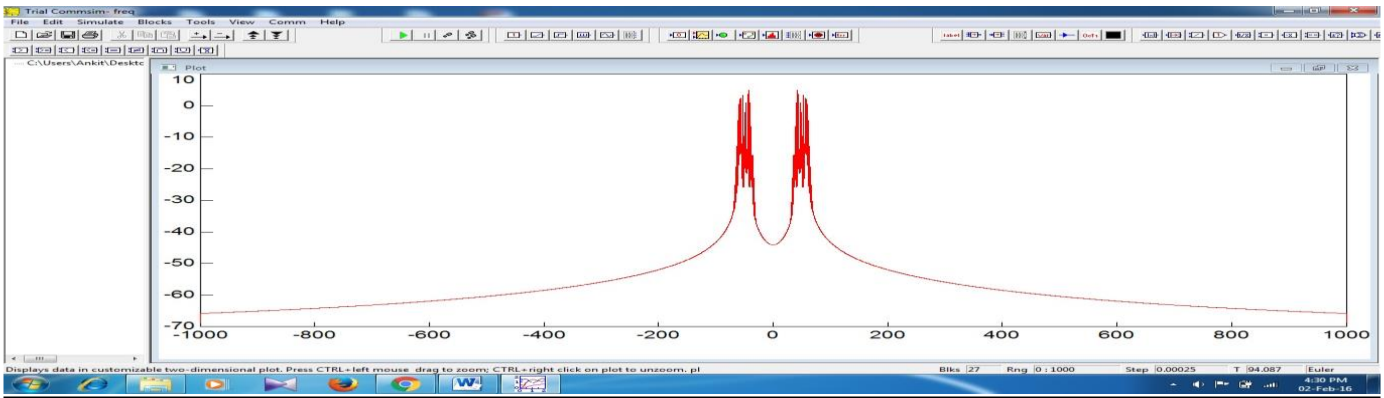
DEMODULATION



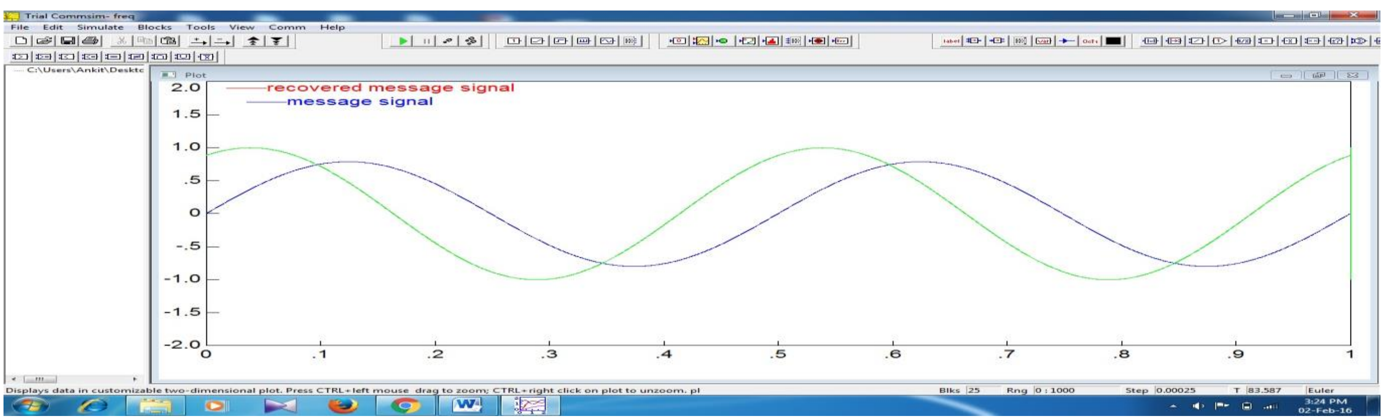
OUTPUT WAVEFORM:-



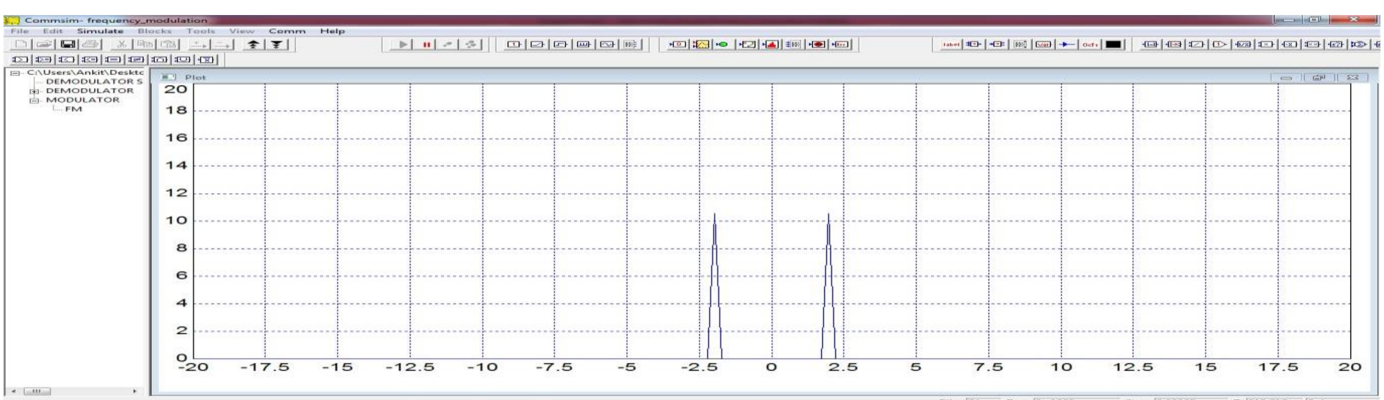
<u>Modulation in Time Domain</u>



<u>Spectrum Analysis of Modulated Signal</u>



<u>Recovered Message Signal</u>



<u>Spectrum analysis of Recovered Message Signal</u>

CONCLUSION:-

It has been clearly seen that after the modulation of input message signal the frequency of the modulated signal changes in accordance with the carrier signal. And after demodulation we are getting the original message signal whose frequency is exactly same as the input message signal frequency.

EXPERIMENT:-5

AIM:-To simulate Phase modulation and demodulation

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

PM changes the phase angle of the complex envelope in direct proportion to the message signal.

Suppose that the signal to be sent (called the modulating or message signal) is m(t) and the carrier onto which the signal is to be modulated is

$$c(t) = A_c \sin(\omega_c t + \varphi_c).$$

Annotated:

carrier(time) = (carrier amplitude)*sin(carrier frequency*time + phase shift)
This makes the modulated signal

$$y(t) = A_c \sin(\omega_c t + m(t) + \varphi_c).$$

This shows how m(t) modulates the phase - the greater m(t) is at a point in time, the greater the phase shift of the modulated signal at that point. It can also be viewed as a change of the frequency of the carrier signal, and phase modulation can thus be considered a special case of FM in which the carrier frequency modulation is given by the time derivative of the phase modulation.

The modulation signal could here be

$$m(t) = \cos\big(\omega_c t + k\omega_m(t)\big).$$

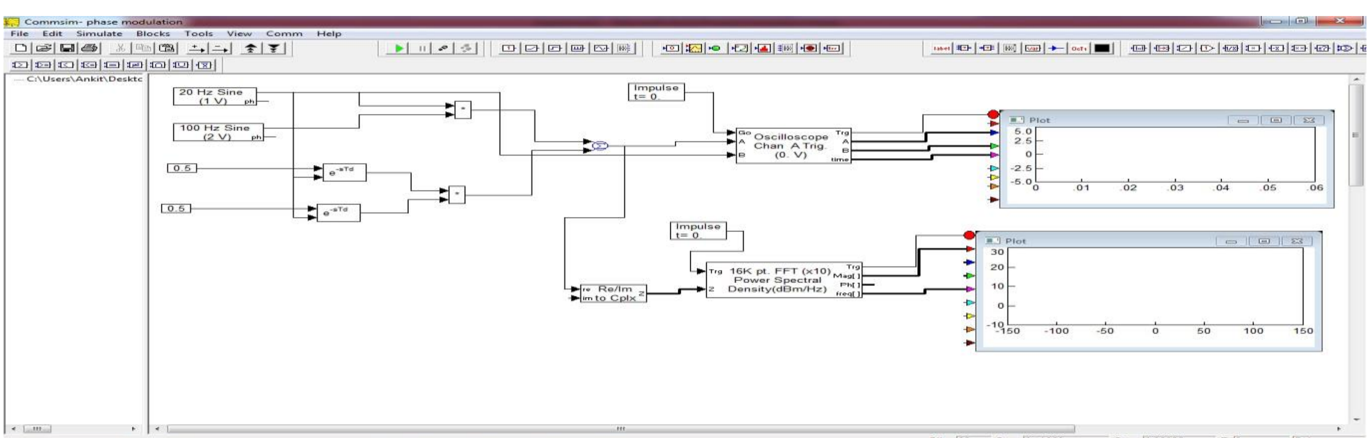
The mathematics of the spectral behavior reveals that there are two regions of particular interest:
For small amplitude signals, PM is similar to amplitude modulation (AM) and exhibits its unfortunate doubling of baseband bandwidth and poor efficiency.
For a single large sinusoidal signal, PM is similar to FM, and its bandwidth is approximately $2(h+1)f_M$,
Where $f_M = \frac{\omega_m}{2\pi}$ and h is the modulation index defined below. This is also known as Carson's Rule for PM.

DEMODULATION

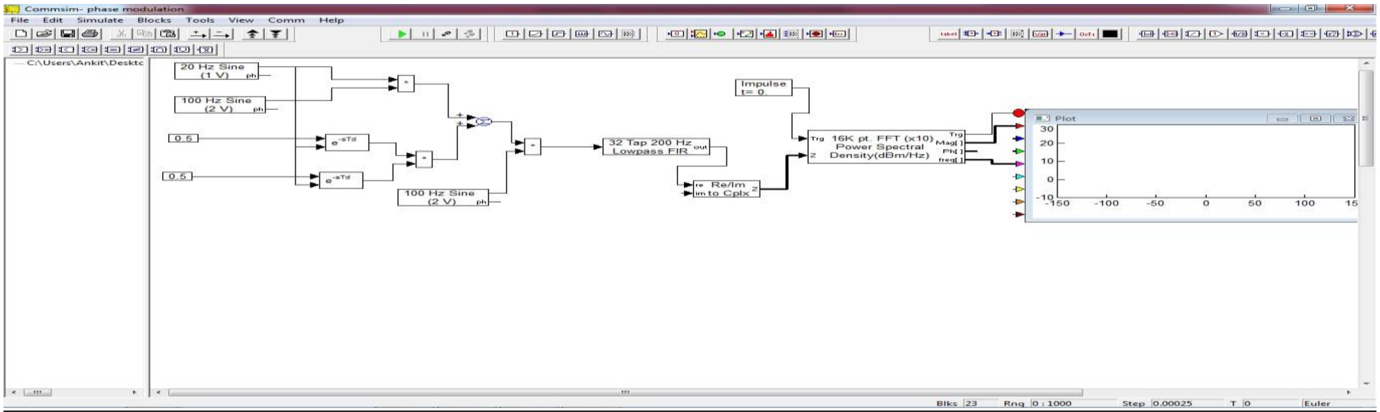
The quadrature detector, which phase shifts the signal by 90 degrees and multiplies it with the unshifted version. One of the terms that drop out from this operation is the original information signal, which is selected and amplified. The signal is feed into a PLL and the error signal is used as the demodulated signal.
A **phase-locked loop** or **phase lock loop** (**PLL**) is a control system that generates an output signal whose **phase** is related to the **phase** of an input signal. While there are several differing types, it is easy to initially visualize as an electronic circuit consisting of a variable frequency oscillator and a **phase** detector.

BLOCK DIAGRAM:-

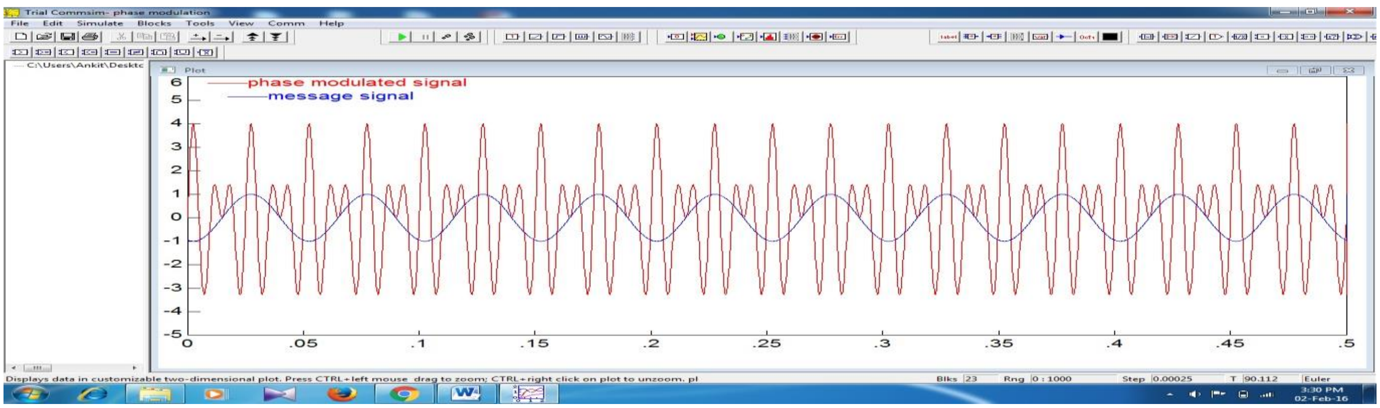
MODULATION



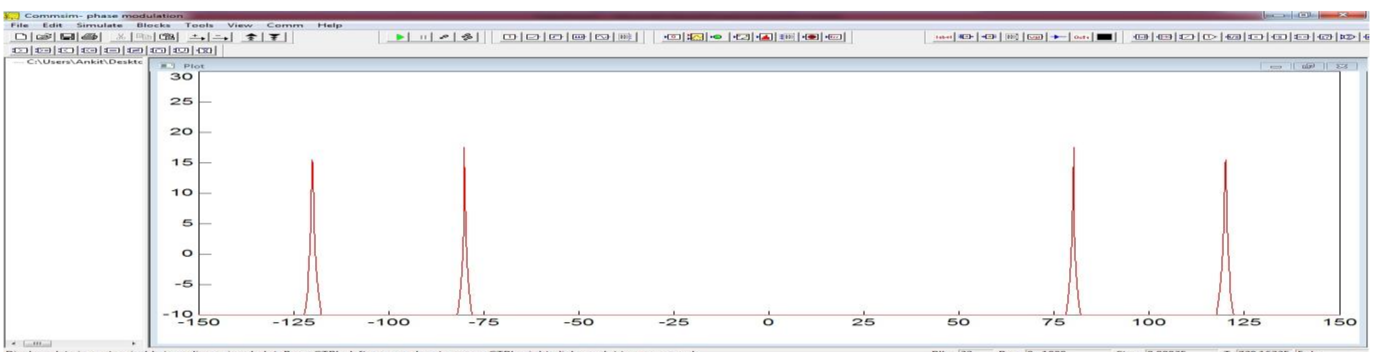
DEMODULATION



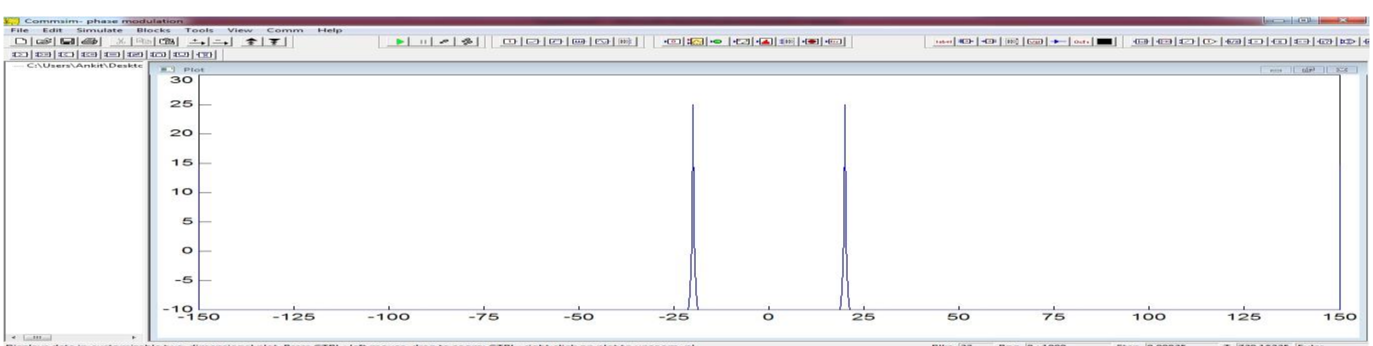
OUTPUT WAVEFORM:-



Modulation in Time Domain



Spectrum analysis of Modulated Signal



Spectrum analysis of Recovered Message Signal

CONCLUSION:-

We are taking a message signal having frequency 20 Hz and a carrier signal having a frequency 100 Hz. So after modulation we see that modulated signal changes their phase and we are getting frequency at 100±50 and at -(100±50). But after demodulation we are getting the message signal exactly at 20 Hz.

EXPERIMENT:-6

AIM:-To simulate Pulse Code modulation and demodulation

SOFTWARE USED:-Commsim 6.0

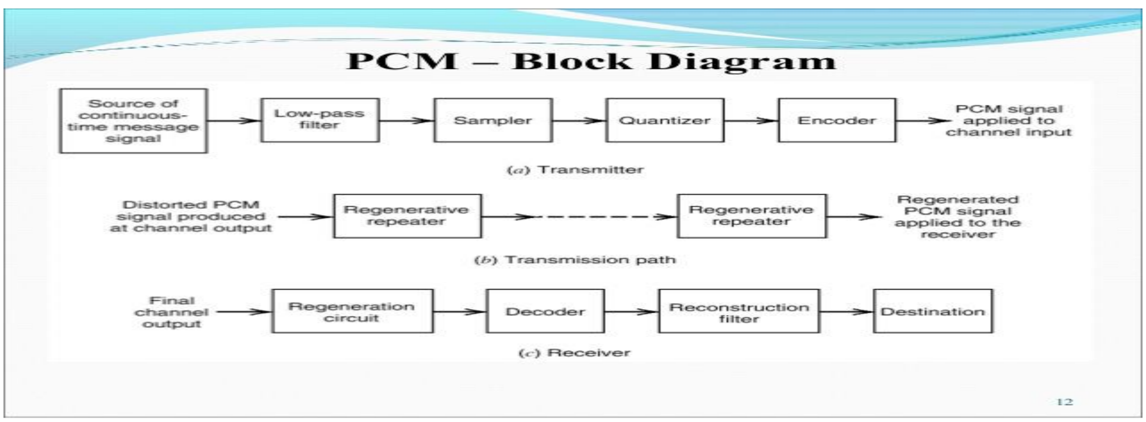
THEORY:-

MODULATION

Pulse-code modulation (**PCM**) is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, Compact Discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

Linear pulse-code modulation (**LPCM**) is a specific type of PCM where the quantization levels are linearly uniform.[5] This is in contrast to PCM encodings where quantization levels vary as a function of amplitude (as with the A-law algorithm or the μ-law algorithm). Though PCM is a more general term, it is often used to describe data encoded as LPCM.

A PCM stream has two basic properties that determine the stream's fidelity to the original analog signal: the sampling rate, which is the number of times per second that samples are taken; and the bit depth, which determines the number of possible digital values that can be used to represent each sample.



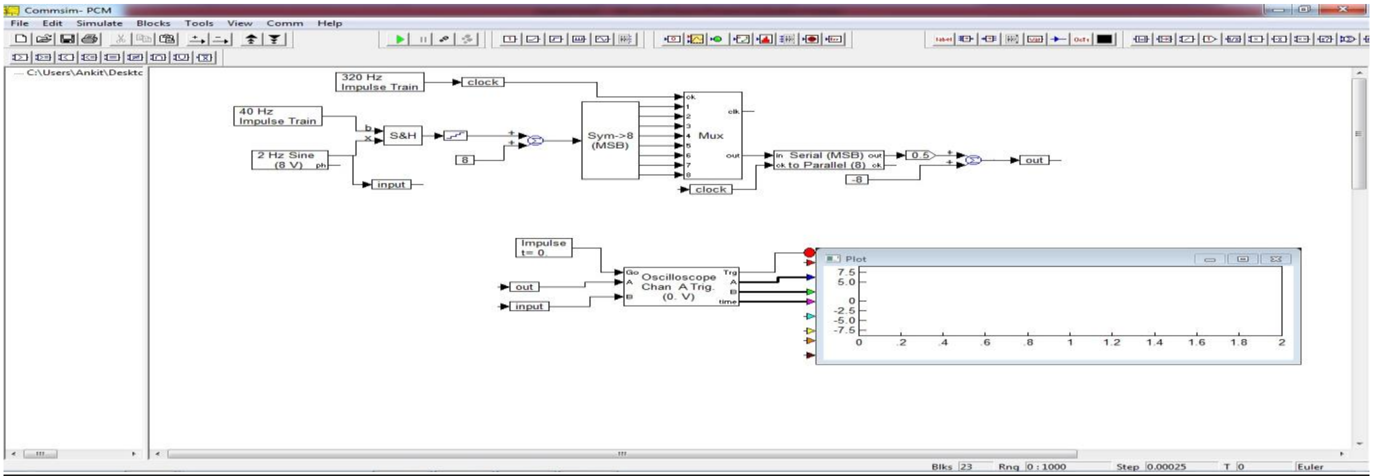
DEMODULATION

To recover the original signal from the sampled data, a "demodulator" can apply the procedure of modulation in reverse. After each sampling period, the demodulator reads the next value and shifts the output signal to the new value. As a result of these transitions, the signal has a significant amount of high-frequency energy caused by aliasing. To remove these undesirable frequencies and leave the original signal, the demodulator passes the signal through analog filters that suppress energy outside the expected frequency range (greater than the Nyquist frequency $f_s/2$).[note 1] The sampling theorem shows PCM devices can operate without introducing distortions within their designed frequency bands if they provide a sampling frequency twice that of the input signal. For example, in telephony, the usable voice frequency band ranges from approximately 300 Hz to 3400 Hz. Therefore, per the Nyquist–Shannon sampling theorem, the sampling frequency (8 kHz) must be at least twice the voice frequency (4 kHz) for effective reconstruction of the voice signal.

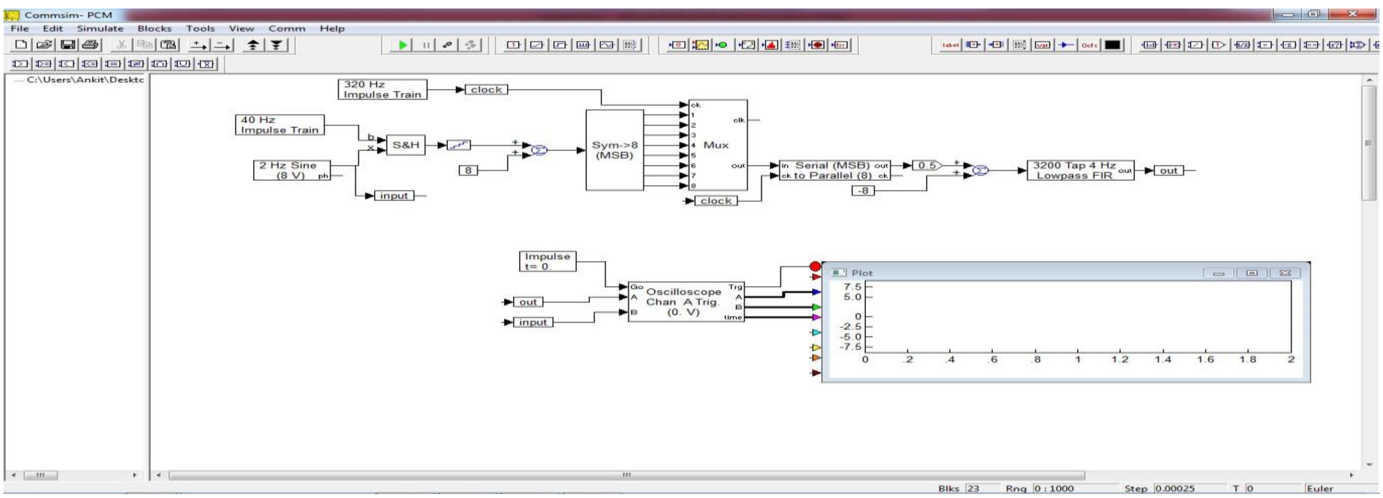
The electronics involved in producing an accurate analog signal from the discrete data are similar to those used for generating the digital signal. These devices are Digital-to-analog converters (DACs). They produce a voltage or current (depending on type) that represents the value presented on their digital inputs. This output would then generally be filtered and amplified for use.

BLOCK DIAGRAM:-

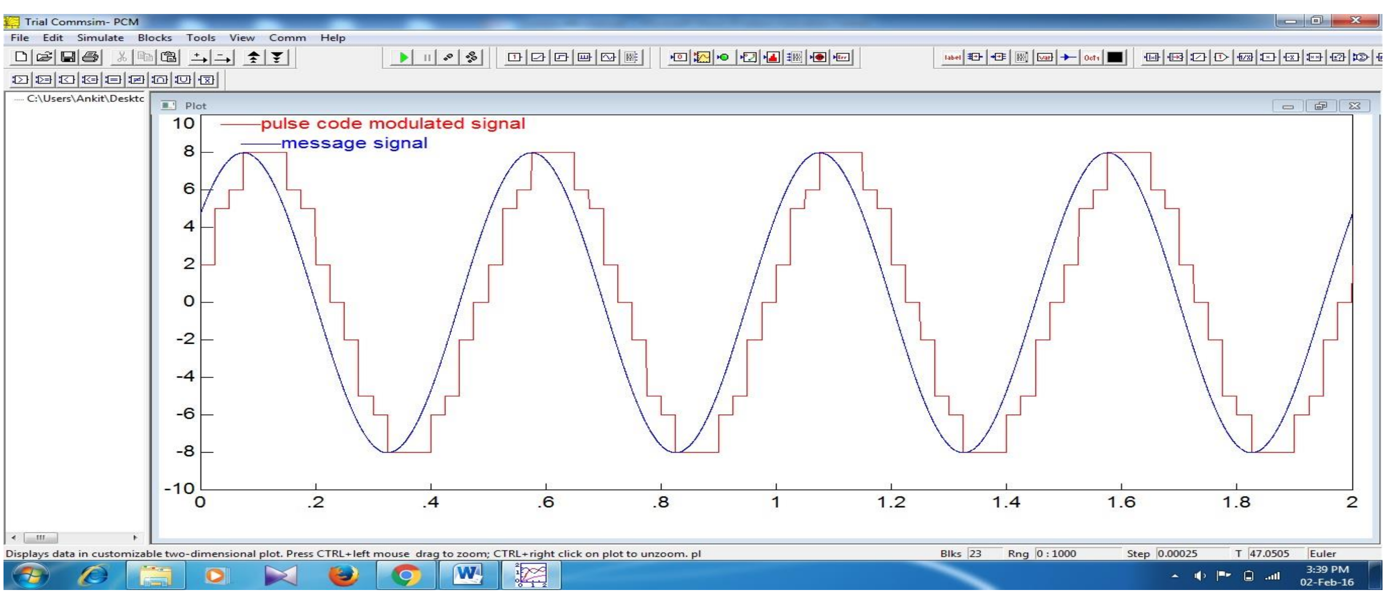
MODULATION



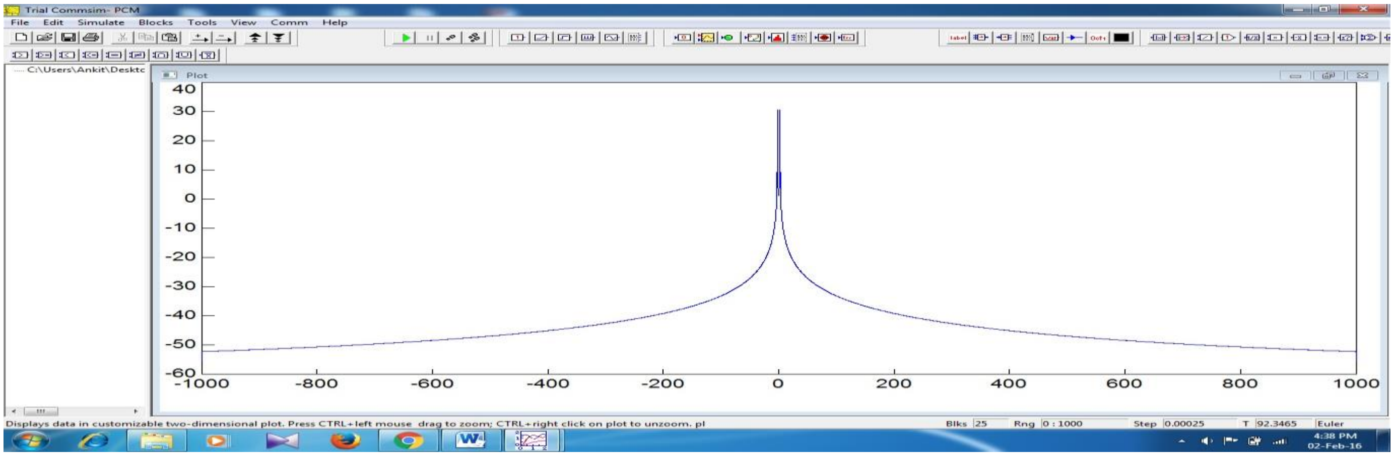
DEMODULATION



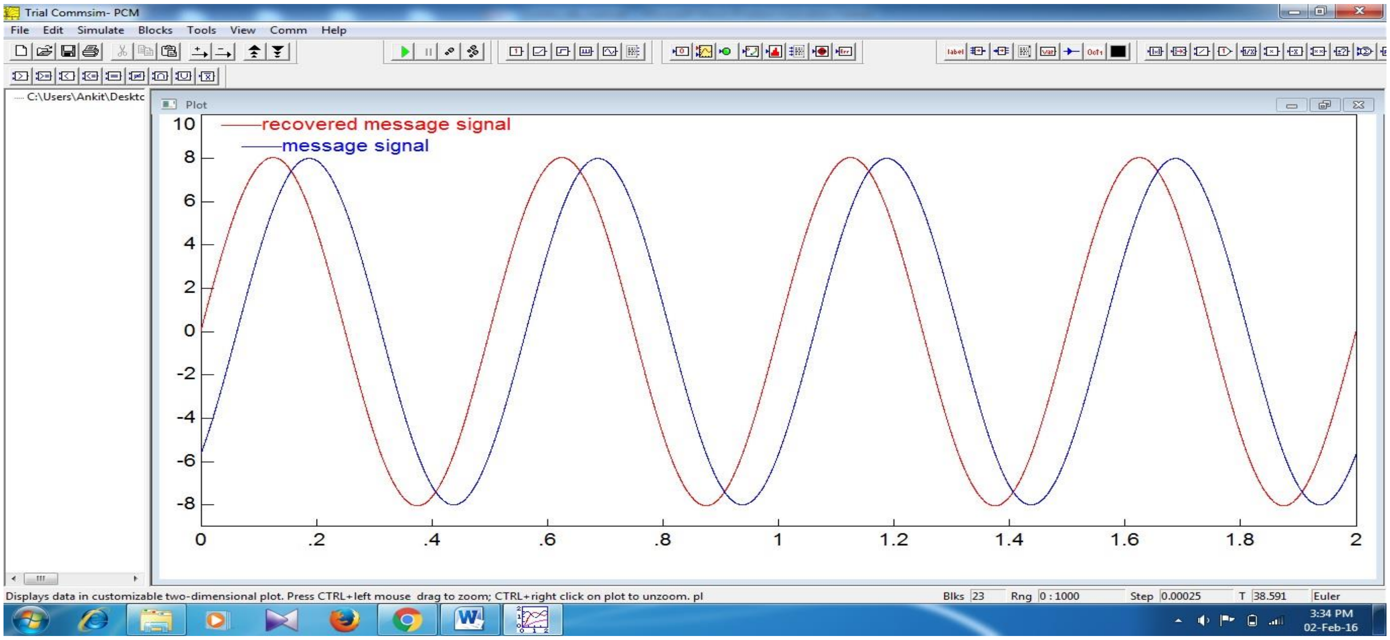
OUTPUT WAVEFORM:-



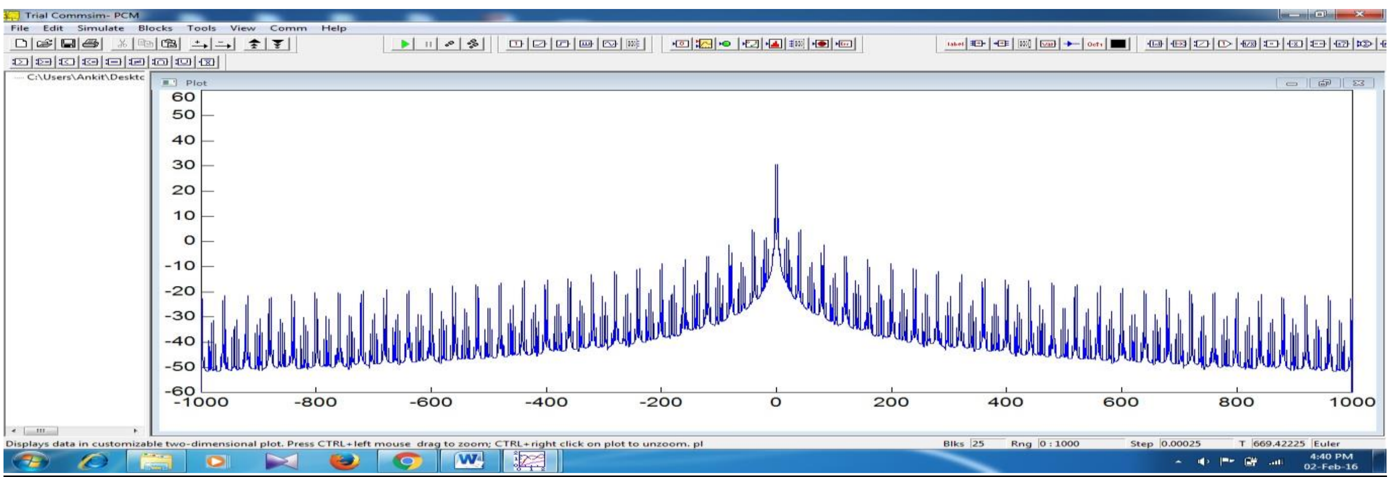
Modulation in Time Domain



Spectrum Analysis of Modulated Signal



<u>Recovered Message Signal</u>



<u>Spectrum Analysis of Recovered Signal</u>

CONCLUSION:-

In this experiment, we acquired a better understanding of pulse code modulation by further probing into sampling and quantization. It has been clearly seen that after modulation the input analog message signal is converted into the digital signal. And after demodulation again it is converted into analog signal similar to the input analog message signal.

EXPERIMENT:-7

AIM:-To simulate Delta modulation and demodulation

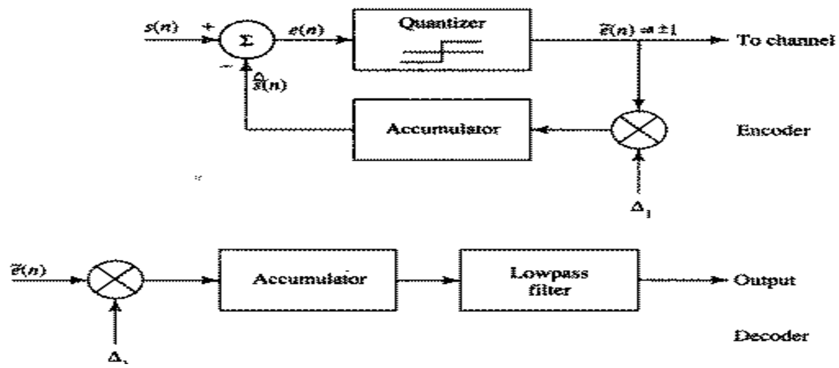
SOFTWARE USED:-Commsim 6.0

THEORY:-

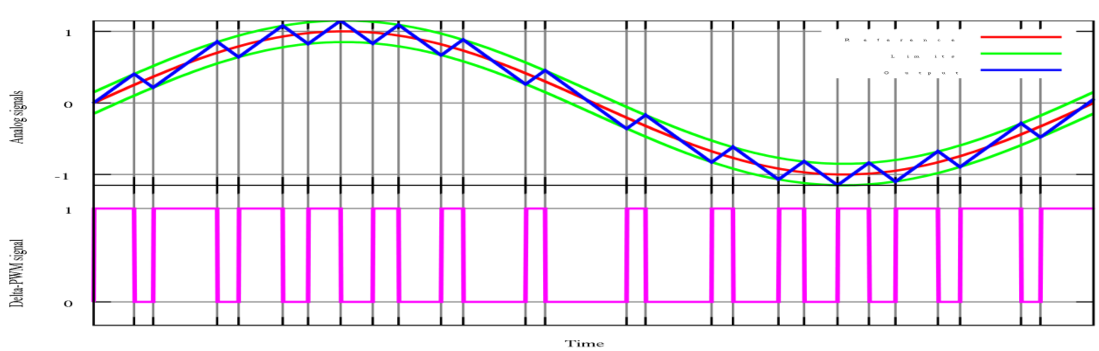
MODULATION

Delta Modulation (DM) is a simplified PCM. In some type of signals, the neighboring samples are closely correlated with each other. Therefore, once a sample value is known this enables the determination of the following sample values most probably. Thus, instead of sending the real value of each sample at each time, differences (variances) between adjacent samples are sent in DM. In DM, two-level quantizer and one-bit coding is used. Transmitted code pulses do not carry the data related to the message signal itself; instead they carry data regarding the differentials of the message function. The output of a delta modulator is a bit stream of samples at a relatively high rate, the value of each bit being determined according to whether the input message sample amplitude has increased or decreased relative to the previous sample.

The operation of a delta modulator is: i) periodically sample the input message, ii) make a comparison between the current sample and the preceding one and, iii) give a single bit as output which indicates the sign of the difference between two samples.



The system is in the form of a feedback loop. It is a continuous-time to discrete-time converter. In fact, it is a form of analog to digital converter. After the sampler is clocked, the resulting signal is the delta modulated signal. The output from the sampler is a bipolar signal, in block diagram being either ± Δ volts. If the output of 'Summer' (or comparator) is positive than the sample value of DM signal is +Δ, otherwise it is -Δ. The waveform of the DM signal is shown in bottom of Figure 2. It is fed back, in a feedback loop, via an integrator, to a summer. The integrator output is a saw tooth like waveform as shown in



The saw-tooth waveform is subtracted from the message and the difference – called as error signal – is the signal appearing at the summer output. An amplifier can also be used in the feedback loop (though not drawn in Figure 1) to control the loop gain and the size of the 'teeth' of the saw-tooth waveform. Signal from the integrator, which is a saw-tooth approximation to the message, is adjusted with the amplifier to match it as closely as possible.

The binary waveform illustrated at the bottom of Figure 2 is the signal transmitted. This is the delta modulated signal as stated above. The integral of the binary waveform is the saw-tooth approximation to the message. Low pass filtering of the saw-tooth (from the demodulator) gives a better approximation process.

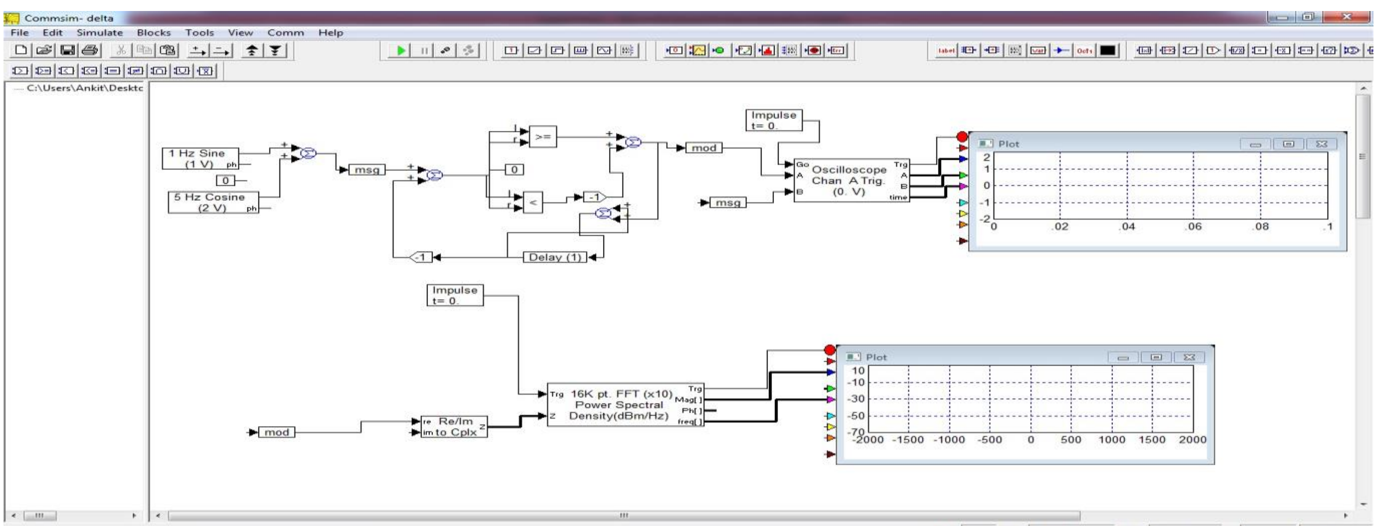
The unwanted products of the modulation process, observed at the receiver, are of two kinds. These are due to 'slope overload' and 'granularity'; those will not be examined in the content of this experiment.
In order to prevent some inappropriate modulation, the pace Δ should be selected according to the following equation.
$\Delta f_s > 2\pi f_x$; here fs is the sampling frequency, and fx is the greatest frequency component of the input signal.

DEMODULATION

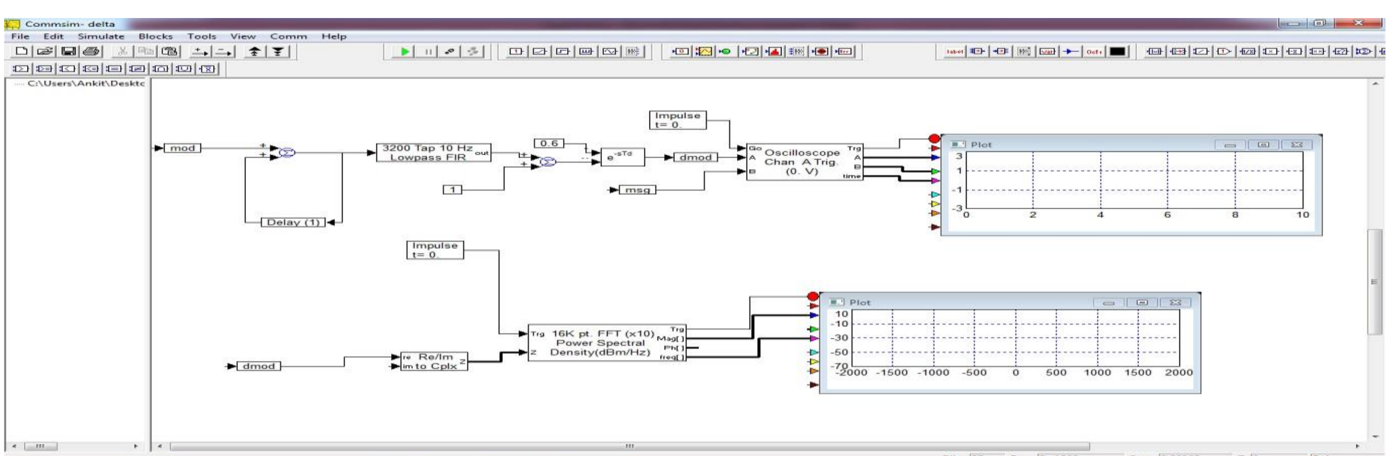
As a result of the delta modulation excludes the encoder, therefore, the structure of delta modulation is simpler than the structure of PCM. On the other hand, the DM signal only consists of a single bit of estimated error value (Ɛq(k)), so, the required transmitted bandwidth of DM signal is smaller than the PCM system.
we know that the DM signal (Xq(t))is a series diversity signal (Δ(t)), therefore, the structure of the delta demodulator will be easier to achieve. Figure 8-1 is the block diagram of delta demodulation. As a result of DM signal is a series diversity signal, so we use the integrator to accumulate the series signal, then we get
yq (t) = Δ(t) + Δ(t – Ts) + Δ(t-2Ts) + Δ(t – 3Ts)+……. (8-1)
where Δ(t) : The diversity signal, i.e. the magnitude of step value.
However, the accumulated series signal consists of high frequency harmonics, therefore, we use the low-pass filter to remove the high frequency parts. Then we can demodulate the DM signal and recover the low frequency signal, as shown in equation (8-2).
yD(t)=Lp{yq(t)} = x(t) …….(8-2)
From figure 8-1, the bipolar square wave will pass through the integrator and obtain a waveform, which is similar to the audio signal. Then the output signal will pass through a low-pass filter and finally, we can obtain the audio signal.

BLOCK DIAGRAM:-

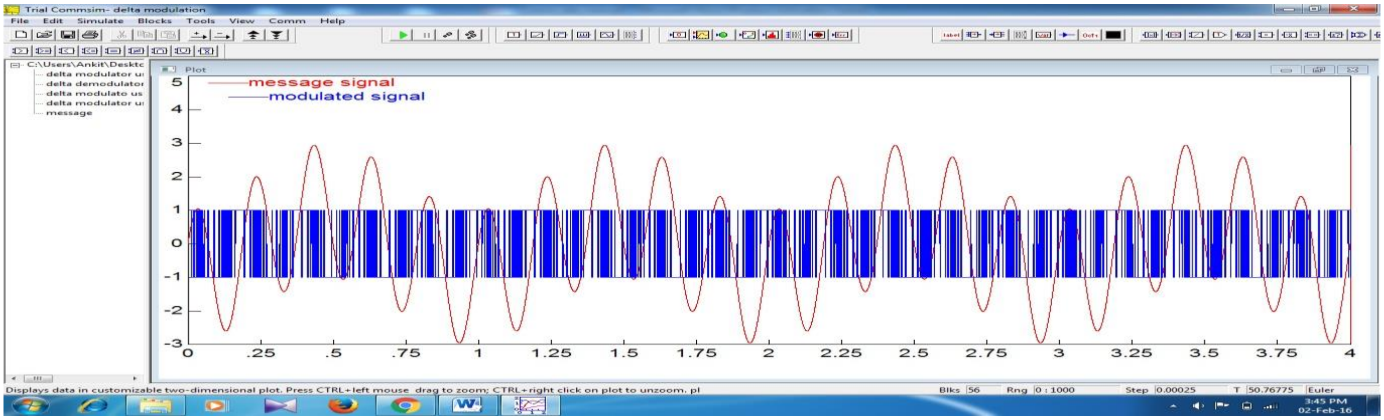
MODULATION



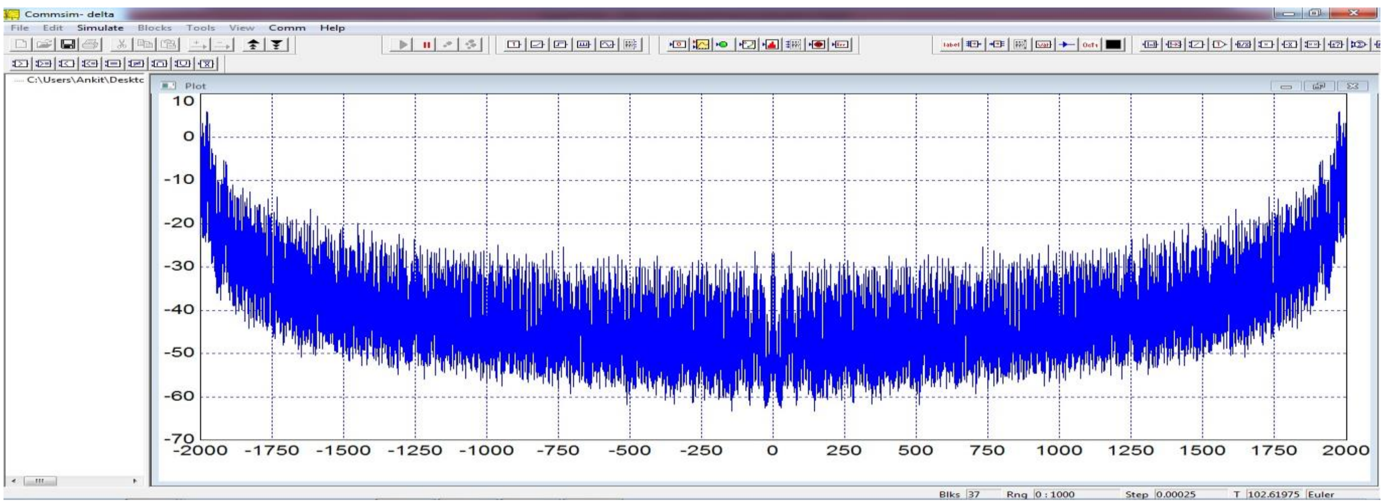
DEMODULATION



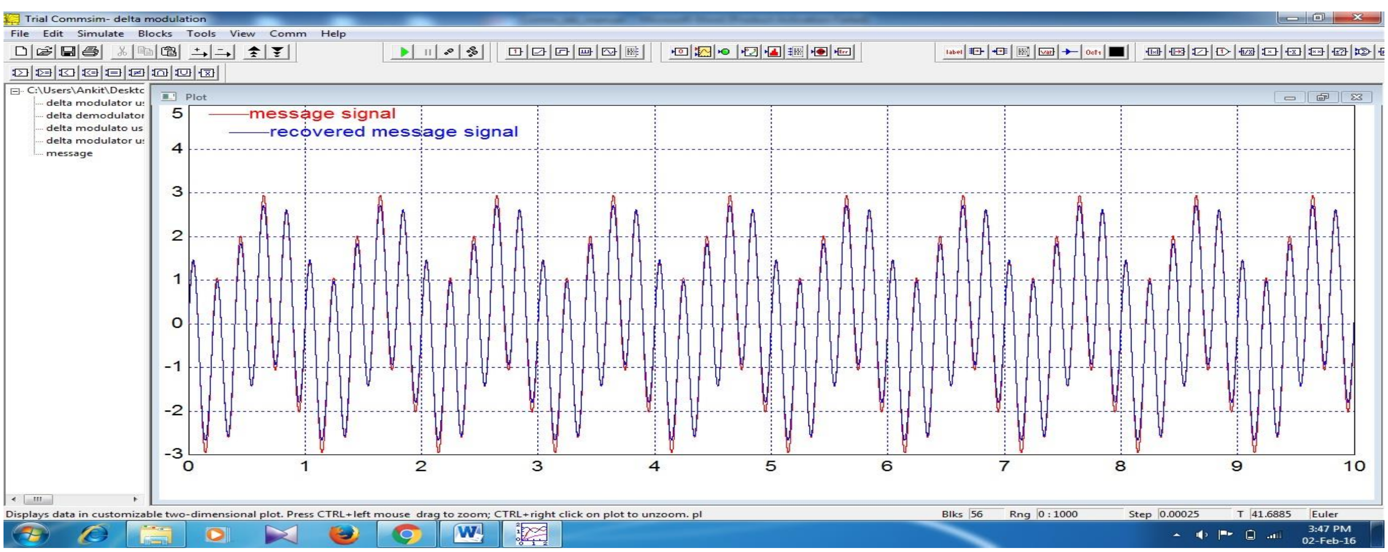
OUTPUT WAVEFORM:-



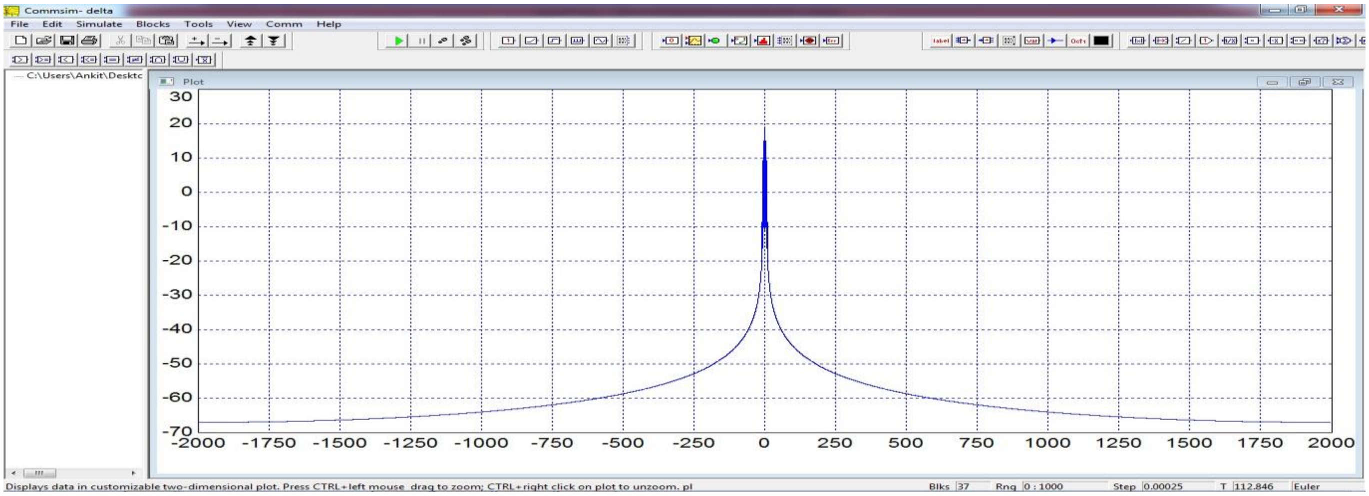
Modulation in Time Domain



Spectrum analysis of Modulated Signal



Recovered Message Signal



Spectrum analysis of Recovered Message Signal

CONCLUSION:-

In this simulation it has been clearly seen that the input signal is converted into digital signal and after demodulation the original message signal is recovered.

EXPERIMENT:-8

AIM:-To simulate Pulse Amplitude modulation and demodulation

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

In pulse modulation method, the carrier is no longer a continuous signal but consists of a pulse train. Some parameter of which is varied according to the instantaneous value of the modulating signal. There are two types of pulse modulation systems as under:-

i) Pulse amplitude Modulation (PAM).
ii) Pulse Time Modulation (PTM).

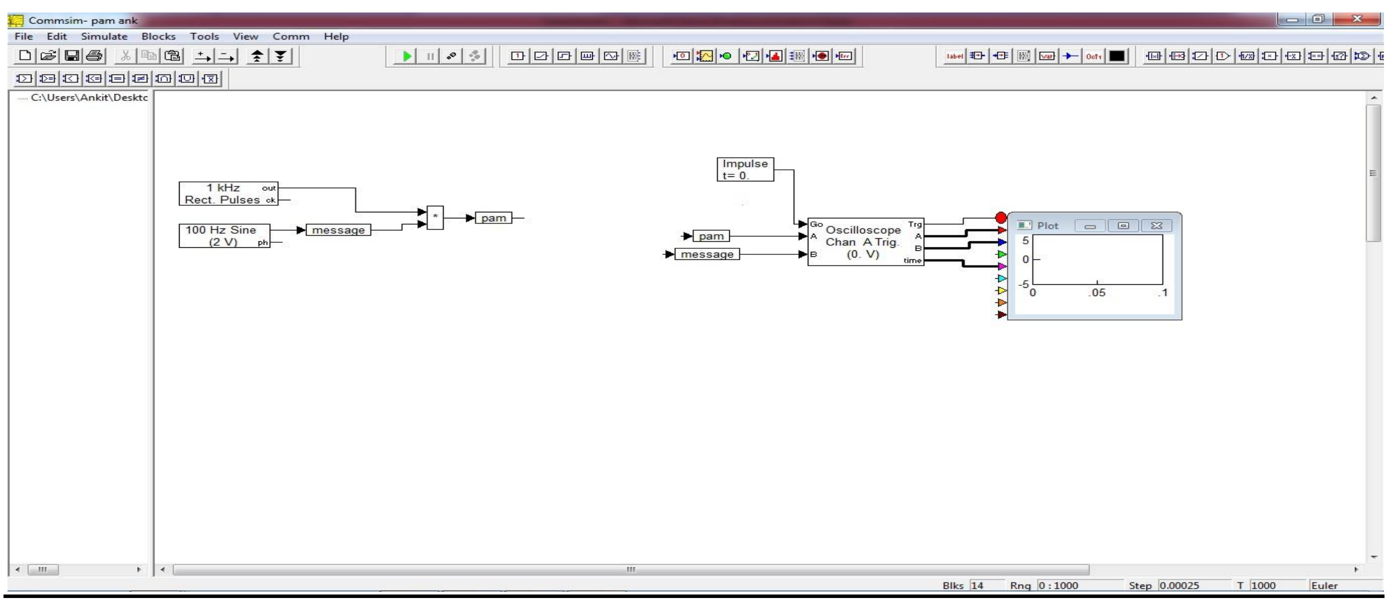
Pulse amplitude modulation may be defined as that type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to the instantaneous value of the modulating or message signal. In fact, the pulses in a PAM signal may be of flat top type or natural type or ideal type. The reason of using flat top PAM is that during the transmission, the noise interferes with the top of the transmitted pulse and this noise can be easily removed if the PAM pulse has flat top.

DEMODULATION

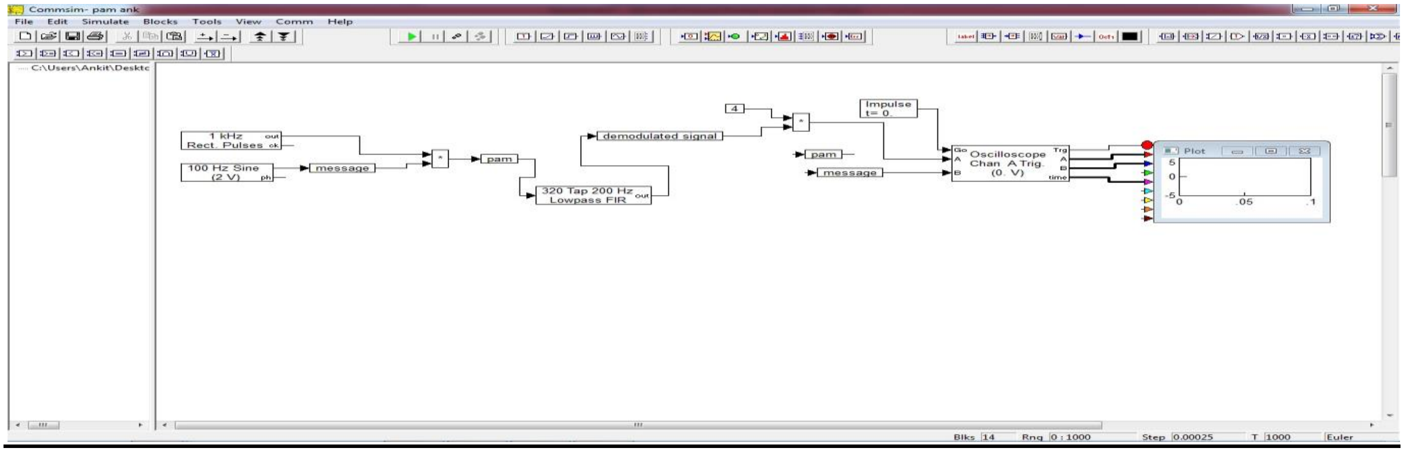
For pulse amplitude modulated signals, the demodulation is done using a hold circuit. In this method, the received PAM signal is allowed to pass through a holding circuit and a low pass filter(LPF).

BLOCK DIAGRAM:-

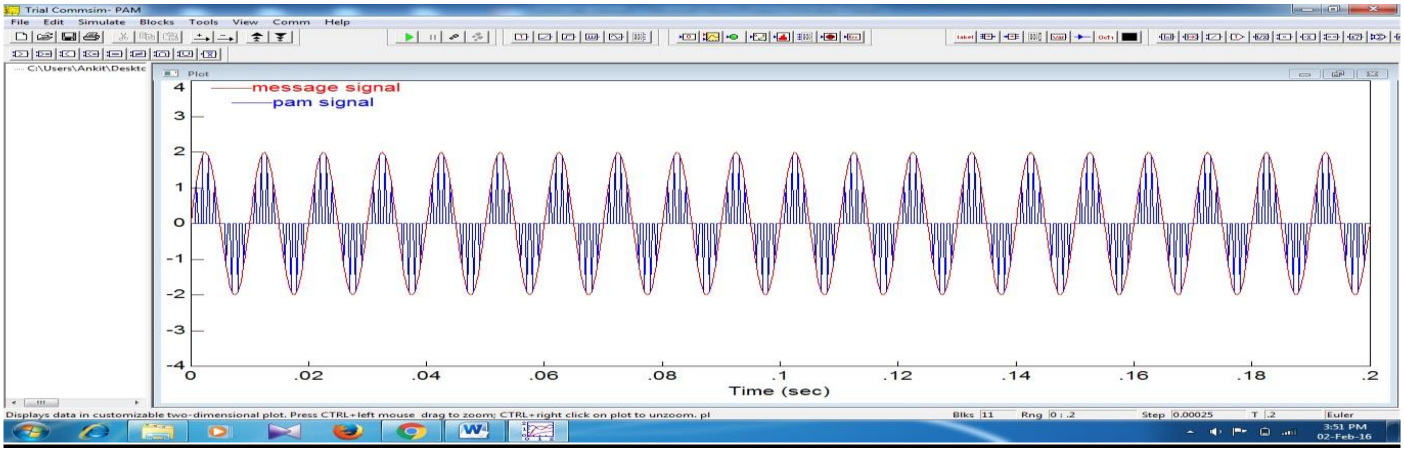
MODULATION



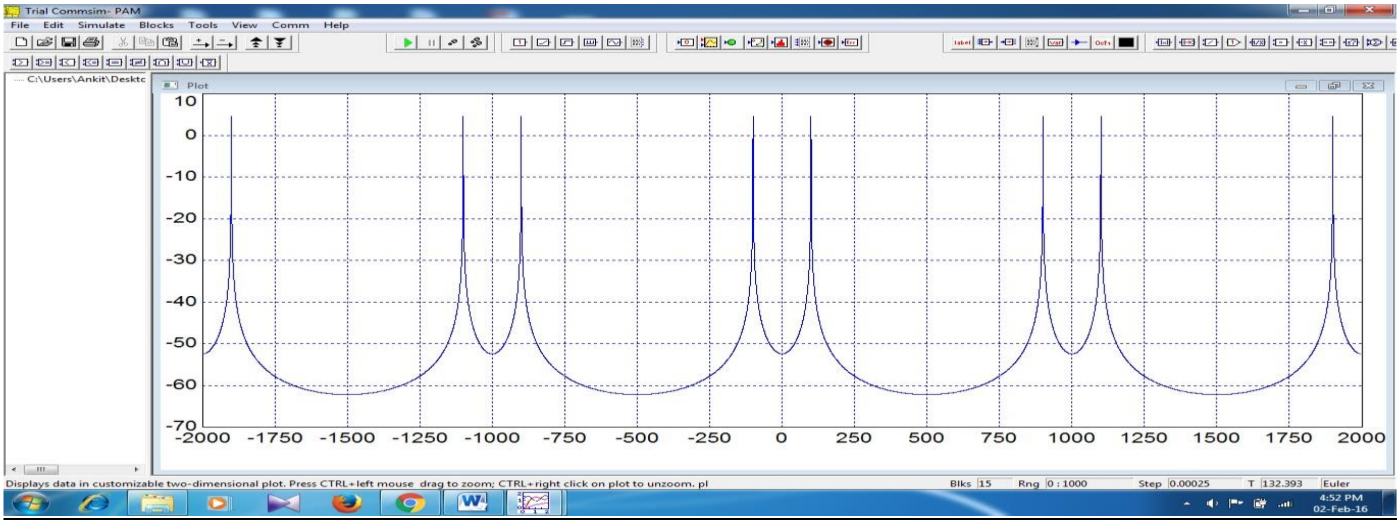
DEMODULATION



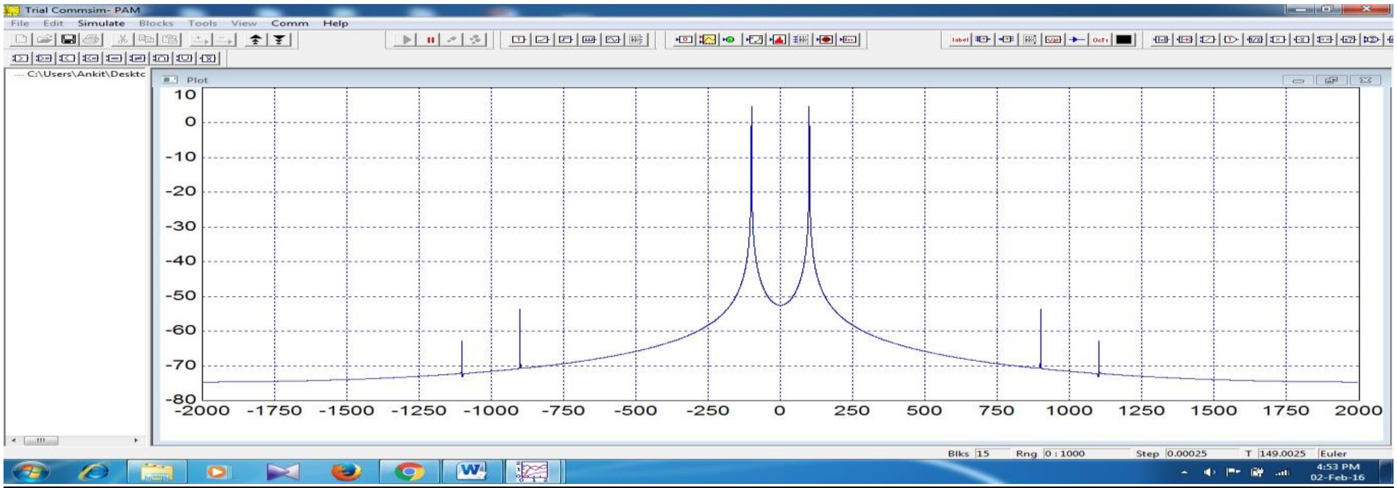
OUTPUT WAVEFORM:-



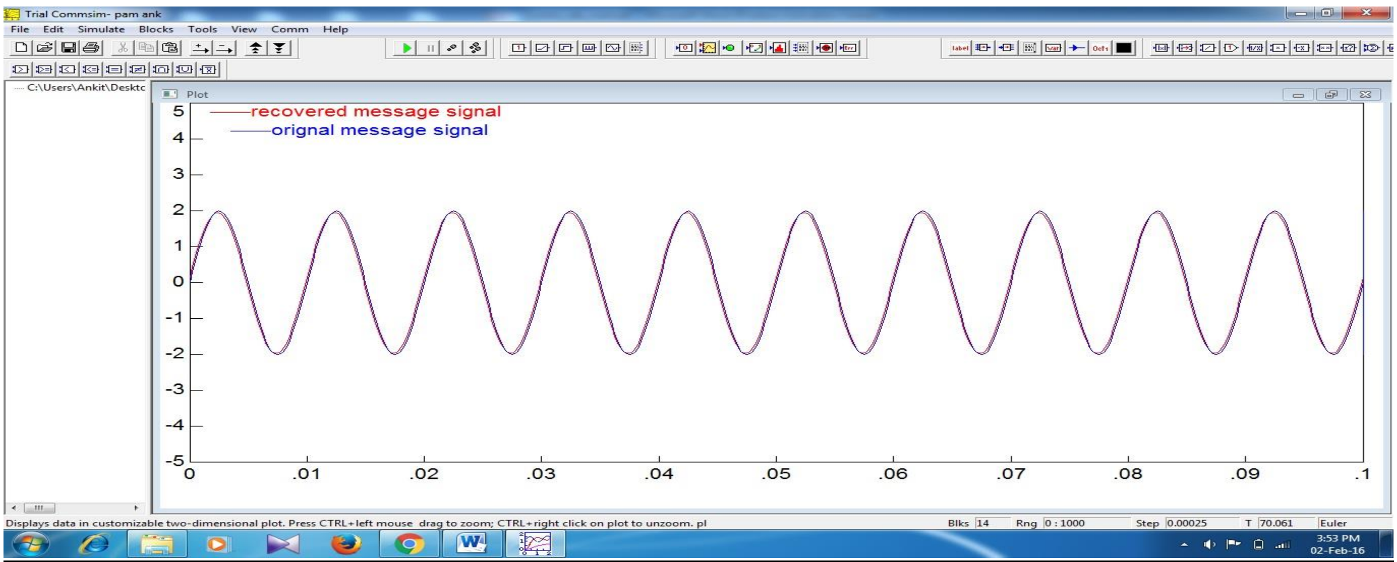
Modulation in Time Domain



Spectrum Analysis of Modulated Signal



<u>Spectrum Analysis of Recovered Signal</u>



<u>Recovered Message Signal</u>

CONCLUSION:-

In this experiment it has been clearly seen that the rectangular pulse is varied in accordance with message signal and after demodulation the input message signal is recovered.

EXPERIMENT:-9

AIM:-**To simulate Pulse Width modulation and demodulation**

SOFTWARE USED:-**Commsim 6.0**

THEORY:-

MODULATION

Pulse-width modulation uses a rectangular pulse wave whose pulse width is modulated resulting in the variation of theaverage value of the waveform. If we consider a pulse waveform f(t), with period T, low value y_{min}, a high value y_{max} and a duty cycle D (see figure 1), the average value of the waveform is given by:

$$\bar{y} = \frac{1}{T}\int_0^T f(t)dt.$$

As f(t) is a pulse wave, its value is y_{max} for $0 < t < D.T$ and y_{min} for $D.T < t < T$ The above expression then becomes:

$$\begin{aligned}\bar{y} &= \frac{1}{T}\left(\int_0^{DT} y_{max}dt + \int_{DT}^{T} y_{min}dt\right) \\ &= \frac{1}{T}(D.T.y_{max} + T(1-D)y_{min}) \\ &= D.y_{max} + (1-D)y_{min}.\end{aligned}$$

This latter expression can be fairly simplified in many cases where $y_{min} = 0$ or $\bar{y} = D.y_{max}$ From this, it is obvious that the average value of the signal ($\bar{y}$) is directly dependent on the duty cycle D.

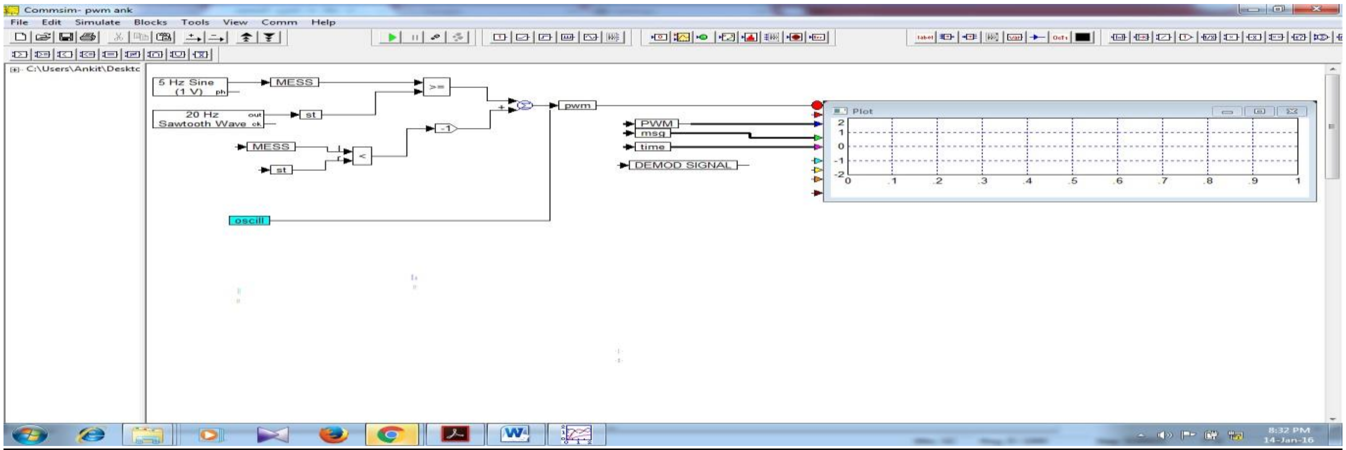
The simplest way to generate a PWM signal is the intersective method, which requires only a sawtooth or triangular waveform (easily generated using a simple oscillator) and a comparator. When the value of the reference signal (the red sine wave in figure 2) is more than the modulation waveform (blue), the PWM signal (magenta) is in the high state, otherwise it is in the low state.

DEMODULATION

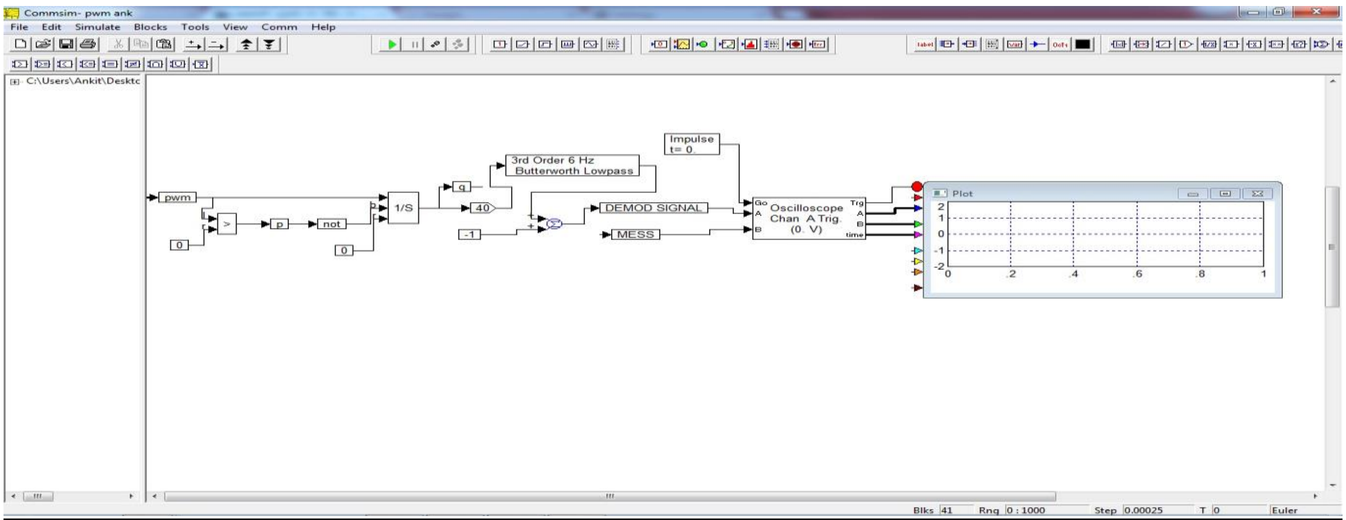
In the demodulation of PWM signal the received PWM signal is applied to the Schmitt trigger circuit. This Schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse detector. The ramp generator produces ramps for the duration of pulses such that heights of ramps are proportional to the width of PWM pulses. The maximum ramp voltage is retained till the next pulse. On the other hand synchronous pulse detector produces reference pulses with constant amplitude and pulse width. These pulses are delayed are specific amount of delay. The delayed reference pluses and the output of a ramp generator is added with the help of adder. The output of adder is given to the level shifter. Here, negative offset shifts the waveform. Then the negative part of the waveform is clipped by rectifier. Finally the output of rectifier is passed through the low pass filter to recover the modulating signal.

BLOCK DIAGRAM:-

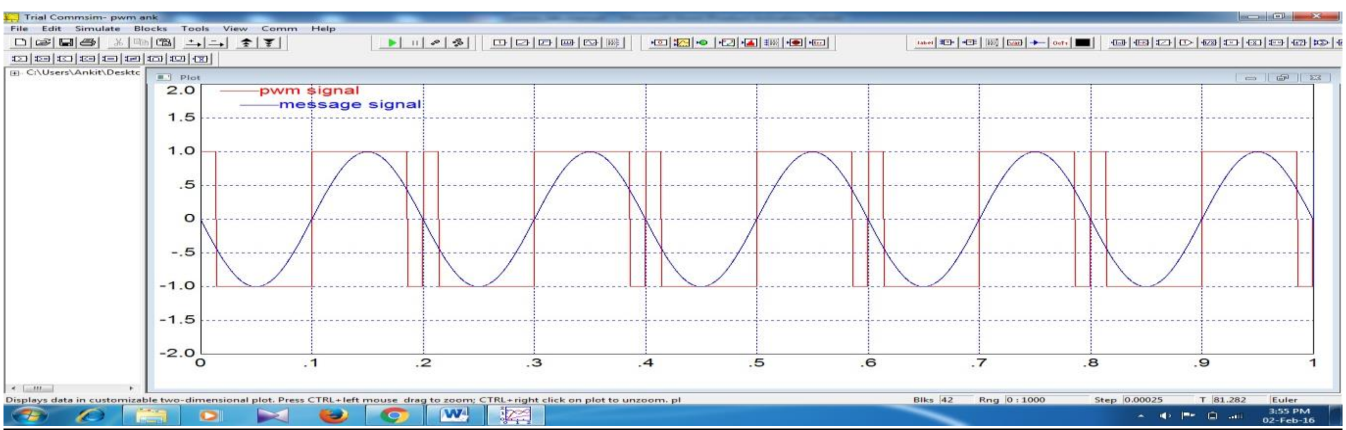
MODULATION



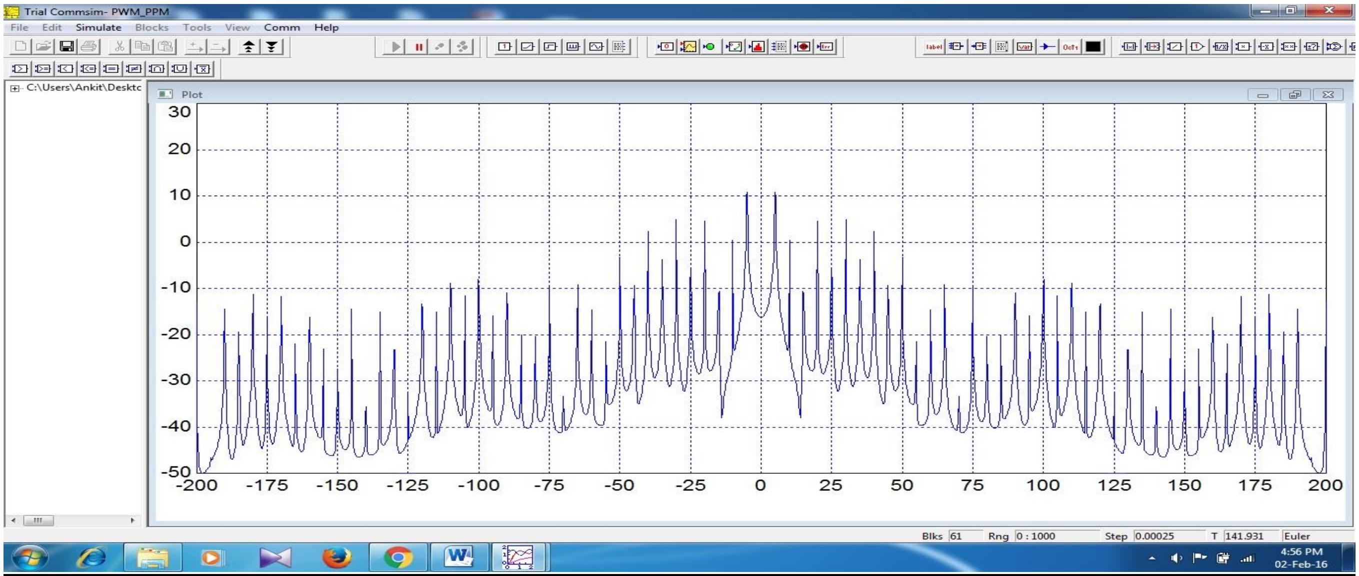
DEMODULATION



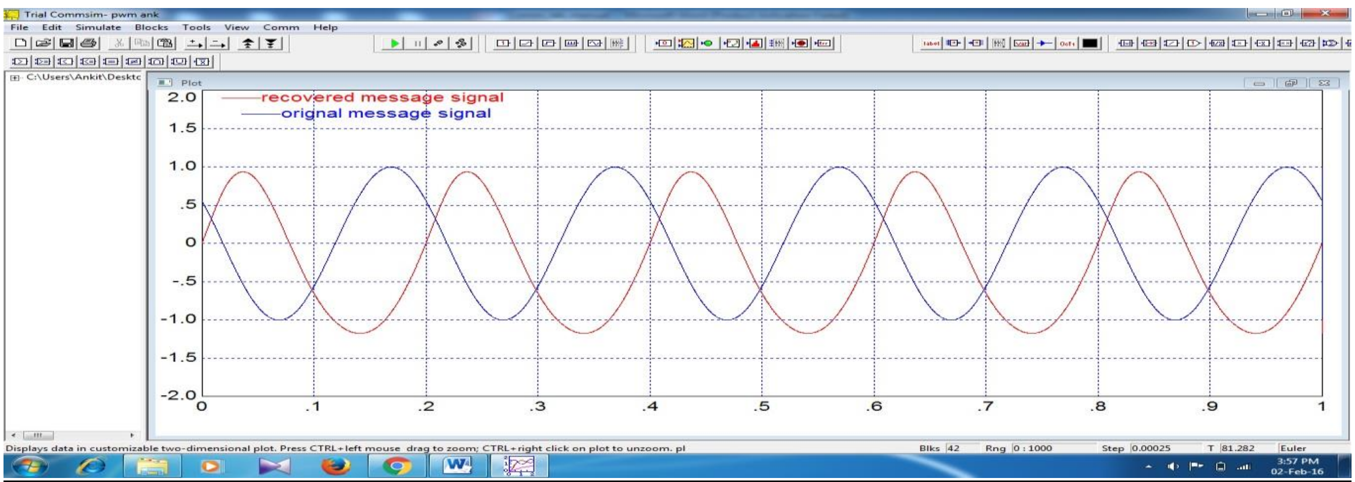
OUTPUT WAVEFORM:-



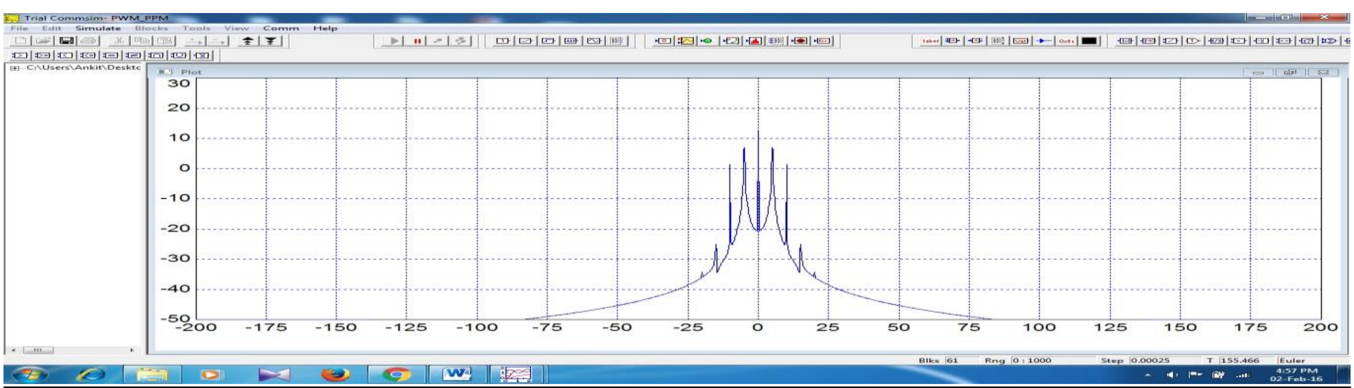
Modulation in Time Domain



<u>**Spectrum Analysis of Modulated Signal**</u>



<u>Recovered Message Signal</u>



<u>**Spectrum Analysis of Recovered Signal**</u>

CONCLUSION:-

In this experiment the width of the rectangular pulse is changes in accordance with the message signal and after demodulation we are getting the original message signal.

EXPERIMENT:-10

AIM:-**To simulate Pulse Position modulation and demodulation**

SOFTWARE USED:-**Commsim 6.0**

THEORY:-

MODULATION

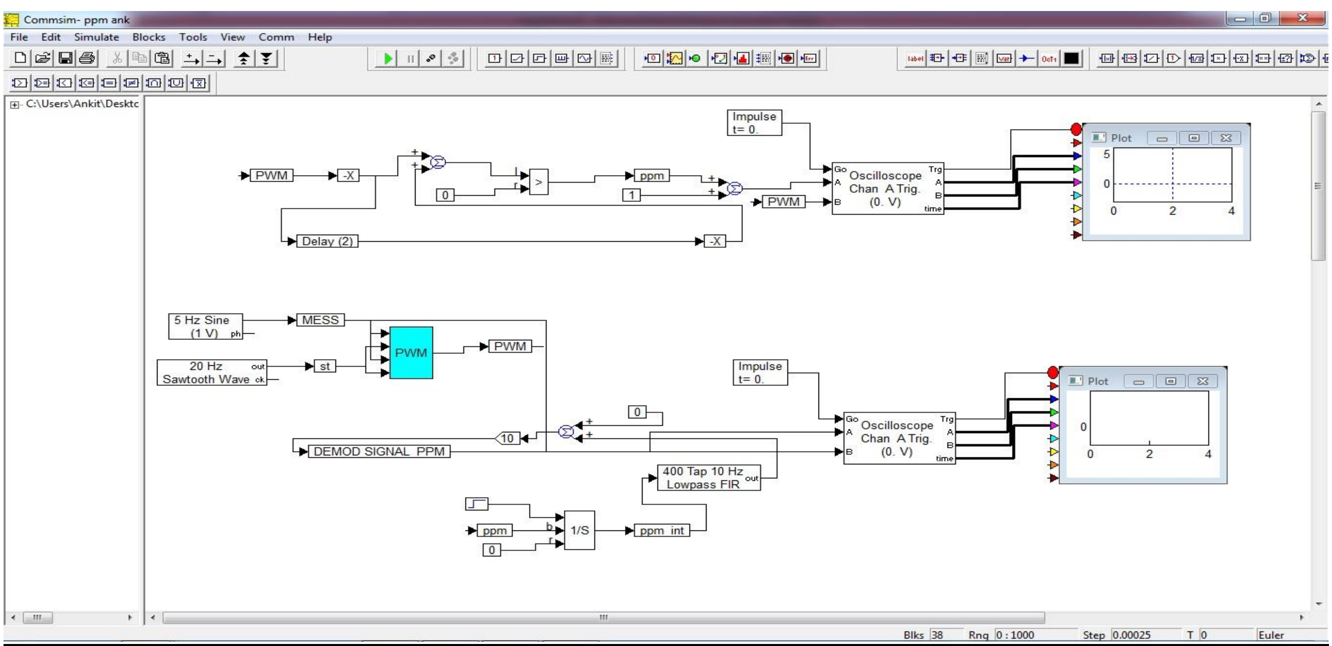
In this system, the amplitude and width of the pulses are kept constant,while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the modulating signal. Thus, the transmitter has to send synchronizing pulses to keep the transmitter and receiver in synchronism. As the amplitude and width of the pulses are constant, the transmitter handles constant power output, a definite advantage over the PWM. But the disadvantage of the ppm system is the need for transmitter – receiver synchronization. Pulse position modulation is obtains from pulse width modulation. It consists of differentiator and a monostablemulti-vibrator. The input to the differentiator is the PWM waveform. The differentiator generates positive and negative spikes corresponding to leading and trailing edge of the PWM waveform. Diode is used to bypass the positive spikes. The negative spikes are used to the trigger the monostablemulti-vibrator. The monostablemulti-vibrator then generates the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform**.**

DEMODULATION

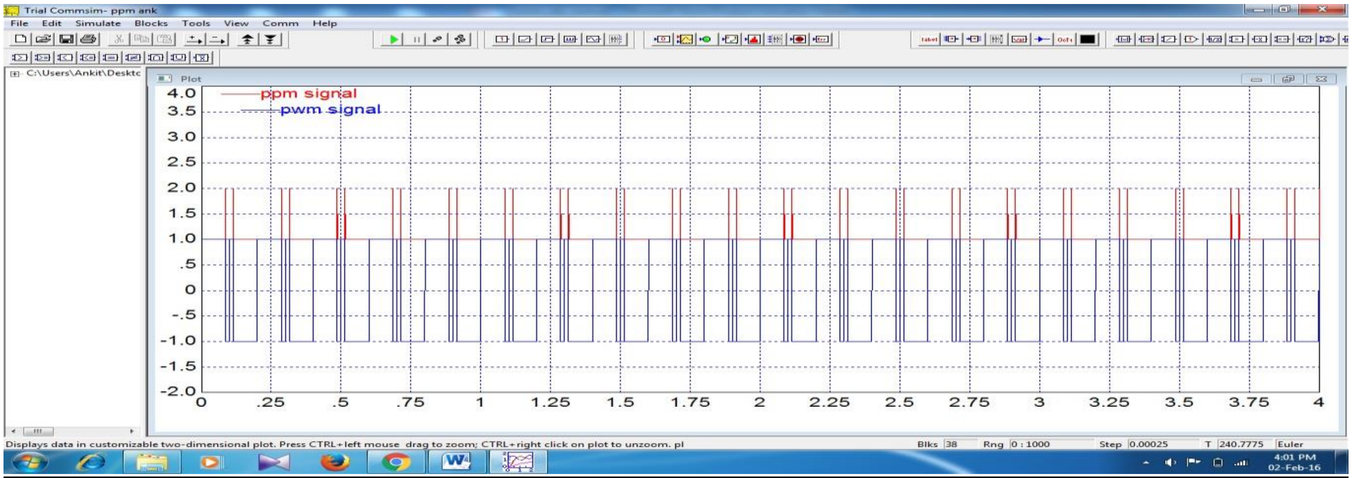
In case of pulse position modulation, it is customary to convert the received pulses that vary in position to pulses that vary in length. Flip flop circuit is set or turned on when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip-flop circuit is reset or turned off at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip-flop. The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

BLOCK DIAGRAM:-

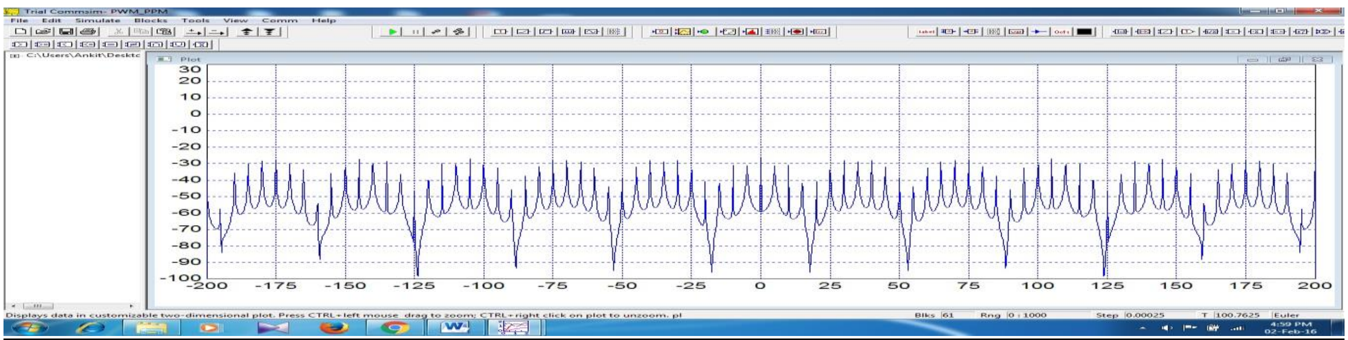
MODULATION& DEMODULATION



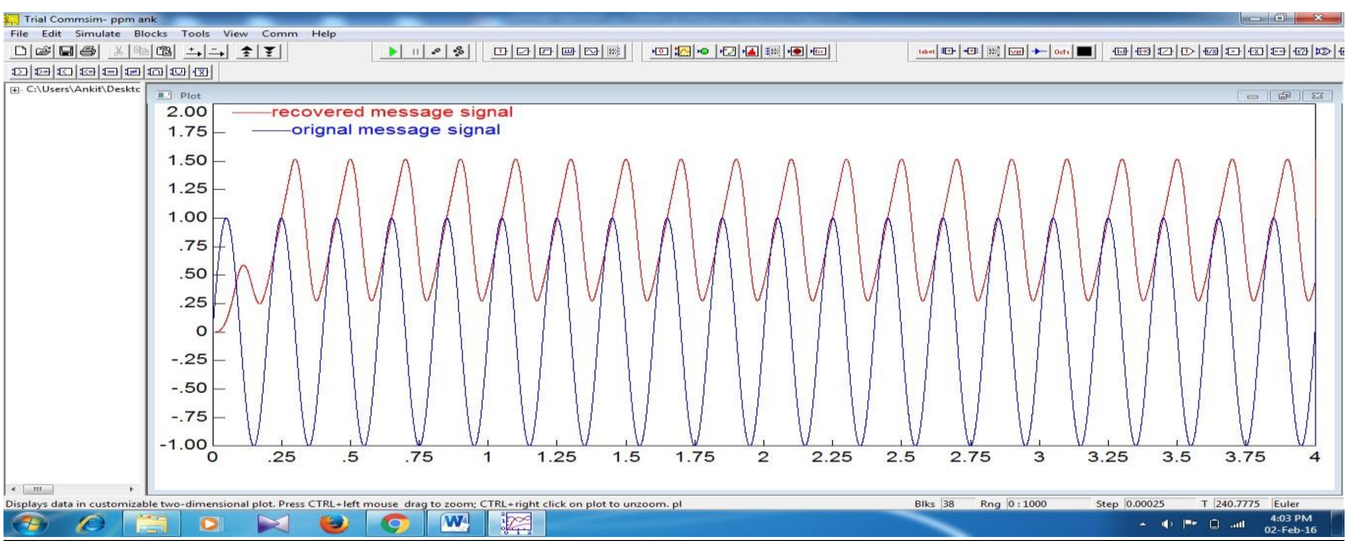
OUTPUT WAVEFORM:-



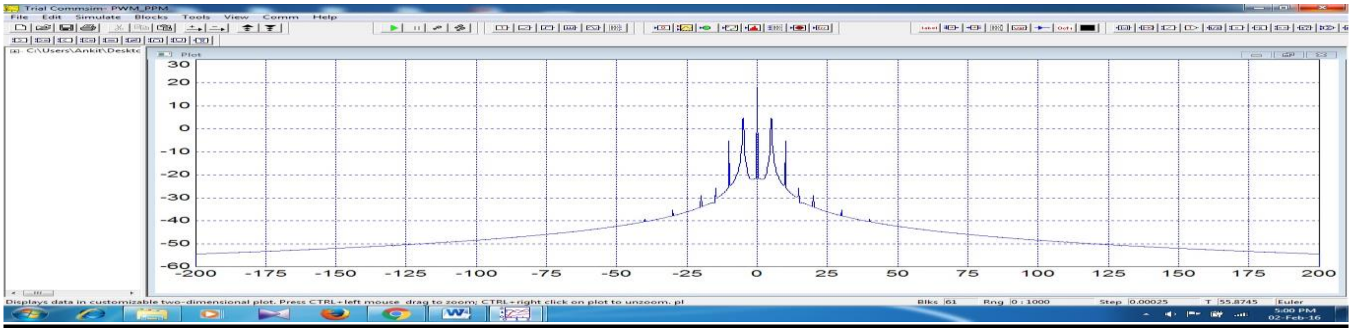
Modulation in Time Domain



Spectrum Analysis of Modulated Signal



Recovered Message Signal



Spectrum Analysis of Recovered Message Signal

CONCLUSION;-

In this experiment it has been clearly seen that the position of PWM is shifted. And after demodulation we get the original message signal.

EXPERIMENT:-11

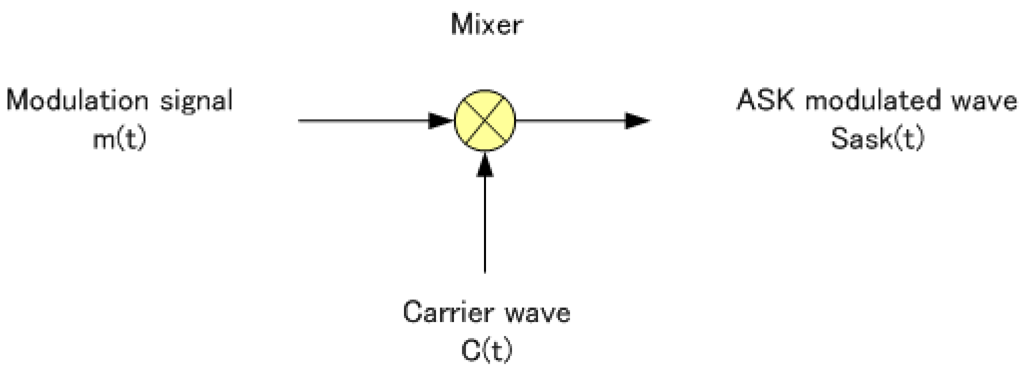
AIM:-To simulate Amplitude Shift Keying(ASK)

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

Amplitude modulation is often associated with the modulation method of AM radio. When the carrier wave is C(t) and the modulating signal is m(t), the modulated wave Sam is as follows.With ASK/OOK modulation, a modulator is used to shift the amplitude of the carrier wave proportionally to the amplitude of the modulating signal (digital signal). In the case of a digital signal, the modulating signal applied to the modulator is called digital modulation. The modulator is the same as for analog modulation. With an ASK modulating signal, the digital signal value is either -1 or +1, and when it is changed to 0 and +1 it is called OOK. With OOK, the amplitude direction of the modulated wave is indicated by the presence or absence of a carrier wave. In other words, when the signal is 0, there is no carrier wave and when it is 1, there is a carrier wave. The spectrum of the ASK modulated wave is centered on the carrier frequency and the square wave spectrum which is the modulating signal takes a spread-out form.



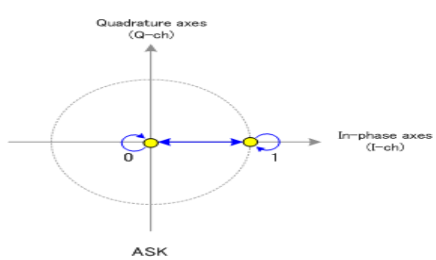
Expressed as a formula, carrier wave C(t) takes the following form.

$$C(t) = A_c.\cos(2\pi.f_c.t)$$

The ASK modulated wave is modulating signal m(t) multiplied by carrier wave C(t), and is expressed as a formula as follows.

$$S_{ask}(t) = m(t).c(t) = m(t).A_c.\cos(2\pi.f_c.t)$$

◇ASK ASK can also be expressed as the constellation in the figure below, with the information at amplitude point 0 and 1 at phase 0 rad corresponding to 0 and 1. 0 rad means that even if the information signal changes, there is no phase shift.



◆**Demodulation**

the receivers use envelope detection, a type of asynchronous detection, to simplify their structure and reduce their cost. Envelope detection is a detection method that can only perform demodulation when the envelope of the modulated signal indicates a modulating signal. The applet can be set with a modulation factor higher than 1, but in this case, since the envelope of the modulated signal does not indicate a modulating signal, envelope detection is not possible and it's necessary to use synchronous detection. The applet allows you to switch between synchronous detection and asynchronous detection. You can confirm that demodulation can be carried out properly even with a modulation index over 1 if you use synchronous detection.

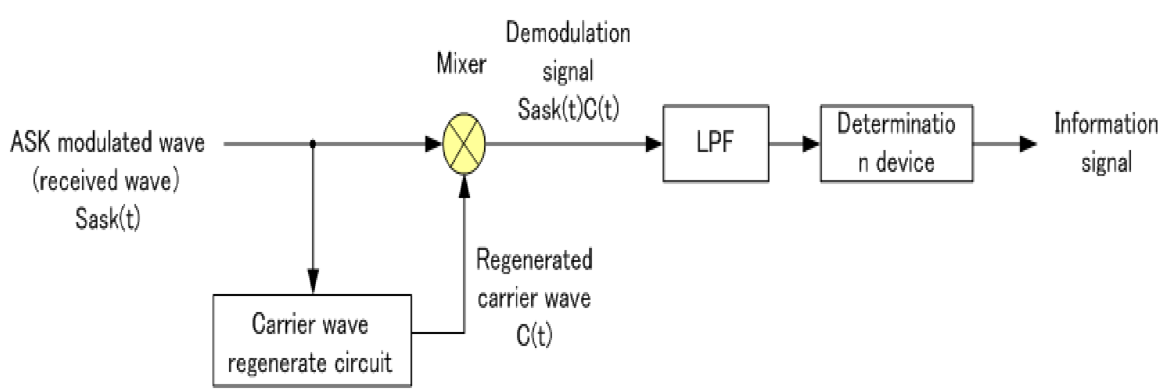
Synchronous detection:

The modulated wave is multiplied with the carrier frequency which is exactly the same frequency and phase as the transmission carrier wave.

Asynchronous detection:

The received signal is multiplied by a squarer. Synchronous detection When the ASK modulated wave is multiplied to become a regenerated carrier wave, it is as follows. The second term in the braces is an unwanted component, so only the LPF signal component is added. The determination device determines the level of the signal and the transmitting end information signal is obtained.

◆**Synchronous detection**



When the ASK modulated wave is multiplied to become a regenerated carrier wave, it is as follows.

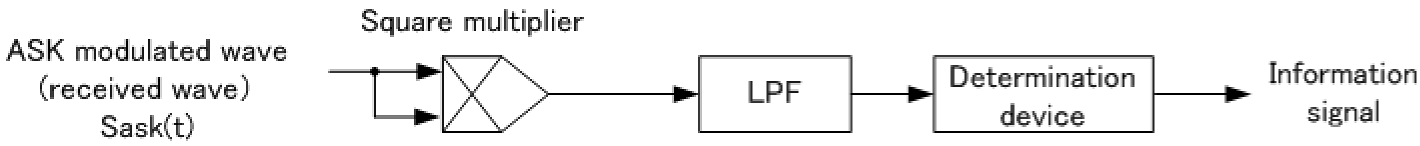
$$S_{ask}(t).c(t) = m(t).A_c.cos^2(2\pi.f_c.t)$$
$$= m(t).A_c.\tfrac{1}{2}\{1 + \cos(4.\pi.f_c.t)\}$$

The second term in the braces is an unwanted component, so only the LPF signal component is added.

$$(S_{ask}(t).C(t))_{LPF} = \frac{A_c}{2}m(t)$$

The determination device determines the level of the signal and the transmitting end information signal is obtained.

◆Example of asynchronous detection



The ASK modulated wave is multiplied by a squarer as follows.

$$S_{ask}{}^{2}(t) = m^2(t).cos^2(2\pi.f_c.t)$$
$$=m^2(t).\frac{A_c^2}{2}\{1 + \cos(4.\pi.f_c.t)\}$$

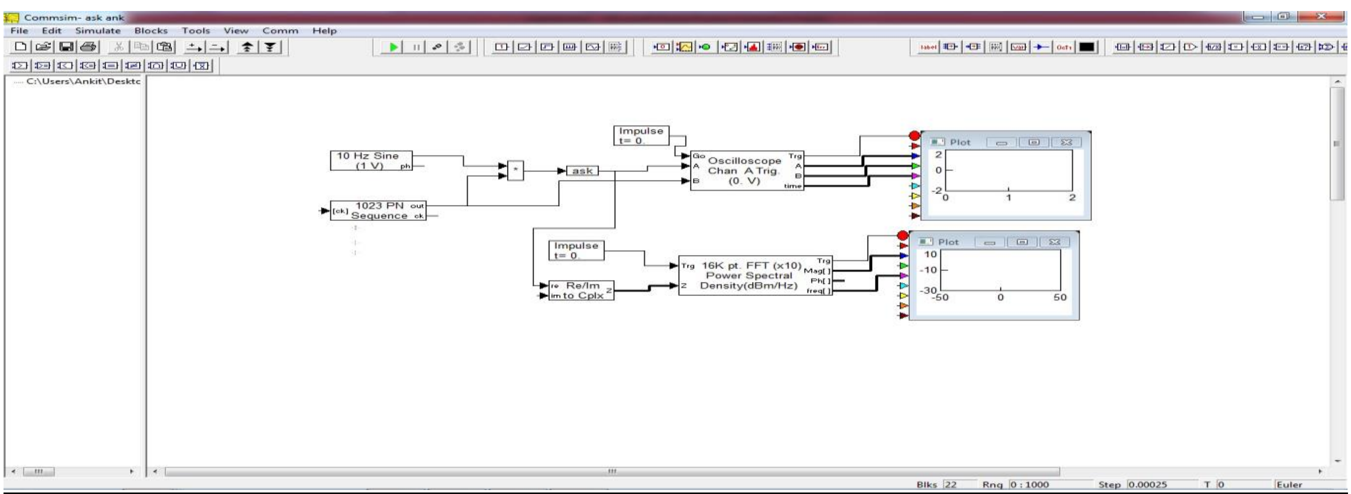
The second term in the braces is an unwanted component, so only the LPF signal component is added.

$$(S_{ask}{}^{2}(t))_{LPF} = \frac{A_c^2}{2}m^2(t)$$

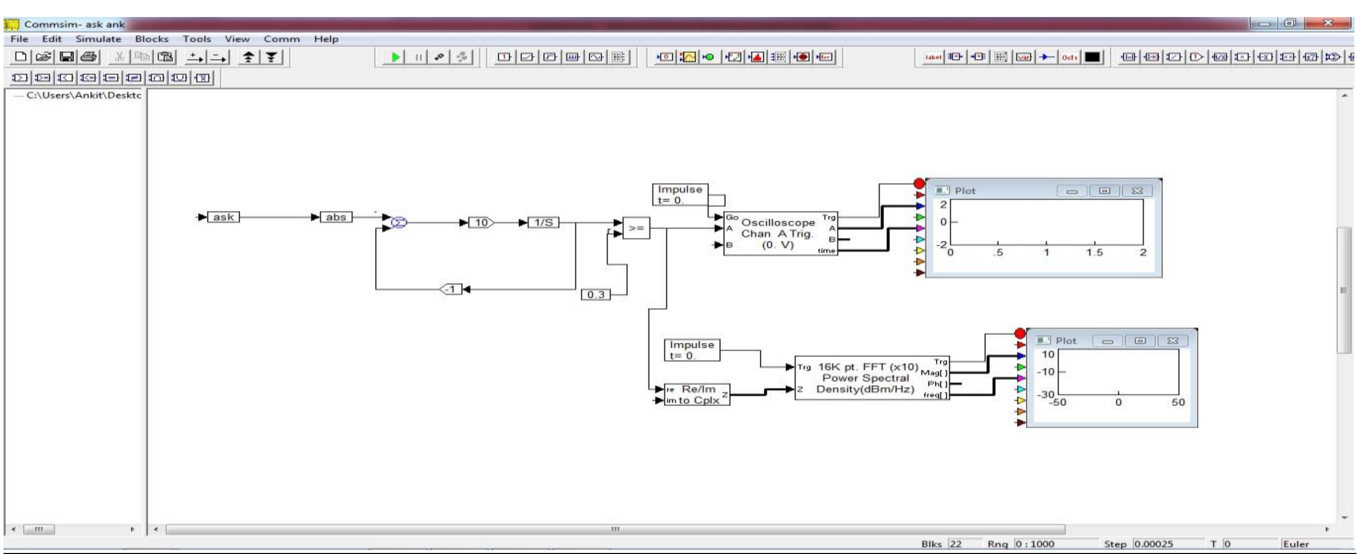
The determination device determines the level of the signal and the transmitting end information signal is obtained.

BLOCK DIAGRAM:-

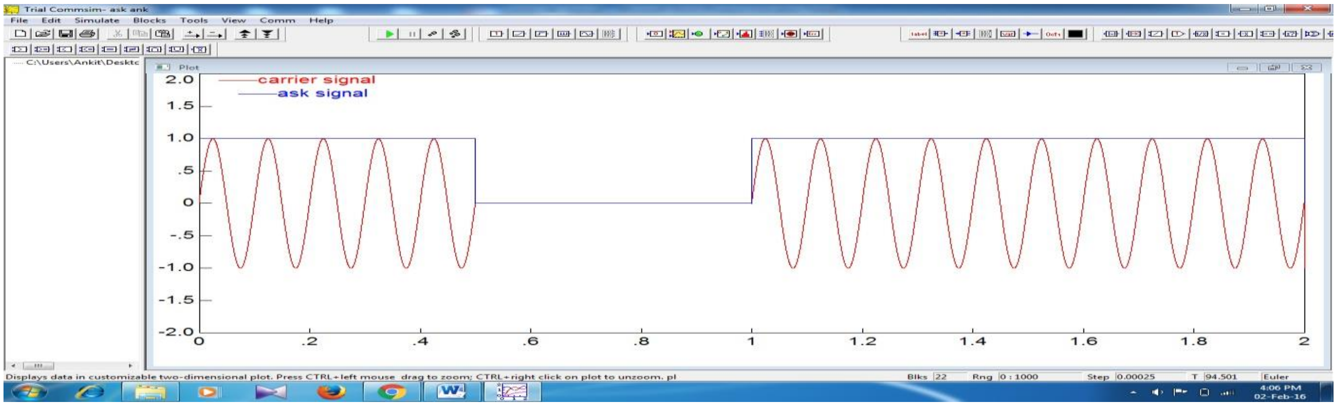
MODULATION



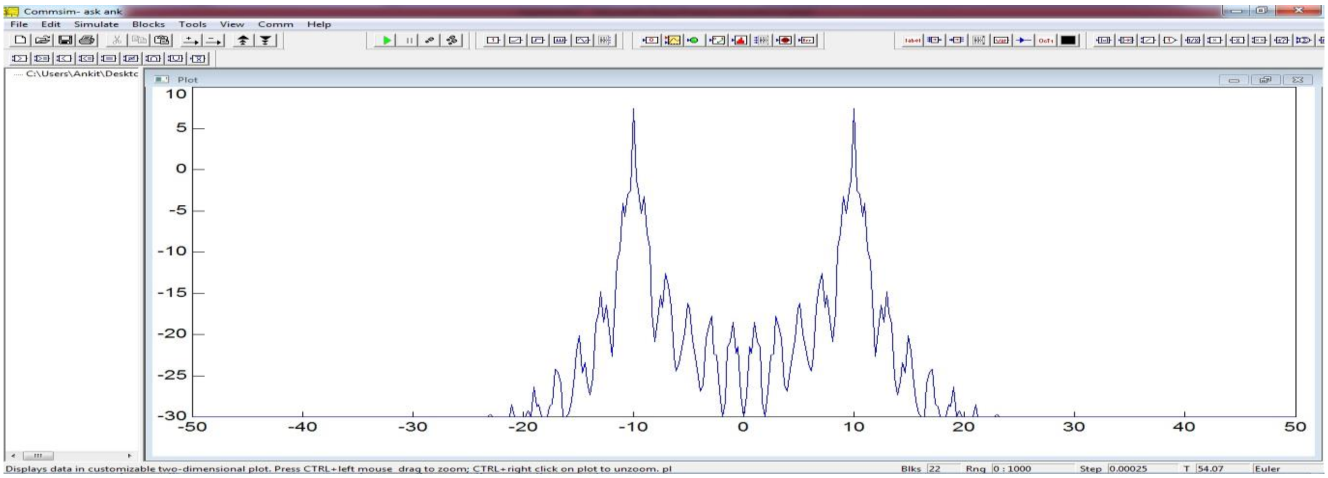
DEMODULATION



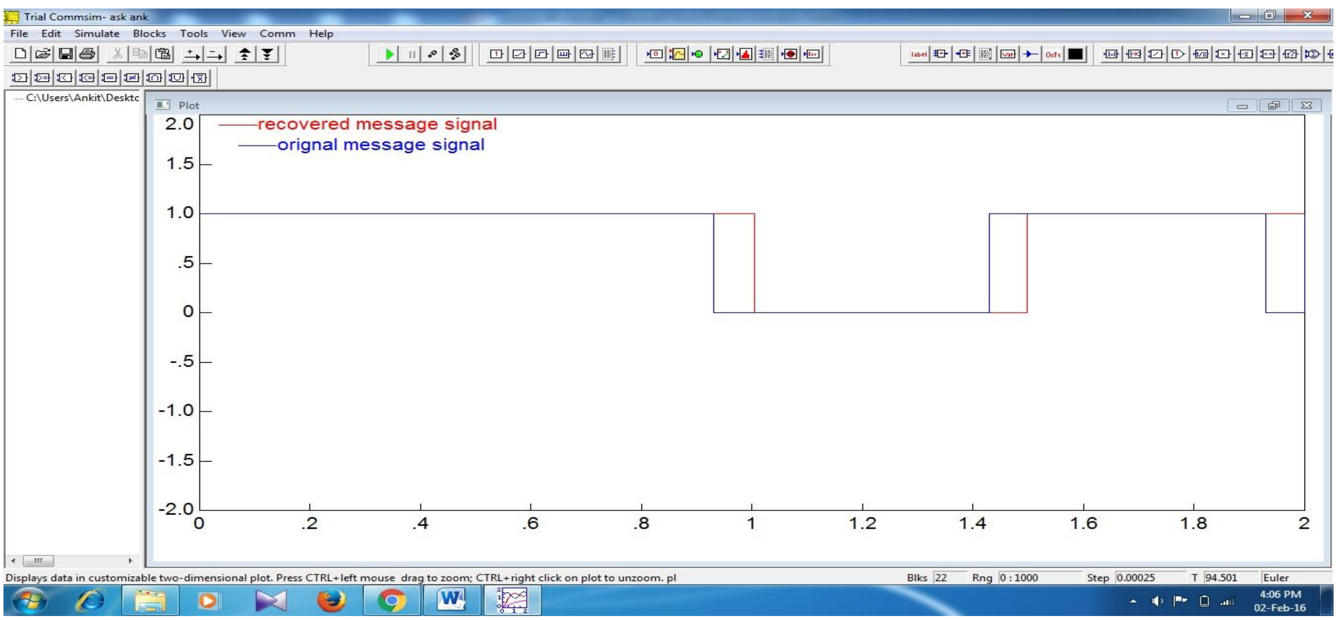
OUTPUT WAVEFORM:-



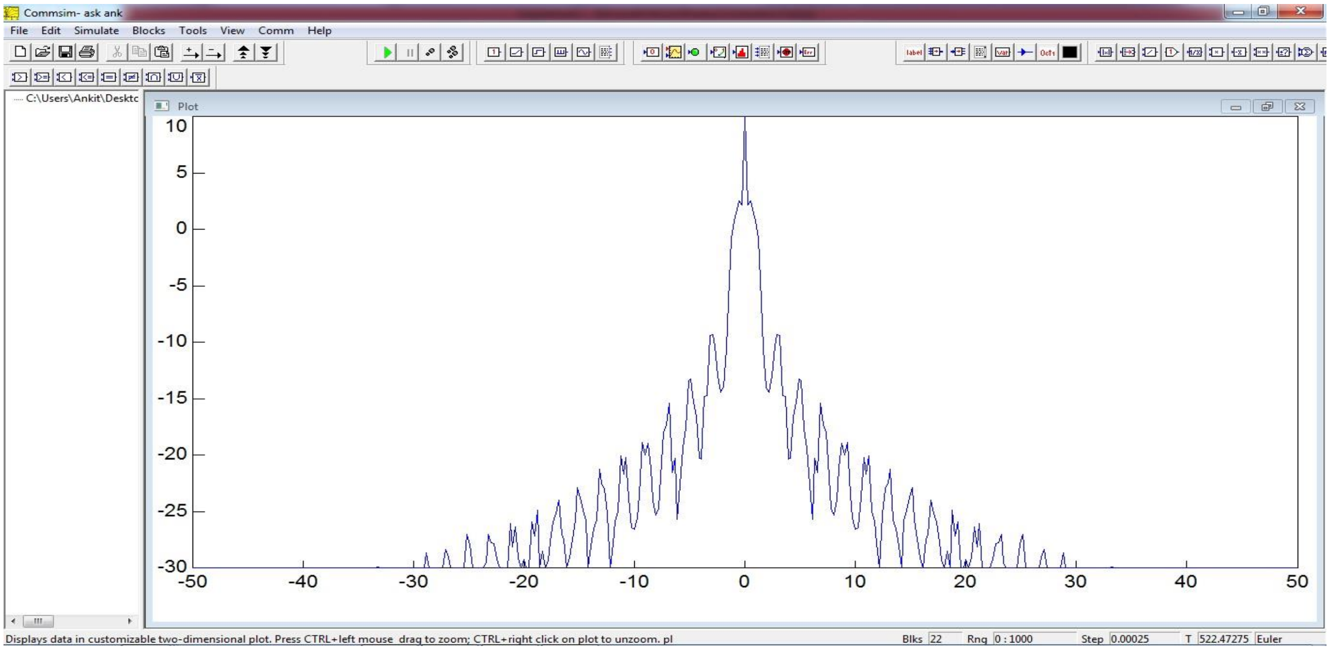
Modulation in Time Domain



Spectrum analysis of Modulated Signal



Recovered Message Signal



Spectrum analysis of Recovered Message Signal

CONCLUSION:-

In this experiment the input carrier signal is varied in accordance with the rectangular pulse and after demodulation we are getting the same input rectangular pulse.

EXPERIMENT:-12

AIM:-To simulate Phase Shift Keying(PSK)

SOFTWARE USED:-Commsim 6.0

THEORY:-

MODULATION

With BPSK modulation, the level of the modulating signal (code 0, 1) is changed to a dipolar NRZ signal, and the signal and carrier wave are multiplied with a mixer. The modulating signal spectrum is shifted directly to the carrier frequency band.

BPSK modulation changes the phase of the carrier wave C(t) proportionally to the information signal. The carrier wave C(t) is a sine wave with the following properties.

$$C(t) = A_c.\cos(2\pi.f_c.t + \emptyset_c)$$

BPSK changes the phase of the carrier wave C(t) by 0 degrees or 180 degrees with regard to the 1-bit 2-status information, resulting in the following formula.

$$S_{psk}(t) = A_c.\cos\{2\pi.f_c.t + \emptyset_c + \pi.m(t)\}$$

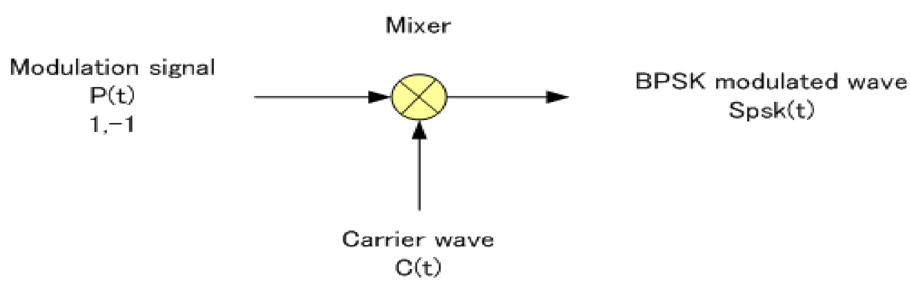
When the initial phase of the carrier wave is Φc = 0;

$$S_{psk}(t) = A_c.\cos\{\pi.m(t)\}.\cos(2\pi.f_c.t)$$

If p(t) is;

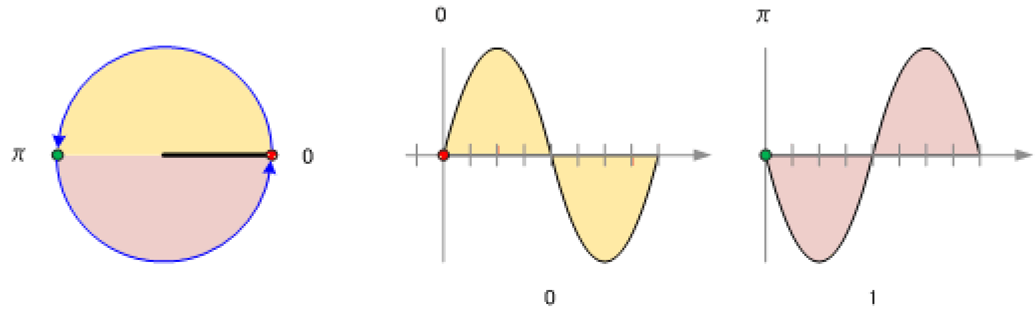
$$\mathrm{P(t) = cos\{\pi.m(t)\}}$$

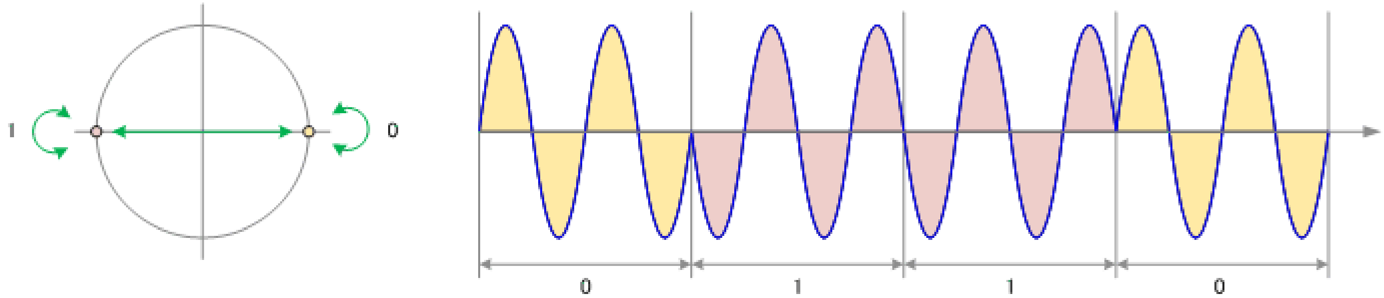
Which is the modulated wave, and the BPSK modulated wave Spsk(t) is multiplied by a squarer as follows.



$$S_{psk}(t) = P(t).c(t) = p(t).A_c.\cos(2\pi.f_c.t)$$

The modulating signal p(t) follows the information signal m(t)(0 or 1), taking the values 1 and -1. With BPSK, 1 bit of the information signal can be expressed with 1 symbol. As shown in the figure below, information is assigned to the 0 and π phases of the carrier wave as 0 and 1 respectively.



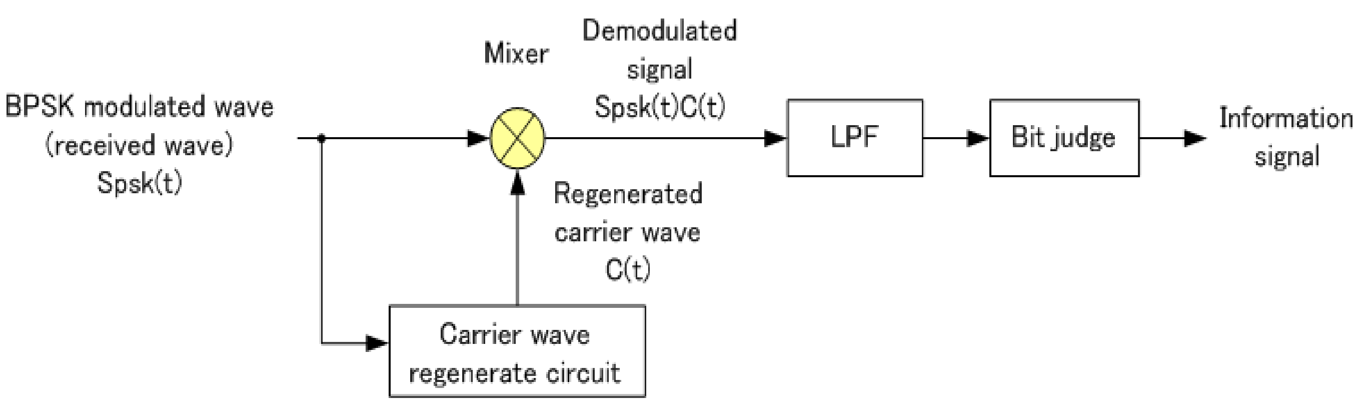


◆BPSK demodulation

The BPSK modulating signal is demodulated with a synchronous detection system. The synchronous detection system uses a modulator to multiply the received signal and regenerated carrier wave. The frequency and phase of the regenerated carrier wave must match (synchronize with) the carrier wave used on the transmitting end. If multiplication is performed with a regenerated carrier wave that is not synchronized, the amplitude level may vary, the signal polarity may be reversed, and many errors may occur, making it unusable. Frequency multiplication and other methods are used to regenerate the carrier wave.

When the received signal is regenerated, the result is as follows.

$$S_{psk}(t).c(t) = p(t).A_c.cos^2(2\pi.f_c.t)$$
$$=p(t).A_c.\tfrac{1}{2}\{1 + \cos(4.\pi.f_c.t)$$

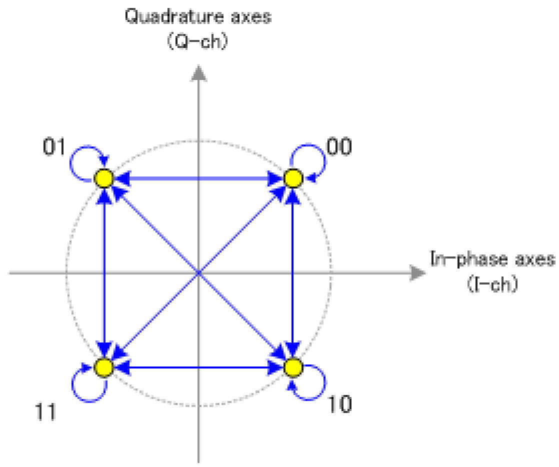


The second term in the braces is an unwanted component, so LPF processing is performed to recover the signal component only.

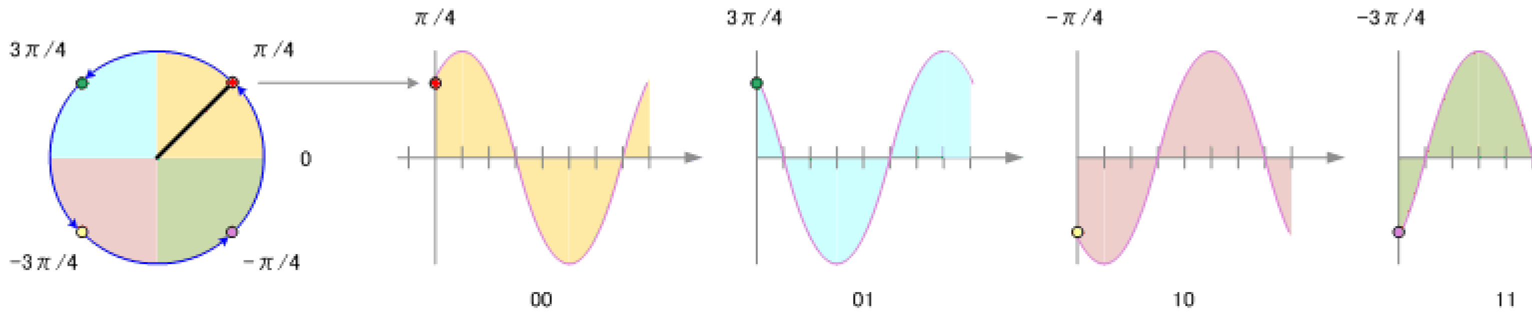
$$\langle S_{psk}(t).C(t)\rangle_{LPF} = \frac{A_c}{2}p(t)$$

*Note: This applet does not actually regenerate the carrier wave.

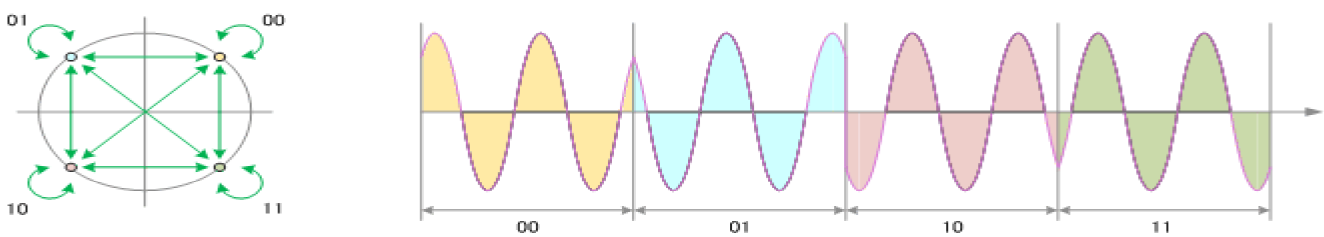
With QPSK, 2 bits of the information signal can be expressed with 1 symbol. The constellation of QPSK modulation and phase transition is as shown in the figure below. As shown in the figure below, information is assigned to the π/4, 3π/4, -π/4 and -3π/4 phases of the carrier wave as 00, 01, 10 and 11 respectively.



QPSK



QPSK constellation and transition of the information signal is shown in the diagram below. The information signal stream input to the modulator is assigned so that two bits represent one symbol. On the right is the QPSK modulated waveform, with one symbol and two wave lengths.



QPSK modulation can be expressed as changing the phase of the I and Q carrier wave C(t) proportionally to the information signal. The carrier waves Ci(t) and Cq(t) are sine waves with the following properties.

$$C(t) = A_c.\cos(2\pi F_c t + \varphi_c)$$

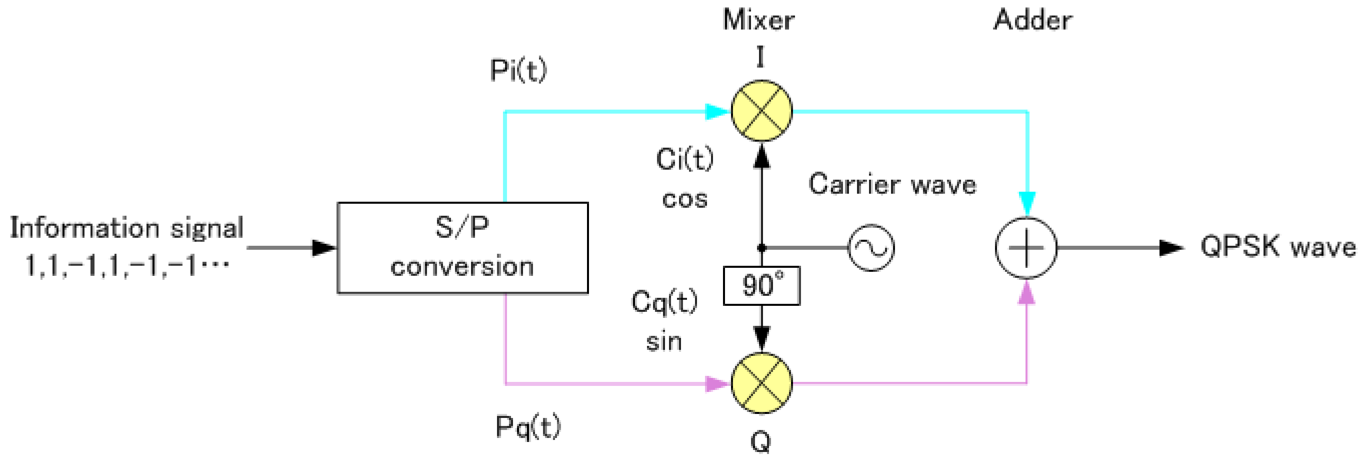
As in the figure below, QPSK modulation is achieved by multiplication of the I and Q components of the dipolar NRZ modulating signal and the I and Q carrier waves with a mixer based on the information, and adding the two signals. The information signal is divided into I component signal and Q component signal by the P/S conversion unit and both are input in the mixer.

$$C_i(t) = A_c.\cos(2\pi.f_c.t)$$

$$C_q(t) = A_c.\sin(2\pi.f_c.t)$$

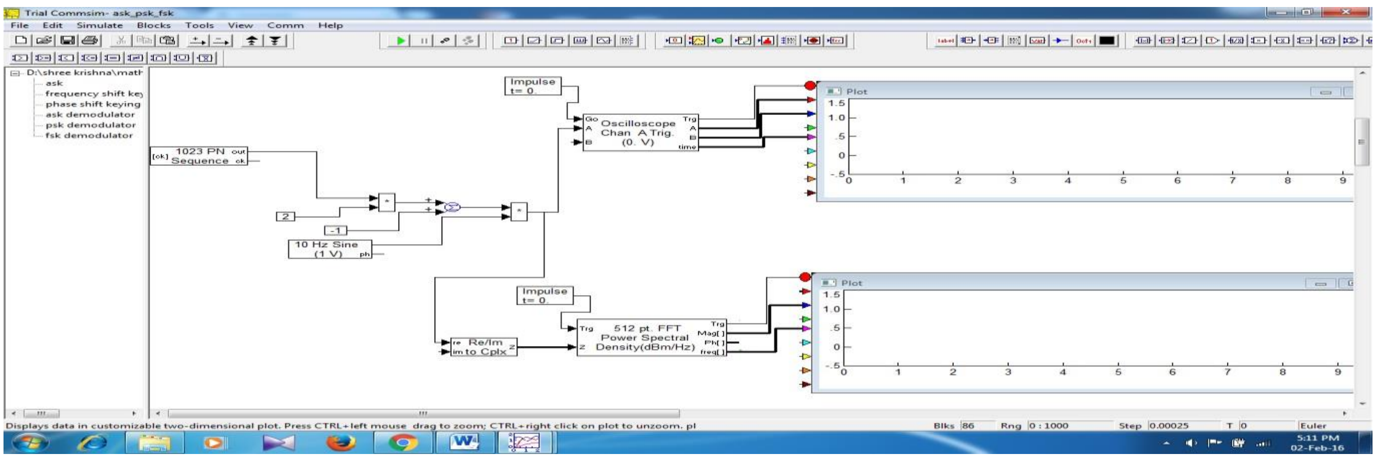
$$S_{qpsk}(t) = P_i(t).C_i(t) + P_q(t).C_q(t)$$

$$= P_i(t).A_c.\cos(2\pi.f_c.t) + P_q(t).A_c.\sin(2\pi.f_c.t)$$

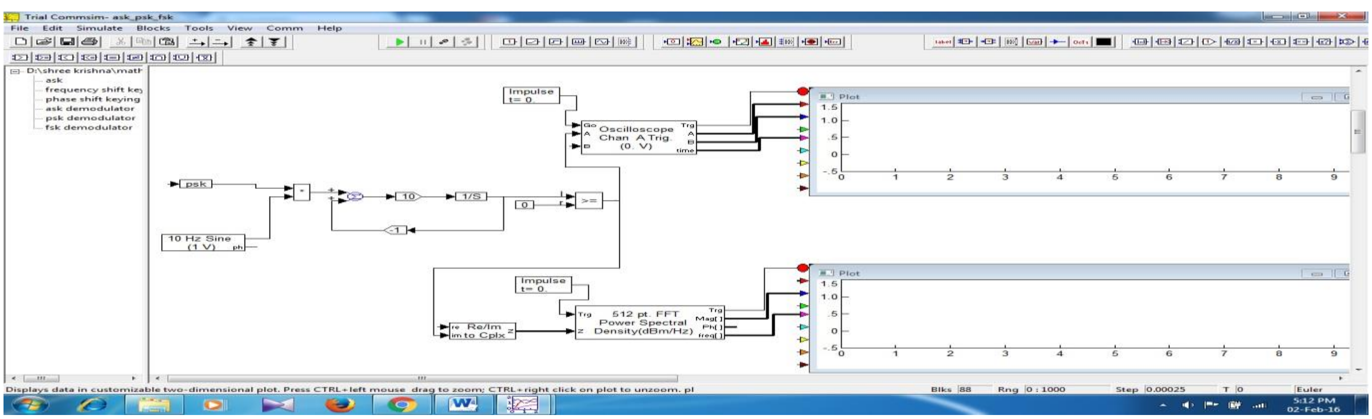


Block Diagram:

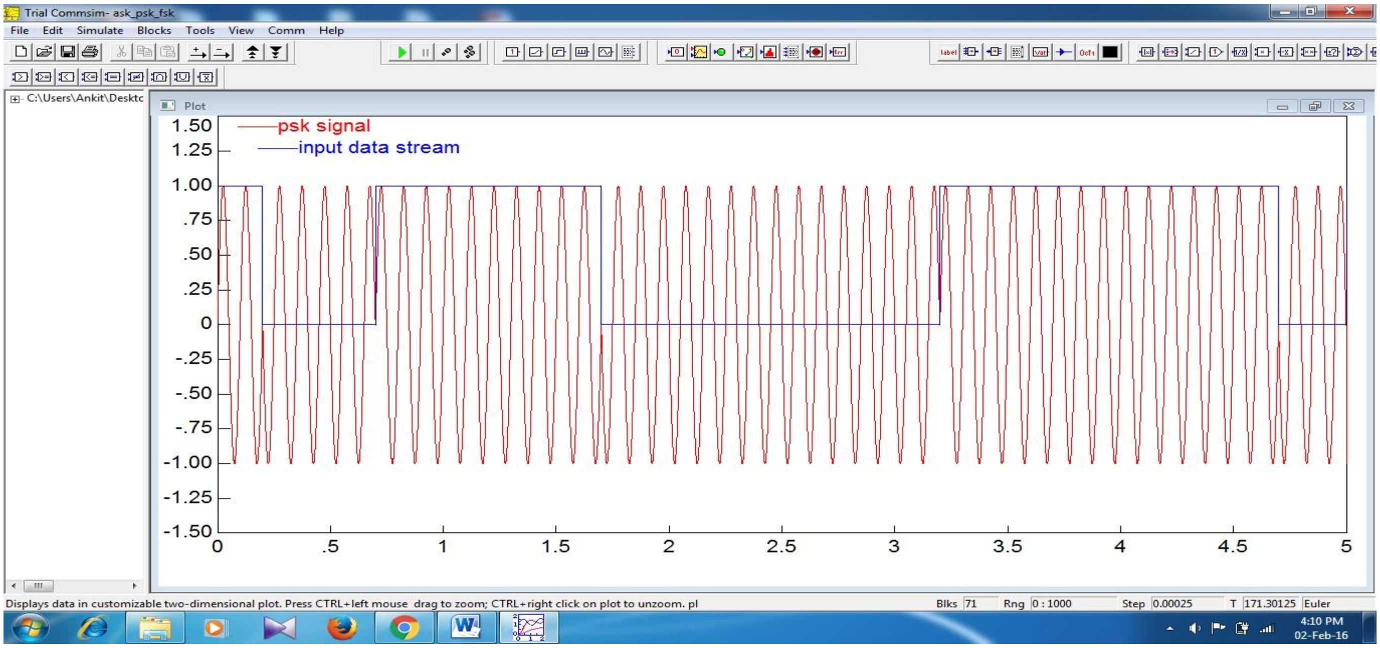
Modulation:-



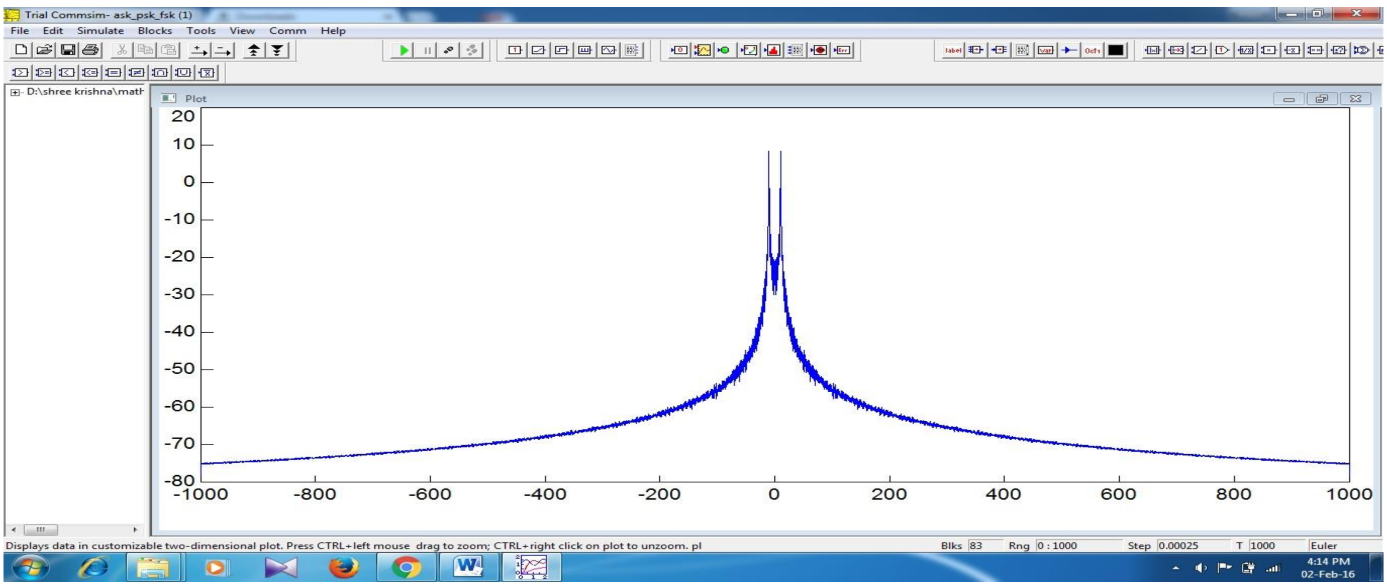
Demodulation:



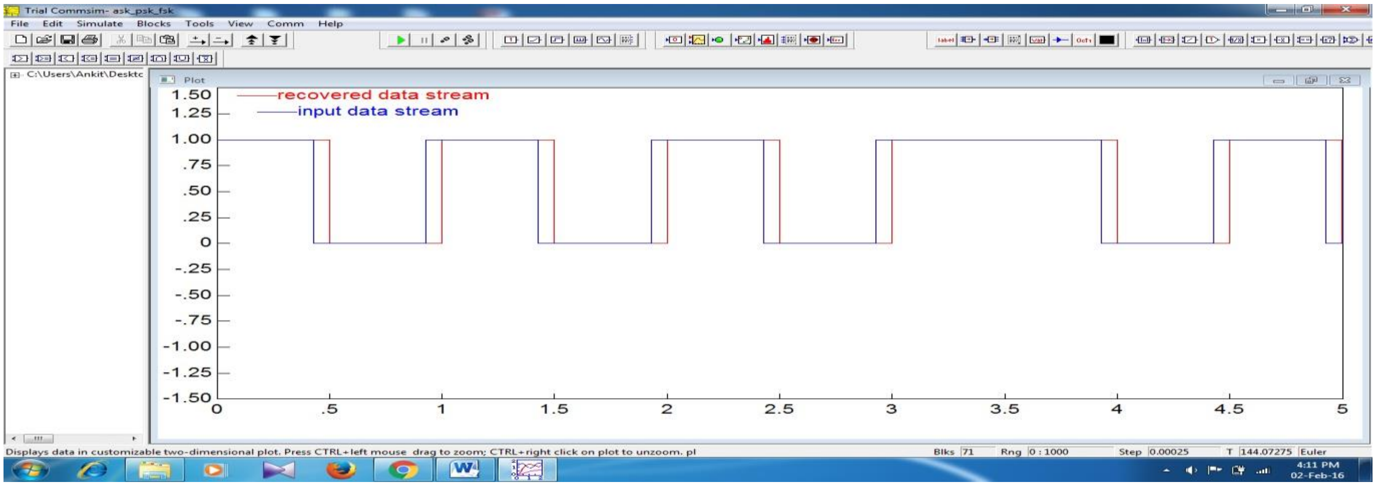
OUTPUT WAVEFORM:-



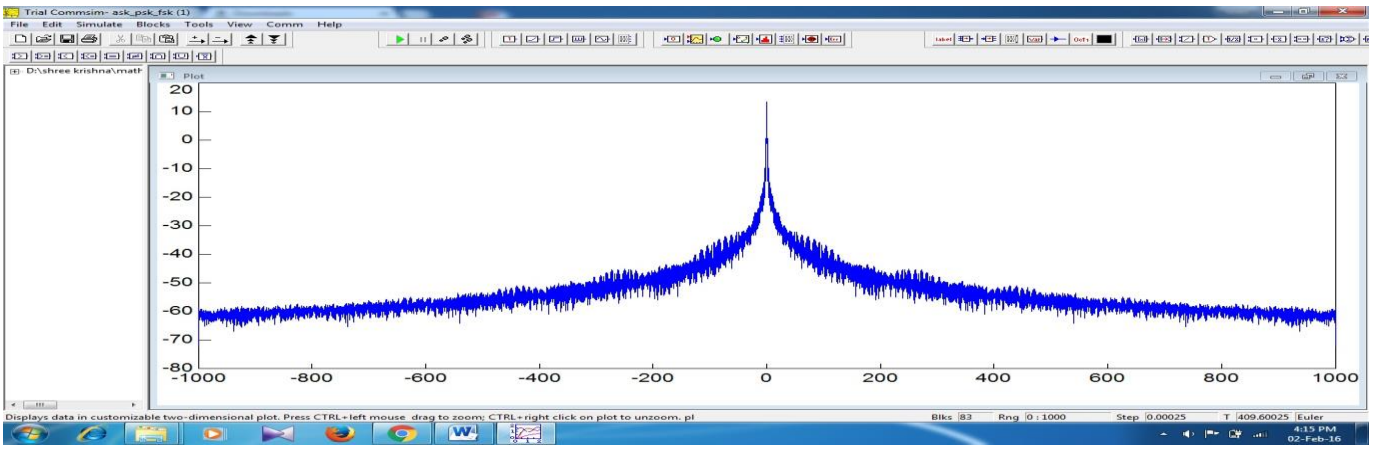
PSK IN TIME DOMAIN



Spectrum Analysis of PSK



Recovered Message Signal



Spectrum Analysis of Recovered Message Signal

Conclusion:

In this simulation phase of the carrier signal changes according to the message signal. And after demodulation we recovered the original message signal. Simulated graphs are tested and verified.