EECS306

Project-1

Submitted To:

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Submitted By:

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Requirement 1

1-Fig.1 represents the modulating signal that is required in point 1 the steps to generate this signal are as follows:

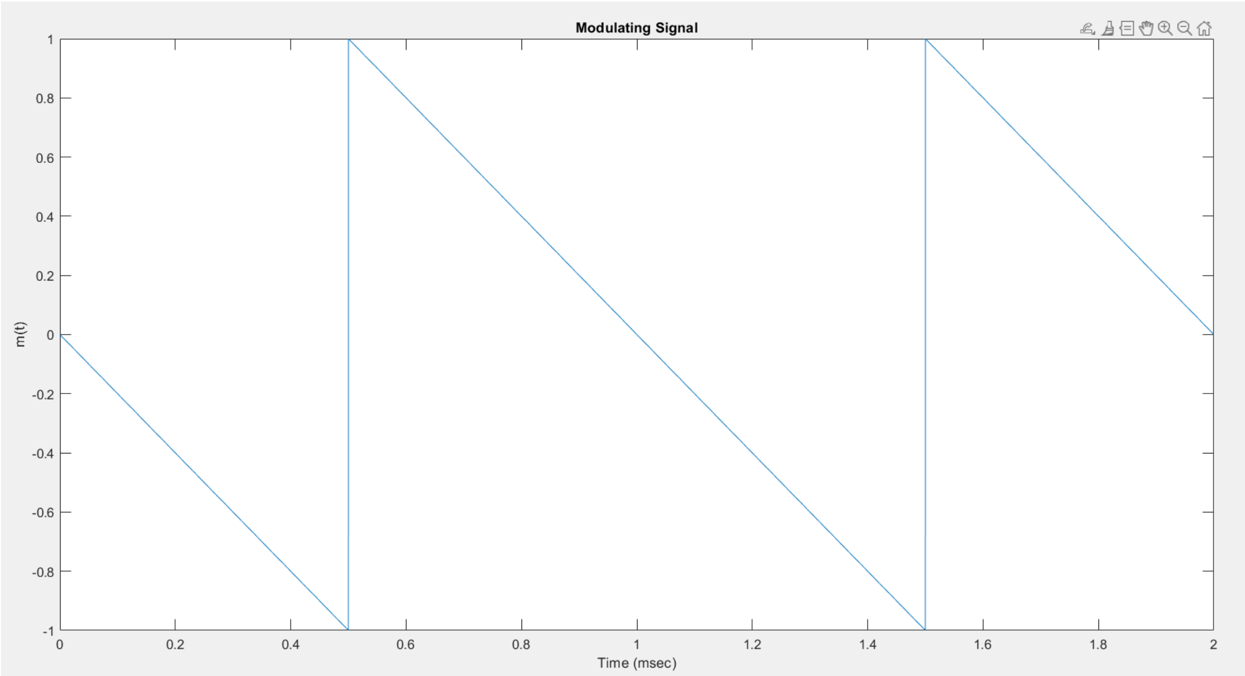

Fig.1

internal variables and equation are clarified at fig.2

```
sampling_rate = 10000;   % Sampling rate
dt = 1/sampling_rate;
num_of_periods=2;         %number of periods of m(t)
t = 0:dt:(1-dt)*num_of_periods ;     %time interval

% Define the shift value
shift_value = 0.5;

% Calculate the shifted time axis
t_m = t + shift_value;

% Calculate the shifted sawtooth wave
modulating_signal = -2*(t_m - floor(t_m)) + 1;
```

Fig.2

the modulating signal is mainly consist of mirrored sawtooth ,so the equation of the sawtooth is equal to t-floor(t) and to get it's mirror multiple this equation by -1 ,this mirrored sawtooth starts from 0 to negative 1 ,but this wasn't the required modulating signal ,so by performing some mathematical operations on the mirrored sawtooth equation to obtain the required modulating signal as specified in fig.2 the equation of modulating_signal.

2-Generating an AM DSB-LC

```
%parameters of the carrier
fc=10;                      %carrier frequency=10 KHz
Ac=1;                        %carrier amplitude
Wc=2*pi*fc;                  %carrier frequency=2*pi*fc rad/sec

unmodulated_signal= Ac.*cos(Wc*t);
Ka=input('Enter the modulation index: ');
Am=Ka*Ac; %to get the new amplitude of the
AM_modulated_signal=(Ac+Am*modulating_signal) .* cos(Wc*t) ;
                              %the DSB-LC modulated signal
```

Fig.3

As shown in fig.3 the parameters of the carrier are defined ,and the carrier equation is clarified as unmodulated_signal ,taking the modulation index from the user and obtain from it the new amplitude of the modulating signal (Am) ,and write the equation of the DSB-LC which is AM_modulated_signal and there was another way to write it in terms of modulating_signal and unmodulated_signal as follows:

AM_modulated_signal = (1+Ka*modulating_signal) .* unmodulated_signal

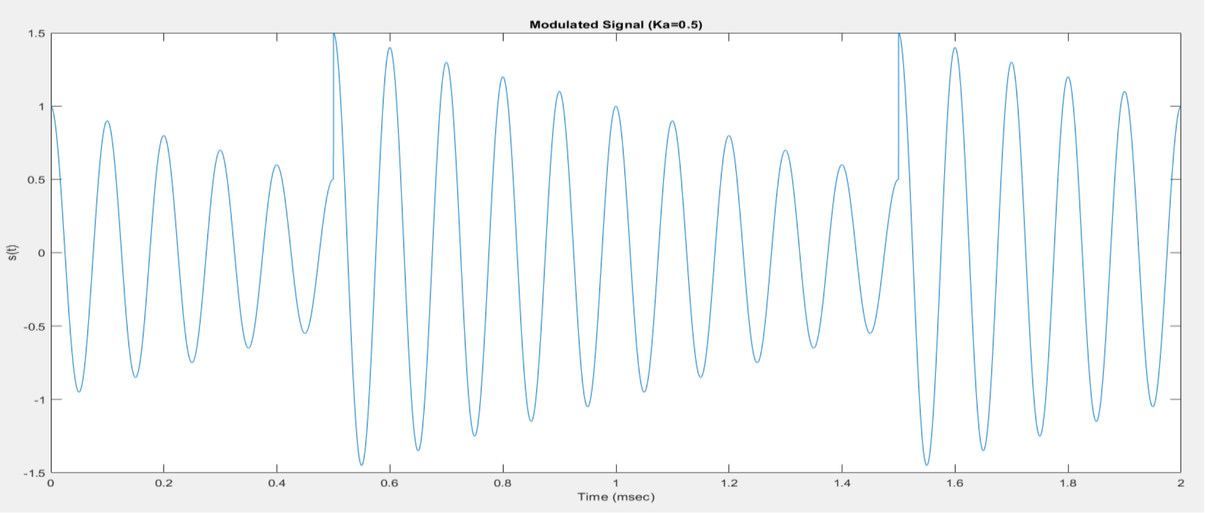

Fig.4

Fig.4 represents the modulated signal (s(t)) at modulation index(Ka=0.5)

At this modulation index the amplitude of the modulating signal Am=0.5 and since that the modulation index is less than or equal to 1 ,thus the modulating signal can be detected using envelope detector as a demodulation technique , because there is no interference between the shifted m(t) and -m(t) as it clarified at fig.4.

3- case Ka=1 ,as shown in fig.5 , Am=1 , and since that the modulation index is less than or equal to 1 there will be no interference between the shifted m(t) and -m(t) but they will meet only at the minimum value of modulating signal as shown at points 0.5 and 1.5 (msec) , thus the envelope detector is still can be used at this Ka .

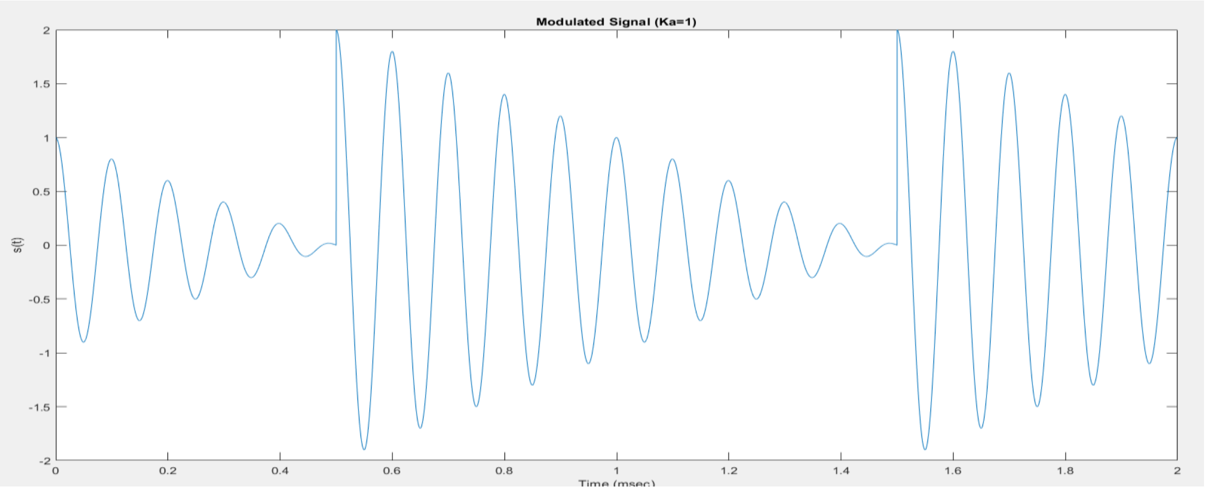

Fig.5

Case Ka=2 , as shown in fig.6 , Am=2 and since that the modulation index is more than 1 there will be interference between the shifted m(t) and -m(t) ,thus the envelope detector can not be used in this case ,such that the signal in the positive side of y-axis will not be the sended message as shown in fig.6.

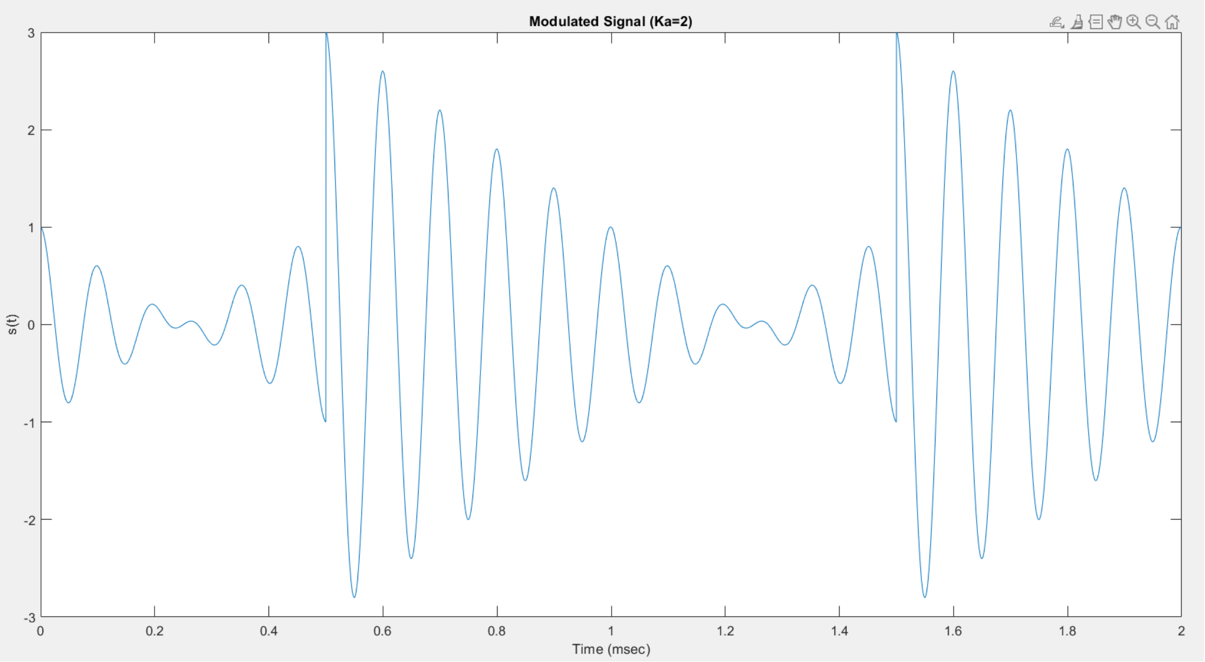

Fig.6

4-Generate an FM signal for the same carrier is by using a build in function fmmod

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%generating an FM signal for the same carrier then Plot the FM signal

Kf=input('Enter the frequency deviation: ');
sampling_rate=100000;
FM_modulated_signal=fmmod(modulating_signal,fc,sampling_rate,Kf);
```

Fig,7

This function to generate FM signal needs the following parameters the modulating signal in the time domain and the carrier frequency (fc) and the sampling rate which modified to change the frequency of the FM signal and the last parameter is the frequency deviation(Kf).

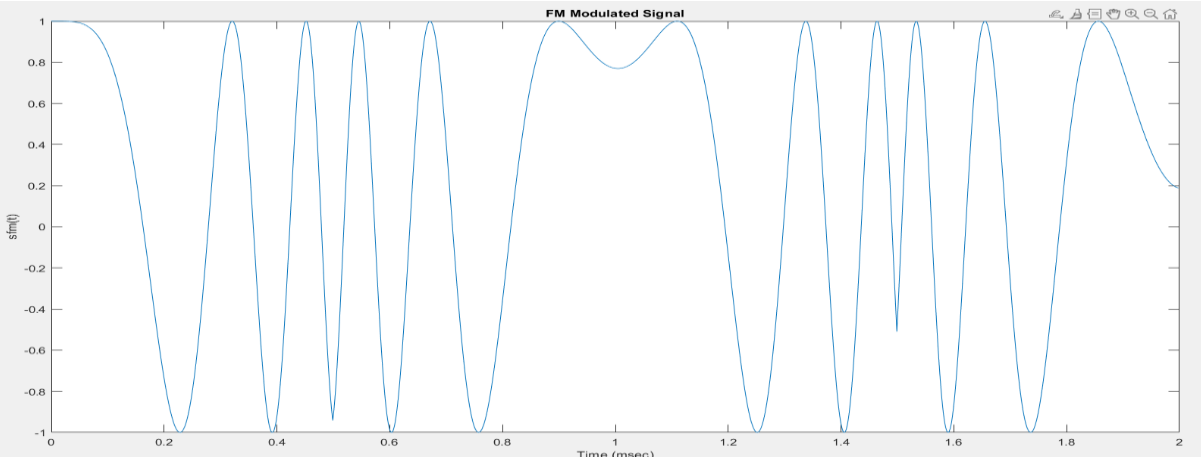

Fig.8

In fig.8 it shows the FM signal at Kf=1000 , it shows at points 0.5 and 1 and 1.5 that there is a sudden change , this is due to the intersection of the modulating signal with zero at the mentioned time but at 0.5 and 1.5 the change is sharp than that at time 1 because the sudden change at 0.5 and 1.5 in the modulating signal rather than at 1 which was passing by zero.

5- at Kf=2000 the FM signal is as follows at fig.9

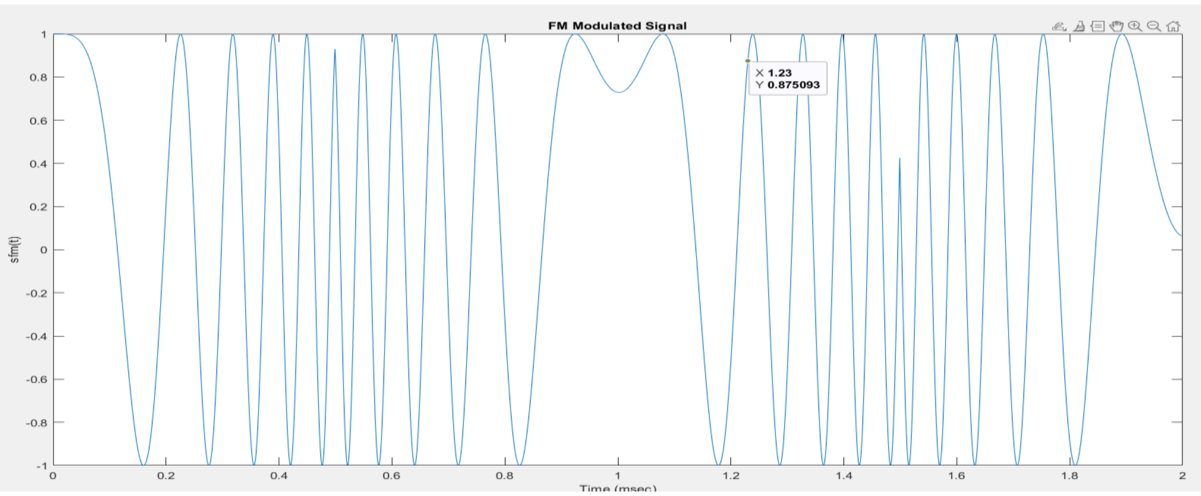

Fig.9

It is clear that the FM signal become more sensitive to the change in the amplitude of the modulating signal rather than that generated at fig.8 and the FM signal has higher frequency than that at fig.8 and at time 0.5 ,1 ,1.5 happens also what mentioned at Kf=1000.

-At Kf=5000

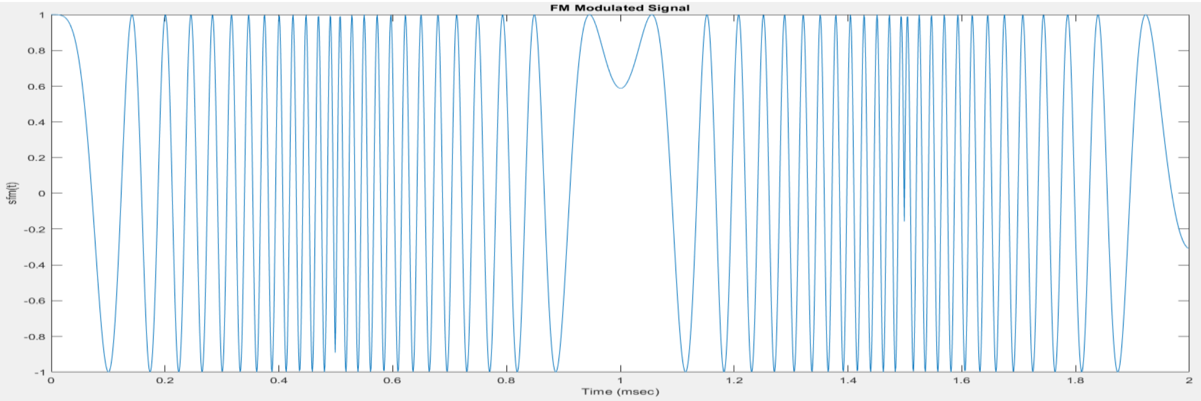

Fig.10

It is clear that the FM signal become more sensitive and more responsive to the change that happens in the modulating signal and also at time 0.5,1,1.5 happens what mentioned at Kf=1000 and has higher frequency.

Requirement 2

1-generating a sinusoidal modulating signal with Am=4 and frequency of 2KHz by using the following code on fig.11

```
Fs=1000;
dt=1/Fs;
t=0:dt:(1-dt);
fm=2;
Wm=2*pi*fm;
Am=4;
%modulating signal
m_t= Am .* cos(Wm *t);
```

Fig.11

The generated modulating signal graph is on fig.12

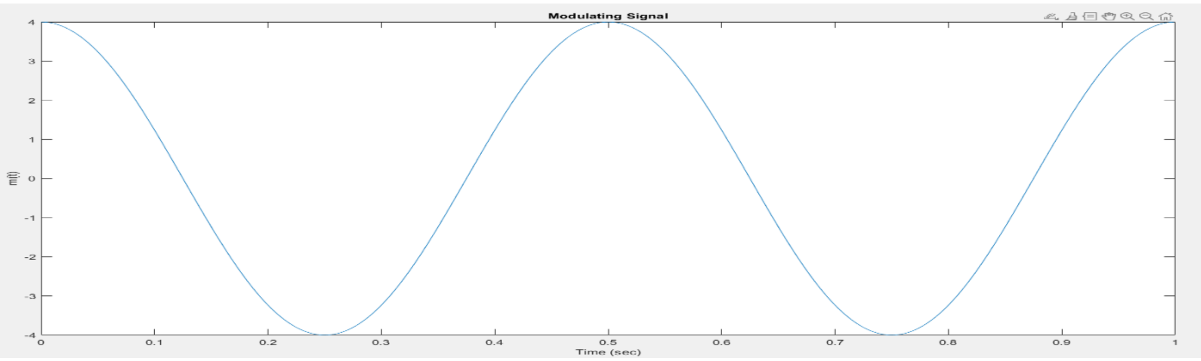

Fig.12

2-as illustrated in fig.13 Ac =2V and the frequency of the unmodulated signal is 10KHz and the s_t_USB and s_t_LSB variables are the modulated signal from the single side band modulator

```
Ac=2;
fc=10;
Wc=2*pi*fc;
%Unmodulating signal
c_t=Ac .* cos(Wc*t);
%generating the modulated signal
m_t_hat= Am .* sin(Wm*t);%quadrature of the Modulating signal
c_t_hat=Ac .*sin(Wc *t);%quadrature of the Unmodulating signal
%generating the USB and plot it
s_t_USB=m_t .*c_t - m_t_hat.* c_t_hat;
%generating the LSB
s_t_LSB=m_t .*c_t + m_t_hat.* c_t_hat;
```

Fig.13

Plot of USB (fig.14) and LSB (fig.15) are as follows

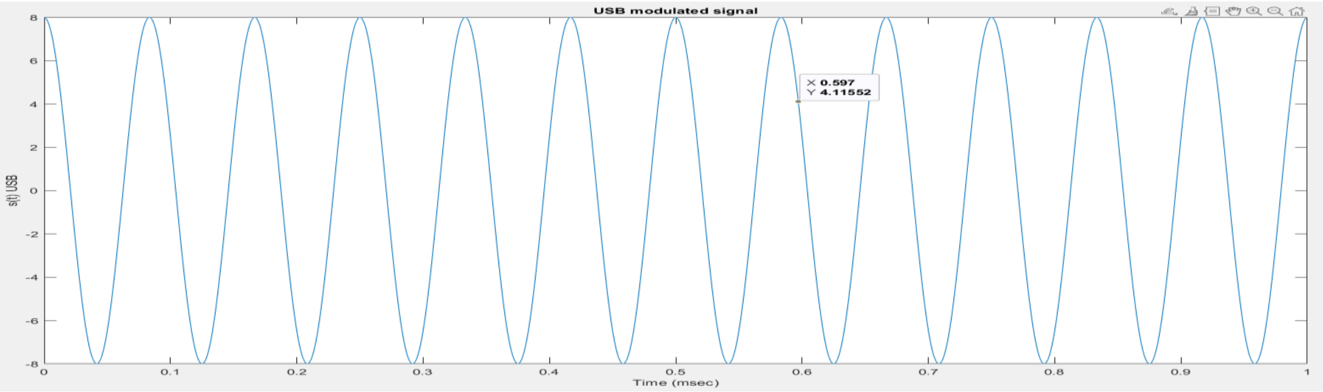

Fig.14

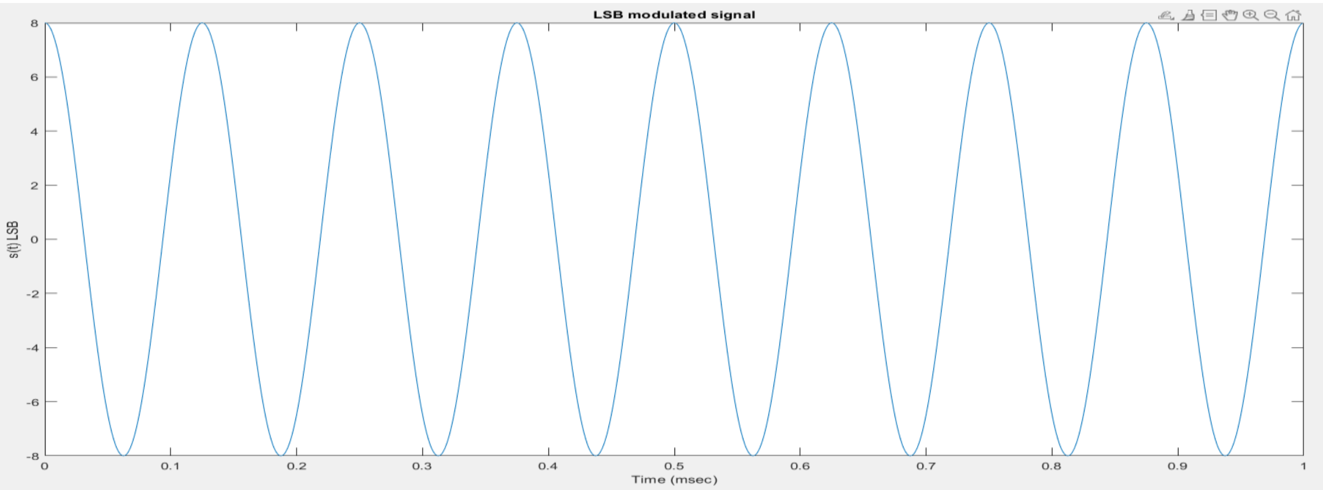

Fig.15

It is clear that the USB has higher frequency than the LSB because the frequency of the USB is the carrier frequency add to it the message frequency but the LSB we minus from the carrier frequency the message frequency.

3- the code used to obtain the frequency spectrum of the modulated signal (fig.16)

```
f = -Fs/2:1:Fs/2 -1;  % Frequency vector

spectrum_LSB = fftshift(abs(fft(s_t_LSB/Fs))); % Shift zero frequency component to center
spectrum_USB = fftshift(abs(fft(s_t_USB/Fs))); % Shift zero frequency component to center
```

Fig.16

We get the frequency vector (f) and get the spectrum of the USB and the LSB using fast fourier transform (fft) this function takes the modulated signal in the time domain and perform fourier transform to get it in the frequency domain and function fftshift is to shift the zero component to the center.

The spectrum of the USB (fig.17) and LSB (fig.18) are as follows

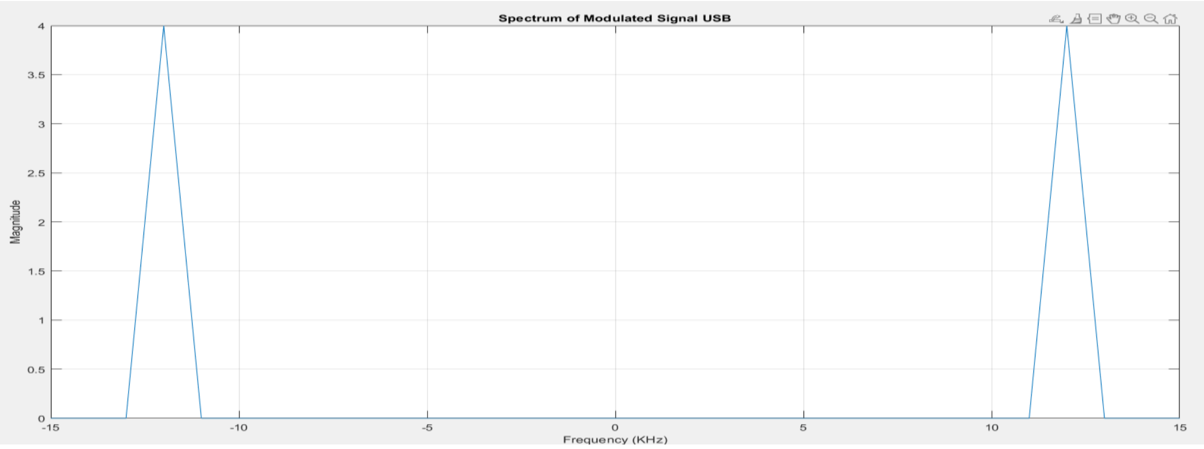

Fig.17

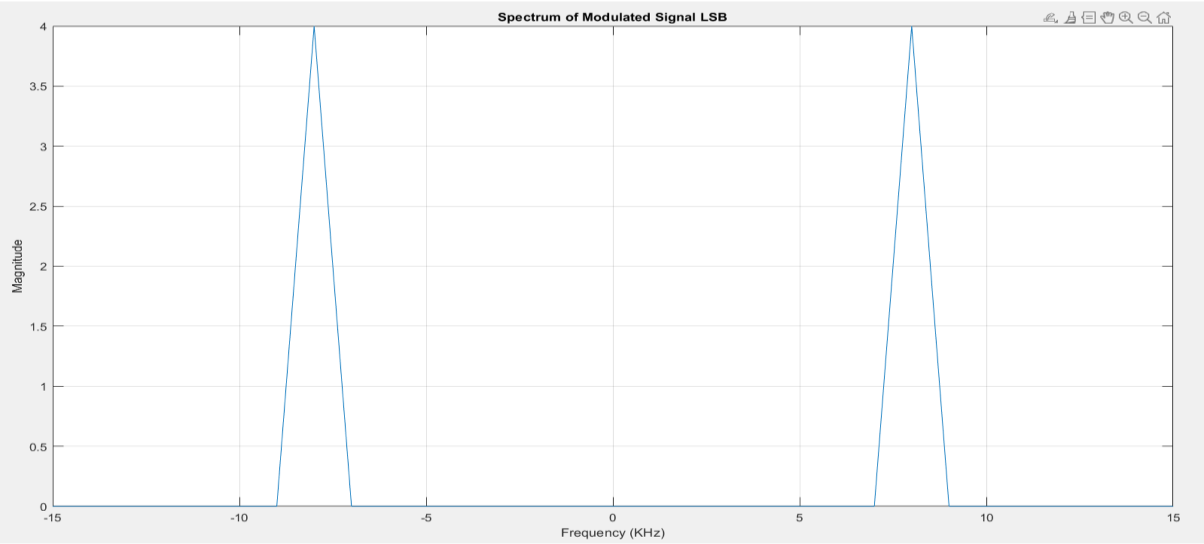

Fig.18

The USB has two deltas one at +12 KHZ and at -12 KHz which is the carrier frequency added to it the message frequency, while at the LSB it has two deltas one at 8 KHz and at -8 KHZ which is the carrier frequency subtracted from it the message frequency.

Why the shape of the triangle appear as we expect that it is a delta at exactly the frequency mentioned , this happens when you take the Fourier Transform of a cosine wave, you'll typically get two peaks in the frequency domain: one at the positive frequency component and one at the negative frequency component. These peaks are located at the frequency of the cosine wave. However, due to the finite length of the sampled data the frequency components may spread out or leak into adjacent frequencies, causing some spectral leakage.

4-a suitable demodulator to extract m(t) from s(t) is a coherent demodulator, can be implemented using the following code at fig.19.

```
%demodulator (coherent demodulator)

Demod_signal = s_t_LSB.*cos(Wc*t);
[b,a] = butter(4,0.01,'low');
Filtered_Demod_signal= filter(b,a,Demod_signal);
Spectrum_Filtered_Demod_signal = fftshift(fft(Filtered_Demod_signal));
```

Fig.19

First we multiplied the USB or the LSB modulated signal with the carrier frequency to obtain the spectrums of the m(t) at the center and then apply LPF which is the function butter and filter, the returned values from function butter is used in function filter and the multiplication of the USB or LSB with the carrier, the parameter Filtered_demod_signal is the m(t) extracted from the modulated signal.

5- if the generator carrier wasn’t perfectly synchronized with the used one in modulator as there is the phase shift.

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% Introducing a phase offset between carrier and local oscillator
phase_offset = pi / 4;  % Example phase offset of pi/4 radians
demodulated_signal_phase_offset = s_t_LSB .* cos(Wc * t + phase_offset);

% Demodulated signal after low-pass filtering with phase offset
demodulated_signal_filtered_phase_offset = filter(b, a, demodulated_signal_phase_offset);
```

Fig.20

We repeat steps in point 4 but by adding a phase shift with pi/4 ,the code in fig.20 perform this task.

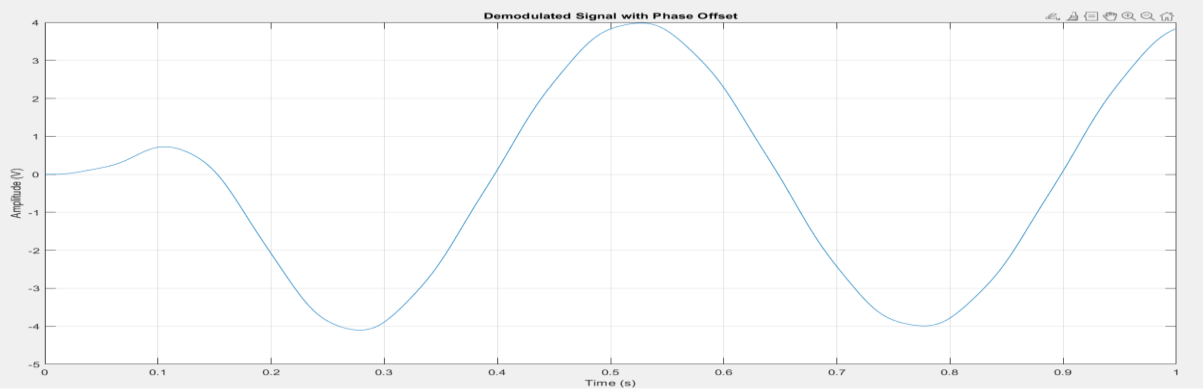

Fig.21

As shown in fig.21 the demodulated signal isn’t exactly the modulating signal that’s due to asynchronization between the oscillator at the modulator and demodulator , so it leads that the demodulated m(t) has attenuation in parts of it due to the phase shift.