Lab1

Name: Norhan Reda Abdelwahed Ahmed
Sec: 2
B.N: 32

1-Choose reasonable values for A and ω to achieve DSB-LC modulation with your previous audio signal and explain how you choose them?

- Let ($Ac \geq$absolute -vepeak amplitude, $|(t)min|$)

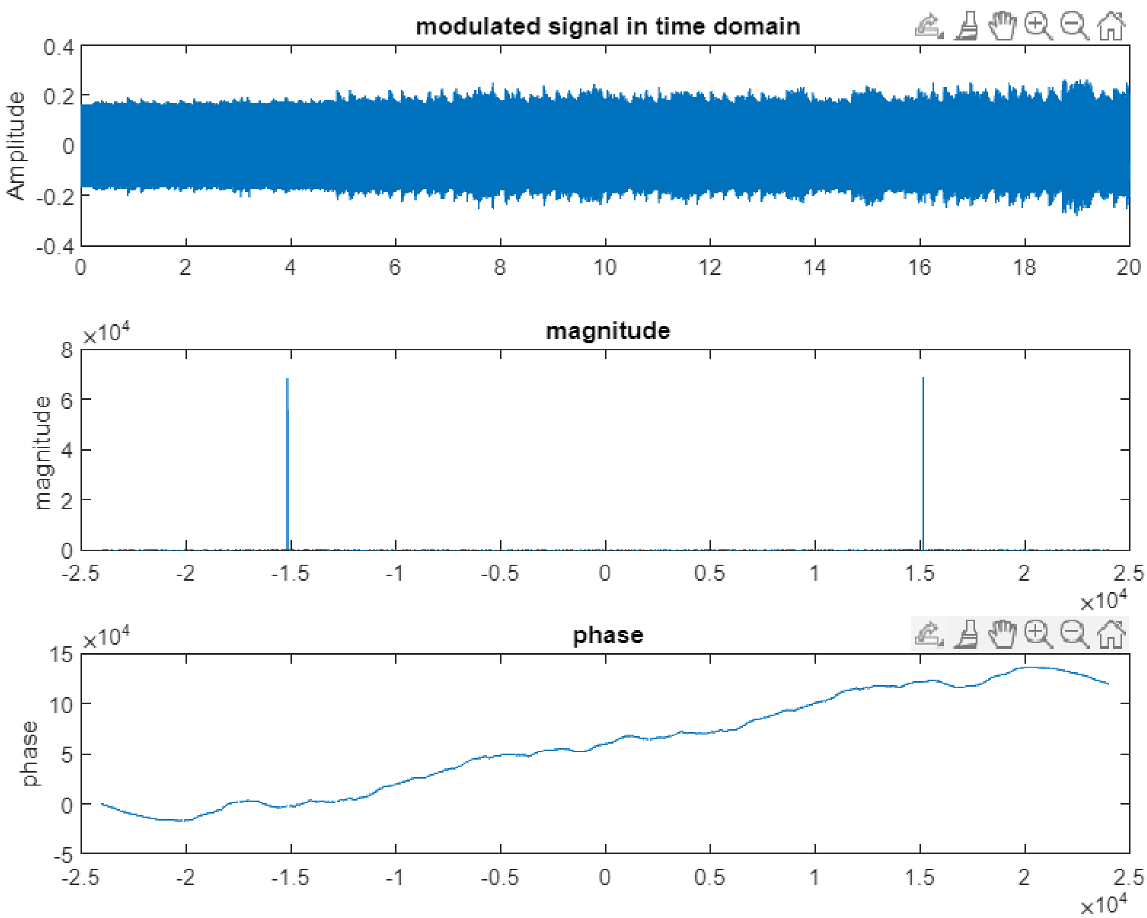
$Ac \geq |m(t)min|$

So from the graph Ac=0.1625 at least to avoid over modulation so it will achieve power efficiency maximum =33.3%

- Let $\omega c = \omega s/2 - \mathrm{bw}$

  to achieve Nyquist criteria for sampling

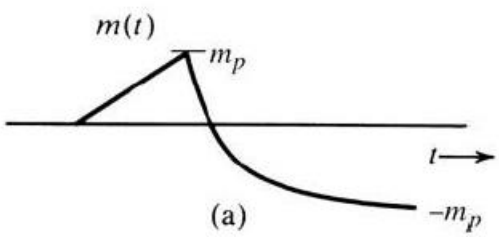
2-Plot the modulated signal waveform in time domain and the modulated signal amplitude and phase in frequency domain.

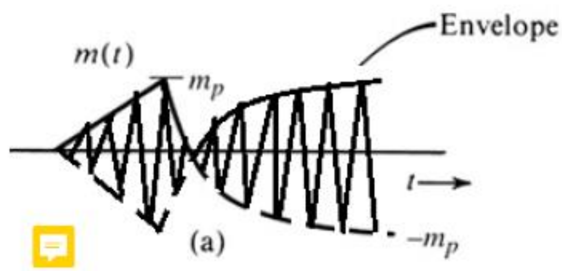


3-What do you think is a carrier’s minimum Amplitude (A) to avoid over modulation? What is the problem with the AM signal when it is over-modulated?

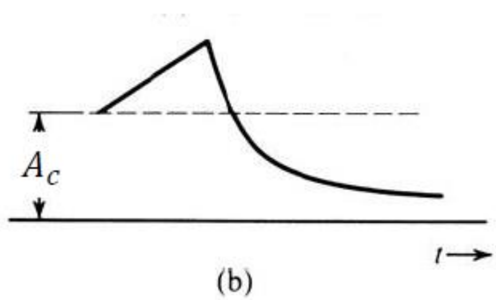
- Let ($Ac \geq$absolute -vepeak amplitude, $|(t)min|$)

$Ac \geq |m(t)min|$

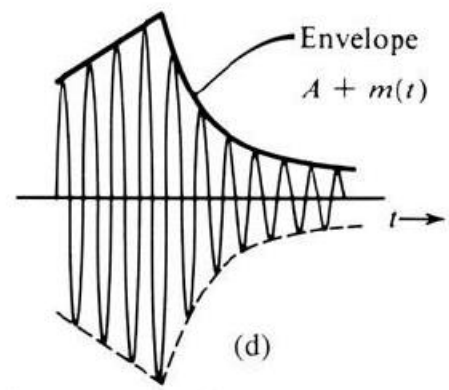




$m(t)\cos\omega_c t$



$A_c + m(t)$



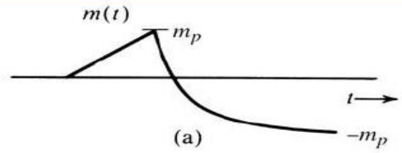
$(A_c + m(t))\cos\omega_c t$

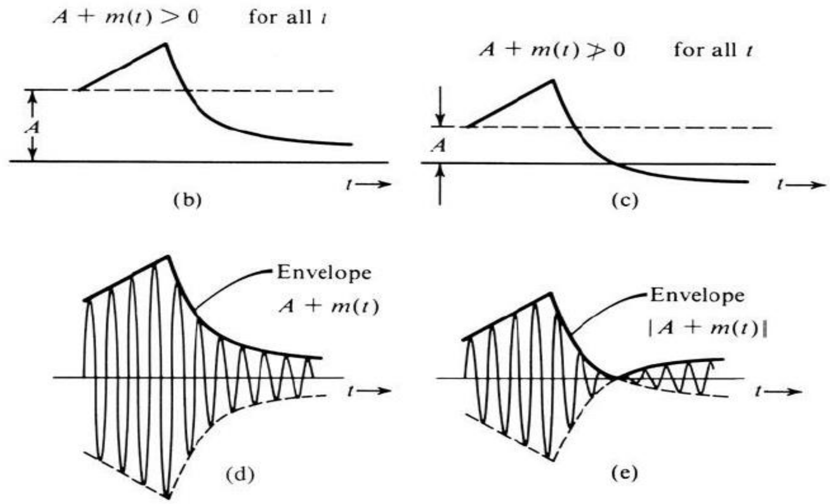
If Ac is less than $|m(t)min|$ this will cause over-modulation

μ=$|mtmin|$ / Ac

If μ> 1 it is over modulation

$\pm m_p$: maximum and minimum values of $m(t)$

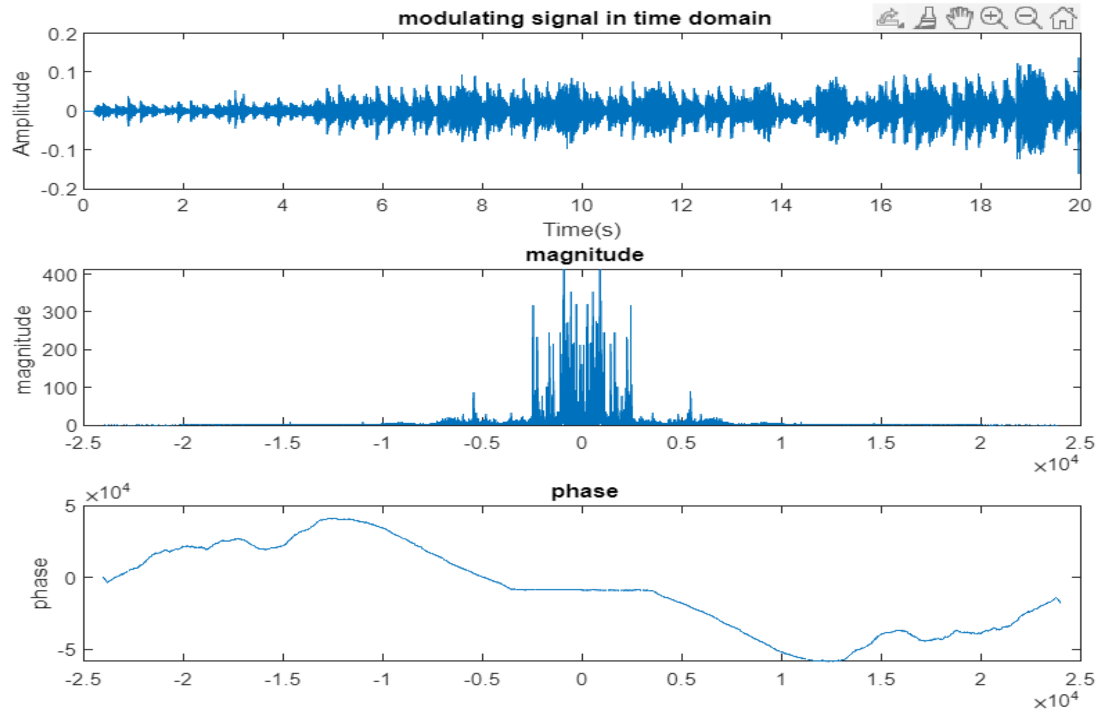


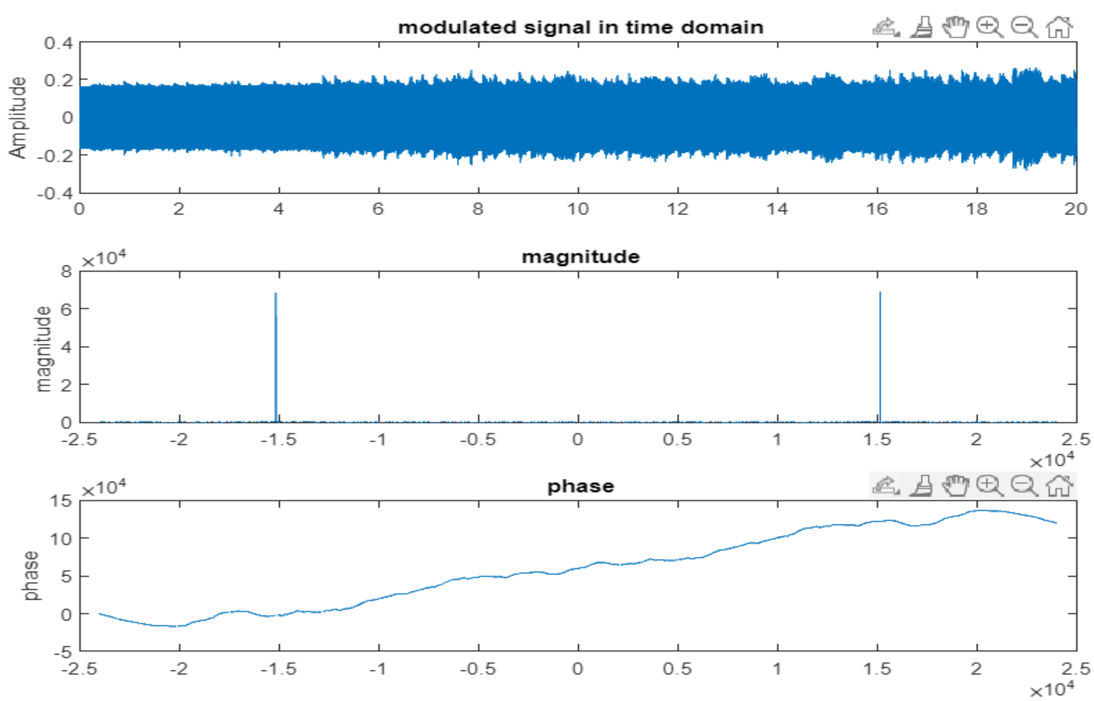


9

In over modulation the envelope is wrong and we can't get the same signal after demodulation (envelope detection not viable) and the signal get loss

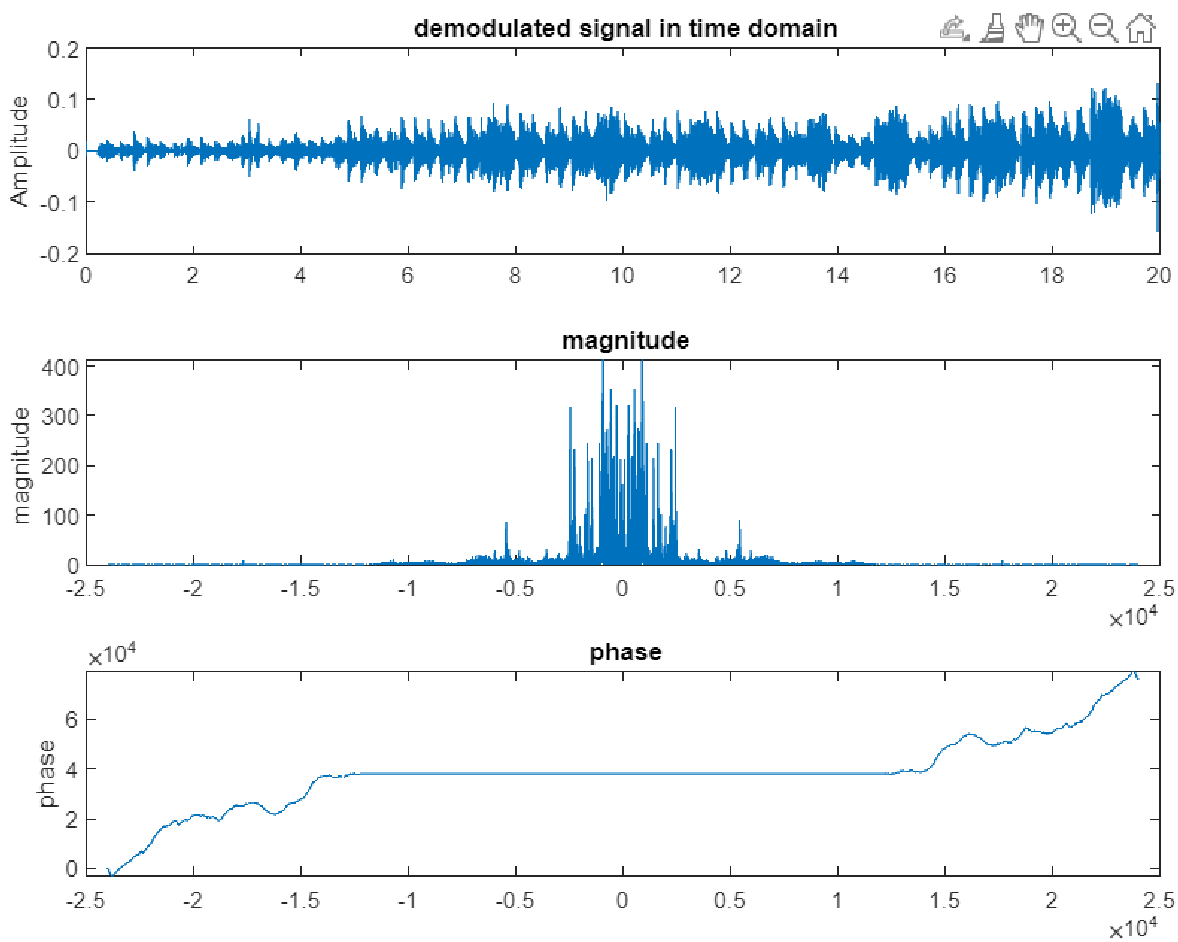
4-Compare between the bandwidth of the audio signal and the modulated one by plotting both signal in the frequency domain.





from the graph it is clear that the bandwidth of the modulated signal is double the bandwidth of the modulating signal (audio signal) and the modulated signal is shifted at the fc frequency and its band width is doubled but It's not very clear because some magnitudes have smaller values that is not appear at the graph because of the scale

5-Do synchronous demodulation to obtain x(t) , then plot the final signal in time and frequency domain as previous.



Wow it looks like the same signal ☺

6- Hear the demodulated signal and compare it with the original one. Are the two signals the same? Explain why?

Yes ,they are same in the sound and everything except the phase but when I asked the doctor why they are different in phase he said "This is probably due to the phase response of the filter that you use." , we apply the Coherent detection in which we synchronize the carrier with the signal and the wc used in demodulation is the same used in modulation and also I have chosen Ac $= |m(t)min|$ to achieve less power loss and achieve maximum power efficiency which we can get which is

33.3% and you can hear the signal after running the code and you will find that they are the same ☺

code :

```
% Read audio files
clc;
clear;
[y,fs] = audioread('fly.mp3');
y =y(:,1) + y(:,2);
% Plot signal in time domain
 N = size(y,1);
t = (0:N-1)/fs;
figure(1);
subplot(3,1,1);
plot(t,y);
title(" modulating signal in time domain");
xlabel('Time(s)');
ylabel('Amplitude ');

%%
%  Frequency domain
% Plot signal in frequency domain
freq_y= fft(y);
dfs=fs/length(y);
fre_range=-fs/2:dfs:fs/2-dfs;
magnitude1=abs(fftshift(freq_y));
phase1=unwrap(angle(freq_y));
subplot(3,1,2);
plot(fre_range,magnitude1);
title(" magnitude");
ylabel('magnitude ');
subplot(3,1,3);
plot(fre_range,phase1);
title(" phase");
ylabel('phase');

%%
%carrier and modulation

bw=bandwidth(y)./(2.*pi);
fc=(fs/2)-bw;
wc=fc*2*pi;

% Ac = max(abs(y));
Ac = abs(min(y));
carrier= cos(wc*t).';

modulated = (y+Ac).*carrier;
figure(2);
subplot(3,1,1);
```

```
plot(t,modulated);
title(" modulated signal in time domain");
ylabel('Amplitude');

freq_y2=fft(modulated);


magnitude2=abs(fftshift(freq_y2));
phase2=unwrap(angle(freq_y2));
subplot(3,1,2);
plot(fre_range,magnitude2);

title(" magnitude");
ylabel('magnitude ');
subplot(3,1,3);
plot(fre_range,phase2);
title(" phase");
ylabel('phase');

% demodulation
%%
demodulated_signal =  modulated.*carrier;

LPF = lowpass(demodulated_signal,10000,fs);
% ('Fp,Fst,Ap,Ast',0.1,0.8,1,80);
% d = fdesign.lowpass('Fp,Fst,Ap,Ast',0.1,0.2,0.07,1e-7,'linear');
% designmethods(d);
% Hd = design(d, 'equiripple');
% LPF = filter(Hd,demodulated_signal);
LPF=LPF*2;
LPF = LPF -Ac;

figure(3);
subplot(3,1,1);
plot(t,LPF);
title(" demodulated signal in time domain");
ylabel('Amplitude');

freq_y3=fft(LPF );
magnitude3=abs(fftshift(freq_y3));
phase3=unwrap(angle(freq_y3));
subplot(3,1,2);
plot(fre_range,magnitude3);
title(" magnitude");
ylabel('magnitude ');
subplot(3,1,3);
plot(fre_range,phase3);
title(" phase");

ylabel('phase');

sound(LPF,fs);
```