Lab 1: Amplitude Modulation

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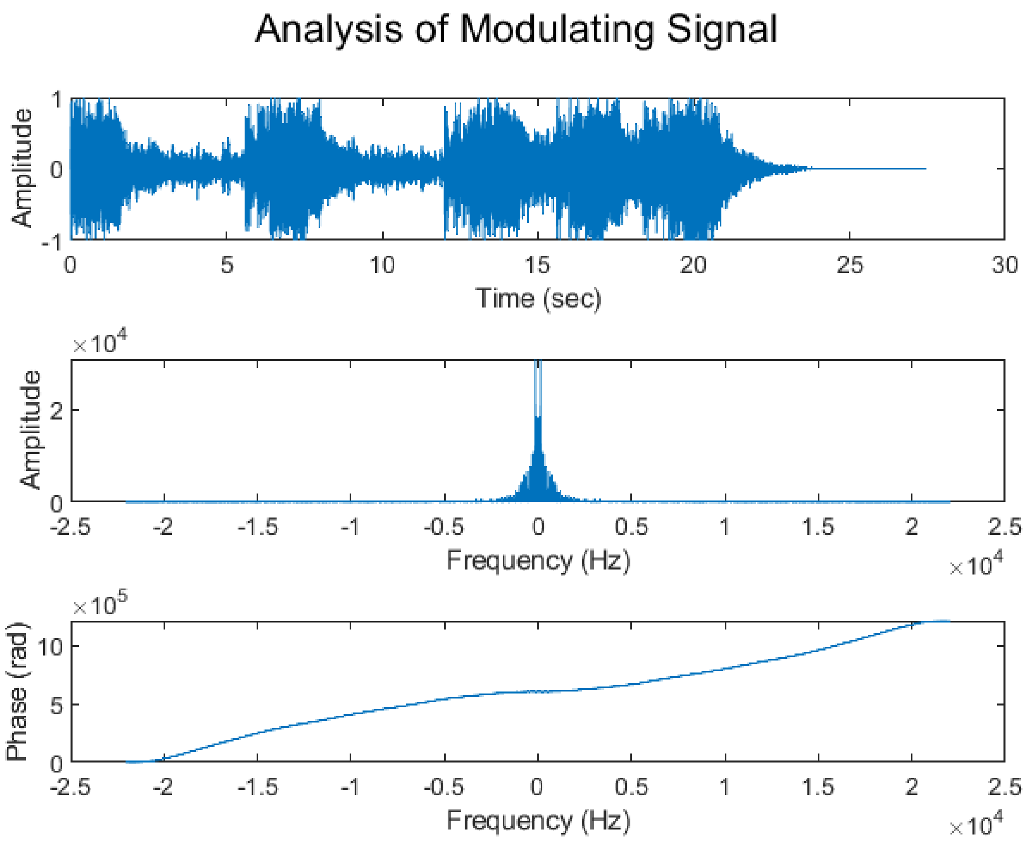
sec: 1

attachment: lab files

1. Analyse Audio



2. Perform DSB-LC Modulation

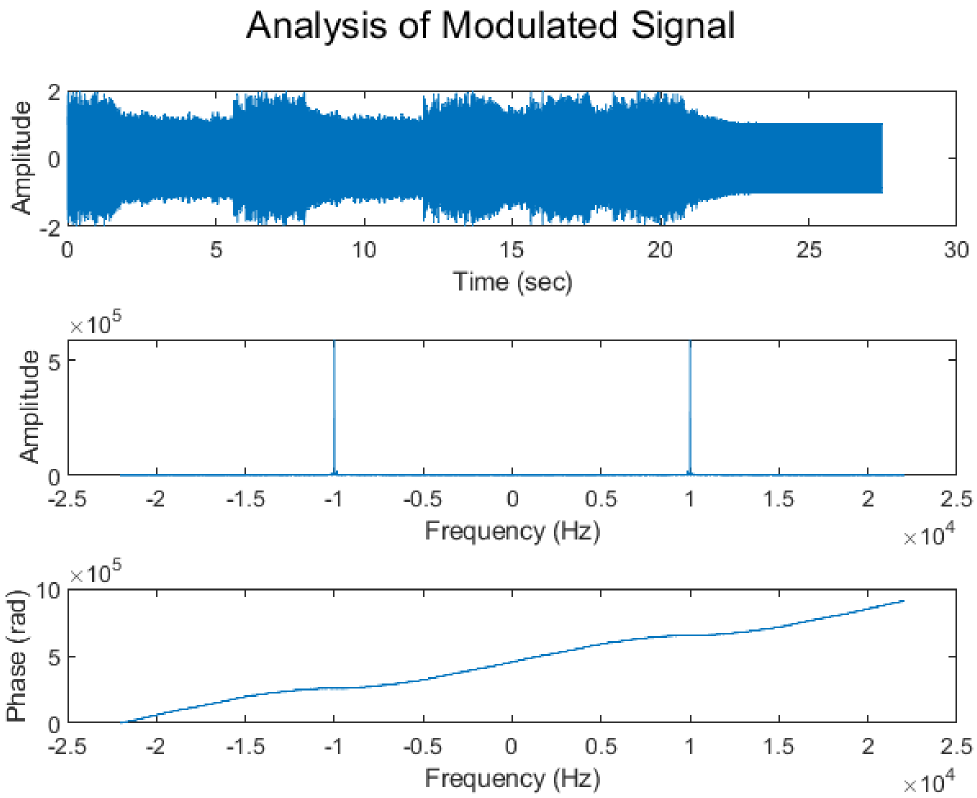
1. Choose reasonable values for $\boldsymbol{A}$ and $\boldsymbol{\omega}$ to achieve DSB-LC modulation with your previous audio signal and explain how you choose them?
    - for choosing $Fc \;\Rightarrow w_c = 2.\pi.F_c$
        - from first plot (signal in frequency domain) $\;\Rightarrow \text{bandwidth} \approx 5000 \text{ hz}$
        - from variables in code $\Rightarrow F_s = 44100 \text{ hz} \;\Rightarrow \frac{F_s}{2} = 22050 \text{ hz}$
        - $\text{bandwidth} < F_c < \frac{F_s}{2}$
        - $F_c = 2 * \text{bandwidth} = 10000 < \frac{Fs}{2}$

- usually it more than <u>2</u> *for antenna and other reasons*

- for choosing A_c

  let $\mu = 1$ so $\therefore \eta = 33.33\%$ & $\therefore A_c = \frac{|min(m(t))|}{\mu} \approx 1$

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

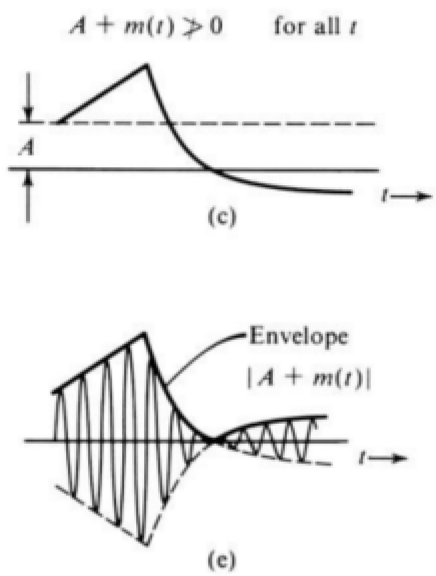


3. What do you think is a carrier's minimum Amplitude (A) to avoid over modulation? What is the problem with the AM signal when it is over-modulated?

- $A_c > |\text{minimum } m(t)|$

- so all signal become above access and when modulated I can extract envelope easily first by my eye then by a simple cheap circuit

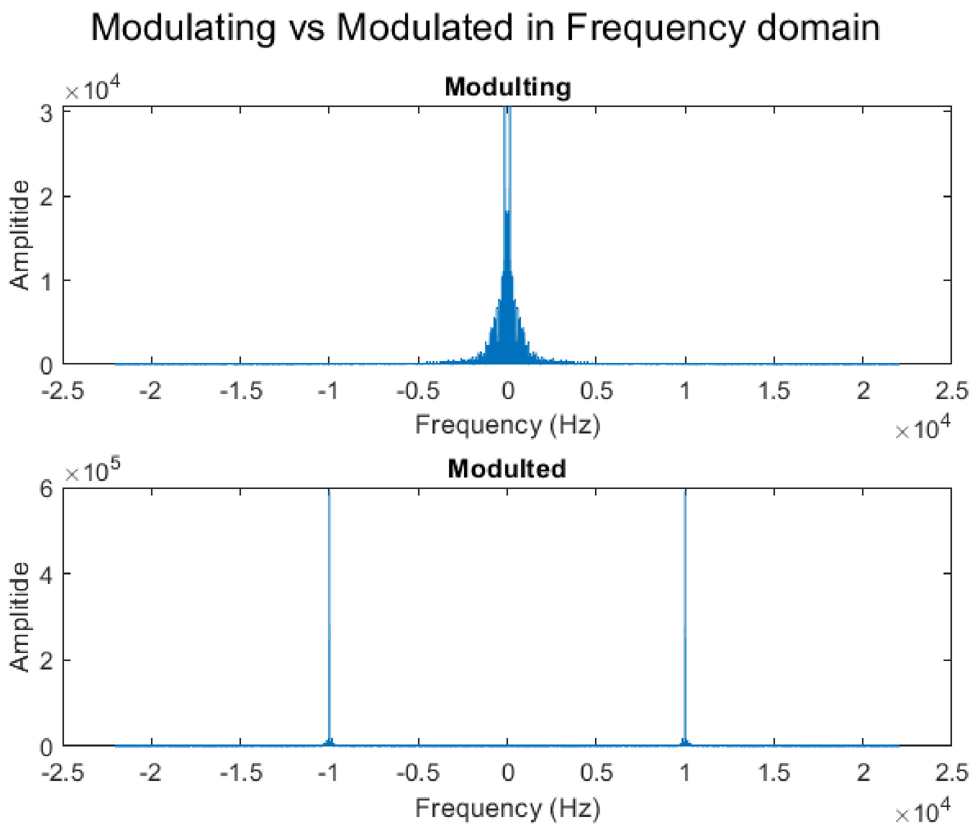
- if the signal toggle between negative and positive x-axis **we won't able to know modulating signal from envelope**



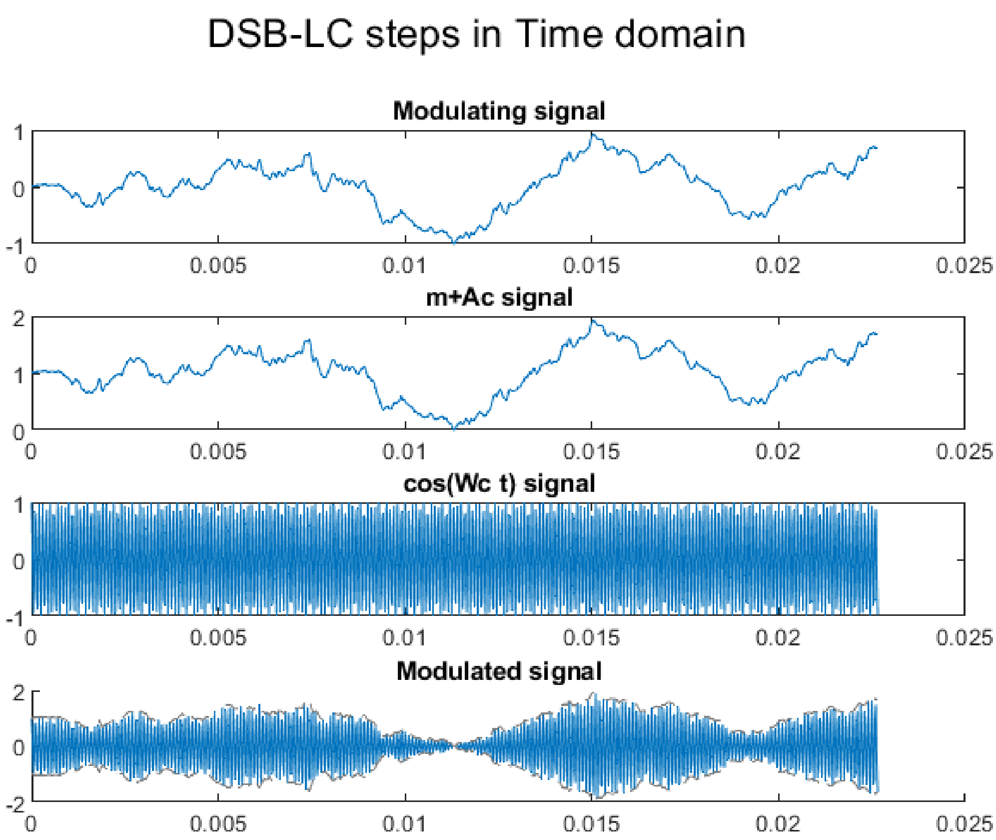
4. Compare between the bandwidth of the audio signal and the modulated one by plotting both signal in the frequency domain.

comment:

- it's clear that **amplitude have peaks at** F_c **&** $-F_c$
- and **bandwidth** of modulated signal is **double** modulating signal *not clear in plot*
- modulated its mirror (even) because modulating signal is real

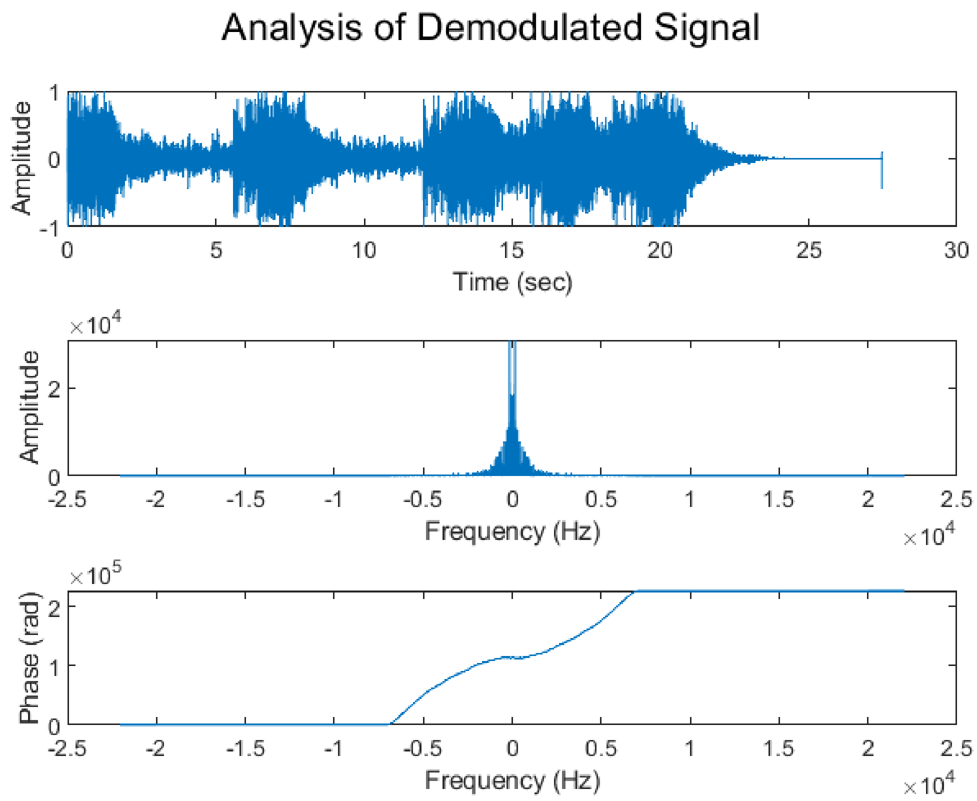


Bonus: steps of DSB-LC modulated signal in time domain



3. Perform DSB-LC Demodulation

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



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

comment:

😲 they Hear the same (for left channel *as I removed right channel to ease calculations)

small change in phase : I think it may be the Low pass filter function because after all I approximate bandwidth by my eye

why:

- carrier signal used in modulation and demodulation process is the same we know exactly ϕ_c, A_c, w_c
- achieve max power efficiency 33%

Code

📊 Utility function to make it easy to analyse signals

analyzeSignal.m

```
function X =  analyzeSignal(time_vector, x, Fs, title)
    figure;
    subplot(3,1,1);

    plot(time_vector, x);
    xlabel('Time (sec)');
    ylabel('Amplitude');

    % plot frequency domain
    % n = 512 % to avoid computation that harm my device
    X = fft(x);
    X = fftshift(X); % shift value to middle at zero
    X_mag = abs(X);
    X_phase = unwrap(angle(X));
    % can make it till 20k hz because we only hear that
    freq_vector = linspace(-Fs/2, Fs/2, length(X));

    subplot(3,1,2);
    plot(freq_vector,X_mag);
    xlabel('Frequency (Hz)');
    ylabel('Amplitude');

    subplot(3,1,3);
    plot(freq_vector,X_phase);
    xlabel('Frequency (Hz)');
    ylabel('Phase (rad)');

    sgtitle(title);
end
```

🎵 Task 1: read audio file and analyse it

assignment_sol.m

```
clear all;
close all;

%% ========= task 1 ======

% 1. read audio and get modulting signal
[audio, Fs] = audioread("song.wav");
m = audio(:,1); %get first channel only because its stero
m = reshape(m, 1, []); %reshape to ensure dimension correct

% 2. define time vector
audio_time = length(m)/Fs; % get numbers of seconds in the audio
time = linspace(0, audio_time, length(m));

% 3. analyse it
M = analyzeSignal(time, m, Fs, 'Analysis of Modulating Signal');
```

📡 Task 2: DSB-LC Modulation

assignment_sol.m

```
%% ========= task 2 ======
% 1. get helper variables
M_mag = abs(M);
mp = abs(min(m));

mue = 1; % mp/Ac
bandwidth = 5000; % approx from the plot
% 2. define carrier properties
Ac = mp / mue;
Fc = 2* bandwidth;
Wc = 2*pi*Fc;
% 3. define signals
c = cos(Wc .* time);
m_added = m + Ac; % intermidate value for later plotting
y = m_added .* c;
% 3.* plot steps
figure;
sgtitle("DSB-LC steps in Time domain");


subplot(4, 1, 1);
plot(time(1:1000), m(1:1000));
title("Modulating signal");

subplot(4, 1, 2);
plot(time(1:1000), m_added(1:1000));
title("m+Ac signal");

subplot(4, 1, 3);
plot(time(1:1000), c(1:1000));
title("cos(Wc t) signal");

subplot(4, 1, 4);
title("Modulated signal");
hold on;
plot(time(1:1000), y(1:1000));
%print envolope around modulted signal
plot(time(1:1000), m_added(1:1000), 'color', [0.5, 0.5, 0.5], 'LineStyle', '--');
plot(time(1:1000), m_added(1:1000)*-1, 'color', [0.5, 0.5, 0.5], 'LineStyle', '--');



% 4. analyse
Y = analyzeSignal(time, y, Fs, 'Analysis of Modulated Signal');

% 5. compare two signals in frequency domain
frequency = linspace(-Fs/2, Fs/2, length(M));

figure;
subplot(2, 1, 1);
sgtitle("Modulating vs Modulated in Frequency domain");

plot(frequency, M_mag);
title("Modulting");
xlabel("Frequency (Hz)");
ylabel("Amplitide");

subplot(2, 1, 2);
plot(frequency, abs(Y));
title("Modulted");
xlabel("Frequency (Hz)");
ylabel("Amplitide");
```

📻 Task 3: DSB-LC Demodulating

assignment_sol.m

```
%% ========= task 3 ======
% 1. get demodulate signal
x = y .* c;
x = lowpass(x, bandwidth ,Fs);
x = 2*x - Ac;
% 2. analyze it
X = analyzeSignal(time, x, Fs, 'Analysis of Demodulated Signal');
% 3. write to another file to hear it
audiowrite('output.wav',x,Fs);
```