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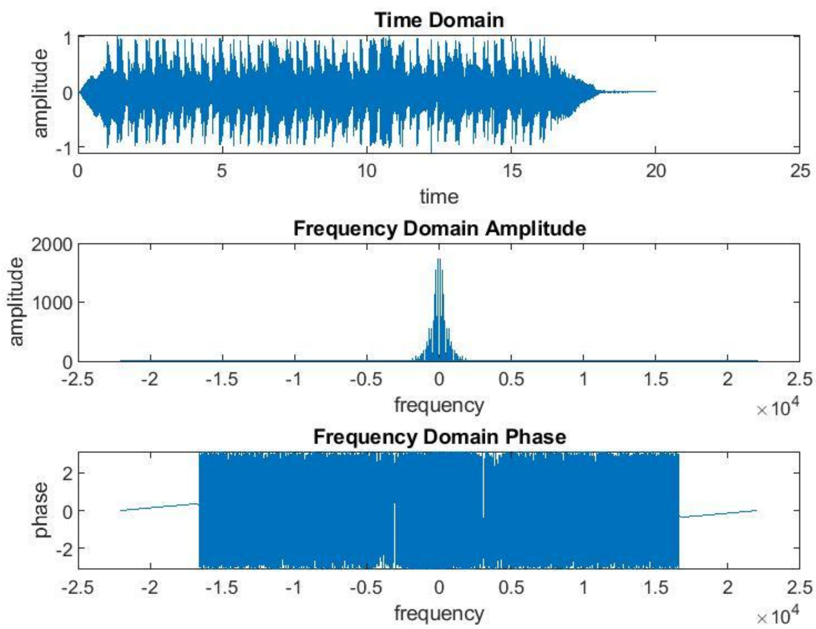
Lab1 Report

Amplitude Modulation

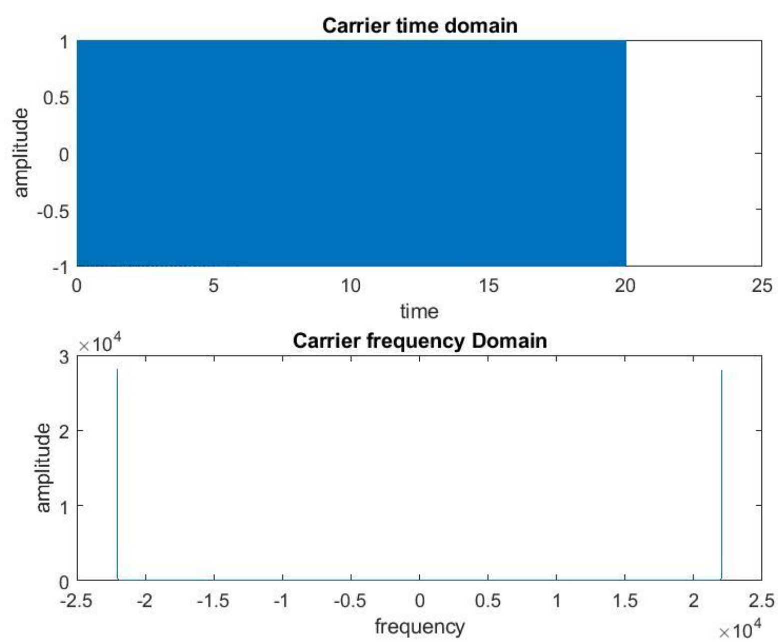
DSB-LC

SIMULATION RESULTS

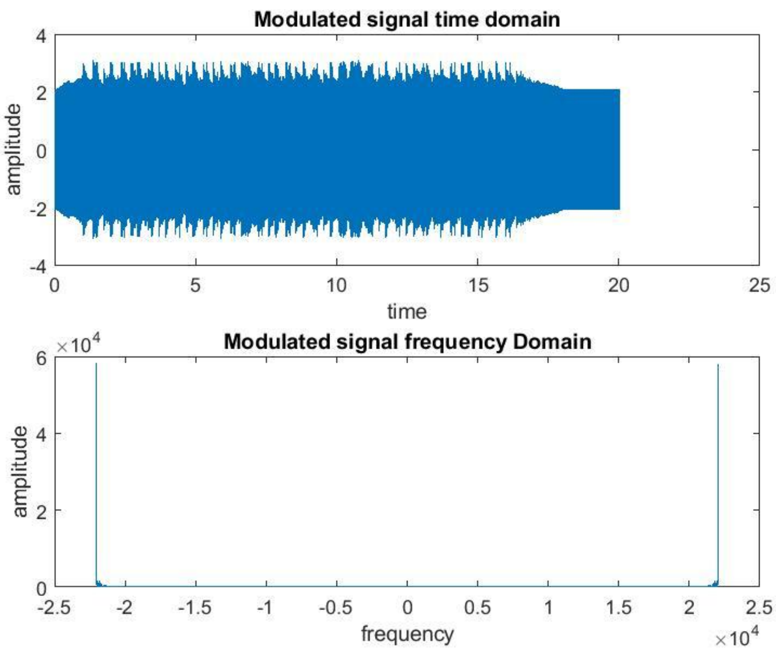
Message Signal:



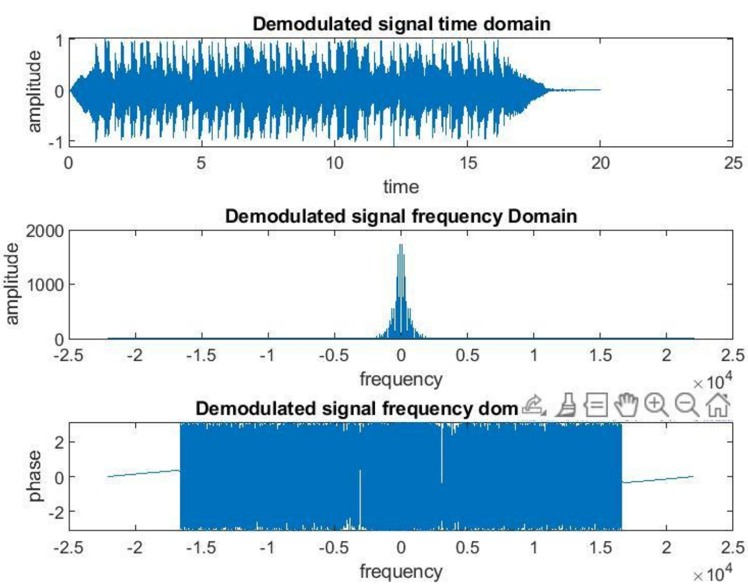
Carrier Signal:



Modulated Signal:



Demodulated Signal:



CODE

```
%1-Read audio signal
[mt,fs] = audioread('audio_signal.m4a');            %mt: sampled audio,fs: sampling frequency
mt = mt(:,1);                                        %to handle sizes
audio_length = length(mt);

%initialize parameters
ts = 1/fs;                                           %sampling interval
fc = (fs/2);                                         %carrier frequency
t = 0:ts:(audio_length-1)*ts;                        %time interval of signal
f = -fs/2:fs/2;
mp = max(mt);                                        %maximum value of signal
m = 0.5;                                             %modulation index
A = mp/m;                                            %DC value

%plot audio signal in frequency domain
mt_fft_amplitude = abs(fftshift(fft(mt,fs+1)));      %obtain ft of modulating signal
mt_fft_phase = angle(fftshift(fft(mt,fs+1)));

%generate carrier
ct = cos(2*pi*fc*t);

%obtain and plot carrier in frequency domain
ct_fft = abs(fftshift(fft(ct,fs+1)));                %obtain ft of carrier signal

%modulate the signal
st = (A + mt) .* ct';

%plot modulated signal in frequency domain
st_fft = abs(fftshift(fft(st,fs+1)));                %obtain ft of modulating signal

%demodulation
wt = st .* ct';                                      %multiply by same carrier
wt_lpf = lowpass(wt, fc, fs, 'Steepness', 0.95) - A;
wt_fft_amplitude = abs(fftshift(fft(wt_lpf,fs+1)));  %obtain ft of modulating signal
wt_fft_phase = angle(fftshift(fft(wt_lpf,fs+1)));    %obtain ft of modulating signal

audiowrite('demodulated_signal.m4a', wt_lpf, fs);    %Export audio
```

COMMENTS

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

ω = 2*pi*fc, to achieve good modulation fc <= fs/2 so I set fc to fs/2 which I found to generate the most accurate results.

I set the modulation index(m) to 0.5 to avoid under or over modulation and got the maximum value (mp) from the received samples and got A = mp/m.

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?

When the amplitude of the modulating signal exceeds that of the carrier wave, the modulation index exceeds one, resulting in information loss. Hence the carrier's minimum amplitude is mp.

Are the two signals the same? Explain why?

Yes, because the modulation and demodulation were done with the same carrier frequency and the DC value subtracted so any change that occurred while modulating was reversed in the demodulating process.

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

In the above figures