Cairo University- Faculty of Engineering
Computer Engineering Department
Communications Engineering – Fall 2022

Modulation project

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Communication project

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1/5/2023

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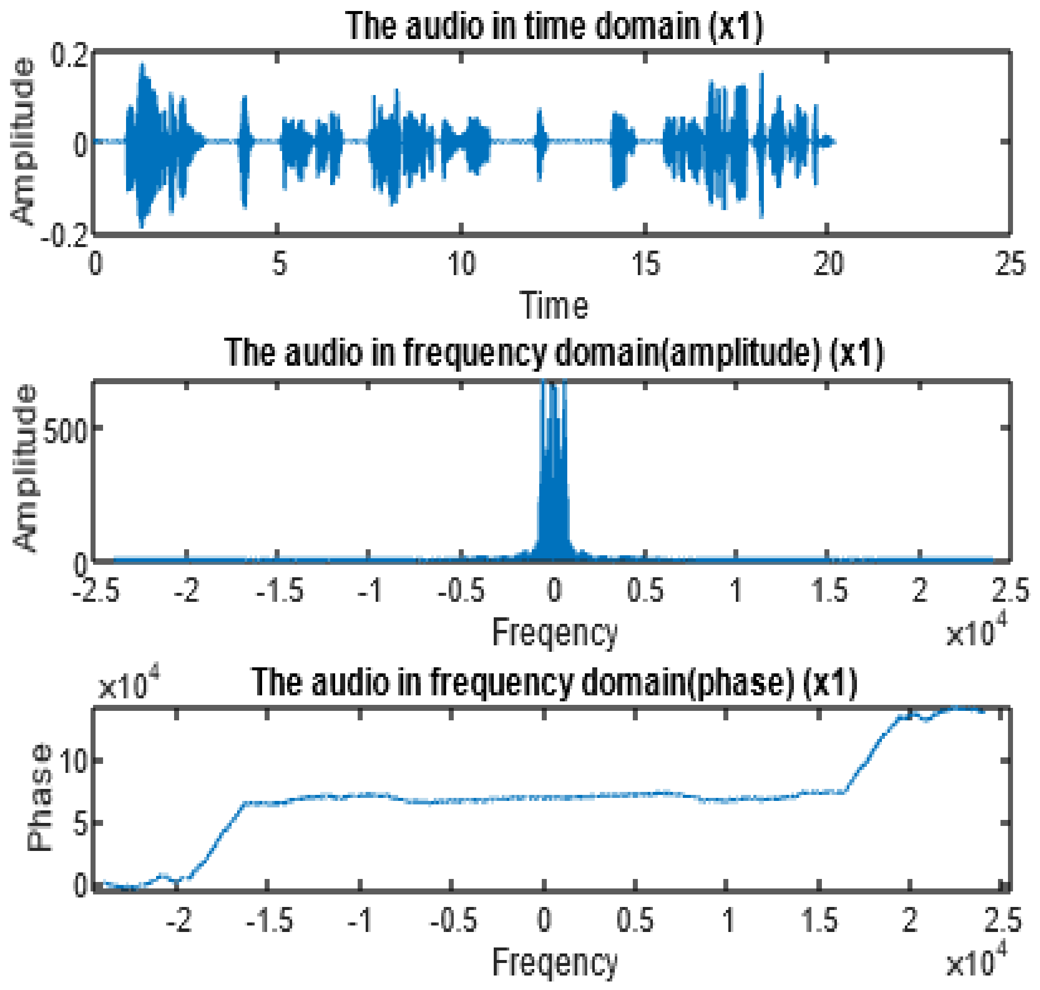
Modulation project

<u>Q1) Obtain the modulated signal. Plot it in time domain. Plot its magnitude spectrum.</u>

Signals before modulation

<u>First Signal:</u>



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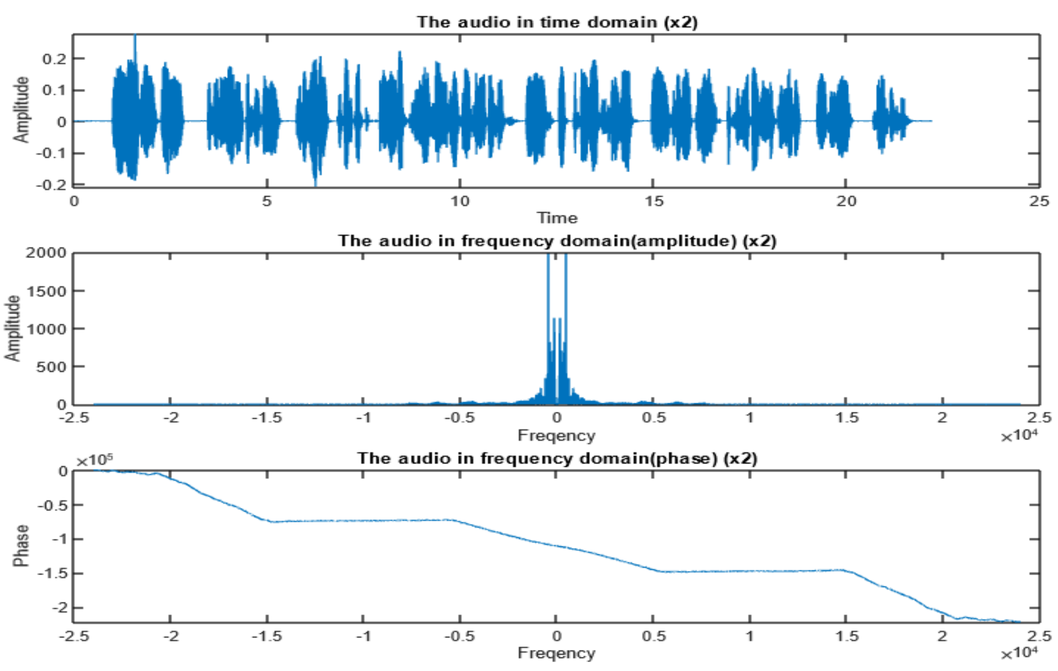
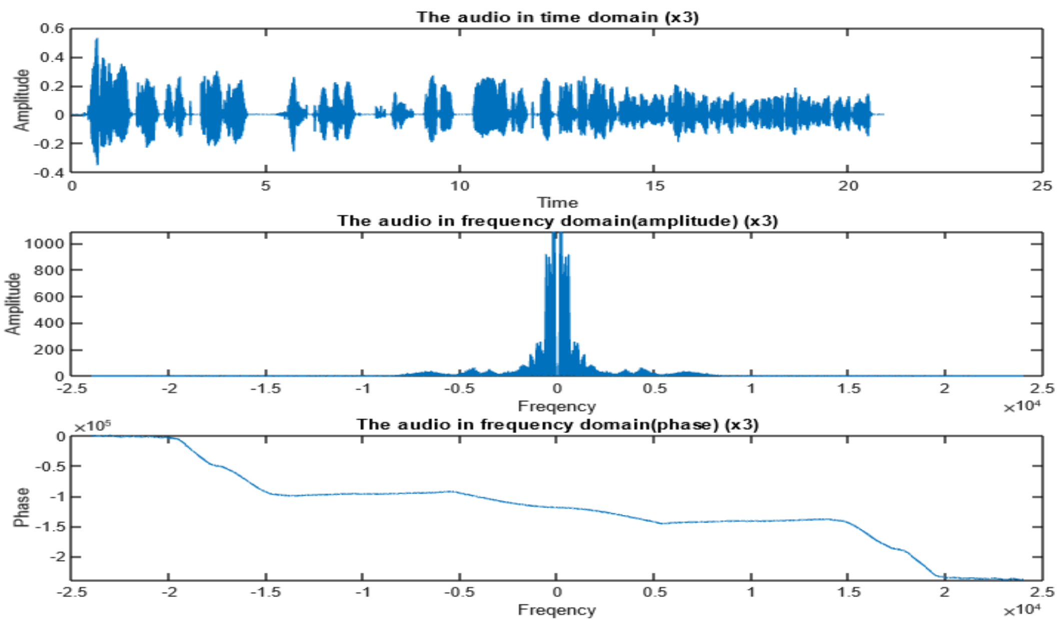
Second Signal:



Third signal:



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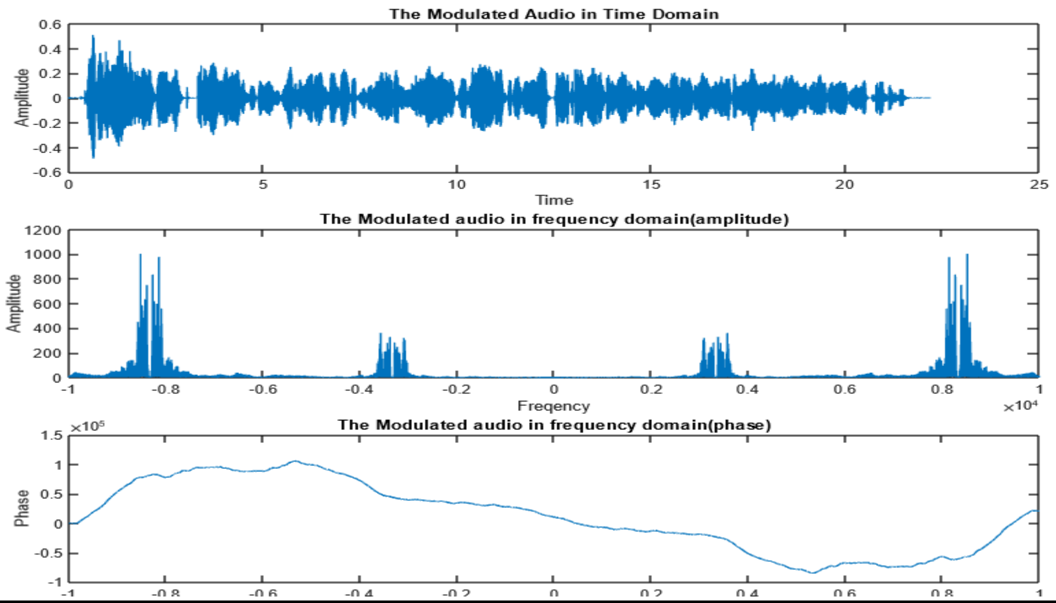
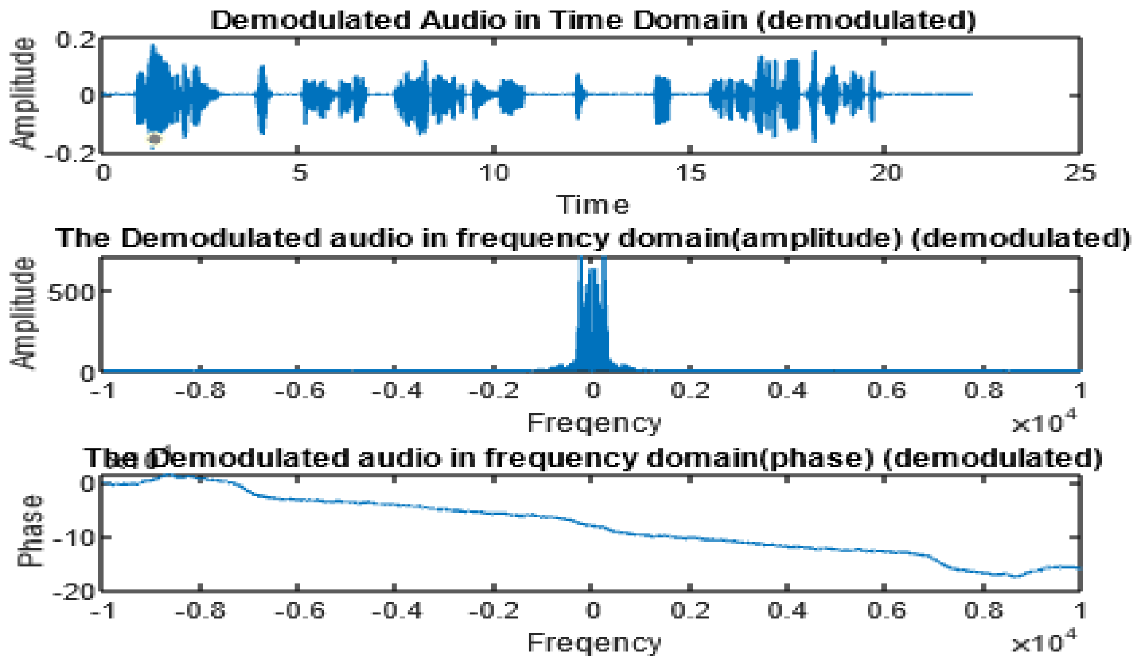
Signals after modulation



Q2) Perform synchronous demodulation to restore the three signals.

First Signal:



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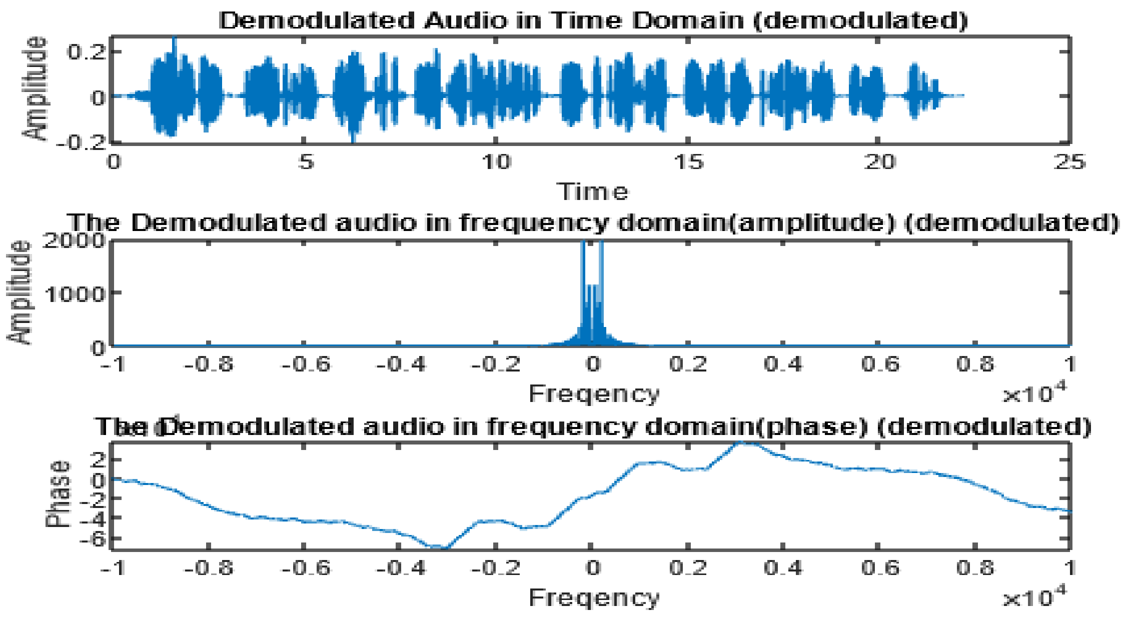
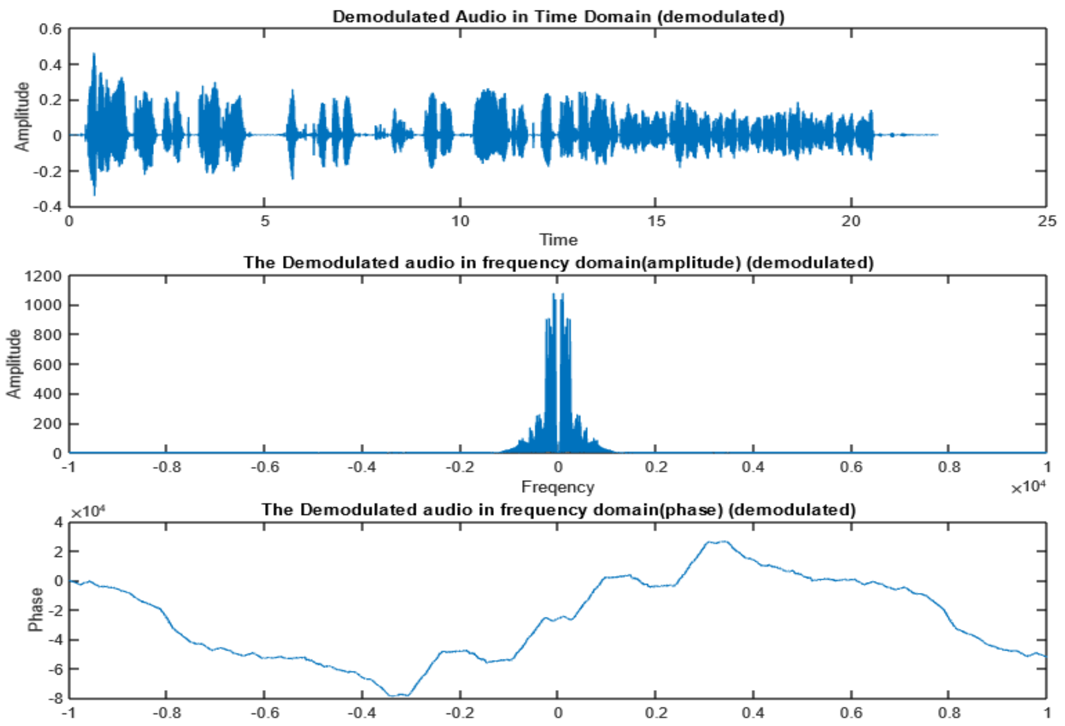
Second signal:



Third signal:



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Justification:

This happened because we have chosen a carrier frequency for the first signal which equals to 8000 Hz and we assigned the second frequency to 2.5 times of the first one(20000 Hz), so after applying the synchronized demodulation we multiplied by the locally generated carrier with the same frequency by which we have applied the modulation and after that we applied a low pass filter to discard the high frequencies, which we don’t want to take.

But notice that we got the same signals for each one although we have transmitted all signals together on the same spectrum, but that happens because during the demodulation we can select which signal we want to hear by multiplying by a local-generated carrier with the same frequency as that which we have modulated the signal on, so in case of the first signal we multiplied by cos(2*pi*t * 8000) and after applying the LPF (low-pass filter) we can only extract the signal on that frequency only, while for the other signals, notice that we used the QAM idea that we modulated two signals on the same frequency, but we made a trick which is sending them on different 90 phase, so by applying some trigonometric equations we know that this phase shift will cause no interference between the two signals, so by this idea we transmitted two signals, on the same carrier frequency, and by applying the same logic we discussed before we achieved the same signals by the demodulation.

Moreover, during the modulation no interference happened that is because the frequency on which we modulate the signals are greater than the bandwidth of both of them, so none of them has affected the other.

Also, notice that we have two frequencies (8 KHZ, 20 KHZ) for each carrier to be able to achieve that the carrier frequency is less than half of the sampling rate (48kHZ).

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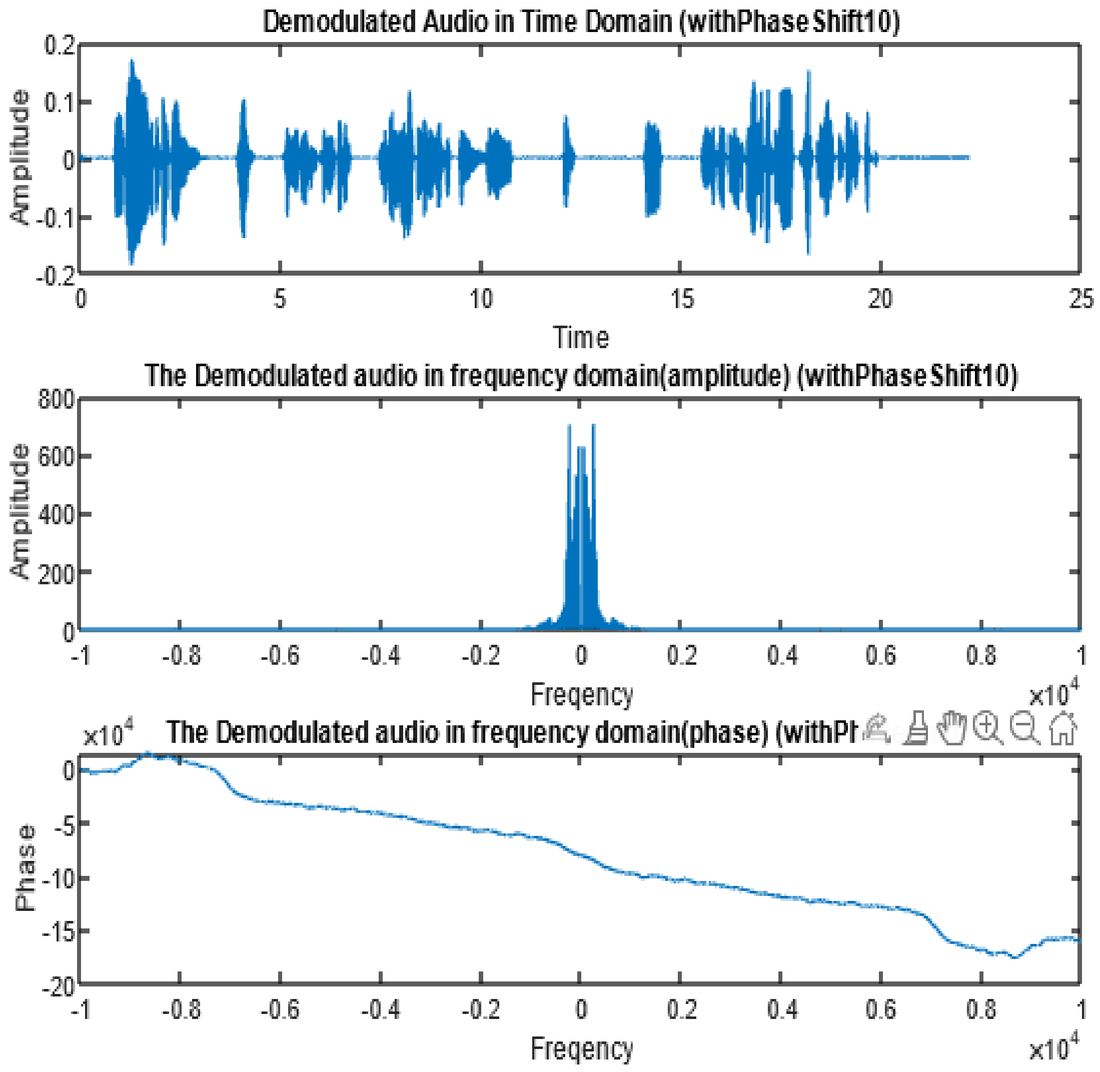
Q3) Perform demodulation three times with phase shifts of 10, 30, 90 degrees for both carriers.

Q3) a- with phase shift 10:

First Signal:



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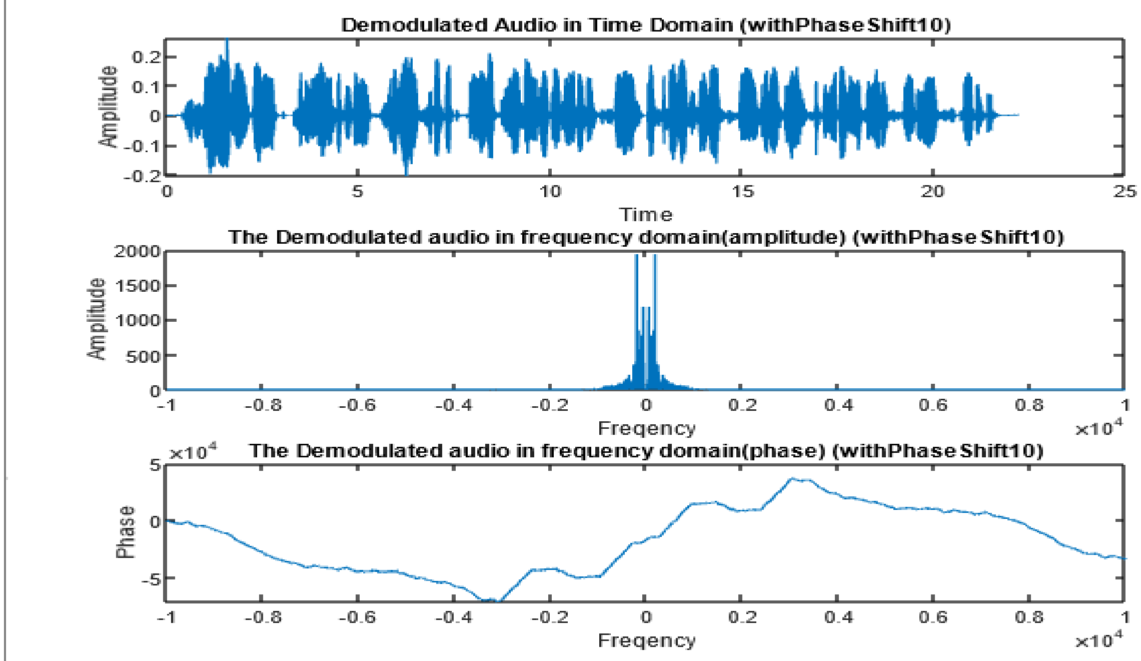
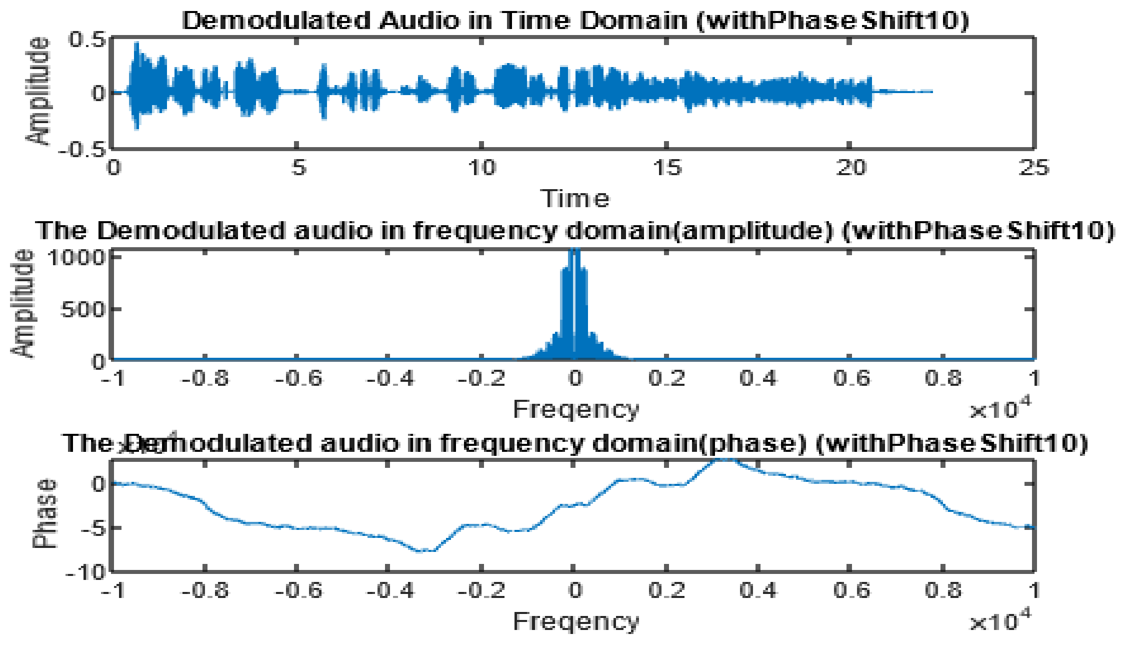
Second Signal:



Third Signal



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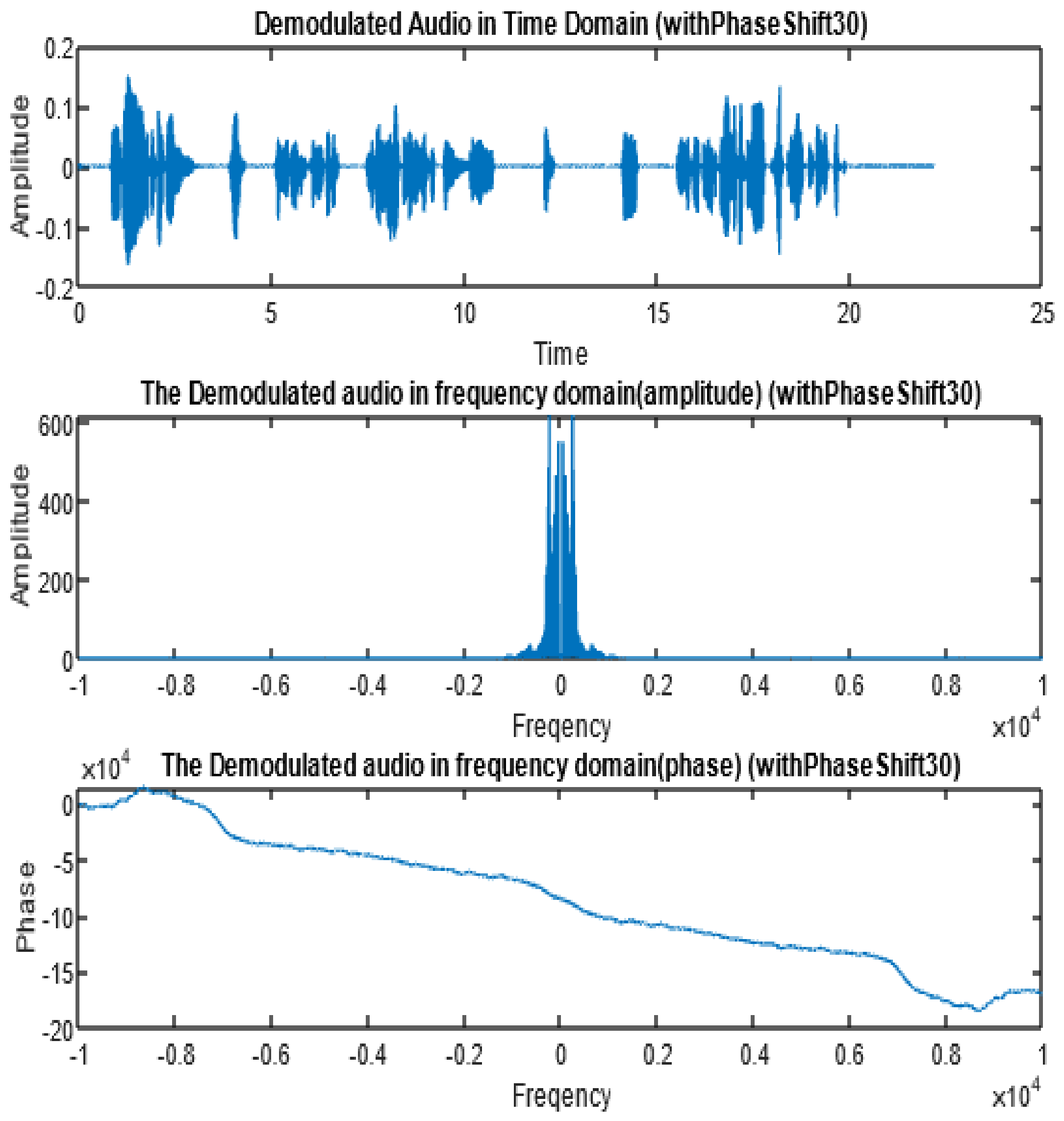
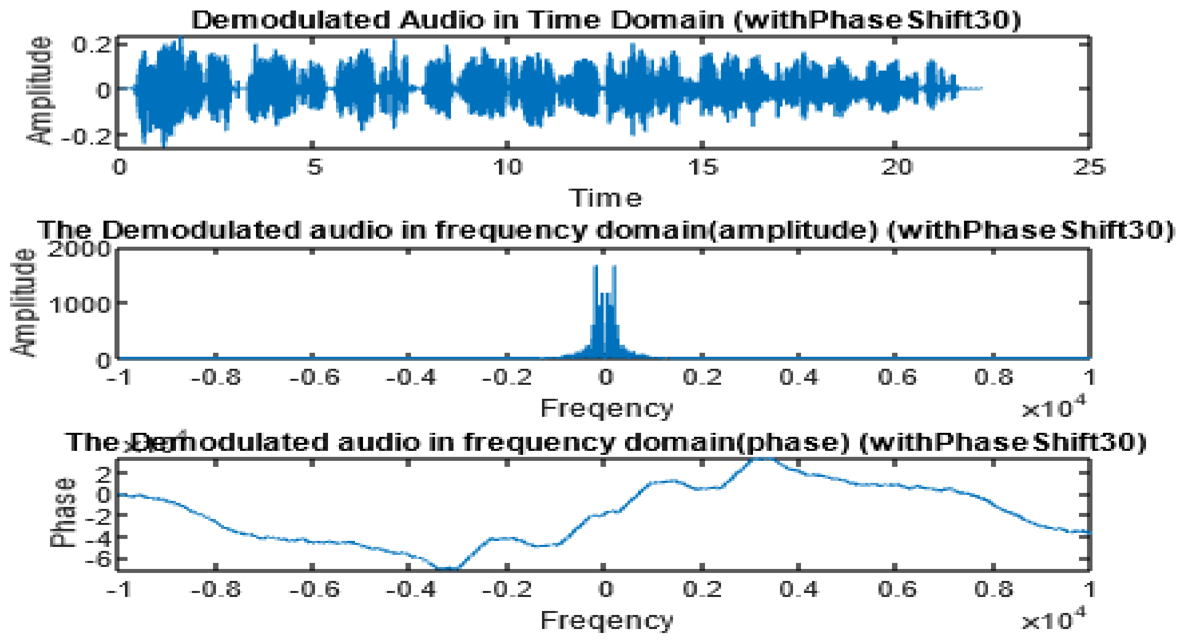
Q3) b- with phase shift 30:

First Signal:

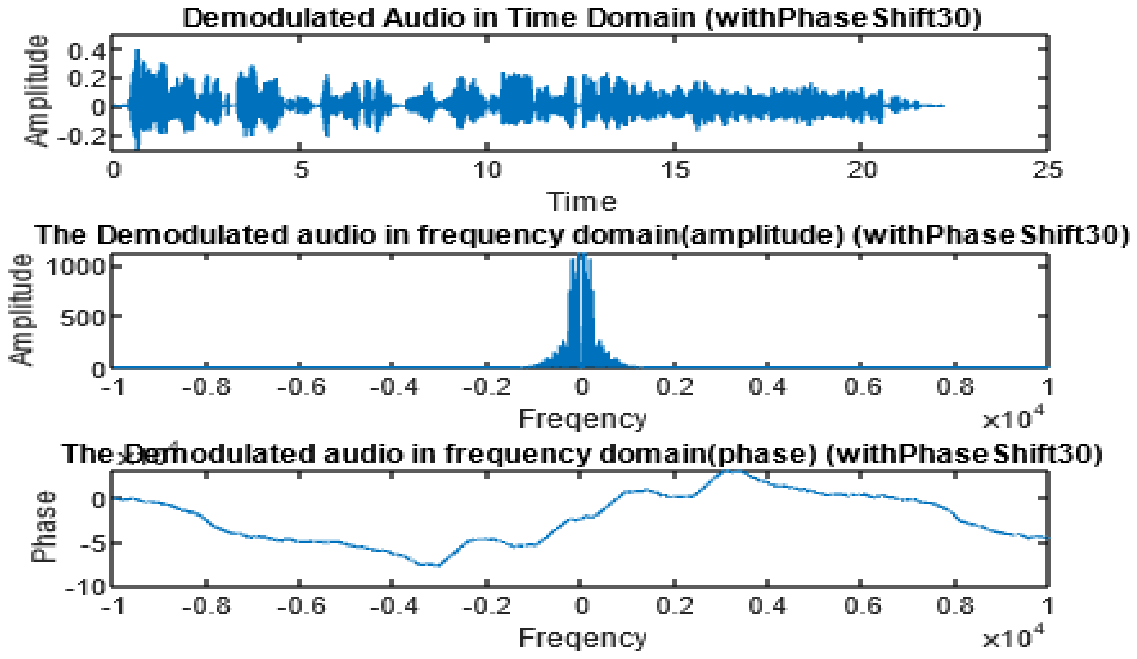


Modulation project

Second Signal:



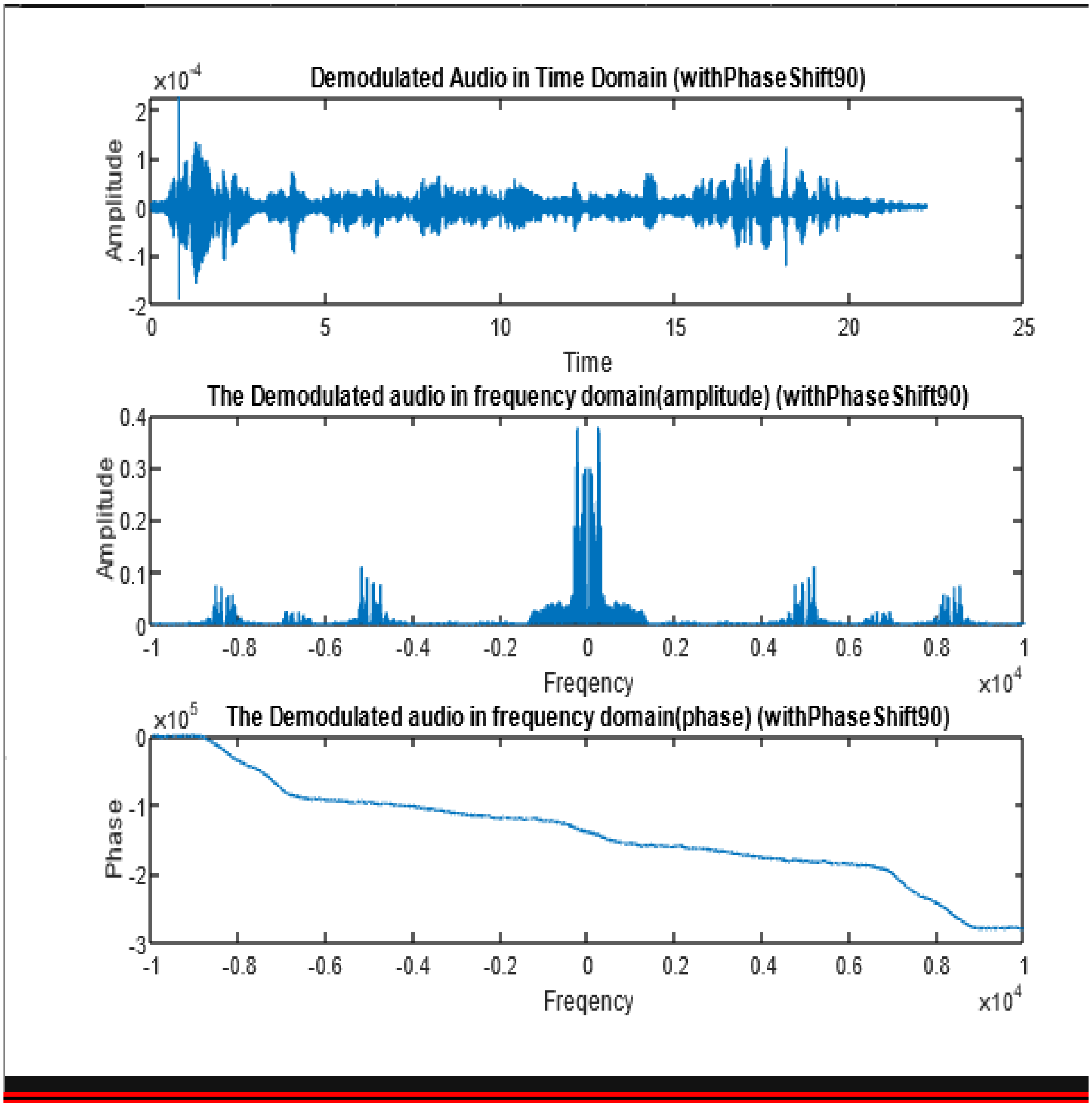
Third Signal:



Modulation project

Q3) c- with phase shift 90:

First Signal: (notice that the amplitude is multiplied by 10^(-4))



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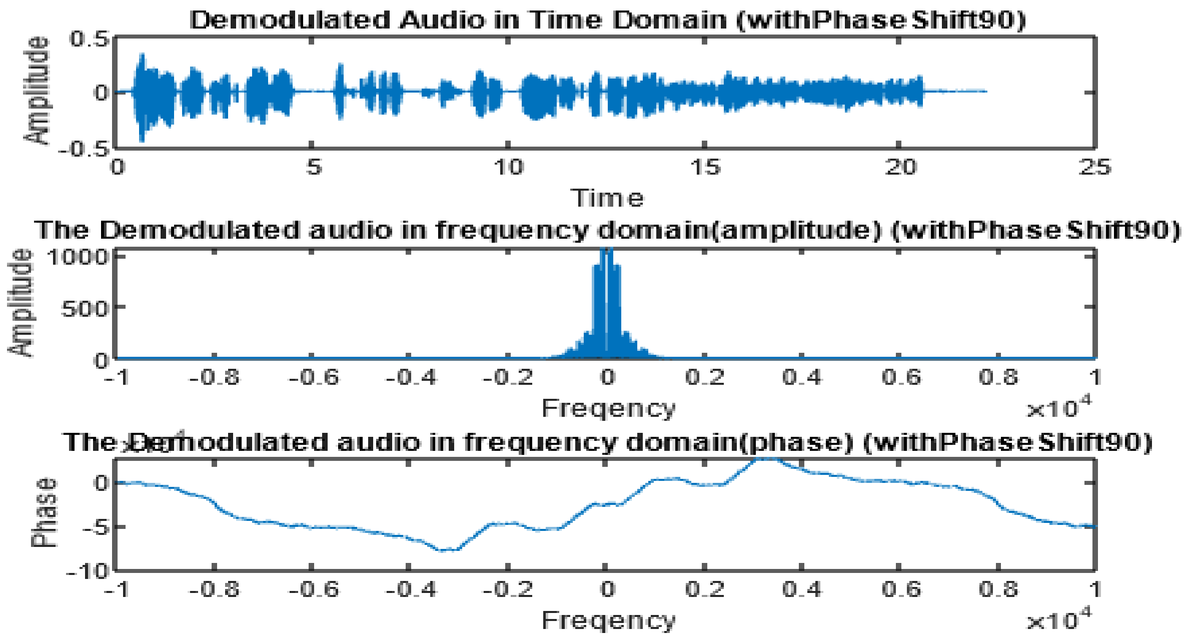
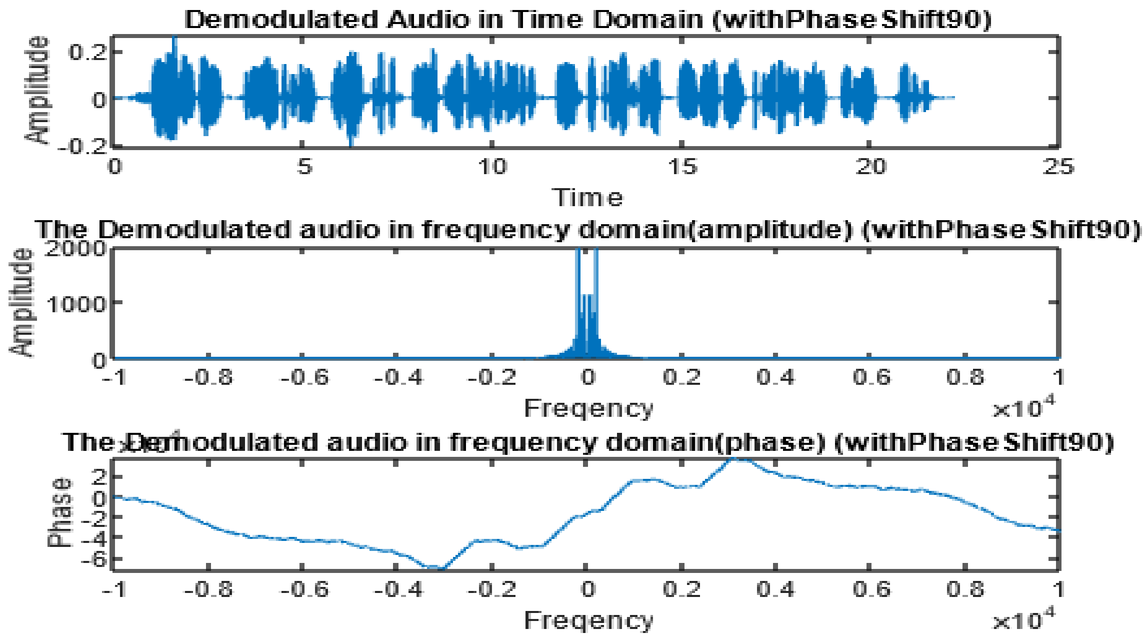
Modulation project

Second Signal:



Third Signal:



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Modulation project

Justifications:

In case of 10 and 30:

Any phase shift case will cause **attenuation** on the signals, so in case of the first signal It will cause only **attenuation** but with small value as the result signal will be the original signal * cos(the phase shift), but in the second and third signals, since they are sent on the same bandwidth and we use the idea of **QAM**, so we depend on the idea that the sin and cos with the same frequency to can avoid any component from any of the two signals on each other, but since we have changed the phase, so we will find some components of each of them affecting the other, so it will cause:

- a **distortion** because the two signals will affect on each other, and this will happen in both cases (10 and 30) (interference between the two channels, so we can hear both voices on both files due to the interference.)
- **Attenuation**.

But in case of 90 phase shift:

- For the first signal it will be **obliterated** till it reach approximately 0, so we will hear nothing in the audio file.
- While for the second and third signals no **attenuation** will take place, that is because we will just replace the signals, as the cos will turn into sin and vice versa so we will hear the second audio instead of the third and vice versa.

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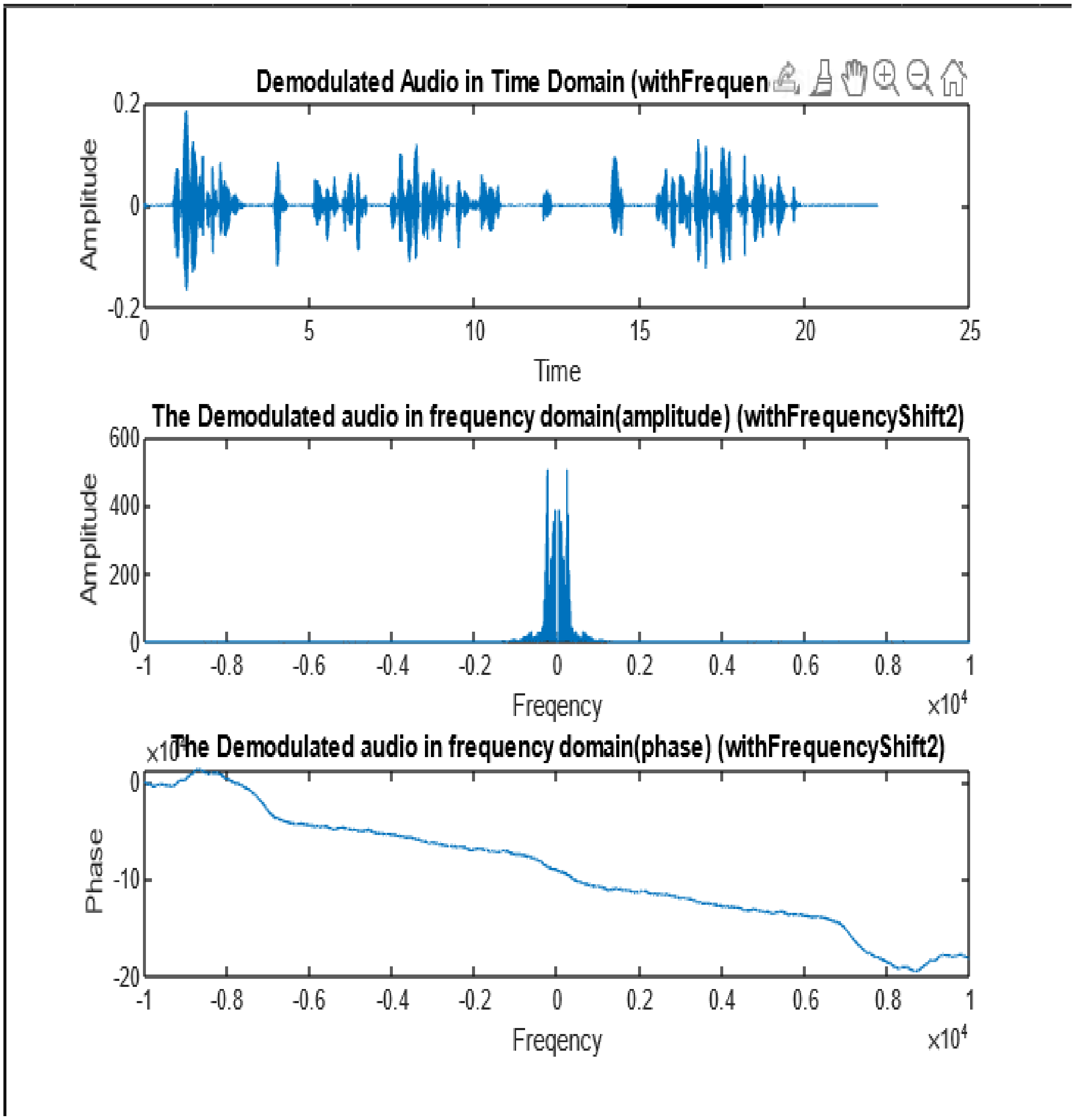
Modulation project

Q4) Perform demodulation two times with a local carrier frequency that is different by 2 Hz and 10 Hz from its carrier frequency.

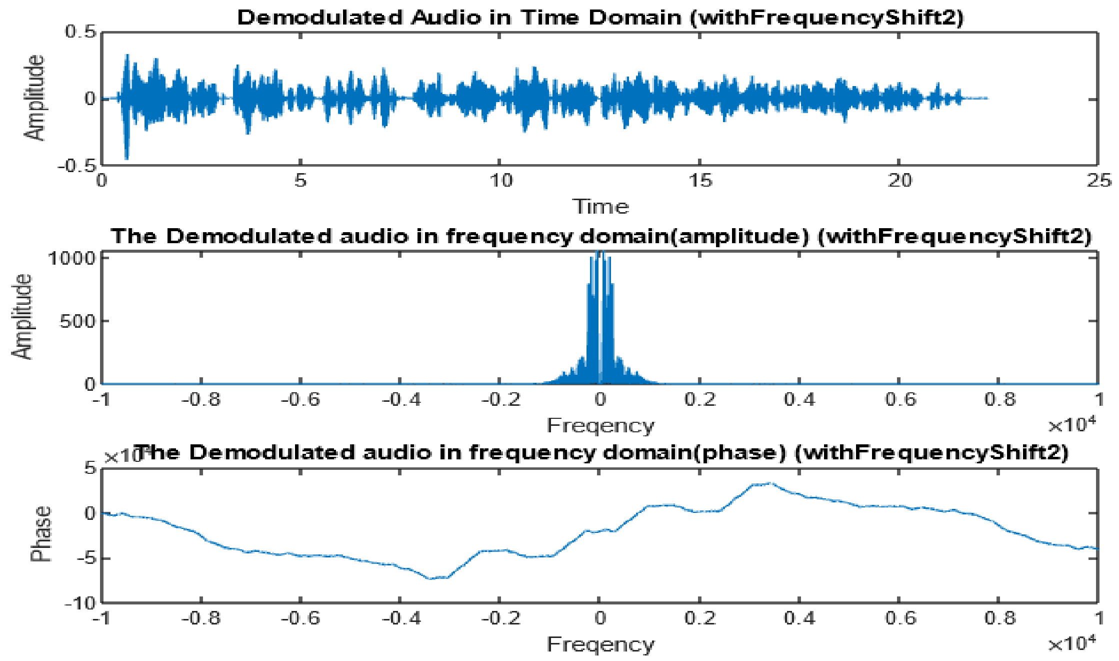
Q4) a- by 2 Hz

First Signal:

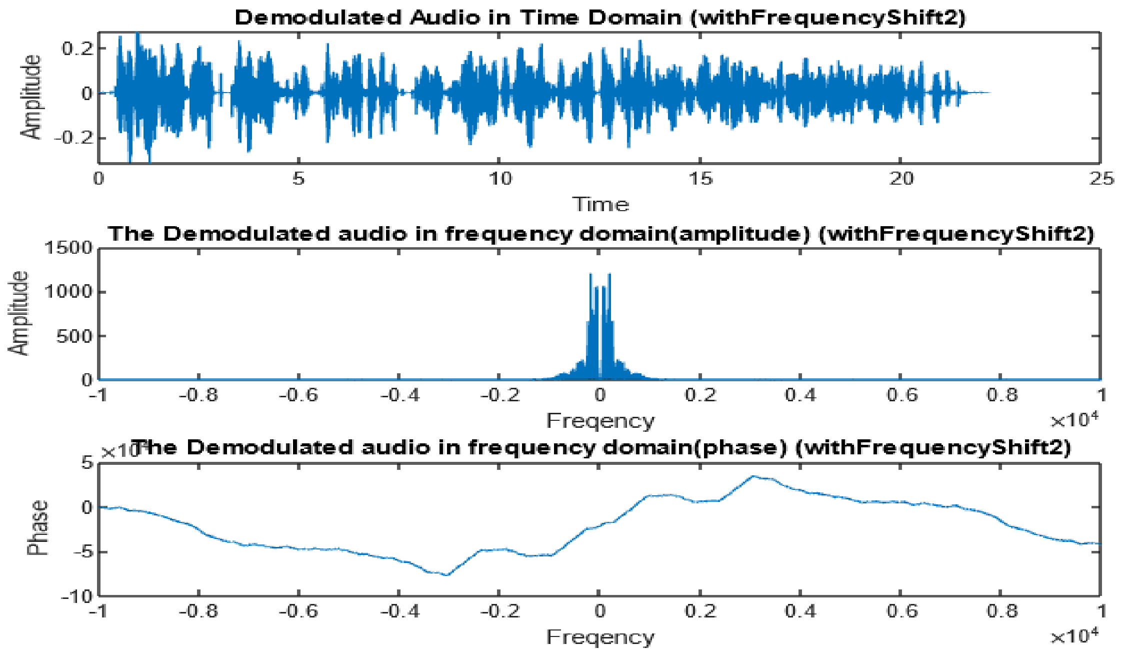


Modulation project

Second Signal:



Third Signal:



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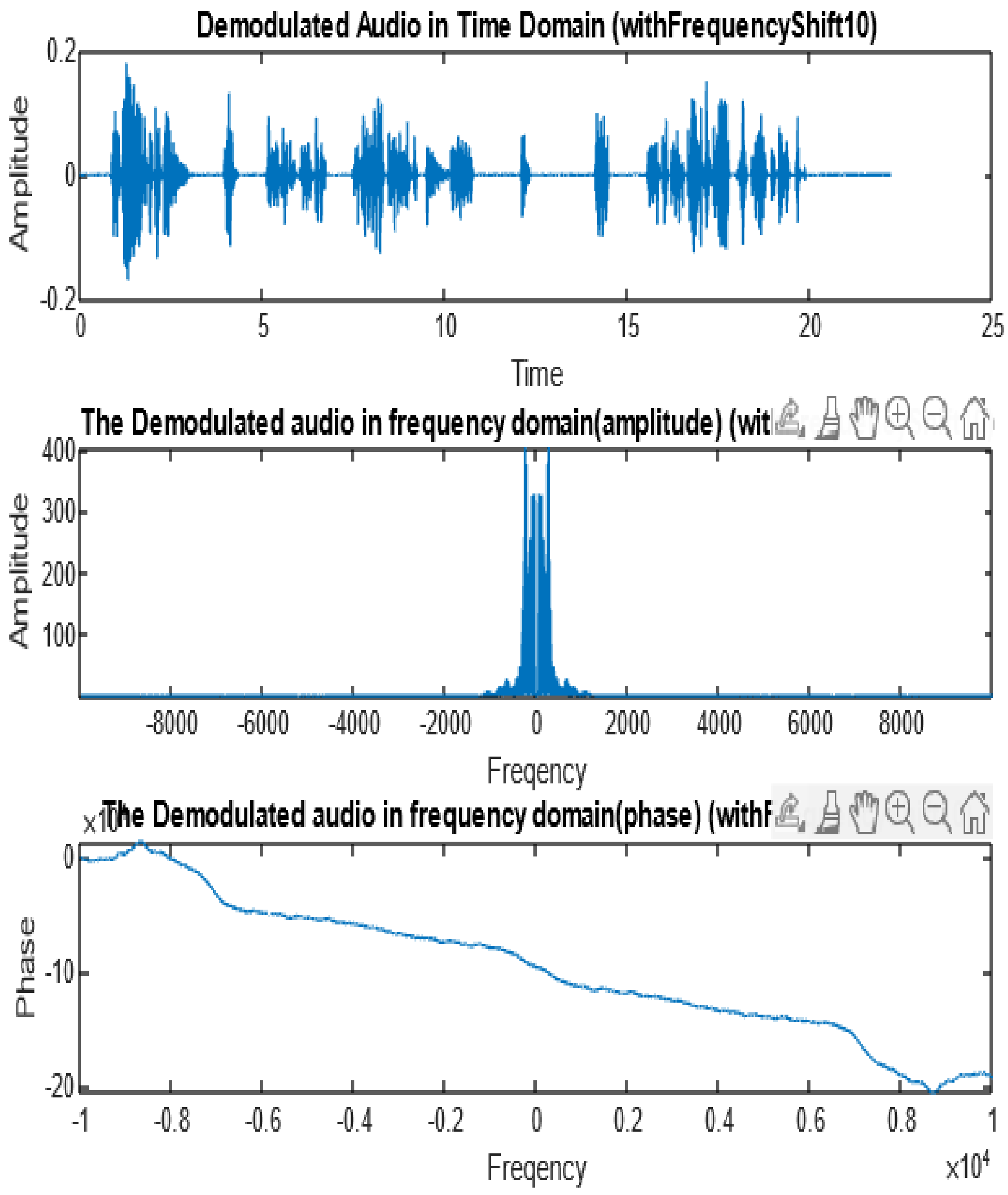
Modulation project

Q4) b- by 10 Hz

First Signal:



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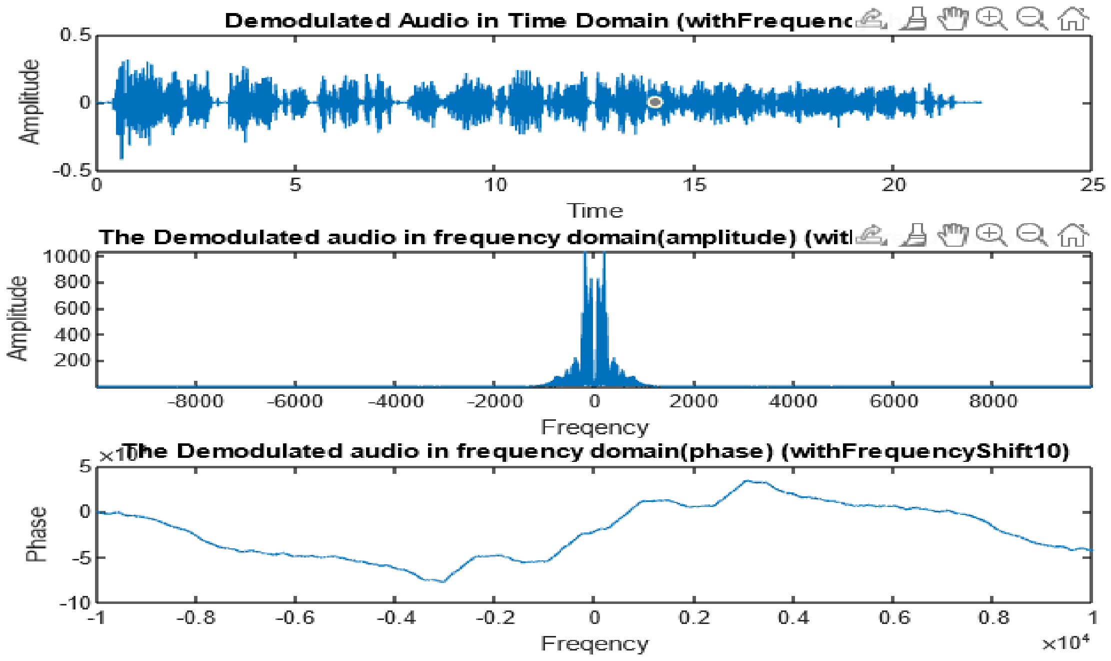
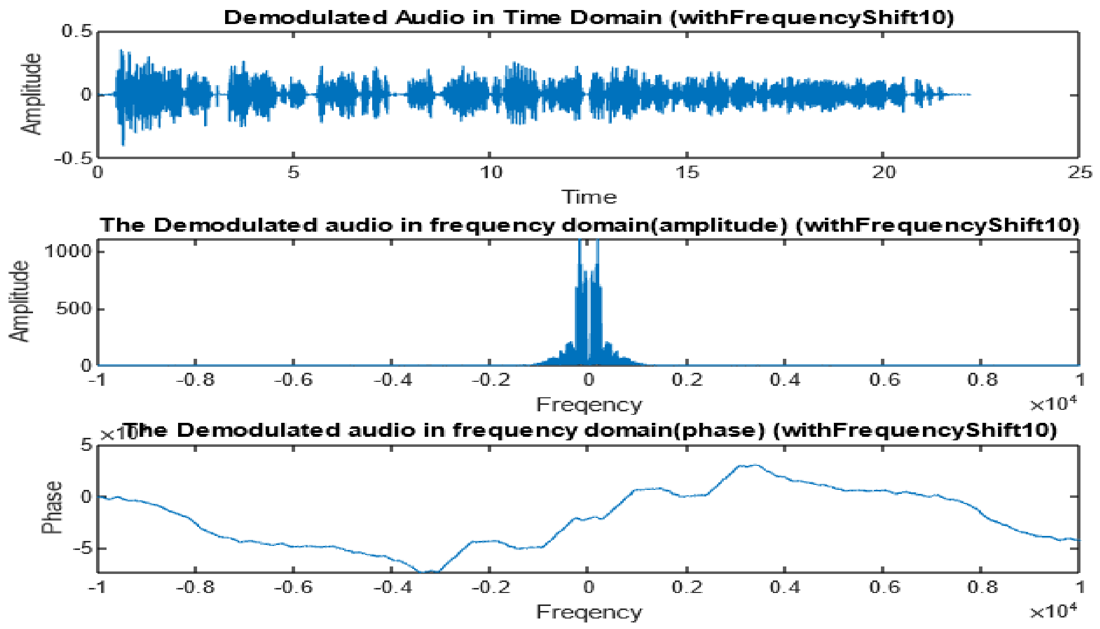
Second Signal:



Third Signal:



Modulation project

For frequency shift, this will cause attenuation on the signals and also it will cause distortion, and this is applied for both cases shift by 2 and also shift by 10.

But as the shift increases the attenuation and distortion increase, so the distortion and attenuation in case of 2HZ will be less than that of 10 HZ.

Moreover, in the case of the signals which are modulated using the QAM idea, there will be an interference between the two signals due to the high required synchronization which will be violated due to the shift in the frequency, so we can hear both voices on both files due to the interference.

Modulation project

Code Snippets:

```
[x1,fs1] = audioread('Adham.m4a');
[x2,fs2] = audioread('abdelaziz.m4a');
[x3,fs3] = audioread('third.m4a');
x1=x1(:,1);
x2=x2(:,1);
x3=x3(:,1);

%===============================Plot The Three
Signals=====================================

plot_all(x1,fs1,'x1');
plot_all(x2,fs2,'x2');
plot_all(x3,fs3,'x3');
fs=min([fs1,fs2,fs3]);
max_len=max([length(x1),length(x2),length(x3)]);
t = linspace(0,max_len/fs,max_len);
x1 = [x1; transpose(zeros(1, max_len - length(x1)))];
x2 = [x2; transpose(zeros(1, max_len - length(x2)))];
x3 = [x3; transpose(zeros(1, max_len - length(x3)))];

%===============================Modulation==================
===================
carrier_one_freq=8000;
carrier_two_freq=2.5*carrier_one_freq;

carrier1=cos(2*pi*carrier_one_freq*t);
carrier2=cos(2*pi*carrier_two_freq*t);
carrier3=sin(2*pi*carrier_two_freq*t);

s=x1.*carrier1.'+x2.*carrier2.'+x3.*carrier3.';

%Draw The Modulated Audio
figure;
subplot(3,1,1);
plotting(t,s,'Time','Amplitude','The Modulated Audio in
Time Domain');

%Calculate Fourier Transform
[m,phase,f]=calc_fft(s,carrier_two_freq);
```

Modulation project

```
%Draw The Modulated Audio Amplitude in Freqency Domain
subplot(3,1,2);
plotting(f,m,'Freqency','Amplitude','The Modulated audio in
frequency domain(amplitude)');

%Draw The Modulated Audio Phase in Freqency Domain
subplot(3,1,3);
plotting(f,phase,'Freqency','Phase','The Modulated audio in
frequency domain(phase)');

%==============================Demodulation
1=====================================
phase_shift=0;
frequency_shift=0;
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,'demodulated');

%==============================Demodulation 2(with phase
shift)=====================================
phase_shift=10;
frequency_shift=0;
title='withPhaseShift10';
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)

phase_shift=30;
frequency_shift=0;
title='withPhaseShift30';
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)

phase_shift=90;
frequency_shift=0;
title='withPhaseShift90';
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)

%==============================Demodulation 3(with
frequency shift)=====================================
phase_shift=0;
frequency_shift=2;
title='withFrequencyShift2';
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)

phase_shift=0;
frequency_shift=10;
title='withFrequencyShift10';
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)


function [m,phase,f] = calc_fft(x,fs)
    N=length(x);
    ftx=fft(x);
    m=abs(fftshift(ftx));
    phase=unwrap(angle(ftx));
    f=(0:N-1)*fs/N;
    f=f-fs*(N-1)/(2*N);
end

function plotting(x,y,labelx,labely,ptitle)
    plot(x,y);
    title(ptitle);
    ylabel(labely);
    xlabel(labelx);
end

function plot_all(y,fs,title)
% Set The Time Vector
t = linspace(0,length(y)/fs,length(y));

figure();
% Draw The Audio in Time Domain
subplot(3,1,1);
plotting(t,y,'Time','Amplitude',strcat('The audio in time
domain (',title,')'));

% Calculate Fourier Transform
[m,phase,f]=calc_fft(y,fs);
```

Modulation project

```
% Draw The Audio Amplitude in Freqency Domain
subplot(3,1,2);
plotting(f,m,'Freqency','Amplitude',strcat('The audio in
frequency domain(amplitude) (',title,')'));


% Draw The Audio Phase in Freqency Domain
subplot(3,1,3);
plotting(f,phase,'Freqency','Phase',strcat('The audio in
frequency domain(phase) (',title,')'));
end

function plot_demodulated(demodulated,fc,t,title)
    % Draw The Demodulated Audio
figure;
subplot(3,1,1);
plotting(t,demodulated,'Time','Amplitude',strcat('Demodulat
ed Audio in Time Domain (',title,')'));

% Calculate Fourier Transform
[m,phase,f]=calc_fft(demodulated,fc);

% Draw The Demodulated Audio Amplitude in Freqency Domain
subplot(3,1,2);
plotting(f,m,'Freqency','Amplitude',strcat('The Demodulated
audio in frequency domain(amplitude) (',title,')'));

% Draw The Demodulated Audio Phase in Freqency Domain
subplot(3,1,3);
plotting(f,phase,'Freqency','Phase',strcat('The Demodulated
audio in frequency domain(phase) (',title,')'));
end



function
demodulation_function(carrier_one_freq,carrier_two_freq,s,f
s,t,phase_shift,frequency_shift,title)
d_carrier1=cos(2*pi*(carrier_one_freq+frequency_shift)*t+ph
ase_shift/180*pi);
```

Modulation project

```
d_carrier2=cos(2*pi*(carrier_two_freq+frequency_shift)*t+ph
ase_shift/180*pi);
d_carrier3=sin(2*pi*(carrier_two_freq+frequency_shift)*t+ph
ase_shift/180*pi);

demodulated_x1=s.*d_carrier1.';
demodulated_x1=2*demodulated_x1;
demodulated_x1=lowpass(demodulated_x1,2000,fs,'Steepness',0
.95);
audiowrite(strcat(title,'x1.wav'),demodulated_x1,fs);

demodulated_x2=s.*d_carrier2.';
demodulated_x2=2*demodulated_x2;
demodulated_x2=lowpass(demodulated_x2,2000,fs,'Steepness',0
.95);
audiowrite(strcat(title,'x2.wav'),demodulated_x2,fs);

demodulated_x3=s.*d_carrier3.';
demodulated_x3=2*demodulated_x3;
demodulated_x3=lowpass(demodulated_x3,2000,fs,'Steepness',0
.95);
audiowrite(strcat(title,'x3.wav'),demodulated_x3,fs);



plot_demodulated(demodulated_x1,carrier_two_freq,t,title);
plot_demodulated(demodulated_x2,carrier_two_freq,t,title);
plot_demodulated(demodulated_x3,carrier_two_freq,t,title);
end
```