EEE321 - Signals & Systems

Lab Assignment III

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Part 1:

Part 1.1 (DTMF Transmitter)

-Code for DTMF

```
function [x] = DTMFTRA(Number)

%DTMFTRA Creates the transmission signal of Dual Tone Multi Frequency

%(DTMF), assuming each button is pressed for 0.25 seconds.

%   Number: 1xN vector of integers, contains the phone number that is

%   dialed.

%   x: 1x(0.25*N)/Ts vector, contains the samples of the transmitted

%   signal.

%   (internal) Ts: sampling period.

Ts = 1/8192; %sampling period

N = length(Number); %length of the Number

t = 0:Ts:(0.25*N-Ts); %time vector

T = 0.25/Ts; %length of each tone

%frequency lookup matrix

freq = [941 1336; %0            %look up table
```

```
697 1209; %1
697 1336; %2
697 1477; %3
770 1209; %4
770 1336; %5
770 1447; %6
852 1209; %7
852 1336; %8
852 1477];%9
x = zeros(size(t)); %initialize x(t)
for ii = 1:length(Number)
ni = Number(ii) + 1; %get the right index for freq
t_current = t(1+(ii-1)*T:ii*T); %isolate current time interval
%add DTMF cosines to x(t) at the corresponding t interval
x(1+(ii-1)*T:ii*T) = cos(2*pi*freq(ni,1)*t_current) + ...
    cos(2*pi*freq(ni,2)*t_current);
end
end
```

Comments

The sound I listened to is the same sound I hear when dialing a phone number on my cellular. Each digit has a different sound, and that is because the transmitted signal (digit) has different frequencies (fr_i , fc_i).

Part 1.2 (DTMF Receiver):

-The calculations of the Fourier transforms are uploaded in a separate pdf file.

Codes:

```
x = DTMFTRA([8 0 6 3 2]);

omega=linspace(-8192*pi,8192*pi,10241);

omega=omega(1:10240);

plot(omega, abs(X));

xlabel('omega');

ylabel('Magnitude of Signal X')

x1 = DTMFTRA(8);

x2 = DTMFTRA(0);

x3 = DTMFTRA(6);

x4 = DTMFTRA(3);

x5 = DTMFTRA(2);

X1 = FT(x1);

X2 = FT(x2);

X3 = FT(x3);

X4 = FT(x4);

X5 = FT(x5);

omega2=omega(1:2048);

figure
```

```
plot(omega2, abs(X1));

xlabel('omega');

ylabel('Magnitude of Signal X1')

figure

plot(omega2, abs(X2));

xlabel('omega');

ylabel('Magnitude of Signal X2')

figure

plot(omega2, abs(X3));

xlabel('omega');

ylabel('Magnitude of Signal X3')

figure

plot(omega2, abs(X4));

xlabel('omega');

ylabel('Magnitude of Signal X4')

figure

plot(omega2, abs(X5));

xlabel('omega');

ylabel('Magnitude of Signal X5')
```

Comments:

Looking at the graph of the magnitude of the fourier transform of the 5-digit number, it is not easy to detect the frequencies of each digit due to the overlap between the cosine signals. To get a better understanding of the frequency content, the signal of each digit was plotted separately.

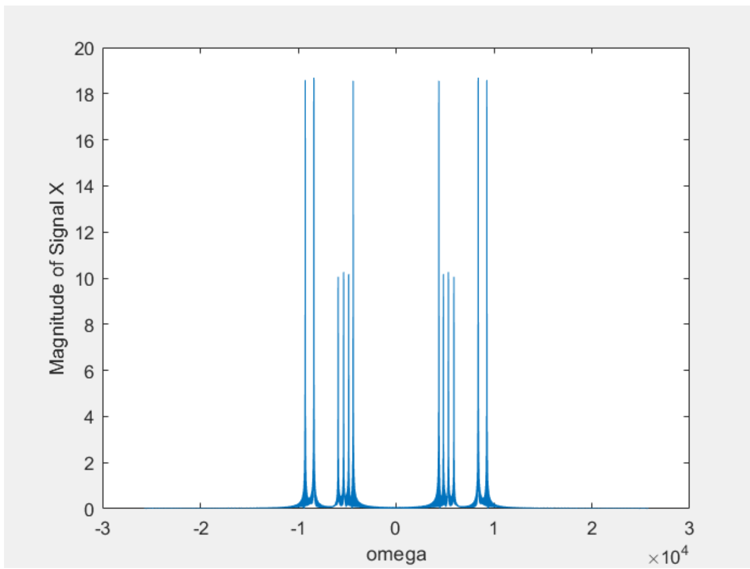

Figure 1: The Fourier Transform of the Signal of the Number 80632

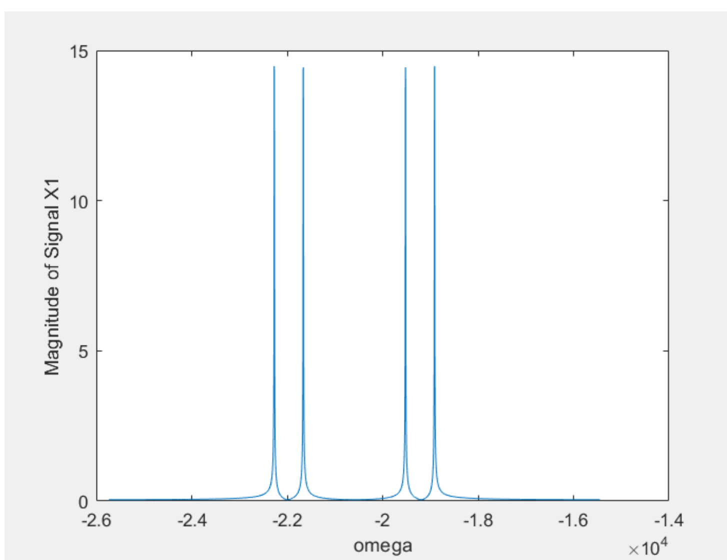

Figure 2: The FT of the Signal of the Digit 8

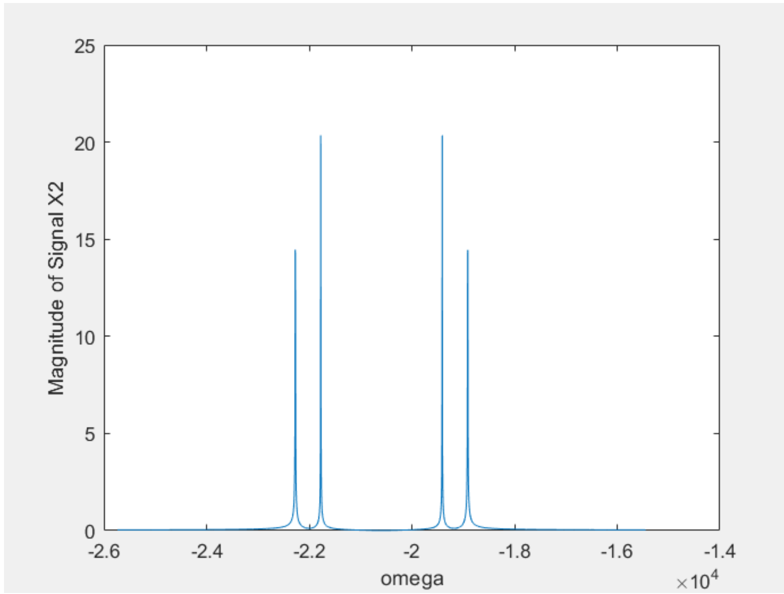

Figure 3: The FT of the Signal of the Digit 0

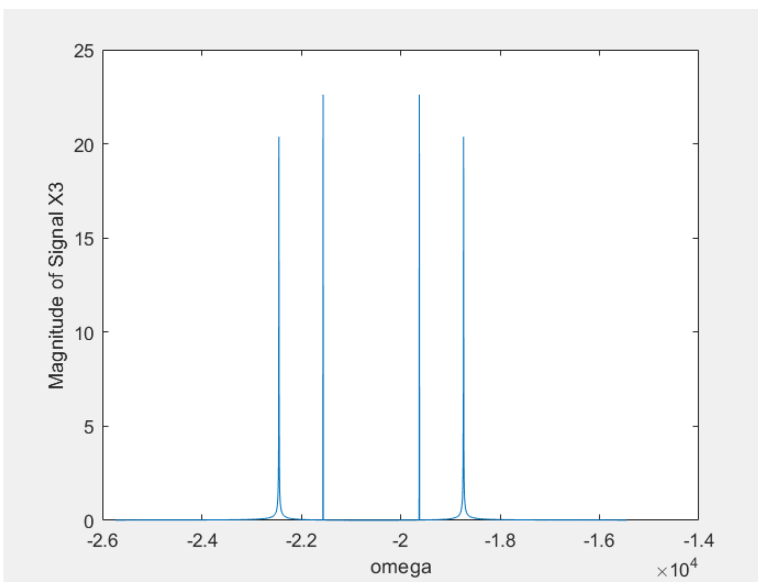

Figure 4: The FT of the Signal of the Digit 6

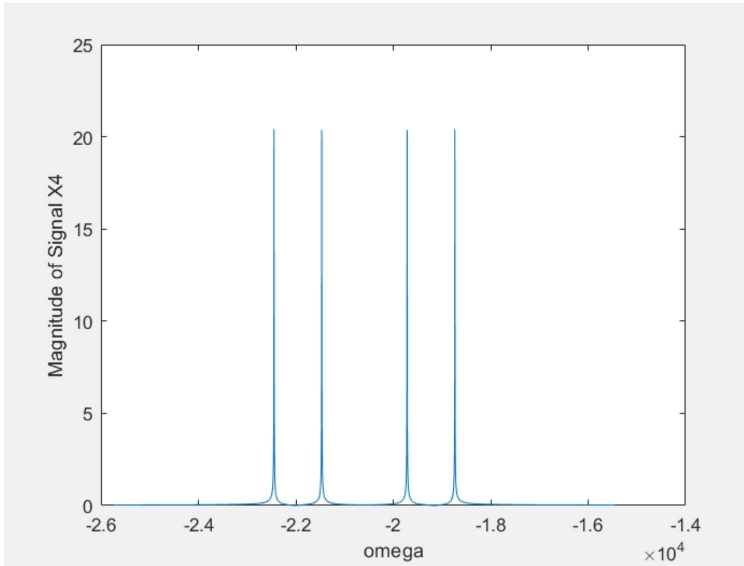

Figure 5: The FT of the Signal of the Digit 3

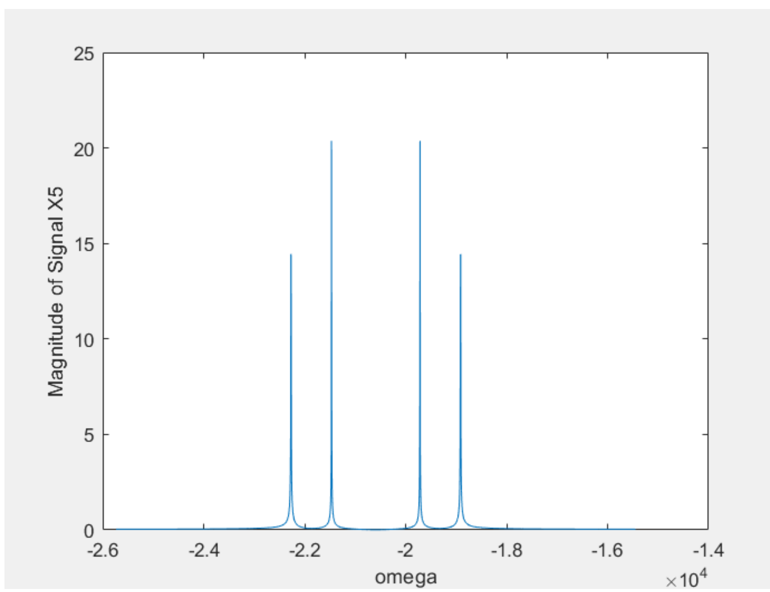

Figure 6: The FT of the Signal of the Digit 2

Part 2:

-The answers of the analytical questions are in the separate pdf mentioned earlier

Codes

```
%% part 2

% recording = audiorecorder(8192, 8, 1);

% duration = 10;

% disp("Start")

% recordblocking(recording,duration);

% disp("Stop")

play(recording);

x = getaudiodata(recording);

x = x.';

figure

plot(x);

xlabel('t')

ylabel('x(t)')

title("Pure Speech Signal")

echo1 = 0.65*[zeros(1,8192*0.25) x zeros(1,8192*3)];

echo2 = 0.50*[zeros(1,8192*0.75) x zeros(1,8192*2.5)];

echo3 = 0.30*[zeros(1,8192*1) x zeros(1,8192*2.25)];

echo4 = 0.22*[zeros(1,8192*1.25) x zeros(1,8192*2)];
```

```
echo5 = 0.15*[zeros(1,8192*2) x zeros(1,8192*1.25)];

echo6 = 0.1*[zeros(1,8192*3.25) x];

x = [x zeros(1,8192*3.25)];

y = x + echo1 + echo2 + echo3 + echo4 + echo5 + echo6;

y_resized = y(1:98304);

% soundsc(y)

% soundsc(y_resized)

figure

plot(y_resized)

xlabel('t');

ylabel('y(t)')

title("Echoed Speech Signal")

Y = FT(y_resized);        %Fourier Transform of Echoed Speech

t = 0:1/8192:12-1/8192;

omega = linspace(-8192*pi,8192*pi,98304);

omega = omega(1:98304);

H = 1 + 0.65*exp(-1j*omega*0.25) + 0.50*exp(-1j*omega*0.75) + 0.30*exp(-... %Frequency
Response

1j*omega*1) + 0.22*exp(-1j*omega*1.25) + 0.15*exp(-1j*omega*2) + 0.1*exp(-...

1j*omega*3.25);

H_magnitude = abs(H);
```

```
figure
plot(omega,H_magnitude)
xlabel('Omega')
ylabel('Magnitude of H(jw)')
title("Magnitude of Frequency Response")
h = IFT(H);
figure
plot(t,h)
xlabel('t')
ylabel('h(t)')
title("Impulse Response Function")
X = Y./H;
xe = IFT(X);
soundsc(xe)
figure
plot(xe)
xlabel('t')
ylabel('x_e')
title("Filtered Speech Signal (Estimated X)")
```

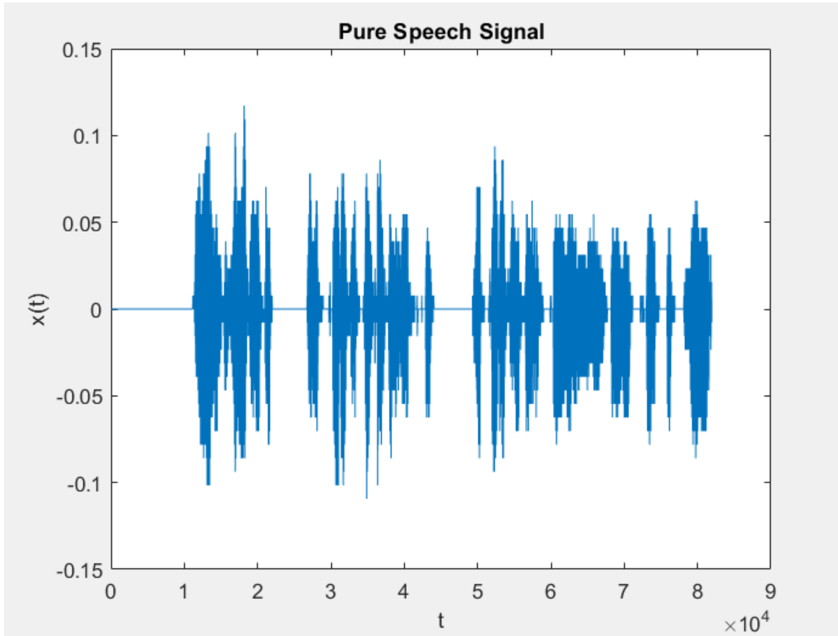

Figure 7: Speech Signal

Recording: “My name is Mohammed Abed, I’m a student at Bilkent University, and my Student ID is 21903608”

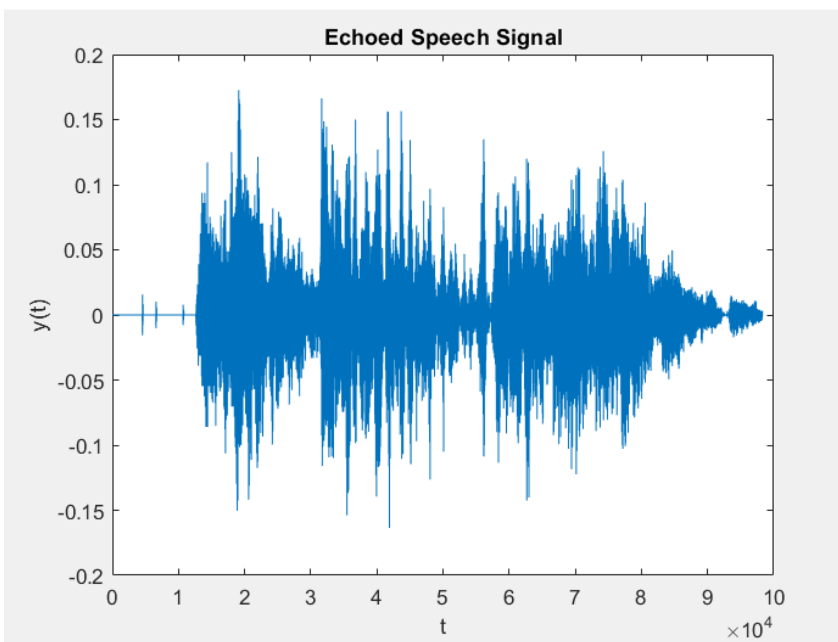

Figure 8: Echoed Version of the Speech Signal

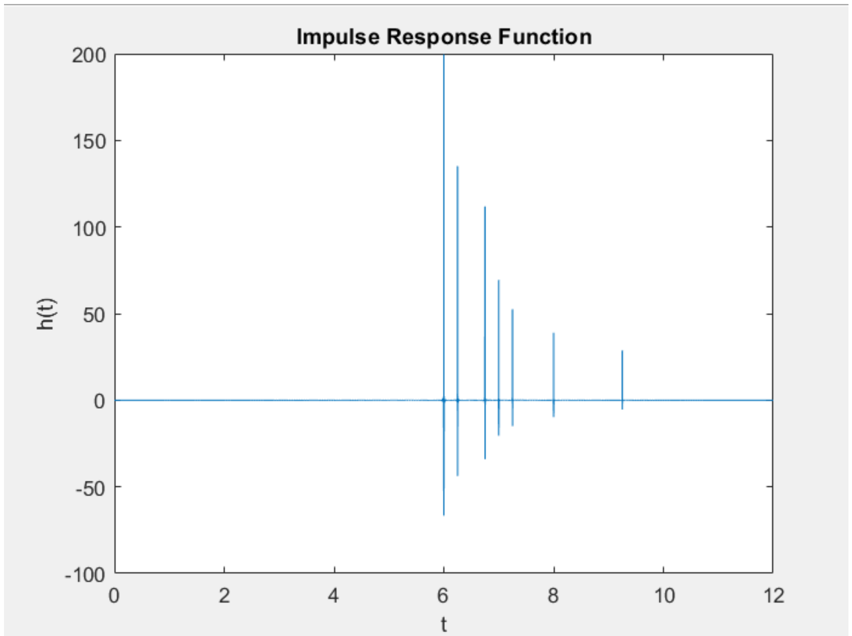

Figure 9: The Impulse Response Used to Process the Speech Signal

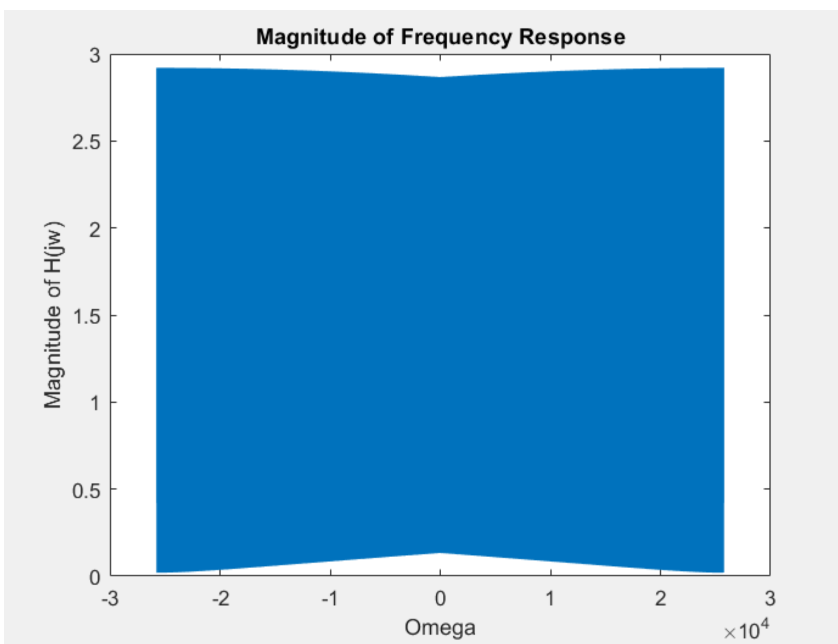

Figure 10: Frequency Response

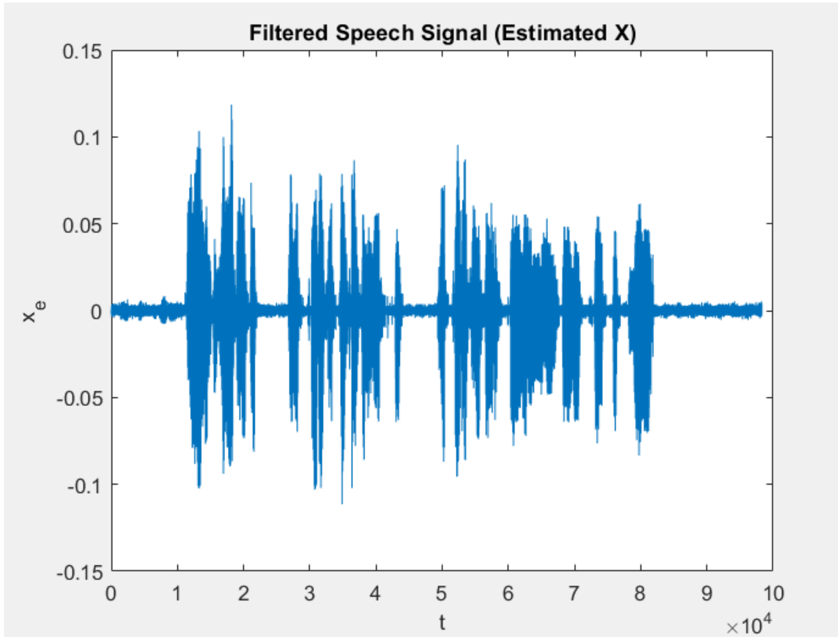

Figure 11: Retrieved (Estimated) Speech Signal

Comments:

The speech signal was not completely recovered as it was estimated; there was very minor echo when I listened to the retrieved version of the echoed speech.