EE 321 Lab 3

Part 1

DTMF Transmitter Code

```
Number = [2 0 1 6 4 5 7 5 1 2];
x = DTMFT(Number)
soundsc(x)
```

```
function x = DTMFT(Number)
frequency = [941 1336; 697 1209; 697 1336;
              697 1477; 770 1209; 770 1336;
              770 1477; 852 1209; 852 1336;
                         852 1477];
nb_size = size(Number);
character = [0 1 2 3 4 5 6 7 8 9];
duration = 0:1/8192:0.25;
x = zeros(1, nb_size(1) * 2049);
i = 0;
    for digit = Number
        index = find(character == digit);
        if digit <= 9 && digit >= 0
             freq_of_digit = frequency(index, :);
signal = cos(2 * pi * freq_of_digit(1) * (duration)) + (cos(2 * pi *
freq_of_digit(2) * (duration)));
             i = i + 1;
             x = horzcat(x, signal);
        end
    end
end
```

The sound of the signal was just like dialing numbers on the phone. It is familiar voice. At here I just put some random numbers. But the sound of the every number is like dialing a number on the phone.

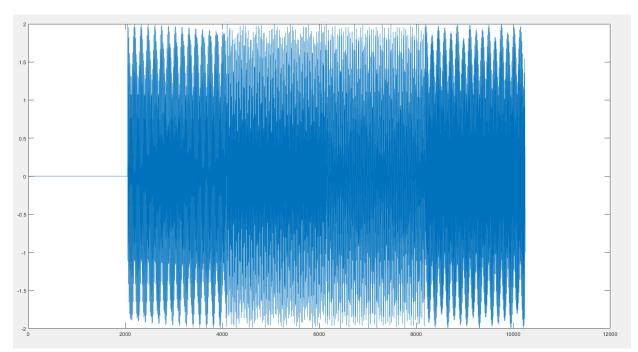


Figure 1 – Graph of the DTMF Transmitter

DTMF Receiver Code

```
function output = FT(input)
M=size(input,2);
t=exp(j*pi*(M-1)/M*[0:1:M-1]);
output=exp(-j*pi*(M-1)^2/(2*M))*t.*1/(M)^0.5.*fft(input.*t);
```



```
Number = [5 7 0 2 1];
x = DTMFTRA(Number);
soundsc(x)

X = FT(x);
w = linspace(-10240, 10240, 12294);
w = w(1:12294);
plot(w,real(X))
```

2)
$$x(u) = \exp(j2\pi [+1])$$
 $X(u) = \int_{0}^{\infty} \exp(j2\pi [+1) \exp(-jut)) dt = \int_{-\infty}^{\infty} e^{j2\pi [+1-jut]} dt = \int_{-\infty}^{\infty} e^{j1[\pi [+-uu]} dt = 2\pi \int (2\pi [+-uu]) dt$

Figure 2- Answers of the FFTs

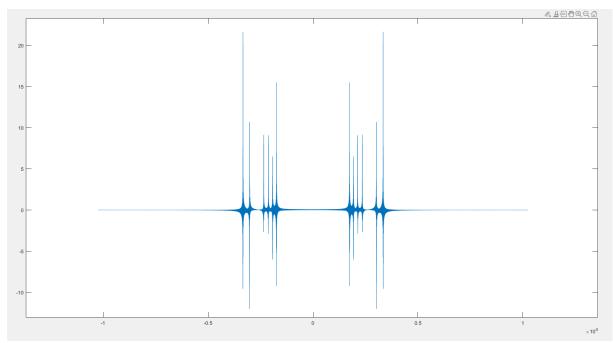
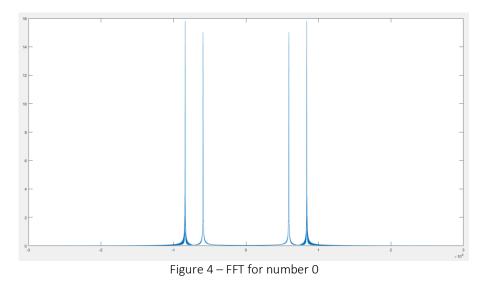


Figure 3 – Output of the FFT

From only looking this figure, we cannot understand which numbers are dialed. Also, the sequence of the numbers cannot be detected. 1-5 or 2-4 are not differentiated. In the figure 4 the numerical values represent the angular frequency not normal frequency. Therefore, these values should be divided 2π before comparing the frequency of each number.



From figure 4, it can be seen that 667 and 1336 Hz frequencies are effective only. The amplitude of the frequency should be the same but there is a small difference which can be observed with naked eyes. Maybe it is about the sampling rate so the peak points of the frequencies cannot be detected as same.

Also, second method is shown every number one by one. But at the first method, the sample cannot be detected because of the frequency domain does not show the sequence

Part 2

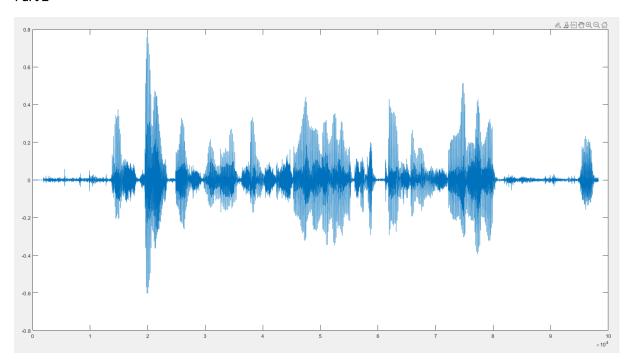


Figure 5- Voice signals in time domain

When it is listened, the signals sound like slower than its normal version. It is probably about sample ratio of the audioread() function.

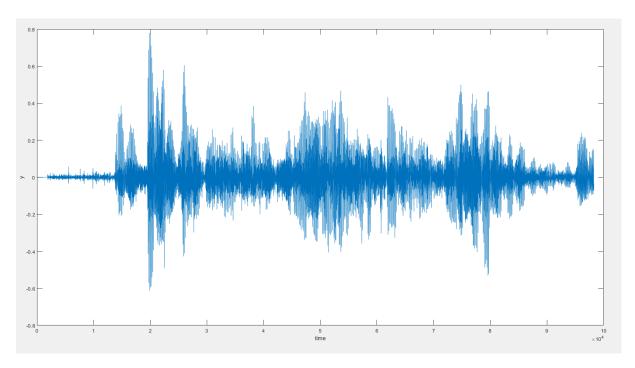


Figure 6- y(t) signal over time

y(t) signal almost the same as the original voice recording but it also has some echoes more than the original one. Because of the extra signal which has time shifted and smaller amplitude we added to original signal, echo effect is created.

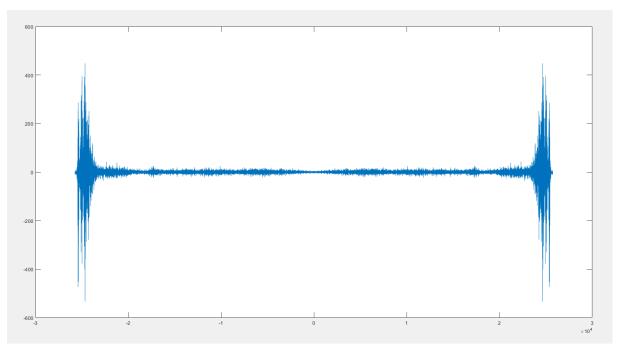


Figure 7- The FT result of the y(t)

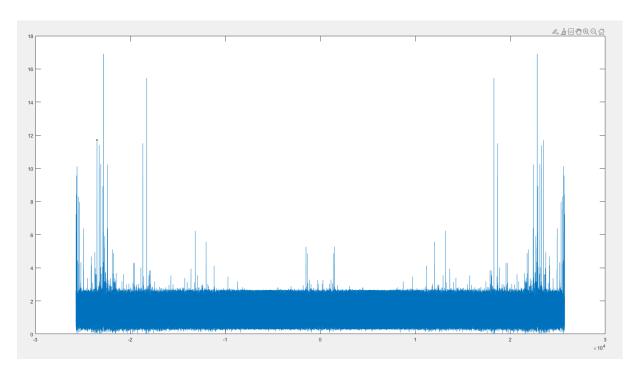


Figure 8- The graph of H(w)

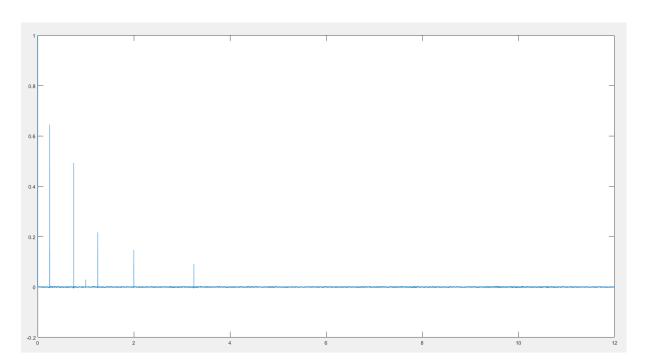


Figure 9- The graph of h(t)

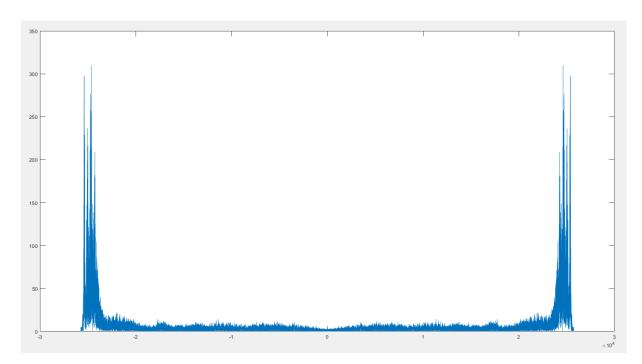


Figure 10- The graph of Xe

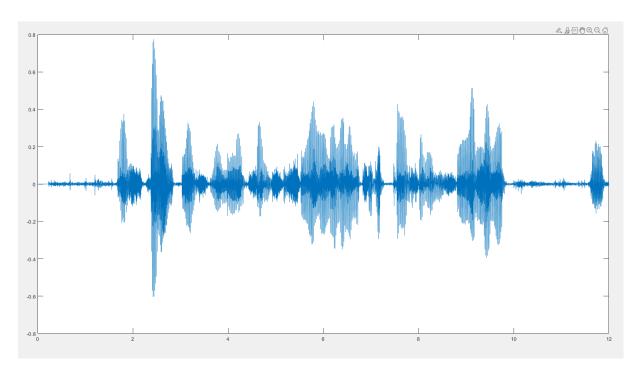


Figure 11- The xe graph

When I listened the Xe sound, I did not realize any difference between the original audio because, respectively we add echoes to the sound and we clear the effect of the echoes by detected the Fourier domain of the echoes. Then, we subtracted the echoes from the sound signal and we reached to original sound.

Matlab Code

```
x = audioread('ses.opus')
x_10 = x(1:98304)
t = 0:1/8192:12-1/8192;
% plot(x)
% xlabel('time')
% ylabel('voice signal amplitude')
x1=circshift(x_10, 2048);
x1(1:2047)=zeros;
x2=circshift(x_10, 6144);
x2(1:6143)=zeros;
x3=circshift(x_10, 8192);
x3(1:8191)=zeros;
x4=circshift(x_10, 10240);
x4(1:10239)=zeros;
x5=circshift(x_10, 16384);
x5(1:16383)=zeros;
x6=circshift(x_10, 26624);
x6(1:26623)=zeros;
y = x_10 + 0.65 * x1 + 0.5 * x2 + 0.03 * x3 + 0.22 * x4 + 0.15 * x5 + 0.1 * x6;
% plot(y)
% xlabel('time')
% ylabel('y')
omega=linspace(-8192*pi,8192*pi,98305);
omega=omega(1:98304);
Y = FT(y);
X_{10} = FT(x_{10});
plot(omega, X_10)
H = Y./X_10;
h = IFT(H);
plot(omeag, abs(h))
Xe = Y./H;
xe = IFT(Xe);
```

```
plot(t, xe)
soundsc(xe)
```

```
function output = IFT(input)
M = size(input,2);

t = exp(-j*pi*(M-1)/M*[0:1:M-1]);
output=real(exp(j*pi*(M-1)^2/(2*M))*t.*(M)^0.5.*ifft(input.*t));
```

```
function output = FT(input)
M = size(input,2);

t = exp(j*pi*(M-1)/M*[0:1:M-1]);
output=exp(-j*pi*(M-1)^2/(2*M))*t.*1/(M)^0.5.*fft(input.*t);
```

Answer for the part 2 questions

a)
$$y(t) = x(t) + \sum_{i=1}^{M} A_i \cdot x(t-t_0) = \int_{-\infty}^{\infty} x(t) h(t-t) dt = x(t) * h(t)$$

Since it is a Sunction which includes time shifted version of the original version.

It can be inferred that if $x(t) = S(t)$, $y(t) = h(t)$ which is impulse response to impulse function.

$$h(t) = S(t) + \sum_{i=1}^{M} A_i \cdot S(t-t_0)$$

$$h(t) = S(t) + \sum_{i=1}^{M} A_i \cdot S(t-t_0)$$

$$h(t) = S(t) + \sum_{i=1}^{M} A_i \cdot S(t-t_0)$$

$$h(t) = h(t) + \sum_{i=1}^{M} A_i \cdot S(t-t_0)$$

Figure 12- Answers of the questions at part 2