Part 2

In order to record my voice I used the MATLAB's built in function "audiorecorder" in the following MATLAB code:

```
clc
clear all
Fs = 8192;
record=audiorecorder(Fs,16,1);
disp('started')
recordblocking(record,12);
disp('ended');
x=transpose(getaudiodata(record));
```

My speech was:

"Hala kapkara su
Hep aynı derinlikte
Gökyüzü kadar soğuk
Sesin duyulmaz olduğunda
Yüreğim kadar hareketsiz
Cehennemde donacağım"

Answers to the questions are below:

$$y(t) = x(t) + \sum_{i=1}^{m} A_i x(t-t_i)$$

a) $h(t)$ is the impulse response of the system. So $x(t) = \delta(t)$

$$h(t) = \delta(t) + \sum_{i=1}^{m} A_i \delta(t-t_i)$$

b) $F\{\delta(t)\} = 1$ and time shift in time domain is multiplying complex exponential in frequency domain. FT is linear. Therefore, using this properties FT of $h(t)$ can be found as:

$$H(\omega) = 1 + \sum_{i=1}^{m} A_i e^{-j\omega t_i}$$

c) Since $y(t) = x(t) * h(t)$ means $Y(\omega) = X(\omega) \cdot H(\omega)$ in frequency domain. i.e. convolution in time domain corresponds to multiplication in freq. domain.

d) So from the result in part c $X(\omega) = \frac{Y(\omega)}{H(\omega)}$, this means if we know $Y(\omega)$ and $H(\omega)$ we can find $X(\omega)$.

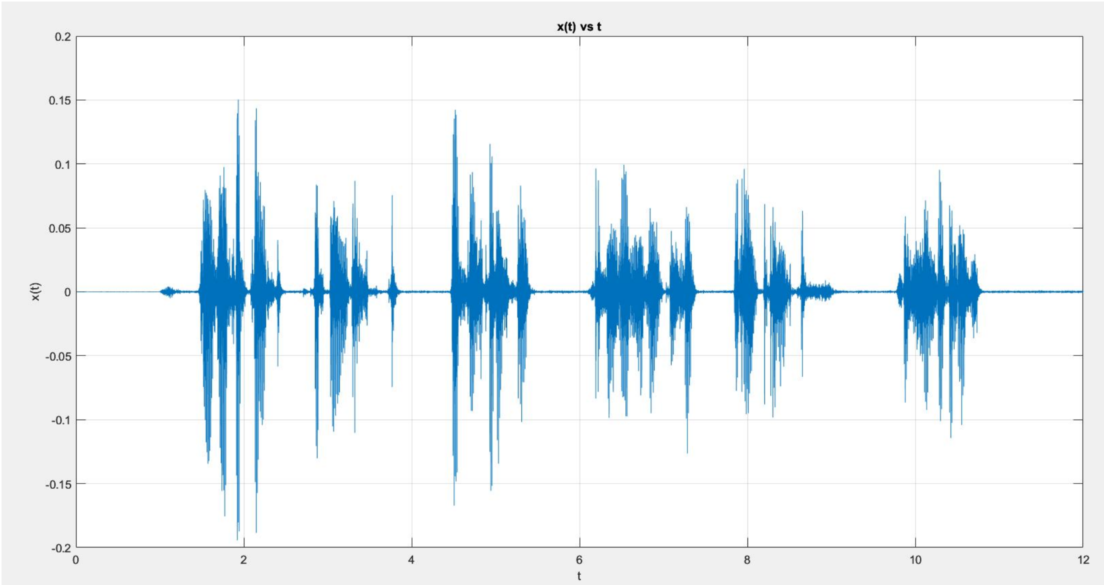

Figure 4: Plot of array of my speech x(t) vs t

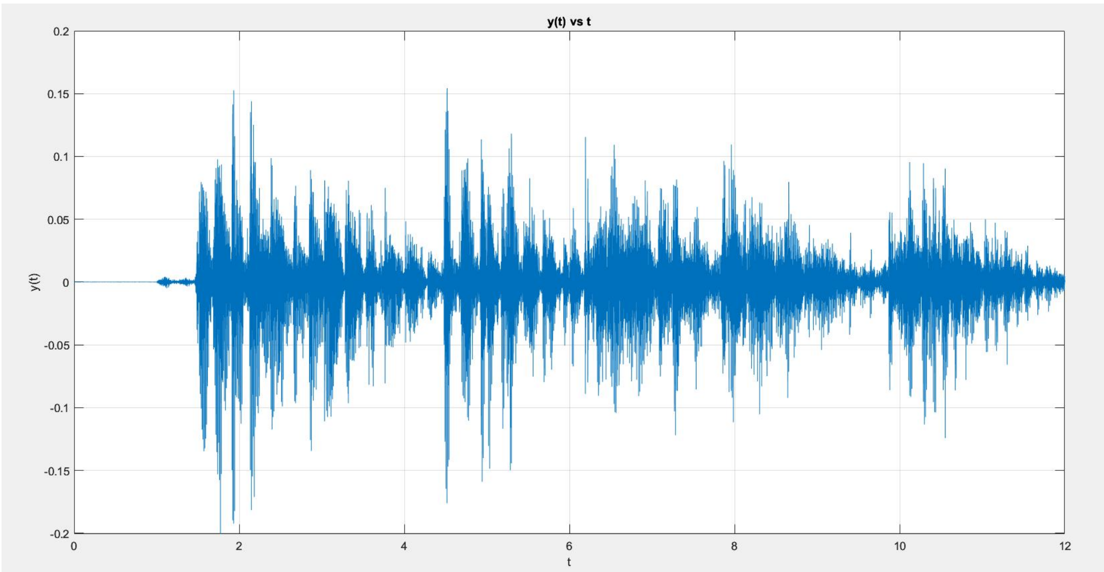

Figure 5: Plot of array of my speech with echo y(t) vs t

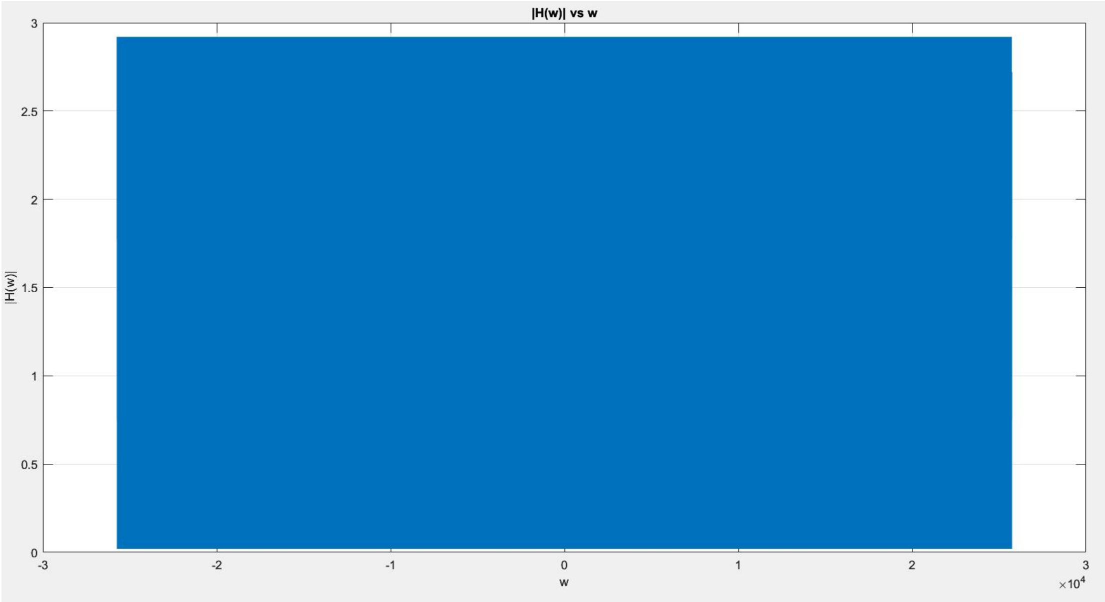

Figure 6: Magnitude of frequency response computed over omega

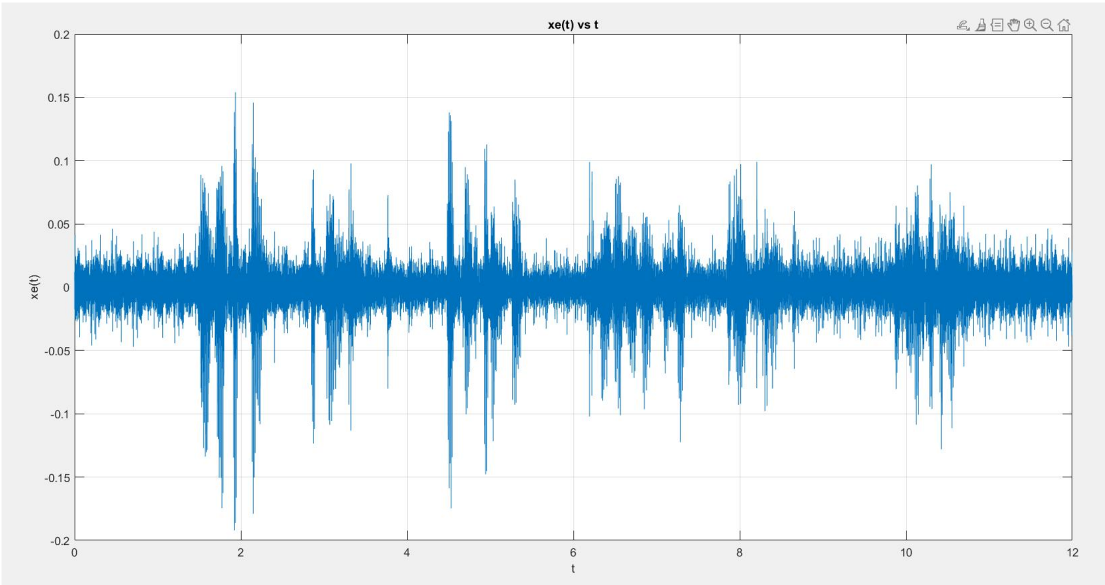

Figure 7: Plot of array of my extracted speech $x_e(t)$ vs t

When y(t) is listened, it was the echoed version of my speech, it was hard to understand. When $x_e(t)$ was listened, even though there were some little noise in the background after the processes, I could understand what I said and the echo and noise were mostly gone. If we compare *Figures 5&7* the general layout of the signal is very similar, also the white noise is also visible in the plot.