Mini project: *Z transform and sampling process* **Due Date: 11/12/19**

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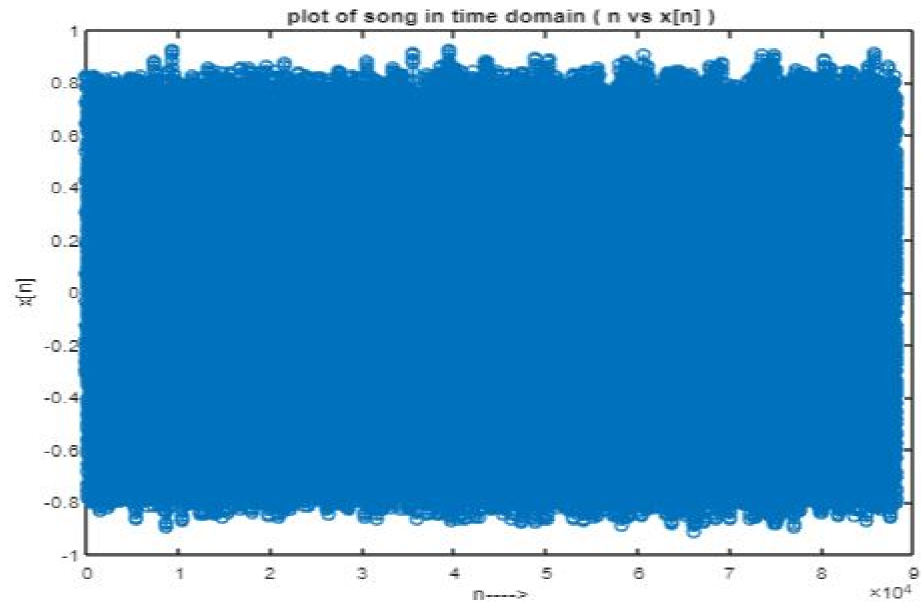
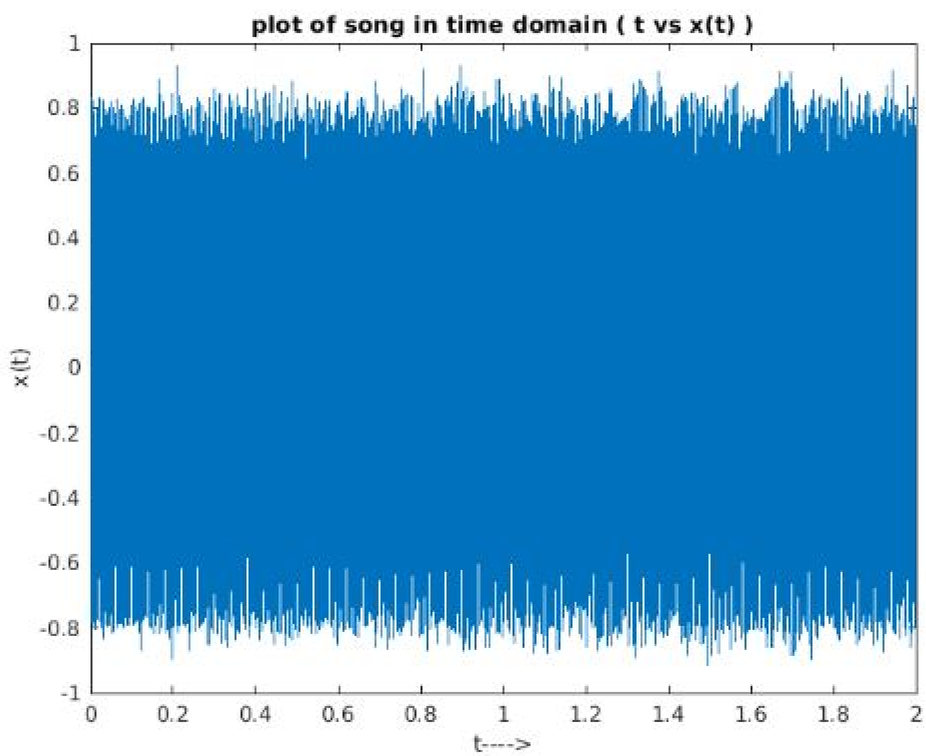
a. By listening given music we can say that, there is strong beep sound in the signal because of which we are not able to hear song properly.

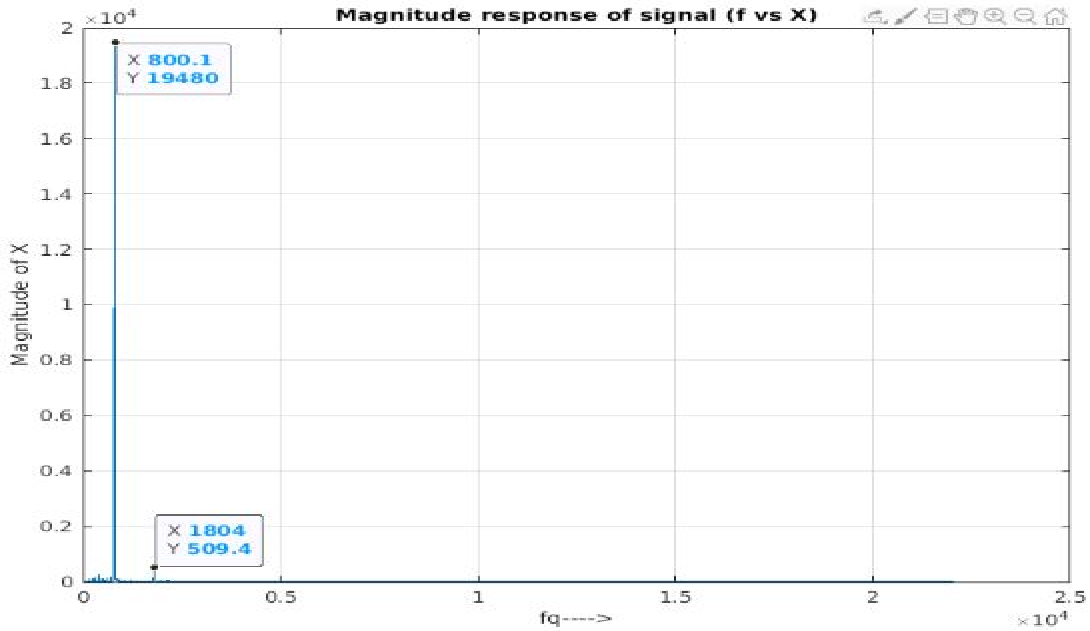
b. Time domain plot

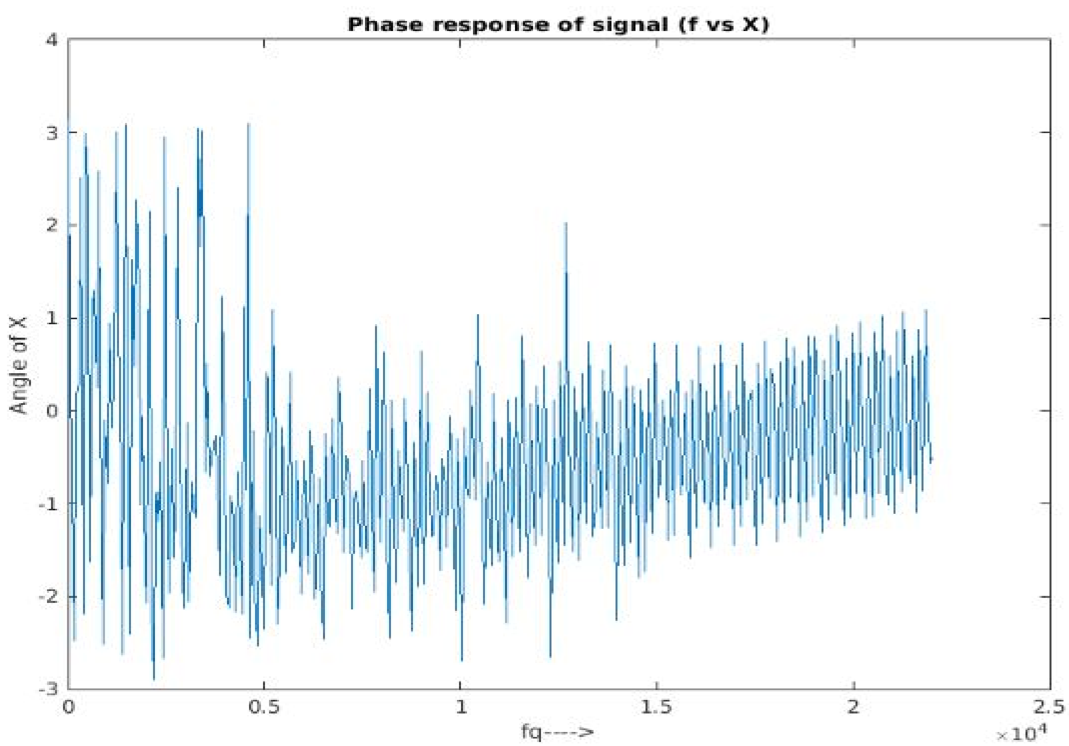


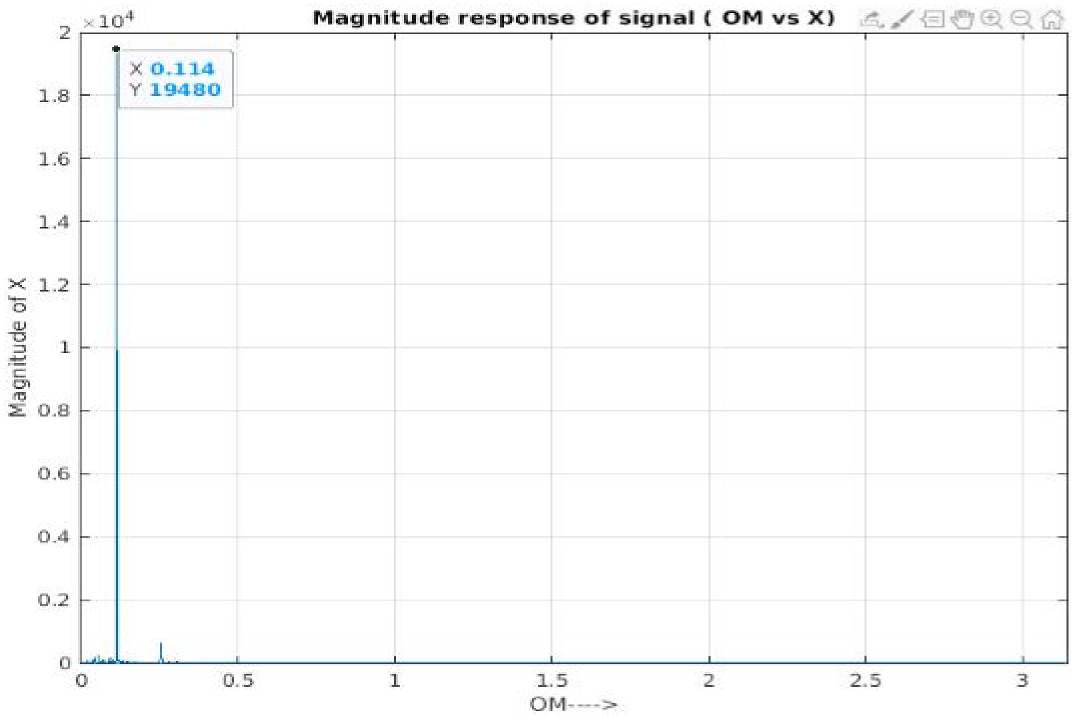


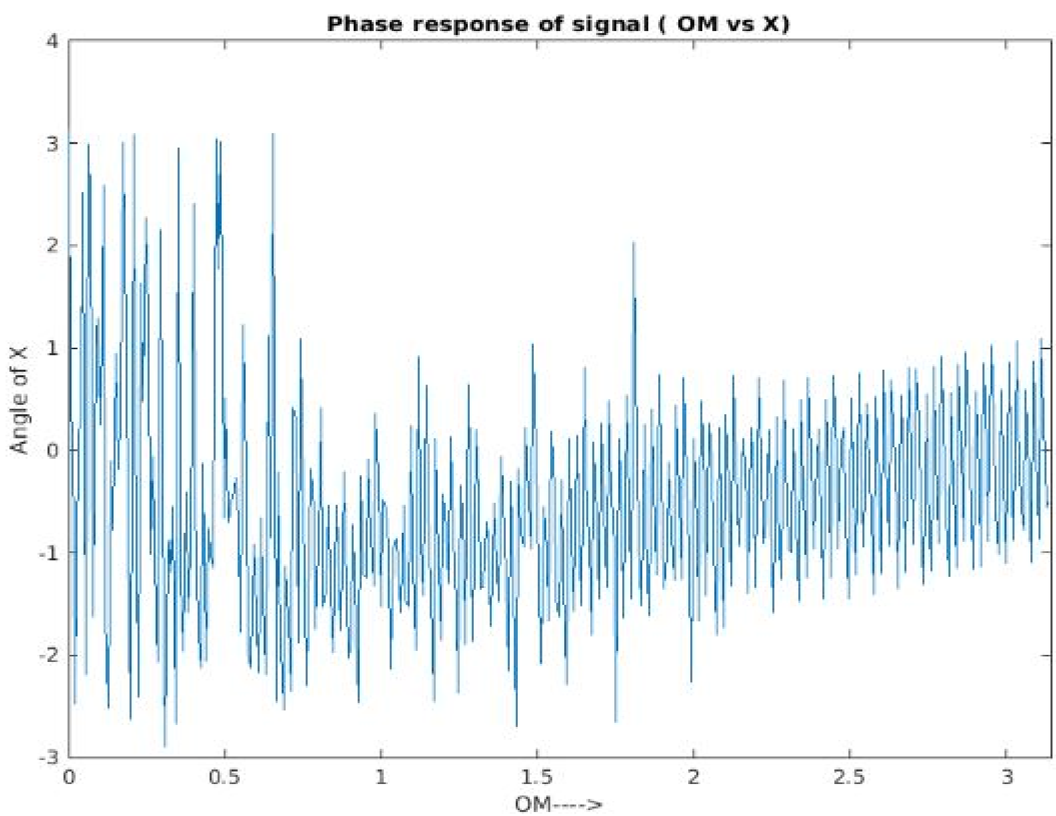
Ans : Fs is 44100 HZ and length is 2 Sec.

c. DTFT plot


Magnitude response of signal (f vs X)
×10^4
X 800.1
Y 19480
X 1804
Y 509.4
Magnitude of X
fq---->
×10^4


Phase response of signal (f vs X)
Angle of X
fq---->
×10^4


Magnitude response of signal (OM vs X)
×10^4
X 0.114
Y 19480
Magnitude of X
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.5
1
1.5
2
2.5
3
OM---->


Phase response of signal (OM vs X)
Angle of X
4
3
2
1
0
-1
-2
-3
0
0.5
1
1.5
2
2.5
3
OM---->

d. There are two disturbing frequencies.

1. f1 = 800.1 Hz
2. f2 = 1804 Hz

Equivalent digital frequencies are

3. OM = O.114 radian
4. OM = 0.256 radian

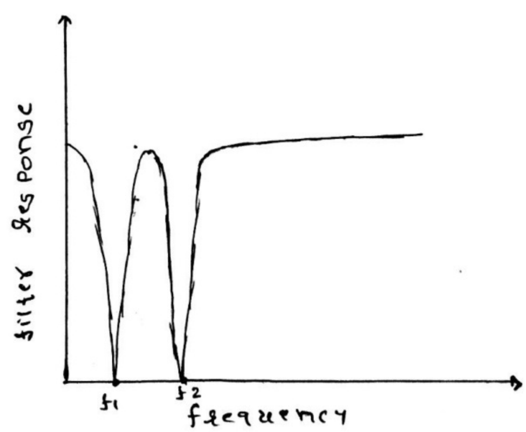
e. Required IIR notch filter to remove disturbing frequencies.

We have to design IIR notch filter to remove disturbing frequencies.

f_1 = 800.1 Hz and

f_2 = 1804 Hz

→ Response of filter should be,



f. Defined IIR notch filter

$$\omega_1 = \frac{800.1}{22050}\pi \quad , \quad \omega_2 = \frac{1804}{22050}\pi$$

→ let numerator and denominator coefficients

$nm = 0.999999$

$dm = 0.99$

→ H(z) can be defined as

$$H(z) = \frac{N(z)}{D(z)}$$

$$H(z) = \frac{(z - nm\,e^{j\omega_1})(z - nm\,e^{-j\omega_1})(z - nm\,e^{j\omega_2})(z - nm\,e^{-j\omega_2})}{(z - dm\,e^{j\omega_1})(z - dm\,e^{-j\omega_1})(z - dm\,e^{j\omega_2})(z - dm\,e^{-j\omega_2})}$$

→ If we simplify the terms,

$nm\,e^{j\omega_1} = 0.9935 + 0.1137j$

$nm\,e^{-j\omega_1} = 0.9935 - 0.1137j$

$nm\,e^{j\omega_2} = 0.9671 + 0.2542j$

$nm\,e^{-j\omega_2} = 0.9671 - 0.2542j$

$dm\,e^{j\omega_1} = 0.9836 + 0.1126j$

$dm\,e^{-j\omega_1} = 0.9836 - 0.1126j$

$dm\,e^{j\omega_2} = 0.9575 + 0.2517j$

$dm\,e^{-j\omega_2} = 0.9575 - 0.2517j$

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g. Defining filter coefficient from transfer function

→ Transfer function can be given as

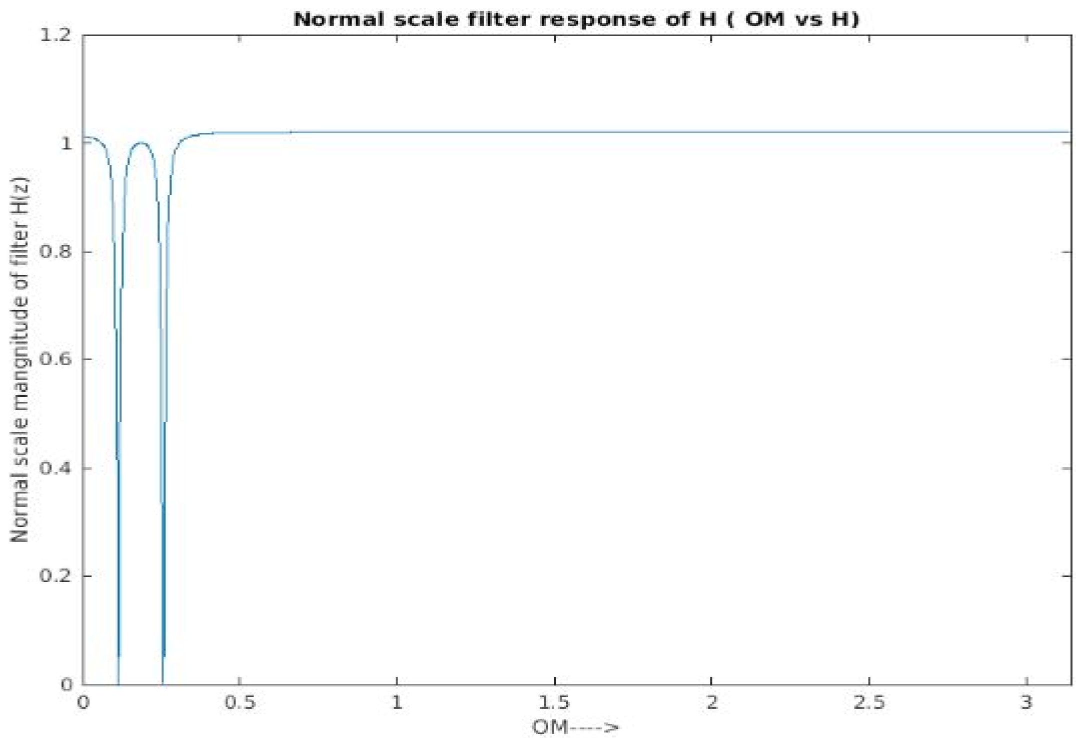
$$H(z) = \frac{z^4 - 3.921z^3 + 5.843z^2 - 3.921z + 1}{z^4 - 3.882z^3 + 5.727z^2 - 3.905z + 0.9606}$$

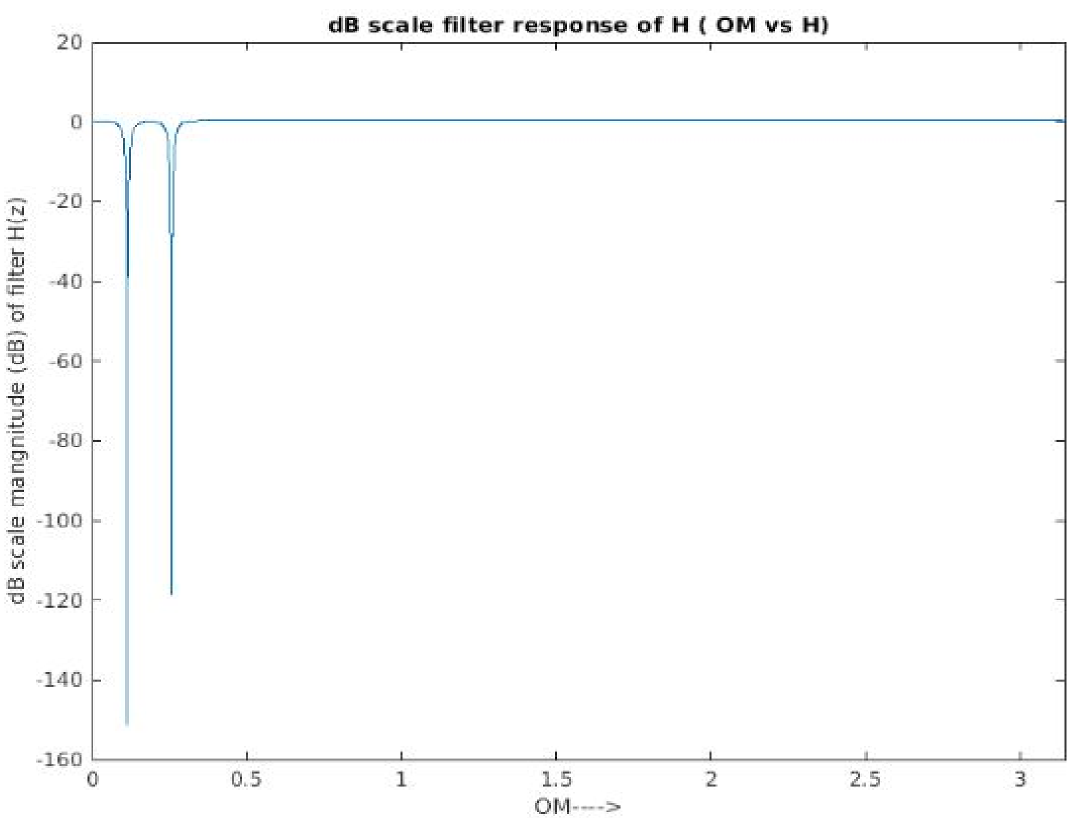
→ filter coefficient will be

num = [1 ; 3.921 ; 5.843 ; -3.921 ; 1]

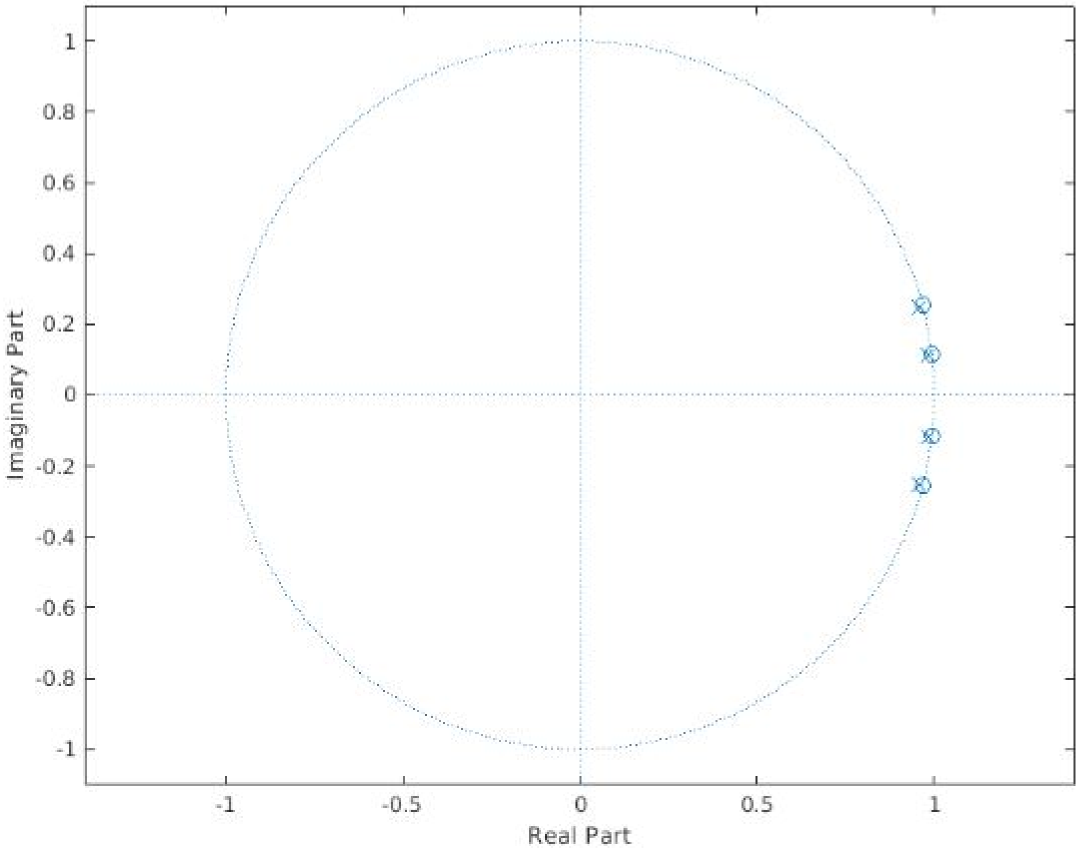
den. [1 -3.882 5.727 -3.805 0.9606]

h. **Filter response in normal and dB scale.**





i. Pole Zero plot of the defined filter.

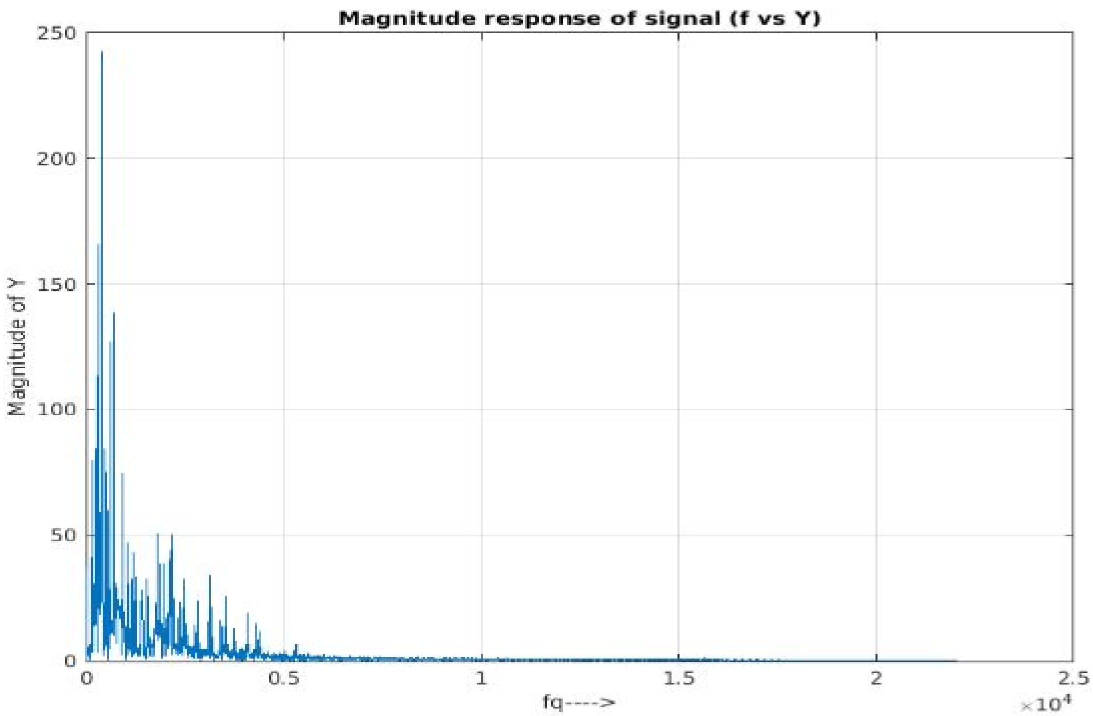


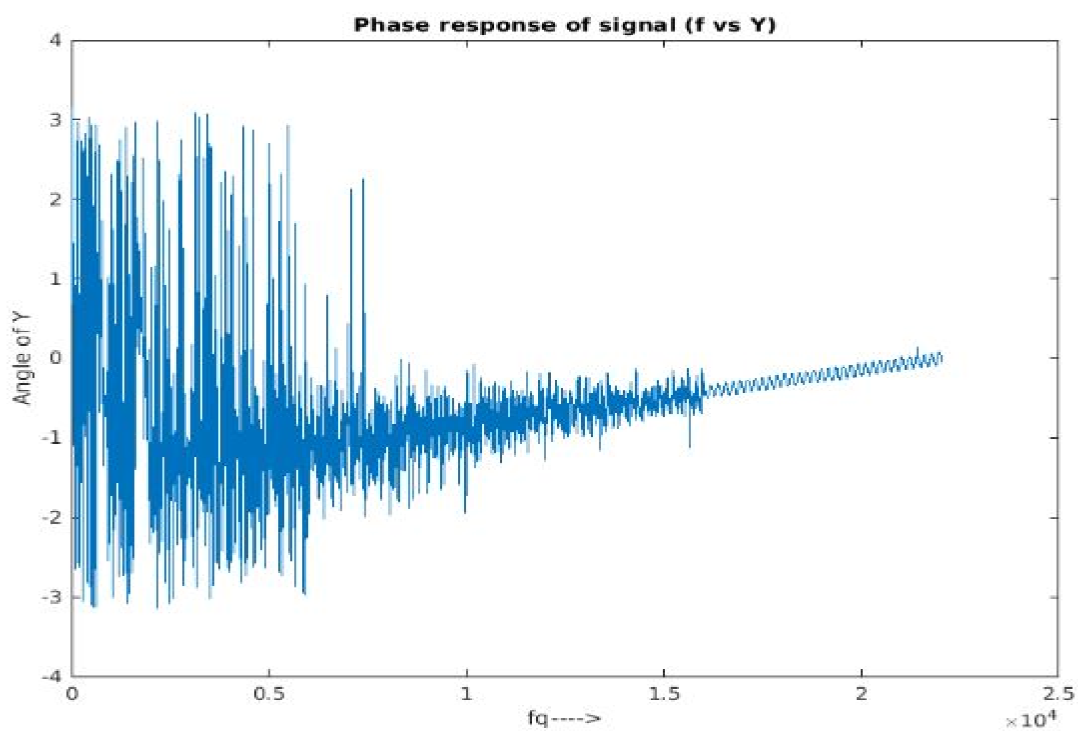
j. Filter the signal with defined filter
k. Listen to the filtered song.

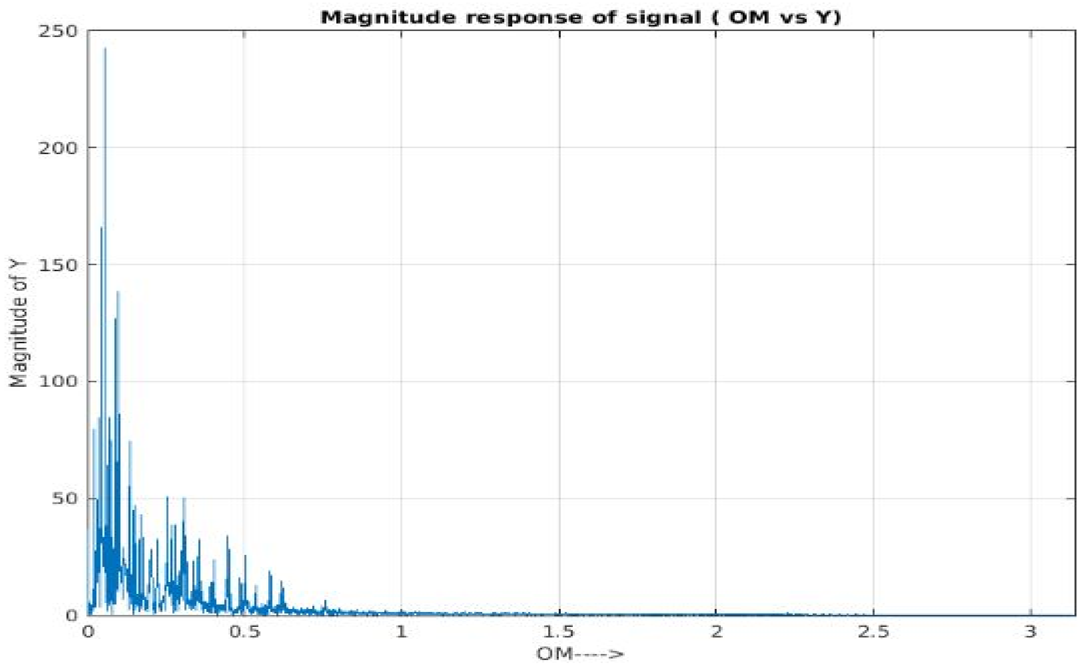
```
y = filter(num, den, x);
sound(x, fs)        %original signal
pause(5);
sound(y, fs)        %filtered signal
```

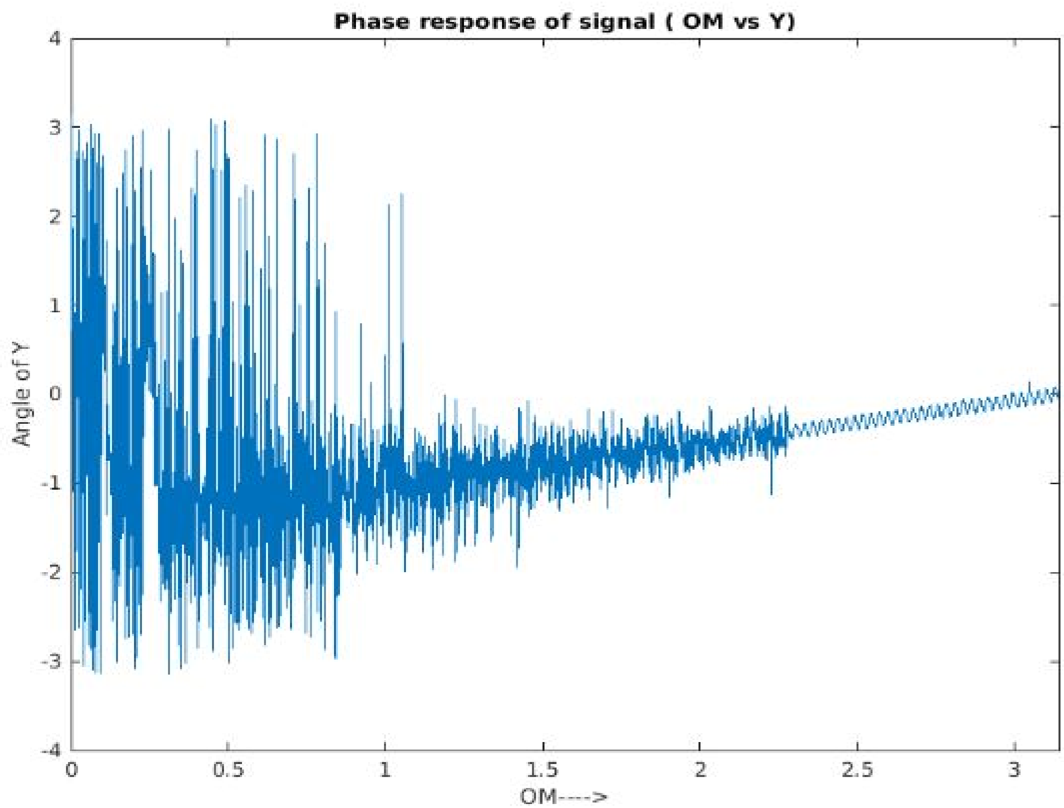
matlab code for j & k

I. Frequency response of filtered signal






Magnitude response of signal (OM vs Y)
Magnitude of Y
OM---->
250
200
150
100
50
0
0
0.5
1
1.5
2
2.5
3


Phase response of signal (OM vs Y)
Angle of Y
OM---->
4
3
2
1
0
-1
-2
-3
-4
0
0.5
1
1.5
2
2.5
3

m. After filtering, amplitude of beep sound which was attenuating signal earlier was decreased. We were able to hear the actual song with negligible disturbance.

n. Observation

1. Filtered output was nor as clear as expected. To remove disturbing frequencies completely we need to design ideal notch filter. To design ideal notch filter we will have to have poles and zeros of the filter on the unit circle but it will make filter unstable.
2. Another reason is we cannot remove all the disturbing frequency as they are placed very nearer to each other in frequency response. We can remove most disturbing frequencies not all. That is why we cannot get completely clear output.

o. Observation

1. We had music file containing beep sound. Because of that beep sound we were not able to hear the song completely. Using sampling process and Z transform method we can design the IIR notch filter to remove disturbing frequencies and to enjoy the song. In the same manner with the help of Z transform and sampling theorem we can design any filter to remove disturbing frequency and have original signal/music file back.
2. Also in communication network we can use Z transform and sampling process to filter out the attenuated signal.