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Department of Information Engineering - DEI

ICT (Internet and Multimedia)

Digital Signal Processing Project

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Outline:

1. Introduction.
2. Compute the spectrum of the input signal.
3. Carriers:
   3.1. Find an estimate of the amplitudes A1 and A2 of the carriers.
   3.2. Find the frequencies f1 and f2 of the sinusoidal carriers.
   3.3. Extracts the two carriers by filtering the input signal.
      - Design an FIR second order filter
      - Use "ellip" function
4. Demodulate the two information signals.
5. Design filter:
   5.1. Design the filters so that the demodulated signals do not present audible distortions.
   5.2. Report the design parameters of the filters and plot their frequency responses.
6. Write the file containing the two output signals.

1. Introduction:

There is an audio file with sample frequency of 48^{Kh} which consist of two real audio information signals. The procedure is to first of all demodulate the pass-band signals to base-band by using the two sinusoidal carriers of the audio file. Second of all filter the demodulated signal and extract the real audio information. Finally store the signals and combine them in channel.

2. Compute the spectrum of the input signal.

The input signal is:

$y(nT) = (x_1(nT) + A1)\cos(2\pi f_1 nT) + (x_2(nT) + A2)\cos(2\pi f_2 nT)$

After reading the file by calculating the length of the signal and compute the spectrum of the signal by using "***fft***" function the result is stored in y.

3. Carriers

3.1 *Find an estimate of the amplitudes A_1 and A_2 of the carriers*

Due to the magnitude of the carriers which are greater than the signals, the real part of y is sorted and very first two component would show the estimate amplitude of A_1 and A_2 respectively. Note that to multiply the maximum value of y to 2/n (which n refer to the number of samples). Regarding this Procedure the values of A_1 and A_2 are the following:

$A_1 = 0.0497$

$A_2 = 0.0398$

3.2 *Find the frequencies f1 and f2 of the sinusoidal carriers*

To find the frequencies of the carriers, we should find the "n" which is pointed to the A_1 and A_2. Then we should multiply them by sample frequency (fs) and divide by length of the signal (n) to convert them in frequency domain. The results are the following:

$f_1 = 5.9^{KHz}$

$f_2 = 17.1^{KHz}$

3.3 *Extracts the two carriers by filtering the input signal*

To extract the carriers after finding their frequencies, we should design an FIR filter. First of all we designed a second order FIR filter by finding the coefficients of the filter and then we also use "***ellip***" function to generate the FIR filter. The coefficients of second order FIR filter are the following: (Note: the sign of "a_i" coefficients in Matlab© are opposite of the notes in class.)

- **FIR second order coefficients:**

To extract the f_2

Nominator coefficient Vector: a = [1 -1.43212854611753 0.999345608622564]

Denominator coefficient vector: b = [0.000456704979713089 0 -0.000456704979713089]

To extract the f_1

Nominator coefficient Vector: c = [1 1.23779004285708 0.999345608622564]

Denominator coefficient vector: d = [0.000513986835039530 0 -0.000513986835039530]

The normalized frequency response of the second order FIR ([figure1], [figure 2]) and the filter which is designed by "***ellip***" function ([figure3], [figure 4]) are shown respectively.

4. Demodulate the two information signals

After extracting the carriers now we can demodulate the input signal to the Base-Band. We should use a gain to compensate the attenuation of signal, we consider the G = 50 to hear the output signal clearly. By multiplying the input signal with the carriers (separately and store them) we would demodulate the input signal to Base-Band.

5. Design filter:

Regarding the project guide, we should design a Low-Pass filter (Avoid aliasing) which is cascaded by a notch-filter (Remove DC) to avoid audible distortion. For the Low-pass filter we use "***firpm***" function with Fpass = 4^{KHz} , Fstop = 4.2^{KHz} and the order of filter is N = 610. The frequency response of the Low-Pass filter is shown in [figure 5]. For the notch-filter the coefficient of the filter are the following:

Nominator coefficient Vector: an = [1 -1.99738200612201 0.997383719594995]

Denominator coefficient vector: bn = [1 -2 1]

The normalized frequency response of notch filter is shown in [figure 6].

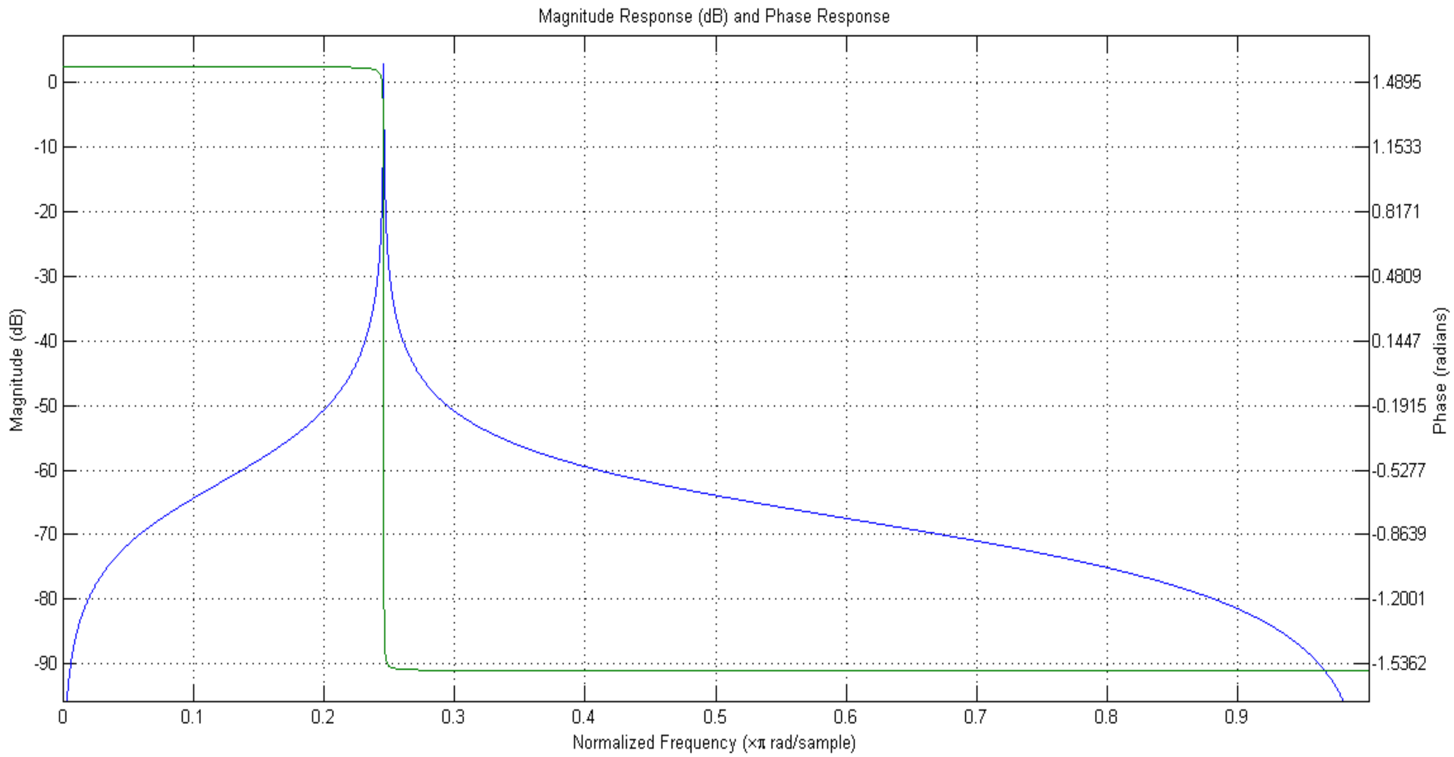

Figure [1]: Designed Second Order FIR filter to extract first carrier with frequency f_1

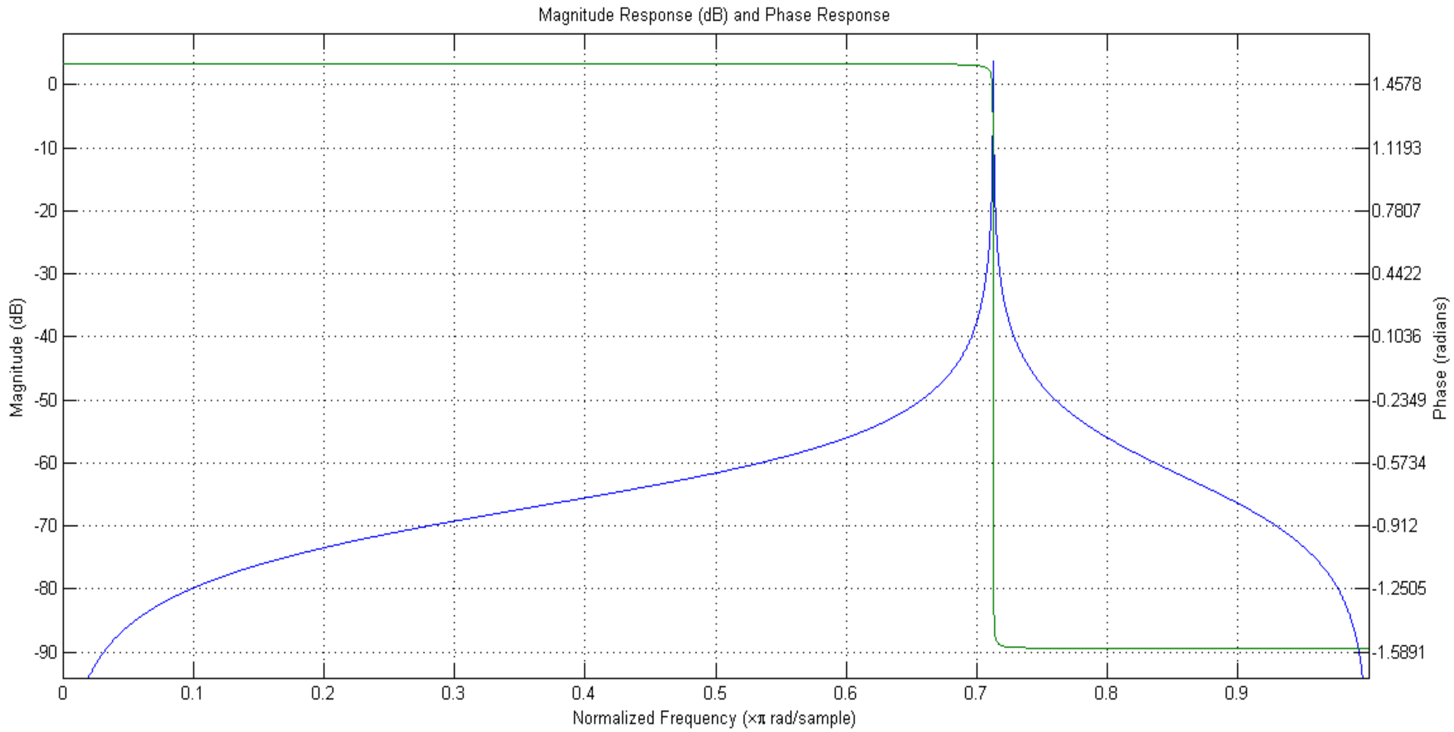

Figure [2]: Designed Second Order FIR filter to extract second carrier with frequency f_2

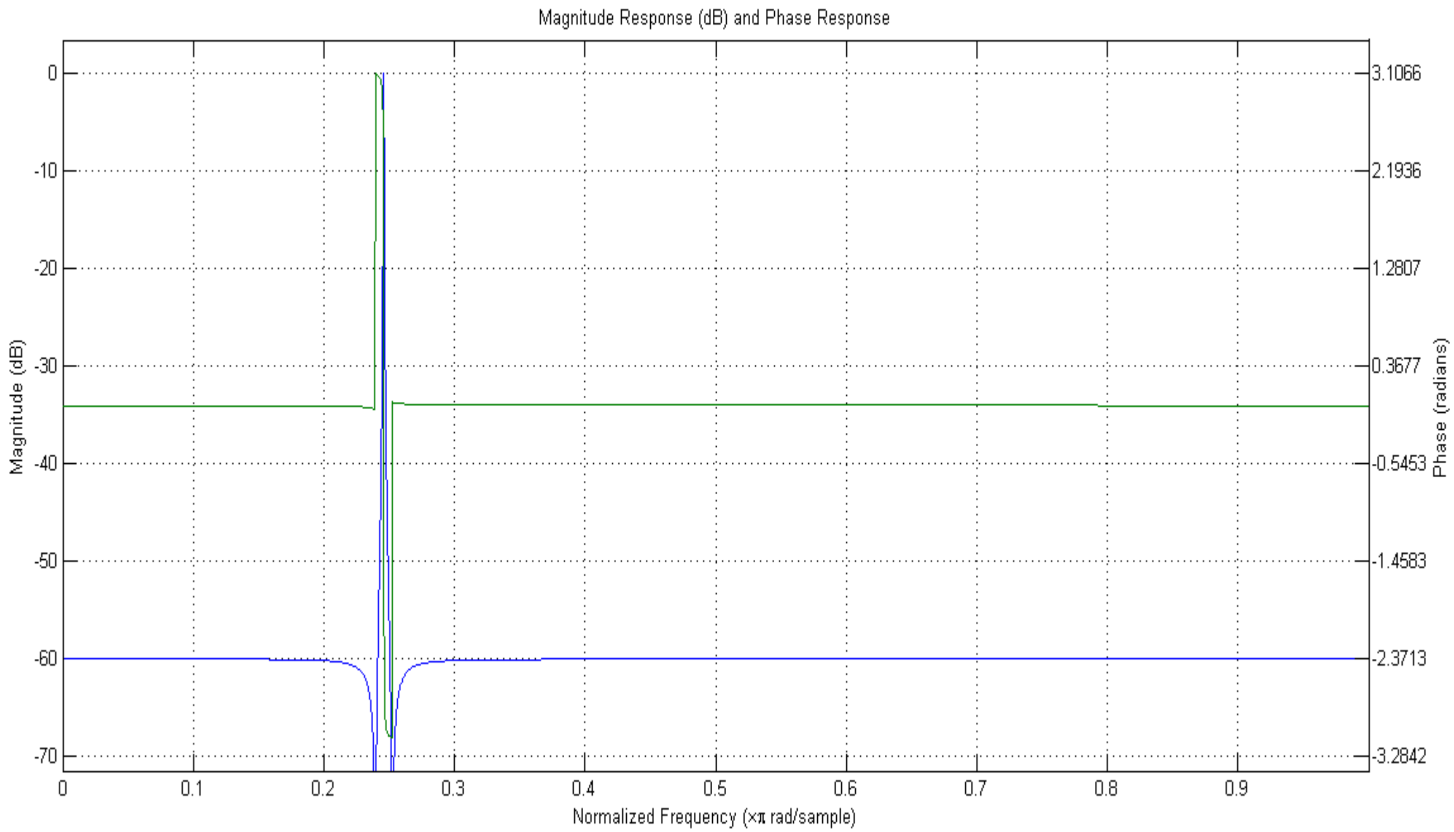

Figure [3]: FIR filter designed by "***ellip***" function to extract first carrier with frequency f_1

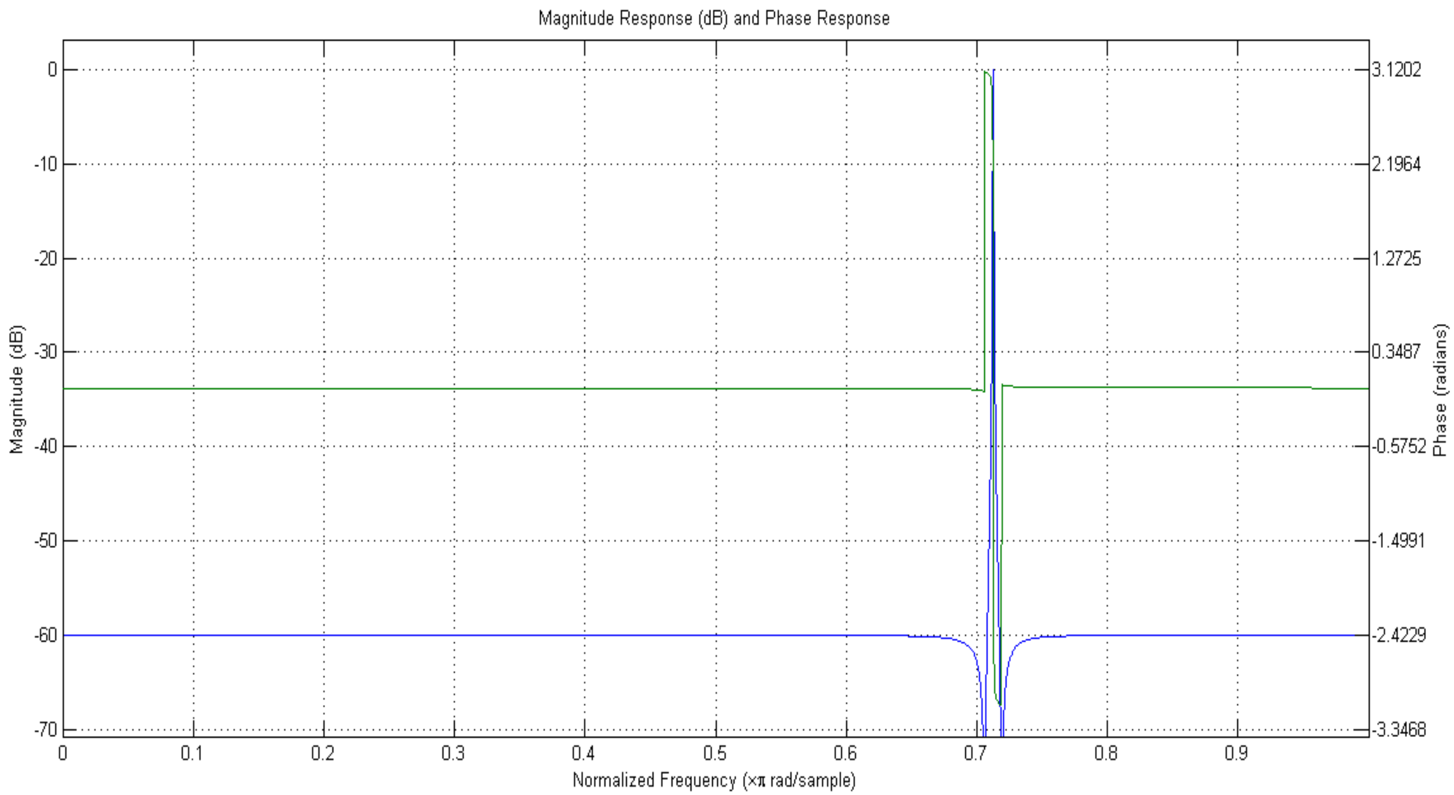

Figure [4]: FIR filter designed by "***ellip***" function to extract second carrier with frequency f_2

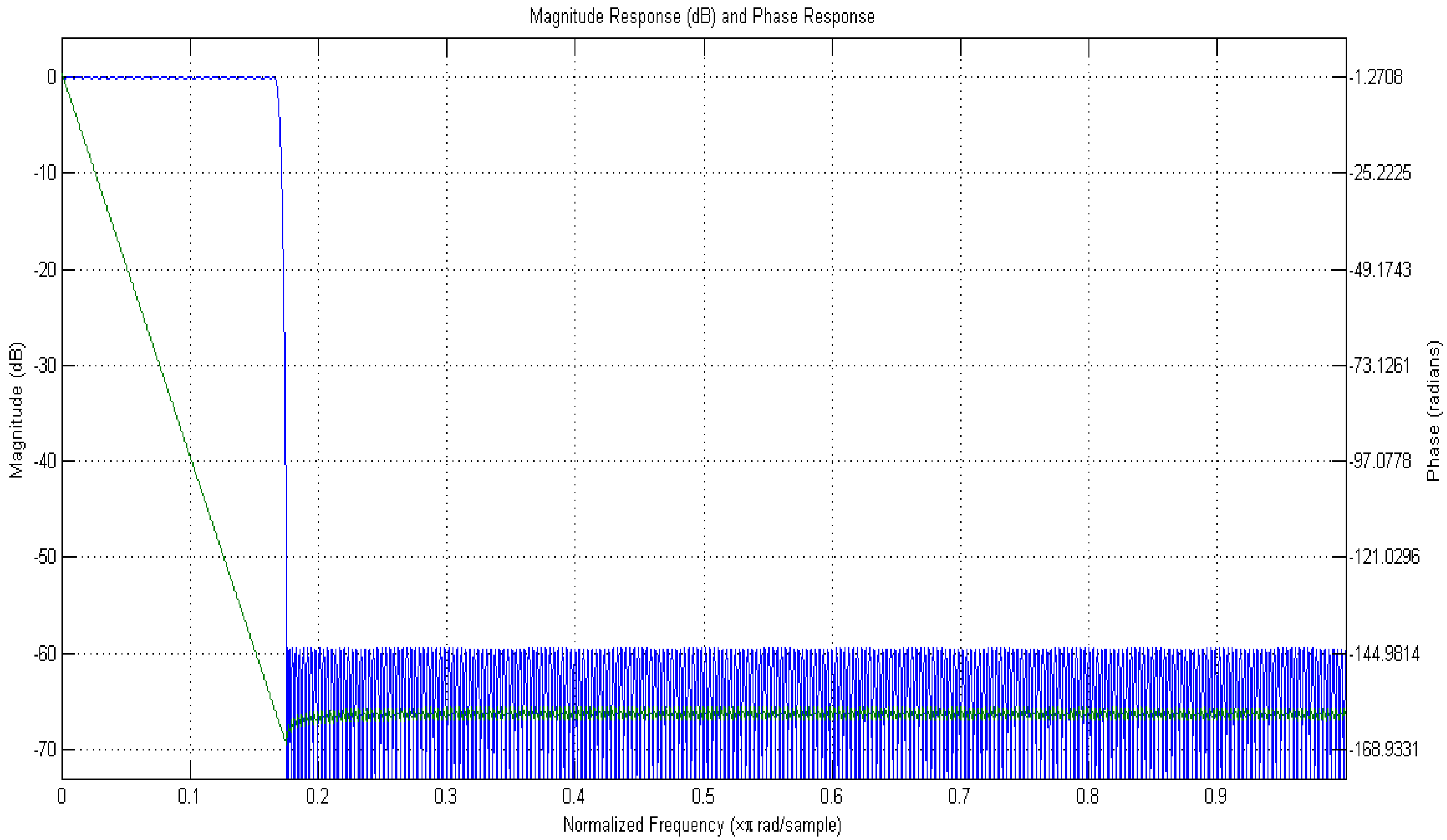

Figure [5]: Low-Pass filter designed by "***firpm***" function to extract first and second audio signal

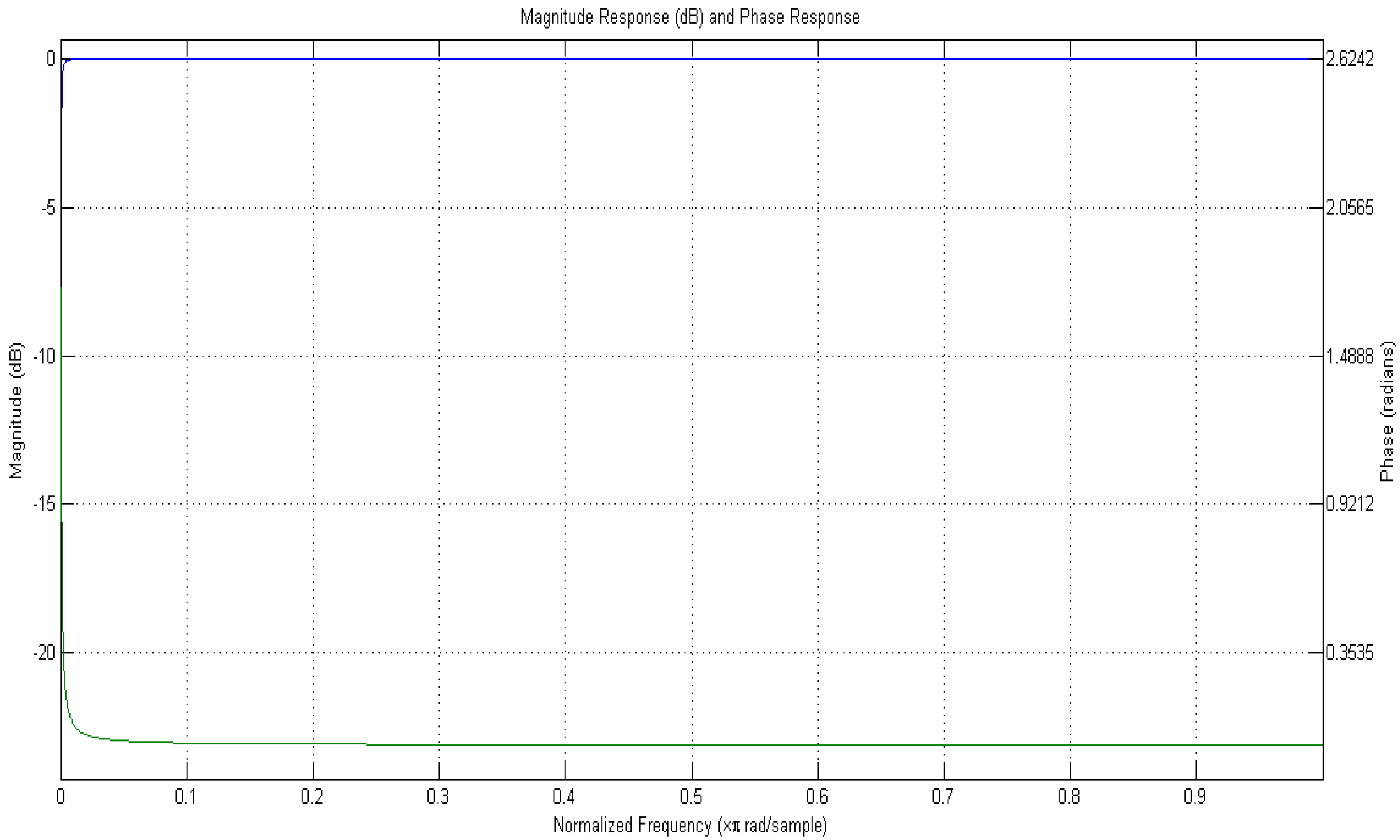

Figure [6]: Second order FIR notch-filter to remove DC regarded to carriers

6. Write the file containing the two output signals.

The two filtered signal is stored as an audio file with the name of Kameli_Mohammad.wav