ICT FOR INTERNET AND MULTIMEDIA - LIFE & HEALTH

DSP Final project 2018/2019

Task

Given an input signal, *signal_102.wav*, of the form

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

the requirede task is to extract a stereo audio file, using the filters described in the corrisponding sections of this paper.
Known information:

- x_1(nT) and x_2(nT) are two real audio information signals with frequency band [20,8000] Hz;
- The sampling frequency is 96000 Hz;
- The audio file is 42 seconds long;
- f_1 and f_2 are the frequencies of the carriers, their relation is: 10000 Hz $\leq f_1 < f_2 \leq$ 38000 Hz; $f_2 - f_1 \leq$ 17000 Hz. A1 and A2 are the amplitudes of the carriers.

The plot of the raw data and its spectrum, in figure 1, clearly shows two main peaks and they are the frequencies of the carriers: f_1 at 18800 Hz and f_2 at 37600 Hz.

An estimation for the amplitude A_i of the carriers is given by the formula $\frac{A_i * n}{2}$, where n is just the number of the points in the FFT. In this case A1 = 0.06, A2 = 0.06.

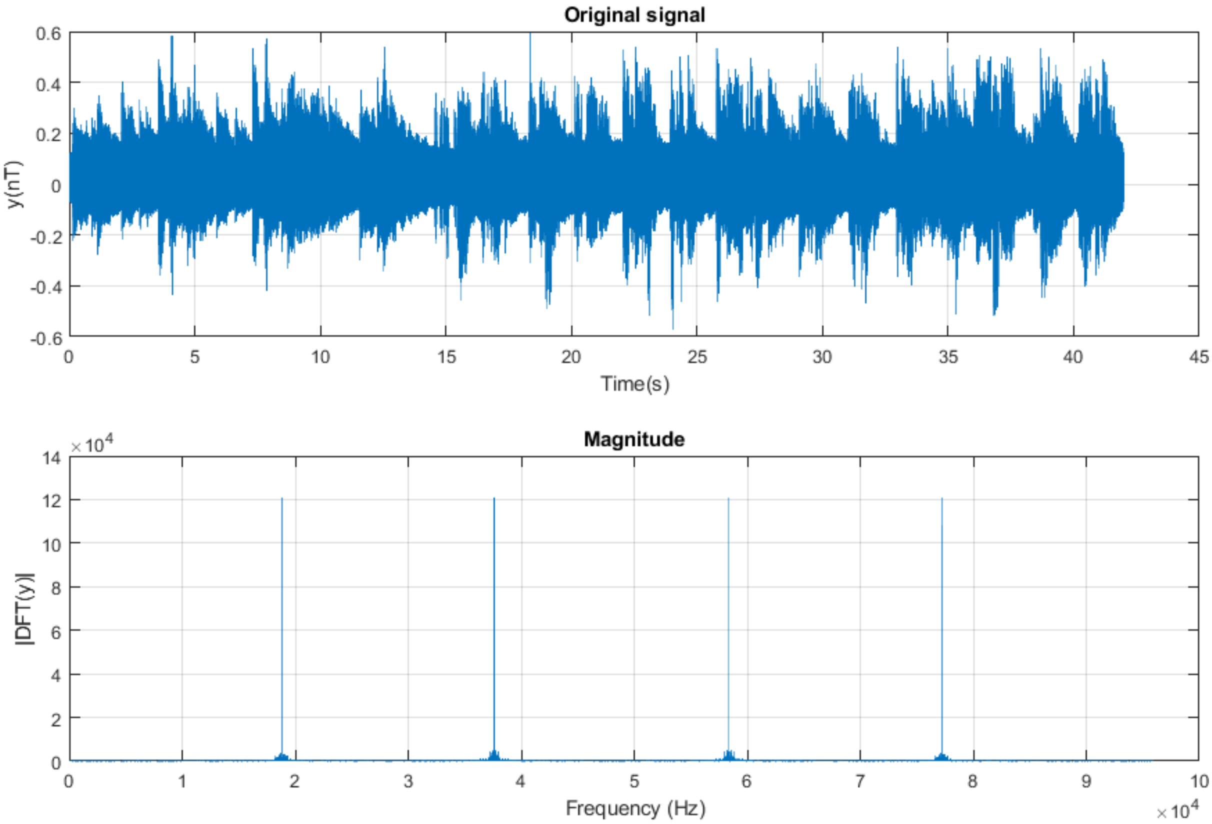

Figure 1: *Plot of the raw data and its spectrum.*

The carriers

Two second order IIR narrow bandwidth band-pass filters are used to extract the carries. Design paramiters for the denominator [1 a_1 a_2] and numerator [b_0 0 -b_0]:

- $\triangle f_{3dB}$ = 10 Hz
- r = 1 - π $\triangle f_{3dB}$ / F_s
- b_0 = 1 - r
- a_1 = -2cos(2π f_0 / F_s)
- a_2 = r^2

with f_0 = f_1 or f_2.
In figure 2 there is the plot for the frequency response for each filter.

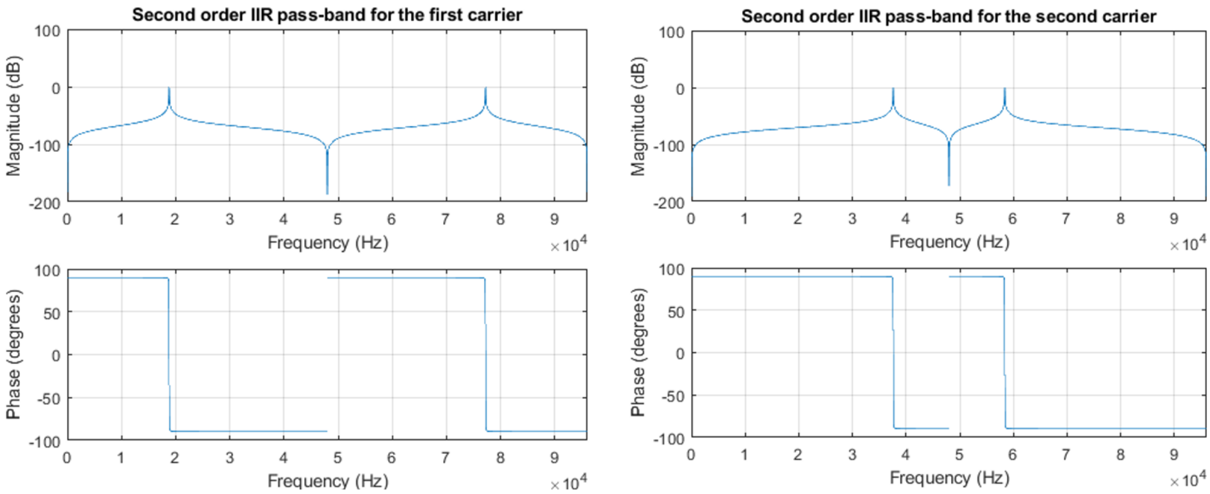

Figure 2: *Second order IIR filters.*

Demodulation

As suggested, the demodulation of the two audio information signals is given by multiplying the modulated signal by the previously extracted carriers (and amplitude) and then the result is filtered with a band-pass [20,8000] Hz. In particular, the band pass is designed as a cascade of a low-pass and a high-pass filter.
The low pass is a linear phase IIR filter, parameters used:

- Pass band frequency = 8000 Hz
- Stop band frequency = 9000 Hz
- Stop band attenuation = 80 dB
- Pass-band ripple = 3 dB
- Order N = 498 dB

In figure 3 is plotted its frequency response.

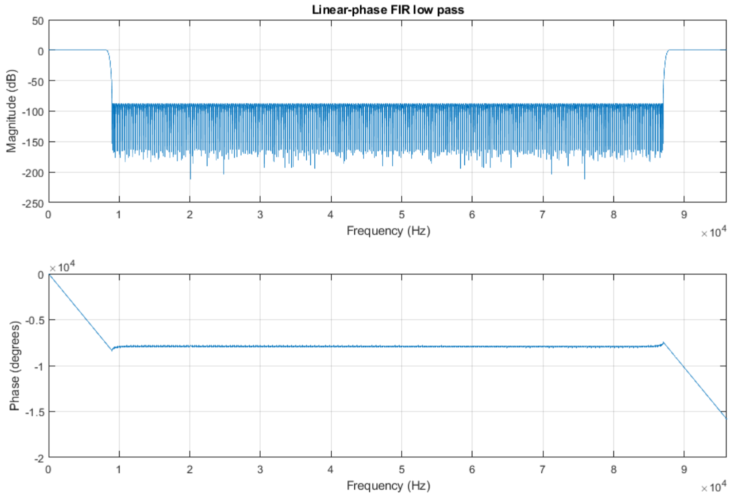

Figure 3: *Frequency response of the low-pass filter.*

The second filter is a high pass notch IIR filter with numerator [1 b_1 1] and denominator [1 a_1 a_2]; in figure 4 it's reported its frequency response. Parameters used:

- $\triangle f_{3dB}$ = 40 Hz
- f_0 = 0 Hz
- r = 1 - π $\triangle f_{3dB}$ / F_s
- b_1 = -2cos(2π f_0 / F_s)
- a_1 = -2 r cos(2π f_0 / F_s)
- a_2 = r^2

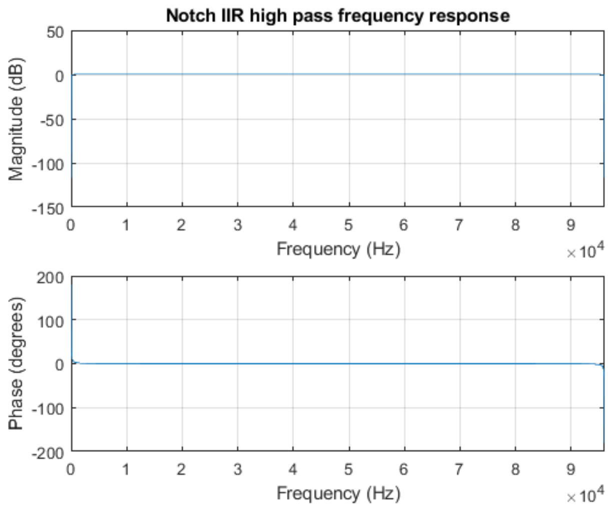

Figure 4: *Frequency response of the high-pass filter.*

Conclusion

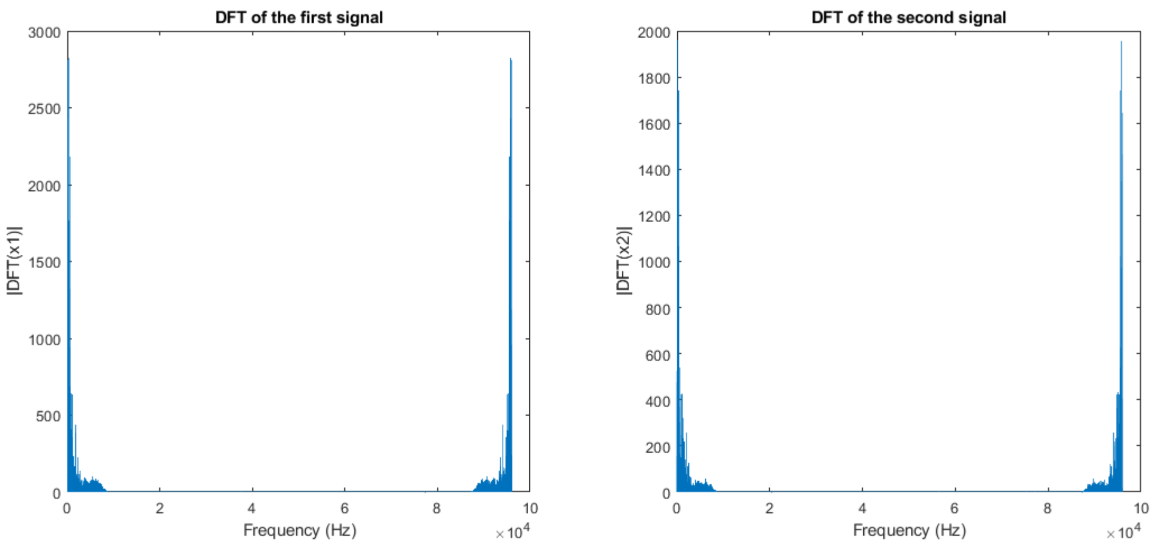

Figure 5: *Spectrum of the two signals.*

In the final spectrum of the two demodulated signals, figure 5, we can see that only the frequencies until 8000 Hz are relevant, so, as we wanted, there is no high frequency presence outside the filtered range. A final audio file is produced, it's a segment from "Father and Son" by Cat Stevens.

Matlab report:

```
*** Peaks ***
First peak found at 18800 Hz
Second peak found at 37600 Hz
Estimated amplitude A1 = 0.06
Estimated amplitude A2 = 0.06
*** second order IIR ***
3dB Bandwidth for both the IIR = 10 Hz
Target frequency first filter = 18800 Hz
Target frequency second filter = 37600 Hz
*** Low pass linear phase ***
Pass band frequency = 8000 Hz
Stop band frequency = 9000 Hz
Desired amplitude of pass band = 1
Desired amplitude of stop band = 0
Stop band attenuation = 80 dB
Pass-band ripple = 3 dB
Order N = 498
*** High pass IIR ***
Center frequency = 0 Hz
3dB Bandwidth = 40 Hz
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