EC3400: Computer Assignment 2

Introduction. This project has several goals:

1. identify a disturbance from the frequency spectrum of a signal;
2. design a recursive filter to eliminate the disturbance;
3. determine the overall frequency response of the filter, including the Sampler and the Zero Order Hold.

The signal DATA.WAV contains a piece of a popular song . The signal is corrupted by a disturbance composed by two sinusoids, which are the same throughout the whole file.

Problem 1. By taking the FFT of part of the signal, determine the frequencies of the disturbance in terms of digital frequency ω (in radians) and analog frequency F (in Hz). In matlab you need to use the command *fft* (for Fast Fourier Transform).

Problem 2. Design a recursive filter

$$y[n] = -a_1 y[n-1] - ... - a_N x[n-N] + b_0 x[n] + ... + b_N x[n-N]$$

to filter out the disturbance. Place the zeros and the poles and make sure the filter is stable and the disturbance is eliminated. The frequency response in magnitude should be close to ideal (zero at the disturbance, flat everywhere else). Test it on the signal and show in the spectrum that it has been eliminated.

Problem 3. Sketch the frequency response of the overall filter, including the ADC and the ZOH. Plot both magnitude and phase as a function of the analog frequency F (Hz).

<u>Hand in</u> :

- a short write up on the design showing where you place the poles and the zeros;
- the plots for the frequency response of the filters (magnitude and phase).

<u>*Matlab Commands of Interest*</u>:

wavread to read a .WAV audio file into a matlab vector;
wavwrite to store a matlab vector into a .WAV audio file;
fft to compute the Discrete Fourier Transform (DFT) of a signal
sound to listen to a file in your sound card
filter to filter a signal;
freqz to generate the frequency response of a LTI system.