Signal & Data Analysis in Neuroscience
Bar Ilan, Spring 2018 Due: June 18 8:00 AM.

Assignment 10

<u>1)</u>
The file q1data.mat contains the representation of two filters (filtVec – 2*50, each filter is one line) and the data (dataVec – 1*10000, 1 second sampled at 10,000 samples/sec).

The filter is defined by:

$$y(n) = \sum_{k=1}^{50} filtVec(i,k) \cdot x(n-k+1)$$

i – number of the filter (either 1 or 2)

For each of the two signals:
1. Plot the impulse response.
2. Plot the original signal overlaid with the two filtered signals.
3. Plot the discrete Fourier transform of the original signal overlaid by the discrete Fourier transform of the filtered signal.
4. Is it a high pass/low pass/band pass filter? Is it an IIR/FIR filter?
5. Compare the two filters to each other and to an optimal filter.

<u>2)</u>
Load the audio file 'audio.au' into MATLAB and listen to it using MATLAB (Sampling rate: 16,384 Hz).
[use auread & sound]
a) Compute and plot the discrete Fourier transform of the signal, displayed in DB units.
b) The signal includes some white noise at high frequencies, and an additional noise at a narrow range of frequencies. What are the noisy frequencies/frequency-ranges?
c) Create a low-pass filter to reduce the high frequency noise and filter the signal. Plot the frequency response of the filter. Re-plot the filtered signal in the frequency domain.
d) Create a notch filter to reduce the narrow-range noise and filter the signal again. Plot the frequency response of the filter. Re-plot the filtered signal in the frequency domain.

For both filters, specify the filter design method, order and the stop and pass frequencies that you used.

Filters specifications:
•Signal magnitude (at the non-desired frequencies) should be reduced by at least 40 dB.
• Filter order: IIR of order <= 15