IIR and FIR Filtering

Part 1: IIR filter as a noise cancelling filter

The signal to use is: Don_Giovanni_1.wav

Connect your headphones to the sound card output of your PC and listen to this signal. You can also use the ".wav" matlab function.
Based on this listening and on the analysis of signal spectrum, you should identify two "noising" frequencies. The target of this work is to develop dedicated FIR/IIR filters, to reject these frequencies.
You are free in the choice of the automatic identification technique. Also, your filtering can be done in time or frequency domain.
One solution is to design an IIR notch filter with one zero placed on the unit circle corresponding to the noising frequency and a pole placed at the same frequency within the unit circle.
Check the validity of the signal processing you proposed by a comparative listening test between the sounds before and after filtering.

Part 2: FIR filter as moving average filter

It is a classic filter whose applications are multiple according to its parameters.
For this filter, the convolution $y_n = \sum_{k=0}^{N-1} x_{n-k}.h_k = \frac{1}{N}\sum_{k=0}^{N-1} x_{n-k}$ gives the average of the N values of x_{n-N+1} to x_n, hence the name of the moving averaging filter.

- Compute, analytically, the transfer function of this filter and the corresponding frequency responses: magnitude and phase.
- Write a Matlab code, simulating the impulse and frequency responses of this filter for any given value of the filter order N. Discuss, the shape of the filter response with increasing value of N.

Download to your working directory the files containing the original audio signal:
Don_Giovanni_2.wav

Connect your headphones to the sound card output of your PC and listen to this signal.

- Apply the moving average filter to the sound "Don_Giovanni_2.wav", with N=3, 11 and 31 and conclude, again, on the behavior of the filter, with increasing order N.