Digital Signal Processing - Lab 3
Infinite Impulse Response (IIR) Filters

1 Simple IIR filters

The goal of this lab is to implement a filter identical to the FIR windowed-sinc filter designed at Lab 2. Using an IIR filter, you have to separate a speech signal from a pure sine signal. Again, the frequencies to be isolated or rejected are respectively 1600 Hz and 5000 Hz. Use second order recursive filters. The transfer function is the following.

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

Specifically, the transfer functions for band-pass and band-reject filters are

$$H_{BP}(z) = \frac{1-\alpha}{2} \frac{1 - z^2}{1 - \beta(1+\alpha)z^{-1} + \alpha z^{-2}}$$

$$H_{BS}(z) = \frac{1+\alpha}{2} \frac{1 - 2\beta z^{-1} + z^{-2}}{1 - \beta(1+\alpha)z^{-1} + \alpha z^{-2}}$$

These functions make use of two parameters α and β, which must be inferior to 1. These parameters control two distinct characteristics of the frequency response of each filter.

2 Assignments

1. First, analyze one filter using Matlab. Set the value of one parameter to a fixed value and change the value of the second parameter in order to find out which characteristics are controlled by α and β. Use *freqz* to visusalize the results of modifying the parameters.
2. Compute α and β for the cutoff frequency and bandwidth that you need and replace them in the transfer function. From there, compute the impulse response of your filter and implement it using a straightforward algorithm on the DSP.
3. Improve your algorithm by implementing Direct Form II (as shown in Figure 1). Listen to the result and measure the frequency response of the filter.
4. Design a new filter by cascading two identical filters, that is, send the samples in the first filter, then take the output of the first filter and send it again to the same filter. Analyze the result and measure the frequency response. What do you conclude ?

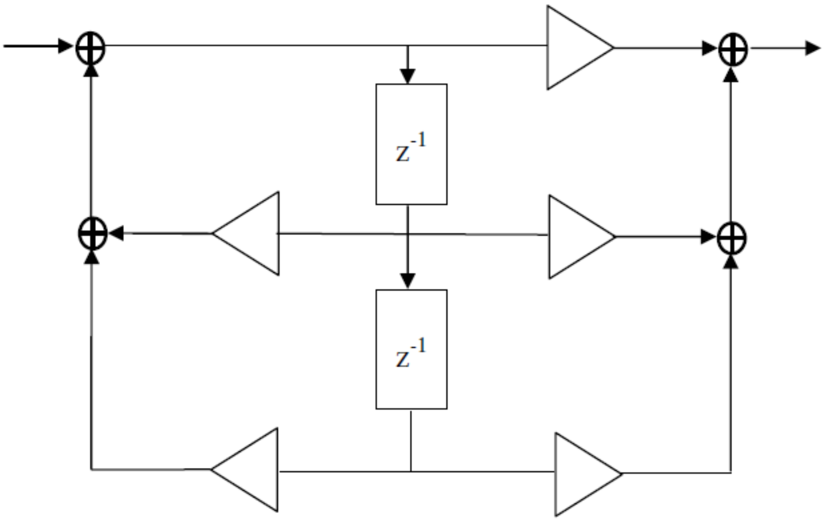

FIGURE 1 – Recursive filter in Direct Form II (DF-II)