Digital Signal Processing - Lab 2
FIR Filtering

1 Designing an FIR filter

The goal of this lab is to split a voice signal from a sinus signal in an audio file. In the folder $C:\backslash DSP\backslash audio$, you will find two audio files containing spoken voices mixed with pure sinus signals (respectively 1600 Hz and 5000 Hz). Chose an input audio file and design an FIR filter to reject the sinus frequency. Then do the same with the second input audio file, but this time, isolate the sinus frequency.

Chapter 4 in the textbook describes how to design a *Windowed Sinc* FIR filter. Use that kind of filter to design a band-pass filter and a band-reject filter. In Matlab, write a script to compute the coefficients of your filters. The scripts takes the length of the filter as an input and returns the corresponding impulse response.

In summary, you have to :

1. Write a Matlab script yo compute the impulse responses of your band-pass and band-reject filters ;
2. Use your script to compute your filter's coefficients ;
3. Generate a C header file containing your coefficients in an array using the *L138_fir_ coeffs* Matlab function ;
4. Write the code implementing your filter.

We suggest that you first write a straightforward basic version of your filter that implements a simple convolution. Then, try to optimize it with a *circular buffer*. Once the filters are up and running, measure their frequency response with the AudioPrecision measurement device.

2 Reminder : Windowed-Sinc FIR Filters

The impulse response of a low-pass and high-pass filter is computed as

$$h[i] = \pm\frac{sin(2\pi f_c i)}{i\pi}$$

Two parameters are needed : the cutoff frequency f_c, which is expressed as a fraction of the sampling rate, and the length of the filter M. The length of the filter can be computed as

$$M \approx \frac{4}{BW}$$

where BW corresponds to the width of the transition band, also expressed as a fraction of the sampling rate. Don't forget to window your filters with a Blackman or Hamming window.

$$w_{Hamming}[i] = 0.54 - 0.46cos(2\pi i/M)$$

$$w_{Blakman}[i] = 0.42 - 0.5cos(2\pi i/M) + 0.08cos(4\pi i/M)$$

Important note : A *band-pass* filter is obtained by **convolving** a high-pass filter with a low-pass filter, and a *band-reject* filter is obtained by **adding** both filters.