Software : MATLAB.
Load the files gamme.m, transffourrier.m and sonbruite.wav from the “serveur pedagogique” : https://pedagogie.ec-nantes.fr/

1 – Synthesis and filtering of an audio signal

Let us consider a note of music as a pure tone produced by a sine wave.

gamme ([s,t] = gamme(duree,fs)) is a function (serveur pedagogique) that generates in *s* successively the 8 notes C, D, E, F, G, A, B, C, corresponding to the following frequencies (in Hz) : 262, 294, 330, 349, 392, 440, 494, 523 Hz. The duration of each note is *duree* and the sampling frequency is *fs*.

Take *duree* = 1 second and *fs* = 8192 Hz.

Listen to the notes (**sound**).

Compute the magnitude spectrum of the signal (using **Transffourier**) and observe it. (Using the function **axis**, limit the **frequency axis to 250-550Hz**)

2 – Design of FIR and RII filters and comparison of the frequency responses

2.1 Specifications for a low-pass filter

fc = 420 Hz (fc corresponds to the middle of the transition band limited by fp and fa)
Delta_f= 200 Hz (transition width)
Ap = 3 dB (passband ripple)
Aa = 40 dB (stopband attenuation)
with fs = 8192 Hz

2.2 Design of FIR filters

Design the following FIR filters respecting the previous specifications using the window design method

- with a rectangular window (**rectwin, fir1**)
- with a Hamming window (**hamming, fir1**)
- with a Kaiser window (**kaiserord, kaiser, fir1**)

for the two first method, the filter order will be first calculated from the table of the windows characteristics ; for the last one, it will be computed using kaiserord.

Give the order of the filters. Plot their impulse responses.

Plot on the same figure (superimposed) the gain in dB of the 3 frequency responses (**freqz**) and the specifications. (`plot ([0 fp fp], [-Ap -Ap -Aa],'k',[fa fa fe/2], [-Ap -Aa -Aa],'k');`)
Using the function **axis**, limit the **frequency axis to 0-1000Hz**.
Verify that the specifications are respected. Comment.

2.3 Design of IIR filters

With the same specifications, design the RII filters using :

- the Butterworth model (**buttord, butter**)

- the Chebyshev model of type 1 (**cheb1ord, cheby1**)
- the Chebyshev model of type 2 (**cheb2ord, cheby2**)

Give the order of the filters. Plot on the same figure (superimposed) the gain of the 3 frequency responses (**freqz**). Verify that the specifications are respected. Comment.

2.4 Compare the phases of the filters

Plot the phase spectrum (using the function **phase**) of the 2 following filters : hamming FIR filter and Cheb1 IIR filters.

2.5 Change the transition width

Take now Delta_f= 80 Hz.

Give the order of the filters and plot on the same figure the gain of the 2 frequency responses (**freqz**) for the the FIR filters Hamming and Kaiser and the IIR filters Chebyshev 1 and 2.

2.6 Change the passband ripple

Take now Delta_f= 80 Hz and Ap = 1 dB.

Give the order of the filters and plot on the same figure the gain of the 2 frequency responses (**freqz**) for the the FIR filters Hamming and Kaiser and the IIR filters Chebyshev 1 and 2.

3 – Filtering an audio signal

3.1 Suppress the 3 last notes with an IIR filter

With the following specifications : fc = 420 Hz, Delta_f= 80 Hz, Ap = 3 dB, Aa = 40 dB

- design a lowpass Chebyshev 2 filter ; which is its order ?
- filter the signal (**filter**)
- plot the filtered signal
- compute and plot (below the signal) the magnitude of the frequency spectrum of the filtered signal
- interpret the results, regarding the frequency response of the filter
- listen to the filtered signal.

Repeat the same steps with Delta_f= 20 Hz.

3.2 Suppress the note F using IIR and FIR filters

With the following specifications :
fcl = 340 Hz, fch = 360 Hz, Delta_f= 10 Hz, Ap = 3 dB, Aa = 40 dB

- design a bandstop Chebyshev 2 filter ; which is its order ?
- plot the gain of the frequency response
- filter the signal (**filter**)
- plot the filtered signal
- compute and plot (below the signal) the magnitude of the frequency spectrum of the filtered signal
- verify if the specifications are respected and interpret the results
- listen to the filtered signal.

Repeat the same steps with the FIR filter based on the Kaiser window method (**kaiserord, kaiser, fir1**). Observe that the filtered signal is delayed, measure the delay. How is it related to the filter order ?

4 - Filtering a noisy audio signal

The signal **sonbruite.wav** (serveur pedagogique) is corrupted by a strong noise (stronger than the useful signal). We want to filter the signal to suppress the noise.
In this aim :

- read the signal .wav : **[y, fs]=wavread ('sonbruite.wav')**
- listen to the signal,
- observe its spectrum using your function transffourier,
- deduce the type of filter (lowpass, highpass ?) and its specifications (fc, Delta_f, Ap Aa),
- choose a type of filter
- filter the signal, listen to the filtered signal and plot its spectrum.

Some complements on Matlab functions to use. For more details, use matlab help under the command window.

To listen to the signal y :

sound (y,fs)

/Values in y are assumed to be in the range -1.0 <= y <= 1.0. Values outside that range are clipped./

Filter design :

FIR and window design method :

h = fir1 (n, 2*fc/fs, window (n+1)) / h : impulse response of a n order filter, with fc/fs the normalized cutoff frequency, and window the type of the truncating window (hamming, boxcar, blackman , ….)

IIR and model based method

buttord : finds the parameters (order, cutoff frequency) of the method according to the specifications

butter : computes the coefficients of A(z) and B(z), using the output parameters of butterord as input of butter

idem for **cheb1ord, cheby1** and for **cheb2ord, cheby2**

Frequency response :

[H, fh] = freqz (b, a, npf, fs) / b and a are the vectors of the coefficients of B(z) and A(z), (for FIR filters, b=h and a=1), fs the sampling frequency and npf the number of points for which H is calculated ; H : frequency response, fh : corresponding frequencies /

S = 20*log10 (abs(H)) / Gain in dB /

plot (fh, S) / plots the Gain in dB /

phi = phase(H) / phase /

To filter the signal y :

z = filter (h,1,y) / in the case of a FIR filter of impulse response h /

z = filter (b,a,y) / in the case of an IIR filter B(z)/A(z) /