# MATLAB Project

## Goals

The goals of this project is to learn how to design and apply a digital filter to remove unwanted content from a signal.

## Filter Configurations

All the filter design functions in the Signal Processing Toolbox operate with normalized frequencies. The normalized frequency is always in the interval $0< f <1$. To convert the normalized frequency back to Hertz, multiply by half the sample frequency (Fs/2). To convert normalized frequency to angular frequency, multiply by π/sample.

## Procedures

(1) Load the audio file ***craudio*** (sample frequency Fs=8192 Hz), playback the audio and plot the power spectrum of the signal (figure1).

(2) To suppress the noise in the audio signal, design a IIR Butterworth low-pass filter with no more than 3 dB of ripple in the passband, and at least 60 dB of attenuation in the stopband. The transition band has a bandwidth of 2 kHz.

   (a) Specify the cutoff frequency and find the filter order.
   (b) Determine the system function H(z) of the filter. Print the coefficients **b** and **a**.
   (c) Plot the magnitude response of the filter (figure2).
   (d) Draw the filter structures (Direct Form II).

(3) Filter the ***craudio*** using the filter designed in step (2), playback the filtered audio and plot the power spectrum (figure3). Has the noise been suppressed by the filter?

(4) Plot the waveforms of ***craudio*** and the filtered signal in the same window (using ***subplot***) for comparison. (figure 4).