Assignment 3, Digital Signal processing: IIR filters

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This assignment covers IIR filters which as before can be low/high/band or stopband filters. Here, we deal with a more realistic scenario: the noise is broadband and the signal bandlimited or in other words the frequency spectra of signal and noise overlap.

The biomed students will tidy up a biosignal (ECG, ...) with artefact contamination (EMG in particular). The final result could be the signal itself or a detection, for example the heartrate.

Anybody else processes a sound recording with background noise – recorded not in the lab but at a proper location. We also have a pro sound recorder and a USB interface for pro microphones if you want to borrow it from Tom O'Hara.

Again you work in teams of two students.

In both cases you need to determine the difference between signal and noise which can be done in different ways. For example, in speech we record "silence" (for example the street noise which the pros call "wild track") to determine which frequencies annoy most. In medical applications these are the artefacts, here the EMG noise and voltages generated by cables. The EMG noise can be recorded by measuring just the muscle activity (for example, between your wrist and shoulder to include the arm muscles or from the hip to the ankle when doing exercises with your legs). For the voice scenario choose a noisy environment, for example outside or a pub and record your voice there. Record it outside of the lab in a realistic environment.

In summary you need to measure the following:

1. your clean speech / biosignal (record both as clean as possible)
2. the noise ("wildtrack" or EMG again in it's cleanest form)
3. the signal recorded under normal conditions (background noise, ECG while doing an exercise . . .)

This task requires planning/initiative before you come to the lab. Think of a scenario before the lab starts. It's not the task of the lab demonstrators / technicians to come up with solutions and they need to come genuinely from you. You need to do it and you need to be creative. I'd like to see different approaches from every team. Enter you project ideas on the WIKI on moodle so that others can see what's already taken.

1. Present an experiment which shows the difficulties of filtering when the spectra of signal and noise are overlapping. Marks are given for initiative, inventiveness and originality (= ideas which haven't come straight from the lecturer, lab demonstrators or other groups). Document the experiment with photos of the setup, diagrams, dataflow diagrams or even shoot a YouTube clip in addition to your report. [30%]

2. Determine the ideal filter response(s) which remove as much noise as possible while leaving the signal intact (i.e. maximises signal to noise ratio). Feel free to do a dynamic approach where different filters are enabled or disabled depending on the input signal. Plot for this purpose the noise, noise+signal and signal. Note that human hearing perception is *logarithmic* in both the frequency domain so you should plot audio logarithmically. Justify your choice of filter parameters. On the other hand ECG, for example, is evaluated by looking at the trace or when you do heartrate detection on the amount of right/wrong detections.

   Generate the coefficients for the filter(s) with the help of Python's high level functions. In case you have higher order filters then implement the filter(s) as a chain of 2nd order filters and cascade them. Feel free to write an IIR class which does everything from filter design, splitting it into 2nd order filters and then calling IIR2Filter. [20%]

3. Write a class `IIR2Filter` which implements a 2nd order IIR filter which takes the coefficients in the constructor and has a method called:

   ```
   y=IIR2Filter.filter(x)
   ```

   where `y` and `x` are simple scalars (no arrays) as before. [30%]

4. Compare your filtered results with the original recordings and discuss if you have been successful. Do a critical analysis. [20%]

Submit your report in *online only* form via moodle. Deadline is 8th January 3pm. In case you have audio upload it to moodle. As before I expect sharp figures in vector format in the report.

Bonus: If you can demonstrate a working python class on github which directly implements a cascaded bandpass/lowpass/bandstop/highpass filter which just takes the cutoff frequencies and has a filter function then you get full marks guaranteed. See for example the IIRJ library.