Exercise in INF4480: Adaptive filter

The LMS-filter will be used to cancel noise and disturbance on speech signals.

Use the code that comes with the Hayes book, both for the LMS filter and the normalized LMS filter: `lms.m` and `nlms.m`. The code can be found on Matlab Central: https://se.mathworks.com/matlabcentral/fileexchange/2183-statistical-digital-signal-processing-and-modeling. Remember to download the file `convm.m` also.

- There is an error in the main loop of `lms.m`. Change ”`for k=2:M;`” to ”`for k=2:M-nord+1;`” just as is already done in `nlms.m`
- The code will also run more than 100 times faster if the matrix A is initalized. Put this code in an appropriate place in each routine: `A = zeros(M-nord+1,N);`

The first speech examples are from http://www.w9gr.com/#Sound Clips. As these examples originally consisted of signal+noise followed by the filtered signal, they have been cut in two so that only the first part can be processed.

Make the result from the filtering of the audio available (OneDrive, Dropbox etc) and send the links with your hand-in for this exercise. Give an overview in the form of a table of algorithms and the best set of parameters that you found for cancellation of tone/noise for each example.

1. Removal of tone from broad band signal

Audio clip:

- Undesired carrier on a short wave link: “HF SSB automatic notch filter example #1”: notch1-a.wav
- 3 consecutive tones on a clean recording of female and male voice: kvinne-mann-3toner.wave
- Chirp tone on a clean recording of female and male voice: kvinne-mann-440-4400.wav

a. Explain how a reference signal can be found in this case
b. Experiment with parameters for the LMS and the normalized LMS filters for the best possible result. In each case estimate the value of the maximum value of μ in order to know how to set its value (the maximum value will change as the order of the filter changes). Make a table with an overview of the ’best’ values for the parameters that you find.
c. Illustrate the effect of the filter with plots showing the difference before and after filtering. Justify why you illustrate it in the time-domain, frequency-domain or the time-frequency domain (spectrogram)

2. Removal of broad band noise from broad band signal

2.1 Sound clip, communications system:

- Atmospheric noise that disturbs a short wave communications link: ”HF SSB noise reduction example #1” hfqrn1-A.wav (QRN = atmospheric noise, see http://en.wikipedia.org/wiki/Q_code)
- VHF FM link which is close to the lower limit for usability in terms of signal to noise ratio: ”2 meter FM noise reduction example #1”
- VHF FM weather service which also is close to the lower limit for usability: ”VHF FM (NOAA weather) noise reduction example #1”

a. Explain how a reference signal can be found in this case
b. Experiment with parameters for the LMS and normalized LMS-filters for best possible result. Generate a table with an overview of the ’best’ values for parameters that you find. Notice that the objective is to get a best possible readability of what is being said, not noise-free hi-fi.
c. Illustrate the effect of the filter with plots showing the difference before and after filtering. Justify why you illustrate it in the time-domain, frequency-domain or the time-frequency domain (spectrogram)

2.2 . Improvement of old audio recordings

Audio clips, old audio:

- BBC recording of King Edward's abdication, 1936 (from NRK archives)
- [Extra: Introduction to Rørospols (national dance, from Nasjonalbiblioteket) from the 1920/30s]

a. Experiment with parameters for the LMS and normalized LMS-filters for best possible result. Generate a table with an overview of the 'best' values for parameters that you find.

b. Why is it much harder to improve the signal compared to section 2.1?

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