EEL 6502

Adaptive Signal Processing

Project I

Due March 12, 2020

The purpose of this project is to design and evaluate the performance of a machine learning algorithm using the normalized LMS algorithm (or one of its variants) that will clean the desired input (speech plus noise) from the machine noise (input). This is an example of the interference canceling problem explained below.

Speech data is collected in a noisy room (loud vacuum cleaner) by two microphones: one in a table that captures speech with the vacuum cleaner noise (d(n)) and the other very close to the vacuum cleaner (n(n)) that basically has no speech. Even if you listen to d(n) the speech is barely audible, and the message is not understandable. The goal is to denoise d(n) and be able to understand the speech. I suggest that you use the signal n(n) as the input to the LMS algorithm and use d(n) as the desired response.

You will find the data set project1.mat in Canvas. This file contains a .mat file with two channel data labeled desired (d) and input (n). The sampling frequency is 21 KHz.

The project requires a report explaining the experimental procedures you select and you must include tables and plots to support your conclusions. Please use the format of an IEEE Transactions paper (limited to 7 double column pages). This means you have to write a brief intro to the theory, explain well the methods and present carefully the results (see below) and conclude. Remember that any scientific paper should, by definition, contain sufficient information such others can replicate your results. A scientific paper must also contain ORIGINAL material only. If you happen to use text or equations from other source you have to reference what you cut and paste (this is not allowed in a normal publication, but here it is OK provide you reference). Of course, I expect the results to be done by the student alone. I would like to see in the report (at least) the following:

Start with a 2-tap filter

1- Plots of the performance surface contours for the two weights filter case.
2- Plot the weight tracks
3- Plot the learning curve and interpret it.
4- Estimate the frequency response from the desired signal to the error when the filter is adapted.
5- Estimate the SNR improvement in dB by the ERLE=$10\log(E\{d^2\}/(E[e^2])$.

Increase the filter order based on an analysis of performance. Explain your choice for the filter order.

1- Estimate again the frequency response from the desired signal to the error.
2- Compute the SNR improvement in dB
3- Evaluate the filter performance as a function of the stepsize.
4- Estimate the misadjustment.
5- Comment on the results obtained and address issues related to the convergence of the algorithm in non-stationary environments.

You can use a FIR filter, a gamma filter or the KLMS to compare performance.