EEL 6502

Adaptive Signal Processing

Project I

Due March 15, 2012

You will find the data set project1.mat in the course website. This file contains a .mat file with two channel data labeled desired (d) and input (n). The sampling frequency is 16 KHz.

The purpose of this project is to design and evaluate the performance of an adaptive filter 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. Please consult Chapter 5 of the book if you need further theoretical explanations about interference canceling.

The project requires a report explaining the experimental procedures you followed and you must include data 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=$(E\{d^2\}/(E[e^2])$.

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.