EEL4930/EEL5840 Fall 2016 – Project 1
Normalized LMS Algorithm – Interference Canceling

October 4, 2016

Due: October 18, 2016, 11:59 PM

This project will be graded with a letter grade with respect to presentation (25%), methods (35%) and results (40%).

You will find the data set *project1.mat* in the course website. This file contains two channel data labeled desired (d) and input (n). The sampling frequency is 21 KHz. The purpose of this project is to design and evaluate the performance of an adaptive FIR 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 interference canceling problem shown in the Figure 1.

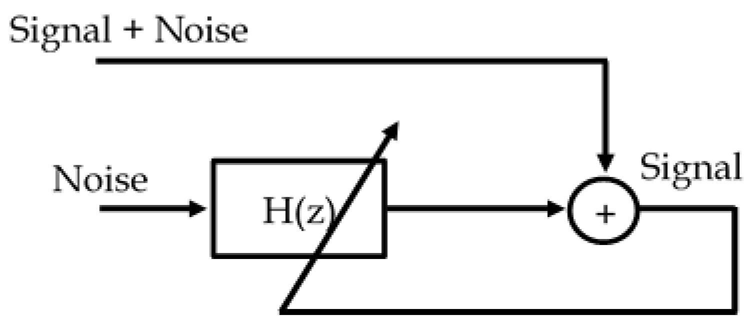

Figure 1: Diagram of an interference canceling problem.

The adaptive filter will always try to make its output as close as possible to the desired given the input. In this case the input is mostly the vacuum cleaner noise. So the filter will learn an output that will match the vacuum cleaner noise added to the speech in the desired. Therefore, the speech will appear as the ERROR!

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). You can download the format from the IEEE website. This means you have to write a brief intro to the theory, explain well the methods and present carefully the results (see below) and conclusions. 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 (one delay) filter

1. Plot the performance surface contours with the weight tracks for the two weights filter case. Weight tracks is the plot of the weights over iterations. Explain what the plot shows.
2. Plot the learning curve and interpret it. Experiment with different learning rates and describe the effect of different learning rates in the learning curve. Did your filter converge? In how many samples?
3. Estimate the SNR improvement in dB by the ERLE= $10\log(E[d^2]/E[e^2])$. Can you understand what the person is saying?

One way to improve the error is to select higher filter orders. I suggest that you use filters of order 10, 20, and 50. Can you use cross-validation based on the ERLE? Explain why or why not.

1. Compute the SNR improvement in dB by the ERLE for the selected filter orders.
2. Select the best step-size by plotting the learning curves.
3. Verify if the best filter, in terms of ERLE, agrees with your assessment of speech intelligibility. Can you explain the difference (if there is one)?