LUNDS TEKNISKA HÖGSKOLA
Inst. för Elektro- och Informationsteknik

Home assignment in
ADAPTIVE SIGNAL PROCESSING (ETTN05)
2013-11-19–2013-11-30

Rules: Since the results of the home assignment may improve your course grading, no form of collaboration is allowed regarding written solutions or computer code. You are, however, allowed to ask each other or the teacher questions about how to solve the problems.

Grading: There are eight problems (4 theoretical and 4 matlab). Two correctly solved problems in each part are mandatory. Additional correct solutions gives additional 0.1p each on the exam summing up to the maximum possible add-on of 0.4p on the exam result. Each solution must be well-motivated and completely correct in order to give you 0.1p. A few of you may be asked to orally explain your solutions.

T1 (i) Explain the methodological differences between following algorithms - the wiener filter, the method of the steepest descent and the LMS algorithm? (ii) Show in level plots the two main differences in the convergence behavior between the SD and the LMS? (iii) Indicate $J(n)$, J_{min} , $J_{ex}(\infty)$ and $\mathbf{w}_o$ in an error surface (bowl) for the LMS algorithm? (iv) Explain the relationship between the orientation of the error surface bowl and the eigenvalues and eigenvectors and explain how the shortest axis of the bowl is related to the stability of the algorithm

T2 (i) Design an LMS-based 1-sample predictor of length 3 of a stochastic process with $r_u(0) = 4, r_u(1) = 2, r_u(2) = 1$ and $r_u(3) = r_u(4) = 0$, i.e., $d(n) = u(n+1)$. What is w_o, J_{min} and $J_{ex}(\infty)$ if calculations are based on $\mu = \frac{1}{4}\mu_{max}$? (ii) What would happen with w_o if we instead want to calculate 3-sample predictor?

T3 (i) Give an example of when each of the normalized LMS, the Leaky LMS, and the sign LMS should be used. (ii) Give two advantages of the Fast LMS. (iii) What is the principal difference between the LMS and the RLS when it comes to their strategy of finding the optimum solution. (iv) Which of the LMS and the RLS is best at convergence and tracking respectively?

T4 (i) Explain the block diagram of the equalizer completely if the channel is a length-5 minimum phase low-pass filter and the equalizer is of length 9. (ii) Explain the block diagram of the identifier completely in terms of how to connect it, how to choose the model order and what happends with additive noise when measuring the output of the unknown system.

M1 (1) A white noise process of length $N = 500$ and unit variance (normal distribution) is used to identify an unknown system. Physical considerations indicate that the delay of the slowest reflexion in the system is 5. The output signal from the system (d(n)) can be downloaded from the homepage (asbhomeM1.mat, use load). Select a suitable filter length and step size for the LMS algorithm. Use the LMS algorithm from computer exercise 2 to identify the system. What is the unknown system (filter)? What is the error and why?(0.1p)

(2) Use M = 2 and plot the convergence for $\mathbf{w}$ in a level curve plot (See computer exercise 1 and modify the LMS function such that it saves all old $\mathbf{w}$:s like the sd function does). Why is the resulting error different from above? Hint: In order to plot level curves around $\mathbf{w}_o$, $\mathbf{R}$ and $\mathbf{p}$ needs to be calculated, see example on lecture 3 with a = 0 and use the jmat command from computer exercise 1. You need to use the result from (1) when calculating $\mathbf{p}$.

M2 Use an adaptive line enhancer to clean the message in asbhomeM2.mat (use load and then sound at 8kHz sampling rate). Explain how you do it (filter order, step size, etc) and plot the error function

M3 Record a sentence (speech signal of 20 seconds) of your own (or download a sentence from the Internet) and generate an echo version of it (i.e. create d) consiting of one echopath with attenuation to 25% of the amplitude and a 100ms delay and one echopath with attenuation to 16% of the amplitude and a delay of 130ms. Run the FastLMS from computer exercise 4 for these two signals (original, u, and only echo d). Hand in the sentence (written as text), the settings you chose to run the algorithm with, and the resulting ERLE curve. Is the echo reduced to an acceptable level? Also, plot the impulse response of the adaptive filter.

M4 Repeat problem M3 but also add another speakers voice to the returning echo signal (d). Plot the ERLE curve and explain the problem. Is it only the new speakers voice remaining in e?

Good luck!