SGN-2206 Adaptive Signal Processing

Project Work

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The solution of this problem is compulsory to be presented in the form of a short report, containing your experimental results. Return the report to the box 322, third floor, Department of Signal Processing.

The application addresses the system identification problem from figure plotted at the top of page 12 in Lecture 1. The LMS algorithm will be used to find the taps $\underline{w}$ of the Adaptive Filter.

The tasks are the following:

(a) Generate the Plant impulse response

$$H(z) = \sum_{j=0}^{N-1} p^j z^{-j}, \quad p = 0.80025, \quad N = 32$$

(b) Three different signals will be used as input signal $\underline{u}$:

Signal 1: zero-mean white Gaussian noise with unitary variance.

Signal 2: AR(5) process for which the poles are 0.5, $0.85\exp(\pm j\pi/3)$ and $0.7\exp(\pm j2\pi/3)$.

Signal 3: AR(5) process for which the poles are 0.5, $0.9\exp(\pm j\pi/6)$ and $0.85\exp(\pm j\pi/2)$.

For Signal 2 and Signal 3:

- In the definition of poles, we have $j = \sqrt{-1}$.
- The driven noise is zero-mean white Gaussian with unitary variance.
- The Matlab function **poly** is useful to get the coefficients of a polynomial from its roots.
- Both AR models are taken from B. Farhang-Boroujeny, "Fast LMS/Newton algorithms based on autoregressive modeling and their application to acoustic echo cancellation", IEEE Tr. Signal Processing, vol. 45, no. 8, p. 1987-2000, Aug. 1997.

(c) Generate 25000 samples for Signal 1, Signal 2 and Signal 3, respectively. In each case, pass the signal $\underline{u}$ through the filter $\underline{H}$ to obtain the signal $\underline{d}$.

Solve the estimation problem by applying the Standard LMS, Normalized LMS and Sign Algorithm. Write the Matlab code for recording (in graphical or table form) the main results pertaining to the following experimental situations:

* two different adaptation steps
* two different choices for the number of taps of the adaptive filter.

Show the learning curves by using the same graphical conventions like in the lecture notes.

(d) In the case when $\underline{u}$ is Signal 1, add zero mean, white Gaussian noise with variance σ^2 to the signal obtained at the output of filter $\underline{H}$. This is the new $\underline{d}$ signal. Repeat the experiments from the point (c) for various values of σ^2. Include in the report the experimental results obtained for one single value of σ^2.