BIRZEIT UNIVERSITY

Faculty of Engineering and Technology
Electrical and Computer Engineering
DSP Semester I – 2022/2023 | Course project

Project submission deadline: 10/2/2023 23:55 through Ritaj
Project discussion: To be defined later
Project description:

Project: System Identification

The problem that arises is several applications is the identification of the system or, equivalently, finding its input-output response relationship. To succeed in finding the filter coefficients that represent a model of the unknown system, we set the system configuration as shown in Fig. 1.

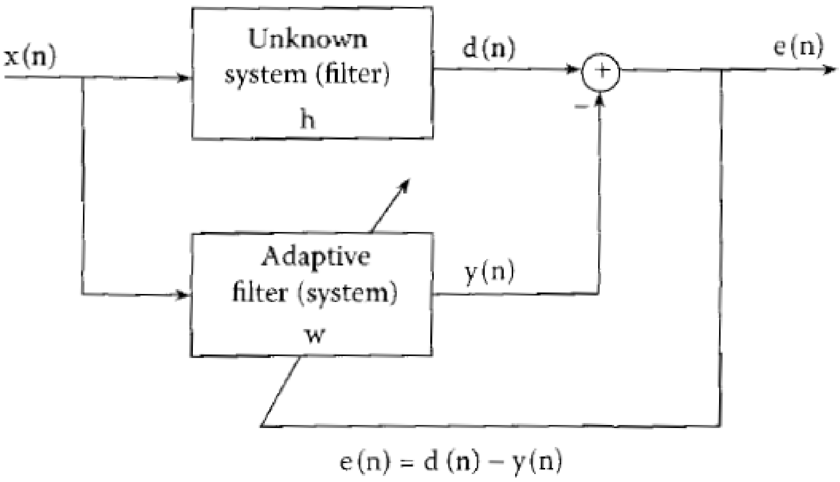


Fig. 1 System Identification

Assuming that the unknown system is time invariant, which indicate that the coefficients of its impulse response are constants and finite such that the desired response is given by

$$d[n] = \sum_{k=0}^{M-1} h[k]x[n-k]$$

The output of an adaptive FIR filter with the same number of coefficients, M, is given by

$$y[n] = \sum_{k=0}^{M-1} w[k]x[n-k]$$

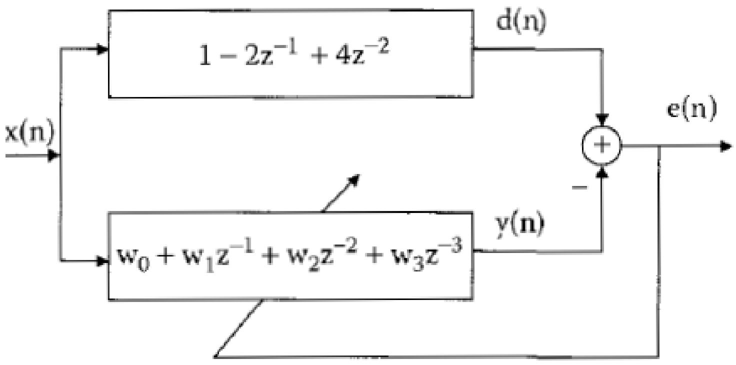
For these two systems to be equal, the difference e[n]=d[n]-y[n] must be equal to zeros. It is the method of adaptive filtering that will enable us to learn the system coefficients and produce an error e[n], approximately equal zero.

Least mean squares (LMS) adaptive filter is an example of the well-known adaptive algorithms. The pseudo LMS adaptive algorithm is (Mth order FIR adaptive filter):

Inputs:M:	filter length
	μ:step-size factor
	x(n): input data to the adaptive filter of length N (Vector)
	w(0):initialization filter (vector) =zeros of length M
Outputs at each iteration (n)	$y(n)=w^T(n)x(n)$
	$e(n)=d(n)-y(n)$
	$w(n+1)=w(n)+2\mu e(n)x(n)$, the updated filter coefficients

<u>Part 1</u>

A) Generate and plot the input x[n]
   $x[n]=\cos(0.03\pi n)$ for N=2000 samples.



B) Plot the amplitude and phase response for the given FIR system.
C) Plot the spectrum for the input signal x[n].
D) Implement the LMS algorithm to estimate the filter coefficients $w_0, \cdots. w_3$
   μ is assumed to be very small (try μ =0.01).

   Plot the learning curves:

   1. The error e(n).
   2. (*J* vs iteration steps). Where *J* is defined to be $J=e^2(n)$.
   3. (10log10(*J*) vs iteration steps).

E) Plot the amplitude and phase response for the estimated FIR system at the end of the iterations. Compare it with the given FIR system.
F) Decrease the value of μ.
   What is the effect of changing μ on the speed of the learning process and on the steady state error?

G) Add 40dB of zeros mean white Gaussian noise to x[n] (hint: use awgn() mathlab function). Repeat parts (D)-(F). Give your conclusions.
H) Repeat part (G) for 30dB. Try to modify the step size value.
I) Ensemble averaging: Repeat part (G) for 1000 trials and average the obtained *J* over the number of trials. plot the averaged *J* (10log10(*J*) vs iteration steps)

Part 2:

Use a different adaptive algorithm from your own, repeat part 1; compare the results for the two algorithms.

Project deliverables by each group:

1- Mini-report as described below in *IEEE template*.

2- System demonstration of each part as described above.

You can use any programming language you prefer for implementing your project. However, we highly recommend MATLAB, Octave, or Python because they have many useful functions.

Dr. Qadri Mayyala & Dr. Alhareth Zyoud
20 – 1 – 2023

About the project:

Teams of three students must do this project. The best arrangement is to choose a division of the project so that each of you can work on separate but interlocking parts. Teams of two or individual work will not be accepted.

Learning teamwork is also one of the more general goals of this course, so team projects will pick up points for demonstrating a successful ability to work with others.

The projects will be graded based on a project report (of around 3-4 pages) as well as in-class short presentations or discussion in the TA office.

Project submission must be via Moodle only, but please use PDF format and **not** Word .DOC files if possible, since we often have formatting problems with Word files.

Your report must have the following structure, using these section headings and using IEEE paper format [will be attached]:

Introduction: A general description of the area of your project and why you are doing it.
Problem Specification: A clear technical description of the problem you are addressing.
Data: What are the real-world and/or synthetic signals you are going to use to develop and evaluate your work?
Evaluation Criteria: How are you going to measure how well your project performs? The best criteria are objective, quantitative, and discriminatory. You want to be able to demonstrate and measure improvements in your system.
Approach: A description of how you went about trying to solve the problem. Sometimes you can make a nice project by contrasting two or more different approaches.
Results and Analysis: What happened when you evaluated your system using the data and criteria introduced above? What were the principal shortfalls? (This may require you to choose or synthesize data that will reveal these shortcomings.) Your analysis of what happened is one of the most important opportunities to display your command of signal processing concepts.
Development: If possible, you will come up with ideas about how to improve the shortcomings identified in the previous section, and then implement and evaluate them. Did they, in fact, help? Were there unexpected side effects?
Conclusions: What did you learn from doing the project? What did you demonstrate about how to solve your problem?
References: Complete list of sources you used in completing your project, with explanations of what you got from each.
The reason for this somewhat arbitrary structure is simply to help you avoid some of the more problematic weaknesses we have seen in past years. If you are having trouble fitting your work into these sections, you should probably think more carefully about your project.