Interactive Teaching of Adaptive Signal Processing [*]

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Abstract. Over the last 30 years adaptive digital signal processing has progressed from being a strictly graduate level advanced class in signal processing theory to a topic that is part of the core curriculum for many undergraduate signal processing classes. The key reason is the continued advance of communications technology, with its need for echo control and equalisation, and the widespread use of adaptive filters in audio, biomedical, and control applications. In this paper we will review the basic theory and applications of adaptive signal processing and also present information on the current multimedia teaching methods on an adaptive signal processing 24 hour module course. In particular we will present some recent world wide web JAVA programmes that were developed for teaching and self-study. We also provide information on how to download and run the JAVA applet.

Keywords. Teaching, Adaptive Signal Processing, JAVA.

1 Introduction

In this paper we review the key theory and applications of adaptive signal processing. Based on the experience of teaching a Master's level Adaptive Signal Processing module over the last 8 years, we focus on our recent development of internet based simulation learning tools. This development follows from our previous efforts at Strathclyde for computer based teaching of adaptive signal processing using a custom windows package [5] and from our knowledge of the pragmatic issues of teaching DSP to practising engineers [6].

Recently we have developed a custom suite of adaptive signal processing algorithms that was written in JAVA to run from the world wide web (WWW). The result is an adaptive filtering learning tool that has proven to be extremely effective in presenting basic and advanced adaptive signal processing concepts. The key aim of the software is to bridge the gap between the theory and mathematics of textbooks and the practical application and implementations of adaptive DSP. The students are able to run the JAVA program from anywhere with Internet access such as University workstations, PCs, or even from home using a modem and PC connected to the WWW. The traditional computer laboratory problems of machine availability, software licensing, portability and so on are clearly circumvented using this approach. The JAVA applet has been used for three years at the University of Strathclyde to teach the Adaptive Signal Processing Master's class and as an example of its portability, in the summers of 1997, 1998, and 1999, the JAVA adaptive suite was successfully used on a course taught by Strathclyde academic staff using a computer laboratory at the University of Los Angeles, UCLA over the internet [7].

In Section 2, we briefly revise the applications of adaptive filtering, describe the outline of the adaptive (JAVA) class and comment on the course. Then, in Section 3, we present the developed JAVA applet and conclude in Section 4. The JAVA applet is freely available on the WWW at `http://www.spd.eee.strath.ac.uk/users/~moritz/algorithmDemo/index.html`.

2 Adaptive Signal Processing

2.1 Applications of Adaptive Signal Processing

Adaptive signal processing is one of the most important classes of algorithms for modern communication systems. Telephone line modems for example now communicate at rates of 56k baud and above as a result of the integration of adaptive echo cancellers and adaptive equalization algorithms. Similarly, the new generation of mobile multimedia systems and set-top boxes will also require the use of adaptive DSP as will adaptive acoustic echo cancellation, arguably the next key “plug-in card” for PCs. Adaptive active noise cancellation is another hi-tech & mature technology found in the cabins of some airliners to reduce the level of noise. More generally, adaptive DSP can be found in biomedical systems, telecommunications systems, industrial control and so on.

Fig. 1 shows the general architectures for the key application areas of (a) noise cancellation, (b) system identification, (c) inverse system identification and (d) prediction. Note the common element in these structures is the general adaptive signal processor, as depicted in Fig. 2, with the input signal $x(k)$, the output signal $y(k)$, the desired signal $d(k)$ and the error signal $e(k)$.

The structure shown in Fig. 1(a) would be employed if a signal $s(k)$ and a corrupting noise $n(k)$ had to be separated, exploiting the presence of a reference of the noise signal $n'(k)$. The system identification structure in Fig. 1(b) can be applied to a number of control problems or to acoustic echo cancellation, where the unknown system would be the transfer function of the teleconferencing room. The set-up given in Fig. 1(c) is a typical inverse system identification scenario which is used, for example, in the equalization problem of telephone lines, where the unknown system is the transfer function of the telephone channel and the adaptive filter should reduce inter symbol interference and other distortions as much as possible. Finally, Fig. 1(d) depicts the adaptive filter in a predictor set-up. There, the filter tries to predict a sample by using a set of past observations. This set-up is commonly used in coders to reduce the redundancy of a data stream, and thereby increases the coding efficiency. Another widepsread example for the application of a predicitor is adaptive line enhancement, where a periodic interference, emanating e.g. from the mains, can be removed from a signal.

2.2 Course Outline

The aim of the Adaptive Signal Processing course is to present its algorithms and architectures and explain their use in real world applications. As prerequisites it is assumed that students have studied discrete and continuous signals and systems, and introductory linear algebra. The syllabus of the course can be summarized as:

- **Introduction to Adaptive Filtering:** a historical perspective; a state of the art perspective.

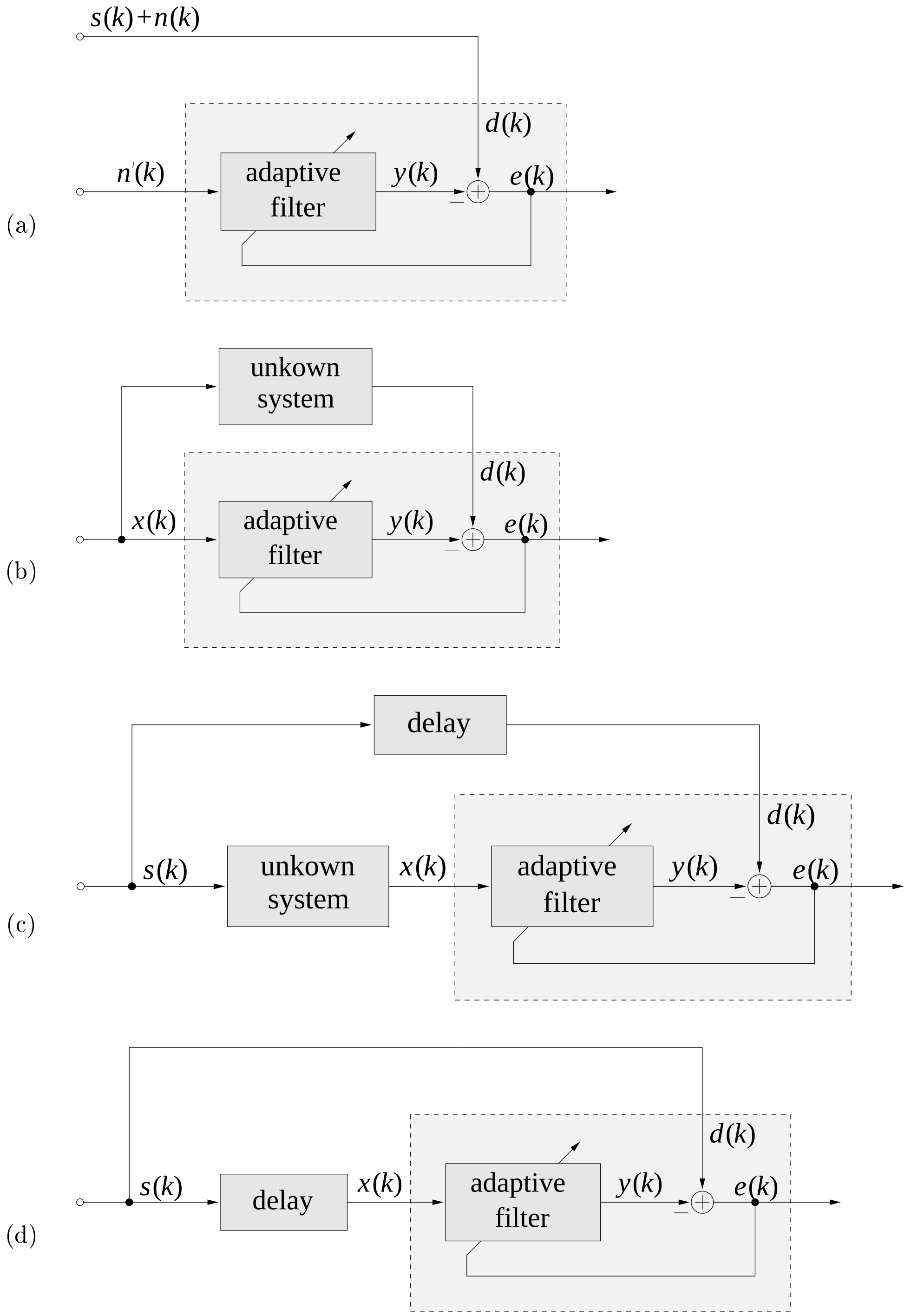
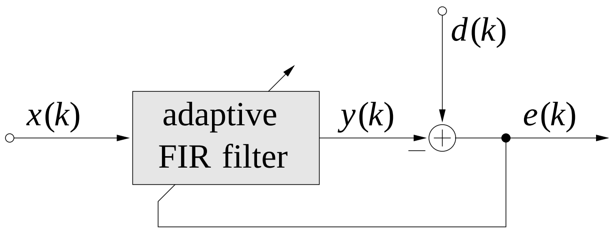


Figure 1: General adaptive filter applications: (a) noise cancellation, (b) system identification, (c) inverse system identification, and (d) prediction.

- **Statistical Signal Processing Revision:** correlation; ergodicity; means, variances; stationarity; wide sense stationarity; periodogramme; frequency response vs. power spectrum.
- **Matrix Algebra Revision:** addition, multiplication and matrix inverses; properties of the correlation/covariance matrix; eigenvalues and eigenvectors; QR algorithm.
- **Wiener Filter Theory:** normal equations; error performance surfaces; orthogonality; minimum mean square errors.
- **The Least Mean Squares (LMS) algorithm:** formulation; convergence; stability criteria. Algorithm variations: normalized algorithm, sign error algorithm, sign data algorithm, leaky algorithm, filtered-X algorithm, variable step-size algorithm.
- **Applications of the LMS:** System identification; room acoustics, control systems; inverse system modeling; modems, telecommunications adaptive equalization, echo cancelation; adaptive beamforming (radar, sonar, hearing aids, listening devices); active noise cancelation systems in cars, airplanes, medical systems, communication systems.
- **Recursive LMS-IIR Algorithms:** output error formulation; equation error formulation; full gradient, simplified gradient, SHARF, Feintuch's algorithm; applications of recursive LMS algorithms.
- **Frequency Domain Adaptive LMS:** Architectures, advantages, and disadvantages.
- **General Least Squares Solution:** Least squares solution of general adaptive system; QR algorithm solution.
- **Recursive Least Squares (RLS) algorithm:** RLS formulation; forgetting factors; practical implementations; QR based RLS; numerical stability and integrity issues.
- **Adaptive Lattice Filters:** gradient lattice, RLS lattice.
- **Non Linear Adaptive Filters:** simple LMS neuron, adaptive polynomial filters (Volterra).
- **Comparative Analysis:** Wiener; LMS-FIR, LMS-IIR; RLS, lattice; frequency domain and neural networks for adaptive signal processing applications
- **Implementation of Adaptive Filters:** DSP microprocessor implementation; software; custom hardware.
- **Application Examples:** Examples for the use and performance of adaptive filters are given and demonstrated by audio demonstrations.

The above list outlines the various adaptive algorithms that are presented on the course, and Figure 1 the different applications considered.

Students are referred to the well known texts [2], [8], [1], [4] for further background in adaptive filter techniques, however none of the texts are compulsory purchases.



$$
\begin{aligned}
y(k) &= \sum_{n=0}^{N-1} w_n(k)x(n-k) = \mathbf{w}^T(k)\cdot\mathbf{x}(k) \\
\mathbf{w}(k+1) &= \mathbf{w}(k) + f(\mathbf{x}(k), \mathbf{w}(k), e(k)) \\
\mathbf{w}(k) &= \begin{bmatrix} w_0(k) & w_1(k) & \cdots & w_{N-1}(k) \end{bmatrix}^T \\
\mathbf{x}(k) &= \begin{bmatrix} x(k) & x(k-1) & \cdots & x(k-N+1) \end{bmatrix}^T
\end{aligned}
$$

Figure 2: General adaptive signal processor.

2.3 Course Presentation and Chronology

Students are first introduced to the general adaptive model shown in Fig. 2. The standard aim of an adaptive signal processing algorithms is to minimize the power of the error signal $e(k)$. This is done by adapting the coefficients of a filter excited by the input signal $x(k)$, such that the filter output $y(k)$ is as similar as possible to some desired signal $d(k)$. It is straightforward to show students that the mathematically most tractable way forward is to minimize the squared error or the mean squared error. From this model, the main adaptive filtering algorithms described in Section 2.2 can be derived and applied to the four main (single channel) adaptive applications in Fig. 1(a)-(d). These applications are then linked and applied to a number of real world adaptive signal processing probems, wfor which a few examples are briefly overviewed in Figs. 3 – 7.

When teaching this course, although many students appear to understand the mathematics of adaptive signal processing, their awareness of how and when to use the various adaptive algorithms were not well developed. When students taking this course were to later undertake an adaptive DSP project, the problems they had highlighted showed the lack of practical and real world experience. For example referring back to Figs. 3–7, although the architectures are all quite intuitive, questions such as the following arise in almost every case:

- How important is the length of the adaptive filter?
- What value should the step size be?
- When do I use LMS-FIR, LMS-IIR or RLS?
- What is the MMSE value?

< Bob's system identification figure >

Figure 3: Adaptive system identification of room acoustics.

< Bob's echo cancellation figure >

Figure 4: Adaptive echo cancellation.

< Bob's noise cancellation figure >

Figure 5: Adaptive noise cancellation.

< Bob's inverse system identification / equalisation figure >

Figure 6: Adaptive inverse system identification / equalisation.

< Bob's active noise cancellation figure >

Figure 7: Adaptive active noise cancellation.

< Bob's adaptive line enhancer figure >

Figure 8: Adaptive prediction / line enhancer.

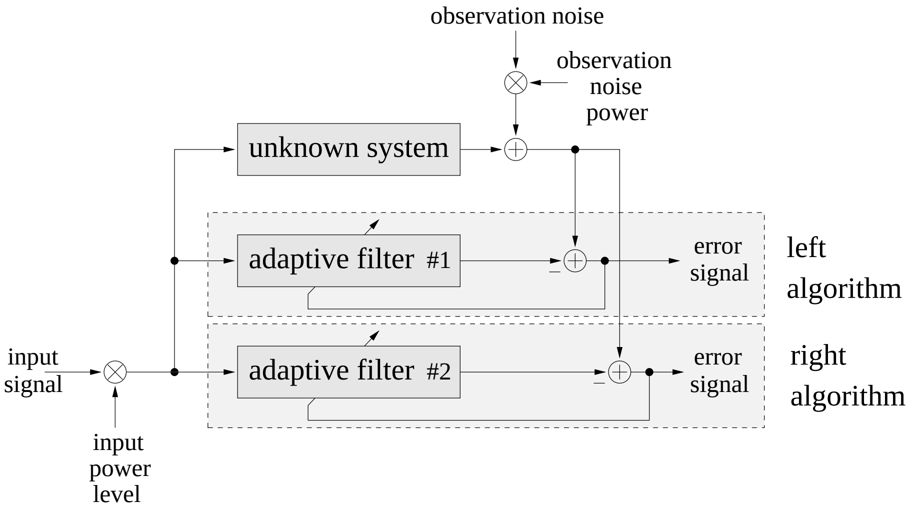


Figure 9: System identification set-up.

- What adapts faster, an LMS, RLS or IIR LMS? How can I see this?
- Which algorithm can track faster changes of the unknown system?
- What happens when a filter goes unstable?
- Does the error really go to zero?

and so on. While a single sentence and mathematical answer is always possible for a specific situation, it was clear the learning process would be greatly enhanced by a software tool that should allow most questions to be answered by the student within just a few minutes of reasoned simulation. Hence the requirement for a software simulation tool is clear. Unfortunately, no such learning tools were readily available.

3 JAVA Applet

To demonstrate the way adaptive filters perform and to give the student an insight into how the different parameters affect the convergence and steady-state behavior of the algorithms, a JAVA applet has been developed which compares two adaptive algorithms for FIR filters in a system identification set-up, c.f. Fig. 1(b). The JAVA environment was chosen for this task as it is platform independent, i.e. the applet can be used on every computer system where a JAVA capable browser is available. Furthermore, it can be downloaded easily via the Internet so that computing facilities at the university, at remote sites or even at home can be used for teaching purposes. In our teaching, we have also stored the applet on portable PCs and presented it directly in class for demonstrative purposes.

The internal set-up of the demonstration and teaching applet is shown in Fig. 9. It consists of the unknown system, which is represented by a 10 tap finite impulse response (FIR) system, and two


Unknown System
Left Algorithm
Right Algorithm
Sampling Frequency