

RAPID PROTOTYPING LIBRARY FOR ADAPTIVE SIGNAL PROCESSING APPLICATIONS

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ABSTRACT

In this paper we present a library for the rapid prototyping of adaptive signal processing algorithms, architectures and applications. The library is hosted by the DSP simulation software *SystemView* and covers virtually the complete spectrum of linear, and non-linear adaptive algorithms currently in use in contemporary DSP and communications applications. The library can be easily used with real signals, with variable system wordlengths, sampling frequencies and so on. Therefore in this paper we will briefly discuss the design philosophy behind the library and overview the various algorithms and applications that are implemented. The paper will show an implementation of an adaptive multiuser CDMA receiver/ decision feedback equaliser (DFE) as an example of the relevance and rapid development that is possible. Copies of the library and example files can be downloaded from the web following the instructions in the paper.

1. INTRODUCTION

Over the last 25 years the original work of Widrow [3] in defining the least mean squares (LMS) algorithm and architecture has lead to literally tens of thousands of conference, journal and industrial reports on performance, application and usefulness of the LMS. The LMS algorithm is in daily use in modems, landline and digital mobile telephones, teleconferencing, and even in Windows95 (ADPCM audio wav file compression uses an IIR-LMS for prediction). Looking to the next 5 years, the relevance of adaptive signal processing for the emerging W-CDMA standards (in terms of data equalisation, echo control and multiuser adaptive receivers), and in the implementation of acoustic echo controllers/noise suppressors for hands-free communication means that the LMS is likely to continue to be researched and developed in academia and in industry. However, it is arguable if there is much more research and development to be performed, and the key requirement is for efficient deployment and implementation of the adaptive signal processing.

Surprisingly there do not seem to be any comprehensive commercial adaptive libraries available for the main DSP simulation packages used within industry. Therefore we have developed a very flexible adaptive signal processing library in order that rapid prototyping and implementation of adaptive algorithms can be undertaken.

The algorithms in the library cover the key algorithmic strategies found in the literature over the last few years, namely [6]:

- Least Mean Squares (LMS)
- Recursive Least Squares (RLS) / QR
- Kalman
- Newton Gradient Techniques
- “Open Loop” (Wiener Hopf, Least Squares)

The structure of the library elements is such that FIR, IIR, and multichannel implementations using any of the above classes of algorithms is straightforward to implement. Figure 1 provides an overview of the various algorithms that can be directly implemented via parameter and variable specification.

The objectives in developing the library were to allow a user to:

- Rapidly develop a real world relevant simulation using appropriate adaptive signal processing algorithms;
- Investigate systems for stability & algorithm performance;
- Setup algorithm variants (variable step size, updating rate etc.);
- Consider alternative to LMS algorithms (RLS, Kalman, QR etc);
- Undertake a *bit true* real world relevant simulation.

This paper briefly reviews the library components, and presents a number of developed applications. A comprehensive set of adaptive applications and an evaluation version of the library can be downloaded from <http://www.entegra.co.uk/icassp99.htm>.

2. DSP SIMULATION WITH SYSTEMVIEW

The adaptive library was developed to run within the DSP simulation tool, SystemView by Elanix [1]. SystemView is a token based simulator for Windows95/98/NT and has the capability to allow professional development of complex DSP and communication system simulations, and to provide support for post-simulation data analysis. A demonstration version of the tool is available on the web at <http://www.elanix.com>.

By interconnection of block diagram type tokens SystemView allows the interconnection of more complex systems typically including digital filters, transform coders, quantisers, samplers,

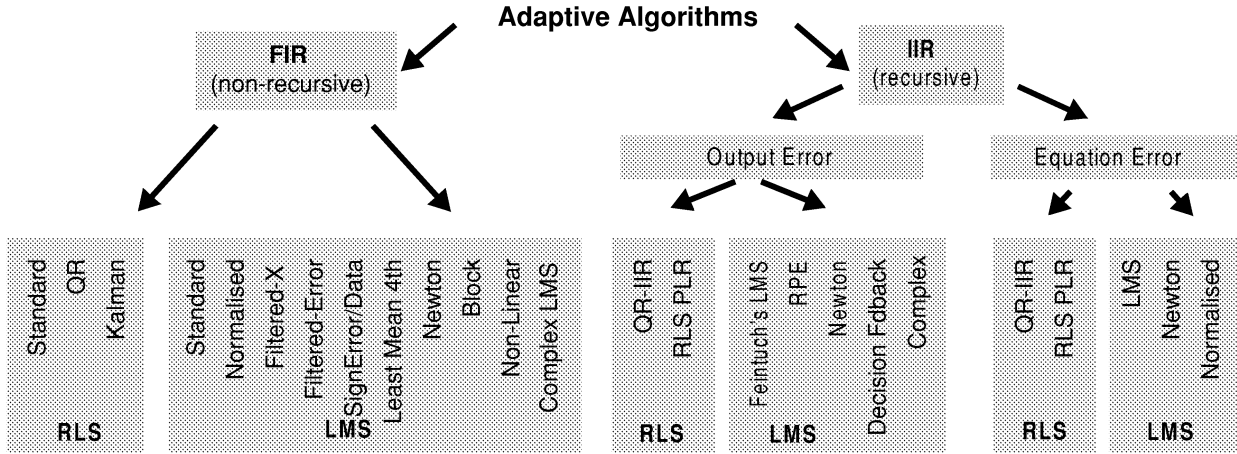


Figure 1: Adaptive library algorithms.

non-linear elements, modulators, and so on. Simulation source data can be generated by source tokens (sine, Gaussian, PRBS etc) and system outputs sent to suitable sink tokens whereupon signals can be post-analysed statistically, using frequency domain analysis tools and so on. Figure 7 (discussed later) presents a screenshot of a SystemView simulation with a number of interconnected tokens.

3. THE ADAPTIVE LIBRARY

In this section we will summarize the algorithm capability of the adaptive library, briefly review the philosophy behind implementation, and thereafter present implementations of the library.

3.1 The LMS Algorithm

The most general tokens in the adaptive library are the Least Mean Squares (LMS) tokens. These tokens allow straightforward implementation of the LMS and its many variants.

For the generic FIR LMS architecture [2] shown in Figure 2, the general filter weight update equation is given by:

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu(-\hat{\nabla}_k) \quad (1)$$

where $\mathbf{w}(k) = [w_0, w_1, \dots, w_{N-1}]^T$

The LMS gradient estimate [3] is well known to be:

$$\hat{\nabla}_k = \frac{de^2(k)}{d\mathbf{w}(k)} = 2e(k) \left[\frac{de(k)}{d\mathbf{w}(k)} \right] = -2e(k) \left[\frac{dy(k)}{d\mathbf{w}(k)} \right] = -\mathbf{x}(k) \quad (2)$$

and therefore the LMS algorithm is:

$$\mathbf{w}(k+1) = \mathbf{w}(k) + 2\mu e(k)\mathbf{x}(k) \quad (3)$$

where $\mathbf{x}(k) = [x(k), x(k-1), \dots, x(k-N+1)]^T$. The LMS algorithm and its many variants are of course used in literally hundreds of applications ranging from simple linear prediction for speech coding to complex arithmetic decision feedback equalisers (DFE) for quadrature channel modulation.

3.2 The General LMS Adaptive Token

The general LMS (titled “GenLMS”) adaptive token of the library implements:

$$\mathbf{w}(k+1) = c\mathbf{w}(k) + 2\mu e(k)\mathbf{g}(k) \quad (4)$$

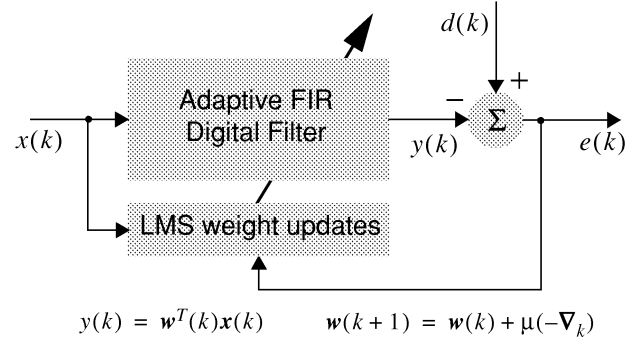


Figure 2: The Generic LMS processor

Figure 4 illustrates the internal signal flow graph connections for one of the general LMS tokens. $\mathbf{g}(k)$ is an N element vector, i.e. the same dimension as the no. of weights in the filter. (The constant c is the leakage factor [2].) In order to implement the standard LMS of Eq. 3 we simply set $\mathbf{g}(k) = \mathbf{x}(k)$ and calculate $e(k)$ as shown in Figure 3 which shows the token interconnects of SystemView required to produce the generic LMS of Figure 2.

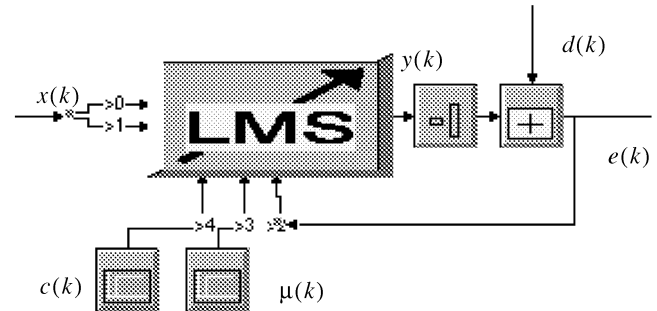


Figure 3: General LMS SystemView implementation.

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System Identification	Inv Sys Identification	Prediction	Noise Cancellation
• Telephone channel ident. • Modem Echo Canc. • Acoustic echo control • Radio channel ident. • Room acoustic ident.	• Channel equalisation • Decision Directed Equ. • Blind Equalisation • Decision Feedback • Fractional Equalisers	• CDMA Intefrence Sup • LPC Speech Coding • Period Noise Suppress • Spectral Whitening • Adaptive CDMA receiver	• Active noise control • Background noise • ECG noise control • Acoustic beamforming • Multimedia noise control

Figure 6: Applications simulated with the SystemView adaptive library.

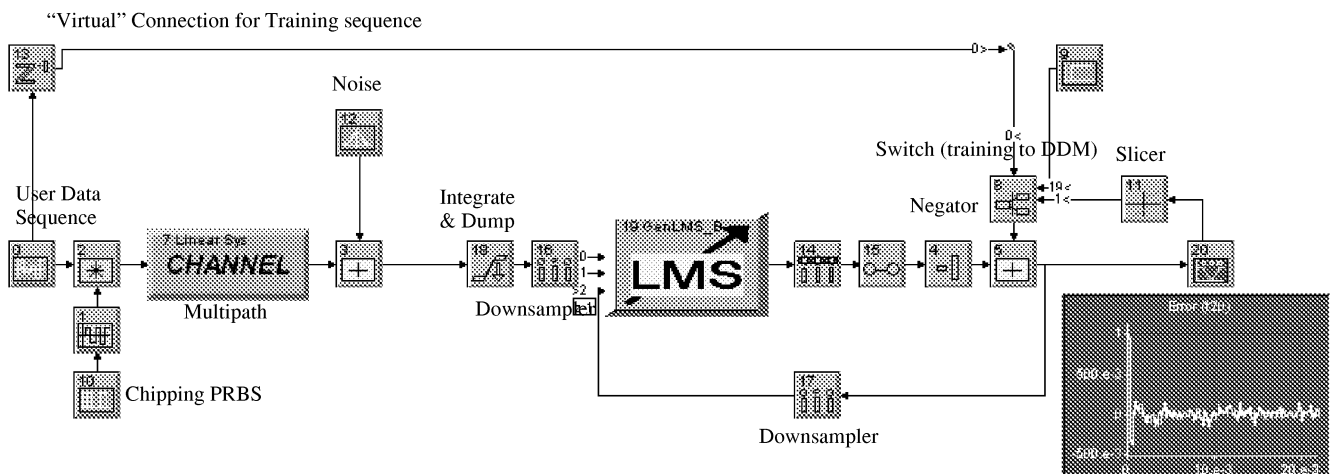


Figure 7: Adaptive multiuser receiver for single user CDMA.