"I hereby agree that I have read this article and in my opinion it is sufficient in terms of scope and quality to be awarded Bachelor Degree of Electronic Engineering (Industrial Electronic)."

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Date	:	21st MARCH 2005

ADAPTIVE INFINITE IMPULSE RESPONSE (IIR) FILTER

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This Report Is Submitted In Partial Fulfillment Of Requirements For The Bachelor Degree of Electronic Engineering (Industrial Electronic)

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March 2005

"I hereby declare that this thesis is based on my effort except for some short notes in which every references has been given".

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Date : 21st MARCH 2005

DEDICATED

To

My Greatest GOD, APPA and AMMA,

My Best Brother, NARENDRAN,

My Supportive Sisters & Sister In Law, DEEPA, SANNA, DURGA & REVATHI.

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ABSTRACT

This paper presents an implementation of Infinite Impulse Response (IIR) Least Mean Squares (LMS) algorithm. The purpose of this project is to design a program to prove that by using the IIR-LMS algorithm will eliminate/reduce the Inter-Symbol Interference (ISI) problem. The principal cause of ISI is delay distortion. Because the individual frequency components propagate at different speeds over the transmission media, they become dispersed, causing changes in amplitude and phase that produce pulse distortion. Meanwhile, the learning algorithm of the adaptive IIR filter is used to adjust the feedback and feed forward coefficients for a particular input and output to optimize a performance criterion that generates a suitable estimate based on a desired response. The convergence of the LMS algorithm depends on the choice of the step-size values for the filter coefficients. The implementation of the IIR-LMS algorithm had minimized the mean-squared error. It is achieved to reduce the ISI problem in the system through simulation results which implicated the LMS algorithm.

ABSTRAK

Laporan ini mendedahkan tentang pelaksanaan penapis suai *Infinite Impulse Response* (IIR) algoritma *Least Mean Squares* (LMS). Tujuan projek ini adalah untuk merekabentuk perisian untuk membuktikan bahawa penggunaan IIR-LMS algoritma akan menyelesaikan masalah ISI. Punca utama terjadinya ISI ialah lengah masa sesuatu perambatan. Disebabkan oleh setiap komponen frekuensi yang tersebar pada kelajuan yang berbeza signal yang dihantar melalui media transmisi akan tersebar kepada beberapa komponen frekeunsi yang mempunyai kelajuan yang berbeza, di mana ini menyebabkan perubahan pada amplitud dan fasa dan seterusnya menghasilkan pengherotan denyut. Algoritma penapis suai IIR digunakan untuk melaraskan *feedback* dan *feedforward* koefisien bagi masukan dan keluaran penapis untuk menjanakan anggaran yang sesuai berdasarkan sambutan yang dikehendaki. Penumpuan algoritma LMS bergantung kepada pemilihan saiz langkah dan nilai permulaan bagi koefisien penapis. Penglibatan algoritma LMS-IIR dapat meminimumkan ralat purata kuasa dua (MSE). Daripada simulasi yang melibatkan algoritma LMS dijalankan, didapati ia dapat mengurangkan masalah ISI pada sistem.

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LIST of ABBREVIATIONS

CLMS	-	constrained LMS
CDMA	-	code division multiple access
CRLS	-	constrained RLS
FIR	-	finite-duration impulse response
IIR	-	infinite-duration impulse response
ISI	-	intersymbol interference
LMS	-	least-mean squares
LS	-	least-squares
MMSE	-	minimum mean-squared error
MSE	-	mean-squared error
NCLMS	-	normalized constrained LMS
NLMS	-	normalized least mean squares
RLS	-	recursive least-squares

LIST of SYMBOLS

$x(k)$	-	input signal sequence
x(k)	-	input signal vector
y(k)	-	output signal sequence
$d(k)$	-	desired signal sequence
$e(k)$	-	output error sequence
$n(k)$	-	noise sequence
w(k)	-	vector of coefficients of the adaptive filter
N	-	Number of filter coefficients
Jw	-	objective function
$?_i$	-	direction of arrival of user i
μ	-	step size
d(k)	-	desired signal vector
$E[e[n]^2]$	-	mean squared error

LIST of APPENDIX

CHAPTER I

INTRODUCTION

1.1 Digital Signal Processing

Digital signal processing become known since middle of the decade 1960 where high speed of computers was widely been used. These developments have brought to a few concept of basic digital signal processing for example z-transform, Fourier analysis, modeling and ext. which exist from those days where could be practice now easily.

Digital signal processing involves signals and discrete time system. This field is divided into two categories, that is, digital filter and spectrum analysis. Digital filter could function as analog filter but analog approach is not suitable to perform in certain critical cases.

Today, there are many commercial products around, utilizing the advantages of digital signal processing, namely:

- an essentially perfect reproducibility
- a guaranteed accuracy (no individual tuning and pruning necessary)
- Well suited for large volume production.

Digital signal processing is heavily used in many telecommunications systems today. For instance, they are used in telephone systems for DTMF (dual-tone multi-frequency) signaling, echo canceling of telephone lines and equalizers used in high- speed telephone modems. Further, error-correcting codes are used to protect digital signals from bit errors during transmission (or storing) and different data compression algorithms are utilized to reduce the number of data bits needed to represent a given amount of information.

1.2 Objective

The objective of this project is to design a program for adaptive Infinite Impulse Response (IIR) filter. IIR-LMS algorithm is used to adjust/update the filter weight to gain an optimum filter. Moreover it is to prove that the IIR-LMS algorithm can be used to eliminate the ISI problem. MATLAB software had been used as a tool in developing the program. Last but not the least; it is to obtain knowledge and skills on the implementation and application of adaptive filters.

1.3 Scope of Work

The scope of work involves implementation of adaptive IIR filter using adaptive algorithm, which will be the LMS algorithm for ISI elimination. Besides that, the scope also involves creating an input signal, corrupted with ISI signal, system of adaptive IIR filter and output through simulation.

1.4 Thesis Outline

The thesis is presented with five chapters. Chapter two presents the literature review which covers all the basics theory and concept of this project.

Chapter three presents the implementation of the project. In this chapter it has been describe in detail about the methodology and the implementation of the project.

Chapter four displays the results gained through simulation using MATLAB.

Finally, chapter five presents discussions and conclusions for the whole project. It also outlines future work and recommendations.

CHAPTER II

LITERATURE REVIEW

2.1 Adaptive Filters

The adaptive filters can have either finite-duration impulse response (FIR) or infinite duration impulse response (IIR). In this thesis, IIR filter had been utilized. An adaptive filter is useful whenever the statistics of the input signals to the filter are unknown or time varying and the design requirements for fixed filters cannot easily be specified [1]. Examples of such applications are: system identification [2], channel equalization/identification and interference suppression in communications systems [3, 4, 5, 6, 7, 8], and acoustic echo cancellation [9]. The adaptive filter measures the output signal of the filter, and compares it to a desired output signal dictated by the true system. By observing the error between the output of the adaptive filter and the desired output signal, an adaptation algorithm updates the filter coefficients with the aim to minimize an objective function. Figure 2.1 shows the basic schematic diagram of an adaptive filter, where $x(k)$, $y(k)$, $d(k)$, and $e(k)$ are the input, output, desired output and error signals of the adaptive filter for time instant k. As can be seen from Figure 2.1, the adaptive filter is a nonlinear filter through its dependence on the input signal, although, at a given time instant it will act as a linear filter.

The adaptive filter can have either finite-duration impulse response (FIR) or infinite duration impulse response (IIR). For an adaptive FIR filter the output is obtained as a linear combination of the present and the $N - 1$ past input signal samples, N being the number of filter coefficients. An adaptive FIR filter is many times preferred over an adaptive IIR filter due to its simplicity and robustness. Furthermore, many practical problems can be accurately modeled by an FIR filter, e.g., channel identification in communications systems [5, 9]. The adaptive IIR filter can serve as a viable alternative to the FIR in applications where the required order of the adaptive filter is very high, since an IIR filter in general requires fewer filter coefficients than its FIR filter counterpart [1]. Drawbacks of the adaptive IIR filter include possible stability problems and, in certain problems, lack of a unique solution [1].

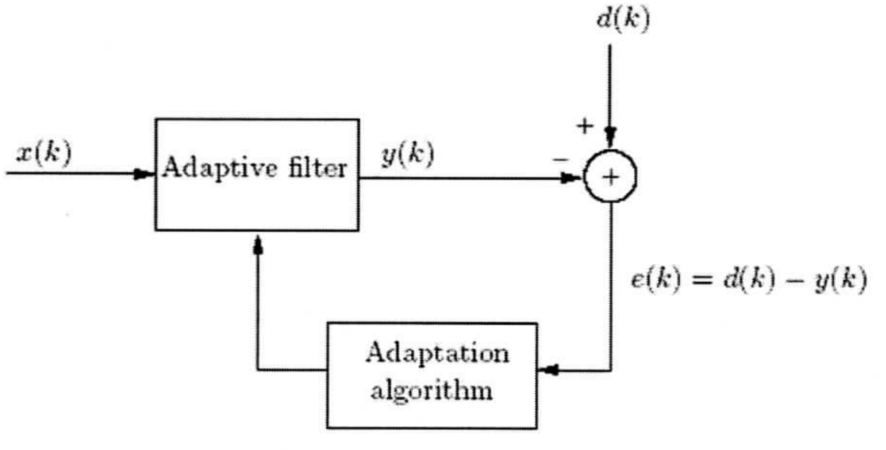

Figure 2.1: Schematic diagram of an adaptive filter

As already mentioned, the adaptation algorithm tries to minimize an objective function J_w, which is often related to the output error. Among the most common objective functions that are used for derivation of adaptation algorithms are:

- The mean-squared error (MSE) having $J_w = E[e^2(k)]$;
- The least-squares (LS) having $J_w = 1/k \cdot \sum_{i=1}^{k} e^2(i)$;

Choosing among the many different objective functions often involves a trade-off between certain conflicting performance measures. Some of the most important performance measures related to adaptive filters are [1]:

- The convergence rate, i.e., the number of algorithm iterations required to converge to the vicinity of a steady-state solution.
- The accuracy of the obtained solution as compared to the optimal obtainable solution. An often used measure is the excess MSE, or the misadjustment, which quantifies how close the adaptive filter coefficients are to the ones of the optimal filter.
- The computational complexity of the algorithm.
- Robustness to quantization when implemented in finite-precision.
- Tracking ability, i.e., the performance of the filter when operating in a non stationary environment.

As previously stated, these performance measures are often conflicting and as a consequence, specifications on the adaptive filter in terms of these measures cannot in general be met simultaneously. For example, fast convergence rate usually implies computationally demanding implementation. On the other hand, if low misadjustment is desired, an algorithm of low computational complexity would most likely suffer from slow convergence.

2.2 Adaptive IIR Filter

Infinite impulse response filter gets its name because its impulse response extends for an infinite period of time. This is because they are recursive, i.e., they utilize feedback [10]. Each element of the set of the output numbers is calculated by weighted summation of a certain number of elements of the input set and of the previous output set. [11] The basic concept of an adaptive IIR filter is shown in Figure 2.2. The objective is to filter the input signal, x(k) with an adaptive filter in such a manner that it matches the desired signal, d(n). The d(n) is subtracted from the filtered signal, y(k) to generate an error signal, e(k). The error signal drives an adaptive LMS Algorithm which generates the filter coefficients in a manner which minimize the error signal. The convergence of the LMS algorithm very much depends on the choice of the step-size and the choice of initial values for the filter coefficients.

The algorithm is used to adjust the feed-back and feed-forward coefficients. The algorithm used in this project is defined as IIR-LMS algorithm where the weights are updated with Feintuch's algorithm. This algorithm is discussed in the following chapter.

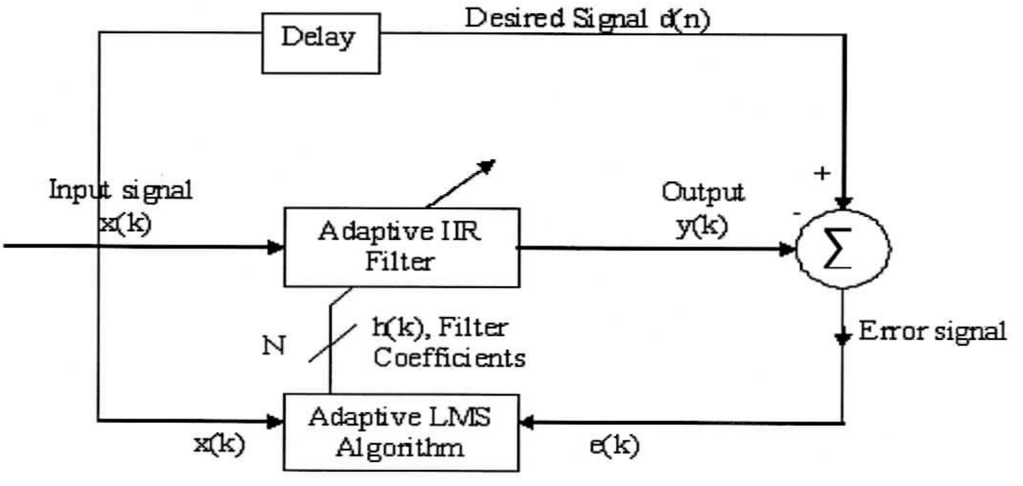

Figure 2.2: Adaptive IIR filter

2.3 Adaptive DSP Applications

Adaptive filters are widely used in our daily live nowadays. Adaptive filters can be used for the design and simulation of a wide range of adaptive algorithms. Figure 2.3 shows the standard adaptive architectures of System Identification, Inverse System Identification, Noise Cancellation and Prediction [12]. Even though each of them is different structures, they still use the same basic concept of adaptive. Input and desired vector is used in determining the estimate error.

x = input filter

y = output filter

d = desired signal

$e = d - y$ = error

Below are some examples of the adaptive applications:

System Identification

- Telephone channel identification
- Modem (V32) Echo Cancellation
- Hands-free acoustic echo control
- Mobile radio channel identification

Inverse System Identification

- Inverse channel identification
- Adaptive data equalization
- (Modem) Decision directed equalization (blind equalization)
- Mobile Radio Decision feedback equalization (DFE)
- Adaptive CDMA Multiuser Receiver

Noise Cancellation

- ECG mains hum noise suppression
- Car engine background noise suppression
- Active noise control