IMPLEMENTATION OF RECURSIVE MAXIMUM LIKELIHOOD (RML) ALGORITHM IN DESIGNING ADAPTIVE NOTCH FILTER

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ABSTRACT

Digital filter is an electronic filter that works by performing digital mathematical operations on an intermediate form of a signal. There are two types of filters such as Infinite Impulse Response (IIR) filter and Finite Impulse Response (FIR) filter. In order to keep tracking the frequency changes in the input signal, the adaptive filter parameter must be updated recursively. However, the parameter estimation in conventional algorithm must be monitored to enforce convergence. Thus, in this project, the Recursive Maximum Likelihood (RML) algorithm is proposed for implementation due to its ability to enforce faster convergence without monitoring. The main objectives from this project are to investigate the use of Recursive Maximum Likelihood (RML) algorithm in designing adaptive filter. Besides, this project implements the algorithm in MATLAB. Furthermore, it is aimed to produce a working program code in MATLAB

ABSTRAK

Penapis berdigit adalah penapis elektronik yang dipersembahkan dalam operasi matematik berdigit dalam isyarat. Ini adalah dua jenis penapis iaitu tindakbalas desakan terhad dan tindakan desakan yang tidak terhad. Untuk mengesan salah satu frekuensi dalam pelbagai frekuensi pada isyarat masukan, parameter penapis yang sesuai mestilah dikemaskinikan. Walaubagaimanapun, penaksiran parameter dalam logaritma biasa mestilah di kawal untuk menguatkuasakan penumpuan. Oleh sebab itu, dalam projek ini RML (Recursive Maximum Likelihood) digunakan bertujuan untuk melaksanakan kebolehan logaritma untuk mengawal penguatkuasakan penumpuan dengan pantas. Tujuan utama projek ini dijalankan adalah untuk menyiasat penggunaan RML logaritma dalam merekabentuk penapis takuk. Selain itu, projek ini boleh dilaksanakan ke dalam perisian MATLAB. Dan juga untuk menghasilkan kod program yang betul di dalam MATLAB.

CONTENT

LIST OF ACRONYMS AND SYMBOLS

ADC	-	Analog to Digital Converter
BIBO	-	Bounded Input-Bounded Output
DAC	-	Digital to Analog Converter
DSP	-	Digital Signal Processing
FFT	-	Fast Fourier Transform
FIR	-	Finite Impulse Response
FRLS	-	Fast Recursive Least Square
IIR	-	Infinite Impulse Response
LMS	-	Least Mean Square
RF	-	Radio Frequency
RML	-	Recursive Maximum Likelihood
RPE	-	Recursive Prediction Error

LIST OF FIGURES

CHAPTER I

INTRODUCTION

The project title is Implementation of Recursive Maximum Likelihood (RML) Algorithm in Designing Adaptive Notch Filter using MATLAB.

1.1 Objectives

There are the objectives of this project:

i) To investigate the use of RML algorithm in designing adaptive filter.
ii) To implement the algorithm in MATLAB.
iii) To produce a working program code in MATLAB.

1.2 Problem Statement

In order to keep tracking the frequency changes in the input signal, the adaptive filter parameter must be updated recursively. However, the parameter estimation in this algorithm must be monitored to enforce convergence. Thus, in this project, the RML algorithm is proposed for implementation due to its ability to enforce faster convergence without monitoring.

1.3 Scopes of Work

There are the scopes of work from this project:

i) Study and understanding on digital filter, adaptive filter and RML algorithm theories.
ii) Implementation of the RML algorithm in designing adaptive filter using MATLAB.
iii) Performance analysis of the implemented algorithm.

1.4 Thesis Layout

Chapter 2 presents about the literature review on the Time Domain, Frequency Domain, Fast Fourier Transform (FFT), Analog and Digital Filters, Infinite Impulse Response (IIR) Filters, Finite Impulse Response (FIR) Filters and RML Algorithm.

Chapter 3 is the project methodology which is the flow of the project. Start from literature review for background knowledge and theory of filters, digital filter, adaptive filter and RML algorithm. After that, the FIR and IIR filter are implemented in MATLAB. Furthermore, the RML algorithm in designing adaptive notch filter is implemented. After the result and analysis is done, came thesis writing. Lastly, project presentation is performed in last session.

Chapter 4 is about the result and discussion for FFT, IIR, FIR and when applying RML algorithm in adaptive filter using MATLAB. The frequency that used is 40Hz and 100Hz which is to compare the different result.

Chapter 5 is about conclusion and the suggestions for future works on this project.

CHAPTER II

LITERATURE REVIEW

For this chapter literature review about the Time Domain, Frequency Domain, FFT, Analog and Digital Filters, IIR Filters, FIR Filters and RML Algorithm are done.

2.1 Time Domain

The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. Filtering generally consists of some transformation of a number of surrounding samples around the current sample of the input or output signal. There are various ways to characterize filters; for example:

- A *linear* filter is a linear transformation of input samples; other filters are *non-linear*. Linear filters satisfy the superposition condition, i.e. if an input is a

- weighted linear combination of different signals; the output is an equally weighted linear combination of the corresponding output signals.

- A *causal* filter uses only previous samples of the input or output signals; while a *non-causal* filter uses future input samples. A non-causal filter can usually be changed into a causal filter by adding a delay to it.

- A *time-invariant* filter has constant properties over time; other filters such as adaptive filters change in time.

- Some filters are *stable*, others are *unstable*. A stable filter produces an output that converges to a constant value with time, or remains bounded within a finite interval. An unstable filter produces output which diverges.

- A FIR filter uses only the input signal, while an IIR filter uses both the input signal and previous samples of the output signal. FIR filters are always stable, while IIR filters may be unstable [1].

2.2 Frequency Domain

Signals are converted from time or space domain to the frequency domain usually through the Fourier transform. The Fourier transform converts the signal information to a magnitude and phase component of each frequency. Often the Fourier transform is converted to the power spectrum, which is the magnitude of each frequency component squared.

The most common purpose for analysis of signals in the frequency domain is analysis of signal properties. The engineer can study the spectrum to get information of which frequencies are present in the input signal and which are missing.

There are some commonly used frequency domain transformations. For example, the converts a signal to the frequency domain through Fourier transforms, takes the logarithm, and then applies another Fourier transform. This emphasizes the frequency components with smaller magnitude while retaining the order of magnitudes of frequency components [1].

2.3 FFT

FFT based measurements are subject to errors from an effect known as leakage. This effect occurs when the FFT is computed from of a block of data which is not periodic. To correct this problem appropriate windowing functions must be applied. The user must choose the appropriate window function for the specific application. When windowing is not applied correctly, then errors may be introduced in the FFT amplitude, frequency or overall shape of the spectrum [2].

The FFT is the Fourier Transform of a block of time data points. It represents the frequency composition of the time signal. Figure 2.1 shows a 10 Hz sine waveform (top) and the FFT of the sine waveform (bottom). A sine wave is composed of one pure tone indicated by the single discrete peak in the FFT with height of 1.0 at 10 Hz.

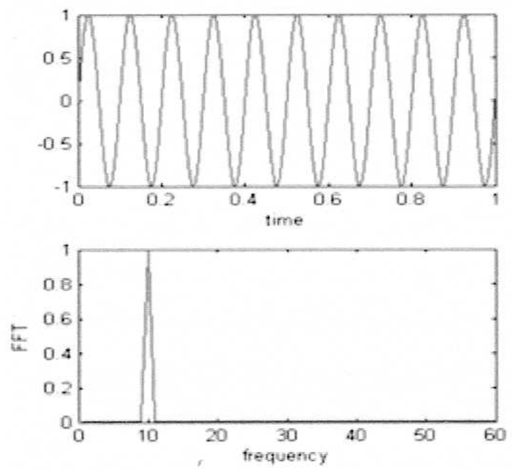

Figure 2.1: Time Waveform of Sine Function (top) and FFT (bottom) [2].

2.4 Analog and Digital Filters

In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range [3].

The following block diagram illustrates the basic idea.

Figure 2.2: Block diagram of a filter [3].

There are two main kinds of filter, *analog* and *digital*. They are quite different in their physical makeup and in how they work.

An analog filter uses analog electronic circuits made up from components such as resistors, capacitors and op amps to produce the required filtering effect. Such filter circuits are widely used in such applications as noise reduction, video signal enhancement, graphic equalizers in hi-fi systems, and many other areas.

There are well-established standard techniques for designing an analog filter circuit for a given requirement. At all stages, the signal being filtered is an electrical voltage or current which is the direct analogue of the physical quantity (e.g. a sound or video signal or transducer output) involved [3].

2.4.1 Digital Filter

In electronics, a digital filter is any electronic filter that works by performing digital mathematical operations on an intermediate form of a signal. This is in contrast to older analog filters which work entirely in the analog realm and must rely on physical networks of electronic components (such as resistors, capacitors, transistors, etc.) to achieve the desired filtering effect [3].

Digital filters can achieve virtually any filtering effect that can be expressed as a mathematical function or algorithm. The two primary limitations of digital filters are their speed (the filter can't operate any faster than the computer at the heart of the filter), and their cost. However as the cost of integrated circuits has continued to drop over time, digital filters have become increasingly commonplace and are now an essential element of many everyday objects such as radios, cell phones, and stereo receivers.

The analog input signal must first be sampled and digitized using an analog to digital converter (ADC). The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output through a digital to analog converter (DAC) to convert the signal back to analog form [3].

Note that in a digital filter, the signal is represented by a sequence of numbers, rather than a voltage or current.

The following diagram shows the basic setup of such a system.

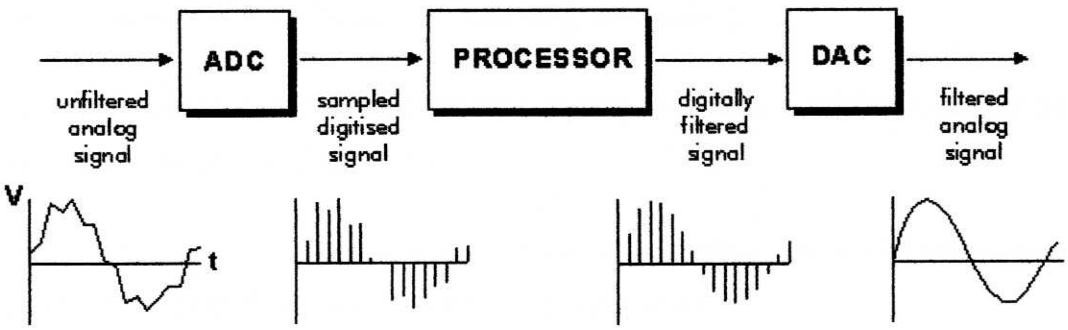

Figure 2.3: The basic setup of such a system [3].

2.4.2 Operation of Digital Filter

In the next few sections, we will develop the basic theory of the operation of digital filters. This is essential to an understanding of how digital filters are designed and used. First of all, we need to introduce a basic notation.

Suppose the "raw" signal which is to be digitally filtered is in the form of a voltage waveform described by the function

$$V = x(t) \qquad (2.1)$$

where *t* is time.

This signal is sampled at time intervals *h* (the sampling interval). The sampled value at time $t = ih$ is

$$x_i = x(ih) \qquad (2.2)$$

Thus the digital values transferred from the ADC to the processor can be represented by the sequence

$x_0, x_1, x_2, x_3, \ldots$corresponding to the values of the signal waveform at times $t = 0, h, 2h, 3h, \ldots$ (where $t = 0$ is the instant at which sampling begins).

At time $t = nh$ (where *n* is some positive integer), the values available to the processor, stored in memory, are $x_0, x_1, x_2, x_3, \ldots, x_n$

Note that the sampled values x_{n+1}, x_{n+2} etc. are not available. The digital output from the processor to the DAC consists of the sequence of values

$$y_0, y_1, y_2, y_3, \ldots, y_n \qquad (2.3)$$

In general, the value of y_n is calculated from the values $x_0, x_1, x_2, x_3, \ldots, x_n$. The way in which the y's are calculated from the x's determines the filtering action of the digital filter [3].

2.4.3 Advantages of Digital Filters

The following list gives some of the main advantages of digital over analog filters.

i) A digital filter is programmable, i.e. its operation is determined by a program stored in the processor's memory. This means the digital filter can easily be changed without affecting the circuitry (hardware). An analog filter can only be changed by redesigning the filter circuit.

ii) Digital filters are easily designed, tested and implemented on a general-purpose computer or workstation.

iii) The characteristics of analog filter circuits (particularly those containing active components) are subject to drift and are dependent on temperature. Digital filters do not suffer from these problems, and so are extremely stable with respect both to time and temperature.

iv) Unlike their analog counterparts, digital filters can handle low frequency signals accurately. As the speed of Digital Signal Processing (DSP) technology continues to increase, digital filters are being applied to high frequency signals in the radio frequency (RF) domain, which in the past was the exclusive preserve of analog technology.

v) Digital filters are very much more versatile in their ability to process signals in a variety of ways; this includes the ability of some types of digital filter to adapt to changes in the characteristics of the signal.

Fast DSP processors can handle complex combinations of filters in parallel or cascade (series), making the hardware requirements relatively simple and compact in comparison with the equivalent analog circuitry.