DEVELOPMENT OF DIGITAL FILTER DESIGN TOOL

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ABSTRACT

The main idea of this project is to design an easy configuration for the filter courses as tool for teaching aid using the Graphical User Interface (GUI) in MATLAB Software. In simple interpretation, the focus is to develop a filter design tool with user-friendly interface. Because there are large scopes of filter design method, thus in this project, the IIR (Infinite Impulse Response) digital filter design was chosen as the primary scope of the project. The filter design tool consists of three types of filters or design methods that are Butterworth filter, Chebyshev Type I, Elliptic filter. Each filter above can be design in three types of responses that are Lowpass response, Highpass response, Bandpass response. The digital filter tool will run in two windows. First, it will run in Menu window and after the filter and response type chosen, it will continue to run in Design window for filter design. After design and evaluation, the Design window will show the plot four graph responses for filter analysis. The graph responses are Magnitude Response, Phase Response, Impulse Response, and Pole/Zero Plot.

ABSTRAK

Matlamat utama projek ini ialah mencipta sebuah alat untuk mereka *filter digital* yang berasaskan interaksi pengguna grafik (GUI) yang juga salah satu *tool* dalam MATLAB. Disebabkan skop *filter* teramat luas, hanya *filter Infinite Impulse Response* (IIR) yang ditumpukan dalam projek ini. Setelah filter IIR dikaji dan diteliti, alat pereka *filter digital* yang dibangunkan ini terdiri daripada tiga jenis *filter* iaitu Butterworth, Chebyshev, dan Elliptic. Setiap satu daripada *filter-filter* tersebut mempunyai tiga jenis *response* iaitu *Lowpass, Highpass* dan *Bandpass*. Alat pereka *filter digital* ini akan beroperasi dalam dua buah *window,* yang pertama dikenali sebagai *MENU window* dan yang kedua pula adalah *DESIGN window*. *MENU window* hanya digunakan untuk memilih *filter* yang hendak direka bersama dengan jenis respon yang hendak digunakan dan selepas dipilih, proses mereka filter dijalankan di *DESIGN window*. Selepas perekaan dan pengiraan, *DESIGN window* akan menunjukkan empat buah graf analisis iaitu *response* Magnitud, *response* Fasa, *response* Impulse, Pole/Zero plot.

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LIST OF ABBREVIATIONS

DSP	Digital Signal Processing
DFD	Digital Filter Design
FIR	Finite Impulse Response
IIR	Infinite Impulse Response
GUI	Graphical User Interface
GUIDE	GUI Design Environment
MATLAB	Matrix Laboratory
Fs	Sampling Frequency
Fpass/Fpb	End of the passband frequency
Fstop/Fsb	Beginning of the stopband frequency
Apass/Rpb	Passband ripple
Astop/Rsb	Stopband attenuation

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CHAPTER I

INTRODUCTION

1.1   Project Background

Electronic filters have a relevant importance in electronic systems because they are present in almost any electronic system. For example, communication systems, as many other electric systems, make intensive use of filtering to separate unwanted noise from the desired signal. Analog filter design is one of the most important areas of electronic design. Although some analog filter design books include simple, well-tested filter designs, filter design is often reserved for specialists because it requires advanced mathematical knowledge and understanding of the processes involved in the system affecting the filter. Modern sampling and digital signal processing tools enable you to replace analog filters with digital filters in applications requiring flexibility and programmability. These applications include audio, telecommunications, geophysics, and medical monitoring.

Digital filters have the following advantages over their analog counterparts they are software programmable, stable and predictable, do not drift with temperature or humidity. Besides, they are also do not or require precision components and have a superior performance-to-cost ratio. Unfortunately, filter design is an intensive

computational task requiring a significant amount of numerical calculations to obtain either the parameters of a filter transfer function or the element values for a filter circuit realization

The main idea of this project is to design an easy configuration for the filter courses as tool for teaching aid using the Graphical User Interface (GUI) in MATLAB Software. The advantage of the filter design software package is it makes use of one of the MATLAB toolboxes, the signal toolbox but use with an interface that makes possible for a beginner to readily design a filter. So, in this project, the initial task is choose a type of filter as a prototype for this project and as a result, digital filter design was chosen as the primary study of the project.

1.2   Objectives

To accomplish a project, the objective of the project must be cleared and can be understand to achieve. Hence, the objective of this project is:

- To implement an GUI-based digital filter design tool
- To provide a tool to be used in teaching aid besides to assist the novice user in digital filter design course.
- To discover GUI toolkit in MATLAB in order to design a user friendly digital filter design tool with low budget.

1.3   Problem statement

Filter design is an intensive computational task requiring a significant amount of numeral calculation to obtain either the parameters of a filter transfer function or element values for a filter circuit realization which is really hard for education progress. The cost of available filter design software is normally very high.

Therefore, the main idea of this project is to design an easy configuration for the filter courses as tool for teaching aid using the Graphical User Interface (GUI) in MATLAB Software. This tool will be helpful and encourage the novices to further explore the filter design courses because of its user friendly.

1.4   Scope

The project is focus on developing the tool for designing digital filter. The main tool use in this project is the MATLAB Software.

- The project is focusing on IIR digital filter design that consist of:
    - Butterworth Filter
    - Chebyshev Type I Filter
    - Elliptic Filter

- Each filter above has three responses implemented:
    - Lowpass
    - Highpass
    - Bandpass

- Every filter evaluated will display four graph response:
    - Magnitude Response
    - Phase Response

– Impulse Response

– Pole/Zero Plot

• The implementation of filter design toolkit is heavily involved GUI tool in MATLAB software.

The result of this project will be shown in simulation where GUI tool in MATLAB are use as an interface for the filter design. The target of this project is to encourage the students or novice users to design a filter in short period of time without having any knowledge about programming in MATLAB. The project will be focus on digital filter design.

1.5   Thesis Outline

This report is divided into five chapters. Chapter Two presents the literature review relating to the development of GUI-based digital filter design tool. It gives the reader basic theory of digital filter design within the scope of the project and relates some GUI implementation theory. Chapter Three discusses the tool design, work flow, system flow, programming algorithm and characterization of all the important methodology used in this study. Chapter Four is Results and Analysis where the results are shown and analyzed. This chapter deals with the field testing of the digital filter design tool that implemented from the methodology that was presented in Chapter Three. Finally, Chapter Five summarizes the study, the significant findings, and provides recommendations or suggestion for future studies regarding to this project.

CHAPTER II

DIGITAL FILTER

This chapter explains the necessary background information to understand the research presented in this report. The first part of this chapter will introduce some of the fundamental ideas involved in the design and development of this project. It will also include the current knowledge relating to Digital Filter, GUI, and MATLAB.

2.1 Digital Filter Concept

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 to achieve the desired filtering effect.

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 cost and their speed because the filter can't operate any faster than the computer at the heart of the filter. 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.

2.1.1   Types of digital filters

Many digital filters are based on the Fast Fourier transform, a mathematical algorithm that quickly extracts the frequency spectrum of a signal, allowing the spectrum to be manipulated (such as to create pass-band filters) before converting the modified spectrum back into a time-series signal.

The transfer function for a typical linear digital filter can be expressed as a transform in the *Z*-domain, as:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 + z^{-1} + b_2 z^{-2} + \ldots + b_N z^{-N}}{1 + a_1 z^{-1} + a_2 z^{-2} + \ldots + a_M z^{-M}} \qquad (2\text{-}1)$$

where *M* is the order of the filter.

This form is for a recursive filter, which typically leads to ***infinite impulse response (IIR)*** behaviour, but if the denominator is unity, then this is the form for a ***finite impulse response filter (FIR).***

2.2   Infinite Impulse Response Filters (IIR)

The digital filters designed using the DFD applications are either Infinite Impulse Response (IIR) or Finite Impulse Response (FIR) filters. IIR filters process both input and output samples. This allows IIR filters to have very sharp transitions in their frequency response characteristics, which is very useful in many applications. The nonlinear phase distortion associated with IIR filters sometimes can limit their selection in many phase sensitive systems.

Infinite impulse response filters (IIR) filters are digital filters with impulse responses that can theoretically be infinite in length (duration). The general difference equation characterizing IIR filters is

$$y_i = \frac{1}{a_0}\left(\sum_{j=0}^{N_b-1}b_j x_{i-f} - \sum_{k=1}^{N_a-1}a_k y_{i-k}\right) \tag{2-2}$$

where Nb is the number of *forward* coefficients (bj) and Na is the number of *reverse* coefficients (ak). In most IIR filter designs, coefficient $a0$ is 1. The output sample at the present sample index i consists of the sum of scaled present and past inputs (xi and $xi\text{-}j$ when j 0) and scaled past outputs ($yi\text{-}k$). The response of the general IIR filter to an impulse ($x0 = 1$ and $xi = 0$ for all i 0) is called the impulse response of the filter. The impulse response of the filter described by equation (2-2) is indeed of infinite length for nonzero coefficients. In practical filter applications, however, the impulse response of stable IIR filters decays to near zero in a finite number of samples.

The advantage of digital IIR filters over finite impulse response (FIR) filters is that IIR filters usually require fewer coefficients to perform similar filtering operations. Thus, IIR filters execute much faster and do not require extra memory, because they execute in place. The disadvantage of IIR filters is that the phase response is nonlinear. If the application does not require phase information, such as simple signal monitoring, IIR filters may be appropriate. You should use FIR filters for those applications requiring linear phase responses. IIR filters are also known as recursive filters or autoregressive moving-average (ARMA) filters.

2.2.1   Cascade Form IIR Filtering

Filters implemented using the structure defined by equation 2-2 directly is known as *direct form* IIR filters. Direct form implementations are often sensitive to errors introduced by coefficient quantization and by computational precision limits. Additionally, a filter designed to be stable can become unstable with increasing coefficient length, which is proportional to filter order. A less sensitive structure can be obtained by breaking up the direct form transfer function into lower order sections, or

filter stages. The direct form transfer function of the filter given by equation 2-2 (with $a0$ = 1) can be written as a ratio of z transforms, as follows:

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + ... + b_{N_b-1} z^{-(N_b-1)}}{1 + a_1 z^{-1} + a_2 z^{-2} + ... + a_{N_a-1} z^{-(N_a-1)}}$$

(2-3)

By factoring equation 2-3 into second-order sections, the filter's transfer function becomes a product of second-order filter functions

$$H(z) = \prod_{k=1}^{N_a} \frac{b_{0k} + b_{1k} z^{-1} + b_{2k} z^{-2}}{1 + a_1 z^{-1} + a_{2k} z^{-2}}$$

(2-4)

This new filter structure can be described as a *cascade* of second-order filters.

Figure 2.1: Cascaded Filter Stages

Each individual second-order stage can be implemented using the direct form filter equations:

y[i] =b0x[i] + b1x[i-1] + b2x[i-2]- a1y[i-1] - a2y[i-2]

(2-5)