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A GUI FOR IIR DIGITAL FILTER DESIGN USING ANALOG TO DIGITAL MAPPING

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This Paper describes a simple computer-aided design approach for designing Infinite impulse response (IIR) digital filters. IIR filter is essentially a digital filter which depends not only upon the Input data but also upon one or more previous output values. A variety of techniques are available for the designing of IIR Filters. This Paper is based only upon Analog to digital mapping techniques for fast convergence of results. The Simulation Program for digital filter is discussed by an example, LP_1K_0.2K_0.4K_0.2dB_0.4dB, presented in GUI frame and the results are found to be very encouraging. Numerical results are presented to illustrate the performance of proposed method and compared with Matlab filter design toolbox.

Keywords: Digital Filter, IIR Filter, Butterworth, Chebyshev and Elliptic Filters.

1. Introduction

During the last few decades the field of Digital Signal Processing (DSP) has grown to important both theoretically and technologically. For example, DSP has obtained wide applications in several engineering areas from communication, biomedical, and control to meteorology. In DSP, there are two important types of Systems. The first type of systems performs signal filtering in time domain and hence it is known as DIGITAL FILTERS. The second type of systems provide signal representation frequency domain and are known as Spectrum Analyzer. Digital filtering is one of the most important and powerful tool of DSP. Digital filters are capable of performance specifications that would, at best, be extremely difficult, if not impossible, to achieve with an analog implementation. In addition, the characteristics of a digital filter can be easily changed under software control. Digital filters are classified either as Finite duration unit pulse response (FIR) filters or Infinite duration unit pulse response (IIR) filters, depending on the form of unit pulse response of the system. In the IIR system, the impulse response is of infinite duration. During design of digital filters it is important to fulfill accepted design specifications such as: the width of pass-band, width of stop-band, value of pass-band ripples and value of stop-band ripples. The preceding design methods have been based on discretization of frequency range. The discretization of frequency range is automatic in this approach. It is shown that the use of GUI can substantially help the design of IIR digital filter. Nevertheless, it is a useful noniterative design method.

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This Paper is organized as follows: In Section II, IIR digital filter design aspects are discussed. In section III, analog to digital mapping is briefly mentioned. The designed GUI frame related to filter design is proposed in Section IV. The simulation results of designed examples used is briefly described in Section V. The Conclusion and future scope is described in Section VI.

2. Problem Formulations

Digital filters are classified as Recursive and Non-Recursive filters. The response of Non Recursive or IIR filters depends not only upon Input data but also upon one or previous data. IIR filters have the following advantages:

- They require less number of arithmetic operations so small memory is required.
- There are shorter time delays in these filters.
- IIR Filters have resemblance with analog filters.
- These filters depend not only upon the input but also upon previous output values.
- They are more susceptible to noises.

The Digital IIR filters have various stages for their design. The flow chart of the Design of Digital filter is shown in Figure 1.

The IIR system has an infinite number of non zero terms, i.e., its impulse response sequence is of infinite duration. The system function of an IIR filter is given by:

$$H(z) = \frac{B(z)}{A(z)} = \frac{\sum_{n=0}^{M} b(n)\, z^{-n}}{1+\sum_{n=1}^{N} a(n) z^{-n}}$$

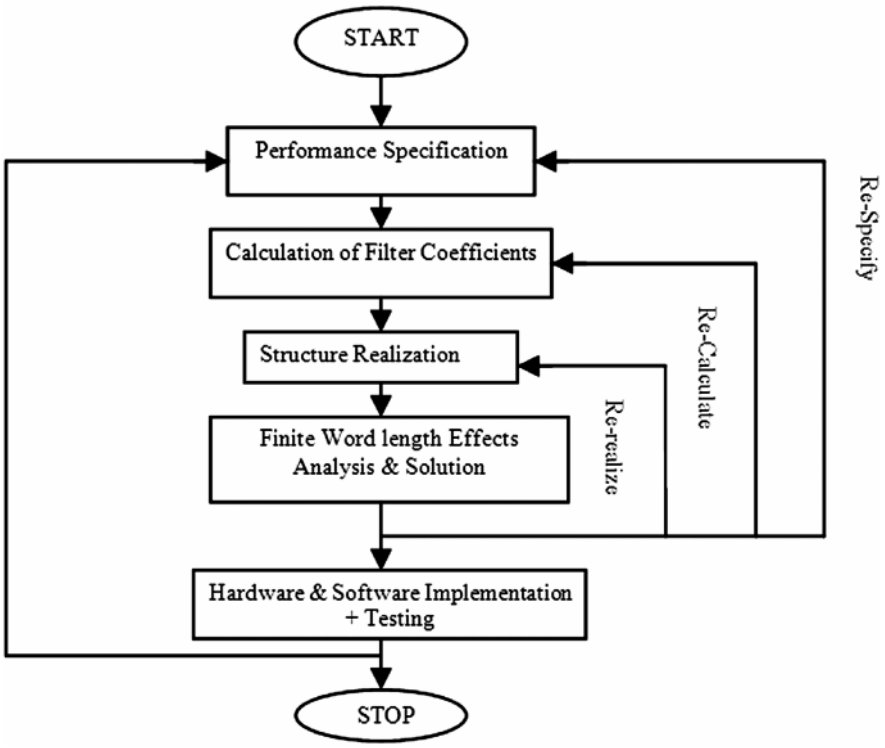

Fig.1

$$= \frac{b(0)+b(1)z^{-1}+.....+b(M)z^{-M}}{1+a(1)z^{-1}+.....+a(N)z^{-N}}$$

where b(n) and a(n) are the coefficients of the filter. The difference equation representation of an IIR filter is expressed as

$$y(n) = \sum_{m=0}^{M} b(m)x(n-m) - \sum_{m=1}^{N} a(m)y(n-m)$$

3. ANALOG TO DIGITAL DOMAIN MAPPING TECHNIQUES

Digital Filters are designed by using the values of both the past outputs and the present input [4], an operation brought about by convolution. If such filters subjected to an impulse then its output need not necessarily become zero. The infinite impulse response of such a filter implies the ability of the filter to have an infinite impulse response. This indicates that the system is prone to feedback and instability.

IIR filters have Infinite-Duration Impulse Response; hence they can be matched to analog filters, all of which generally have infinite long impulse response. Therefore the basic technique of IIR Filter design transforms well-known analog filters into Digital Filters. Hence IIR filter designs will be treated as the characteristic of three widely used analog filters, namely, Butterworth, Chebyshev (Type1 and Type2) and Elliptic filters [6,7].

Butterworth of maximally flat filters have a monotonic amplitude frequency response which is maximally flat at zero frequency response and the amplitude frequency response decreases logarithmically with increasing frequency. The Butterworth filter has minimal phase shift over the filter's band pass when companied to other conventional filters. Chebyshev Filters are of two type's i.e Chebyshev I and Chebyshev II filters. Chebyshev I filters are all pole filters which are equiripple in the pass band and are monotonic in the stop band .Chebyshev II filters contain both poles and zeros exhibition a monotonic behavior in the pass band and equiripple in the stop band. Elliptic filters are characterized by equiripple the both their pass bands and stop band. They provide a realization with the lowest order for a particular set of conditions.

4. DESIGNED GUI FRAME

This section presents the simulation framework for the Design of IIR filter using Analog to Digital transformation. A front panel of Graphical User Interface (GUI) is designed for easiness and good design of filter. GUI stores the data in two files; A FIG-file, with extension .Fig, that contains a complete description of the GUI layout and the GUI components, such as push buttons, and edit buttons. An M-file, with extension .m, that initially contains initialization code and templates for some callbacks that are needed to control GUI behavior. The Butterworth, Chebyshev (Type1

and Type2) and Elliptic filters callbacks have been added for GUI components. The FIG-file is a binary file. The frame construct digital filter transfer functions by conversion from classical analog transfer functions (Butterworth, Chebyshev I, Chebyshev II, elliptic) with definite characteristics such as monotonic or equiripple magnitude response. Nevertheless, it is a useful noniterative design method. Numerical results are presented to illustrate the performance of GUI frame and compared with the MATLAB filter design toolbox.

5. SIMULATION RESULTS

The MATLAB simulation is carried for certain specifications such as f_s = 1000Hz, f_{pb} = 200 Hz, f_{sb} = 400 Hz, R_{sb} = 1dB and R_{sb} = 15dB for IIR filters. The various plot for Magnitude, Impulse Responses and Pole-Zero for Butterworth and Elliptic LP_1K_0.2K_0.4K_1dB_15dB are shown below in Fig. 3 to Fig. 8. Table 1 gives the results of coefficients of Butterworth and Elliptic filter by proposed method. Table 2 illustrates the coefficients obtained by the filter design toolbox.

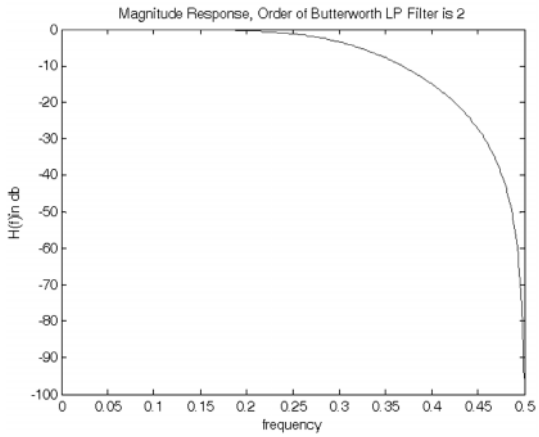

Fig. 2

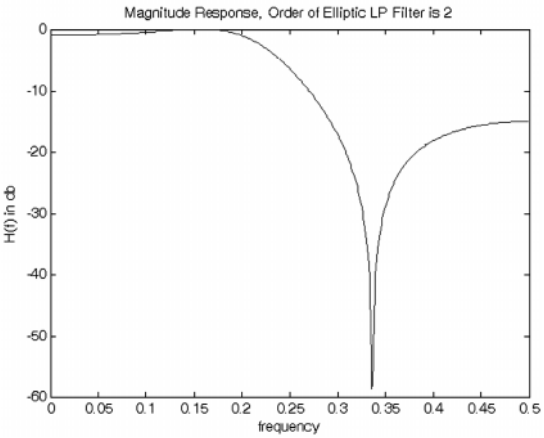

Fig. 5

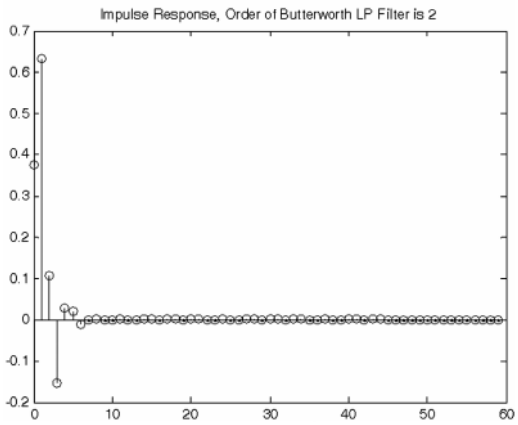

Fig. 3

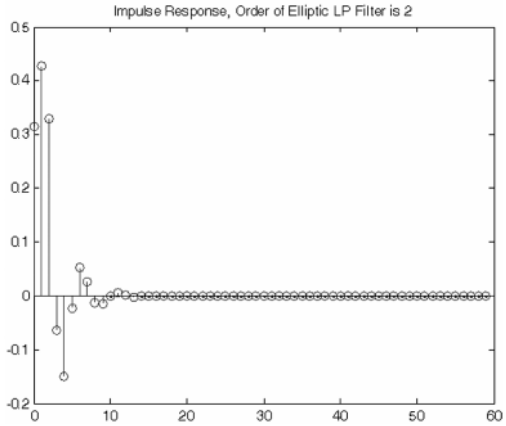

Fig. 6

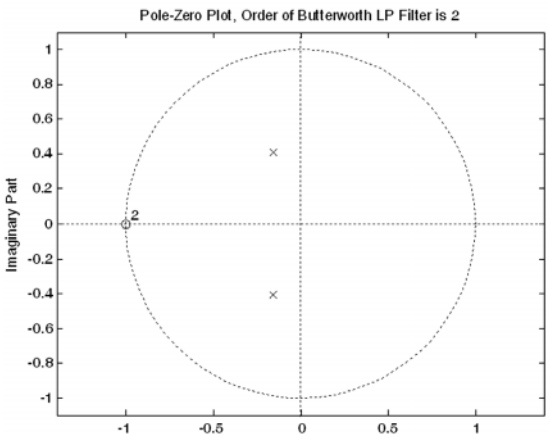

Fig. 4

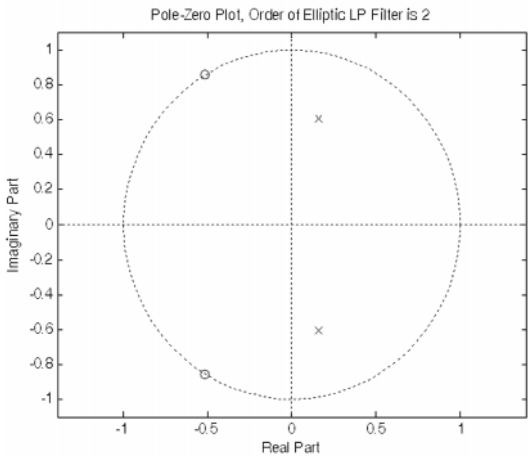

Fig. 7

Table 1
Results Obtained by Proposed Algorithm

Name of filter	Order of filter	Coeff. of Numerator	Coeff. of Denominator
Butterworth	3	0.3752, 0.7504,	1.0, 0.3120, 0.1888
Elliptic	3	0.3157, 0.3260, 0.3157	1.0, -0.3218, 0.3961

Table 2
Results Obtained Using Filter Design Toolbox

Name of filter	Order of filter	Coeff. of Numerator	Coeff. of Denominator
Butterworth	3	1.0, 2.0,1.0	1.0, 0.0214, 0.17165
Elliptic	3	1.0, 1.03388, 0.9999	1.0, -0.3164, 0.3945

6. CONCLUSION

In this paper, IIR digital filter using analog to digital transformation is discussed. It is shown that the use of GUI can substantially help the design of IIR digital filter. The applicability of the GUI frame has been demonstrated by Low pass filter. The Simulation Program for design of IIR digital filters using an example is presented to illustrate the effectiveness of GUI frame and the results are found to be very encouraging. Numerical results are compared with the Matlab filter design tool box which shows the effectiveness of proposed algorithm. The stability is presented by Pole-Zero plot of the IIR filter. Further research will focus on High pass and Band pass filter.

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