

# Studying on Digital Filters in MATLAB Environment

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**Abstract** – Digital filters are modeled as discrete-time linear systems. They can be used for many tasks: to separate signals and noise added during different processes, to separate signals into their constituent frequencies, to demodulate signals, to restore signals that have been degraded by some process, known or unknown. With help of the computer simulation it's made a trial in the paper to illustrate their basic characteristics and possibilities. It can be used by deeper studying the type of digital filters, their characteristics and their properties. The computer simulation is realized in MATLAB environment. The paper can be used in engineering education in studying on these systems.

**Keywords** – Digital filters, linear-time systems, noise separation, signal restoration, signal demodulation, computer simulation.

## I. INTRODUCTION

The digital filtration account for influence to the spectrum of the processing signals by definite computing algorithm. Choosing the best filter structure depends on the task the filter will perform. Some structures are more suited to or may be more computationally efficient for particular tasks. For example, often it is not possible to build recursive (IIR) filters to run at very high speeds and instead, we can use a nonrecursive (FIR) filter. FIR filters are always stable and have well-behaved round off noise characteristics.

The goal of filter design is to perform frequency dependent alteration of a data sequence. A very precise specification could ask to achieve the performance goals with the minimum filter order, or it could call for an arbitrary magnitude shape, or it might require an FIR filter. Filter design methods differ primarily in how performance is specified. There are 3 forms of realization: direct, cascade and parallel. In the practice often can be used cascade and removal cascade form [1]. The goal of the synthesis is to obtain a transfer characteristic and the respectively realization. This characteristic can be obtained by analogue low-pass filter prototype or by bilinear  $z$ -transform. The transfer characteristic can be achieved by polynomial approximation. Dependency from the type of the polynomial approximation, practically applications have following low-pass filter prototypes: Butterworth, Chebyshev

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Type I, Chebyshev Type II, and elliptic filters.

It's described in the paper a laboratory exercises for the undergraduate course on Signals and Systems of students from faculty of communications and from faculty of computer systems. The students can synthesize digital filters and analyze different filter responses such a magnitude response, phase response, impulse response and pole-zero plots on the base of definite filter coefficients. The filter they design is then computed using the default filter design method and filter order. They can investigate the change of the signal by filtration in time and frequency domain, too.

The computer simulation can be realized in the program environment of MATLAB with using the system for visual modelling SIMULINK and FDA TOOL.

## II. PROBLEM FORMULATION

The problem on studying on digital filters can be presented with following features:

-There 4 type of filters: low-pass, high-pass, band-pass and rejector. They are described by the transfer characteristics, respectively by their coefficients. For the IIR filters is given in Eq.1[1].

$$\begin{aligned}
 H(z) &= \frac{Y(z)}{X(z)} = \frac{a_0 + a_1 z^{-1} + \dots + a_M z^{-M}}{1 - [b_1 z^{-1} + \dots + b_N z^{-N}]} = \\
 &= \frac{\sum_{m=0}^M a_m z^{-m}}{1 - \sum_{m=1}^N b_m z^{-m}} \quad (1)
 \end{aligned}$$

The complex frequency coefficient can be obtained in Eq.2 from Eq.1 by the substitution  $z = e^{j\omega T}$ .

$$\begin{aligned}
 H(j\omega) &= \frac{a_0 + a_1 e^{-j\omega T} + \dots + a_M e^{-j\omega T M}}{1 - [b_1 e^{-j\omega T} + \dots + b_N e^{-j\omega T N}]} = \\
 &= \frac{\sum_{m=0}^M a_m e^{-j\omega T m}}{1 - \sum_{m=1}^N b_m e^{-j\omega T m}} \quad (2)
 \end{aligned}$$

The magnitude and phase responses are given in Eq.3 and Eq.4.

$$|H(j\omega)| = \sqrt{\frac{A^2 + B^2}{C^2 + D^2}} \quad (3)$$

$$\varphi(\omega) = \arctg \frac{B}{A} - \arctg \frac{D}{C} \quad (4)$$

where  $A$  and  $C$  are the real part of  $H(j\omega)$ ,  $B$  and  $D$  are the imaginary part. The impulse response  $h(n)$  can be obtained by Eq.5.

$$Y(n) = \sum_{m=-\infty}^n X(m)h(n-m) = \sum_{m=0}^{\infty} h(m)X(n-m) \quad (5)$$

By fulfillment of condition  $h(n) = 0$  for  $n < 0$ , the recursive filter is causal. It is stable by the condition, given in Eq.6 [3].

$$\sum_{n=-\infty}^{\infty} |h(n)| < \infty \quad (6)$$

So on base of computer simulation we can formulate following problems:

1. To synthesize digital filters in the 4 types by the polynomial approximation of Butterworth, Chebyshev Type I, Chebyshev Type II, and elliptic filters.
2. To choose the best filter design method and filter order.
3. To analyze a magnitude response, phase response, impulse response and pole-zero plots of the synthesized digital filters.
4. To investigate the change of different signals by filtration in time and frequency domain, too.

By the simulation we need to see the going processes. The characteristics can be given in graphical mode.

### III. EXPERIMENTAL PART

The formulated problems are solved by computer simulation in MATLAB, version 6.5 with using the Filter Design & Analysis (FDA) TOOL and SIMULINK TOOLBOX. FDA TOOL is a powerful user interface for designing and analyzing filters. It enables the students to quickly design digital FIR or IIR filters by setting filter performance specifications, by importing filters from MATLAB workspace, or by directly specifying filter coefficients [2]. FDA TOOL also provides tools for analyzing filters, such as magnitude and phase and impulse response plots and pole-zero plots. It can be shown the best form of realization. The students can change only one or some parameters. In this case FDA TOOL can design the filter with the determinate parameters and compatibility optimized characteristics. It can be choose the filter transform, the type of the filter and the frequency, too. There are the following options:

- Type of the filter by 'Response Type'
- IIR or FIR filters and the method of approximation by 'Design Method'
- Filter order
- Frequency specifications
- Magnitude specifications
- Form structure and coefficients by 'Filter Coefficients'
- Filter transform by 'Frequency Transformations'

On the other side, models can be created by the students without experience in using of the other programming tools. It's may be a great priority, because visual presentation of the fulfillment simulation can be realized. So the students can create a model to filtration of different signals. It's possible to simulate the process and to investigate the change of the signals by the intuitive easy graphical interface of SIMULINK.

The running time can be set in constant or variable step. The students can observe a running process by the block 'Scope', which is included in the model, too. They can analyze the signal spectrum by the block PSD. The realized model of filtration is given on Fig. 1.

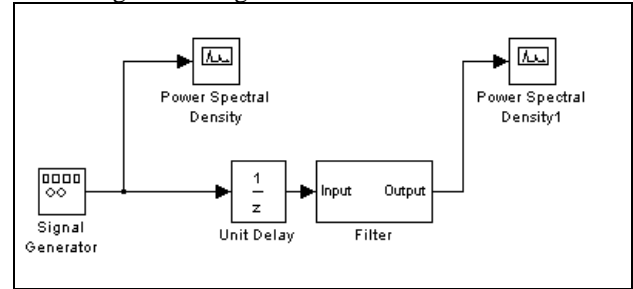


Fig.1. A model of filtration in SIMULINK

The Magnitude response, Phase response, Impulse response and Pole/Zero Plot for low-pass filter with  $f_s = 8000$  Hz,  $f_p = 1600$  Hz,  $f_c = 2000$  Hz,  $A_{\min} = 40$  dB,  $A_{\max} = 1$  dB by polynomial approximation of Butterworth, Direct Form II, order 17, stable is given on Fig.2.

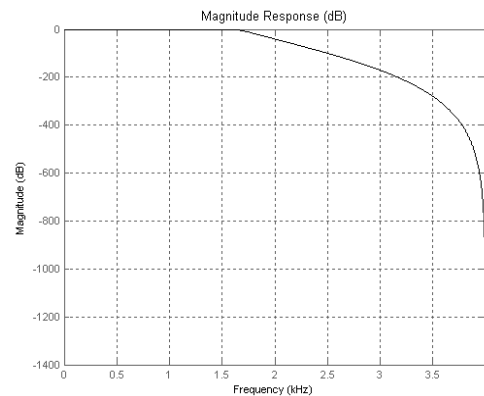


Fig.2a. Magnitude response

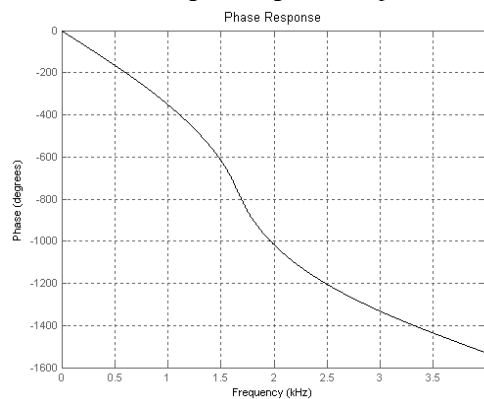


Fig.2b. Phase response

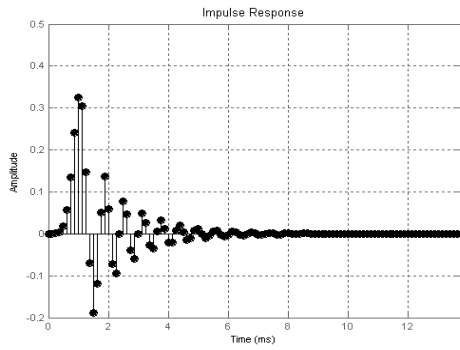


Fig.2c. Impulse response

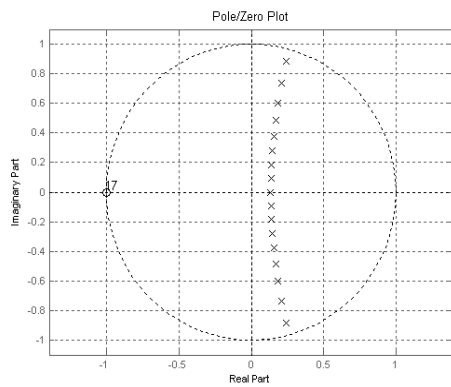


Fig.2d. Pole/Zero Plot

On Fig.3 we can see the change of the signal form and its spectrum by the same filter.

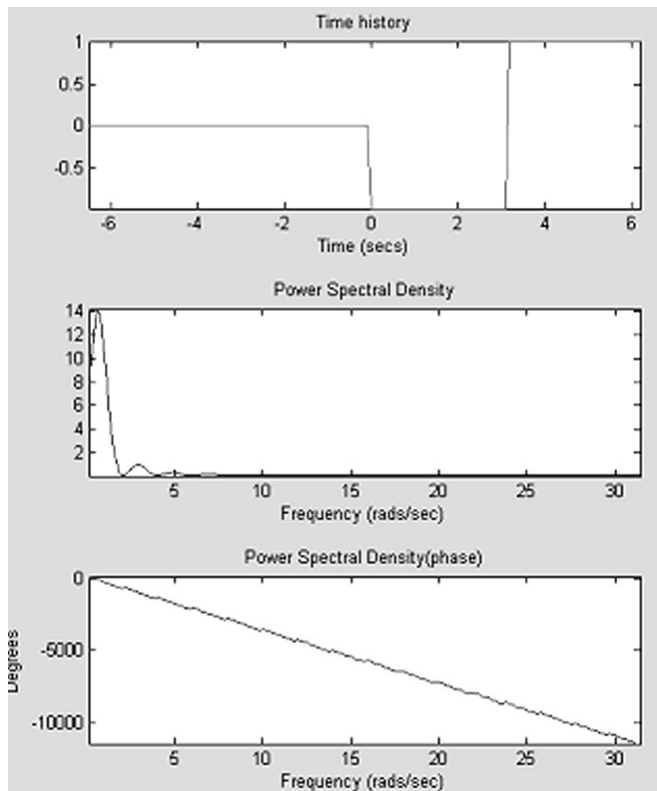


Fig.3a. Input signal

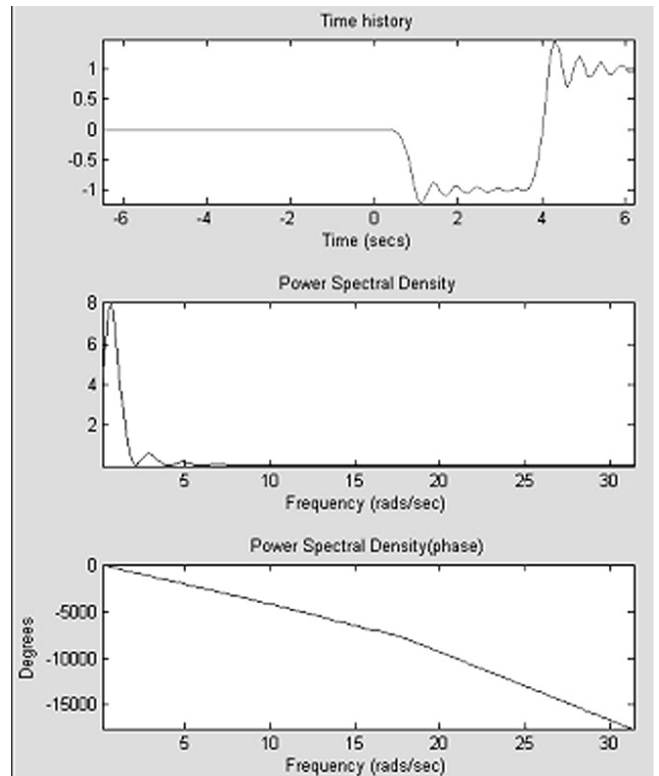


Fig.3b. Output signal

#### IV. CONCLUSION

The studying of digital filters is one of the very important problems in the theory of the signals. It's made a trial in the paper to present the possibility of the computer simulation in MATLAB environment to get deeper insight on the syntheses of digital filters and analyzes of different filter responses such a magnitude response, phase response, impulse response and pole-zero plots on the base of definite filter coefficients. The filter they design is then computed using the default filter design method and filter order. By help of simulation they can investigate the influence of the filtration on the signal form and its spectrum. The method can be developed and the described exercise can be used for web based distance education.

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