

Development "Filtration" Software for the Digital Processing of the Measuring Signals

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Abstract— The paper covers “Filtration” software developed by author. “Filtration” software allows the design of low-pass filters up to an 8th order filter with Chebychev, Bessel, Elliptic, Hamming, Kaiser, RMS or Butterworth responses from frequencies of 0.1 Hz to 1 MHz. “Filtration” software also can be used to design band-pass and high-pass filters with Chebychev, Bessel, Elliptic, Hamming, Kaiser, RMS or Butterworth responses. Options (specify order of filter, enter the value of signal to noise ratio) are available. Once the filter response has been identified, “Filtration” software generates the frequency response and the parameters of design filter. Software also will recalculate all values to meet the desired response, allowing real values to be substituted or changed as part of the design process. “Filtration” software allows to filtered noise on the base of the data, which corresponds to some measured signal, and to choose the desired type of digital filter, which eliminated unnecessary frequencies (noises) from input signal the best way. Software package was created by using Visual C++ environment.

Index Terms— Software, filtration algorithm, digital filter, low-pass filter, high-pass filter, bandpass filter, impulse response, noise cancellation, signal to noise ratio.

I. INTRODUCTION

Digital filters are able to remove (filter out) or enhance certain frequency components in a signal. Digital filters are a key technology in many audio processing applications and are fundamental to communication systems. The idea of a digital filter is to either reduce or increase the strength of a signal but only over specific frequency ranges. Digital filters are used in various applications, a few of which are audio recording-playback-enhancement applications, noise removal, bandpass filtering for audio and image storage and/or transmission, and narrow-band filtering for selecting a communication channel [1].

In connection with the spread of digital information processing methods the recent trend is to increase the role of computer modeling for the analysis and synthesis filters. Software implementation of algorithms for digital filtering is stable, versatile and accurate. And now it competes with analog implementations.

II. SOFTWARE DESCRIPTION

C++ is a powerful and relatively versatile tool for implementing a variety of mathematical algorithms, in this case the filtering algorithms, so I apply object-oriented

programming in Visual C++ for digital processing of noisy measuring signals.

The "Filtration" software package contains a huge database of existing digital filters with finite and infinite impulse response [2]. Therefore, it is possible to select the optimal digital filter based on the results of the program for a given input signal. This software is a tool that simplifies filter design for a given input signal.

“Filtration” software allows the design of low-pass filters up to an 10th order filter with Chebychev, Bessel, Elliptic, RMS, Finite Difference, Modified Finite-Difference, Hamming, Kaiser or Butterworth responses from frequencies of 0.1 Hz to 1 MHz. “Filtration” software also can be used to design band-pass and highpass filters with different response functions.

To provide convenience of communication with the software the user interface has been designed. When you run the program *Filter Design* dialog window appears on the screen and this window enables the user to create a filter by specifying all aspects of the filter. There are three tabs in this window: *Filter Information*, *Filter Parameters* and *Noise Cancellation* (Fig. 1). The *Filter Information* tab enables the user to specify the design method (Batterworth, Bessel, Chebishev and so on) and response type (Lowpass, Highpass, Bandpass). After specifying the Filter Information, user should select “Design Filter” button or second *Filter Parameters* tab. *Filter Parameters* window is shown below (Fig. 2).

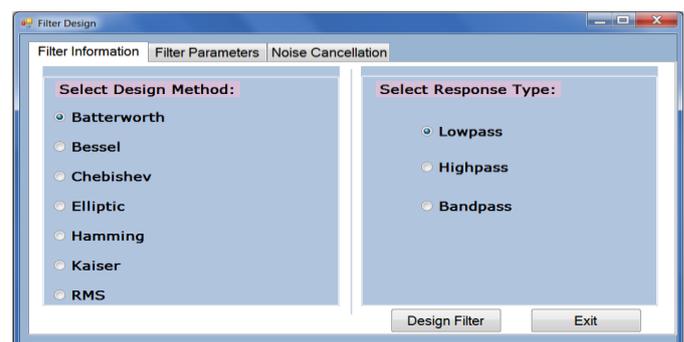


Fig. 1. Filter Design dialog window

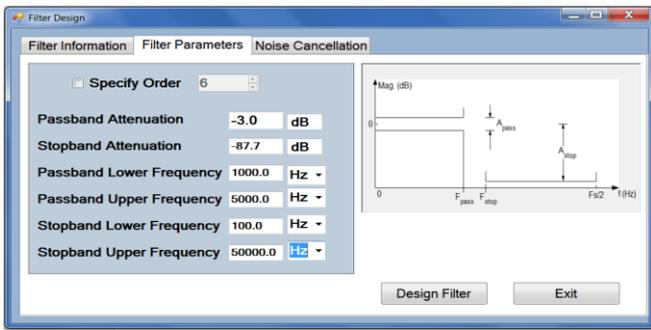


Fig. 2. Filter Parameters tab

This window contains main parameters of filters. These parameters vary depending on the selected filter type. If you select lowpass filter, then you will see passband and stop band upper and lower frequencies. For all type of filters there will be the passband and stopband attenuations, the passband and stopband frequencies [3, 4]. Parameters for the synthesis of lowpass filters are shown in Fig. 3.

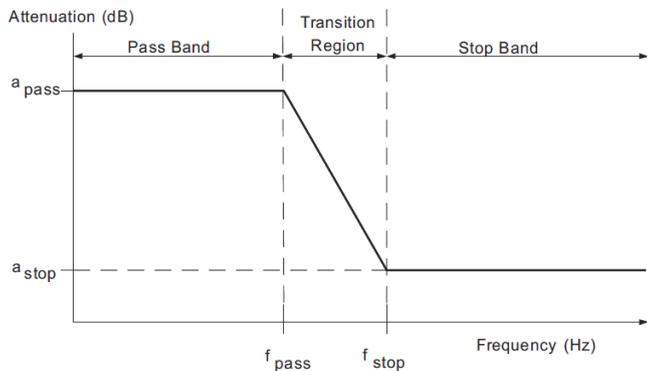


Fig. 3. Specifications for Low-pass filter

For highpass filters there will be the passband and stopband frequencies. For low-pass selectivities, f_{pass} upper and f_{pass} lower represent the passband and upper and lower frequencies, while f_{stop} upper and f_{stop} lower represent the stop band and upper and lower frequencies (Fig. 4).

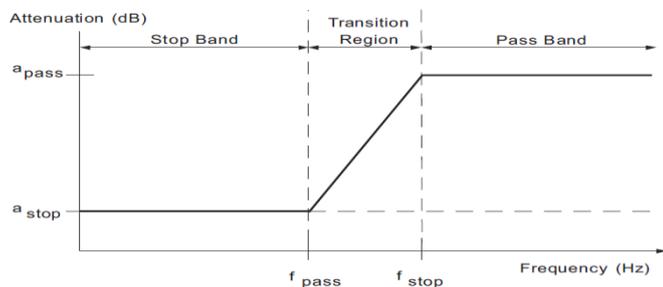


Fig. 4. Specifications for High-pass filter

The main parameters of bandpass filters are shown in Fig.5.

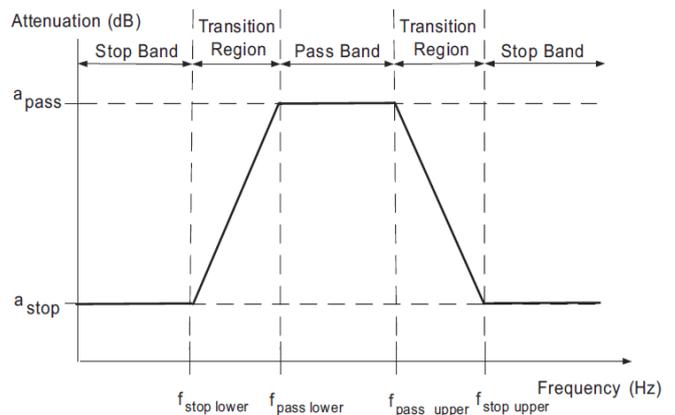


Fig. 5. Specifications for Bandpass filter

There is the *Specify Order* option in the Filter Parameters dialog box. It enables the user to specify the filter order or have the program calculate the filter order based on the dialog entries. To specify the filter order, select the *Specify Order* checkbox. When the checkbox is selected, the user specifies only the *Pass Band Attenuation* and the *Pass Band* frequencies. Program automatically calculates the *Stop Band Attenuation* and *Stop Band* frequencies based on the order. When the checkbox is not selected, the user specifies the attenuation and all frequency values and “Filtration” package calculates the order of designed filter based on the values of attenuation and frequency. The *Pass Band Attenuation* is the change in magnitude of the frequencies in the pass band. The *Stop Band Attenuation* is the minimum reduction in magnitude of the frequencies in the stop band relative to the pass band. The *Pass Band Frequency* is the starting point of the pass band. The *Pass Band Frequency* must be lower than the *Stop Band Frequency* for low-pass filters. If a value is entered in the *Filter Parameters* tab is larger than the *Stop Band Frequency*, the error message will appear on the screen. The *Pass Band Frequency* must be greater than the *Stop Band Frequency* for high-pass filters. If a value is entered in the *Filter Parameters* tab is smaller than the *Stop Band Frequency*, the error message also will appear on the screen.

The *Stop Band Frequency* is the starting point of the stop band and it must be greater than the *Pass Band Frequency* for low-pass filters. For *high-pass filters* it must be greater than the *Stop Band Frequency*.

In order to implement the filter after finishing all selections you should press “Design Filter” or “Cancel” button. If there is not any error, the next window with user selections appears on the screen (Fig. 6). If you want filtering some input data from unnecessary frequencies, you should select third tab *Noise Cancellation*.

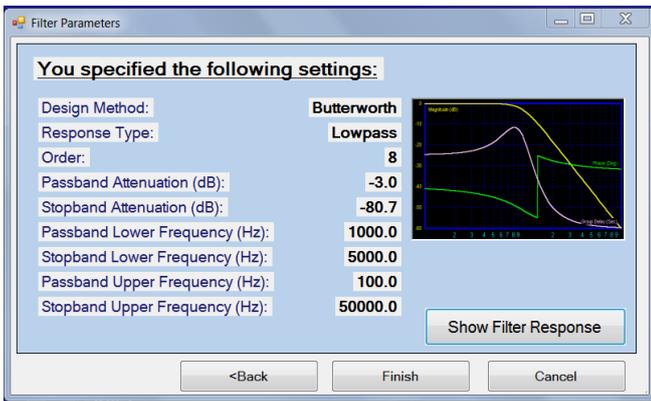


Fig. 6. Specified by user Filter parameters

“Show Filter Response” button allows you to see filter frequency response graphically. The left axis displays the attenuation of the filter. The frequency range is automatically set to selected measurement units (Fig. 7).



Fig. 7. Filter Response window

The “Noise Cancellation” tab (Fig. 8) allows you to check selected filter type on input data. The main purpose of this is noise cancellation, which means the reduction of noise from input data.

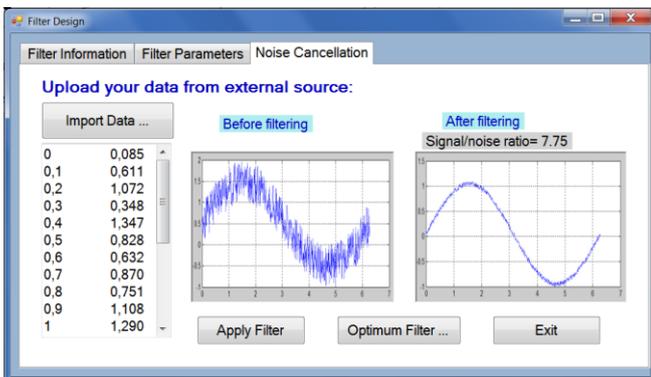


Fig. 8. Noise Cancellation tab

Input data are exported from .xls or .dat file by clicking “Import data” button. The input file should consist of two columns. The first column represents independent variable and the last one represents dependent variable for some measured quantity. After pressing “Apply Filter” button input data passing through selected filter, the output filtered data appears on the screen. You can see the data before and after filtering.

At the same time the value of signal to noise ratio is calculated and is shown on the screen (Fig. 8). The program also has the ability to use the signal to noise ratio and the value of the parameters which are entered by user to calculate the optimal low-pass filter: the filter order, passband and stopband frequencies, and the type of frequency response. For this case you should press “Optimum Filter” button and the following window appear on the screen (Fig. 9). Optimal filter parameters are determined on base of solving optimizing problem. The main criterion is a minimum noise variance on the output of filter.

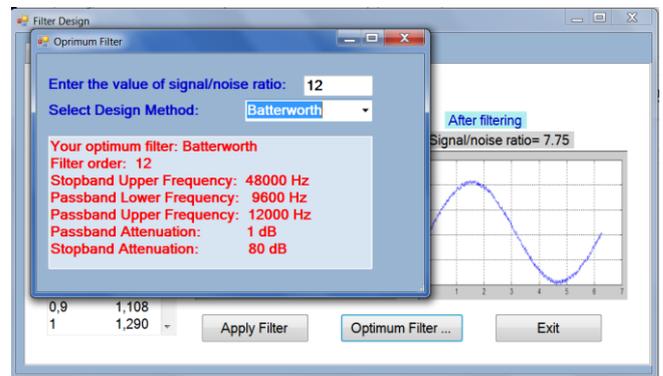


Fig. 9. Optimum Filter dialog window

Developed “Filtration” software allows to filtered noise on the base of the data, which corresponds to some measured signal, and to choose the desired type of digital filter, which eliminated unnecessary frequencies (noises) from input signal the best way.

III. CONCLUSION

The advantage of this software package is that by obtaining the optimal filter characteristics for a given input measurement signal, it is possible to physically implement the filter and to use it for pre-filtering of noise at the output of the measuring device. Filtering the signal with the best signal / noise ratio can be achieved only after the recording of the experimental data.

Based on the graphic and numeric information given by designed software package and value of signal to noise ratio you can make a conclusion about the effectiveness of the selected filter type.

Thus, the developed “Filtration” software allows the design of low-pass filters up to an 8th order filter with Chebychev, Bessel, Elliptic, Hamming, Kaiser, RMS or Butterworth responses from frequencies of 0.1 Hz to 1 MHz. “Filtration”

software also can be used to design band-pass and high-pass filters with Chebychev, Bessel, Elliptic, Hamming, Kaiser, RMS or Butterworth responses. Options (specify order of filter, enter the value of signal to noise ratio) are available. Once the filter response has been identified, "Filtration" software generates the frequency response and the parameters of design filter. Software also will recalculate all values to meet the desired response, allowing real values to be substituted or changed as part of the design process. "Filtration" software allows to filtered noise on the base of the data, which corresponds to some measured signal, and to choose the desired type of digital filter, which eliminated unnecessary frequencies (noises) from input signal the best way.

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