COMPARISON BETWEEN ANALOG AND DIGITAL FILTERS

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Abstract: *Digital signal processing(DSP) is one of the most powerful technologies and will model science and engineering in the 21st century. Revolutionary changes have already been made in different areas of research such as communications, medical imaging, radar and sonar technology, high fidelity audio signal reproducing etc. Each of these fields developed a different signal processing technology with its own algorithms, mathematics and technology, Digital filters are used in two general directions: to separate mixed signals and to restore signals that were compromised in different modes. The objective of this paper is to compare some basic digital filters versus analog filters such as low-pass, high-pass, band-pass filters. Scientists and engineers comprehend that, in comparison with analog filters, digital filters can process the same signal in real-time with broader flexibility. This understanding is considered important to instill incentive for engineers to become interested in the field of DSP. The analysis of the results will be made using dedicated libraries in MATLAB and Simulink software, such as the Signal Processing Toolbox.*

1. INTRODUCTION

Analog filters are a first layer block in signal processing, often used in electronics. Passive filters have been the base of communications since the 1920's and are of considerable importance for frequencies situated between 100 and 500 kHz. Hundreds, if not thousands of types of passive filters have been developed in order to satisfy the needs of different applications. However most filters can be described by few common charactheristics. First of them is the frequency domain of their bandpass. The bandpass of a filter is frequency domain over which an input signal will pass. Signals of frequencies that are not in the bandpass will be attenuated.

2. LOWPASS FILTER

Lowpass filters allow low frequency signals to pass, while they block high frequency signals. The concept of lowpass filter exists in various forms, including electronic circuits, digital algorithms used to process data sets, acoustic barriers, image processing etc. Low-pass filters play the same role in signal processing that moving averages do in some other fields, such as finance; both tools provide a smoother form of a signal which removes the short-term oscillations, leaving only the long-term trend. In equation (1) you can see the break frequency, also called the turnover frequency or cutoff frequency (in Hz).

$$f_c = \frac{1}{2\pi\tau} = \frac{}{2\pi} \qquad (1)$$

where R is the resistor, with a value in ohms, and C is the capacitor, with a value in Farads.

In figure 1 you can see the analog realization of a lowpass filter and in figure 2 you can see the output waveform of a lowpass filter.

Same filter response can be obtained using a digital filter as shown by the system block diagram in figure 3. The analog signal x(t) is converted into a discrete-time signal x(n), which is processed by the digital filter, to yield a discrete-time output y(n). Finally, the discrete output y(n) is converted into an analog form y(t). The cutoff frequency of digital filter response $H(e^{j\acute{\omega}})$ is related to the analog cutoff frequency through the important analog-digital frequency relation

$$\acute{\omega}=\acute{\Omega}T \qquad (2)$$

where T(s) is the sampling interval of discrete-time system.

Hence, the unit of analog frequency, $\acute{\Omega}$, is radians/s and while the unit of digital frequency, $\acute{\omega}$, is radians.

The digital filter can be realized using a Digital Signal Processor (DSP). The DSP can be programmed to act as any kind of filter. This is one of the main advantages of digital systems.

In digital processing the signal is represented by a signal of numbers that are stored and then processed.

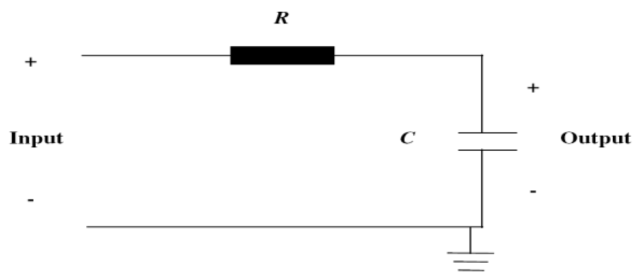

Figure 1. Analog realization of lowpass filter

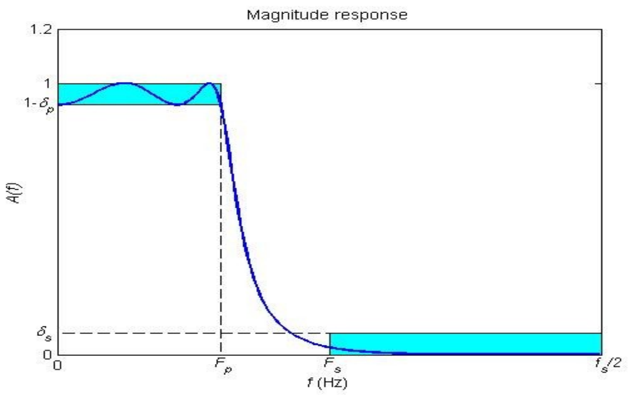

Figure 2. Output waveform of a lowpass filter

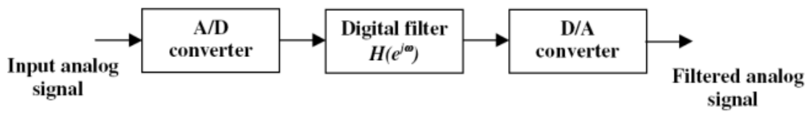

Figure 3. System block diagram of a digital lowpass filter

3. EXAMPLE

Using Mathlab the model is built in the Simulink and implemented on the ECG signal. In the model the Designed Low pass, high pass and notch filter has been cascaded figure shows the basic model used.

ECG amplifier gives the unfiltered output which contains the noise artifacts. The Power spectrum in the Figure Shows In the unfiltered signal the power line interferences as well as the high frequency noise is present. This nose is to be eliminated so that no information in the ECG signal missing. Figure 4 shows the Basic block diagram for the system used for the filtration of the ECG signal.

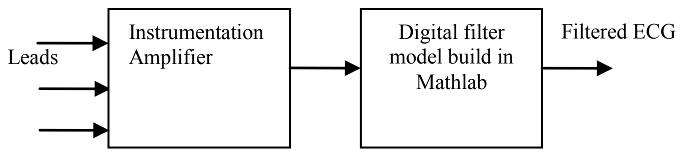

Figure 4 Block Diagram for system used for the noise reduction in the ECG

Below are presented the different lead combinations clearly showing the Noise reduction due to different filters. The filters work satisfactorily.

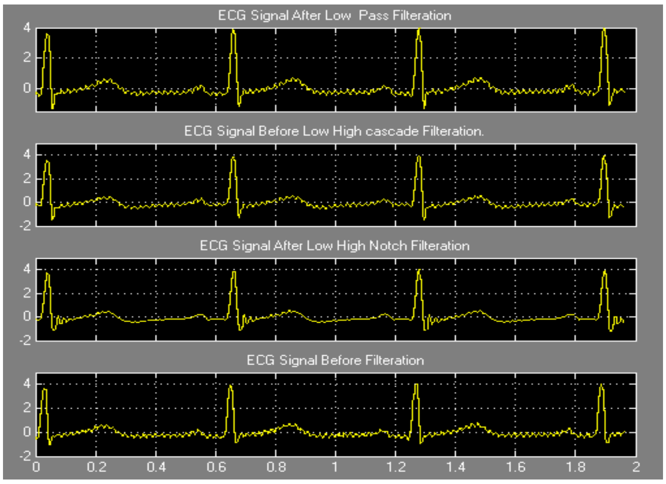

Figure 5.a ECG lead I signal of Chebyshev- II cascade filter.

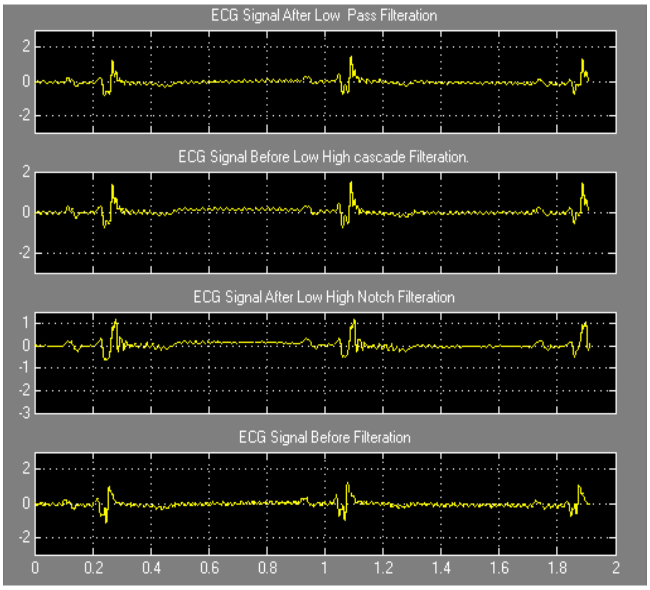

Figure 5.b ECG lead III signal of Chebyshev-II cascade filter.

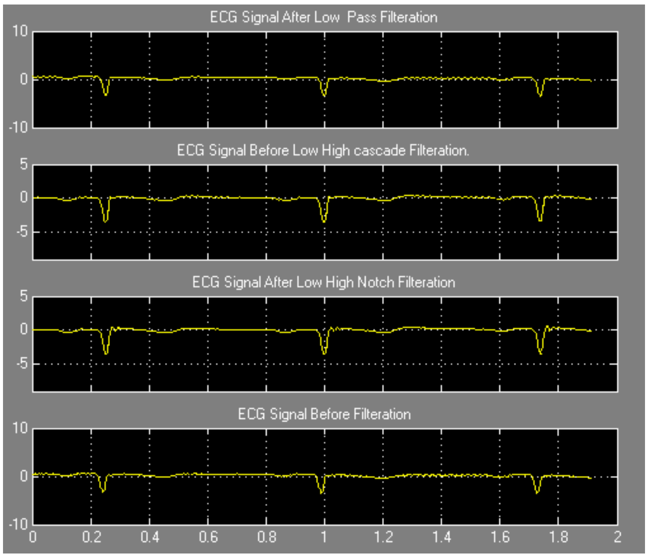

Figure 5.c ECG lead aVR signal of Chebyshev-II cascade filter.

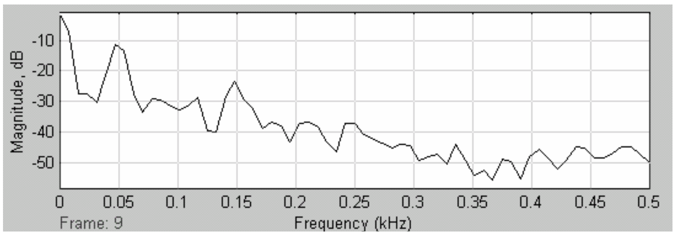

Figure 6.a Frequency Spectrum for Chebyshev II Filter Before Filtration of the ECG Signal

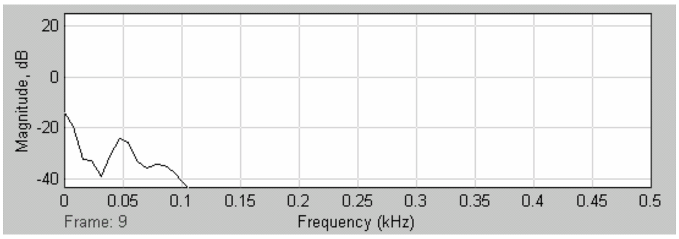

Figure 6.b Frequency Spectrum for Chebyshev II Filter after Filtration of the ECG Signal.

4. CONCLUSIONS

With visualization of above results by appropriate design if the Digital Chebyshev type II Filter the noise in the ECG signal can be effectively reduced.

Throughout our experiments we used MATLAB and SIMULINK in order to simulate de filter and process the input signal.

Some of the key features that distinguish digital filters from analog filters are: digital filters are programmable, implemented using software packages, can easily be changed or updated without affecting their circuitry (hardware), can be designed, tested and implemented on a general purpose computer, are independable of physical variables such as temperature, tolerances of elements, noise, fluctuations or inteferences. Analog filters use elements that are environment dependent, and any filter change is usually hard to implement and a complete design is often required.

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