EPJ Web of Conferences **318**, 05010 (2025) https://doi.org/10.1051/epjconf/202531805010
*AAPM-III 2025*

# Utilizing the LabVIEW software environment for investigating frequency characteristics of electrical circuits

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**Abstract.** This article explores the feasibility of digital bandpass filters within the LabVIEW environment. To facilitate the study of digital filters using virtual instruments, a dedicated digital filter simulation program was developed. This program provides users with various test mode views to aid in examining the Butterworth digital filter and modifying input actions and parameters. These parameters include throughput, order, rank, and slope; amplitude, offset, and frequency; the addition of noise types, such as white noise, over the input signal; selection of digital filter types, specifically bandpass filters; waveforms; graphical representations of input and output signals; and graphical representations of the harmonic spectrum in the filtered signal.

## 1 Introduction

When analyzing electrical circuits using the LabVIEW development environment, certain modeling limitations are present. However, the resulting errors are minimal and can generally be ignored. Additionally, it's important to consider the drawbacks of digital filters compared to analog filters when studying digital filter models. The primary objective of this research is to evaluate the pros and cons of digital filters within LabVIEW's virtual environment in comparison to their analog counterparts [1].

To achieve this, we established a virtual workplace and performed a series of tests on a Butterworth digital band-pass filter simulated within the setup. We adjusted the frequency, amplitude, and noise level of the incoming sinusoidal signal across a broad range to test the digital filter's performance, focusing on the following disadvantages compared to analog filters:

- Speed limitation: The maximum frequency range of signals processed by real-time digital filters is considerably smaller than that of analog filters.
- Real-time scenarios: The conversion time from analog to digital signals limits the maximum frequency that can be processed in real-time situations.
- Finite word-length effects: Digital filters are affected by ADC quantization noise, which must be considered in calculations during noisy approximations.

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- Longer design and production time: Designing and manufacturing digital filters and their hardware can take significantly longer than the design and assembly of analog filters.

The operating principle of a digital filter involves sampling the input signal through an ADC, processing it using a specific algorithm, and generating the output signal via a DAC [2]. This filtering principle is illustrated in Figure 1.

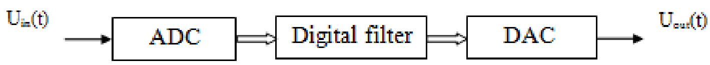


**Fig. 1.** Implementation principle of a digital filter.

A digital filter is a mathematical algorithm that can be implemented in hardware and/or software to process a digital input signal, producing an output digital signal to achieve the desired filtering effect. The term "digital filter" encompasses both the hardware and software components that perform this algorithm.

Digital technologies provide several advantages over analog filters, making digital filters often more favorable. These advantages include:

- Achieving high performance levels that are sometimes not possible with analog filters, such as attaining an ideal linear phase characteristic in mass production.
- Stability in performance, unaffected by environmental changes like temperature fluctuations, thus eliminating the need for frequent instrument calibration.
- The ability to handle a wide range of input signals without needing additional hardware resources for different channels [3].

## 2 Implementation of a LabView virtual workspace for studying a digital bandpass filter

To study digital filters using virtual hardware, a simulation diagram was developed specifically for this purpose. Figure 2 shows the front panel of the virtual test bench.

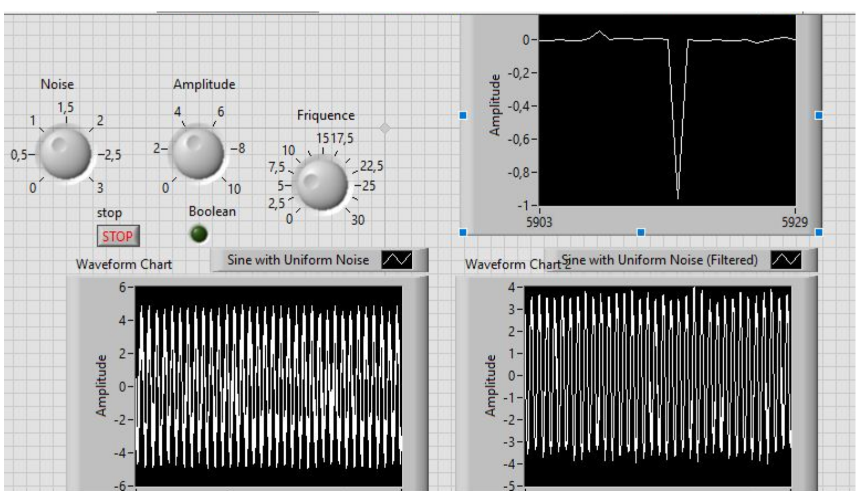


**Fig. 2.** Measurement Panel.

The block diagram of the program designed to study the Butterworth digital bandpass filter is shown in the image below. During testing, the input stimulus parameters for the digital filter, such as "Amplitude," "Noise," and "Frequency," can be adjusted. To conveniently display the signal shapes in the time domain, a dynamic scaling property was selected for the graphical indicator. Once execution begins, users can alter the signal type and parameters using the control elements on the front panel [4].

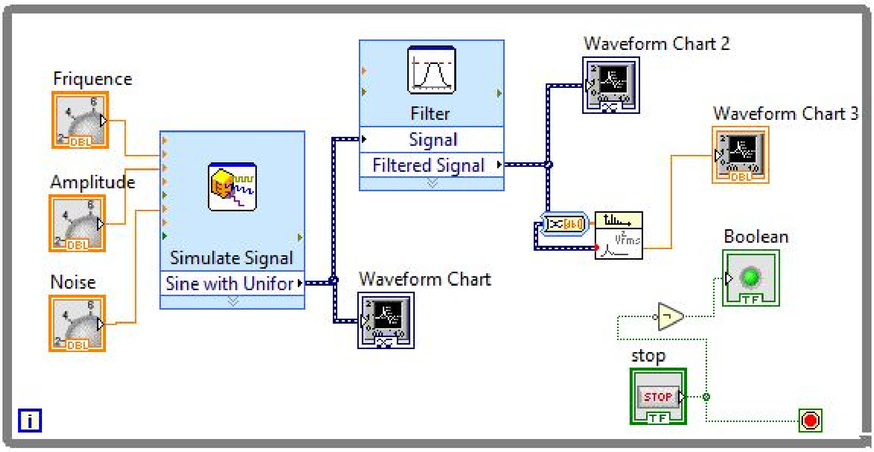


**Fig. 3.** Virtual Workspace Diagram.

In the initial stage of testing, a simple sinusoidal input stimulus without any "noise" component was applied. Both screens of the virtual oscilloscopes displayed (see Figure 4) the presence of approximately identical input stimuli.

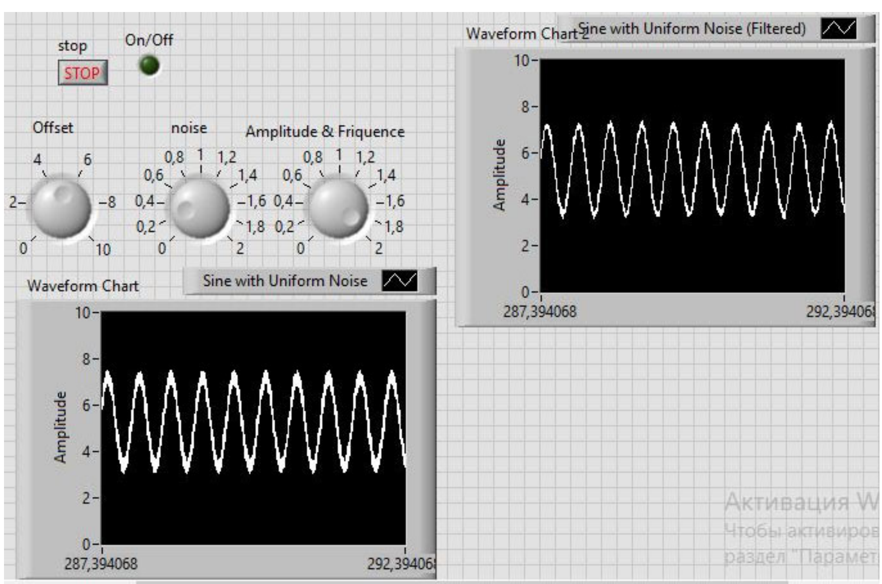


**Fig. 4.** Result of the First Experiment.

Note: The lower left screen of the oscilloscope demonstrates the input signal, while the upper right one shows the shape of the filtered input stimulus. In the second stage of the study, the amplitude of the noise component was increased. The result is shown in Figure 5.

In the result of this experiment (with a noise component of the input signal equal to 2V), a significant difference in the shape of the input signal with and without filtering was observed. It is evident from the figure that the digital filter largely preserved the sinusoidal shape of the input stimulus.

In the third stage of the study, both the amplitude and frequency of the input signal were reduced. The result is shown in Figure 6.

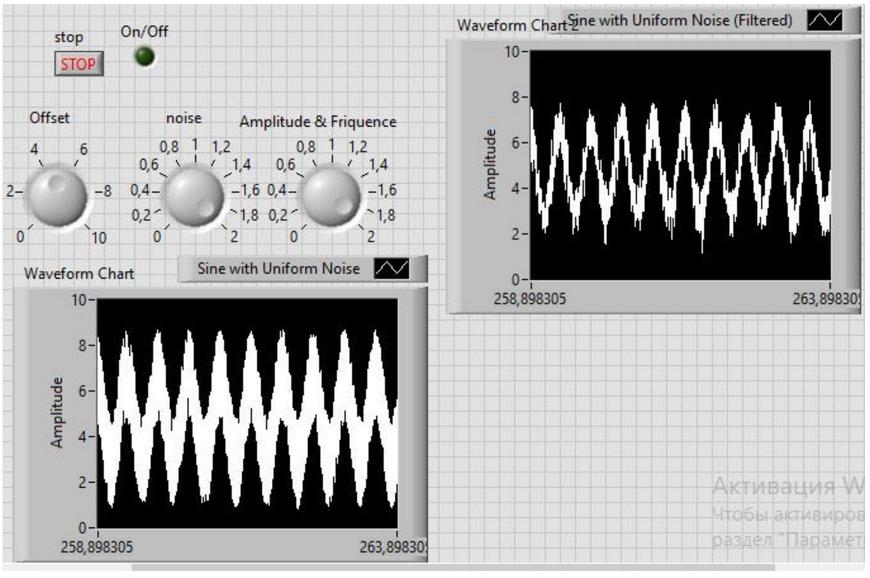


**Fig. 5.** Result of the Second Experiment.

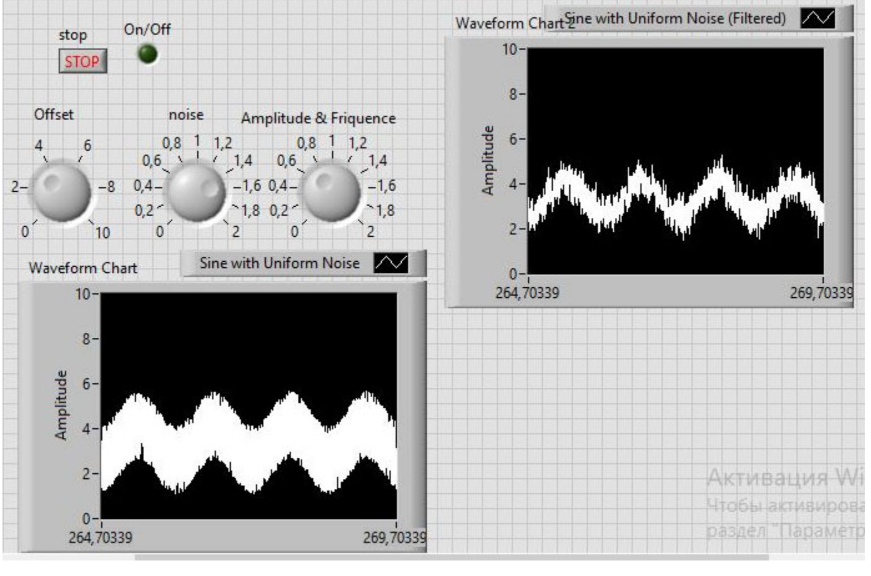


**Fig. 6.** Result of the 3rd Experiment.

In the third experiment, the signal shapes remained approximately the same as in the previous test. This indicates that if the incoming stimulus remains within the passband of the digital filter, it is capable of restoring the original signal shape.

In the subsequent experiments, extremely low frequency (see Figure 7) and then extremely high frequency (see Figure 8) of the input signal were sequentially applied [6].

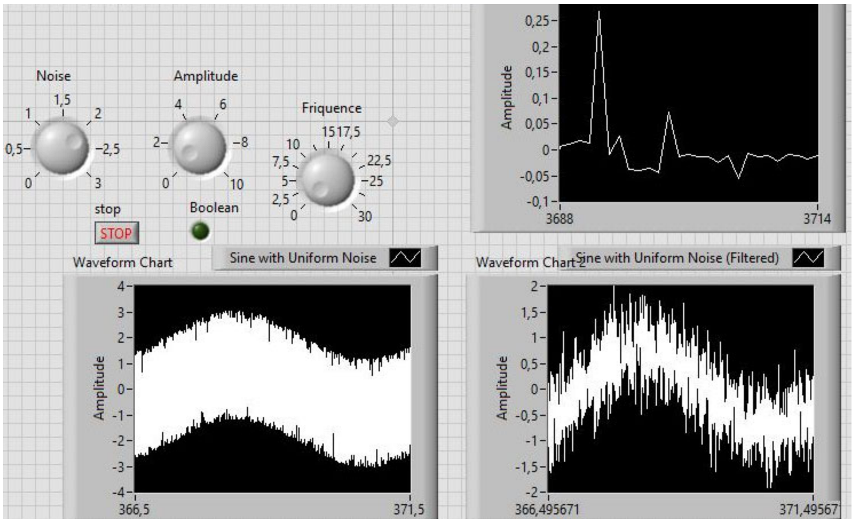


**Fig. 7.** Result of the 4th Experiment. Lower frequency limit of the input signal.

Note: The upper right screen (refer to Figure 7 and Figure 8) displays the spectrum of the input signal after passing through the digital filter.

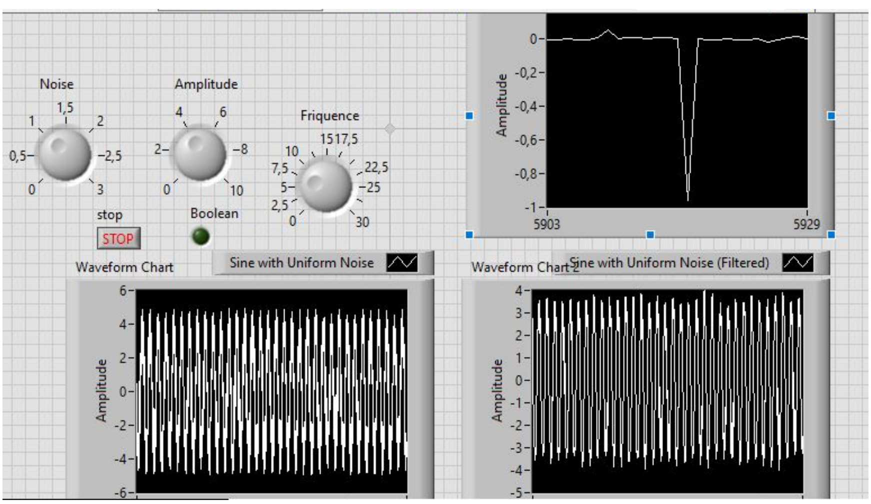


**Fig 8.** Result of the 4th Experiment. Upper frequency limit of the input signal.

The results of the final stage of the study showed that the signal shapes with or without filtering were practically indistinguishable, as the input stimulus exceeded the frequency bandwidth of the digital filter.

The parameters of the tested filter model are provided in Figure 9.

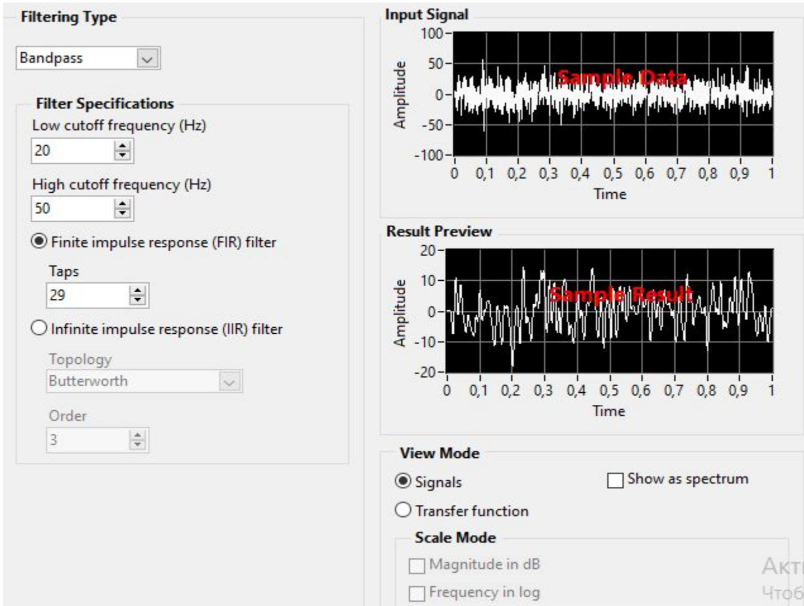


**Fig. 9.** Parameters of the Digital Filter Model.

The parameters of the input stimulus generator model are provided in Figure 10.

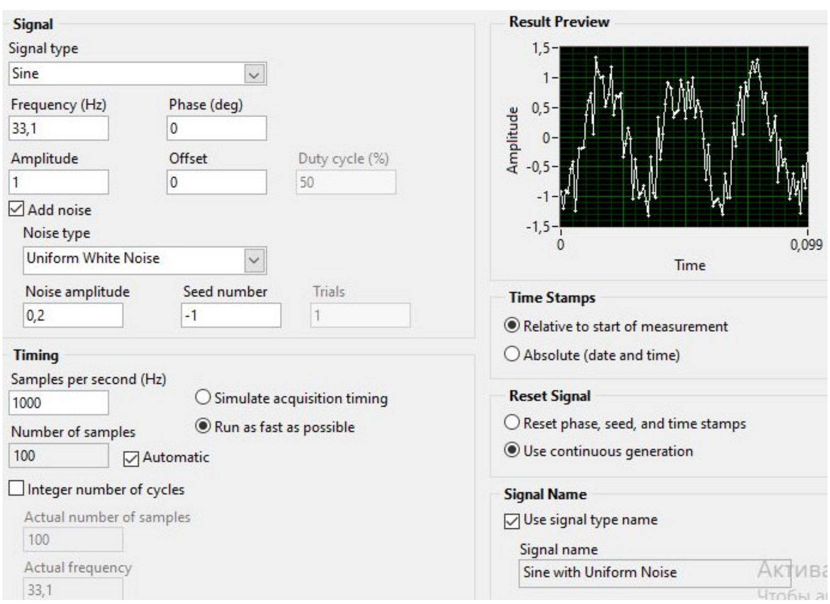


**Fig. 10.** Parameters of the Input Signal Generator Model.

## 3 Conclusion

Using the controls and indicators available in the LabVIEW programming environment, users can design their own virtual instruments essential for studying electrical circuits during the design phase. Each application can be configured so that any change in filter parameters (or any other device) or input signal parameters is reflected in the updated signal shape at the filter output, along with its frequency characteristics.

In the realm of digital filters, LabVIEW offers valuable functions for determining the coefficients of finite impulse response (FIR) or infinite impulse response (IIR) filter polynomials. The library of functions dedicated to signal analysis and data processing enables users to create their own applications for studying digital filters, as well as models of other electrical circuits.

By leveraging these LabVIEW tools, we created a specialized virtual test bench to conduct a series of tests on the digital filter. We compared the filtering results across various input signals with different noise components. The digital filter models perform almost identically to their analog counterparts. In LabVIEW, the characteristics of a digital filter can be easily adjusted through dynamic software control of the command processor.

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