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Designing a frequency bandpass filter using a universal active filter

Bachelor's thesis

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Abstract

This paper describes what universal active filters are, their advantages and disadvantages compared to passive ones, as well as the method of realization for the purposes of obtaining a 50Hz frequency-pass filter. Attached is the procedure for theoretical and software determination of the parameters required for the realization of the requested filter, analysis of filter properties on real equipment in real conditions, as well as a comparison of the obtained results with the theoretical ones. The filter was implemented as a second- and fourth-order filter, a comparison was made and a conclusion was drawn about the price-performance ratio for the two systems. The filter used is UAF42(*Universal Active Filter*) manufactured by Texas Instruments.

Keywords: Filter42, LTspice, Notch filter, UAF42, Universal filter

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Chapter 1

Introduction

A filter is a device that removes unwanted signal components, which means that certain frequencies can be removed and others used in order to suppress the interfering signal and reduce noise. Because of this, filters play an important role in communication. With the advancement of the industry, many manufacturers are looking for new ideas to produce filters that can be a band pass filter, a low pass filter and a high pass filter at the same time. In addition to those mentioned, it is possible to implement a frequency-pass filter as well as an all-pass filter by combining the three previously mentioned filters. This type of filter, which can be implemented as any of the above, is called a universal filter. The motive for studying the given topic is given in the next paragraph.

Namely, alternating current flows through every mains socket in Europe, which always has the same oscillation frequency of 50Hz. The so-called nominal frequency of alternating current on our continent and in most of the world is 50Hz. This number indicates that the current, relatively speaking, changes direction fifty times per second, and that the rotors of all generators in the power plant rotate at a speed of fifty times per second. The question that arises is why exactly fifty times? Today, there are no special technical reasons for choosing that particular frequency, except for the obvious necessity that everyone uses the same standard. Therefore, the question arises, how to remove the network frequency signal in situations where we do not need it? As one of the main problems, I would single out the influence of the network frequency on medical instruments when performing measurements, where the measurement results would vary significantly if the network frequency was not suppressed. The answer to the mentioned question as well as the solution to the problem and the method of realization is attached in this paper.

Chapter 2

Methodology

2.1 Filter Features

The use of filters today plays a very important role in various spheres of electrical engineering. They are used for two purposes - for signal separation as well as for its restoration. Signal separation is required when the signal is subject to interfering signals, noise or other unwanted signals. Let's imagine a device for measuring the activity of the baby's heart in the mother's womb. The measured signals would be corrupted by the mother's breathing and heartbeat. Therefore, a filter would be useful, which would enable the extraction of a given signal as well as its individual analysis.

Signal restoration is used when the signal is corrupted in some way, as can be the case when an audio signal is recorded with poor quality equipment or also to improve the quality of a poorly recorded image. The solution to these problems can be obtained by using an analog or digital filter. The question is which one is better? Analog filters are cheap, fast and have a large dynamic range for both amplitude and frequency. Digital filters are significantly superior in terms of the level of performance they can achieve. This leads us to the conclusion that analog filters have their limitations based on electronic components such as the tolerances of passive components as well as their accuracy, while with digital filters the limitation is reflected in the signal, i.e. the theoretical possibilities that the signal and its processing satisfy.

2.2 Description of the universal active filter UAF42

Depending on the components they contain, filters can be passive or active. Passive ones contain only passive components, while active ones also have some active component. The filter that was used to process the given topic contains operational amplifiers that are active components, which means that the UAF42 filter is a universal active filter. Also, these filters provide a gain and the limit of strictness of passing is based on the Q-factor (*Quality factor*). The higher the value of the Q-factor, the narrower the limit of the pass and stop bands of the filter.

From all of the above, I conclude that universal filters can be of great use and find their application in different spheres. Apart from the applications mentioned in the previous sections, these filters are also used in sound systems where they can be used to obtain a clean output signal without noise. In noisy areas, buildings and offices may have filters to remove outside noise. Also, there are special earplugs with a filter in them for people who work in extremely noisy places.

Some of the disadvantages of universal active filters are that they are not operational at high power and that by changing the value of external passive components in order to change certain parameters of one filter, the parameters of another filter can be affected. Figure 2.1 shows the block diagram of the universal active filter UAF42.

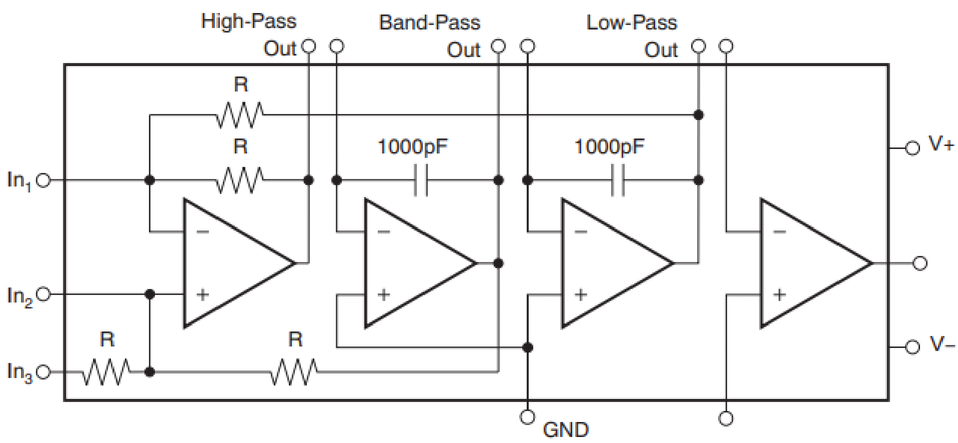

Figure 2.1: *Block diagram of the universal active filter UAF42.*

The universal active filter UAF42 from Texas Instruments was used for the implementation of the 50Hz frequency stop filter. It is based on the classic analog architecture and consists of three operational amplifiers - two integrators and one inverting amplifier. The chip also contains a fourth operational amplifier which is there for the flexibility of the chip. Internal passive components have a tolerance of 0.5% to ensure stable performance. The filter is compatible to implement multiple types of filters, such as Butterworth, Bessel and Chebyshev. Available in 14-pin plastic DIP and SOIC-16 versions.

The filter contains two built-in capacitors of 1000pF each and four resistors of resistance of 50kΩ each. There are two input pins for 20V power supply ($V+$ and $V-$), an input signal pin V_{IN}, an output signal pin V_{OUT} while the other pins are used to connect external components. The Band-Pass Out, High-Pass Out, and Low-Pass Out pins can also be used as output pins, which provide the output of a band-pass filter, a high-pass filter, and a low-pass filter, respectively.

Figure 2.2 shows the pin numbering and configuration scheme of the analyzed filter in the 14-pin plastic DIP version.

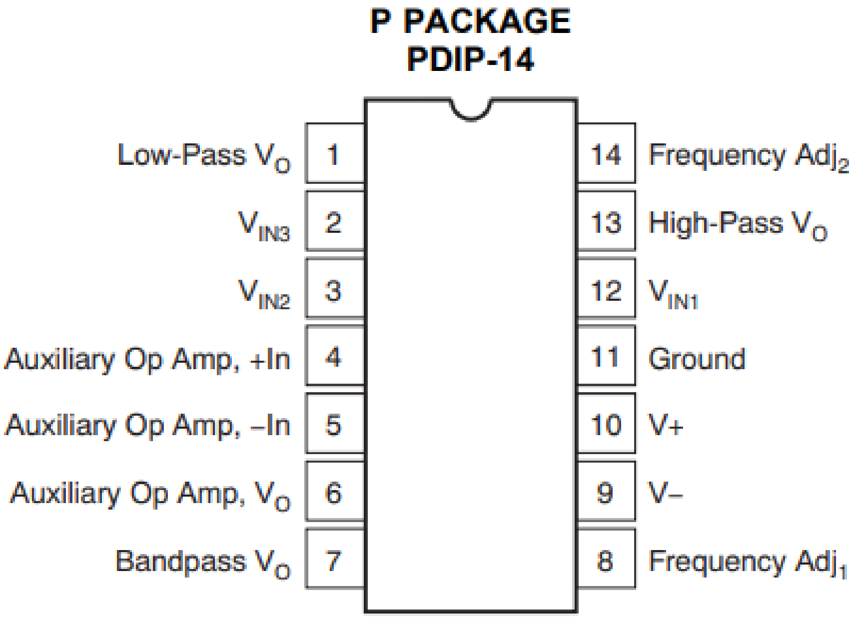

Figure 2.2: *Numbering scheme and pin configuration of the universal active filter UAF42.*

2.3 Calculation of second-order filter parameters

A second-order band-pass filter can be easily implemented by adding six resistors to the circuit. Figure 2.3 shows a block diagram of a universal active filter configured as a frequency-pass filter.

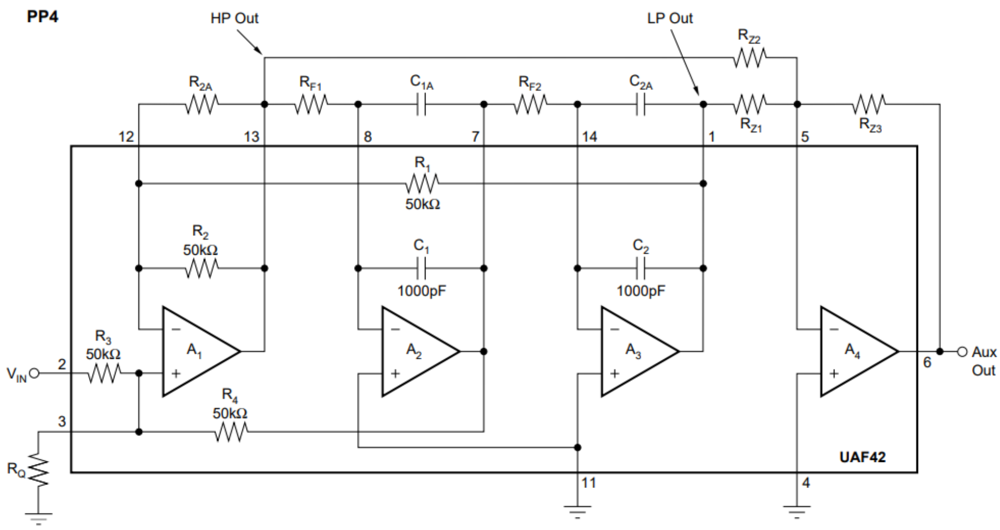

Figure 2.3: *Block scheme of the universal active filter UAF42 implemented as a frequency filter.*

Next follows the calculation of the filter parameters for the requested frequency $f_{NOTCH} = 50Hz$, goodness factor $Q = 10$ and filter order $n = 2$.

The notch frequency is obtained from the expression:

$$f_{NOTCH} = \sqrt{A_{LP}/A_{HP} * R_{Z2}/R_{Z1}} * f_0, \tag{2.1}$$

where : A_{LP} - low-pass output gain for $f = 0$, A_{HP} - gain on High-pass output for $f >> f_0$.

The expression $A_{LP}/A_{HP} * R_{Z2}/R_{Z1}$ is set to one for simplicity of calculation which gives the expression for f_{NOTCH}:

$$f_{NOTCH} = f_0, \tag{2.2}$$

for f_0 given as:

$$f_0 = \frac{1}{R_F * C * 2\pi}, \tag{2.3}$$

where $R_F = R_{F1} = R_{F2}$ and $C = C_1 = C_2$.

Note that f_{NOTCH} can be modified by changing the resistance R_F and adding new resistors or capacitors. The three-decibel bandwidth can be adjusted according to the equation:

$$BW_{-3dB} = \frac{f_{NOTCH}}{Q}, \tag{2.4}$$

where $BW_{-3dB} = f_H - f_L$. The Q factor can be adjusted based on the resistance R_Q using the equation:

$$R_Q = \frac{25k\Omega}{Q-1}, \tag{2.5}$$

The attenuation in the bandwidth of the filter can be adjusted by the factor Q according to the formula:

$$Q = \frac{R_{Z3}}{R_{Z1}} = \frac{R_{Z3}}{R_{Z2}}, \tag{2.6}$$

Note that the parameters f_0 and Q can be adjusted independently with an adequate choice of R_{F1}, R_{F2} and R_Q. By substituting the value of the notch frequency of 50Hz and the value of the adopted goodness factor into the previous equations, the necessary parameter values for the realization of the second-order frequency-stop filter are obtained:

$$f_{NOTCH} = f_0 = 50Hz,$$

$$f_0 = \frac{1}{R_F * C * 2\pi} \implies R_F = R_{F1} = R_{F2} = 3.18M\Omega,$$

$$BW_{-3dB} = f_H - f_L \implies BW_{-3dB} = 5Hz,$$

$$R_Q = \frac{25k\Omega}{Q-1} \implies R_Q = 2.78k\Omega,$$

$$Q = \frac{R_{Z3}}{R_{Z1}} = \frac{R_{Z3}}{R_{Z2}} \implies R_{Z1} = R_{Z2} = 2k \text{ ohm}, R_{Z3} = 20k\Omega.$$

In order to check the obtained results, a calculation was also made in the software tool FILTER42, which adjusts the resistor and capacitor parameters for a given frequency f_{NOTCH} and goodness-of-fit factor Q. Figure 2.4 shows the calculated values of the parameters, while Figure 2.5 shows the obtained amplitude spectrum for the adopted parameters.

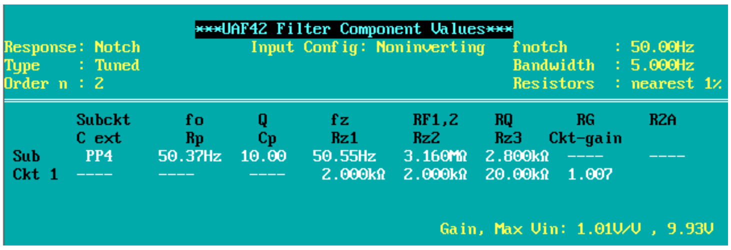

Figure 2.4: *Second-order filter parameters obtained with the FILTER42 software tool.*

Let's note that the software calculation does not fully match the theoretical one, which will result in minor deviations that will not manifest themselves to a large extent on the given orders of magnitude. The FILTER42 software tool provides parameter values of passive components that need to be connected in a circuit in such a way that they are close to the required theoretical ones, but also that they can be purchased in retail.

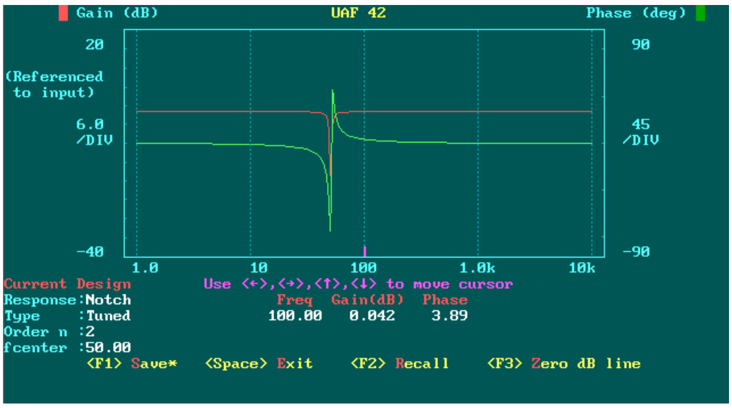

Figure 2.5: *Amplitude characteristic of the second-order filter obtained with the software tool FILTER42.*

2.4 Calculation of fourth-order filter parameters

Due to the more complex calculation of the parameters for the fourth-order filter, the parameters were obtained by software. Increasing the order of the filter increases the complexity of the implementation, so the given calculation concludes that a fourth-order frequency-pass filter can be realized by connecting two filters in series, where the output of the first filter represents the input of the second. Figure 2.6 shows the values of the required filter parameters calculated with the FILTER42 software tool.

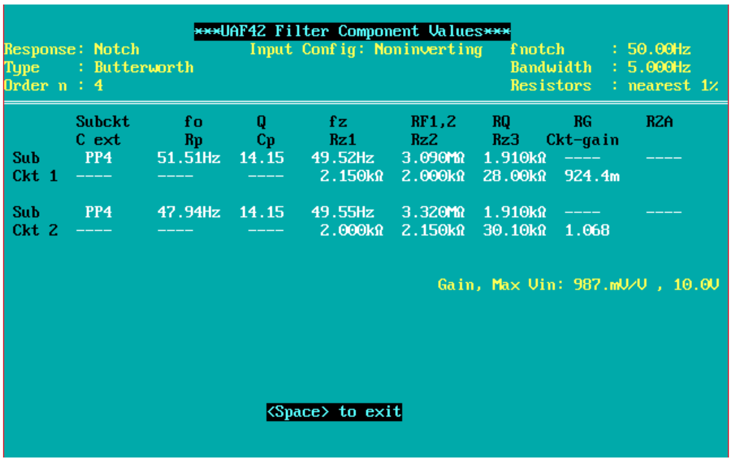

Figure 2.6: *Fourth-order filter parameters obtained with the FILTER42 software tool.*

It is expected that an increase in the order of the filter will result in a higher attenuation in the impervious part of the band, as well as a smaller fluctuation of the attenuation value in the permeable part of the band. Figure 2.7 shows the amplitude characteristic of the fourth-order filter obtained with the FILTER42 software tool.

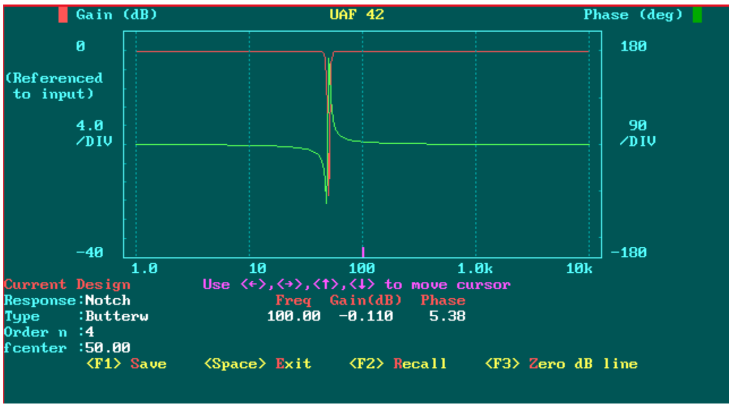

Figure 2.7: *Amplitude characteristic of the fourth-order filter obtained with the software tool FILTER42.*

By comparing it with the amplitude characteristic of the second-order filter, it is concluded that the attenuation is more pronounced with the fourth-order filter, which is the expected conclusion, and that the transition from the permeable to the impervious part of the band is narrower.

Chapter 3

Results

3.1 Design and implementation

First, I would like to present the results obtained for the calculation of the second-order frequency bandpass filter. After the calculation and connection of the given components, the block diagram of the filter is given in the Figure 3.1.

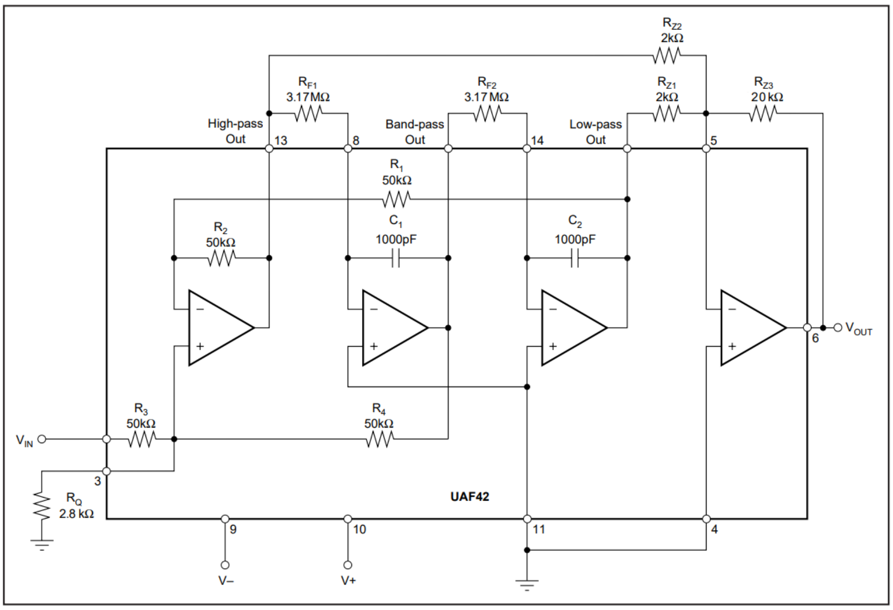

Figure 3.1: *Block diagram of a second-order universal active filter set as a frequency stop at 50Hz.*

Capacitances C_{1A} and C_{2A} are zero, so they represent an open connection and therefore do not appear in the picture. Pin V_{OUT} is connected to the input of the oscilloscope where the time form of the output signal is read. Pin V_{IN} represents the signal generator input. The input signal is of the form $U * Sin(2 * \pi * f * t)$, where the amplitude U is set to 1V and the frequency f at the initial moment is 20Hz and it will change in the range of 20Hz to 90Hz with 1Hz step. At each point, the maximum values of the output and input signals are measured, dividing and logarithmizing the value of the component of the amplitude characteristic at the given point is obtained.

Using the Python software tool, the amplitude characteristic of the filter was determined on real components and in real conditions, which should have the appearance of the curve given in the Figure 2.5 obtained by simulation. It is noticeably weakened at a frequency of 50Hz, which was the goal of the gain. Outside the impervious range, the attenuation varies around zero, which is a consequence of the imperfection of the components as well as the impossibility of realizing the filter according to the ideal parameters obtained in the theoretical calculation in section 2.3. The amplitude characteristic obtained by the software tool Python is given in the Figure 3.2:

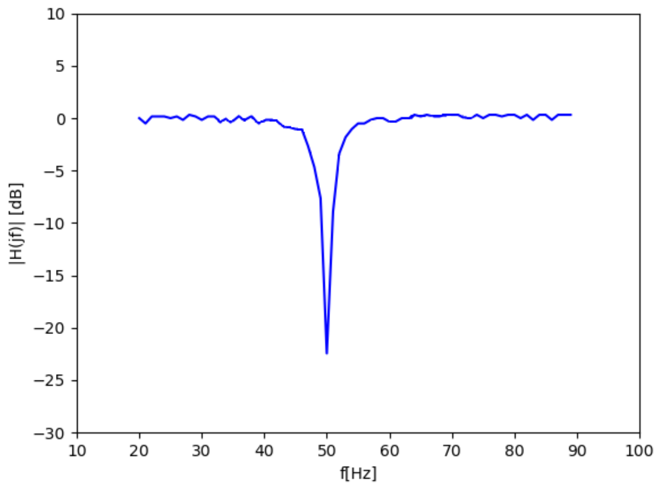

Figure 3.2: *Amplitude characteristic of the second-order filter obtained with the Python software tool.*

The results for the fourth-order filter are shown below. The block diagram of the fourth-order filter is obtained by serial connection of the two blocks shown in Figure 3.1 with the parameter values from Figure 2.6. The signal generator is connected to pin 2 of the first filter, pin 6 of the first filter represents the input signal of the second filter on pin 2, while an oscilloscope is connected to pin 6 of the second filter, which displays the time form of the output signal. The amplitude characteristic of the fourth-order filter obtained by the software tool Python is given in Figure 3.3.

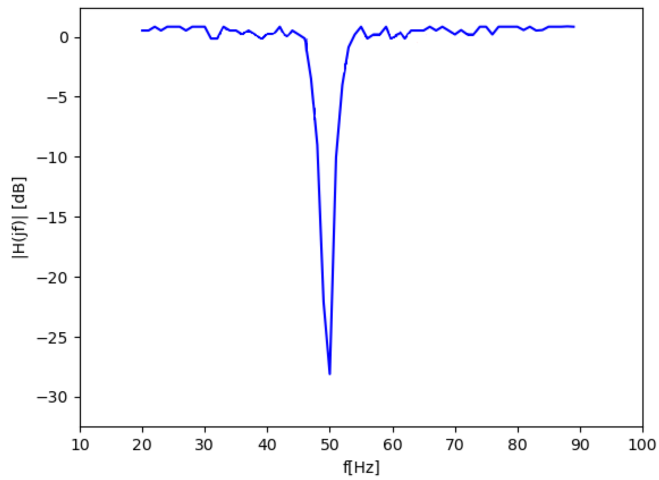

Figure 3.3: *Amplitude characteristic of the fourth-order filter obtained with the Python software tool.*

By comparing the amplitude characteristics of the second and fourth order filters, we come to the conclusion that increasing the order of the filter results in a greater attenuation in the impervious part of the range of about 5dB and that the fluctuation in the permeable part of the range is still pronounced. The transition from the impervious to the permeable part of the band is narrower, which leads us to the conclusion that the increase in the order of the filter led to a greater attenuation in the part of the band that is of interest, but also that this increase is not drastic compared to the filter of the second order.

3.2 Project-realization comparison

Let's then compare the previously obtained results. It is clear that the values obtained with the FILTER42 software tool will differ from the results obtained by working on real equipment. Figures 3.4 and 3.5 show the amplitude characteristic obtained in real conditions in blue, while the values of the amplitude characteristic in software predefined points and with a certain step from the FILTER42 program are marked in red.

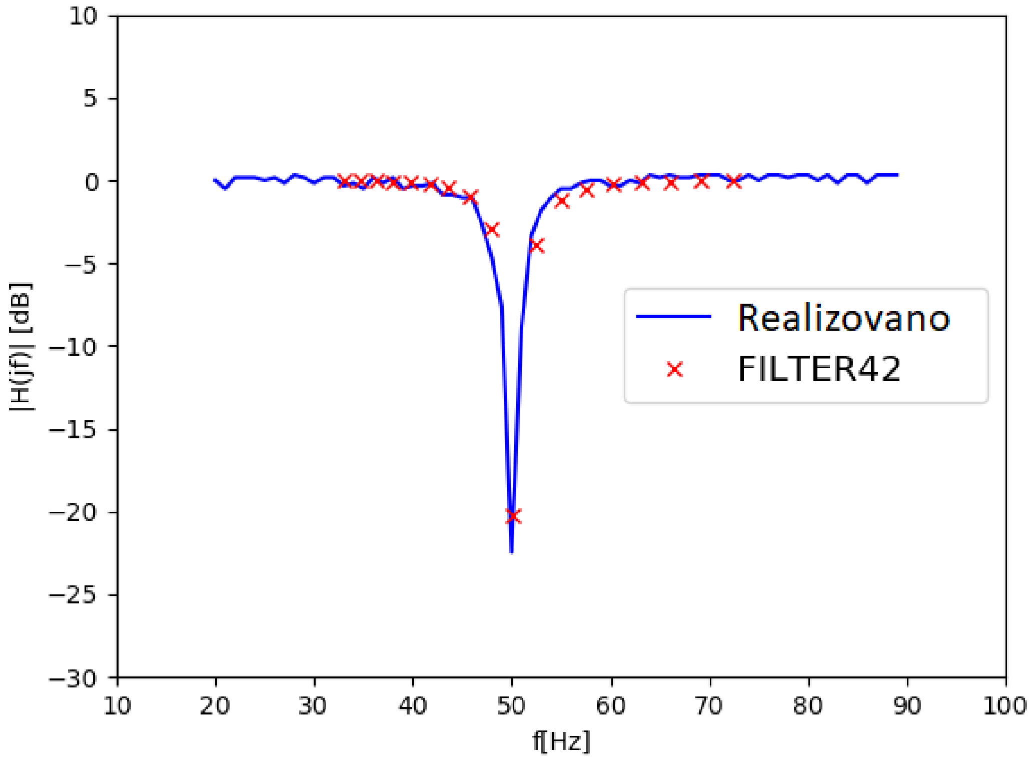

Figure 3.4: *Comparison of amplitude characteristics for a second-order filter - also realized by FILTER42.*

From the pictures it can be concluded that for the second-order filter there are no major deviations in the attenuation value at the frequency of 50Hz, while for the fourth-order filter the software-obtained maximum attenuation value is unreachable in real conditions. Attenuation in the impervious part of the band for both filters has pronounced fluctuations in the attenuation values, but they are close to the software calculation values and have values around zero.

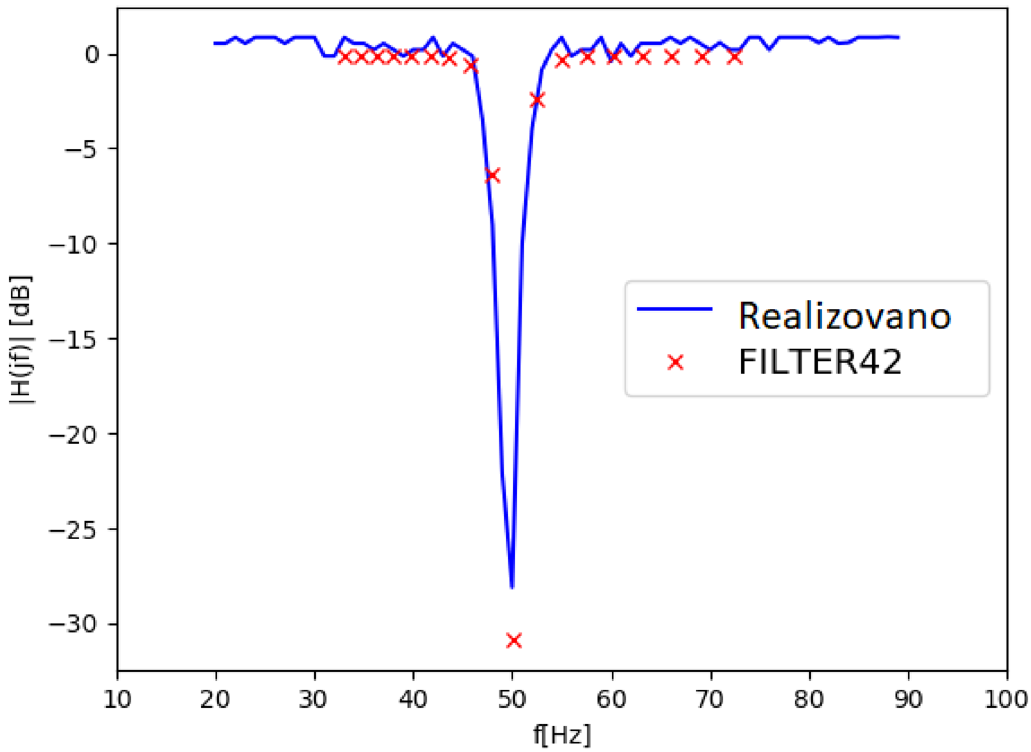

Figure 3.5: *Comparison of amplitude characteristics for a fourth-order filter - also realized by FILTER42.*

3.3 Comparison ideal-designed

Let us now compare the results obtained for the theoretical values of the passive components calculated in sections 2.3 and 2.4. Figure 3.1 shows a block diagram of a second-order universal active filter set as a frequency stop at 50Hz. First of all, it is necessary to realize the given block diagram in a software tool in order to perform the simulation. Figure 3.6 shows the block diagram of the second-order filter implemented in the LTspice software tool.

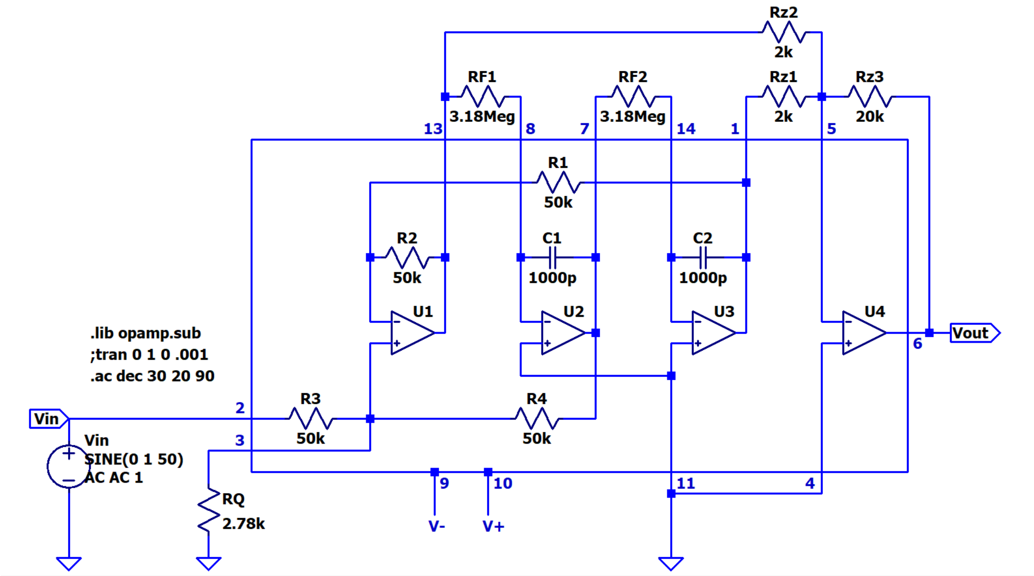

Figure 3.6: *Block diagram of a second-order filter set as a 50Hz bandpass obtained using the LTspice software tool.*

Figure 3.7 shows the amplitude characteristic of the filter. Let's note that in ideal circumstances, such as the Spice simulation, the attenuation value in the bandwidth part of the band is constant and equals 0, while for the frequency value of 50Hz, a considerable attenuation of about 23dB is expressed.

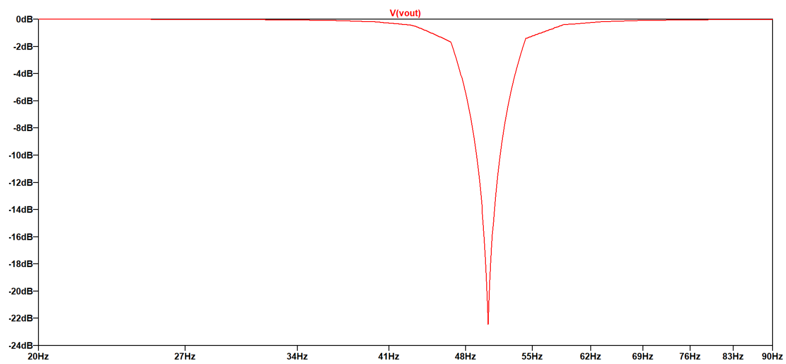

Figure 3.7: *Amplitude characteristic of the second-order filter obtained with the LTspice software tool.*

Let's compare the results obtained for the fourth-order filter. Figure 3.8 shows the realization of the fourth-order filter set as a frequency stop at 50Hz, and it is represented by the series connection of two blocks from Figure 3.6.

It is expected that, as in the case of the implementation of the second-order filter, the value of the attenuation in the impervious part of the range will be more pronounced, while the attenuation in the permeable part of the range should also be 0. Figure 3.9 shows the amplitude characteristic of the fourth-order filter.

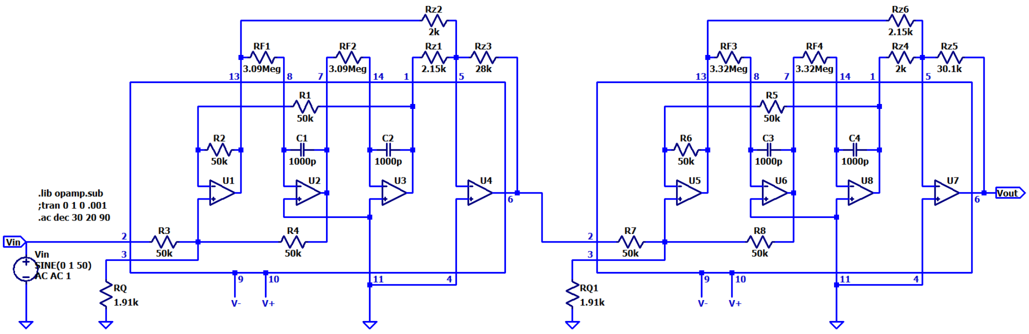

Figure 3.8: *Block diagram of a fourth-order filter set as a frequency stop at 50Hz obtained using the LTspice software tool.*

From the given image, it can be seen that a greater attenuation has been achieved, which is close to 27dB, which is approximately 4dB more than the second-order filter. The difference of 4dB is not big, but it certainly cannot be ignored. Depending on the sphere of application, this difference in attenuation can be of great use, but in the application discussed in this paper it is not of great importance.

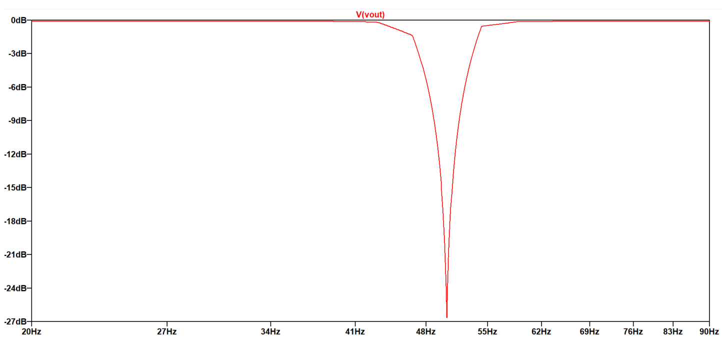

Figure 3.9: *Amplitude characteristic of the fourth-order filter obtained with the LTspice software tool.*

Chapter 4

Conclusion

The paper presents the procedure of how it is possible to obtain the parameters necessary for the implementation of a 50Hz frequency-stop filter with simple calculations. It is easy to calculate the values of the parameters required for the implementation of other types of filters, which also indicates that the implementation is extremely simple. By replacing the values of external passive components, different types of filters can be obtained according to needs.

In the presented work, although the components are not of identical values as those calculated theoretically, the error obtained at the output is negligible and it is possible to remove it to a large extent by an adequate selection of components that will fully satisfy the above equations as well as by increasing the order of the filter, which is also shown.

Also, it was concluded that the price-performance ratio of the filter implemented as a fourth-order filter in the treated topic is not of great importance, but it should definitely be considered when applied in other areas more sensitive to the given problem.

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