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RESEARCH ARTICLE

A Digital Filter Design for Reactive Power Measurement

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ABSTRACT

The electric meters used to measure the amount of electricity consumption measure both active and reactive power consumption in the distribution line with current transformers. As in every analog measurement process, in every analog measurement, existing of undesired components such as noises, interferences and harmonics are caused mistakes. In order to remove this undesirable information included in the measurement from the measurement results, digital filters are used in applications because of their performance and their easily replaceable features. In this study, we aimed to remove unwanted components such as noise and interference from the measurement signal. The subject of a commercial project is to design a digital filter in MATLAB® environment, which allows filtering the measurements of reactive power consumption in power distribution lines with less than 1% error. To ensure whether designed filter satisfies for mentioned criteria or not, we used the Gaussian model as a noise model because Gauss noise is spreaded whole frequency spectrum. While the power signal modeled in the design is a sine wave of $I_{pp} = 2$ A 50 Hz, Gaussian noise with different variances and mean values is used in modeling the unwanted components.

Index Terms—Digital filters, digital signal processing, IIR, reactive power measurement

I. INTRODUCTION

Electric meter is used to calculate the electrical energy consumption. In order to determine the energy consumption, the voltage of the distribution line and the current drawn from the power distribution line need to be measured. Current transformers are used to measure information of the current drawn from the power distribution line. Because the current measurements are analog, the results obtained from the current transformer are converted into digital by the analog-digital converter units in the microcontroller and the consumption amount is calculated. As in every analog measurements processes, in every analog measurement, existing of undesired components such as noises, interferences and harmonics are caused to corruption on the measurement. In many analog measurement applications, it is used coupling capacitors to remove the DC components and, also use anti-aliasing filters to limit the frequency band of the measurement. But there are still noise, interference, and harmonic. Filters are used to remove these components from the measurement. Digital filters are basically of two types:

1. Infinite impulse response (IIR)
2. Finite impulse response (FIR)

The mathematical representation of these filters is shown below. The IIR filter equation is given in (1) and the FIR equation is given in (2).

$$y(n)=\sum_{k=0}^{N} b_k x(n-k)-\sum_{i=1}^{M} a_k y(n-i) \tag{1}$$

$$y(n)=\sum_{k=0}^{N} b_k x(n-k) \tag{2}$$

There is a difference between the IIR and the FIR and its feedback mechanism. The IIR filter is a digital filter that has only feedback mechanism and this advantage give an ability to the IIR filter like to take the same filtered result in low filter order than the FIR filter.

There are many realization types of digital filters such as direct form-I, direct form-II, cascade, and parallel in literature. In this research, we used the direct form-I one because of its user-friendly usage. The illustration of this realization type for the IIR and the FIR is in Fig. 1 and Fig. 2, respectively.

In the literature review, the researchers who studied about the same field used six different filter design techniques to filter the measurements in power consumption. These are as follows:

1. Serial IIR low-pass filter [1],
2. IIR band-pass filter [2],

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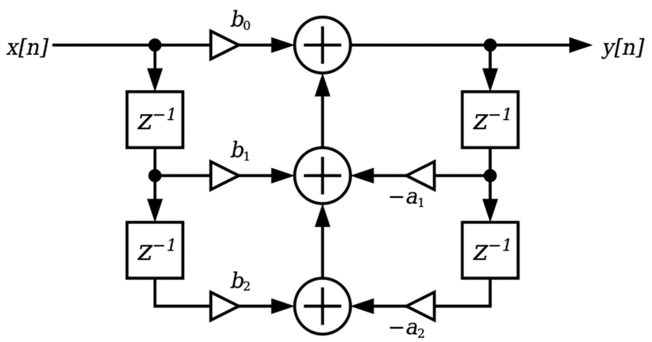

Fig. 1. Realization of the IIR filter in direct form-I type. IIR, infinite impulse response.

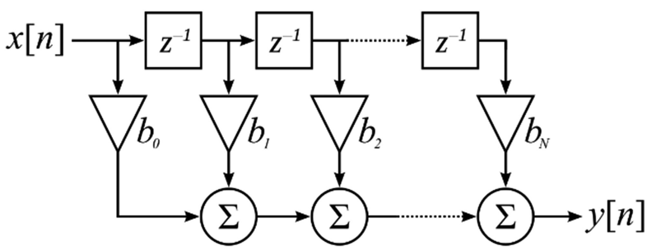

Fig. 2. Realization of the FIR filter in direct form-I type. FIR, finite impulse response.

3. Parallel IIR low-pass filter [3],
4. Adaptive low pass filter [4],
5. IIR low-pass filter [5],
6. FIR low-pass filter with phase-orthogonal approach [6].

The common aspect of the studies is that the designed filters are run on either digital signal processing (DSP) microprocessors or field-programmable gate array. The cost of the designed filters is as important as their performance. Therefore, the designed digital filter is intended to work in a low-power microcontroller. Also, this filter will produce results with less than 1% probability of error.

When the measurement results are examined, the current value transferred from the power distribution line to the system is $I_{PP} = 2.08$ A. As it is known, the city network is a 50 Hz sine wave. So, a 50 Hz sine wave was generated in MATLAB to model the measurement results and perform error analysis. A random number generating Gaussian noise was added to this generated sine wave. In the other steps, the filtered and unfiltered results were compared and their compliance with the determined criteria was checked.

Main Points

- Current value transferred from distribution line to the system is peak to peak 2.08A.
- The city network is 50 Hz sine wave so that we set a model which is pure sine wave with peak to peak current value 2A.
- Also we used Gaussian noise as the noise model.
- To check the filters performance, we compared difference between unfiltered and filtered results.

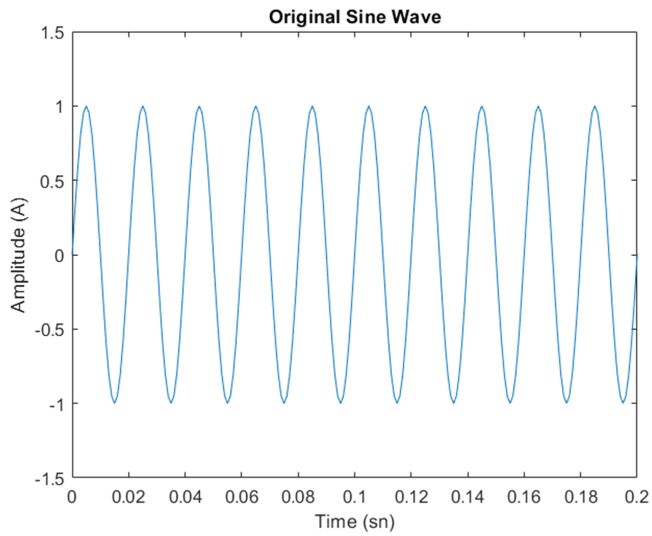

Fig. 3. 10 Periot 50 Hz $I_{pp} = 2$ A Sine wave in time domain.

II. GENERATION OF NOISE SIGNAL AND DESIGNING THE FILTERS

A. Generation of the Noise Signal

Power systems are 50 Hz sine wave in our country. Therefore, the system is modeled as 50 Hz sine. Gaussian noise model was used as a noise model because Gaussian noise is spreaded in whole frequency band as equally. Besides, where the noise power that we interested is not known where it is. For this reason, we used the Gaussian noise in our study as the noise model.

The outputs of the generated sine wave in the time and frequency domain are shown in Fig. 3 and Fig. 4, respectively.

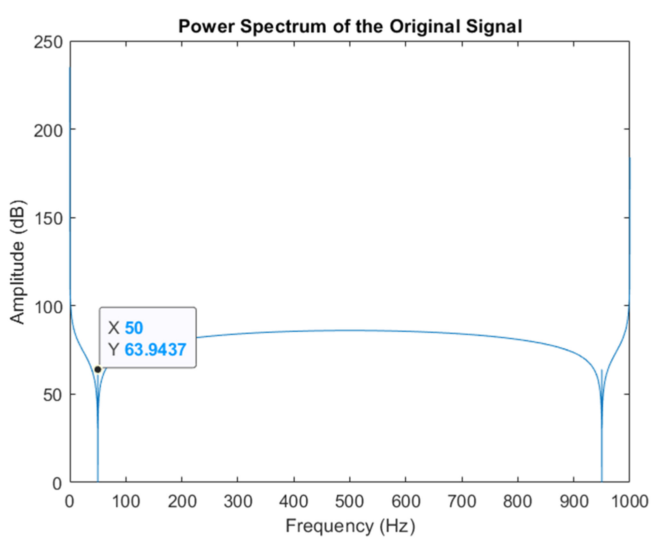

Fig. 4. 50 Hz $I_{pp} = 2$ A Sine wave in frequency domain.

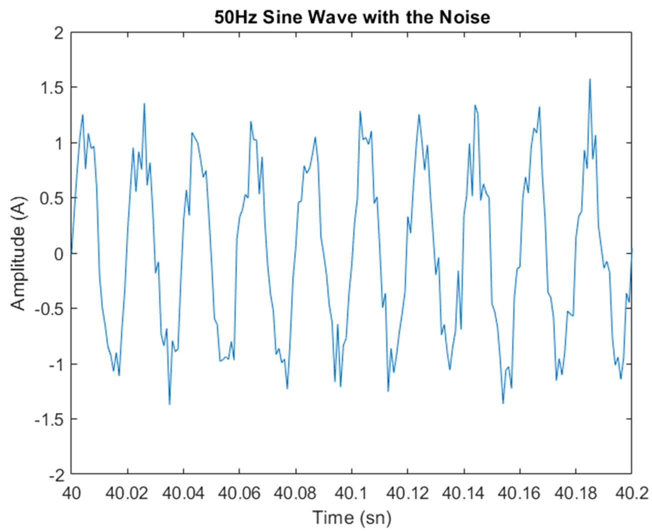

Fig. 5. 50 Hz $I_{pp} = 2$ A Sine wave with $\sigma = 0.2$, $\mu = 0$ Gaussian noise in time domain.

Gaussian noise has three parameters that can be changed. The amplitude, mean, and variance of Gaussian noise can be changed. In this study, only the mean value and the variant were emphasized. Gaussian noise with 0.2 variance and 0 mean was added to the sine wave. The result in the time domain is shown in Fig. 5, also the one in frequency domain is in Fig. 6.

The variance of Gaussian noise was fixed at 0.04, and at the same time, the mean value is increased to 2 to examine the DC component effect. Obtained results for the time domain and frequency domain are shown, respectively, in Fig. 7 and Fig. 8.

To have a deep understanding of the effect of the DC and the variance, as the DC was fixed at 2 and the variance is scaled up to 100.

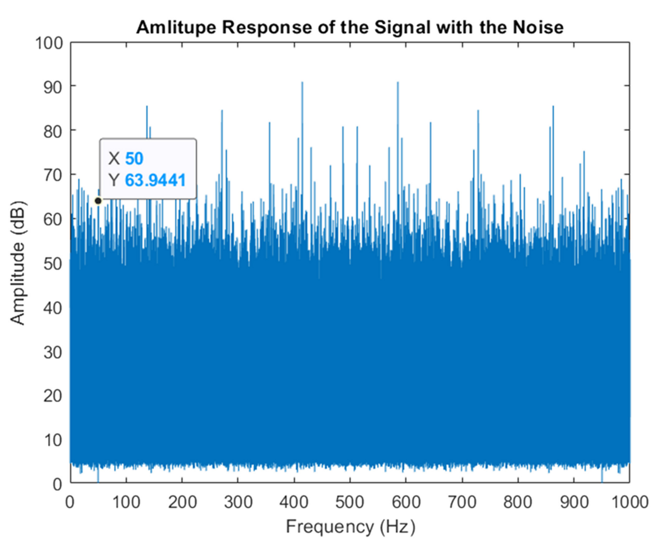

Fig. 6. 50 Hz $I_{pp} = 2$ A Sine wave with $\sigma = 0.2$, $\mu = 0$ Gaussian noise in frequency domain.

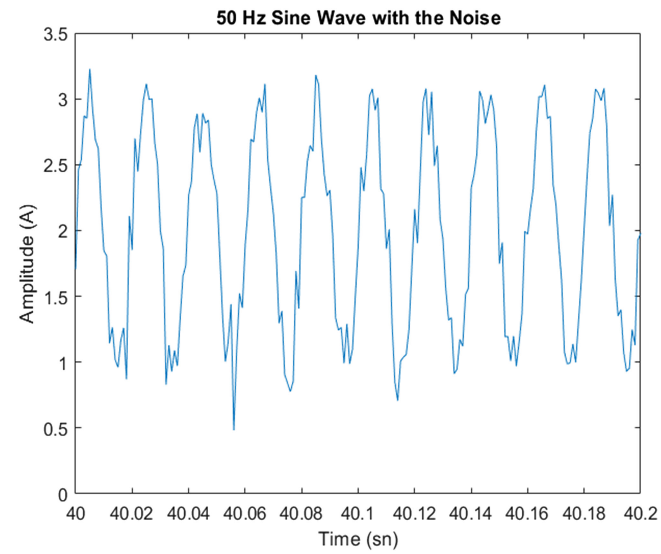

Fig. 7. 50 Hz $I_{pp} = 2$ A Sine wave with $\sigma = 0.2$, $\mu = 2$ Gaussian noise in time domain.

The time domain and the frequency domain are shown in Fig. 9 and Fig. 10, respectively.

Comparing the last eight figures (from Fig. 3 to Fig. 10):

1. Increasing the mean value of the Gaussian noise caused an increase in the DC level. The DC value is 13 dB bigger than the original value (Fig. 8).
2. As that is known from probability, the Gaussian distribution is called the normal distribution. This is because the model affects the entire frequency domain equally. As explained, when the variance of the noise is increased, the generated noise is found to be approximately equal in the whole frequency domain.

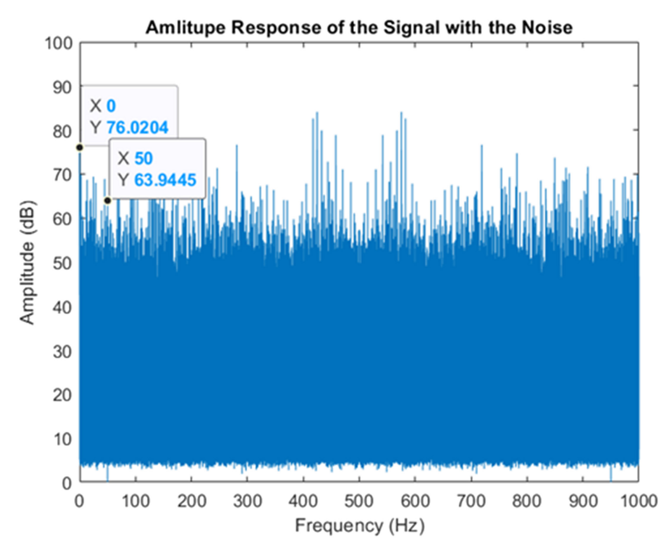

Fig. 8. 50 Hz $I_{pp} = 2$ A Sine wave with $\sigma = 0.2$, $\mu = 2$ Gaussian noise in frequency domain.

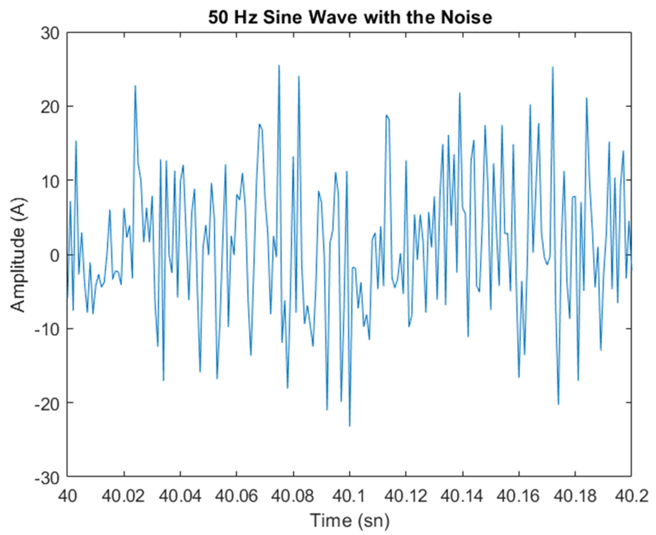

Fig. 9. 50 Hz $I_{PP} = 2$ A Sine wave with $\sigma = 10$, $\mu = 2$ Gaussian noise in time domain.

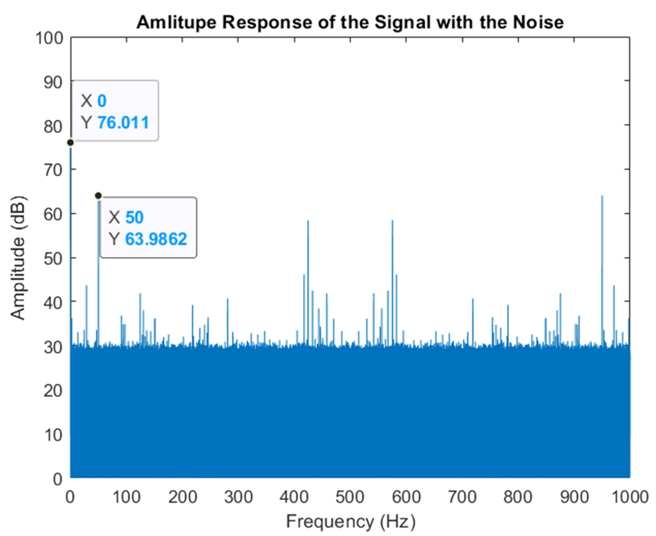

Fig. 10. 50 Hz $I_{PP} = 2$ A Sine wave with $\sigma = 10$, $\mu = 2$ Gaussian noise in frequency domain.

3. As a result of the increase in the variance, the noise resembled the normal distribution (Fig. 10).

B. Designing the Filters

The DC component was assumed to be isolated when designing the filter. Thus, an undesirable important value was removed from the spectrum. As it is known, the Fourier transform of a sine signal is a unit impulse. The unit impulse is a theoretical signal whose amplitude is infinity and width 0. Since the model we designed is a sinusoidal filter, the filter must be a band-pass filter with a bandwidth close to 0 Hz and at least 50 dB attenuation to eliminate the noise (Fig. 8). The filter to be designed is intended to run on a low-power microcontroller. Therefore, it was predicted that it would be more appropriate to design an IIR filter because IIR filters have a feedback mechanism [in (1)]. Unlike other filter types such as Elliptic, Chebyshev type I, Chebyshev type II, and the Butterworth filter will be an appropriate choice because the Butterworth filter does not generate oscillation in the pass band or in the stop band.

As a result of these filter requirements, the filters to be designed were selected Butterworth IIR band-pass filter. For this reason, four band-pass filters were designed in different orders and bandwidths. We used Bilinear Transform Method [7] to determine of the filters' orders and cutoff frequencies. Although the Bilinear Transform Method is briefly mentioned, further information is not provided here to avoid unnecessary lengthening of the paper. Let f_{C1} be lower cut-off frequency and f_{C2} be the higher cut-off frequency. Thus, the specifications of the designed filter are as follows:

1. $f_{C1} = 49.5$ Hz, $f_{C2} = 50.5$ Hz Butterworth IIR sixth-order band-pass filter,
2. $f_{C1} = 49.8$ Hz, $f_{C2} = 50.2$ Hz Butterworth IIR eighth-order band-pass filter,
3. $f_{C1} = 49.9$ Hz, $f_{C2} = 50.1$ Hz Butterworth IIR eighth-order band-pass filter,
4. $f_{C1} = 49.85$ Hz, $f_{C2} = 50.25$ Hz Butterworth IIR eighth-order band-pass filter.

The magnitude responses of these filters are shown in, respectively, Fig. 11.

C. Simulation Results

In the simulation, we assumed that isolation was provided between the measurement circuit and the microcontroller. So, the DC component was subtracted from the measurement results. For this reason, the Gaussian noise that was used as the model was set to a mean value of 0 and a variance value of 0.04. The noise model generated was included in $I_{PP} = 2$ A 50 Hz sine wave signal and the result is as obtained in Fig. 5.

The obtained result using the designed first filter is shown in Fig. 12.

The first filter can be considered to work as desired according to Fig. 12. There is just about 0.8% error, which is negligible. However, it was observed that there were oscillations in long-term output. The related figure is shown in Fig. 13.

If there is an oscillation in a system output, it may cause an unstable condition. In order to prevent the system from unstable conditions, we cannot use this filter. The output of the designed second filter is shown in Fig. 14(a). It is seen that the second filter partially provides the specified criteria.

It was used "partially" definition because there is about 1.7% error. This error is higher than the criteria. The output of the designed third filter is shown in Fig. 14(b).

According to Fig. 14(b), there is almost 41% fault. Therefore, it cannot be used. When the ten-period output of the designed fourth filter in Fig. 14(c) is examined, it is observed that it generates results with an error rate of 0.3%. The results meet the error rate criterion of 1%.

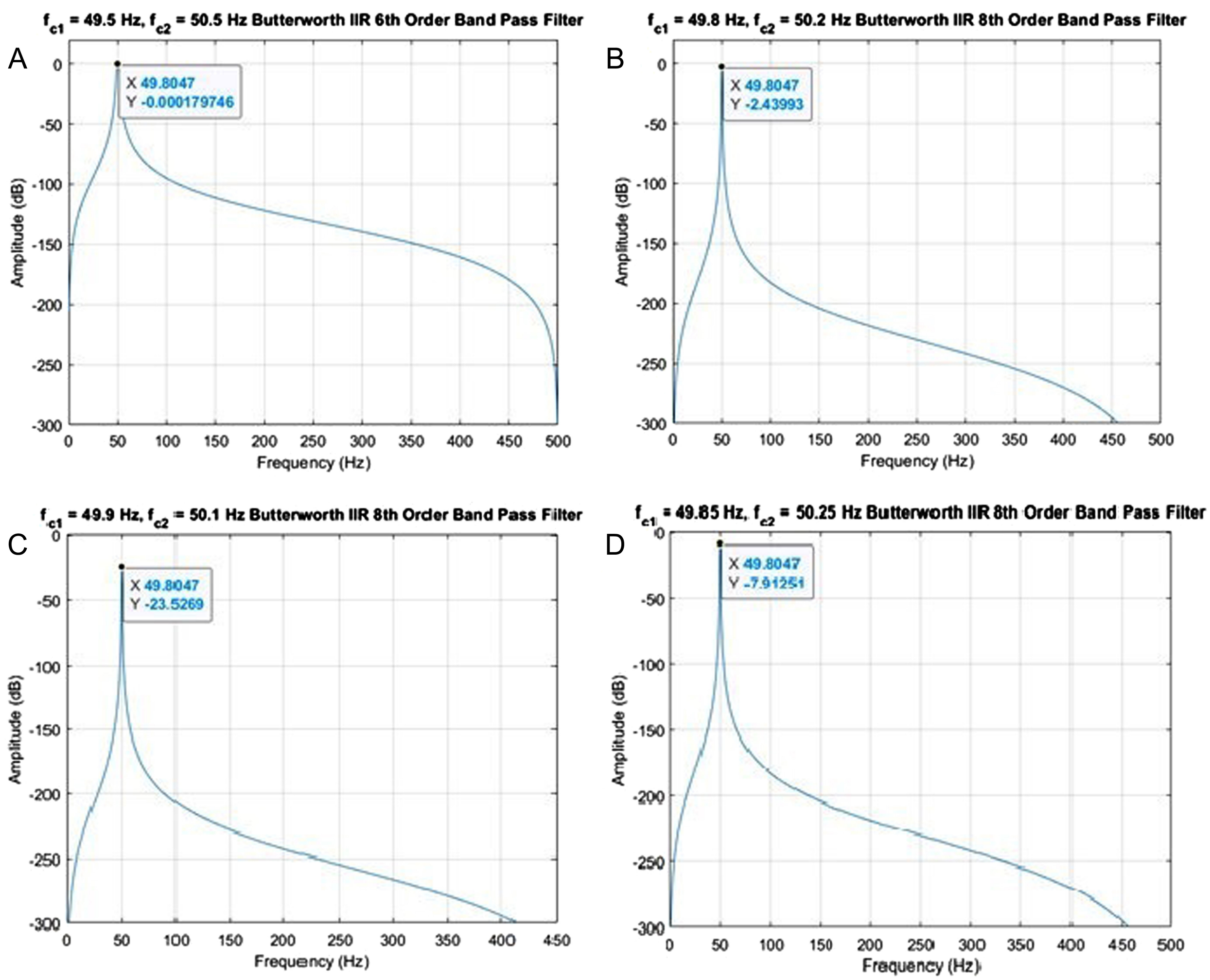

Fig. 11. (a) Designated first filter. (b) Designated second filter. (c) Designated third filter. (d) Designated fourth filter.

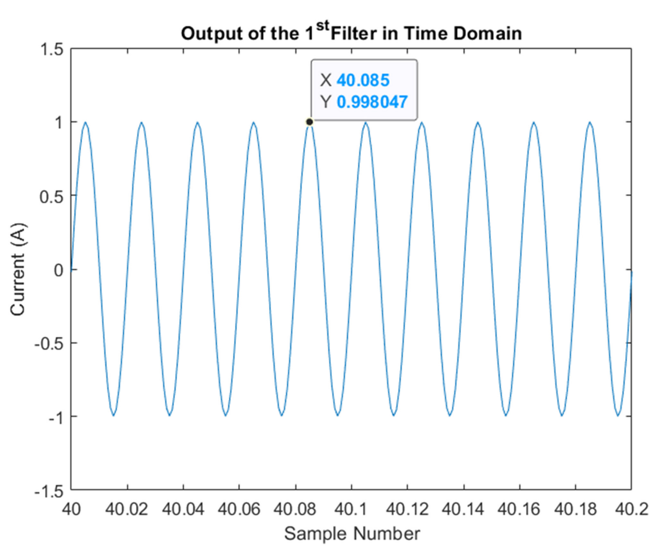

Fig. 12. f_{C1} = 49.5 Hz, f_{C2} = 50.5 Hz 6th Order IIR Butterworth band-pass filter output. IIR, infinite impulse response.

To make sure that the fourth filter is enough for this study, it plotted the long-term output and the results are obtained as in Fig. 15.

The results show that the fourth filter met the error rate criterion of 1%. According to the long-term filtering output in Fig. 15, a 2% overshoot and 3 seconds settling time are observed. Therefore, the designed fourth filter is preferred.

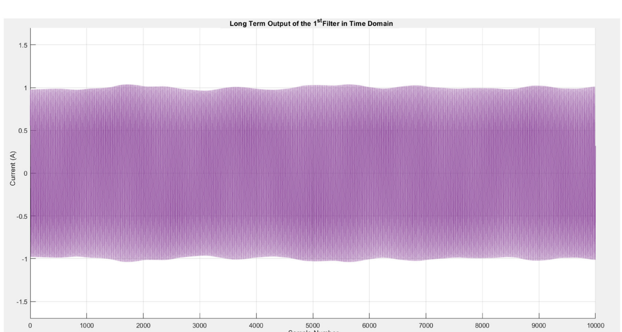

Fig. 13. Long-term output of the first filter.

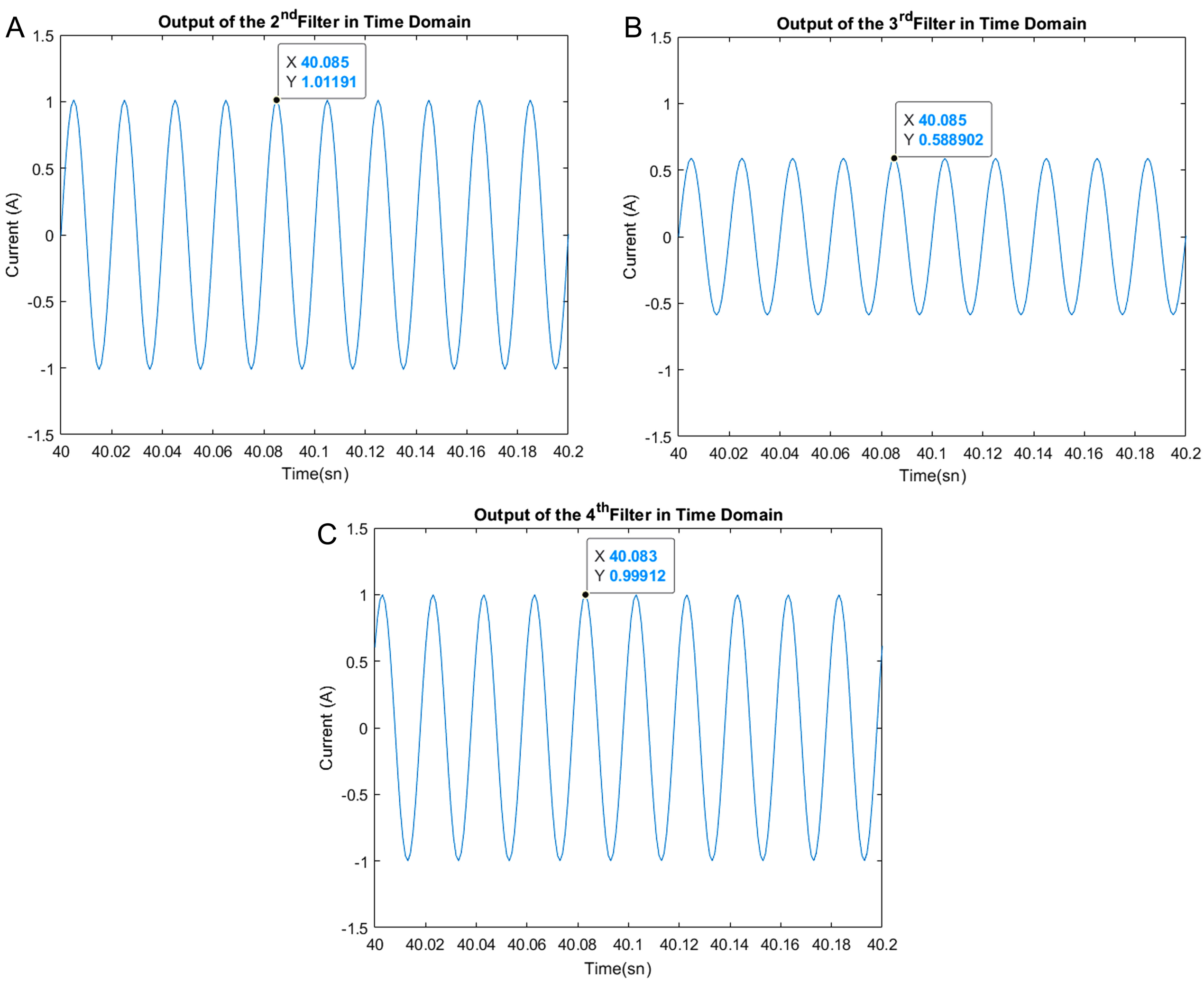

Fig. 14. (a) f_{C1} = 49.8 Hz, f_{C2} = 50.2 Hz 8th Order IIR Butterworth band-pass filter output. IIR, infinite impulse response. (b) f_{C1} = 49.87 Hz, f_{C2}=50.17 Hz 8th Order IIR Butterworth bandpass filter output. IIR, infinite impulse response. (c) f_{C1} = 49.85 Hz, f_{C2}=50.25 Hz 8th Order IIR Butterworth bandpass filter output. IIR, infinite impulse response.

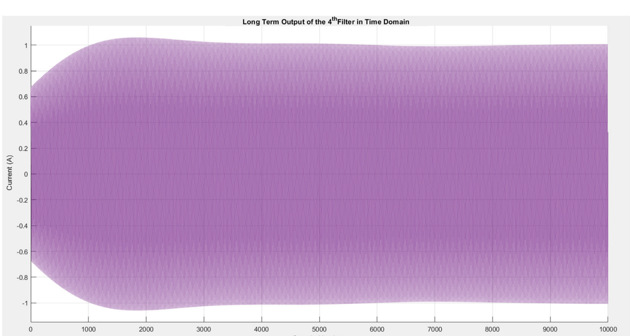

Fig. 15. Long-term output of the last filter.

III. CONCLUSION

Considering the entire filters designed, it is seen that the first filter met the criteria, but it worked in an unstable state, the second filter designed had about 10% overshoot and it had a very long settling time, and the third filter had an error rate of 49%. According to the criteria determined for our system, the most suitable filter is the fourth filter because the error rate was very low and the settling time was very short. On the other hand, it was understood that the DC component should be removed from the filtering process. In Fig. 8, the power of the DC component is much more than the power of the signal. This will complicate the filtering process.

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REFERENCES

1. A. Eski, O. Zavalani, and D. Komici, “The implementation of a digital filter-based algorithm for active power measurement using an ARM microcontroller,” *EJECE*, vol. 1, no. 3, 2017.
2. H. Zhenhua, C. Jianye, S. Ling, W. Zanji, and W. Yi, “A digital controller of hybrid filter based on pseudo-inverse filtering technique,” 2005 IEEE/PES Transmission & Distribution Conference & Exposition: Asia and Pacific, Dalian, 2005, pp. 1–6. [CrossRef]
3. P. Hao, W. Zanji, and C. Jianye, “A measuring method of the single-phase AC frequency, phase, and reactive power based on the Hilbert filtering,” *IEEE Trans. Instrum. Meas.*, vol. 56, no. 3, pp. 918–923, 2007. [CrossRef]
4. R. R. Pereira, C. H. da Silva, L. E. B. da Silva, and G. Lambert-Torres, “Application of adaptive filters in active power filters,” 2009 Brazilian Power Electronics Conference, Bonito-Mato Grosso do Sul, Brazil, pp. 770–774. [CrossRef]
5. A. N. Serov, N. A. Serov, and A. A. Lupachev, “The application of digital filtration for the active power measurement,” 2018 IEEE International Conference on Electrical Engineering and Photonics (EExPolytech), St. Petersburg, Russia, 2018, pp. 88–92. [CrossRef]
6. P. Duraiswamy, and S. K. Selvaraja, “Phase-orthogonal FIR filter based reactive power measurement for power meters,” *Analog Integr. Circuits Signal Process.*, vol. 108, no. 2, pp. 317–322, 2021. [CrossRef]
7. L. Tan, and J. Yiang, *Digital Signal Processing: Fundamentals and Applications*, 2nd ed. Amsterdam: Elsevier Press, 2014, pp. 301–391.