Adaptive Filtering for Noise Reduction in the Input Circuits of Protection Relay Systems

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***Abstract*—This paper describes an adaptive filtering to reduce noise in the analog input circuits of microprocessor-based protection relay (digital relay) systems. Considering that digital relays tend to have multiple functions, it is important to raise signal to noise ratios (SNR) in analog input circuits to a level that will satisfy the requirements of these functions. We have realized a noise reduction method employing an adaptive filter which can be generally applied; not only within digital relays but also for IEDs in substations and independent of protection schemes. This paper presents the concept of this method and illustrates this effectiveness using a number of examples and experimental results.**

***Index Terms*—Protection Relay, Adaptive Filter, IED, Noise Reduction, MATLAB/Simulink, LMS algorithm**

I. INTRODUCTION

The vital performance indices of a digital relay, such as accuracy and response, mainly depend upon analog input processing. Recently, in view of cost-efficiency, some digital relays have not only protection functions but also a wide area monitoring function employing synchrophasor techniques, a fault locating function and a disturbance recording function. It means that these multiple functions share the same analog input circuit. These functions have such differing requirements in terms of the dynamic range of the analog input that the accuracy of measurement functions such as fault location may be compromised when the requirement for protection is given priority.

On the other hand, for many years, the accuracy of analog input processing in a digital relay has mainly depended upon the resolution of the analog to digital converter (ADC). In recent years, the 16-bit successive approximation ADC has commonly been used. Moreover, the delta-sigma ADC has become the ADC of choice because of the higher resolution and simple configuration. However, even if the specific resolution of analog to digital conversion becomes higher, it may still not be possible to achieve the resolution required due to the presence of white and colored noises generated in an analog input circuit and noises induced by power supply lines, etc.

The purpose of this study is to reduce the noise in the analog input circuits of digital relays. The noise reduction method proposed allows higher accuracy analog input processing and enables higher performance protection, control and measurement IEDs.

Many studies on analog and digital filters for digital relays have been conducted [1]. However, most of these studies have focused on the optimization of filter designs and algorithms for a specific protection application for the purpose of improvement in response time and removal of harmonics generated in power systems. Some studies have focused on analog signal processing. For example, one study analyzed the relationship between ADC resolution, sampling rate and the accuracy of measurement for power quality [2]. To the best of our knowledge, however, there has been little study on noise reduction in the analog input circuits of digital relays.

For the purpose of the evolution of analog signal processing in digital relays, we have proposed an adaptive filter algorithm that will reduce the noise found in analog input circuits. Today, practical applications of adaptive filters can be found in widespread use in the fields of communication and measurement, however, for the field of protection and control, there is but only the proposal for an improvement in the accuracy of measurement of frequency [3]. Although the Kalman filter is similar to an adaptive filter, the solutions proposed also depend on each protection application and seem to be too complex for practical use [4].

We have found that an adaptive filter employing a sinusoidal signal as a desired signal is particularly useful for reducing noise within the analog input circuits of digital relays. Moreover, we have explored their actual implementation in practical protection relays and an application for self checking of analog input circuits. For these discussions and verification purposes we have constructed an accuarte simulation model of the analog input process using MATLAB/Simulink.

In Section II, we present the noise model of an analog input circuit, and the configuration of our proposed adaptive filter. In Section III, we deal with the simulation model using MATLAB/Simulink. Section IV shows the experimental results using the data obtained in a substation. Finally, Section V summarizes our conclusion.

II. NOISE REDUCTION IN THE INPUT CIRCUITS OF PROTECTION RELAY SYSTEMS

Fig. 1 shows a noise model for an analog input circuit, which has white noise and colored noise such as the nonlinear error of input transformers, thermal noise from the analog filters, multiplexer switching noise, ADC quantization error and noise induced by power supply lines. In this study, the output of the ADC is fed to an adaptive filter.

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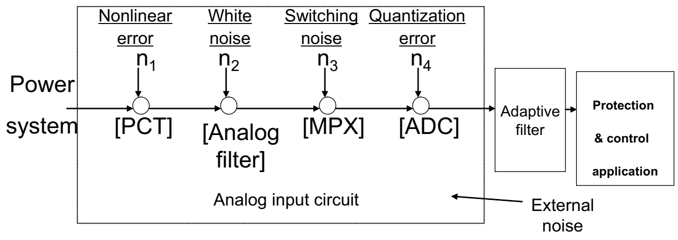

Fig. 1. Noise model of an analog input circuit.

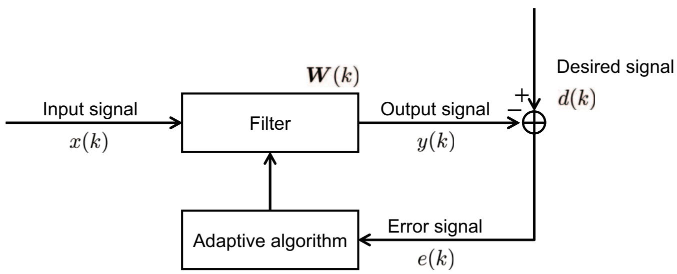

Fig. 2. Block diagram of an adaptive digital filter.

In this paper, we have proposed adaptive filtering for noise reduction in the input circuits of protection relay systems. The adaptive digital filters can converge to the optimum Wiener solution in a stationary environment and can track time variations in the statistics of the input data in a non-stationary environment.

Fig. 2 shows the block diagram of an adaptive digital filter. The adaptive digital filter consists of two basic blocks: a filtering block which has a variable coefficient filter and an adaptive processing block which has the automatic adjustment of the coefficients of the filter in accordance with the estimation error.

In the filtering block, $x(k)$ is the input signal to the adaptive digital filter (the output of the ADC) and $y(k)$ is the output signal of the adaptive digital filter. In this paper, the desired signal $d(k)$ for the adaptive digital filter is defined as follows:

$$d(k) = A_0 \sin\left(2\pi \frac{f_0}{f_\mathrm{s}} k + \theta_0\right), \tag{1}$$

where A_0 is the amplitude of the desired signal and can be found in the protection relay systems separately. The sampling frequency f_s is 4800 Hz. The nominal frequency f_0 is 50 Hz. The phase difference θ_0 between the input signal and the desired signal is 0 rad. The output signal $y(k)$ is given by

$$y(k) \quad = \quad \boldsymbol{W}^T(k)\boldsymbol{X}(k) = \sum_{i=0}^{L} w_i(k)x(k-i), \tag{2}$$

where the column vector $\boldsymbol{X}(k)$ denotes the input signal vector and the column vector $\boldsymbol{W}(k)$ denotes the filter coefficient of the adaptive digital filter at time k, i. e.

$$\boldsymbol{X}(k) \quad = \quad [x(k), x(k-1), ..., x(k-L)]^T, \tag{3}$$
$$\boldsymbol{W}(k) \quad = \quad [w_0(k), w_1(k), ..., w_L(k)]^T. \tag{4}$$

The order L of the adaptive digital filter is 95 for a window which covers one period of the input signal. The estimation

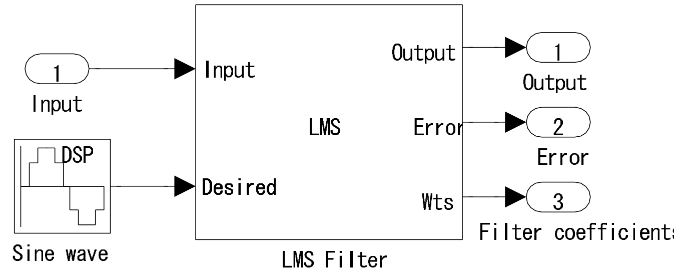

Fig. 3. Simulation model of the noise reduction system using an adaptive filter.

error at time k is given by

$$e(k) \quad = \quad d(k) - y(k). \tag{5}$$

In the adaptive processing block, the requirement is to make the estimation error $e(k)$ "as small as possible" in some statistical sense. To satisfy the requirement, in this paper, the coefficient vector $\boldsymbol{W}(k)$ is updated using the LMS algorithm [5]

$$\boldsymbol{W}(k+1) = \boldsymbol{W}(k) + 2\mu e(k)\boldsymbol{X}(k), \tag{6}$$

where μ is the step size parameter.

III. Simulation Model

In this section we have introduced a simulation model for a protection relay system for the purpose of analyzing the behavior of our proposed system. We have constructed a simulation model using MATLAB/Simulink [6]. As stated in the previous section, the adaptive digital filter plays the most important role in realization of our proposed protection relay system. Therefore, construction of a simulation model of the adaptive filter is the most important task. However, in order to analyze the behavior of the entire system shown in Fig. 1, it is necessary to construct simulation models of not only the adaptive filter, but also other systems such as the analog input circuit and the protection application.

Taking this into account, this section initially presents a simulation model of our proposed noise reduction system using the adaptive filter, and then provides some of the other necessary simulation models.

A. Simulation Model of the Noise Reduction System

Figure 3 shows our noise reduction system based on the adaptive filter proposed in the previous section. In order to realize the adaptive filter on Simulink, we have used the "LMS Filter" block that is available in the Signal Processing Blockset. In this model, the input signal of the adaptive filter is the output of the ADC and this signal is contaminated with various kinds of noise, as discussed in the previous section. The desired signal, which is indicated as the other input signal to the LMS Filter block, is the discrete-time sinusoidal signal that is defined by (1) and generated by the "Sine Wave" block. With this specification the simulation model yields an output signal with reduced noise. In addition, the LMS Filter block provides not only the output signal but also information on

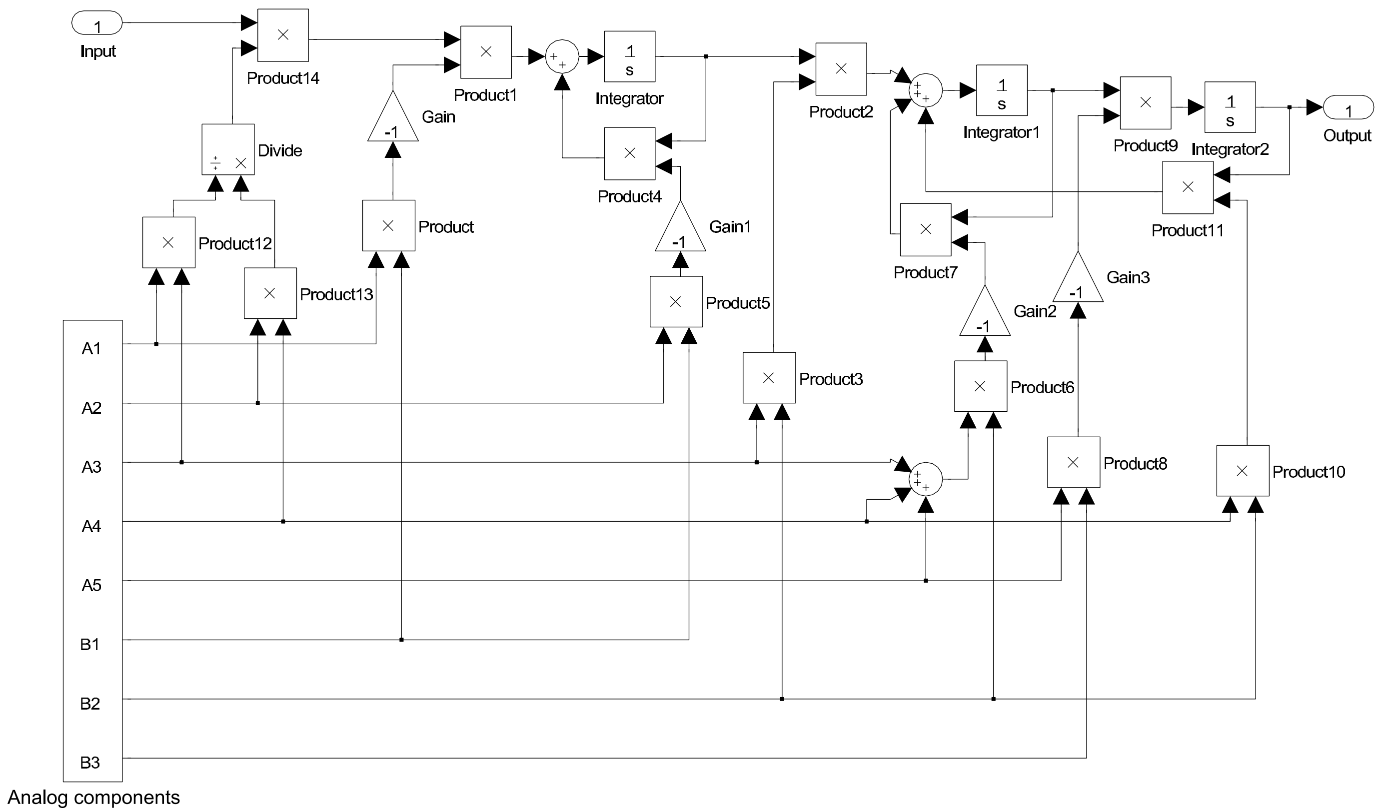

Fig. 4. Simulation model of the analog filter.

the error signal and the coefficients of the adaptive filter, which enables us to analyze the convergence behavior of our proposed noise reduction system in more detail.

B. Simulation Model for the Analog Input Circuit

This subsection overviews a part of our simulation model for the analog input circuit, where we introduce simulation models for the analog filter and the ADC.

The simulation model for the analog filter is shown in Fig. 4. This filter is a third-order low-pass filter. In the model shown in Fig. 4, the values of the filter parameters can be arbitrarily chosen by the user. Another feature of this simulation model is that the state-space representation is used to construct this model because the state-space representation is known to be very effective for analysis of the performance of analog filters [7], [8].

Figure 5 shows the simulation model for the ADC. In this model, the "Zero-order hold" block is used to convert the analog input signal into the discrete-time signal. This discrete-time signal is then quantized into a digital signal of finite "wordlength" by the "ADC" block. This ADC block makes use of a successive approximation technique for quantization of signals, and we have realized this block by using the Embedded MATLAB Function. The other parameters denoted by "Vscale" and "Wordlength" correspond to the full-scale amplitude range and the "wordlength" number of the ADC respectively. These parameters can be arbitrarily specified by the user. In the model shown in Fig. 5, they are set to 1 V and 16 bits respectively.

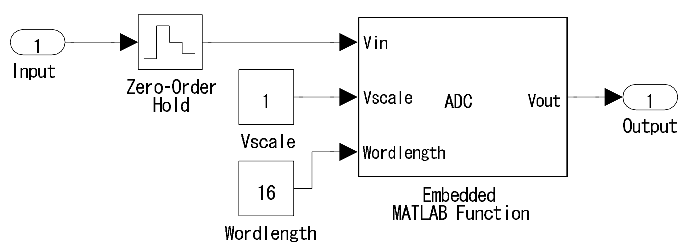

Fig. 5. Simulation model of the A/D converter.

In this paper, the simulation models introduced for the analog input circuit have been restricted to the analog filter and the ADC. Realization of the other models that are required to constitute the entire system shown in Fig. 1 is one of our future tasks.

IV. Experimental Results

To demonstrate the effectiveness of the adaptive digital filter proposed in Section II, we added the new noise reduction filter to the input circuit of a protection relay system.

Figures 6 and 7 show the input signal $x(k)$ and the desired signal $d(k)$ from the adaptive digital filter. In this section, the input signal $x(k)$, which is the output signal of the ADC, is the data obtained from within a substation. In this paper, we use $A_0 = 63.18$, which is the constant value derived from the maximum value of the input sequence in Fig. 6. Figures 8 and 9 show the signals which expand the beginning of the signals

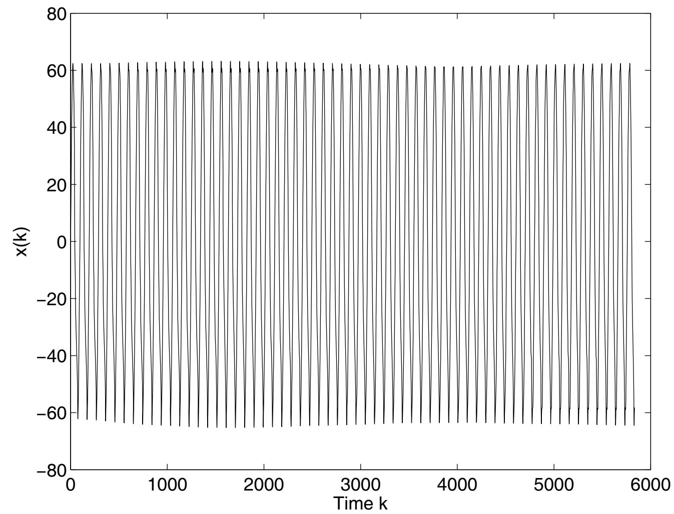

Fig. 6. Input signal $x(k)$.

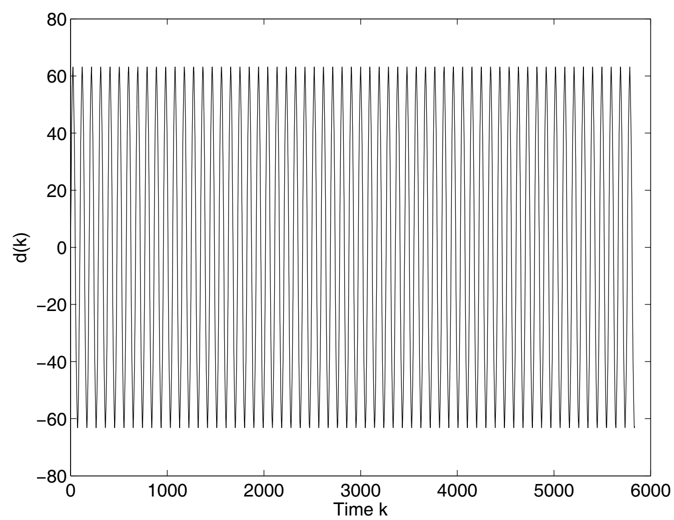

Fig. 7. Desired signal $d(k)$.

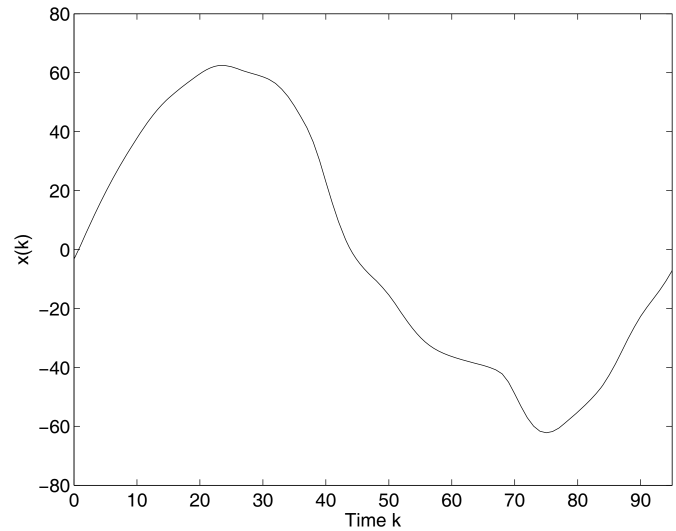

Fig. 8. Input signal $x(k)$ (detailed).

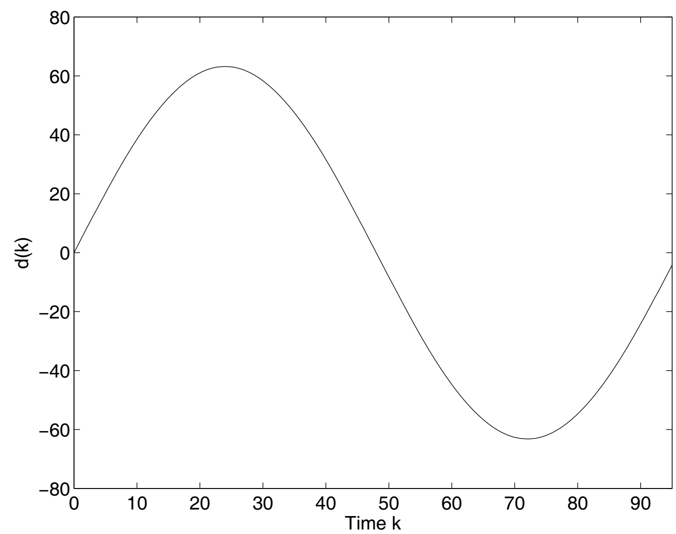

Fig. 9. Desired signal $d(k)$ (detailed).

in Figs. 6 and 7. From Fig. 8, we can see that the input signal $x(k)$ includes the noise found in the analog input circuit.

Figure 10 shows the learning curve (square error $e^2(k)$) of the proposed adaptive digital filter. Figure 10 shows that the square error of the proposed adaptive digital filter converges at the small adaptation noise. Figure 11 shows the coefficient $w_i(k)$ of the adaptive digital filter. Figure 11 shows that the coefficients converge at the values required to reduce the noise. Figures 12, 13 and 14 show the input, desired and output signals when the coefficients converge. Figures 12, 13 and 14 show that the noise of the output signal is reduced. Figures 15 and 16 show the frequency amplitude response of the input signal and the output signal. We can observe from Figs. 15 and 16 that the signal of the nominal frequency $f_0 = 50$ is maintained and the others are reduced.

To demonstrate the effectiveness of the proposed adaptive digital filter, we define the normalized noise level as follows $|X_{\mathrm{noise}}|$, i. e.

$$|X_{\mathrm{noise}}| = \sum_{f=0}^{f_0 - f_0/2} |X(f)| + \sum_{f=f_0+f_0/2}^{f_s/2} |X(f)|. \tag{7}$$

Using the proposed method, the normalized noise level of the input signal is $|X_{\mathrm{noise}}| = 0.4308$ and that of the output signal is $|Y_{\mathrm{noise}}| = 0.0056$. These results show that the proposed method can adequately reduce the noise, that is, the normalized noise level of the output signal is $0.0056/0.4308 = 1/78.21$ times that of the input signal. On the other hand, in a conventional filter with fixed coefficients, the normalized noise level of the input signal is $|X_{\mathrm{noise}}| = 0.3859$ and that of the output signal is $|Y_{\mathrm{noise}}| = 0.0546$. These results show that with the conventional method the normalized noise level of the output signal is $0.0546/0.3859 = 1/7.07$ times that of the input signal.

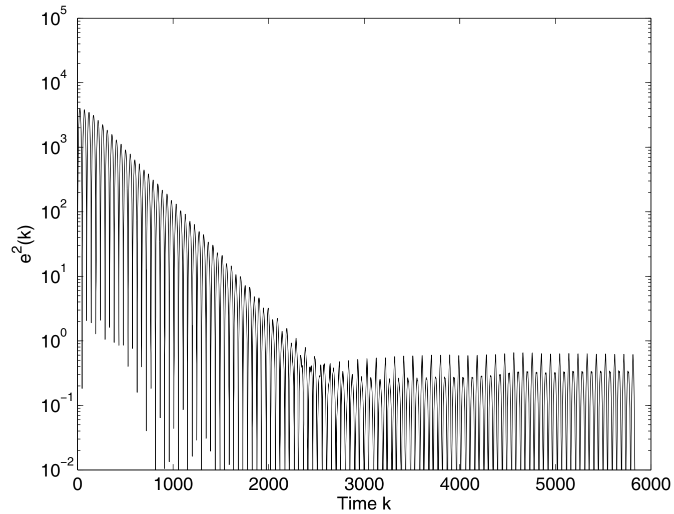

Fig. 10. Learning curve of the proposed adaptive digital filter.

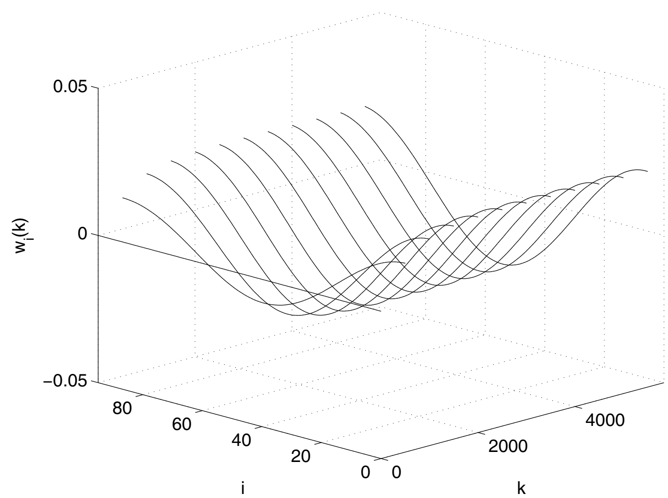

Fig. 11. Coefficient $w_i(k)$ of the adaptive digital filter.

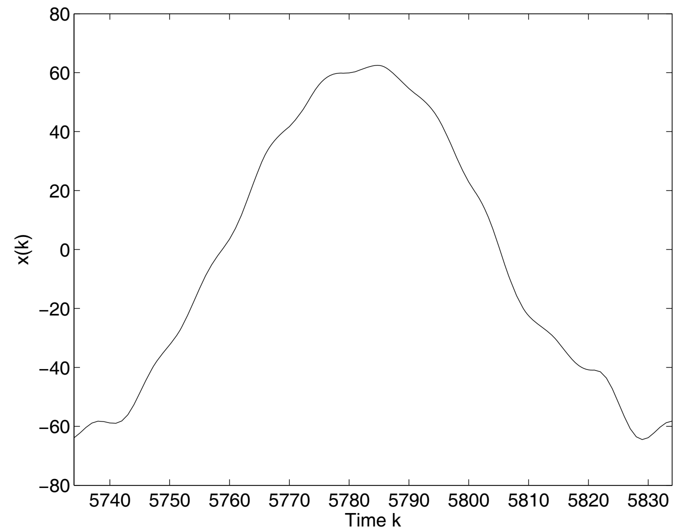

Fig. 12. Input signal $x(k)$.

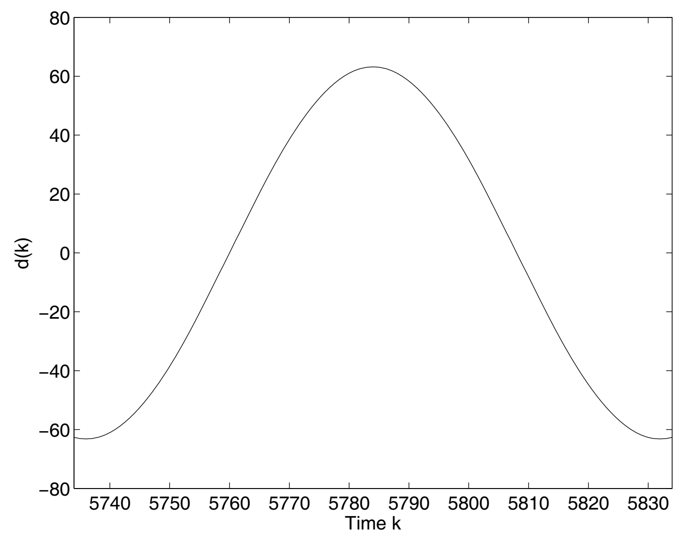

Fig. 13. Desired signal $d(k)$.

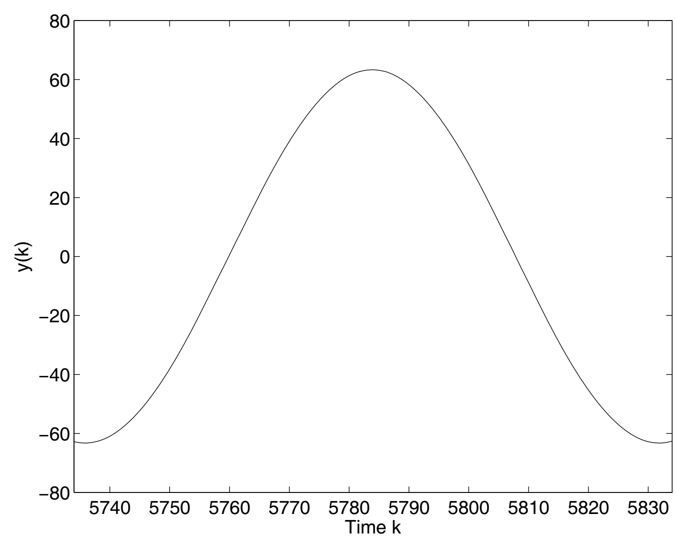

Fig. 14. Output signal $y(k)$.

V. CONCLUSION

Adaptive filtering has shown to be very useful when applied to reduce the noise in the analog input circuits of digital relays; our study provided us with an extensive improvement in signal to noise ratios (SNR). The results of this study suggest that higher accuracy measurement functions can be realized and the accuracy for fault location can be higher using our proposed algorithm rather than when using conventional methods. The new method can commonly be adopted within digital relays because it is independent of the protection application. Moreover, we introduced an accurate simulation model to explore the analog input process. This simulator will enhance current and future studies in the evolution of analog signal processing. Digital signal processing for analog signals of protection relay systems is a key technology within digital relaying because the next generation protection and control systems, including IEC61850-9-2 (process bus) will need a much wider dynamic range. The development of self-diagnosis techniques for analog input circuits using variations

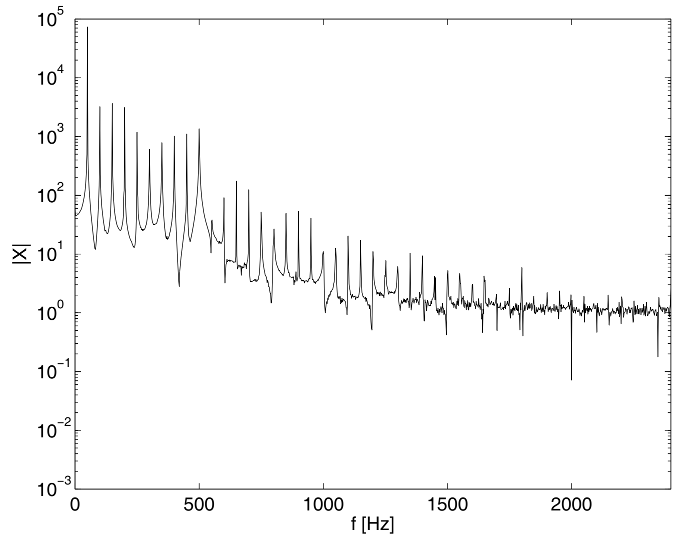

Fig. 15. Frequency amplitude response of the input signal.

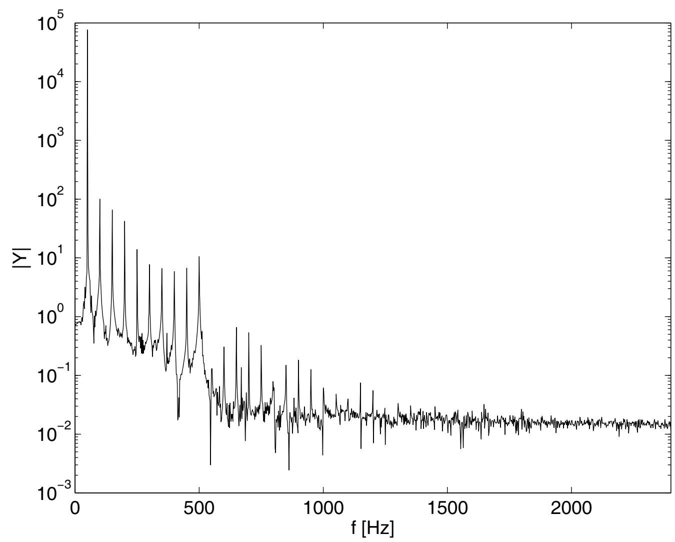

Fig. 16. Frequency amplitude response of the output signal.

in the coefficient of the adaptive filter and the implementation method necessary for high speed execution of the algorithm require further study.

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