# ON-LINE ADAPTIVE LINE FREQUENCY NOISE CANCELLATION FROM A NUCLEAR POWER MEASURING CHANNEL

by

### Javed QADIR, Masood IQBAL\*, and Hameed QAISER

Nuclear Engineering Division, Pakistan Institute of Nuclear Science and Technology (PINSTECH), Nilore, Islamabad, Pakistan

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On-line software for adaptively canceling 50 Hz line frequency noise has been designed and tested at Pakistan Research Reactor 1. Line frequency noise causes much problem in weak signals acquisition. Sometimes this noise is so dominant that original signal is totally corrupted. Although notch filter can be used for eliminating this noise, but if signal of interest is in close vicinity of 50 Hz, then original signal is also attenuated and hence overall performance is degraded. Adaptive noise removal is a technique which could be employed for removing line frequency without degrading the desired signal. In this paper line frequency noise has been eliminated on-line from a nuclear power measuring channel. The adaptive LMS algorithm has been used to cancel 50 Hz noise. The algorithm has been implemented in labVIEW with NI 6024 data acquisition card. The quality of the acquired signal has been improved much as can be seen in experimental results.

Key words: adaptive cancellation, line frequency noise, nuclear power measuring channel, PARR-1

#### INTRODUCTION

Adaptive techniques have been widely used in variety of applications in last couple of decades. The main reason for this wide application is the achievement of much improved results as compared to other techniques. In adaptive technique the gain or parameter coefficients are adapted dynamically according to the time dependent variations of a plant. In adaptive control, the controller gains are adjusted as the plant mass, like airplane mass due to fuel consumption or any other parameter is changed.

Similarly in adaptive filters, the filter coefficients are adjusted according to signal variations. This adaptive behavior is the key to improved results as the parameter variation is automatically catered for. An adaptive filter is a computational device that iteratively models the relationship of two signals: the input signal and the desired signal as shown in fig. 1. The filter input is x(n) at time n. An adaptive algorithm controls the coefficients of the linear filter, and the coefficients can change over time. y(n) is the output of the adaptive filter and d(n) is the desired signal. The difference between d(n) and y(n) is called error signal e(n). The adaptive algorithm adjusts the filter coefficients iteratively to minimize the error signal e(n). As shown in fig. 1, an adap-

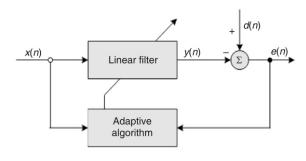


Figure 1. Adaptive filter structure

tive filter consists of two parts: a linear filter and an adaptive algorithm. Based on different structures, linear adaptive filters have two groups: finite impulse response (FIR) adaptive filters and infinite impulse response (IIR) adaptive filters. FIR filters are most widely used because of their simplicity and stability. IIR filters with smaller filter orders can achieve nearly the same performance as FIR filters with much higher filter orders. The main drawback of IIR filters is the possible instability. An adaptive algorithm aims at adjusting the parameters of an adaptive filter to minimize the error signal e(n). Adaptive algorithms can be classified in the following two categories: statistical optimization-based algorithms, such as the least mean squares (LMS) algorithm and its derivatives and deterministic optimiza-

 $<sup>\</sup>hbox{$*$ Corresponding author; e-mail: masiqbal@hotmail.com}\\$ 

tion-based algorithms, such as the recursive least squares (RLS) algorithm and its derivatives.

In the past adaptive filtering technique has also been applied in related applications. Kumaravel and Nithiyanandam [1] have used genetic algorithm (GA) based method for removing sinusoidal power line noise in electrocardiograms. GA has been used for removing two types of noises, namely power line noise with frequency drift and interference with frequency drift and third harmonic distortion. The results were taken by generating simulated interference signal on electrocardiogram on actual noisy electrocardiogram records. Ramos et al. [2] have shown implementation of single noise suppressor on a field programmable gate array (FPGA). The adaptive suppressor automatically adapts its parameters to eliminate noise. Author used this technique for the elimination of power line noise. Thakor and Zhu [3] have proposed many adaptive filter structures for noise cancellation and arrhyth-

The filter structures are presented to eliminate different forms of noise, such as, baseline wander, 60 Hz power line interference, muscle noise, and motion artifact. Keenan and Grossman [4] have described an adaptive filter for separating low frequency sympathetic and high frequency parasympathetic components from an electrocardiogram (ECG) interval signal. This helps in attaining more accurate heart rate variability measurements. The results show improvement over standard heart rate variability spectral measurements. Saxena and Ganesan [5] implemented improved adaptive Wiener filter which provides signal to noise ratio (SNR) improvement from 2.5 to 4 dB. Wang and Trussela [6] presented constrained filter which greatly reduces computation without reducing its effectiveness. Zeng et al. [7] used least squares (RLS) based adaptive noise canceling approach to eliminate the maternal ECG and hence to extract the fetal electrocardiogram. Nimunkar and Tompkin [8] studied empirical mode decomposition (EMD) for filtering power line noise in electrocardiogram signals. They added a pseudo noise at a frequency higher than the highest frequency of the signal to filter out just the power line noise in the first intrinsic mode function (IMF). Ziarani [9] proposed non-linear adaptive electromagnetic interference (EMI) filter for elimination of power line interference on ECG signals. Gao and Jinfeng [10] proposed neural network based PSO algorithm (particle swarm optimization), which shows the better noise cancellation capability compared to the traditional adaptive noise canceller [10]. In this paper an LMS adaptive filter has been used to eliminate power line noise from a nuclear measuring channel.

#### PROBLEM FORMULATION

The main purpose of this paper is to address a system, developed for on-line elimination of line frequency noise. For conducting reactor experiments, a huge amount of data is acquired from reactors. If this data is corrupted or is not to the required standard, then results produced are also ambiguous. Thus for performing any calculation, the data should be as accurate as possible. As mentioned before, line frequency noise corrupts data to a great extent and if the frequency of interest is in close vicinity of power line frequency then accurate measurements becomes nearly impossible. Different methods and techniques have been employed in the past to eliminate power line noise, but adaptive technique outperforms these with fastest convergence rate. Adaptive filtering is an advanced method for eliminating any frequency or noise pattern. The main advantage of this technique is that original signal is recovered without any distortion and the signal can be extracted even if noise level is much higher than signal level. Due to these advantages, authors have used LMS, an adaptive filtering technique for removing power line noise from nuclear power measuring channel. The details of this technique and results are discussed in next sections.

#### ADAPTIVE FILTER STRUCTURE

#### Mathematical description of adaptive filter

The most commonly used structure for adaptive filters is the transversal structure shown in fig. 2. The following equation defines the filter coefficients vector w(n)

$$w(n) [w_0(n), w_1(n), \dots, w_{M-1}(n)]^T$$
 (1)

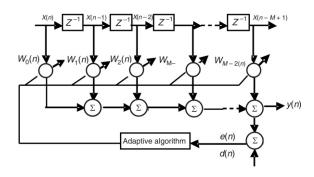


Figure 2. FIR adaptive filter

Because an adaptive filter is a self adjusting filter, the filter coefficient w(n) changes over time. The output is calculated by using following relation.

$$v(n) \quad x^{T}(n)w(n) \tag{2}$$

LMS algorithm is used widely for finding optimum value by descending on a performance surface [11]. This algorithm is important because of its simplicity and ease of computation, and also it does not require off-line gradient estimation or repetitions of data.

For adaptive linear combiner, we can write

$$\varepsilon(n) d(n) X_n^T W(n)$$
 (3)

where  $X_n$  is the vector of input samples.

Normally for developing an adaptive algorithm, the gradient of mean square error, denoted as  $\xi(n)$   $E[\varepsilon(n)^2]$  is estimated by taking differences between short-term averages of  $\varepsilon_k^2$ , but in LMS,  $\varepsilon(n)^2$  is used itself as an estimate of  $\xi(n)$ . Thus at each iteration, the gradient estimate of the following form is obtained

With this simple estimate of gradient, the weight update equation for LMS can be written as

$$W(n-1)$$
  $W(n)$   $\mu$   $(n)$   $W(n)$   $2\mu\varepsilon(n)X(n)$   $(5)$ 

Here  $\mu$  is the gain constant, which determines speed and stability of adaptation. From last equation it is clear that LMS algorithm can be implemented without squaring, averaging and differentiation and can be implemented easily with improved efficiency.

### LABORATORY BASED EXPERIMENTAL RESULTS

Laboratory set-up consists of a computer with NI 6024E data acquisition card, labVIEW 8.5 control software, noise generator, and a signal generator for generating 50 Hz signal. The noise generator signal was considered as a desired signal (FFT shown in fig. 3) and it was corrupted by adding 50 Hz signal, which

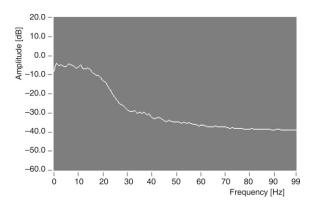


Figure 3. FFT of original signal for laboratory test

is considered as a noise signal. After mixing of 50 Hz in the signal, the FFT is shown in fig. 4. The adaptive LMS algorithm was adapted with 50 Hz signal by using analog channel 0 of data acquisition card. The corrupted signal was given to channel 1. The adaptive algorithm running in labVIEW removes the noise signal after the adaptation and that was achieved in few milliseconds. Laboratory tests were performed extensively to assure the fidelity of the system at different conditions and varying noise signals. FFT of input signal, signal + 50 Hz signal and filtered signals are shown in figs. 3-5, respectively. Tests were also performed by changing the amplitude of desired and training signals. After completely verifying laboratory results, the system was implemented on actual system.

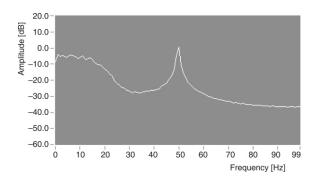


Figure 4. FFT of original + 50 Hz mixed signal for laboratory test

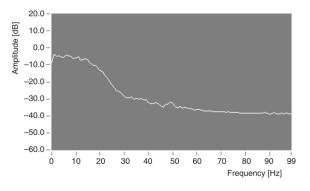


Figure 5. FFT of signal after adaptation for laboratory test

#### FIELD TEST RESULTS

After successful testing of laboratory based experimental set-up, same system was tested on actual reactor, Pakistan Research Reactor (PARR-1) which is a 10 MW pool type research reactor having a parallelepiped core comprising material testing reactor (MTR) type silicide LEU fuel [12]. The input signal was taken from PARR-1 linear nuclear channel, in which 50 Hz was also present. The data acquisition system at PARR-1 has same NI 6024 data acquisition card and labVIEW control software. The training sig-

nal for adapting 50 Hz noise signal was generated within the software by using built in signal generator while actual plant signal was given to the system as input. Here also the adaptive filter removed the 50 Hz within few milliseconds. The FFT of the input signal from linear channel is shown in fig. 6. After adaptation, the signal is depicted in fig. 7. As shown in fig. 7, the 50 Hz peak reduced substantially, which fulfills our requirements.

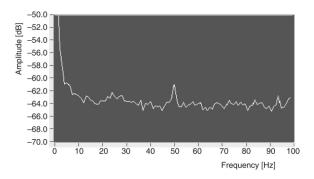


Figure 6. FFT of original signal from the linear channel of the reactor

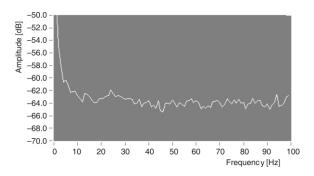


Figure 7. FFT of the signal after adaptation

#### **CONCLUSION**

An on-line system for adaptively canceling 50 Hz noise signal has been designed and tested at PARR-1.

The adaptive LMS algorithm has been implemented in labVIEW 8.5 with NI 6024 data acquisition card. The system successfully removed the power line noise after adaptation of 50 Hz training signal, generated within the software. Field test results show much improvement in the filtered signal quality, which helps in getting good quality results for experiments involving these signals.

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#### REFERENCES

- [1] Kumaravel, N., Nithiyanan, N., Genetic-Algorithm Cancellation of Sinusoidal Power Line Interference in Electrocardiograms, *Medical and Biological Engi*neering and Computing, 36 (1998), 2, pp. 191-196
- [2] Ramos, R., et al., Applications by Means of FPGA of an Adaptive Canceller of 50 Hz Interference in Electrocardiography, Proceedings, 18<sup>th</sup> IEEE Instrumentation and Measurement Technology Conference, 2001, 1, pp. 32-37
- [3] Thakor, N. V., Zhu, Y. S., Application of Adaptive Filtering to ECG Analysis: Noise Cancellation and Arrhythmia Detection, *IEEE Transactions on Biomedi*cal Engineering, 38 (1991), 8, pp. 785-794
- [4] Keenan, D. B., Grossman, P., Adaptive Filtering of Heart Rate Signals for an Improved Measure of Cardiac Autonomic Control, *International Journal of Signal Processing*, 2 (2006), 1, pp. 52-58
- [5] Saxena, G., Ganesan, S., Das, M., Real Time Implementation of Adaptive Noise Cancellation, IEEE International Conference on Electro/Information Technology, Ames, Ia., USA, 2008
- [6] Wang, J., Trussela, H. J., Adaptive Harmonic Noise Cancellation with an Application to Distribution Power Line Communications, IEEE Transaction on Communications, 36 (1988), pp. 875-884
- [7] Zeng, Y., Liu, S., Zhang, J., Extraction of Fetal ECG Signal via Adaptive Noise Cancellation Approach, The 2<sup>nd</sup> International Conference on Bioinformatics and Biomedical Engineering, ICBBE 2008, Shanghai, China, 2008, pp. 2270-2273
- [8] Nimunkar, A. J., Tompkins W. J., EMD-Based 60-Hz Noise Filtering of the ECG, *Proceedings*, 29<sup>th</sup> Annual International Conference of the IEEE EMBS Cité Internationale, Lyon, France, 2007, pp. 1904-1907
- [9] Ziarani, A. K., A Nonlinear Adaptive Method of Elimination of Power Line Interference in ECG Signals, IEEE Transactions on Biomedical Engineering, 49 (2000), pp. 540-547
- [10] Gao, L. X., Jinfeng, H. L., Adaptive Noise Canceller Based on PSO Algorithm, *Proceedings*, IEEE International Conference on Automation and Logistics, Qingdao, China, 2008, pp. 1759-1762
- [11] Widrow, B., Stearns, S. D., Adaptive Signal Processing, Pearson Education, Inc., Upper Saddle River, N. Y., USA, 1985
- [12] Muhammad, A., et al., Calculation and Measurement of Kinetic Parameters of Pakistan Research Reactor-1 (PARR-1), Annals of Nuclear Energy, 38 (2011), 1, pp. 44-48

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#### Џавид КАДИР, Масуд ИКБАЛ, Хамид КАЈЗЕР

## АДАПТИВНО УКЛАЊАЊЕ У РЕАЛНОМ ВРЕМЕНУ ФРЕКВЕНЦИЈСКОГ ШУМА У ПРОВОДНИКУ СА МЕРНОГ КАНАЛА СНАГЕ РЕАКТОРА

Дизајниран је софтвер за адаптивно уклањање у реалном времену фреквенцијског шума проводника од 50 Hz и тестиран на Пакистанском истраживачком реактору-1. Фреквенцијски шум проводника ствара многе проблеме при аквизицији слабих сигнала. Понекад је шум толико доминантан да је оригинални сигнал потпуно прекривен. Иако се за елиминисање шума може корисити филтер непропусник опсега, уколико је жељени сигнал у околини 50 Hz тада и оригинални сигнал бива смањен и долази до пада укупних перформанси. Адаптивно уклањање шума је техника која се може примењивати за уклањање шума на проводнику без снижавања жељеног сигнала. У овом раду приказан је поступак уклањања у реалном времену шума проводника са мерног канала снаге реактора. Адаптивни ЛМС алгоритам коришћен је за елиминисање шума од 50 Hz. Алгоритам је имплементиран у LabVIEW код са NI6204 аквизиционом картицом. Квалитет прибављеног сигнала је значајно побољшан, што се може видети из приказаних резултата.

Кључне речи: адайшивно уклањање, фреквенцијски шум йроводника, мерни канал сна*те* реакшора, Пакисшнски исшраживачки реакшор-1