2-Coefficient Fixed Point Hardwired Adaptive Filtering Based on LMS Algorithm

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Abstract

Adaptive filtering techniques have a wide spectrum of utilization, such as in acoustic and electrical echo cancellation, interference cancellation, cancellation of harmonics, among others. Among the various strategies of adaptation, the Least Mean Square algorithm (LMS) is arguably the most popular, mainly because of its low computational cost, robustness and tracking ability. We are proposing the study of tradeoffs in the design of dedicated structures of fixed point and float point adaptive filters based on LMS algorithm. While the fixed point structure enables the implementation of simpler structure, and therefore speeding up the development time of the project, on the other hand the float-point structure enables higher precision in the results, which allows for less error in stead-state operation of the adaptive filter. However, this higher precision is obtained at the cost of higher complexity in the hardware structure. This work will explore the study of these tradeoffs in the development of the fixed point and float point structures, and after that we intend to apply them to the harmonics cancelling application. The developed architectures were described in hardware description language VHDL and the validation of the structure shows that it is able to cancel the interference from a sinusoidal signal efficiently.

1. Introduction

The important advancements in the design of integrated circuits have enabled complex Digital Signal Processing (DSP) techniques to be applied in traditional signal processing areas such as, audio, video, and image processing, communications, and data compression and transmission [1]. Since DSP systems are often well suited to VLSI implementation, the problem of designing application-specific DSP systems is an intensive studied research topic and has significant industrial and commercial relevance.

In Digital Signal Processing - DSP applications, adaptive filtering is one of the most widely used strategy. In this work, adaptive filter computation, based on Least Mean Square (LMS) algorithm is addressed, where the main goal is the development of efficient fixed-point and float-point architectures for cancelling harmonic power line interference application [2], [3]. For the removal of sinusoidal interference, notch filters can be used with fixed coefficients tuned to the frequency of the interference at each harmonic. However, if the frequency of the interference is not known beforehand with accuracy, or even if the frequency is previously known, but some variations has occurred, the adaptive filter is certainly the best alternative.

The architectures of adaptive filter using LMS algorithm proposed in this work will processes fixed point data by using Q15 format, in order to reduce the complexity/cost of arithmetic operators. In the fixed point of implementation, the convergence of adaptive filter, hence your effectiveness in the cancelling interference process, could become difficult, because the needed rounding of samples. On the other hand, the float point structure should be able to cancel the interferences more efficiently, because the higher precision in the results, which allows for less error in stead-state operation. But, this higher precision in the results is obtained at cost of a more complex hardware. However, our architecture should be able to cancel the interferences efficiently, even by operating on fixed-point. This work proposes the study of these tradeoffs between fixed point and float point architectures. In this point of the work we are able to show a 2-coefficient fixed point adaptive filter architecture based on LMS algorithm. The filter is implemented with a reduced number of coefficients, because according to [4], only two coefficients are needed to represent a sinusoidal signal. The synthesized architecture is validated in order to reduce the minimum error between both the signals from the proposed architecture and the signals from software implementation using floating point data from the simulation of the structure by using MATLAB tool.

The rest of the paper proceeds as follows. Section 2 presents the adaptive filtering background and the proposed architecture is introduced in Section 3. The validation of the structure is given in Section 4 and finally, Section 5 concludes the paper.

2. Background

The adaptive filtering application can be categorized in four classes: identification, inverse modeling, prediction, and interference cancellation [1]. In this work, the last class is addressed, and its block structure is shown in Fig.1.

An adaptive system consists of an adaptive algorithm that aims to adjust the values of the weights of a digital filter that can be composed of FIR (Finite Impulse Response) or IIR (Infinite Impulse Response) filter. In this context, the FIR filter is widely used in adaptive structures, due to its inherent stability, so that it is the best choice for real-time applications [5], [6]. In the block of Fig. 1, *s(n)* is the signal corrupted by additive noise $x_2(n)$. A distorted signal $x_1(n)$, but correlated with $x_2(n)$, is also available. The goal of the adaptive system in this class of application is to produce an output, *y(n)*, that closely resembles $x_2(n)$. Therefore, the output *e(n)* will closely resemble *s(n)*.

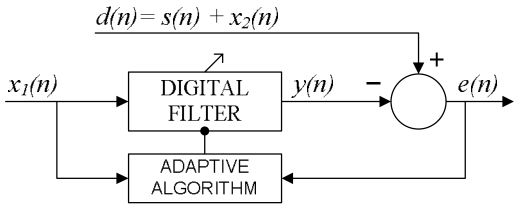

Fig. 1 - Adaptive system: interference cancelling

3. Developed Adaptive Filter Structure

For the harmonic canceller architecture application, the desired input signal *d*(*n*) is operated on 20 bit-width. It occurs, because we have observed that it is the minimum number of bits to suppress the interference with only two coefficients in the filters stages. The structure of one 2-coefficient fixed point adaptive filter architecture is presented in Fig. 2 (we have omitted the control part in order to minimize the complexity of the figure). This architecture was developed for two coefficients, because it will be replied in the future structure of interference canceller in order to suppress the power line interference and its high order harmonics. The process of updating the two coefficients is realized at seven clock cycles, as can be seen in Table I.

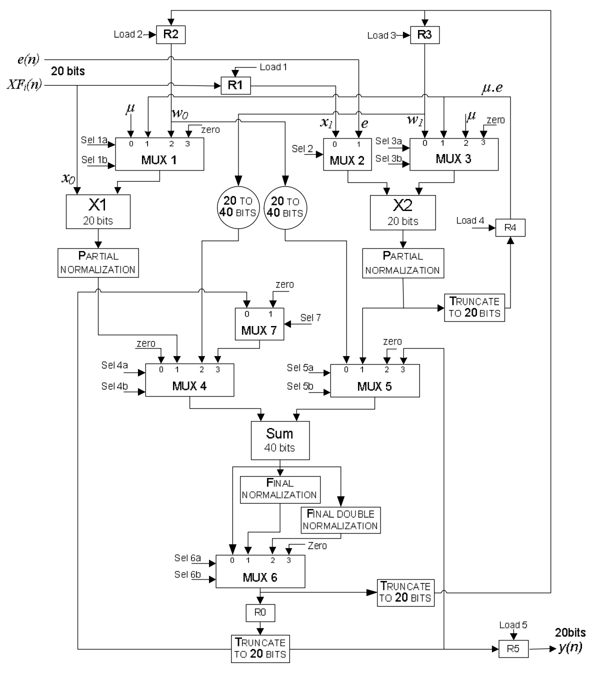

Fig. 2 - Adaptive filter architecture based on LMS algorithm

To operate the necessary signals, the architecture of Fig. 2 includes registers, multiplexers, multipliers, adders and truncation blocks. Also, normalization blocks can be identified. With these blocks the signals are adjusted in order to reduce some error caused by the fixed point operation. The normalization can be divided into two blocks: the partial normalization (where the values from the multipliers outputs *X1* and *X2* are shifted seven bits right), and the final normalization (where the values from the adder output *Sum* are shifted right by eight bits). Hence, it represents a total of fifteen bits shifted right, what leads to the division of the signals by a normalization factor of 32768. The reference signal $XF(n)$ is obtained from the harmonics generator block and from the error signal $e(n)$.

TABLE I. OPERATIONS OF THE LMS ALGORITHM ARCHITECTURE

Clock cycles	***Operation***
1°	$w_0(n).x_0(n)+w_1(n).x_1(n)$
2°	$e(n)=d(n)-y(n)$
3°	$\mu.e(n)$
4°	$w_1(n)+\mu.e(n).x_1(n)$
5°	$w_1(n+1)$
6°	$w_0(n)+\mu.e(n).x_0(n)$
7°	$w_0(n+1)$

As the main goal of the work is to observe the tradeoffs between the fixed and float point architectures, in the future, it will be necessary to adapt the structure shown in Fig. 2 to float point. For this purpose the float point adder and multipliers have already been developed by the group based on [7], [8]. As should be observed, in the float point architecture, the stages of normalization and truncation (presented in Fig. 2 for the fixed point operation), will be avoided. However, the implementation of float point arithmetic operators is more complex than the fixed point ones.

4. Validation of the Proposed Architecture

To validate the developed architecture, the simulation results from the Altera Quartus II were compared with the results from the simulation of the structure by using MATLAB. Fig. 3 shows a sinusoidal signal operating on 50Hz, whose samples were acquired with 4 KHz sampling frequency. This signal contains interference of a sinusoidal frequency of 1 KHz, sampled at a rate of 4 KHz. The simulations were realized using 10,000 samples of the input signal. In the simulation of the adaptive filter, the adaptation step μ= 0.0015 was used. It was chosen, because among all the simulations, it represented the optimal value to the filtering of the desired signal. The obtained signals from the model from MATLAB and from the hardwired adaptive filter and are presented in Fig. 4 and Fig. 5, respectively.

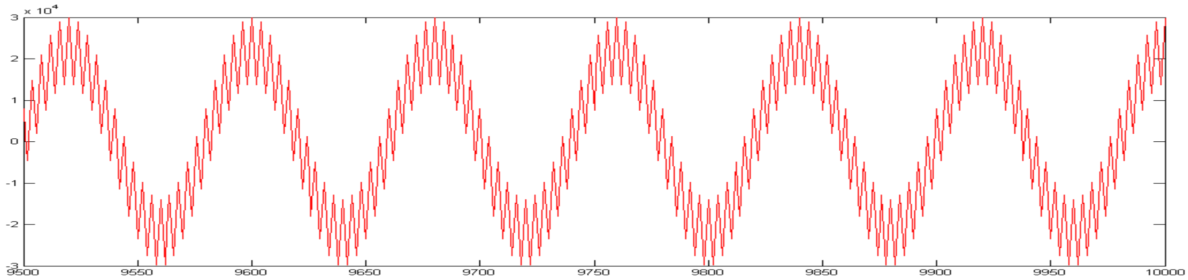

Fig. 3 – Input signal with interference

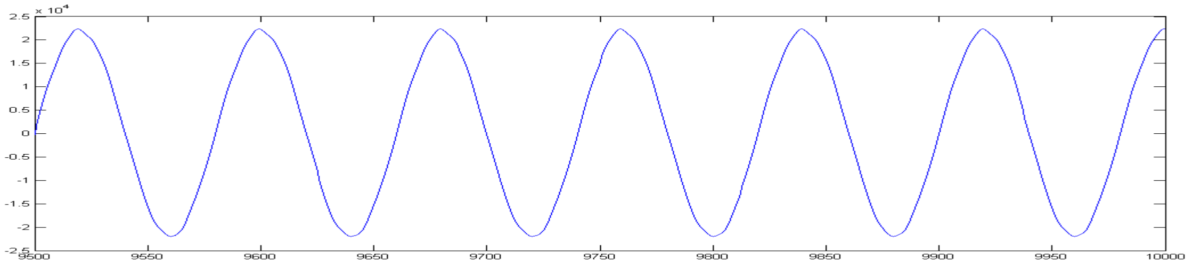

Fig. 4 - Signal obtained from the MATLAB model

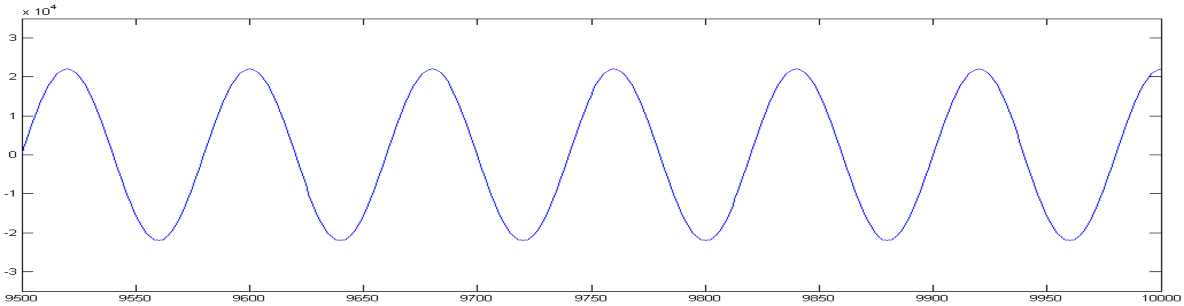

Fig. 5 - Signal obtained from the 2-coefficient hardwired adaptive filter

The graphics presented in Fig. 6 and Fig. 7 show the efficiency of the proposed adaptive filter structure, where two main points should be observed: i) the proposed structure was able to cancel the interferences from the input signal; ii) the filter result from the proposed fixed-point architecture is close to that obtained from the float-point MATLAB model.

5. Conclusion and Future Works

We introduced a hardwired structure for 2-coefficient adaptive filter based on LMS algorithm. The efficiency of the proposed structure could be proved by filtering the interferences from a sinusoidal signal with almost the same quality results as the MATLAB model. Although we have presented only the structure of the adaptive filter, we hope as future work we are able to present a fully structure for cancelling interferences from the power line in fundamental frequency and its high order harmonics. Moreover, we hope to explo*re* the study of the tradeoffs in the development of the fixed point and float point structures, and after that we intend to apply both structures to the harmonics cancelling application.

6. Bibliography

[1] MANOLAKIS, D.G. “Statistical and adaptive signal processing”. Norwood: Artech House, 2005.

[2] COSTA, M. H.; TAVARES, M. C. “Removing harmonic power line interference from biopotential signals in low cost acquisition systems.” Computers in Biology and Medicine 39, 2009. pp. 519 - 526.

[3] CHIMENO, M.F.; PALLÀS-ARENY, R. “A comprehensive model for power line interference in biopotential measurements”, IEEE Transactions on Biomedical Engineering 49 (3) (2000) 535–540.

[4] HAYKIN, S. “Adaptive Filter Theory”. 4.ed. Prentice-Hall, 2002.

[5] KUO, S.M.; LEE, B.H. “Real-Time Digital Signal Processing: Implementations, Applications, and Experiments with the TMS320C55X”. John Wiley & Sons, 2001.

[6] WIDROW, B.; GLOVER, J.R.; MCCOOL, J.M.; et al. “Adaptive noise cancelling: principles and applications”, Proceedings of the IEEE 63 (12) (1975) 1692–1716.

[7] PATTERSON, D., HENNESSY, J. “Organização e Projeto de Computadores”, Rio de Janeiro: Campus, 2005.

[8] LOISEAU, L. at al. “A Flexible Floating-Point Format for Optimizing Data-Paths and Operator in FPGA Based DSPs”: Montréal, 2002.