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Variable Constraint based Least Mean Square algorithm for power system harmonic parameter estimation

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ABSTRACT

This paper presents the maiden application of a novel signal processing algorithm called Variable Constraint based Least Mean Square (VCLMS) for power system harmonic parameter estimation. The amplitude, phase and frequency of a power signal containing harmonics, sub-harmonics, inter-harmonics are estimated using this algorithm in the presence of white Gaussian noise under simulating environment. Four Least Mean Square (LMS) based algorithms, reported in the literature are considered for judging the comparative performance with the proposed algorithm. These algorithms are applied and tested for both stationary as well as dynamic signals containing harmonics. Practical validation is made with the experimentation of the algorithms with real time data obtained from a solar connected inverter system used for supplying electrical energy during power cut at National Institute of Technology (NIT) Silchar through a power quality analyzer and estimation are performed in MATLAB simulation. Comparison of the results amongst LMS, Normalized LMS, Complex Normalized LMS, Variable Leaky LMS and VCLMS algorithms reveals that proposed VCLMS algorithm is the best in terms of accuracy and computational time.

Introduction

The ideal sinusoidal nature of a voltage or current waveform deteriorates due to the presence of nonlinear loads and power electronic devices in power system. The increase in the use of this type of loads causes much more harmonic pollution in the entire electrical power network [1]. The presence of harmonic pollution in power network significantly deteriorates the power quality also. So under this circumstance, it is necessary to estimate the parameters of the harmonics for reducing the harmonic pollution from the power network. After estimating harmonic parameters, such as amplitudes, phases and frequencies, the harmonic components can be compensated by injecting appropriate compensating devices into a power system [2,3].

Various approaches have been proposed to estimate the parameters of these harmonics in the literature. One of the suitable approach is the Fast Fourier Transform (FFT) for stationary signals, but it loses accuracy under time varying frequency conditions and also posses picket and fence problems [4,5]. The International Electro-technical Commission (IEC) [3] has recommended some standard drafts for a specified signal processing method and also the definitions for harmonic, sub-harmonic and inter-harmonic measurement. IEC has suggested to use of Discrete Fourier Transform (DFT) over a rectangular window of exactly 12 cycles for 60 Hz (10 cycles of 50 Hz) and frequency resolution of 5 Hz [6]. DFT is known for most widely used algorithm for harmonic analysis in power system as it is a fast executable algorithm. However, the DFT-based algorithms do not perform stably for systems with time varying frequency conditions [7,8].

The Recursive Least Square (RLS) group and Kalman filtering (KF) group are known as a better approach for its simplicity and robustness for the estimation of harmonics and frequency in power system. However, these methods require the knowledge of a prior statistics of the signal and the state matrix needs to be defined accurately as well. If the initialization of the parameters of these algorithms are not done accurately then the estimation performance degrades for these methods [8,11,12,15–17,21–23].

The conventional Least Mean Square (LMS) algorithms based on adaptive linear filtering process have the capability of frequency analysis of static as well as dynamic signals because of their adaptive features [9,18]. LMS algorithms are computationally efficient and robust with respect to the dynamic variations in the system. The advantage of its simplicity is in its underlying structure. However, it suffers from the problem of poor convergence rate if the step size for adaptation is fixed. This algorithm requires an

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understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable [18].

The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm, which bypasses this issue by calculating maximum step size value [13,14]. The noise amplification becomes smaller or less size due to the presence of normalized step size [14]. It has minimum steady state errors and faster convergence, but the NLMS algorithm requires more number of computations for evaluation when compared to a conventional LMS algorithm due the presence of the reference signal power, which will lead to more computations in NLMS algorithm [14,15]. The NLMS algorithm requires $3M + 1$ multiplications, which is M times more than the LMS algorithm [14]. A Least Mean Square (LMS) algorithm in complex form has been presented in [18] to estimate the power system frequency.

A complex signal for LMS algorithm is derived from three-phase signal using $\alpha - \beta$ transformation. This algorithm suffers from poor convergence rate as the step size of the LMS is fixed [10]. If the LMS of the error is only considered as the cost function to be minimized then with respect to the dynamic variation the linear weights of the filter may go unbounded or take a longer time to respond because of the stalling effect [18–20].

The variable leaky least-mean-square (VLLMS) algorithm is one of the improved LMS-based algorithms that use a leakage factor to control the weight update of the LMS algorithm [19,20]. This leakage factor solves the problem of drifting in the LMS algorithm by bounding the parameter estimate. It also improves the tracking capability of the algorithm, convergence and the stability of the LMS algorithm. But one of the main drawbacks of the VLLMS algorithm is its low convergence rate as compared to the other improved LMS based algorithms [25].

However, the performance of the reported algorithms of this class is limited under real time environment for a solar PV system. Recently, Bouzelata et al. [26] have investigated a new unified power quality conditioner (UPQC) supplied by photovoltaic system to realize the limits of standards of a solar PV system. The authors have proposed a PV system with 64 PV panels, boost converter, PI controllers and MPPT algorithm have been designed to generate the maximum active power, which is exported to the network through the shunt APF function of the UPQC [26]. The simulation results performed by MATLAB/Simulink. The authors have proved that UPQC fed by PVs has offered a promising way to annihilate the current and voltage harmonics [26].

Another work carried out by the same authors Bouzelata et al. for a solar PV system referred in [27]. In this paper, a multifunctional Active Power Filter (APF) fed by a PV system is proposed in order to remove these harmonics problems. The design and the analyses were carried out in simulink software. The simulation system is considered to have a nonlinear load which causes a harmonic disturbance and increases the Total Harmonic Distortion (THD) in the grid line. The proposed APF removes the most leading harmonics by using two different current detection algorithms and the obtained results are compared in terms of current THD level and the power factor [27].

In view of the above, an urge is felt to investigate the performance of Variable Constrained based Least Mean Squares (VCLMS) algorithm for power system harmonic parameter estimation under simulating and real time environment. VCLMS is a simple stochastic gradient-descent algorithm, which requires only a variable constraint adjustment technique to impose on the solution for minimizing the Mean Square Error (MSE) and avoiding the drifting of weights in the estimation mechanism [24,28,29]. A major advantage of the VCLMS algorithm is that it has a self-correcting feature permitting it to operate for arbitrarily long periods of time in a digital computer implementation without deviating from its constraints because of cumulative round off or truncation errors [28]. This is achieved by employing the sum of exponentials of the error as the cost function. A leakage factor is added to the sum of exponential cost function, which makes the proposed algorithm a combination of the generalization of the mixed norm stochastic gradient algorithm with a leaky factor. The algorithm is applicable to array processing problems in Geosciences, sonar, and electromagnetic antenna arrays in which a simple method is required for adjusting an array in real time to discriminate against noises impinging on the array side lobes [29–34].

The main objectives of the present work reported in this paper are

(a) Maiden application of VCLMS algorithm is proposed for estimating harmonic parameters such as amplitudes, phases and frequency of the fundamental, harmonics, inter and sub harmonics in the presence of various noises in power system signal.
(b) To evaluate the comparative performance of the proposed algorithm as compared to other LMS based algorithms like LMS, NLMS, CNLMS and VLLMS for finding the best harmonic estimator.
(c) To evaluate the performance of the algorithms for accurately estimating harmonic parameters on the data obtained from a real time experimental setup at NIT Silchar through solar connected inverter system for finding the best and appropriate method for harmonic parameter estimation.

Variable Constraint Least Mean Square (VCLMS) algorithm

The VCLMS algorithm presented in this section is referred from [34]. If we consider x_n as the input harmonic signal corrupted by white Gaussian noise and w_n be the weight coefficient of the filter, which is used to estimate the output of the filter termed as estimated output y_n and is given by

$$y_n = w_n^T x_n \tag{1}$$

Also the error generated during estimation represented by e_n, which is the difference between the desired signal d_n and estimated output of the filter y_n, will be given by

$$e_n = d_n - w_n^T x_n \tag{2}$$

Now the objective of the approach is to minimize this error in the mean square sense, subject to the constraint

$$c^T w = a \tag{3}$$

where a is a constant and c is a fixed vector.

Now using the Lagrange multiplier method the new objective function is as follows

$$J_c = E\{e_n^2\} + \lambda(c^T w - a) \tag{4}$$

where λ is the Lagrange multiplier. Hence, the following relation becomes

$$\nabla_w J_c = 0 \text{ and} \tag{5}$$

$$\frac{\partial J_c}{\partial \lambda} = 0 \tag{6}$$

This must be satisfied simultaneously. The term $\frac{\partial J_c}{\partial \lambda}$ produces the constraint $c^T w = a$. Next by substituting the error e_n in Eq. (4) to obtain

$$J_c = J_{\min} + \xi^T R_X \xi + \lambda(c^T \xi - a') \tag{7}$$

where