Electrical Power and Energy Systems 75 (2016) 91–98

Contents lists available at ScienceDirect

Electrical Power and Energy Systems

journal homepage: www.elsevier.com/locate/ijepes

Real time harmonics estimation of distorted power system signal

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ARTICLE INFO

Article history:
Received 3 January 2015
Accepted 28 August 2015
Available online 11 September 2015

Keywords:
Variable Leaky Least Mean Square (VLLMS)
Least Mean Square (LMS)
Fast Fourier Transform (FFT)
Kalman Filter (KF)
Recursive Least Square (RLS)

ABSTRACT

Power system harmonics should be estimated properly for the maintenance of power quality. So, there is a need for development of more accurate and computationally efficient estimation technique. In this context, this paper presents a harmonics estimation technique using a Variable Leaky Least Mean Square algorithm. In this proposed method drifting of parameters is avoided using a leak adjustment method. To improve the rate of convergence, step size is also adapted in this method. Real time power system is also imitated using different cases through simulation, which enables the superiority of the proposed method over the other existing techniques. Studies made on experimental data, processed in arduino due microcontroller, also support its superiority.

Introduction

Estimation of power system harmonics is now a days one of the important factors for power system researchers. Because of significant growth of power electronics based loads in power systems, power system signals become highly distorted. Main sources of inter harmonics and sub harmonics in power system are power electronics devices (e.g., cycloconverters), arcing loads (e.g., welding machines and arc furnaces), which are widely used in rolling mills and linear motor drives [1]. Recently, grid integration of Renewable energy sources involving various types of power electronics based converters also introduce more amount of harmonics to the power system [2,3]. The objective of this paper is on developing efficient harmonics estimation technique and after that to design filter for removal of unwanted harmonics distortion in power system.

In literature [4–8] so many algorithms have been applied for power system harmonics estimation. Out of these Fast Fourier Transform (FFT) is the very fundamental. Frequency is not constant, because of leakage and picket-fence effects [9,10]. Some methods [11–13] are provided to improve these drawbacks. Then Kalman Filter is the right choice for harmonics estimation in a power signal [14]. However, to optimize estimations, the higher-order terms in the Taylor's expansion for this method were neglected because of a nonlinear function including the formulation of measurements. As a matter of fact, estimation of distorted signals may occur incorrectly or take longer to converge and even diverge. A method which is mostly used for parameter estimation is Least Square (LS) algorithm [6]. Most researchers used this technique for estimation of power system harmonics. But online estimation is not possible in this method because it is batch processing method.

A power system digital harmonic analyzer [15] having built in FFT, where 68,000 microprocessor-based instrument samples the waveform and calculates the harmonics of signal for both current and voltage waveforms. Method based on Taylor–Kalman–Fourier [16] was also used for instantaneous phasor and harmonics estimation of power system signals. A method based on filter bank (FB) with adaptive filtering [17] was used for estimation of parameters describing harmonics and interharmonics in a power signal. Each harmonic and interharmonic component is estimated recursively. Advantages and disadvantages of adaptive filters such as cascade FIR comb filters and resonator based filters for estimation problem were discussed and an improved & suitable combined algorithm [18] based on above two filters using decoupled modules and multirate sampling was proposed for frequency and harmonics estimation of power signal. Using wavelet transform [19], analysis on harmonics distortion was carried out by many researchers. A fast and accurate approach for real time estimation for time varying harmonics of voltage and current signal based on rotational invariance technique (ESPIRIT) [20,21] assisted adaptive wavelet neural network (AWNN) is presented. AWNN provides quick estimate where as ESPIRIT handles for time varying signal with higher accuracy.

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http://dx.doi.org/10.1016/j.ijepes.2015.08.017
0142-0615/

Now a days soft computing methods such as fuzzy logic, artificial neural networks (ANN), genetic algorithm and Bacterial Foraging Optimization (BFO) techniques are applied for harmonics estimation by many authors [24–26]. But these methods have drawbacks involving computational complexity and convergence rate. In ANN, learning parameter and initialization of weight vector affect its convergence characteristics.

Due to involvement of fixed step sixe in conventional LMS technique, its rate of convergence is slower. Poor convergence problem of LMS can be overcome using time varying step size [23]. As Least Mean Square of error is taken as the cost function in LMS algorithm, weights are not bounded and it takes more time to respond because of stalling effect [23]. To overcome this, leaky LMS algorithm is employed in which magnitudes of weights are also included in cost function to eradicate the stalling and parameter drifting effect [27].

In this paper, a harmonics estimation technique based on Variable Leaky LMS (VLLMS) [28–31] is developed. So, power system signal is modeled in a parametric form and then the proposed algorithm is applied on this signal. Estimation performance of proposed algorithm is compared with other algorithms applied by researchers, e.g., RLS [22] and LMS [23] for different cases that may arise in a power system, e.g., sudden rise or fall in amplitude of signal, inclusion of sub-harmonics and inter-harmonics in the signal and also for dynamic signal. Finally, harmonics estimation of laboratory generated signal i.e. across the output of a single phase PWM inverter, using arduino due microcontroller is also presented. In each case of estimation, proposed VLLMS outperforms over the other two algorithms.

The remaining of the paper is organized as follows. Proposed VLLMS algorithm and its mathematical analysis are presented in Section 'VLLMS based harmonics estimation'. Section 'Simulation results' discusses on the simulation results obtained using proposed algorithm with comparative estimation results using other existing algorithms. Section 'Conclusions' presents the hardware implementation of the proposed algorithm using arduino due microcontroller and the obtained estimation results. The conclusion of the paper is given in Section 'Conclusions'.

VLLMS based harmonics estimation

Consider the voltage or current waveforms of the known fundamental angular frequency ω as the sum of harmonics of unknown magnitudes and phases. General form of the waveform can be expressed as

$$y(t) = \sum_{n=1}^{N} A_n \sin(\omega_n t + \phi_n) + A_{dc}\exp(-\alpha_{dc} t) + \varepsilon(t) \tag{1}$$

where N = number of harmonics; $\omega_n = n2\pi f_0$; f_0: fundamental frequency; $\varepsilon(t)$: noise; $A_{dc}\exp(-\alpha_{dc}t)$: dc offset decaying term.

After discretization of Eq. (1) with a sampling period, T one obtains the following expressions

$$y(k) = \sum_{n=1}^{N} A_n \sin(\omega_n kT + \phi_n) + A_{dc}\exp(-\alpha_{dc} kT) + \varepsilon(k) \tag{2}$$

Invoking Taylor series expansion of the dc decaying term, $A_{dc}\exp(-\alpha_{dc}t)$ and retaining first two terms and deleting higher order terms of the series.

$y(k)$ obtained as

$$y(k) = \sum_{n=1}^{N} A_n \sin(\omega_n kT + \phi_n) + A_{dc} - A_{dc}\alpha_{dc} kT + \varepsilon(k) \tag{3}$$

In order to estimate amplitudes and phases Eq. (3) can be rewritten as

$$y(k) = \sum_{n=1}^{N} [A_n \sin(\omega_n kT)\cos\phi_n + A_n\cos(\omega_n kT)\sin\phi_n] + A_{dc} - A_{dc}\alpha_{dc} kT + \varepsilon(k) \tag{4}$$

Eq. (4) can be rewritten in parametric form as follows

$$y(k) = H(k)X \tag{5}$$

$$H(k) = [\sin(\omega_1 kT)\ \ \cos(\omega_1 kT)\ \ \ldots\ \ \sin(\omega_N kT)\ \ \cos(\omega_N kT)\ \ 1\ \ -kT]^T \tag{6}$$

Unknown parameter vector

$$X = [A_1\cos(\phi_1)\ \ A_1\sin(\phi_1)\ \ \ldots\ \ A_n\cos(\phi_n)\ \ A_n\sin(\phi_n)\ \ A_{dc}\ \ A_{dc}\alpha_{dc}]^T \tag{7}$$

The VLLMS algorithm [27] is applied to estimate the state. The algorithm minimizes the square of the error recursively by altering the unknown parameter X_k at each sampling instant using Eq. (8) given below

$$X_{k+1} = (1 - 2\mu_k\gamma_k)X_k + 2\mu_k e_k \hat{y}_k \tag{8}$$

where the error signal is

$$e_k = y_k - \hat{y}_k \tag{9}$$

Step size μ_k is varied for better convergence of the VLLMS algorithm in the presence of noise.

$$\mu_{k+1} = \lambda\mu_k + \gamma_k R_k^2 \tag{10}$$

where R_k represents the autocorrelation of e_k and e_{k-1}. It is computed as

$$R_k = \beta R_{k-1} + (1-\beta)e_k e_{k-1} \tag{11}$$

where β is an exponential weighting parameter and $0 < \beta < 1$, and $\lambda(0 < \lambda < 1)$ and $\gamma > 0$ control the convergence time.

The variable leakage factor γ_k [27] can be adjusted as

$$\gamma_{k+1} = \gamma_k - 2\mu_k \rho e_k \hat{y}_k X_{k-1} \tag{12}$$

After the updating of the unknown parameter vector using LMS algorithm, amplitudes, phases of the fundamental and nth harmonic parameters can be derived as

$$A_n = \sqrt{(X_{2N}^2 + X_{2N-1}^2)} \tag{13}$$

$$\phi_n = \tan^{-1}\left(\frac{X_{2N}}{X_{2N-1}}\right) \tag{14}$$

Fig. 1 shows the flow chart of the estimation scheme of VLLMS algorithm.

Simulation results

Static signal in presence of random noise and DC component

A discretized power system signal having 50 Hz fundamental frequency is generated in MATLAB. The signal also contains higher order harmonics such as 3rd, 5th, 7th, 11th and a DC decaying component [9]. Noise value taken as $\mu(t) = 0.05$ randn.

$$y(t) = 1.5\sin(\omega t + 80^\circ) + 0.5\sin(3\omega t + 60^\circ) + 0.2\sin(5\omega t + 45^\circ) + 0.15\sin(7\omega t + 36^\circ) + 0.1\sin(11\omega t + 30^\circ) + 0.5\exp(-5t) + \mu(t) \tag{15}$$

Different parameters [27–30] used in the simulation studies are selected on the basis of series of experiments performed during the implementation of this method and are given in Table 1.