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Improved Recursive Newton Type Algorithm based power system frequency estimation

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ABSTRACT

This paper presents power system frequency estimation by using an Improved Recursive Newton Type (IRNTA) algorithm. The proposed approach uses Jacobian and covariance matrices for updating the unknown parameters. The recursive form of unknown parameters and covariance matrix are incorporated in the algorithm to have faster convergence. The performance of the proposed algorithm is studied through simulations and experiments for several critical cases that often arise in a power system. Efficacy of the proposed algorithm is also compared with other signal processing techniques such as Recursive Least Square (RLS) and Kalman Filter (KF). Studies made on industrial data also support for the superiority of the proposed algorithm.

Introduction

Power Quality (PQ) has become an important research subject due to the increased number of non-linear components in modern power systems. The power electronic equipment introduces harmonic distortions that can deteriorate power system quality and affect the system performance [1]. In order to maintain the specified PQ level, it is crucial to control the quality of the supplying voltage. Therefore, harmonic filters must be used to reduce the level of harmonic distortions. The operation of these filters relies on the quality of the monitoring of the harmonic distortion. The monitoring relies on the quality of transducers used in the measurement scheme, as well as on the methodology for the assessment of signal distortions. In the past a number of methods have been developed to estimate harmonic distortions in power systems. One of widely used methods is the Fast Fourier Transform (FFT). This method is simple and efficient; however it is sensitive to the changes of the fundamental signal frequency [2]. Furthermore, to achieve a fast convergence, the FFT data window must be short enough, which might lead to limitations in terms of extraction of frequency components. Least Square (LS) method has also been widely used as a parameter estimator [3,4]. It minimizes the sum of the squares between measured and observed values [5]. In different applications, LS has shown the estimation accuracy with the presence of large disturbance and frequency deviations. Another method, Kalman Filter (KF), improves the quality of estimation by considering more rigorously the understanding of the signal noise [6–8]. In a number of cases LS and KF have shown their advantages compared to FFT based techniques in processing signals consisting of variation in frequency. However, the dynamics involved in LS and KF raises concern since it exhibits poor performance [10] with respect to sudden change in amplitude, phase or frequency of signal i.e. during dynamic changes in power system parameters, convergence of the algorithm is slow means it takes more time to track the parameters of the signal.

So far as identifications of non stationary systems are concerned, [11,12] discusses on time frequency signal filtration applied to recursive method of modal parameter identification. Model order of the system is reduced and signal components are separated by using adaptive wavelet filtering. For identification of Linear Time Varying (LTV) systems [13] having non stationary properties and small magnitude vibration, a typical subspace based technique is used to extract the observability range space using the Singular Value Decomposition (SVD) of a general Hankel matrix.

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The varying transition matrix is estimated at each moment through the SVD of two successive Hankel matrices. A vector vibration response measurement based [14] parametric identification for time varying structure is reported, where the identification is based on three simultaneously measured vibration response signal obtained during a single experiment. An ARMA (Auto Regressive Moving Average) [15–17] with varying co-efficients is also a very familiar choice in using the adaptive estimate methods. As these models were initially proposed for scalar (single-input and single-output) systems, the use of them is quite difficult in multivariable systems. So, to address the more complex problem, state space model is preferred. Popularity of the state space model is due to the latest development in the subspace-based methods for state space model realization. Accurate spate space model of multi variable systems can be obtained directly from input–output data using subspace methods.

In this paper, an IRNTA [19] based frequency estimation [20] technique is developed. The performance of such a formulation is studied for several critical cases that often arise in a power system, e.g., sudden change in frequency, amplitude and phase of signal, in presence of harmonics in the signal. Efficacy of the proposed algorithm is verified through industrial data obtained from a Ferro Alloys plant. Finally, frequency estimation of the laboratory data collected from normal working day of a Laboratory is also presented.

The remaining of the paper is organized as follows. Section 'Signal model presentation and algorithm development' presents the proposed algorithm and formulation of equations. Section 'IRNTA based frequency estimation' describes the frequency estimation based on the proposed algorithm. Section 'Simulation studies' discusses the simulation results of the proposed algorithm. Section 'Experimental studies' presents some schematic for industrial data collection as well as experimental setups in laboratory and test results for the data obtained from them. Section 'Conclusion' concludes the paper.

Signal model presentation and algorithm development

Let us assume the following observation model of the input signal (arbitrary voltage or current);

$$A(t) = h(x,t) + \eta(t) \tag{1}$$

where $A(t)$ is an instantaneous signal at time t, $\eta(t)$ is a random noise, X is a suitable parameter vector, and $h(.)$ is expressed as follows:

$$h(x.t) = A_0 e^{-\delta t} + \sum_{k=1}^{M} A_k \sin(k\omega t + \varphi_k) \tag{2}$$

For the generic model (2), a suitable vector of unknown parameters is given by

$$x = [A_0, \delta, \omega, A_1, \ldots.., A_M, \varphi_1, \ldots.., \varphi_M]^T \tag{3}$$

where A_0 is the magnitude of the decaying dc component at $t = 0$, $\delta = 1/T$, T being the time constant, M is the highest order of harmonics present in the signal, ω is the fundamental angular velocity,

Table 1
Parameters used for simulation studies (RLS, KF and IRNTA).

Algorithms	$\delta (P = \delta I)$	λ
RLS	100	0.96
KF	100	–
IRNTA	100	0.96

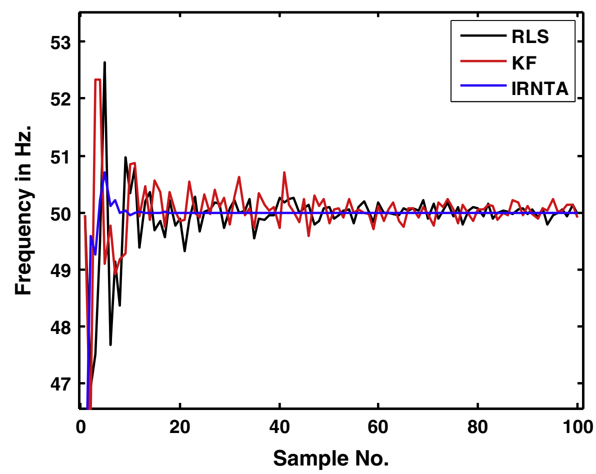

Fig. 1. Estimation of frequency from noisy signals with SNR 40 dB.

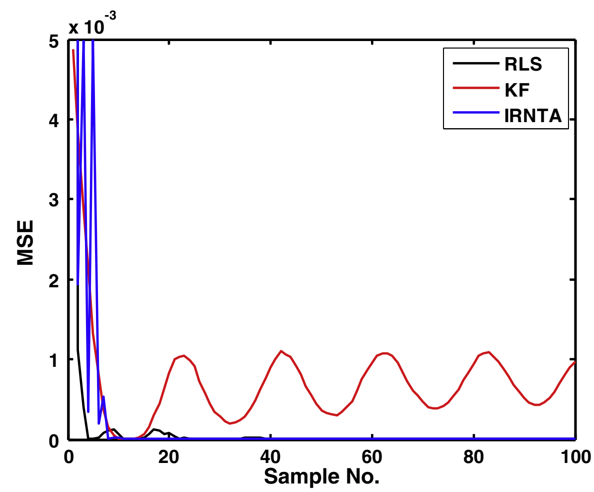

Fig. 2. Estimation performance in MSE of frequency of signal.

Table 2
Comparative assessment of RLS, KF and IRNTA algorithms.

Parameter	RLS	KF	IRNTA
Estimated frequency	49.84	49.78	49.997
Frequency deviation, %	0.32	0.44	0.006
Computational time (seconds)	0.1024	0.1375	0.0994

equal to $2\pi f$, f being frequency, A_K is the magnitude of the kth harmonics. The number of unknowns, i.e., the model order, is

$$n = 2M + 3 \tag{4}$$

The model (2) can be simplified, e.g., containing only the fundamental harmonic. This is due to the fact to reduce the order of the system and our requirement i.e. determination of frequency can be met with that simplification. The model selection depends on the application, i.e., on the features of the input signal processed.

The vector of unknown model parameters (3) can be estimated by applying non recursive NTA numerical algorithm given by

$$\hat{x}_{k+1} = \hat{x}_k + (J_k^T J_k)^{-1} J_k^T (A - h(\hat{x}_k)) \tag{5}$$

where k is an iteration index, J is an (m, n) jacobian matrix, A is an $(m.1)$ measurement vector, h is an $(m.1)$ vector of nonlinear functions determined by the model assumed, and m is the number of