Group B : *Engineering Educations, Applications, and Techniques*

Reviewing Processing Algorithm Adopted in Power System Harmonics Measurement

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Manuscript received March 10, 2012
Revised March 31, 2012

ABSTRACT

Harmonics measurement in power system plays an important part in measuring and improving power quality. Many methods have been proposed to determine the amplitude and phase of fundamental and harmonic components of time-varying waveform. As this information is essential in several applications including protection, control and monitoring in power system. The three basic considerations when selecting the suitable method are; the response time, implementation complexity and accuracy. This paper presents a brief review of different advanced signal processing techniques proposed for used in power systems applications. The basic requirements of harmonic measurements in power system are generally described. The method applied for measuring harmonics are analyzed and discussed.

Keywords: *DFT, least squares algorithm, Kalman filter, Fuzzy logic and Power system Disturbance Quantities*

1 INTRODUCTION

Preferably, an electricity power supply should provide a perfectly sinusoidal voltage signal at every customer location. However, for a number of reasons, utilities often find it is hard to preserve such desirable condition. Power quality problems can described as any variation in the electrical power waveform, such as voltage dips, voltage swell and sag, and harmonic distortion. Waveform distortion, generally expressed as harmonic distortions, describe the deviation of voltage and current waveforms from sinusoidal shape that may contain time-varying components [1]. Waveform distortions are caused by nonlinear devices in power systems. The dramatically increase in the use of nonlinear equipments and loads for efficient voltage control and energy utilization has created a situation where waveform distortions are generated, injected and propagated in lines and feeders in power systems.

Waveform distortion in power systems are inherently time-varying due to continuous changes in system configurations, load types and load levels. The time-varying nature of waveform distortions requires a real-time tool to process that is essential in several applications including protection, control and monitoring in power systems [2] – [5].

In this paper presents a comprehensive review on the recent real-time signal processing algorithms of harmonic component measurement of time-varying waveform distortion in power systems. Reviewing of evolving real-time signal processing algorithms provide understanding into problems in power system measurement and insight to the solutions that can be put into practical applications. A time-varying waveform identification method that is suitable for real time monitoring should satisfy the following criteria:

- Mathematical simplicity so that it can be practically implemented;
- Fast detection and tracking of disturbances;
- Accuracy in extracting disturbances signals;
- Robustness for reliability.

2 REAL TIME PROCESSING TECHNIQUES FOR TIME-VARYING HARMONICS ESTIMATION

This section briefly explains several recent methods for identification of amplitude and phase-angle of the fundamental and its harmonic components that is used in power systems applications. The principles of identifying harmonics using the discrete Fourier transform (DFT) based algorithms, least squares method, wavelet transforms (WT) and Kalman Filter (KF) methods are explained. Their advantages and shortcomings are mentioned.

Equation (1) represents a general power system signal which consists of a fundamental and harmonics and is used as the input signal for the various methods to be studied in the following literature review.

$$y(t) = X_1 \cos(\omega_1 t + \theta_1) + \sum_{i=2}^{K} X_i \cos(\omega_i t + \theta_i) + e(t) \quad (1)$$

where, X_i is the unknown magnitude of the signal; where subscript 1 refers to the fundamental component; $\omega_i t + \theta_i$ is the unknown instantaneous angle of the i^{th} component; and ω_1 is a known fundamental angular frequency, usually fixed at 50/60 Hz. The i is an integer value and usually $\omega_i = i\omega_1$, where ω_i represents the frequency of the i^{th} component. However, in some power systems, inter-harmonics exist and for inter-harmonics $\omega_i \neq i\omega_1$. K is the total number of harmonic components. The variable $e(t)$ represents the additive a stationary noise possesses of zero mean with variance σ^2

2.1 Discrete Fourier Transform

Harmonics are steady-state concept where the waveform to be analyzed is assumed to repeat itself forever. The discrete Fourier transform (DFT) is a discrete complex valued series for calculating the Fourier coefficients. The DFT is the most well-known tool for estimation of amplitude and phase angle of the fundamental and harmonic components in a power system signal [6] - [7].

The basic equation that describes the DFT is

$$x_h = \sum_{k=0}^{N-1} x(k) e^{-j\frac{2\pi hk}{N}} \quad (2)$$

where

k is the present data point,

h is order of each harmonic component,

N is the window length (i.e. number of captured data points),

$x(k)$ is the input signal at point n , and

x_h is the DFT evaluated as complex series at h^{th} harmonic.

The exponential component of (2) is described as follows,

$$e^{-j\frac{2\pi hk}{N}} = \cos\left(\frac{2\pi hk}{N}\right) - j\sin\left(\frac{2\pi hk}{N}\right) \quad (3)$$

In general, the transform into the frequency domain will be a complex valued function that includes magnitude and phase angle by plugging (3) into (2) as follows;

$$x_h = \sum_{k=0}^{N-1} x(k)\cos\left(\frac{2\pi hk}{N}\right) - j\sum_{k=0}^{N-1} x(k)\sin\left(\frac{2\pi hk}{N}\right) \quad (4)$$

Equation (3) can be expressed in their complex form by

$$x_h = X_{hr} + jX_{hi} \quad (5)$$

where

X_{hr} is the real value of x_h ,

X_{hi} is the imaginary value of x_h ,

The magnitude and phase angle for the h^{th} harmonic can be given as follows.

$$\text{magnitude} = X_h = \sqrt{X_{hr}^2 + X_{hi}^2} \quad (6)$$

$$\text{phase angle} = \theta_h = \tan^{-1}\left(\frac{X_{hi}}{X_{hr}}\right) \quad (7)$$

However, the DFT method needs data points sampled over one cycle of signal to calculate the harmonic components accurately. In other words, the minimum possible detection time of the DFT is one fundamental cycle. Therefore, the DFT based algorithms are known to be unsuitable for harmonic analysis with time-varying signals [9]. Also, the corruption of the DFT would yield inaccurate results due to leakages/ or resolution limit of the DFT (leakages occur if the number of periods sample is not an integer) and picket-fence (occur if the analysed waveform includes inter-harmonics) [8]. Though several techniques can be adopted to resolve the pitfalls of DFT applications, additional computational burden may be introduced [9]. Beside, the usage of DFT cannot display the dynamic characteristics of measured signals over time because of the assumption of stationary measured signals. Another shortcoming of the DFT based methods is their sensitivity to noise. Moreover, the DFT result in significant estimation errors for changes in fundamental frequency, even for slight changes. To help solve this problem, recent studies have reported improvements with the DFT based procedures which first estimate the frequency and then apply the DFT using a window which is a multiple of the estimated period in order to obtain accurate harmonic magnitudes [10]. However, these methods have more complicated structures due to the frequency estimation requirement. Attempts to analyze time-varying harmonics using the filter banks based on DFT have been proposed in [11]. This methodology has some drawbacks associated with analysis of inter-harmonics and with real-time application.

Therefore, DFT gives accurate results under the following conditions:

- The signal is stationary
- The sampling frequency is greater than two times the highest frequency within the signal
- The number of samples per period is an integer
- The waveform does not contain frequencies that are non-integer multiples (i.e. interhermonics) of the fundamental frequency.

2.2 Wavelet Transform

Literature has reported on the application of wavelet transform (WT) on power studies, such as power quality disturbance detection and classification [12] – [14], and harmonic extraction [15] – [17]. The wavelet transform is a Multi-resolution Analysis, which analyzes the signal at different frequencies with different resolutions. This analysis gives good time resolution at high frequencies and poor time resolution at low frequencies. Although the wavelet is a tool that can extract fast transients, it also has some shortcomings. It requires digital filters and extensive computational efforts to handle time-varying waveform distortion detection. The performance is also greatly dependent upon the mother wavelet. There are recent literatures that claim various mother wavelets as the most effective one. Among the most recent are the Daubechies 8 (Db8) [12] and Daubechies 9/7. However, the practical implementation has some problems, which include a delay of about 1 fundamental cycle [18]. There are also some suggestion for using sine wave as a first level approximation of reference and comparing it to the values of the line voltage extracted by the wavelet. This will obviously add to the complexities for hardware implementation of wavelet. Due to these drawbacks, wavelet is mainly utilized as an offline waveform analyzer or power quality classification tool that will be applied on stored data.

2.3 Least squares method

In least squares method the main harmonic of signal is determined by finding the coefficients of Fourier transform which make the error between the Fourier series and the actual data minimum. The least square technique possesses simplicity and robustness. Owing to these features, the power system harmonic estimation can be improved dramatically by using the least square algorithm and it has found applications in areas such as power quality monitoring [19] - [21].

This process is usually done by matrix manipulation and it requires extensive calculations. The most prominent limitation is the high computation cost due to the matrix inversion and number of matrix multiplications. The conventional least squares method requires $(2K)^3 + 8LK^2 + L$ number of multiplications /additions operations to be performed. where L represents the total number of captured samples (this should agree with the Nyquist theory) and K is the total number of harmonics,

Consequently, the processing requirement for real-time implementation of the conventional least squares algorithm is very high. Moreover, real-time implementation of the conventional least squares algorithm is rather susceptible to round-off error, which is a characteristic of computer hardware [22], due to the matrix inversion operation. This round-off error may cause failure in the numerical process.

In order to solve the problem an efficient least square method is proposed [23]. This method utilizes some algebraic manipulations to simplify the complex calculations of least square method. The approach manipulates the general Digital Fourier Transform matrix and decomposes it to two matrixes and uses rotation matrix to avoid complex calculations. Therefore, the number of matrix calculations is dramatically decreased. It requires a pre-calculated matrix that will be a constant coefficient for fundamental and harmonic component manipulation. The size of the matrix is $2K \times L$. This matrix is subjected to changes by altering the sampling parameters. It requires only $2K \times L$ multiplication and additions for each output. The required detection time is half a cycle of fundamental.

2.4 Kalman Filtering Algorithm

Kalman Filter (KF) has been proposed in recent years to improve the accuracy of harmonic detection in time-varying conditions. KF give an optimal estimate of basic component of the signals with random noises via recursive solution structure [24]. The Kalman filter is a set of mathematical equations that provides an efficient computational (recursive) means to estimate the state of a process, in a way that minimizes the mean of the squared error. The KF can be used as an estimator to find the amplitude, frequency and phase angle of the power system signals[25-29]. KF -based algorithms have the ability to identify and track time varying harmonic components. In addition to harmonic amplitude and phase angle identification, a modified KF technique that can track the system frequency, has also been proposed [26] - [27].

Kalman filtering has been recognized as powerful state estimation technique [24-30]. It is a model based on the optimal estimator with minimum error covariance. Given the observed data, a KF is described by a set of dynamic process (i.e., state space model) equation and a set of measurement (i.e., observation) equation as follow:

$$\mathbf{x}_k = \Phi_k \mathbf{x}_{k-1} + \mathbf{W}_{k-1} \tag{8}$$

$$Z_k = \mathbf{H}_k \mathbf{x}_k + V_k \tag{9}$$

where

k - present time step,

$\mathbf{x}_k$ is the state vector at present time step k ,

Φ_k is the state transition matrix,

$\mathbf{W}_k$ represents the discrete time variation of the state variables due to an input noise sequence

Z_k is a scalar input signal measurement at time step k ,

$\mathbf{H}_k$ is the observation matrix,

v_k is a scalar noise signal,

The model structure required by KF is flexible to allow the measured signal to be represented in many ways based on different assumptions. The detail of the sequential recursive computational processing in KF can be found in [27].

In order to obtain the good performance of KF, it is necessary to know both the dynamic process and the measurement model. In the power system, the measured signal can be expressed by the sum of sinusoidal waveforms and the noise. If $\mathbf{W}_k$ and v_k are white Gaussian noise with zero mean value and are uncorrelated, then the KF provides an optimal estimate of the real-time signal parameters. Therefore $\mathbf{W}_k$ and v_k can express as follow:

The average of $\quad w = E[w] = 0 \qquad (10)$

The variance of $w = E\left[w_k w_j\right] = \begin{cases} \mathbf{Q}, k = j \\ 0 \quad , k \neq j \end{cases} \qquad (11)$

The average of $v = E[v] = 0 \qquad (12)$

The variance of $v = E\left[v_k v_j\right] = \begin{cases} R, k = j \\ 0 \quad , k \neq j \end{cases} \qquad (13)$

The covariance of w and v is $E\left[w_k v_j\right] = 0 \qquad (14)$

To update the estimates $\mathbf{x}_k$ as follows

$$\hat{\mathbf{x}}_k = \hat{\mathbf{x}}_k^- + \mathbf{K}_k \left(z_k - \mathbf{H}_k \hat{\mathbf{x}}^- \right) \tag{15}$$

Where

$\mathbf{K}_k$ is the Kalman gain

$\hat{\mathbf{x}}_k$ is the state estimation

$\hat{\mathbf{x}}_k^-$ is the state estimation before measuring signal, (i.e. $\hat{\mathbf{x}}_k^- = \mathbf{\Phi}\hat{\mathbf{x}}_{k-1}$)

Therefore, to calculate the Kalman gain $\mathbf{K}_k$ as follow,

$$\mathbf{K}_k = \mathbf{P}_k^- \mathbf{H}^T [\mathbf{H}\mathbf{P}_k^- \mathbf{H}^T + \mathbf{R}]^{-1} \tag{16}$$

where $\mathbf{R}$ is the process noise covariance matrix

$\mathbf{P}_k^-$ is the error covariance matrix

The estimation of the error process covariance in the next time step $k-1$ ($\mathbf{P}_{k-1}$) can be obtained by the following equation:

$$\mathbf{P}_k^- = \mathbf{\Phi}\mathbf{P}_{k-1}\mathbf{\Phi}^T + \mathbf{Q} \tag{17}$$

$\mathbf{Q}$ is the process noise covariance matrix

$\mathbf{P}_{k-1}$ is the error covariance matrix in the next step time $k-1$

The error covariance can be update according to:

$$\mathbf{P}_k = \mathbf{P}_k^- - \mathbf{K}_k \mathbf{H}\mathbf{P}_k^- \tag{18}$$

The harmonic components of the signal can be identified in state variables and measurement frame. Here, the state variables frame is utilized by applying the following conditions to the KF algorithm [30].

$$\begin{bmatrix} x_1 \\ x_2 \\ \cdots \\ x_{2n-1} \\ x_{2n} \end{bmatrix}_{k+1} = \begin{bmatrix} \mathbf{M}_1 & \cdots & 0 \\ \cdots & \cdots & \cdots \\ 0 & \cdots & \mathbf{M}_n \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_{2n-1} \\ x_{2n} \end{bmatrix}_k + \mathbf{w}_k, \quad (19)$$

where $\mathbf{M}_1 ... \mathbf{M}_n$ is the sub-matrix relating the state variables show below as:

$$\mathbf{M}_h = \begin{bmatrix} \cos(h\omega\Delta t) & \sin(h\omega\Delta t) \\ -\sin(h\omega\Delta t) & \cos(h\omega\Delta t) \end{bmatrix} \quad (20)$$

The measurement equation can be then expressed as:

$$z_k = \begin{bmatrix} 1 & 0 & \cdots & 1 & 0 \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \\ \cdots \\ x_{2n-1} \\ x_{2n} \end{bmatrix}_k + v_k \quad (21)$$

The state variables $\mathbf{x}_n$ represent the in-phase and in-quadrature components of the harmonics. Therefore, the magnitude and phase angle of each harmonic component can be obtained as follows.

$$V_h = \sqrt{x_{(2h-1)}^2 + x_{(2h)}^2} \quad (22)$$

$$\theta_h = \tan^{-1}\left(\frac{x_{(2h-1)}}{x_{(2h)}}\right) - h\omega t_k \quad (23)$$

The block diagram of the above described KF algorithm for harmonic detection is illustrated in Fig.1.

The main advantage of the KF-based algorithm is its ability to track harmonics in highly noise polluted power system signals. This method can also detect time-varying harmonics. However, the potential of the KF as a tool for harmonic analysis practically has been limited by implementation difficulties.

The response of the filter is governed by the noise covariance matrices $\mathbf{Q}$ and R, which act as "*tunning*" parameters for the estimation, balancing between accuracy, speed of tracking and filter divergence. In practice, choosing appropriate parameters for desired filter operation can be an arduous task, limiting the success of the application. Also, the performance of KF approach is the relatively complex mathematics requiring extensive calculations.

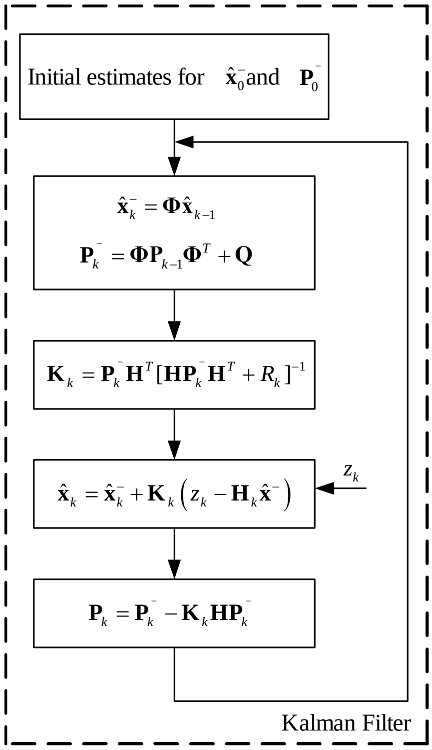

Fig.1 *Standard KF harmonic extraction algorithm*

2.5 Fuzzy Logic Based Adaptive Kalman Filter

The performance of KF is usually affected by the mathematical model used in the solution procedure. Fig.1 shows that the standard KF algorithm procedure requires a significantly great number of computational steps. The computation cost of standard KF can be reduced by estimating the Kalman gain in advanced as below:

$$\mathbf{K}_k = a_k\mathbf{H}^T \quad (24)$$

To overcome implementation limitations in the KF methods, as may be seen, a_k is a multiplier of Kalman gain which (i.e. constant) and can be pre-calculated which depends on parameter matrices $\mathbf{Q}$ and R. The standard Kalman filter algorithm can be modified in order to reduce computational step as shows in Fig.2.

As the accuracy and the time response of the KF is governed by the estimated error covariance matrix R . Therefore, [31] proposed the Fuzzy logic based adaptive the matrix R referred to R_k that corresponds to noise of input signal.

Considering the state variables of standard KF as the following:

$$r_k = z_k - \mathbf{H}_k \hat{\mathbf{x}}^- \tag{25}$$

From (25), the residual error r_k can be obtained. Apply this into (26) to obtains variance, V_k :

$$V_k = \frac{1}{N}\sum_{i=i_0}^{k} r_i^2 \tag{26}$$

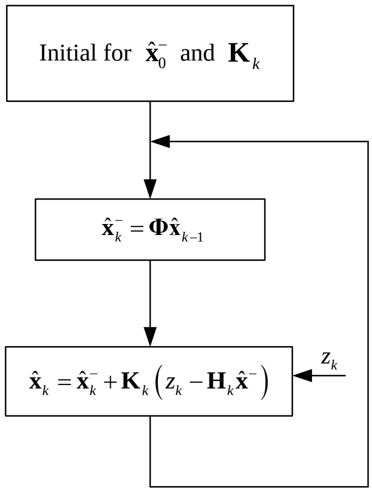

Fig. 2 *Reduction computation step of Standard KF.*

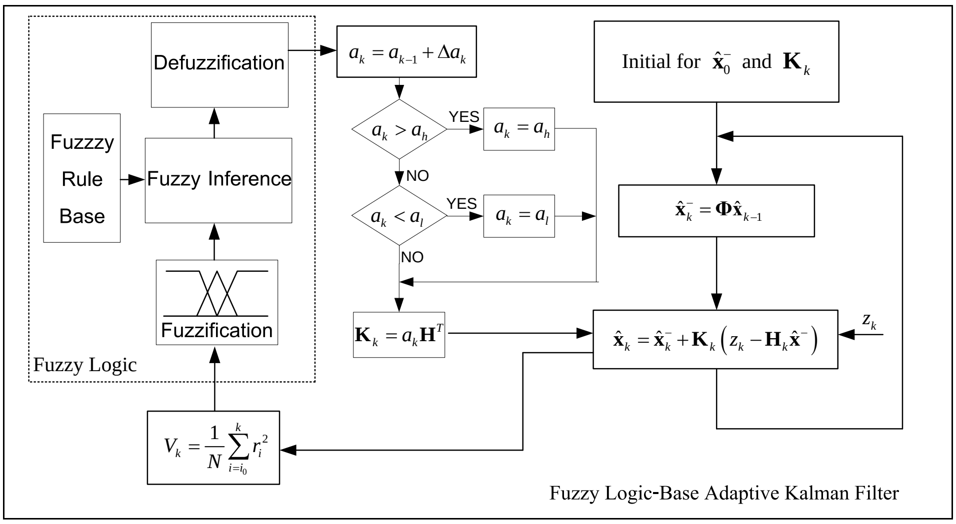

Fig. 3 *Fuzzy Logic-Base adaptive Kalman Filter algorithm*

The condition to tune Kalman Filter gain $\mathbf{K}_k$ have the following:

If $\mathbf{K}_k$ is large , then V_k is large and provides fast response.

If $\mathbf{K}_k$ is small then V_k is small and becomes less sensitive to noise.

The adjustment of $\mathbf{K}_k$ can be done with adjustment to a_k . Which the each calculation can be increase or decrease the value of a_k as:

$$a_k = a_{k-1} + \Delta a_k \tag{27}$$

As you may see, Δa_k is tuning factor, that can be add or subtract from a_k that is step time calculation from the Fuzzy logic output, where a_{k-1} is a_k at $k-1$ step time.

According to the calculation above, it is clear that the implementation will require less calculation burden compare to standard Kalman filter and wavelet transform. It also does not have the drawbacks of DFT and least square method in presence of noise.

Therefore, a well-designed filter for elimination of the effect of covariance matrix selection in the KF can be briefly described as Fig.3 [31].

In [31] studies show that the proposed signal processing system offers many advantages such as simple structure, high estimation accuracy, higher noise immunity and less sensitivity to marginal fundamental frequency changes compared to that of the basic DFT method. The required detection time is only half a cycle of fundamental. Compare to this, the standard Kalman filter may have 50 times slower response for highly polluted signal.

3 CONCLUSTION

A review of the most significant real time processing algorithm of harmonic detection in Power Systems has been presented. This classification may help researchers and engineers to find suitable techniques for their applications in the power system area.

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