Global Journal of researches in engineering
Electrical and electronics engineering
Volume 12 Issue 1 Version 1.0 January 2012
Type: Double Blind Peer Reviewed International Research Journal
Publisher: Global Journals Inc. (USA)
Online ISSN: 2249-4596 & Print ISSN: 0975-5861

Linear Kalman Filter Algorithm with Clarke Transformation for Power System Frequency Estimation

By Soliman Abdel-Hady, A.M. Al-Kandari, R.A.Alammari
Misr University for Science and Technology

Abstract - This paper presents a new application for linear Kalman filter algorithm for power system frequency estimation. The filter uses the digitized samples of the three-phase voltages or current waveform signals. These three phases are transformed into two αβ-phases, using the well-known αβ-transformation matrix (Clarke Transformation). Having obtained the signal of the two new phases, a complex phasor is constructed using the new two-phase voltages. Kalman filter is then applied to extract the frequency and phase angle of the fundamental component of the complex phasor. The proposed algorithm is tested on simulated and actual recorded data at different conditions for the three phase signals; noise free and balanced three phase signals, unbalanced three-phase system and harmonics contaminated three phase signals. It has been shown that the proposed filters with the proposed transformation are succeeded to estimate the signal frequency at off nominal, near nominal and at nominal frequency.

Keywords : Linear Kalman filter, frequency estimation, αβ-transformation.

GJRE-F Classification: FOR Code: 090601

LINEAR KALMAN FILTER ALGORITHM WITH CLARKE TRANSFORMATION FOR POWER SYSTEM FREQUENCY ESTIMATION

Strictly as per the compliance and regulations of:

Linear Kalman Filter Algorithm with Clarke Transformation for Power System Frequency Estimation

Soliman Abdel-Hady [α], A.M. Al-Kandari [σ], R.A.Alammari [ρ]

Abstract - This paper presents a new application for linear Kalman filter algorithm for power system frequency estimation. The filter uses the digitized samples of the three-phase voltages or current waveform signals. These three phases are transformed into two $\alpha\beta$-phases, using the well-known $\alpha\beta$-transformation matrix (Clarke Transformation). Having obtained the signal of the two new phases, a complex phasor is constructed using the new two-phase voltages. Kalman filter is then applied to extract the frequency and phase angle of the fundamental component of the complex phasor. The proposed algorithm is tested on simulated and actual recorded data at different conditions for the three phase signals; noise free and balanced three phase signals, unbalanced three-phase system and harmonics contaminated three phase signals. It has been shown that the proposed filters with the proposed transformation are succeeded to estimate the signal frequency at off nominal, near nominal and at nominal frequency.

Keywords : Linear Kalman filter, frequency estimation, $\alpha\beta$-transformation.

I. Introduction

An accurate measurement for power system frequency as well as the voltage amplitude is essential in power system digital control and protection. The deviation of the system frequency from nominal frequency is an indication for the system unbalance. If the system frequency is below the nominal frequency, the load-shedding relay must shed the unimportant loads on the system to retain the system frequency to its nominal value. If the system frequency is above nominal frequency the generator governors must decrease the generator-input power to retain the frequency to its nominal value. As such, the power system frequency can be considered as a measure to the power balance in the system.

Many techniques have been proposed over the last twenty-five years for measuring digitally the power system frequency. Recently Ref. [1] presents an approach for measuring the system frequency using the complex extended Kalman filter and the well-known $\alpha\beta$-transformation, by this transformation a nonlinear state space formulation is obtained for extended Kalman filter.

Author [α] : Misr University for Science and Technology, Electrical Power and Machines Department, 6th of October City, Giza, EGYPT.
E-mail : sasoliman@must.edu.eg
Author [σ] : College of Technological Studies, Electrical Engineering Department, Shewiekh, Kuwait, Kuwait. E-mail : kandari1@yahoo.com
Author [ρ] : Qatar University, Electrical Engineering Dept., Doha, Qatar.

The orthogonal FIR digital filter with least error squares algorithm are applied in Ref.[2] for measuring the operating frequency of a power system. The proposed algorithm in this reference was able to produce a correct and noise free estimate for near nominal, nominal and off-nominal frequency in about 25 ms.

The same algorithm with some modifications to reduce the computing time and improve the resolution is implemented in Ref. [3]. This algorithm has beneficial features including fixed sampling rate, fixed data window size and easy implementation.

Refs.. [4-6] present two orthogonal filters for power system frequency estimation. The essential property of the algorithms proposed in these references is that their outstanding immunity to both signal orthogonal component magnitudes and FIR filter gains variations. All the algorithms use the per-phase digitized voltage samples.

Proney's estimation is applied in Ref. [7] for measuring the system frequency together with the discrete Fourier transform (DFT) with a variable data widow to filter out the noise and harmonics associated with the signal. The static state estimation algorithms have been used in power systems measurements. The static least error squares algorithm is implemented in Refs.{15] and [16], while the dynamic estimation algorithms have been implemented in Ref.[17], these including linear and nonlinear Kalman filtering algorithms. Most but all of these algorithms use the per-phase digitized samples to estimate the system frequency. Each algorithm implemented up till now has its own figure of merit, and is only suitable for the system it works with. Also, all the available algorithms are either tested off-line or on-line, and they produce, in most cases, good estimates for the purposes they are designed for.

The design of an extended complex Kalman filter for the measurement of power system frequency has been presented in Ref. [1]. A complex model has been developed to track a distorted signal that belongs to a power system. The model inherently takes care of the frequency measurement along with the amplitude and phase of the signals. The proposed algorithm is suitable for real-time applications where the measurement noise and other disturbances are high. Ref. [2] proposes a precise digital algorithm based on Discrete Fourier Transforms (DFT) to estimate the frequency of a sinusoid with harmonics in real-time. This algorithm is called the Smart Discrete Fourier Transforms (SDFT) smartly avoids the errors that arise when frequency deviates from the nominal frequency, and keeps all the advantages of the DFT e.g., immune to harmonics and the recursive computing can be used in SDFT.

Ref. [3] presents an analysis of methods for tracking the fundamental power frequency to see if they have the performance necessary to cope with the requirements of future protection and control equipment and are robust enough to cope with the more demanding nature of modern power system conditions.

An approach based on extended Kalman filter (EKF) for the measurement of power system frequency has been presented in Ref. [4]. The design principles and the validity of the model have been outlined in this reference. The performance of the proposed filter has been compared with some of the existing methods for estimating the frequency of a signal under noisy conditions. The feasibility of the proposed filter has been tested in the laboratory under worst-case measurement and network conditions, which might occur in a typical power system. Also, the proof of the stability for the proposed filter has been discussed for a single sinusoid. It has been found that the proposed algorithm is suitable for real-time applications especially when the frequency changes are abrupt and the signal is corrupted with noise and other disturbances due to harmonics.

Ref. [5] presents a unified power signal processor (PSP) for use in various applications in power systems. The introduced PSP is capable of providing a large number of signals and pieces of information which are frequently required for control, protection, status evaluation, and power quality monitoring of power systems. The PSP receives a set of locally measured three phase voltage and current signals and provides their fundamental components, amplitudes, phase angles, frequency, harmonics, instantaneous and stationary symmetrical components, active and reactive currents and powers, power factor, and the total harmonic distortion. Simplicity and integrity of its structure as well as its robustness with respect to internal parameters and external disturbances and noise render the proposed scheme very attractive for practical implementations.

An improved recursive Newton-type algorithm suitable for various measurement applications in electric power systems is presented in Ref. [6]. It is used for the power system frequency and spectra estimation. The recursive algorithm form is improved with a strategy of sequential tuning of the forgetting factor. By this, the proposed algorithm convergence and accuracy are significantly improved. A method for estimation of power frequency and its rate of change is presented in Ref.[7]. Unlike conventional methods which are based on the concept of linearization, the proposed scheme accommodates the inherent nonlinearity of the frequency estimation problem. This makes the method capable of providing a fast and accurate estimate of the frequency when its deviation from the nominal value is incremental or large. The estimator is based on a newly developed quadrature phase-locked loop concept. The method is highly immune to noise and distortions. The estimator performance is robust with respect to the parameters of its structure. Structural simplicity and performance robustness are other salient features of the method.

Ref. [8] introduces a phase-locked loop (PLL) system. The proposed system provides the dominant frequency component of the input signal and estimates its frequency. The mechanism of the proposed PLL is based on estimating in-phase and quadrature-phase amplitudes of the desired signal. The PLL provides a superior performance for power system applications. Advantages of the proposed PLL over the conventional PLLs are its capability of providing the fundamental component of the input signal which is not only locked in phase but also in amplitude to the actual signal while providing an estimate of its frequency. An approach to the design of a digital algorithm for local system frequency estimation is presented in Ref. [9]. The algorithm is derived using the maximum likelihood method. One sinusoidal voltage model was assumed. FIR digital filters are used to minimize the noise effect and to eliminate the presence of the harmonics effect. This technique provides accurate estimates with error in the range of 0.005 Hz in about 25 ms and requires modest computations.

A numerical differentiation-based algorithm for high-accuracy, wide-range frequency estimation of power systems is proposed in Ref. [10]. The signal includes up to 31st-order harmonic components. Using the central numerical differentiation of multi-points on the basis of Lagrange interpolation, the estimated frequency of nonsinusoidal signals of power systems can be measured at an accuracy of 99.999% over a wide range of from 10 to 100 Hz with fundamental amplitude varying from 100 to 300 V and with a fundamental phase angle varying from 0 to 360. The proposed algorithm needs at most two cycles or 40 ms for estimation of frequency over 40 to 60 Hz and needs at most three cycles or 60 ms over 10 to 40 Hz or 60 to 100 Hz.

The design of an extended complex Kalman filter for the measurement of power system frequency has been presented in Ref. [17]. A complex model has been developed to track a distorted signal that belongs to a power system. The model inherently takes care of the frequency measurement along with the amplitude and phase of the signals. The proposed algorithm is suitable for real-time applications where the measurement noise and other disturbances are high. Ref. [18] proposes a precise digital algorithm based on Discrete Fourier Transforms (DFT) to estimate the frequency of a sinusoid with harmonics in real-time. This algorithm is called the Smart Discrete Fourier Transforms (SDFT) smartly avoids the errors that arise when frequency deviates from the nominal frequency, and keeps all the advantages of the DFT e.g., immune to harmonics and the recursive computing can be used in SDFT.

Ref. [19] presents an analysis of methods for tracking the fundamental power frequency to see if they have the performance necessary to cope with the requirements of future protection and control equipment and are robust enough to cope with the more demanding nature of modern power system conditions.

An approach based on extended Kalman filter (EKF) for the measurement of power system frequency has been presented in Ref. [20]. The design principles and the validity of the model have been outlined in this reference. The performance of the proposed filter has been compared with some of the existing methods for estimating the frequency of a signal under noisy conditions. The feasibility of the proposed filter has been tested in the laboratory under worst-case measurement and network conditions, which might occur in a typical power system. Also, the proof of the stability for the proposed filter has been discussed for a single sinusoid. It has been found that the proposed algorithm is suitable for real-time applications especially when the frequency changes are abrupt and the signal is corrupted with noise and other disturbances due to harmonics.

Ref. [21] presents a unified power signal processor (PSP) for use in various applications in power systems. The introduced PSP is capable of providing a large number of signals and pieces of information which are frequently required for control, protection, status evaluation, and power quality monitoring of power systems. The PSP receives a set of locally measured three phase voltage and current signals and provides their fundamental components, amplitudes, phase angles, frequency, harmonics, instantaneous and stationary symmetrical components, active and reactive currents and powers, power factor, and the total harmonic distortion. Simplicity and integrity of its structure as well as its robustness with respect to internal parameters and external disturbances and noise render the proposed scheme very attractive for practical implementations.

An improved recursive Newton-type algorithm suitable for various measurement applications in electric power systems is presented in Ref. [22]. It is used for the power system frequency and spectra estimation. The recursive algorithm form is improved with a strategy of sequential tuning of the forgetting factor. By this, the proposed algorithm convergence and accuracy are significantly improved. A method for estimation of power frequency and its rate of change is presented in Ref.[23]. Unlike conventional methods which are based on the concept of linearization, the proposed scheme accommodates the inherent nonlinearity of the frequency estimation problem. This makes the method capable of providing a fast and accurate estimate of the frequency when its deviation from the nominal value is incremental or large. The estimator is based on a newly developed quadrature phase-locked loop concept. The method is highly immune to noise and distortions. The estimator performance is robust with respect to the parameters of its structure. Structural simplicity and performance robustness are other salient features of the method.

Ref. [24] introduces a phase-locked loop (PLL) system. The proposed system provides the dominant frequency component of the input signal and estimates its frequency. The mechanism of the proposed PLL is based on estimating in-phase and quadrature-phase amplitudes of the desired signal. The PLL provides a superior performance for power system applications. Advantages of the proposed PLL over the conventional PLLs are its capability of providing the fundamental component of the input signal which is not only locked in phase but also in amplitude to the actual signal while providing an estimate of its frequency. An approach to the design of a digital algorithm for local system frequency estimation is presented in Ref. [25]. The algorithm is derived using the maximum likelihood method. One sinusoidal voltage model was assumed. FIR digital filters are used to minimize the noise effect and to eliminate the presence of the harmonics effect. This technique provides accurate estimates with error in the range of 0.005 Hz in about 25 ms and requires modest computations.

A numerical differentiation-based algorithm for high-accuracy, wide-range frequency estimation of power systems is proposed in Ref. [26]. The signal includes up to 31st-order harmonic components. Using the central numerical differentiation of multi-points on the basis of Lagrange interpolation, the estimated frequency of non-sinusoidal signals of power systems can be measured at an accuracy of 99.999% over a wide range of from 10 to 100 Hz with fundamental amplitude varying from 100 to 300 V and with a fundamental phase angle varying from 0 to 360. The proposed algorithm needs at most two cycles or 40 ms for estimation of frequency over 40 to 60 Hz and needs at most three cycles or 60 ms over 10 to 40 Hz or 60 to 100 Hz.

This paper presents a new application for linear Kalman filter algorithm for power system frequency estimation. The filter uses the digitized samples of the three-phase voltages or current waveform signals. These three phases are transformed into two $\alpha\beta\gamma$ -phases, using the well-known $\alpha\beta\gamma$ -transformation matrix. Having obtained the signal of the two new phases, a complex phasor is constructed using the new

Global Journal of Researches in Engineering (F) Volume XII Issue I Version I January 2012

two-phase voltages. Kalman filter is then applied to extract the frequency and phase angle of the fundamental component of the complex phasor. The proposed algorithm is tested on simulated data at different conditions for the three phase signals; noise free and balanced three phase signals, unbalanced three-phase system and harmonics contaminated three phase signals. It has been shown that the proposed filters with the proposed transformation are succeeded to estimate the signal frequency at off-nominal, near nominal and at nominal frequency. Also, the proposed algorithm is tested on actual recorded data generated from EMTP.

II. Modeling the Voltage Signals [1]

The three phase voltages of the power systems can be written at any sample k, k=1,..,..,m, m is the total number of samples available, as:

$$
\begin{aligned}
v_a(k) &= V_{ma}\sin(\omega k\Delta T+\phi_a) + \zeta_a(k) \\
v_b(k) &= V_{mb}\sin(\omega k\Delta T+\phi_b -120) + \zeta_b(k) \\
v_c(k) &= V_{mc}\sin(\omega k\Delta T+\phi_c +120) + \zeta_c(k)
\end{aligned} \tag{1}
$$

V_m	is the signal amplitude
ω	is the signal nominal frequency
ΔT	is the sampling time= $\frac{1}{F_s}$, F_s is the sampling frequency
k	is the sampling step; $k=1,..,m$
ϕ	is the signal phase angle
$\zeta_a(k),\zeta_b(k),\zeta_c(k)$	are the noise terms which may contain harmonics

The well-known αβ - transformation (Clarke Transformation) for the three-phase signal is given as:

$$
\begin{bmatrix} v_\alpha(k) \\ v_\beta(k) \end{bmatrix} = \sqrt{\frac{2}{3}}\begin{bmatrix} 1 & -0.5 & -0.5 \\ 0 & \frac{\sqrt{3}}{2} & \frac{-\sqrt{3}}{2} \end{bmatrix}\begin{bmatrix} v_a(k) \\ v_b(k) \\ v_c(k) \end{bmatrix} \tag{2}
$$

The above equation can be rewritten as:

$$
v_\alpha(k) = \sqrt{\frac{2}{3}}\left[v_a(k) - 0.5v_b(k) - 0.5v_c(k)\right] \tag{3}
$$

$$
v_\beta(k) = \sqrt{\frac{2}{3}}\left[\frac{\sqrt{3}}{2}v_b(k) - \frac{\sqrt{3}}{2}v_c(k)\right] \tag{4}
$$

For m samples of the three-phase signal, then m samples for $v\alpha(k), v\beta(k)$ are obtained. By using this transformation, the harmonics of order three and their multiple are suppressed. Equation (3) and (4) give the transformed voltage $v_\alpha(k)$ and $v_\beta(k)$ at any sample k. The complex voltage formed from these two voltages which having the same frequency ω and phase angle Φ as the original three phases is:

$$
\begin{aligned}
v(k) &= v_\alpha(k) + jv_\beta(k) \\
&= V(k)e^{j(\omega k\Delta T+\phi)}
\end{aligned} \tag{5}
$$

Where the amplitude $V(k)$ of the complex signal is calculated at any sample k as:

$$
V(k) = \left[v_\alpha^2(k) + v_\beta^2(k)\right]^{\frac{1}{2}} \tag{6}
$$

and its rms value is

$$
V^2(k) = \frac{1}{m}\sum_{k=1}^{m}\left[v_\alpha^2(k) + v_\beta^2(k)\right] \tag{7}
$$

Equation (6) gives the value of complex voltage at any instant k, while e1uation (7) can be used to calculate the rms value of the complex voltage signal, while.

The phase angle of this transformed voltage is

$$
\theta(k) = \tan^{-1}\frac{v_\beta(k)}{v_\alpha(k)} \tag{8}
$$

Comparing equation (8) with equation (5), we can calculate the frequency of the voltage signals as;

$$
\theta(k) = 2\pi kTf(k) + \phi(k) \tag{9}
$$

Now, m samples are available for the phase angle of the transformed voltage $\theta(k)$. Define

$$
\begin{aligned}
&X_1(k) = f(k); \\
&X_2(k) = \varphi(k) \\
&and \\
&h_1(k) = 2\pi k\Delta T \\
&h_2(k) = 1
\end{aligned} \tag{10}
$$

Then

$$
\theta(k) = \begin{bmatrix} h_1(k) & h_2(k) \end{bmatrix}\begin{bmatrix} X_1(k) \\ X_2(k) \end{bmatrix} \tag{11}
$$

For m samples available for $\theta(k)$, equation (11) can be written as:

$$\underline{Z}(k) = H(k)\underline{X}(k) + \underline{\zeta}(k) \qquad (12)$$

Where $\underline{Z}(k)$ is $mx1$ vector of phase angle $\theta(k)$ at any instant k, $H(k)$ is $mx2$ measurement matrix whose elements depend on the sampling frequency and can be calculated off-line for a pre-specified sampling frequency. $\underline{X}(k)$ is $2x1$ state vector to be estimated and $\underline{\zeta}(k)$ is $mx1$ noise vector to be minimized.

The state space equation for the two-state model can be written as:

$$\underline{X}(k+1) = \phi(k)\,\underline{X}(k) + \underline{v}(k) \qquad (13)$$

Where $\phi(k)$ is a $2x2$-identity matrix, and is called the state transition matrix, $\underline{v}(k)$ a is $2x1$-noise vector associated with the states when they are moving from k to k+1 sample. Now, equation (12) and (13) are suitable for Kalman filter implementation. Before going further let us interpret equations (6) and (8). Equation 6 gives the amplitude of the complex phasor formed from the two orthogonal voltages V_α and V_β. If the three phase voltages are balanced and are noise and harmonics free signals, this amplitude will be constant amplitude. Also, equation 8 gives the phase angle of this complex phasor. For the same balanced conditions, the rate of change of this phase angle, which is the speed of rotation of the complex phasor, will be constant and equals to the nominal angular velocity of the original three-phase signals. As such, the frequency of the will be constant during the data window size, and the locus of the complex phasor is a circle as shown in Figure1, by the dashed curve. However, if the three phases are unbalanced, then the locus of the complex Phasor will be just a closed curve as shown in Figure1. By plotting the complex Phasor the following advantages could be obtained

- It provides information about the system balance.
- It provides information about noise and harmonics contamination to the three-phase system.
- It provides information about the system frequency.
- Also, due to the nature of the $\alpha\beta$ -transformation, harmonics of order three and its multiple are suppressed

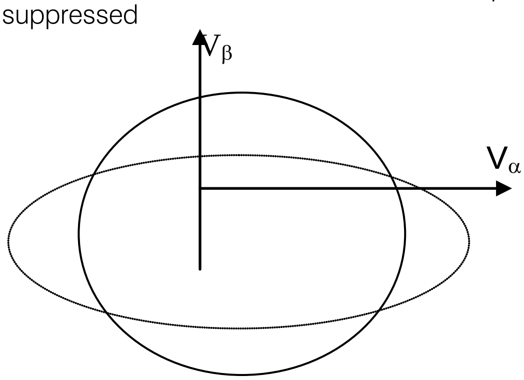

Figure 1 : The locus of the complex phasor

III. Testing the Algorithm Using Simulated Data

a) A balanced and noise free system

The proposed algorithm is tested using samples of a balanced and noise free three-phase system. The voltage amplitude is 1.0 p.u and the phase angle is -2.1 radian. The signal frequency is varied from 45-51 Hz. The three phase voltage signal are sampled at 3000 Hz (60 sample per cycle) and a data window size of 48 sample (0.8 cycle) is used in the estimation process. Figures 2 - 4 give the estimated frequency, phase angle and the amplitude of the complex phasor. Examining these figures reveals the following remarks.

- The proposed technique produces accurate estimates for the system frequency ranging from 45 to 51 Hz, i.e. accurate estimates are obtained for off-nominal, near nominal and at nominal frequency.
- The estimated phase angle is accurate and constant independent of the signal frequency.
- The amplitude of the complex phasor is also accurate and equals to the actual value at all the assumed frequency range.

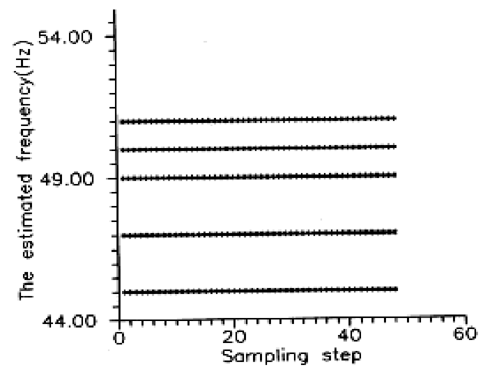

Figure 2 : Estimated frequencies

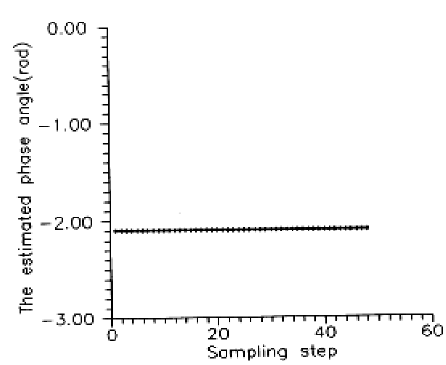

Figure 3 : The estimated phase angle

b) Unbalanced Three-phase Voltage Signal

The proposed algorithm is tested when there is unbalance between phases. Here, we use V_a = 1.0 p.u, V_b= 1.1 p.u and V_c=0.9 p.u, while the phase angle is -2.1 radian. The same sampling frequency, 3000 Hz, and number of samples 48 samples are used. Figures 5-7 give the results obtained. Examining these figure reveals the following remarks, solid lines are the actual values,

- Good estimates are obtained when the signal frequency is 45, 48, 50, 51 Hz. Although there is about 0.5 Hz errors for the signal frequency of 51 Hz, but it is acceptable error for such unbalanced system.

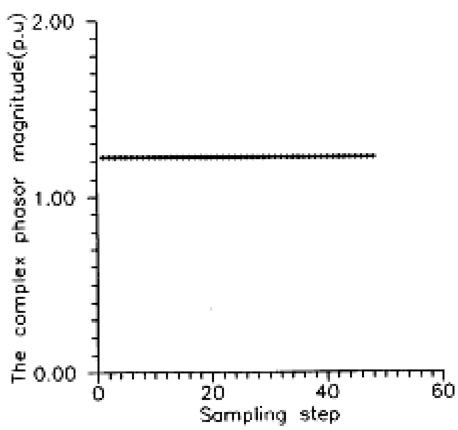

Figure 4 : Amplitude of the complex phasor

- The phase angle estimate is a good quality estimate at all the four frequencies of the signal, and there is a maximum error of about 0.1 radian (4.76 %), which is acceptable error in this unbalanced case.
- Figure 7 gives the amplitude of the complex phasor when the signal frequency is 50 Hz. It can be noticed that the amplitude is a time varying (figure 1 solid curve), but the rms. value over the estimation period gives almost the actual value of the amplitude.

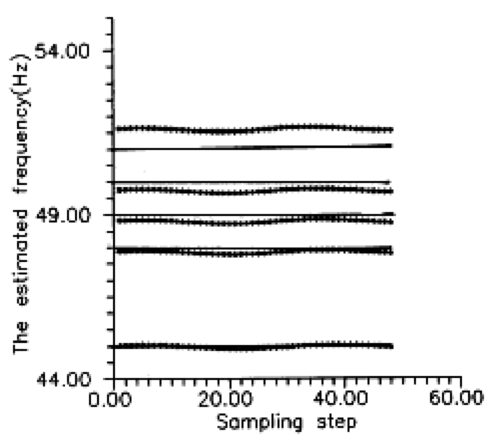

Figure 5 : Estimated frequencies

As such, we can conclude that the system unbalance and the degree of this unbalance have a great effect on the system estimates. Actually, that is true, if one examines Figure 1 carefully, he can notice that the amplitude as well as the argument of the complex phasor will be changed from instant to instant, if the phasor rotates on the solid curve.

IV. Harmonics Pollution

In this test, the proposed algorithm is tested with three-phase voltage signals contaminated with harmonics. The signals contain 1 p.u fundamental, 0.12 p.u second harmonic, 0.3 p.u third harmonic and 0.05 p.u fifth harmonic, all are balanced in the three phases. Figure 8 gives the locus of the complex phasor, which is not a circle; this means that this phasor has different amplitudes and arguments from instant to instant. Figure 9-11 give the estimated frequency, phase angle, and voltage amplitude. The sampling frequency used in this test is 3000 Hz and the data window size is 48 samples. Examining these figures reveals the following:

- ♣ The estimated frequency of the fundamental component is affected by the presence of harmonics in the voltage signal.

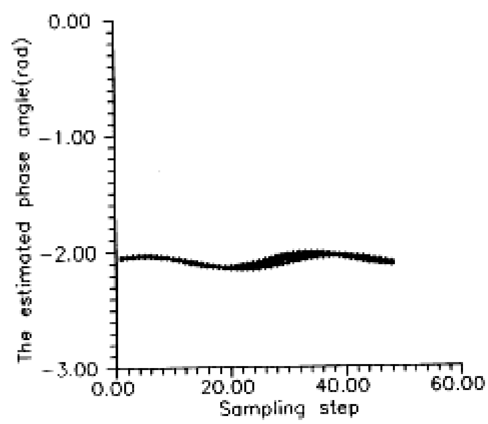

Figure 6 : Estimated phase angle

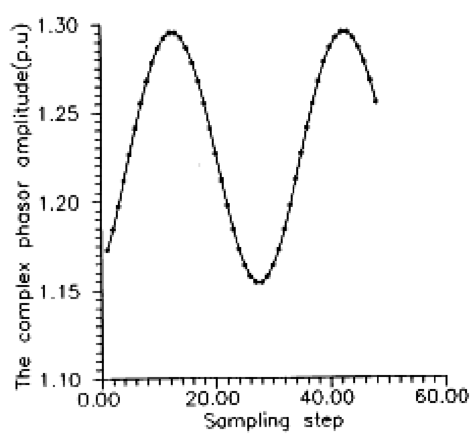

Figure 7 : Estimated complex phasor amplitude

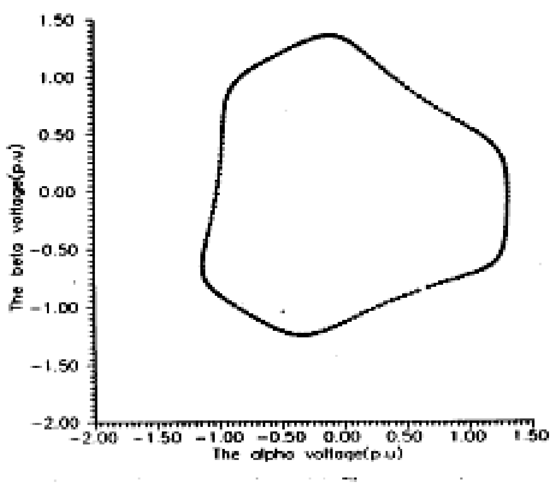

Figure 8 : Locus of complex phasor

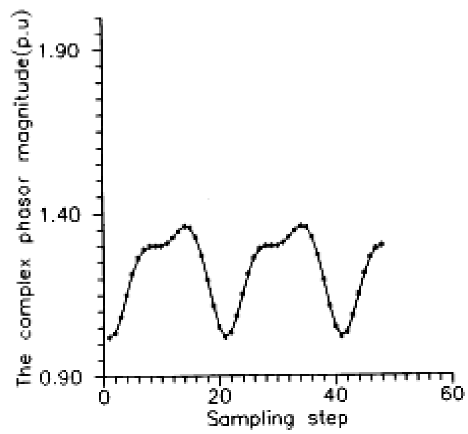

Figure 11 : Estimated complex phasor amplitude

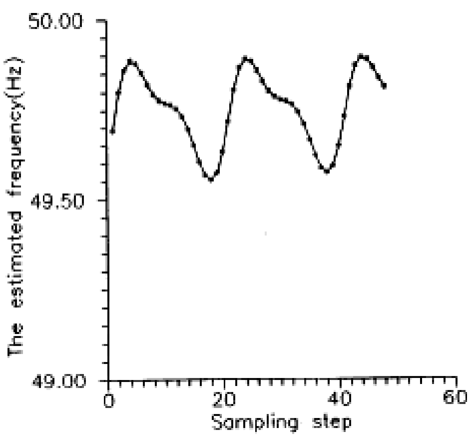

Figure 9 : The estimated frequency

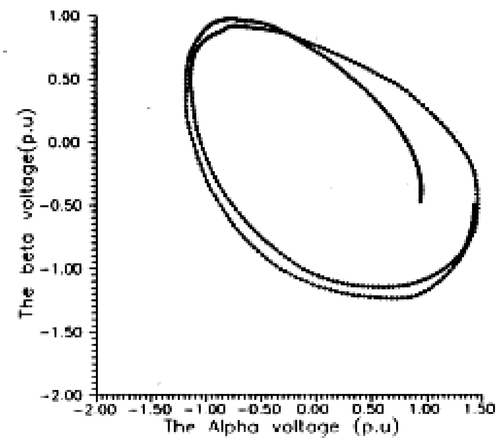

Figure 12 : Locus of the complex phasor

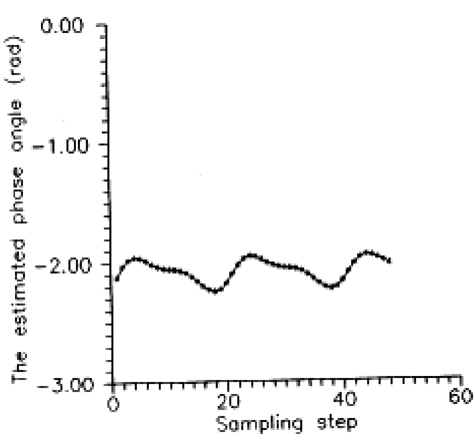

Figure 10 : Estimated phase angle

- ♣ The minimum frequency estimate for the fundamental component is about 49.6 Hz (error = -0.4 Hz, or -0.8 %), while the maximum frequency estimate is 49.9 (error=-0.1 Hz, or -0.2 %), which are considered good estimates for such polluted signals. However, due to the nature of $\alpha\beta$-transformation, the third harmonic has nothing to do with the estimates
- ♣ The average frequency estimate over the data window size used in this test is about 49.75 Hz, which is still a good estimate (error = -0.25 Hz, or -0.5 %)
- ♣ The phase angle estimate can be considered as good as the signals are contaminated with harmonics, and the average value is about -2.05 radian (error = -0.05 radian, or -2.3 %)
- ♣ As we said earlier, the complex phasor amplitude could be considered as a time varying amplitude, with rms value almost equals to the actual value.

V. Testing the Algorithm Using Actual Recorded data

The algorithm has also been tested using actual recorded data for a practical system in operation. The EMTP program is used to compute the three phase voltages at a certain bus on a system during the switching on of 500 kV transmission systems. Figure 12 gives the locus of the complex phasor formed by using the three phase signals and the proposed transformation. It can be noticed that, such a system is unbalanced and highly contaminated with harmonics system. These harmonics are not periodical waveforms. The sampling frequency used in this study is 10 kHz (ΔT = 0.1 ms) and a number of samples of 55 samples is used. Figures 13-15 give the estimated frequency, phase angle and the amplitude of the complex phasor. Examining these figures reveals the following remarks.

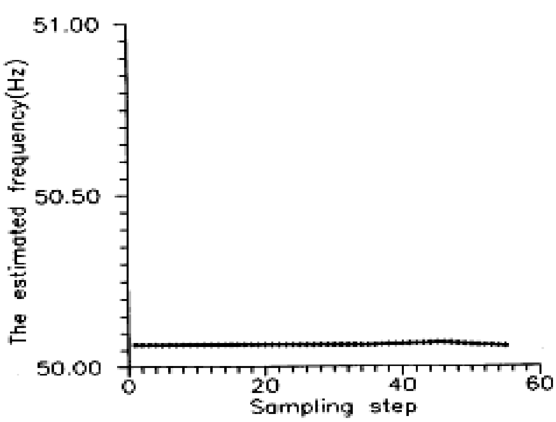

Figure 13 : Estimated frequencies

- The estimated frequency of the fundamental component is about 50.06 Hz, which could be considered as an accurate estimate (error=0.06 Hz, or 0.12%).
- The phase angle estimate can be considered as a good, and is a constant during the data window size used in this analysis.
- Although the amplitude of the complex phasor voltage is constant up to 79 % of the data window size, but its value is about 0.95 p.u. As we said, this value varies from instant to instant and does give induction to the effect of distortion of the three-phase signals amplitude.

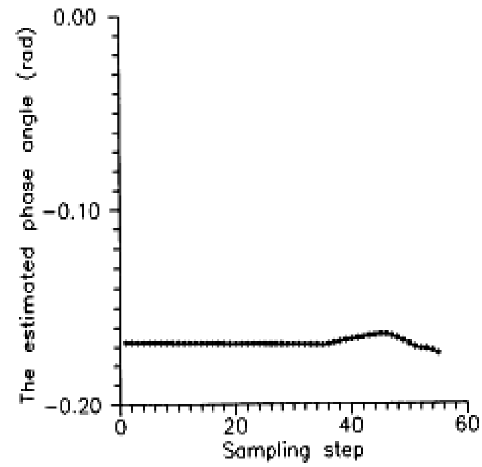

Figure 14 : Estimated phase angle

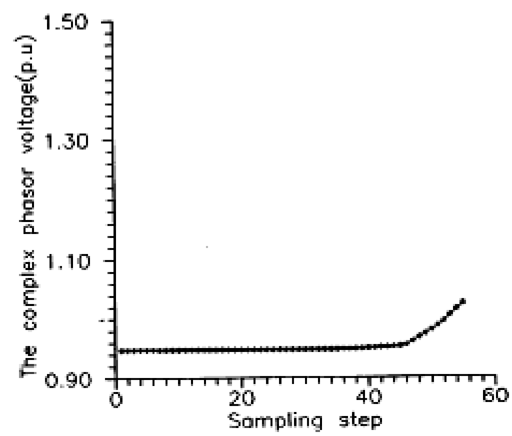

Figure 15 : Estimated complex phasor amplitude

VI. Conclusions

The main contributions of this paper are:

1. A complex phasor is developed which can be used to provide information about the system balance and harmonics contamination.
2. The linear Kalman filter is used to extract the frequency and phase angle of the fundamental component, using the argument of the complex voltage phasor.
3. The three phase voltage signals are used to estimate the system parameters, frequency, phase angle and voltage amplitude, using the well-known$\alpha\beta$-transformation. Unlike the available techniques, which use the per-phase voltage samples to estimate the system parameters?
4. The estimate using this technique is slightly affected by the noise and harmonics contamination.
5. Direct estimates are obtained for the system parameters, without modeling the three phase voltage signals.

References References Referencias

1. P. K. Dash, A. K. Pradhan, and G.Panda, "Frequency Estimation of Distorted Power System Signal Using Extended Complex Kalman Filter" IEEE Trans. on Power Delivery, Vol. 14, No.3, pp.761-766, July 1999.
2. T. S. Sidhu, "accurate Measurement of Power System Frequency Using a Digital Signal Processing" IEEE Trans. On Power Delivery, Vol. 14, No.1, pp.75-81, 1999.
3. T.S. Sidhu, and M. S. Sachdev, "An Accurate Technique for Fast and Accurate Measurement of Power System Frequency", IEEE Trans. on Power Delivery, Vol. 13, No.1, pp.109-115, 1998.
4. Jose A.dela O, H.J.Altave, and I.Diaz, " A New Digital Filter for Phasor Computation, Part I: Theory", IEEE Trans. on Power Systems, Vol.13, No.3, pp.1026-1031, 1998.
5. Jose a. dela O, H. J. Altave, and I. Diaze"A New Digital Filter for Phasor Computation, Part II: Evaluation", IEEE Trans. on Power Systems, Vol.13, No.3, pp.1032-1037, 1998.
6. J. Szafran, and W. Rebizant, "Power System Frequency Estimation", IEE Proc. Gener. Trans. Disturb., Vol.145, No.5, pp.578-582, 1998.
7. T. Lobos and J. Rezmer, "Real-time Determination of Power System Frequency", IEEE Trans. on Inst. and Measurement, Vol.46, No.4, pp.877-881, 1997.
8. P.J.Moore, R.D.Carranza and A.T.Johns, "Model System Test on a New Numeric Method of Power System Frequency Measurement", IEEE Trans. on Power Delivery, Vol.11, No.2, pp.696-701, 1996.
9. P.J.Moore, J.H.Allmeling and A.T.Johns, "Frequency Relaying Based on Instantaneous Frequency Measurement", IEEE Trans. on Power Delivery, Vol.11, No.4, pp.1737-1742, 1996.
10. V.Terzija, M.Djuric and B.Kovacevic, " A New Self-Tuning Algorithm for the Frequency Estimation of Distorted Signals" IEEE Trans. on Power Delivery, Vol.10, No.4 pp.1779-1785, 1995.
11. V.Terzija, M.Djuric and B.Kovacevic, " Voltage Phasor and Local System Frequency Estimation Using Newton-Type Algorithms"" IEEE Trans. on Power Delivery, Vol.14, No.3 pp.1368-1374, 1994.
12. M.M.Begovic, et-al, " Frequency Tracking in Power Network in the Presence of Harmonics", " IEEE Trans. on Power Delivery, Vol.8, No.2 pp.480-486, 1993.
13. M.Kezunovic, et-al, " New Digital Signal Processing Algorithm for Frequency Measurement"," IEEE Trans. on Power Delivery, Vol.7, No.3 pp.1563-1572, 1992.
14. S.A.Soliman, et-al, "Dynamic Tracking of the Steady State Power System Voltage Magnitude and Frequency Using Linear Kalman Filter: A Variable Frequency Model", Electric Machines and Power Systems, Vol. 20, pp.593-611, 1992.
15. S. A. Soliman, et-al ,"Estimating of Steady State Voltage and Frequency of Power Systems from Digitized Bus Voltage Samples", Electric Machines and Power Systems, Vol. 19, pp.555-567, 1991.
16. M. E. El-Hawary, et-al, "A Study of Estimation Techniques for Frequency-Relaying Applications", Can. J. Elect. & Comp. Eng., Vol.21, No.1, pp.9-20, 1996
17. P. K. Dash, R. K. Jena, G. Panda, and A. Routray," An Extended Complex Kalman Filter for Frequency Measurement of Distorted Signals", IEEE TRANSACTIONS ON INSTRUMENTATION AND MEASUREMENT, VOL. 49, NO. 4, AUGUST 2000
18. Jun-Zhe Yang and Chih-Wen Liu," A Precise Calculation of Power System Frequency", IEEE TRANSACTIONS ON POWER DELIVERY, VOL. 16, NO. 3, JULY 2001.
19. D W. P. Thomas, and M. S. Woolfson," Evaluation of Frequency Tracking Methods", IEEE TRANSACTIONS ON POWER DELIVERY, VOL. 16, NO. 3, JULY 2001
20. A. Routray, A. K. Pradhan, and K. P. Rao," A Novel Kalman Filter for Frequency Estimation of Distorted Signals in Power Systems" IEEE TRANSACTIONS ON INSTRUMENTATION AND MEASUREMENT, VOL. 51, NO. 3, JUNE 2002.
21. M. Karimi-Ghartemani and M. R. Iravani," A Signal Processing Module for Power System Applications", IEEE TRANSACTIONS ON POWER DELIVERY, VOL. 18, NO. 4, OCTOBER 2003.
22. Vladimir. V. Terzija," Improved Recursive Newton-Type Algorithm for Frequency and Spectra Estimation in Power Systems" IEEE TRANSACTIONS ON INSTRUMENTATION AND MEASUREMENT, VOL. 52, NO. 5, OCTOBER 2003
23. H. Karimi, M. Karimi-Ghartemani, and M. R. Iravani," Estimation of Frequency and its Rate of Change for Applications in Power Systems" IEEE TRANSACTIONS ON POWER DELIVERY, VOL. 19, NO. 2, APRIL 2004.
24. H. Karimi, M. Karimi-Ghartemani, and M. R. Iravani, " Magnitude/Phase-Locked Loop System Based on Estimation of Frequency and In-Phase/Quadrature-Phase Amplitudes" IEEE TRANSACTIONS ON INDUSTRIAL ELECTRONICS, VOL. 51, NO. 2, APRIL 2004.
25. M. D. Kuˇsljevic," A Simple Recursive Algorithm for Frequency Estimation", IEEE TRANSACTIONS ON INSTRUMENTATION AND MEASUREMENT, VOL. 53, NO. 2, APRIL 2004.
26. Wu Jiekang, Long Jun, and Wang Jixiang," High-Accuracy, Wide-Range Frequency Estimation Methods for Power System Signals under Nonsinusoidal Conditions", IEEE TRANSACTIONS ON POWER DELIVERY, VOL. 20, NO. 1, JANUARY 2005.

Appendix

III. Linear Kalman Filter Algorithm [14]

In the implementation of Kalman filter, the mathematical model for the system under consideration should be in the state form as:

$$\mathbf{x}(k+1) = \phi(k)\,\mathbf{x}(k) + \boldsymbol{\omega}(k)$$

and the observation(measurement) of the process is assumed to occur at discrete points of time in accordance with the relation

$$\mathbf{z}(k) = H(k)\,\mathbf{x}(k) + \mathbf{v}(k)$$

Assume that we have a prior estimate $\mathbf{x}^-(k)$, and its error covariance matrix $P^-(k)$; then the general recursive filter equations are as follows

a) Compute the Kalman gain filter, $\mathbf{K}(k)$*, as*

$$K(k) = P^-(k)H^T(k)[H(k)P^-(k)H^T(k) + R(k)]^{-1}$$

b) Compute the error covariance for the update

$$P(k)=[I - K(k)H(k)]\,P^-(k)$$

c) Update the estimate with the measurement $\mathbf{z}(k)$ *as*

$$x(k) = x^-(k) + K(k)[z(k)-H(k)x^-(k)]$$

d) Project ahead, the error covariance and the estimate

$$P^-(k+1) = \phi(k)\,P(k)\,\phi^T(k) + Q(k) \quad x^-(k+1) = \phi(k)\,x^-(k)$$

a) Initialization of the Kalman Filter

For off-line application, it is necessary to initialize the recursive process of the Kalman filter, with an initial vector $\mathbf{x}_o^-$ and its initial covariance matrix $\mathbf{P}_O$. Also, the system and measurement noise variances are needed.

A simple deterministic procedure is implemented to calculate the initial process vector, as well as its covariance matrix, using the static least error squares estimate of the previous measurements. Thus, in general the initial process vector may be computed as

$$x_o^- = [H^TH]^{-1}\,H^T z$$

and the corresponding covariance error matrix is

$$P_o = [H^TH]^{-1}H^T$$