POWER SYSTEM HARMONICS ESTIMATION USING DIFFERENT SIGNAL PROCESSING TECHNIQUES

A thesis submitted in partial fulfillment of the requirements for the award of the degree of

Master of Technology

In

Electrical Engineering

By

Rishikesh Kumar Jaiswal

Roll No: 212EE3229

Department of Electrical Engineering

National Institute Of Technology, Rourkela

Rourkela, Odisha, 769008

2014

POWER SYSTEM HARMONICS ESTIMATION USING DIFFERENT SIGNAL PROCESSING TECHNIQUES

A thesis submitted in partial fulfillment of the requirements for the award of the degree of

Master of Technology

In

Electrical Engineering

By

Rishikesh Kumar Jaiswal

(Roll No: 212EE3229)

Under the guidance of

Prof. Pravat Kumar Ray

Department of Electrical Engineering

National Institute Of Technology, Rourkela

Rourkela, Odisha, 769008

2014

National Institute of Technology

Rourkela

Certificate

This is to certify that the thesis titled ***"Power System Harmonics Estimation Using Different Signal Processing Techniques"*** submitted by ***Mr. Rishikesh Kumar Jaiswal*** in partial fulfillment of the requirements for the award of Master of Technology degree in ***Electrical Engineering*** with specialization in ***"Control and Automation"*** during session 2013-2014 at ***National Institute Of Technology, Rourkela*** is an authentic work by him under my supervision and guidance. To the best of my knowledge, the matter embodied in this project report has not been submitted to any other university/institute for the award of Degree or Diploma.

Date:

Prof. Pravat Kumar Ray

Department of Electrical Engineering

National Institute of Technology

Rourkela, 769008

ACKNOWLEDGEMENT

The project provided in this thesis is the most significant procurement of my career so far, fulfillment of this project and thesis would not have been attainable without the assistance of numerous individuals, whom I am exceptionally grateful. They continually helped me to do the project with full vitality.

Above all else, I might want to thank my guide **Prof. Pravat Kumar Ray** for his direction, persuasion and creative technical discussions. He is not just a good lecturer with significant vision additionally is an effectively agreeable kind individual. He empowered, backed and persuaded me all through the work. I always had the freedom to take over my thoughts for which I am extremely thankful.

Beside my advisor, I might additionally want to thank the majority of my teachers **Prof. B. Subudhi, Prof. A.K. Panda** and **Prof. Sandip Ghosh** for giving string learning to my study.

Last but not the least, I am obliged to my guardians for their adoration, sacrifice, and support during the entire period.

Rishikesh Kumar Jaiswal

ABSTRACT

Harmonics have been available for quite a while and its presence shapes the execution of a power system. Consequently, harmonics estimation is of principal vitality while analyzing the power system. Emulating the beginning of harmonics, different filters have been formulated to attain an ideal control methodology for constant rejection.

This thesis acquaints different algorithms to dissect harmonics in the power system. The target is to estimate the voltage magnitude and phase plot of the power system in the proximity of noise by using various estimation approaches. This thesis has centered the consideration towards the investigation of Kalman filter (KF), Recursive Least squares (RLS), least mean square (LMS) and Variable leaky least mean square (VLLMS) based filter for estimation of harmonics. For a test signal KF, RLS, LMS and VLLMS based calculations have been examined and the results have been looked at. The several algorithms are compared for various signals to noise ratio. The SNR used here are 40 dB, 30dB and 20dB. The proposed estimation methodologies are implemented on a typical power system signal acquired from mechanical burden embodying power electronic converters and arc furnaces.

Contents

List of Figures

List of Tables

List of Abbreviations

- RLS Recursive Least Square
- LMS Least Mean Square
- KF Kalman Filter
- SNR Signal to Noise Ratio
- LMS Least Mean Square
- VLLMS Variable Leaky Least Mean Square
- AMP1 Amplitude of fundamental harmonic component
- AMP3 Amplitude of 3rd harmonic component
- AMP5 Amplitude of 5th harmonic component
- AMP7 Amplitude of 7th harmonic component
- AMP11 Amplitude of 11th harmonic component

CHAPTER 1

INTRODUCTION

1.1 Background

The vast use of power electronic devices such as variable speed drives, computerized processing lines, PCs and nonlinear electronic gadgets in power system have provided increase in a kind of deformation in current and voltage waveform called as 'harmonics'. Harmonics could be defined as the undesirable parts of a deformed periodic waveform. The deformed waveform frequencies are the integer multiple of the basic frequency. The closeness of these harmonics to the noise brings power loss and equipment warming, and obstruction with protection and control circuits and also with customer loads.

1.1.1 What are Harmonics?

Harmonics can be described as a sinusoidal section of a periodic wave having a frequency that is a integral multiple of basic frequency (1). A distorted periodic wave of any possible shape might be generated by utilizing distinctive harmonic frequencies with different amplitudes (1). Any distorted periodic wave could be disintegrated into a basic wave and a set of harmonic waves. This disintegration process is called Fourier series expansion. By this expansion, the influence of nonlinear loads in power system could be systematically investigated.

Different harmonics have different magnitudes and phases separately. The basic power frequency in India is 50 Hz & 60 Hz in U.S.A. There are basically two types of harmonics. These are odd and even harmonics. Repeatedly, when curve is symmetrical to origin then only odd harmonics are available in the power system. So, harmonics are indication of failure or some faulty operation.

1.1.2 What are causes of harmonics?

Nonlinear loads cause harmonics. A nonlinear load in power system is a gadget in which current has nonlinear relation with the voltage. It implies current has same wave shape as voltage. The nonlinear loads are continuous power supply (UPS), rectifiers, PCs and solid states variable velocity speed motor drives. Those transformers, motors and generators which are saturable, likewise produce harmonics. Arc furnaces, fluorescent and mercury lights likewise create harmonics. To identify harmonics in power system is a tough task.

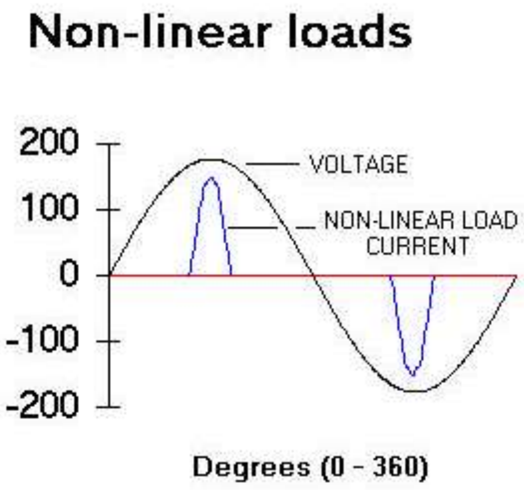

Fig1.1 Waveforms of voltage and current for non-linear loads

1.1.3 Effects of Harmonics

The harmonics causes failures of electrical gadget, overheating of neutral wires, transformer heating. The currents produced due to harmonics can create a lot of problems. Such as gadget heating, device malfunction, gadget failure, communications interference, fuse and breaker failure, problems in processes and heating of the conductor.

1.2. Literature review

G. K. Singh [1] has discussed about harmonics present in the power system. He has discussed about the nonlinear loads due to which harmonics are generated .In this paper exhaustive study of causes and effects of harmonics generation has been done.

Maamar Bettayeb [2] exhibits the provision of familiar recursive estimation strategies to the critical issue of the power system. On-line estimation of harmonics amplitudes and phases is obtained utilizing recursive least square (RLS) calculations, known for their effortless of calculation and great convergence properties. The assessments are upgraded recursively as specimens of the harmonic signals are obtained. The majority of the estimates are in 3–4% error. The maximum deviation is in 9–10%. The results acquired from the recursive estimation are reasonably near the evaluations got by the batch calculations. This finding empowers the utilization of recursive calculations to be actualized on-line and provide estimates with time varying tracking capability.

LMS algorithm [4] is used when the formulated structure seems simple. It possesses property of robustness. The Poor convergence rate of LMS algorithm is a big problem when the step size is fixed. The step size is inversely proportional to input power, when the step size is small.

To solve this problem variable step LMS algorithm has been given by Raymond h. [5]. P.K. Ray [6] has given an algorithm which has faster convergence and poor computational burden. Keren Kennedy [7] uses a Kalman filter for analysis of harmonics in power systems. There are many cases which have been discussed in it. Amplitudes and phases of harmonics have been estimated for different conditions. One condition is fixed frequency. Harmonics estimation 4 is important to lessen harmonic line currents.

1.3. Harmonics estimation problem

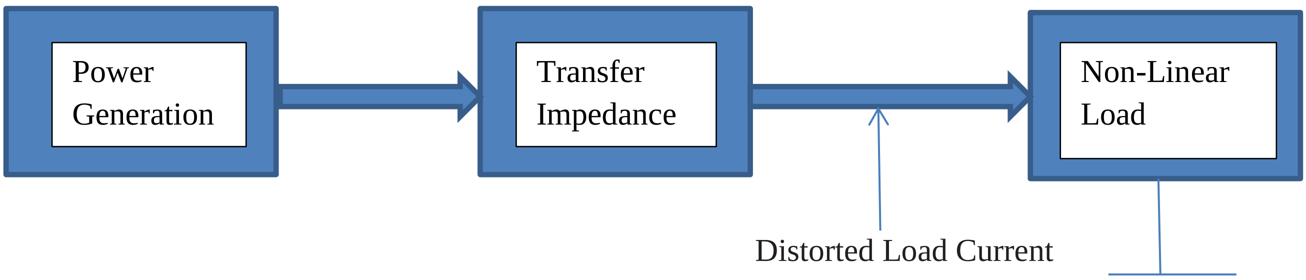

Fig.1.2 Schematics of the harmonics estimation problem

More use of nonlinear loads, for example, power gadgets presents more amount of harmonics into the power system. In actuality, power system voltage or current sign strays from the periodic sinusoidal waveform and specifically the distortion of the current waveform gets more mind boggling. Without suitable filtering there is a great loss of power and inter harmonics and sub harmonics will be introduced in the system. Both harmonics and inter harmonics have unfriendly impacts of the power system.

1.4. Objective of the thesis

The objectives of the thesis are following:

- To apply recursive least squares (RLS) algorithm to the power system signal obtained from industrial load. This load is comprised of power electronic converter and arc furnaces [4].
- To apply least mean square (LMS) algorithm for the same signal and estimation of amplitudes and phases have been done with MATLAB.
- Variable leaky least mean square (VLLMS) has been proposed to avoid less speed of convergence.
- To compare of RLS, LMS, VLLMS with Kalman filter (KF) results obtained from applying KF algorithm to the same load signal in the MATLAB.

1.5. Organization of the thesis

The thesis is structured as follows

Chapter 1 contains the basics of the harmonics in the power system. It includes definition of the harmonics and effects of the harmonics. It also includes harmonics estimation problem and objective/motivation of the thesis.

Chapter 2 provides different signal processing techniques such as KF, RLS, LMS and VLLMS. These techniques have been explained and these have been applied to the power system signal.

Chapter 3 consists of simulation results. Various results obtained from simulation have been compared in this chapter. Comparison of amplitudes and phases by different algorithms for a particular signal to noise ratio have been done. The different harmonics term present in the signal has been estimated by these algorithms.

Chapter 4 gives estimated values of amplitudes for various SNR and also the conclusions of various estimation methods for power system harmonics.

CHAPTER 2

ESTIMATION OF HARMONICS

2.1 Power system signal

We have taken a corrupted static signal. The signal is altered with random noise and decreasing DC component to estimate the performance of power system parameters such as amplitudes and phases using different signal processing techniques. MATLAB programming has been performed for implementation of these techniques for harmonic estimation. The power system signal utilized for the estimation, contains fundamental frequency and higher order harmonics. The higher order harmonics are III, V, VII, XI and DC decreasing component.

2.1.1 DC decreasing offset

In transient state, electrical signal consists of harmonics and DC decreasing component. The goal is to remove these components while estimation of signal at initial stages. These are in the form of exponential terms. The current signal consists of decreasing component when there is a short circuit in the power system.

2.2 Discretization of the standard power system signal for harmonic estimation

Assume the waveform (current or voltage waveform of the power system) consists of the fundamental angular frequency ω as a harmonics' addition of unknown magnitudes and phases. The waveform's general form is

$$y(t) = \sum_{m=1}^{M} A_m \sin(\omega_m t + \phi_m) + A_{dc} \exp(-\alpha_{dc} t) + \varepsilon(t) \tag{2.1}$$

Where M is the number of harmonics, $\omega_m = m2\pi f_0$

f_0 is the fundamental frequency; ε (t) is the additive noise; $A_{dc} \exp(-\alpha_{dc} t)$ is the DC offset decreasing term.

After discretization of Eq. (2.1) with a sampling time T, one gets the following equations

$$y(t) = \sum_{m=1}^{M} A_m \sin(\omega_m nT + \phi_m) + A_{dc} \exp(-\alpha_{dc} nT) + \varepsilon(k) \tag{2.2}$$

Expansion of the DC decreasing term using Taylor series for small value of α_{dc} giving

$$y_{dc} = A_{dc} - A_{dc}\alpha_{dc} nT \tag{2.3}$$

Using Eq. (2.4) in Eq. (2.3), $y(n)$ can be calculated as

$$y(t) = \sum_{m=1}^{M} A_m \sin(\omega_m nT + \phi_m) + A_{dc} - A_{dc}\,\alpha_{dc} nT + \varepsilon(n) \tag{2.4}$$

For estimation of amplitudes and phases, Eq. (2.4) can be modified as

$$y(t) = \sum_{m=1}^{M} A_m \sin(\omega_m nT)\cos\phi_m + A_m \cos(\omega_m nT)\sin\phi_m + A_{dc} - A_{dc}\,\alpha_{dc} nT + \varepsilon(n) \tag{2.5}$$

Further, the Eq. (2.5) can be written in parametric form as

$$y(n) = H(n)X$$

$$\text{H(n)}=[\ \sin(\omega_1 nT) \quad \cos(\omega_1 nT) \quad \ldots\ldots\ldots \quad \sin(\omega_m nT) \quad \cos(\omega_m nT) \quad 1 \quad \text{nT}\ ]^T \tag{2.6}$$

The vector whose parameter is not known

$$\text{X (n)}=[\ X_1(n) \quad X_1(n) \ldots\ldots\ldots X_{2M-1}(n) \quad X_{2M}(n) \quad X_{2M+1}(n) \quad X_{2M+2}(n)\ ]^T \tag{2.7}$$

$$\text{X}= [\ A_1\cos(\phi_1) \quad A_1\sin(\phi_1) \ldots\ldots\ldots\ldots A_m\cos(\phi_m) \quad A_m\sin(\phi_m) \quad A_{dc} \quad A_{dc}\alpha_{dc}\ ] \tag{2.8}$$

2.3 Least mean square for the power system signal

By using a least mean square algorithm, state estimation is done. In this algorithm, square of error is minimized recursively by changing the obscure parameter X_n each and every sampling instant using equation (2.10) given below

$$X_n = X_{n-1} + \mu_n e_n \hat{y}_n \tag{2.9}$$

Where e_n - error signal

μ_n - Step size

Error signal

$$e_n = y_n - \hat{y}_n$$

For better convergence, the step size is updated in the presence of noise.

$$\mu_{n+1} = \lambda \mu_n + \gamma R_n^2 \tag{2.10}$$

Where R_n is autocorrelation of e_n and e_{n-1}. It is obtained as

$$R_n = \rho R_{n-1} + (1-\rho) e_n e_{n-1} \tag{2.11}$$

Where ρ - exponential weighting factor

$0 < \rho < 1$

$0 < \gamma < 1$

Amplitudes and phases of basic and nth harmonic parameters are derived by updating vector X_n with the help of LMS algorithm as follows.

$$A_m = (\sqrt{X_{2M}^2 + X_{2M-1}^2}) \tag{2.12}$$

$$\phi_m = \tan^{-1}(\frac{X_{2M}}{X_{2M-1}}) \tag{2.13}$$

DC decreasing parameters are derived as

$$A_{dc} = X_{2M+1} \tag{2.14}$$

$$\alpha_{dc} = (\frac{X_{2M+2}}{X_{2M+1}}) \tag{2.15}$$

2.3.1 Steps to estimate harmonics in least mean square (LMS) algorithm

1. Initialize λ, ρ, R, X
2. Generate power system signal
3. Discretize the signal with a sampling period and estimate it using initial state vector
4. Obtain: Estimation error = Actual signal - Estimated signal
5. Update auto correlation, and step size using (2.11) and (2.10).
6. Update the unknown vector X using (2.9)
7. Go to step 4, if last iteration is not obtained.
8. Estimate amplitude and phase for fundamental and nth harmonics from (2.12) and (2.13).

2.3.2 Flow chart of LMS algorithm

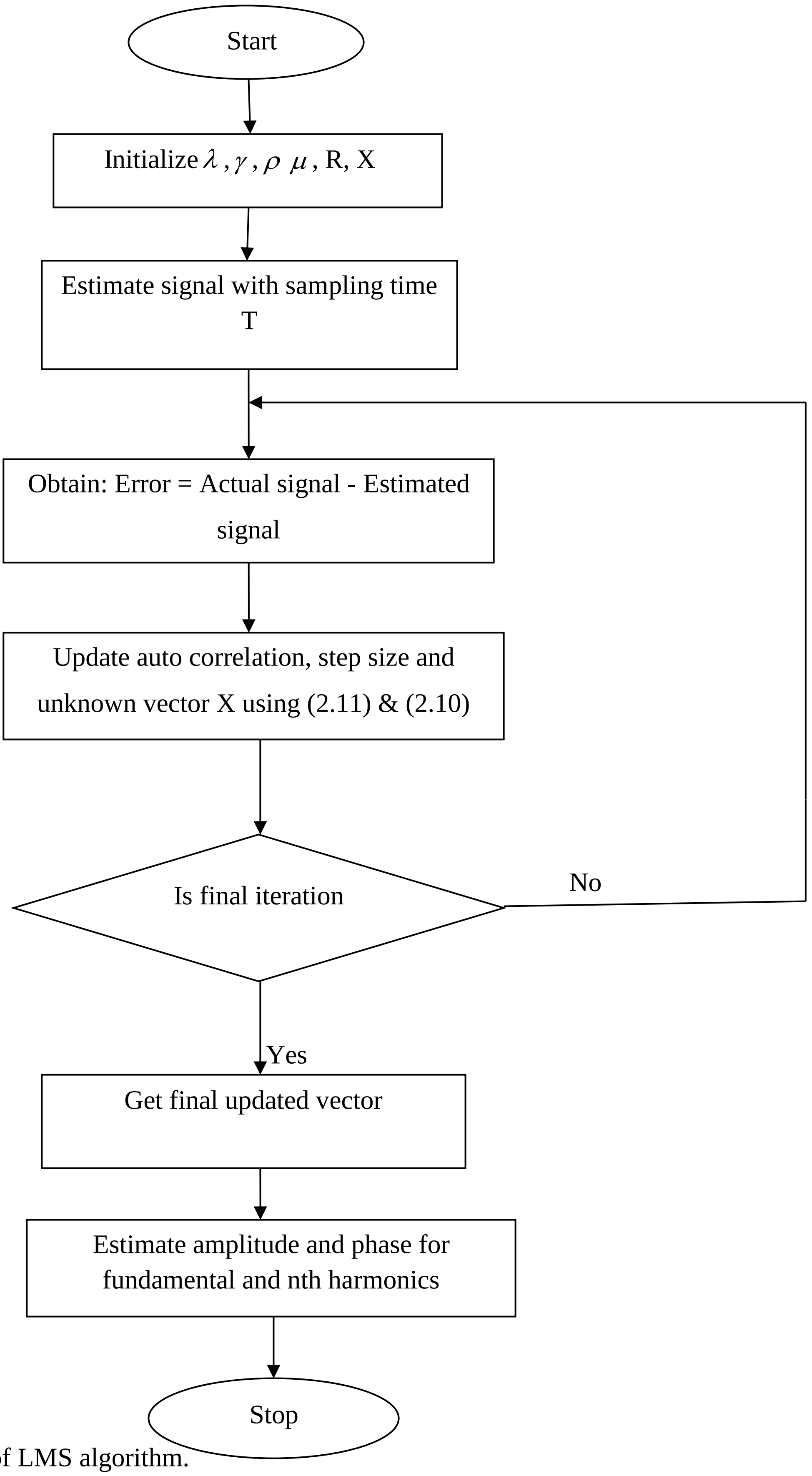

Fig. 2.1Flowchart of LMS algorithm.

2.4 Kalman filter for harmonic estimation

2.4 Introduction of Kalman filter

The Kalman filter is a set of equations. It uses predictor-corrector kind methodology that minimizes the estimated error covariance once some assumed conditions are met.

Kalman filter estimates the state $x \varepsilon R^n$ of the process. Here the process is in the form of linear difference equation. This equation is stochastic.

$$x_n = Ax_{n-1} + Bu_n + w_{n-1}$$

There is a measurement vector $z \varepsilon R^n$.

$$y_n = Hx_n + v_n$$

Here u is the input vector. A, B & H are matrices of proper dimensions. w Is process noise and v is the measurement noise vector. Here w and v are independent of one another and white noise having zero mean. The covariances of w & v are obtained as

$$E[w_n w_n^T] = Q$$

$$E[v_n v_n^T] = R$$

2.4.1 Computational origin of the filter

A priori state estimate is $\hat{x}_n^-$ and A posteriori state estimate is $\hat{x}_n$. Priori and posteriori error estimates are defined as

$$e_n^- = x_n - \hat{x}_n^- \quad \text{And}$$

$$e_n = x_n - \hat{x}_n .$$

Then a priori estimate error covariance $P_n^- = E[e_n^- e_n^{-T}]$

And posteriori estimate error covariance is $P_n = E[e_n e_n{}^T]$

For Kalman filter, posteriori state estimate is identical to priori state plus the weighted difference between actual measurement and measurement prediction.

$$\hat{x}_n = \hat{x}_n^- + K(y_n - H\hat{x}_n^-)$$

Where K is Kalman gain

$$K_n = P_n^- H^T (HP_n^- H^T + R)^{-1}$$

2.4.2 Discrete Kalman filter algorithm

The estimation by Kalman filter is done by a type of feedback form. Firstly process state is estimated at some time by filter, then filter get feedback as noisy measurements. Kalman filter equations are splitted into two groups such as time update and measurement update. By time update current states are being projected in time and a priori estimate is obtained by error covariance for next time step. The feedback is being decided by measurement update. Time update is also called as a predictor and measurement update is called as corrector equations. So, we can say that it is a predictor and a corrector type algorithm.

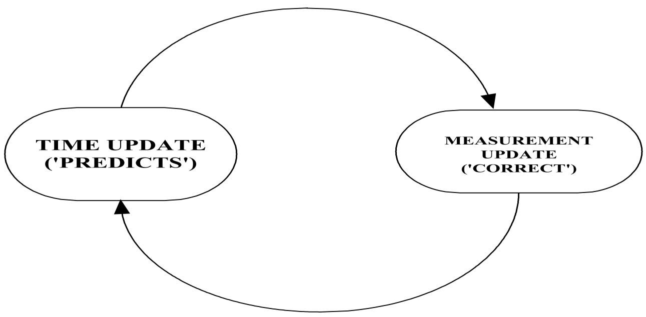

Fig. 2.2 Kalman filter cycle

Time update equations:

$$x_n = Ax_{n-1} + Bu_n$$

$$P_n^- = AP_{n-1}A^T + Q$$

Measurement update equations : $K_n = P_n^- H^T (HP_n^- H^T + R)^{-1}$

$$\hat{x}_n = \hat{x}_n^- + K(y_n - H\hat{x}_n^-)$$

$$P_n = (I - K_n H)P_n^-$$

2.4.4 Application of Kalman filter for estimation of harmonics

The signal as given in section 2.1 is taken; the vector of unknown parameters *X*, as in (2.10) is updated using the Kalman Filter algorithm as

$$K(n) = P(n/n-1)H(n)^T (H(n)P(n/n-1)H(n)^T + R)^{-1} \tag{2.16}$$

Where

K – Kalman Gain

H - Observation vector

P – Covariance matrix

R – Noise Covariance of the measurement noise

The covariance matrix *P* depends on the Kalman gain *K* and previous covariance matrix *P(n-1)* as follows

$$P(n/n) = P(n/n-1) - K(n)H(n)P(n/n-1) \tag{2.17}$$

The state vector is updated according to the formula written below. The state vector depends on the previous vector as follows

$$\hat{X}(n/n) = \hat{X}(n/n-1) + K(n)(y(n) - H(n)\hat{X}(n/n-1)) \tag{2.18}$$

The updated state vector *X* is obtained from the above formulas. Then amplitude and phases of the basic harmonics as well as nth harmonics and DC decaying terms are obtained by using (2.12) - (2.15).

2.5 Recursive least square (RLS) for harmonics estimation

2.5.1 RLS method

The signal as given in equation 2.1 is used; the unknown vector parameter X as in (2.8) is upgraded using RLS as

X is updated using the RLS algorithm as

$$\hat{X}(n+1) = \hat{X}(n) + K(n+1)e(n+1) \quad (2.19)$$

Measurement error is

$$e(n+1) = y(n+1) - H(n+1)^T \hat{X}(n) \quad (2.20)$$

The relation between gain K with covariance of parameter vector is

$$K(n+1) = P(n)H(n+1)[1 + H(n+1)^T P(n)H(n+1)]^{-1} \quad (2.21)$$

Using matrix inversion lemma, the updating of covariance of parameter vector

$$P(n+1) = [I - K(n+1)H(n+1)^T]P(n) \quad (2.22)$$

The above equations are calculated by using some beginning values to estimate at instants n. The beginning covariance matrix P is taken as $P = \alpha I$, where α has a large integer value and I is a square identity matrix.

After the updating of the vector of unknown parameters using Recursive Least Square (RLS) algorithm, amplitudes, phases of the basic and not harmonic parameters and DC decreasing parameters can be derived using (2.12) -(2.15).

2.5.3 Flow Chart of RLS Filter

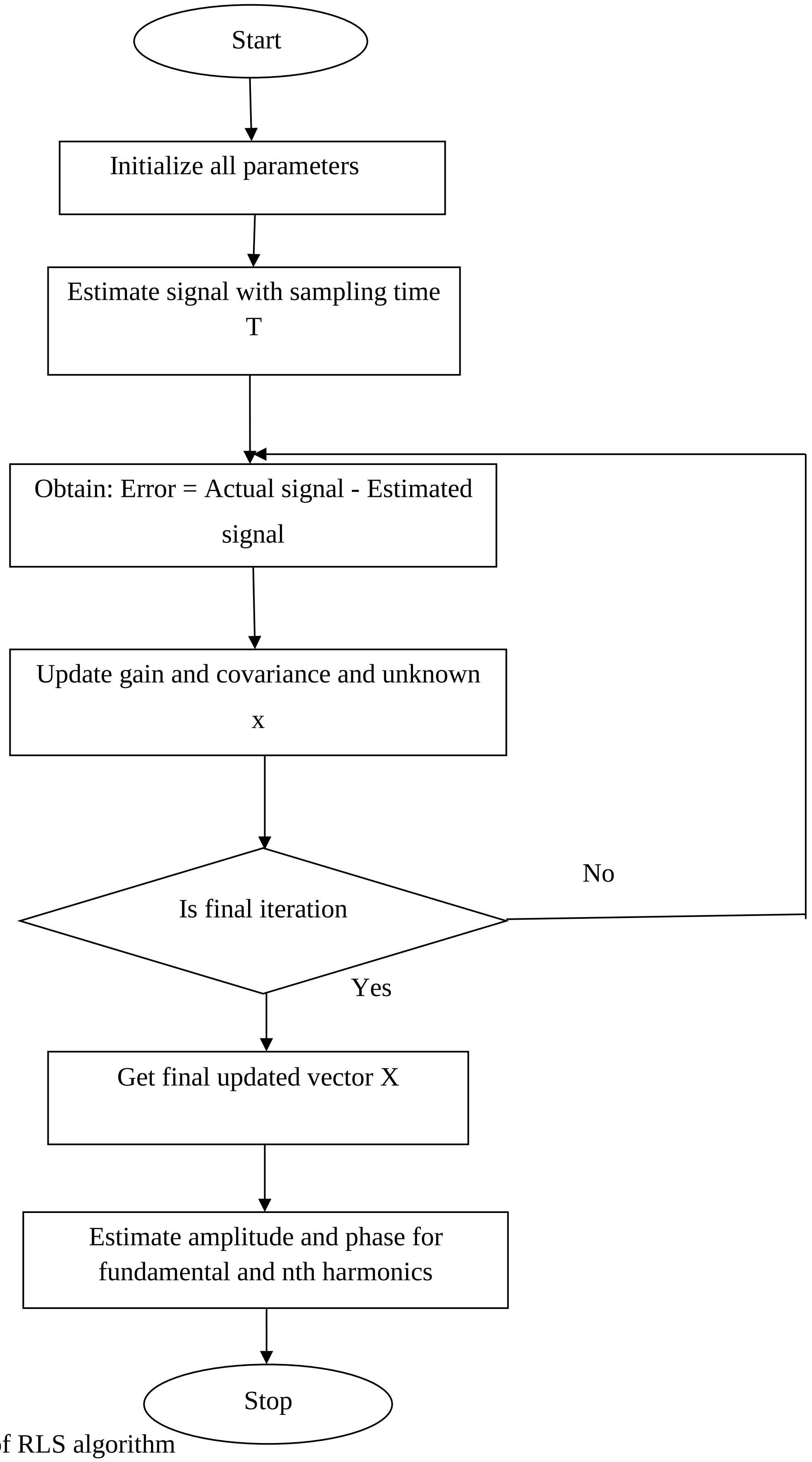

Fig.2.4 Flowchart of RLS algorithm

2.6 Variable leaky least mean square algorithm for harmonics estimation

2.6.1 VLLMS Algorithm

In VLLMS algorithm step size is varied along with leakage factor. It has faster convergence. During estimation process, the computational burden is less. It has higher accuracy. It is based on adaptive linear filtering shown in figure.

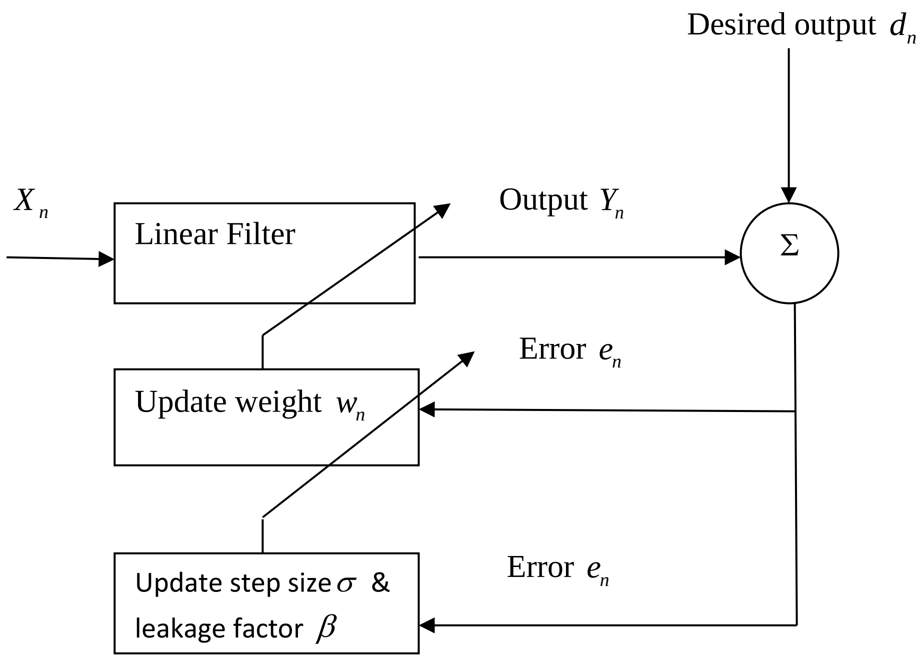

Fig.2.5 Adaptive linear estimation scheme

The linear filter is written in mathematical equation

$$Y_n = w_n^T X_n \tag{2.23}$$

$X_n \varepsilon R$ is input to the filter ,n is Discrete time constant and w_n is filter gain . Filter gain w_n maps input $X_n \varepsilon R$ to the output $Y_n \varepsilon R$ which is matched with desired output d_n .

The aim is to tune w_n so that error converges to minimum.

$$e_n = d_n - Y_n \tag{2.24}$$

There are several methods to tune w_n.One of these is LMS. In LMS, tuning is done to minimise instantaneous cost function shown below to zero.

$$J_n = e_n^2 \tag{2.25}$$

Drifting of w_n during external disturbance results in higher filter gains. This is disadvantage of LMS. Leaky algorithm by MAYYAS is proposed in which cost function is

$$J_n = e_n^2 + \beta w_n^T w_n \tag{2.26}$$

$$0 < \beta < 1$$

β is chosen as per above range so that drifting is avoided. $w_n^T w_n$ is called regularisation component. Constant β may lead to over or under parameterisation. So, we use variable β .Hence, cost function becomes

$$J_n = e_n^2 + \beta_n w_n^T w_n \tag{2.27}$$

This cost function is utilized in updating of VLLMS algorithm.

The adaptive laws for updating w_n and β_n are following.

For Steepest descent rule, we can write

$$w_{n+1} = w_n - \sigma \frac{\delta J_n}{\delta w_n} \tag{2.28}$$

Where $\sigma > 0$

is a step size parameter. Fixed step size will cause problem to convergence. So, we use variable step size for faster convergence.

$$w_{n+1} = w_n - \sigma_n \frac{\delta J_n}{\delta w_n} \tag{2.29}$$

Where

σ_n -Variable step size parameter

σ_n is updated as

$$\sigma_{n+1} = \lambda\sigma_n + \beta_n e_n^2 \tag{2.30}$$

For harmonics estimation, we have to deal with a lot of external disturbances. To take care of external disturbances in, updating of σ_n is involved to use of autocorrelation of error to square of error.

$$\sigma_{n+1} = \lambda\sigma_n + \beta_n p_n^2 \tag{2.31}$$

Where p_n denotes autocorrelation of e_n and e_{n-1}.It is computed as

$$p_n = \xi p_{n-1} + (1-\xi)e_n e_{n-1} \tag{2.32}$$

Where, 0< ξ <1

To obtain final update equation of w_n and β_n, we write

$$e_n^2 = d_n^2 + (w_n^T w_n)^2 - 2d_n w_n^T x_n \tag{2.33}$$

From above equation

$$\frac{\delta e_n^2}{\delta w_n} = 2(w_n^T x_n - d_n)x_n = -2e_n x_n \tag{2.34}$$

Using equation (2.27) in (2.29) and then (2.34) in that, we obtain

$$w_{n+1} = (1 - 2\sigma_n\beta_n)w_n + 2\sigma_n e_n x_n \tag{2.35}$$

Further, to update β_n , the steepest descent method is applied to cost function (2.25) since the drift of parameter is no longer required to control. Hence

$$\beta_{n+1} = \beta_n - \frac{\tau}{2}\frac{\delta e_n^2}{\delta\beta_{n-1}} \tag{2.36}$$

τ >0, the derivative term is also written as

$$\frac{\delta e_n^2}{\delta \beta_{n-1}} = [\frac{\delta e_n^2}{\delta w_n}]^T \frac{\delta w_n}{\delta \beta_{n-1}} \quad (2.37)$$

Using equation (2.35) we may also write

$$\frac{\delta w_n}{\delta \beta_{n-1}} = -2\sigma_n w_{n-1} \quad (2.38)$$

By using (2.34) & (2.38) into (2.37), and that in (2.36), we get

$$\beta_{n+1} = \beta_n - 2\sigma_n \tau e_n x_n^T w_{n-1} \quad (2.39)$$

This is complete VLLMS algorithm.

2.6.2 Variable leaky least mean square (VLLMS) algorithm implementation for power system harmonics estimation

Weight W is state vector X.

X_n is updated at each sampling instant as

$$X_{n+1} = (1 - 2\sigma_n \beta_n) X_n + 2\sigma_n e_n \hat{y}_n \quad (2.40)$$

Where $e_n = y_n - \hat{y}_n$ (2.41)

For better convergence step size is varied as

$$\sigma_{n+1} = \lambda \sigma_n + \beta R_n^2 \quad (2.42)$$

Where $R_n = \xi R_{n-1} + (1-\xi) e_n e_{n-1}$ (2.43)

Variable leakage factor is updated as

$$\beta_{n+1} = \beta_n - 2\sigma_n \tau e_n \hat{y}_n X_{n-1} \quad (2.44)$$

After updating X_n, phases and amplitudes of fundamental and nth harmonics are obtained.

2.6.3 Flow chart of estimation by VLLMS algorithm

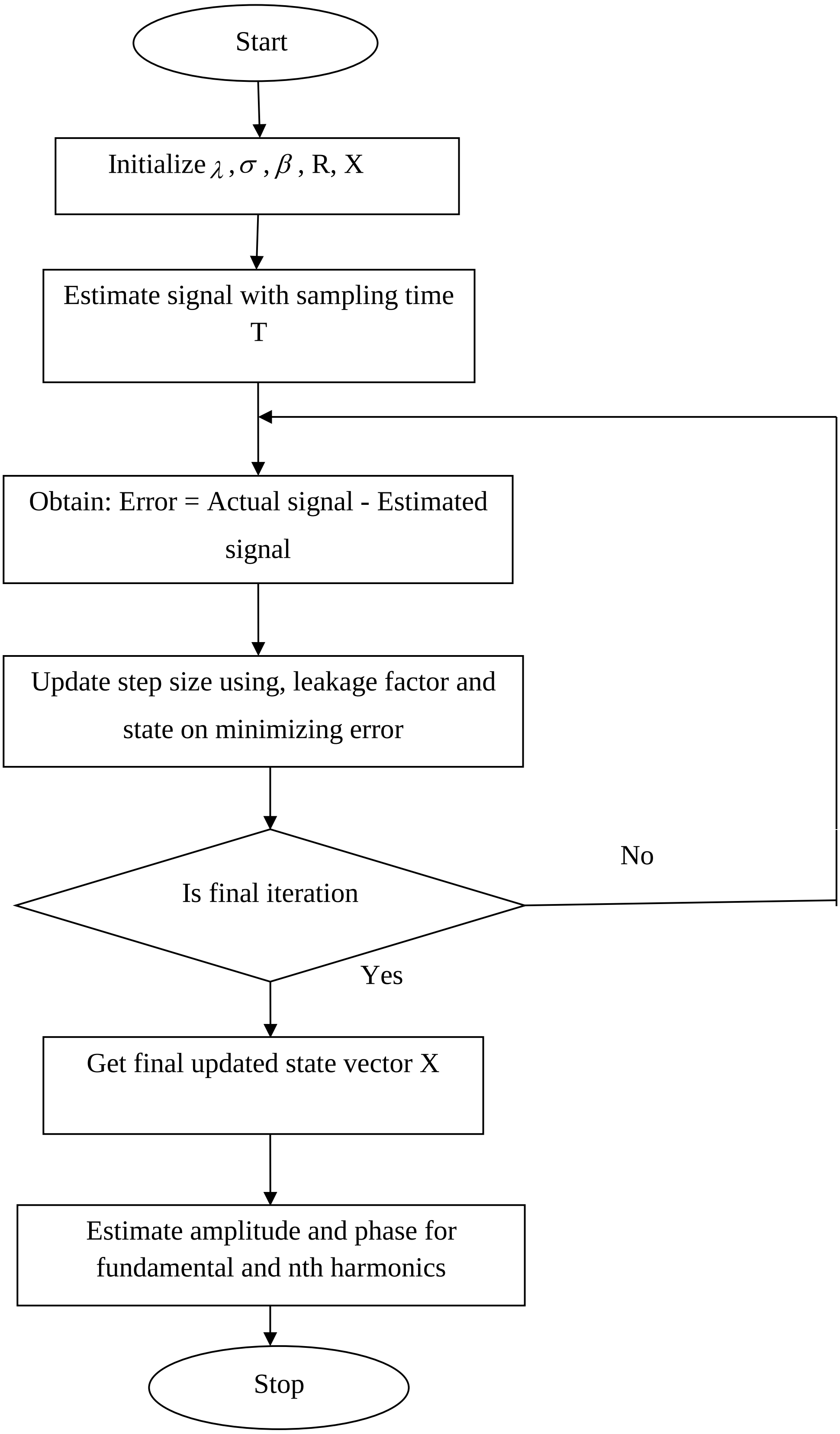

Fig. 2.6 Flowchart of estimation by VLLMS algorithm

CHAPTER 3

SIMULATION

RESULTS

3.1 A Signal Used For Simulation

We have taken a typical signal from industrial load consisting of power electronics converters and arc furnaces. This typical signal has been used in MATLAB. By application of different algorithms on the signal we have been able to estimate magnitudes and phases of fundamental harmonic as well as nth harmonics terms.

$$y(t) = 1.5\sin(\omega t + 80^0) + 0.5\sin(3\omega t + 60^0) + 0.2\sin(5\omega t + 45^0) +$$
$$0.15\sin(7\omega t + 36^0) + 0.1\sin(11\omega t + 30^0) + 0.5\exp(-5t) + \varepsilon(t)$$

The parameters used for LMS algorithm are following.

$\rho = 0.99$; $\gamma = 0.001$

$\lambda = 0.97$; $\mu(1) = 0.001$

The parameters used for VLLMS algorithm are following.

$\sigma(1) = 0.001$; $\xi = 0.99$; $\tau = 1.1$;

$\lambda = 0.97$; $\beta = 0.001$

3.2. Comparison of amplitudes of harmonics by different algorithm

Sampling time has been taken 0.001 seconds. So, 10 samples means = 10*0.001=0.01 sec.

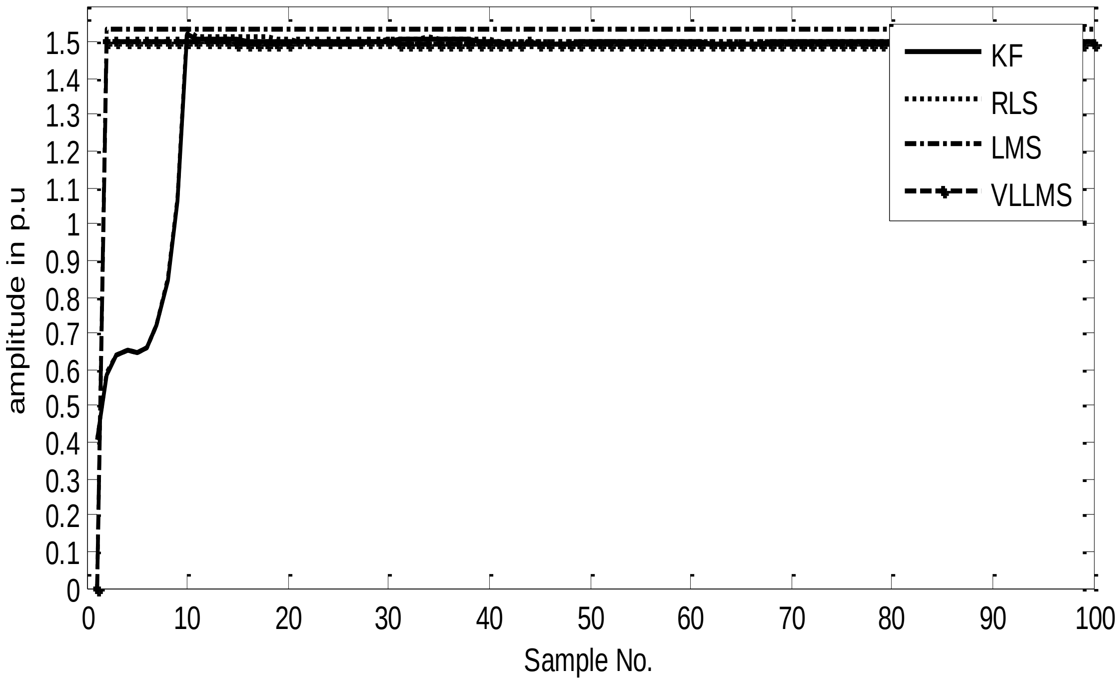

Fig. 3.1 Amplitudes of fundamental component

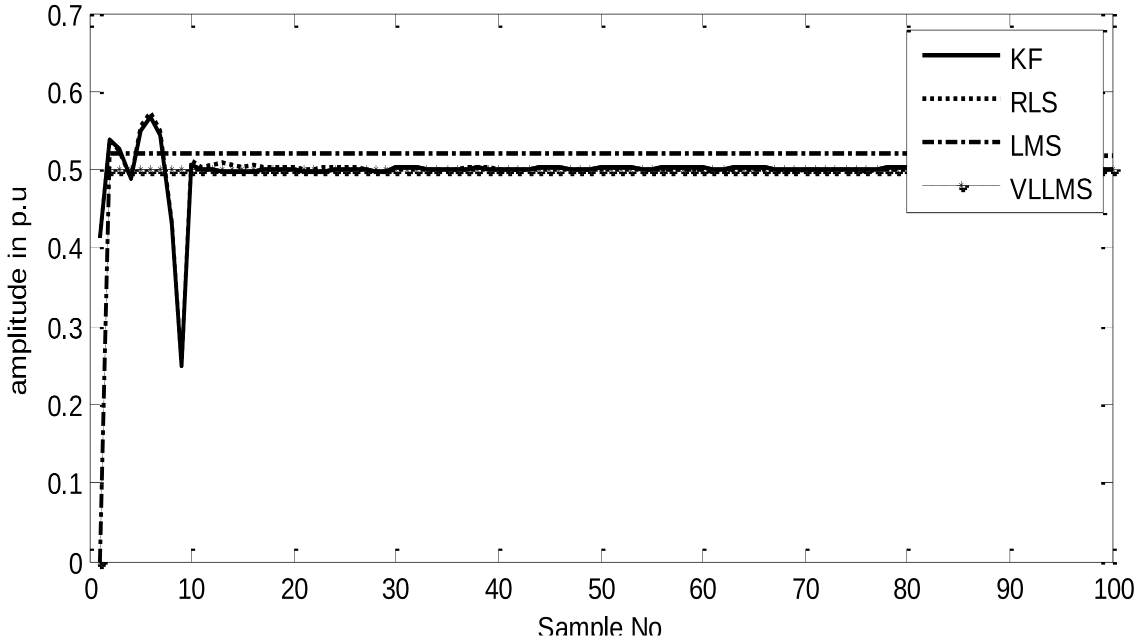

Fig. 3.2 Amplitudes of 3rd harmonic component

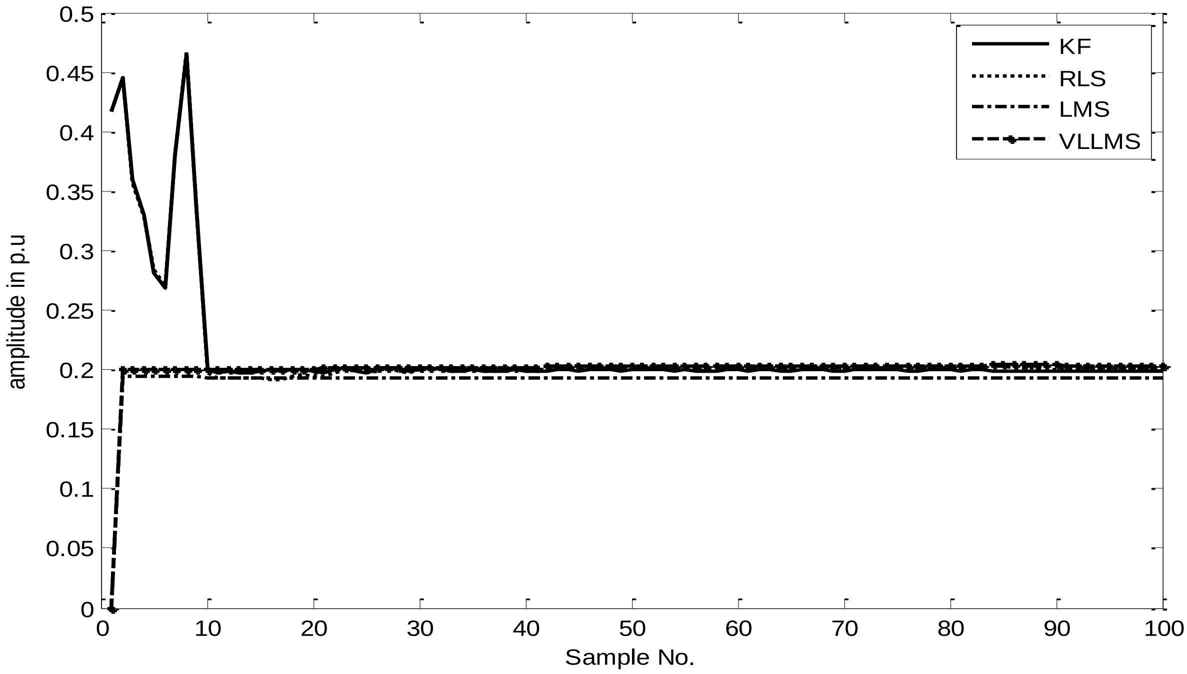

Fig. 3.3 Amplitudes of 5th harmonic component

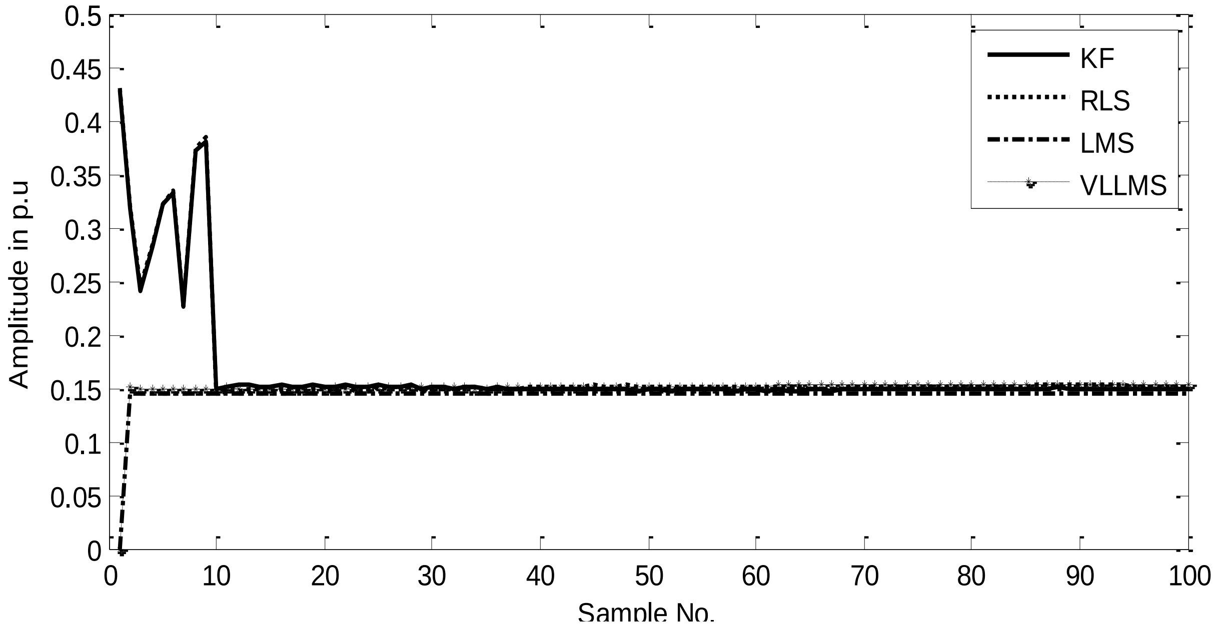

Fig. 3.4 Amplitudes of 7th harmonic component

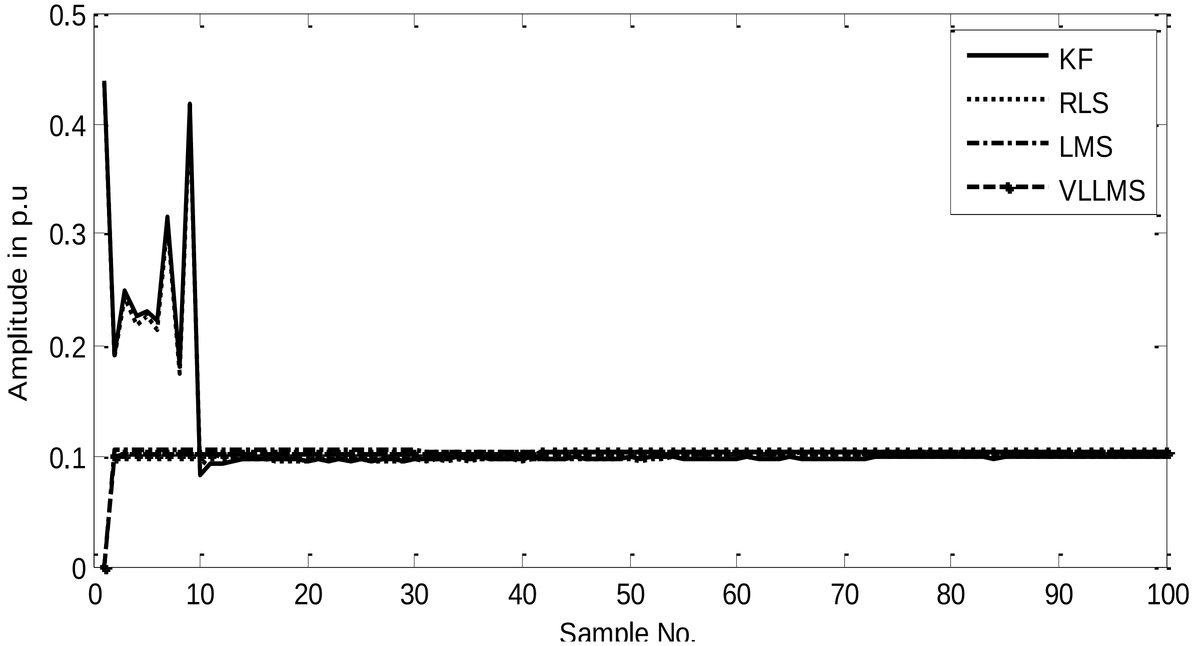

Fig. 3.5 Amplitudes of 11^{th} harmonic component

3.3. Comparison of Phases of harmonics by different algorithm

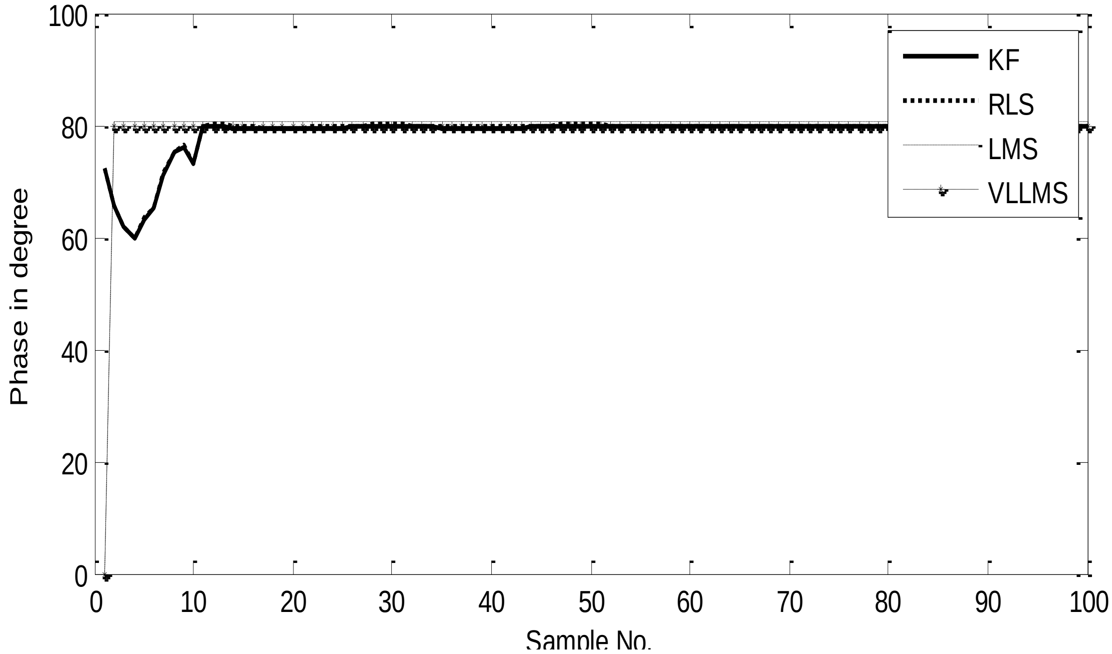

Fig. 3.6 Phase of fundamental component

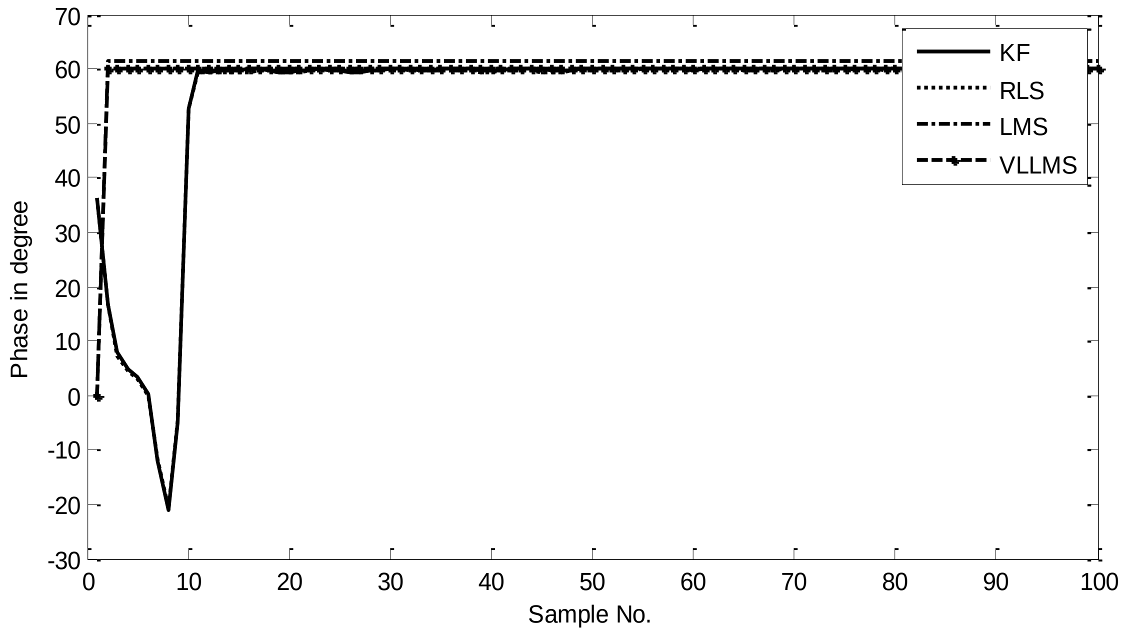

Fig. 3.7 Phase of 3rd harmonics component

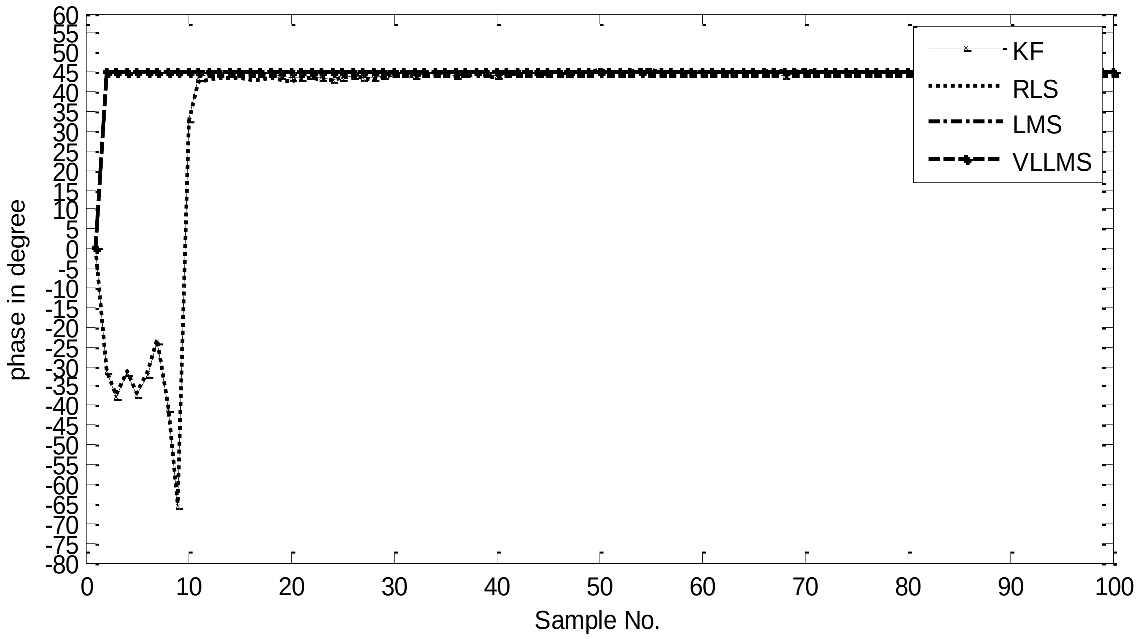

Fig. 3.8 Phases of 5th harmonics component

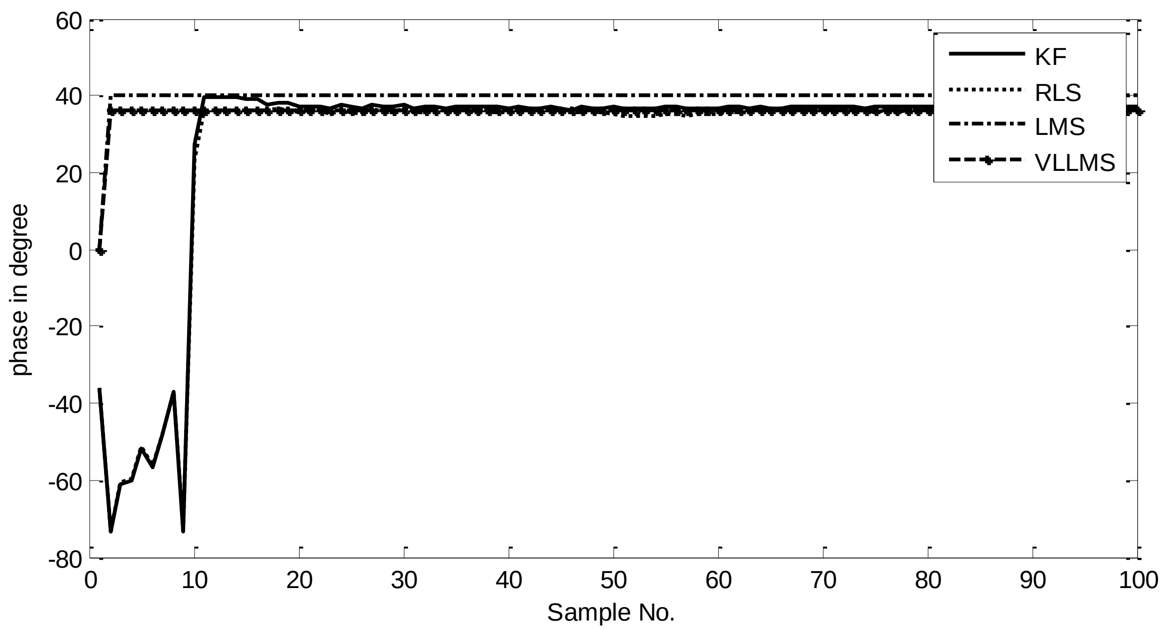

Fig. 3.9 phase of 7th harmonics component

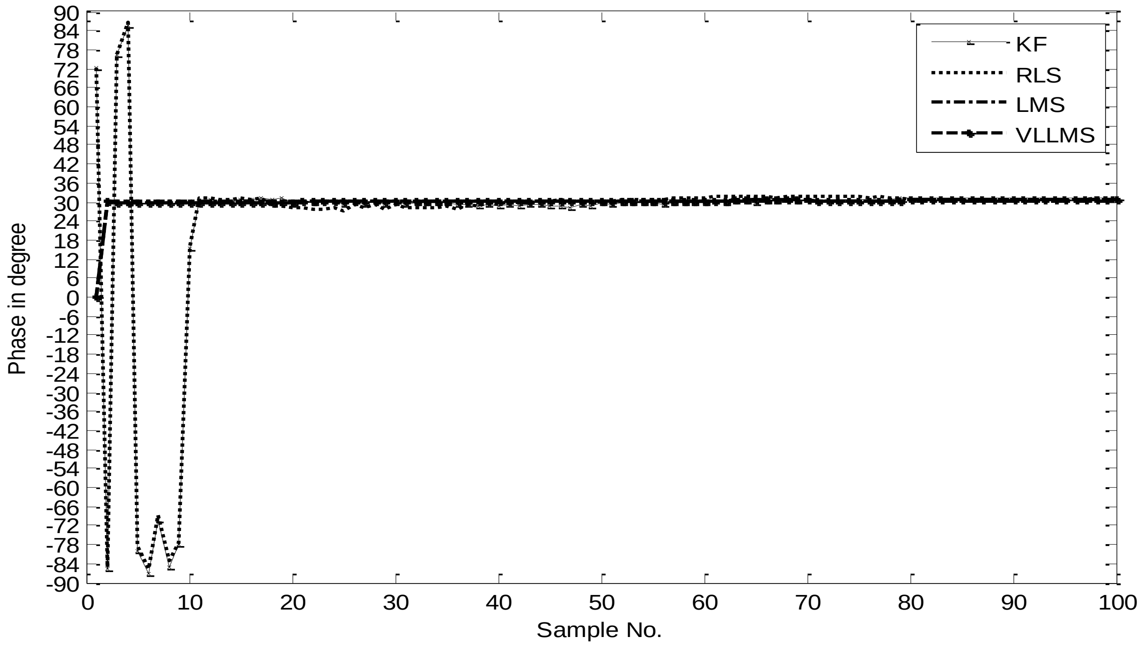

Fig. 3.10 Phase of 11th harmonics component

3.4 Estimation of fundamental and harmonics components

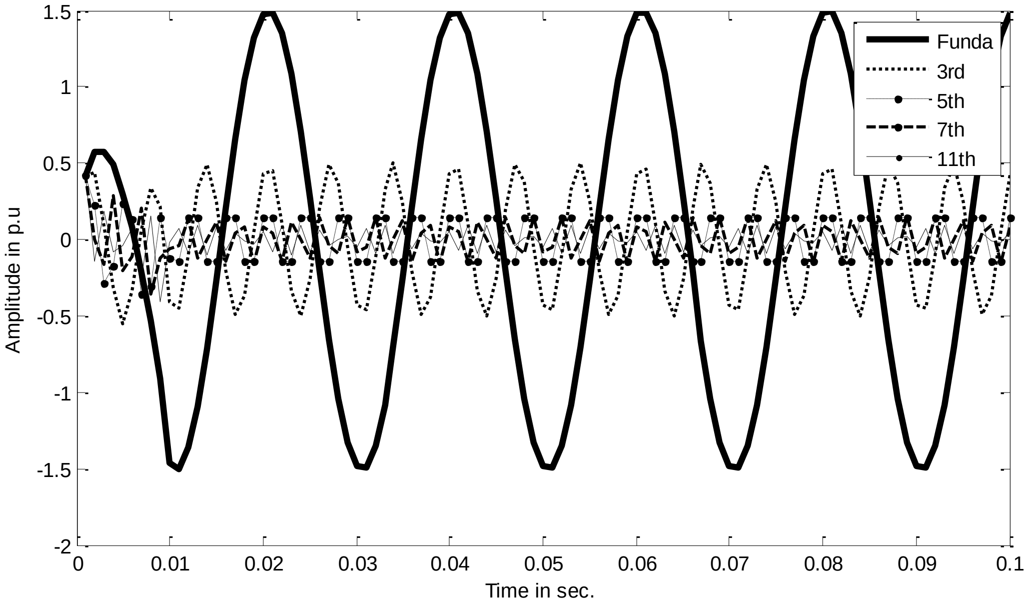

Fig 3.11 Estimation of various harmonics components of signal by RLS

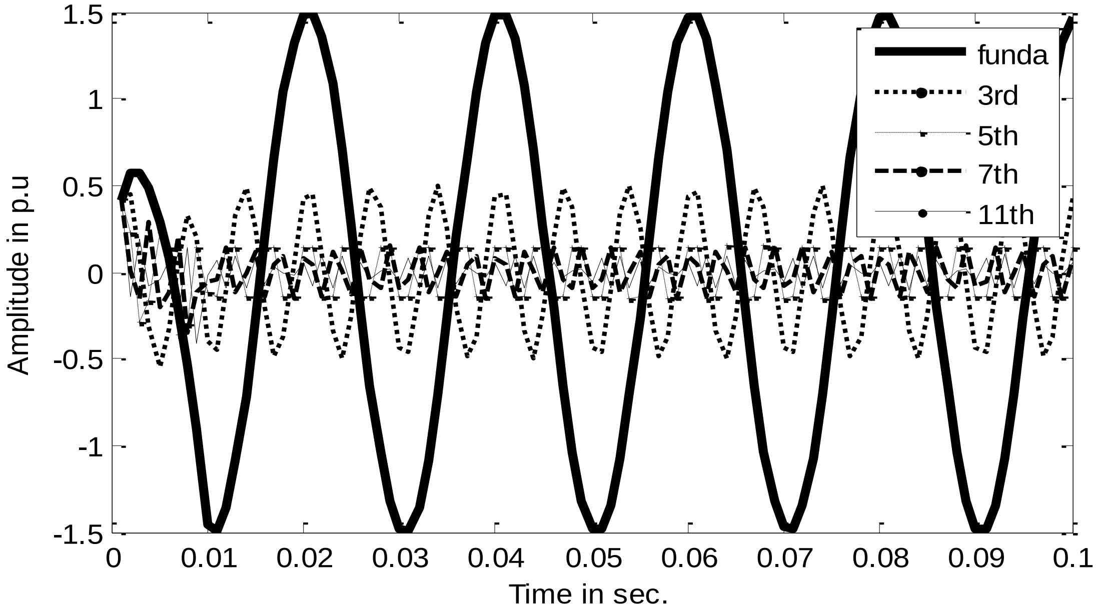

Fig 3.12 Estimation of various harmonics components of signal by KF

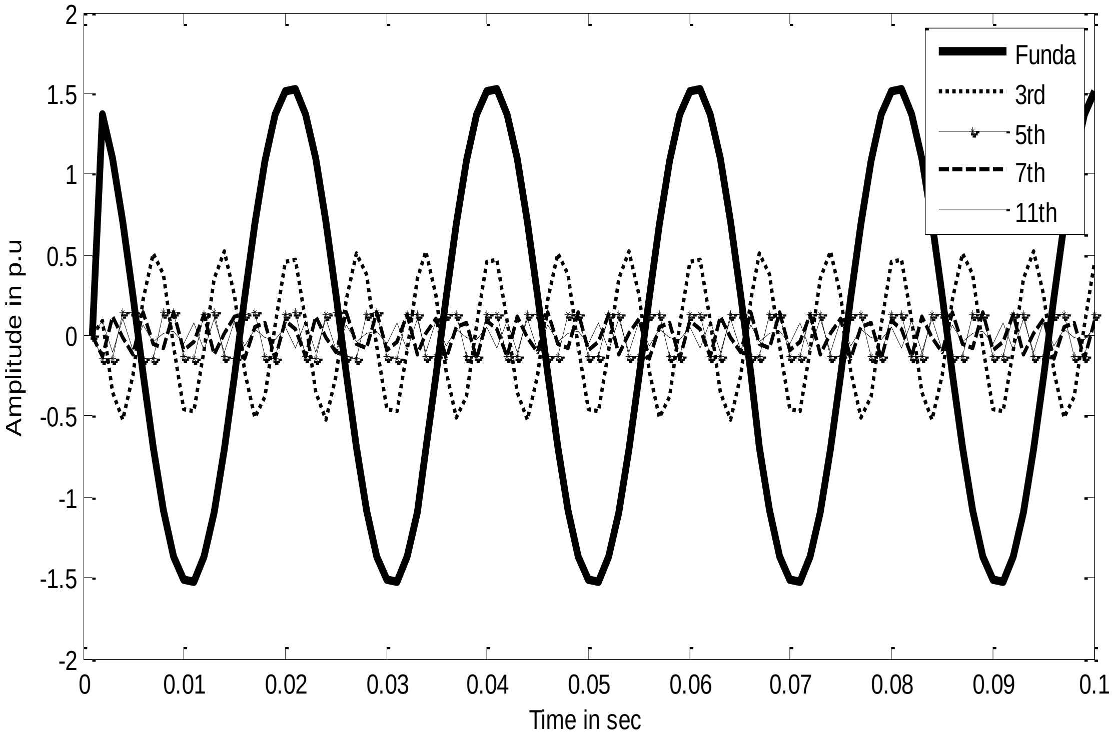

Fig 3.13 Estimation of various harmonics components of signal by LMS

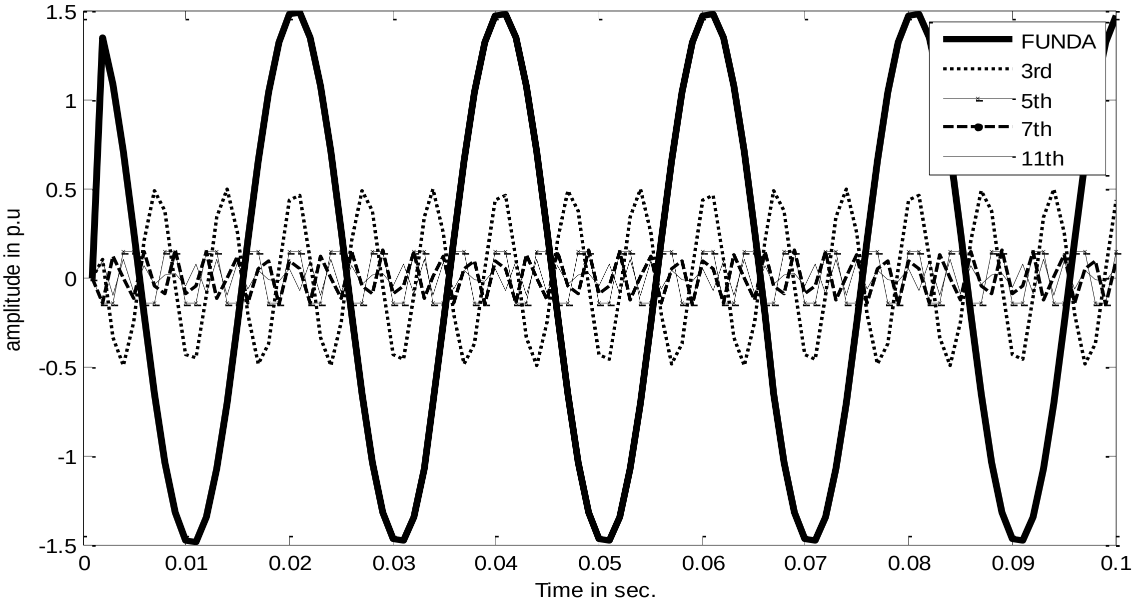

Fig 3.14 Estimation of various harmonics components of signal by VLLMS

CHAPTER 4

CONCLUSION

CONCLUSION AND FUTURE WORK

Time taken to reach steady state for the Kalman filter algorithm is a half cycle while RLS take slightly more than half cycle. The Kalman filter method gives a more accurate estimate as compared to the recursive least square method. The elapsed time for RLS algorithm for 40 dB noise is 0.114 seconds. The elapsed time for Kalman filter algorithm for 40 dB noise is 0.360 seconds. The elapsed time is minimum for VLLMS. It is 0.03180 seconds.

VLLMS gives faster convergence in comparison with RLS, KF AND LMS algorithms. It has been shown by several simulations that estimation error is less for VLLMS in comparison with RLS, KF and LMS. It is slightly computational complex as compared to the LMS. From table 4.4 VLLMS is most accurate as compared to other algorithms and convergence time of VLLMS is best among all the algorithms discussed in the thesis.

Many hybrid algorithms can be proposed in the future for minimizing the deviation error and elapsed time.

Table 4.1

Performance comparison of RLS, KF, LMS and VLLMS for 40 dB SNR

	Standard Value	RLS	KF	LMS	VLLMS
AMP1	1.5000	1.5097	1.5048	1.5296	1.4891
AMP3	0.5000	0.5016	0.4999	0.5168	0.4980
AMP5	0.2000	0.2168	0.2157	0.1962	0.2007
AMP7	0.1500	0.1698	0.1649	0.1439	0.1512
AMP11	0.1000	0.1164	0.1139	0.1036	0.1015

Table 4.2

Performance comparison of RLS, KF, LMS and VLLMS for 30 dB SNR

	Standard Value	RLS	KF	LMS	VLLMS
AMP1	1.5000	1.5136	1.5098	1.5295	1.4905
AMP3	0.5000	0.5046	0.5016	0.5166	0.4997
AMP5	0.2000	0.2167	0.2217	0.1920	0.2024
AMP7	0.1500	0.1645	0.1636	0.1438	0.1528
AMP11	0.1000	0.1152	0.1085	0.1035	0.1031

Table 4.3

Performance comparison of RLS, KF, LMS and VLLMS for 20 dB SNR

	Standard Value	RLS	KF	VSLMS	VLLMS
AMP1	1.5000	1.5199	1.5104	1.5289	1.4891
AMP3	0.5000	0.4925	0.4973	0.5160	0.4981
AMP5	0.2000	0.2021	0.1862	0.1913	0.2007
AMP7	0.1500	0.1696	0.1804	0.1431	0.1512
AMP11	0.1000	0.1115	0.1394	0.1028	0.1015

Table 4.4

Performance comparison of VLLMS in presence of harmonics

METHODS	PARAMETERS	FUNDAMENTAL	3rd	5th	7th	11th
ACTUAL	f(Hz)	50	150	250	350	550
	A(V)	1.5000	0.0005	0.2000	0.1500	0.1000
LMS	A(V)	1.5296	0.5168	0.1962	0.1439	10.360
	Deviation (%)	1.9700	3.3600	1.9000	4.7000	3.6000
RLS	A(V)	1.5097	0.5016	0.2168	0.1698	0.1164
	Deviation (%)	0.6466	0.3200	8.4000	13.200	16.400
KF	A(V)	1.5048	0.5030	0.2157	0.1649	0.1139
	Deviation (%)	0.3200	0.6810	7.8500	9.9333	13.900
VLLMS	A(V)	1.4960	0.4975	0.2001	0.1506	0.1009
	Deviation (%)	**0.2700**	**0.2500**	**0.0500**	**0.4000**	**0.9000**

REFERENCES

1. Singh, G. K. "Power system harmonics research: a survey." European Transactions on Electrical Power 19.2 (2009): 151-172.
2. Bettayeb, Maamar, and Uvais Qidwai. "Recursive estimation of power system harmonics." Electric power systems research 47.2 (1998): 143-152.
3. Subudhi, B., et al. "Parameter estimation techniques applied to power networks." TENCON 2008-2008 IEEE Region 10 Conference. IEEE, 2008.
4. Bernard Widrow, John McCool, and Michael Ball “The Complex LMS Algorithm” Proceedings of the IEEE, vol.63, no.4, pp.719-720, 1975.
5. Raymond H. Kwong, Edward W. Johnston “A Variable Step Size LMS Algorithm” IEEE Trans. on Signal Processing, vol. 40, no.2, pp.1633-1642, 1992.
6. B. Subudhi, P.K.Ray and S. Ghosh, Variable Leaky LMS Algorithm Based Power System Frequency Estimation, IET Science, Measurement & Technology, Vol. 6, issue 4, pp. 288-297, 2012.
7. Karen Kennedy, Gordon Lightbody, Robert Yacamini, “Power System Harmonic Analysis Using the Kalman Filter” IEEE Power Engineering Society General Meeting, Vol.2, pp.752-757, 13-17th July 2003.
8. E.A. Abu Al-Feilat, I. El-Amin, M. Bettayeb “Power System Harmonic Estimation a Comparative Study” Electric Power System Research, vol.29, issue 2, pp91-97, 1994.
9. B.Subudhi and P.K.Ray, “A Comparative Study on Estimation of Power System Harmonics”7th International R & D Conference, Bhubaneswar, Feb. 4-6, 2009.
10. Ray, Pravat Kumar,and B. Subudhi , “Ensemble Kalman filter based power system harmonics estimation.”Instrumentation And Measurement, IEEE Transactions on 61.12(2012):3216-3224.
11. Jin Jiang , Youmin Zhang, “A revisit to block and recursive least squares for parameter estimation” *Computers and Electrical Engineering,* Vol. 30, pp. 403-416,2004.
12. P. K. Ray, “ Signal processing and soft computing approaches to power signal frequency and harmonics estimation,” Ph.D. Dissertation, Dept. Electical Eng., National Institute of Technology, Rourkela, India, 2011.