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Least Error Square based frequency and amplitude estimation of power signal

Ku. Rohini Pradip Haridas

Assistant Professor,Electrical Engineering Department, S.S.G.M.C.E, Shegaon, India

Abstract:. An algorithm based on the least squares error method is proposed for frequency tracking and phasor estimation. The mathematical aspects of the technique are described and factors affecting the algorithm are discussed. To demonstrate the performance of the developed algorithm, computer simulated data records presented. To establish the feasibility of the proposed algorithm, it has been tested in a laboratory in offline and online environment and results of sample tests are also presented in this paper.

Keywords: LES, Window size, Phasor Technology, Pseudo inverse

I. INTRODUCTION

Frequency is the most important quantity in power system analysis, operation and control because it reflect the power system situation. Deviation of frequency from its rated value is an indication of imbalance between real power generation and load demand. Frequency higher than the nominal value indicates that the load generation is more than the load. Under such condition over frequency relays provided at the generator terminals are used to protect the generator from over speeding during starts-up. On other hand, when frequency is lower than its nominal value, indicate that the generation is less than demand. Under frequency relays are used to detect these conditions and disconnect load blocks to restore the frequency to its nominal value. Thus, frequency shows the energy balance between load and generating power. Hence accurate frequency estimation is gaining more importance ue to the increasing levels of distortions and also due to the larger variations of frequency. Furthermore, the frequency can be used as a base for estimating other parameters including the amplitude and phase of power signals. Thus reliable frequency estimation is necessary for many applications in power system.

Several digital algorithms applied to calculating frequency during recent years, for example Zero crossing based techniques [1]-[3]. Level crossing technique[4], Least squares error technique[5],[6],7Newton method[],Kalman filter[8]-[10],Prony method[11],[12] and Discrete Fourier transform[13], Adaptive neural network is the basis for another set of approaches for the estimation of power system frequency[14]-[16] , and adaptive notch filters [17] are other signal processing techniques proposed for frequency estimation in power systems. The concept of three phase locked loop (PLL) is also widely used for phase and frequency estimation [18],[19], etc. In Real time most of the aforementioned methods have tradeoff between accuracy and speed. This paper discussed practically the least error square method for estimation of frequency and magnitude of Voltages.

The organization of this paper is as follows, we described the basic principle of LES algorithm in section III. To evaluate the performance of the proposed algorithm, the voltage signal is captured from the mains power supply. To provide the isolation between the computer and power system supply, the gain control circuit is used. Data Acquisition card by Advantech is used to capture the voltage signals. These signals are recorded at a sample rate of 1000 samples /second and the algorithm was tested for different

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sampling rate by varying the time duration in section and the factors affecting the performance of the algorithm are described in section IV and experimental results are given in section V. Finally, we give a conclusion in section VI.

II. PHASOR MEASUREMENT

There are many types of transforms in engineering and all of them have one thing is common: they are used to simplify physical calculations. The phasor transform is no different .It replaces a time harmonic physical quantity with a single complex constant that can be manipulated easily by the engineer. The concept of a phasor is more than a century old and dates back to Steinmetz, who proposed complex numbers for analysis and synthesis of electrical networks with sinusoidal sources under steady state conditions. A sinusoidal signal of a known frequency f is fully described by its magnitude Vm and angular position θ with respect to an arbitrary time reference by AC waveform can be mathematically represented the equation:

$$\mathrm{V(t)} = \mathrm{Vm}\cos(wt + \theta)$$

Where;

Vm is the maximum value of the voltage.

w is the radian frequency and is equal to 2πf.

t is time in seconds.

θ is an arbitrary phase angle.

The phasor representation V of the sinusoid of (1) is given by

$$V = (\frac{Vm}{\sqrt{2}})e^{j\theta}$$

The magnitude of the phasor is the rms value of the sinusoid, and the frequency does not appear explicitly in the phasor representation; but it is an implied property of the phasor.

III. ALGORITHM FOR FREQUENCY AND AMPLITUDE ESTIMATION

Least square technique is used to estimate the amplitude and frequency of voltages and currents. It is based on the minimizing the mean square error between the actual and assumed waveforms. This section presents the basics of LES algorithm that estimate the frequency and from a voltage signal. Consider a sinusoidal signal is given by

$$v(t) = Vm\sin(wt + \theta)$$

(1)

Where; V_m is the maximum value of the voltage

w is the radian frequency and is equal to 2πf

t is time in seconds.

θ is an arbitrary phase angle.

Here aim is to estimate the time varying frequency w. suppose we sample the signal with sampling frequency of F_s So the time interval between two successive samples is equal to $\Delta t = 1/F_s$

Using the trigonometric identity,

$$\sin(2\pi ft + \theta) = \cos\theta\sin(2\pi ft) + \sin\theta\cos(2\pi ft)$$

Equation (1) can be expanded as follows:

$$v(t) = V_m\cos(\theta)\sin(2\pi ft) + V_m\sin(\theta)\cos(2\pi ft) \quad (2)$$

The Taylor series of function $f(x)$ in the neighborhood of number a is given by

$$f(x) = f(a) + \frac{f'(a)}{1!}(x-a) + \frac{f''(a)}{2!}(x-a)^2 + \frac{f'''(a)}{3!}(x-a)^3 + ... \quad (3)$$

Expanding $\sin(2\pi ft)$ and $\cos(2\pi ft)$ using Taylor series in the neighborhood of an expected value f_0 .So the first three terms of these series are given as-

$$\sin(2\pi ft) = \sin(2\pi f_0) + 2\pi t\Delta f\cos(2\pi f_0 t) - \frac{(2\pi t)^2}{2}(\Delta f)^2\sin(2\pi f_0 t) \quad (4)$$

$$\cos(2\pi ft) = \cos(2\pi f_0) + 2\pi t\Delta f\sin(2\pi f_0 t) - \frac{(2\pi t)^2}{2}(\Delta f)^2\cos(2\pi f_0 t) \quad (5)$$

Putting $\Delta f = f - f_0$ in equations (4) and (5) we get

$$\sin(2\pi ft) = \sin(2\pi f_0) + 2\pi t(f - f_0)\cos(2\pi f_0 t) - \frac{(2\pi t)^2}{2}(f - f_0)^2\sin(2\pi f_0 t) \quad (6)$$

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$$\cos(2\pi ft) = \cos(2\pi f_0) + 2\pi t(f - f_0)\sin(2\pi f_0 t) - \frac{(2\pi t)^2}{2}(f - f_0)^2\cos(2\pi f_0 t) \quad (7)$$

Substituting Equations (6) and (7) in Equation (2) we get

$$v(t) = V_m(\cos\theta)\left[\sin(2\pi f_0 t) + 2\pi t(f - f_0)\cos(2\pi f_0 t) - (f - f_0)^2\sin(2\pi f_0 t)\right] + V_m(\sin\theta)\left[\cos(2\pi f_0 t) - 2\pi t(f - f_0)\sin(2\pi f_0 t) - \frac{(2\pi t)^2}{2}(f - f_0)^2\cos(2\pi f_0 t)\right] \quad (8)$$

Now, putting $(f - f_0)^2 = (f^2 - 2ff_0 + f_0^2)$ in Equation (8) and rearranging the terms we get,

$$\begin{aligned} v(t) &= [\sin(2\pi f_0 t)]V_m\cos\theta + [2\pi t\cos(f_0 2\pi t)](f - f_0)V_m\cos\theta + [\cos(f_0 2\pi t)]V_m\sin\theta + \\ &[[-2\pi t\sin(f_0 2\pi t)](f - f_0)V_m\sin\theta] + [t^2\sin(f_0 2\pi t)]\left(-\frac{(2\pi)^2}{2}f^2 + (2\pi)^2 ff_0 - \frac{(2\pi)^2}{2}f_0^2\right) \\ &V_m\cos\theta + [t^2\cos(f_0 2\pi t)]\left(-\frac{(2\pi)^2}{2}f^2 + (2\pi)^2 ff_0 - \frac{(2\pi)^2}{2}f_0^2\right) + V_m\sin\theta \end{aligned} \quad (9)$$

If samples of voltage are obtained at equal time interval, Δt seconds. a set of m samples may be designated as $v(t_1), v(t_1 + \Delta t), v(t_1 + 2\Delta t), \ldots\ldots\ldots v(t_1 + m\Delta t)$ where t_1 is an arbitrary time reference.The voltage sampled at $t = t_1$ can now be expressed by substituting t_1 for t ,similarly the next voltage sample,taken at time $t_2 = (t_1 + \Delta t)$ and so on…in Equation (9) and rearranging the terms we get;

$$v(t_1) = a_{11}x_1 + a_{12}x_2 + a_{13}x_3 + a_{14}x_4 + a_{15}x_5 + a_{16}x_6 \quad (10)$$

$$v(t_2) = a_{21}x_1 + a_{22}x_2 + a_{23}x_3 + a_{24}x_4 + a_{25}x_5 + a_{26}x_6$$

…

$$v(t_m) = a_{m1}x_1 + a_{m2}x_2 + a_{m3}x_3 + a_{m4}x_4 + a_{m5}x_5 + a_{m6}$$

Here all the x's are unknowns and are the functions of V_m, θ, f, f_0.Also t_1 is the time reference and the voltage is sampled at a pre-selected rate Δt is known $v(t_1)\ldots\ldots v(t_m)$ are measured inputs and all a 's can be calculated.

We can rewrite equation (10) in the matrix form as follows:

$$[A]_{m*6}[X]_{6*1} = [V]_{m*1} \quad (11)$$

To determine six unknowns, at least six equations are needed.In other words, at least six samples of voltage would required. The values of x 's unknown can be find by pre-multiplying both the sides of equation (11) with left pseudo-inverse of $[A]$

$$[X] = [A]^{\dagger}[V] \quad (12)$$

Where $[A]^{\dagger} = [[A]^T[A]]^{-1}[A]^T$

Out of all the elements of the vector $[X]x_1, x_2, x_3, x_4$ are of interest for calculating the system frequency. There are two possible approaches for calculating the change in frequency, first one is is to use x_1 and x_2

$$\frac{x_2}{x_1} = \frac{(f - f_0)V_m\cos\theta}{V_m\cos\theta} = f - f_0 \quad (13)$$

Another possible approach is to estimate frequency change using variables x_3 and x_4 as follows.

$$\frac{x_4}{x_3} = \frac{(f - f_0)V_m\sin\theta}{V_m\sin\theta} = f - f_0 \quad (14)$$

Frequency of the sampled signal is therefore given by

$$f = f_0 + \frac{x_2}{x_1} \quad \text{or} \quad f = f_0 + \frac{x_4}{x_5}$$

The pseudo-inverse matrix $[A]^{\dagger} = [[A]^T[A]^{-1}][A]^T$ can be considered a filter,which will amplify or suppress noise depending on its coefficients.Both the sampling rate and the window size play rather important roles in determining the pseudo-inverse matrix

IV. EXPERIMENTAL RESULTS

A. Experimental setup

The block diagram of experimental set up for study is as shown in figure 1

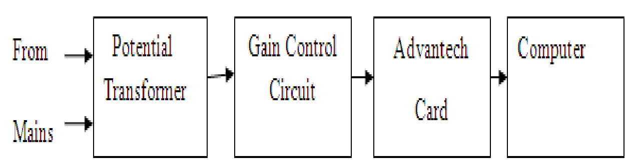

Figure 1 Block diagram for experimentation

The potential transformer is used to step down the voltage level. The voltage ratio of 230V/6V is used for potential

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transformer. This gain control circuit reduces the input voltage by ½. Data acquisition card by Advantech instrument is used to capture the voltage signal. These signals are recorded at a sample rate of 850,1000,1200,2500,and 5000 samples/sec. The two important components used in the hardware are described as follows:

B. *Gain control circuit*

The gain control circuit is necessary to prevent the clamping of input voltage signal and A simple closed loop op-amplifier used as an inverting amplifier as shown in figure 2

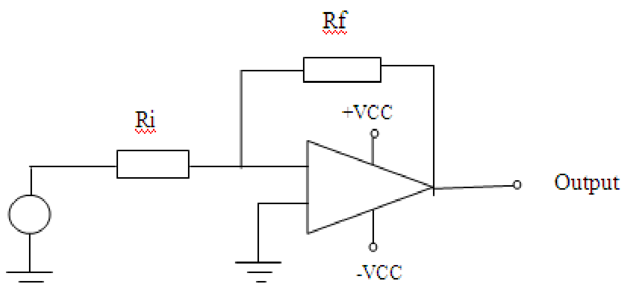

Figure 2 A simple closed loop op-amplifier used as an inverting amplifier

The output of a inverting amplifier is given as

$$V_0 = \frac{-R_f}{R_i} * V_{in}$$

(15)

Where,

R_f = Feedback resistor

R_i = Input resistor

Gain = $\frac{-R_f}{R_i}$

V_{in} = input voltage

$V_{(0)}$ = output voltage of operational amplifier

The gain of 0.5 is achieved by choosing the values of $R_f = 5k\Omega$ and $R_i = 10k\Omega$. The input voltage at the operational amplifier is 9V and voltage available at output is 4.5 V. The experimental setup is as shown in figure 3

Figure 3 Experimental Set Up For Online Implementation

C. *Data collection*

To evaluate the developed algorithm, the voltage signal is taken from the mains. The potential transformer is used to reduce the voltage level. Further the gain control circuit is used to reduce the voltage by the ratio of ½.The output of the gain control circuit is given to Advantech card. These signals are recorded at various sampling rate such as 850, 1000, 1200, 2500, 5000, samples/sec as shown in figure 4

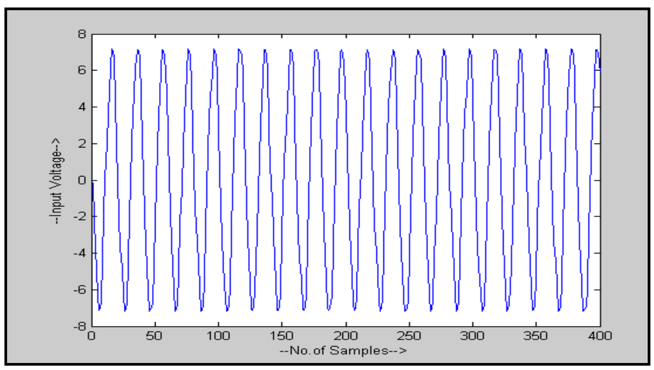

Figure 4 Data captured for offline analysis for sampling rate of 1000 samples/sec

D. *Implementation of algorithm*

The algorithm is tested in offline as well as in online condition by varying the sampling rate such as 850 Hz, 1000Hz, 1200Hz, 2500Hz and 5000Hz for durations of 0.04, 0.08 and 0.12 sec. Thus, the number of samples per cycle is given by N=Fs*duration. The matrix A for six unknowns for sampling rate of 100Hz and duration is 0.04 sec is obtained Then the pseudo inverse of matrix A is calculated. The matrix x is calculated and frequency and magnitude of the signal is estimated by using given formulas.

The various factors affect the performance of the algorithm like data window size, sampling rate and level of truncation

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of Taylor series expansion. For this purpose the coefficients of left pseudo-inverses were calculated for different cases. The sum of the squares of the rows of the left pseudo-inverse of coefficient matrix is given by

$$\sum S_i^2 = (S)'(S)$$

Where,

S is the $N \times i$ column vector of matrix Ainv

$\sum S_i^2$ is the sum of squares of squared values from vector

E. Data window size

The effect of varying the data window size is investigated. The coefficients of left pseudo-inverses were calculated for different cases. Here the first three terms of the Taylor series expansion of sine and cosine functions are used. The combinations of parameters used in analysis is given in table no.1

Table No. 1 List of The Combination Of Parameters Used In Testing Of An Algorithm

Sr.No	Sampling Frequency (Hz)	Data window (Sec)
1	1000	0.02
2	1000	0.04
3	1000	0.06
4	1000	0.08

The sum of the squares of the rows of the left pseudo-inverse of coefficient matrix for the above combination are given in table no. 2 From this table it is shown that all most all the elements of the 1st and 3rd rows are numerically less than 1.0. If the filter would suppress or amplify the noise is determined by the magnitudes of coefficients of filter. Sum of the squares of filter coefficients is the measure of noise amplification. If the sum is greater than the noise is amplified however if the sum is less than 1.0 the noise present in the input is suppressed. The graph is plotted between no. of samples and sum of squares of the 1st row elements and 2nd row elements as shown in figures 5.

Table No.2 Sum of Squares of the Row Elements

No. Of Sample	S1	S2	S3	S4
20	6.74	1889.66	2.56	10056.6
40	0.59	158.54	0.55	206.37
60	0.33	45.73	0.33	51.17
80	0.21	19.32	0.23	19.55

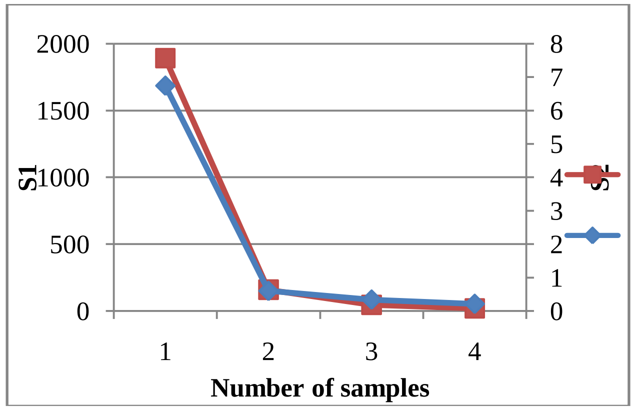

Figure 5 S1 and S2 as a function of window size; Sampling rate=1000Hz

From the above figures, it shows that the sum of the squares of the row elements (S1 and S2) reduce as the number of samples are increased i.e. the size of data window size is increased. Thus we can say that the effect of noise on the frequency measurement would be reduced if more number of samples in other words wider data window is used. One disadvantage of increasing the number of samples per window increases the computation time.

F. Sampling rate

In this, the effect of different sampling rates on the performance of algorithm is studied. For this purpose the pseudo-inverse matrices for different sampling rates such as 1000,1200,2500,5000 Hz are calculated. Also the sums of squares of row elements are calculated. Again here the first three terms of the Taylor series expansion of sine and cosine functions are used .The number of samples used is 20.Here the graph is plotted between sum of squares of 1st row of pseudo inverse matrices for different sampling rates. The graph is as shown in figure 6. From the figure, it shows that increasing the sampling rate improves the noise immunity characteristics. However the suitable combination of sampling rate and data window size is needed to be selected.

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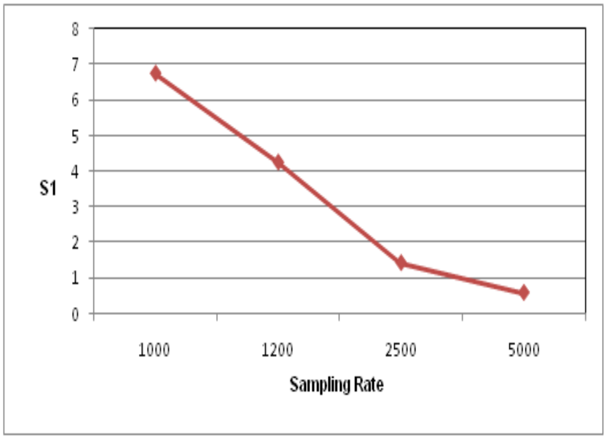

Figure 6 S1 as a function of sampling rate

G. Expansion of Taylor series

the algorithm is developed for 3 terms of Taylor series expansion. However the algorithm is tested for more number of Taylor series expansions like 4 and 5. Thus the numbers of unknowns are increased to 8 and 10 respectively. The few terms of Taylor series expansion will give reasonable accuracy if the data window size used is small. However if the window used is wider then the more number of Taylor series expansion terms are needed to get reasonable accuracy. The values for sum of squares of the first four rows for different unknowns are given in the table no.3. Here the constant window size 40 is used. From the table it is shown that the sum of squares of the first four rows for each row increases as the numbers of Taylor series expansion terms are increased. If the more terms of Taylor series expansion is used then the size of window size may have to be increased in order to reduce the susceptibility to noise.

Table No.3 Sum of squares of the row elements for different unknowns

No.of Unknowns	S1	S2	S3	S4
6	0.11	2.46	0.10	2.6
8	0.24	17.20	0.24	16.94
10	0.40	80.56	0.44	80.02

V. EXPERIMENTAL RESULTS

The offline and online analysis results of LES-basic approach method for estimating the frequency and peak value of voltage by varying the number of samples per cycle, sampling rate such as 850Hz,1000Hz,1200Hz,2500Hz and 5000Hz and the number of Taylor series expansion terms is presented below. Here only the results with number of unknown six is presented.

Table No. 4 List of the combination of parameters used in testing of LES algorithm

Case No.	Number Of Unknowns	Duration (WW)		
I	6	0.04	0.08	0.12
II	8	0.04	0.08	0.12
III	10	0.04	0.08	0.12

Table No.5 Tabular form for Case No.I-Online, Number of Unknowns=6,duration=0.04

Sampling Frequency (Hz)	Estimated Frequency (Hz)			Estimated Magnitude (V)		
	Min	Max	Avg	Min	Max	Avg
850	42.92	81.8	50.03	6.6	7.18	6.92
1000	45.34	60.2	49.83	6.6	7.21	6.94
1200	45.76	60.6	49.92	6.7	7.24	6.98
2500	33.30	49.8	49.85	6.96	7.22	6.96
5000	-165.2	64.6	48.17	6.95	7.44	7.20

Table No.6 Tabular form for Case No. I-Online, Number of Unknowns=6,duration=0.08

Sampling Frequency (Hz)	Estimated Frequency (Hz)			Estimated Magnitude (V)		
	Min	Max	Avg	Min	Max	Avg
850	36.01	49.93	50.72	6.84	7.08	6.95
1000	15.58	61.84	49.63	6.85	7.12	6.98
1200	47.47	51.68	49.97	6.85	7.10	6.97
2500	46.48	54.84	49.95	7.08	7.34	7.21
5000	40.25	50.08	49.83	7.11	7.21	7.34

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Table No. 7 Tabular form for Case No I-Online, Number of Unknowns=6,duration=0.12

Sampling Frequency (Hz)	Estimated Frequency (Hz)			Estimated Magnitude (V)		
	Min	Max	Avg	Min	Max	Avg
850	45.74	161.11	51.20	6.94	7.14	7.02
1000	49.89	52.91	50.34	6.91	7.20	7.04
1200	49.10	51.2	50.13	6.93	7.09	7.01
2500	49.0	51.0	49.88	7.04	7.27	7.15
5000	49.4	50.8	49.92	7.07	7.22	7.14

The results in tabular form are given in table No.5.6 and 7 respectively. From the table No.5.It is observed that, for the duration of 0.04 at sampling rate of 1200Hz, gives better results. It gives minimum peak at 45.76Hz and maximum at 60.63 Hz but average value of estimation is 49.92 Hz. Similarly for magnitude estimation, it gives minimum value at 6.70 V and maximum value at 7.22 V but average value of magnitude estimation is 6.98V. Similar results are obtained for duration 0.08 and 0.12. Some of the Matlab results are as shown in following figures:

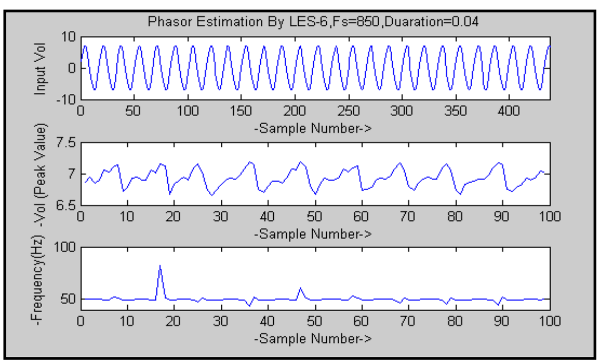

Figure 7 LES, Number of Unknowns=6, Duration=0.04 Fs=850 Hz

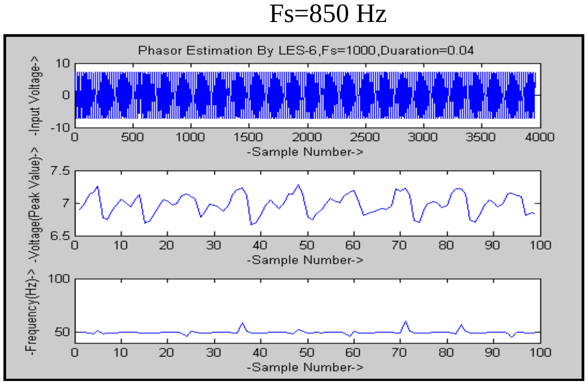

Figure 8 LES, Number of Unknowns=6, Duration=0.04 Fs=1000 Hz

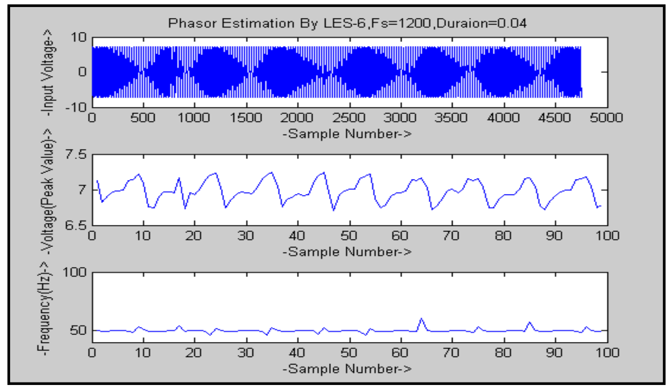

Figure 9 LES, Number of Unknowns=6, Duration=0.04 Fs=1200 Hz

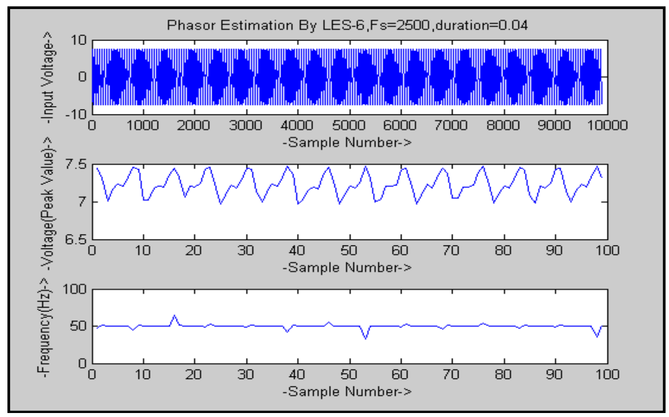

Figure 10 LES, Number of Unknowns=6, Duration=0.04,Fs=2500Hz

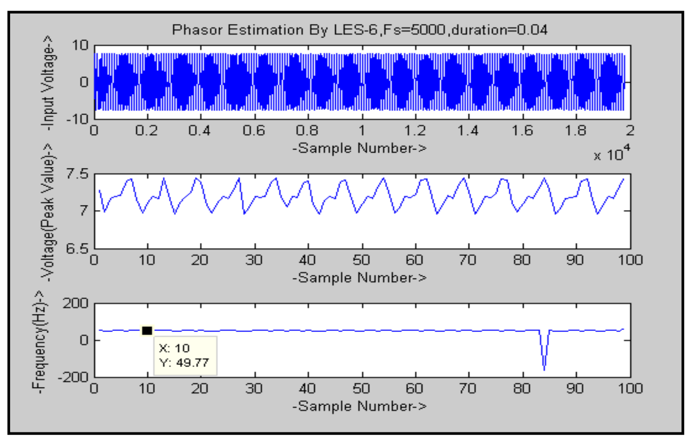

Figure 11 LES, Number of Unknowns=6, Duration=0.04 Fs=5000 Hz

VI. CONCLUSION

The Least Error Square (LES) method gives information about the magnitude and frequency of input signal but does not give information about phase angle. It assumes that

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signal is pure sine wave of fundamental frequency. The factors such as data window size, sampling rate, truncation of Taylor series affects the performance of this algorithm. The algorithm is tested for more number of Taylor series expansions like 4 and 5. Thus the numbers of unknowns are increased to 8 and 10 respectively. The few terms of Taylor series expansion will give reasonable accuracy if the data window size used is small. However if the window used is wider then the more number of Taylor series expansion terms are needed to get reasonable accuracy. From experimental results it is observed that the effect of noise on the frequency measurement would be reduced if more number of samples in other words wider data window is used. One disadvantage of increasing the number of samples per window increases the computation time. Again it shows that increasing the sampling rate improves the noise immunity characteristics. However the suitable combination of sampling rate and data window size is needed to be selected. From the experimental results it is observed that sampling rate of 1200Hz for the duration 0f 0.12 sec when number of unknowns used are 6 gives good results. To obtain the better results filtering is must.

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BIOGRAPHY

Rohini Haridas received her B.E. and M.E. from the S.G.B.University of Amravati, India in 2011 and 2013 respectively in Electrical Power system Engineering. She is Assistant Professor in Departmaent of Electrical Engineering S.S.G.M.C.E.SHEGAON. She is Life Member of ISTE.