ESTIMATION OF
POWER SYSTEM FREQUENCY

SAURAV KUMAR BENGANI (109EE0261)

Department of Electrical Engineering
National Institute of Technology Rourkela

ESTIMATION OF POWER SYSTEM FREQUENCY

A Thesis submitted in partial fulfillment of the requirements for the degree of
Bachelor of Technology in "Electrical Engineering"

By
SAURAV KUMAR BENGANI
109EE0261

Under guidance of
Prof. Pravat Kumar Ray

Department of Electrical Engineering
National Institute of Technology
Rourkela-769008 (ODISHA)
May-2013

DEPARTMENT OF ELECTRICAL ENGINEERING
NATIONAL INSTITUTE OF TECHNOLOGY, ROURKELA- 769 008
ODISHA, INDIA

CERTIFICATE

This is to certify that the draft report titled "**Estimation of Power System Frequency**", submitted to the National Institute of Technology, Rourkela by **Mr. Saurav Kumar Bengani, Roll No**: **109EE0261** for the award of **Bachelor of Technology** in **Electrical Engineering** is a bona fide record of research work carried out by him under my supervision and guidance.

The candidate has fulfilled all the prescribed requirements.
The draft report which is based on candidate's own work has not been submitted elsewhere for a degree/diploma.

In my opinion, the draft report is of standard required for the award of a Bachelor of Technology in Electrical Engineering.

Prof. P. K. Ray
Supervisor
Department of Electrical Engineering
National Institute of Technology
Rourkela – 769 008 (ODISHA)

ABSTRACT

Frequency is an important parameter in power system for indicating the dynamic balance between amount of power generated and amount of power consumed. Small changes in system frequency can affect the operation of a power system in a great manner. Thus, estimating the frequency is one of the most important tasks in the power system operation. A least mean square (LMS) algorithm with varying step size to estimate the frequency is studied in this report and its performance under the influence of harmonics, noise, unbalanced conditions, step changes in frequency, etc is observed through various simulations. Next, Frequency estimation is carried out with an improved Non-linear Least Squares (NLS) method in which frequency is estimated through a 1-D search over a specified range. The validity of method under different conditions including line to ground fault is also tested. This method is also verified using the data obtained from an experimental setup consisting of a non-linear load.

ACKNOWLEDGEMENT

Any project is incomplete without mentioning the names of those persons without whom the project would not have been a success. First of all, I would like to take this opportunity to express my profound gratitude to Prof. Pravat Kumar Ray, Department of Electrical Engineering, National Institute of Technology Rourkela for his valuable guidance and continuous support that helped a lot in completion of this project.

I would also like to express my gratitude to all the teaching and non-teaching staff for their help and co-operation at various stages of the project.

I am highly indebted to M-tech and PhD Scholars, Miss Raseshwari Pradhan, Miss Rosy Pradhan and others who were always there to help me out, no matter whatever the problem be. Last, but not the least I would like to thank my parents for always supporting and encouraging me.

Saurav Kumar Bengani

109EE0261

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF FIGURES (CONTD.)

CHAPTER 1

INTRODUCTION

Frequency is the number of cycles per second. It gives the per second power output from stator to rotor (in case of a motor) or from rotor to stator (in case of a generator). In most parts of the world, 50Hz frequency is used while in USA, a frequency of 60Hz is used. But due to technological developments and increasing power demand, frequency is deviated from its nominal value and therefore, there is a need to control the frequency. But to control the frequency, one needs to determine the frequency first. Although it is difficult to accurately determine the frequency, one can estimate the approximate frequency by considering various constraints.

1.1 Motivation

Frequency is one of the most important and sensitive parameter of power system. Any variation in power system is eventually reflected in the change of system frequency.

1. A change in frequency leads to a change in system reactance and the operation of several relays such as reactance relays is affected.
2. The electric clocks are driven by synchronous motors whose speed depend on the frequency of the system.
3. Reduced frequency may lead to damage of turbines, stalling of generators, shut down of power plants, damaging of transformers, etc.
4. Frequency is also a measure of mismatch between power generation and load demand. if demand is greater than generation, under-frequency situation arises. If generation is greater than demand, over-frequency situation arises. In either case, change in frequency poses a threat to efficiency and safety of entire system and the chances of collapse of system increase.

Thus, Frequency is an integral part of power system protection, power quality monitoring and operation and control of devices using digital technologies. Hence, the accurate estimation and tracking of system frequency is of utmost importance.

1.2 Thesis Objectives

1. To study the Least Mean Square (LMS) technique of frequency estimation and observe its performance under different practical conditions.
2. To study the Non-Linear Least Squares (NLS) technique of frequency estimation and observe its performance under different practical conditions.
3. To modify the NLS technique by overcoming its disadvantages.

4. To validate the modified NLS method by observing its performance under different practical conditions and comparing the results obtained by those of the older NLS method.
5. To acquire data from an experimental setup and observe the performance of the methods.
6. To compare both the linear and non-linear techniques and conclude.

1.3 Organization of Thesis

The entire thesis has been divided into five chapters including the chapter of Introduction.

Chapter2 gives an idea about the existing techniques of frequency estimation. An in-depth knowledge about the concept of the Least Mean Square (LMS) technique and the Non-Linear Least Squares (NLS) technique is provided.

Chapter3 describes the methodology adopted, the steps followed, the mathematical calculations involved in both the Least Mean Square (LMS) and Non-Linear Least Squares (NLS) technique. The proposed modification is also discussed.

Chapter4 presents the various MATLAB simulations that have been performed. The behavior of LMS, NLS and modified NLS techniques in presence of noise, harmonics, step changes in amplitude, sudden frequency jumps and normal condition is presented. The performance of modified NLS method under fault condition is studied. The validity of modified NLS method is also confirmed by carrying out the frequency estimation from data obtained through a laboratory setup.

Chapter5 discusses the advantages of NLS method over LMS method. The performance of the old as well as modified NLS method is compared. The effect of choice of window length on the performance of NLS method is also discussed. Finally, the work done is concluded.

CHAPTER 2

BACKGROUND AND LITERATURE REVIEW

Due to the development of several electronic and other non-linear devices, the present system is subjected to several undesirable conditions. The system is subject to harmonics, noise, etc. The demand for more and more power is forcing the power systems to operate much closer to their limits and system is prone to several transient and abnormal conditions, noise, harmonics, etc; offline studies are not of much help as every operation needs to be done on the go. Keeping in mind all the undesirable conditions, several methods have been proposed for estimating the frequency online. But all the methods have a trade-off between estimation accuracy, speed of convergence, robustness to noise and sampling rate. In the past forty years, several frequency estimation techniques have been developed with each having its own advantage and disadvantage. Most of these techniques use digitized sample of system voltages. Traditionally, the frequency is estimated by the time between two zero crossings[1] as well as by the calculation of the number of cycles. The voltage waveform was assumed to be pure sinusoid and accordingly, the time between consecutive zero crossings gave the frequency. However, this method fails for distorted signals. Some of the other techniques that have been developed include:

1. Variance reduction method[2] in which a stable band-pass second degree digital integrator (BPSDDI) is used with variance reduction algorithm to estimate the frequency.
2. Orthogonal filters method[3] in which the voltage signal is decomposed into two orthogonal components and using mathematical calculations and simplifications, system frequency is estimated.
3. Kalman filtering[4] which includes both linear and non-linear approaches to accurately estimate the frequency in presence of harmonics and noise.
4. Soft computing techniques like neural network and genetic algorithms are also used for frequency estimation[5,6].

Among these methods, some methods strictly depend on the performance of filters for filtering out the harmonic content. Some methods are based on the static sinusoid signals and hence, cannot perform well under dynamic conditions while other methods which rely on zero-crossing techniques are unable to perform under unbalanced and fault conditions. Methods are divided into two groups:

- Time domain based approach, like zero-crossing technique.
- Frequency domain based approach, like Fourier transform.

Following sections describe a Least Mean Square (LMS) technique[5] of frequency estimation. It is structurally simple, computationally efficient and robust. Another method employing a 1-D search with a small modification is also discussed. This method is Non-Linear Least Squares (NLS) method.

2.1 Least Means Square (LMS) Algorithm[7]:

The LMS approach of signal estimation is shown in the fig.1, where X_k is the input data vector at the *k*th instant, y_k is to be the desired signal, and $\hat{y}_k$ represents its estimate. The signal can be estimated correctly by the filter with a suitable value of its coefficient W_k, which is obtained through minimizing the square of the error signal e_k. Hence, at each iteration the filter coefficient is updated. So, this is an adaptive filter. The weight vector at each iteration is given by:

$$W_{k+1} = W_k + 2\mu e_k X_k$$

Where μ is the adaptation parameter or step size.

The algorithm is initialized by setting all coefficients to zero or any other starting value. e_k is found out. Consequently, W_k is modified and this cycle is performed until steady state conditions are obtained. It has been seen that a time varying step size provides better and faster convergence. Hence, in this technique we use a time varying step size.

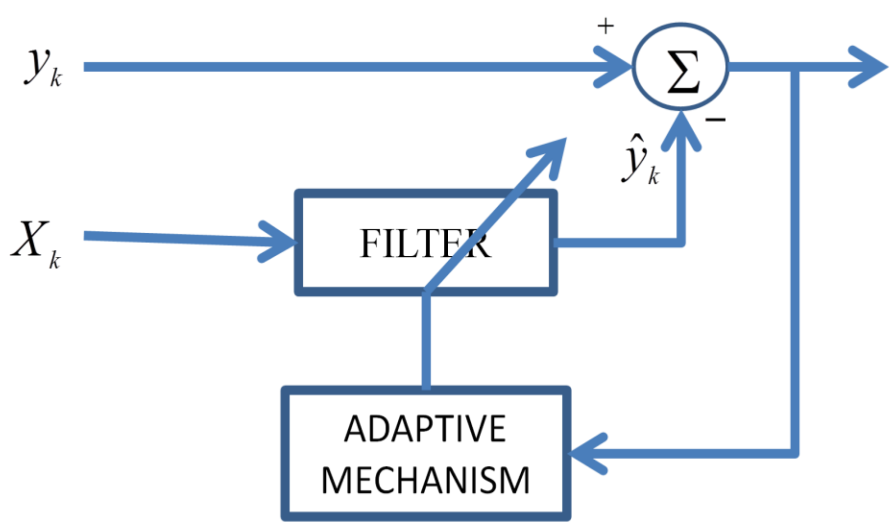


Fig 2.1 LMS Filter structure

In this technique, three phase voltages are first converted into complex form by $\alpha\beta$-transformation and LMS algorithm is applied upon them. Additional changes introduced are:

1. A Variable step size is selected so as to overcome the convergence problem and perform estimation faster.
2. To reduce noise disturbance, the step size is adjusted in accordance with the square of time averaging estimate of consecutive error samples.

Assumptions made are:

1. Signal remains constant during observation period.
2. Effect of negative sequence component is not considered.

2.2 Non-Linear Least Squares (NLS) Technique[11]:

In this technique, the estimate of frequency is obtained by minimizing the squared error norm between the actual signal and the estimated signal. First, the signal is sampled at a frequency higher than the frequency of highest harmonic present. The signal is modeled by Fourier series and a signal model is developed in such a way which eliminates the amplitude terms of Fourier series, thereby leaving only one unknown parameter which is the frequency. Then, the frequency is estimated by performing a 1-D search over a range of frequencies. The value which gives the lowest error norm is taken to be the estimated frequency. The fineness of the search range can be decided by the accuracy desired. In this technique, the dc offset component and triplen harmonics have not been considered while developing the signal model.

CHAPTER 3

METHODOLOGY

3.1 LMS Method[7]:

Three phase voltages can be discretized and represented as:

$$V_{a_k} = V_m \cos(\omega k \Delta T + \Phi) + \varepsilon_{a_k}$$
$$V_{b_k} = V_m \cos(\omega k \Delta T + \Phi - \frac{2\pi}{3}) + \varepsilon_{b_k} \quad (1)$$
$$V_{c_k} = V_m \cos(\omega k \Delta T + \Phi + \frac{2\pi}{3}) + \varepsilon_{c_k}$$

Where V_m = peak value of fundamental component.

ε_k = noise term.

ΔT = sampling interval.

$\Phi =$ Phase of fundamental component.

$\omega = 2\pi f =$ Frequency of fundamental signal.

Converting into complex form by $\alpha\beta$-transform:

$$\begin{bmatrix} V_{\alpha_k} \\ V_{\beta_k} \end{bmatrix} = \sqrt{\frac{2}{3}} \begin{bmatrix} 1 & -0.5 & -0.5 \\ 0 & 0.866 & -0.866 \end{bmatrix} \begin{bmatrix} V_{a_k} & V_{b_k} & V_{c_K} \end{bmatrix}^T \quad (2)$$

A complex voltage can be obtained from the above as:

$$V_k = V_{\alpha_k} + jV_{\beta_k} \quad (3)$$

or $\quad V_k = Ae^{j(\omega k \Delta T + \Phi)} + \varepsilon_k$

or $\quad V_k = \hat{V}_k + \varepsilon_k \quad (4)$

Where $\hat{V}_k$ is the estimated value of voltage at *k*th instant.

ε_k is the noise component.

Now,

$$\hat{V}_k = Ae^{j(\omega k\Delta T+\Phi)} = Ae^{j(\omega(k-1)\Delta T+\Phi)}e^{j(\omega\Delta T)}$$

$$\Rightarrow \hat{V}_k = \hat{V}_{k-1}e^{j(\omega\Delta T)}$$
$$\Rightarrow \hat{V}_k = W_k\hat{V}_{k-1} \quad (5)$$

This model has 1 input vector element and 1 weight matrix element.

For next iteration,

$$W_{k+1} = W_k + \mu_k e_k \hat{V}_k^* \quad (6)$$

Where * represents the complex conjugate.

μ_{k+1} is found out by: $$\mu_{k+1} = \lambda\mu_k + \gamma p_k p_k^* \quad (7)$$

Where $$p_k = \rho p_{k-1} + (1-\rho)e_k e_{k-1} \quad (8)$$

Where $0<\rho<1$; $0<\lambda<1$; $\gamma>0$ and $\mu_{\min} < \mu_k < \mu_{\max}$ for better convergence.

At any instant,

$$W_k = e^{j\omega\Delta T}$$

$$\Rightarrow \hat{f}_k = \frac{1}{2\pi\Delta T}\sin^{-1}[\mathrm{Im}(W_k)] \quad (9)$$

3.2 NLS Method[11]:

Any periodic signal can be represented by Fourier series. Assuming the dc component to be zero and the signal *f(t)* is known at *M* uniformly sampled points, we get the following set of *M* equations:

$$f(t_k) \approx \sum_{n=1}^{N}(a_n\cos n\omega_0 t_k + b_n \sin n\omega_0 t_k) \quad (10)$$

Where k=0,1, … , M-1.

We know that waveforms with half-wave symmetry do not contain even harmonics. Also, in three-wire systems, triplen harmonics are absent. Hence, $n \in \{1,5,7,11,\ldots,n_h\}$. Let N_a denote the total number of harmonics present. Then, we can write *f(t)* in matrix notation as follows:

$$Hx \approx y \tag{11}$$

where

$$H = [H_a \quad H_b]$$

$$H_a = \begin{bmatrix} \cos\omega_0 t_1 & \cos 5\omega_0 t_1 & \ldots & \cos n_h \omega_0 t_1 \\ \cos\omega_0 t_2 & \cos 5\omega_0 t_2 & \ldots & \cos n_h \omega_0 t_2 \\ \vdots & \vdots & \vdots & \vdots \\ \cos\omega_0 t_M & \cos 5\omega_0 t_M & \ldots & \cos n_h \omega_0 t_M \end{bmatrix}_{M \times N_a} \tag{12}$$

$$H_a = \begin{bmatrix} \sin\omega_0 t_1 & \sin 5\omega_0 t_1 & \ldots & \sin n_h \omega_0 t_1 \\ \sin\omega_0 t_2 & \sin 5\omega_0 t_2 & \ldots & \sin n_h \omega_0 t_2 \\ \vdots & \vdots & \vdots & \vdots \\ \sin\omega_0 t_M & \sin 5\omega_0 t_M & \ldots & \sin n_h \omega_0 t_M \end{bmatrix}_{M \times N_a} \tag{13}$$

$$x = \begin{bmatrix} a_1 & a_5 & \ldots & a_{n_h} & b_1 & b_5 & \ldots & b_{n_h} \end{bmatrix}^T_{1 \times 2N_a}$$

$$y = \begin{bmatrix} f(t_1) & f(t_2) & \ldots & f(t_M) \end{bmatrix}^T_{1 \times M} \tag{14}$$

Thus, it can be easily seen that minimum number of samples must be equal to $2N_a$. But normally, M should be greater than this value for better results. Now, x is given by:

$$x \approx (H^T H)^{-1} H^T y$$

The coefficients and frequency are unknown. We need to estimate only the frequency. The coefficients can be eliminated in the following manner.

$$H(H^T H)^{-1} H^T y \approx y \tag{15}$$

And the error vector is given by:

$$e = [I - H(H^T H)^{-1} H^T]y \tag{16}$$

This error is a function of frequency only. The value of frequency which minimizes the square of error norm is taken to be the estimated frequency. Let,

$$I - H(H^T H)^{-1} H^T = A \tag{17}$$

According to the modified NLS method, this ***A*** matrix is made to adjust according to the sampling instant. The order of this matrix depends on the number of samples being considered. One cannot go on increasing the samples due to computational and storage considerations. Hence, a window of optimum length is selected to accommodate the required samples. Let the window length be "$\boldsymbol{W_l}$ " and sampling frequency be " $\boldsymbol{f_s}$ " (frequency greater than highest harmonic frequency for proper sampling). So, sampling instant, $\boldsymbol{t} = 1/f_s$ and optimum number of samples, $\boldsymbol{k} = W_l / t = W_l * f_s$.

This must also be the final size of matrix ***A***. The older method begins to estimate frequency once the minimum number of samples is equal to ***k***. This causes delay.

The modified method makes the matrix ***A*** a variable one. As a signal is applied, the numbers of samples begin to increase and so is the size of ***A*** till its order becomes equal to ***k***. Once this happens; at the next instant, a row of new values is added to ***A*** and the row of oldest instant values is deleted, thereby, keeping the order of ***A*** at constant value ***k***. Thus, frequency can be estimated with lesser delay.

CHAPTER 4

SIMULATION RESULTS

4.1 LMS Method:

In the MATLAB programming environment with a sampling rate of 5 kHz, different simulations were performed. The values taken were ρ = 0.99, λ=0.97, γ =0.000765, $p_{initial} = 0$ and $0 < \mu_k < 0.1$. In each case, the signals assumed are standard 3 phase sinusoidal signals.

1. Estimation of 49.5 Hz signal contaminated by a noise signal with SNR of 40dB:

The response of the method in presence of noise was observed by adding a white Gaussian noise of 40dB to the signal. The filter was initialized at 50Hz with μ_k = 0.00445. It was seen that the frequency was estimated within 0.04 seconds. Estimated frequency = 49.5052 Hz. Error = 0.0052 Hz. The performance of the method is shown in Fig 4.1.

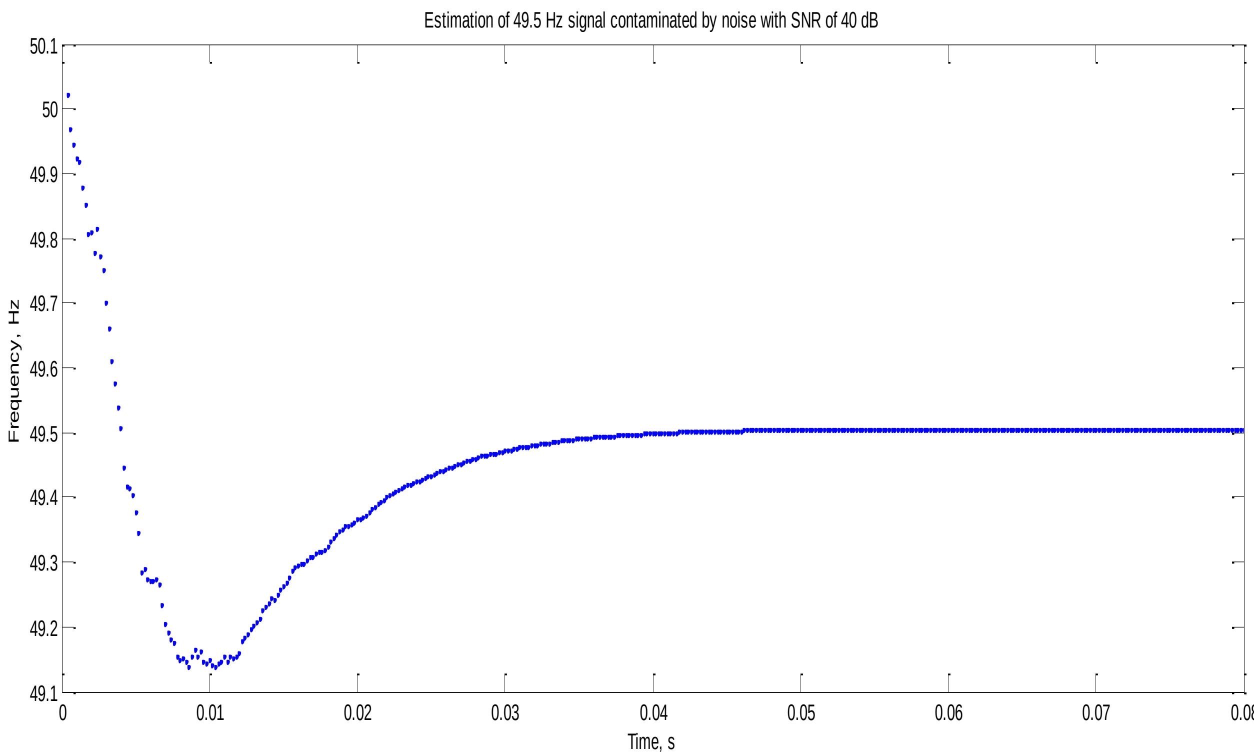


Fig 4.1 Estimation of 49.5 Hz signal contaminated by noise with SNR of 40dB

2. Estimation of 50.5 Hz signal with LMS initiated at 50 Hz:

Under normal conditions, a 50.5 Hz signal was estimated within 0.04 seconds with μ_k =0.00445. Estimated frequency = 50.4917 Hz. The signal was initialized at 50 Hz and supplied to the LMS filter. The performance is shown in Fig 4.2 Error in estimation was found out to be around 0.0083 Hz.

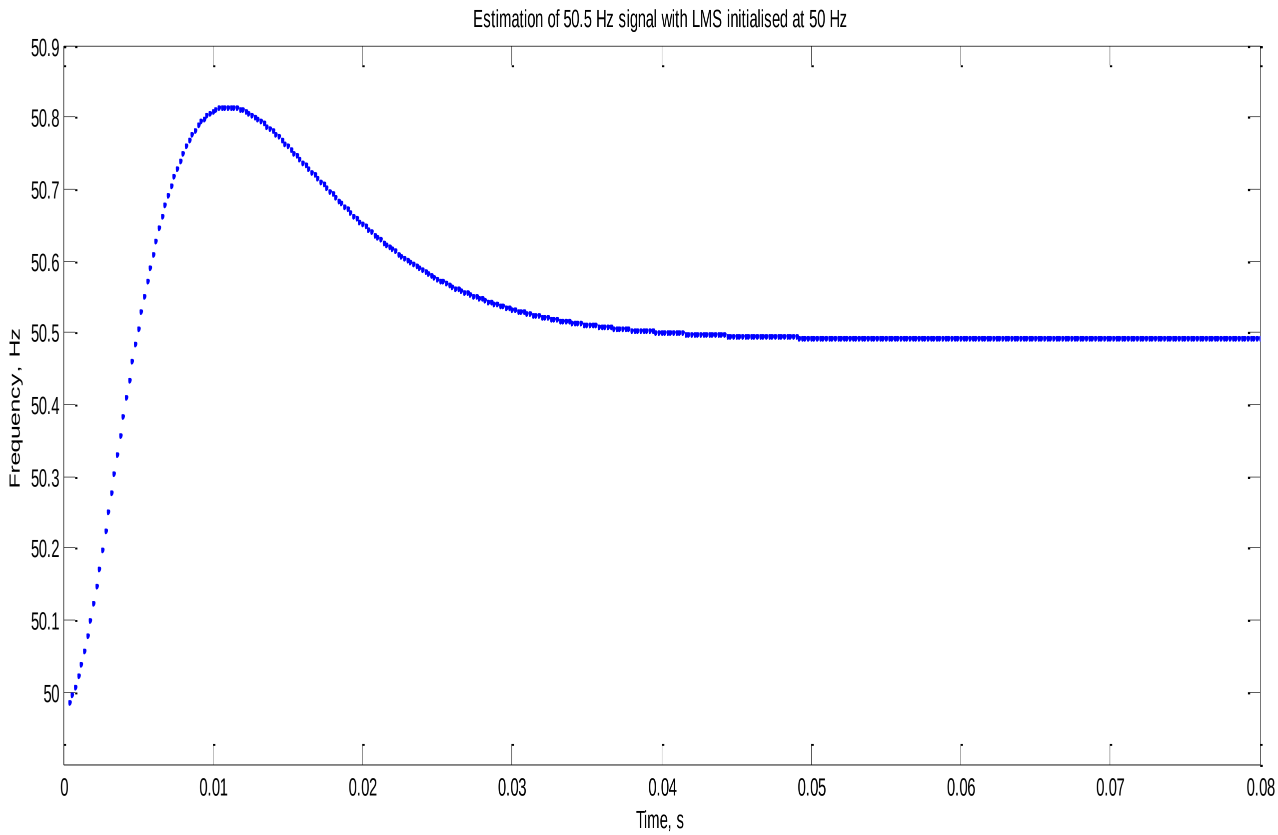


Fig 4.2 Estimation of 50.5 Hz signal with LMS initiated at 50 Hz

3. Estimation during a sudden frequency change from 50 Hz to 48 Hz:

Whenever a frequency change occurs, the voltage signal is affected and so is the performance of the algorithm. The algorithm gave satisfactory results within one cycle when used during sudden frequency jump condition. The frequency was changed from 50 Hz to 48 Hz at 0.0156 s. μ_k = 0.00461 and Estimated frequency = 48.003 Hz. The result obtained is shown in Fig 4.3,

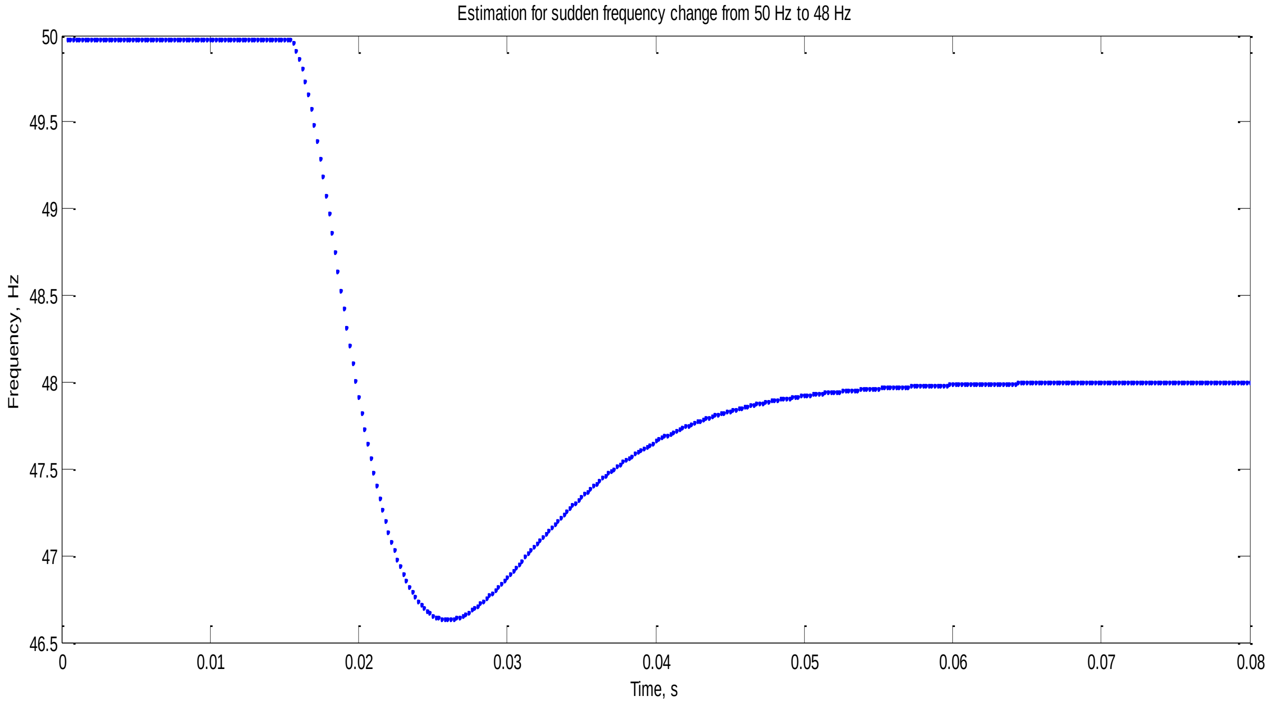


Fig 4.3 Estimation during a sudden frequency change from 50 Hz to 48 Hz

4. 50 Hz frequency Estimation in presence of 10% 3rd harmonics and 5% 5th harmonics:

The presence of harmonics distorts the voltage signal. The performance of the method in presence of harmonics is shown in Fig 4.4. In presence of harmonics also, the frequency was estimated within 0.04 seconds.

μ_k = 0.00465 and Estimated frequency = 49.9962 Hz

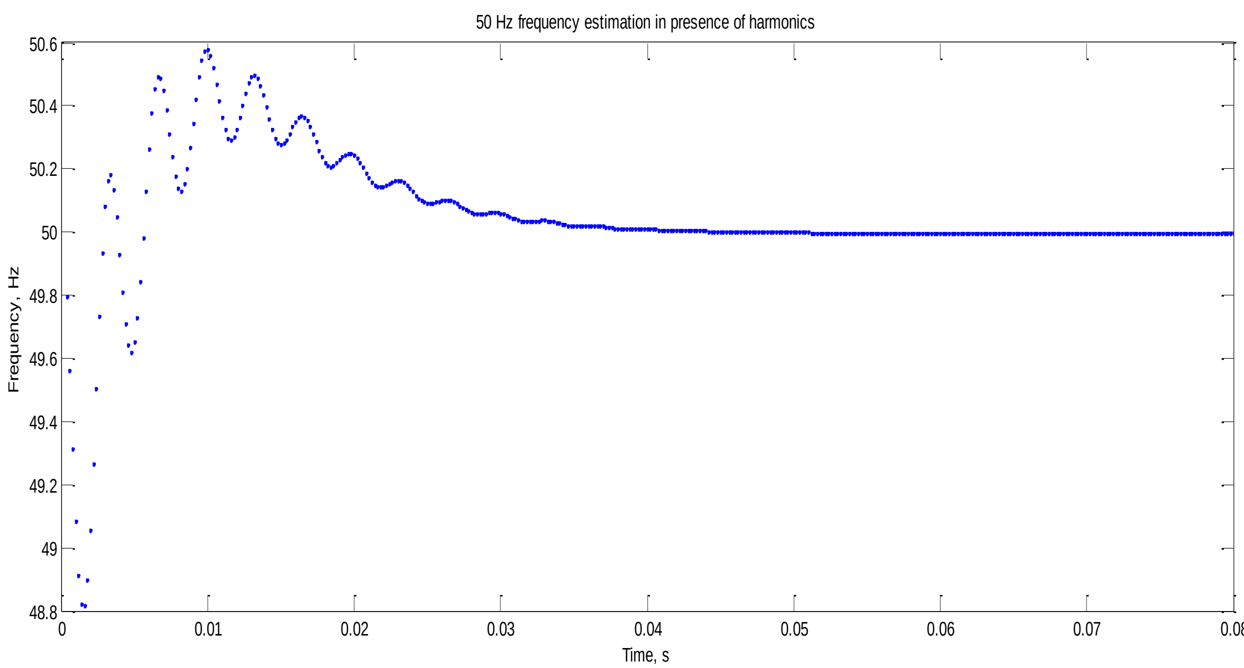


Fig 4.4 50 Hz frequency Estimation in presence harmonics

5. 50 Hz frequency estimation during unbalanced condition:

Here, the voltages of the three phases were made unequal by setting the maximum amplitude to 1.1, 0.9 and 0.8 of each phase respectively. The frequency was then estimated using these signals by LMS algorithm. μ_k = 0.0051 and the frequency was estimated within 0.04 seconds with estimated value being 50.0007 Hz. The plot of the performance of this method under unbalanced condition is shown in Fig 4.5.

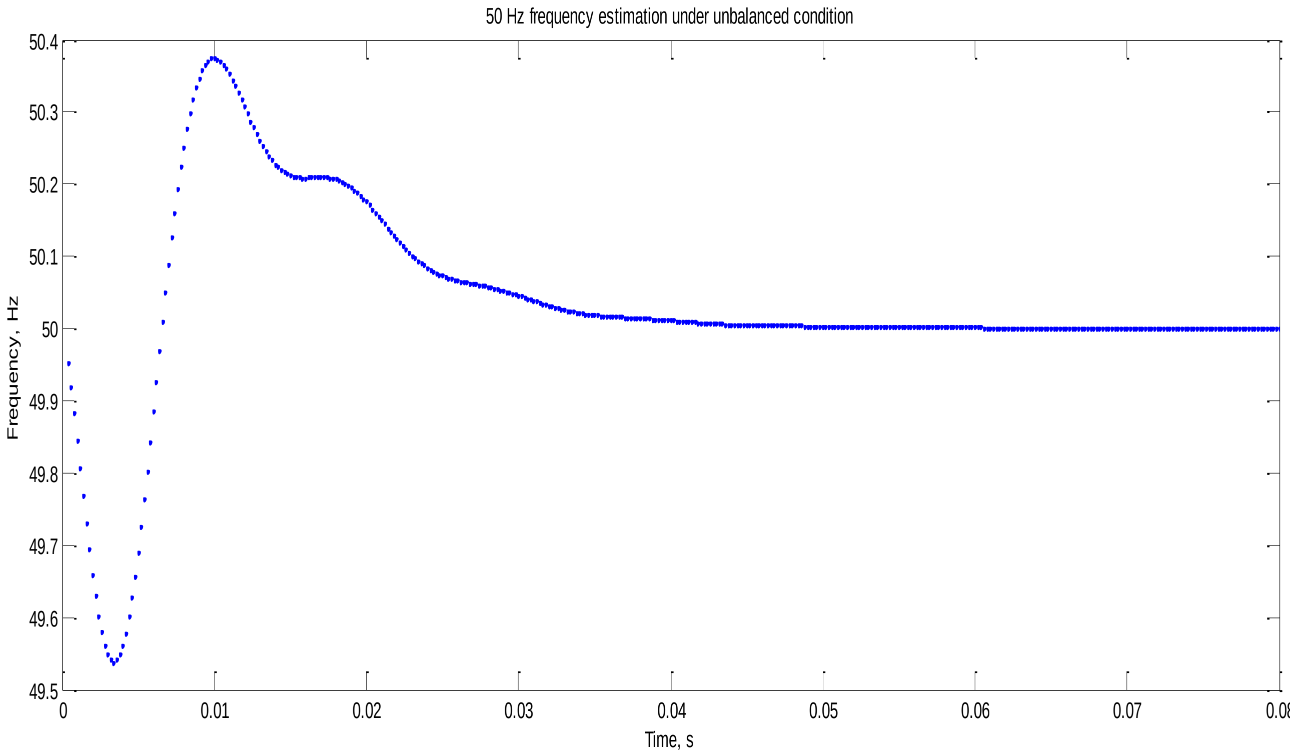


Fig 4.5 50 Hz frequency estimation under unbalanced condition

4.2 NLS Method:

The following curve (Fig 4.6) is an illustration of how search is carried out in NLS method. The signal taken was a pure sinusoidal signal and frequency was varied from 48.5 Hz to 51.5 Hz in steps of 0.1 Hz. The frequency at which minimum norm occurred was the estimated frequency. The sampling frequency was taken to be 1.6 kHz. As is seen from the curve, at 50 Hz, minimum norm occurred and hence, the estimated frequency is 50 Hz which matches with the actual frequency.

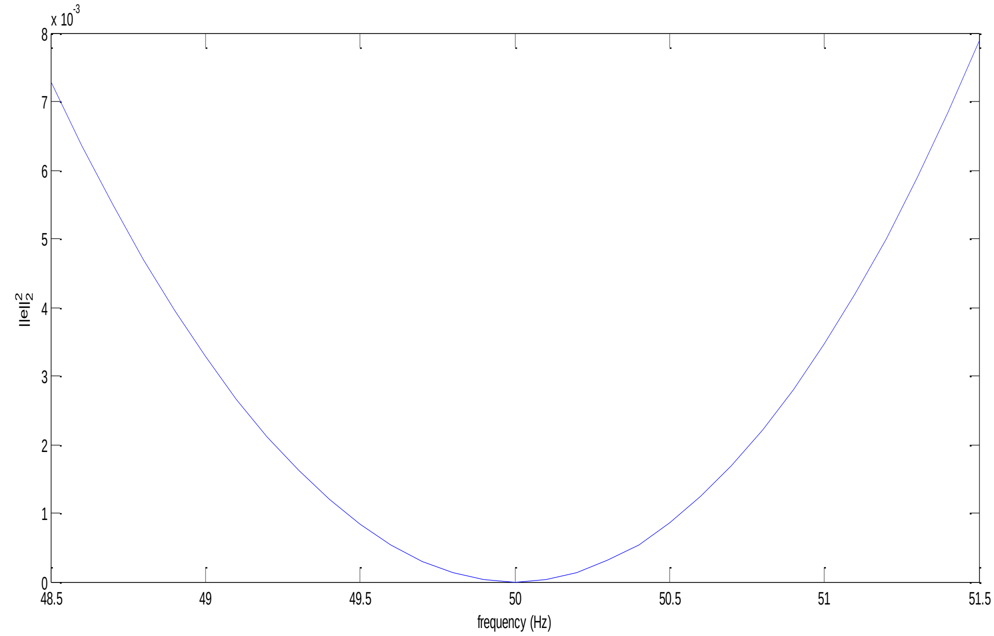


Fig 4.6 A Simple NLS 1-D Search

Let us consider a signal[11] with 10% THD:

$$v(t) = \sin \omega_0 t + 0.0856 \sin 5\omega_0 t + 0.0428 \sin 7\omega_0 t + 0.0306 \sin 11\omega_0 t + 0.0183 \sin 13\omega_0 t$$

The signal is shown in Fig 4.7. Let the sampling frequency used be 3.2 kHz and the frequency of this signal be 50 Hz. In the algorithm, the frequency was varied from 48.5 Hz to 51.5 Hz in steps of 0.05 Hz. The frequency which gives the lowest norm is the estimated frequency.

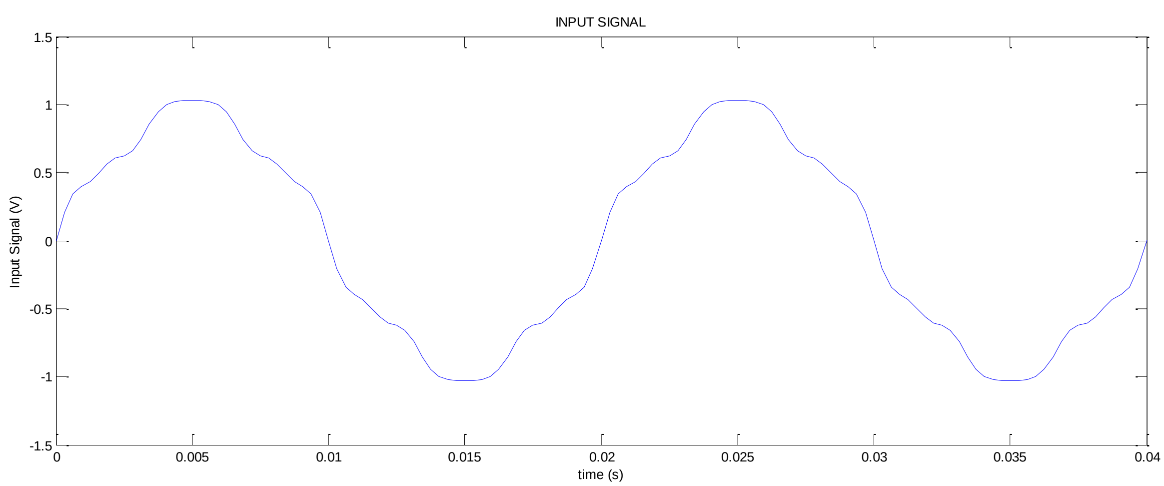


Fig 4.7 Input Signal with a constant frequency of 50 Hz

A window length of 10ms was used. So, Number of samples per window length = 3.2*10 = 32.

1. Estimation of 50 Hz frequency for a voltage signal of arbitrary peak value:

A voltage signal of arbitrary peak value was fed to both the algorithms. It was seen that the older method began to estimate frequency only when the number of samples became equal to 32 while the modified method worked independent of the window length and was able to estimate the frequency in 0.005s. It is known that the minimum number of samples required to estimate the frequency must be greater than or equal to two times the number of harmonics present. Here, the assumed signal contains 4 harmonics. So, frequency can be estimated if approx. 10 samples are given. So, the frequency estimation process in modified method begins much earlier. The performance of the two methods is presented in Fig 4.8.

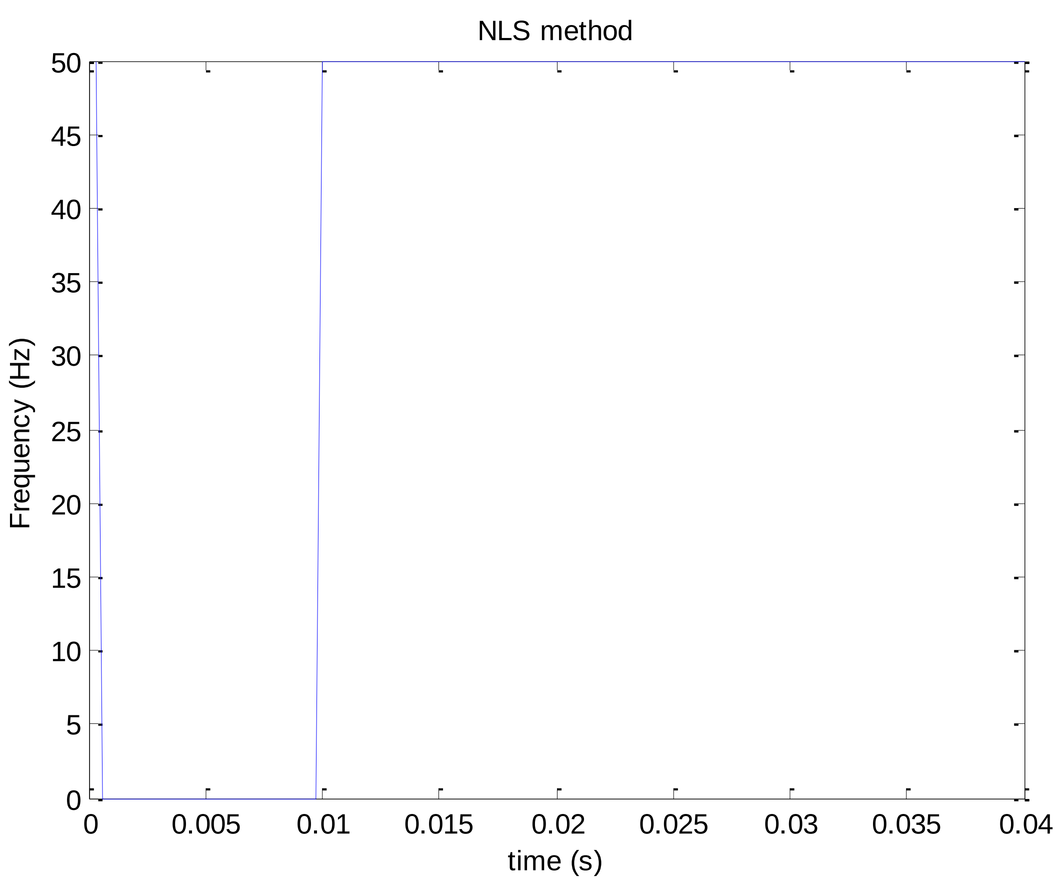


Fig 4.8(a) Frequency estimation using old NLS method

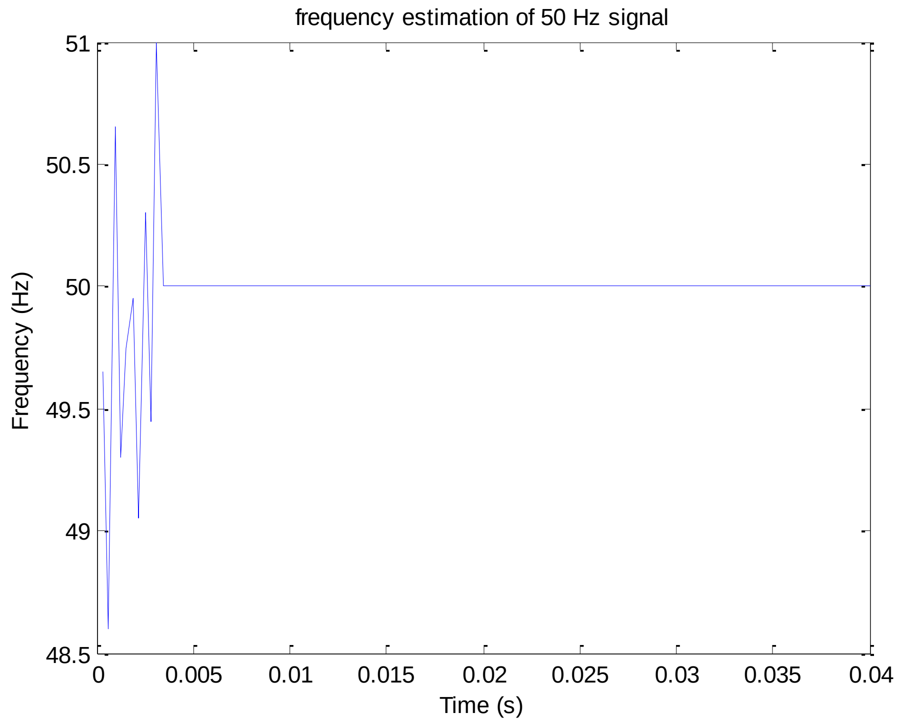


Fig 4.8(b) Frequency estimation using modified NLS method

2. Frequency estimation during frequency jump from 49 Hz to 51 Hz:

The performance of the old and modified NLS methods during sudden frequency jumps have been presented in Fig 4.9. Frequency was suddenly changed from 49 Hz to 51 Hz at 0.02s in the assumed signal. The behavior of both the methods was almost the same in this case.

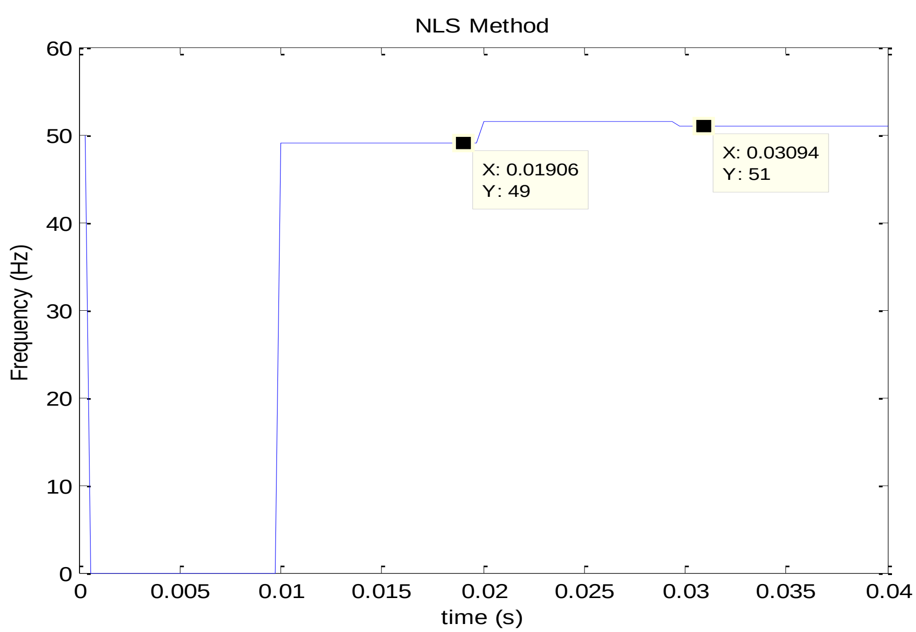


Fig 4.9(a) Estimating frequency during frequency jump using old NLS method

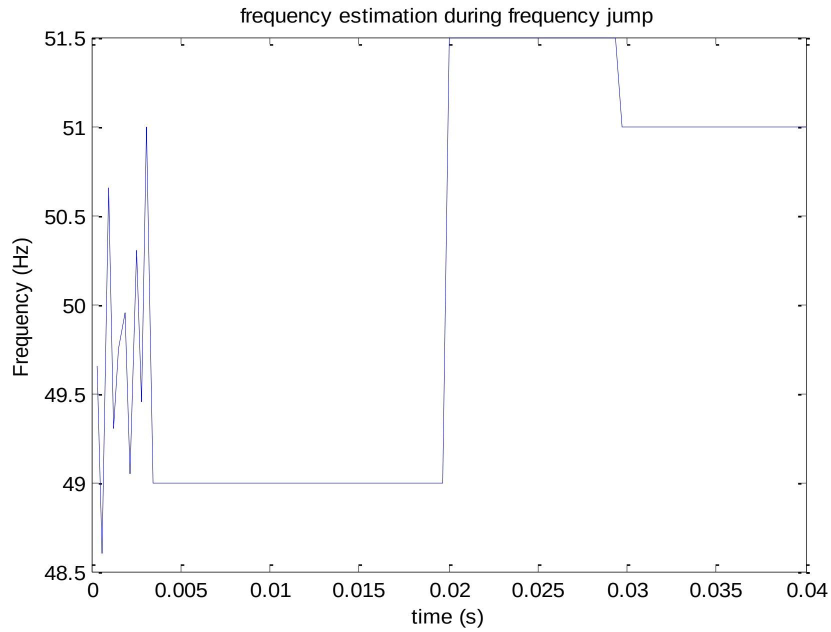


Fig 4.9(b) Estimating frequency during frequency jump using modified NLS method

3. Frequency estimation during 15% step change in magnitude:

A sudden change in magnitude of voltage signal can affect the performance of NLS method. The magnitude of the signal was suddenly raised by 15% at 0.02s and fed to the algorithms. The performance of the two methods is shown in Fig 4.10. The behaviour of both the methods was almost the same in this case. Each of the two methods was able to track the original frequency at a delay equal to the window length.

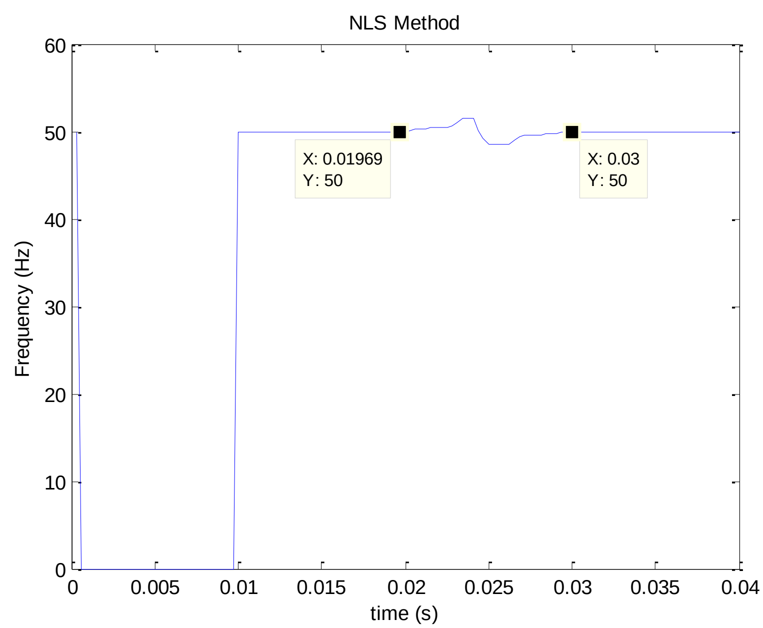


Fig 4.10(a) Estimating frequency during magnitude change using old NLS method

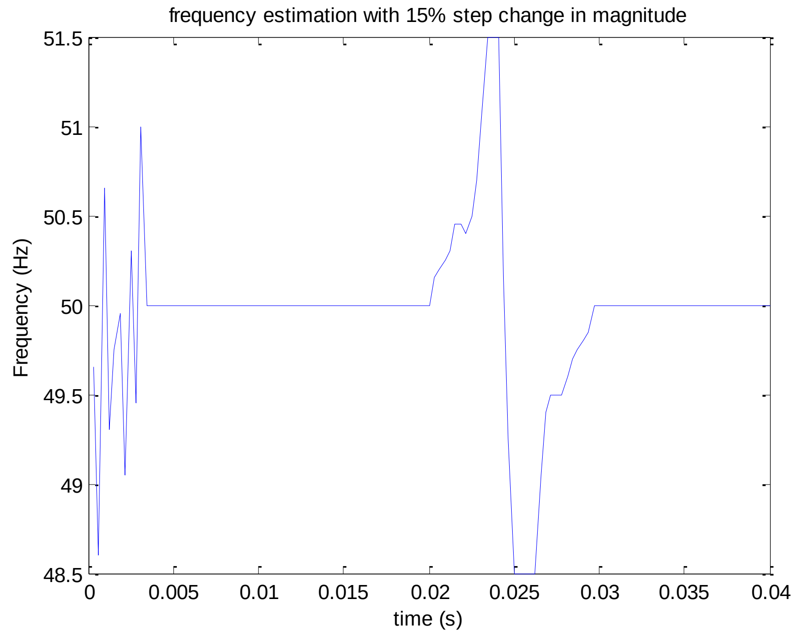


Fig 4.10(b) Estimating frequency during magnitude change using modified NLS method

4. Frequency estimation under 60dB noise:

Addition of noise to any signal distorts its waveform and the performance of the algorithm to which the signal is fed may be affected. The performance of both the methods was observed by adding a white Gaussian noise of 60dB to the signal. As seen from Fig 4.11, the modified method was more immune to noise than the older one.

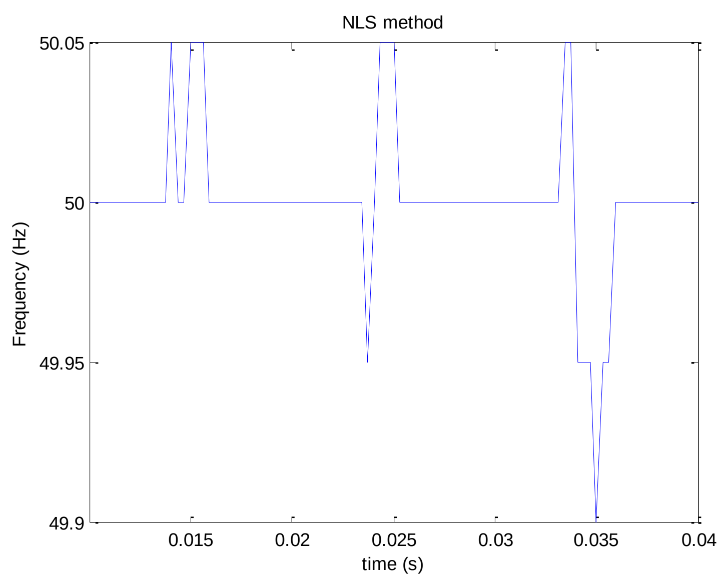


Fig 4.11(a) Estimating frequency under noise using old NLS method

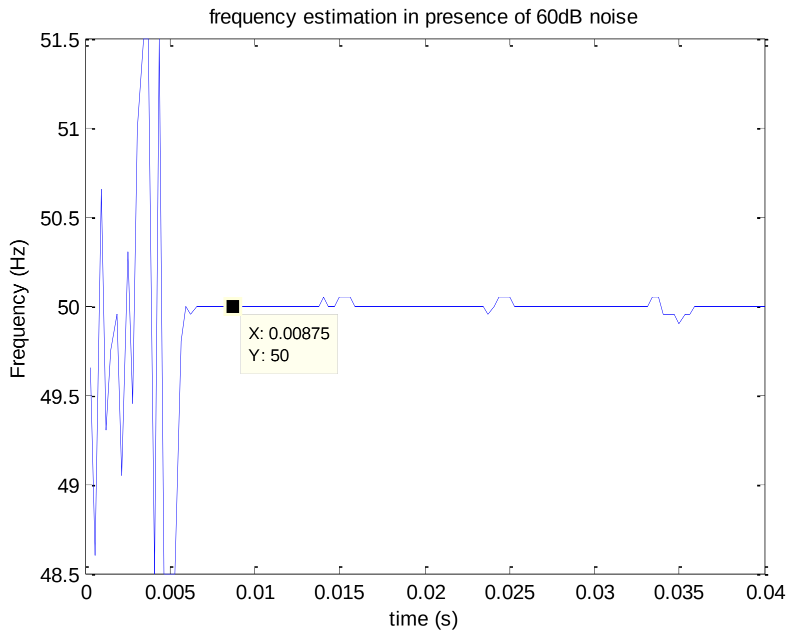
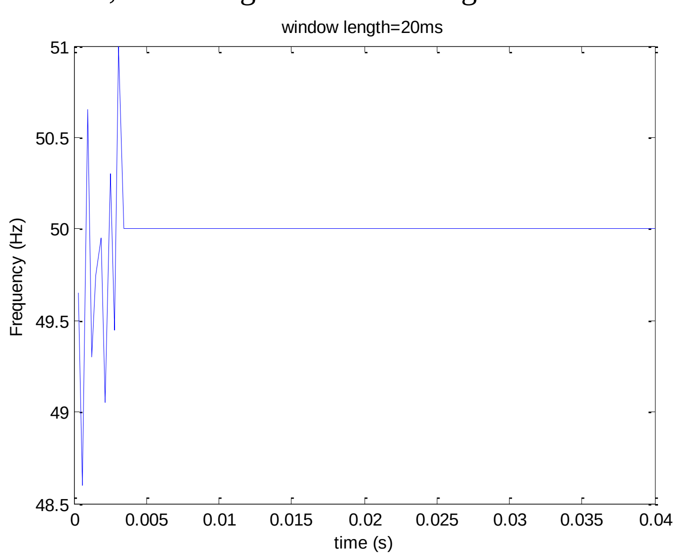
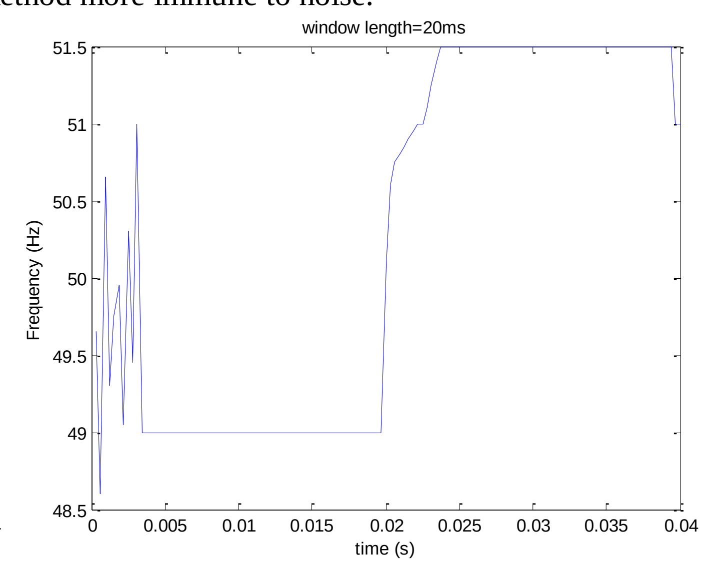


Fig 4.11(b) Estimating frequency under noise using modified NLS method

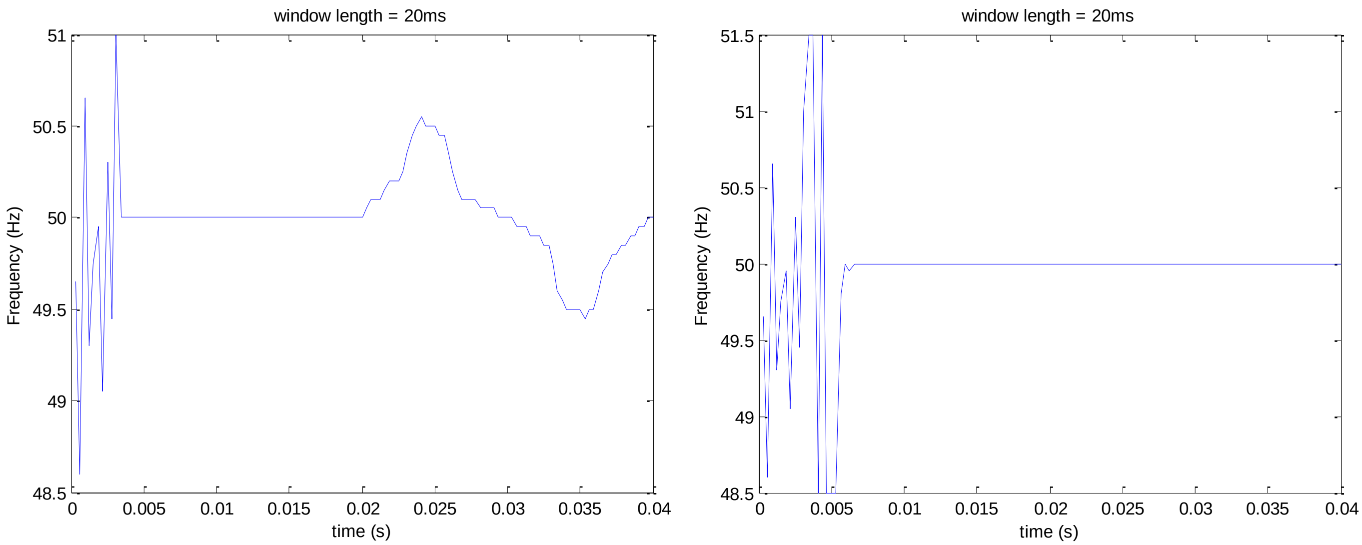
The performance of the modified method upon changing its window length is presented in Fig 4.12. The initial delay in estimating the frequency is independent of the window length in modified method. Whenever there is a frequency jump or step change in magnitude, the time taken to re-estimate the frequency is equal to the window length which is not advantageous if the size of window length is greater because this introduces a delay in estimation process. But a greater window length lowers down the overshoot of frequency in case of magnitude jumps. Also, increasing the window length makes the method more immune to noise.



(a) 50 Hz frequency estimation



(b) Frequency jump from 49 to 51 Hz



(c) 15% step change in magnitude (d) under 60dB noise

Fig 4.12 Behavior of modified method when window length = 20ms

5. Performance of the modified method under fault condition:

A single line to ground fault was created using PSPICE, the schematic of which is shown in Fig 4.13(a). The voltage source has a frequency of 50 Hz and peak value of 230V and is supplying a R-L load. An L-G fault occurs at time = 20ms and the system is restored at time = 100ms.

:

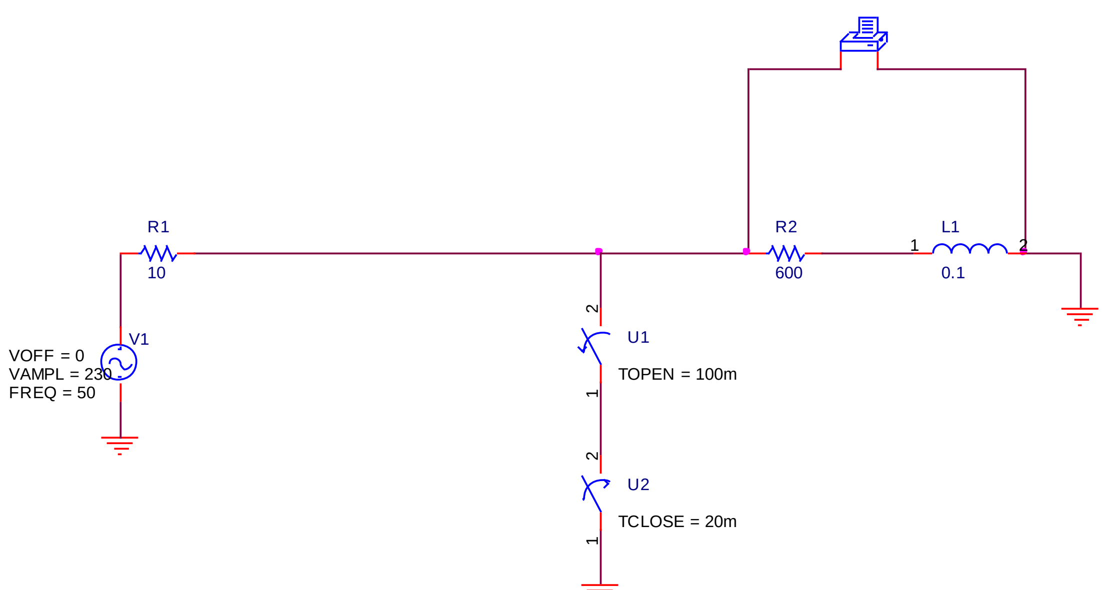


Fig 4.13(a) Schematic L-G fault

During the fault, the voltage dips as a result of which the performance of the modified method is affected. The fault voltage waveform is shown in Fig 4.13(b). The behavior of method during this time interval is plotted in Fig 4.14. During fault, variation in frequency is less when sampling frequency = 6.4 KHz. Also, a fault causes the frequency to dip which is clearly satisfied by this method.

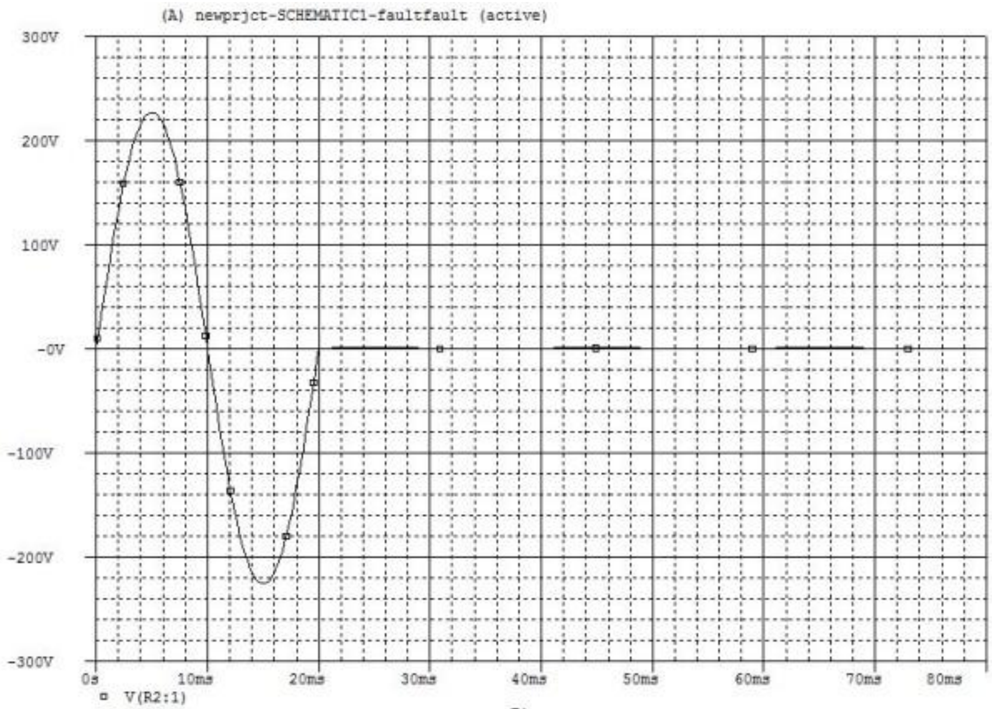
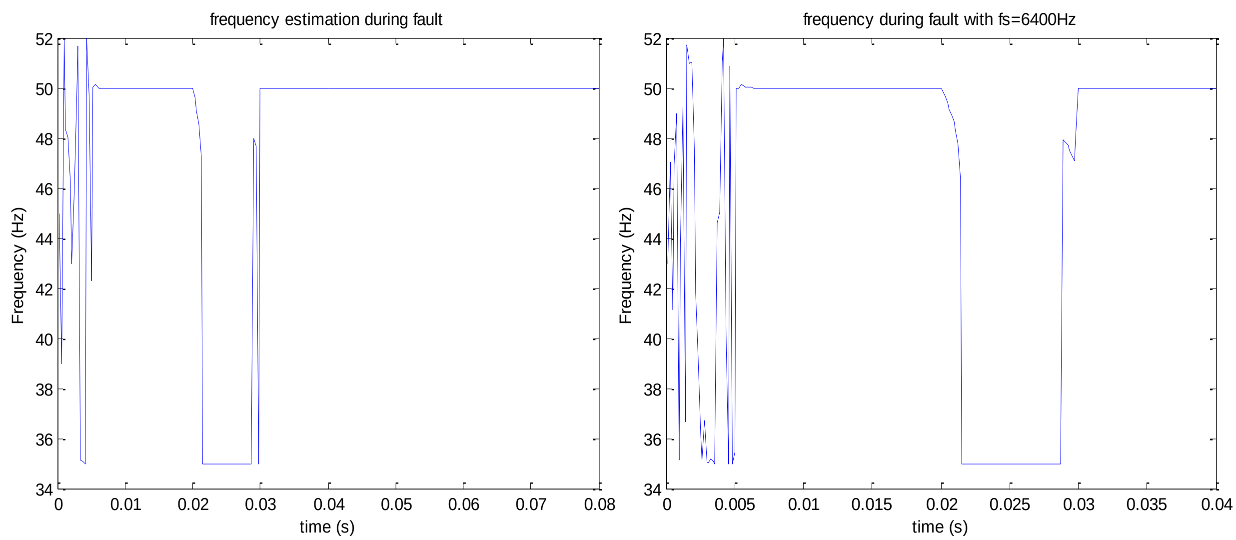


Fig 4.13(b) Fault voltage waveform



(a) Sampling frequency = 3.2 KHz

(b) Sampling frequency = 6.4 KHz

Fig 4.14 Performance of the method under fault for different sampling frequencies

5. Performance of the method under Practical conditions:

The block diagram of the experimental setup is shown in Fig 4.15. 230 V supply was given to the Auto-transformer input and required voltage was supplied to the R-L load using an Isolation transformer. Rectifier makes the load non-linear and is the source of harmonics because of switching action of diodes. Voltage waveform was taken at the terminals of isolation transformer through a Digital Storage Oscilloscope (DSO) and captured in PC using a PC-PC Communication Software. DSO and PC were connected using a 9-pin female port.

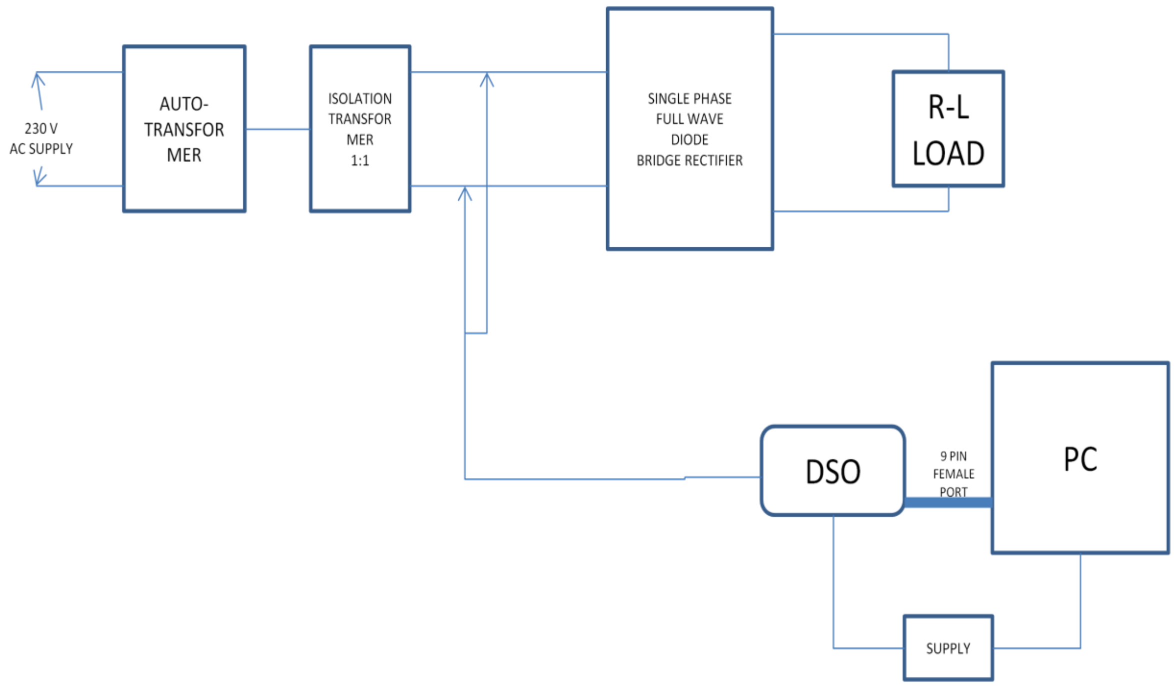


Fig 4.15 Block Diagram of the Experimental Setup

A) FIRST SET OF DATA:

Auto-transformer output = 33V

R=300Ω; L=50mH; frequency=50.1Hz

B) SECOND SET OF DATA:

Auto-transformer output = 120V

R=300Ω; L=250mH; frequency=50.1Hz

The captured voltage waveforms for the first and second set of data are shown in Fig 4.16.

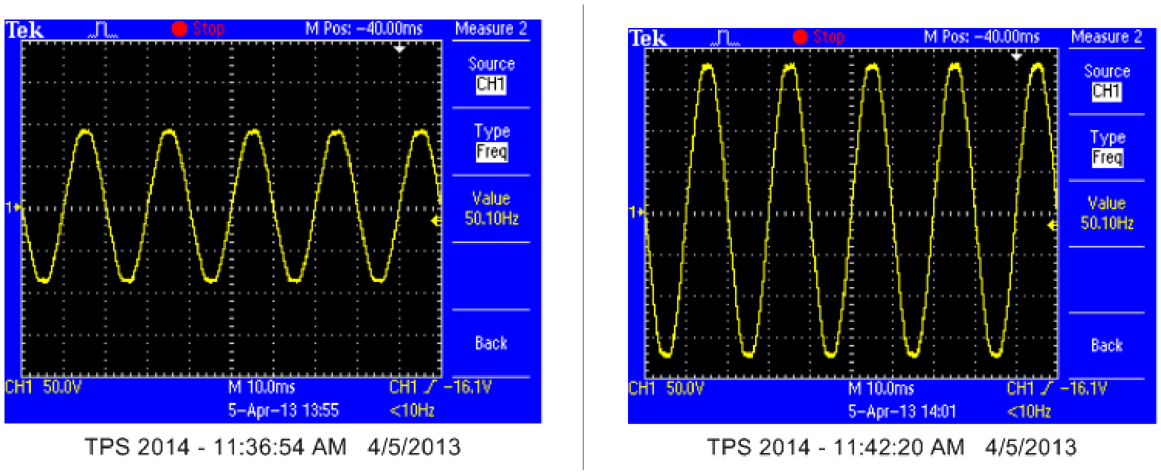


Fig 4.16 Captured Voltage waveforms

For the Simulation,

Sampling Frequency = 5 kHz

Window length for simulation = 30ms

Total data points used = 400 points.

The performance of the NLS method for the data collected is presented in Fig 4.17 and Fig 4.18. It is observed that the old method takes nearly one and a half AC cycles to estimate the frequency while the modified method estimates the frequency within one AC cycle.

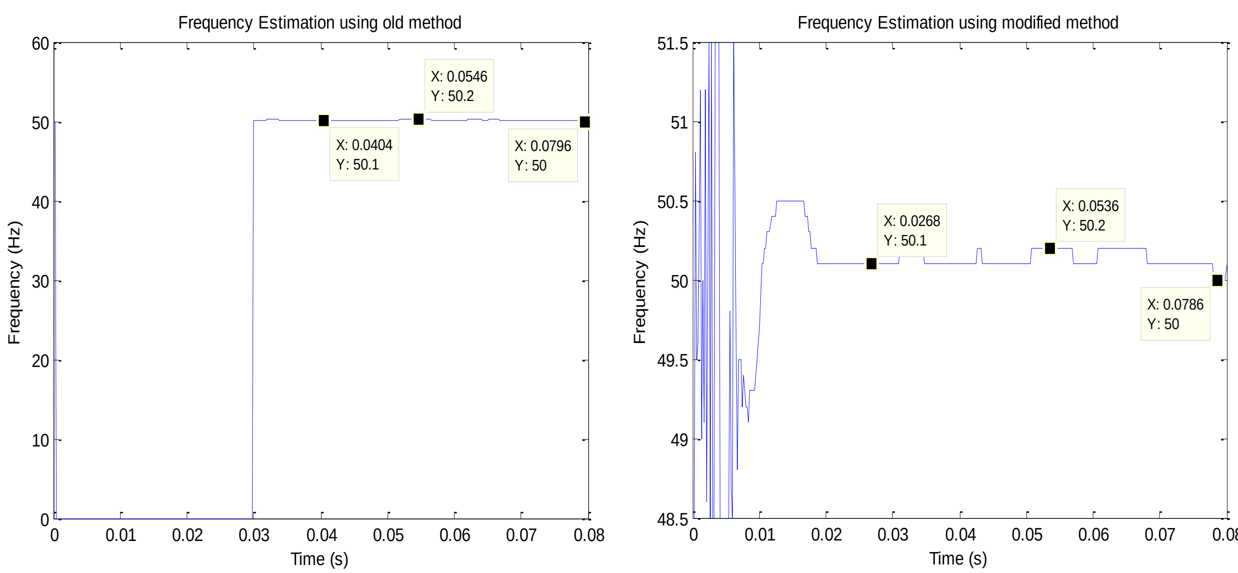
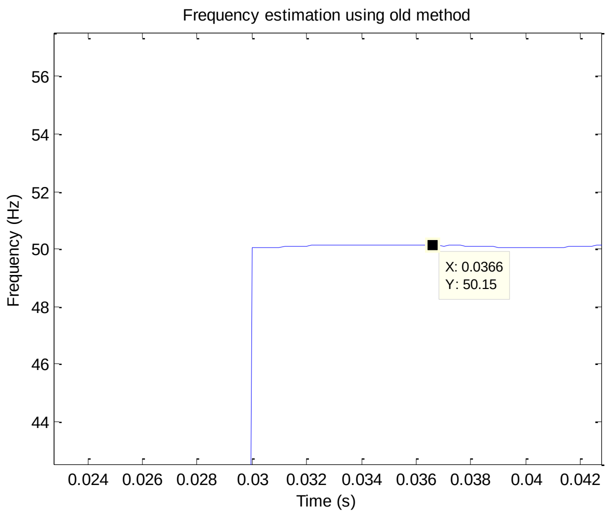


Fig 4.17 Frequency estimation for the first set of data



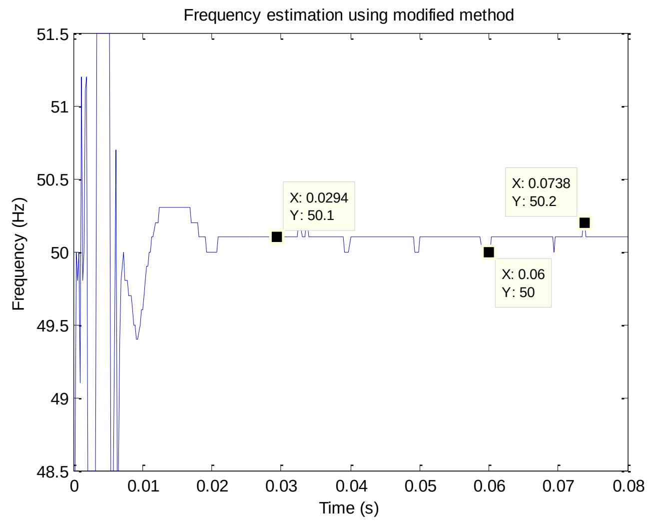


Fig 4.18 Frequency estimation for the second set of data

It is, thus, quite evident that the modified method estimates the frequency much faster as compared to the old method. In both the methods, frequency oscillation is found to be less than 0.2 Hz in the steady state. The rating of equipments used is given in appendix. The experimental setup is shown in Fig 4.19.

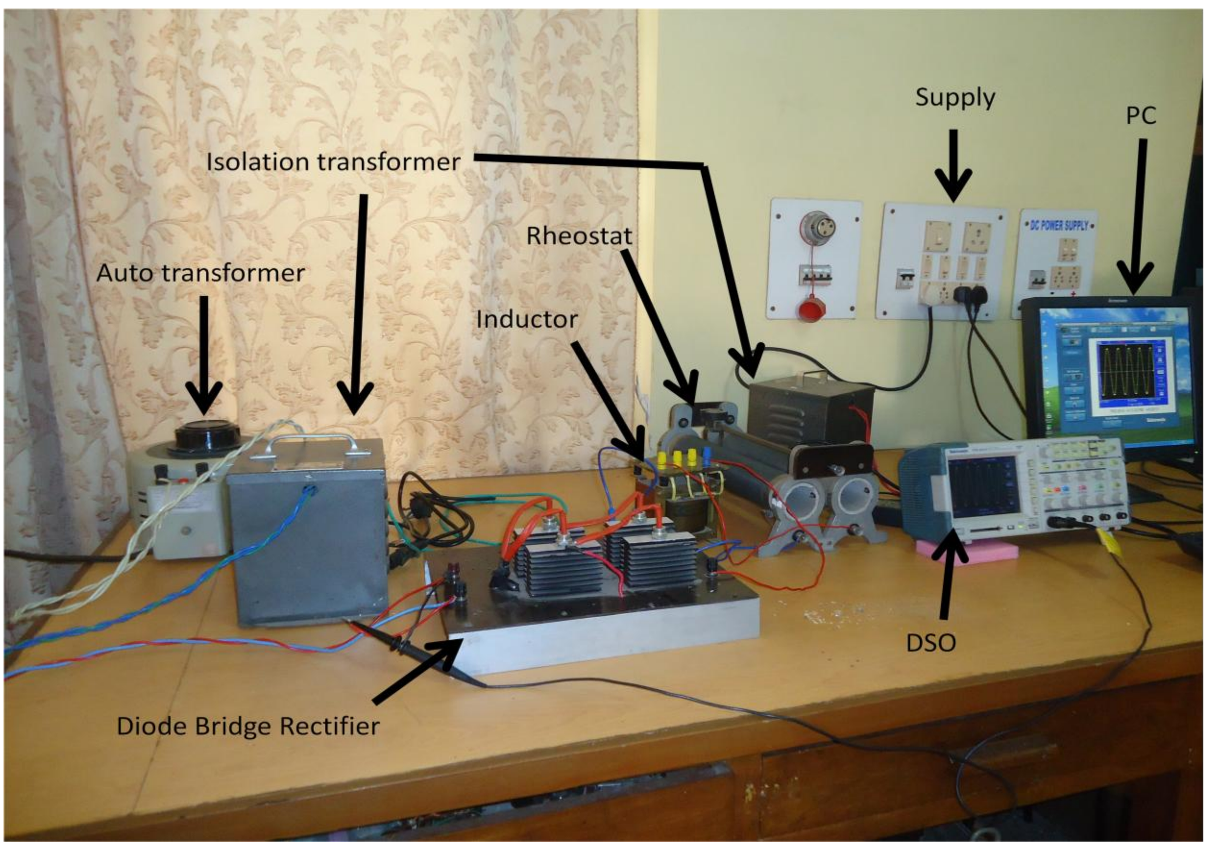


Fig 4.19 Experimental Setup for Estimation of frequency

CHAPTER 5

DISCUSSION AND CONCLUSION

5.1 Superiority of modified NLS method over the older method:

Modified method is faster in estimating frequency. Within the quarter of a cycle, frequency can be estimated using this method. Modified method provides better performance as compared to old NLS under noise. ***A*** matrix is constant and pre-computed in older method whereas it is of variable size and dynamically computed in modified method. Thus, storage of ***A*** matrix requires more memory in older method than modified method.

5.2 Choice of window length:

It depends on the type of performance needed. As is seen, more the window length, more accurate is the frequency estimation, better is the noise immunity and lesser is the overshoot in case of step change in magnitude. But the disadvantage is the increased delay in estimating frequency and the increased size of ***A*** matrix.

5.3 Comparison between LMS method and NLS method:

Delay in estimating frequency is more in LMS than NLS. LMS takes up to two cycles to estimate the frequency which is done in a quarter cycle by NLS. Accuracy and Performance under noise of LMS method is poor than NLS method. The steady state frequency estimated in LMS method is closer to the true value but the frequency estimated in NLS method is equal to true value. Three phase voltages are required for estimating frequency in LMS method but for estimating frequency using NLS method, voltage of any one phase is sufficient. Parameters of LMS algorithm need to be adjusted under different conditions which is not so in case of NLS algorithm. Parameter such as initial step size is different for different conditions. LMS is a linear method while NLS is non-linear due to presence of sine and cosine terms in the matrix. LMS uses an adaptive filter with varying step size while the modified method makes use of a time varying matrix.

5.4 Conclusion

Thus, the performance of modified method under all conditions is satisfactory as compared to the old NLS method and is much better than its linear counterpart, the LMS method although the modified method involved rigorous matrix multiplication and addition. With the modified method, Frequency is estimated within one AC cycle and the frequency oscillations are also very less in the steady state. The modified method holds good for all practical conditions, even during faults with a slight modification in its parameters.

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APPENDIX

Auto transformer	Type : 15D-1P Max. load = 15A; KVA = 4.05 Input = 240V 50/60Hz ; Output = 0-270V
Isolation transformer	Capacity = 2KVA Primary volt = 230 V Secondary volt = 115 et 115 V
Bridge Rectifier	4 Diodes 25A, 1200V
Rheostat Inductor	300Ω, 5A 0-250mH (0-50, 0-100, 0-250mH)
DSO	Tektronix TPS 2014 (Four channel DSO) 100MHz, 1GS/s Sampling frequency = 25000Hz