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A Novel Power Conditioning Unit (PCU) using Adaptive Signal Processing for Low THD

Debanjan Mukherjee, Asim Kumar Jana, Malay Kumar Pandit

Abstract— The presence of harmonic in power system is a major concern to power engineers. With the heavy usage of non-linear loads in power systems, the harmonic effect becomes more serious. One of the most popular computation algorithms for harmonic analyzer is Fast Fourier Transform (FFT). In this paper ,single phase current waveform is taken from a three phase supply fed to motor through power analyzer interfaced with PC. FFT is done using MATLAB program on the imported data. After that, same waveform like the current waveform obtained from the hardware setup is created in MATLAB SIMULINK window. That created waveform is filtered by Normalized LMS Filter to reduce Total Harmonic Distortion (THD) in the filtered output. Comparison is done between the hardware results with the software simulated results. THD value is reduced from at max 12.03% to at min 0.17%, which has set a new record to our best knowledge.

Index Terms—FFT, SIMULINK, NLMS Filter, Total Harmonic Distortion (THD).

I. INTRODUCTION

Harmonic is an integer multiple of the fundamental system frequency of an electrical signal. In power system , harmonic analysis is used to determine the influence of harmonic producing load on a power system . Nonlinear loads affect the quality of supply by drawing harmonic currents and reactive power from electrical distribution system. Active power filters are the most viable solution for solving such power quality problems in compliance with the harmonic standards. This article presents a NLMS algorithm for elimination of harmonics and reactive power compensation under distorted voltage without the use of conventional phase-locked-loop or low-pass filter blocks. The effect of harmonics on power system can be in the form of power efficiency reduction, overheating in wire, ageing of electrical insulation, etc [1] [2]. Many algorithms have been proposed for harmonic study [3]-[7] and Fast Fourier Transform (FFT) is the most widely used computation algorithm [7]-[9]. FFT is an efficient algorithm used to compute Discrete Fourier Transform (DFT). DFT uses a finite set of discrete-time sample of an analogue signal and produces a finite set of discrete-frequency magnitude spectrum values.

Power converters are of flexible in control, high efficiency and reduced cost. In spite of above advantages, these nonlinear loads are drawing harmonic current and more reactive power from the utility. Power quality pollutions distort the voltage and current waveforms. This causes severe deterioration of power-factor and other adverse effects such as, increases R.M.S. value of supply current, overheating of distribution transformer, interference to communication lines, power loss and poor system efficiency. This harmonic contaminated supply distorts the supply voltage profile at the point of common coupling (PCC) and increases the distortions in supply voltage. This distorted voltage affects the performance of nearby connected consumers. If the supply voltage is already distorted, which is more common case in electrical distribution system, then the current distortions will be more and can be classified as (a) customer generated harmonics, (b) utility generated harmonics.

Filtering of harmonic contained signal is needed to eliminate the harmonics. There are a lot of filters to filter out the harmonics. Here in this paper, Normalized LMS Filter has been used. Actually adaptive filters can adapt their filter coefficients to the environment according to existing conditions. The filters are capable of learning from the statistics of current conditions and change their coefficients to achieve desired signal. To design a filter, a clear knowledge of the desired response is required. When such knowledge is not available, in that case adaptive filter is used [10]-[15]. The adjustment is directly proportional to the tap input vector u(n). Therefore, when u(n) is large, the LMS filter suffers from a gradient amplification problem. To overcome this difficulty, normalized LMS filter is used. In particular the adjustment applied to the tap-weight vector at iteration n+1 is "normalized" with respect to the squared Euclidean norm of the tap-input vector u(n) at iteration n-hence the term "normalized"[15],[16]. This paper is presented as follows; II. Fast Fourier Transform, III. Adaptive Filter Theory, IV. Hardware experimental setup, V. Observation of hardware setup data, VI. Software simulation, VII. Simulation result observation and analysis VIII. Discussion, IX Conclusion, X. Acknowledgement,XI. References, XII. About the authors.

II. FAST FOURIER TRANSFORM

Fast Fourier Transform (**FFT**) is an effective algorithm to compute the discrete Fourier transform (DFT) and it's inverse. There are many distinct FFT algorithms having a wide range of mathematics, from simple complex-number arithmetic to number theory. A DFT decomposes a sequence of values into components of different frequencies. This operation is essential in many fields but computing it directly from the definition is often too time consuming to be practical. An FFT is a way to compute the same result more quickly: computing a DFT of N points in the native way, using the definition, takes $O(N^2)$ arithmetical operations, while an FFT can compute the same result in only O(N log N) operations. The difference in speed can be substantial, especially for long data sets where N may be in the thousands or millions—in practice, the computation time can be reduced by several orders of magnitude in such cases, and the improvement is roughly proportional to N / log(N).

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This massive amendment made many DFT-based algorithms practical; FFTs are of great importance to a wide variety of applications, from digital signal processing and solving partial differential equations to algorithms for quick multiplication of large integers [17],[18].

The most FFT algorithms depend upon the factorization of N, but there are FFTs with O(N log N) complexity for all N, even for prime N. Many FFT algorithms only depend on the fact that $e^{-\frac{2\pi i}{N}}$ is an N[th] primitive root of unity, and thus can be applied to analogous transforms over any finite field, such as number-theoretic transforms. Since the inverse DFT is the same as the DFT, but with the opposite sign in the exponent and a 1/N factor, any FFT algorithm can easily be adapted for it. Thus using this tool , the supply harmonics created by nonlinear loads is easily identified. Nonlinearity can be introduced by using three phase induction motor coupled to DC generator connected with load circuit. Here one time signal corrupted with zero mean random noise is shown below in figure 1. Along with the time signal, corresponding FFT spectra is shown in figure 2.

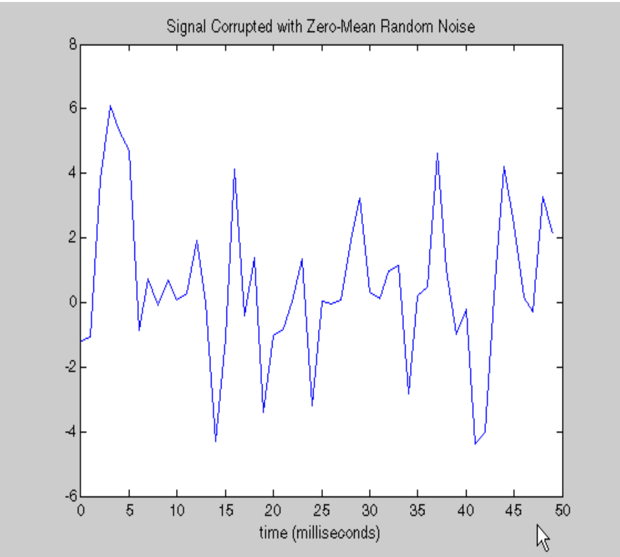

Figure 1. Time varying signal

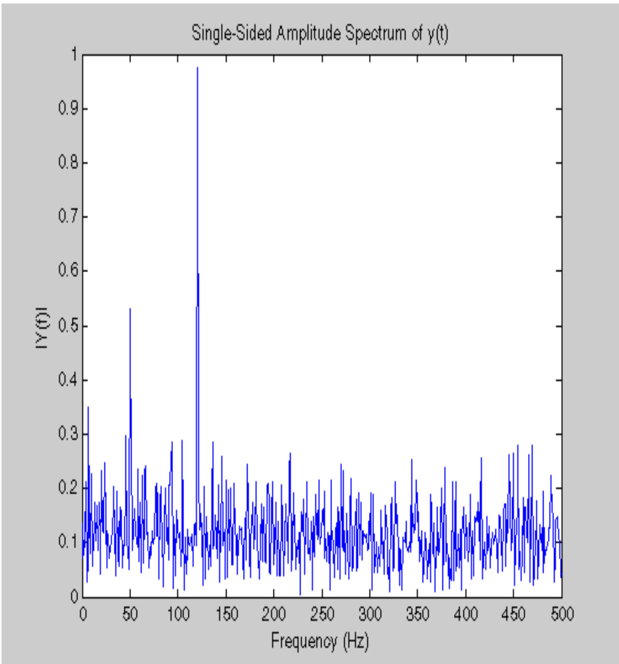

Figure 2. Frequency spectrum of the above signal

III. ADAPTIVE FILTER THEORY

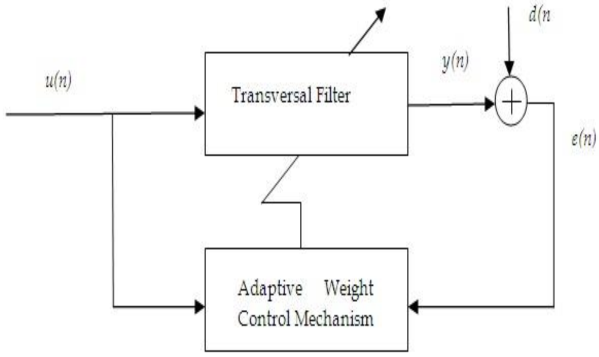

Figure 3. Block diagram of Adaptive Filter

In the standard form of an LMS filter, the adjustment applied to the tap-weight vector of the filter at iteration n+1 consists of the product of three terms:

- The step-size parameter μ, which is under the designer's control.
- The tap-input vector u (n), which is supplied by a source of information.
- The estimation error e (n) for real-valued data, or its complex conjugate $e^*(n)$ for complex-valued data, which is calculated at iteration n.

The adjustment is directly proportional to the tap-input vector u (n) is large; the LMS filter suffers from a gradient noise amplification problem. To overcome this difficulty, the normalized LMS filter is used. The adjustment applied to the tap-weight vector at iteration n+1 is normalized with respect to the squared Euclidean norm of the tap input vector $u(n)$ at iteration n hence the term "normalized."

SUMMARY OF NORMALIZED LMS ALGORITHM

Parameters: M= number of taps (i.e. filter length)

μ= adaptation constant

$0<\mu<2*[\,E[|u(n)|^2]\,\mathcal{D}(n)/E[|e(n)|^2]]$

Where

$E\,[|e(n)|^2]$ =error signal power

$E\,[|u(n)|^2]\,\mathcal{D}(n)$ =input signal power

$\mathcal{D}(n)$ =mean-square deviation

Initialization. If prior knowledge about the tap weight vector w(n) is available, use that knowledge to select an appropriate value for w(0).Otherwise, set W(0)=0.

Data

(a)Given: u (n) =M-by-1 tap input vector at time n

d (n) =desired response at time step n

(b)To be computed: W (n+1) =estimate of tap weight vector at time step n+1

Computation: For n=0, 1, 2, 3..........compute

$e\,(n) = d(n) - W^H(n)u(n)$,

$W\,(n+1) = W\,(n) + [\,\mu\,/\|u(n)\|^2]\,u\,(n)\,e^*(n)$.

IV. HARDWARE EXPERIMENTAL SETUP

In the hardware experimental set-up, there is one MFS (Machinery Fault Simulator). It is an innovative tool to study the signatures of common machinery faults without compromising production schedule or profits.

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The bench top system has a spacious modular design featuring versatility, operational simplicity, and robustness. Each component is machined to high tolerances so it can be operated without conflicting vibration. In MFS, intentionally faults are created for increasing nonlinearity within the system.

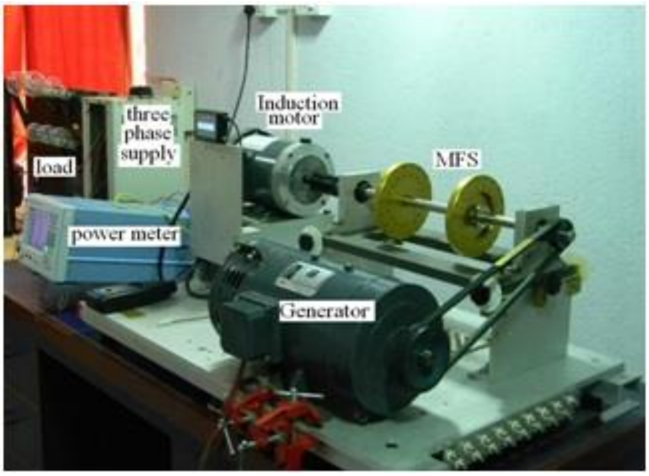


Figure 4. Experimental Setup

In the set-up shown in figure 4, there is a three phase induction motor (with bearing fault) of 1/3 HP,2850 rpm,50Hz rating .Induction motors with rotor fault are nonlinear load to the supply. The said motor is coupled to a long shaft, housed in two bearing housings. On that shaft there are two metal discs. Each discs having 36 slots on the periphery for inserting nut and bolts to create different loading as well as to create different degree of unbalance also.

At first, supply wires are connected to power analyser. Power Analyzer (WT500) is a mid range product for single-phase and three-phase power measurements. It is designed into a compact half-rack size chassis which can also be double rack mounted. Standard features include a color TFT display and USB interface for communications and memory. The instrument has a basic accuracy of 0.1%, Maximum input of 1000 $V_{rms,}$ 40 A_{RMS} and a measurement bandwidth of DC to 100 kHz. Then power analyser's output wires are connected to the motor input terminal. There are belt pulley arrangement to couple one (2.2KW, 230V, 12.1Amp, 1500 rpm) DC generator with the said long shaft of MFS (in figure 5).

Figure 5. DC generator coupled to motor shaft.

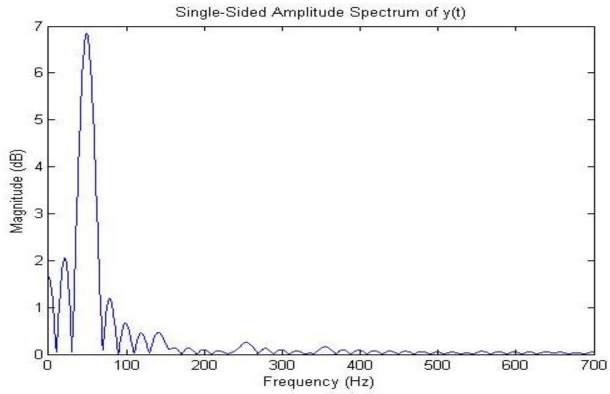
Generator generates dc voltage, which is fed to the load circuit.

Figure 6. Bulb loading circuit.

There are four 110Volt bulbs on that load board for creating four different loading conditions shown in figure 6. Wave data is taken through PC , interfaced with power analyser.

V. OBSERVATION OF HARDWARE SETUP DATA

Four 60 watt bulbs are used in the load circuit for varying load on motor. Here it is seen that magnitude of fundamental frequency is also being increased with the increase in load. FFT spectrums of current waveforms in different loading conditions are shown below (in figure 7):



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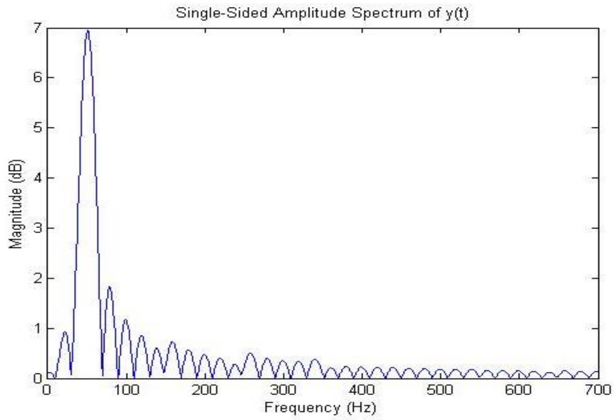

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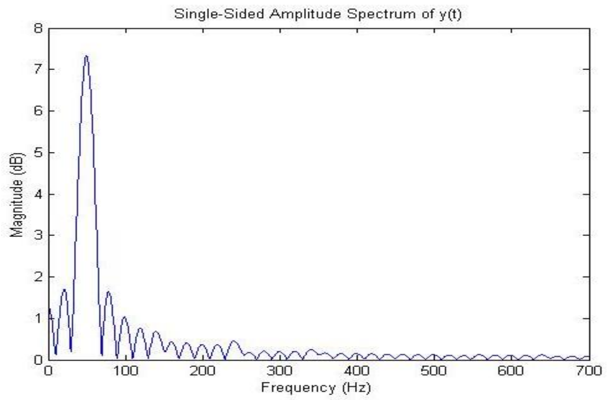

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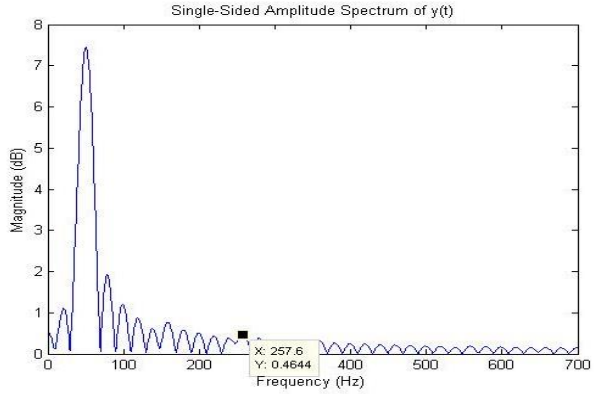


(d)

Figure 7. FFT of R-phase current waveforms in different loading conditions;(a) Noload, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

Here it is seen that the peak of fundamental frequency is largest in amplitude and harmonics are also present within the range of frequency from 0Hz to 700Hz.Amplitude of fundamental, harmonic frequencies increases when the nonlinearity in load increases, in a word total harmonic distortion (THD) gradually is being varied.

VI. SOFTWARE SIMULATION

Using MATLAB program, the obtained wave data (taken from hardware set up) is processed to have its FFT spectrum. Here one SIMULINK model is made where input waveform is created resembling the waveform obtained from the hardware set up. That generated waveform is then fed to normalized LMS filter block. Filtered output is got from the output terminal of the Normalised LMS filter block. Block diagram of the whole simulink model is shown in figure 8.

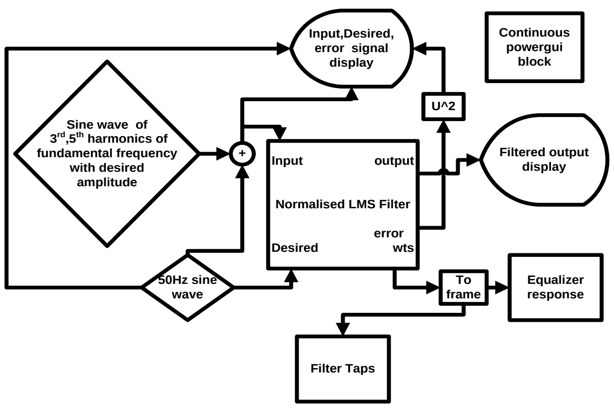
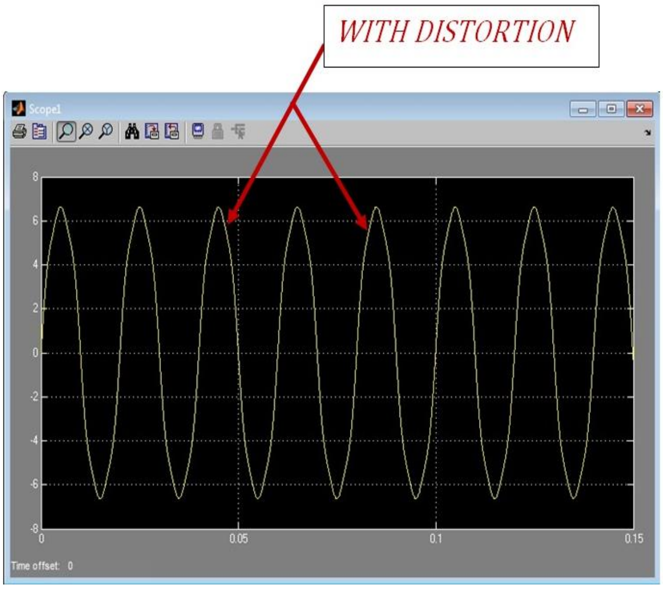


Figure 8. Block diagram of filtering technique for eliminating harmonics from input signal through simulink.

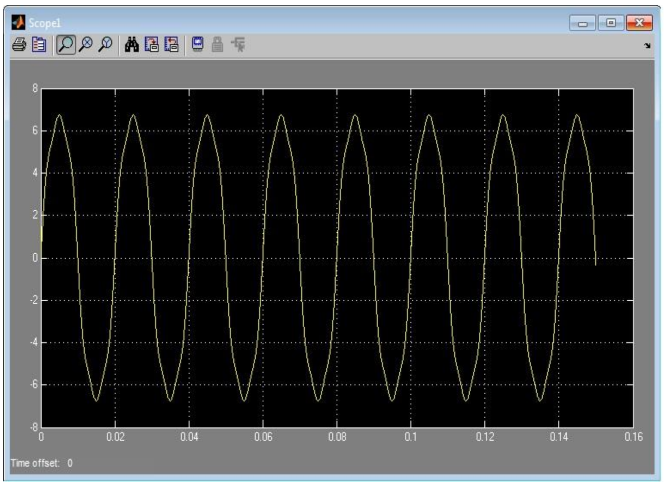
At first 50Hz sine wave has been contaminated with 3rd (150Hz), 5th (250Hz) sine waves, then the summed signals are fed to the Normalized LMS Filter as input. Simultaneously one 50Hz sine wave is fed to the desired input terminal as desired signal. On the other side of the filter block, there are output, error & wts terminals. From output terminal, desired output is displayed on scope. From error terminal the undesired signal components in squared form along with actual signal, desired signal are displayed on a three input scope at a time. Equalizer response display & filter tap (response of coefficient index Vs amplitude) display are connected with wts terminal through to frame block. Different current signals are captured for different loading conditions from the said hardware setup. Like those current waves of different load, similar waves are created using sine wave blocks in simulink. Those waves were inputs of the normalized LMS filter.

VII. SIMULATION RESULTS OBSERVATION AND ANALYSIS

From the simulation model, different input waveforms for different loads are obtained and are given below:



(a)



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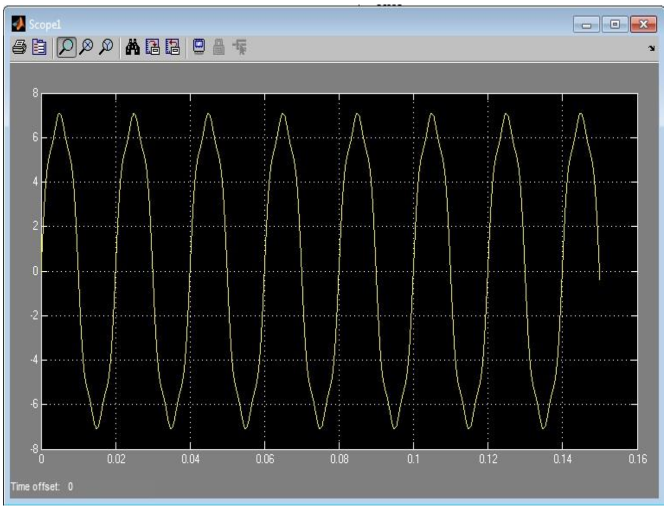

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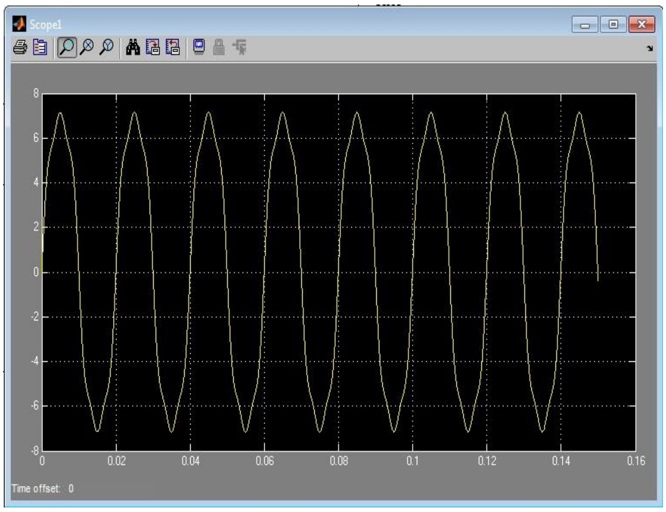
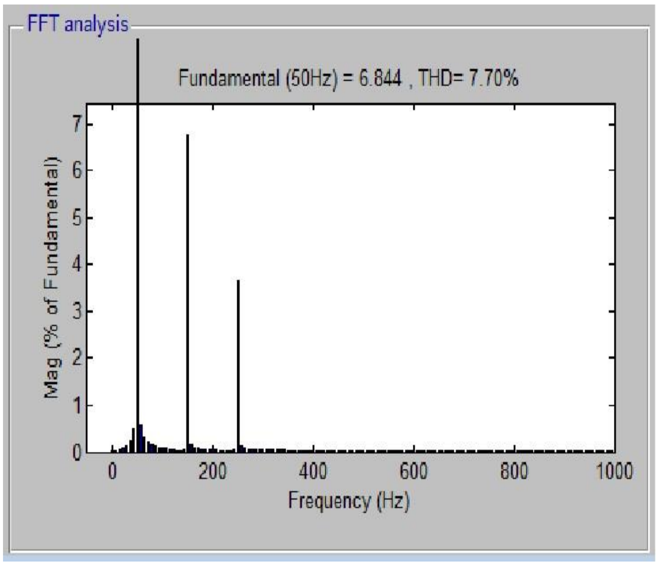


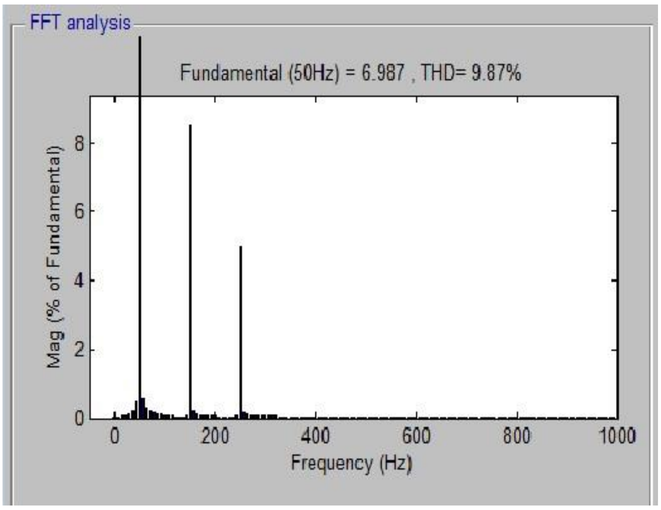
(d)

Figure 9. Input waveforms to the NLMS filter ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

In figure 9, it is seen that peak values of different input wave forms are gradually increasing with the increase in load. The waveforms are not purely sinusoidal because there are harmonics. If FFT is done on the input waveforms individually, it is seen that the magnitude of fundamental harmonic is increasing with the increment of electrical load. This phenomenon is very clear in figure 10 shown below:



(a)



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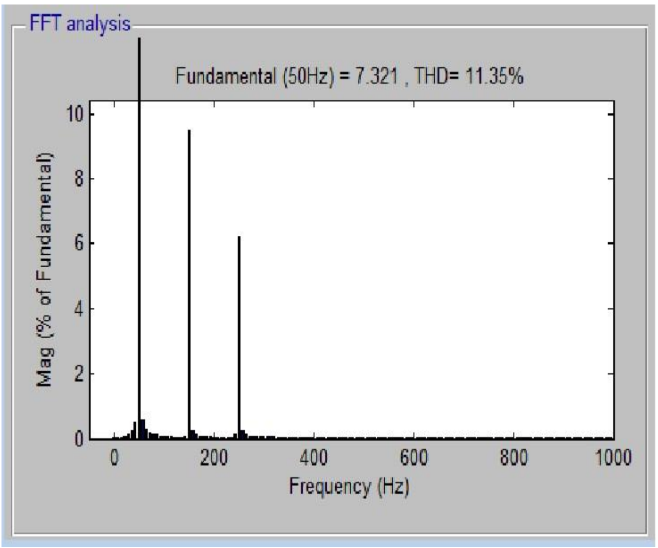

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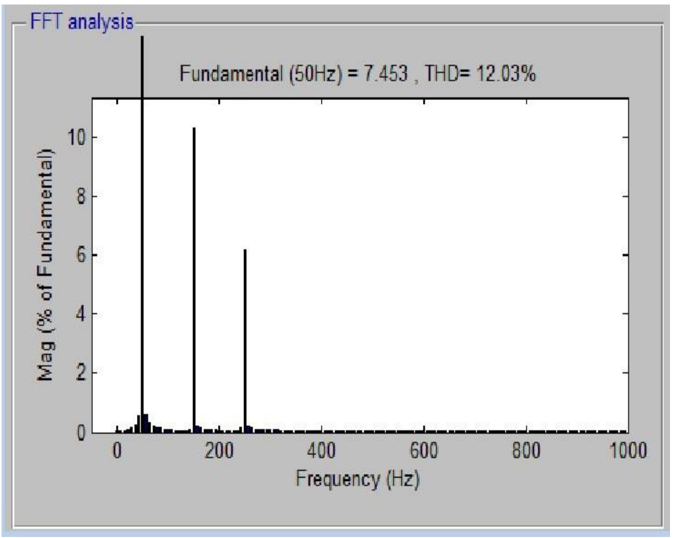

(d)

Figure 10. FFT spectrum of input waveforms ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

Here it is also observed that total harmonic distortion (THD) is varying with variation in load.

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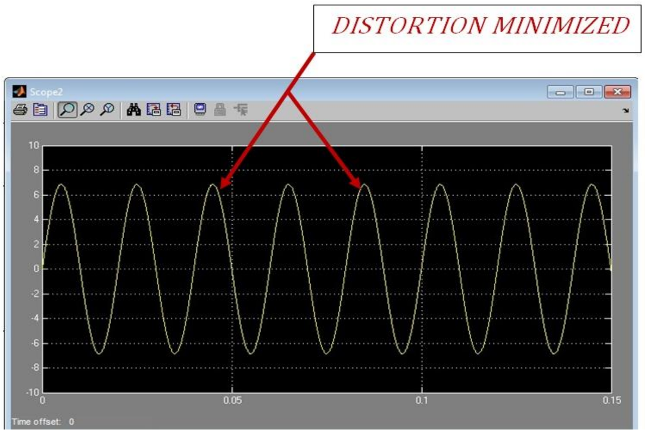
After filtration through the NLMS filter, output waveforms of corresponding inputs for four different loading conditions (shown in figure 9) have been shown below in figure 11.



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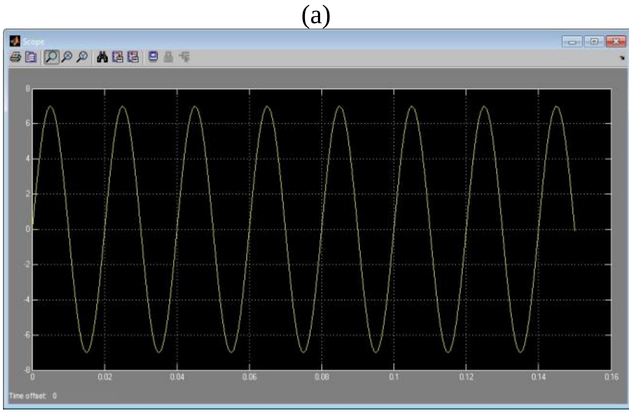

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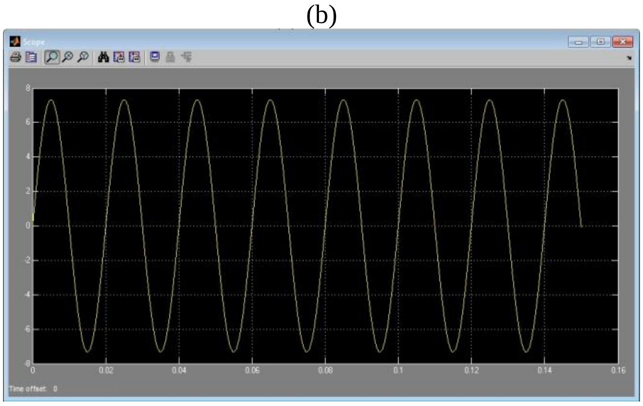

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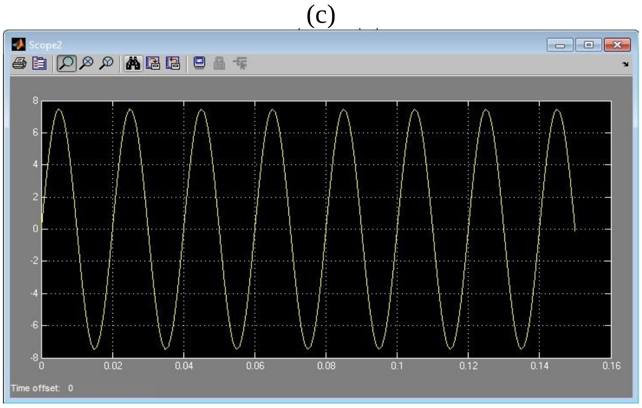
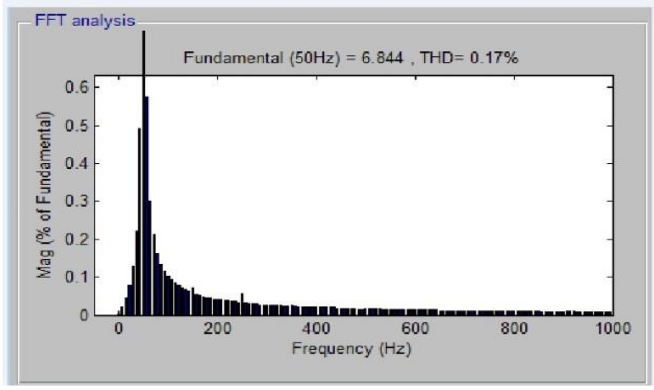


(d)

Figure 11. Output waveforms from the NLMS filter ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

From figure 11, it is seen that the output waveforms from NLMS filter are sinusoidal. That means most of the harmonic contents have already been eliminated from the input waveforms after filtration. FFT spectra of the above showed output waveforms demonstrate that **THD has come down to near 0.2%, which has set a new record to our best knowledge.**



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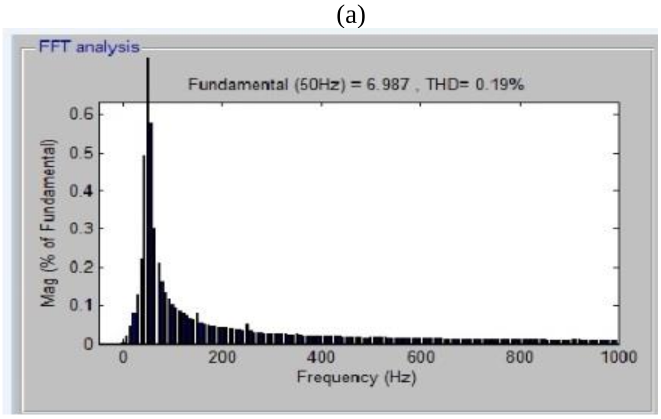

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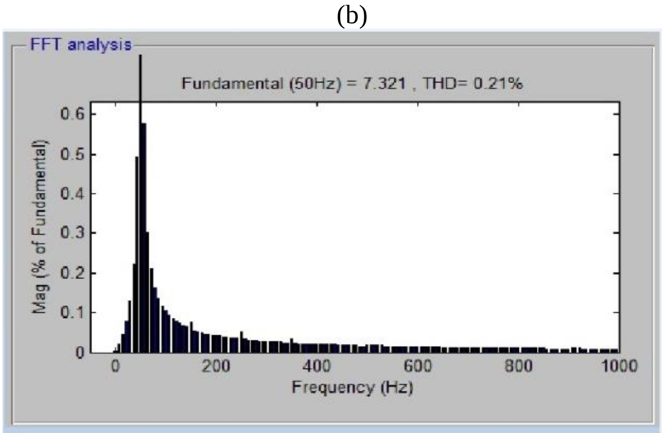

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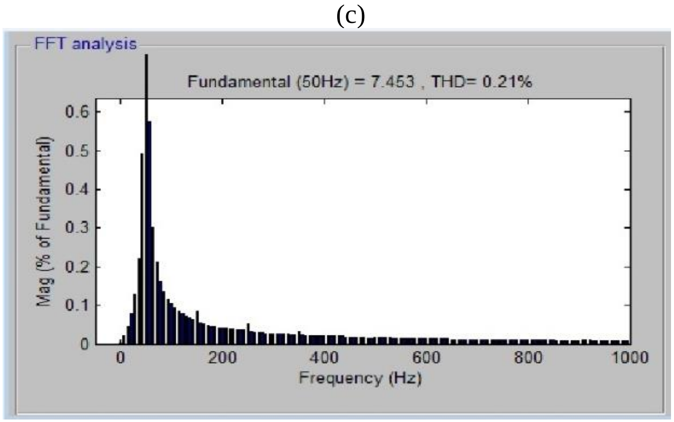

(d)

Figure 12. FFT spectrum of output waveforms ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

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After filtration through the NLMS filter, FFT of output waveforms for four different loading conditions are shown above in figure 12. Reduction in THD value means that influences of harmonic contents are less in amount after filtration.

VIII. DISCUSSION

Performance of NLMS filter is evaluated in terms of harmonic elimination and in reduction of THD values.

Table: 1 THD of input & output spectrums obtained from simulation model for different loads.

Electrical load	THD of input to filter	THD of output from filter
No load	7.7%	0.17%
One bulb	9.87%	0.19%
Three bulbs	11.35%	0.21%
Four bulbs	12.03%	0.21%

Simulation result shows that THD of FFT spectrum after filtration reduces up to 0.17%, which is a great deal in percentage. Therefore filtering technique using NLMS adaptive filter to eliminate harmonic contents is prominently effective.

IX. CONCLUSION

This paper presents NLMS filter for harmonic elimination in power line. This method is applicable in single phase as well as in three phase system. Use of high-tech instruments like MFS, WT500 gives reliability in data acquisition. This is due to the fact that instantaneous variation in harmonic contents has been taken into consideration for weight adjustment of the filter. Here from table 1, it is seen that THD value reduces from at max 12.03% to at min 0.17%, which is very much desirable.

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Malay K. Pandit received his B.E and M. E degrees in Electronics Engineering from Electronics and Telecom Engg Dept, Jadavpur University, India in 1989 and 1991, respectively. He received his PhD from UK’s renowned Cambridge University in 1996. He did his post-doc from the Optoelectronics Research Centre, City University of Hong Kong till 2002 where he pioneered the use of polymers for optical waveguide applications. .He then took a corporate career where he worked in a fibre optic company “FONS (I) Ltd” in the domain of optical networking. Now he is a full Professor in the Electronics Engg Dept and Dean of School of Engineering of the reputed Haldia Institute of Technology where he focuses on embedded systems and grid computing, including their usage in WDM optical networks. He currently leads a MODROBS project of the AICTE, Govt of India on embedded electronic systems. He has 45 international publications in this area.Dr. Pandit was awarded the National Scholarship of the Government of India in 1983 and the Nehru Scholarship of the Government of India during his Ph.D. studies