Detailed Investigation on Power Conditioning Unit (PCU) using Intelligent Signal Processing

Debanjan Mukherjee, Asim Kumar Jana, Malay Kumar Pandit

Abstract— Harmonic in power line communication is an important matter now a days. With the extensive usage of non-linear loads in power systems, the harmonic effect becomes more prominent. Fast Fourier Transform (FFT) is one of the most popular computation algorithms for harmonic analysis. 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 designed in SIMULINK window of matlab7.5 version. Those designed waveforms are filtered by Recursive Least Square (RLS) Filter to reduce Total Harmonic Distortion (THD) in the filtered output and the response of Normalised Least Mean Square (NLMS) for same input signal is discussed in detail in author's previous publication, given as reference. Adaptive signal processing to eliminate harmonics is replicated again in Code Composer Studio (CCS) using TMS320C6713 SIMULATOR .Here comparison is done between the responses of RLS filter with Normalised LMS filter. At last, the conclusion is drawn as NLMS filter is superior than RLS filter in the field of power line harmonic elimination to author's best knowledge.

Index Terms—RLS Filter NLMS Filter, FFT, SIMULINK, Total Harmonic Distortion (THD), TMS320C6713 SIMULATOR, CCS.

I. INTRODUCTION

Harmonic is nothing but integer multiple of the fundamental system frequency of an electrical signal. 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 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 comparison between RLS (Recursive Least Square) algorithm & NLMS (Normalized Least Mean Square) algorithm for elimination of harmonics form current spectrum and reactive power compensation without the use of traditional filter blocks. 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. There are a lot of filters to filter out the harmonics. Here in this paper, RLS filter response & Normalized LMS Filter Response has been compared. Actually adaptive filters can adjust their filter coefficients to the changing signal 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 [1]-[5],[7]. The adjustment is directly proportional to the tap input vector u (n).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"[6],[17],[18].

Effect of harmonics on power system can be in the form of power efficiency reduction, overheating in wire, ageing of electrical insulation, etc [8] [9]. Many algorithms have been proposed for harmonic study [10]-[14] and Fast Fourier Transform (FFT) is the most widely used computation algorithm [14]-[16]. 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.

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. Work on code composer studio, IX.Discussion, X Conclusion, XI. Acknowledgement, XII. References, XIII. About the authors.

II. FAST FOURIER TRANSFORM

Fast Fourier Transform (FFT) is a method for computing the Discrete Fourier Transform (DFT) with reduced execution time. Fourier Transform (FT) gives the frequency information of a signal, which means that how much of each frequency exists in the signal. In this paper, FFT of the current signal has been implemented using MATLAB. It is an effective algorithm to compute the discrete Fourier transform (DFT) and it's inverse.

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There are many distinct FFT algorithms having a wide range of mathematics, from simple complex-number arithmetic to number theory.[17]

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 $N / \log(N)$. [18]

III. ADAPTIVE FILTER THEORY

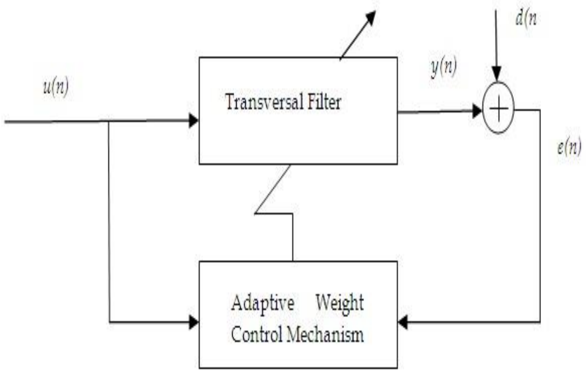

Figure 1. Block diagram of Adaptive Filter

Adaptive filtering algorithm consists of two basic processes:

- A filtering process, which involves (a) Computing the output of filter in response to an input signal and (b) generating an estimation error by comparing this output with a desired response.
- An adaptive process, which involves the automatic adjustment of the parameters of the filter in accordance with the estimation error.

The combination of these two processes working together constitutes a feedback loop, as illustrated in figure 1.

SUMMARY OF RECURSIVE LEAST SQUARE & NORMALIZED LMS ALGORITHM

RLS:

Initialize the algorithm by setting

$W(0)=\mathbf{0}$,

$\mathbf{P}(0)=\delta^{-1}\mathbf{I}$,

And

$$\delta = \begin{cases} \text{Small positive constant for high SNR} \\ \text{Large positive constant for low SNR} \end{cases}$$

For each instant of time, n= 1, 2, 3... compute

$\pi(n) = \mathbf{P}(n-1)\, u(n)$,

$\mathbf{k}(n) = \dfrac{\pi(n)}{\lambda + u^H(n)\,\pi(n)}$,

$\xi(n) = d(n) - w^H(n-1)\, u(n)$,

$w(n) = w(n-1) + k(n)\, \xi^*(n)$,

and

$P(n) = \lambda^{-1}P(n-1) - \lambda^{-1}k(n)\, u^H(n)\, P(n-1)$.

NLMS:

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

μ= adaptation constant

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

Where

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

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

$\mathfrak{D}(n)$ =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

Machinery Fault Simulator (MFS), a setup for simulating various types of induction motor faults initially fitted with a motor. The experimental test setup, shown in Fig. 2 consists of MFS, 1/3 HP, 3-phase, 190V, 2 pole, 50 Hz, 2850 rpm induction motors, power meter, and 3 phase auto transformer. Motor line currents are measured with YOKOGAWA power meter [Model WT500], and then transferred to the PC for further analysis.

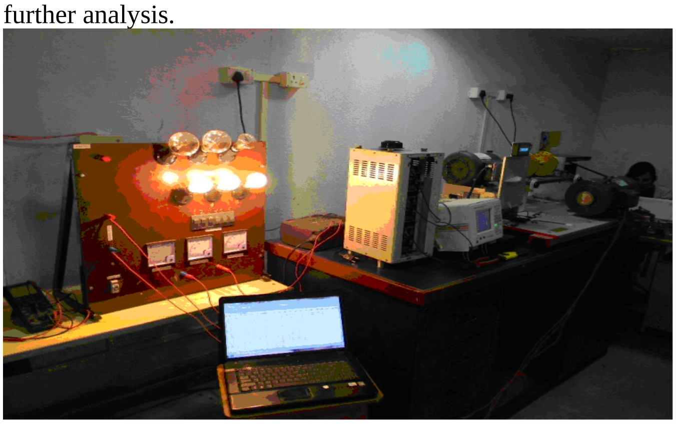

Figure 2. Experimental Setup

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 its periphery for inserting nut and bolts to create to create different degree of unbalance .

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.

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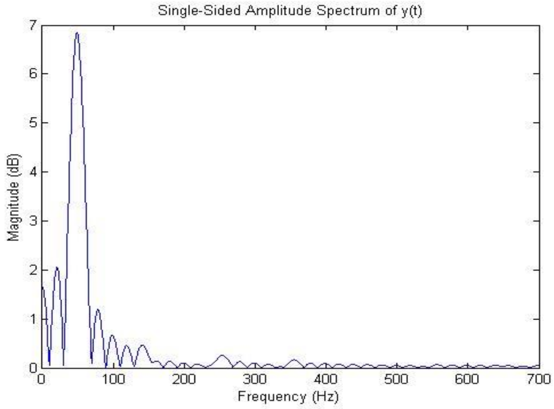
It is designed into a compact half-rack size chassis which can also be double rack mounted. Standard features include a color TFT display and USB interface for communications and memory. The instrument has a basic accuracy of 0.1%, Maximum input of 1000 V_{rms}, 40 A_{RMS} and a measurement bandwidth 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 3).

Figure 3. DC generator coupled to motor shaft.

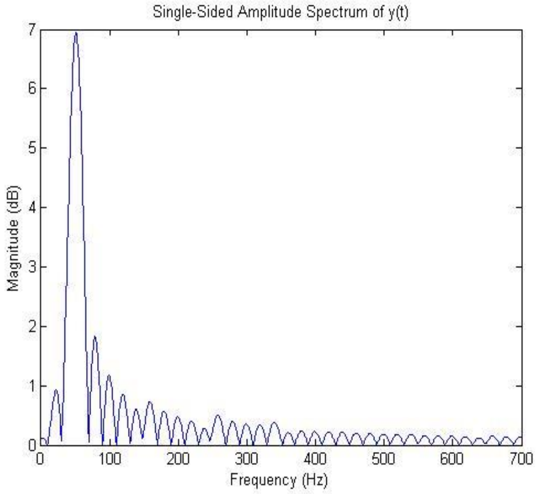
Generator generates dc voltage, which is fed to the load circuit. There are four 60watt bulbs on that load board for creating four different loading conditions shown in fig 2. Wave data is taken through PC, interfaced with power analyser.

IV. OBSERVATION OF HARDWARE SETUP DATA

In this paper, four 60 watt bulbs are used in the load board for varying loading condition. 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 4):



(a)



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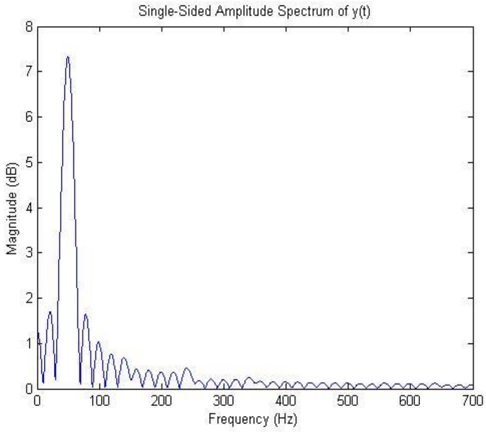

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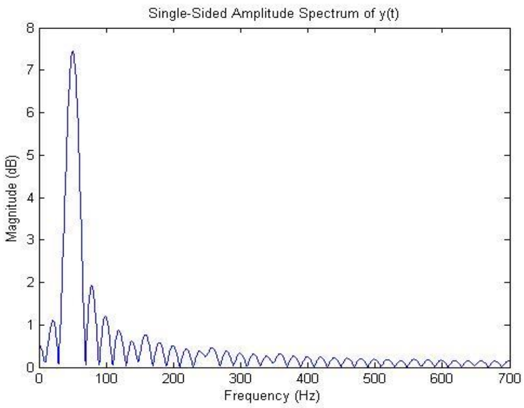


(d)

Figure 4. FFT of a phase current waveforms in different loading conditions ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

It is seen that the fundamental frequency amplitude is largest and 3rd, 5th harmonics are also prominent within the range of frequency displayed here. Amplitude of fundamental, harmonic frequencies increases when the nonlinearity in load increases, so, total harmonic distortion (THD) also varies.

V. SOFTWARE SIMULATION

Implementing MATLAB program needs the obtained wave data 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 The RLS filter at first then the same signals are fed to the Normalized LMS filter block. Filtered output is got from the output terminal of the said filter blocks respectively. Simulink model is shown in figure 5.

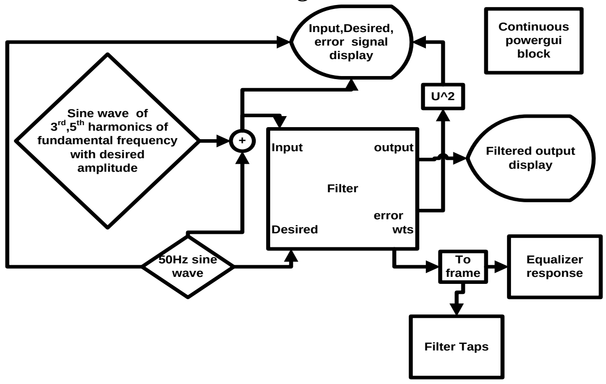
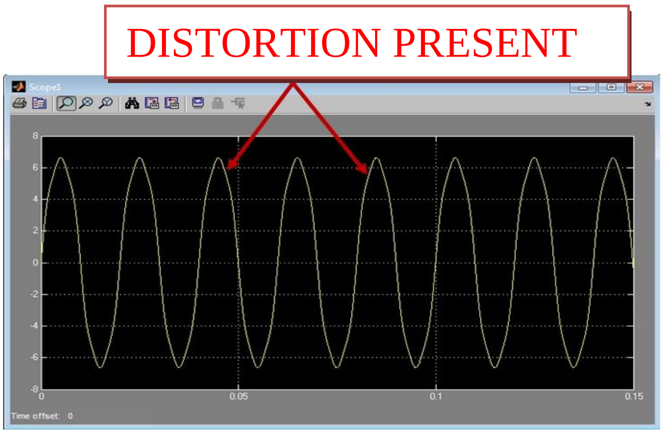


Figure 5. Simulink model of filtering technique for eliminating harmonics from input signal.

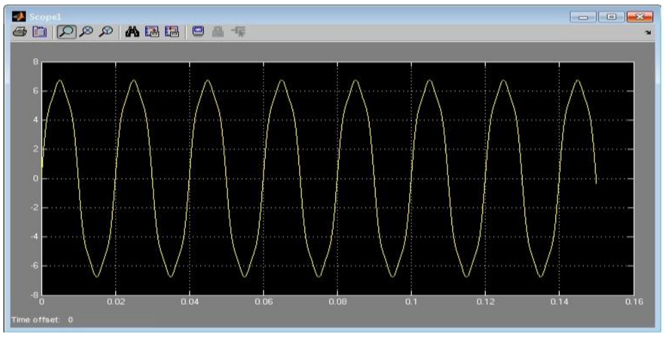
At first 50Hz sine wave has been contaminated with 3rd (150Hz), 5th (250Hz) sine waves, then the summed signals are fed to the said filter blocks respectively as input. Simultaneously, one 50Hz sine wave is fed to the desired input terminal as desired signal. On the other side of the filter block, there 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 at a time on a three input scope. Equalizer response display & filter tap (response of coefficient index Vs amplitude) display are connected with wts terminal through to frame block. Different current signals were 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 are inputs to the RLS filter as well as Normalized LMS filter.

VI. SIMULATION RESULTS OBSERVATION AND ANALYSIS

Different input waveforms for said loading conditions are obtained from the simulation model are shown below:



(a)



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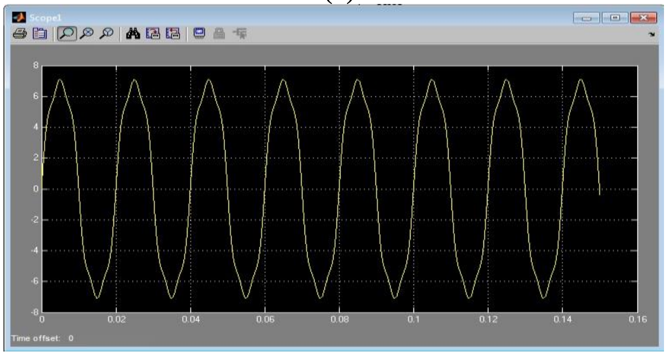

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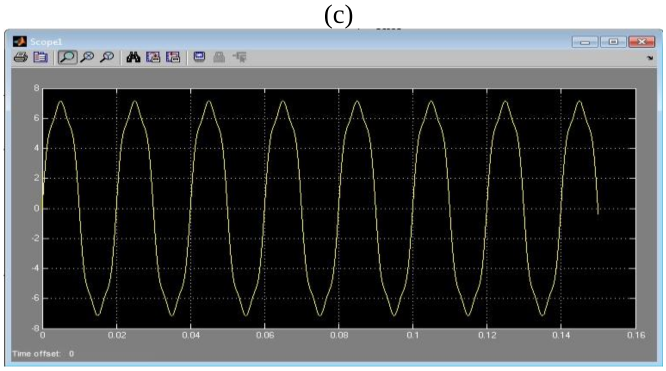
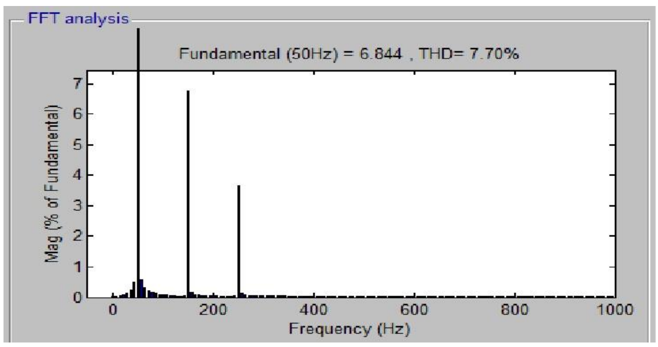


(d)

Figure 6. Input waveform to both filters ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

In figure 6, 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 clear in figure 7 shown below:



(a)

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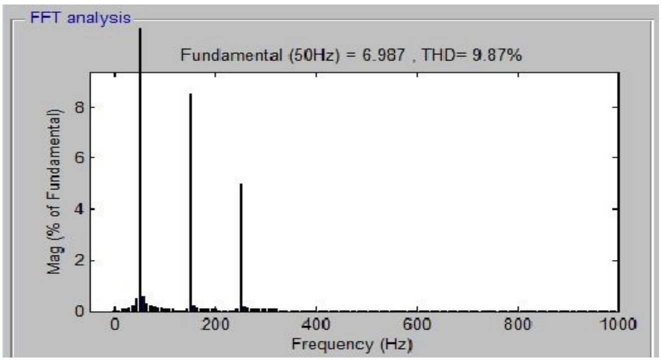

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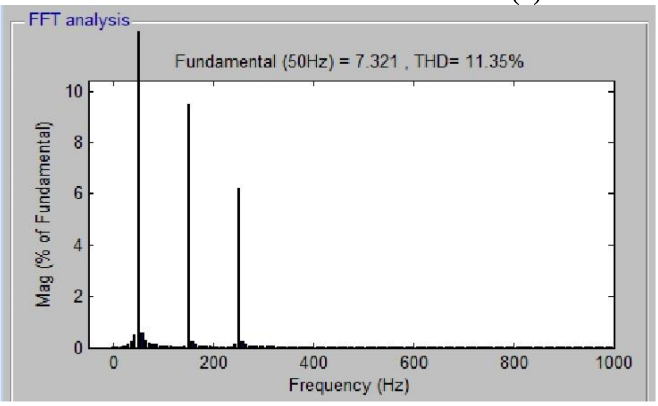

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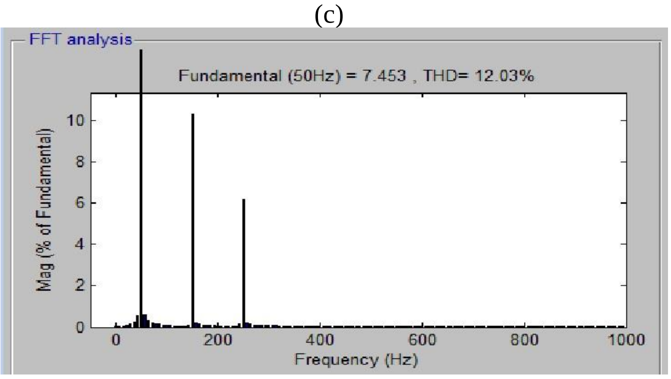
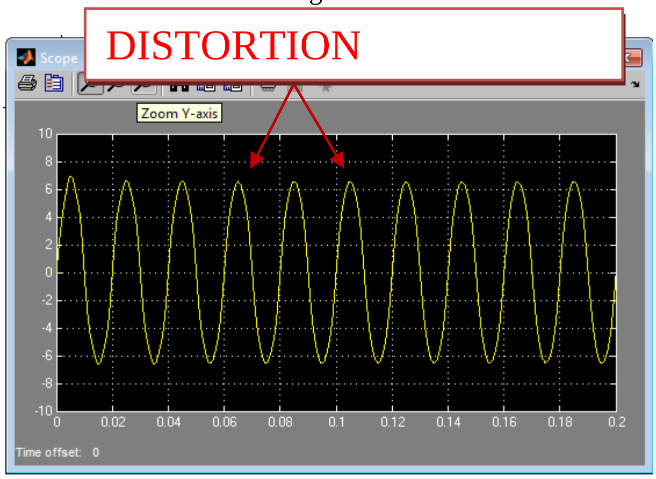


(d)

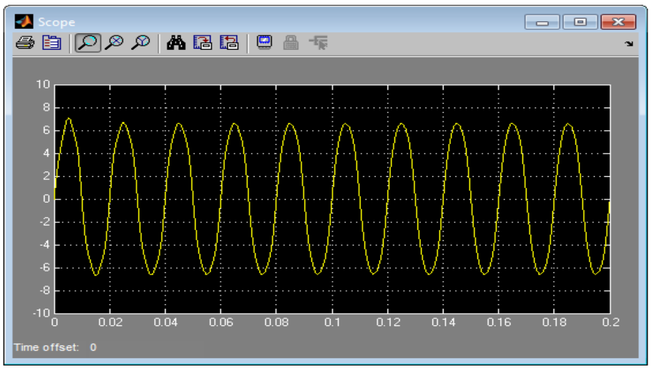
Figure 7. FFT spectrum of input waveforms ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

Here it is also observed that total harmonic distortion (THD) is varying with variation in load [19].

After filtration through the RLS filter, output waveforms of corresponding inputs for four different loading conditions have been shown below in figure 8.



(a)



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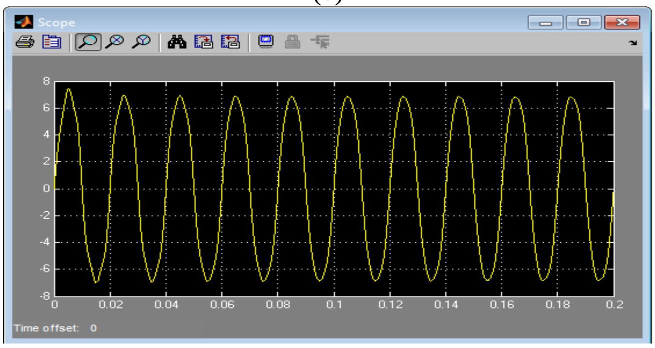

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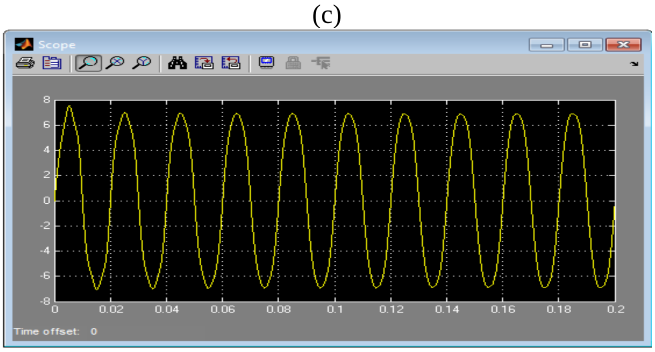
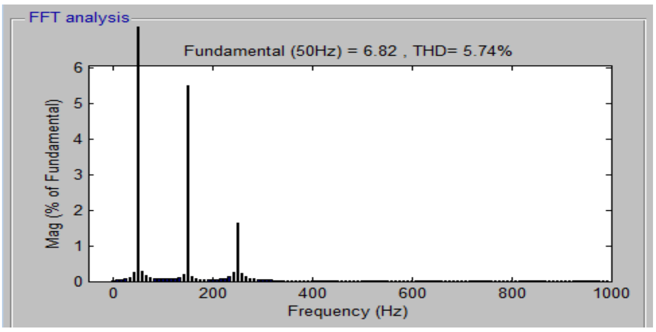


(d)

Figure 8. Output waveforms from the RLS filter ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

After filtration through the NLMS filter, output waveforms of corresponding inputs for four different loading conditions have been shown below in author's previous publication [19]. From figure 8, it is seen that the output waveforms from RLS filter are almost sinusoidal. That means most of the harmonic contents have been eliminated from the input waveforms after filtration. FFT spectrum of output waveforms from RLS filter are shown in figure 9.



(a)

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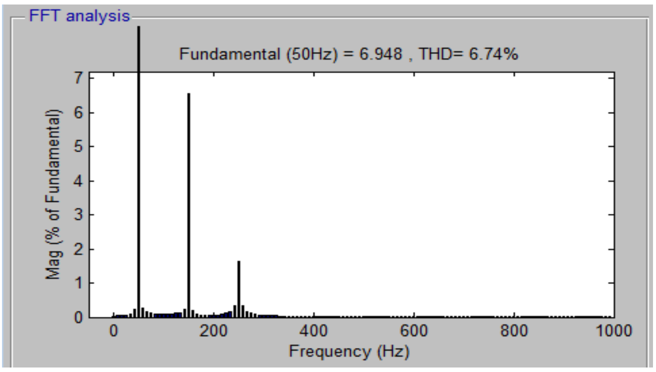

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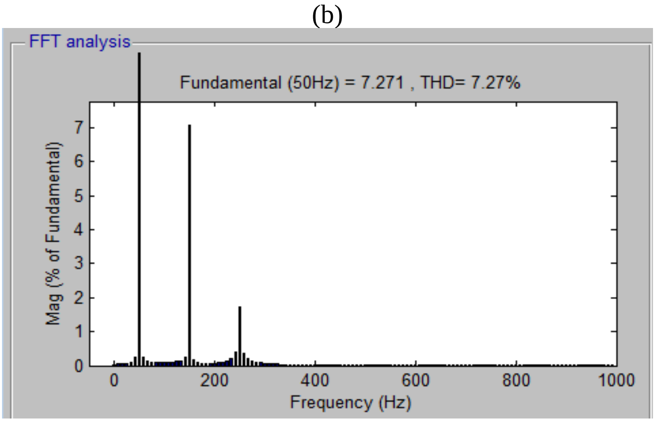

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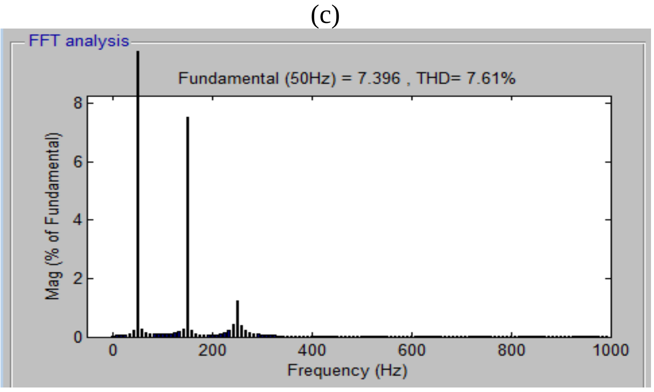
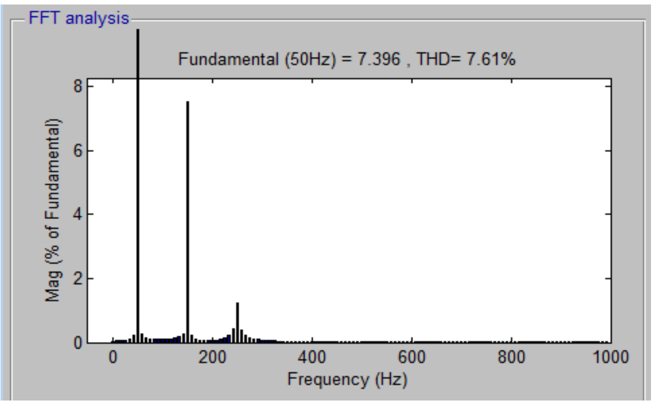


(d)

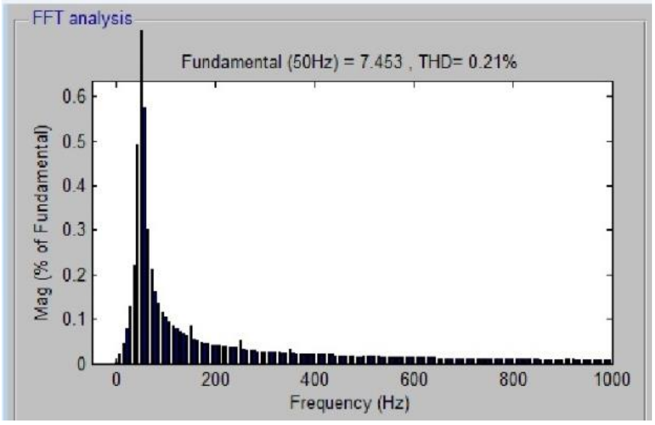
Figure 9. FFT spectrum of output waveforms ;(a) No load, (b) One bulb, (c) Three bulbs, (d) Four bulbs.

After filtration through the NLMS filter, FFT of output waveforms for four different loading conditions are discussed in detail in author’s previous journal [19].

THD comparison of output waveforms for four bulb loading case between RLS and NLMS filter is enough to distinguish their output response. Two THD output windows are given below in figure 10 to compare the responses of the said filters.



(a)



(b)

Figure 10. FFT spectrums of output waveforms for four bulb loading case ;(a) RLS filter, (b) NLMS filter.

VII. WORK ON CODE COMPOSER STUDIO

On main.c window, 50Hz sine wave with 3rd harmonic is designed. After debugging and running the program generated output array of the said designed wave is then fed to the C program which implements adaptive filter algorithm. Sample snap shot is shown in figure 11, where the said generated array is displayed within the red colored rectangular box.

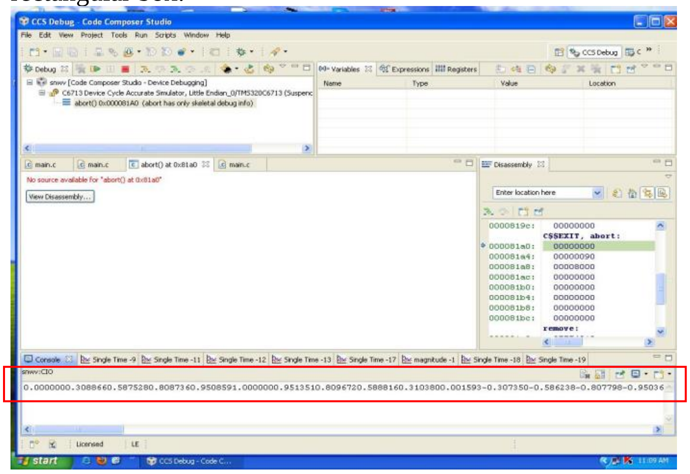


Figure 11. Screen shot to array of data generated after running 50Hz sine wave with 3rd harmonic in CCS .

Entire job is done in TMS320C6713 SIMULATOR. Waveform as input to the filter and corresponding FFT spectrum are shown below in figure 12:

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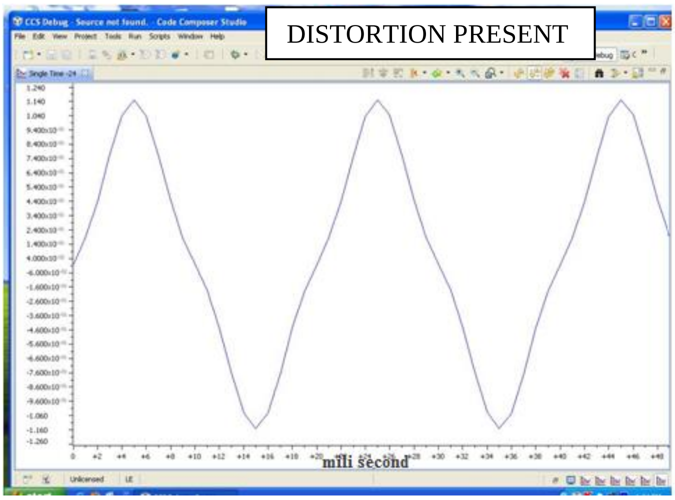
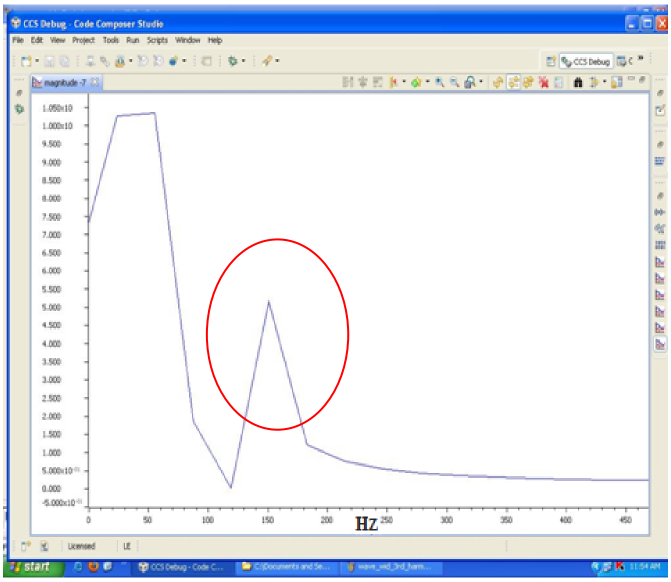


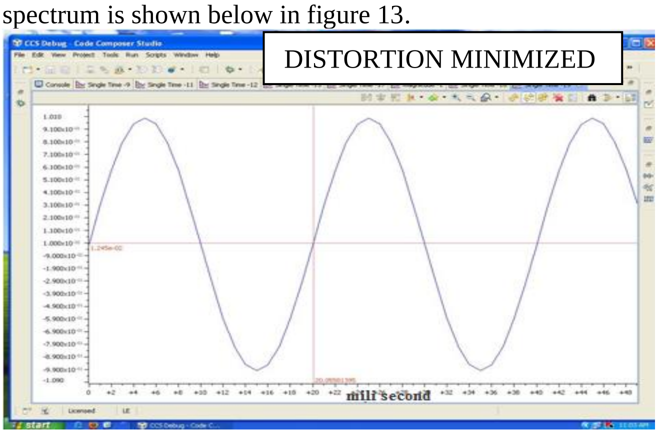
(a)



(b)

Figure 12. (a) Input to the filter in TMS320C6713 SIMULATOR (b) corresponding FFT spectrum.

After filtration, the output wave form and corresponding FFT spectrum is shown below in figure 13.



(a)

(b)

Figure 13. (a) Filter output in TMS320C6713 SIMULATOR (b) corresponding FFT spectrum.

If figure 12(b) and figure 13(b) are compared, it is observed that red marked 3rd harmonic in figure 12(b) is not present in figure 13(b). That means, 3rd harmonic has been eliminated after filtration in CCS simulator.

VIII. DISCUSSION

Here the author has taken the four bulb loading case to distinguish the performance of the two filters respectively. Because, on full loading condition motor draws maximum current and it is already proved that THD is maximum under full load condition [19]. So, four bulb case is considered as absolutely right for analysis.

Table: 1 THD% of output spectrums of NLMS & RLS filter obtained from simulation model for four bulb loading case are shown.

TIME (min)	NLMS	RLS
	THD%	THD%
0	12.14	12.14
0.15	0.18	8.69
1.5	0.18	6.18
2.5	0.18	5.89
4	0.16	5.28
5	0.14	4.75
7	0.11	3.84
10	0.09	1.74
12	0.08	0.92
15	0.06	0.96
18	0.05	0.95
20	0.05	0.94

Simulation result shows that THD of FFT spectrum after filtration reduces with the increase in time for both filtering techniques. To justify the values given in table 1, some responses of RLS filter for four bulb loading case in simulation time 0.15min, 5min, 15min, 20min respectively are shown below in fig.14.

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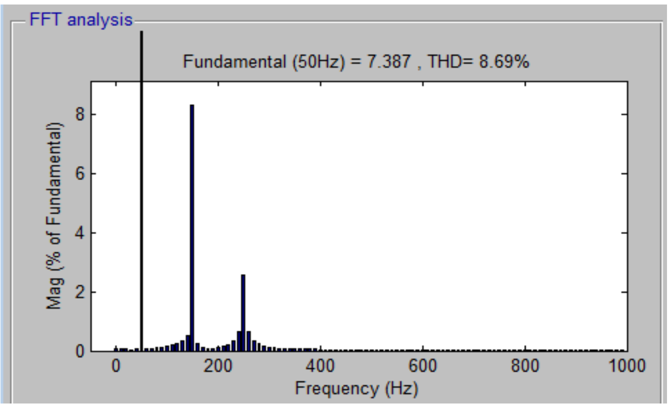
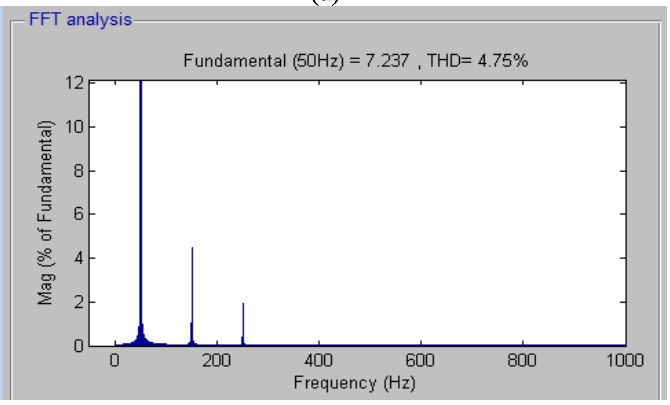


(a)



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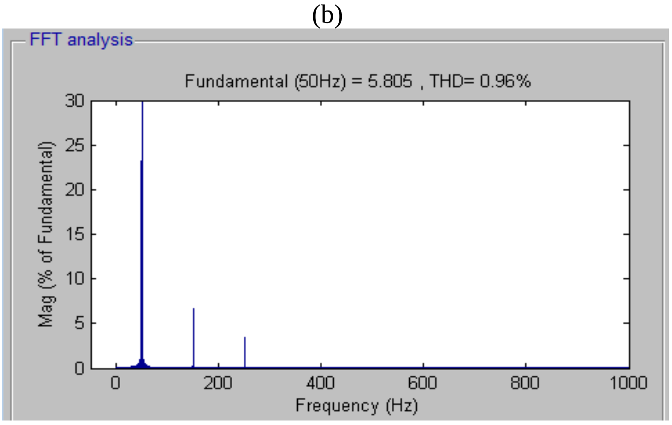

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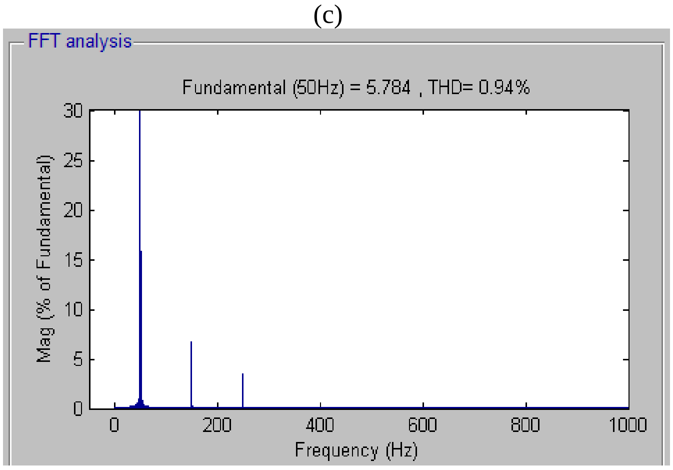
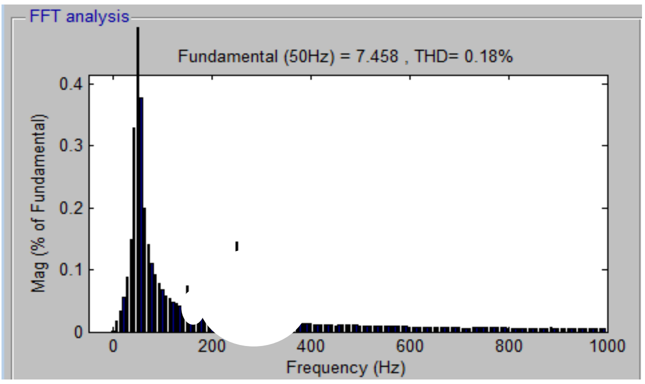


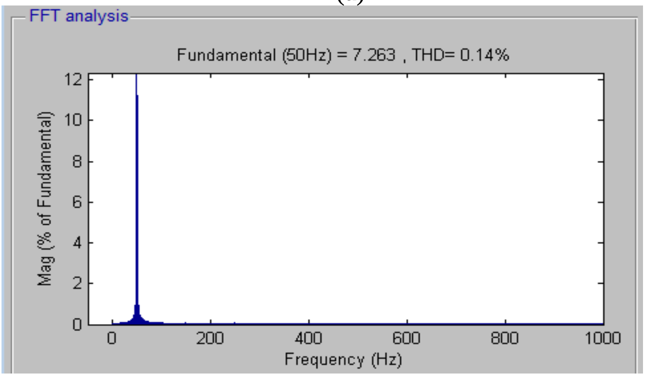
(d)

Figure 14.FFT of output spectrum from RLS filter (a) after 0.15 min, (b) after 5min, (c) after 15min & (d) after 20min.

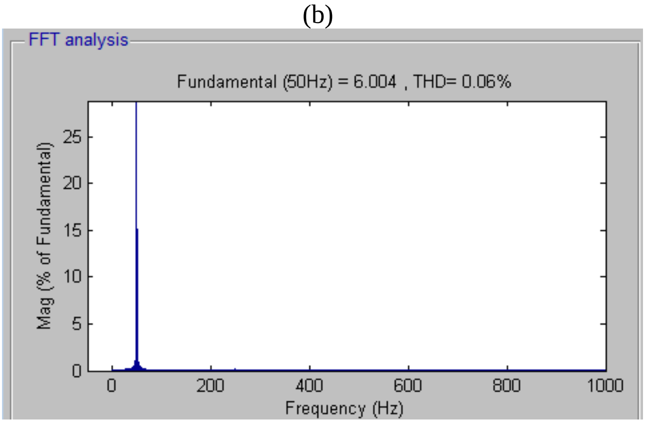
Similarly to justify the values given in table 1, some responses of NLMS filter for four bulb loading case in simulation time 0.15min, 5min, 15min, 20min respectively are shown below in fig.15.



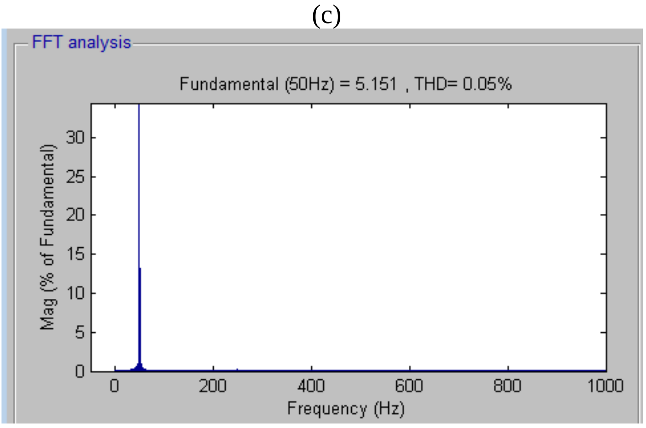
(a)



(b)



(c)



(d)

Figure 15.FFT of output spectrum from NLMS filter (a) after 0.15 min ,(b) after 5min, (c) after 15min & (d) after 20min.

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IX. CONCLUSION

Graphical representation of the contents of Table 1 shown above will help to distinguish the performance of both filters very clearly in figure 16 .

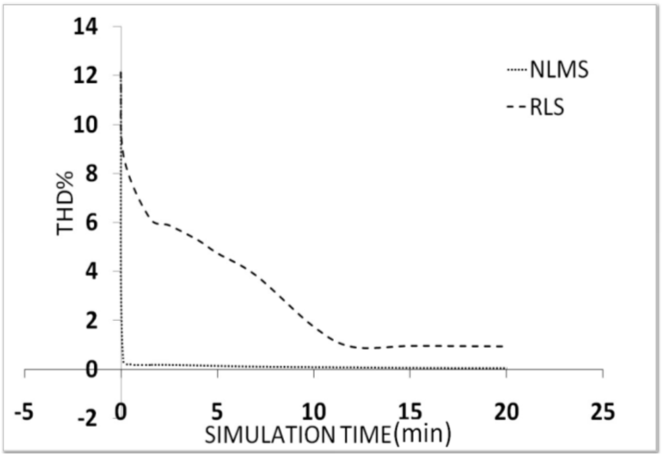

Figure 16. Plot of Simulation Time Vs. THD (%) for NLMS and RLS filter.

Percentage of THD reduction with increase in time for RLS filter is not so well like the THD reduction by NLMS filter response. Therefore, filtering technique using NLMS adaptive filter to eliminate harmonic contents is prominently effective.

This paper presents comparison between RLS & 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. At last, the conclusion is drawn as **NLMS filter is superior than RLS filter in the field of power line harmonic elimination to author's best knowledge.**

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