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HARMONICS AND INTERHARMONICS MEASUREMENT USING A NEW ADAPTIVE FILTERING METHOD BASED ON LEAST MEAN SQUARE ALGORITHM

BY

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Abstract. The paper presents a new algorithm for process the signals delivered by an inverter, necessary to power an electrical induction motor. The proposed algorithm, based on a Least Mean Square (LMS) adaptive filtering method, estimates in real-time the harmonics and interharmonics that affect the fundamental frequency of power supply voltage. The application in which the algorithm is involved is designed to monitor ten bands of harmonics along with their interharmonics. The presented method reveals better performances than other conventional measurements methods, mainly due to its possibility of real-time implementation.

Key words: adaptive filter; signal with harmonics; signal processing; signal spectrum.

1. Introduction

In recent years one can observe an important growing of nonlinear loads connected to power grids. Several examples of traditional nonlinear loads

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are: arc furnaces, rectifiers, converters, inverters, etc. To these, an increasing number of applications that use power electronic converters may be added, such as colour televisions, fluorescent tubes, automatic washing machines or voltage variators. All of these lead to appearance and propagation in the electrical networks of distorted current or voltage waveforms (Albu, 2007). The frequency of appearance of these waveforms and their amplitude, in relation with the fundamental amplitude of the voltage supply, represent the most important analysis type in electrical systems. By connecting with the power system, the distorted waveforms propagate in the electrical network where, in some cases, may occur resonance phenomena (Beites *et al.*, 1998). In order to study the periodical nonsinusoidal signals, their development in trigonometric series of sine waves (Fourier series) can be used

$$f(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} (a_n \cos n\omega t + b_n \sin n\omega t) . \qquad (1)$$

In practice, the analytical expression of the function $f(t)$ is not known, but the deformed signal shape is available by acquiring it using various devices: oscilloscopes, data acquisition boards, spectrum analysers, etc. By applying the Fourier transform, the values of a_n and b_n coefficients for the input signal can be computed and displayed.

A possible source of high frequency harmonics and interharmonics, is an electrical drive system which uses a Pulse-Width Modulation (PWM) technique. This paper presents some studies regarding the voltage waveforms picked-up across an electrical drive system that powers an induction motor with short circuit rotor having an adjustable speed. The performed study regarding the harmonics and interharmonics occurring in this system is used to quantify the distortions affecting the voltage and current waveforms in different points of the power system in order to determine where the resonance conditions are dangerous.

The Fourier transform provides the spectral content of a signal, but does not give us information about the time when the frequencies occur in the spectrum of investigated signal. Modern harmonic analysers have a drawback in data processing, resulting errors like mutual compensation of some harmonics (Kumar & Kannan, 2004; Rifai *et al.*, 2000). A harmonic analysis can be performed concerning an electrical periodic signal, in order to determine the amplitude, frequency or phase of the spectral components as well as for building the power spectrum (Kumar & Kannan, 2004). From the spectral composition of the signal, the level and frequency of the harmonic components, the frequency response of the device through which the signal passes, the nonlinearity and intermodulation distortion, the frequency stability, spectral purity and signal attenuation can be assessed (Beites *et al.*, 1998; Kumar & Kannan, 2004; Rifai *et al.*, 2000). All of these aspects reveal the practical importance of the spectral analysis for electrical signals.

In this paper, the results of an original research undertaken in order to estimate the amplitude and phase of harmonics and interharmonics components of a distorted signal using a new adaptive filtering technique are presented.

2. Experimental Test Stand

The experimental test stand is shown in Fig. 1. The electrical motor is connected to a DC circuit through a PWM inverter, which contains a three-phase bridge structure carried out using IGBT transistors (Albu, 2007). The transistors are controlled by PWM signals at a frequency of 5,060 Hz, slightly variable. The PWM inverter powers a squirrel cage induction motor having a rated power at 0.37 kW and the nominal speed of 1,320 rpm. The voltage and current flowing through a phase of the motor are acquired through a measurement system using Hall-type sensors (LEM modules) (Albu, 2007). The collected data was stored on a portable PC to be analysed off-line using the MATLAB software. For acquiring the data an NI USB-6251 acquisition board was used, working at a sampling rate of 500 kSamples/s. Also, this experimental set-up was used to test the proposed algorithm in real-time.

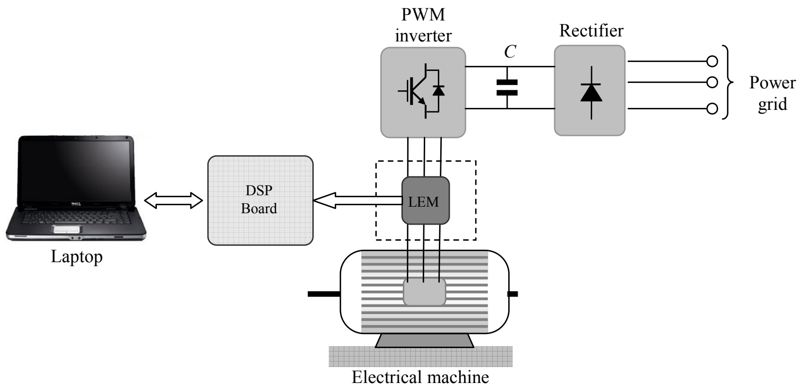

Fig. 1 - Experimental test stand structure

2.1. Experimental Data

The voltage and current waveforms, acquired from the power supply, are presented in Fig. 2.

After acquiring the signals, a Discrete Fourier Transform (DFT) has been performed using MATLAB in order to obtain the signal spectrum. The results are presented in Fig. 3. Also, the statistical analysis concerning the probability of occurring of harmonics and interharmonics in the measured signal was performed according to literature indications. From these analyses it can observe that the appearance of harmonics in signal spectrum is around the dotted line (Fig. 4). Another type of analysis that was performed off-line is the computing of the spectral power density, as depicted in Fig. 5.

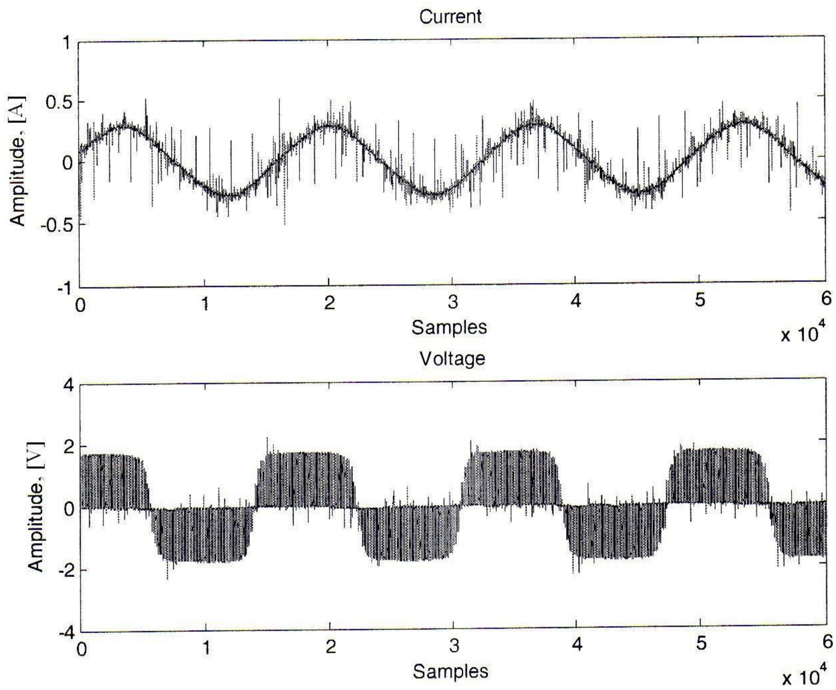

Fig. 2 – Waveform of the acquired voltages and currents: *a* – current, *b* – voltage.

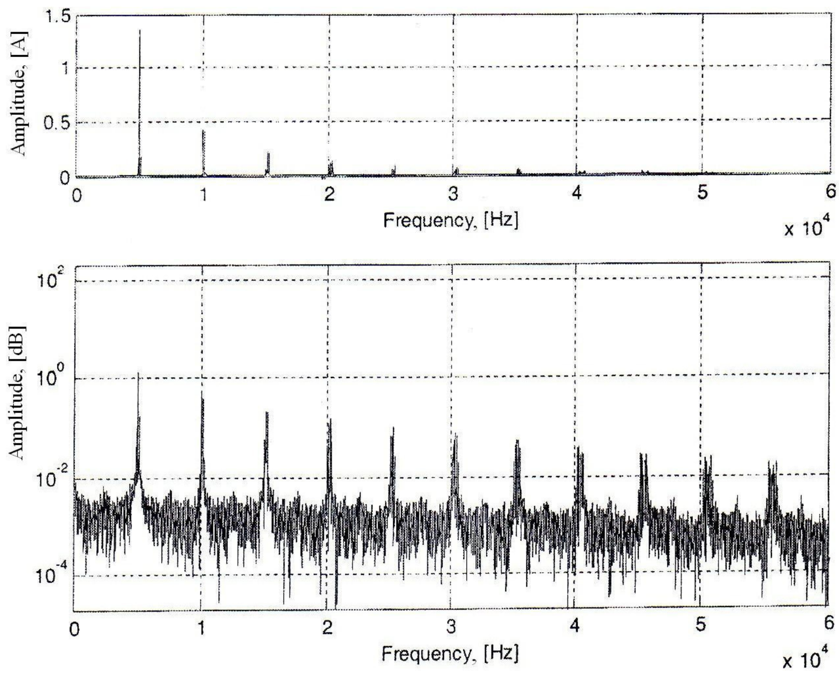

Fig. 3 – Representation in frequency domain of the voltage signal.

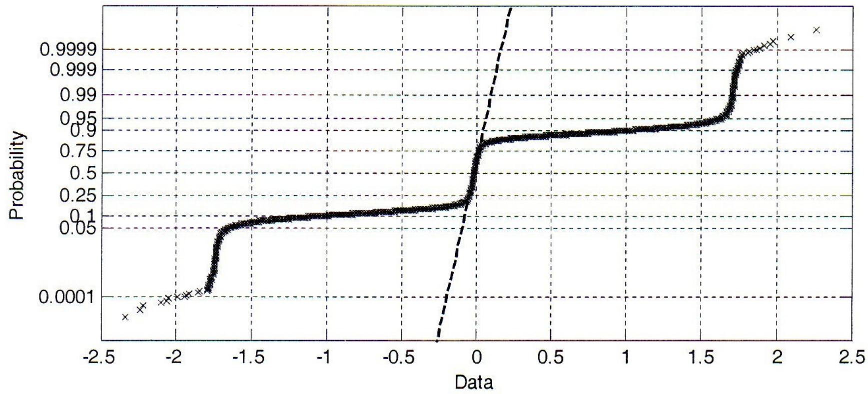

Fig. 4 – Probability of harmonics and interharmonics appearing in the measured signal.

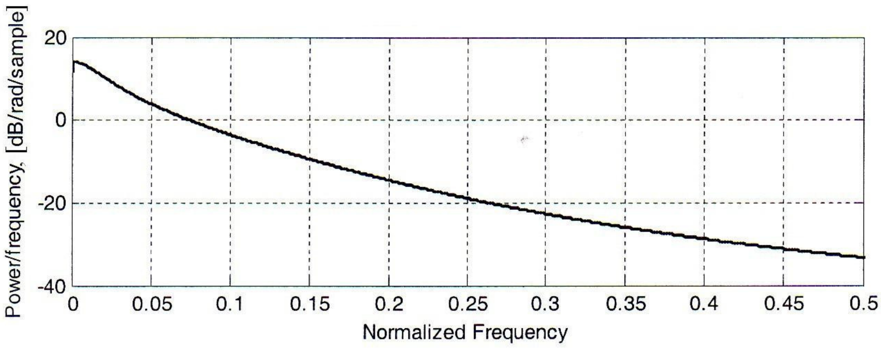

Fig. 5 – Representation of spectral power density.

3. Algorithm Presentation

The proposed algorithm has been employed in our research for evaluating the amplitude and phases of harmonics and interharmonics for the signal delivered by the PWM inverter, as an alternative to the DFT with the aim of improving the time cost in detecting the input signal frequencies. The algorithm uses an adaptive filter structure which is implemented using the Least Mean Square (LMS) method as in the papers published by Dash *et al*. (1996), Sarkar *et al.* (2011). These works were reported for estimating the harmonics and interharmonics, but with more time cost in detecting the input signal frequencies. The proposed idea in this paper is to develop an adaptive structure working with an *a priori* known commutation frequency for the PWM and also knowing the fundamental frequency of the signal. Let us assume f_c to be the commutation frequency of the PWM inverter, and f_1 the fundamental (nominal) frequency of the motor power supply. In Fig. 6 is presented the harmonics spectrum of the PWM voltage delivered by the inverter working in linear domain.

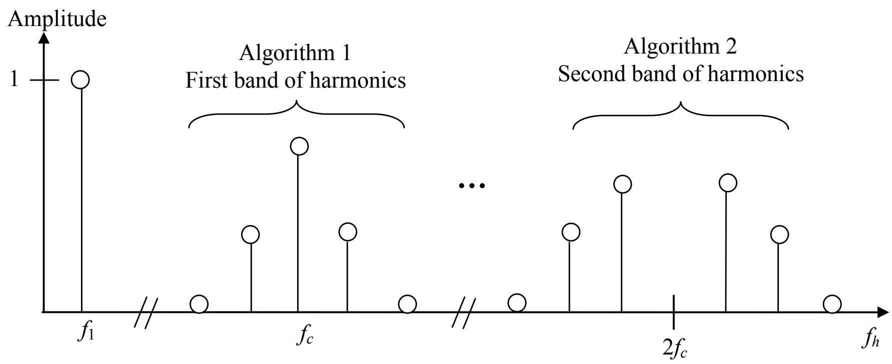

Fig. 6 – Harmonics spectrum of voltage of PWM inverter working in linear domain.

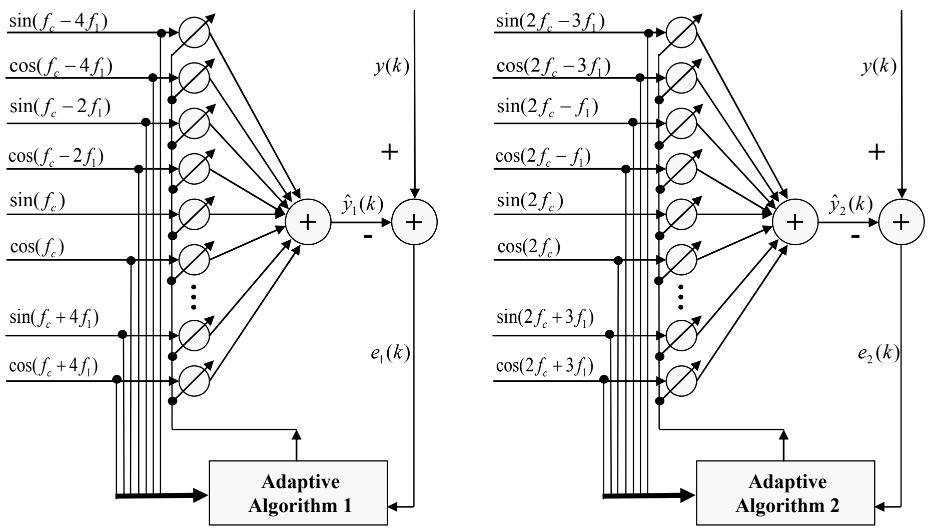

Fig. 7 – The proposed structure for odd and even interharmonics evaluation.

In Fig. 7 the proposed structure for odd and even interharmonics evaluation is presented. The input signal in discrete form is

$$y(k)=\sum_{l=1}^{N}A_l\sin\left(\frac{2\pi lk}{N_s}+\phi_l\right)=\sum_{l=1}^{N}A_{ld}\cos\phi_l\sin\frac{2\pi lk}{N_s}+\sum_{l=1}^{N}A_{lq}\sin\phi_l\cos\frac{2\pi lk}{N_s},\quad(2)$$

where: N_s is the sampling rate, defined by $N_s = f_s/f_1$, f_1 represents the nominal frequency and f_s is the sampling frequency. From the Fig. 7 it may be easily recognized that the input tap in adaptive filter is composed by

$$X(k)=\left[\sin\frac{2\pi k}{N_s}\quad \cos\frac{2\pi k}{N_s}\quad \sin\frac{4\pi k}{N_s}\quad \cos\frac{4\pi k}{N_s}\quad ...\quad \sin\frac{2N\pi k}{N_s}\quad \cos\frac{2N\pi k}{N_s}\right]. \quad (3)$$

The algorithm which adapts the filter weights, depicted in Fig. 7, follows the rule given by

$$w(k+1)=w(k)+\frac{\mu e(k)X(k)}{X^T(k)X(k)}, \quad (4)$$

where k represents the index of algorithm iterations.

The vector representing the filter coefficients at the moment k is

$$W(k)=[w_1(k)\ w_2(k)\ w_3(k)\ w_4(k)\ ...\ w_{2N-1}(k)\ w_{2N}(k)\ w_{2N+1}(k)\ w_{2N+2}(k)]^T. \quad (5)$$

Considering $X(k)$ the input vector at the moment k, one may calculate the error, $e(k)$, with the following relation:

$$e(k)=y(k)-\hat{y}(k), \quad (6)$$

where: $\hat{y}(k)$ represents the estimated amplitude at the moment k, $y(k)$ is the actual amplitude at the same moment and parameter μ – a learning constant that is necessary in the algorithm convergence.

Once the error signal converges to zero, the vector containing the filter coefficients becomes

$$W_0=[A_1\cos\phi_1\ \ A_1\sin\phi_1\ \ A_2\cos\phi_2\ \ A_2\sin\phi_2\ ...\ A_N\cos\phi_N\ \ A_N\sin\phi_N]^T. \quad (7)$$

The amplitude and the phase of the estimated components for N-th harmonics or interharmonics are given by

$$\begin{cases} A_N=\sqrt{W_0^2(2N-1)+W_0^2(2N)}, \\ \phi_N=\tan^{-1}\left[\dfrac{W_0(2N-1)}{W_0(2N)}\right]. \end{cases} \quad (8)$$

4. Algorithm Implementation on Digital Signal Processor

In order to test the proposed algorithm, we implemented it to run on a Digital Signal Processor (DSP) type TMS320C6713 produced by Texas Instruments. This processor is embedded onto an application development board, DSK 6713, produced by Digital Spectrum. This implementation is suitable for testing real-time signal processing algorithms. In Fig. 8 the flowchart of adaptive estimation of filter coefficients using the LMS algorithm (4) is presented.

First , the program performs some hardware initializations for setting-up the ADC/DAC parameters, the timer, the sampling frequency, the adaptation coefficient, μ, the weight vector, the estimated output from filter and sampling order, *k,* the switching frequency and also the fundamental frequency. Each moment when a sample is acquired, a timer interrupt occurs, after which the timer counter is reset and the algorithm is computed to estimate actual weights, following the rule (4). In order to create the error signal (6), it is necessary to acquire the measurement voltage from PWM converter, as shown in Fig. 1. The input vector (3) is computed by applying the lookup table to create the sine and the cosine components of harmonics and interharmonics signals, presented in

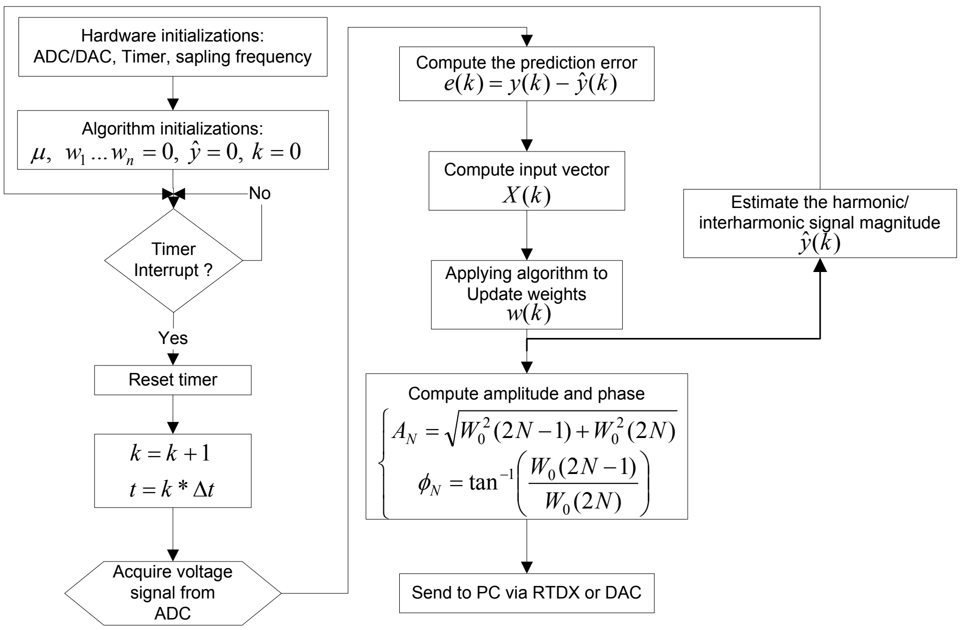

Fig. 8 – Flowchart of adaptive estimation of filter coefficients using LMS algorithm.

Figs. 6 and 7. The estimated component for *N*-th harmonics or interharmonics (8) are transmitted by RTDX protocol to PC, where a visual interface built-in LabVIEW or MATLAB is used to present the estimation of these parameters. A more detailed implementation flowchart is depicted in Fig. 9. This flowchart is an original point of view of the proposed implementation, to estimate in real-time using DSP hardware. The signal is sampled and A/D converted using an ADS8361 ADC working at 500 kHz sampling rate and 16 bit resolution. The DSP has been programmed using the Code Composer Studio (s. § **3.1**).

From this figure it is possible to see that the application is designed to estimate 10 bands of harmonics and their interharmonics. It was necessary to implement 10 different lookup tables, one for each adaptive structure, that estimate a band of interharmonics.

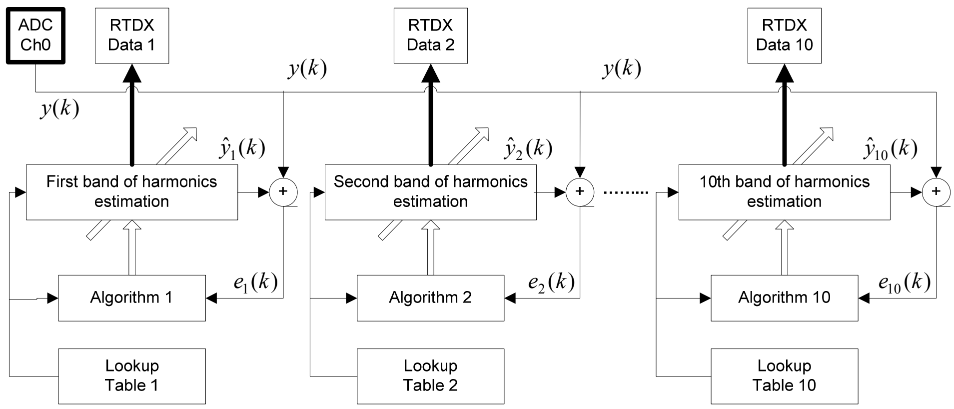

Fig. 9 – The proposed application flowchart.

4. Results and Discussions

In Figs. 10 and 11 are depicted four bands of harmonics and their inter-harmonics, determined with the proposed algorithm in real time. Around of switching frequency, the interharmonic components have been estimated (Fig. 10). In Fig. 11 there were estimated the components from the 2-nd, 3-rd and 4-th band of interharmonics.

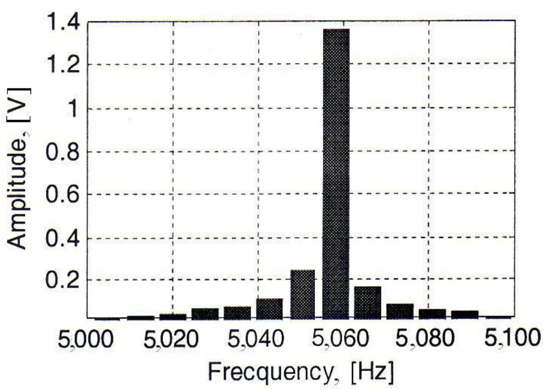

Fig. 10 – Detected fundamental frequency.

The estimated values of the components are updated in real time, which makes this analysis by using the algorithm proposed to be very proper. The algorithm is the independent of switching frequency and it can track small or large variations of frequency value in real time. Also the algorithm can be implemented on FPGA circuits quite easily.

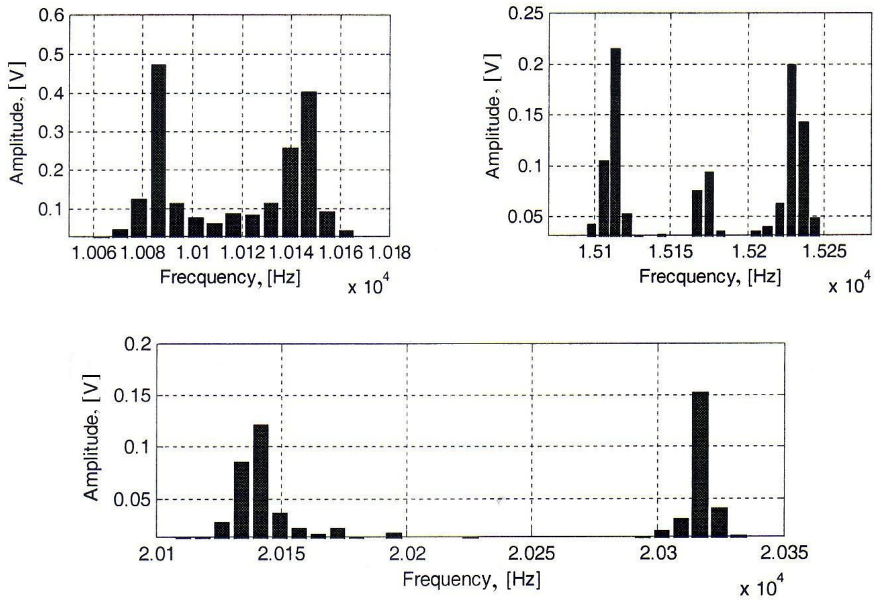

Fig. 11 – Results obtained with the proposed algorithm.

6. Conclusions

Nowadays, with existing modern technology, can be implemented easily the real-time measurement systems, with great utility in industrial measurements. So, the main novelty presented in this paper is the adaptive estimation algorithm of harmonics and interharmonics from a distorted signal. Estimation is done on tapes of even and odd interharmonics. Working speed of algorithm is due to the real-time implementation on a DSP processor. The proposed algorithm has good results in estimation of harmonics and interharmonics in real-time. The obtained results show that this measurement solution is powerful and easily to implement on DSP processors.

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*** http://www.mathworks.com/help/techdoc/ref/fft.html

*** http://www.mathworks.com/matlabcentral/answers/507-normal-distribution-range

MĂSURAREA ARMONICILOR ŞI INTERARMONICILOR UTILIZÎND O NOUĂ METODĂ DE FILTRARE ADAPTIVĂ BAZATĂ PE ALGORITMUL LMS

(Rezumat)

Se prezintă un punct de vedere original cu privire la prelucrarea semnalelor provenite de la un invertor PWM care este necesar pentru alimentarea cu energie electrică a unui motor de inducție. Algoritmul propus estimează în timp real componentele de armonici si interarmonici care afectează frecvența fundamentală a tensiunii de alimentare. Aplicația este proiectată pentru a monitoriza 10 benzi de armonici împreună cu interarmonicile lor pe baza algoritmului LMS într-o structură nouă. Sistemul de măsurare prezentat este mai bun decât alte sisteme clasice. Monitorizarea armonicilor şi a interarmonicilor este utilă în control şi în procesele industriale.