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Original Contribution

STUDY OF APPLICATION OF ACTIVE LC FILTER FOR DETECTION AND REDUCTION OF HARMONICS IN POWER SYSTEM

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ABSTRACT
Taking into account the increased rate of nonlinear loads in power systems, recognition and extraction of harmonics and noise seems necessary. The duty of active filter is to identify these non sinusoid currents and take action to cancel them. In this paper, adaptive method is examined and it is observed that adaptive filter might develop appropriate performance in various conditions. Generally speaking, adaptive filter is defined as a filter that its specifications might be changed to achieve to a specific goal and have this change or adaptation automatically and without the need for any intervention. Among the features of this filter, one might refer to its simplicity and powerful structure.

Key words: invertor, Hysteresis method, power system, adaptive filter, harmonic

INTRODUCTION

In power systems we need to recognize and extract harmonics and noise simultaneously. Now, there are many methods for this purpose which the majority of them are complex and are difficult to adjust and implement. Discrete Fourier Transfer function (DFT) which has accepted because of its simplicity, is well-known for its low accuracy in transient states.

Instantaneous Reactive Power Theory (IRPT) is another method which is difficult to adjust and implement. Its primary limitation is that it defines Instantaneous reactive power only in three phase systems and is not applicable in single phase systems. Both of analog and digital implementation of this method, has a complex structure. Digital implementation of this method requires powerful microprocessors and quick analog-to-digital and vice versa invertors, due to the huge amount of mathematical operations which is needed.

Notch filter is used as well for extraction of the main component of the noise signal. A notch filter with a bandwidth of 50 *Hz* has a good characteristic, but is impossible to have a quick response and good harmonic cancellation with the use of this filter. The effectiveness of adaptive method has done by simulation and experimental implementation [1-4].

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THE STRUCTURE OF ADAPTIVE FILTER

The structure of adaptive filter is shown in **Figure 1.**

Assuming voltage signal hasn't been disturbed the regarding filter can be set as the delete current signal reactive. The construction can be used for the detection of harmonic in active filter successfully.

Convergence of algorithm speed can be increased by using of a simple low path filter that will improve frequency response.

When μ_i increases, transient speed of filter increase also and in the same condition vibrations create the stability response. Filter LIP has been considered in **figure 1** so the measurement of vibration without any decrease delete in transient response. In fact correct drawing LIP can improve stable state response well. **Figure 2** shows main current copmonent in two circuts with LPF and without LPF.

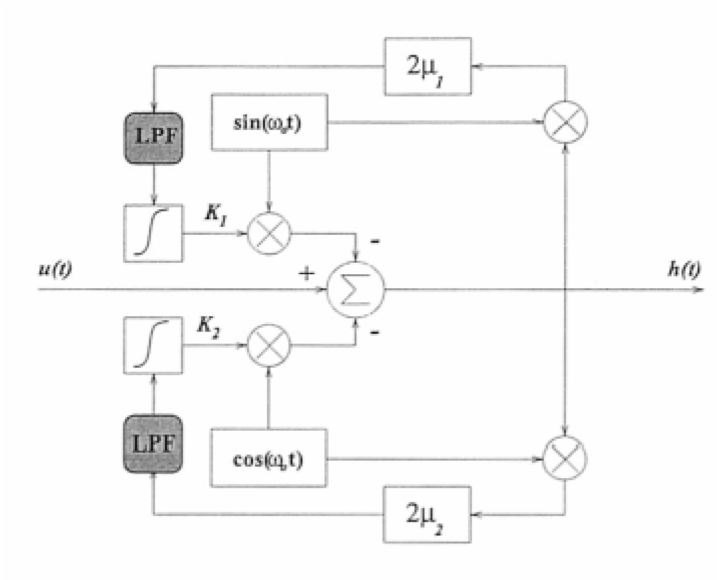

Figure 1. Adaptive filter block diagram for the detection of harmonics

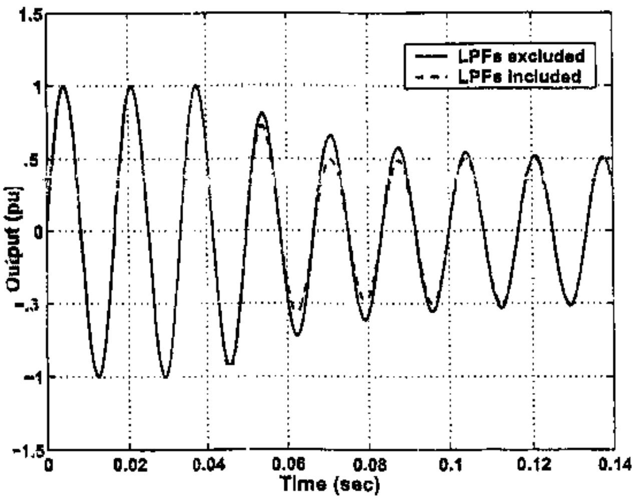

Figure 2. Main current copmonent with LPF and without LPF

FILTER MODEL BASED ON SQUARES MINIMUM METHOD

To assume *u(t)* is a continuous function in figure 1 that can be defined as follow.

$$u(t)=\sum_{n=0}^{\infty}[A_n \sin(n\omega_o t)+B_n \cos(n\omega_o t)]. \quad (1)$$

The basic component of this sigral is as follow:

$$u(t)=A_1 \sin(n\omega_o t)+B_1 \cos(n\omega_o t) \quad (2)$$

In practical condition where signals are instable, all of the parameters Eq.1 may become variables in time unit. According to squares minimum method, function J should be have its minimum valve to estimate major signal.

$$J(K_1,K_2)=(u-y)(K_1,K_2))^2 \cong h^2 \quad (3)$$

K_1 and K_2 are approximate of A_1 and B_1

$$y(t)=K_1(t)\sin(\omega_o t)+K_2(t)\cos(\omega_o t) \quad (4)$$

Function $h(t)=y(t)-i(t)$ is used to estimate harmonic *u(t).*

Vectors K_1 and K_2 can estimate by using of decreasing gradient method

$$\frac{d}{dt}(K_1,K_2)(t)=-\mu\frac{\partial J(K_1,K_2)}{\partial(K_1,K_2)} \quad (5)$$

The matrix $\mu(2\times 2)$ is learning rate algorithm and show algorithm convergence.
Vectors K_1 and K_2 are specified by following equations.

$$\dot{K}_1(t) = 2\mu_1 \sin(\omega_o t)h(t)$$

$$\dot{K}_2(t) = 2\mu_2 \cos(\omega_o t)h(t) \quad (6)$$

This dynamic system can be introduced as an according filter that it`s input and output, are *u(t)* and *h(t)*.

By using of equation $h = u - y$ and replacement of (4) in (6):

$$\dot{K}(t) = \begin{bmatrix} -2\mu_1 \sin^2(\omega_o t) & -2\mu_1 \sin(\omega_o t)\cos(\omega_o t) \\ -2\mu_2 \sin(\omega_o t)\cos(\omega_o t) & -2\mu_2 \cos^2(\omega_o t) \end{bmatrix}$$

$$K(t) + \begin{bmatrix} 2\mu_1 \sin(\omega_o t) \\ 2\mu_2 \sin(\omega_o t) \end{bmatrix} u(t)$$

Which *K(t)* is determined like $[K_1(t), K_2(t)]^T$ .The filter structure of Figure 1 is simple and is simply dismountable by mathematics and trigonometry relation , both in the form of software (DSP) and hardware [5].

Another aspect, is the non-sensitivity of filter to the parameters of μ_1 and μ_2. These parameters in algorithm without any rigid effect on it`s proficiency increase the transient speed. In the structure of given filter, if we increase the parameter μ , it is possible that we gain faster respone. Nevertheless, this act generates vibrations in a constant state in the cases that noises and harmonics. Thus, there should be coordination between the rate of convergence and vibrations.

In some application, the transient response isn`t more than two given cycles and the error of constant state versus the riot of input signal, shouldn`t is more than *2%*. Now, we consider the resistive rectified load current and investigate the performance of filter. **Figure 3** shows phase current of the three phases rectifier and **Figure 3a and Figure 3b** show phase current after removing waste components and identified waste components respectively.

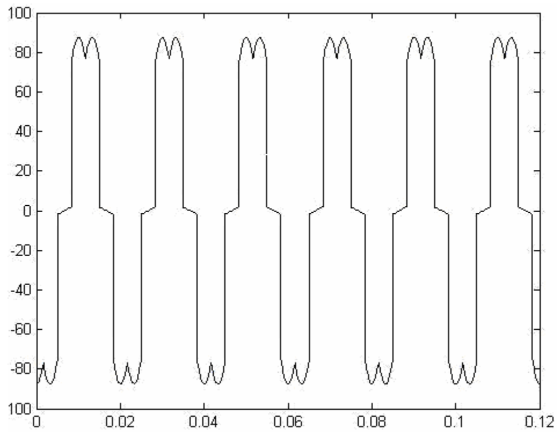

Figure 3. Phase current of the three phases Rectifier

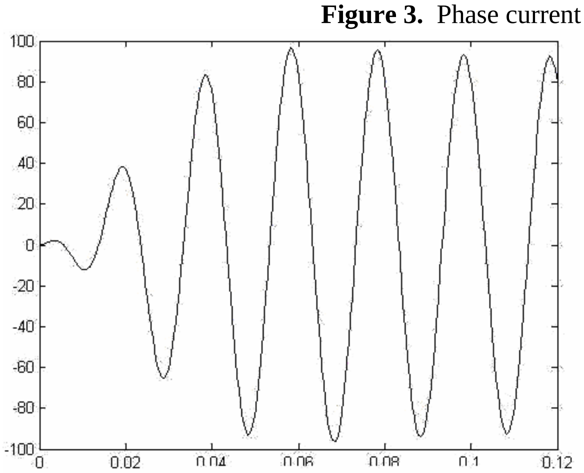

Figure 3a. Phase current after removing waste components

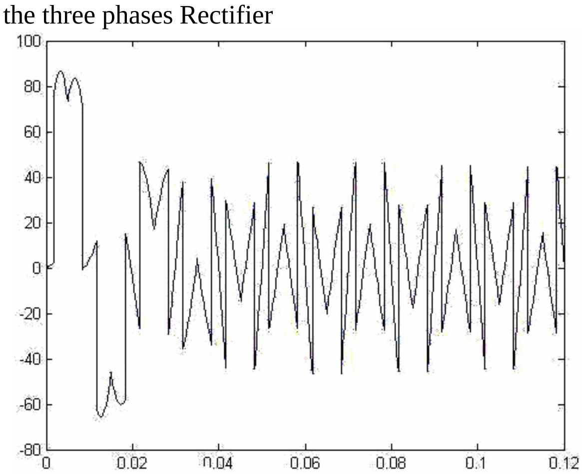

Figure 3b. Identified waste components

Now, if in the amplitude of input current, a change is gained this method accommodates itself to these changes rapidly. **Figure 4** show load current range with a 50 % reduction in amplitude. **Figure 4a and Figure 4b** show waste component of the load current and the fundamental components of the load current respecticely.

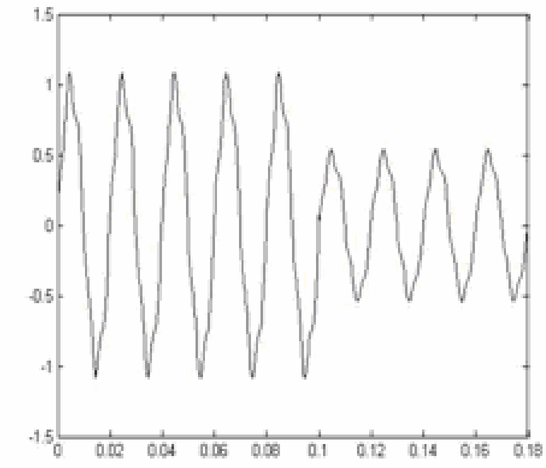

Figure 4. Load current range with a 50 % reduction in amplitude

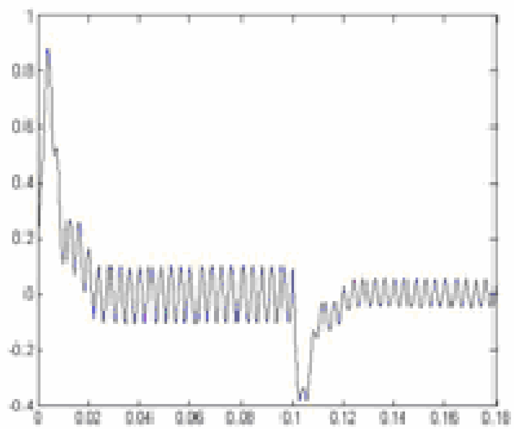

Fig. 4a Waste component of the load current

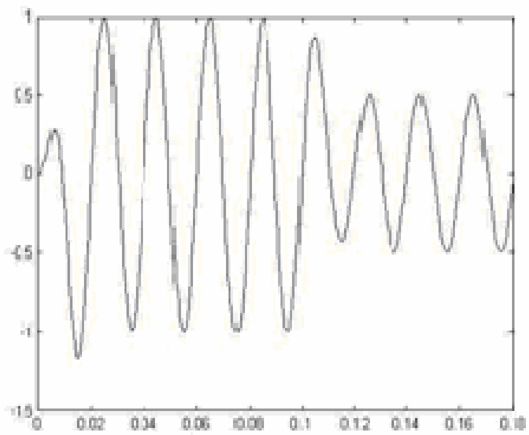

Fig. 4b. The fundamental components of the load current

Figure 5 shows the actual current associated with the phase that is measured by oscilloscope and the performance of the adaptive structure to remove harmonics and disturbances. **Figure 5a and Figure 5b** waste component of the current and current phases, after removing harmonics respectively.

SIMULATION AND RESULTS

In this section, the three-phase active powers filter (APF) is simulated that is used in the identification of waste components with the adaptive method. The given load is considered as a three-phase diode rectifier and assumed rectifier load must be an ohmic resistance which would result higher THD. The desired filter is considered a shunt with VSI inverter and hysteresis control switching is selected that is one kind of PWM switching. For planning an active filter to eliminate the line current harmonics and simulate line current to a pure sinusoidal, two stages are done as follow:

- Sampling of the line current
- To obtain the reference current using adaptive method

To create pulse for switching inverter was used hysteresis method which is extracted from comparing the obtained source current from identification methods and inverter output current.

In the last part of inverter output, to improve the quality of active power and remove harmonic of switching, is used a passive filter. LC filter has a simple design and reduces PWM harmonics which is selected $L=0.1\ mh$ and $C=10^{-5}\ F$ for inductor and capacitor, respectively. Meanwhile, the hysteresis bandwidth is considered 5 (h=5).

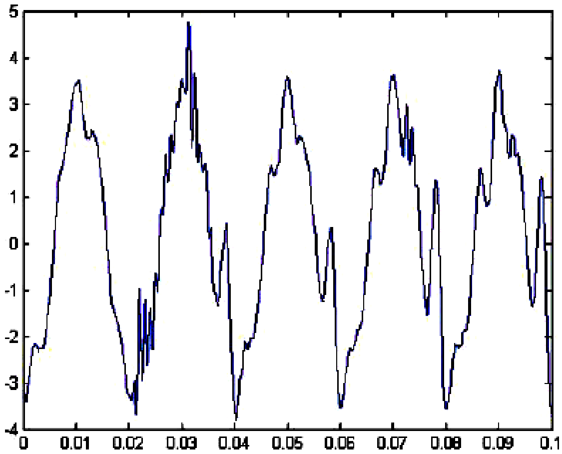

Figure 5. The actual current related to the single-phase

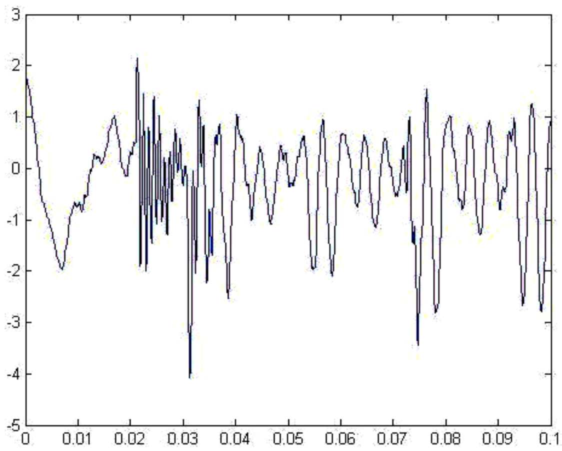

Figure 5a. waste component of the current

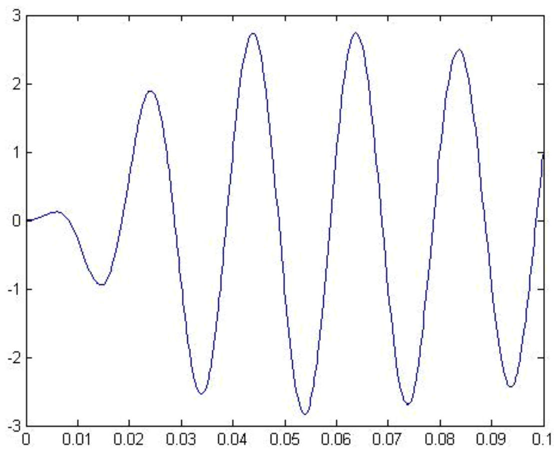

Figure 5b. current phases, after removing harmonics

In the VSI inverter, source voltage is used instead of a capacitor and the size of the source should be approximately 1.5 times the maximum voltage phases because filter active should always be having a good performance. This value is obtained according to the following calculation. (Conduction time is considered 180°).

V_{ab} voltage will develop in the following fourier series:

$$V_{ab} = \sum_{n=1,3,5}^{\infty} [\frac{4V_S}{n\pi}\cos(\frac{n\pi}{6})\sin n(\omega t + \pi/6)]. \quad (7)$$

Consequently, line voltage to rms line at inverter output:

$$V_{ab,rms} = \frac{4Vs}{\sqrt{2\pi}}\cos\frac{n\pi}{6} \quad (8)$$

The amount of main component of line voltage to rms line at inverter output:

$$V_{1,ab,rms} = \frac{4Vs}{\sqrt{2\pi}}\cos\frac{\pi}{6} \quad (9)$$

Maximum phase voltage at output inverter:

$$V_{ph-n} = \frac{\sqrt{2}}{\sqrt{3}}\frac{4V_s\cos 30}{\sqrt{2\pi}} \Rightarrow V_S = V_{dc} = 1.56V_{ph} \quad (10)$$

In the **Figure 6**, is used the adaptive method in I_{ref} part for all three phases.

Figures 7 and 8 show the load current of single phase and Main component of current respecticely.

CONCLUSION

An adaptive filter for detection and extraction of harmonics is provided. It is observed that the filter can result a good performance in different conditions and update itself with changes of input. Among the advantages of this filter are its easily and robust structure which lends itself to easy implementations in hardware and software environments.

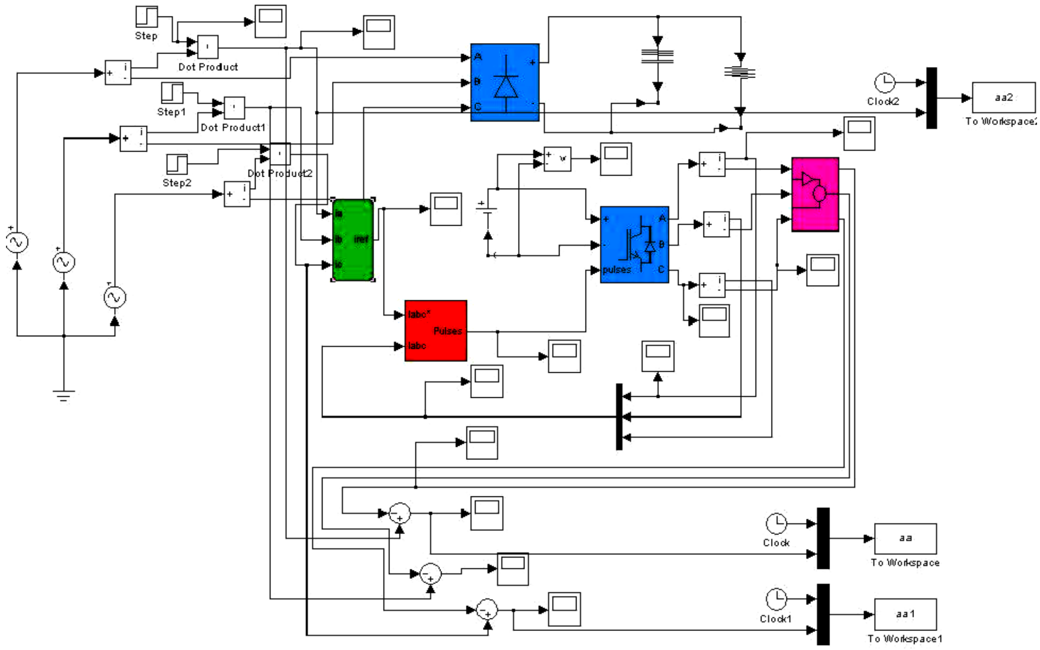

Figure 6. Simulation of active filter with diode rectifier load

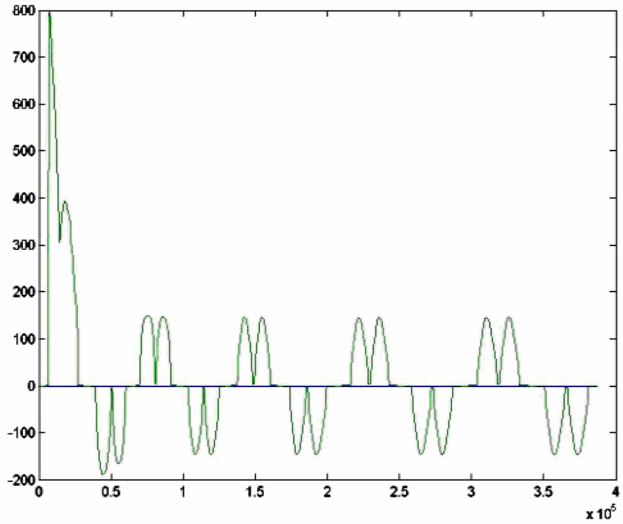

Figure 7. Load current of single phase

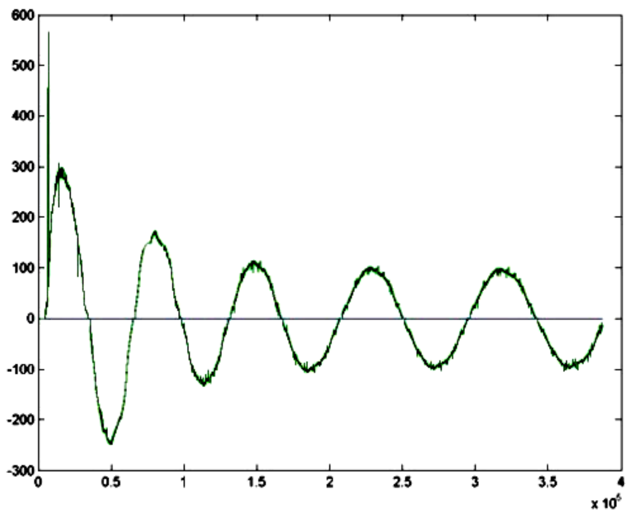

Figure 8. Main component of current

REFERENCES

1. S. Lou and Z. Hou, "An Adaptive detecting Method for harmonic and reactive currents", *IEEE Trans*. Lnd.Electron, vol.42, pp.88-89, 1995.
2. A. A.Girgis and F. M. Ham, "A quantitative study of pitfalls in the FFT", *IEEE Trans.Aerosp Elecytron syst*. vol, AES-16, pp.434-439, 1980.
3. T .Tanaka and H.akagi,"A new method of harmonic power detection based on the instantaneous active power delivery. vol.10, pp. 1742-1937, 1995.
4. Mohsen Mojiri,Masoud Karimi, "Estimation of Power System Frequency Using An Adaptive Notch Filter", *IEEE Trans* ,vol 56, no. 6 , 2007.
5. Hoshang karimi, Masoud karimi ghatermani, N. Reza iravani,"An adaptive filter for synchronous Extraction of Harmonics and Distortion", *IEEE Transaction on power delivery*. vol.18, 2003.