

Performance of adaptive filter for absorption of harmonics and noise in power systems

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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. Specifications of adaptive filters can be changed suiting to a specific goal and such change or adaptation should be automatic and without the need for any intervention. Among such features of this filter, one might refer to its simplicity and powerful structure.

Keywords: Inverter, 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 but majority of them are complex and are difficult to adjust and implement. Discrete Fourier Transfer function (DFT) which has been accepted because of its simplicity and 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, besides it demands complex mathematical operations (Lou & Hou,1995).

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 it 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 (Girgis & Ham, 1980; Tanaka & Akagi, 1995; Mojiri *et al.*, 2007).

The structure of adaptive filter

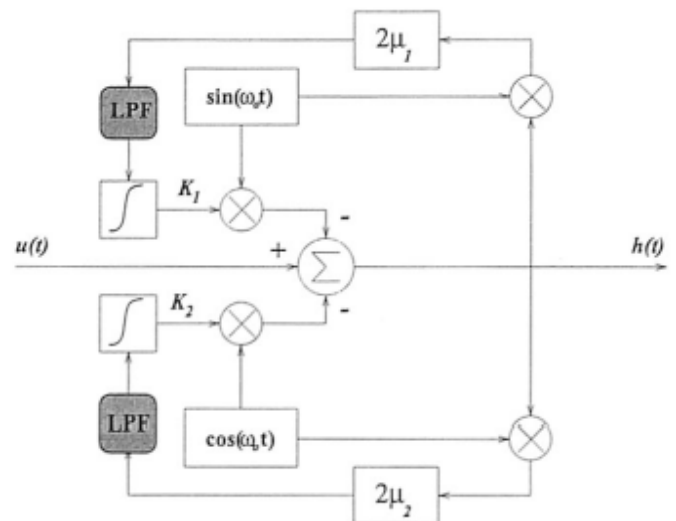
The structure of adaptive filter is shown in Fig.1. Assuming voltage signal hasn't been disturbed, the regarding filter can be set as the delete current signal reactive.

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 Fig.1 for the measurement of vibration without any decrease delete in transient response. In fact, correctly drawing LIP can

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



improve stable state response well. Fig. 2 shows main current component in two circuits one with LPF and another without LPF.

Fig.2.Main current component with LPF and without LPF Prove of filter model based on squares minimum method

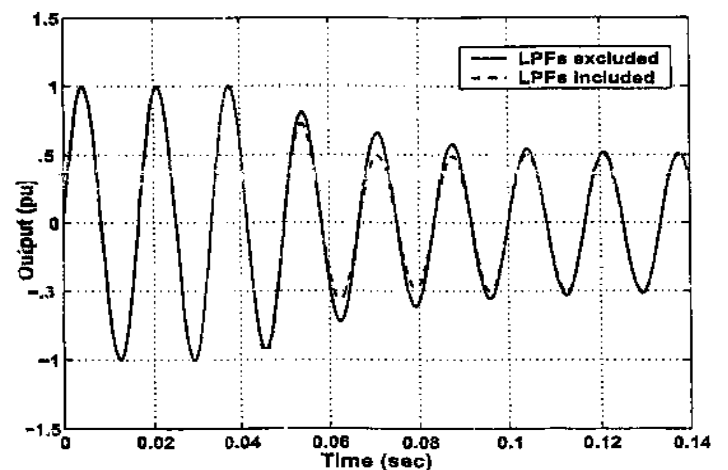
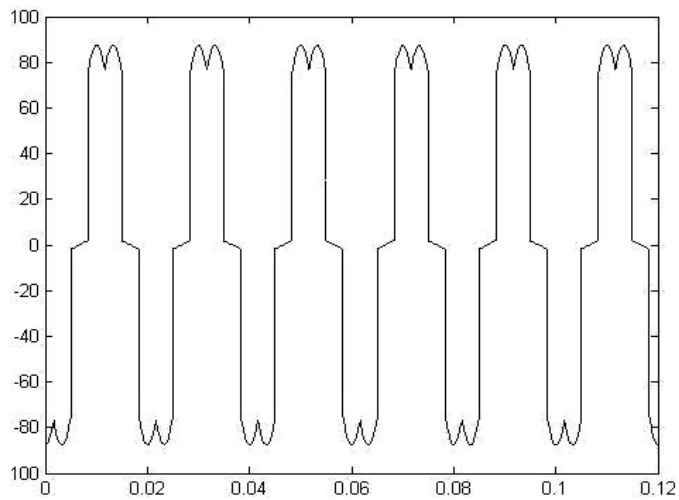
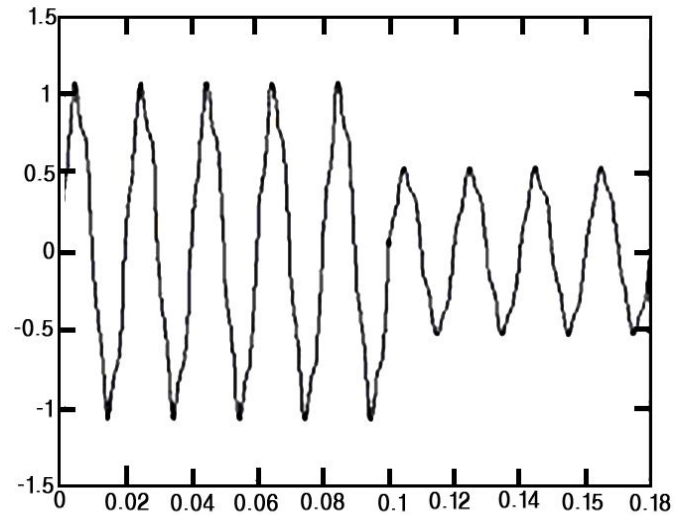
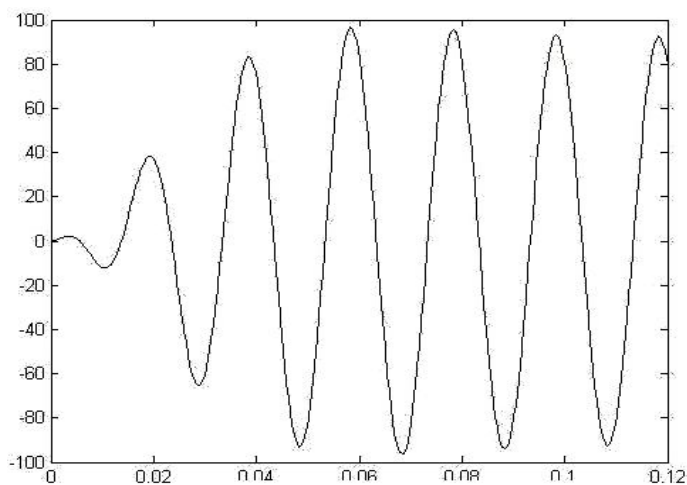
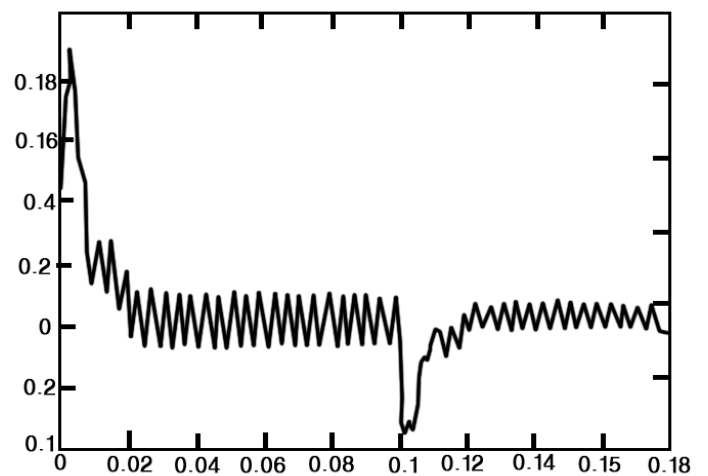
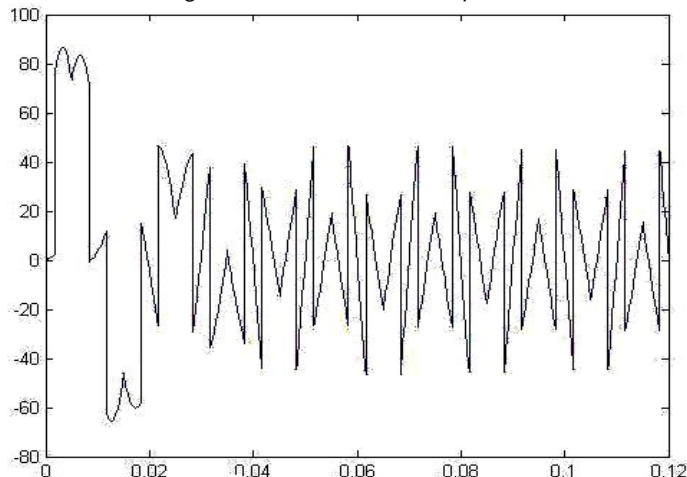
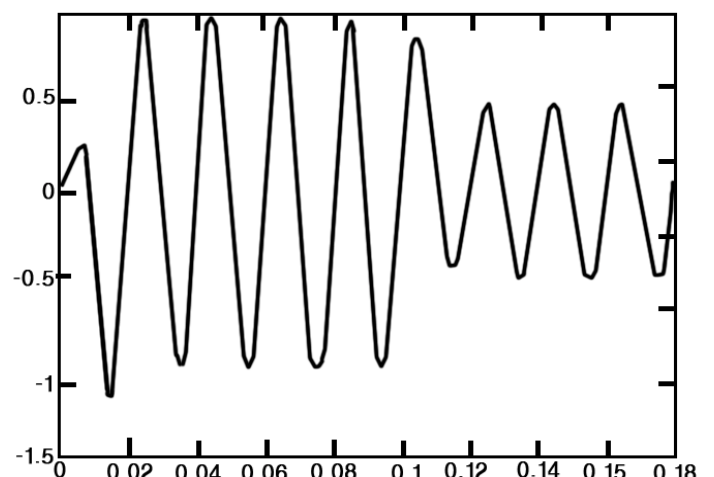


Fig.3.Phase current of the three phases Rectifier*Fig.4.Load current range with a 50 % reduction in amplitude**Fig.3a.Phase current after removing waste components**Fig.4a.Waste component of the load current**Fig.3b. Identified waste components**Fig.4b.The fundamental components of the load current*

To assume $u(t)$ is a continuous function in Fig.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 signal 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 have its minimum value 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 its input and output, are $u(t)$ and $h(t)$. By using of equation $h = u - y$ and replacement of (4) in (6):

$$\dot{K}(t) = \begin{bmatrix} -2\mu_1 \sin^2(\omega_o t) & -2\mu_1 \sin(\omega_o t) \cos(\omega_o t) \\ -2\mu_2 \sin(\omega_o t) \cos(\omega_o t) & -2\mu_2 \cos^2(\omega_o t) \end{bmatrix} K(t) + \begin{bmatrix} 2\mu_1 \sin(\omega_o t) \\ 2\mu_2 \sin(\omega_o t) \end{bmatrix} u(t) \quad (7)$$

Which $K(t)$ is determined like $[K_1(t), K_2(t)]^T$. The filter structure of Fig. 1 is simple and is simply dismantable by mathematics and trigonometry relation, both in the form of software (DSP) and hardware (Karimi, *et al*, 2003).

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 its proficiency increase the transient speed. In the structure of given filter, if we increase the parameter μ , it is possible that we gain faster response. 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 be more than 2%. Now, we consider the resistive rectified load current and investigate the performance of filter.

Now, if a change is gained in the amplitude of input

current, this method accommodates itself to these changes rapidly. Fig.3. shows phase current of the three phases rectifier. Fig.3a. and Fig.3b. show Identified waste components and phase current after removing waste components respectively. Also Fig..4. shows load current range with a 50 % reduction in amplitude. Fig.4a. and Fig.4b. show waste component of the load current and

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

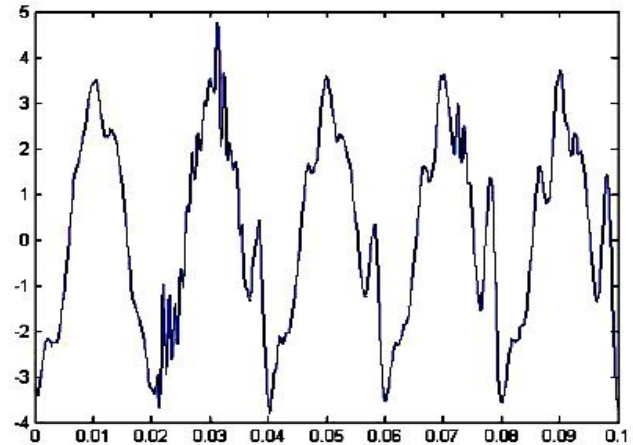


Fig.5a. Waste component of the current

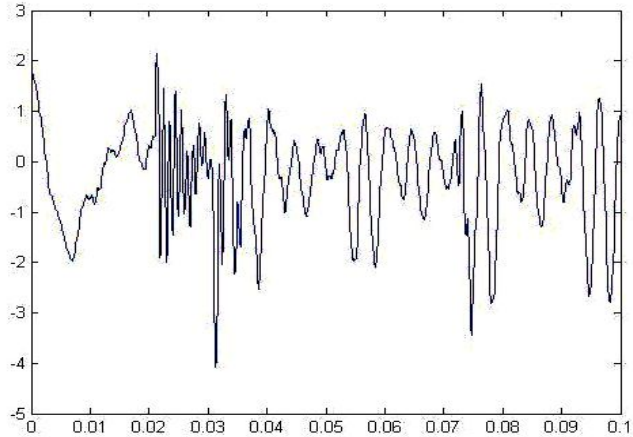
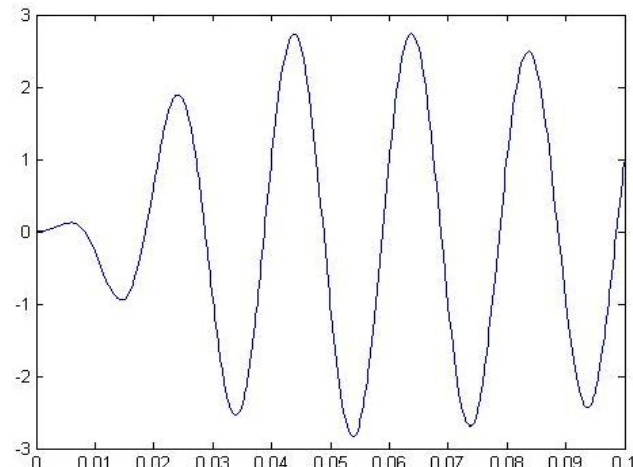


Fig.5b. Current phases, after removing harmonics



the fundamental components of the load current respectively.

Fig.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.

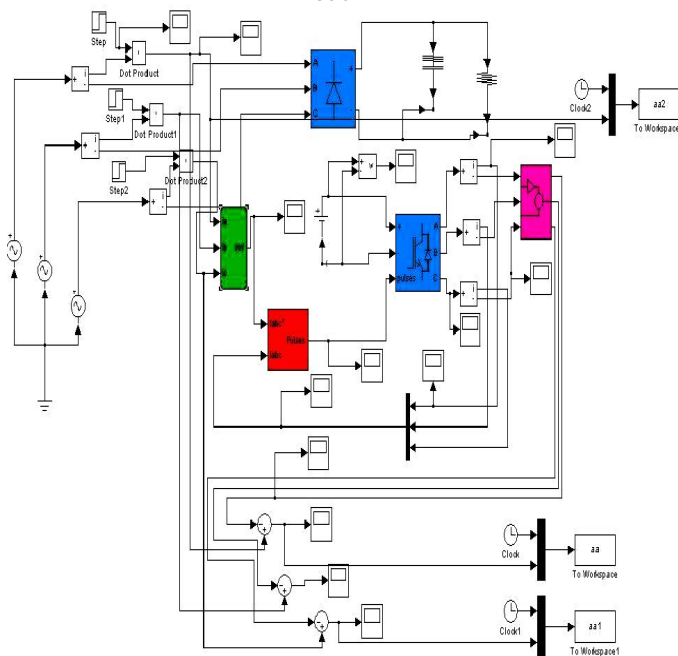
In this section, the three-phase active powers filter (APF) is simulated that is used in the identification of waste components with the adaptive method. The given load is considered as a three-phase diode rectifier and assumed rectifier load must be an ohmic resistance which would result higher THD. 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 $L=0.1\text{ mh}$ and $C=10^{-5}\text{ F}$ for inductor and capacitor, respectively. Meanwhile, the hysteresis bandwidth is considered 5 (h=5).

Fig.6. Simulation of active filter with diode rectifier load



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:

$$V_{ab} = \sum_{n=1,3,5}^{\infty} \left[\frac{4V_s}{n\pi} \cos\left(\frac{n\pi}{6}\right) \sin n(\omega t + \pi/6) \right]. \tag{8}$$

Consequently, line voltage to rms line at inverter output:

$$V_{ab,rms} = \frac{4V_s}{\sqrt{2\pi}} \cos \frac{n\pi}{6} \tag{9}$$

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

$$V_{1,ab,rms} = \frac{4V_s}{\sqrt{2\pi}} \cos \frac{\pi}{6} \tag{10}$$

Fig. 7. Load current of single phase

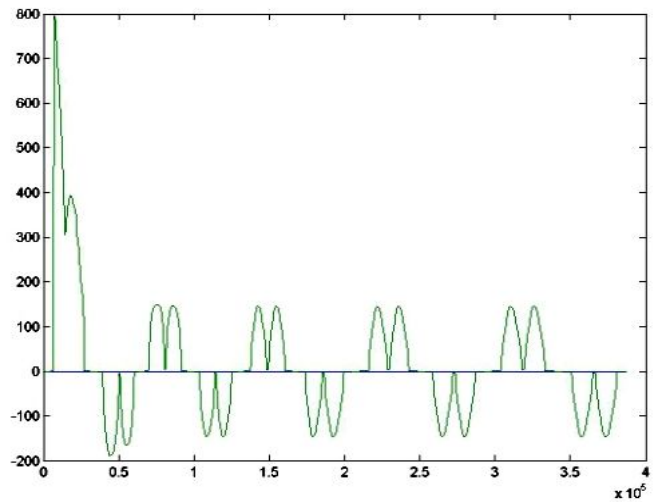
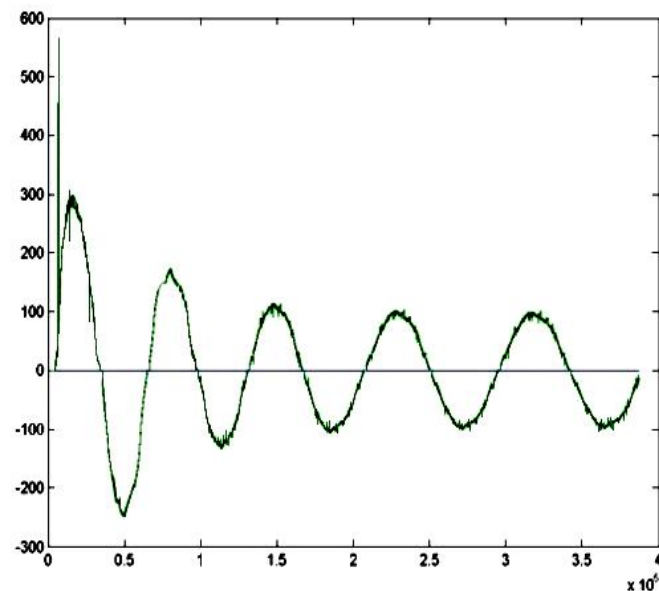


Fig. 8. Main component of current



Maximum phase voltage at output inverter:

$$V_{ph-n} = \frac{\sqrt{2}}{\sqrt{3}} \frac{4V_s \cos 30}{\sqrt{2\pi}} \Rightarrow V_S = V_{dc} = 1.56V_{ph} \quad (11)$$

In the Fig. 6, is used the adaptive method in I_{ref} part for all three phases. Fig. 7. and Fig. 8 show load current of single phase and main component of current.

Conclusion

An adaptive filter for detection and extraction of harmonics is provided. It provides desirable performance in different conditions and updates itself with changes of input. Its robust structure lends itself to easy implementations in hardware and software environments.

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