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PREDICTIVE STRUCTURE FOR CURRENT REFERENCE GENERATION OF ACTIVE POWER FILTER

Tomislav Kilić, Stanko Milun

Faculty of Electrical Engineering, Machine Engineering and Naval Architecture
University of Split, Split, Croatia

Abstract – The shunt active power filters are used to attenuate the harmonic currents in power systems by injecting equal but opposite compensating currents. Successful control of the active filters requires an accurate current reference. In this paper the current reference determination based on predictive filtering structure is presented. Current reference is obtained by taking the difference of load current and its fundamental harmonic. For fundamental harmonic determination with no time delay a combination of digital predictive filter and low pass filter is used.

Keywords: active filter, harmonics, predictive filter

1. INTRODUCTION

The shunt active power filter is an inverter driven to generate compensating currents that attenuate the harmonic components generated by the nonlinear loads. Therefore, only the fundamental current would be delivered by the mains supply. The performance of active filters is based on the inverter parameters, control method, and the method for current reference determination. Current reference can be obtained by using the bandpass filters [1] or the instantaneous power theory [2]. Bandpass filters have the disadvantage that they cause delay of filtered signal. The instantaneous power theory, on the other hand, is based on complex voltage and current transforms and their inverse transforms. Further, if the voltage source is distorted by harmonics the instantaneous power theory does not provide an accurate basis for active filters.

The objective of this paper is to introduce an efficient method to obtain the current reference for the active power filter. The method for current reference determination is based on extracting the fundamental harmonic from load current waveform without its phase shifting. In such way the current reference can be obtained simply by subtracting the fundamental harmonic from the measured load current. The current reference is calculated for each phase current separately. The predictive filtering structure consists of a cascade of digital Chebyshev low pass filter and digital predictive filter. Chebyshev low pass filter extracts the fundamental harmonic from distorted load current waveform and predictive filter cancel out the time delay introduced by low pass filter.

2. ACTIVE POWER FILTER

The shunt active power filters are inverters driven to attenuate harmonic currents generated by the nonlinear loads. As can be seen from Fig. 1, their operation principle is based on the injection of compensating current in the network so that only the fundamental current would be delivered by the mains supply. The harmonics of all orders can be compensated by one piece of equipment and better harmonic compensation characteristics can be obtained due to the use of the high controllability and quick response of switch-mode power electronic converters. However, successful control of the shunt active power filter requires an accurate and delayless current reference. The distorted current i_l that nonlinear load draw from power supply can be written as sum of fundamental current harmonic i_{l1} and its current harmonics

$$i_l = i_{l1} + \sum_{\nu=2}^{N} i_{l\nu} \tag{1}$$

where ν is order of harmonic and $i_{l\nu}$ current harmonic component.

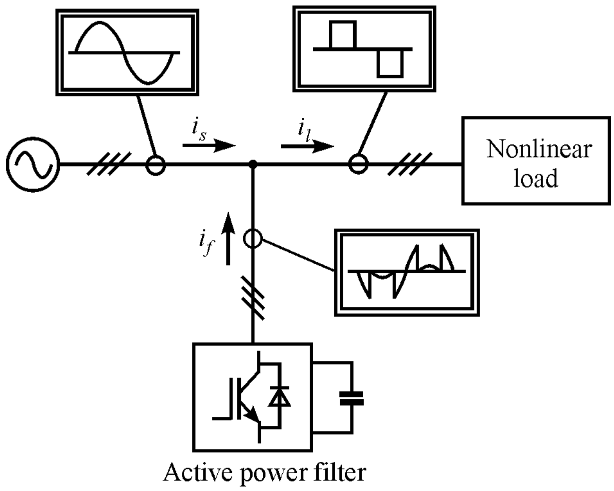

Fig. 1. Block diagram of an active power filter

According to Fig. 1 load current is

$$i_l = i_s + i_f \tag{2}$$

where i_f is current that active power filter injects in the network. Since the sinusoidal current of nominal frequency from power supply is desired the current that active power filter should produce is

$$i_f = i_l - i_{l1} = \sum_{\nu=2}^{N} i_{l\nu}\,. \tag{3}$$

The active power filter uses a current controlled inverter to inject corrective current waveform according to reference current to attenuate undesirable harmonics. The inverter should produce a current as close as possible to the reference one. Successful control of the active filters requires an accurate current reference. As can be seen from (3) desired or reference current of active filter is equal to sum of load current harmonics and it can be obtained as difference of load current and its fundamental harmonic.

3. DETERMINATION OF REFERENCE CURRENT

Fig. 2 shows a block diagram of the current reference determination. The signal proportional to load current i_l is converted to digital signal $i_l(n)$ and filtered with low-pass filter. On the output of low-pass filter the fundamental harmonic component $i_{l1}(n-k)$ with time delay is obtained. This harmfully phase shift is compensated by predictive filter and fundamental harmonic component $i_{l1}(n)$ without phase shift is obtained Therefore, the reference current $i_{ref}(n)$ can be obtained simply by subtracting the fundamental harmonic $i_{l1}(n)$ from the measured load current $i_l(n)$.

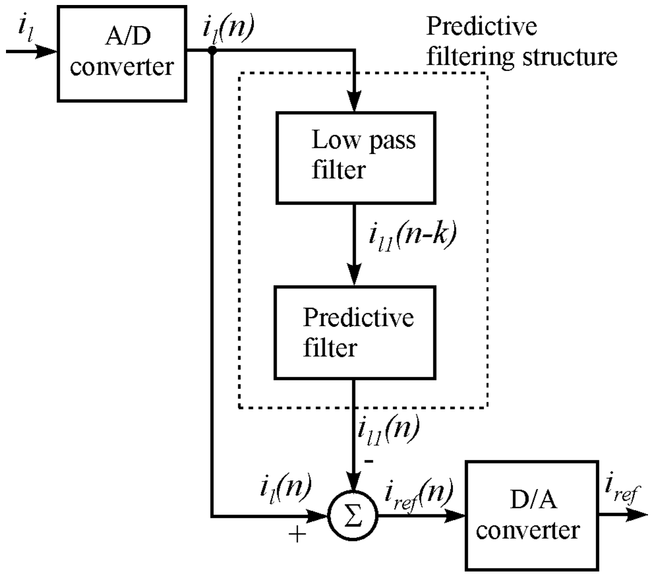

Fig. 2. Block diagram for current reference determination

3.1. Digital low-pass filter

We used a sixth-order Chebyshev type II infinite impulse response (IIR) filter with a passband cutoff frequency f_c=140 Hz and stopband ripple 50 dB. The active power filter should eliminate the third and above harmonics and because of that the passband cutoff frequency f_c=140 is chosen. A sampling rate of the A/D converter is 10 kHz. This filter is designed to efficiently attenuate the undesired harmonic components of the input signal. The transfer function of used Chebyshev filter is

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + b_3 z^{-3} + b_4 z^{-4} + b_5 z^{-5} + b_6 z^{-6}}{1 - a_1 z^{-1} - a_2 z^{-2} - a_3 z^{-3} - a_4 z^{-4} - a_5 z^{-5} - a_6 z^{-6}} \tag{4}$$

and coefficients are:

b_o=0,00292777 b_1=-0,01716804 b_2=0,04233044
b_3=-0,05618029 b_4=0,04233044 b_5=-0,01716804
b_6=0,00292777

a_1=-5,77704520 a_2=13,9099598 a_3=-17,8676483
a_4=12,91371986 a_5=-4,97910578 a_6=0,80011960

The realised Chebyshev filter introduces on nominal frequency 50 Hz a phase shift of 113° or a delay of 6,27 ms.

3.2. Predictive digital filter for sinusoidal signals

The predictive digital FIR (Finite Impulse Response) filter is used to compensate delay introduced by low-pass Chebyshev filter. Generally the predictive digital filter predict an input signal few discrete steps ahead. Since the delay of Chebyshev filter on 50 Hz is 6,27 ms and sampling rate is 10 kHz the prediction step is p=62,7.

FIR filters can be designed for prediction p-step ahead of input signal using Lagrange multipliers method [3]. Output value $\hat{u}(n+p)$ of p-step ahead predictor is obtained by multiplying past input samples $u(n-k)$, k = 1, ..., N with predictor coefficients h_k, k = 1, ..., N

$$\hat{u}(n+p) = \sum_{k=1}^{N} h_k u(n-k)\,. \tag{5}$$

Depending on the nature of the incoming signal to be predicted, the FIR predictor coefficients h_k are based on a polynomial model, a sinusoidal model or some other time-domain signal model which represents the important signal trends. Since the input signal of this filter is sinusoidal waveform we have been used p-step ahead FIR predictor for sinusoidal signals.

There are two preconditions for the task of determining the coefficients h_k. The first, exact prediction of sinusoidal signal of the nominal frequency ω_o is required, i.e.,

$$\sin\left[\omega_o(n+p)+\phi\right] = \sum_{k=1}^{N} h_k \sin\left[\omega_o(n-k)+\phi\right]. \tag{6}$$

The second, the noise attenuation is optimized with respect to the white noise. Because the noise components in each sample are assumed to be independent, the noise power gain $F(h_1, ..., h_N)$ is given by [4]

$$F(h_1,\ldots,h_N) = \sum_{k=1}^{N} \left(h_k\right)^2 \tag{7}$$

In order to achieve correct sinusoid prediction regardless of signal phase it is required that:

$$cos\left[\omega_o(n+p)\right] = \sum_{k=1}^{N} h_k \, cos\left[\omega_o(n-k)\right]$$

$$sin\left[\omega_o(n+p)\right] = \sum_{k=1}^{N} h_k \, sin\left[\omega_o(n-k)\right]. \qquad (8)$$

On the base of (8) two parameters g_o and g_1 for optimisation can be defined

$$g_o = \sum_{k=1}^{N} h_k \, cos\left[\omega_o(n-k)\right] - cos\left[\omega_o(n+p)\right]$$

$$g_1 = \sum_{k=1}^{N} h_k \, sin\left[\omega_o(n-k)\right] - sin\left[\omega_o(n+p)\right]. \qquad (9)$$

The optimization of noise power gain $F(h_1, ..., h_N)$ and parameters g_o and g_1 can be carried out analytically using the method of Lagrange multipliers [3]. The Lagrange function is

$$L\left(h_1, \ldots, h_N, \lambda_o, \lambda_1\right) = \sum_{k=1}^{N} \left(h_k\right)^2 + g_o\lambda_o + g_1\lambda_1 . \qquad (10)$$

where λ_o and λ_1 are Lagrange multipliers. Setting the partial derivatives of Lagrange function $L(h_1, ..., h_N, \lambda_o, \lambda_1)$ with respect to all arguments equal to zero:

$$\frac{\partial L}{\partial h_k} = 2h_k + \lambda_o\left(cos\,\omega_o\,cos\,k\omega_o + sin\,\omega_o\,sin\,k\omega_o\right)$$

$$+ \lambda_1\left(sin\,\omega_o\,cos\,k\omega_o - cos\,\omega_o\,sin\,k\omega_o\right) = 0$$

$$k = 1,2,\ldots,N \qquad (11)$$

$$\frac{\partial L}{\partial \lambda_0} = g_0 = 0 \qquad (12)$$

$$\frac{\partial L}{\partial \lambda_1} = g_1 = 0. \qquad (13)$$

the coefficients h_k can be determined. From (11) coefficients h_k, are

$$h_k = -\frac{1}{2}\lambda_o\left(cos\,\omega_o\,cos\,k\omega_o + sin\,\omega_o\,sin\,k\omega_o\right)$$

$$-\frac{1}{2}\lambda_1\left(sin\,\omega_o\,cos\,k\omega_o - cos\,\omega_o\,sin\,k\omega_o\right)$$

$$k = 1,2,\ldots,N. \qquad (14)$$

Substituting (14) and (9) in (12) and (13) and solving the pair of equations the multipliers λ_o and λ_1 are obtained. Finally substituting λ_o and λ_1 in (14) the coefficients h_k, k = 1, ..., N for the predictive FIR filter are obtained.

Generally, the filter characteristics depend on prediction step p and the filter length N. A long filter has narrow-band characteristics and high noise attenuation. For line frequency signals we must take into account the possible frequency variation. If the predictor is designed to have too narrow a prediction band (N is too large), then the slightly frequency variation will cause prediction error. Thus the design of a predictor is always a compromise between noise attenuation and bandwidth. As a good compromise between noise attenuation and bandwidth we have chosen predictive FIR filter with N = 22. The prediction step p=62,7 is determined by delay of Chebyshev filter. The transfer function of this predictive FIR filter is:

$$\begin{aligned} H(z) = \; & 0{,}25956014 + 0{,}23264099z^{-1} + 0{,}20549226\,z^{-2} \\ & + 0{,}17814072z^{-3} + 0{,}15061339z^{-4} + 0{,}12293741z^{-5} \\ & + 0{,}09514011z^{-6} + 0{,}06724892z^{-7} + 0{,}03929137z^{-8} \\ & + 0{,}01129504z^{-9} - 0{,}01671243z^{-10} - 0{,}04470342\,z^{-11} \\ & - 0{,}07265029z^{-12} - 0{,}10052546z^{-13} - 0{,}12830143\,z^{-14} \\ & - 0{,}15595077z^{-15} - 0{,}18344622z^{-16} - 0{,}21076062z^{-17} \\ & - 0{,}23786703z^{-18} - 0{,}26473869z^{-19} - 0{,}29134909z^{-20} \\ & - 0{,}31767196\,z^{-21} \end{aligned} \qquad (15)$$

The amplitude response of the predictive filtering structure (cascade of digital Chebyshev low pass filter and digital predictive filter) is shown in Fig. 3, and phase response is shown in Fig. 4. On the nominal frequency 50 Hz, the amplitude response is equal unity and there is not a phase shift.

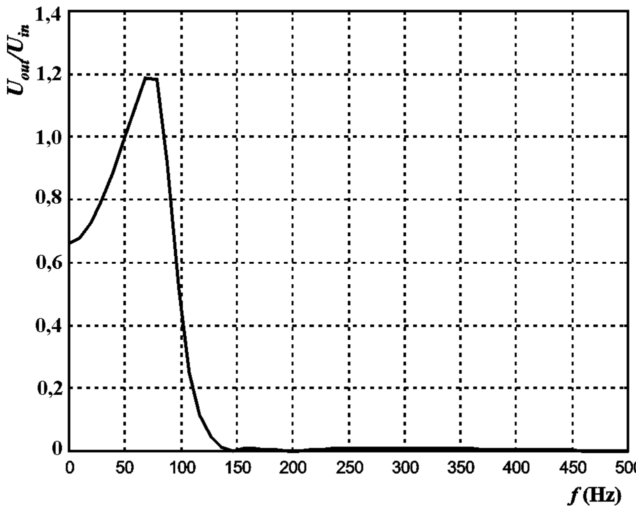

Fig. 3. Amplitude response of the predictive structure

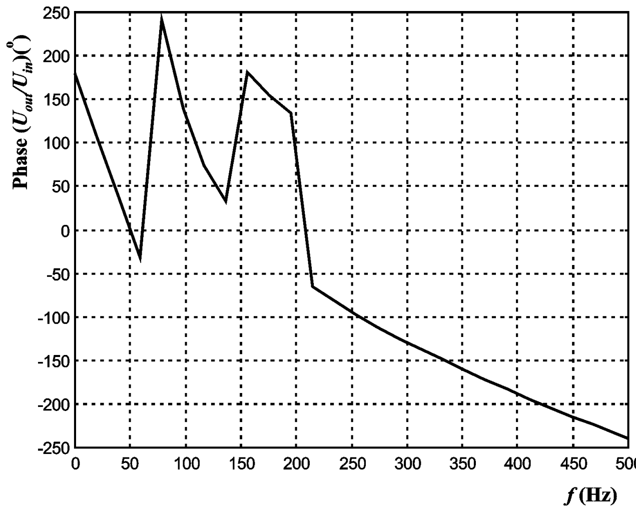

Fig. 4. Phase response of the predictive structure

For evaluation of the predictive filtering structure a complete model of the proposed filtering system has been built using Matlab simulation software. As a input signal we used a square wave signal and on the output we obtained sinusoidal signal of nominal frequency, illustrated in Fig. 5.

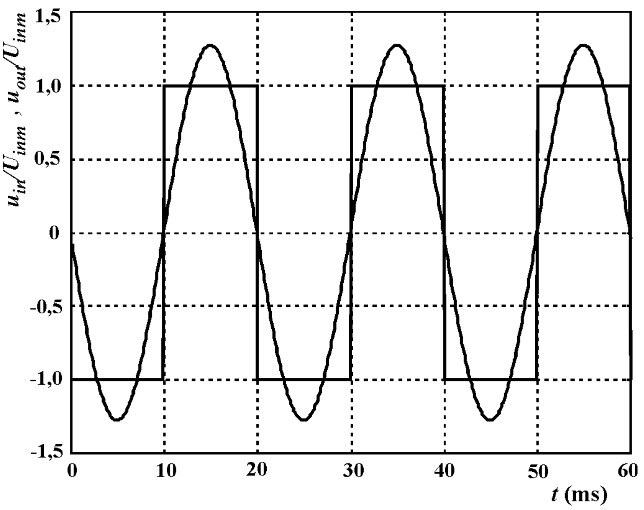

Fig. 5. Input square wave and output sinusoidal signal

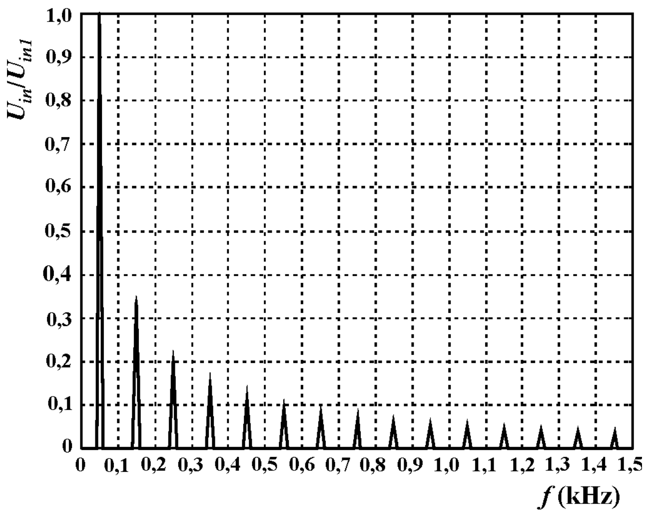

Fig. 6. Frequency spectrum of input square waveform

Fig. 6 shows the frequency spectrum of input signal and Fig. 7 shows frequency spectrum of difference between input and output signal. The frequency spectrum of difference between input and output signal is the same as frequency spectrum of input signal without the fundamental harmonic. According to (3) the reference current is equal to sum of high order harmonic components. As we can see from Fig. 7, the difference between input signal of predictive filtering structure and its output signal is equal to sum of high order harmonic components of input signal. On the base of simulation results it is obvious that proposed predictive filtering structure can be used for current reference determination of active power filter.

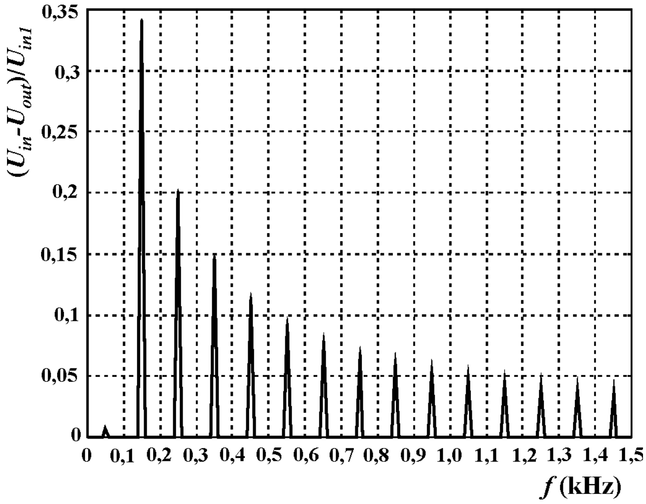

Fig. 7. Spectrum of the difference between input signal of predictive filtering structure and its output signal

4. IMPLEMENTATION OF PREDICTIVE FILTERING STRUCTURE

A laboratory prototype of a 16,5 kVA three-phase active power filter for experimental purpose has been set up and tested [5]. Algorithm for current reference determination was adapted and implemented on DSP controller ADMC300 from Analog Devices. The ADMC300 integrates a 25 MIPS, fixed-point DSP core with peripherals for power electronics control. It integrates five completely independent analog-to-digital converters (ADC) based on sigma delta conversion technology. The sigma delta converters consist of two stages, a modulator and a sinc filter, that combine to produce a 16-bit conversion. For each channel, signal-to-noise ratios (SNR) of 76 dB may be achieved, corresponding to greater than 12 bits of resolution from each converter. The sampling rate is up to 32,5 kHz.

The load currents are measured by means of Hall effect current sensors and the voltage signals proportional to the load currents are obtained. These signals proportional to load currents are converted to digital signals using ADCs of ADMC300. The sampling rate is 10 kHz. After analog-to-digital conversion the described procedure for current reference determination is applied. The complete software for current reference determination is written in assembly language.

The validity of proposed predictive filtering structure for current reference determination of active power filter was confirmed experimentally on laboratory prototype. Fig. 8 and Fig. 9 show the waveform and frequency spectra of the load current. Fig. 10 shows the spectrum of reference current that is almost the same as the spectrum of load current without fundamental harmonic. Fig. 11 shows the spectrum of filter current that active power filter on the base of reference current inject in network. In such way the active power filter attenuate current harmonics generated by non linear load and almost sinusoidal current will flow from power supply. Fig. 12 and Fig. 13 show waveform and frequency spectra of the supply current with active power filtering. Comparing the waveforms and frequency spectra of load and supply current, it can be seen that with active filtering all harmonic components in supply current are reduced.

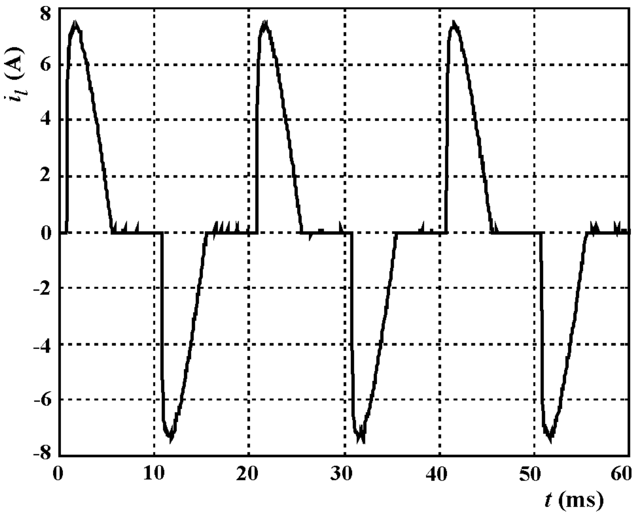

Fig. 8. Load current waveform

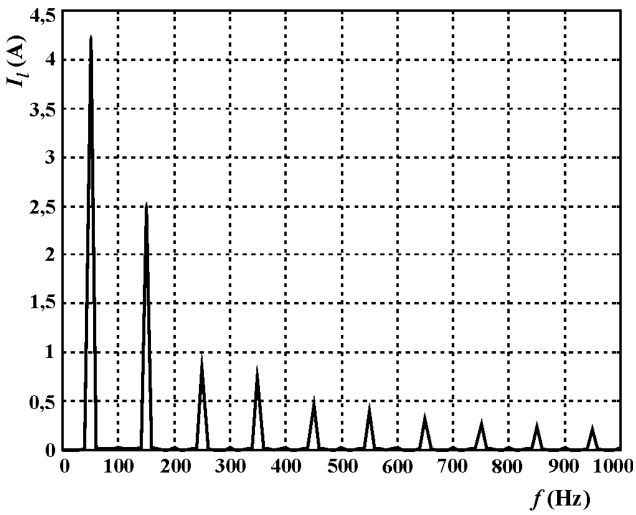

Fig. 9. Frequency spectrum of load current

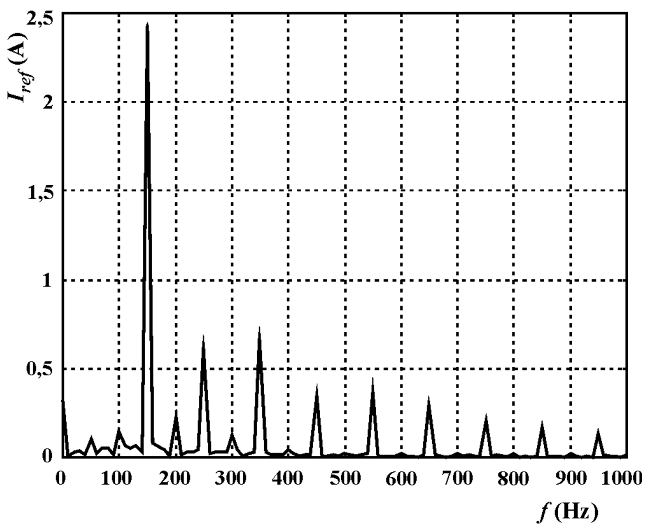

Fig. 10. Frequency spectrum of reference current

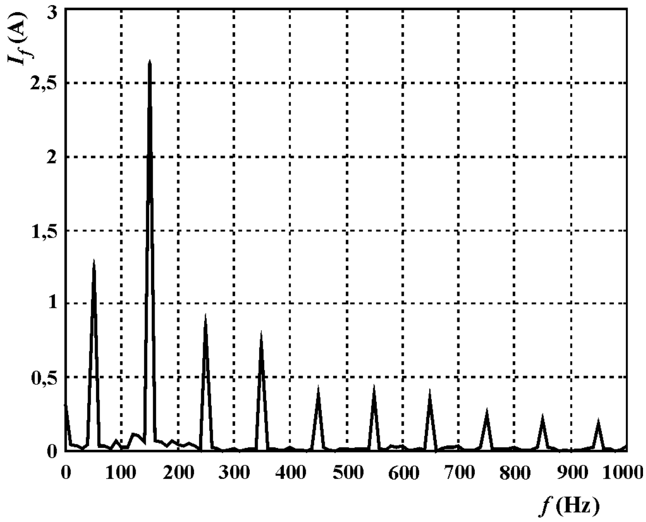

Fig. 11. Frequency spectrum of filter current

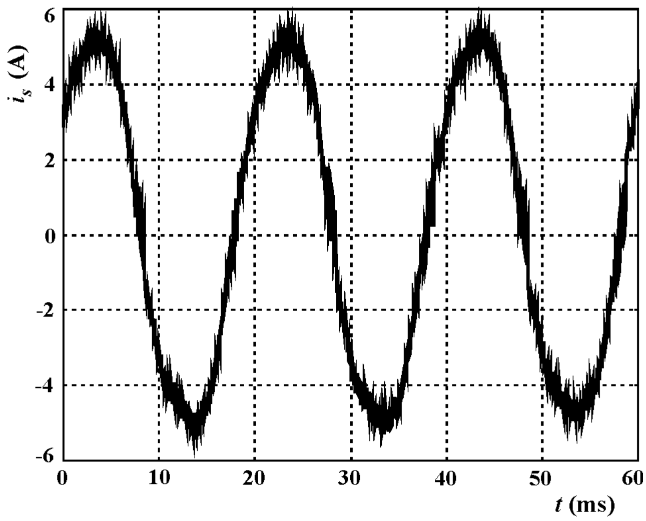

Fig. 12. Source current waveform with active power filtering

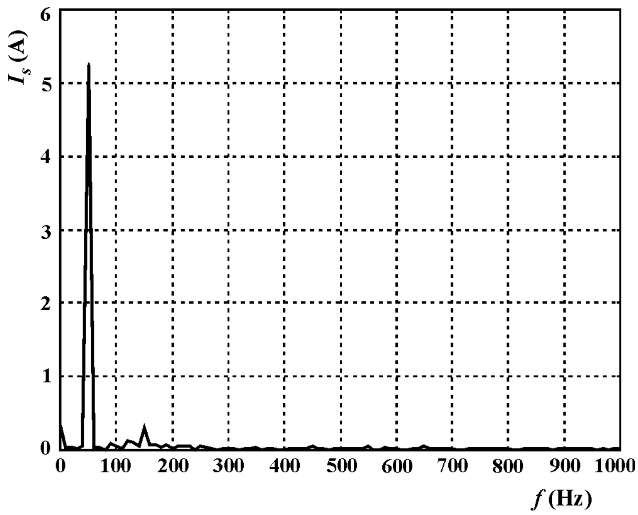

Fig. 13. Frequency spectrum of source current

5. CONCLUSIONS

A digital signal processing system for current reference generation of active power filter based on the predictive FIR filter was presented. The predictive filtering structure is capable of extracting the fundamental harmonic from highly disturbed signals without phase shift, even if there are considerable frequency and amplitude variations. Practically no prior knowledge about network and load characteristics is necessary for the generation of the compensating reference current. Thus, the proposed method can be used in various applications with only moderate modifications. The proposed method can easily be adapted and implemented in digital signal processor for different system specifications and noise characteristics.

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Authors: Dr. sc. Tomislav Kilić, Dr. sc. Stanko Milun, Faculty of Electrical Engineering, Machine Engineering and Naval Architecture, University of Split, R. Boškovića bb, 21000 Split, Croatia, tel. 385 21 305653, fax. 385 21 463877, E-mail: tkilic@fesb.hr, smilun@fesb.hr