ON THE HARMONICS REDUCTION USING WAVELET BASED SIGNAL PROCESSING

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Abstract: This paper presents a method for calculating the reference currents needed for the command of an active filter. By using the Discrete Wavelet Transform (DWT) the high-frequency components of the currents are eliminated. Also, a reference is delivered to the control block of the active filter and a comparison is made between the DWT and the classical Fourier transform.

Keywords: harmonic reduction, wavelet analysis, active filter, signal processing.

1. INTRODUCTION

Until recently power system harmonics have created few problems except in certain special cases such as traction substations, large rectifier/inverter installation for high voltage d.c. transmissions, and in the electrochemical industry. However, with the growth in the number and rating of disturbing loads the situation has changed. Quality of distributed power to the customers is now in question as a result of harmonics. Hence reduction of harmonics also means improvement of power quality.

Insular systems are also becoming increasingly popular. These hybrid wind-diesel-photovoltaic systems act as finite impedance sources, thus the harmonics' effects are more important [4]. Moreover, the freqency within these systems can have shifts around the main value, which raise several signal processing problems.

Started from this reality the paper presents some results on research concerning the reduction of the harmonics in power systems based on signal processing techniques. The first one is the classic Fourier method and it is used in order to have a reference signal for an active filter [3]. The second technique is an alternative but approximate one and it is based on wavelet theory.

In the following some general concepts are presented in order to define the context of the research. In section 2 the method of harmonic reduction using Discrete Wavelet Transform (DWT) is described. Some simulation results are presented in section 3 as well as a short comparison between the results obtained by DWT and Discrete Fourier Transform (DFT). The comparison is made using signals which presents small shifts in the fundamental's frequency.

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1.1 1.1 Some problems in Harmonic Analysis

In order to eliminate or minimize harmonic distortion it is necessary to identify harmonic components and determine their magnitudes and phase angles. Fourier transform is the traditional yet most precise method of determining harmonic components of any repetitive distorted waveform. For speed computation Fast Fourier Transform (FFT) can be used. However, the Fourier transform is affected by the presence of noise in the signal. Although high frequency noise can be eliminated through low pass filtering before applying the Fourier transform it is still trouble-some to eliminate possible sub-harmonics that exist in the distorted waveform. The main problem of the Fourier transform is the number of points in the observation window, which should be a multiple of the numbers of samples per period. When the fundamental's frequency varies around the 50Hz value, this corresponds to a modification in the number of samples per period. Thus, the number of points in the observation window is not a multiple of the number of sample per period. As a result, the accuracy of the de-noising is reduced. Usually, the signal is decomposed into its harmonic components, using the Fourier transform and then a reference signal is created using the sum of the superior harmonics. The reference signal, added to the original one, eliminates the unneeded harmonic components.

Although wavelet de-noising is used in other applications, like speech processing [1, 2], it was not used in power electronics due to superior results of FFT. However, in certain cases, such as isolated sites, the main frequency can have small variations around 50Hz, and the Fourier transform is sensitive to frequency shifts. In these cases wavelet de-noising can have better results. Moreover, wavelet de-noising can also eliminate interharmonics, which appear in isolated sites [4]. Interharmonics can be thought as inter-modulation of the fundamental and harmonic components of the line system with any other frequency components of e.g. the electrical machine. They can be observed by using loads as e.g. cycloconverters or current source inverters for speed controlling of electrical machines. These frequency parts of voltage or current spectrum are not an integer of the fundamental but discrete frequencies or a wide-band spectrum. Standards for defining measurement algorithm or maximum levels are still in discussion.

1.2 1.2 A brief discussion on wavelets

A wavelet, in the sense of DWT, is an orthogonal function which can be applied to a finite group of data [1]. Functionally, it is very much like the Discrete Fourier Transform, in that the transforming function is orthogonal, a signal passed twice through the transformation is unchanged, and the input signal is assumed to be a set of discrete-time samples. Whereas the basis function of the Fourier transform is a sinusoid, the wavelet basis is a set of functions which are defined by a recursive difference equation

$$f(x) = \sum_{k=0}^{M-1} c_k f(2x-k) \qquad (1)$$

where the range of the summation is determined by the specified number of nonzero coefficients c_k. The number M is arbitrary, and it will be reffered as the order of the wavelet. The values of the coefficients are not arbitrary, but are determined by constraints of orthogonality and normalization. In many cases no explicit expression for f is available, but there are fast algorithms that use the refinement equation to evaluate the scaling function at dyadic points $x = 2^{-j}k,\ k, j \in Z$. Generally, the area under the wavelet "curve" over all space should be unity, which requires that

$$\sum_k c_k = 2 \qquad (2)$$

Function f is orthogonal to its translations; i.e. $\int f(x) f(x-k) dx = 0$. What is also desired is an equation which is orthogonal to its dilatations, or scales; i.e. $\int y\,(x)\,y\,(2x-k)dx = 0$. Such a function y does exist, and is given by

$$y\,(x) = \sum (-1)^k c_{1-k} f(2x-k) \qquad (3)$$

which is dependent upon the solution of $j\,(x)$. The function y is a wavelet, and (3) defines the "discrete wavelet transform". Normalization requires that

$$\sum c_k c_{k-2m} = 2 d_{0m} \qquad (4)$$

which means that the above sum is zero for all m not equal to zero. Another important equation which can be derived from the above conditions and equations is

$$\sum_k (-1)^k c_{1-k} c_{k-2m} = 0 \qquad (5)$$

1.3 1.3 The pyramid algorithm

The pyramid algorithm operates on a finite set of N input data, where N is a power of two; this value will be referred to as *input block size* (Fig. 1). These data are passed through two convolution functions, each of which creates an output stream that is half the length of the original input. These convolution functions are filters; one The pyramid algorithm half of the output is produced by the "low-pass" filter function, related to equation (1):

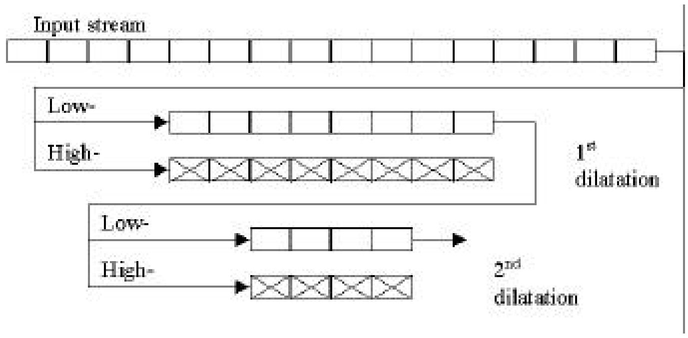


Figure 1: Description of the pyramid algorithm

$$a_i = \frac{1}{2}\sum_{j=1}^{N} c_{2i-j+1} f_j , \quad i = 1,...N/2 \qquad (6)$$

and the other half is produced by the "high-pass" filter function, related to equation (3):

$$b_i = \frac{1}{N}\sum_{j=1}^{N} (-1)^{j+1} c_{j+2-2i} f_j , \quad i = 1,...,N/2 \qquad (7)$$

where N is the input block size, c are the coefficients, f is the input function, and a and b are the output functions [2].

In many situations, the low-pass output contains most of the "information content" of the original input signal. In general, higher-order wavelets (i.e. those with more non-zero coefficients) tend to put more information into the low-pass output, and less into the high-pass output. If the average amplitude of the high-pass output is low enough, then the corresponding part of the signal can be discarded without greatly affecting the quality of the reconstructed signal. An important step in wavelet-based data compression and de-noising is finding wavelets functions which cause the high-pass functions to be nearly zero.

2. HARMONIC REDUCTION IN POWER ELECTRONICS

2.1Active filters

Active filters (AF) are commonly used today because they are flexible (one AF can compensate multiple harmonics) and due to the drop in power semiconductors' prices, economically viable. The typical implementation scheme of an active filter is presented in Fig.2, where Z_L is the line's impedance, i the line currents, i_F the filter currents and i_L load currents. The aim of the active filter is to remove all harmonics caused by the non-linear load from the common voltage point v_s. This is accomplished by controlling the active filter in order to generate harmonic currents which match in magnitude and phase with those existing in the load current i_L. This leads to a sinusoidal current in the ac source and consequently no harmonic distortion due to Z_L may occur.

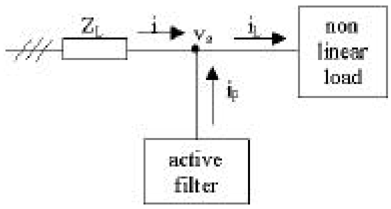


Fig.1. Fig.2 The active filter principle

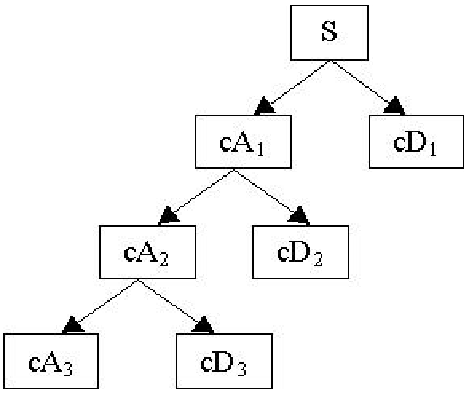
An AF is an inverter, and its control algorithm is its most important part. It involves the determination of a reference signal, then, based on this reference, the command sequence for the semiconductor devices is generated. While some elements of the control algorithm are well studied, getting the reference signal is always a difficult task. Using the wavelets analysis we present here an alternative method for obtaining the reference signal.

2.2 Noise suppression

The method consists in applying the pyramid algorithm to a given waveform and to neglect the high-pass output while reconstructing the signal. Thus, the high-frequency "information" contained within the signal is discarded.

For instance, after applying the pyramid algorithm to a signal, this can be separated in an approximation of the signal and details of the signal. Next the pyramid algorithm is applied to the approximation, and so on [5]. Following the decomposition tree presented in Fig.3, the initial signal can be recomposed using a sum of signal's approximations and details:

$$S = cA_3+cD_3+cD_2+cD_1 \qquad (8)$$



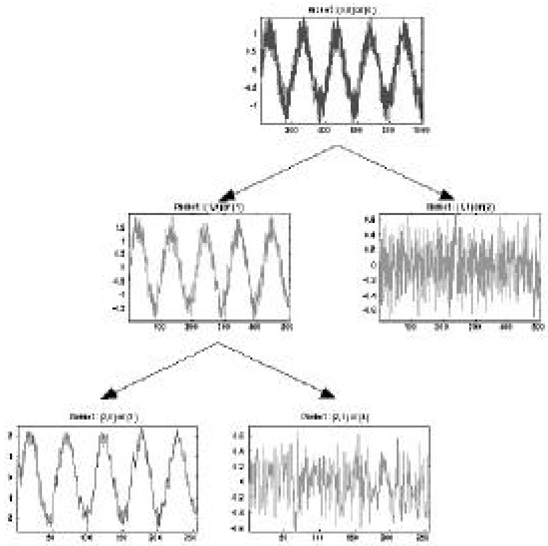


Fig.2. Fig. 3 The wavelet decomposition tree

Moreover, if a threshold is applied to the detail coefficients, the signal will be reconstructed without most of the frequency information contained in these coefficients, which is related to high-frequency components. Thus the noise within the signal is eliminated. For sub-harmonics the same method can be applied to the approximation coefficients.

In a three-phase system, where current harmonics are to be eliminated, the pyramid algorithm needs to be applied for each phase separately. Considering the amount of numerical operations, it is more interesting to reduce the three-phase system to a two-phase system using the Clarke transform. By using the Clarke transform, we can reduce the amount of calculation needed, because the pyramide algorithm is applied only twice, and instead its third application only the direct, respective the inverse Clarke transform need to be calculated.

3 SIMULATION RESULTS

Some simulations results are presented in fig.6. The signal used consists in a sum of different harmonics and white noise. We have considered a test signal, in which the fundamental's frequency is 50Hz and it's amplitude is 10. The harmonic components of the signal are 5, 7, 11 and 13 and theirs amplitudes are randomly selected to be maximum 15% of the fundamental. Considering the Total Harmonic Distorsion Factor (THD) to be:

$$THD = \sqrt{\sum_{i=2} I_i^2 \Big/ \sum_{j=1} I_j^2} \text{ e} \qquad (9)$$

where I_n is the amplitude of the n^{th} harmonic, then for the given signal the THD is 0.09 after de-noising, while normally the THD should be less than 10%. The magnitude spectrum of the test signal used before and after de-noising is presented in figure 5.

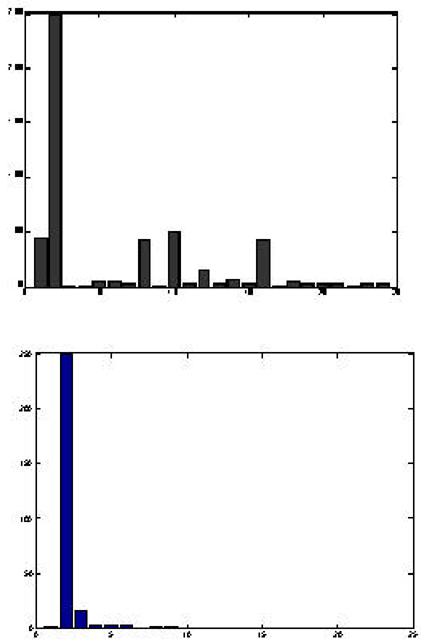


Fig 4 The magnitude spectrum of the current before and after de-noising

Although some harmonic components are still present, after applying the pyramidal algorithm their amplitude is negligible. The signal obtained after de-noising is subsequently used to generate the reference for the AF.

3.1 COMPARISON WITH THE CLASSICAL DE-NOISING METHOD

We have considered the case of the fundamental frequency of 49Hz, with all the other elements of it being the same as in the previous simulations. The signal was analyzed using DFT, but the number of points in the observation window was a multiple of the number of samples per a 0.02s period, corresponding to a 50Hz fundamental. The high-order harmonics were subtracted from the original signal and the resulting signal, presented in fig.6, is clearly not sinusoidal. Moreover its THD is very large. The DWT, however, is insensitive to small frequency variation, thus the superior elimination of the noise. The results are presented in fig. 7, and clearly show an improvement in the signal quality.

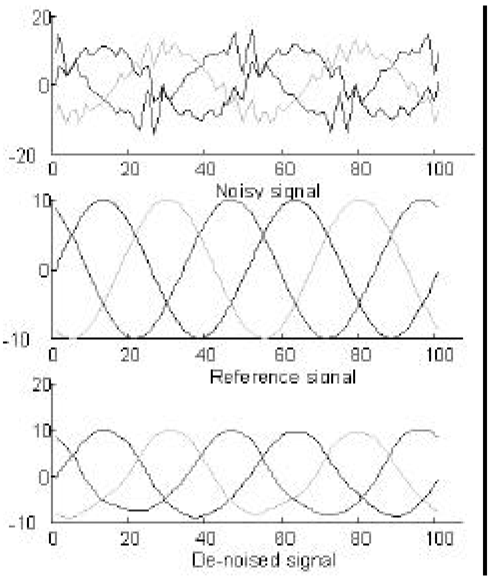


Fig.5 Some simulation results

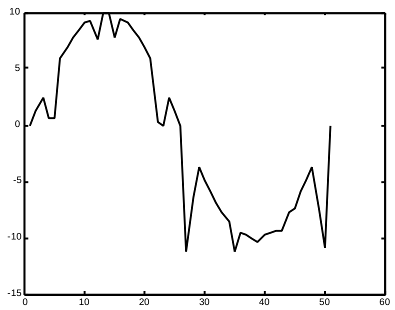


Fig. 6 De-noising of a 49Hz current using DFT

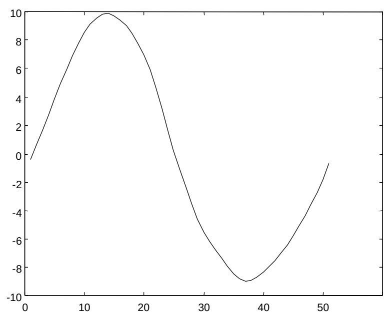


Fig.7. De-noising of a 49Hz current using DWT

Further tests were carried out, considering test-signals composed from fundamental, harmonics and interharmonics. Yet again, the DWT de-noising was satisfactory so that THD was within the imposed limits.

4 CONCLUSIONS

The results obtained proved that DWT can be used to eliminate the superior harmonics. Although the Fourier transform perform better in ideal condition the DWT proved to be insensitive small frequency shifts. Thus, it provides an interesting alternative to the classical Fourier transform, especially for currents analysis in systems where the frequency has repeated shifts around the main value.

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