SIGNAL ANALYSIS IN POLLUTED POWER NETWORKS

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Abstract: In the last decades, commercial electric customers have become increasingly interested in the relative quality of the power they purchase as their businesses rely more heavily on modern high-tech processes. The utilization of power electronic control elements causes unwanted grid perturbations, and therefore a reduction of the power supply quality. Spreading of harmonics is one of the main concerns, caused by switching actions of the power electronic control elements, and by voltage variations in the electrical grid. The mentioned grid perturbations lead to a functional impairment of the connected power electronic as well as electrical devices. In many applications, even small loads reduce power quality and lead to thermal overload of the whole appliance and a reduction of life time of single components, e.g. filter capacitors in converters. Power disturbances found on typical electric utility distribution systems degrade product quality, increase process downtime, and dissatisfy commercial customers. For this reason, monitoring the quality of the supply networks has become an issue of international interest.

This paper describes the main power quality issues in electrical networks, aiming to investigate harmonic and interharmonic pollution from a modern approach. The beginning shows a theoretical description of the distortions occurring in power networks. The paper continues with a virtual instrument that detects harmonics, interharmonics and the fundamental frequency of the processed signal. The virtual instrument is based on signal processing techniques like Fourier and Hartley transforms and a fundamental frequency approximation algorithm. In the end, the test made with simulated perturbations will be described and the results will be discussed.

Key words: **Fundamental Frequency, Harmonics and Interharmonics, Virtual Instrument, Wavelet Denoising, Zero Crossings**

1. INTRODUCTION

The problem of the quality of electrical energy is one of the most vital problems of today electrical power engineering, theory of non-sinusoidal mode being part of it . Nonsinusoidal modes cause the presence of different harmonic components in the current and voltage curves: higher harmonics; subharmornics; interharmonics.

The power must be supplied without interruption, but also the voltage and current waveforms must maintain nearly sinusoidal shape, constant frequency, and amplitude at all times to ensure continuous equipment operation. Low power quality can cause serious problems for the affected loads, such as short lifetime, malfunctions, instabilities, interruption, and etc. The key reason for our increasingly keen interests in power quality lies in the great economic value directly associated with those disturbances [1].

Poor power quality is normally characterized by the presence of disturbances such as harmonics distortion, capacitor switching, high impedance faults, transformer inrush currents, lightning pulses, and motor starting transients. In order to improve the quality of service, electrical utilities must provide real-time monitoring systems that are able to identify the signatures of different events and make proper decisions for switching and maintenance.

The analysis of power quality in electrical power systems includes the study of transient disturbances as frequency variations, sags, swells, flicker or interruptions.

According to the periodicity of power disturbances, they can be classified as stationary or non-stationary signals. For many stationary signals, Fourier analysis is a useful analysis tool because the frequency content is of great importance. Practical measurements using Fourier Transform (FT) assume infinite periodicity of the signal to be transformed. Furthermore, the time-domain information in the signal would be spread out on the whole frequency axis and become unobservable following the transformation. Therefore, this method is not suitable for analyzing non-stationary signals.

Digital control and protection of power systems require the estimation of supply frequency and its variation in real-time. Variations in system frequency from its normal value indicate the occurrence of a corrective action for its restoration. A large number of numerical methods is available for frequency estimation from the digitized samples of the system voltage[2].

The algorithm proposed in this paper is based on a Discrete Wavelet Transform filter that attenuates the high frequency harmonics which would create false zero crossings near the real one, an adaptive window of search and an algorithm that tracks the fundamental frequency and approximates it each period.

2. FUNDAMENTAL FREQUENCY ESTIMATION ALGORITHM

A. Discrete Wavelet Transform

The following signal contains a fundamental component of 50hz and two interharmonics, a low frequency interharmonic (rank 3.3) and a high frequency one (rank 25.7).

$$f(t) := 230\sin(\omega t) + 23\sin\left(3.3\omega t + \frac{\pi}{2}\right) + 17\sin(25.7\omega t) \quad (1)$$

This signal will be sampled at 512 points and it will be stored in vector F.

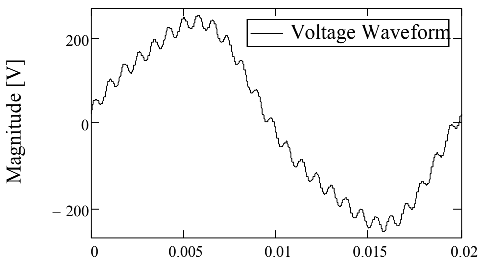

Fig. 1. Voltage Waveform

The discrete wavelet transform of function f(t):

$$DWT(f,n,x) = \int_{x-2^n}^{x+2^n} f(t)\cdot W(t-x,n)dt \quad (3)$$

The mother wavelet chosen is Daubechies 14:

Fig. 2. Daubechies 14 wavelet and scaling function

Where S is the scaling function and W is the wavelet function.

The transform coefficients are computed :

$$p = 0 \ldots n-2 \quad A_{p,i} = DWT_{2^{p+1}+\left[\frac{i}{2^{n-p-1}}\right]} \quad (4)$$

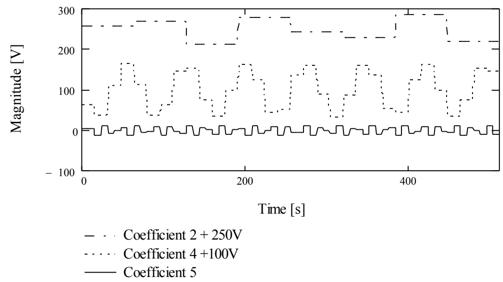

Fig. 3. Discrete Wavelet Transform coefficients (2, 4 and 5)

The signal is now decomposed in 10 levels of coefficients, each representing a frequency band. The first two elements, 0 and 1, are called *approximation* coefficients. The remaining elements are the *detail* coefficients. The transform DWT contains 8 levels of detail. The last 256 entries represent information at the smallest scale, the preceding 128 entries represent a scale twice as large, and so on.

The parameter *scale* in the wavelet analysis is similar to the scale used in maps. As in the case of maps, high scales correspond to a non-detailed global view (of the signal), and low scales correspond to a detailed view. Similarly, in terms of frequency, low frequencies (high scales) correspond to a global information of a signal (that usually spans the entire signal), whereas high frequencies (low scales) correspond to a detailed information of a hidden pattern in the signal (that usually lasts a relatively short time).

The large scale coefficients will contain the fundamental frequency and the low frequency harmonics and the high frequency harmonics will be stored in the low scale coefficients. Therefore, if the low scale coefficients are truncated, the remaining coefficients will contain only low frequencies. The signal is reconstructed by taking the Inverse Discrete Wavelet Transform of the remaining coefficients.

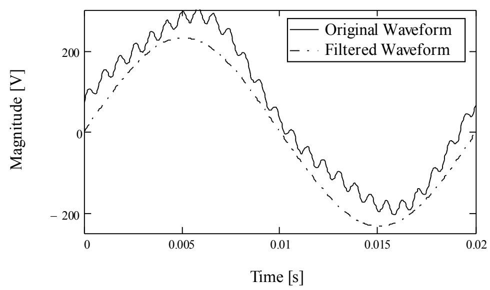

Fig. 4. Original voltage waveform and filtered voltage waveform

B. The "Zero Crossings" frequency estimation method

The zero crossing detection algorithm is based on the sign difference between two consecutive samples. When a group of t_i and t_{i+1} samples are found, a linear interpolation is performed in order to find an approximate coordinate in time when the signal value is 0.

Zero crossing condition:

$$sign(t_i) + (sign(t_{i+1}) - 2)^2 = 0 \qquad (5)$$

Linear interpolation method :

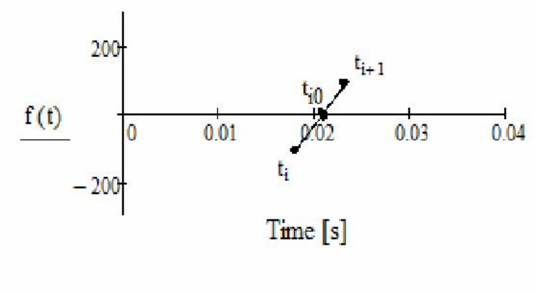

Fig. 5. Linear interpolation

$$t_{i0} = t_i - \frac{f(t_i)\cdot(t_{i+1} - t_i)}{f(t_{i+1}) - f(t_i)} \qquad (6)$$

After the first zero crossing, the axes are translated into that location in time so the first non integer cycle is truncated. The frequency estimation is determined by dividing the number of zero crosses counted and the total duration of the integer cycles.

$$f = \frac{number\ of\ crosses}{total\ time} \qquad (7)$$

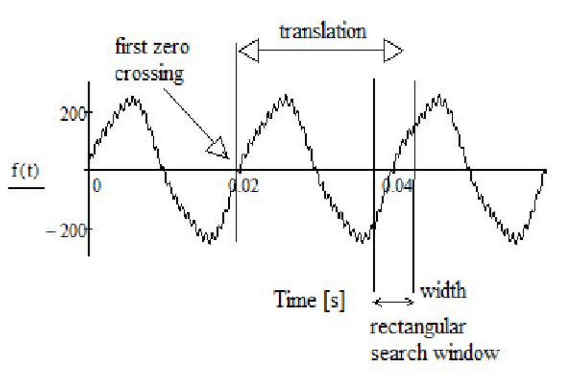

Fig. 6. First zero crossing and the adaptive search window

The rectangular search window is dimensioned after the first zero crossing when the width and the translation are defined. This window will be then shifted along the whole signal.

Multiple zero crossing can occur on a single period duration leading to major measuring errors. Usually these problems are caused by harmonics and the most difficult ones to detect are the ones very close to the fundamental zero crossing point. Luckily, if the "false" zero crossing are very close to the "real" crossing points, the harmonic frequency that produces them is also very high. Since the harmonic frequency is high, it will not pass through the low pass wavelet filter. If the false zero crossings are further than the real ones, they are produced by low rank harmonics with high amplitudes. These effects are solved by adjusting a rectangular window of search around the expected fundamental frequency value. Furthermore, the window of search will considerably reduce the number of iterations per period since the samples outside the window are not being read.

3. INTERHARMONICS

Electrical energy quality standards used in different countries identify the norms for higher harmonics levels and distortion coefficient of voltage (current) sinusoidal curve, voltage (current) harmonic component coefficient as well as permissible values of the power of valve converters. Normalizing of interharmonics is in most cases not carried out; it is either being developed or exists in way of recommendations. This can be explained by the fact that the theory of interharmonics is a relatively new field of knowledge and consequently has not been thoroughly studied as compared with the theory of high harmonics. However, in today conditions of continuous increase of different powerful nonlinear loads analysis of the quality of electrical energy without analyzing interharmonics distortions seems to be incomplete and insufficient.

Accurate harmonics/interharmonics analysis and measurement in electrical power systems are of particular importance since a true and exact spectrum of a waveform provides a clear understanding of the causes and effects of waveform distortion. In this regard, discrete Fourier transform (DFT) is still the basic tool. DFT is of great interest because it approximates the continuous Fourier transform of the time-domain signal. However, validity of this approximation is strictly a function of the wave form being analyzed and the signal sequence covered by a window width. If the window width of DFT is not properly chosen or if a fixed window width is used uniformly for a spread of frequencies, there will be a negative aspect; the spectral leakage of an analysis of a periodic wave form using DFT may reach an unacceptable level, which sometimes totally blurs the true line spectrum . To tackle the spectral leakage problem, new algorithms/schemes have been proposed in the literature.

Interharmonics can be thought of as the inter-modulation of the fundamental and harmonic components of the system with any other frequency components and can be observed in an increasing number of loads. These loads include static frequency converters, cycloconverters, sub-synchronous converter cascades, induction motors, arc furnaces and all loads not pulsating synchronously with the fundamental power system frequency.

IEC-1000-2-1 [6] defines interharmonic as follows:

"Between the harmonics of the power frequency voltage and current, further frequencies can be observed which are not an integer of the fundamental. They can appear as discrete frequencies or as a wide-band spectrum."

Harmonics and interharmonics of a waveform can be defined in terms of its spectral components in the quasi-steady state over a range of frequencies. The following table provides a simple, yet effective mathematical definition:

Table 1.

Harmonic	$f = nf_1$ where n is an integer >0
DC	$f = nf_1$ for n= 0
Interharmonic	$f \neq nf_1$ where n is an integer >0
Subharmonic	$f > 0$ Hz and $f < f_1$
f1 - fundamental power system frequency	

The interharmonic analysis is based on the discrete Fourier transform and the signal is windowed at 5Hz or 0.5Hz.

$$F(k) = \sum_{n=0}^{N-1} f(n) W_N^{-nk} \quad (8)$$

Where $$W_N = \exp\left(j2\pi / N\right) \quad (9)$$

The Fourier coefficients, which represent the real and the imaginary part of the transform, lead to the magnitude and phase spectrum.

$$a_0 = \frac{2}{T}\int_0^T f(t)\cdot dt \cong \frac{1}{2p}\sum_{i=1}^{2p} f_i \cdot \cos\left(\frac{k\cdot\pi\cdot i}{p}\right) \quad (10)$$

$$a_k = \frac{2}{T}\int_o^T f(t)\cdot\cos(k\cdot\omega_1\cdot t)dt \cong \frac{1}{p}\sum_{i=1}^{2p} f_i\cdot\cos\left(\frac{k\cdot\pi\cdot i}{p}\right) \quad (11)$$

$$b_k = \frac{2}{T}\int_o^T f(t)\cdot\sin(k\cdot\omega_1\cdot t)dt \cong \frac{1}{p}\sum_{i=1}^{2p} f_i\cdot\sin\left(\frac{k\cdot\pi\cdot i}{p}\right) \quad (12)$$

The same result can be obtained by using the Hartley Transform which, in its fast version is even faster than the Fast Fourier Transform.

$$H_k = \sum_{n=0}^{N-1}\left[x_n\cdot\left(\cos\left(2\frac{\pi}{N}\cdot n\cdot k\right) + \sin\left(2\frac{\pi}{N}\cdot n\cdot k\right)\right)\right] \quad (13)$$

The odd and even coefficients are given by :

$$O_k = \frac{H_k - H_{N-k}}{2} \qquad E_k = \frac{H_k + H_{N-k}}{2} \quad (14)$$

The amplitude and phase spectrum :

$$A_k = \sqrt{(O_k)^2 + (E_k)^2} \qquad \phi_k = \arctan\left(\frac{-O_k}{E_k}\right) \quad (15)$$

or

$$A_k = \sqrt{(a_k)^2 + (b_k)^2} \qquad \phi_k = \arctan\left(\frac{-a_k}{b_k}\right) \quad (16)$$

Considering the signal given in figure 1, the magnitude and the phase spectrum computed with a 5Hz window are presented below.

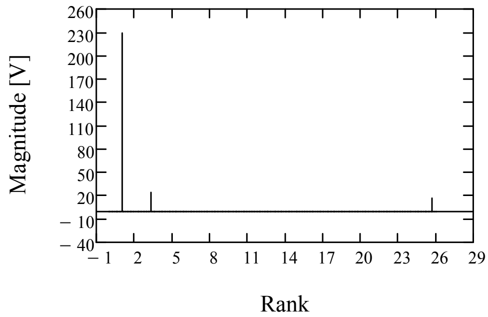

Fig. 7. Magnitude spectrum

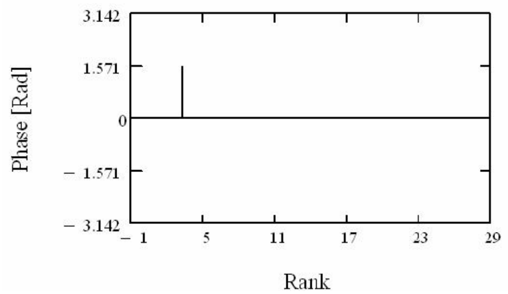

Fig. 8. Phase spectrum

The results in figure 7 and 8 do not contain any spectral leakage because the sampled signal has perfectly round components. If the signal components are real, spectral leakage appears. This leakage can be reduced by selecting an appropriate window, like the Hanning window for example.

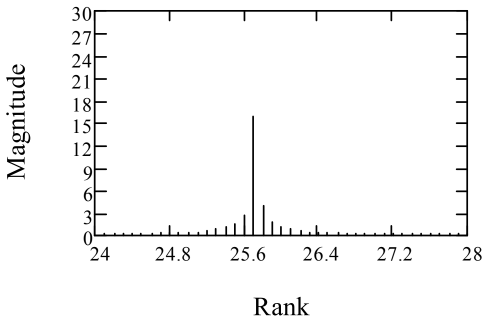

Fig. 9. Interharmonic rank 25.72 with a 5Hz rectangular window

Using the Hanning window the spectral leakage is significantly reduced. The low amplitude interharmonics are no longer visible in the next figure.

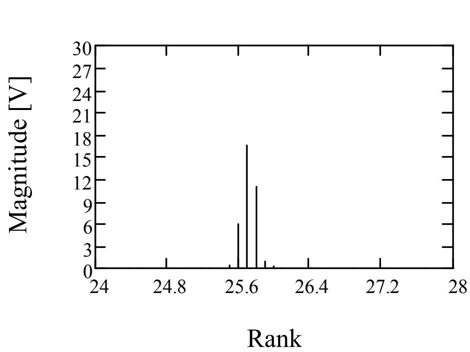

Fig. 10. Interharmonic rank 25.72 with a 5Hz Hanning window

4. VIRTUAL INSTRUMENT

A virtual instrument has been implemented with the help of LabView. The instrument has two modules, one for accurate fundamental frequency estimation and the second for interharmonic identification.

Within the frequency estimation module, the current or voltage waveform can be either inputted as an array of samples or as an analytical expression. The instrument requires the width of the search window, the total analysis time (if the waveform is an analytical expression) and the sampling frequency. The instrument returns a figure of the frequency approximation in time, the frequency being approximated every cycle, the measurement error (analytical case), the total integer cycles time, the number of detected false zero crossings and the total analysis duration. The input waveform and the wavelet filtered waveform are also displayed.

The analytical test waveform contains a 50hz fundamental frequency, a low frequency interharmonic (rank 3.3) and a high frequency interharmonic (rank 25.7), each component with a phase shift of 0.36 rad, 0.7 rad and 1.6 rad.

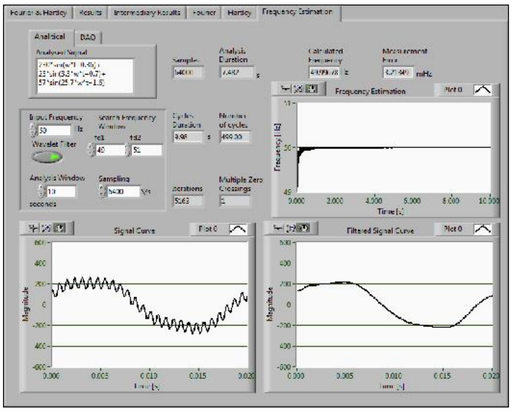

Fig. 9. Virtual instrument front panel. Frequency estimation

The total duration chosen was 10 seconds which corresponds to 500 cycles of 20ms each and the sampling time 6400 samples/second, therefore 128 samples per cycle.

The total numbers of samples was 64000 but the program performed only 5163 iterations. This reduced calculation complexity is given by the rectangular search window. The window slides along the sampled signal and the virtual instrument reads only the samples that are inside the window.

Figure 10 presents a set of measurements on the interval 42.5Hz to 58Hz, comparing the analytically input frequency and the estimated frequency. The frequency deviation is calculated as the difference between the estimated frequency and the real frequency. As seen from the table, the maximum error for the analyzed domain is 3.72mHz. According to the 61000-4-30 standard, the instrument complies with class A equipment where the maximum allowable deviation for a 10 seconds analysis is ±10mHz [5]. The total analysis duration for a 10 seconds waveform is about 5 seconds so the instrument is able to perform a real time frequency measurement.

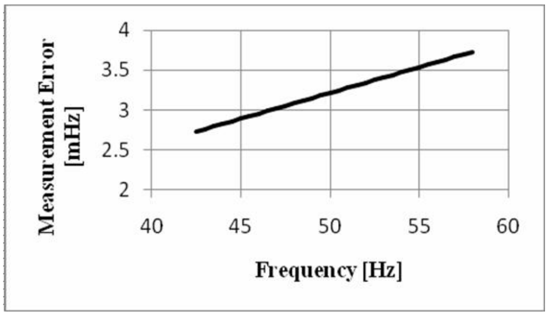

Fig. 10. Measurement error on the 42.5Hz – 57.5Hz domain

The interharmonic identification module performs the Fourier and the Hartley transform.

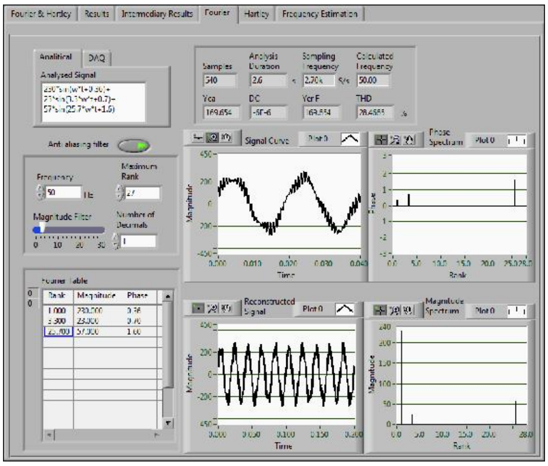

Fig. 11. Virtual instrument front panel. Interharmonic identification.

The window width chosen in the above situation was 5Hz.

CONCLUSIONS

In the last decades power quality estimation has become a very important issue in power systems so the necessity of accurate parameters measurement has grown in importance too.

The algorithm presented and the results obtained are satisfactory. The new signal processing techniques, like the discrete wavelet transform offered important accuracy improvements and solved most of the false zero crossings instances.

The paper describes briefly the wavelet denoising process which can be useful in many other applications in electrical engineering and also two different ways of interharmonics identification. The Fourier algorithm, which is the classic approach and the Hartley Transform.

With these techniques, a virtual instrument containing two modules was developed, one for the fundamental frequency estimation and the other one for interharmonics identification.

The virtual instrument developed performed well in high harmonic polluted conditions. The interharmonic identification module computed the waveform components correctly, displaying the magnitude and phase spectrum. The frequency estimation module computed the fundamental frequency of a signal with high frequency harmonics accurately. The total analysis duration was less than the total waveform duration so the instrument can be used online.

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