Smart Power Quality Monitoring System for Single-Phase Loads

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***Abstract-* Power quality (PQ) is the description of voltage, current and frequency variation on the power system. Thus, the power quality problem is a problem manifested in voltage, current or frequency deviations that results in failure or misoperation of customer equipment. The increased use of power electronic controlled equipment (such as SMPS, AC and DC motor drives, UPS and electronic lighting ballast) in the industrial equipment has led to growing concern for the quality of power and its impact on system equipment and operation. This paper addresses the issues of equipments used and software designed in simple PQ monitoring for single-phase loads and the techniques involved.**

I. Introduction

Electric power quality (PQ) has been a topic of consideration for a few decades now, but it has acquired intensified interest and importance during the last decade [1]. Power quality monitoring is the process of gathering some data about voltages and currents; transporting that data to somewhere useful; and converting that data into decision-making information. Using digitized waveform data and the Fast Fourier (FFT) algorithm, the power engineer can make fast and accurate harmonic measurements on the plant power system. However, the engineer must be careful in both the application of the equipment and the analysis of the results or the conclusions may not be correct.

In this paper, the techniques available for harmonics measurement are reviewed and the hardware design and software development for simple PQ monitoring of single-phase loads are described.

II. Harmonics

A. Methods available for measuring harmonics [2]

Engineering experiences and judgments can play an important role in detecting harmonics. Knowing the fundamental frequency, it can be roughly estimated at which frequencies the harmonics will occur, since the latter are usually multiples of the fundamental, except in the case where interharmonics exist.

Firstly, if harmonics exist, then the voltage and current spectra should show correlation. Since the harmonics will generate voltage drop along the system impedance, both the voltage and current spectra should display similar trends at these frequencies. Hence, plotting the current and voltage at the same frequency will indicate presence of harmonic at that frequency if the current and the voltage trends have similar shape, indicating correlation between them.

Secondly, a practical method is to use longer window size. This method is applicable if, and only if, the signal is almost stationary and enough data is available. In this case, longer window size can improve frequency resolution and help to locate the harmonics more accurately.

Thirdly, zero padding method is an improved technique to locate harmonics. By observing the true spectrum through limited 'fences', zero padding appends dummy samples with value of zero to the end of effective data. Zero padding smoothens the spectrum and thus helps to get a better idea of the true spectrum. This method is helpful in interpreting the spectrum but it should be noted that it cannot improve frequency resolution. The effective data is still the original ones, and there is no new information introduced by adding zeros to a signal

B. Method used for measuring harmonics [3,4,5]

The method used to measure the harmonics induced due to the load being measured was the most commonly used one. This method involves recording the sampled values of the continuous-time signals of voltage and current across the single-phase load. Once the discrete-time sampled data is obtained, a window of it is plotted against time in MATLAB, whereby a FFT (Fast Fourier Transform) algorithm is performed upon the data and hence the harmonics are extracted.

Any continuous repetitive waveform can be represented as a fundamental sinusoid with some higher frequency sinusoids. The Fourier transform takes a waveform, which is continuous in the time domain (Figure 1) and maps it on the frequency domain (Figure 2). The Fast Fourier Transform algorithm, or FFT, has been developed to make fast and accurate Fourier transforms on waveforms that are digitized and discrete in time (that is, they are not continuous to infinity).

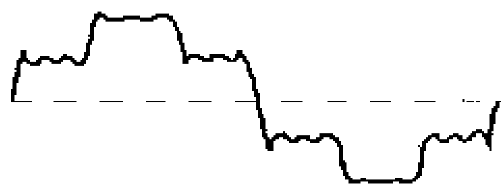

Figure 1: Distorted waveform

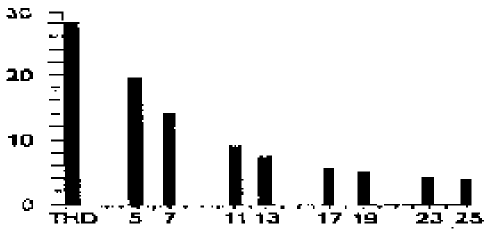

Figure 2: Example of a harmonics-ridden waveform and its harmonic spectrum

The microprocessor that performs the FFT can be that of a personal computer which is connected to an Analog-to-Digital Converter Card (ADC). However, the use of the FFT algorithm should not induce the reader to think that measuring power system harmonics is trivial. This is because of the inherent characteristics of the FFT algorithm that introduces complications.

The sampled waveform is captured and stored as discrete points separated by equal time intervals. The time intervals required is set using the Nyquist Sampling Theorem as mentioned further. The FFT is then taken on this data afterwards for extracting the harmonics. As such, the FFT is not taken real-time, and the input is not the actual continuous waveform but rather a collection of discrete data points. The behavior of the waveform in between the points can only be approximated.

The FFT assumes steady-state input (no variance in signal amplitude with time) and if the waveform is changing during the sampling interval, the FFT will display the average magnitudes for each harmonic. Therefore transient harmonic analysis is, at best, a guess for any given instant.

In order to simplify mathematical calculations for the FFT algorithm, the number of input data points is limited to a power of two. For example, the number of samples can be 512 (2^9) points. Hence, the FFT will use a block of 512 continuous points. The output from the FFT algorithm will be in terms of magnitude against frequency, and will contain exactly half the number of input points. These output points will be equally spaced in frequency and have to be adjusted (multiplied by a multiple). The first output point will be at the minimum resolvable frequency. Each additional output point will fall at integer multiples of the minimum resolvable frequency. In other terms, the first point will give the magnitude at the fundamental frequency and each additional point will be at an increasing harmonic of the fundamental. For the FFT algorithm to resolve a frequency, one complete cycle must occur during the sample period. Therefore, the minimum resolvable frequency is 1/(period length). In this case, since the fundamental frequency is 50 Hz, the time for 1 cycle is 20 milliseconds. However, it is better to sample for more than 1 cycle if waveform display is important.

C. Nyquist Sampling Theorem and Aliasing

An interpretation of the sampling theorem states that the sampling frequency must be at least twice the highest frequency contained in the original signal for a correct transfer of information to the sampled system [5,6]. The frequency component at half the sampling frequency is referred to as the Nyquist frequency.

If the software has the task of measuring, say, up to the 50^{th} harmonic, which represents 2500 Hz, given the fundamental frequency of 50 Hz, thus, the sampling frequency should be twice the highest frequency being measured. Hence, a sampling frequency of 5 kHz is required, which gives a sampling time of 0.2 milliseconds.

The representation of frequencies above the Nyquist frequency as negative frequencies means that should the sampling rate be less than twice the highest frequency present in the sampled waveform then these higher frequency components can mimic components below the Nyquist frequency, introducing error into the analysis. It is possible for high frequency components to complete many revolutions between samplings; however, since they are only sampled at discrete points in time, this information is lost.

This misinterpretation of frequencies above the Nyquist frequency, as being lower frequencies, is called ‘aliasing’ and is illustrated in Figure 3 below:

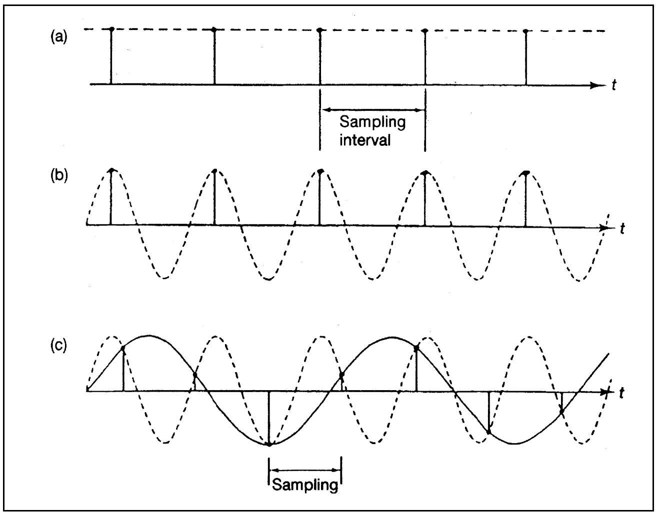

Figure 3: The effect of aliasing

(a) $x(t) = k$;
(b) $x(t) = k \cos 2\pi n f t$. For (a) and (b) both signals are interpreted as being dc.
(c) The sampling can represent two different signals with frequencies above and below Nyquist or sampling rate.

There are two methods to prevent aliasing. Firstly, to prevent aliasing, it is necessary to pass the time domain signal through a band limited low pass filter, with a cut-off frequency, f_c, equal to the Nyquist frequency. Thus, if sampling is undertaken on the filtered signal and the Discrete Fourier Transform is applied, the frequency spectrum has no aliasing effect and is an accurate representation of the frequencies in the original signal that are below the Nyquist frequency. However, information on those frequencies above the Nyquist frequency is lost due to the filtering process.

Another method to prevent aliasing is simply to make sure that the sampling time is fast enough to capture all input frequencies. For example, if there is significant harmonic distortion at the 50th harmonic, which is at 2500 Hz, the sampling rate according to the Nyquist Sampling Theorem to prevent aliasing, would be 5 kHz or 200 µs between points.

III. Software Developed

In general, the software to be developed should ideally be an intelligent power quality monitoring system with the following characteristics [7,8]:

a) It should gather all the required data, which is not restricted to current and voltage, but also to time.
b) It should provide a useful display or monitoring of the data processed, possibly offering options of softcopy or hardcopy output.
c) It should combine the power quality data with other sources of data, as results of power quality monitoring is very important
d) Finally, it should convert the data obtained into information that can be used to take action. The action may be writing a standard, turning off a Power Factor correction capacitor or even assessing if the quality of power adheres to the standards set out.
e) Moreover, it should possess a database for storage of data and enable sharing of information as well as updating data.

In order to simulate/reconstruct the digitized waveform from the discrete, sampled values obtained from the ADC and extract the required harmonics contents, many techniques are available and have been investigated. However, the Fast Fourier Transform from MATLAB 6.1 has been selected as it offers the ability to treat data which is available in the form of a sampled time function, represented by a time series of amplitudes, separated by fixed time intervals of limited duration.

The software used for data acquisition is C, the processing and extraction of harmonics is done in MATLAB 6.1. For the Graphical User Interface (GUI), report generation and database manager parts, Visual Basic 6.0 program is used. An internal module using the DDE (Dynamic Data Exchange) in Visual Basic is designed to interface with the MATLAB platform and browse for the required processed data. Finally, a Microsoft Access platform is used for hosting the database and can be modified at will.

From the specifications of the ADC, it offers a minimum sampling time of 3µs, which is largely enough to read and plot the values up to the 50th harmonic, with frequency 25000 Hz (period of 1 cycle = 400µs). As such, the data acquisition software must offer a minimum sampling time in order to capture the analog signals of current and voltage. In this perspective, Turbo C was deemed the best software for data acquisition because it allows for a sampling time in microseconds (µs) as compared to the milliseconds (ms) offered by Visual Basic.

Furthermore, MATLAB has in-built capabilities to extract harmonics from the digitized waveform, using tools provided in signal analysis, such as the Fast Fourier Transform (FFT). Visual Basic 6.0, being an object-oriented software, reduces the lines of codes required for the program, and thus eases the programming task. Moreover, the in-built features for database generation and interfacing with both MATLAB and Microsoft Access makes it the best choice for the Graphical User Interface (GUI).

Graphical User Interface (GUI)

The GUI has several forms, which perform different functions by acquiring data from the Turbo C program. The recorded data is then passed on to MATLAB for extracting harmonics, using FFT algorithm, and performing the different calculations required.

For example, the frequency analyzer form displays the frequency spectrum of both current and voltage of the load being investigated when the data acquisition process is launched. An additional feature is the display of the total harmonic frequency distortion as a percentage of the fundamental frequency at 50 Hz, the fundamental frequency having, of course, a value of 100%. This feature is available for both the current and voltage waveforms as shown in Figure 4 below.

Similarly, the power analyzer form displays the different graphs associated with power (Active, Reactive, Apparent) as well as their respective waveforms, and also display the power factor while stating whether it is lagging or leading. This form is shown in Figure 5.

Also, the program lists the relevant information such as important parameters as the form factors and amplitude (crest) factors for both current and voltage. Moreover, it also gives the voltage and current values for the highest magnitude harmonic after the fundamental. Finaly, the percentage Total Harmonic Distortion as a percentage of the fundamental frequency is given for both current and voltage. This form is shown in Figure 6.

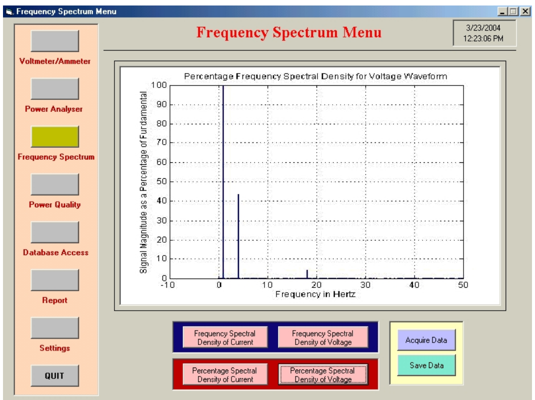

Figure 4: Frequency Spectrum

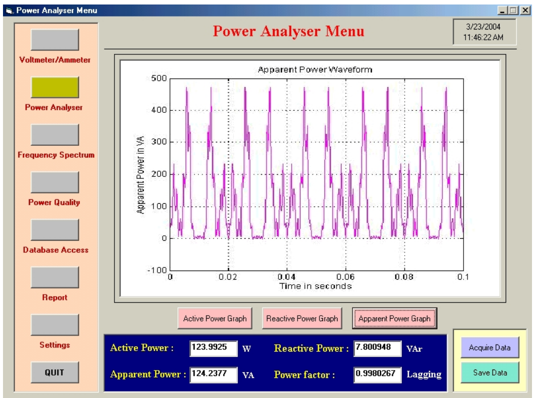

Figure 5: Power Analyser Menu

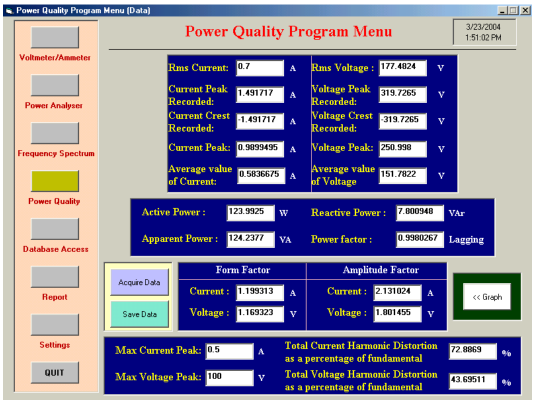

Figure 6: Power Quality Program Menu

Finally, a database including a robust collection of tools for accessing data from a database is provided, using ActiveX Data Objects (ADO). It provides an easy-to-use application-level interface to OLE DB, which provides the underlying access to data.

IV. Hardware Equipment

The parameters, which are to be monitored, being current and voltage; these are termed to be analog variables. For analog variables, the precise value of the quantity carrying the information is significant. For data acquisition techniques [9,10] where a computer is being used, the computer will understand only digital data. A digital signal is basically a binary (on/off) nature and variations in numerical value are associated with changes in the logical state (true/false) of some combination of switches. So, before converting the analog data to digital data, there are a lot of processes, which must be applied to the analog variables (current and voltage) for it to be suitable for software processing.

A. Analog-to-Digital Converter

The Analog-to-Digital Converter Card (ADC) is vital in cases where the microprocessor of a personal computer is being used to perform the Power Quality monitoring. The computer is a device, which processes only discontinuous digital signals. On the other hand, real world data are in analog form, which are continuous. Thus, proper converters were designed to enable conversion of analog-to-digital and digital-to-analog signal so that the microprocessor can communicate with the real world.

The electronic circuit that converts an analog signal into a digital signal is called an analog-to-digital (A/D) converter. Likewise, the electronic circuit that converts a digital signal into an analog signal is called a digital-to-analog (D/A) converter. Both A/D and D/A converters are known as data converters.

The A/D conversion is a quantizing process whereby an analog signal is represented by equivalent binary states. A/D converters can be classified into 2 general groups:

1. *Successive-approximation , Counter and Flash-type Converters*
   The given analog input signal is compared with the internally generated equivalent signal.

2. *Integrator converters and voltage-to-frequency Converters*
   The analog input signal is converted into time or frequency and these new parameters are compared to known values.

When comparing those two techniques, accuracy is at the expense of speed for each case. The successive-approximation type is faster but less accurate than the integrator or voltage-to-frequency type converters. The integrator converters possess many characteristics, with the foremost being their excellent noise-rejection characteristics. This is due to the input voltage being integrated over a period of time, hence any high-frequency noise riding over the input signal gets cancelled out. This is the reason why the integrator converter is not suitable for PQ monitoring, since the converter will reject high frequency harmonics. Hence, the best choice of ADC is the Successive Approximation Register (SAR) converter.

Figure 7 illustrates the block diagram of a successive approximation A/D converter, as only the Analog-to-Digital conversion technique is required.

The above process includes the following major elements:

1. The digital-to-analog (D/A) converter.
2. The comparator
3. The control unit (i.e. the microprocessor)
4. The successive approximate register (SAR) with clock (CLK)
5. The output register

The basic principle used in the Successive Approximation A/D conversion (SA ADC) is divide and search approach in converting the analog value to digital form. In particular, a binary search approach is used where the search space is reduced by half in each clock cycle. Therefore, a SA ADC will require 6 clock cycles to convert an analog signal to 6 bit values.

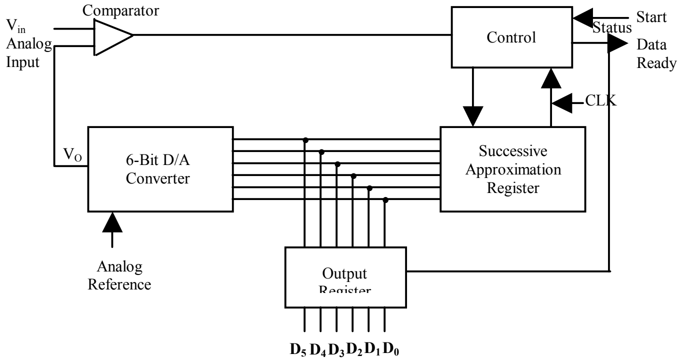

Figure 7: Detailed structure of ADC

Table 1: Comparison of different methods of measuring current

Methods for measuring current	Accuracy	Galvanic Protection	Power Dissipation Capacity	Relative Cost	Current measurement Specifications
Sense Resistor	Higher than 95%	No protection	High	Low	Works for dc as well as for AC up to 100kHz for less than 20A
Current Transformer	Around 95%	Provides protection	Moderate	Medium	Works only for AC up to 1000A, usually used for mains AC measurement
Open loop Hall sensor	Around 95%	Provides protection	Low	Medium	Works for DC as well as for AC up to 20kHz for up to 1000A
Closed loop Hall Sensor	Higher than 95%	Provides protection	Moderate to High	High	Works for dc as well as for AC up to 150kHz for up to 500A

B. Description of the working principle of ADC

A sample and hold circuit, with zero-order hold, is used to sample the analog input signal and retain its value during the data conversion process. In the first clock cycle, the most significant bit (MSB) of the successive approximation register (SAR) is set to 1. The contents of the SAR are the input to the DAC; this converts the stored value in the SAR to an analog signal value that is applied to the comparator. The comparator compares the output of the DAC with the sampled analog signal. If the output of the DAC is smaller than the sampled analog value, then the MSB of the SAR remains unchanged during the entire data conversion process; otherwise it is reset to 0 during the entire data conversion process. Thus the contents of the SAR in the first clock cycle reduce the search space by half. The same process is repeated in the second clock cycle starting with the second MSB to 1. Repeating this process for N clock cycles results in converting the analog signal to its equivalent N-bit digital value. The content of the SAR at the end of N clock cycles is the converted value of the analog input signal.

C. Signal Conditioning [11,12]

Voltage Measurement

Testing discussed in previous articles has established that relaying and metering accuracy instrument transformers have sufficient accuracy up to 5 kHz for harmonic measurement. In order to measure the instantaneous voltage across any single-phase load being applied, a voltage transformer devoid of harmonic reduction characteristics was chosen. This was a commercially available 230-6 V, 500 mA voltage transformer, with outputs of 6-0-6 V. The output 0-6 V was then chosen to feed the ADC output.
However, due to the harmonics present in the waveform being recorded, the voltage output of the transformer may exceed the peak voltage of 340 V from the primary (peak voltage = rms voltage x √2 = 240 x √2, given a voltage supply of 240 V rms). Hence the peak output of the secondary of the transformer may also exceed 8.9 V (primary peak voltage x 6/230) and thus apply a higher voltage than the 10 V maximum that the Analog-to-Digital Card connected in the PC can accept.

As such, the secondary potential had to be dropped before being fed to the ADC. The method to do this without removing any harmonics in the circuit was using linear devices such as a potentiometric division method using passive resistors. Consequently, the secondary output was halved by using two 10 kΩ resistors in series and connecting the first input channel of the ADC in parallel with one of them.

Current Measurement

In order to measure the instantaneous value of current as consumed by a single-phase load, a current transformer in the form of a current clamp is used. This is because in terms of availability and cost, as well as other aspects considered, the current clamp is the better choice. This transducer allows the ADC card to measure low electrical or/and electronic current up to 60 amperes AC/DC with a high frequency response up to 20 kHz. Another advantage is that when measuring current with the clamp, there is no need to break a circuit or to affect the isolation or sheathing. The extended measurement jaws allow performing measurements in a narrow space.
It should be ensured however, that the maximum output of the current clamp to the ADC is within the range of the latter and is easily read and input in a channel of the Analog-to-Digital Card. Since relatively small currents are to be measured by the ADC from the current clamp, a simple shunt resistor was used to convert the current into a voltage, which the ADC can then measure. This can be done providing the signal can be grounded.

V. Conclusions

Harmonic measurements made by the personal computer, which is a very powerful and versatile tool in power quality assessment, is very important in any plant where harmonic sources are a significant part of the load or where problems have been encountered with power-factor correction capacitors. The software and hardware selected to measure these high-frequency harmonics as well as other parameter have to be properly chosen. It should also be kept in mind that the FFT algorithm and digitized data present limitations which the engineer must consider when taking and analyzing data.

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