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ALGORITHM FOR EVALUATING THE BASIC CHARACTERISTICS OF THE ELECTROMAGNETIC COMPATIBILITY OF ELECTRICAL POWER SYSTEMS BY DIGITAL ANALYZER

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ABSTRACT
An algorithm for evaluating the basic characteristics of electromagnetic compatibility (EMC) such as harmonic and asymmetrical components, nonsinusoidal, unbalance coefficients (factors) of voltage and current in specific nodes of the electrical power systems, providing for some specific distorting big loads is presented in this paper.
This algorithm is designed for digital analyzer for control of electromagnetic compatibility of electrical power systems.
In addition, it will allow a certain research work to be carried out, concerning the methods and the precision of evaluation of electromagnetic compatibility.

An approach for determination the characteristics of electromagnetic compatibility (EMC) in electrical power system are discussed in the paper. This approach is used in digital analyzer developed by the authors. The structural scheme and functional abilities of the analyzer are discussed in retile in Сидеров *и др.* (2003).

BASIC CHARACTERISTICS OF EMC

The basic EMC characteristics are subject of many standards and publication such as Шидловски *и др.* (1977). These characteristics are critical for electrical power quality and are related to evaluation of non-sinusoidal current and voltage, unbalance in three phase power systems, voltage variation (deviation) and fluctuation.

The basic characteristic of non-sinusoidal current and voltage are:
- Harmonic ratio of current and voltage components

$$k_{I\nu} = I_{\nu*} = \frac{I_\nu}{I_1}.100, \%; \qquad (1)$$

$$k_{U\nu} = U_{\nu*} = \frac{U_\nu}{U_1}.100, \%; \qquad (2)$$

- Voltage (Current) total harmonic distortion (THD)

$$k_{\text{нс U}} = \frac{\sqrt{\sum\limits_{\nu=2}^{50} U_\nu^2}}{U_1}.100 = \sqrt{\sum\limits_{\nu=2}^{50} k_{U\nu}^2}, \%; \qquad (3)$$

$$k_{\text{нс I}} = \frac{\sqrt{\sum\limits_{\nu=2}^{50} I_\nu^2}}{I_1}.100 = \sqrt{\sum\limits_{\nu=2}^{50} k_{I\nu}^2}, \%; \qquad (4)$$

- Normalized THD of inductions and capacitors:

$$k_{нс,ind} = \frac{\sqrt{\sum\limits_{\nu=2}^{50} \left(\frac{U_\nu^2}{\nu^\alpha} \right)}}{U_1} 100, \%; \qquad (5)$$

$$k_{нс,cap} = \frac{\sqrt{\sum\limits_{\nu=2}^{50} \left(\nu U_\nu \right)^2}}{U_1} 100, \%, \qquad (6)$$

where
U_ν and I_ν are voltage and current harmonics;
ν - harmonic order;
α - a parameter taking value 1 or 2.

The current and voltage harmonics ratios are used for choosing power filters, estimating the overload power system elements and for control of the units that generate harmonics.

Harmonic distortions are intended for evaluating of heating and extra active power losses in the elements of electric power system: $k_{\text{нс U}}$ and $k_{\text{нс I}}$ - for power transformers, cables and airlines; $k_{нс,ind}$ - for asynchronous motor coils; $k_{нс,cap}$ - for directly connected capacitors, without protective inductors.

There are some more coefficients (factors), defined in the electrical engineering, which characterize the form of current and voltage:

- Form factor (sine wave - k_f =1.11)

$$k_{fU} = \frac{U}{U_{cp}}; \quad k_{fI} = \frac{I}{I_{cp}}; \tag{7}$$

- Crest factor (sine wave $k_a = \sqrt{2}$)

$$k_{aU} = \frac{U_{\max}}{U}; \quad k_{aI} = \frac{I_{\max}}{I}; \tag{8}$$

- Fundamental factor

$$k_{DU} = \frac{U_1}{U}; \quad k_{DI} = \frac{I_1}{I}, \tag{9}$$

where

U and *I* are r.m.s. value of the voltage and current taken for one phase and period $T=2\pi f$ (*f* is the fundamental frequency);

U_{cp} , I_{cp} – average voltage and current values for the fundamental component for one phase and period T;

U_1 , I_1 – voltage and current r.m.s. values for fundamental component;

U_{max} , I_{max} – the maximal (peak) voltage and current values in period T.

The distortion of the current and voltage curves (forms) can be evaluated by comparing form and crest factors derived from measurements and these calculated for sine wave.

The unbalance in three-phase system is characterized by unbalance factor that can be determined to be:

$$\varepsilon_u = \frac{U_{neg}}{U_{pos}}.100 = \frac{\sqrt{\sum_{\nu=1}^{40} U_{neg,\nu}^2}}{\sqrt{\sum_{\nu=1}^{40} U_{pos,\nu}^2}} 100,\%; \tag{10}$$

$$\varepsilon_i = \frac{I_{neg}}{I_{pos}}.100 = \frac{\sqrt{\sum_{\nu=1}^{40} I_{neg,\nu}^2}}{\sqrt{\sum_{\nu=1}^{40} I_{pos,\nu}^2}} 100,\%, \tag{11}$$

where

U_{pos} and I_{pos} are r.m.s. values of positive sequence components of voltage and current;

U_{neg} и I_{neg} - are r.m.s. values of negative sequence components of voltage and current;

$U_{pos,\nu}$, $U_{neg,\nu}$, $I_{pos,\nu}$, $I_{neg,\nu}$ - are r.m.s. values of positive and negative sequence components of voltage and current harmonics with order ν.

The degree of unbalance of a three-phase system is often characterized by unbalance factor of voltage and current which is determined as:

$$\alpha_u = \frac{U_0}{U_{pos}}.100 = \frac{\sqrt{\sum_{\nu=0}^{40} U_{0,\nu}^2}}{\sqrt{\sum_{\nu=1}^{40} U_{pos,\nu}^2}} 100,\%; \tag{12}$$

$$\alpha_i = \frac{I_0}{I_{pos}}.100 = \frac{\sqrt{\sum_{\nu=1}^{40} I_{0,\nu}^2}}{\sqrt{\sum_{\nu=1}^{40} I_{pos,\nu}^2}} \%, \tag{13}$$

where

U_0 and I_0 are r.m.s. values of direct sequence components of voltage and current;

$U_{0,\nu}$ и $I_{0,\nu}$ - r.m.s. values of direct sequence components of voltage and current harmonics with order ν.

The voltage deviation amplitude for each phase can be defined by the expression

$$V(t) = \frac{U(t) - U_H}{U_H} 100,\%, \tag{14}$$

where

U(t) is wrapping curve of the voltage r.m.s. values;

U_H – the nominal voltage of the power system.

The average and r.m.s. amplitude as well as average and r.m.s voltage deviation can be determined from *V(t)* .

In analogical manner the fundamental frequency deviation can be determined. The voltage and frequency fluctuation are not subject of this paper.

The voltage and current harmonic calculation are necessary to determine the basic EMC characteristics. Direct positive and negative component are estimated in three-phase unbalance systems.

When current and voltage are non-sinusoidal and unbalance, first the Fourier transformation has to be done and after that the direct, positive and negative sequence component are calculated for each harmonic. The algorithm discussed in this paper considers the way of determination of the above quantities.

OPERATION ALGORITHM

This paragraph considers the main algorithm and it's heaviest part – harmonic analysis that will be discussed in details. All possible operation modes are described in Сидеров *и др.* (2003).

General operation algorithm

Generalized flowchart of the operation algorithm is given in *Figure 1*. The following basic steps are shown:

- Initializing the system resources assigning initial values to all variables, configuration I/O interface and data memory check.
- Operation mode choosing. The desired mode has to be selected in accordance with the client task.
- Measurement. At this point all quantities related to chosen mode are measured and acquired data are put into computation routines.
- Visualization (Show) of the results in the LCD.
- Returning to operation mode choice.

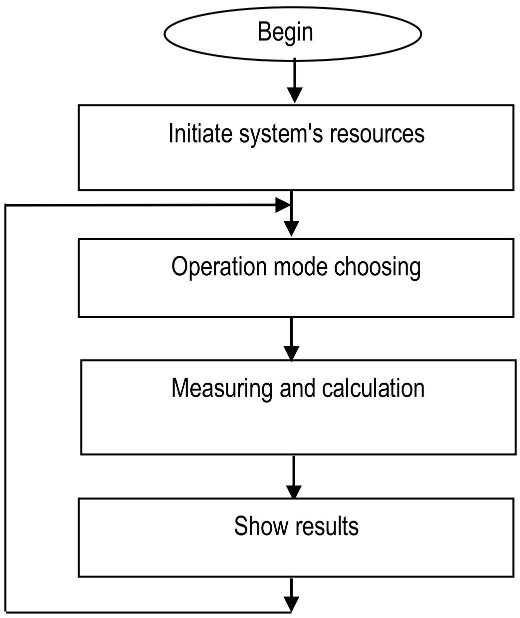

Figure 1. Flowchart of general operation algorithm

Algorithm in harmonic analysis mode

A flowchart describing working sequence is given in *Figure 2*. The main steps are:

- Choice of the external connection scheme. After each change a checking subroutines is started to reassume absence of mistakes.
- Setup of the basic measurement and calculating parameters. The user can select some measurement parameters as: sample rate, the kind of window function, the kind of anti-aliasing filter, the maximum harmonic order (the upper bound of the harmonic order depends on the previously setup parameters)
- Performing the actual measurement with ADC.
- Filtering the acquired data and saving in the memory.

As it is well known the input signals has to be limited in the frequency domain in order to receive correct results from Fast Fourier Transform (FFT). A digital filter is used to prevent the aliasing of out-of-band noise and interference. A Finite Impulse Response (FIR) filter is chosen because of its simplicity and suitable characteristics.

The digital FIR filter is linear discrete system with single input and single output. It has constant (according to time) parameters. In that way the system reactions is independent from the input signal initial time. Any linear system can be unambiguously determined by its reaction to the series of normalized pulses, each of which can be defined by the following expression Иванов (1997):

$$\delta(\kappa) = \{1,0\}, k = 0, k \neq 0. \quad (15)$$

If we mark the filter input with $\{u_n\}$ and its reaction (i.e. output) with $\{y_n\}$, the input can be presented as a sum of normalized pulses

$$u(n) = \sum_{k=-\infty}^{\infty} u(k)\delta(n-k) \quad (16)$$

And the filter output is expressed by

$$y(n) = L[\{\sum_{k=-\infty}^{\infty} u(k)\delta(n-k)\}]. \quad (17)$$

For the linear, time invariant systems, equation (17) can be written as

$$y(n) = \sum_{k=-\infty}^{\infty} u(k)h(n-k), \quad (18)$$

where $\{h_k\}$ is impulse response of the filter. In practice the input sequence is constrained, so if the number of input points is *N,* the FIR filter output will be

$$y(n) = \sum_{k=0}^{N-1} u(k)h(n-k) = \sum_{k=0}^{N-1} h(k)u(n-k). \quad (19)$$

This sum is called convolution and can be written in the next way

$$\{y_n\} = \{u_n * h_n\}. \quad (20)$$

If the number of discrete input values is equal to the number of filter coefficient, it is realized by the mathematical operation

ring shrinkage (circular shrinkage) but if not aperiodic shrinkage is used.
The filter response in the z-domain is

$$Y(z) = H(z)U(z)\,, \qquad (21)$$

Where *H(z)* is the filter transfer function

$$H(z) = \sum_{k=-\infty}^{\infty} h(k) z^{-k} \qquad (22)$$

The phase response of this kind of filters is a linear function of the frequency, which is a big advantage of such kind of applications. The linearity is achieved only for the filter coefficients are symmetrical around the central coefficient (i.e. the first and the last coefficients has to be equal). The linear filter delay is *(N – 1)/Fs*, where *Fs* is the sampling frequency, and *N* is the number of points. For instance for 21 point and 1 kHz sampling rate the delay will be 20 ms. In practice the frequency response is given by (22).

The "windowing" approach is accepted in filter design. The initial impulse response is multiplied by "window" function. The convolution realized in that way leads to smoothing in the frequency response. The user can chose between rectangular window and Hanning window. The last one is required by standard EN 61000-4-7.

In filter design the following sequence is accepted: first the sampling rate is set; the pass band and the hold band are set, after that the transition band with and normalized cutting frequency are calculated. Finally filter order and filter coefficient are defined. The results after filtration can be obtained by using equation (19).

- Performing Fast Fourier Transformation (FFT) and determine current and voltage harmonic ratio.

When the signals are captured experimentally but are not given as analytical expressions, there are two common methods for their processing:

- In the first one a function approximation is done, followed by numerical integration.;
- In the second method continuous Fourier Transformation is replaced by Discrete Fourier Transformation (DFT). This approach suits better to our tasks. The FFT is always preferred than directly applying DFT because of reducing mathematical operations. For example DFT with 512 samples requires $1{,}5.10^6$ mathematical operations, while by FFT this number can be reduced to about $2{,}5.10^4$ mathematical operations for same number of samples. In general FFT requires *N.log2N* complex multiplications, where *N* is the number of samples Доневска *и др.* (1999).

The abbreviation FFT represents not single method but set of algorithms that speed up the process. One widely used algorithm is called "time decimation". It can be generally described with the following equations:

$$F(n) = \sum_{k=0}^{N-1} f(\kappa) W^{nk}, \;\; n = 0,1,2,\ldots,(N-1); \qquad (23)$$

where
F(n) is coefficients row of DFT;
f(k) – input data sequence (the index k associated with the time), $W = e^{-j2\pi/N}$.

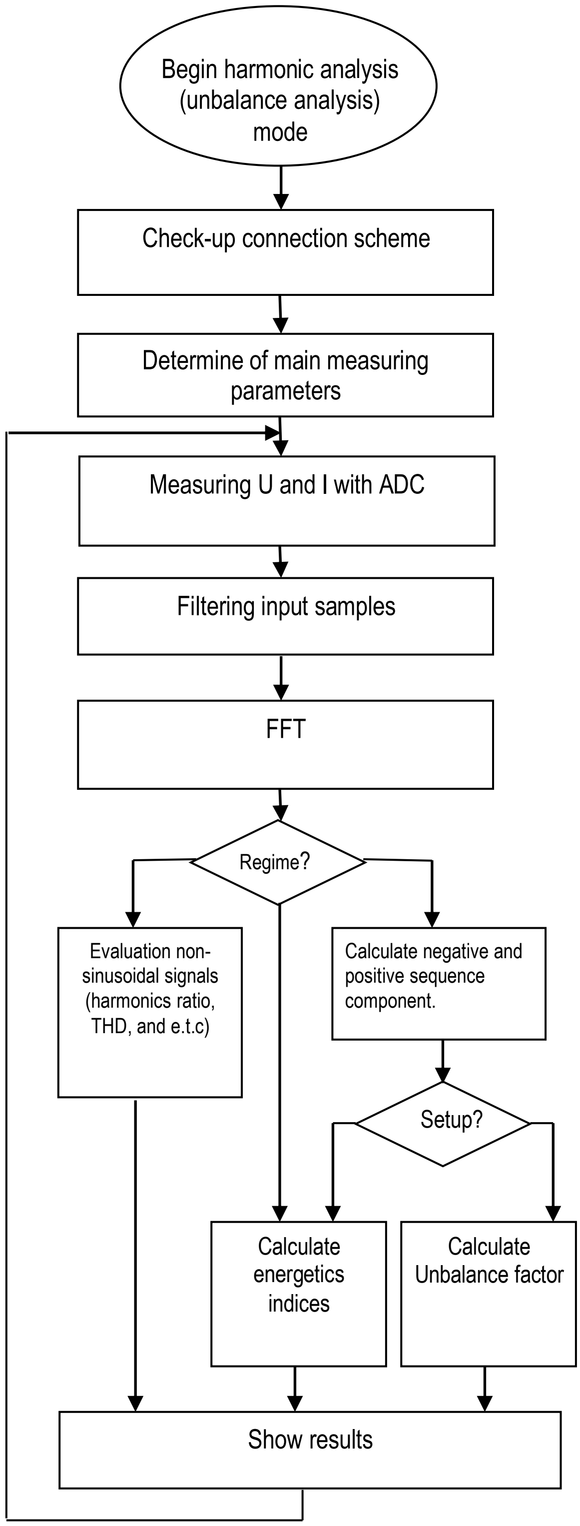

Figure 2. Flowchart for harmonic analysis mode

If the number of samples *N* can be divided by 2, the initial DFT can be decomposed in two shorter subsequences – the first one containing all samples with even indexes *f(2k.),* and the second one comprised by all samples with odd indexes *f(2k+1)*, where *k=0,1,2,…,(N/2-1)*. Each of these two new rows of data is decomposed in to parts in its turn. In that way the current and voltage harmonics are evaluated for all three phases at the same time. In this case the sum (23) can be represented by four subsumes given by (24).

$$\begin{aligned} F(n) = & \sum_{k=0}^{N/4-1} f(4k)(W^n)^{2k} + \\ & + \sum_{k=0}^{N/4-1} f(4k+1)(W^n)^{(4k+1)} + \\ & + \sum_{k=0}^{N/4-1} f(4k+2)(W^n)^{(4k+2)} + \\ & + \sum_{k=0}^{N/4-1} f(4k+3)(W^n)^{(4k+3)}. \end{aligned} \tag{24}$$

This formula can be writhen in other way

$$\begin{aligned} F(n) &= \sum_{m=0}^{3} \sum_{k=0}^{N/4-1} f(4k+m)(W^n)^{(4k+m)} = \\ &= \sum_{m=0}^{3} \sum_{k=0}^{N/4-1} f(4k+m)(W^n)^{4k}(W^n)^m \end{aligned} \tag{25}$$

The time decimation process continues until initial and final values for *к* become equal to 0 and 3.

- In dependence on the current mode of operation either non-sinusoidality characteristics are calculated by expressions (1) - (9) or unbalance parameters are calculated in accordance to the equations (10) - (13).

- The analyze is completed by visualization of the results. In stand-alone regime of operation the results are shown in graphics LCD module. If the analyzer is connected to PC specially developed software capture the data and gives much more possibilities for data processing and visualization.

- The algorithm loops with measurement of the next current and voltage periods.

The analyzer can evaluate a wide variety of EMC characteristics. In addition active and reactive power for a user-supplied interval can be measured. In such way observation of the load variation could be performed.

During the last few years the requirements concerning EMC became stronger and stronger. Such requirements are included in the new standards. This means that the question for fast and cheep measuring devices is its importance.

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