REAL-TIME IMPLEMENTATION OF ADAPTIVE AMPLITUDE ESTIMATORS FOR PROTECTION AND CONTROL PURPOSES

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In the paper a new fast and adaptive scheme for measurement of signal amplitude for protection and control purposes is presented. The algorithm developed is intended for amplitude tracking of the distorted sinusoidal signals with varying frequency. Wide frequency band features of the estimator are obtained by adaptive on-line changes of the orthogonal filter parameters (data window length) and coefficients in the measurement equations according to the observed actual frequency being tracked with special but simple digital procedure.
The developed algorithms have been tested with MATLAB and EMTP-ATP generated signals. Practical implementation of the adaptive scheme on the TMS signal processor is described in the paper. The real-time tests have shown that the time required for execution of all equations of the adaptive estimator within one time-step was equal to 0.14ms, which allows working with the sampling frequency set to 7kHz. That means that real-time application of the adaptive algorithm for estimation of amplitude of both currents and voltages is utterly attainable.

1. INTRODUCTION

Most of the digital measurement algorithms used for power system relaying purposes are designed to operate at fixed nominal frequency of input signals since frequency deviations in normal power system operating conditions are very small, allowing to get proper accuracy of estimation. There are situations, however, when protections have to operate in serious off-nominal frequency conditions (e.g. during start up of a generator) and, if their measuring algorithms are not designed properly, they have sometimes to be interlocked [1], thus impairing protection system reliability. Similarly, the relaying systems based on fault impedance estimation methods may maloperate if a fault occurs during off-nominal frequency conditions.

In digital protection systems the signals which the decision whether to trip the faulty power line is based on are usually determined from orthogonal components of current and voltage phasors obtained by use of non-recursive digital filters. Selective frequency response of the filters and resulting selective features of the estimators make them inadequate for application in off-nominal frequency conditions. Contradictory requirements to be selective (suppression of noise) and unselective (to have all-pass unity frequency response insensitive to frequency changes) call for adaptive features of the estimators. The idea of adaptive estimation has already been suggested in papers [1-5].

In this paper a new approach to the problem of signal amplitude measurement in off-nominal frequency conditions is presented. Adaptive estimators based on signal period estimation are described. According to the frequency assessment results the orthogonal filters are tuned up to the new frequency band by modification of their data window length and coefficients. The proposed method of adaptive measurement is simple and effective in use.

Practical implementation of the adaptive measurement scheme in real-time is further described. The algorithm feasibility within single sampling period resulting from the sampling frequency applied in the protection relay is tested. The testing signals are defined in MATLAB or obtained from EMTP simulations. Real-time measurements are discussed and compared with simulation study results.

2. MEASUREMENT ALGORITHM DESCRIPTION

2.1. Basic equations

For measurement of power system signal amplitudes (e.g. voltage |*U*|) the following digital algorithms are usually used:

$$\left|U\right|^2 = u_s^2(n) + u_s^2(n) \tag{1}$$

$$\left|U\right|^2 = \frac{1}{\sin(k\omega_1 T_s)}\left(u_c(n-k)u_s(n) - u_s(n-k)u_c(n)\right) \tag{2}$$

where u_c, u_s are voltage phasors at the orthogonal filter outputs (for instance full-wave Fourier cosine and sine ones), ω_1 is the angular fundamental frequency, T_s is the sampling period and k is a number of delay samples (chosen from the range 1 ... $N/4$, with N being number of signal samples in one period of the actual fundamental frequency component).

The features of algorithms (1) and (2), including their frequency responses are thoroughly discussed in [6]. It is obvious that the estimators deliver correct values of measured quantities only when the signal frequency is equal to its nominal value. When the frequency changes, certain measurement errors appear, either constant or oscillatory, depending on the estimator type.

2.1. Adaptive solution

In order to minimize estimation errors in case of application of estimators (1) and/or (2) for wide frequency band measurements (e.g. for generator protection), it is necessary to adapt the orthogonal filters to the actual frequency of the signals. It is realized by the following procedure:

- determine the number of samples N in one period of input signals (directly or by signal frequency estimation),
- set the filter data window length to N and modify filter coefficients (i.e. the filter impulse response).

Additionally, in case of estimator (2), the value of k resulting from the currently estimated signal frequency is forwarded to the measurement block, thus on-line updating the previously used delay value. The procedure of adaptive measurement can be represented by the block diagram shown in Fig. 1.

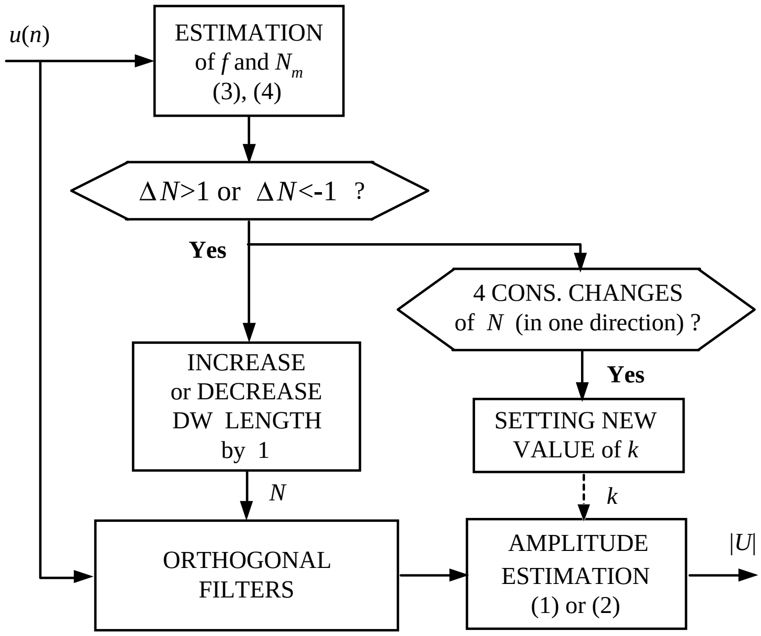

Fig. 1. Block scheme of the ΔN – based adaptive estimator.

The input signal frequency is calculated at each discrete time step according to:

$$f = \frac{1}{2\pi k T_s} arc\cos\left\{\frac{1}{2}\frac{x(n-2k)x(n-k)-x(n)x(n-3k)}{x(n-k)x(n-k)-x(n)x(n-2k)}\right\} \tag{3}$$

With the value of f from (3), the actual length of fundamental frequency period (in samples) can easily be obtained:

$$N_m = \frac{T}{T_s} = \frac{f_s}{f} \tag{4}$$

Determined number N_m is usually not a discrete value and thus can not be directly used for updating the filters' DW length N. The direction of frequency changes can be known by calculating the difference between previously set discrete value of N and the number of N_m:

$$\Delta N = N - N_m \tag{5}$$

The procedure of upgrading filter DW length is then as follows:

$$\text{If } \begin{Bmatrix} \Delta N < -1 \\ |\Delta N| \le 1 \\ \Delta N > 1 \end{Bmatrix} \text{then } \begin{Bmatrix} \text{increment } N \text{ by } 1 \\ \text{no change of } N \\ \text{decrement } N \text{ by } 1 \end{Bmatrix} \tag{6}$$

Provided the algorithm (2) is further used for amplitude calculation, the actual value of delay k has to be newly set as well. Keeping in mind that k should be an integer number, its updating is only possible after four consecutive changes of N in one direction (up or down) - i.e. when $N/4$ becomes integer.

3. VALIDATION AND TESTS

A number of various test signals have been prepared with MATLAB and EMTP-ATP programs. Both accuracy and dynamics of the adaptive estimators have been tested. In this paper just a few examples of algorithm operation for EMTP generated testing signals are presented.