NEW METHOD FOR DYNAMICAL CORRECTION OF FREQUENCY INSENSITIVE IMPEDANCE MEASURING ALGORITHMS

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Abstract – New method for dynamical correction and simultaneous frequency adaptation of measurements of protection criterion values is presented in the paper. Thanks to the new algorithms worked out and their special arrangements it is possible to get very fast and frequency insensitive estimators of impedance and its components. The simulation results confirmed theoretical considerations and shown advantages of the new algorithms as well as their possible application in fast digital protection and control systems.

Keywords: digital protection, protection criterion values, dynamical correction, frequency adaptation

1 INTRODUCTION

Safe and reliable operation of power system and continuous delivery of electrical energy require application of adequate control and protection systems. Especially important are protections used to fast and reliable elimination of faults occurring in power system. They protect power system equipment from damage or destruction and prevent from fault propagation. Proper operation of protection schemes depends on their two main functional blocks: the measurement of criterion values and decision making ones. Requirements concerning measurements are often contradictory – for instance speed and accuracy. This conflict is solved by using dynamical correction or, in general, corrective and adaptive means [1].

No doubt that new protection systems require very fast estimators of criterion values. It can be realised with recognising fault inception instant and using fast estimation algorithms, among them those applying dynamical correction. With such an approach flexible solutions can be obtained assuring both fast estimation during transients and high accuracy in the steady state [2,3,4].

Additional problem appears when the signal parameters change that are usually assumed to be constant for designed estimators. An example could be signal frequency. When it changes certain measurement errors appear due to nonorthogonality of signal components (issued at the outputs of used finite impulse response filters (FIR filters)) and worse noise reduction. To avoid the problem it is necessary to work out adaptive estimation schemes, where the estimator parameters are matched to currently measured frequency variation [5].

There are, however, some cases when it is necessary to apply very fast estimators, e.g. when signal parameters change dynamically. In such situations it is necessary to develop special algorithms, which allow for simultaneous dynamical correction and frequency adaptation. The methods and algorithms of such type have been worked out and are presented in the next sections of the paper.

2 FUNDAMENTALS OF SIMULTANEOUS DYNAMICAL CORRECTION AND FREQUENCY ADAPTATION OF CRITERION VALUES ESTIMATORS

2.1 Known methods of dynamical correction

Methods for dynamical correction mentioned in introduction are based on one of the two fundamental principles: unique response of reactive power estimator or considering transient behaviour of applied digital filters [2,3,4]. The first approach is simple and attractive, however, it requires phase shift delay of $\pi/2$ or equivalent number of samples in order to realise different than reactive power estimators. The second method is numerically complex. In both cases it is necessary to identify the fault instant. Filtering and measurement procedures are then started at this moment.

It is important to recall here the principles of the more general method, since its features will be applied later. Let samples of a signal be given by the equation:

$$x(n) = X\cos(n\Omega + \varphi)\,, \qquad (1)$$

where: $\Omega = 2\pi f / f_S$, f – frequency of the signal, f_S – sampling frequency.

A FIR filter stimulated with such an input signal (1) produces the following output signal:

$$y_S(n) = \sum_{i=0}^{n} x(n-i)h_S(i)\,. \qquad (2)$$

where: $h_S(i) = \sin((i+0{,}5)\Omega)$ – filter coefficients.

One can notice that:

$$x(n-i) = X\cos[(n-i)\Omega + \varphi] = \\ = x_C(n)\cos(i\Omega) + x_S(n)\sin(i\Omega)$$

$$\text{where:}\ x_C(n) = X\cos(n\Omega + \varphi) \\ x_S(n) = X\sin(n\Omega + \varphi)$$

and then the output signal of the filter can be given in the form:

$$y_S(n) = a(n,\Omega)x_C(n) + b(n,\Omega)x_S(n)\,, \qquad (3)$$

where: $a(n,\Omega)=\sum_{i=0}^{n} h_C(i)\cos(i\Omega)$

$$b(n,\Omega)=\sum_{i=0}^{n} h_S(i)\sin(i\Omega)$$

It is seen that coefficients a and b depend on the time instant n and frequency Ω. Taking the other digital filter (usually orthogonal) with the impulse response $h_C(i)$ one can get the output of this filter similar to eqn. (3) (say with coefficients c, d). This allows calculating orthogonal signal components $x_S(n), x_C(n)$, for each n starting from time instant equal to one. As a result one can estimate magnitude and phase of the signal $x(n)$.

When time index n reaches value of filter window length N then usually one of the coefficients a, b in eqn. (3) becomes constant and the other is equal to zero; at the output of one filter we get one orthogonal component and in the second filter the other one. It is then possible to calculate signal magnitude and phase directly and in quite simple way.

It is seen that the coefficients a, b in eqn. (3) are functions not only of the time instant n but also of the frequency of the signal. Having two different filters it is possible to estimate orthogonal signal components for any input signal frequency at every time instant. No doubt, however, that such calculations would be much more complex than with use of standard measuring procedures. That is why we are looking for different simpler methods. Some of them are presented in [3, 4].

2.2 Adaptation of estimators to changes of signal frequency

Looking at eqn. (3) we can notice that even at steady state, i.e. when time index n is equal to filter window length N, the coefficients a, b depend on frequency. The very fact of frequency dependence can also be seen on frequency response of any filter. It means that when the signal frequency differs from nominal value then outputs of a pair of orthogonal filters are not orthogonal. It is then possible to calculate these orthogonal components from linear combination of the filter outputs – according to eqn. (3) and similar equation for the other filter. With that in mind we can further estimate magnitude and phase of the signal. However, the problem of adaptation of magnitude estimator to frequency variations has much simpler solution [3, 4, 6]. This result we obtain using following algorithm of magnitude measurement:

$$X_m^2=\frac{y_S(n)y_C(n-k)-y_S(n-k)y_C(n)}{F_C(\Omega_1)F_S(\Omega_1)\sin(k\Omega_1)}, \qquad (4)$$

where: y_S, y_C – outputs of orthogonal filters in steady state, $\Omega_1=2\pi f_1/f_S$, f_1 – frequency of fundamental component, $F_C(\Omega_1)$, $F_S(\Omega_1)$ – gains of the filters at frequency f_1.

However, when the frequency of the signal changes, then the measured value is given by the equation:

$$X_m^2=\frac{F_C(\Omega)F_S(\Omega)}{F_C(\Omega_1)F_S(\Omega_1)}\frac{\sin(\Omega)}{\sin(\Omega_1)}X^2, \qquad (5)$$

where: X_m – measured magnitude, X – real, proper value of magnitude,
and it is possible to calculate real value of signal magnitude (squared) as follows:

$$X^2=\frac{1}{W_{Cor}(\Omega)}X_m^2, \qquad (6)$$

where: $W_{Cor}=\frac{F_C(\Omega)F_S(\Omega)}{F_C(\Omega_1)F_S(\Omega_1)}\frac{\sin(\Omega)}{\sin(\Omega_1)}$

For large frequency changes it is necessary to use both adaptation coefficient W_{Cor} and adequate filter window N (to get better noise filtering). The block scheme of adaptive measurement is presented in Fig. 1.

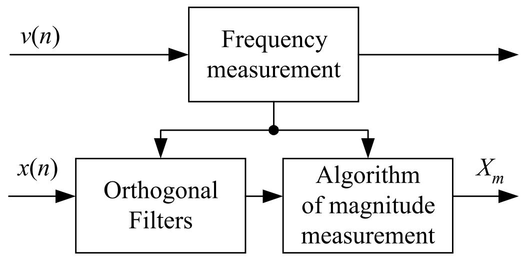

Figure 1: Scheme of magnitude measurement with adaptation to frequency variation

2.3 Dynamical correction of frequency adaptive estimators

Theoretical considerations and simulation results clearly show that algorithm (4) is the best to realise both frequency adaptation and simultaneous dynamical correction. The simplest way to reach that is addition of dynamical correction to frequency adaptive scheme described above – see eqn. (4). To get very simple form of final result it is necessary to solve the problem of adequate realisation of delayed signals of the filters: $y_C(n-k)$, $y_S(n-k)$. It can be made either at the input or at the output of the filters as shown in Fig. 2.

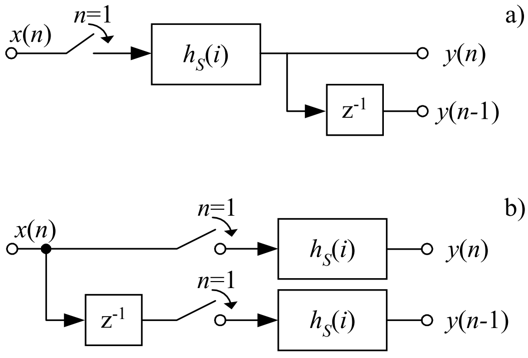

Figure 2: Two possibilities of realisation of delay of the filter output.

During steady state it gives the same result, however, during transients we have following equations (describing delayed output signal of FIR filter):

- for the scheme from Fig. 2a

$$y(n-1) = a(n-1,\Omega)x_C(n-1) + b(n-1,\Omega)x_S(n-1) \quad (7a)$$

- for the scheme from Fig. 2b

$$y(n-1) = a(n,\Omega)x_C(n-1) + b(n,\Omega)x_S(n-1)\,. \quad (7b)$$

Solution from Fig. 2b is advantageous in such a meaning that coefficients *a*, *b* are the same for signals at given instant and for signals delayed. It results in especially simple correction factor. If such switch control is used the switches are closed in time equivalent to fixed delay value. It is counted from fault instant. Then following dynamical correction factor can be found:

$$C^2(n) = \frac{1}{N^2}\left((n+1)^2 - \frac{\sin^2(n\Omega_1)}{\sin^2(\Omega_1)}\right) \quad \text{for } 0 \le n \le 19\,. \quad (8)$$
$$C^2(n) = 1 \;\; for \;\; n > 19$$

Now the equation of magnitude of voltage or current measurement with application of dynamical correction and adaptation to frequency changes can be given in the form

$$X = \frac{X_m(n,\Omega)}{C(n)W_{Cor}(\Omega)}\,. \quad (9)$$

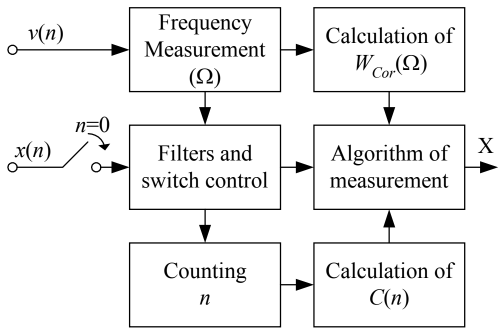

Figure 3: Scheme of simultaneous frequency adaptation and dynamic correction for magnitude measurement.

Block scheme of its realisation is shown in Fig. 3. One can notice that dynamical correction factor (8) is equal to one in steady state. The same is valid for the frequency adaptation factor at nominal frequency. Otherwise they are not equal to unity and either correction or adaptation is active (or both of them)

2.4 Impedance estimator

If the same algorithm (4) is applied to measurement of voltage and current and when in addition the same methods for dynamical correction and frequency adaptation are used then very simple algorithm of impedance estimation is obtained:

$$Z = \frac{V}{I} = \frac{V_m(n,\Omega)}{I_m(n,\Omega)}\,. \quad (10)$$

It means that in the case of impedance measurement we can obtain very fast frequency insensitive algorithm applying neither correction nor adaptation (under condition that eqn. (4) and switch control from Fig. 2b are used). Of course, to suppress signal noise adequately the filter window length should be controlled by matching it to actual frequency.

2.5 Estimators of active power and impedance components

It was proved that none of known algorithms of power can be adapted and simultaneously corrected in such a simply way as magnitude. However, it was found that similar possibility exists when we use a sum of chosen algorithms of active power. Let the equations of active power calculations be given in the form:

$$P_{m1}(n,\Omega) = \frac{u_S(n)i_C(n-1) - u_S(n-1)i_C(n)}{F_S(\Omega_1)F_C(\Omega_1)\sin(\Omega_1)}\,, \quad (11a)$$

$$P_{m2}(n,\Omega) = \frac{u_C(n)i_S(n-1) - u_C(n-1)i_S(n)}{F_S(\Omega_1)F_C(\Omega_1)\sin(\Omega_1)}\,. \quad (11b)$$

Then the same coefficients of dynamical correction and frequency adaptation as in magnitude measurement may be used:

$$P = \frac{P_{m1}(n,\Omega) + P_{m2}(n,\Omega)}{2}\,\frac{1}{C(n)W_{Cor}(\Omega)}\,. \quad (12)$$

It allows calculating resistance according to equation:

$$R = \frac{2P}{I^2} = \frac{P_{m1}(n,\Omega) + P_{m2}(n,\Omega)}{I_m^2(n,\Omega)}\,. \quad (13)$$

It is seen that coefficients $C(n)$ and $W_{Cor}(\Omega)$ were cancelled. Then in this case we manage to obtain very fast frequency insensitive resistance estimator – similarly to impedance measurement.

Having such estimators of impedance and resistance it is possible to get reactance estimator:

$$X = \sqrt{Z^2 - R^2}\,. \quad (14)$$

And finally we can say that certain special family of criterion value estimators was worked out. Estimators of impedance and its components have distinctive features. They are very fast and frequency insensitive without application of dynamical correction and frequency adaptation.

3 SIMULATION STUDIES

3.1 Scope of simulative analyses

Described above algorithms of impedance components measurement with application of dynamical correction and frequency adaptation were put to thorough simulative tests. The tests were performed in two stages. Firstly, the signals generated with MATLAB package were used for testing of algorithms (10), (13) and (14). The considered signals did not contain any disturbances, therefore they could serve as a base for checking conformity of the results with theoretical expectations. Next the other signals contaminated with higher harmonic and decaying DC components were prepared (currents and voltages), with the aim to determine the influence of such disturbances on accuracy of measurement.

In the second stage of analyses the voltage and current signals generated with use of EMTP-ATP programme were used. The model of test power system is shown in Fig. 4. It consists of two subsystems A and B represented by electromotive forces behind certain impedances (both positive and zero sequence) connected with 400 kV 150 km overhead transmission line. Parameters of the line and subsystems are given in Fig. 4. Numerous fault cases on the line connecting substations A and B were simulated, with such parameters being changed as fault type, fault resistance, fault duration etc. The purpose of this stage of analyses was to check correctness of the impedance components measurement (|Z|, R, X) as well as performing comparative studies with the standard algorithms, i.e. when correction of measurement equations is not applied.

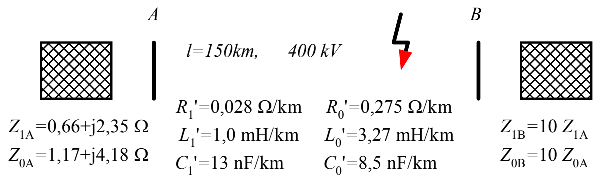

Figure 4: EMTP-ATP model of test power transmission system

3.2 *Investigation results*

The investigation results obtained in the fist stage, i.e. for pure sinusoidal signals defined in MATLAB, fully confirmed the expectations as to accuracy and dynamical features of the measurement algorithms with combined dynamical-frequency correction. Thanks to applied amendments, as described in previous section, all the algorithms proposed were able to deliver proper values of measured parameters as fast as in the second sampling step after sudden change of signal shape. Such results were an effect of compensation of frequency dependence and correction of dynamical trajectory of FIR filters used.

In Fig. 5 the measurement outcomes of impedance |*Z*| and its components *R* , *X* for the signals with simultaneous sudden change of amplitude and frequency are shown. It is seen that the steady state of algorithm response after sudden signal jump was obtained almost immediately, which is not the case for algorithms without correction applied.

In Fig. 6 a comparison of measurement results (impedance absolute value |*Z*|) is presented for the following algorithms:

- with correction, eqn. (10),
- without correction, using simple relationship with orthogonal components of voltage and current signals, in the form:

$$|Z| = \frac{U}{I} = \frac{\sqrt{u_C^2 + u_S^2}}{\sqrt{i_C^2 + i_S^2}}, \tag{15}$$

- without correction, using the algorithm (4) for both current and voltage magnitude calculation, which yields the following algorithm:

$$|Z| = \frac{U}{I} = \sqrt{\frac{u_S(n)u_C(n-1) - u_S(n-1)u_C(n)}{i_S(n)i_C(n-1) - i_S(n-1)i_C(n)}} \tag{16}$$

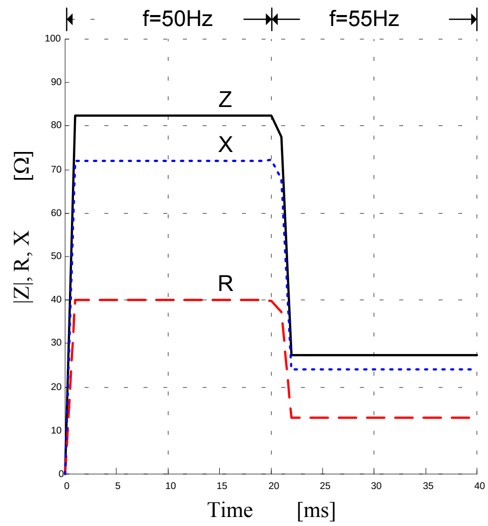

Figure 5: Results of impedance absolute value |*Z*| (solid line), resistance *R* (dashed line) and reactance X (dotted line); measurement with use of eqn. (10), (13), (14); at time instant *t*=21ms sudden change of both signal amplitude and frequency took place.

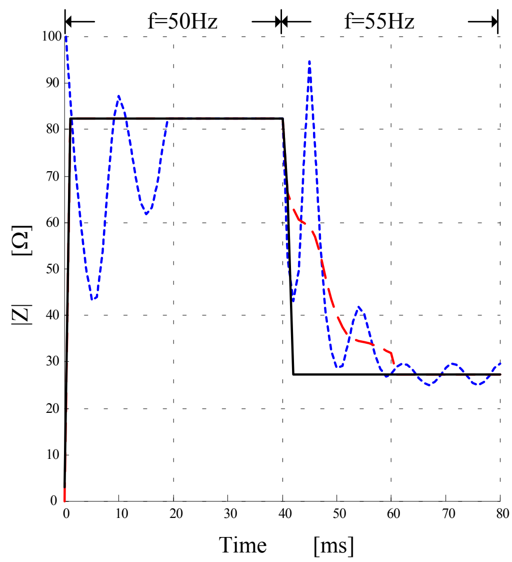

Figure 6: Results of impedance absolute value |*Z*| measurement with use of algorithms: with correction (10) (solid line), standard algorithm (15) (dotted line), standard algorithm (16) (dashed line); at time instant *t*=41ms sudden change of both impedance and frequency took place.

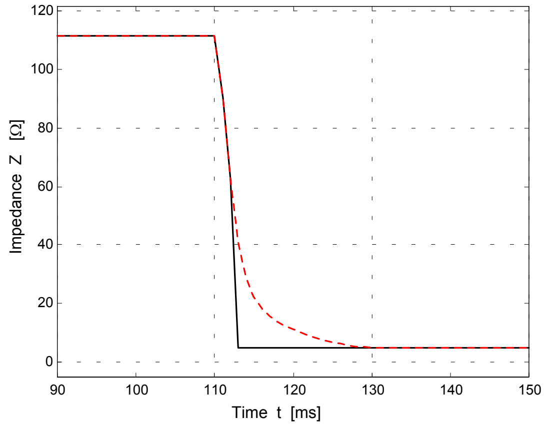

Figure 7: Results of measurement of $|Z|$ with use of algorithms: with correction (10) (solid line), without correction (16) (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in faulty phase (at subsystem A side).

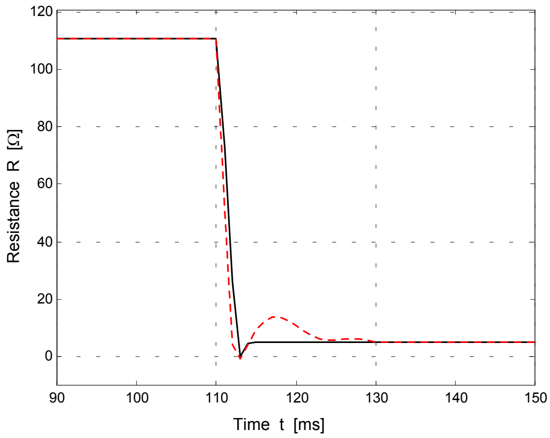

Figure 8: Results of measurement of R with use of algorithms: with correction (13) (solid line), without correction (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in faulty phase (at subsystem A side).

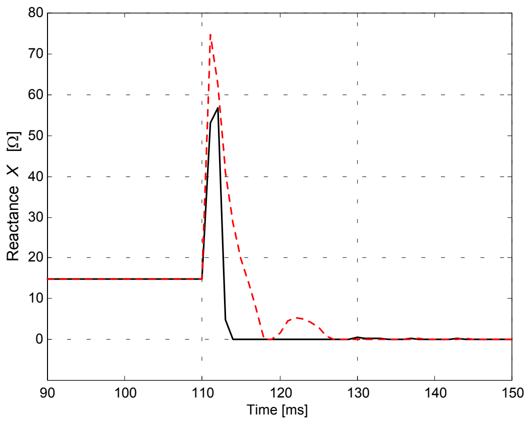

Figure 9: Results of measurement of X with use of algorithms: with correction (14) (solid line), without correction (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in faulty phase (at subsystem A side).

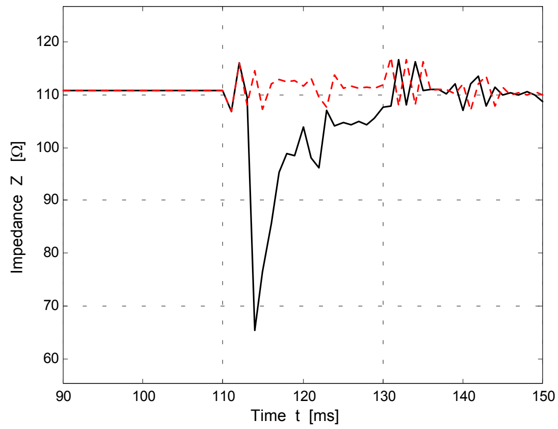

Figure 10: Results of measurement of $|Z|$ with use of algorithms: with correction (10) (solid line), without correction (16) (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in healthy phase (at subsystem B side).

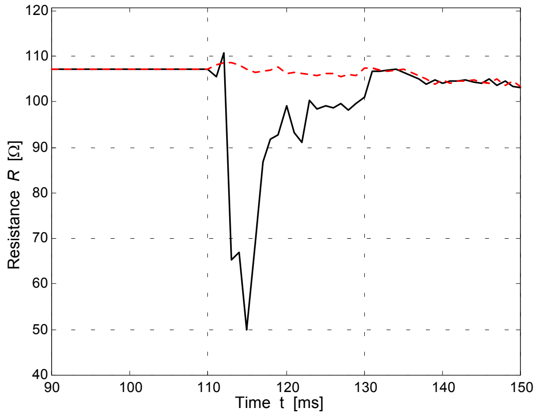

Figure 11: Results of measurement of R with use of algorithms: with correction (13) (solid line), without correction (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in healthy phase (at subsystem B side).

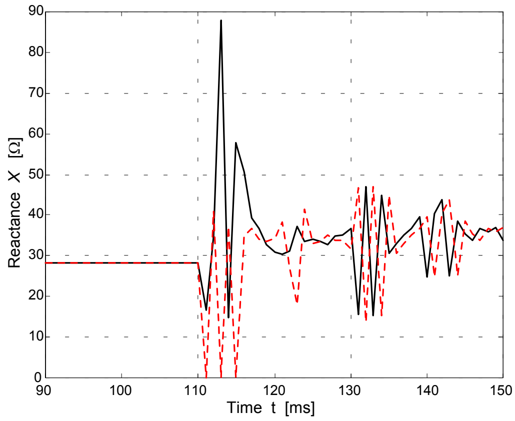

Figure 12: Results of measurement of X with use of algorithms: with correction (14) (solid line), without correction (dashed line); at time instant t=111ms phase-to-ground fault begins; measurement in healthy phase (at subsystem B side).

In the Fig. 6 significant advantages of the algorithm with correction (10) over the other standard measurement methods are visible. Thanks to introduced correction both dynamical and frequency variation induced estimation errors were eliminated, thus enabling fast and smooth answer of the algorithm (10). The algorithms (15) and (16) delivered outcomes characterised by measurement transients lasting ca. 20 ms. Additionally (in case of eqn. (15)), not fading oscillations in the steady state are observed, which is an effect of FIR filter windows and gains being unmatched to current signal frequency.

On the basis of simulative studies with use of the signals from EMTP-ATP (contaminated with various disturbance components), the influence of signal disturbances on measurement accuracy was determined. It was affirmed that higher harmonics present in voltage signal cause appearing of fading oscillatory components in transient state of measurement. In case of current signal, presence of decaying DC component is responsible for underestimating of measured impedance components (lower values are delivered than they are in reality).

Some results of $|Z|$, R and X measurement that were obtained for signals generated with use of EMTP-ATP (phase-to-ground fault case in the middle of the line) are presented in Figs. 7 – 12. It is seen that proposed algorithms with correction assure much faster and more accurate results when the measurement is performed in the phase involved in the fault (Figs. 7 – 9). However, because of high relative content of disturbance components in the signals from healthy phases the results delivered with algorithms (10), (13), (14) under such conditions are worse than with algorithms without correction (Figs. 10 – 12). Such features of the algorithms with correction result from the very fact that their equations have been derived for a simple single-component signal model and that dynamical correction coefficients $C(n)$ have very strong amplifying effect on all disturbance components.

The best operation of the algorithms with combined dynamical-frequency correction was observed for measurement of impedance components for the cases of phase-to-phase faults.

4 CONCLUSIONS

In the paper a new method for combined dynamical and frequency dependence correction intended for improvement of the transient and steady state features of measurement algorithms is presented. Taking advantage of the approach proposed new effective algorithms of impedance measurement have been designed. The algorithms developed are characterised by very fast, yet accurate response to dynamically changing parameters of the signal to be measured, especially for currents and voltages from the phases involved in the fault.

The most important advantage of the correction method described in the paper is its very simple realisation with correction coefficients $C(n)$ that are quite easy to be determined for given set of used filters and measurement algorithms. The negative feature of presented method is its high sensitivity to signal components (disturbances) that have not been included in the measurement model, which is characteristic for all algorithms with introduced corrective means.

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