SIMULATIVE ANALYSIS OF A NEW TYPE WIDE RANGE ESTIMATOR OF PROTECTION CRITERION VALUES

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ABSTRACT - *In the paper a new adaptive estimator of protection criterion values insensitive to wide range power system frequency changes is proposed and analysed. Its operation is based on a coarse calculation of signal frequency and appropriate correction of used orthogonal filters frequency responses as well as adaptation of chosen parameters of the measurement algorithm. The main features of the new method has been compared with standard criterion values estimators both theoretically and by simulations. The proposed estimators assure good accuracy of measurement within the frequency range from 5 to 95 Hz.*

INTRODUCTION

The calculation of such protection criterion values as current and voltage magnitudes, power and impedance components is an essential part of the relaying procedure leading to the final decision concerning state of the protected plant. A great number of algorithms has been proposed lately to measure the values by means of digital methods and technique. Among these are algorithms applying Fourier transform or FFT as well as different variants of least square error technique. Many of the methods apply orthogonal signal components obtained by means of nonrecursive orthogonal filters, Kalman filters or signal delaying [1-4]. A common feature of these algorithms is their design for fixed nominal frequency (50 or 60 Hz in most cases). Deviations of the frequency small but met in power system under normal and abnormal operating conditions are the source of errors of criterion values measurement.

Significant errors of estimation may occur in cases when power system frequency changes in a wider range. Such situation can be observed e.g. for reversal hydro plants which go continuously from turbine to pump operating conditions and the other way round, during warm-up of cross-compound generators etc. To avoid consequences of possible protection system maloperations some protection functions are usually temporarily blocked. On the other hand, additional relays used for low-frequency protection must be removed from service for normal operation, i.e. after generator synchronisation.

In this contribution two groups of estimators are discussed and their performance for sample input signals obtained by simulation are described. First, a set of frequency independent estimators is given for which no adaptation is needed if the measured signals do not contain noise. The algorithms are developed in such a way that the gains of applied orthogonal filters (varying with frequency) are cancelled. Such opportunity exists for quotient-type estimators like impedance components or phase shift, contrary to the second group of estimators (signal magnitudes, power components) for which an adaptation procedure is necessary. If the signals are contaminated by noise, some corrective means are indispensable for all the estimators. Both filters frequency responses and delay values in the estimator equations are adjusted to the actual value of signal frequency which is calculated by additional simple and stable estimation procedure [6].

The proposed estimators offer good accuracy and dynamics for wide range frequency changes allowing to measure the protection criterion values at frequencies from almost zero up to approximately one hundred Hz. Numerous simulation tests (with use of MATLAB and EMTP) allowed to characterise the estimators under different power system operating conditions. The simulation results of proposed adaptive estimators operation are compared with those obtained for standard estimation algorithms. The results confirmed that the developed estimators can be successfully applied to different tasks of power system control and protection.

STANDARD MAGNITUDE, POWER, IMPEDANCE AND FREQUENCY ESTIMATORS

Let a noise-free current or voltage phasor sampled with the rate T is given in the form:

$$\begin{aligned} y(n) &= Y\exp[\,j(n\omega T+\varphi)] = y_C(n) + jy_S(n) = \\ &= Y\cos(n\omega T+\varphi) + jY\sin(n\omega T+\varphi) \end{aligned} \tag{1}$$

Assuming that a pair of nonrecursive filters (having sine and cosine impulse responses, respectively) is used for the purpose of signals orthogonalization the filter outputs may be expressed as:

$$\begin{aligned} y_{CF}(n) &= \sum_{k=0}^{N-1} Y\cos[(n-k)\omega T+\varphi]\cos(k\frac{2\pi}{N}+\frac{\pi}{N}) = \\ &= F_C(\omega)\,y_C(n) \end{aligned} \tag{2a}$$

$$\begin{aligned} y_{SF}(n) &= \sum_{k=0}^{N-1} Y\cos[(n-k)\omega T+\varphi]\sin(k\frac{2\pi}{N}+\frac{\pi}{N}) = \\ &= F_S(\omega)\,y_S(n) \end{aligned} \tag{2b}$$

where: N - number of samples of filter window,
$F_C(\omega),\ F_S(\omega)$ - filter gains at frequency ω.

The signals (2a), (2b) are orthogonal for all frequencies, however, their magnitude, initially equal to Y, is now modulated by varying filter gain coefficients due to frequency changes. The simplest estimators of sought criterion values will be as follows:

$$Y = \sqrt{\left[\frac{y_{CF}(n)}{F_C(\omega)}\right]^2 + \left[\frac{y_{SF}(n)}{F_S(\omega)}\right]^2} \tag{3a}$$

$$P = 0.5\left[\frac{u_{CF}(n)i_{CF}(n)}{F_C^2(\omega)} + \frac{u_{SF}(n)i_{SF}(n)}{F_S^2(\omega)}\right] \tag{3b}$$

$$Q = 0.5\left[\frac{u_{SF}(n)i_{CF}(n)}{F_C^2(\omega)} - \frac{u_{CF}(n)i_{SF}(n)}{F_S^2(\omega)}\right] \tag{3c}$$

$$R = \frac{\dfrac{u_{CF}(n)i_{CF}(n)}{F_C^2(\omega)} + \dfrac{u_{SF}(n)i_{SF}(n)}{F_S^2(\omega)}}{\dfrac{i_{CF}(n)i_{CF}(n)}{F_C^2(\omega)} + \dfrac{i_{SF}(n)i_{SF}(n)}{F_S^2(\omega)}} \tag{3d}$$

$$X = \frac{\dfrac{u_{SF}(n)i_{CF}(n)}{F_C^2(\omega)} - \dfrac{u_{CF}(n)i_{SF}(n)}{F_S^2(\omega)}}{\dfrac{i_{CF}(n)i_{CF}(n)}{F_C^2(\omega)} + \dfrac{i_{SF}(n)i_{SF}(n)}{F_S^2(\omega)}} \tag{3e}$$

$$f = \frac{1}{2\pi}\,\frac{\dfrac{y_{CF}(n)\Delta y_{SF}(n)}{F_C(\omega)F_S(\omega)} - \dfrac{y_{SF}(n)\Delta y_{CF}(n)}{F_C(\omega)F_S(\omega)}}{\dfrac{y_{CF}(n)y_{CF}(n)}{F_C^2(\omega)} + \dfrac{y_{SF}(n)y_{SF}(n)}{F_S^2(\omega)}} \tag{3f}$$

where: $\Delta y_{CF}(n),\ \Delta y_{SF}(n)$ - finite difference representation of the first derivatives of filter outputs.

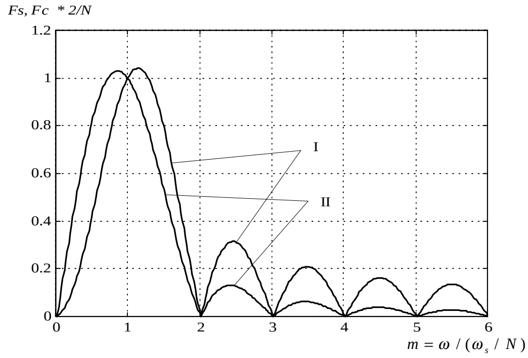

Fig.1. Frequency responses of full-cycle orthogonal filters having: I - cosine, II - sine data windows.

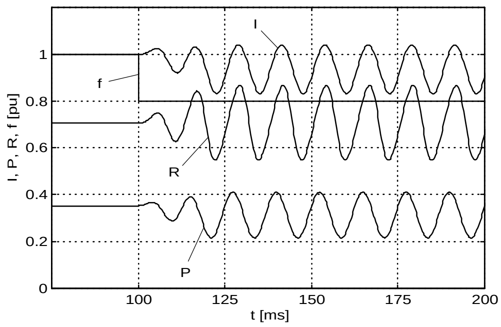

Fig.2. Sample estimation results for purely sinusoidal voltage and current signals; f=40Hz, $\varphi_u - \varphi_i = \pi/4$.

It is seen that the estimators (3) deliver correct values of measured quantities only when the signal frequency is equal to its nominal value for which the estimators were designed. In the case of frequency deviations increasing measurement errors appear due to varying (with signal frequency) algorithm coefficients. The errors are especially high when the frequency varies in a wide range.

A discussion of frequency dependent errors of estimation may be done taking into consideration the filters frequency responses, shown in Fig.1. It is seen from the figure that when the frequency deviates from its nominal value then the filter gains change and are different than those fixed in eq. (3). This is a source of errors increasing with the frequency deviation. Moreover, if the input signals are contaminated by harmonic noise, additional errors may occur due to imperfect signal filtration. In this case the noise can not be rejected completely since the points when filter gains equal to zero appear only for frequencies being a multiple of the nominal value. An example of estimation process for purely sinusoidal input signals and sudden frequency drop from 50 to 40Hz (at t=100ms) is shown in Fig.2. The calculated current magnitude, active power and resistance oscillate around the real values. This is a result of inadequate filter gains which had been initially set for other signal frequency.

WIDE FREQUENCY RANGE ESTIMATORS

To achieve suitably accurate measurement algorithms one has to give special consideration to mentioned above effects. The first one can be removed by either development of adequate algorithms or/and recursive correction of frequency-dependent filter gains.

Looking at the estimation eqn. (3) one can say that the first solution is possible for the quotient-type algorithms, i.e. for impedance components, phase shift and frequency, contrary to such quantities as power components, current and voltage magnitudes. Such simple frequency independent estimators may be obtained when all the coefficients in the numerator and denominator are identical, i.e. of the type $1/(F_C F_S)$ instead of $1/F_C^2$ or $1/F_S^2$, allowing for their cancellation.

Applying voltage and current signals to eqn. (2), after simple rearrangements, the following fundamental relationship for filter outputs is obtained [6]:

$$\begin{aligned} &u_{SF}(n)i_{CF}(n-k) - u_{SF}(n-k)i_{CF}(n) = \\ &= UI\cos(\varphi_U - \varphi_I)F_C(\omega)F_S(\omega)\sin(k\omega T) \end{aligned} \tag{4}$$

Rearrangement of eqn. (4) leads to the following quotient-type algorithms independent of filter gains:

$$f = \frac{\cos^{-1}\left(\dfrac{y_{SF}(n)y_{CF}(n-2k) - y_{SF}(n-2k)y_{CF}(n)}{y_{SF}(n)y_{CF}(n-k) - y_{SF}(n-k)y_{CF}(n)}\right)}{2\pi k} \tag{5a}$$

$$R = \frac{u_{SF}(n)i_{CF}(n-k) - u_{SF}(n-k)i_{CF}(n)}{i_{SF}(n)i_{CF}(n-k) - i_{SF}(n-k)i_{CF}(n)} \tag{5b}$$

$$Z^2 = \frac{u_{SF}(n)u_{CF}(n-k) - u_{SF}(n-k)u_{CF}(n)}{i_{SF}(n)i_{CF}(n-k) - i_{SF}(n-k)i_{CF}(n)} \tag{5c}$$

$$X = \sqrt{Z^2 - R^2} \tag{5d}$$

$$\varphi_U - \varphi_I = \varphi = arctg\left(X / R\right) \tag{5e}$$

It is worth to note here that the value of delay samples k should be chosen in such a way that $\sin(k\omega T) \neq 0$. The remaining criterion values resulting from eqn. (4) are not filter gains independent functions. Relevant estimation algorithms of voltage (current) magnitude as well as active and reactive power are given in the form:

$$U = \frac{\sqrt{u_{SF}(n)u_{CF}(n-k) - u_{SF}(n-k)u_{CF}(n)}}{\sqrt{F_C(\omega)F_S(\omega)\sin(k\omega T)}} \tag{6a}$$

$$P = \frac{u_{SF}(n)i_{CF}(n-k) - u_{SF}(n-k)i_{CF}(n)}{2F_C(\omega)F_S(\omega)\sin(k\omega T)} \tag{6b}$$

$$Q = \frac{u_{SF}(n)i_{CF}(n) - u_{SF}(n)i_{CF}(n)}{2F_C(\omega)F_S(\omega)} \tag{6c}$$

In this case the product of filter gains in eqn. (6) must be corrected according to the actual value of frequency which should be estimated by additional procedure. The filter gains for various frequencies may be either pre-calculated and stored in look-up tables or updated on-line, depending on the capability of applied microprocessor system.

Concluding one can say that it is possible to get filter gains independent algorithms (5) which could be applied for wide range of frequencies. However, to get small estimation errors due to noise, the filter frequency responses should be at least coarsely matched to actual signal frequency. Such solution is necessary for estimators (6) and improves performance of algorithms (5).

ADAPTIVE METHOD

Remarkable extension of the accurate estimation over a wide frequency range can be realised by use of the adaptive method presented. The principle of the method is illustrated by Fig.3. At the first stage a coarse estimation of the input signal frequency is carried out (prior to digital filtration) and the approximate value of the actual frequency is determined with resolution which depends on the used sampling rate. Any of the input signals available from power system can be used for the purpose but preference should be made for the voltage ones, in particular for those which are less prone to distortions (for example, line-to-line voltages instead of line-to ground, etc.). Then the following function (here: for voltage signal) is calculated [6]:

$$\cos(k\omega T) = \frac{U(n-2k)U(n-m) - U(n)U(n-2k-m)}{2[U(n-k)U(n-m) - U(n)U(n-k-m)]} \tag{7}$$

where: $k = N/4$ and N is the number of samples in one cycle of the coarsely estimated frequency ω_k; m is a certain arbitrary time delay ($m>0$).

It can be shown that the function (7) is proportional to frequency deviation, i.e.: $\cos(k\omega T) \cong -kT(\omega - \omega_k)$

The value of k is updated acc. to the procedure:

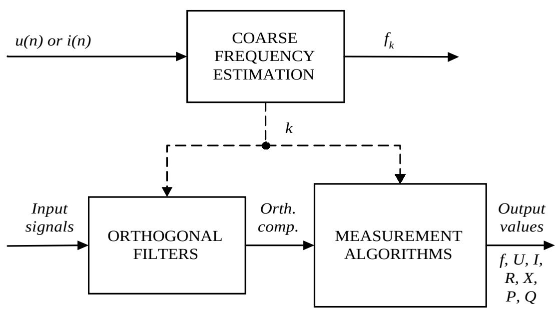

Fig.3. Block scheme of proposed wide-frequency band adaptive estimators of protection criterion values.

$$If \begin{Bmatrix} -kT(\omega-\omega_k)<-\varepsilon \\ |kT(\omega-\omega_k)|\leq\varepsilon \\ -kT(\omega-\omega_k)>\varepsilon \end{Bmatrix} \text{then} \begin{Bmatrix} increment\ k\ by\ 1 \\ no\ change\ of\ k \\ decrement\ k\ by\ 1 \end{Bmatrix} \quad (8)$$

where ε is a discrimination threshold assumed for the expected range of frequency changes.

According to the result of coarse frequency estimation the orthogonal filters are adjusted and their data windows modified to ensure optimal filtration of the measured signal. The value of delay k in (5), (6) is always set to be equal to a quarter of the actual frequency cycle. The data window length of the orthogonal filters is extended (or compressed) to the value of $N=4k$ (full cycle of the actual frequency). The filter coefficients (and by the same their frequency response) are also modified respectively.

SIMULATIONS AND PERFORMANCE ANALYSIS

The operation of proposed adaptive estimators was extensively tested in simulative way for input signals both undistorted and distorted by harmonics (2nd up to 4th - 0.1 pu) and decaying DC component. First, using MATLAB program, the test signals were generated (at sampling rate of 4 kHz) with two scenarios of frequency changes: step change from 50 to 40 Hz, linear change at the rate of +20 Hz/s.

A pair of full-cycle cosine and sine filters was used to obtain the orthogonal components of the input signals. The performance of proposed adaptive estimators (Fig.3) applying base algorithms (5) and (6) was compared with the non-adaptive estimators designed for the nominal frequency of 50 Hz.

In Fig.4a the results of frequency estimation obtained by use of the adaptive estimator (5a) are shown for distorted input signals and step change of frequency. Very good dynamics of measurement may be observed for both coarse and fine frequency estimates. The coarse frequency estimate reaches its quasi-steady state in app. 15ms. After that time the estimate reveals 1.5Hz periodic deviation from the actual frequency since, despite of filters adaptation, the higher harmonics are not rejected entirely. The fine estimation monothonically reaches the correct value of frequency in app. 35ms giving the steady state error equal to 0.2Hz (this can be reduced to 0.05Hz by additional averaging).

The tracking properties of the estimator are shown in Fig.4b. The frequency changes are tracked perfectly with the time delay which is equal app. to one period of currently measured frequency.

The next figure (Fig.5) illustrates the performance of the algorithm (6a) with adaptation for the step (Fig.5a) and linear (Fig.5b) change of frequency and distorted as before input signal. The simulation results are compared with those obtained by use of standard estimator (3a). The performance of the proposed adaptive estimator is almost errorless over the whole simulated frequency range (the instantaneous estimation errors do not exceed the level of 2%) while the errors of the standard one reaches the unacceptable level of 30-40%. Proper adjustment of the orthogonal filter windows results in almost perfect suppression (rejection) of the noise contrary to the case when no adaptation is realised.

The curves in Fig.5b may be understood as „dynamic” frequency characteristics of the examined estimation algorithms (the same also for Fig.6c,d).

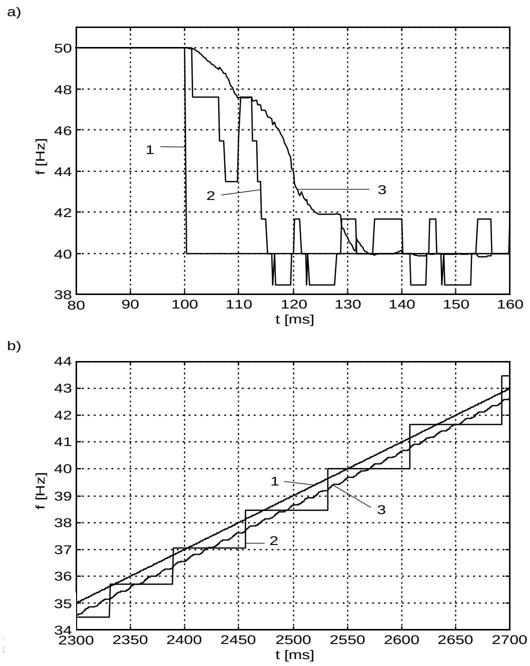

Fig.4. Frequency estimation results for contaminated by noise input signal and: a) step, b) linear change of frequency; 1 - the reference frequency, 2 - the coarse estimate, 3 - the fine estimate.

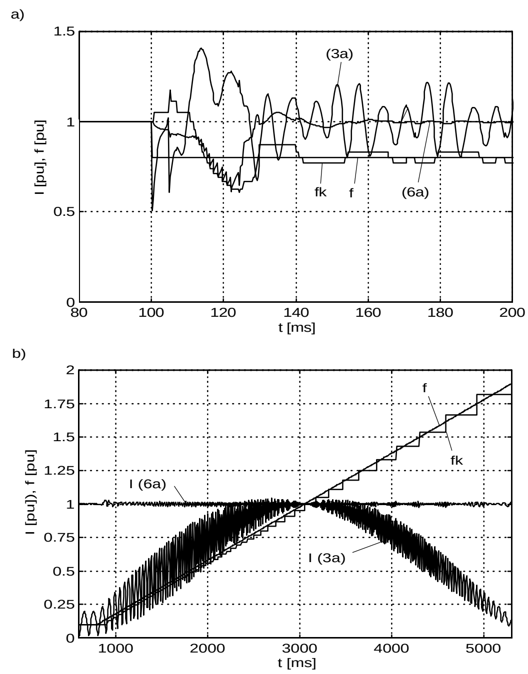

Fig.5. Estimation of magnitude for distorted current signal for: a) step change of frequency, b) linear change of frequency.

However, it should be noted that the used orthogonal filters did never see the full cycle of measured signal since the current frequency was continuously varying. The magnitude estimate obtained with the algorithm (3a) oscillates between the frequency responses of used orthogonal filters while in the case of the adaptive estimator (6a) the unique measurement curve is observed. At time t=3050 ms at which the signal frequency f crosses value of 50 Hz a node of lowest estimation errors can be observed for both algorithms which is understandable having in mind the filters frequency responses.

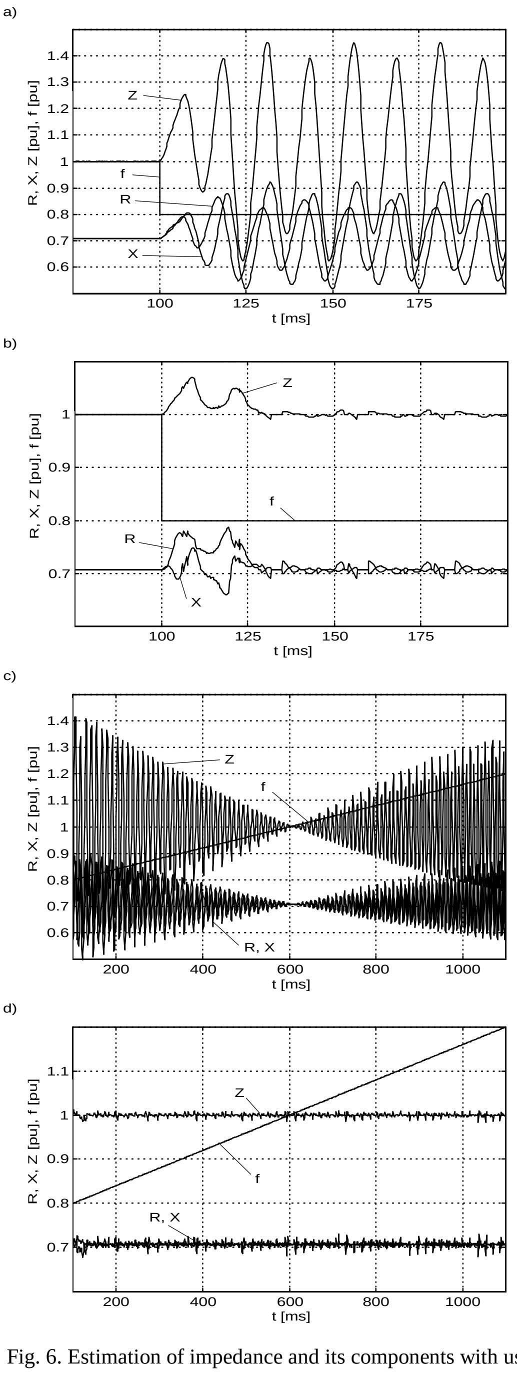

Fig. 6. Estimation of impedance and its components with use of standard algorithm (3) (a, c) and adaptive algorithm (5) (b, d) for: a, b) step change of frequency, c, d) linear change of frequency.

The similar performance features as for magnitude estimators are observed for the algorithms of impedance measurement (Fig.6). In the case of step change of frequency the acceptable results are obtained only with use of the adaptive estimators (5b-d) for which the steady state measurement errors are not greater than 2%. Advantages of the adaptive method are most clearly visible during frequency tracking (Fig.6c, 6d). It is seen that again only the adaptive algorithm is able to ensure the acceptable accuracy of estimation in the assumed range of frequency changes due to adaptive „on-line" updating of the orthogonal filters frequency responses. The outstanding features of proposed adaptive estimation method remain valid also in much wider range of frequency (between 5 and 95 Hz).

The other tests of the proposed adaptive estimation algorithms were done with use of EMTP package. In order to test the proposed estimators performance under dynamic conditions close to those that may happen in real power systems, an example of quite complicated system configuration shown in Fig.7 was modeled. Main parameters of the system are placed at the figure, the details can be found in [5]. The system under study consists of two generation points (P, Q) feeding common busbars R via two 400 kV transmission lines. The system is initially connected to the infinite bus and loaded to 1128 MVA, pf=0.88 (the generators take only a part of the total load). At time t=200 ms the switch to the infinite bus is opened (the system becomes isolated). Additionally at t=1.0 s a 3-phase fault in the middle of the line Q-R was modeled. Transients in generators P and Q rotors' speed accompanying this sudden load variation and following fault conditions are shown in Fig.8a. The effects of voltage regulator operation can be observed in the voltage waveform (Fig. 8b), while in the current signal some fluctuations due to power swing between the two generating points are visible (Fig. 8c). In consequence of new loading conditions the system frequency decreases. At the moment of fault inception the system frequency was equal to 48.5 Hz. Since in the simulated case the fault is not cleared within a reasonable period of time, both generators accelerate reaching speed values far beyond acceptable limits.

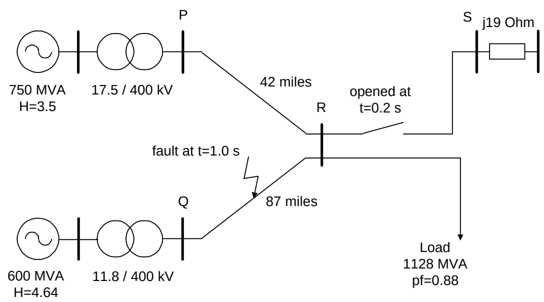

Fig.7. Fragment of power system modeled in EMTP.

The proposed adaptive scheme was used to measure the magnitude of current (phase A) flowing from the station Q to the busbars R. The coarse frequency estimation was based on phase A voltage. Figure 8d presents the results of current magnitude estimation with use of algorithms (3a) and (6a). It may be noticed from the figure that acceptable accuracy of estimation is obtained for the algorithm (6) only. Considerable differences are observed for the time range 1.0 - 1.7 s, i.e. after fault inception. Poor performance features of the estimator (3a) result from the very fact that the input signals are not very well filtrated and the noise is not completely rejected. The EMTP simulations confirmed the excellent properties of the adaptive estimator (6) which turned out to be the best also for current signal generated using MATLAB and frequency changing within a very wide range.

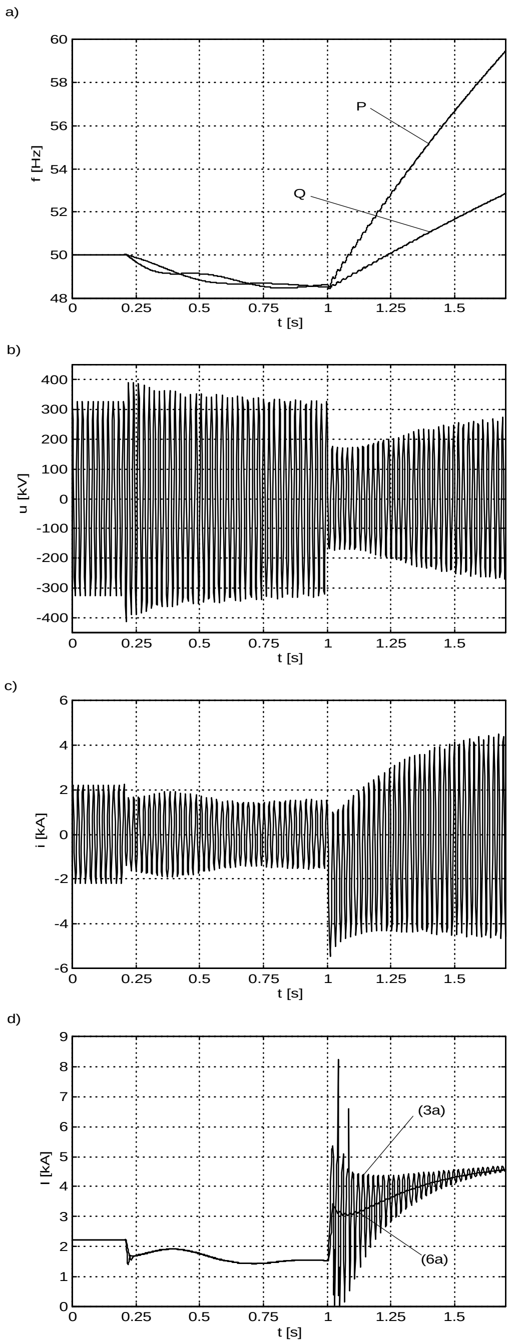

Fig.8. Transient signals due to load change and 3-phase fault: a) rotor's speed of generators P and Q, b) phase A voltage, c) phase A current (signals picked up at busbars Q) d) magnitude measurement with use of standard (3a) and adaptive (6a) estimators.

CONCLUSIONS

In the paper new adaptive estimators insensitive to wide range power system frequency changes are proposed and analysed. The proposed estimators assure good accuracy of protection criterion values measurement within the frequency range from 5 to 95 Hz. This range can be even greater since the only limitations are due to zero filter gains. Adaptive features of the estimator are obtained by on-line modification of digital filter data windows (and their frequency responses) as well as relevant parameters of the measuring algorithm according to results of a coarse measurement of the actual frequency.

Simulative analysis and tests with use of MATLAB and EMTP packages confirmed that proposed estimation algorithms have good accuracy and dynamics adequate for measurement during transient phenomena in power systems. The measurement errors due to frequency changes in the system are much less than those appearing when standard algorithms without adaptation are applied.

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