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Signal Processing Algorithms for Intelligent Electronic Devices Using Phasor Measurement Technology

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KEYWORDS

Intelligent electronic devices, phasor measurements, relay protection, robust and adaptive signal processing algorithms, digital filters.

1 INTRODUCTION

Intelligent electronic devices (IEDs), using phasor measurement technology, along with traditional application area of WAMS are becoming in recent years more and more common in WACS, WAPS and as basic management elements in intelligent electrical networks [1,2].

Measurement algorithms for complex current amplitude and fundamental harmonic voltage, based on complex amplitude orthogonal component reshaping or orthogonal current component as well as fundamental harmonic voltage forming, are the most prevailing in IEDs, which use phasor measurement technology.

Issues about algorithm synthesis of IED signal processing, using robust or adaptive algorithms of signal filtering, are considered in the following paper. The first type of the algorithms is founded on robust filter usage, that have low sensitivity to disturbance parameters changing and frequency deviation in power system. The application of signal processing adaptive algorithms allows to realize IED with maximum speed.

2 IED SIGNAL PROCESSING ALGORITHMS

2.1 Algorithms for orthogonal components of complex amplitudes

A brief description of a basic algorithm for signal processing by example of an analog system-prototype is given in the tab. 1. An IED input signal $\chi(t) = \text{Re}(\dot{\mathbf{X}}^T e^{pt})$ is represented by a set of complex amplitudes $\dot{\mathbf{X}}$ and frequencies $\mathbf{p}$. Let us constrain the signal models to one model only as an exponential component, useful sinusoidal component of commercial frequency ω_1 (nominal value is $\omega_0 = 2\pi 50 \text{ rad/sec}$) and higher harmonics. More complicated signal models are considered in the papers [3-5]. An average FIR filter $K_a(p)$ should isolate the constant ($\dot{y}(t) = \dot{X}_1$ at $\omega_1 = \omega_0$) or low-frequency component ($\dot{y}(t) = \dot{X}_1(t)$ at $\omega_1 \neq \omega_0$) and have low sensitivity to the changing exponential disturbance damping coefficient β_0 , as well as to the changing power system frequency $\omega_1 = 2\pi(50 \pm 5) \text{ rad/sec}$.

An average FIR filter with a rectangular time window is used for the popular Fourier algorithm in IEDs. The Fourier algorithm improvement is related to usage of filters with more complicated time windows; an instantaneous spectrum analysis on bigger quantity of frequencies; an application of adaptive algorithms [6,7].

№	Items	Diagram/expressions
1.	Block scheme of algorithm	$\chi(t) = \text{Re}(\dot{\mathbf{X}}^T e^{pt}) \rightarrow \begin{array}{c} \times \\ \uparrow e^{-j\omega_0 t} \end{array} \rightarrow \dot{x}(t) \rightarrow K_a(p) \rightarrow \dot{y}(t) = \dot{X}_1(t)$
2.	Signal description $\chi(t)$	$\dot{\mathbf{X}} = [X_0 \quad \dot{X}_1 \quad \dot{X}_2 \quad \dots \quad \dot{X}_N]^T, \mathbf{p} = [-\beta_0 \quad j\omega_1 \quad j2\omega_1 \quad \dots \quad jN\omega_1]^T$
3.	Input signal of filter	$\dot{x}(t) = 2\dot{\mathbf{X}}^T e^{(\mathbf{p}-j\omega_0)t} + 2\bar{\mathbf{X}}^T e^{(\bar{\mathbf{p}}-j\omega_0)t},$ $\bar{\mathbf{X}}, \bar{\mathbf{p}} - \text{are complex - adjoint vectors}$
4.	Algorithm	$\dot{X}_1(t) = k_1 X_T(j\omega_0, t), T_0 = 2\pi / \omega_0,$ $X_T(j\omega_0, t) = \int_{t-T_0}^t \chi(\tau) e^{-j\omega_0 \tau} g(t-\tau) d\tau = \int_{t-T_0}^t \dot{x}(\tau) g(t-\tau) d\tau$

Table 1: A basic algorithm of IED

An adaptive algorithm using robust filters [7] is shown in fig. 1.

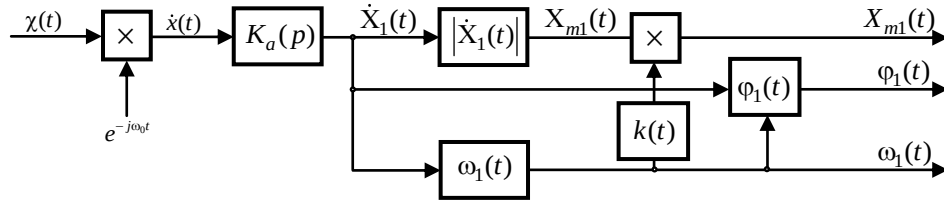


Fig. 1: Adaptive algorithm with usage of robust average FIR filters

To meet the accuracy requirements in signal processing, average FIR filters should ensure efficient elimination of higher harmonics and damped oscillatory components within complex frequency $p = -\beta + j\omega_0$. For amplitude-frequency characteristic of an average filter over the frequency range 0 to 5 Hz, whereas the assurance of the gain coefficient being close to one, is not needed. In this case a power system frequency is determined according to the following algorithm

$$\omega_1(t) = \omega_0 - \frac{X'_c(t)X_s(t) - X'_s(t)X_c(t)}{|\dot{X}_1(t)|^2},$$

where $\dot{X}_1(t) = X_c(t) + jX_s(t)$.

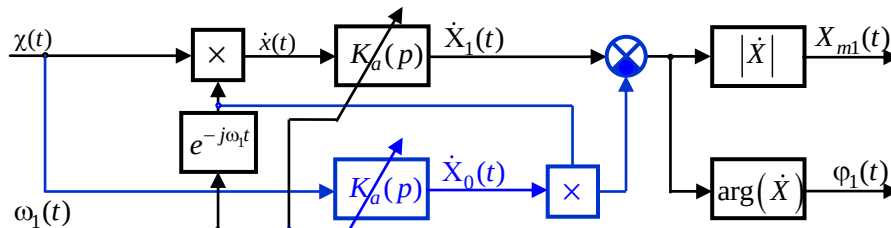


Fig. 2: Adaptive algorithm

An algorithm using adaptive average FIR filters, which adjustment is defined by the preliminarily measured value of a power system frequency $\omega_1(t)$, is given in the fig. 2. The additional module, marked with blue color, is ensuring reduction of signal processing errors over the specified range of the changing exponential disturbance damping coefficient by using signal instantaneous spectrum values on zero frequency [7].

2.2 Algorithms of useful signal orthogonal components

The block schemes of algorithms, based on useful sinusoidal signal orthogonal components shaping, are given in the fig. 3. In case of using a filter with transfer function $\underline{K}_1(p) = 100(p + j\omega_0)^{-1}(1 - e^{-pT_0})$ for the algorithm, shown in fig. 3 a), one will get the result, analogous to the Fourier algorithm. The algorithm from the fig. 3 b) at $K_a(p) = 50p^{-1}(1 - e^{-pT_0})$ is equivalent to the algorithm from the fig. 2 [7].

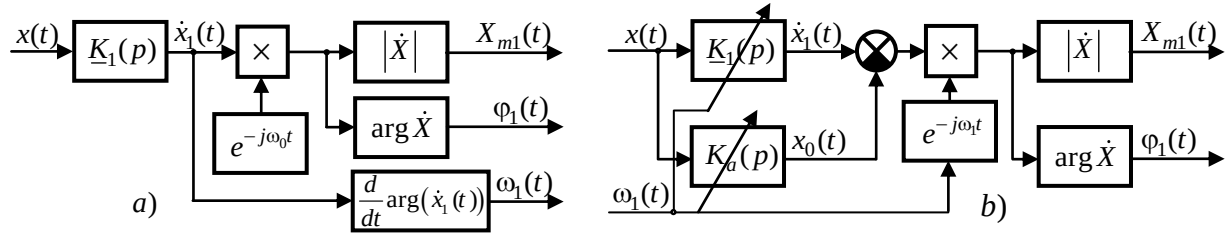


Fig. 3: Algorithms of useful signal orthogonal components

As well as for the algorithms from the tab. 1, fig. 1 and 2, efficiency of the algorithms can be increased by optimal synthesis of frequency filters.

3 FREQUENCY FILTER SYNTHESIS

3.1 Synthesis of superhigh-speed average FIR filters

Synthesis methods for IED filters with maximum possible speed are considered in the paper [7]. Amplitude-frequency responses of filters, that, comparing to the Fourier algorithm average filter (curve 1) at similar (curve 2) or more selective amplitude-frequency response (curve 3), do not delay signal processing, are shown in the fig. 4. The essential fault of synthesized filters is related to unrecorded high-frequency disturbance and noise gain, since amplitude-frequency response of filters out of the concerned on the fig. 4 frequency range with frequency growth will go to infinity. The algorithms of power system IED signal processing using high-speed filters can be regarded as preliminary in two-stage synthesis methods [7].

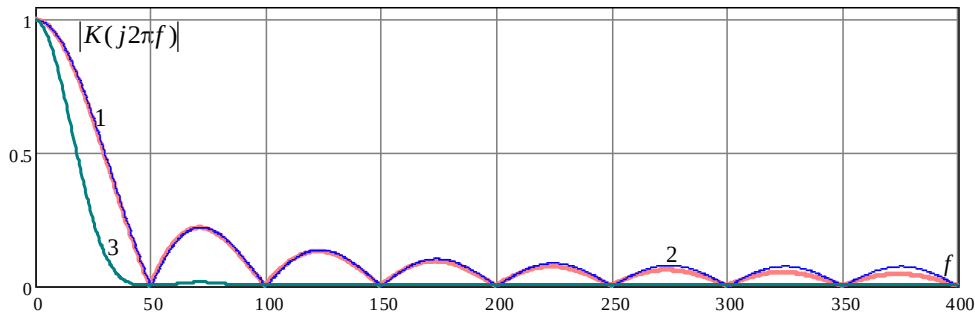


Fig. 4: Amplitude-frequency responses of high-speed filters

3.2 Robust filter synthesis

The application of spectral representations using the Laplace transform enables to efficiently solve problems of frequency filter analysis and synthesis at non-stationary input signals described by a set of semi-infinite or finite damped oscillatory components [4,5,7]. Impulse functions of analog filters-prototypes can be also specify likewise

$$\dot{g}(t) = \dot{\mathbf{G}}^T e^{\mathbf{q}t} - \dot{\mathbf{G}}^T e^{\mathbf{q}(t-T_1)}, \quad g(t) = \text{Re}(\dot{g}(t)), \quad (1)$$

To describe FIR filters, it is enough to use the vectors of impulse function complex amplitudes $\dot{\mathbf{G}} = [\dot{G}_m]_M$ and $\dot{\mathbf{G}}' = [\dot{G}_m e^{\rho_m T_1}]_M$, complex frequency vector $\mathbf{q} = [\rho_m]_M$ and filter length T_1 .

This approach significantly simplifies filter synthesis: instead of setting requirements to a

frequency characteristic, easy dependencies for filter transfer function on complex frequencies of input signal components in consideration of its changing, are needed to be done. The synthesis is carried out with growth of impulse function (1) component number till the specified quality performance of signal processing will be achieved [7]. The optimal parameters of filter impulse functions were detected by the Optimization Toolbox of the MATLAB system using specified target function minimization with limitations or methods of nonlinear optimization with limitations, as well as by specially elaborated program FFIR [7].

3.3 Average FIR filter synthesis

The impulse function (1) for average FIR filters is equivalent to a time window. Let us consider an average FIR filter synthesis for algorithms according to the tab. 1 and fig. 1.

In this case the synthesis for two tasks:

1. the filter length is about 40 ms, accuracy class is 0,2, $X_0 = 0$, simplification of the time window (1) to cosine time window at $M = 5$: $\dot{\mathbf{G}} = \dot{\mathbf{G}}^* = [\mathbf{G}_m]_M$, $\mathbf{q} = [j(m-1)\omega_1]_M$, $T_1 = 2\pi / \omega_1$;
2. the filter length is about 50 ms, accuracy class is 0,1, $X_0 = |\dot{X}_1|$, $M = 2$. The additional errors: at 45÷55 Hz frequency the deviation error is less than 0,15%, at maximum level of the exponential disturbance at $\beta_0 = 20 \div 200 \text{ c}^{-1}$ the error should not be more than 3%.

The filter synthesis results for the first task at different filter lengths are given on fig. 4 and 5. The filters 1, 2 are used for the algorithm from the tab. 1, and the filters 3÷6 – for the algorithm from the fig. 1. Comparative analysis with average FIR filters (time windows Raised cosine, Hann, Hamming, Blackman, Bartlett, Rectangular, Flat Top, Kaiser, Nutall 4-term), which are applied in one of the most developed PMU 1133A Power Sentinel made by american company Arbiter [8], shown, that synthesized filters significantly excel PMU 1133A filters in all basic quality performance of signal processing.

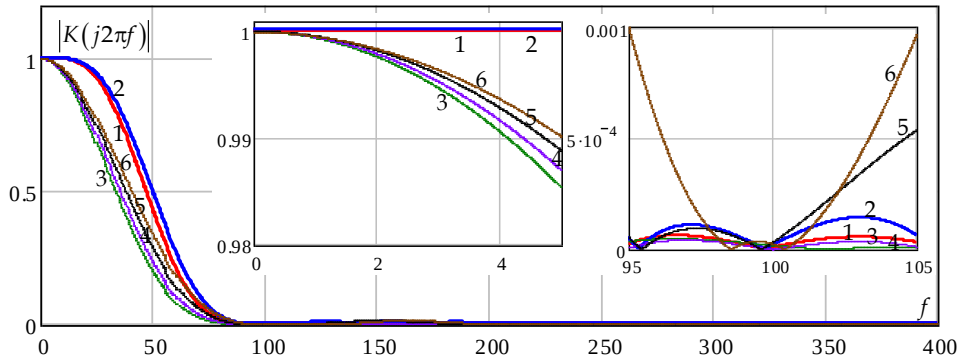


Fig. 4: Amplitude-frequency responses of average FIR filters

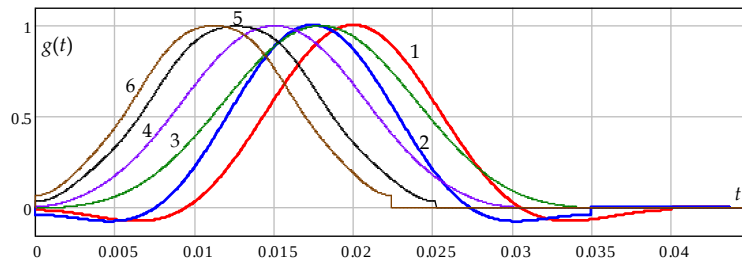


Fig. 5: Impulse functions (time windows) of average FIR filters

Using different target functions for the second task, two average FIR filters with practically identical amplitude-frequency responses in section $p = j2\pi f$ were obtained:

$$\dot{\mathbf{G}} = \begin{bmatrix} 80,48e^{j4,232} & 37,93e^{j0,5887} \end{bmatrix}^T, \quad \mathbf{q} = \begin{bmatrix} -22,99 + j62,30 & -23,26 + j186,9 \end{bmatrix}^T, \quad T_1 = 0,051\text{c};$$

$$\dot{\mathbf{G}} = \begin{bmatrix} 42,26e^{j6,024} & 38,36e^{j2,938} \end{bmatrix}^T, \quad \mathbf{q} = \begin{bmatrix} -4,668 + j42,69 & -23,28 + j178,7 \end{bmatrix}^T, \quad T_1 = 0,050 \text{ c.} \quad (2)$$

Amplitude-frequency responses of filters in sections $p = j2\pi f$ and $p = -\gamma + j\omega_0$ (curves 1 and 2) are given in fig.s 6 and 7 a). Filter impulse characteristics (fig. 7 b), comparing to the ones considered before (fig. 5), are asymmetric. Therefore, filters with vertically mirrored impulse characteristics (curves 3 and 4) will have the same amplitude-frequency responses in section $p = j2\pi f$. However, amplitude-frequency responses of filters 3 and 4 in the section $p = -\gamma + j\omega_0$ significantly differ from analogous amplitude-frequency responses of the parents – filters (filters 1 and 2). This implies the important conclusion about inefficiency of traditional filter amplitude-frequency responses usage (section $p = j2\pi f$) for filter analysis at nonperiodic input signals.

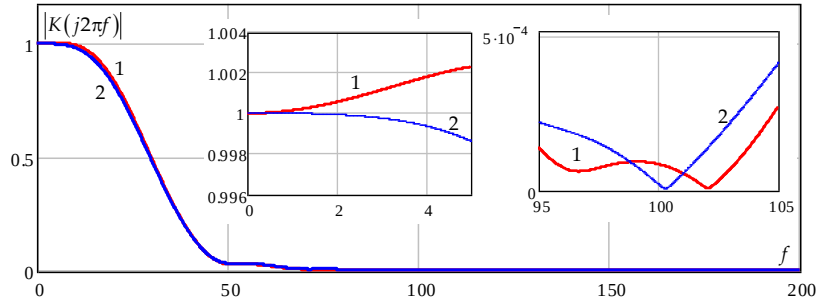


Fig. 6: Filter amplitude-frequency responses in section $p = j2\pi f$

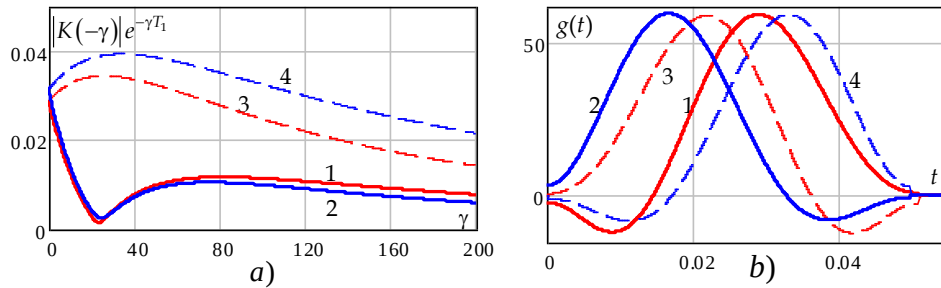


Fig. 7: Amplitude-frequency responses in section $p = -\gamma + j\omega_0$ and filter impulse responses

Shared filter 1 and 2 operation allows to increase accuracy of IED control by transient detection in the filters [4]. The filters ensure combination of high speed and high precision of signal processing. The speed of the second filter is about 20 ms (accuracy class 10) for relay protection and about 40 ms (accuracy class 0,2) for measurement of power system mode parameters.

The considered examples of filter synthesis shown the following important regularities of time windows for average FIR filters [4]:

1. in case of using cosine time windows or time windows (1) at harmonic input signals, the shape of synthesized windows is close to simmetrical bell-shaped or to the form of a “hat” (fig. 5);
2. using time windows (1) in case of need to suppress the aperiodic component, the windows with clearly defined asymmetric forms were obtained (fig. 7).

3.4 FIR filter synthesis

The FIR filter synthesis for the algorithm in the fig. 3 a) can be implemented on the basis of the achieved solutions for average FIR filters of the algorithm from the tab. 1: $\dot{g}_1(t) = 2e^{j\omega_0 t} g_a(t)$. In that case two filters with the impulse functions $g_1(t)$ and $g_2(t)$, $\dot{g}(t) = g_1(t) + jg_2(t)$, will be obtained.

Let us consider the transfer from the average FIR filter in the fig. 7 b (curve 2) to two FIR filters of the algorithm in the fig. 3 a). The amplitude-frequency characteristics of synthesized filters (curves 1 and 2) and amplitude-frequency characteristics of average filter-prototype (curve 3), are shown in the fig. 8. The filter phase-frequency characteristics are given in the fig. 9 a). The impulse characteristics of filters and an average filter-prototype (curves 3 and 4) are presented in the fig. 9 b).

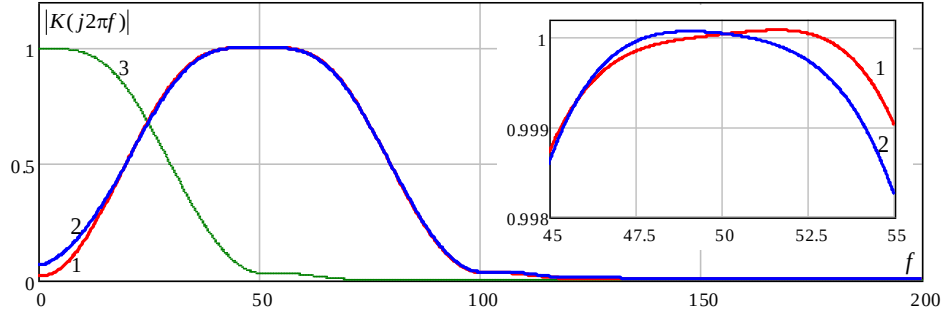


Fig. 8: Amplitude-frequency responses of FIR filters

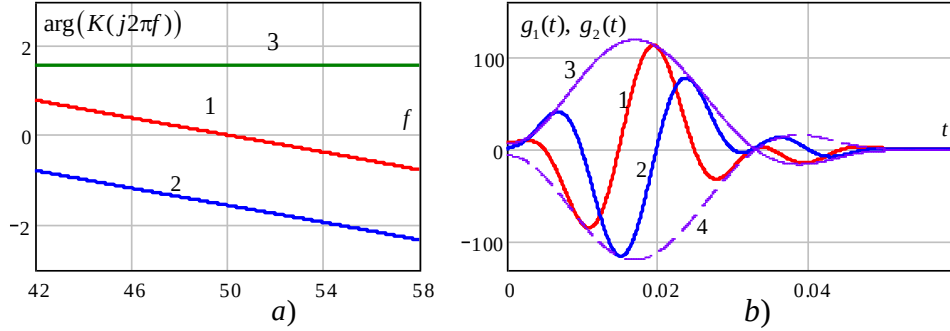


Fig. 9: Phase-frequency responses and impulse responses of FIR filters

As follows from the fig. 8, amplitude-frequency responses of the filters are very close to each other, and the difference of the filter phase-frequency response values over the frequency range 45÷55 Hz is practically constant and close to $\pi/2$ (curve 3 on the fig. 9 a). Thereby, synthesized filters assure high quality of signal processing at the specified range of power system frequency changing, in case of higher harmonics and aperiodic disturbances. Synthesized filters are applicable for the relay protection devices, because they ensure speed less than 30 ms (accuracy class 10). This is much better than most of the relay protection devices 6-35 kV have, made by Russian and foreign producers. The application of synthesized filters is reasonable also for current (voltage) symmetric component shaping.

In the case of lowering of the requirements to signal processing accuracy, filters, assuring higher speed, can be synthesized. For instance, the synthesis of three FIR filters [7] is carried out for the algorithm in the fig. 3 a).

Amplitude-frequency responses of the synthesized filters (curves 1, 2 and 3) are shown in the fig. 10 a), and corresponding to them impulse responses – in the fig. 10 b).

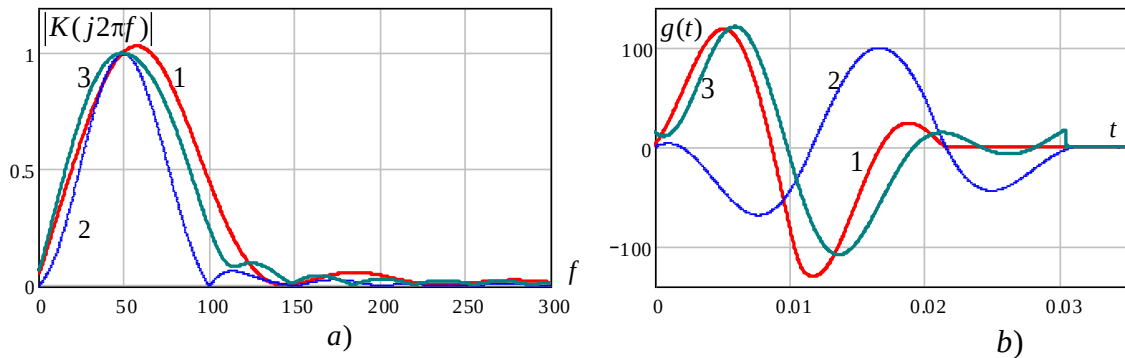


Fig. 10: Amplitude-frequency responses and impulse responses of FIR filters

The first filter ensures higher speed about 10 ms at 10% of admissible error. The second filter has lower errors at power system frequency changing (about 2 %), and its speed is about 20 ms at 10% and about 31 ms at 1% error of signal processing.

4 ALGORITHM SYNTHESIS USING SPECTRUM ANALYZERS

The algorithms for power system IED signal processing, that ensure instantaneous spectrum analysis on several frequencies, are considered in the paper [7]. The adaptive algorithm according to the fig. 11 was designed for the phasor measurement realization.

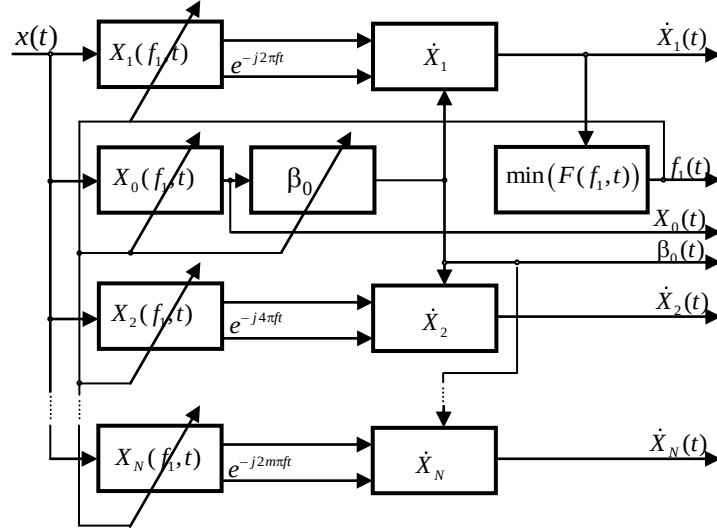


Fig. 11: IED algorithm on the basis of vector measurement technology

The following designations are used in the block scheme of the fig. 11 : $n=1..N$ – is the quantity of accountab harmonics; $\dot{X}_n(t) = X'_n(f_1, t)$ – is the amplitude estimation of the n -th harmonic on the corrected value of the instantaneous spectrum on frequency $2\pi f_1$, at the observation period of $1/f_1$. The result of the measurements is a set of estimations of the harmonic complex amplitudes $\dot{\mathbf{X}}(t) = [\dot{X}_1(t) \ \dot{X}_2(t) \ \dots \ \dot{X}_N(t)]^T$.

In that case complex amplitude of the n -th harmonic is computed on the basis of the corrected value of instantaneous spectrum on the frequency of the specified harmonic

$$\dot{X}_n(t) = 2nf_1 X_n(f_1, t) - \frac{2nf_1 \beta_0}{\beta_0 + j2\pi n f_1} X_0(f_1, t) e^{-j2\pi n f_1 t},$$

The measurements of exponential disturbance damping coefficient $\beta_0 \rightarrow \beta_0(t)$ and power system frequency $f_1 \rightarrow f_1(t)$ are taken by using the specially designed algorithms [7]. The algorithm speed is about 20 ms for accurate measurements.

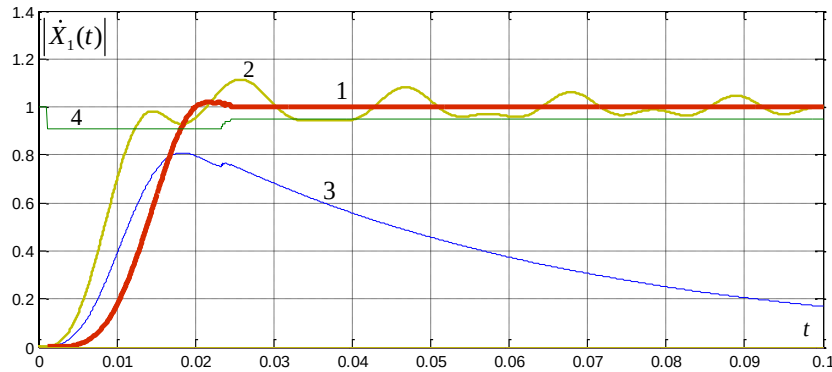


Fig. 12: Output signal diagram

The diagrams for output signal modules $|\dot{X}_1(t)|$ (curve 1), $2f_1|X_1(f_1, t)|$ (curve 2), $X_0(t)$, as well as the diagram $f_1(t)/50$ (curve 4) at input signal as additive mixture of useful signal and

exponential disturbance at $\omega_1 = 2\pi 47,5$ rad/s, $X_2 = X_{m1} = 1$, $\beta_2 = 20$ c⁻¹, are given in the fig. 12.

The application of spectrum analyzers in complex frequency coordinates enables even more possibilities for algorithm synthesis of power system IED signal processing [7].

5 IED FAST ALGORITHM SYNTHESIS

The advantage of using analog filters-prototypes with finite impulse response (1) is related to FIR filter fast algorithm synthesis. The transfer from an analog filter-prototype description to a digital filter is performed by the known methods in consideration of the analog FIR filter specific [7].

In case of using the invariant impulse response method for signaling processor with operation support over the complex numbers with fixed comma (fourth decimal digit) on the basis of analog filter-prototype (2) at sampling increment $T = 500$ microseconds, the following filter fast algorithm was obtained [4]

$$\begin{bmatrix} \dot{y}_1(k) \\ \dot{y}_2(k) \end{bmatrix} = \begin{bmatrix} -0,0171 - j0,0364 \\ 0,0158 + j0,0105 \end{bmatrix} x(k) - \begin{bmatrix} 0,0049 + j0,0115 \\ -0,0045 - j0,0037 \end{bmatrix} x(k-102) + \begin{bmatrix} (0,9881 + j0,0308)\dot{y}_1(k-1) \\ (0,9841 + j0,0922)\dot{y}_2(k-1) \end{bmatrix}.$$

The issues about fast algorithm synthesis for digital filters with integer coefficients, fast algorithms of adaptive filters, fast algorithms of spectrum analyzers and wavelet transforms are examined in the papers [3,4,7].

6 CONCLUSION

The application of the methods offered above allows to get efficient IED algorithms using phasor measurement technology. The designed algorithms of signal processing were realized in IEDs of the ENIP family, produced by JSC "Engineering Centre "Energoservice". ENIP-2 has up to five digital interfaces, as well as up to two Ethernet ports with the possibility of independent and flexible setting for every port and as well as the possibility of remote readout and the changing ENIP-2 configuration. Record supporting of IEC 60870-5-104 and IEC 61850 is provided. OPC UA technology supporting works are carried out. Experimental models of intelligent devices ENIP-3 with phasor measurement technology support were built.

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