ESTIMATION OF POWER-LINE FREQUENCY WITH LOW UNCERTAINTY FOR HIGH HARMONIC DISTORTION OF SIGNALS

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Abstract – *The paper compares four methods (zero-crossing with pre-filtration, integrated zero crossing, windowed and interpolated FFT, and dynamic parameter estimation with pre-filtration) of instantaneous power-line frequency estimation from the point-of-view of their sensitivity to higher-order harmonic content and to the additive noise and from the point-of-view of their response time. Comparison is based on computer simulations.*

Keywords –Power-line frequency measurement, interpolated FFT, zero-crossing method, integrated zero-crossing method

1. INTRODUCTION

Knowledge of the (instantaneous) power system frequency f_{inst} is important for control of energy generation/consumption balance in both global and local power nets and for achieving highest values of the series-mode rejection ratio in integrating laboratory digital voltmeters. Its knowledge is very useful also for measurement of basic parameters of digitized periodic signals (RMS value, rectified mean value, active power) if measurement should be performed using low numbers of signal periods. The estimation of the instantaneous power-line frequency is therefore investigated in many laboratories (see e.g. [1] – [12]). Methods used are for example classical zero-crossing method and its modifications [1], dynamic parameter estimation method [2], [3], method of orthogonal filtration [4] – [6], STFT [7], DFT in combination with Prony method [8], and other methods (e.g. [9] and [10]). Overviews of methods usable not only for power-line frequency is presented in [11], eighth basic methods are compared in [12].

Real-time measurement is important in some applications (especially for load shedding in maintaining the energy balance in power nets). Using the shortest possible series' of samples and DSP-chips algorithms implementation is of much importance here. Using short measuring times implies higher sensitivity of these methods to additive noise.

Massive use of nonlinear and switched loads and control circuits leads to non-sinusoidal waveforms of currents and voltages. Compatible levels of higher-order harmonic components up to 50^{th} harmonic are given in international standards for various types of systems, and these levels might be surprisingly high (e.g. [15] for THD = 8 %).

It is therefore important to know the sensitivity of individual methods to harmonic distortion and also to additive noise. Apart from external noise there is always the quantization noise of the ADC used (usually 12-bit ADC on common DAQ PC plug-in boards and 16-bit ADC on boards with DSP chips). Immunity of individual methods to both harmonic distortion and additive noise can be increased by pre-filtration (and sometimes also post-filtration – [5] , [6]) of the digitized signals.

2. SELECTED METHODS OF FREQUENCY ESTIMATION

The aim of this paper is to verify the achievable accuracy of estimation of f_{inst} using three methods which were found to be the most accurate from the eight compared in [12], namely *classical zero crossing* (ZCR), *integrated zero-crossing* (IZC) [1] and *FFT windowed and interpolated in frequency domain* (IFFT)[13], [14]. Pre-filtration is used before application of the ZCR algorithm. We compare also method of *dynamic parameter estimation* (DPE) [2], [3] as a representative of methods allowing finding f_{sig} in time shorter than signal period. Pre-filtration is used also with the DPE method. All methods were investigated in frequency band 48 Hz to 52 Hz, surpassing the maximum possible frequency deviation in many national global power nets. Since rates of change of frequency are in global power systems usually below 1 Hz/s (as a consequence of high inertia of large electromechanical generators), signals can be considered to be quasi-stationary and measured frequency can be supposed constant during the data window. Other methods must be used for rapidly changing frequencies (e.g. STFT as in [7] or various time-frequency distributions).

The *ZCR method* is well known - time interval between two consecutive crossings of zero level with the same sign of the derivative is taken as signal period time. Accuracy increases with the sampling frequency increase. Linear interpolation at the end of the measuring interval was included in the algorithm. Method is insensitive to harmonic distortion unless this causes more than two zero-crossings during signal period. For SNR above 50 dB errors caused by additive noise might be unacceptably high. For large values of SNR the ZCR algorithm sometimes collapsed. Therefore we used the pre-filtration of the signal (par.3).

The IZC method [1] is the ZCR method applied on integrated signal (with subtracted mean value). Since the integration acts as a sort of a low-pass filtering, sensitivity to noise and harmonics is better than that of the ZCR without pre-filtering.

The *FFT windowed and interpolated in the frequency domain* (IFFT) [13], [14] eliminates the leakage caused by non-synchronization of signal and sampling by using data windowing and interpolation allowing to find the signal fundamental frequency even if this does not lie on one line of the FFT grid. Knowledge of the spectral window is used and some approximations [13] allow to keep the computation time low even if the complete theory behind the method is not quite trivial. Since the method is used here for finding the fundamental frequency only, harmonic distortion plays no role here. Because of the Hann window used by us, at least three (better four) signal periods should be sampled. This means that the method has slower response that the other compared, but is also the least sensitive to the noise.

The *dynamic parameter estimation method* (DPE method) [2], [3] uses as the model of measured signal a sinusoid with three parameters – amplitude, frequency and phase – of which only the frequency is relevant here. It is therefore similar to one method used nowadays extensively for the ADC testing. Parameter values are found iteratively by minimizing the total square error between the signal samples and outputs of the algorithm. A formula for initial parameter guess is given in [2] securing algorithm convergence in three steps. Low number of samples used for one frequency estimate (thirteen or even four) assures fast response. Sensitivity of the DPE to noise and harmonic distortion can be reduced using pre-filtering.

3. PRELIMINARY SIGNAL FILTRATION

Uncertainty [16] of frequency estimation can be reduced in most cases by signal pre-filtration. Pre-filtration reduces influence of both additive noise and harmonic distortion. Additional filter increases response time of the algorithm. Suitable types of filters are low-pass (LP) and band-pass (BP), the pass-band of which includes suitable frequency band around power-line nominal frequency. In this filter application the filter gain at 50 Hz need not be one, quantity of interest is output signal *frequency* of the filter.

We have examined several LP and BP filters, both IIR and FIR type. The LPs chosen for experiments were Chebyschev 1st type, N=3 (Fig.1) or N=4,passband ripple 0.1 dB, f_C=53Hz and *FIR low-pass,* N = f_S/f_{SIG}, f_C=55 Hz, (Fig.1) designed by windowing method (group delay of this linear-phase FIR filter is 10 ms). BPs examined were Chebyschev1, pass-band ripple 0.8 dB, N=4, 40-60 Hz, and Cauer (elliptic), N=3, pass-band ripple 0.8 dB, stop-band attenuation 50 dB, 45-55 Hz.

Amplitude frequency response, input and output signals, and dependence of filter output signal frequency on time was studied for each filter. *Band-pass filters* were found unsuitable because of their long transient times (output frequency settled to nominal frequency with error 0.01 Hz after about 600 ms). FIR *LP filter* output frequency settles to 0.01Hz in 30 ms regardless of the sampling frequency (Fig.2 and Fig.3). LP Chebyschev1, 3rd order, settled to 0.01 Hz in 45 ms for f_S=8 kHz (Fig.2 and Fig.3) and in 57 ms for f_S=800 Hz, 4th order settled in about 80 ms. So the FIR LP filter of the order depending on the sampling frequency (N=f_S/*50*) was chosen for signal pre-filtration in connection with classical zero-crossing (ZCR) method and with the dynamic parameter estimation (DPE) method.

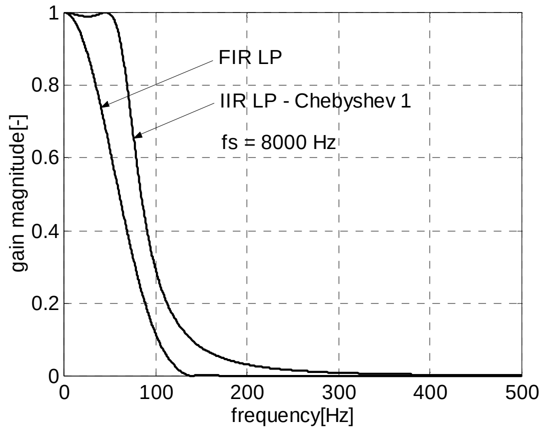

Fig.1 Examples of LP filters - amplitude frequency responses.

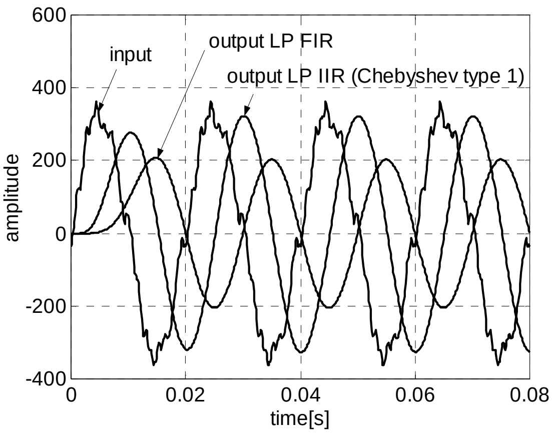

Fig.2 Waveforms of filters input and output signals.

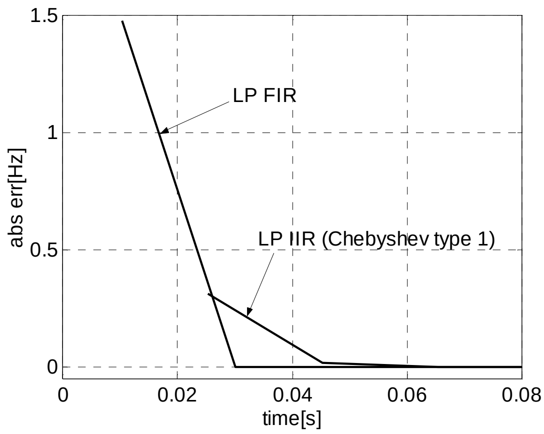

Fig.3 Settling times of filter signals output frequencies.

4. SIMULATION RESULTS

Simulations were performed in MATLAB environment for frequencies 48 to 52 Hz, SNR values from 40 dB to 70 dB and sampling frequencies 800Hz, 1600 Hz, 3200 Hz, 8000 Hz and 12800 Hz (i.e. for numbers of samples per period 16, 32, 64, 160, and 256).

.ZCR method with pre-filtration is referred to as *ZCRF* and the DPE method with pre-filtration is referred to as *DPEF* further in the text.

Other filters then the FIR DP described above were examined in ZCRF and DPEF methods only for sampling frequency 8000 Hz.

Numbers of harmonic components of signal from [15] were included in simulations so that the sampling theorem was not violated. RMS values of the harmonic components (in percent of fundamental) corresponding to [15] are shown in Table 1. Since no information about phase shifts of harmonic components is given phase-shifts were chosen to be uniformly distributed random numbers in the interval $-\pi$, π , which is in agreement with recommendations from [16].

Table 1 – RMS values of harmonic components in percent of fundamental (compatibility levels given in [15], THD=8%)

Harmonic	Level (%)	Harmonic	Level (%)
1st	100	26th	0.2
2nd	2	27th	0.63
3rd	5	28th	0.2
4th	1	29th	0.6
5th	6	30th	0.2
6th	0.5	31st	0.2
7th	5	32nd	0.2
8th	0.5	33rd	0.56
9th	1.5	34th	0.2
10th	0.5	35th	0.54
11th	3.5	36th	0.2
12th	0.2	37th	0.2
13th	3	38th	0.2
14th	0.2	39th	0.49
15th	0.3	40th	0.2
16th	0.2	41st	0.2
17th	2	42nd	0.2
18th	0.2	43rd	0.47
19th	1.5	44th	0.2
20th	0.2	45th	0.46
21tst	0.2	46th	0.2
22nd	0.2	47th	0.2
23rd	1.5	48th	0.2
24th	0.2	49th	0.2
25th	1.5	50th	0.2

It was found from simulations that pre-filtration used in connection with the integrated zero-crossing method did not help to reduce measurement uncertainty, the estimation uncertainty was even a bit higher than without (external) pre-filtration. Integration of the signal and subtracting the DC component of the integrated signal before application of the zero-crossing algorithm acts as a sort of inherent pre-filtration of this algorithm.

No pre-filtration was also used in the FFT windowed and interpolated in the frequency domain, since this method (the output of which is here only the value of the fundamental frequency) is inherently not sensitive to either the higher-order harmonic components or the additive noise. The important condition of getting the low uncertainty estimations of frequency using this method is to sample at least four signal periods. This condition is valid because we used the Hann window ("hanning") for long-range leakage reduction. For windows with broader main lobes more signal periods should be sampled.

Following set of 3D figures is an example of results for sampling frequency 8000 kHz. It shows expanded uncertainties of type A, coverage factor 2 [16] of frequency estimates gained from 100 repetitions of simulation for the four investigated methods. As can be seen from the figures, uncertainty of the estimations using IZC, ZCRF and IFFT methods depend only slightly on frequency, but increase smoothly with decrease of the SNR. Uncertainty of the DPEF method does not show any observable trend in dependence on either frequency or SNR.

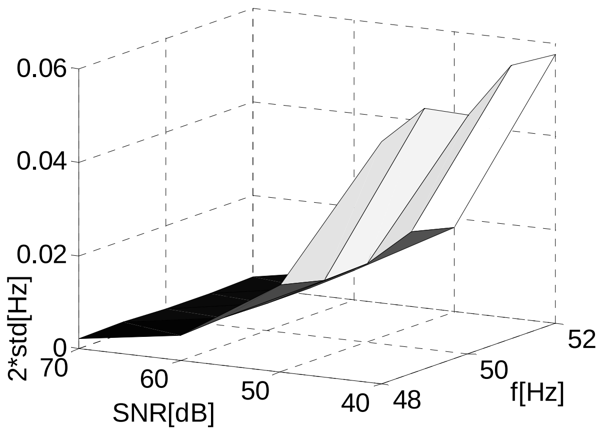

Fig.4 Type A expanded uncertainties of the IZC method

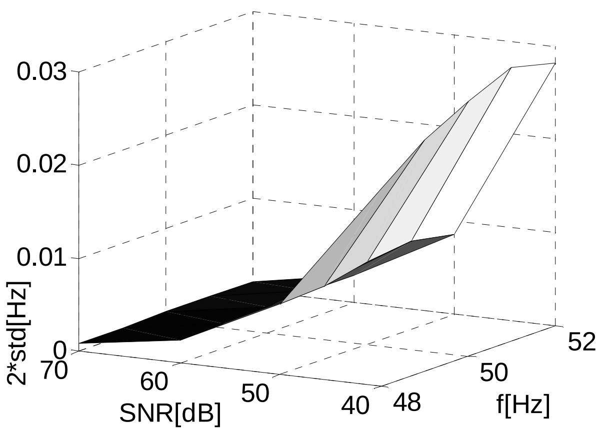

Fig.5 Type A expanded uncertainties of the ZCRF method

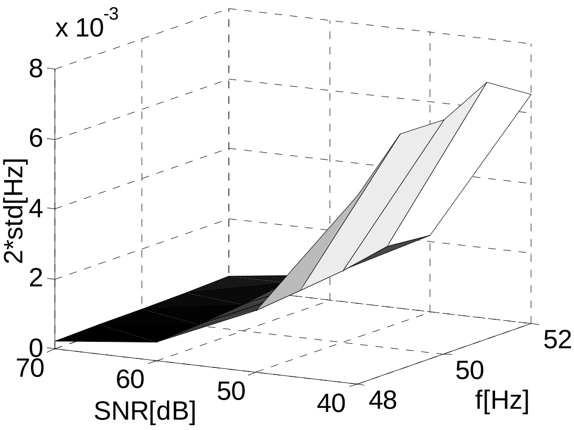

Fig.6 Type A expanded uncertainties of the IFFT method

Since numbers of processed samples (samples per period) is increasing proportionally to sampling frequency by ZCRF and IZC methods, their uncertainty decreases with increase of f_S.

Accuracy of IFFT method increases with FFT legth N. The FFT grid becomes more dense and interpolation errors are lower.

On the contrary, the uncertainty of the DPEF method using given number of samples (in our case 13 for each estimate) increases with f_S increasing, since shorter part of the signal period is used for estimation and shorter time for noise averaging is at disposal. That is why the uncertainty of the DPEF method is relatively high in Fig.7 (f_S=8000 Hz) – much better results are achieved for lower f_S, as can be seen from Table 2.

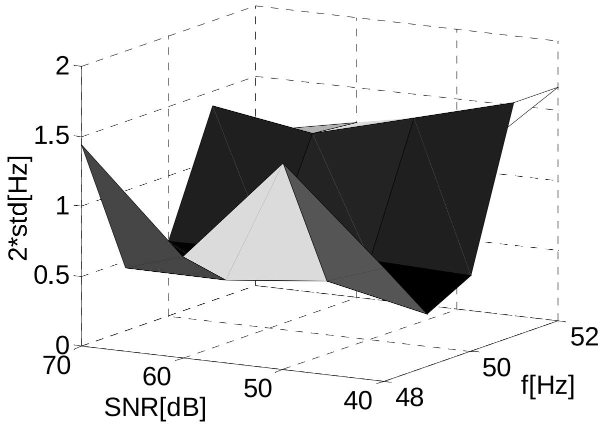

Fig.7 Type A expanded uncertainties of the DPEF method

Table 2 - Ranges of type A expanded uncertainties (coverage factor 2) in Hz of the four investigated methods for different f_S and different SNR. Numbers of samples are chosen to cover four periods of the signal. Pre-filtration in ZCRF and DPEF with FIR LP (Fig. 1) with group delay 10 ms. N is total number of samples

	SNR	70 dB	60 dB	50 dB	40 dB
fs = 800 Hz	*IZC*	0.01-0.11 Hz	0.02-0.12	0.06-0.13	0.13-0.22
N = 64	*ZCRF*	0.005	0.01	0.025-0.03	0.07-0.09
16 Samples/period	*IFFT*	0.003-0.005	0.004-0.006	0.009-0.011	0.034-0.035
	DPEF	0.14-0.18	0.14-0.16	0.14-0.2	0.22-0.25
fs = 1600 Hz	*IZC*	0.01-0.03	0.02-0.035	0.04-0.05	0.12-0.14
N = 128	*ZCRF*	0.005	0.007	0.018-0.02	0.055-0.07
32 Samples/period	*IFFT*	0.002-0.004	0.003-0.005	0.008-0.009	0.023-0.027
	DPEF	0.3-4.5	0.3-0.6	0.3-0.6	0.3-0.6
fs = 3200 Hz	*IZC*	0.005-0.001	0.01-0.015	0.025-0.035	0.08-0.1
N = 256	*ZCRF*	0.0025	0.005	0.013-0.016	0.037-0.042
64 Samples/period	*IFFT*	0.001-0.004	0.002-0.004	0.004-0.006	0.017-0.018
	DPEF	0.40 - 01.3			
fs = 8000 Hz	*IZC*	0.002	0.006	0.019	0.055
N = 1024	*ZCRF*	0.001	0.003	0.008	0.025
160 Sa/period	*IFFT*	0.0003	0.0007	0.002	0.006
	DPEF	0.5 –1.5			
fs = 800 Hz	*IZC*	0.002	0.005	0.015	0.04-0.048
N = 1024	*ZCRF*	0.001	0.0025	0.007	0.022
256 Sa/period	*IFFT*	0.0003-0.004	0.001-0.004	0.003-0.004	0.008-0.009
	DPEF	0.7-3.2	0.8-3.1	0.5-2.5	0.5-2.2

Table 3 - Type A expanded uncertainties (coverage factor 2) in Hz of frequency estimation for two different filters used for signal pre-filtration (detailed filters parameters see text above)

	SNR	70 dB	60 dB	50 dB	40 dB
FIR LP filter (Fig.1)	*ZCRF*	0.001	0.003	0.008	0.025
	DPEF	0.5 –1.5			
IIR LP filter (Chebyshev 1)	*ZCRF*	0.001	0.0025	0.0065	0.018-0.022
	DPEF	1 - 7			

5. CONCLUSIONS

The classical zero-crossing method with pre-filtration behaved surprisingly well, its errors being about two-times lower than errors of the integrated zero-crossing method. Without pre-filtration there were comparatively many estimations with excessively large uncertainties (caused by additional zero-crossings due to noise and higher-order harmonic components), so results for this method are not given here.

Dynamic parameter estimation method has even after pre-filtration the largest measurement uncertainty of all the four investigated methods. Lower values of errors of it were reported in [3], but we were not able to reach them in our experiments. The cause may be that in [3] no noise was added to the measured signal. This method is on the other hand able to find the frequency estimate in the much shorter time than the other three methods. It needs only about 20 samples for both initial guess of parameters and the input set of samples for one estimate. Its computation algorithm is more complicated than algorithms of other investigated methods. As can be seen from Table 2, its estimation uncertainty increases with sampling frequency. It is caused by the fact that the model parameters are found from shorter part of the signal period and the additive noise is worse filtered out in shorter time interval used for signal sampling in case of higher sampling frequency used.

The zero-crossing method with pre-filtration and the integrated zero-crossing method need at least 30ms for signal sampling plus time needed for computation. The windowed and interpolated FFT method using Hann window needs at least 60 ms signal sampling (three signal periods) plus time needed for computation. For shorter signal sampling time main lobes of the Hann spectral windows overlapp and the errors of instantaneous frequency estimate become considerably larger.

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