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FAST FREQUENCY TRACKING

A method of periodical signal frequency tracking by the frequency-locked loops is proposed. Increasing of frequency adjustment accuracy is achieved by using of a new fast frequency discriminator, based on estimates of an instantaneous frequency. Reasonability of an input signal pre-filtering in case of nonlinear distortions, harmonics interferences and strong noise is proved.

Keywords: FLL, speed, frequency estimation, interference, adaptive filter, open loop.

Periodic signals processing is an important part of electronic support measures (**ESM**) technologies on which a variety of different modern technical systems are based. Thus, the problem of frequency synchronization in telecommunication systems is still relevant [1]. The same applies to panoramic receivers, the main feature of which is fast detection of signals with a priori unknown parameters [2]. Coherent processing algorithms that are used in the receiver require measurements of phase and frequency of weak radio signals in the presence of noise. Signals in Global Satellite Navigation Systems besides have a large frequency (Doppler) uncertainty at the receiver that is consequence of the high relative satellite-to-receiver velocity [3]. The specificity of power systems of grid-connected converter type is high-precision frequency tuning to known nominal value and ensuring phase synchronization [4—5].

In the most of above-mentioned systems, the signal (x) can be considered as a single-tone (s) with additive white Gaussian noise (η):

$$x_i = s_i + \eta_i = \rho \sin\left(\sum_{i=0}^{j} \omega_i \tau + \varphi_0\right) + \eta_i,\ i = \overline{1, j},$$

where j — the current sample index;
ρ — the amplitude;
φ_0 — the initial phase;
ω — the unknown frequency that can vary in time;
τ — the sampling period.

The measurement of these parameters is considered in [6] the main point of which is an algorithm of instantaneous frequency estimation. It was this work which became the basis of the present study, where we solve the problem of improving performance of harmonic signal synchronization systems. We propose a new frequency tracking method based on estimation of instantaneous frequencies and fast frequency-locked loop (**FLL**) system with a new fast frequency discriminator (**FD**) and an open loop to enhance the frequency tracking with nonlinear element in the closed loop. We also propose to implement the input signal pre-filtering using an adaptive low-pass filter (**ALPF**).

At first we consider general principles of frequency tracking with the use of phase-locked loops (**PLL**) and frequency-locked loops. Then we provide detailed description of the proposed system and its structural elements. System effectiveness is researched by computer simulations and analysis of frequency tracking transient processes for different versions of the FLL. Finally, we prove it necessary to use an adaptive filter for reduction of noise, interferences and higher harmonics.

Basic methodology

The conventional synchronization technique is based on the application of PLL which also provides phase synchronization of reference and generated signals. These systems generally include the three typical structural blocks: phase discriminator, control unit (**CU**) and controlled oscillator (**CO**).

The typical examples of such systems are three-phase PLL-systems [4]. Although these systems are fast and accurate under balanced conditions, they become inapplicable when the utility voltage is unbalanced. This circumstance leads to system decomposition onto three independent channels with individual parameters tracking [4]. The usage of the single-phase PLL is typical for the above-mentioned ESM-systems.

As it is said in [5], PLLs synchronize with the phase of the input signal, and hence, the accuracy and dynamical response of its estimation under transient conditions are highly influenced by phase jumps. An FLL, on the other hand, estimates the frequency of the input signal, which does not experience such sudden changes and can acquire and track signals which are at higher frequency offsets than a PLL can. A significant improvement of measurement ability in FLL is achieved by reducing the parametric dimension of the problem.

The general approach to designing the FLL is to adjust the output signal frequency to the reference signal frequency, which may be constant or changed by an unknown law. It is similar to the PLL, but a phase discriminator is replaced by FD (see **Fig. 1**).

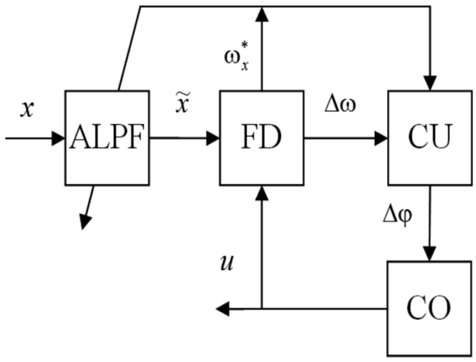

Fig. 1. Proposed fast FLL structure

There is also an open loop of an instantaneous frequency ω_x^* estimation of the reference signal besides the closed loop and the ALPF of the reference signal. The digital harmonic output signal (u) with the desired frequency is generated by the CO, which is schematically shown in **Fig. 2**.

In order to approximate this model to real technical systems, it is considered that the dependence of CO on the control signal adjusting characteristic $\Delta\varphi_j$ is nonlinear and generally can be represented by functional transformation

$$\Delta\tilde{\varphi}_j = G(\Delta\varphi_j),$$

where $\Delta\tilde{\varphi}_j$ is an actual generator phase growth at j-th step.

The instantaneous frequency of the output signal equals

$$\omega_{u,j} = \Delta\tilde{\varphi}_j / \tau.$$

The current phase $\tilde{\varphi}_j$ of the output signal $u_j = \sin\tilde{\varphi}_j$ is formed in the block Φ (see Fig. 2) as the sum of all phase growths between the adjacent samples

$$\tilde{\varphi}_j = \sum_{i=1}^{j} \Delta\tilde{\varphi}_i.$$

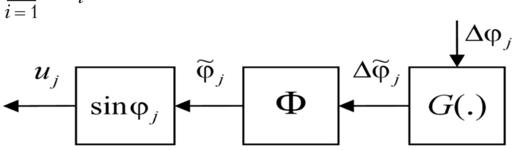

Fig. 2. Controlled oscillator block diagram

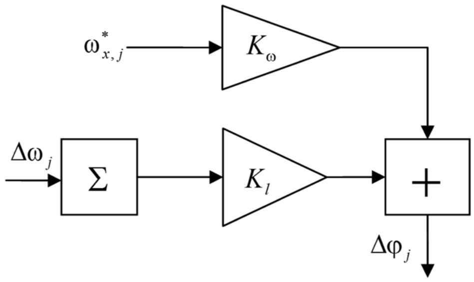

Fig. 3. Control unit structure

The CU considered in the paper (**Fig. 3**) is the simplest first order unit, which provides astatism by frequency. The corresponding mathematical model of the CU can be written as

$$\Delta\varphi_j = K_l \sum_{i=1}^{j} \Delta\omega_i + K_\omega \omega_{x,j}^*,$$

where K_l, K_ω — gains of close and open control loops, respectively;
$\Delta\omega_j$ — the difference between instantaneous frequency estimates of the reference $\omega_{x,j}^*$ and output $\omega_{u,j}^*$ signals.

Fast frequency discriminator

Frequency tracking speed in the FLL system is largely determined by the inertia of the FD. Usually the FD includes a mixer (multiplier) of two signals connected in series with a low pass filter [7] without direct frequency estimation. The fundamental need for a filter to isolate low-frequency component leads to considerable inertia of a closed loop control. A transient process may exceed approximately ten cycles of the harmonic signal. Construction of the FD by zero-crossing digital method also reduces adjustment time because data appearance tempo is only half of the signal period.

The problem of FLL performance improvement is solved in this study by using a new FD, the block diagram of which is shown in **Fig. 4**. Instantaneous frequency estimates of reference and output signal are obtained independently in fast frequency estimation (f_FE) blocks. It also allows us to use the value of the instantaneous frequency estimation of the reference signal in the open loop control. Another considerable feature is absence of a filter in opposite to the classical FD.

Digital instantaneous frequency measurement receivers have been used for wideband monitoring of radar environments in naval, airborne and

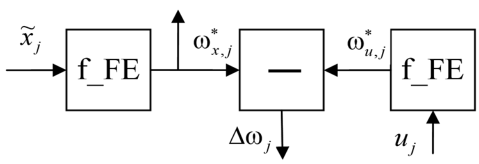

Fig. 4. The structure of the proposed fast frequency discriminator

ground-based ESM-systems all over the world for over 50 years [8]. There are a lot of researches on algorithms improvement at present time, but they usually provide sufficient noise immunity only on condition of significant observation interval and can be based, for example, on Fourier and Hilbert transforms. It is necessary to use algorithms that can work with a short sample of signal, in particular, the one developed by authors of [6, 9].

These algorithms are based on an auto regression model of sine wave:

$$s_n = \alpha s_{n-1} - s_{n-2},\ n = \overline{3, M},\ \alpha = 2\cos(\gamma),$$

where γ is a phase shift between adjacent samples of the signal. This phase shift is named normalized frequency.

Auto regression model allows building the phase shift estimate:

$$\gamma^*_{1(2)} = \arccos\left(\left(B(\overline{x}) \pm \sqrt{B(\overline{x})^2 + 2}\right)/2\right),$$

where $B(\overline{x})$ is calculated in M-size running window as

$$B(\overline{x}) = 0.5\frac{\sum\limits_{k=j-M+1}^{j}[(x_{k+1}+x_{k-1})^2 - 2x_k^2]}{\sum\limits_{k=j-M+1}^{j}[x_k(x_{k+1}+x_{k-1})]}.$$

The next step is to select the value of γ^* located in the zone of the method unambiguity $(0, \pi/2)$. And finally, the real frequency is calculated as

$$f^*_s = \gamma^*/(2\pi\tau).$$

The single instantaneous frequency value at a certain point j of discrete time is calculated by f_FE algorithm on the basis of several (M in number) previous consecutive signal samples (the so-called “window”). The size of this window must be at least 4 samples. Larger window sizes in real systems increase stability in noise conditions.

For the current time j window models of reference and output signals, which are processed parallel in two f_FE blocks, can be written in a vector form:

$$\mathbf{X}_j = \{x_{j-M+1}, x_{j-M+2}, \ldots, x_j\},$$

$$\mathbf{U}_j = \{u_{j-M+1}, u_{j-M+2}, \ldots, u_j\}.$$

A model of the fast harmonic signal frequency discriminator can be written as

$$\Delta\omega^*_j = \omega^*_{x,j} - \omega^*_{u,j},\ \omega^*_{x,j} = \mathrm{E}_\omega(\mathbf{X}_j),\ \omega^*_{u,j} = \mathrm{E}_\omega(\mathbf{U}_j).$$

Thus, sequential evaluation of the input process instantaneous frequencies is performed by the f_FE in running window mode step by step for each point in discrete time.

The appearance of the proposed fast FD leads to a necessity of carrying out a specific research on the influence of open loop and adaptive filtration on effectiveness of frequency adjustment. Quality of the frequency adjustment is determined by the stability and duration of the transition process and steady-state error.

Because the fast FD is a nonlinear element, the behavior of the frequency closed loop control cannot be accurately described in the framework of classical control theory. Therefore, initial research of the f_FLL, as the new system, is implemented by computer simulation.

Open loop

First of all, it is necessary to point out that in case of linear adjusting characteristic ($K_\omega \cdot G(\Delta\varphi_j) \equiv 1$), only the open loop is enough to carry out the frequency adjustment in an FLL-system. Thus, close loop becomes unnecessary. So from now on we shall consider the nonlinear adjusting characteristic. As an example, we have chosen the following expression:

$$\Delta\widetilde{\varphi}_j = (\Delta\varphi_j)^{3/4},$$

and the following general conditions for computer simulations: the reference signal frequency range is 25—400% of the nominal value of 1 MHz; sampling frequency 50 MHz (τ = 20 μs); running window size M=50, which corresponds to 1 cycle of the nominal signal.

Fig. 5 demonstrates acceleration of the transient process by the open loop when the closed loop gain is invariable K_I = 0.012. It should be noted, that this coefficient is almost proportional to the τ value.

The maximum value of the open loop gain K_ω=0.9, with which the best result was obtained, is close to the stability boundary for the given frequency range (16-fold frequency variation). In different situations, the duration of the aperiodic transient process (up to 5% deviation level) is 3 to 5 signal cycles. There is a possibility to shorten this time by simultaneously decreasing the frequency range by means of increasing K_ω coefficient. It was found that nonlinearity of quadratic and square root functions leads to considerable dynamic range narrowing from the point of view of its stability.

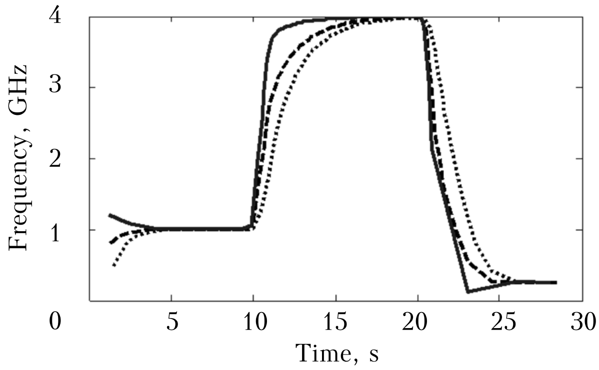

Fig. 5. Transient processes of the fast FLL with an open loop for different K_w values:
····· 0; ----- 0,5; —— 0,9

Adaptive filtering

The main disadvantage of the estimator [10] is its sensitivity to interferences (in particular higher harmonics) and noises, especially in low frequency range. It is reasonable to use preliminary filtering of input (reference) signal to reduce the influence of such factors [10—12]. It is the possibility to estimate the instantaneous frequency in the proposed fast FLL that allows the filter bandwidth adaptation to be carried out. It means that coefficients $\{a_j\}$, $\{b_j\}$ of the transfer function $H_j = H_b(\{b_j\})/H_a(\{a_j\})$ should be modified at each step j by filter synthesis laws

$$\{a_j\} = \Im_a(\omega^*_{x,j}), \quad \{b_j\} = \Im_b(\omega^*_{x,j}).$$

Hence, the bandwidth depends on the obtained frequency estimate. The adaptive filter as an element of the fast FLL is shown in **Fig. 6**.

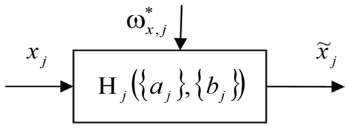

Fig. 6. Adaptive filter structure

As the preliminary research has shown, it is preferable to use the first order Butterworth filter as a low-pass IIR due to the advantages of the former in operating speed and stability. Thus, it is this filter we focus on hereafter.

The first fundamental reason to use the filter is that it allows maintaining the maximum signal-to-noise ratio **(SNR)** which can be reached because the filter cutoff frequency (f_{co}) equals the signal frequency. This can be seen from the graph of the transient processes in **Fig. 7** for SNR=10 dB. The figure clearly shows that the system is virtually inoperable with such noise level without the filter.

Using the nonadaptive filter adjusted to nominal frequency significantly reduces the frequency tracking error for the low-frequency signal, but worsens the precision for the high-frequency signal by suppressing it. The adaptive filter decreases the tracking error for the low-frequency signal even more, and significantly improves precision for the high-frequency signal. Minor loss at nominal frequency, as compared to the non-adaptive filter, is caused by instantaneous frequency fluctuations and, respectively, cut-off frequency fluctuations. It should be noted, that the presence of the filter virtually does not delay the transient process of the frequency jump.

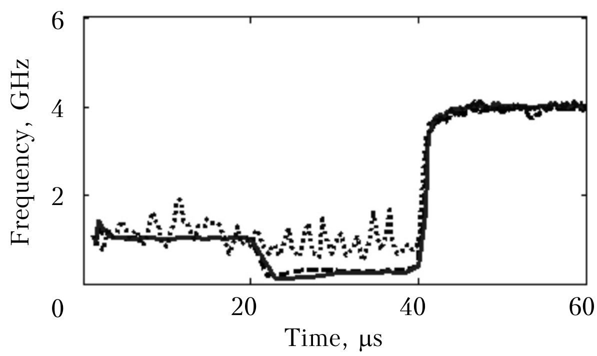

Fig. 7. Transient processes of the fast FLL with sinusoidal reference signal:
····· without filtering; ----- with a nonadaptive filter; —— with an adaptive filter

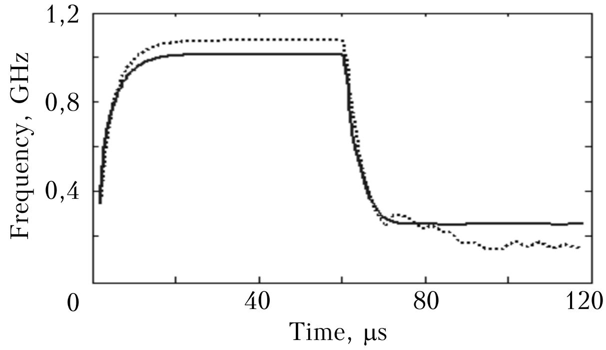

Fig. 8. Transient processes of the fast FLL with triangular reference signal:
····· without filtering; —— with an adaptive filter

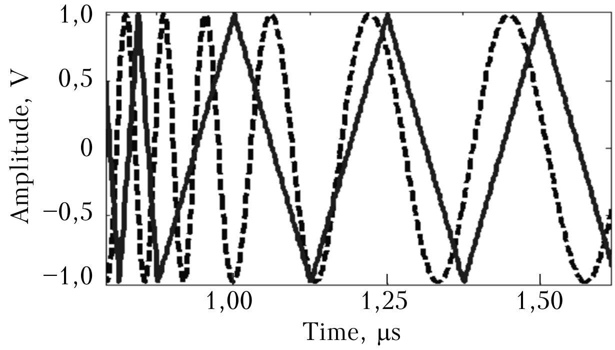

Fig. 9. Visualisation of adaptation process
····· output signal; —— reference signal

The second positive effect of the adaptive frequency filtering is the suppression of higher harmonics, which enables to perform error estimation of the main tone frequency of periodic nonsinusoidal signals. This effect is considered further, on an example of a triangular signal without noise.

Fig. 8 shows transient processes of the system with and without filter. One can see that the adaptive filter provides a sufficiently higher precision of tracking of the first harmonic frequency of the triangular signal. **Fig. 9** shows a fragment of tracking of the output sinusoidal signal to the triangular reference signal. As can be seen from the figure, the thansient process lasts no more than two cycles of lower frequency signal.

Properties of the frequency estimation algorithm with pre-filtering

Above mentioned positive features of the new FLL first of all depend on the precision of the frequency estimation of the disturbed reference signal. Therefore, the behavior of pre-filter and estimator as a pair must be analyzed in more detail for different input processes.

Frequency estimation errors in the presence of the harmonic interference

The appearance of an additional harmonic ξ with different frequency f_ξ and power P_ξ considerably decreases the estimation accuracy because the algorithm does not have any filtering properties. **Fig. 10** shows the graph of the estimation mean which depends on the frequency ratio and the signal to interference power ratio $m_f(f_\xi/f_S, P_\xi/P_S)$ when the noise is absent. Estimations randomness is caused by randomization of the signal and the interference initial phases and the standard deviation lies within 9% zone relative to the nominal frequency. The obtained surface of the estimation mean $m_f(\cdot)$ is characterized by smoothness, and one-dimensional dependencies $m_f(f_\xi/f_S)$ at P_ξ/P_S = const are characterized by high enough linearity.

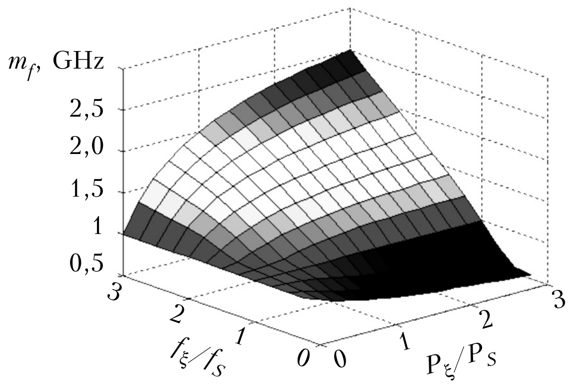

Fig. 10. Estimation mean for the harmonic interference

It was found that frequency estimations are virtually independent of the window size and the sampling frequency, if such frequency is much higher than f_S and f_ξ.

It should be noted, that in this situation there is no point in studying the pre-filtering, because it is quite enough to determine the signal to interference power ratio from amplitude-frequency characteristic of the filter and directly address the function $m_f(f_\xi/f_S, P_\xi/P_S)$.

Influence of frequency deviation on estimation precision

In the case of the locally non-stationary signal, when its actual instantaneous frequency **(IF)** is significantly varied within a single window, the estimation depends on the variation degree. For example, in the case of the linear deviation, the frequency estimation approximately equals to the medium value between the initial (f_b) and the final (f_e) frequencies of the window:

$$f^* = (f_b + f_e)/2 + \Delta f^*.$$

The character of deviation Δf^* is shown in **Fig. 11**. The charts for each window size (8, 16, 32, 64) are different because of the difference in phase distances between the samples.

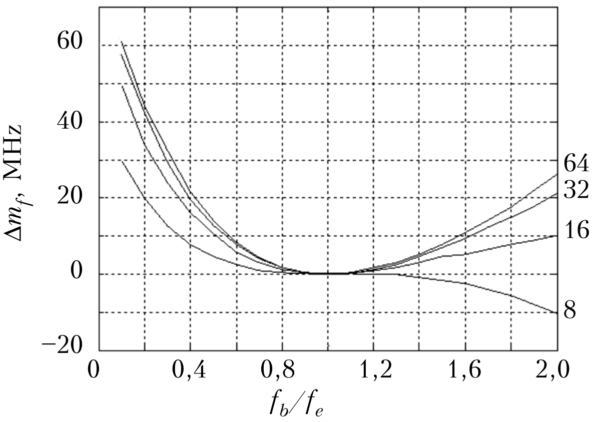

Fig. 11. Frequency estimation with deviation for different window size

Reasonability of the input signal pre-filtering

In actual practice, the correlative noise process is formed by pre-filtering before using the frequency estimation algorithm. An input parameter for research is the signal-to-noise ratio $SNR = P_s/\sigma_g^2$, where σ_g^2 is the variance of the additive white Gaussian noise.

The graph of the estimation mean **(Fig. 12)** for the 1st order low-pass filter **(LPF)** shows a sufficiently larger working area near nominal frequency in comparison to the case, when the LPF is not used.

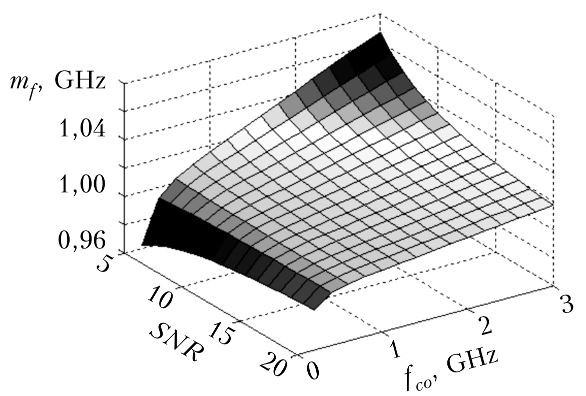

Fig. 12. The estimation mean for the 1st order preliminary LPF (f_{co} — cutoff frequency)

This can also be confirmed by the mean square error **(MSE)** of frequency estimation **(Fig. 13**, *a***)**. The surface is characterized by the reduction of the argument of the MSE function minimum, while SNR is increasing. This means that greater signal suppression by the filter is allowed.

Some decrease of the error mean can be achieved by reducing the sampling frequency and the number of samples in the window. But we must remember that reducing the number of samples generally causes the increase of the MSE.

The 2nd order LPF can be considered as more efficient in use. The MSE surface for such filter is shown in **Fig. 13**, *b*. Such MSE value, in comparison to Fig. 13, *a*, decreases 2 to 3 times at the

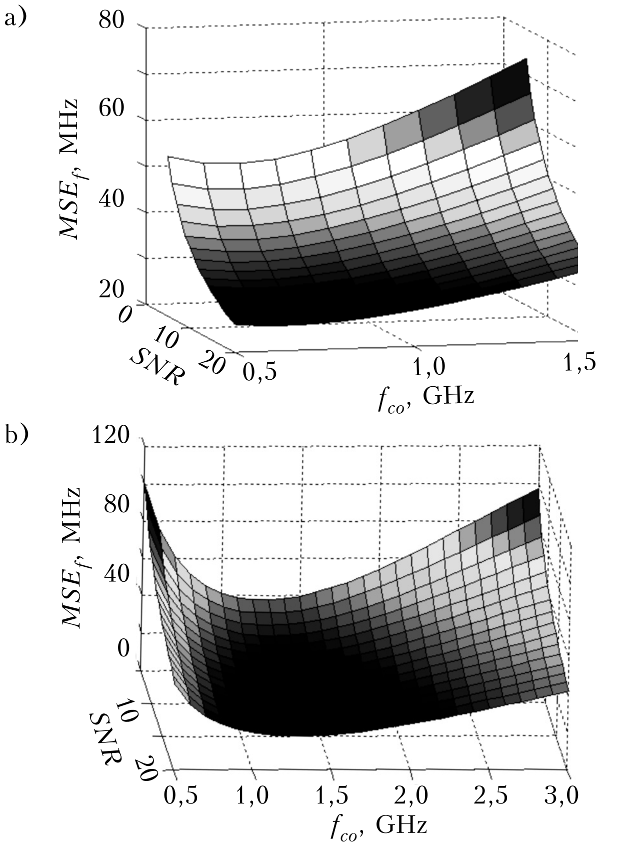

Fig. 13. The mean square error for the 1st (*a*) and the 2nd (*b*) order preliminary LPF

points of optimum, but if the cutoff frequency f_{co} is less than $0.7f_S$, the MSE increases much sharper.

The application of a band-pass filter allows us to further reduce the MSE at optimal points, but requires a precise coordination of frequency tuning to a range of possible signal frequencies. According to research results for the 1st order band-pass filter with a 20% bandwidth, the prior uncertainty range should not exceed 0.8—1.2 relative to the true frequency value. When the 2nd order filter is used or the bandwidth is narrower, the requirements for prior knowledge become stricter.

Improvement of non-harmonic signals estimation

As it was mentioned earlier, pre-filtering is also useful for estimation of frequency of periodic non-harmonic signals, the main feature of which is presence of higher harmonics. For example, the MSE surface of estimation of square wave frequency after the 1st order LPF is shown in **Fig. 14**.

As it can be seen, there is a gradual shift of f_{co}^{opt} towards zero, due to the negative value of the second derivative of response of the LPF in the high frequency range. Without pre-filtering the errors of square wave frequency estimation (even without the noise) exceed 50%, which proves the reasonability of application of pre-filtering.

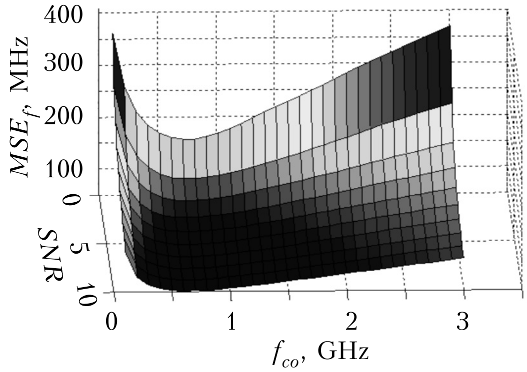

Fig. 14. The MSE for the 1st order preliminary LPF

It was found that the value of the MSE for trapezoidal signals (which have much smaller harmonics), decreases 4—6 times in comparison to the meander.

Detection of the signal frequency modulation

A great feature of the algorithm is a sufficiently accurate estimation at intervals (windows) equal to a period of the signal [6] and even at a half period when the noise level is low, which brings us nearer to the actual IF and provides the opportunity to observe its modulation over time. This property is investigated on the example of a linear frequency-modulated **(LFM)** signal processing with pre-filtering by the 1st order LPF. The results of measuring the IF of such signal with the frequency that varies from 0.9 to 1.8 MHz during the time interval of 10 ms with the sampling frequency of 16 MHz are shown in **Fig. 15**. The MSE is obtained by averaging of differences at each signal sample.

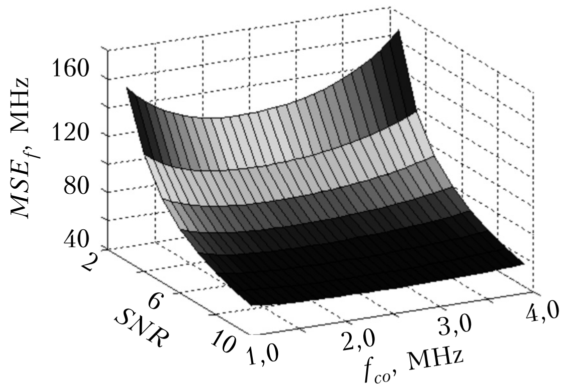

Fig. 15. The MSE for the LFM signal

Conclusions

The application of the new frequency discriminator with an estimation of instant frequencies of reference and generated signals allows adding to the FLL-system an adaptive filter of the reference signal and an open regulation contour. Small lagging of blocks of the instant frequency estimation and the open regulation contour provide fast frequency tracking. The speed of a transient process reaches 3 to 5 cycles of a signal. The adaptive pre-filtering allows increasing the signal to interference ratio at FLL-system input and improves the accuracy of frequency tracking. The application of the 2nd order band-pass pre-filter is only reasonable for a small prior frequency ambiguity range (not more than 20%), while in other cases the 1st order low-pass filter is more preferable.

REFERENCES

1. Xin Jin, Runbo Fu, J. S. Nielsen. Method and apparatus for frequency tracking in a space timetransmit diversity receiver. Pat. USA no 2012288041A1, 2012.

2. Rembovsky A. *Radio Monitoring*. Springer, 2009.

3. Curran J. T., Lachapelle G., Murphy C. C. Improving the design of frequency lock loops for GNSS receivers. *IEEE Trans. on aerospace and electronic systems*, 2012, Vol. 48, no 1, pp. 850-868.

4. Xiao-Qiang Guo, Wei-Yang Wu, He-Rong Gu. Phase locked loop and synchronization methods for gridinterfaced converters: a review. *Przegl d Elektrotechniczny (Electrical Review)*, 2011, Vol. 87, no 4, pp. 182-187.

5. Rodríguez P., Luna A., Candela I., Mujal R., Teodorescu R., Blaabjerg F. Multiresonant frequency-locked loop for grid synchronization of power converters under distorted grid conditions. *IEEE Trans. on industrial electronics*, 2012, Vol. 58, no 1, pp. 127-138.

6. Prokopenko I.G., Omelchuk I.P., Chyrka Y.D. Radar signal parameters estimation in the MTD tasks. *International Journal of Electronics and Telecomunications*, 2012, Vol. 58, no 2, pp. 159-164.

7. Talbot D. *Frequency acquisition techniques for Phase-locked loops*. IEEE Press, 2012.

8. East P. W. Fifty years of instantaneous frequency measurement. *IET Radar, Sonar and Navigation*, 2011, Vol. 6, no 2, pp. 112-122.

9. I.G. Prokopenko, I.P. Omelchuk. Method for harmonic signal frequency estimation. Pat. UA no 2345679, 2009.

10. Weigang Sun, Huainan Ma, Wenshen Wang. Mutual loop circuit device for decimal fraction frequency division lock. Pat. CN no 201004621Y, 2008.

11. Haykin S. *Adaptive filter theory, 3rd edition.* Prentice Hall, 1995.

12. Kušljevic M. D. A simple method for design of adaptive filters for sinusoidal signals. *IEEE Trans. on instrumentation and measurement*, 2008, Vol. 57, no 10, pp. 2242-2249.

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ШВИДКЕ ВІДСЛІДКОВУВАННЯ ЧАСТОТИ

Пропонується метод відслідковування частоти періодичного сигналу. Підвищення точності підлаштування частоти досягається завдяки використанню нового швидкого частотного дискримінатора на основі оцінок миттєвої частоти. Також доводиться доцільність попередньої фільтрації вхідного сигналу у випадку нелінійних спотворень, гармонічних завад та сильного шуму.

Ключові слова: ФАПЧ, швидкість, оцінювання частоти, завада, адаптивний фільтр, розімкнений контур.

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БЫСТРОЕ ОТСЛЕЖИВАНИЕ ЧАСТОТЫ

Предлагается метод отслеживания частоты периодического сигнала. Повышение точности подстройки частоты достигается благодаря использованию нового быстрого частотного дискриминатора на базе оценок мгновенной частоты. Также доказывается целесообразность предварительной фильтрации входного сигнала в случае нелинейных искажений, гармонических помех и сильного шума.

Ключевые слова: ФАПЧ, скорость, оценивание частоты, помеха, адаптивный фильтр, разомкнутый контур.