Identification Approach To Determining Of Radio Signal Frequency

D.P. Kucherov (0000-0002-4334-4175)
Department of Computerized Control System
National Aviation University
Kyiv, Ukraine
d_kucherov@ukr.net

A.L. Berezkin (0000-0003-3087-1184)
NAS of Ukraine, Pukhov Institute for Modeling in Energy Engineering
Kyiv, Ukraine
Abis1999@ukr.net

***Abstract*— The article deals with method of spectral analysis of radio, which can be useful in used for receiving signals by undirected antenna. The main problem of this analysis is find the central of frequency of a sinusoidal oscillation when observed object hidden in noise. Typically this problem is solves by estimation unknown frequency by least square error method. In this paper, proposed iterative technique that can be used for in panoramic receiving set. The advantage of the new estimator is its computational simplicity.**

Keywords—radiosignal; digital processing; frequency; maximum of periodogram; identification

I. Introduction

Modern society cannot be imagined without the means of transmission and reception of radio signals. If twenty years ago, receiving and transmitting radio signals, mostly associated with public radio and television, now the position has changed radically. Radio receivers literally fused with every day of our life, penetrated into various domain of activities. This is a mobile communication and internet, Wi-Fi devices, radio-controlled devices, radio navigation etc.

At the same time, the relevant control measures become important by using of radio devices. In order to ensure continuous operation of government, military and commercial radio systems is radio monitoring ongoing basis. Radio reconnaissance is a part of radio monitoring that acting in military activity.

There are many radio devices of radio monitoring. Among them are scanning radio receiver units, selective microvoltmeter, digital spectrum analyzer, panoramic radio measuring receivers. Their using for monitoring of radio signals in some frequency band, to determining the workload of the observed range and detection of parasitic emitters, measuring of the field strength at the point of reception from different emitters, to determining and identification of the type emitters, and calibration of radio signals generator. The most common device is a radio receiver that performs measurement frequency of radio signals.

Input measuring signal is radio signal having alternative amplitude and different frequencies. Measured frequency located range from hundreds of kilohertz to tens of gigahertz. The accuracy of the measurement results depends on errors of the test and the method of measurement. The relative accuracy of measurement is in the interval up 10^{-3} to 10^{-4} that achieved now. Current requirements to precision of radio measuring is an order higher at least.

Modern approaches to measuring radio signals are based on a digital representation of the input signals. To overlapping wide frequency range is need to except from the structure of panoramic receiver analog processing, and its output on the intermediate frequency is converted into digital form and further it is to process at digital PC.

An efficient method to determine the ranges of valid bandpass sampling frequency for multiband RF signals is presented in [1]. This result is effective to using in multiband SDR receiver front-end. Some practical realizations of digital processing is given enounce in [2]. A number of algorithms for estimating the frequency of sinusoidal vibrations are represented in [3-8]. As the studies show, the search for an effective algorithm for determining the frequency of the adopted oscillations is not completely solved by now.

The article is an investigating the method of quick search of measuring the frequency of radio signals by its identification.

II. Problem formulation

Let receiving measuring system which may include broadband input and converting device, providing reception signal $s(t)$ with the additive noise γ (t)

$$y(t) = s(t) + \gamma(t) \tag{1}$$

where y (t) is a signal at the input of the receiving means, γ (t) is a random signal Gaussian type with zero mean, t is a time that belong in interval of observation T, $t \in [0, T]$. For signal of γ (t) is true

1) γ (t) is strictly stationary and ergodic with $E\ \{\gamma\ (t)\} = 0$, and $E\ \{\gamma^2(t)\}<\infty$.

2) If $\mathscr{F}_t$ being σ-algebra of events generated by $\{\gamma_s;\ s \le t\}$, $E\{\gamma(t) |\ \mathscr{F}_\infty\} = 0$.

3) The spectral density function f (ω) of $\gamma(t)$ is strictly positive at the true frequency.

A useful signal $s(t)$ is characterized by a vector of parameters, among them the largest interest to researchers attractive the amplitude and frequency. Our approach allows representing signal $s\ (t)$ in the form

$$s(t)=\mathrm{Re}\{S(t)\exp[\,j(2\pi ft+\varphi)]\}\,, \tag{2}$$

where $S\ (t)$ is an instantaneous value of amplitude of the signal at time t, and f is measured parameter, the frequency of signal reception. It is assumed that the parameters $S\ (t)$ and f is a priori unknown, but may be in the range

$$\underline{S}\le S(t)\le \overline{S},\ \ \underline{f}\le f\le \overline{f}\,, \tag{3}$$

where $\underline{S}$, $\overline{S}$, $\underline{f}$, $\overline{f}$ is boundaries of measured parameters S and f accordingly.

We will assume that in receiving set performs analog and digital processing. Analog part fulfils preliminary processing, such as: preselecting, amplify, and down converting signal spectrum in intermediate frequency domain. Further, this signal undergoes discretization in time and quantization in level, i.e. transformation to digital form. Digital part consists on getting discrete spectrum, fixation, and definition frequency receiving signal. Thus, we have such architecture receiving set as show on Figure 1.

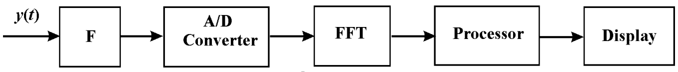

Figure 1. Architecture receiving set

Signal $y(t)$ after filtering F and analog-digital converting in A/D converter (Figure 1) we can represented signal $y(t)$ in discrete form as periodogram of a time series

$$y[n\Delta]=s[n\Delta]+\gamma[n\Delta]\,, \tag{4}$$

where Δ is a simple period, $\Delta = 1\ /\ f_s$, $f_s = \overline{f}$ is a sample rate. The signal $s[n\Delta]$ is describes by the model of harmonic signal

$$s[n\Delta]=U_0+U\cos(\omega[n]\Delta+\varphi)\,. \tag{5}$$

In (5) U_0 is the mean or DC term, U is the amplitude, φ is the initial phase.

Transformation from timing domain to frequency domain performed by Discrete Fourier Transformation (DFT)

$$Y[k]=\sum_{n=0}^{N-1} y[n]e^{-j2\pi kn/N}\,. \tag{6}$$

In (6) N is a quantity of dots DFT. Processor is defines the readouts with maximum amplitude. Typically path to definition maximal frequency is maximize of the square of (6) so that

$$|\,Y[k]\,|^2=\frac{2}{N}\left|\sum_{n=0}^{N-1} y[nT]e^{-j2\pi kn/N}\right|^2. \tag{7}$$

over (0, π). Unavoidable noise are lead to errors in determining of frequency ω.

In this article concerns and solves the problem of determining the assessment methods parameter ω of the received signal in conditions of a priori uncertainty about their actual values are given in the form (3), to build a panorama measuring receiver.

III. Idea of Algorithm's Identification

The proposed identification technology is based on the idea of solving of recurrent inequalities, which in Euclidean space $\{\beta\}$ has general form

$$\eta[n]=|\,(a[n],\ \beta[n])+\alpha[n]\,|\ \le\ \varepsilon\,. \tag{8}$$

In (8) n = 0, 1, 2, …, it is the sample's number from set of a that is estimated; β is the weight vector, $\beta[n]\in\{\ \beta\ \}$; (a [n], β) is a dot product of vectors a and β; $\alpha > 0$, $\varepsilon > 0$ are real numbers.

Inequalities (8) for fixed n is defined zone (stripe) between two parallel plane in space { β }. The bounders of stripe (8) is a priori unknown. We will suppose that is exist an algorithm, which by calculated value $\beta[n]$ that deliveries vectors $a\ [n]$ to zone defined by (8). This algorithm called as a "stripe" [9]. In accordance to discrete values $a[n]$ the correction of values of vector β is performed such as

$$\beta[n]=\begin{cases}\beta[n-1], & if\ |\,\eta[n-1]|\le\ \varepsilon,\\ \beta[n-1]-\lambda[n-1]\eta[n-1]\,|\,a[n-1]\,|^{-2}\ a[n-1] & otherwise.\end{cases} \tag{9}$$

If inequality in (9) is true, the vector $\beta[n]$ is not change by algorithm, in otherwise the vector $\beta[n]$ is projected on the plane ($a[n]$, $\beta[n]$) + $\alpha[n]$ = 0 that is located inside of the stripe (8). The action of algorithm (9) is illustrates on Figure 2.

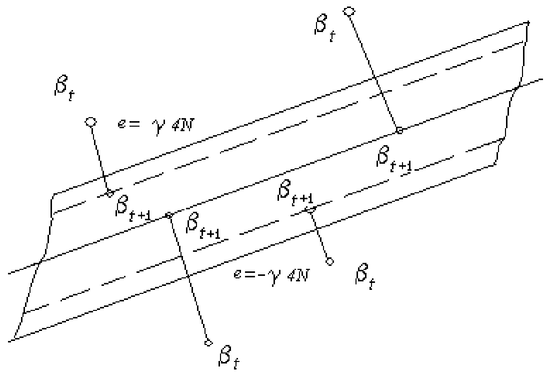

Figure 2. The action of algorithm "stripe"

Thus, algorithm (9) with discrete value $a[n]$, estimating value $\beta[n]$, and setting threshold ε is performs correction $\beta[n]$.

IV. Algorithm of identifikation

We will consider the algorithm of identification in form (9) where

$$\eta[k] = |Y[k]|^2 - |Y[k-1]|^2 . \qquad (10)$$

Here $Y[n]$ is an element of series frequency sequence, is dot of FFT, coefficient of setting

$$\beta[k] = \exp(-2\pi\omega[k]) . \qquad (11)$$

corresponds the value of estimation, $a[k] = Y[k]$, and ε is sufficiently small value. The number γ is given from interval $0< \lambda' \le \lambda[k] \le \lambda'' < 2$ so that to satisfy requirement $\beta[k] > 0$. Algorithm (9) is solves inequality with (11) which is in interval $1 \le k \le [(N - 1) / 2]$. The sought frequency is determined by a relation $\omega = \ln(\beta)$ that does not contradict the results obtained in [3-8].

Theorem. Let $y(t)$ be generated by (1) and $\gamma(t)$ satisfies condition 1)-3) (Section II) and $0 < \omega < \pi$. For $\beta[0] \ne 0$ the algorithm

$$\beta[k] = \begin{cases} \beta[k-1], \ \ if \ |\eta[k-1]| \le \ \varepsilon, \\ \beta[k-1] - \lambda[k-1]\eta[k-1]/Y[k-1] \ \ otherwise \end{cases} \qquad (12)$$

is converging with Lyapunov's function $V(\beta) = |\beta - \beta^*|^2$, where β^* is explicit value.

Proof: This follow from expression (10) – (12) and imagination about $V(\beta) \le c = const.$

In result, we get processing unit shown on Figure 3.

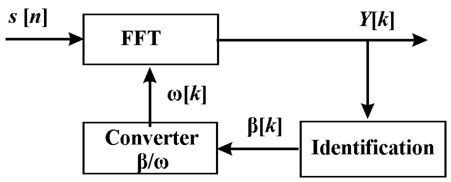

Figure 3. Processing with identification

An identification unit and a converter β[k] to $\omega[k]$ supplement processing for the effective operation of the FFT unit.

V. Simulation

For generation of time series were used (2) in form

$$y(t) = U\cos(j\omega_s t + \varphi) + \gamma(t) \qquad (13)$$

and the algorithm of identification frequency are used from Section IV. In numerical experiments assumes $U = 1$, $\varphi = 0$, f_s=100 c^{-1}, time t is in interval of observation $t \in [0, T]$, where $T = m / f_s$, $m >> 1$, Figure 4.

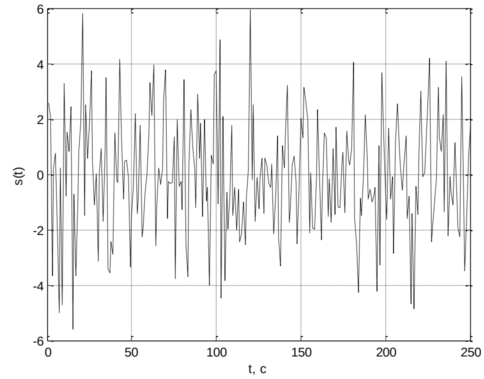

Figure 4. Signal $s(t)$

Further signal $s(t)$ fall under discretization with N dots. Signal noise $\gamma(t)$ (13) is sequence of uniform distributed random numbers that forms by random generator, $|\gamma(t)| \le U$. This additive sum come in to identification set with algorithm (12). The identification parameter of the input signal is unknown. Let we set numbers $\beta[0] = 1$, $\varepsilon = 0{,}01$, $\lambda[0] = 1$ for identification in the beginning. In this identification procedure step, we get the frequency spectrum as on Figure 5.

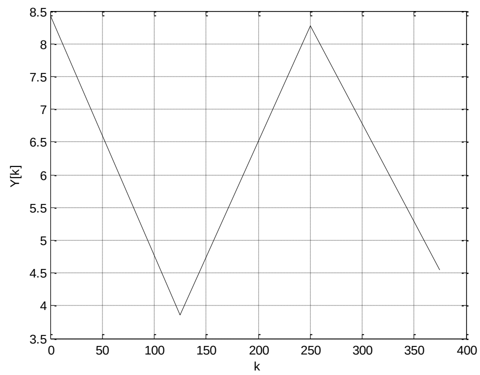

Figure 5. Frequency spectrum on the first step of identification

On the last identification procedure step, we get frequency spectrum as show on Figure 6.

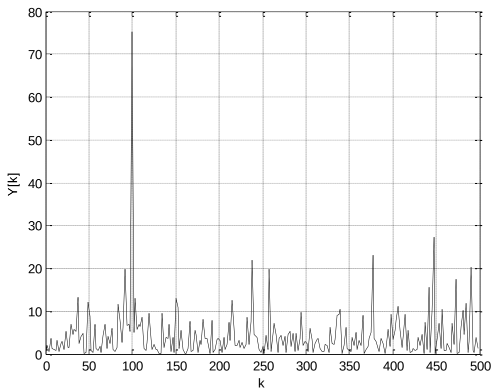

Figure 6. Frequency spectrum on the last step of identification

From the last picture, we can set one main harmonic in spectrum that correspondence signal frequency and that gets for bounded steps number (typically $k \leq 7$).

We researched varying coefficient β and function V(β) behavior in process of identification. It shown on Figure 7, 8 accordingly.

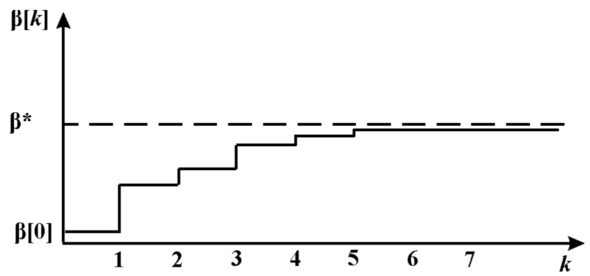

Figure 7. Coefficients β[k]

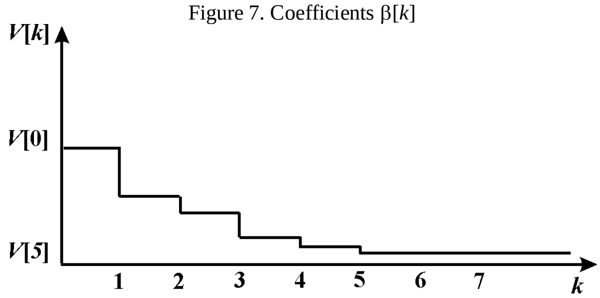

Figure 8. Function V(β)

Figure 10 shows that the identification procedure was completed in 5 iterations.

VI. Conclusions

In this paper, a new technique of estimation of radio signal frequency that is based on identification approach is presented. Our approach considers only digital signal form. The algorithm of estimator, that we propose, has the same order of errors as least square error in sample size N that in accordance to result of [3-5].We assumes that its can used in means of radio monitoring.

Acknowledgment

Authors thank both the authorities of National Aviation University, National Academy of Science of Ukraine, Pukhov Institute for Modeling in Energy Engineering for their support during the preparation of this paper.

References

[1] C.-H. Tseng, S.-C. Chou, "Direct Downconversion of Multiband RF Signals Using Bandpass Sampling", IEEE Trans. on Wireless Communications, Vol. 5, No. 1, 2006, pp. 72-76. http://ieeexplore.ieee.org/document/1576530/

[2] E.C. Ifeachor, B.W. Jervis, Digital Signal Processing: A Practical Approach, 2nd ed., N.Y.: Pearson Education, 2002, pp. 935. https://www.bookdepository.com/Digital-Signal-Processing-Emmanuel-C-Ifeachor/9780201596199

[3] E.J. Hannan, "The estimation of frequency", J. Appl. Prob., Vol. 10, No. 3, 1973, pp. 510 -519. https://www.jstor.org/stable/3212772?seq=1#page_scan_tab_contents

[4] B.G. Quinn, "Estimating Frequency by Interpolation Using Fourier Coefficients", IEEE Trans. Signal Processing, Vol. 42, No. 5, 1994, pp. 1264-1268. http://www.ingelec.uns.edu.ar/pds2803/materiales/articulos/analisisfrecuencial/00295186.pdf

[5] B.G. Quinn, "Esimation of frequency, Amplitude, and Phase from the DFT of a Time Series", IEEE Trans. on Signal Processing, Vol. 45, No. 3, 1997, pp. 814 - 817. https://pdfs.semanticscholar.org/df2e/2b3ae9d784e19ea0840f8bb26ff622b17c22.pdf

[6] G. K. Smyth, "Employing Symmetry Constraints for Improved Frequency Estimation by Eigenanalysis Methods", Technometrics, Vol. 42, 2000, pp. 277 – 289. http://www.statsci.org/smyth/pubs/constrai.pdf

[7] S. Nandi, D. Kundu, "Estimating the fundamental frequency of a periodic function", Elsevier Signal Processing, Vol. 84, 2004, pp. 653 - 661. http://home.iitk.ac.in/~kundu/paper86.pdf

[8] S. Nandi, D. Kundu, "An Efficient and Fast Algorithm for Estimating the Parameters of Sinusoidal Signal", Sankhya, 68(2), 2006, pp. 283 - 306. http://home.iitk.ac.in/~kundu/paper114.pdf

[9] D.P. Kucherov, A.M. Kozub, Metodi sinteza adaptivnih sistem terminalnogo upravlenia, LAP Lambert Academic Publishing, 2013, pp. 388. https://www.morebooks.de/store/gb/book/isbn/978-3-659-40021-6