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Frequency Estimation for Short Realization of Radar Signals

I. The New Algorithm Presentation

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Abstract

The paper is devoted to the problem of short time delays measurement in radar systems. Authors present the algorithm for measurement of short time of delays for linearly frequency modulated signals.

Keywords: radars, chirp, signal processing, frequency estimation

Introduction

Many application of short-range distance measurement by radio- and ultrasonic continuous signals with linearly frequency modulation (chirp signals) used heterodyne signal processing technique. Such systems have the problem of decreasing the minimum time of delay that can be measured with required accuracy [4]. In chirp radars the estimation of time of delay is carried out by computation from estimated frequency of treated signal [8]. Vale of frequency generally evaluated by determination of maximum position in Furies spectra area [1]. However such approach required the number of samples which corresponded more than to 2-3 signal periods [3]. In case of decreasing the number of samples peak of it spectra become flatter that provides the increasing error of the estimation of it maximum position.
In many application of short range radars the development of new algorithms which provided reduction the minimum of the measured time of delay are necessary [4]. In chirp radar such condition implies decreasing the number of periods (N) of continuous beating signal that leads to increasing of error. In instance, all of the above are actual for distance or level measurements, flow measurements, nondestructive testing and act where heterodyne processing for signals with complex form are widely used .
For mentioned applications many algorithms whose provide increasing of extracted information from limited number of samples have been proposed. As a rule enhancing informational content implemented by using a priori information, quadrature decomposition, interpolation and ect. [1, 5-7]. However in each considered articles the issue about minimum limitation of the measured time delay with a given accuracy doesn’t researched. For solution the described the problem the paper propose new algorithm.

Main part

As it was mention above in the continuous-waves radars signals with linear frequency modulation (chirp signals) are generally used. In most cases receiving of signals is carried out by the heterodyne scheme as it shown on the figure 1 [8].

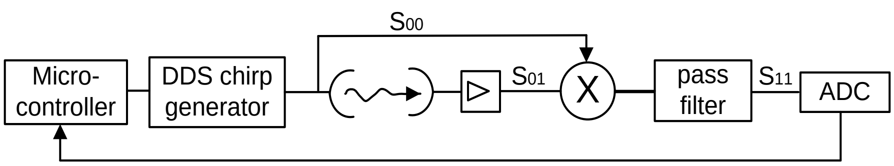

Figure 1 – scheme of receiving and processing of chirp signals. S00 – emitted signal, S01 received signal, s11- beating signal [8]

The difference generalized variant with scheme that shown on the figure 1 in the use of micro scheme for direct digital signal forming (DDS). In such devises chirp signals can be created with high level of linearization frequency changing. This can be implemented if time of discretization of frequency changing is less enough compared with the expected heterodyne signal frequency [9].
The signal processing in the scheme 1 are carried out by multiplexing of received signal with delay (τ) and reference signal (S01 and S00) and followed separation from this mix low frequency component (S11):

$$S_{11} = A\cos\left[\left(\frac{b\tau}{T_M}\right)t + (\omega_0\tau + \frac{b^2\tau}{T_M} + \varphi_{00} - \varphi_{01})\right] = A\cos(\omega_h t + \varphi_h) \quad (1),$$

where A – amplitude, ω_0 – start frequency, b – deviation of frequency, $b = \omega_{max} - \omega_0$, T_M – duration of chirp modulation, τ- delay, ω_h, φ_h – frequency and phase of beating signal, n – white noises. In signal (1) information about time of delay included infrequency and phase. Time of delay basically computed from estimated frequency, which determine by the Fourier transform [8]. However this method can't be used for short observation time due to occurrence of Gibbs phenomena and due to the broadening of spectral peak that decrease accuracy of measurement, the Nyquist frequency is also essential [10].
The extension of spectra by zeros or interpolation is widely uses for decreasing discrete error [3]. Interpolation methods are not applicable if form of peak is different from expected, that have place to be in case when number of samples little enough. Nevertheless when estimated signals consist only from one harmonic, interpolation can be correctly carried out by function of phase changing depend on time. In this case function of phase changing has linear form and a large number of samples are not required. Such relation can be obtained in digital form by Hilbert transform or in analog form by quadrature decomposition [5].
It should be noted that the most part of algorithms that offered for increasing of accuracy of frequency measurement can't be used for processing of short realization. For instance methods [5, 7] have been proposed for estimation of frequency by computing meaning value of slope on each interval of phase changing from -π to π. This cannot be implemented if realization has just 1-2 period of continuous signal or less than 1 full period in the time of observation. Nevertheless the relation of phase can be unwrapped up to infinity that used in [6]. But authors [6] have proposing method of maximum likelihood estimation, which requires infinity length of sample [11]. The truncation number of samples is leading to rise of error.
Based on analysis of the considered materials the new algorithm for estimation of frequency in conditions of short number of period and samples in continuous signal (on short time of observation) is proposed. Implemented method includes next stages.

1. For the signal (1) in analog or digital form extracting of signals quadrature are carried out.

$$s_Q(t) = \frac{1}{2}\cos(\omega_6 t + \varphi_6), \text{ and } s_I(t) = \frac{1}{2}\sin(\omega_6 t + \varphi_6),$$

2. From both obtained sampling the function of phase depends on time are obtained.

$$\varphi(t) = \arctan[s_I(t)/s_Q(t)] \ (2).$$

3. Unwrapping relation (2) on the range from 0 to$+\infty$.

As it was noted beating signal (1) consist only form one harmonic, that's why function (2) has mainly linear form, it depend on number of periods (N, NΔt=T-time of observation, Δt – time of discretization step).

4. For unwrapped function (2) performed linear interpolation by least squares method. Obtained function is:

$$\varphi_h(t) = at + b \quad (3).$$

5. The heterodyne frequency estimated as a slope of line (3) as:

$$f_h = \frac{1}{2\pi}\frac{1}{a}.$$

Conclusion

The new algorithm based on linear relation between changing of phase of beating signal from time has been proposed. Also should be noting the relative computation simplicity for proposed method. That is taken place to be when analog quadrature decomposition of beating signal is carried out in analog area, than in digital domain only phase to time relations are processed by least square method.

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References

[1] Kalmykov A. A., Nikitin N. P. The signal processing of autoregessin signals: students book, USTU-UPI, Yekaterinburg, 2002.

[2] Melentev V. S., Batishhev V. I. Approximation methods for systems of measurement and control of parameter of periodic signal, FIZMATLIT, Moscow, 2011.

[3] Antipov S. A., Gnezdilov D. S., Kozmin V. A. and Stopkin V. M. Digital interpolation algorithms for estimating the frequency of the harmonic signal. Comparative analysis, *Radiotekhnika*, **3,** (2014), 42 - 46.

[4] Atayants B. A., Parshin B. C. Measuring the frequency of the harmonic signal received on a background of additive white noise, in his short implementation, *Measuring equipment.* **6**, (2004), 42 - 46.

[5] Kupnik M., Krasser E. and Groschl M. Absolute transit time detection for ultrasonic gas flowmeters based on time and phase domain characteristics, *2007 IEEE Ultrasonic symposium*. (2007), 142 - 145.

[6] N. Roosnek Novel digital signal processing techniques for ultrasonic gas flow measurements , *Flow measurement and instrumentation*, **11** (2000), 89 - 99. http://dx.doi.org/10.1016/s0955-5986(00)00008-x

[7] Kalmykov A. A., Ronkin M. V. Investigation of approximation method for increase the accuracy of phase measurement systems, *2013 23rd Int. Crimean Conference "Microwave & Telecommunication Technology" (CriMiCo'2013),* (2013), 1200 - 1201.

[8] Komarov I. V. *Fundamentals of the theory of radar systems with continuous radiation of frequency-modulated oscillations*, Gorjachaja linija–Telekom, Moscow, 2010.

[9] Karmanov J. T., Povaljaev S. V. Characteristics of radar range finders with LFM-probing signals synthesized by a digital technology DDS, *SUSU. Computer technology, management, electronics,* **14**, №1, (2014), 14 - 22.

[10] Oppenheimer A., Sharf R. *Discrete-time signal processing*, Prentice Hall, New Jersey, 1999.

[11] Van Trees H. *Detection, estimation and modulation theory*, Wiley, New York, 1971.

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