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Estimates of the Carrier Frequency of the Signal received by the Satellite Communication system in Package mode

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ABSTRACT

The process of estimating the carrier frequency of the signal received by the satellite communication system in the packet mode according to the rule of maximum similarity (MP-estimation) is considered. Goal. Development of functional dependencies and, based on them, development of the procedure for estimating the carrier frequency of the signal received by the satellite communication system in a continuous mode according to the rule of maximum likelihood. Method. The proposed procedure and the algorithm developed on its basis allows to perform MP-estimation of the carrier frequency taking into account the condition of uncertainty of all parameters of the signal received by the satellite communication system in packet mode at short observation intervals. Results. Functional dependences are determined, based on them a rule is formed and an algorithm for estimating the carrier frequency of a signal received by a satellite communication system in a packet mode according to the rule of maximum likelihood using a sliding FFT that provides short observation intervals is proposed.
This algorithm makes it possible to estimate the frequency according to the rule of maximum likelihood, taking into account the condition of uncertainty of all parameters of the signal received by the satellite communication system in packet mode at short intervals of observation. Conclusions. A comparative analysis of the results of the evaluation of a complex sinusoidal signal in a noise environment, carried out on the basis of an algorithm using a sliding FFT, which is provided by the dichotomous search procedure and depends on the offset value, showed that this estimate is asymptotically unbiased with its root mean square Cramer-Rao. This confirmed the feasibility and relevance of the approach specified in this paper to estimate the carrier frequency of the signal received by the satellite communication system in packet mode.

Key words: The received signal, packet data transmission mode, signal carrier frequency estimation, fast Fourier transform, Kramer-Rao lower bound.

1. INTRODUCTION

Satellite communication is intensively used in the implementation of important national projects, including for the effective solution of national security problems, with the aim of socio-economic development of states and successful international cooperation. Today, the development of satellite communications is difficult to imagine without its connection to terrestrial networks. Most infrastructure changes have a significant impact on both satellite services and system performance.
The simplest satellite communication channel includes two satellite earth stations and a space communication station located on board the satellite. Note that this channel has at least one signal frequency conversion. This transformation is carried out in the on-board repeater of the communication satellite [1]. The presence of this transformation due to the internal instability of the reference generator-frequency converter of the on-board repeater of the communication satellite leads to accidental and non-stationary displacement of the carrier oscillation of the signal relative to the nominal value. This causes the frequency uncertainty of the signal [2].

1.1 Problem analysis

The entire frequency range in which the satellite repeater operates is divided into some bands (width 27 ... 36, 72 ... 120 MHz), in which the signal is amplified by a separate path - the trunk [2,3]. The trunk, in turn, can transmit signals from many satellite earth stations. Thus, the satellite repeater can provide communication to a large number of subscribers. The organization of access to virtually independent earth stations in the general communication system and the rapid establishment of connections between arbitrary stations and multi-station access are widely used in satellite communication systems [2,3].
In general, there are several different ways in which many users can send information via satellite. Currently, two types of multi-station access are widely used in satellite communication systems (SCS [1,2]:
- multi-station access with frequency division multiplexing (MAFD);
- multi-station access with temporary division of channels (MATD).

The relative simplicity and low cost of equipment, as well as the extensive experience in the development and operation of frequency division systems gained in the development and operation of early communication systems, have led to the use of MAFD in the vast majority of modern satellite communication systems. On-demand satellite communication systems (OCS), which operate with frequency division multiplexing (MAFD - OCS), operate at fairly low information rates. As a result, in such systems it is possible to use relatively cheap satellite terminals with a small aperture class a VSAT (very small aperture terminal) [3,4]. Relatively narrowband channels are used in MAFD - OCS systems. Therefore, the initial shift in the frequency of the carrier oscillation of the signal can be compared with the band of the channel.

Earth stations of the system, built on the technology of MAFD - OCS, include modems of information channels and modems that form a common control channel. This control channel in satellite communication systems with the provision of the channel on demand is called a common synchronization channel (CSC) [2,3,4].

Modems that provide CSC on peripheral earth stations operate in continuous mode for reception and in batch mode for transmission. The CSC modem of the central station operates in a continuous mode for transmission and in a batch mode for reception. In other words, there is a common packet control channel in MAFD - OCS satellite systems. This channel is usually built on the algorithm ALOXA with random access of packets [1,4].

To ensure the operation of modems, signal demodulators are used, which should be distinguished by types: demodulators operating in batch mode and demodulators operating in continuous mode.

Synchronization and demodulators operating in continuous mode on the carrier frequency is carried out from the information (modulated) signal. Synchronization of demodulators operating in batch mode is carried out according to the preamble transmitted at the beginning of each packet. As a rule, a harmonic signal is transmitted at the beginning of the preamble to synchronize the packet demodulator.

One of the features of satellite communication systems such as MAFD – OCS is the predominant use in them during signal reception in both modes of phase modulation of signals intended for the transmission of useful information [2]. The use of this type of modulation requires solving the problem of estimating the carrier frequency of the signal. And the estimation itself is reduced to the problem of estimating the frequency of the maximum in the spectrum of a fragment of a sinusoidal signal against the background of additive Gaussian noise.

2. MAIN MATERIAL

2.1 How to increase the speed of the satellite communication system

To solve the problem of estimating the carrier frequency of the FM signal in conditions of uncertainty about the initial phase of the signal (φ), the value of its delay (τ) and the transmitted information sequence (d) in a number of works substantiates the feasibility of applying the rule of maximum likelihood. It is known that the use of the rule of maximum likelihood to estimate the carrier frequency (MP-estimate) provides asymptotically effective and asymptotically unbiased estimates. [5].

In the presence of information about the parameters (d, φ, τ) MP-estimation of the carrier frequency can provide the minimum limiting variance, which will be determined by the lower Kramer-Rao boundary [4,5].

At large observation intervals ($K \rangle\rangle 1$), the normalized Kramer-Rao boundary for estimating the carrier frequency of the phase-modulated signal can be represented as a functional dependence on the unit signal pulse (E_S), the interval of information pulses of the complex envelope of the received signal (N_0) and the interval on which the estimation is performed. [6].

In turn, the speed of satellite communication systems requires a reduction in the estimation time, which can be achieved by reducing the observation interval. One way to reduce the observation interval is to use the fast Fourier transform function (FFT) in the carrier frequency estimation algorithm [7].

The disadvantage of the procedures for estimating the signal transmitted in packet mode, based on FFT is that their implementation requires prior accumulation of data for subsequent calculations according to a given algorithm. Additional delay may be unacceptable in batch mode, as the length of the processing procedure is strictly limited by the length of the preamble. In addition, it should be borne in mind that the general estimation algorithm should work equally for both continuous and packet data transmission.

One of the ways to solve this problem for satellite communication systems such as MAFD – OCS is the use in the algorithm for estimating the frequency of the signal transmitted by the sliding FFT [8,9].

2.2 Analysis of previous works

The application of FFT to estimate the frequency of the transmitted signal was considered in the following works.

The authors of [10] considered the use of FFT to estimate the frequency of the input signal by the Taylor - K. Kalman - Fourier filter. It is established that the specified filter allows to reduce the time of estimation of frequency of the input signal and to increase its accuracy, but questions of increase of speed of estimation through use of the algorithm based on sliding FFT are not considered in work.

In [11], the use of FFT in the algorithm for analyzing the spectrum of the harmonic input signal is considered. This paper presents a signal filtering design technique that uses the Taylor - Fourier transform and the classical solution of normal least squares approximation equations. Which reduces the computations in the frequency estimation process and, therefore, the total estimation time. The use of sliding FFT in the proposed algorithm was not considered in the work.

In [12] the question of using FFT to estimate the accuracy of the input signal in the background of noise is considered. Evaluation of signal accuracy is provided by the use of Hanning windows when receiving the signal and its subsequent evaluation using FFT. The main objectives of the proposed algorithm, improving the accuracy of the assessment, reducing the interval of assessment and the use of sliding FFT for this purpose in the work were not considered.

In [7, 13] the question of the use of FFT in the evaluation of the input signal using the proposed in the works of the Taylor-Fourier transform algorithm (TFT). This algorithm allows to increase the accuracy of the estimate in conditions of dynamic instability at certain time intervals. The issue of reducing the evaluation interval due to the use of sliding FFT in this work was not considered.

In [14], the issue of reducing the interval for estimating the frequency of the input signal using an algorithm based on FFT is considered. The authors propose to modify the FFT formula at each stage of the evaluation and propose recurrent formulas for this purpose. Questions of application in the specified algorithm of sliding FFT in work are not considered.

In [15], the issue of increasing the accuracy of estimating the frequency of the input signal against the background of noise using an algorithm based on FFT. To this end, the authors propose to use the Khan window (Henning) and an advanced algorithm based on window FFT. Questions of application in the specified algorithm of sliding FFT in work are not considered.

Thus, the development of an algorithm for estimating the packet-transmitted signal based on FFT is an urgent scientific task aimed at improving the accuracy of estimating the frequency of the transmitted signal and increasing the speed of the synchronization system of demodulators of modems of satellite communication systems.

2.3 Statement of the research problem

As defined earlier, the disadvantage of FFT-based evaluation procedures is that their implementation requires prior accumulation of data for subsequent calculations according to a given algorithm. Additional delay may be unacceptable in batch mode, as the length of the processing procedure is strictly limited by the length of the preamble.

In this regard, it seems appropriate to use an algorithm for estimating FFT readings based on a recurrent procedure [16, 17]. Or, what is the same, in the calculation of the so-called sliding FFT (sliding FFT) [16, 17]. The calculation of the sliding FFT does not require the initial accumulation of N - ground sample of the signal received to obtain the corresponding spectral sample of length N.

The advantages of sliding FFT include the fact that the calculation procedure is quite simple to implement at any FFT length.

In this paper, we present the essence of the proposed algorithm MP - estimation of the frequency of a sinusoidal signal using the method of sliding FFT.

2.4 Overview of the sliding fast Fourier transform

Using a sliding FFT, the samples of the amplitude spectrum of the received signal are calculated [18]:

$$\left|I_k(nT_S)\right|$$

where $\left|I_k(nT_S)\right|$ – the current value of the sliding FFT with the number k, $k = 0,1,\ldots, N_f - 1$;

n – current reference number, $n = 1,2,3,\ldots$

T_S – sampling period.

2.5 Stages of signal frequency estimation

At the first stage we make a rough estimate of the frequency:

$$f_0 = arg\max_k \left\{I_k\left(N_f T_S\right)\right\}\frac{1}{N_f T_S}, \qquad (1)$$

where $k = 0,1,2,\ldots, N_f - 1$

The procedure of the second stage is to find the maximum of the function

$$I(\nu) = \left|\sum_{n=1}^{N_f} z_n exp\left(-j2\pi n\nu T_s\right)\right|$$

in the vicinity f_0

Note that the value $I(\nu)$ is nothing more than a peridiogram of the received signal. It is clear that the maximum $I(\nu)$ should be sought around the evaluation of the first stage defined by rule (1). In [19,20] it is proposed to use an algorithm based on an iterative procedure of a dichotomous process to find the considered maximum.

In the process of implementing this procedure, the values are calculated:

$$I\left(f_i^+\right) = \sum_{n=1}^{N_f} z_n exp\left(-j2\pi f_i^+ nT_s\right)$$

$$I\left(f_i^-\right) = \sum_{n=1}^{N_f} z_n exp\left(-j2\pi f_i^- nT_s\right)$$

where $f_1^+ = f_{i-1} + \frac{1}{2^i N_f T_s}$, $f_1^+ = f_{i-1} + \frac{1}{2^i N_f T_s}$

$i = 1,2, \ldots, M_1$

Here the value M_1 determines the number of iterations of the dichotomous search, and is f_0 determined by expression (1).
The considered procedure consists in the following:
if

$$\left| I\left(f_i^+\right)\right| \geq I\left(f_i^-\right)$$

then

$$f_i = f_i^+$$

Otherwise

$$f_i = f_i^-$$

When calculating the final estimate of the carrier frequency of the received signal:

$$\approx\!\!\nu = f_{M_1}$$

The frequency distribution $\Delta f_c T$ provided by this procedure is limited by the number of iterations and can be determined as follows [20]:

$$\Delta f_c T = \frac{T}{T_S\ N_f\ 2^{M_1}} \qquad (2)$$

2.5 Analysis of the effectiveness of the proposed procedure for estimating the frequency

It is known that the effectiveness of the estimates provided by the dichotomous search procedure depends on the number of iterations. In [20,21] it is shown that to implement the procedure of dichotomous search with sufficient efficiency of estimates in any, previously set range of signal-to-noise ratio, it is possible to dwell on such a number of iterations that

$$\left(\Delta f_c T\right)^2 = \frac{1}{4}\left(CRLB(\nu)T^2\right)_{min} \qquad (3)$$

where $\left(CRLB(\nu)T^2\right)_{min}$ is the minimum $\left(E_S/N_0\right)_{max}$ of the limiting variance in a given range of signal-to-noise ratio.
For the considered minimum value of the limiting variance can be written

$$\left(CRLB(\nu)T^2\right)_{min} = \frac{1}{2\pi^2}\frac{1}{K^3}\frac{1}{\left(E_S/N_0\right)_{max}}$$

where $\left(E_S/N_0\right)_{max}$ the maximum signal-to-noise ratio per information symbol in a given range of signal-to-noise ratio. Let the observation interval equal $N_f T_S$ to contain an integer number of clock intervals. That is $N_f T_S = KT$.
In this case, based on expression (2), expression (3) will be rewritten as

$$\frac{1}{K^2 2^{2M_1}} = \frac{1}{4}\frac{1}{2\pi^2}\frac{1}{K^3}\frac{1}{\left(E_S/N_0\right)_{max}}$$

As a result, after simple transformations for M_1 we will receive

$$M_1 = \left\lfloor \frac{1}{2}log_2\left[\frac{8\pi^2 K\left(E_b/N_0\right)_{max} log_2 M_\varphi}{3}\right]\right\rfloor \quad 4)$$

Where $\left(E_b/N_0\right)_{max}$ – is the maximum value of the ratio E_b/N_0 in a given range of signal / noise ratio per bit of information.
Thus, based on (4), the number of iterations M_1 is determined, in which the proposed procedure of MP estimation of the sinusoidal signal frequency provides asymptotically effective estimation at the selected length of the observation interval over a given range of signal-to-noise ratio per bit of information.
To assess the feasibility of the proposed approach and rule (4), consider the results of estimating the frequency of a complex sinusoidal signal in a noise environment, based on an algorithm using sliding FFT, and which are provided by the dichotomous search procedure and depends on the offset value (q).
These results are presented in Figure 1 [21]. The dependences presented in Fig. 1 were obtained using hybrid FFT interpolation at half and offset q, which showed the convergence of the estimate for the two operations. It was shown that this estimate is asymptotically unbiased with its RMS value performed on the lower Kramer-Rao boundary ($CRLB$).
That confirms the conclusion about the relevance of the use of the proposed in this work approach to the use of sliding FFT for MP-evaluation of the signal transmitted in a continuous mode.
It should be noted that the parameters of the noise environment can be formed by various environmental factors, among which are both external and internal noise. Among which some interest may be taken into account in the process of estimating the frequency of internal noise associated with

changes in nonlinear properties of composite materials of the element base of the synchronization system under the influence of increasing the number of additional tracks of charge carriers due to decay in the material structure of radioisotope inclusions [21]. What can affect the growth of internal noise of the synchronization system and requires its consideration in the development of advanced systems.

Also, an important element of reliability, noise immunity and stability of the synchronization system, which directly affects the accuracy of the carrier frequency estimate, is the adopted model of the synchronization system. The use of combined synchronization systems with a broken connection, which have a high order of a statism, can have a good effect on reducing the dynamic errors of the frequency estimation process, as evidenced by the studies presented in [23].

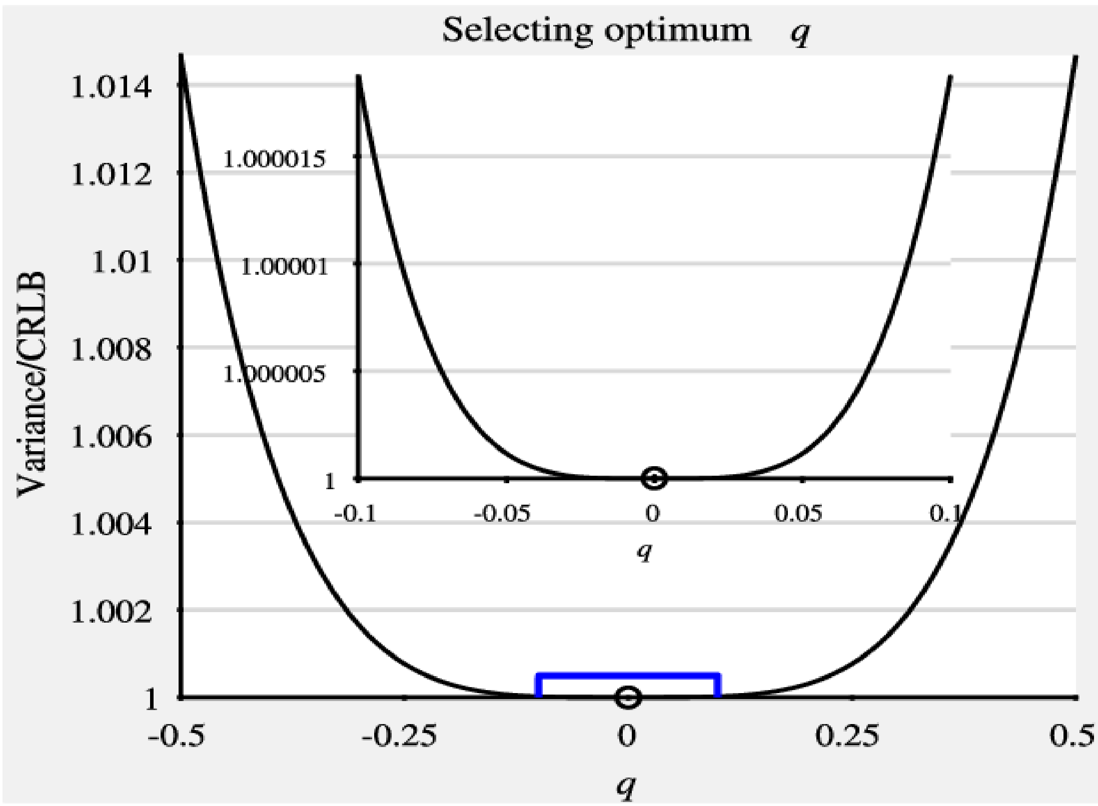

Figure 1:. The ratio of the variance of the sine wave signal to the CRLB compared to the value of the offset number - q . The insert shows magnification around lower q values for better visualization

3. CONCLUSION

The paper determines the functional dependences and, based on them, forms a rule and proposes an algorithm for estimating the carrier frequency of the signal received by the satellite communication system in packet mode according to the rule of maximum likelihood using sliding FFT.

This algorithm makes it possible to estimate the frequency according to the rule of maximum likelihood, taking into account the condition of uncertainty of all parameters of the signal received by the satellite communication system in packet mode at short intervals of observation.

To illustrate the possibility of using this approach to estimate the carrier frequency of the signal transmitted in batch mode, the paper compares the results of estimating a complex sinusoidal signal in a noise environment, based on an algorithm using sliding FFT, and which is provided by the dichotomous search procedure and depends on offset value.

It was determined that this estimate is asymptotically unbiased with its RMS value performed on the lower Kramer-Rao boundary.

Their analysis shows that this estimate is asymptotically unbiased with its RMS value performed on the lower Kramer-Rao boundary. This confirmed the feasibility and relevance of the application of this approach to estimate the carrier frequency of the signal received by the satellite communication system in packet mode.

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