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Method for Transmitting Information under Conditions of Excess of Interference Level over Useful Narrowband Signal

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***Abstract*—There are various methods for suppressing radio interference in communications. In particular, interference exceeding the useful signal level. However, these methods in most cases are based on the parameters of the interference in the spectrums' range, the occurrence's time, the action's direction, polarization, etc. It is known that radio systems with a single receiver do not have effective suppressions' means of powerful broadband interference. In addition, there are fundamental limitations of single-channel information transmission systems formulated in the Shannon-Hartley theorem. We have developed a digital signal processing algorithm. It suppresses radio interference and selects a narrowband useful signal. This algorithm is based on the method of spatially spaced two-channel synchronous reception. The research results show high efficiency. The theoretical model allows error-free transmission of information with a signal-to-noise ratio up to minus 26.5dB. This method will significantly improve information transmission systems for their key parameters. These include the frequency range used and noise immunity.**

Keywords—narrowband signal, broadband interference, two-channel synchronous reception, exceeding the interference level.

I. Introduction

There are a number of important parameters in radio engineering, namely, the frequency span in which information is transmitted, the noise immunity of system, the range. Other parameters of the system are commonly determined by its intended purpose.

Within the scope of this article, a restriction is imposed over the maximum noise immunity and channel bandwidth, which determines the frequency range used for transmitting information.

There is a Shannon-Hartley theorem, in which sets a constraint on the maximum amount of error-free digital data that can be transmitted over such a channel [1].

The value of the spectral efficiency of the channel in accordance with the theorem is defined as:

$$\frac{C}{B} = \log_2(1+\frac{S}{N}), \qquad (1)$$

where C – capacity's channel, B – bandwidth's channel, S – signal's power and N – power of noise.

For the narrowband signal, where the spectral efficiency of information transmission, for instance, is 0.25bits/s/Hz, the signal-to-noise ratio is minus 7.2dB. By modern standards this result is not sufficient for high noise immunity of the system. Therefore, the use of single-channel systems for transmitting information is currently irrelevant!

There are known methods that allow to achieve the system performance with a signal / (interference + noise) ratio in the range from minus 7.2 to minus 20dB. For instance, method suppressing interference using an adaptive spatial filter based on an antenna array [2,3] or matched and adaptive nonlinear filtering methods [3,4]. Or methods based on broadband signals [5].

However, in most cases, aimed at combating noise, whose parameters are known, for example, frequency, impact area, time of occurrence, spatial direction, polarization, etc. and the anti-interference whose parameters are unknown, but sacrificing one of the most important resources radio device – efficient use of radio frequency.

The aim of this work is to show a possibility of solving these problems using a digital signal processing algorithm based on the method of two-channel synchronous reception. According to this algorithm, high-power broadband interference will be suppressed, and a narrowband useful signal will be allocated.

II. Methodology

The modern radio-electronic element base, as well as computing capacity of systems on a chip, make it possible to implement a two-channel synchronous receiver. Figure 1 shows a model of spatially spaced signal reception.

The digital signal processing algorithm is reduced to the following steps:

- Values' calculation of coefficients α_1 and α_2 (Figure 4), which are obtained on the reference signals. These coefficients provide the maximum value of the signal-to-noise ratio.
- Evaluated information signals when α_1 and α_2 are stationary over a certain time interval.

Evaluation of information signals is reduced to the following steps:

- Calculating the coordinates of the information signal vector for its evaluation:

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$$\vec{S} = \alpha_1 \cdot \vec{f_1} + \alpha_2 \cdot \vec{f_2} \tag{2}$$

- Among all the information signals $\vec{S_i}$ to find the signal $\vec{S_k}$ that has the minimum distance d_k to the estimate $\vec{S}$:

$$\vec{S_k} : d_k = \min_{i=1..N} d(\vec{S_i}, \vec{S}) \tag{3}$$

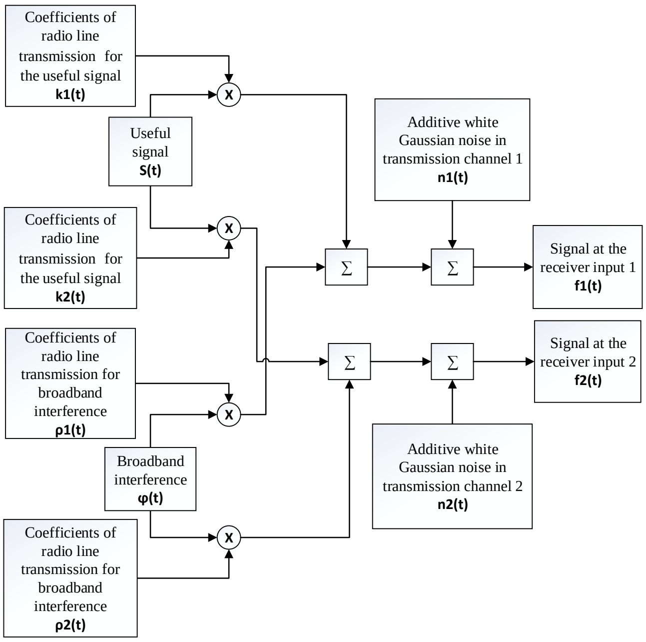

Fig. 1. The model of spatially spaced reception.

To find the coefficients α_1 and α_2, it should be noted that any linear combination input signals $\vec{f_1}$ and $\vec{f_2}$ lies in the plane of these vectors, and the smallest distance from a given point (vector $\vec{S}$) to the plane is reached perpendicular from this point to the plane. So the point on the plane that has the smallest distance to the vector $\vec{S}$ is the projection of that vector on the plane of the vectors $\vec{f_1}$ and $\vec{f_2}$, as shown in Figure 2.

The projection vector $\vec{S_\perp}$ lying in the plane of the vectors $\vec{f_1}$ and $\vec{f_2}$ can be expressed as follows:

$$\vec{S_\perp} = \alpha_1 \cdot \vec{f_1} + \alpha_2 \cdot \vec{f_2} \tag{4}$$

To find the coefficients α_1 and α_2, the following steps are performed sequentially:

- Search of the projections $\vec{p_1}$ and $\vec{p_2}$ of the vector $\vec{S}$ on the vectors $\vec{f_1}$ and $\vec{f_2}$.
- Plotting perpendicular lines to these projections.
- Determination point's intersection of these perpendiculars, which will be the vector's end $\vec{S_\perp}$.

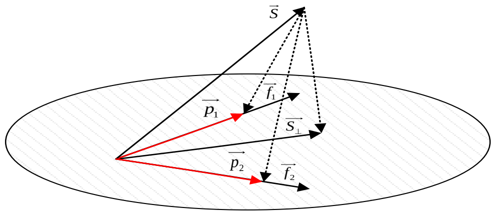

Fig. 2. Projection of the signal vector.

The vector projections of $\vec{S}$ on the vectors $\vec{f_1}$ and $\vec{f_2}$ are equal:

$$\vec{p_1} = \frac{(\vec{f_1}, \vec{S^*})}{\left|\vec{f_1}\right|^2} \cdot \vec{f_1}, \tag{5}$$

$$\vec{p_2} = \frac{(\vec{f_2}, \vec{S^*})}{\left|\vec{f_2}\right|^2} \cdot \vec{f_2}, \tag{6}$$

where vector $\vec{S^*}$ is complex-conjugate to vector $\vec{S}$, and $(\vec{f_1}, \vec{S^*})$, $(\vec{f_2}, \vec{S^*})$ are scalar multiplication of n-dimensional complex vectors.

Then the perpendicular lines to the ends of the vector projections are plotted. These perpendiculars must satisfy the following conditions:

$$(\vec{p_1}, \vec{p_{1\perp}}^*) = 0, \tag{7}$$

$$(\vec{p_2}, \vec{p_{2\perp}}^*) = 0 \tag{8}$$

The required perpendicular lines are written as follows:

$$\vec{p_{1\perp}} = \vec{p_1} - \frac{\left|\vec{p_1}\right|^2}{(\vec{p_1}^*, \vec{p_2})} \cdot \vec{p_2}, \tag{9}$$

$$\vec{p_{2\perp}} = \vec{p_2} - \frac{\left|\vec{p_2}\right|^2}{(\vec{p_1}, \vec{p_2}^*)} \cdot \vec{p_1} \tag{10}$$

The intersection point of the vectors' continuations $\vec{p_{1\perp}}$ and $\vec{p_{2\perp}}$ determines the projection signal's received $\vec{S_\perp}$ on the plane of vectors $\vec{f_1}$ and $\vec{f_2}$, as shown in Figure 3.The intersection point of the vectors' continuations $\vec{p_{1\perp}}$ and $\vec{p_{2\perp}}$ determines the projection signal's received $\vec{S_\perp}$ on the plane of vectors $\vec{f_1}$ and $\vec{f_2}$, as shown in Figure 3.

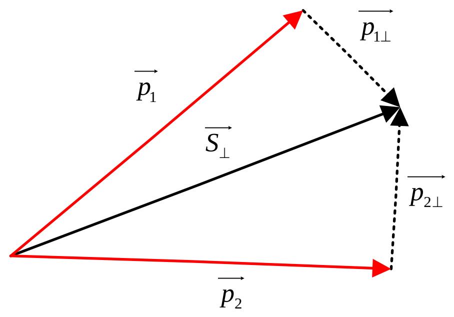

Fig. 3. The intersection of the vectors $\overrightarrow{p_{1\perp}}$ and $\overrightarrow{p_{2\perp}}$

The desired vector can be written as a linear combination of pairs of vectors $\overrightarrow{p_1}$, $\overrightarrow{p_{1\perp}}$ or $\overrightarrow{p_2}$, $\overrightarrow{p_{2\perp}}$ in the following form:

$$\overrightarrow{S_\perp} = \overrightarrow{p_1} + \xi_1 \cdot \overrightarrow{p_{1\perp}} = \overrightarrow{p_2} + \xi_2 \cdot \overrightarrow{p_{2\perp}}, \qquad (11)$$

where ξ_1 and ξ_2 is some complex coefficients that need to be determined.

Given the formulas (9) and (10), (11) can be written as:

$$A = B, \qquad (12)$$

where

$$A = \overrightarrow{p_1} + \xi_1 \cdot \overrightarrow{p_1} - \xi_1 \cdot \frac{\left|\overrightarrow{p_1}\right|^2}{(\overrightarrow{p_1^*}, \overrightarrow{p_2})} \cdot \overrightarrow{p_2}, \qquad (13)$$

$$B = \overrightarrow{p_2} + \xi_2 \cdot \overrightarrow{p_2} - \xi_2 \cdot \frac{\left|\overrightarrow{p_2}\right|^2}{(\overrightarrow{p_1}, \overrightarrow{p_2^*})} \cdot \overrightarrow{p_1} \qquad (14)$$

A vector equality will be true if the coefficients for $\overrightarrow{p_1}$ and $\overrightarrow{p_2}$ in both parts of the equality (12) are equal. Thus, the system of linear equations with regard to ξ_1 and ξ_2 is written in the form:

$$\begin{cases} 1 + \xi_1 + \xi_2 \cdot \dfrac{\left|\overrightarrow{p_2}\right|^2}{(\overrightarrow{p_1}, \overrightarrow{p_2^*})} = 0 \\ 1 + \xi_2 + \xi_1 \cdot \dfrac{\left|\overrightarrow{p_1}\right|^2}{(\overrightarrow{p_1^*}, \overrightarrow{p_2})} = 0 \end{cases} \qquad (15)$$

By entering the notation μ_1 and μ_2 the system (15) can be written as:

$$\begin{cases} 1 + \xi_1 + \mu_2 \cdot \xi_2 = 0 \\ 1 + \xi_2 + \mu_1 \cdot \xi_1 = 0 \end{cases} \qquad (16)$$

The solution of this system is relatively ξ_1 and ξ_2 written as:

$$\xi_1 = \frac{1 - \mu_2}{\mu_1 \cdot \mu_2 - 1}, \qquad (17)$$

$$\xi_2 = \frac{1 - \mu_1}{\mu_1 \cdot \mu_2 - 1} \qquad (18)$$

As a result of further mathematical solutions, substituting (17), (18) in (11), given the replacements in (16), and substituting (5) and (6) in (11), the desired projection signal's received on the plane of the vectors $\overrightarrow{f_1}$ and $\overrightarrow{f_2}$ is written as:

$$\overrightarrow{S_\perp} = \alpha_1(\overrightarrow{f_1}, \overrightarrow{f_2}, \overrightarrow{S^*}) \cdot \overrightarrow{f_1} + \alpha_2(\overrightarrow{f_1}, \overrightarrow{f_2}, \overrightarrow{S^*}) \cdot \overrightarrow{f_2}, \qquad (19)$$

where

$$\alpha_1(\overrightarrow{f_1}, \overrightarrow{f_2}, \overrightarrow{S^*}) = \frac{\left|\overrightarrow{f_2}\right|^2 (\overrightarrow{f_1}, \overrightarrow{S^*}) - (\overrightarrow{f_2}, \overrightarrow{S^*}) \cdot (\overrightarrow{f_1^*}, \overrightarrow{f_2})}{\left|\overrightarrow{f_1}\right|^2 \left|\overrightarrow{f_2}\right|^2 - (\overrightarrow{f_1^*}, \overrightarrow{f_2}) \cdot (\overrightarrow{f_1}, \overrightarrow{f_2^*})}, \qquad (20)$$

$$\alpha_2(\overrightarrow{f_1}, \overrightarrow{f_2}, \overrightarrow{S^*}) = \frac{\left|\overrightarrow{f_1}\right|^2 (\overrightarrow{f_2}, \overrightarrow{S^*}) - (\overrightarrow{f_1}, \overrightarrow{S^*}) \cdot (\overrightarrow{f_1}, \overrightarrow{f_2^*})}{\left|\overrightarrow{f_1}\right|^2 \left|\overrightarrow{f_2}\right|^2 - (\overrightarrow{f_1^*}, \overrightarrow{f_2}) \cdot (\overrightarrow{f_1}, \overrightarrow{f_2^*})} \qquad (21)$$

The program model is implemented in the programming language and consists of software functional blocks that perform digital signal processing.

The model works with a signal at zero intermediate frequency. The signal is sampled at a frequency of 1MHz. Quantization errors are not taken into account.

Figure 4 shows the functional program diagram of the developed model.

The developed information transmission system includes the following models:

- Transmitter's model.
- The model of signal transmission and spaced reception.
- Interference's model.
- The model of Receiver.

The interference and noise model simulates interference and noise signals in the form of white Gaussian noise. The interference generator uses a built-in software Gaussian random number generator, which gives a uniform spectrum and independent samples. Unlike a noise signal, an interference signal is a source from a point coherent signal and the signals are correlated at the receiver inputs.

The model uses codes with a low density of parity checks (LDPC code) with a block length of 2048 bits. The practice of coding based on LDPC has shown a significant increase in noise immunity compared to the protection used in DVB-T. The gain in the signal-to-noise level due to LDPC encoding can be up to 3dB for a typical error level and with the same proportion of control characters.

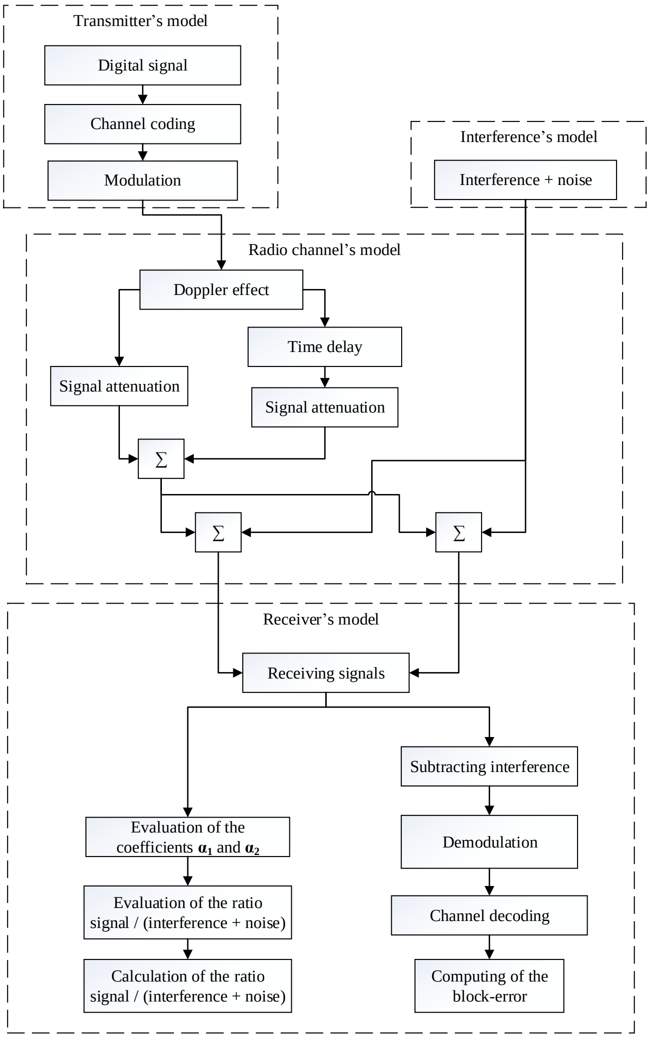

Fig. 4. Functional scheme of the software

The signal-to-noise ratio for the demonstrated model is calculated according to:

$$SNR = \frac{P_S \cdot |\alpha_1 \cdot k_1 + \alpha_2 \cdot k_2|^2}{P_\varphi \cdot |\alpha_1 \cdot \rho_1 + \alpha_2 \cdot \rho_2|^2 + P_{n1} \cdot |\alpha_1|^2 + P_{n2} \cdot |\alpha_2|^2} \quad (22)$$

where P_S and P_φ - signal power and interference, normalized to a single power, P_{n1} and P_{n2} - noise power in the first and second channels, normalized to a single power.

III. Results

The dependence's graph of the block-error probability on the signal / (interference + noise) ratio is shown in Figure 5. According to this graph, the simulated system works effectively at the level of interference exceeding the useful signal up to 26.5dB. A sharp decrease in the probability of a block error at the signal / (interference + noise) ratio of minus 27dB is due to the operation of the LDPC codec.

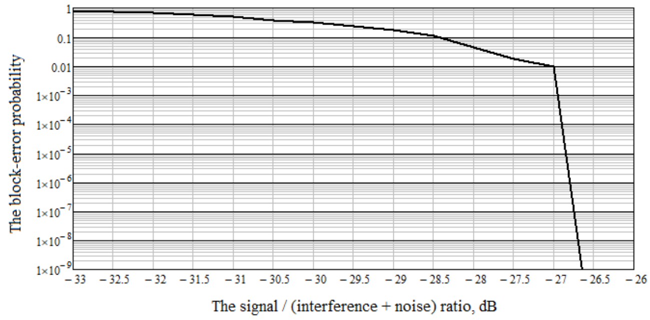

Fig. 5. Block-error rate curve for model.

IV. Conclusions

The problem of resistance's noise of the current systems information transmission is application of methods' and algorithms' with inefficient use of frequency range radio channel's. Commonly, they are focused on the known parameters of interference on the system, such as the spectrums' region, occurrence's time, the action's direction, polarization, etc. The obtained results of a software model show transfer possibility of a narrowband signal when exposed to broadband interference with unknown parameters and exceeding the useful signal level up to 26.5dB.

In contrast to the proposed method, methods using an adaptive spatial filter based on an antenna array [2, 3] are subject to complex calculations due to the operation of calculating the inverse correlation matrix. In addition, when interference acts in the direction of the useful signal, the effectiveness of these methods tends to zero. The proposed method in contrast to the adaptive nonlinear filtering method [4] is able to work in the incoherent band path of the receiver, and in contrast to [5], is resistant to the multipath effect, which leads to signal fading.

A series of articles will be written on this research topic. The results of the research obtained both in laboratory with the use of a radio system layout and under conditions close to real will be published.

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