Conclusion. Fig.5 (b) shows that the radar data matrix has some irrelevant distortions related to the sampling frequency of echoes and the viewing angle change.

SAR programming model allows to simulate the formation of radar images in rectilinear uniform motion of the aircraft over the site of the relief area. The model is the basis for the development of fast algorithms for the synthesis of radar images, for radar data correction algorithms in the drift trajectory of the aircraft, as well as for filtering algorithms. It helps to assess the technical requirements for the SAR functional blocks at its implementation.

REFERENCES

1. Кондратенков, Г.С. Радиолокационные системы дистанционного зондирования Земли : учеб. пособие для вузов / Г.С. Кондратенков ; под ред. Г.С. Кондратенкова. – М. : Радиотехника, 2005. – 368 с.
2. Беляев, А. Современные устройства цифровой обработки сигналов. Вместе или врозь? / А. Беляев, Т. Солохина, В. Юдинцев // Электроника: Наука, Технология, Бизнес. – 2009. – № 1. – С. 28–35.
3. Радиолокационные системы землеобзора космического обзора / В.С. Верба [и др.] ; под ред. В.С. Вербы. – М. : Радиотехника, 2010. – 680 с.
4. Применение цифровой обработки сигналов [Электронный ресурс]. – Режим доступа: http://lib.alnam.ru/book_acos.php?id=86. – Дата доступа: 13.12.2015.

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THE ANALYSIS OF METHODS OF SIGNALS RECEPTION IN TECHNICAL CHANNELS OF INFORMATION LEAKAGE

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According to a digital signal definition an actual problem is to establish numerical values of criterion of speech signal security. The criterion of speech signal security should correspond sufficiently to the criterion of a speech signal in a digital form.The criterion sets limit, which value is a border between presence and absence of a speech signal in a leakage channel.

The transfer of **digital information** has obvious advantages if compare it to transfer of analogue information. Methods of protection from information leakage became more complicated thanks to obtaining of high quality and high level of reliability of **digital information** transfer.

The primary analog speech signal is biological. Thereupon its transformation to the digital form provides naturalness of the restored speech for qualitative perception. Transfer of speech signals in the digital form through communication channels (data transmission) is caused by a number of transformations because digital signals should be transferred through analog channels (tone frequency channel), i.e. correspond to spectral efficiency.

The purpose of the present research is to substantiate the possibility of application of binary signals for security estimation of digital signals with various structural characteristics and parameters in channels of information leakage.

Thereto it is necessary to take the analysis of structural characteristics and parameters of signals in information leakage channels.

General structure of communication system is a complex of message source, transmission channel and addressee (receiver). At this point the channel is a part of transmission system, kind and characteristics of which are specified and their change is complicated or simply impossible. The problem solved by transmission system is to transfer a message m from an originator to an addressee. As a rule, the source message is presented in such form in which its effective channeling is impossible. Therefore transfer and reception devices are usually included in the system. Given devices transform a message m into a signal s and transform an accepted signal r into an accepted message $m!$.

Thus transformation $m \rightarrow s$ carried out by the transmitter is biunique and determined; transformation $s \rightarrow r$ defined by the channel is casual; transformation $r \rightarrow m!$ carried out by the transmitter is determined but not biunique.

If a source generates messages from final set it is called a source of discrete messages, otherwise a source is called a source of continuous messages.

Process of transformation of a signal by message is called modulation. Change of parameters of a signal according to the message which is subject to transfer is carried out during modulation. If a set of signals formed during modulation is final modulation, it is called discrete or digital.

Channel (antinoise) coding-decoding is applied to provide maximum reliability of transfer. When using antinoise coding speed of transfer decreases at the expense of transfer of the redundant symbols, which allows correcting the errors arising in the channel.

The continuous message of a source is exposed to analog-digital transformation. Digital bitstream is formed as a result of this operation and information leakage is possible. After reception of a digital stream the continuous message of a source is restored with the help of digital-to-analogue transformation.

The general structure of the technical channel of information leakage is a set of a signal source, environment of distribution and tool of secret removal of the information (a receiver).

Model of speech signal leakage channel is radiation of a speech signal in the digital form. Direct and reverse transformation of a speech signal is the source of that radiation. The simplified model of leakage channels of a speech signal in the digital form is reflected on Figure 1.

Possible leakage channels of the speech information, formed by extraneous fields of a speech signal are presented in figure 2:

leakage channel 1 – during the moments of time of transformation of a speech signal in a signal in the digital form;

leakage channel 2 – during the moments of time of transformation of pulse group in a harmonic signal (manipulation);

leakage channel 3 – by transfer of a speech signal in propagation medium;

leakage channel 4 – during the moments of time of transformation of a harmonic signal in pulse group (demodulation);

leakage channel 5 – during the moments of time of transformation of an initial speech signal from numerical order;

leakage channel 6 – the leakage channels caused by the higher harmonics of sampling frequency, modulated by modulating signals which are formed by the digital-to-analog converter;

analog leakage channel 1, 2 – information leakage channels, which are characteristic for an acoustic speech signal in the form of magnetic and electric extraneous fields.

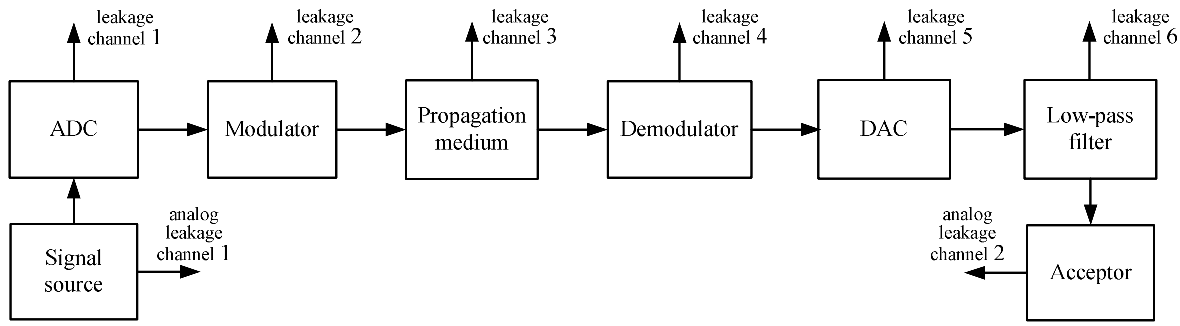

Fig. 1. Model of leakage channels of a speech signal in the digital form

In picture 1«ADC» – analog-to-digital converter, «DAC» – digital-to-analog converter.

The problem consists in a choice of the validated single criterion for analog and digital speech. Complexity of this problem is the variety of forms of presentation of digital signals. It is rational to consider from their variety signal constructions, orthogonal at which the cross-correlation coefficient is equal to zero. To such signals rate FM-signals.

Other large class of signals is that at which the cross-correlation coefficient is not equal to zero. It is necessary to carry PM-signals to such signals.

Data transmission is carried out by transfer of signal elements of a direct current or elements of an elementary sinusoidal signal. Bit timing possesses the important role in formation of data signal.

It is necessary to find out useful signals against the background random noise in a point of reception of extraneous fields of a transferred signal. To distinguish and find out them the more difficultly, than more intensively a noise background.

The noise background masks useful signals, being imposed on them in a random way and doing necessary some "guessing" therefore the signal can be accepted for other signal or in general is passed, and strong emission of noise is accepted for a useful signal. Thus, when the useful signal is found out correctly, dimension of its parameters as a result of noises is made with this or that error.

At reception there are two tasks:

1) optimum detection of a useful signal against hindrances;

2) optimum dimension of some parameters of a useful signal in the presence of hindrances.

For the purpose of definition of presence of information in leakage channels it is offered to establish a source of a measuring signal in a point of placing a source of extraneous fields of a signal data-transmission system, and the receiver – in a point of reception of extraneous fields of a transferred signal. The optimum receiver with the matched filter can serve as the receiver of such signal.

The block diagram of such demodulator is shown in Figure 2.

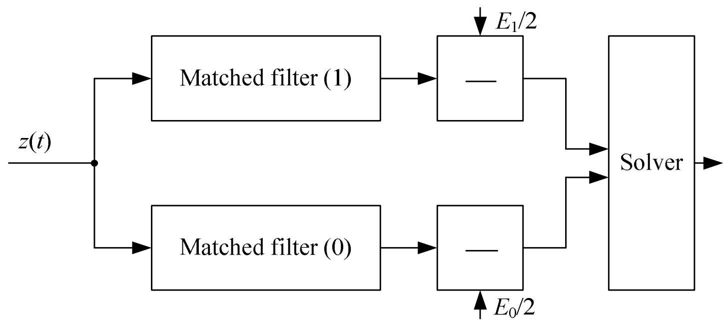

Fig. 2. The optimum demodulator on the basis of the matched filter

In figure 2 «–» – subtractor.

At giving on an input of the matched filter of an accepted signal $z(t)$ voltage on an output of the filter at the moment of time $t = T$ $y(T) = \int_0^T z(\tau) g(T-\tau)\,\mathrm{d}\,\tau$, where $g(\tau)$ – pulse response of the filter.

The frequency characteristic of the matched filter with the pulse response $g(t) = as(t_0 - t)$, is defined by Fourier transform

$$K(j\omega) = \int_{-\infty}^{\infty} g(t)\,\mathrm{e}^{-j\omega t}\,\mathrm{d}t = a\int_{-\infty}^{\infty} s(t_0 - t)\,\mathrm{e}^{-j\omega t}\,\mathrm{d}t = a\int_{-\infty}^{\infty} s(\tau)\,\mathrm{e}^{-j\omega(t_0-\tau)}\,\mathrm{d}\tau = a\hat{S}(j\omega)\,\mathrm{e}^{-j\omega t_0},$$

where $\hat{S}(j\omega)$ – the conjugate complex function to spectral density of a signal $s(t)$.

Consequently, the amplitude-frequency characteristic of the matched filter is defined by amplitude spectrum of a signal $s(t)$ accurate within coefficient a, and its phase-frequency characteristic is inversely on a sign to a phase spectrum of a signal $s(t)$. Owing to it all spectrum component of received signal develop in a phase and give the peak response during the moment t_0.

Decision-making of the optimum demodulator is based on inequality check

$$\int_0^T z(t)s_1(t)\mathrm{d}t - 0,5E_1 > \int_0^T z(t)s_0(t)\mathrm{d}t - 0,5E_0.$$

The symbol «1» is registered at inequality performance, and otherwise «0».

Quality of transfer of the speech signal transformed to the digital form, estimate bit-error probability, and not more low10^{-5}.

Limiting value of bit-error probability (Shannon's limit) is established in work [6] for a case of the analysis of a signal with white Gaussian noise. In addition, the limiting noise immunity for various signals is established in work [7], at such noise immunity of a signal at substitution h = ln2 Shannon's limit is defined.

Normalize numerical value of an indicator of an estimation of security in the digital form probably to establish in the form of bit-error probability and to define its dependence on coefficient of speech intelligibility.

It will allow to establish security of channels of information leakage in the automated mode by single criterion – speech intelligibility. For this purpose information leakage channels are proved, which are inherent for speech signals in the digital form.

REFERENCES

1. Железняк, В.К. Защита информации от утечки по техническим каналам : учеб.пособие / В.К. Железняк. – СПб. : ГУАП, 2006. – 188 с.
2. Савищенко, Н.В. Многомерные сигнальные конструкции: их частотная эффективность и потенциальная помехоустойчивость приема / Н.В. Савищенко ; под ред. Д.Л. Бураченко. – СПб. : Изд-во Политехн. ун-та, 2005. – 420 с.

3. Теория электрической связи : учеб. для вузов / Д.Д. Кловский [и др.] ; под ред. Д.Д. Кловского. – М. : Радио и связь, 1999. – 432 с.
4. Теория электрической связи : учеб. пособие / К.К. Васильев [и др.] ; под общ. ред. К.К. Васильева. – Ульяновск : УлГТУ, 2008. – 452 с.
5. Помехоустойчивость и эффективность систем передачи информации / А.Г. Зюко [и др.] ; под общ.ред. В.И. Борисова. – М. : Радио и связь, 1985. – 272 с.
6. Скляр, Б. Цифровая связь. Теоретические основы и практическое применение / Б. Скляр ; пер. с англ. ; под ред. А.В. Азаренко. – 2-е изд. – М. : Вильямс, 2007. – 1104 с.
7. Варгаузин, В. Вблизи границы Шеннона / В. Варгаузин // ТелеМультиМедиа. − 2005. − № 3. − С. 3−10.

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ANTENNA SIMULATION IN ANTENNA MAGUS SOTWARE PACKAGE

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This article describes the software package Antenna Magus. The article shows the main features of the program, discusses the parameters that can be changed during the simulation of the antenna. Also, it considers the diagram, which the program during the simulation of the antenna.

Product Antenna Magus, developed by Magus (Pty) Ltd and is intended for the design and simulation of various types of antennas, is a database of different antennas, from which the user can select the appropriate parameterized model and export it to the package CST MICROWAVE STUDIO, which is then executed her modeling and optimization.

The package is aimed at the broad masses of developers antennas experts on EMC, as well as system integrators, we have the estimate location of antennas on large objects. The program analyzes the available Antenna Magus design goals and performs intelligent choice lacking initial parameters.

Subsequently, the user can manually adjust the proposed program settings. To implement the technology Smart Design All design goals in Antenna Magus organized into groups. For each of these groups a user is a calculator of parameters, which allows different ways to implement the task, grouping initial targets in the project requirements. Smart Design Technology enables the user to specify only those parameters which are known to him, and fully trust Antenna Magus to get the final results at a design stage.

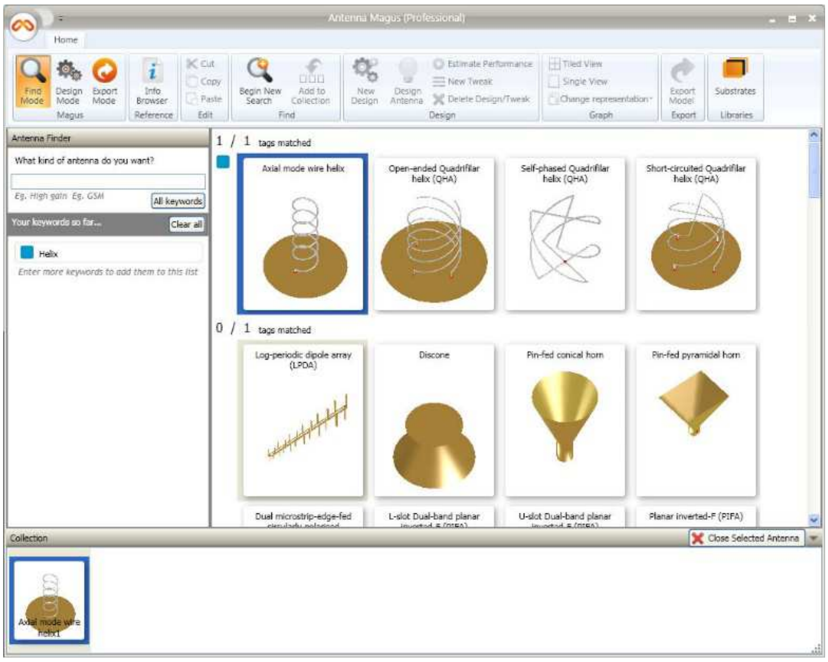

Fig. 1. Cover Page of software package Antenna Magus