E. V. Malinkin
Siberian State University of Telecommunications and Computer Science, Russia, Novosibirsk

INVARIANT SIGNALING WITH PROCESSING IN THE FREQUENCY AREA

A method of sending signals to the subsequent processing in the frequency domain is offered. To implement such a treatment, a structure is proposed based on the direct and inverse Fourier transformation with an element of division, making the division of information signals and training signals possible.

Keywords: immunity, invariant, invariant relative amplitude modulation, the probability of pairing transition, the signal/noise.

The author [1] suggested a relative amplitude modulation (RAM). Its essence lies in the fact that the previous assumption is core to the next, and their attitudes at the reception, despite the impact of multiplicative noise with the channel transmission coefficient K, equal to the ratio of these parcels for transfer. The RAM can significantly reduce the influence of multiplicative noise. However, the influence of additive noise has not been eliminated.

The author of this work used the idea of the RAM, but bearing messaging of equal size allocated in the same sequence of training signals as their subsequent averaging to reduce the impact of additive noise. Information signals of various amplitudes are identified in the information sequence.

The relation of the information sequence signals to value of the average training signals also reduces the influence of a multiplicate hindrance.

This method of signal formation on transfer and their demodulation on the reception party allows improving the noise stability of the offered structure at least by two orders in comparison with the classical RAM systems.

The given work is devoted to the synthesis of the processing of the information algorithm by the invariant system of information transfer in frequency areas; and to the analysis of the received technical characteristics.

We have an analogue communication channel with a pass-band from $f_{\min}$ to $f_{\max}$ on which there is transferred peak-modulated rectangular bending around the signals. The sequence of information part signals with those of the training part is united into blocks.

Transferred signals are treated to actions of hindrances: a multiplicate, described by the change of a channel broadcast of communication factor $k(t)$ at certain frequencies; and additive, representing white noise with no correlated readout and a dispersion σ^2 (while modelling the dispersion accepts a value equal to 1). Besides this, the accepted signal is influenced by peak-frequency and fazo-frequency distortions of the communication channel.

Let's consider that the communication channel is subject to a smooth general fading and consequently, it is possible to allocate the interval's stationary channel properties. Particularly, to allocate intervals at which the factor of a channel broadcast of communication is constant, and the amplitude of the readout AFC is constant (does not vary).

Let's suppose that the duration of the transferred block does not exceed the duration of an interval stationarity.

To resist the additive hindrance accumulation the averaging of training signals is entered. For this purpose, arithmetically there are readouts of training signals with the same name and then this sum is divided by the sum of the composed.

It is necessary to synthesise processing algorithms of the signal steadily against the influence of the hindrance complex (multiplicate and additive) and also to synthesize the system's structure of the information transfer, allowing demodulate transfer signals according to the offered algorithm with the subsequent analysis of quantitative characteristics of the information transfer system.

Synthesis of the signal processing algorithm. To solve the task it is required to first, generate a signal transfer (opposite side) and, secondly – to synthesize the algorithm of processing on reception.

We shall generate the transfer signal in blocks. Each block will consist of a training signal (the pilot of the signal) and an information signal. The training signal will be identical on all subsequent blocks. For the simplicity of training signal formation $S_{tr}(nT)$ we shall present it in the form of readout of an equal amplitude. According to the laws of digital signals to training signal processing $S_{tr}(kT)$ there corresponds a power spectrum $S_{tr}(jk)$. A power spectrum of an information signal shall be presented in:

$$S_{inf}(jk) = S_{tr}(jk) S_{mod}(jk), \quad (1)$$

where $S_{tr}(jk)$ – is the power spectrum of the training signal, $S_{mod}(jk)$ – is the power spectrum of the transfer signal.

In fig. 1. the structure of the transferring part of the invariant system is presented.

Thus, the power spectrum of a transfer signal at input of block OBPF shall look like:

$$\left[S_{tr}(jk), S_{inf}(jk)\right]_1;\ \left[S_{tr}(jk), S_{inf}(jk)\right]_2;\ \ldots;$$
$$\left[S_{tr}(jk), S_{inf}(jk)\right]_i;\ \ldots$$

In block the OBPF transformation of the power spectra readout to the readout of the transfer signal $S_{inf}(nT)$ is accomplished. By means of the modulator the modulation of the information signal by means of AM modulations is done.

The first, during T_{tr}, transfer the radio impulses of identical amplitude representing are signals of training sequence. Values of readout amplitudes of the training signals on a communication channel exit are remembered subsequently in a memory element and serve for the demodulation together with values of readout of information signals amplitudes. Besides, the values of amplitudes of signals of training sequence readout on a communication channel exit are used for the definition of channel broadcast of communication factors. During T_{inf}, the sequence of radio impulses for various amplitudes, representing the sequence of information signals is transferred. Their quantity is equal to the quantity of training signals. The transfer time for information sequence is equal to the time of training sequence transfer and consequently $T_{inf} = T_{tr}$. Then the transfer of training sequence follows its information and so on, until the end of the transfer of the whole block.

After the reception of training signals, the value of the readout of training signals amplitudes in a memory element arithmetically develops and is averaged by the quantity composed for the purpose of resisting to additive hindrance, which raises noise stability in the whole system of information transfer.

On fig. 2 the readout of information and the training sequence of signals (fig. 2, *a*) with their power spectra (fig. 2, *b*) are presented. It is necessary to notice, that the amplitudes of readout for a training signal on a communication channel input are equal among themselves in time space; the values of amplitudes of power spectra readout for training signals are equal to the communication channel input. Amplitudes of signals of information sequence readout in a communication channel input differ in size and in time, and have various values of power spectra readout of signal amplitudes.

The set N of readout for an information part in time space corresponds to N readout of its power spectrum.

The similarly set N of readout for a training signal in time area corresponds to N readout of its power spectrum.

For a signal of information sequence of i block, its power spectrum will be equal to:

$$S_{i\,out}(jk) = S_{i\,in}(jk) \cdot k_i(jk) \cdot H_i(jk) + N_{1i}(jk)\,, \qquad (2)$$

where k – is a harmonic number; $S_{i\,out}(jk)$ – is a power spectrum of a signal on a communication channel exit on i block; $S_{i\,in}(jk)$ – is a power spectrum of a signal on a communication channel input on i block; $k_i(jk)$ – is a power spectrum of a multiplicate hindrance on i block; $H_i(jk)$ – is the transfer characteristic of the channel on i block of processing; $N_{1i}(jk)$ – is the spectral density of capacity of additive noise of a signal of information sequence on i block of processing.

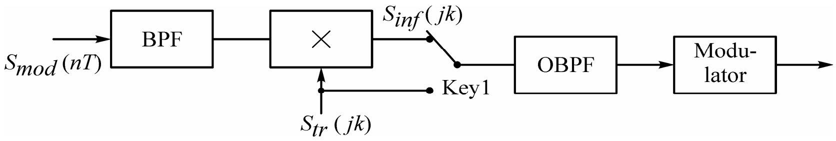

Fig. 1. Sending device structure

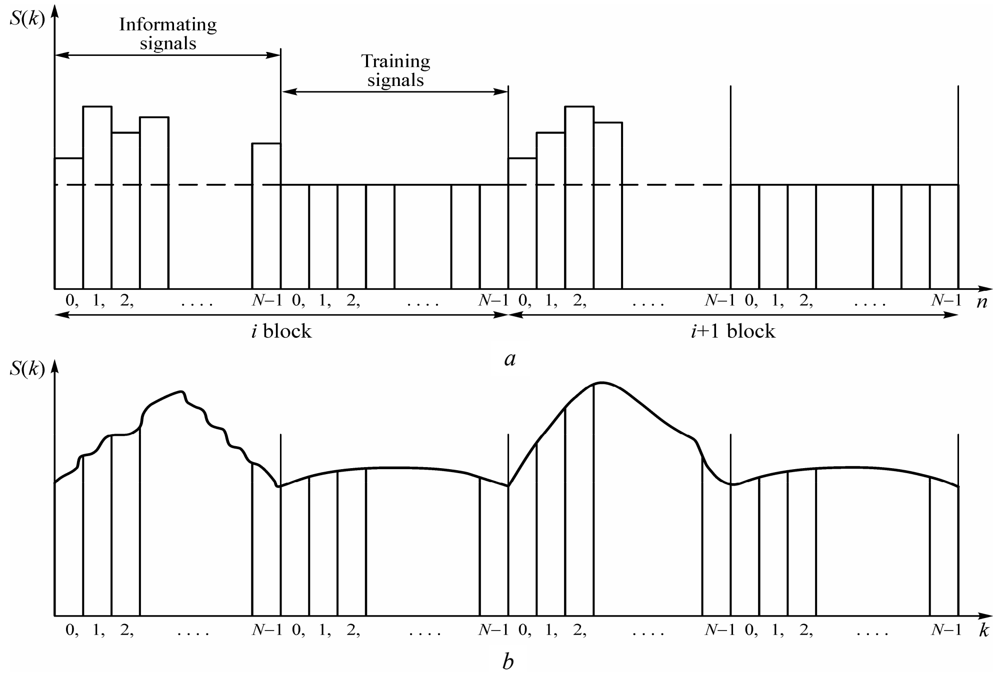

Fig. 2. Information and training sequences and their power spectra

For a signal of training sequence of i block its power spectrum will be equal to:

$$S_{i\,tr.out}(jk) = S_{i\,tr.in}(jk)\cdot k_i(jk)\cdot H_i(jk) + N_{2i}(jk)\,, \qquad (3)$$

where $S_{i\,tr.out}(jk)$ – is a power spectrum of a training signal on a communication channel exit; $S_{i\,tr.in}(jk)$ – is a power spectrum of a training signal on a communication channel input; $N_{2i}(jk)$ – is the spectral density of capacity of additive noise of a signal of training sequence on i block of the processing.

It is necessary to notice, that after the averaging of training signals, the value of the amplitudes of readout $N_{2i}(jk)$ it is much less, than the values of amplitudes of readout $N_{1i}(jk)$ at the expense of the accumulation effect with averaging. Values of the readout amplitudes $N_{1i}(jk)$ are more than values of the amplitudes of readout $N_2(jk)$ by 1 000 times, i.e. $N_{1i}(jk) = 1\,000 N_{2i}(jk)$ as the quantity of averaging equal to 1 000 [2].

As it was said earlier, the modulating parameter on transfer modulates the relation of power spectra of information and training parts. For the demodulation of reception signals it is necessary to divide a power spectrum of an information part into a power spectrum of a training part. In result we will receive:

$$\begin{aligned} S_{i\,mod.out}(jk) &= S_{i\,out}(jk) \big/ S_{i\,tr.out}(jk) = \\ &= \frac{S_{i\,in}(jk)\cdot k_i(jk)\cdot H_i(jk) + N_{1i}(jk)}{S_{tr.in}(jk)\cdot k_i(jk)\cdot H_i(jk) + N_{2i}(jk)}\,. \end{aligned} \qquad (4)$$

Due to the fact that the readout values of amplitudes of a power spectrum of additive noise in training sequence is essentially less than the values of amplitudes of the power spectrum readout of additive noise of information sequence, the size $N_{2i}(jk)$ in expression (4) can be neglected [3].

Then expression (4) after elementary transformations is reduced to:

$$S_{i\,mod.out}(jk) = S_{i\,mod.in}(jk) + \frac{N_{1i}(jk)}{S_{i\,tr.in}(jk)\cdot k_i(jk)\cdot H_i(jk)}\,, \qquad (5)$$

where $S_{i\,mod.out}(jk)$ – is the power spectrum of modulating sequence on a communication channel exit on i block; $S_{i\,mod.in}(jk)$ – is the power spectrum of modulating sequence on a communication channel input on i block.

According to Parseval's theorem the energy of the signal calculated in time space is equal to the energy of a signal calculated in frequency space. Therefore the relation of the information signal energy to the energy of a training signal remains both in time, and in frequency space. So, the results of processing in time and frequency space yield identical result and, consequently noise stability estimation will be made in time space.

For this purpose, look at fig. 3, which shows the principle of signal processing.

At point "a" information and training signals are given in the form of a readout in time area. At exit BPF ("b") this signal is presented by the readout of a power spectrum. At point "c" the signal is presented by readout of the power spectrum of an information part, and in point "d" – is the readout of the power spectrum of a training part. In point "*e*" – is the readout of the power spectrum of modulating sequence together with the readout of a power spectrum of the additive noise, the size of which is estimated by formula (5).

At point "f" is the signal before modulating the sequence of readings presented in the time domain.

To calculate the probability of erroneous reception of the time domain, we introduce a notion of the invariant. Given the use of the relative value of the invariant amplitude modulation; this can be found as [2]:

$$\mathrm{INV}_l = \frac{\sum_{i=1}^{n} \mathrm{INV}_l \cdot S_{tr}}{\sum_{j=1}^{N} S_{tr}} = \frac{N\cdot \mathrm{INV}_l \cdot S_{tr}}{N \cdot S_{tr}}\,.$$

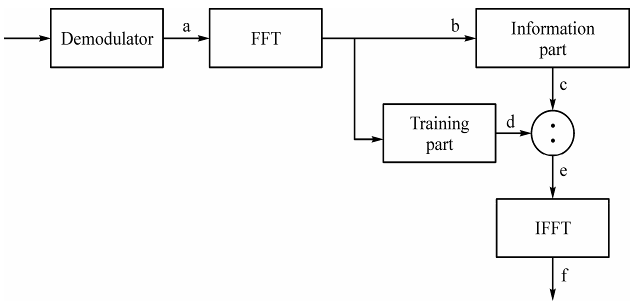

Fig. 3. Block diagram of the ISPR:
FFT – fast Fourier transform, *IFFT* – inverse Fourier transform

We assess the probability of the enumeration invariants by selecting the first pair as the smallest value of the invariant – $\mathrm{INV}_1 = 1$ and the second invariant is taken from 2–6. When passing an invariant by the other there is an error in the reception. Instead of the first invariant of the pair it takes a second, compared to certain probability, and vice versa, instead of the second invariant of the pair being compared invariants is taken first with the same probability.

We perform the calculation of the probability of the erroneous reception. For this we use the well-known approach [2]:

$$P = P_1 \int_{-\infty}^{z_p} W_2(z)\,dz + P_2 \int_{z_p}^{\infty} W_1(z)\,dz\,, \qquad (6)$$

where P – is the transition probability of the first invariant of the second and vice versa; P_1 – is the probability of occurrence of the first invariant, P_2 – is the probability of a second constant, the first integral – is the probability of occurrence of the second invariant, when sent to the first and the second integral – is a probability of occurrence of the first invariant when sent second invariant; z_p – is the threshold needed to calculate P, if known P_1 and P_2.

The value of z_p is determined by using the best Bayesian estimates by minimizing P to z_p. We believe in the equally probable appearance of the first and second invariants, so we choose $P_1 = P_2 = 0{,}5$.

As seen from expression (6), it is necessary to know the analytical expression of $W_1(z)$ and $W_2(z)$.

For coherent reception with a sinusoidal subcarrier calculation, the functions $W_1(z)$ and $W_2(z)$ are known and are given in [2].

In our case, a rectangular envelope of the signal is allocated using the synchronous detector, and hence, the interference has a normal distribution [4].

Therefore, in our case, you can use the same approach for finding the analytical expression of probability density estimates of the invariant [2].

Thus, the value assessment of the invariant in our system is calculated as:

$$\mathrm{INV}_l^* = \frac{\sum_{i=1}^{N}\left(k \cdot \mathrm{INV}_l + \xi(i)\right)}{\frac{1}{L}\sum_{m=1}^{L}\sum_{j=1}^{N}\left(kS_{tr} + \eta(m,j)\right)} \cdot S_{tr}, \qquad (7)$$

where INV_l – l is the transmitted invariant; $\xi(i)$ – i is the value of the Gaussian noise; in the denominator: S_{tr} – is the meaning training signal; $\eta(m, j)$ – j is the value of the Gaussian noise in the m realization of the signal S_{tr}; k – is the coefficient of transmission of the communication channel; N – is the number of samples taken from the envelope INV_l or S_{tr}; L – is the number of training signals.

To calculate P we need to know the expectation and variance of the numerator and the denominator of expression (7).

For their calculation we use the following approach.

The expectation of the numerator (7) will be equal to:

$$m_1 = k \cdot N \cdot \mathrm{INV}_l. \qquad (8)$$

The expectation of the numerator (7) will be equal to:

$$D_1 = N \cdot \sigma^2, \qquad (9)$$

where σ^2 – is the variance of Gaussian noise.

The expectation of the denominator (7) after the transformation will be equal to:

$$m_2 = k \cdot N. \qquad (10)$$

The dispersion of the denominator after transformation will be:

$$D_2 = \frac{N}{LS_{tr}^2} \cdot \sigma^2. \qquad (11)$$

Then the expression of the probability density estimates of the invariant is equal to [3] taking into account expressions (8–11):

$$W(z) = \int_{-\infty}^{\infty} \frac{1}{2\pi\sigma_1\sigma_2} e^{-\frac{(zx - kN\,\mathrm{INV}_l)^2}{2N\sigma_1^2}} e^{-\frac{LS_{tr}^2(x-kN)^2}{2N\sigma_2^2}} |x|\,dx, \qquad (12)$$

where $\sigma_1 = \sqrt{D_1}$; $\sigma_2 = \sqrt{D_2}$; L – is the number of signal processing.

Calculation P performed numerical approximation formula (12).

The proposed system was compared with a classical coherent system.

The probability of pairing transition was calculated in both cases for the same values of h signal/noise ratio, which is calculated by formula:

$$h^2 = \frac{\sum_{i=1}^{N} k^2\,\mathrm{INV}_{li}^2\,\Delta t}{N\Delta t\sigma^2} = \frac{k^2\,\mathrm{INV}_l^2}{\sigma^2}.$$

Thresholds were calculated by minimizing Z_p P in formula (6). For $k = 1$ and $\mathrm{INV}_1 = 1$, $\mathrm{INV}_2 = 2, 3, 4, 5, 6$ calculations give the result Z_p = 1,5; 2; 2,5; 3; 3,5.

For k = 0,7 and INV*1* = 1, INV*2* = 2, 3, 4, 5, 6 calculations give the result Z_p = 1,5; 2; 2,5; 3; 3,5.

The simulation results are shown in fig. 4 and 5.

Invariant feature of any system based on the principle of relativity invariant amplitude modulation is that the channel transmitted amplitude-modulated signals, producing the INV– and S_{tr}.

The transfer of these signals is provided on the basis of classical processing algorithms, which are generally with low immunity [4].

And, only after processing these signals in accordance to the algorithm of private expression (7), we obtain the invariant essential to the number, rather than signal.

As seen in fig. 4 and 5, the probability of pairing transition of one invariant in the other at high signal-noise ratio is defined by (10^{-30}–10^{-80}). The same values of the SNR probability of erroneous reception for a single character in the classical systems lie in the range (10^{-6}–10^{-10}).

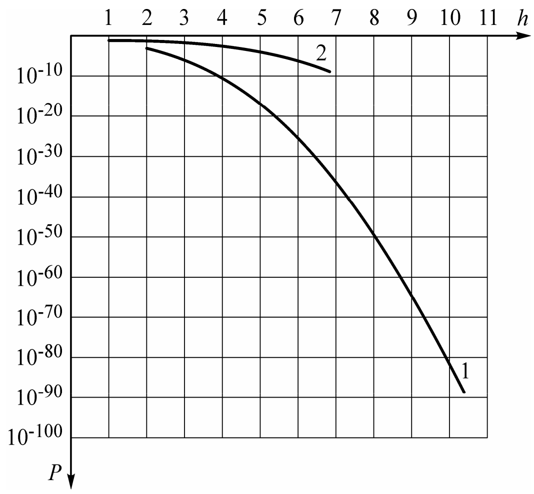

Fig. 4. The probability of pairing transitions for one invariant inside the other inthe following prescribed conditions: $k = 1$; $\mathrm{INV}_1 = 1$; INV_2 = 2, 3, ... , 10 curve *1* – is the coherent reception, invariant relative amplitude modulation, rectangular envelope, curve *2* – is the classical relative amplitude modulation

According to the author, the immunity investigated invariant system should be compared to the noise immunity similar invariant systems, which will be done in subsequent papers.

We have proposed an invariant coherent information transfer system and defined its qualitative characteristics.

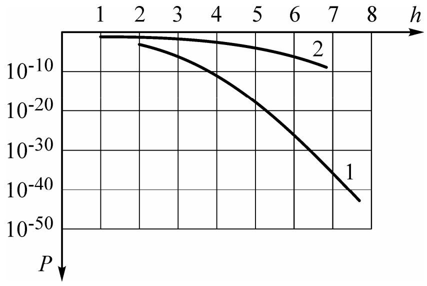

Fig. 5. The probability of pairing transition of one invariant inside the other inthe following prescribed conditions: k = 0,7; INV_1 = 1; INV_2 = 2, 3, ... , 10 curve *1* – is the coherent reception, invariant relative amplitude modulation, rectangular envelope, curve *2* – is the classical relative amplitude modulation.

The analysis of this system has shown that it has a high level of noise immunity. The chance of error in the classical algorithm with a relative amplitude modulation of at least two orders of magnitude is greater than the probability of pairing transition in the invariant system. Therefore, this system can be applied in telecommunication systems, remote control systems, and other systems, which demand high indicators for noise immunity.

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V. B. Malinkin, H. F. Kurash, O. V. Soboleva, E. V. Malinkin
Siberian State University of Telecommunications and Computer Science, Russia, Novosibirsk

INVARIANT METHOD OF INFORMATION TRANSMISSION IN FIBER-OPTIC TRANSMISSION SYSTEMS

We have synthesized a method of controlling the distortions introduced by fiber-optic communication lines. This method is based on the use of invariant equality. Its main technical characteristics have been determined.

Keywords: invariant fiber-optic transmission system.

Classical AM Modulation in fiber-optic transmission systems (FOTS) is used to transfer information signaling in most cases.

The probability of erroneous reception in regenerators is 10^{-10}, according to the recommendations of the International Telecommunication Union (ITU) [1]. More recent ITU [2] recommendations have proposed to apply device error protection (RCD) the performance of which is based on a special transmission signal coding by means of cyclic codes. It is difficult to create a residual operating in real time at a speed of 10 Gb / s and more.

Meanwhile, the reduction of error probability can be achieved in other ways. One is suggested below.

We have the FOTS (fig. 1). A laser is used as a transmitter. A photo detector is used as a receiver. The second window of transparency is used to send the information signal.

The information signal transmission algorithm must be synthesized based on the invariant method of information processing.

Fig. 1 shows the structure of FOTS, which includes the transmitting and receiving devices and the fiber-optic transmission.

It should be noted that the cross-cutting path of the FOTS at the second window of transparency [3] is linear, provided the power output of the transmitter to not exceed the permissible value of 1 mW. The Z-transform signal reception $Y(Z)$ on the output of the FPU will be equal to (for i-volume processing unit):

$$Y_i(Z) = \left[G(Z) \cdot H_0(Z) \cdot H_1(Z) \cdot H_2(Z)\right]_i, \qquad (1)$$

where $G_i(Z)$ – is the Z-transform of the signal transmission at the i-volume processing unit; $H_{0i}(Z)$ – the transfer characteristic of the signal shaper at the i-volume unit; $H_{1i}(Z)$ – is the transfer characteristic of fiber-optic transmission at the i-volume unit; $H_{2i}(Z)$ – the transfer characteristic of the unit (FPU) on the i-unit.

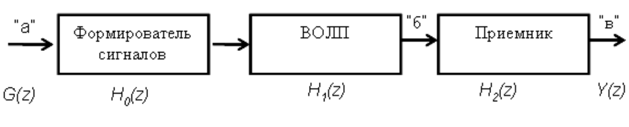

Fig. 1. The FOTS structure