Protection of data transmission systems from the influence of intersymbol interference of signals

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Abstract. The problems of creating a regenerator of an optical system for transmitting discrete messages with the aim of protecting against the influence of intersymbol interference of signals using the methods of statistical theory for a mixture of probability distributions are considered. A demodulator adaptation model that calculates the weighted sum of 4 samples of the input signal and compares the resulting sum with a threshold value has been created and studied. The main operations for adaptation of the regenerator during demodulation of signals distorted by intersymbol interference are assigned under certain transmission conditions, which are formulated as the task of evaluating some parameters of the probability distribution of the mixture. A block diagram of a regenerating optical signal in which an algorithm for digital processing of a mixture is implemented is proposed. The analysis is carried out and the requirements for the parameters of the analog-to-digital converter used in the regenerator are determined. The results of modeling digital processing of a discrete test message to verify the process of adaptation of the device to the influence of intersymbol distortions of signals are presented. The block diagram of a digital processing device is substantiated, in which high-speed nodes that perform real-time processing are combined with a microprocessor, which implements an algorithm for adapting the regenerator to specific message transmission conditions.

Keywords: mixture of probability distributions, fiber optic communication lines, intersymbol interference of signals, demodulator adaptation, optical signal regenerator.

1 Using statistical theory for mixtures of probability distributions to eliminate intersymbol interference of signals

1.1 Basic concepts of mixtures

Protection of information systems for transmitting discrete signals from the effects of intersymbol interference (ISI) remains one of the pressing communication problems. ISI causes distortion of the signal corresponding to the *i*-th transmitted symbol, by signals corresponding to the *i*-1, *i* + 1, ... symbol. This makes it difficult for the demodulator to make the right decision about the meaning of each transmitted symbol. To eliminate the influence of ISI, analog and digital signal correctors are used. Analog linear distortion correctors that compensate for the imperfection of the frequency characteristics of the communication channel are used to neutralize ISI. Operations with discrete samples of the received signal are performed by a demodulator, which actually performs the correction functions in the time domain.

The transmission of discrete information is often accompanied by ISI, which may be caused by imperfection of the frequency characteristics of the channel or is artificial. In any case, the presence of ISI makes signal demodulation difficult and forces the use of rather complex processing algorithms. [1].

The principles of constructing correctors and demodulators to neutralize ISI do not take into account the stochastic nature of the signal, which is caused by additional signal distortions due to random noise and the pseudorandom nature of the sequence of transmitted symbols. The concept of ISI neutralizing using correctors loses its meaning in situations where interference is artificially created using signals with a partial response. Solving the problems of ISI is possible while simultaneously performing the tasks of correction and demodulation with minimization of errors. This is possible when using a protection method based on the probabilistic interpretation of ISI phenomena and using the concepts and methods of statistical theory for mixtures [2].

Let a parametric family $\mathbf{F} = \{f(x; A): A \in \mathbf{A}\}$ of probability distributions of random variables be given. The elements $f(x; A)$ of this family are described by an identical known analytical dependence from x (value of a random variable) and differ only by values of some parameters A, which belong to a set $\mathbf{A}$ of admissible values. In the general case, the probability distribution $f_P(x)$ of an infinite mixture from the family $\mathbf{F}$ is defined by the expression:

$$f_P(x) = \int_{\mathbf{A}} f(x; A)dP(A), \tag{1}$$

where $P(A)$ is a distribution function of the mixture. If the entire mass of the distribution $P(A)$ is concentrated in a finite number Q of points $A_1,...,A_Q$, which correspond to the probabilities $p_1,...,p_Q$, then expression (1) will be transformed into the probability distribution of the final mixture

$$f_P(x; p_1,..., p_Q, A_1,..., A_Q) = \sum_{j=1}^{Q} p_j f(x; A_j). \tag{2}$$

This expression statistically describes a mixed sample $\mathbf{X}_P=\mathbf{X}_1\cup...\cup\mathbf{X}_Q$ with volume $N = N_1 + ... + N_Q$, in which elements of partial samples $\mathbf{X}_1,...,\mathbf{X}_Q$ are mixed in an arbitrary sequence. Each partial sample $\mathbf{X}_j$ consists of N_j random variables that obey the distribution $f(x;A_j)$ and mix in the ratio $p_j=N_j/N$ [3].

In the general case, the statistical theory for a mixture of probability distributions allows us to solve the following problems by processing a specific $\mathbf{X}_P$ sample:

1) to determine the number Q of partial samples from which the mixture was formed;

2) to find estimates of the probabilities $p_1,...,p_Q$ and the parameters $A_1,...,A_Q$ of the partial distributions;

3) to decide on the membership of each element x_i (i = 1,...,N) of the $\mathbf{X}_P$ mixed sample (that is, what partial sample of $\mathbf{X}_1,...,\mathbf{X}_Q$ does it belong to).

The statistical theory for mixtures of distributions determines a set of conditions, the implementation of which provides the possibility of solving problems 1 and 2 (the task 3 reduces to a typical problem of statistical solutions). The main of them are the conditions of recognition, consisting in the requirement of linear independence of the elements of the family **F** [1]. Many families of probability distributions satisfy these conditions, in particular, normal and uniform distributions [2,4].

It is important to note two circumstances. Firstly, the property of identifiability is caused not by the stochastic nature of the function $f(x; A)$, but by the peculiarity of the dependence on the arguments x and A. Secondly, both occurrences – intersymbol interference and mixing samples of random variables – are the result of linear transformations of partial actions. These circumstances justify the formal identification of physical signals with probability distributions of random variables.

1.2 An example of setting the signal demodulation problem

Consider a communication channel with amplitude modulation, through which a sequence of symbols "1", "0" is transmitted at a speed of $1/T$ bit/sec (where T is the duration of the bit interval). Let a impulse $g(t,\theta)$ of a Gaussian form be formed for"1", where the function $g(t,\theta)$ of the Gaussian form is formed, where the function $g(t,\theta)$ is identical to the analytical description of the normal probability distribution with parameters $m = \theta T$, $\sigma = 0.425T$. Moreover, the impulse width at the level of 0.5 is T. The values of the function $g(t,\theta)$ are truncated to zero at $t = m \pm T$. Such a signal occupies 2-bit intervals and reproduces well the effect of dispersion distortion in fiber-optic communication systems. Let $G(t)$ be the signal that will be formed from the transmitted sequence "1" and "0" as a result of superposition of the corresponding impulses $g(t,\theta)$. Additional distortions are caused by random noise $N(t)$ with zero expectation and standard deviation σ_n.

Suppose a demodulator provides signal samples

$$S(t) = G(t) + N(t) \tag{3}$$

at discrete time instants t_m with a periodicity of dt= 0.5T, and this receiver is synchronized with the signal transmitter, and samples are produced at the boundary and in the middle of bit intervals. Denote the samples as $s_m = S(t_m)$.

The elementary method of demodulation is *the comparison of values s_m with a threshold value sd* (CWT method). That is, the solution r_i about the presence of " 1 "or" 0 " in the bit position with the number *i,* should be: r_i = 1, if $s_{2i}+2 > s_d$, otherwise r_i = 0. However, this does not take into account that useful information is contained not in one but in all samples on stretch of an interval 2T of the existence of the signal $g(t,\theta)$, as a result of which the reliability of demodulation is significantly reduced.

The best results can be obtained if to accept *maximum likelihood solutions* (MLS method). For a solution on the position of bit number i it is necessary to analyze $M = 4$ samples

$$s_{2i}, s_{2i+1}, s_{2i+2}, s_{2i+3}, \tag{4}$$

which fall into a " a sliding window " of $2T$ duration, and also to take into account influence of the following i+1-st and previous i-1-st of a symbol. Thus, the values of the samples in the "window" of the analysis depend on the combination "1" and "0" at 3 positions. For all possible $2^3 = 8$ combinations are necessary to generate the etalons of a signal $e_{h,0},...,e_{h,3}$ (h = 0,...,7) as the values of the process $G(t_{2i})$, ..., $G(t_{2i+3})$ in the expression (3). For normal distributions of noise $N(t)$ the solution accepts such aspect that r_i is central bit of 3-digit binary number j

$$j = \arg \min_{h=0,...,7} \sum_{m=0}^{M-1} (s_{2i+m} - e_{h,m})^2 \tag{5}$$

The disadvantage of this method of demodulation is a fairly large amount of computation.

Now we consider a solution to the demodulation problem based on *the statistical theory for mixtures* (method STM). As is known, the number of pulses in the “window” of analysis (Q = 3), their shape and position, it is enough to solve only problem 2, where is necessary to estimate the probabilities p_1, p_2, p_3 . In this case, the reference sequence (4) plays the role of the empirical probability distribution of a sample of random variables from the general population, which obeys the discontinuous distribution of the mixture probabilities

$$f_m(p_1,..., p_Q) = \sum_{j=1}^{Q} p_j f_{m,j}, \tag{6}$$

where p_j is the parameter of the presence of the symbol "1" or "0" at position j (respectively $p_j \neq 0$ or $p_j = 0$), and $f_{m,j}$ are the values of the partial discontinuous distribution equal to the values of the function $g(m \times 0.5, j\text{-}1)$.

It is necessary to find estimates $p_1^{(i)},...,p_Q^{(i)}$ of the parameters $p_1,...,p_Q$ by handling the empirical distribution (4). For the least squares estimation [3] they correspond to the minimum of the criterion

$$D(p_1,..., p_Q) = \sum_{m=0}^{M-1} (s_{2i+m} - f_m(p_1,..., p_Q))^2 = \sum_{m=0}^{M-1} (s_{2i+m} - \sum_{j=1}^{Q} p_j f_{m,j})^2. \tag{7}$$

If we equate the partial derivatives of the criterion (7) on $p_1,...,p_Q$ to zero, we obtain a system of simple equations

$$\sum_{j=1}^{Q} p_j c_{j,h} = b_h^{(i)}, \tag{8}$$

where

$$c_{j,h} = \sum_{m=0}^{M-1} f_{m,j} f_{m,h}, \; b_h^{(i)} = \sum_{m=0}^{M-1} s_{2i+m} f_{m,h} \tag{9}$$

The solution of the system of equations (8) gives all estimates $p_1^{(i)},...,p_Q^{(i)}$ however, it is advisable to use only one of them $p_2^{(i)}$ which can be calculated by the formula

$$p_2^{(i)} = \sum_{j=1}^{Q} ci_{2,j} b_j^{(i)}, \tag{10}$$

where ci - are elements of the matrix inverse to the matrix $\|c_{j,h}\|$. These matrixes can be calculated in advance. The solution rule for this method is to check the inequalities: if $p_2^{(i)} > pd$, then $r_i = 1$, otherwise $r_i = 0$. It is easy to see, that the volume of calculations of magnitudes $b_h^{(i)}$, $p_2^{(i)}$ is much smaller in comparison with the volume of calculations for the MLS method.

1.3 Statistical Modeling Results

For the considered demodulation methods, a statistical experiment is conducted, the results of which are presented in Fig. 1. The abscissa axis corresponds to the ratio SN of the useful signal power $G(t)$ to the noise power $N(t)$ (in decibels). The ordinate axis corresponds to the probability of demodulation errors for the CWT, MLS and STM methods. In the experiment, it was taken into account that the moments of the reference signals are shifted by $0.2T$ relative to the above positions due to phase lag of the synchronization circuit. The threshold values are $sd = pd = 0.5$. As expected, the CWT method is the least efficient, and the MLS method is the most efficient and can be considered as the optimal demodulation method. With an increase in the signal-to-noise ratio, the gain in efficiency increases, and at SN = 18 dB the MLS method reduces demodulation errors by 3 orders of magnitude compared to the CWT method. The effectiveness of the STM method is slightly inferior to the MLS method, but also significantly superior to the CWT method.

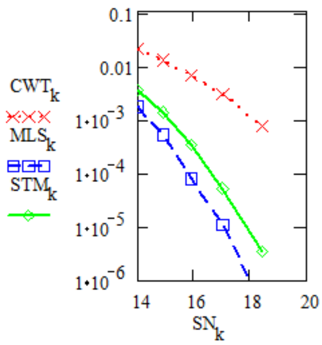

Fig.1. Statistical experiment results

On Fig.2 the diagrams of time of simulation are represented. The lower part of the column of each diagram corresponds to the time spent on signal modeling, the upper part to the time spent on the demodulation operation. The diagrams show that the STM method requires about 8 times less computation, which is an important argument for the benefit of its choice.

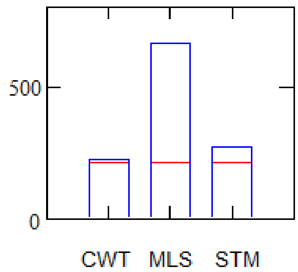

Fig.2. Diagrams of time of simulation

The results of statistical modeling showed that:

- the proposed STM method, based on the use of the statistical theory of mixtures, provides almost the same demodulation efficiency as the MLS, but it requires approximately an order of magnitude less computation;
- the STM method can be extended to more complex cases, for example, when it is necessary to estimate the position of signals in the "window" of analysis (when solving problem 2, it is necessary to evaluate not only the probabilities $p_1,..., p_Q$, but also parameters $A_1,...,A_Q$).

2 Technical solution for eliminating intersymbol interference of signals in a communication channel

2.1 Model of a device for reducing the influence of inter-symbol interference in an optical information transmission channel

The traditional means of combating intersymbol interference of signals are the use of various analog linear distortion correctors that compensate for the imperfection of the amplitude-frequency characteristics of the communication channel. Digital corrections are also used, which directly operate on the signal as a function of time [5].

The general block diagram of an optical signal regenerator with digital processing that uses a conversion algorithm based on a mixture theory is shown in Fig. 3.

The regenerator circuit consists of such elements. The regenerator circuit consists of such elements:

AID – analog input devices that receive Si(t) optical signals, detect them and filter them. Received optical signals Si(t) is converted using a fiber optic converter (OE converter), at the output of which an electric signal S(t) is generated, which contains information about the sequence of transmitted symbols "0" and "1", as well as a synchronization signal C(t).

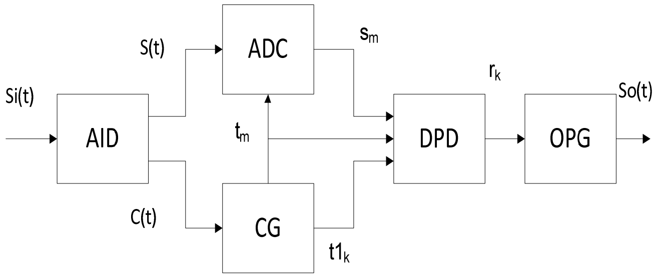

Fig. 3. Block diagram of an optical signal regenerator

CG is a clock generator that generates the clock sequences $t1_k$ (one pulse per bit interval) and t_m (n_b pulses per bit interval) from the signal C(t).

ADC is an analog-to-digital converter that encodes samples $s_m = S(t_m)$ in the form of an n_r-bit binary code.

DPD is a digital processing device that implements adaptation and demodulation operations; in the demodulation mode, it gives a decision $r_k = 1$ or $r_k = 0$ for each k-th bit interval;

OPG is an optical pulse generator that, in accordance with the solution rk, generates the reconstructed optical pulses $S_o(t)$ at the output of the regenerator.

Regardless of the technical implementation of the CPU, it is important to determine the time discreteness of the counts of the signal S(t), that is, the number of readings nb during the bit interval and the discreteness of the readings of the amplitude s_m, that is, the number of n_r encoding bits. The first parameter must satisfy the inequality n_b= 2.

However, the increase in n_b necessitates an increase in the speed of ADC and DSP, which complicates the regenerator. Moreover, the increase in nb is justified only if the independence of the random distortion of the signal S(t) by the noise N(t) is maintained for each sample. If the WUA performs a consistent filtering of the signal S(t), then the random components of the counts are found to be correlated more significantly, the greater the n_b. Given these features, it is advisable to take $n_b = 2$ [5].

Increasing the number of n_r encoding bits complicates ADCs and DTCs if the demodulation algorithm has a technical implementation. Due to low-bit coding, rather rough "images" of the signal are formed during the adaptation phase and this reduces the reliability of demodulation. If, as the number of digits increases, the encoding discreteness decreases to values smaller than the root mean square noise σ_n, then a further increase in n_r becomes useless. Thus, determining the minimum number of encoding bits that provides acceptable demodulation accuracy is one of the important tasks of the technical implementation of the proposed processing algorithm [6,7].

The signal amplitude sampling in the ADC can be described by the following function:

$$\text{kvant}(x)=\begin{cases}\dfrac{Nr-1}{Nr}, & \text{if } \ x>\dfrac{Nr-1}{Nr}\\ \dfrac{[x\cdot Nr]}{Nr}, & \text{if } 0\le x\le\dfrac{Nr-1}{Nr}\\ 0, & \text{if } \ x<0\end{cases}$$

where $[n]$ is the integer part of n; $Nr = 2n_r$ is the number of quantization levels. The operational value of argument x is in the range $0 \le x \le 1$.

When processing digital sample values, the required number of bits must exceed n_r by several units. If the CPU is based on microprocessors, then this requirement is met, since modern signal processors have at least 64 bits. To test the effect of the number of coding bits nr, statistical modeling was performed.

The discretization of the readout values is taken into account in the form of replacement of s_m by quant (s_m) in the calculation of the array of accumulated readings s_m for adaptation of the regenerator, which is performed according to the formula:

$$sm_m = \frac{1}{Nw}\sum_{j=0}^{Nw-1} s_{6j+m},\ m = 0,\ ...,\ 5. \tag{11}$$

Quantum (s_m) is also used to calculate the probability estimate $p_2(k)$ that the symbol "1" is in the kth position when demodulating the signals:

$$p_2(k)=\sum_{m=1}^{M} c1_{2,m}\cdot s_{k\cdot n_b+m-1}. \tag{12}$$

All other calculations were performed with the maximum accuracy possible in MathCAD14 [8].

To adapt the regenerator, a sequence of repeating triples of symbols "010" (TPS1) was used, which consisted of $N_w = 257$ triples of symbols "010". In each demodulation simulation session, a sampling of signals over a 256-bit interval was generated. For the demodulation error statistics set, the simulation sessions were repeated 400 or 4000 times. Thus, the total number of characters that were modeled in each session group reached 100,000 or 1,000,000. The simulation results are shown in Table. 1.

Table 1. Results simulation regenerator test operation

σ_v	S/ N	$n_r = 2$	$n_r = 3$	$n_r = 4$	$n_r = 6$	$n_r = 10$	$n_r = 15$
0.2	14,00	0.0043	0.0030	0.0028	0.0030	0.0030	0.0035
0.18	14,90	0.0019	0.0011	0.0011	0.0011	0.0011	0.0015
0.16	15,90	0.00062	0.00034	0.00024	0.00024	0.00025	0.00041
0.141	17,00	0.00015	0.00008	0.00002	0.00004	0.00004	0.00008
0.12	18,40	0.00003	0.000005	0.000001	0.000001	0.000001	0.000001

The first column presents the noise intensity in the form of a value of σ_n. In the second column, the signal-to-noise ratio in dB is calculated as 20lg (1 / σ_n), given that the maximum amplitude of the useful signal C(t) is approximately 1. Other columns provide estimates of the probability of demodulation errors for different numbers of n_r encoding bits in the ADC.

As we can see, with a small number of digits $n_r = 2$, the errors are, as expected, maximal. However, at $n_r = 4$, they reach a minimum. Further increase in the number of digits of the coding does not actually improve the accuracy of demodulation, but on the contrary, worsens it. For example, for the highest possible accuracy of calculations in MathCAD14, which corresponds to 15 decimal places ($n_r = 15$), the results are even worse than for $n_r = 4$.

This can be explained by the fact that quantization makes the "image" of the signal coarser. On the other hand, quantization negates the effect of noise in the range of ΔS values between quantization levels. If S << n, no leveling occurs. At the same time, it is obvious that, as noise levels decrease, it is advisable to reduce ΔS, that is, to increase n_r. Given that the simulation noise level exceeds the actual noise level in the regenerators (since it is impossible to obtain statistically reliable estimates of error probabilities at low noise), 5 digits of coding in the ADC can be considered sufficient.

Digital processing operations associated with adaptation of the regenerator and demodulation of the signal can be performed using modern signal processors.

Such a microprocessor embodiment of digital processing of the received signal is technically the simplest and cheapest. However, in this case, only sequential execution of operations is possible. This limitation is related to the actual DPD speed. However, the most critical in terms of performance are two groups of operations directly related to the processing of s_m counters. These are operations of accumulation of sum_s (11) at the stage of adaptation and operations of calculating the probabilistic estimate $p_2(k)$ (12).

At the demodulation stage, high speed is required in the decision-making operation r_k, where it is necessary to compare the estimate obtained with the decision threshold p_{por} in terms of:

$$r_k = \begin{cases} 1 & \text{if } p_2(k) > p_{por}, \\ 0 & \text{— else.} \end{cases} \qquad (13)$$

Due to the simplicity of these operations, it is advisable to create specialized nodes for their execution, which provide minimal processing time, in particular due to the organization of parallel calculations. Such a principle and the order of processing can be recommended. Firstly, it is necessary to form a sufficiently long test signal sequence (TPS1) to adapt the regenerator. Moreover, only the initial fragment with a length of $3 \cdot N_w$ characters is actually used to obtain information about the properties of the signals. On this fragment, it is necessary to form partial sums (for each m = 0, ..., 5) in expression (11). Corresponding operations must be performed in real time by the accumulation unit (AU). After the initial fragment, the sequence of TPS1 should continue for a time exceeding the time required by the microprocessor calculator (MPC) to perform all subsequent operations, including the calculation of the coefficients $c1_{2.1}$ ..., $c1_{2.4}$, which are used in expression (12). During this time, which may be called the "adaptation period" (PA), the input signals of the regenerator are ignored. To determine the duration of the PA interval need to carry out simulations of MPC. After the end of the PA interval, regenerator enters the operating mode of signal processing in accordance with expressions (11), (12). This processing is performed by a separate demodulation node (DN).

The block diagram of a possible option for constructing such a hardware-software DPD is shown in Fig. 2. According to the results of simulation, we take the number of bits of the ADC coding $n_r = 5$, and the length of the initial fragment of TPS1 is 768 characters ($N_w = 256$).

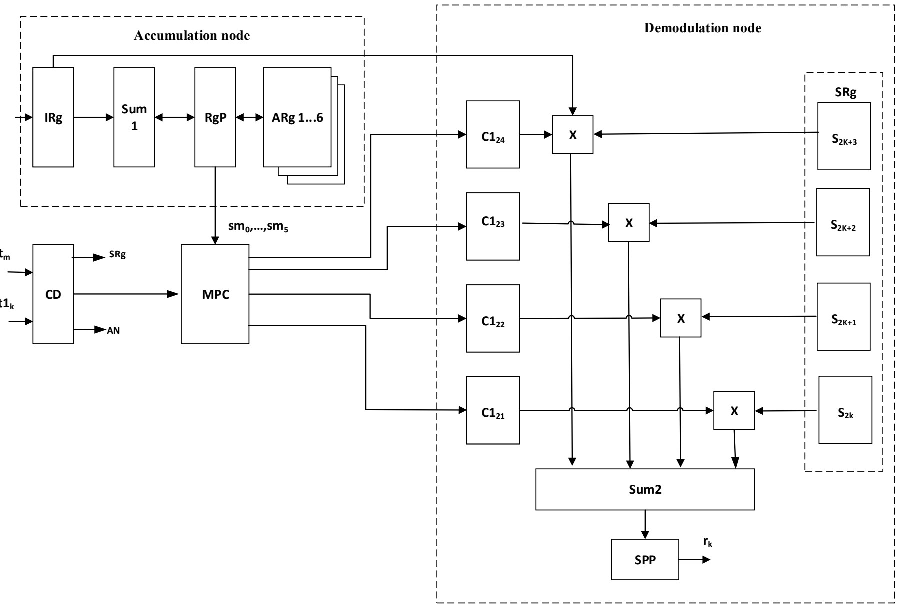

Fig. 2. Block diagram of a digital processing device

The accumulation node (AN) consists of the following elements:

- IRg - input 5 - bit register, which receives s_m signals from the ADC;
- ARg1 ..., ARg6 - 13-bit registers of accumulation of private sums (11). The number of digits is defined as 5 + log2 (N_w) = 5 + 8 = 13;
- RgP - 13-bit intermediate register, with the help of which communication with ARg1 ..., ARg6 and MPC is carried out;
- SUM1-13-bit adder adds the next signal from the ADC to the corresponding partial sum.

The received amounts after completion of accumulation are transferred using RgP to microprocessor calculator. The numbers with which this node operates are treated as non-negative integers (with a point fixed after the least significant bit). The result of the calculations of the operations performed by the MPC are the coefficients $c1_{2.1}$ ...,

$c1_{2.4}$, written to the corresponding registers $C1_{21}$, ..., $C1_{24}$ of the demodulation node (as 16-bit numbers with a fixed or floating point).

The demodulation node (DN) consists of the following elements:

- $C1_{21}$, ..., $C1_{24}$ - registers with weight coefficients, which are determined at the stage of adaptation of the regenerator;
- SRg - shift register from 4 cells to 5 bits, where sm samples are received in the operating mode (each clock pulse tm shifts information by one cell);
- MUX is a 16-bit multiplier that calculates each sum of (12), where s_m is treated as fractional numbers with a point fixed before the highest digit;
- SUM2 is a 16-bit adder that accumulates the sum (12);
- SPP is a comparison scheme with the threshold $p_{por} = 0.5$, which gives the solution r_k in accordance with (13).

The solution is the value of the discharge, which is located to the right of the point separating the integer and fractional part of the number at the output of the adder. The operations of multiplication and summation in the demodulation node are performed once during the bit interval, that is, by the clock cycle $t1_k$.

The control device (CD) synchronizes the interaction of all elements of the circuit using clock pulses t_m, $t1_k$.

According to the logical sequence of signal processing in the DPU, the microprocessor calculator starts working after the accumulation node completes its work. However, it must be taken into account that the MPC and AN start to function simultaneously after turning on the power or the receipt of a special signal to start operation. While AN carries out the accumulation of TPN1, it is advisable to provide for testing MPC. Since the ratio of accumulation time and testing time is unknown, we will proceed from the following interaction principle:

- AN, after completing its work, transfers to the MPC the accumulated amounts sm1, ..., sm6 and the readiness flag, which are recorded in the MPC memory in the interrupt mode;
- After completion of testing, the MPC checks the sign of readiness and, if there is one, proceeds to the processing of information, and in the absence, the signs of readiness are expected.

After completing all the processing associated with the adaptation of the regenerator, the MPC issues the obtained weight coefficients $c1_{21}$ ..., $c1_{24}$ to the demodulation unit and gives the adaptation termination flag to the control device, by which the latter opens the arrival of the s_m signal samples to the shift register in the ID and after 2 bit intervals (when the register is full) allows the operation of a comparison scheme with a threshold that produces a solution r_k.

Conclusions

The basic operations of protecting the regenerator of a signals transmission system that have distortion due to intersymbol interference can be considered from the point of view of the statistical theory of mixtures as a task of estimating some parameters of the probability distribution of the mixture.

Due to a significant decrease in the effective bandwidth of the communication line, as the length of the link increases, a significant pulse expansion occurs and its shape becomes bell-shaped, which causes significant inter-symbol interference of the signals.

It is proposed to adapt the digital corrector to the features of a particular communication line using a special test sequence of characters that is generated at the beginning of the communication session and allows you to create a "image" of the distorted signal in the communication line.

To adapt the regenerator, need to accumulate 6 averaged real-time counts that correspond to three bit intervals (2 counts per bit interval). Then need center and highlight 4 of the 6 averaged counts. Selected readings form an "image" of a real signal in the form of a response of the transmission line to a single character "1", which, as a result of inter-character distortions, takes about 2 bit intervals. Finally - to calculate the values of the weighting coefficients for the obtained "image" of the real signal.

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