**Ministry of Education and Science of Ukraine**
**O.S. Popov Odesa National Academy of Telecommunications**

**A.G. Zuko Telecommunication Theory and Metrology Department**

**Ivaschenko P., Rozenvasser D.**

# EDUCATION MANUAL ON TELECOMMUNICATION THEORY AND THEORY OF INFORMATION AND CODING

**Laboratory Works**

Odesa 2018

UDK 621.391 Plan of Education Materials Publication in 2018

Ivaschenko P. Education manual on telecommunication theory and theory of information and coding: Laboratory Works / P. Ivaschenko, D. Rozenvasser. – Odesa: O.S. Popov ONAT. – 2018. – 41 p.

Editor of the English language version: Kuznietsova G.

The education manual is intended for students majoring in:

172 – Telecommunications and radio technology studying Telecommunication theory;

121 – Software engineering studying Theory of information and coding.

The education manual contains methodical guidelines for fulfilling 17 laboratory works.

The methodical guideline for fulfilling each laboratory work contains objectives, main postulates, questions for checking the knowledge, instructions for home- and Laboratory works and guidelines for conclusions.

APPROVED at the meeting
of A.G. Zuko Telecommunication
Theory and Metrology Department.
Minutes № 4, November 26, 2017

APPROVED
by O.S. Popov ONAT
Publishing committee.
Minutes № 4, December 27, 2017

CONTENTS

# LW 1.0 THE STUDY OF THE TELECOMMUNICATION SYSTEM STRUCTURAL SCHEME

## 1 Objectives

1.1 The studying of telecommunication system structure.
1.2 The researching into telecommunication system accuracy.

## 2 Main postulates

See textbook [1, section 1].

## 3 Questions

3.1 Give the definition of "information", "message", and "signal".
3.2 Give the definition of "communication network", "telecommunication system", "communication channel", "transmission system".
3.3 What element of a telecommunication system converts the message into the baseband telecommunication signal? Give examples.
3.4 Explain “coding” and “code” in telecommunication systems.
3.5 Explain “digital baseband signal”.
3.6 Explain “modulation”.
3.7 Explain the reason of a demodulator usage in the transmission system.
3.8 How do you estimate the accuracy of the telecommunication system?
3.9 What parameters determine the accuracy of the telecommunication system?

## 4 Homework

4.1 Encode the first three letters of your last name in the ITC-2 code.

**Table 1 –** International Telegraph Code # 2 (ITC-2)

| Codeword number | Code-word | Latin letters register | Cyrillic letters register | Numbers register | Codeword number | Codeword | Latin letters register | Cyrillic letters register | Numbers register |
|---|---|---|---|---|---|---|---|---|---|
| 1 | 11000 | A | А | – | 17 | 11101 | Q | Я | 1 |
| 2 | 10011 | B | Б | ? | 18 | 01010 | R | Р | 4 |
| 3 | 01110 | C | Ц | : | 19 | 10100 | S | С | ' |
| 4 | 10010 | D | Д | ! | 20 | 00001 | T | Т | 5 |
| 5 | 10000 | E | Е | 3 | 21 | 11100 | U | У | 7 |
| 6 | 10110 | F | Ф | « | 22 | 01111 | V | Ж | = |
| 7 | 01011 | G | Г | » | 23 | 11001 | W | В | 2 |
| 8 | 00101 | H | Х | Є | 24 | 10111 | X | Ь | / |
| 9 | 01100 | I | И | 8 | 25 | 10101 | Y | Ы | 6 |
| 10 | 11010 | J | Й | ; | 26 | 10001 | Z | З | + |
| 11 | 11110 | K | К | ( | 27 | 00010 | Space | Ш | % |
| 12 | 01001 | L | Л | ) | 28 | 11011 | < | Ю | * |
| 13 | 00111 | M | М | . (point) | 29 | 00000 | > | Щ | № |
| 14 | 00110 | N | Н | , (comma) | 30 | 01000 | Numbers register | | |
| 15 | 00011 | O | О | 9 | 31 | 00100 | Cyrillic letters register | | |
| 16 | 01101 | P | П | 0 | 32 | 11111 | Latin letters register | | |

4.3 Be ready to answer the questions.

## 5 Laboratory work

5.1 **Study the model of a laboratory work at the workplace**. Run the program **1.0** using the icon **TT (English)** on the desktop. Click on the virtual model icon in the Delphi environment to do the laboratory work. Study the block diagram of the virtual model and enter input parameters. The model displays a telecommunication system based on a single transmission system and includes:

1 "Messages source", where you can enter the three letters of the Cyrillic alphabet.

2 "Source encoder". This block code has letters entered by ITC2. It forms the baseband digital signal with a digital signal rate 1000 bit/s.

3 "Modulator" generates a binary amplitude modulated signal (BASK). The "Generator of a carrier oscillation" produces harmonic oscillation required for the modulator.

4 "Transmission line" forms the sum of the signal and noise.

5 "Generator of noise" generates noise, the "on" and "off" buttons manage the generator work. The average power of noise $n(t)$ is equal to the average power of a signal $s(t)$ at the modulator output.

6 "Attenuator" is included in the circuit of the noise generator. It provides noise attenuation, the value of which can be set from 0 dB to 9 dB with 1 dB step. In the virtual model, the ratio of an average signal and noise power at the transmission line output (SNR) is equal to the attenuation level.

7 "Demodulator" recovers the baseband digital signal.

8 "Source decoder" decodes the recovered baseband digital signal.

9 "Oscilloscopes": the time scale measured in milliseconds.

Many blocks are with a symbol . After clicking on it you can see the output voltage of the corresponding unit.

The "Start" button starts the virtual model simulation.

The "Close" button closes the program.

5.2 **The study of messages and signals transformation during the transmission in telecommunication system.** Enter a message, which you used in your homework. Turn noise off. Run the program. Write or draw in the protocol:

– the transmitted message;
– the baseband signal;
– the modulated signal;
– the output of the transmission line;
– the demodulator output;
– the decoder output.

Make sure that the message is undistorted at the decoder output.

Identify the duration of a binary symbol $T_b$ using the oscillogram of the modulating signal. Calculate the digital signal rate $R$

5.3 **The study of messages and signals transformation during the transmission in the telecommunication system with noise**.

Turn on the noise. Set attenuation 3 dB. Describe the changes at the output of the separate blocks of the transmission system with noise.

Running the program, determine the smallest SNR when the error is observed at the demodulator output (and in the message). As errors occur randomly, for each SNR value it is necessary to start a few (3 ... 5) program runs to determine the presence or absence of errors.

**6 Conclusion**

Make a conclusion about each item of the laboratory work, analyze the results (coincidence of experimental and theoretical states, discuss numeric values).

## LW 1.1 THE STUDY OF PERIODIC SIGNALS SPECTRA

**1 Objectives**

1.1 The study of periodic signals spectra: sequences of rectangular pulse, triangle pulse, sawtooth pulse.
1.2 The research into the signal spectrum limited by the signal form.

**2 Main postulates**

See textbook [1, subsection 2.5].

**3 Questions**

3.1 What signals are named "periodic"?
3.2 Write an expression of the Fourier series for a periodic sequence of rectangular pulses.
3.3 Write an expression of amplitudes and initial phases for the Fourier series components of a periodic sequence of rectangular pulses.
3.4 Define the amplitude and phase spectra of the periodic signal.
3.5 What is the fundamental difference between the spectrum of periodic and non-periodic signals?
3.6 How does the spectrum of a periodic sequence of rectangular pulses change, if the pulse duration is reduced?
3.7 How does the spectrum of a periodic sequence of rectangular pulses change, if the period of pulses is reduced?
3.8 Why are there only sine components in the Fourier series for sawtooth pulses and triangular pulses?
3.9 How does the spectrum limitation influence the form of rectangular pulses?

**4 Homework**

4.1 Calculate the amplitude spectrum of a periodic sequence of rectangular pulses

$$s(t) = \frac{A\tau}{T}\left[1 + 2\sum_{n=1}^{\infty}\frac{\sin \pi n f_1 \tau}{\pi n f_1 \tau}\cos 2\pi n f_1 t\right] \tag{1}$$

with the period of pulses $T = 2N$ ms, duration $\tau = T/(N + 1)$ ms, and the amplitude $A = 1$ V, where $N$ is the number of your team. Arrange the results of calculations in the table and plot the spectrum.

4.2 Be ready to answer the questions.

**5 Laboratory work**

**5.1 Study the model of a laboratory work at the workplace**. Run the program using the icon **TT(English)** on the desktop and then the LW **1.1**. Study the block diagram of the virtual model (it is described above) and enter the input parameters.

The virtual model contains: a generator of a periodic signal with three types of signals:

- a periodic sequence of rectangular pulses;
- a periodic sequence of bipolar sawtooth pulses;
- a periodic sequence of bipolar triangular pulses.

It is possible to choose the amplitude and a period of pulses for all the signals, and pulse duration for rectangular pulses.

The key allows to monitor time and spectral diagrams of signals at the generator output or at a lowpass filter (LPF) output. The cut frequency $F_{\text{cut}}$ can be changed.

**5.2 Study the spectrum of a periodic sequence of rectangular pulses at the generator output**. Set the value of the amplitude, a period of pulses and pulse duration from the homework. Put down the time and spectrum diagrams of the investigated signal in the protocol. Compare the experimental spectrum with the calculated one in your homework.

**5.3 Study the spectrum of a periodic sequence of sawtooth pulses at the generator output**. Set the same amplitude and a period of pulses as in the previous task. Determine the theoretical values of amplitudes of spectrum components by the formula

$$s(t)=\frac{2A}{\pi}\left(\sin 2\pi f_1 t-\frac{1}{2}\sin 2\pi 2 f_1 t+\frac{1}{3}\sin 2\pi 3 f_1 t-\frac{1}{4}\sin 2\pi 4 f_1 t+\ldots\right), \qquad (2)$$

compare them with experiment results and with the spectrum of rectangular pulses sequence.

**5.4 Study the spectrum of a periodic sequence of triangular pulses at the generator output**. Set the same amplitude and a period of pulses as in the previous task. Determine the theoretical values of amplitudes of spectrum components by the formula

$$s(t)=\frac{8A}{\pi^2}\left(\sin 2\pi f_1 t+\frac{1}{3^2}\sin 2\pi 3 f_1 t+\frac{1}{5^2}\sin 2\pi 5 f_1 t+\ldots\right), \qquad (3)$$

compare them with experiment results and with the spectrum of rectangular pulses sequence.

**5.5 Study the LPF influence on the spectrum and form of a periodic sequence of rectangular pulses**. Set the amplitude and period of pulses the same as in task 5.2. Perform the experiment for two values of the LPF cut frequency, namely 2/τ and 1/τ (τ – pulse duration). Put down the time and spectral diagrams of filtered signal in the protocol. Make conclusion about changes in the form and spectrum of rectangular pulses sequence.

**5.6 Study the LPF influence on the spectrum and form of periodic sequences of sawtooth and triangular pulses**. Set the amplitude and period of pulses the same as in 5.3 and 5.4. Perform the experiment for the value of the LPF cut frequency $F_{cut}$=4/$T$ ($T$ – period of pulses). Put down the time and spectral diagrams of filtered signals in the protocol. Make conclusion about changes in the form and spectrum of signals.

**6 Conclusion**

Make a conclusion about each item of the laboratory work, analyse the results (coincidence of experimental and theoretical states, the form of signal dependence on the value of cut frequency).

## LW 1.2 RESEARCH INTO RANDOM PROCESSES PROBABILITY DISTRIBUTIONS

**1 Objectives**

The study and experimental research into one-dimensional probability distribution functions and the probability density functions of random processes.

**2 Main postulates**

See textbook [1, section 3].

**3 Questions**

3.1 What processes are called stationary and ergodic?

3.2 Give the definition of the one-dimensional probability distribution function of a random process and prove its properties.

3.3 Give the definition of the one-dimensional probability density function of a random process and prove its properties.

3.4 How can you find the hit probability of process values at the defined interval, by using the probability distribution function or the probability density function?

3.5 Write down the expressions for the expectation and dispersion of a random process. What is their physical meaning?

3.6 Write down the expression for the normal probability distribution function and explain the meaning of values with it used.

3.7 Explain the type of the graphs of probability distribution function of the harmonic oscillation with an accidental phase, fluctuation noise, and the random process with a uniform distribution.

3.8. Describe the principle of a device operation to measure the probability distribution function and probability density function of a random process.

**4 Homework**

4.1 Perform calculations and build a probability distribution function $F(x)$ and a probability density function $p(x)$ graphs of the normal (Gaussian) random process, $a$ = 0 and root-mean-square deviation $\sigma$ = 1 + 0,1$N$ (where $N$ is the number of your team) for the values $-3\sigma < x < 3\sigma$. In the absence of the probability integral table it is possible to use the approximation formula:

$$Q(z) \cong 0{,}65 \exp[-0{,}44(z + 0{,}75)^2] \text{ under } z > 0;$$

$$Q(z) = 1 - Q(|z|) \text{ under } z < 0,\ Q(0) = 0{,}5,\ Q(\infty) = 0.$$

The results of calculations should be presented in the form of tables and graphs.

4.2 Be ready to answer the questions.

**5 Laboratory work**

**5.1 Learn a virtual model at your workplace.** Start the program **1.2**, using the icon **TT(English)** on the desktop. It is necessary to study the structure of a virtual model and master parameters entering. Click on the virtual model icon in the HP VEE environment to do the laboratory work. The model allows to investigate the characteristics of a random process with a uniform probability distribution, the Gaussian random process, and harmonic oscillation.

This virtual model realizes two basic functions for each process:

1. The generation of the $N$ samples of the researched random process $X(t)$. The samples are displayed as “Realization of a process”;

2. Calculations on the basis of the generated samples of values and displaying:

   a) a probability distribution function;

   b) a probability density function;

   c) an average value of a process;

   d) a root-mean-square deviation of a process.

For every studied random process different methods of sample generation, different parameters of processes are used.

The generation of samples of a process with a uniform distribution is performed by the built-in function “randomize”. The values of $x_{\min}$ and $x_{\max}$ are preset in the model.

The generation of the Gaussian process samples is performed by nonlinear transformation of two arrays of samples $u(i)$ and $v(i)$ of random process with a uniform distribution at an interval (0, 1).

The transformation is given by

$$X(i) = a + \sigma \cdot \sqrt{-2\ln(u(i))} \cdot \cos(2\pi v(i)),\ \ i = \overline{1, N},$$

here $i$ is the number of the sample in an array; $a$ and $\sigma$ are the average value and root-mean-square deviation of the studied random process, which a student sets on a model.

A built-in functional generator performs the generating of samples of a harmonic oscillation. The student sets the amplitude, the frequency, and the initial phase of oscillation.

The calculation of values of probability distribution function and probability density function is made in the range of the argument values from the lower-range value $x_{\text{low}}$ and to the upper-range value $x_{\text{up}}$. An interval $(x_{\text{low}}, x_{\text{up}})$ is divided at $M$ of identical subintervals with the length $\Delta x = (x_{\text{up}} - x_{\text{low}})/M$; the quantity of samples $k_j$, where the $j$-th subinterval is calculated ($j$ takes on values from 1 to $M$). The hit frequency of sample values at the $j$-th subinterval $q_j = k_j/N$. In the case of sufficiently large values $M$ and $N$ (in the model $M = 200$, $N = 10000$), the values of a frequency $q_j$ give the probability of getting hits of the sample values at the $j$-th subinterval. Values hit probability at the $j$-th subinterval are $q_j = p(x_j)\Delta x$, where $x_j = j\Delta x$. Therefore,

$$p(x_j) = \frac{k_j}{N\Delta x} = \frac{k_j M}{N(x_{\text{up}} - x_{\text{low}})}, \quad j = \overline{1, M}$$

The arrays of values $p(x_j)$ and $x_j$ are displayed as "Probability density function".

Using the property of the probability distribution function $F(x)$, the array of values is calculated as:

$$F(x_j) = \Delta x \sum_{k=1}^{j} p(x_k), \quad j = \overline{1, M}$$

The arrays of values $F(x_j)$ and $x_j$ are displayed as "Probability distribution function".

The average value of the researched process is calculated by the formula

$$\overline{X(i)} = \frac{1}{N} \sum_{i=1}^{N} X(i),$$

where $X(i)$, $i = \overline{1, N}$ is $i$-th sample of the researched process. The value $\overline{X(i)}$ is displayed. This display is called "Measured average value".

Root-mean-square deviation of the researched process is calculated as

$$\sigma = \sqrt{\frac{1}{N-1} \sum_{i=1}^{N} (X(i) - \overline{X(i)})^2}\,.$$

The value σ is the "Measured root-mean-square deviation".

**5.2 The research into the random process with a uniform distribution probability.** In the menu "Choice of process" click the item "With a uniform distribution". Enter values $x_{\min} = -1$ and $x_{\max} = 1$ into proper windows. Ultimate values of argument at the analysis of distributions are $x_{\text{low}} = -2$ and $x_{\text{up}} = 2$. Write down the measured average value, the root-mean-square deviation, graphs of probability distribution function and the probability density function. Repeat measurements for other values $x_{\min}$ and $x_{\max}$.

**5.3 The research into a Gaussian process.** In the menu "Choice of process" click the item "Gaussian process". Enter in the proper windows values $a$ and σ, obtained in the homework, and choose values $x_{\min}$ and $x_{\max}$ so that they cover a range of values $x$ from $a - 3\sigma$ up to $a + 3\sigma$. Write down the measured average value, and the root-mean-square deviation, graphs of the probability distribution function, and the probability density function. Repeat measurements for other values of average value $a$ and root-mean-square deviation σ.

**5.4 The research into statistical characteristics of a harmonic oscillation.** In the menu "Choice of process" click the item "Harmonic oscillation". In the proper windows enter the value of amplitude $A = 1$, the value of frequency $f$ from 10 to 20 kHz and the value of an accidental phase φ. Establish ultimate values of an argument analyzing distributions so that they cover a range of values $x$ from $-A$ up to $+A$. Write down the measured average value and the root-mean-square deviation, graphs of probability distribution function, and the probability density function. Repeat measurements for other values of $A$, frequency $f$, phase φ.

**6 Conclusion**

Make a conclusion about each item of the laboratory work, analyse the obtained results:

- the coincidence of functions $p(x)$ and $F(x)$ forms each of the researched process compared to theoretical one;
- the implementation of properties $p(x)$ and $F(x)$;
- the coincidence of the measured average value and the root-mean-square deviation with the calculated one, by the given parameters in the research process ($x_{min}$ and $x_{max}$, $A$);
- the dependence of functions $p(x)$ and $F(x)$ on the frequency and the initial phase of harmonic oscillation.

## LW 1.3 RESEARCH INTO THE CORRELATION CHARACTERISTICS OF RANDOM PROCESSES AND DETERMINISTIC SIGNALS

**1 Objectives**

1.1 Study the method of experimental investigation of correlation characteristics of random processes and deterministic signals.

1.2 Research into the connection between correlation functions and spectrums of random processes and deterministic signals.

**2 Main postulates**

See textbook [1, subsections 2.2, 3.4 – 3.7].

**3 Questions**

3.1 Give the definition of the correlation function (CF) of a random process.

3.2 How do you determine CF of the process?

3.3 Enumerate the main properties of the random process CF.

3.4 What random process parameters are possible to define according to CF?

3.5 What does Wiener-Khinchin theorem state?

3.6 Enumerate methods of a correlation time determination.

3.7 How are bandwidth and correlation time of a random process connected?

3.8 What form does the CF of a rectangular video pulse have?

3.9 What form does the CF of a rectangular radio pulse have?

3.10 Why doesn't the initial phase of a rectangular radio pulse influence its CF?

**4 Homework**

4.1 Build a block diagram of the correlation meter for the research of correlation functions of random processes and deterministic signals.

4.2 Perform calculations and build graphs for the CF of a rectangular video pulse and a rectangular radio pulse for such input data: $T_p$ = 2 ms, frequency of oscillation of a radio pulse signal $f_0 = 500(N + 1)$ Hz, where $N$ is the number of your team.

4.3 Be ready to answer the questions.

## 5 Laboratory work

**5.1 Familiarity with a virtual model at the workplace.** Start the program **1.3**, using the icon **TT(English)** on the desktop. It is necessary to study the structure of a virtual model and master parameters entering. Click on the virtual model icon in the HP VEE environment to do the laboratory work. A model contains the following generators:

– a generator of noise, which produces the realization of quasi-white noise with the band (0, $F_{max}$), duration 20 ms, in the form of 5000 samples. It is possible to set the $F_{max}$ value 1000, 2000 and 3000 Hz;

– a generator of a single rectangular video pulse that allows to set pulse duration 0,5; 1 and 1,5 ms and an arbitrary amplitude of a pulse;

– a generator of a rectangular radio pulse, with duration 2 ms, allows to set arbitrary amplitude of a pulse and the frequency $f_0$ of oscillation 1000, 2000 and 3000 Hz. The initial phase of oscillation is a random value, this value is displayed on the indicator $\varphi$.

The switch **S** allows the researched process to be chosen.

If the noise is chosen for the research, displays show:

– the noise realization;

– the value of the measured average power of realization;

– the correlation function;

– the power spectral density of the noise realization, got as the Fourier transformation from the correlation function of the realization. The program generates samples of quasi-white noise. However, because of few samples, the spectrum is far from white in the band (0, $F_{max}$).

If a rectangular video pulse or a rectangular radio pulse is chosen, displays show:

– pulse oscillogramms;

– the measured pulse energy value;

– the correlation function of a pulse;

– the square of an amplitude spectrum of pulse, got as a Fourier transform from the correlation function of pulse.

**5.2 Research into correlation and spectral characteristics of realization of noise.** Set in generator of the quasi-white noise $F_{max}$ = 1000 Hz. After the program execution, analyse the experimental data and write them down. Check up the implementation of properties of a correlation function, determine the maximum frequency of a spectrum, and determine the correlation time $\tau_к$ for a correlation function, find out their product, compare it with the theoretical value $\tau_к F_{max} \approx 0{,}5$. Give the visual estimate of the average value of the power spectral density $N_0$ on an interval (0, $F_{max}$). Multiply the value of power spectral density $N_0$ by $F_{max}$ and compare the product with the value of the measured average power of realization.

Repeat the experiment for other values $F_{max}$.

**5.3 Research into correlation and spectral characteristics of a rectangular video pulse.** Set in the generator of a rectangular video pulse $A$ = 2 V, $T_p$ = 0,5 ms. After the program is run, complete the $K_s(\tau)$ and $S^2(f)$ graphs. Analyse the experimental data and write them down. Compare the experimental dependence $K_s(\tau)$ with the theoretical one; compare the measured value of the pulse energy with the value of $K_s(0)$.

Repeat the experiment for other values $A$ and $T_p$.

**5.4 Research into correlation and spectral characteristics of a rectangular radio pulse.** Set in the generator of a rectangular radio pulse $A$ = 2 V, $f_0$ = 1000 Hz. After the program is run, complete the $K_s(\tau)$ and $S^2(f)$ graphs. Analyse the experimental data and write them down. Compare the experimental dependence $K_s(\tau)$ with theoretical one, and compare the measured value of pulse energy with the value of $K_s(0)$. Write down the value of the initial phase. Run the program and make sure, that a correlation function does not depend on an initial phase.

Repeat the experiment for other values $A$ and $T_p$.

**6 Conclusion**

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of experimental and theoretical states, review the implementation of correlation functions properties, etc.).

## LW 1.5 RESEARCH INTO DIGITAL MODULATED SIGNALS

**1 Objectives**

1.1 The study into transmission methods of digital signals with modulated *M*-ASK, *M*-PSK and BFSK signals.

1.2 The research into time and spectral characteristics of *M*-ASK and *M*-PSK signals for $M$ = 2 and 4 and BFSK signal.

**2 Main postulates**

See textbook [1, subsections 4.1 – 4.8].

**3 Questions**

3.1 What is the aim of using the modulation in the telecommunication systems?

3.2 Give the definition of a digital signal.

3.3 Give the definitions of digital modulation signals: *M*-ASK; *M*-PSK; *M*-FSK.

3.4 Why aren't the radio-frequency pulses with rectangular envelope used for transmitting digital signals through communication channels? What form must a pulse envelope have?

3.5 What are the forms of the spectrums of *M*-ASK; *M*-PSK; *M*-FSK signals?

3.6 What are the multi-level signals transmitting digital signals through communication channels used for?

3.7 What signals of digital modulations are one-dimensional, and what signals are two-dimensional?

## 4 Homework

4.1 Given clock period: $T = 50$ ms. It is necessary to build the time diagrams of elementary radio pulses of frequency $f_0 = 40$ Hz for two cases: with a rectangular envelope and with the Nyquist pulse envelope.

*Note.* It is necessary to take into account that elementary radio pulse is the product of a rectangular pulse of duration $T$ or the Nyquist pulse, and a harmonic wave. In case of the Nyquist pulse it is possible to take a function $A(t) = \sin(\pi t/T)/(\pi t/T)$. Draw the diagram of this function at the interval ($-4T$, $4T$).

4.2 Be ready to answer questions.

## 5 Laboratory work

**5.1 The study a virtual model at your workplace.** Run the program **1.5**, using the icon **TT(English)** on the desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work.

A model is a universal modulator of digital modulated signals. It includes the digital signal generator with a duration that equals $8T_b$, so signal symbols can be changed. Given bit duration is: $T_b = 50$ ms. The modulator consists of the followings blocks: a mapper, shaping filters, carrier generators, two multipliers and an adder. The setting of the modulation type affects the mapping code of an encoder and carrier generators. It permits to set the followings types of modulation: BASK, QASK, BPSK, QPSK, and BFSK. The signals from two mapper outputs enter the filter inputs, shaping the radio pulse envelopes in the form of the Nyquist pulse. The scheme contains a switch, allowing to exclude the shaping filters from the scheme, so a radio pulse has the rectangular envelope. The formed pulses are multiplied by carriers. Given carrier frequency $f_0$ is equal to 40 Hz. The model has oscillographs and a spectrum analyzer. Study the scheme model, using the description above. Discuss the plan of your laboratory work with your teacher.

**5.2 The preparation of a virtual model.** Use the decimal $128 + 10 \cdot N$ ($N$ is a number of laboratory stand) expressed as a binary number. The roll-off factor is equal to $\alpha = 1 - 0{,}1 \cdot N$.

**5.3 The research into the form and spectrum of BASK and QASK signals as the functions of an envelope form.** To do the research it is necessary to set: a type of a modulation – BASK; the envelope form is a rectangular pulse. In your protocol you should present the time diagrams of the following signals one under the other: the digital signal, the output signal of a mapper, the modulated signal. Also present the spectral diagram of the modulated signal. After that it is necessary to set the second envelope form – the Nyquist pulse. In your protocol you should put down the time and spectral diagrams of the modulated signal.

Do the same research for the QASK signal.

In conclusions, on the basis of spectral diagrams, you should indicate the appropriateness of using the radio-frequency pulses with the comparison Nyquist pulse envelope and the appropriateness of using the multi-positional signals for decreasing the occupied frequency band.

**5.4 Research into the form and spectrum of BPSK and QPSK signals as the functions of an envelope form.** Repeat the experiments, performed in 5.3 for the BPSK and QPSK signals. Compare the spectrums of *M*-ASK and *M*-PSK signals.

**5.5 Research into the form and spectrum of BFSK signal as the functions of an envelope form.** Repeat experiments, completed in items 5.3 and 5.4 for the MSK and BFSK signals. Compare the spectrums of BASK, MSK and BFSK signals.

**6 Conclusion**

Make a conclusion about each item of the work, analyse the obtained results (coincidence of theoretical and experimental data showing the properties of signals, etc.).

## LW 2.1 RESEARCH INTO ALGORITHMS OF EFFECTIVE CODING OF DISCRETE MESSAGES SOURCES

**1 Objectives**

1.1 The study of the information characteristics of discrete messages sources and postulates of messages effective coding.

1.2 The study of algorithms and investigation of Huffman and Shannon-Fano effective coding.

**2 Main postulates**

See textbook [2, subsections 2.1 – 2.7].

**3 Questions**

3.1 What is information?
3.2 How is the quantity of information defined in the message?
3.3 What is the source entropy? When is it maximal?
3.4 What is the source redundancy and what are its causes?
3.5 Formulate a Shannon coding theorem for a noiseless channel.
3.6 What is the main principle of an effective code?
3.7 What is a prefix code?
3.8 Describe the algorithm of Huffman coding.
3.9 Describe the algorithm of Shannon-Fano coding.
3.10 List the disadvantages of Huffman and Shannon-Fano effective codes.

**4 Homework**

4.1 The source of discrete messages with alphabet size $M_A = 5$ is set. The number of occurrences of symbols is set in table 1. It is necessary to construct Huffman tree and Shannon-Fano table, write down Huffman code and Shannon-Fano code, define $H(A)$, $K_{\text{red}}$, $\bar{n}$, μ and η.

**Table 1** – The number of symbol hits

| Team number | The number of symbols appearances | | | | |
|---|---|---|---|---|---|
| | A | C | B | F | D |
| 1 | 30 | 25 | 20 | 15 | 10 |
| 2 | 40 | 20 | 18 | 12 | 10 |
| 3 | 48 | 25 | 12 | 10 | 5 |
| 4 | 33 | 30 | 17 | 18 | 2 |
| 5 | 43 | 18 | 14 | 13 | 12 |
| 6 | 37 | 29 | 16 | 10 | 8 |

4.2 Be ready to answer the questions.

## 5 Laboratory work

5.1 **Run the program "Huffman coding" / "Shannon-Fano coding"**, using the folder **TT(English)** on the desktop. Study the model.

The laboratory work is done on a computer using a virtual model that is implemented in MatLab. The structure of laboratory model includes: a source of discrete messages, a generator of random numbers and Huffman and Shannon-Fano coders. Alphabet size $M_A$ can be set from 2 to 16. The encoder works in a step mode, the output is displayed on the screen.

5.2 **Investigate the source of discrete equiprobable messages.** Set the size of the alphabet 5, then create fields, and leave symbols equiprobable. Run the algorithm. Write codewords, the value of entropy, and calculate the average length of codewords. Compare the result with the uniform code.

Press the "Clear" button. Repeat the experiment for equiprobable messages with the alphabet size 8. Make your conclusions and write them down.

5.3 **Investigate the source of discrete not equiprobable messages.** Set the size of the alphabet 5, and then create fields, that have the number of occurrences of symbols from your homework. Run the algorithm. Compare the results with homework. Write down conclusions.

Set the size of the alphabet from 10 to 16. Create a field with random occurrences of symbols by pressing button "random". Run the algorithm. Draw the coding process and write the value of entropy. Write codewords, calculate the average length of codewords, the degree of compression and the compression factor. Write down conclusions about the decoding of messages.

5.4 **Investigate the source of discrete messages with the maximum coding efficiency.** Set the size of the alphabet 6. Set the probability symbols equal to the negative powers of two (16; 8; 4; 2; 1; 1). Run the algorithm. Write codewords, calculate the average length of codewords. Compare it with the entropy. Write down conclusions.

## 6 Conclusion

Make a conclusion about each item of the work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

# LW 2.4 RESEARCH INTO BASEBAND TELECOMMUNICATION SIGNALS SAMPLING

## 1 Objectives

1.1 The study of continuous signal sampling and recovery of a continuous signal from samples.

1.2 The analysis of discrete signals characteristics.

## 2 Main postulates

See textbook [1, subsection 2.7].

## 3 Questions

3.1 Explain the physical essence of a continuous signal sampling.

3.2 What is continuous signals sampling carried out for?

3.3 Explain the dependence of the continuous and discrete signals spectrum.

3.4 Explain the physical essence of a signal on a sample recovery process.

3.5 Formulate Kotelnikov's theorem (Nyquist's–Shannon's sampling theorem).

3.6 Write down Kotelnikov's series for a signal with the band-limited spectrum.

3.7 What are the basic differences of an amplitude response and a phase response of an ideal LPF and real LPF?

3.8 What are the reasons for errors which arise at signal restoration on samples?

## 4 Homework

4.1 Signal $s(t) = A_1\sin(2\pi f_1 t) + A_2\sin(2\pi f_2 t) + A_3\sin(2\pi f_3 t)$ have a sampling with the frequency $f_s$. For the data specified in table 1 (according to the number of your laboratory team), represent a spectrum of the signal $S(f)$ and a discrete signal spectrum $S_d(f)$, constructed in a range of frequencies $0 \le f \le 2f_s$.

**Table 1** – Initial data for homework

| Team number | $A_1$, V | $f_1$, kHz | $A_2$, V | $f_2$, kHz | $A_3$, V | $f_3$, kHz | $f_s$, kHz |
|---|---|---|---|---|---|---|---|
| 1 | 0 | 1 | 1 | 2 | 3 | 1,5 | 5 |
| 2 | 1 | 1 | 5 | 2,5 | 4 | 1,5 | 5,5 |
| 3 | 2 | 0,5 | 4 | 1,5 | 3,5 | 2,5 | 6 |
| 4 | 3 | 1 | 3 | 1,5 | 2,5 | 2,5 | 6,5 |
| 5 | 2,5 | 0,5 | 2 | 2,5 | 1 | 3 | 7 |
| 6 | 1,5 | 1 | 5 | 2,5 | 4 | 3 | 7,5 |
| 7 | 3,5 | 0,5 | 4 | 2 | 3 | 3 | 8 |
| 8 | 1 | 0,5 | 3 | 1 | 3,5 | 2 | 7 |

4.2 Calculate and draw impulse response ideal LPF with the cut frequency $F_{cut} = f_s/2$ for values $t$, belonging to an interval $(-4T_s, 4T_s)$ (it is necessary to take the value $f_s$ from table 1).

4.3 Be ready to answer the questions.

### 5 Laboratory work

5.1 **Study the virtual model.** Run the program **2.4**, using an icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. The virtual model block diagram is given on figure 1.

The laboratory model consists of: a continuous signal source $s(t) = A_1\sin(2\pi f_1 t) + A_2\sin(2\pi f_2 t) + A_3\sin(2\pi f_3 t)$, a sampler, a recovering LPF, the generator of sampling pulses and the δ-pulse generator. It is possible to set the frequencies and amplitudes of a harmonic waveform $A_1, f_1, A_2, f_2, A_3, f_3$, sampling frequency $f_s$ and cut frequency LPF $F_{cut}$ values.

The switch gives the chance to supply a discrete signal $s_d(t)$ or δ-pulse at an input recovering LPF. The time and spectral diagrams can be observed in the three points of a laboratory model: at a source output, at an LPF input and at an LPF output.

5.2 **Investigate sampling process in time and frequency domains.** Set values $A_1, f_1, A_2, f_2, A_3, f_3$ and frequency $f_s$, given in your homework, to designate LPF input from the sampler and run the program. Write the signals at the source and sampler outputs of oscillograms and the spectrograms in the report. Compare the spectrograms calculated and received on a laboratory model. Enter the results of comparison in the report. Increase the sampling frequency by 1 kHz. Write the signal spectrogram at a sampler output in the report, make conclusions.

5.3 **Investigate the characteristics for recovering LPF.** Select the influence on LPF as δ-pulse and set LPF cut frequency value which was given in your homework. Enter the impulse response and AR LPF in the report. Compare impulse response LPF with the calculated one. Set the sampling frequency twice smaller, enter the impulse response and AR LPF in the report, make conclusions.

5.4 **Investigate continuous signal recovery process in time and frequency domains.** Set parameters $A_1, f_1, A_2, f_2, A_3, f_3, f_s$ and $F_{cut}$, given in your homework, enter at the LPF input a signal from the sampler. Compare oscillograms and spectrum at LPF output and at a source output, make conclusions.

Set LPF cut frequency smaller then frequency $f_3$, and then bigger then $f_s - f_3$. In both cases draw oscillograms and spectrograms at LPF output, describe the recovery error character, explain the reasons of errors.

### 6 Conclusion

Make a conclusion about each item of the laboratory work.

## LW 2.5 RESEARCH INTO ANALOG SIGNALS CODING BY PCM AND DPCM

### 1 Objectives

1.1 The study of the digital transmission of analog signals with PCM and DPCM methods.

1.2 The research into the basic characteristics of the transmission system using the PCM and DPCM methods.

## 2 Main postulates

See textbook [2, subsections 2.8 – 2.13].

## 3 Questions

3.1 Explain the digital signal forming principle in the PCM method transmission system.
3.2 How is the sampling interval or the sampling frequency determined?
3.3 What is the quantization step and how can it be defined?
3.4 What does the codeword length depend on in PCM systems?
3.5 What is quantization noise? What is the reason of its origin?
3.6 How is it possible to increase the signal/(quantization noise) ratio in the transmission systems using the PCM methods?
3.7 Explain the principle of forming a digital signal in the DPCM transmission system.
3.8 What is the difference between the transmission systems using PCM and DPCM?

## 4 Homework

4.1 Represent the structure schemes of a PCM encoder and decoder.
4.2 Calculate the characteristics of an analog signal transmission system by the PCM method with a uniform quantization. Given: the sample frequency is equal to 8 kHz; the quantization level quantity $L_1 = 2^{N+2}$ ($N$ is the team number); $L_2 = 2L_1$; $L_3 = 2L_2$. Calculate for three values of $L$: the quantization step $\Delta b$ ($|b(t)|_{\max} = 1$ V$^2$); the code length $n$; the average quantization noise power $\overline{\varepsilon_q^2}$ (the analog signal amplitude coefficient $K_A$ = 2,85); the signal/(quantization noise) ratio $\rho_q$, dB; the digital signal rate $R$. Enter the results of your calculations in table 1. Analyse how $\rho_q$ and $R$ will change if the code length changes by a unit.

**Table 1 –** The characteristics of the transmission system by the PCM method

| $f_s$, kHz | $L$ | $n$ | $R$, kbits/s | $P_b$, V$^2$ | $\Delta b$, V | $\overline{\varepsilon_q^2}$, V$^2$ | $\rho_q$, dB |
|---|---|---|---|---|---|---|---|
| | $L_1$ | | | | | | |
| | $L_2$ | | | | | | |
| | $L_3$ | | | | | | |

4.3 Be ready to answer the questions.

## 5 Laboratory work

**5.1 Study a virtual model.** Start the program **2.5a**, use the icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display.

The analog signal generator forms the signal as a sum of a few harmonic waves. The signal duration is equal to 2 ms, and the maximum frequency of its spectrum is $F_{\max} = 3{,}4$ kHz. The analog signal is normalized so that $|b(t)|_{\max} = 1$ V.

**5.2 Determine the origin of quantization noise for PCM.**

Set the number $L_1$ (you obtained in your homework) of quantization levels. Present the results of doing the program like in table 2: write down the $\Delta b$ and $b(kT_s)$, $i(kT_s)$, $b_q(kT_s)$ for $k$ = 1, 2, 3 and 4. Calculate the values of $\varepsilon_q(kT_s)$, compare them with $\Delta b$, $\overline{\varepsilon_q^2}$, and explain the result. Repeat the experiment for $L_2$.

**Table 2** – Analysis of a quantization error

| $L$ | $\Delta b$ | $k$ | $b(kT_s)$, V | $i(kT_s)$ | $b_q(kT_s)$, V | $\varepsilon_q(kT_s)$, V |
|---|---|---|---|---|---|---|
| | | 1 | | | | |
| | | ⋮ | | | | |
| | | 4 | | | | |
| | | 1 | | | | |
| | | ⋮ | | | | |
| | | 4 | | | | |

**5.3 Determine the signal/(quantization noise) ratio.** Set the quantization level number $L_1$ (from your homework). Write down the results of the program run in the table (like in table 1): the values of $\Delta b$, $\overline{\varepsilon_q^2}$ and $P_b$. Calculate the experimental value of $\rho_q$. Compare the got values $\Delta b$, $\overline{\varepsilon_q^2}$ and $\rho_q$ with the results of the calculations in your homework. Repeat the task for $L_2$ and $L_3$. Having obtained results, build the graphs of signal/(quantization noise) and digital signal rate as the functions of the number of quantization levels.

**5.4 Determine the origin of the quantization noise for DPCM.** Start the program **2.5b**, using an icon **TT(English)** on a desktop.

Set the number of quantization levels $L_1$ (from your homework). Insert the results of the program run in the table (like in table 1): the values of $\Delta b$, $\overline{\varepsilon_q^2}$ and $P_b$. Calculate the experimental value of $\rho_q$. Compare the obtained values $\Delta b$, $\overline{\varepsilon_q^2}$ and $\rho_q$ with the results for PCM. Repeat the task for $L_2$ and $L_3$.

## 6 Conclusion

Make a conclusion about each item of the work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

# LW 3.1 RESEARCH INTO MATCHED FILTERS

## 1 Objectives

Study and experimental verification of the matched filters' properties.

## 2 Main postulates

See textbook [1, subsections 2.2, 5.4, 5.5].

## 3 Questions

3.1 What filter is called matched?

3.2 What parameters of a signal must be known for a matched filter (MF) synthesis?

3.3 Write down expressions for AR and PR of MF. Give physical interpretation for them.

3.4 How is the impulse response of MF determined?

3.5 What is the condition of physical realization of MF?

3.6 What form does the MF response have at its input signal which it is matched with?

3.7 How is the signal/noise ratio determined at the MF output?

3.8 Draw the chart of a filter matched with a rectangular pulse.

3.9 Explain Barker`s sequences (codes) settings and their properties.

## 4 Homework

4.1 Calculate and build the normalized correlation function graph of the set signal (table 1).

**Table 1 –** Basic data to do the homework

| Team number | Signal |
|---|---|
| 1, 5 | Videopulse, $T_s = 2$ ms |
| 2, 6 | Radiopulse with a rectangular envelope, $T_s = 2$ ms, $f_0 = 1000$ Hz |
| 3, 7 | Videopulse, $T_s = 4$ ms |
| 4, 8 | Radiopulse with a rectangular envelope, $T_s = 4$ ms, $f_0 = 1000$ Hz |

4.2 Be ready to answer the questions.

## 5 Laboratory work

5.1 **The study a virtual model.** Start the program **3.1**, use the icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display.

The signal generator is intended for the formation of the signals $s(t)$:

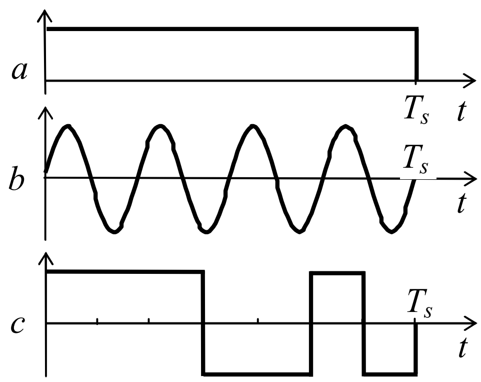


**Figure 1** – Researching signals

a) a single videopulse
b) a radiopulse with a rectangular envelope;
c) a complex signal on the basis of Barker`s 7-elements sequence.

Parameters of the signal generator:

– the signals listed above;
– the duration of a signal $T_s$ 4 or 2 ms;
– the frequency of a radiopulse $f_0$ is 1000 Hz or 1500 Hz.

The noise generator produces the realization of a noise $n(t)$ with an average power $P_{\text{n in}}$ 0,1 or 1 $V^2$, a quasi-white noise in the bandwidth of frequencies $f_{\text{n}} = 35$ kHz.

A model also contains the δ-function generator.

The switch $S$ gives $s(t)$, $n(t)$, $s(t)+n(t)$ or $\delta(t)$ at the matched filter input.

A model contains measuring devices and indicators of average powers of processes $z(t)$ (at the MF input) and $y(t)$ (at the MF output), oscillographs and spectrum analyzers.

5.2 **The research into time and spectral characteristics of the signals used in work.** Draw the time diagrams $s(t)$ and amplitude spectrum $S(f)$ of signals in the report: videopulse, radiopulse and complex signal at the duration either 4 or 2 ms, and the radiopulse at frequencies either 1000 or 1500 Hz. Write down the values of average powers of signals at the MF input $P_{\text{s in}}$.

5.3 **The research into impulse responses and AR of filters.** Feed delta-function at the input of the matched filter. Draw the MF impulse responses $g(t)$ and AR $H(f)$ in the report (by the way, an amplitude spectrum at the output of a filter repeats its AR). Do the task for the signals: from homework and the ones set by your teacher.

5.4 **The research into responses of filters for signals which they are matched with.** Do the task for the signals: from homework and the ones set by your teacher. Draw a graph of the responses of filters on signals which they are matched with in the report. Write down the sampling values of signals at the MF output $y_{\text{s}}(t_0)$. Set the signals and filters matched with them in turn to complete the task.

*Note*. At the MF output time diagrams must repeat the correlation functions of signals displaced at times.

5.5 **Define the gain in the signal/noise ratio, which is provided by MF.** For this purpose:

– feed at the input of filter the sum of the signal $s(t)$ (for example, videopulse) and noise $n(t)$ with the power 0,1 $V^2$ or 1,0 $V^2$; set a filter, matched with an input signal from your homework; when you have run the program compare temporal diagrams at an MF input and output, make sure a filter has considerably weakened the noise;

– start the program at the disconnected signal execution; write down the values of average powers of noise at the MF input $P_{n\,\text{in}}$ and at the MF output $P_{n\,\text{out}}$; power of the initial signal in a sampling moment $P_{\text{s out}}$. It is determined as a square of sampling of the maximum value of the initial signal (task 5.4) $P_{\text{s out}} = y_{\text{s}}^2(t_0)$; the value $P_{n\,\text{in}}$ is certain for 5.2: $P_{n\,\text{in}}$ equals the square of an amplitude for a videopulse and a complex signal, and $P_n$ is half as much as the square of the amplitude for a radiopulse;

– expect the gain in signal/noise ratio, provided by MF,

$$g_{\text{MF}} = \frac{P_{\text{s out}} / P_{\text{n out}}}{P_{\text{s in}} / P_{\text{n in}}}.$$

## 6 Conclusions

Make a conclusion about each item of the work, analyse the obtained results (coincidence of theoretical and experimental data, compare the time and spectral diagrams, obtained as a result of the objective and 5.2 task, find the connection between time and spectral characteristics of signals and filters matched with the signals, analyse the compliance of MF properties, compare the obtained values with the expected value of the gain $2F_nT_s$).

## LW 3.2 RESEARCH INTO OPTIMAL DEMODULATORS OF BINARY MODULATION SIGNALS SCHEMES

### 1 Objectives

1.1 The study the functional diagrams of optimal coherent demodulators of BASK, BFSK and BPSK signals.

1.2 The research into transformations of signals in the separate demodulator blocks.

### 2 Main postulates

See textbook [1, subsections 5.1 – 5.8].

### 3 Questions

3.1 Explain the setting of a modulator and a demodulator during the transmission of digital signals.

3.2 Represent the time diagrams of BASK, BPSK and BFSK signals for a digital signal 101100.

3.3 Formulate the criterion of optimal demodulators of a digital modulation.

3.4 Represent the functional diagrams of BASK, BPSK and BFSK signal demodulators and explain what transformations are executed by every block.

3.5 What is a coherent detector and what is the purpose of a recovering carrier scheme?

3.6 What is the role of the matched filters in the demodulator schemes?

3.7 What is the role of the clock recovery scheme?

3.8 Formulate the process for the decision of BASK, BPSK and BFSK signals demodulation.

### 4 Homework

4.1 Represent the functional diagrams of BASK, BPSK and BFSK signals demodulators.

4.2 Write down a number 48 + $N$ in the binary scale of notation ($N$ is the number of your team). Represent the obtained digital signal, considering that the rate $R$ = 1000 bit/s. Represent the time diagrams of BASK, BPSK and BFSK signals, if $A(t)$ is a videopulse, $f_0$ = 4 kHz, $f_1$ = 5 kHz.

4.3 At the input of a BASK signal demodulator the BASK signal which was obtained in 4.2 is given. Represent the time diagrams of signals in all points of a demodulator functional diagram: at each input and output of a multiplier, at the output of the matched filter, at the output of a sampler, at the output of a decision.

4.4 Be ready to answer the questions.

## 5 Laboratory work

5.1 **The study a virtual model.** Start the program **3.2a**, use the icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display.

A **model 3.2a** is intended for the research into demodulators of BASK and BPSK signals. It contains:

– the generator of a digital signal with a speed of 1000 bit/s, a signal consists of 6 binary symbols (bit), symbols are set on the panel of a model;

– the modulator of signals – BASK or BPSK is set, the envelope radiopulses are rectangular, the frequency of a carrier is 4 kHz;

– a model of a communication channel that summarizes the signal and the noise;

– the noise generator, which gives the possibility to include and turn it off, and also set the level of noise;

– the demodulator of BASK and BPSK signals is placed after a scheme. If the model of a modulation type changes, there is a corresponding change in the rule of decision in a deciding scheme. The carrier recovery and clock recovery schemes create “ideal” oscillations; at BPSK it is possible to change the phase of a picked up carrier oscillation to 180° to imitate an inversion work;

– 5 oscillographs for the supervision of sentinel diagrams in all points of a demodulator functional diagram.

A **model 3.2b** is intended for the research into demodulator of BFSK signal. It contains:

– the generator of a digital signal with a speed of 1000 bit/s, the signal consists of 6 binary symbols (bit), symbols are set on the panel of a model;

– the modulator of a BFSK signal, the envelope radiopulses are rectangular, the frequencies of radiopulses are $f_0 - \Delta f/2 = 4$ kHz and $f_0 + \Delta f/2 = 5$ kHz;

– a model of a communication channel is a undistorted quadripole;

– the demodulator of a BFSK signal is placed after a scheme;

– 5 oscillographs for the supervision of sentinel diagrams are set in all points of a demodulator functional diagram; oscillographs are commuted for the supervision of sentinel diagrams in the upper or the lower subchannel.

5.2 **Signals transformations in BASK signal demodulator researching.** For this purpose it is necessary to set a digital signal, used in your homework to turn off the noise. Start the program. Draw the time diagrams from oscillographs on the model panel. Compare the results with those you got in your homework.

5.3 **Signals transformations in BPSK signal demodulator researching.** The statement of the problem is the same as in 5.2. The phase of a recovering carrier is equal to 0 and 180°. Draw the time diagrams from oscillographs on the model panel. Describe the differences of oscillogramms at the different phases of a recovered carrier.

5.4 **Signals with noise transformations in BASK signal demodulator researching.** The statement of the problem is the same as in 5.2, but the noise generator is on and the noise level is a two conditional unit. Run the program and check, whether there was an error in the accepted digital signal. If it did not arise up, repeatedly start the program to find one error. Draw the time diagrams from oscillographs on the model panel. Explain the reason of the origin of errors.

5.5 **Signals with noise transformations in BPSK signal demodulator researching.** The statement of the problem is the same as in 5.4. The phase of the recovered carrier is equal to 0. Draw the time diagrams from oscillographs on the model panel to explain the reason of errors origin.

5.6 **Signals transformations in BFSK signal demodulator researching**. Start the program 3.2a, use the icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display. Set a digital signal. Draw the time diagrams from oscillographs on the model panel at first in the upper subchannel, and then in the lower.

**6 Conclusions**

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

## LW 3.3 RESEARCH INTO DIFFERENTIALLY ENCODED PHASE MODULATION

**1 Objectives**

1.1 The study of the differential coding and decoding principles.

1.2 The research into the influence of errors over the results of the differential decoding in the communication channel.

**2 Main postulates**

See textbook [1, subsection 5.11].

**3 Questions**

3.1 Why are differential methods of transmission used?

3.2 Explain the rules of binary and quaternary symbols coding and decoding.

3.3 Explain the principle of the BDPSK modulator and demodulator operation.

3.4 Explain the principle of the *M*DPSK ($M > 2$) modulator and demodulator operation.

3.5 What is an error multiplication?

**4 Homework**

4.1 Draw block diagrams of binary and quaternary symbols of differential coders and decoders.

4.2 Write the number 896 + $N$ in the binary system ($N$ is the number of your team). Do the coding and decoding of this digital signal by binary differential code for $b_0^d = 0$ and $p = 0$. Present your results in a table, use [1, table 5] as an example.

4.3 Do the coding and decoding of a digital signal received in 4.2 by quaternary mapping Gray code, and then by a differential quaternary code for $q_0^d = 0$. Do the differential decoding for $p = 0$ mapping decoding. Present your results in a table, use [1, table 9] as an example.

4.4 Be ready to answer the questions.

**5 Laboratory work**

**5.1 The study of a virtual model.**

Run the program **3.3a** (a binary differential coding) and then **3.3b** (a quaternary differential coding), using the icon “Laboratory works” on a desktop, and then the folder TT(English). Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display.

Model **3.3a** is designed to study differential binary coding and decoding. It includes:

– a digital signal generator which generates 10 bits, the bits are set on the model panel;

– a differential coder, $b_0^d$ is set on the model panel;

– a binary channel with uncertainty of the second order (that is the model with BDPSK communication channel); phase shift $p$ takes values 0 or 1 randomly. It is possible to set the transmission channel without an error or with an error, in the mode of a transmission with an error, the error occurs randomly in one of the symbols, the indicator indicates the number of the symbol with an error;

– a differential decoder.

Model **3.3b** is designed to study a differential quaternary coding and decoding. It includes:

– a digital signal generator, which generates 10 bits, the bits are set on the model panel;

– a mapper;

– a differential coder, $q_0^d$ is set on the model panel;

– a quaternary channel with uncertainty of the fourth order (that is the model with QPSK communication channel); phase shift $p$ takes values 0, 1, 2 or 3 randomly. It is possible to set the transmission channel without an error or with an error, in the mode of transmission with an error, the error occurs randomly in one of the symbols, the indicator indicates the number of the symbol with an error;

– a differential decoder;

– a mapping decoder.

**5.2 Research into binary differential coding and decoding.**

It is necessary to set:

– a digital signal which was used in your homework;

– the differential coder initial state $b_0^d = 0$;

– the transmission mode of the communication channel "without errors".

Run the program **3.3a**. If the shift in the communication channel is $p = 1$, then rerun the program until $p = 0$. Write down a sequence of digital signals, filling the table (example: [1, table 5]). Compare it with the results of your homework.

**5.3 Research into binary differential coding and decoding for the $b_0^d = 1$ initial state of the differential coder**.

Other conditions are the same as in 5.2. After running the program **3.5a** with $p$ = 0 write down a sequence of digital signals, filling in the table (example: [1, table 5]). Compare it with the results in 5.2.

**5.4 Research into binary differential decoding with a phase shift in the channel $p = 1$.**

State $b_0^d$ is arbitrary, the transmission mode of communication channel is “without errors”. Run the program **3.3a** until $p = 1$. Write down a sequence of digital signals, filling in the table (example: [1, table 5]). Compare it with the results in 5.2 and 5.3.

**5.5 Research into binary differential decoding in transmission mode of the communication channel "with errors".**

The shift in the channel $p$ and state $b_0^d$ are arbitrary. After running the program **3.3a** write down a sequence of digital signals, filling in the table (example: [1, table 5]). Analyze the results, drawing attention to the errors multiplication in a decoding.

**5.6 Research into quaternary differential coding and decoding.**

Run program **3.3b** in two modes in turn: $q_0^d = 0$ and $q_0^d \neq 0$.

It is necessary to set:

– a digital signal which was used in your homework;

– the differential coder initial state $q_0^d = 0$ or $q_0^d \neq 0$;

– a transmission mode of a communication channel "without error".

Run the program **3.3b**. If the shift in the communication channel $p \neq 0$, then rerun the program to perform until $p = 0$. Write down a sequence of digital signals, filling in the table (example: [1, table 9]). Compare it with the results of your homework.

**5.7 Research into quaternary differential decoding with phase shift in the channel $p \neq 0$.**

State $q_0^d$ is arbitrary, a transmission mode of a communication channel is “without errors”. Run the program **3.3b** until $p \neq 0$. Write down a sequence of digital signals, filling in the table (example: [1, table 9]). Compare it with the results in 5.6.

**5.8 Research into quaternary differential decoding in a transmission mode of the communication channel "with errors".**

The shift in the channel $p$ and state $q_0^d$ are arbitrary. After running the program **3.3b** write down a sequence of digital signals, filling in the table (example: [1, table 9]). Analyze the results, pay attention to the multiplication of errors in decoding.

## 6 Conclusions

Make conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

# LW 3.4 RESEARCH INTO OPTIMAL DEMODULATOR NOISE IMMUNITY OF DIGITAL MODULATED SIGNALS

## 1 Objectives

1.1 The study of the method of experimental research into the digital modulated signals noise immunity.

1.2 The experimental research into noise immunity of the following signals: BASK, BFSK, BPSK, BDPSK, QPSK.

## 2 Main postulates

See textbook [1, subsection 5.9].

## 3 Questions

3.1 What is the error rate, the error ratio, the error probability?

3.2 What is the signal noise immunity?

3.3 Write down and explain expressions for the calculation of the bit error probability at the optimum demodulation BASK, BFSK, BPSK, BDPSK and QPSK signals.

3.4 Explain, how bit error probability depends on the digital signal rate when $P_s$ and $N_0$ are constants.

3.5 Compare BASK, BFSK, BPSK, BDPSK and QPSK signals noise immunity of optimum demodulation at the fixed value $h_b^2$.

3.6 What is the quantitative measure of difference between signals? Compare the difference between BASK, BFSK, BPSK and QPSK signals.

3.7 How do you calculate the necessary communication channel bandwidth for the BASK, BFSK, BPSK, BDPSK and QPSK signals?

3.8 How can we experimentally measure the SNR?

## 4 Homework

4.1 Calculate the table of the bit error probability values for the BASK, BFSK, BPSK, BDPSK and QPSK signals, increasing the value $h_b^2$ from 2 to 10 dB with a step 1 dB.

The calculations of error probability are possible to made in the mathematical packages MathCAD and MatLab, using the error function erf(•):

$$Q(x) = 0{,}5\left(1 - \operatorname{erf}\left(x/\sqrt{2}\right)\right) \tag{1}$$

The following formula provides high accuracy of calculations for values $Q(x) > 10^{-10}$

$$Q(x) = 0{,}65\ \exp(-0{,}44(x + 0{,}75)^2). \tag{2}$$

Using calculation results build graphs of $p$ from SNR $h_b^2$ dependence. Build graphs. Use one page in your copybook for this graph.

4.2 Represent a block diagram for researching into digital modulated signals noise immunity.

4.3 Be ready to answer the questions.

## 5 Laboratory work

5.1 **The study of a virtual model.** Start the program **3.3**, use the icon **TT(English)** on a desktop. Click on the virtual model icon in the HP VEE environment to do the laboratory work. Study the laboratory model scheme on the computer display. The virtual model contains:

- a digital signal source which produces equiprobable symbols 1 and 0; digital signal rate $R$, bit/s, are set on the panel of the model;
- a modulator which forms the BASK, BFSK, BPSK, BDPSK and QPSK signals, the average power of a signal, for all modulation types $P_s = 1$ V$^2$, the modulation type is set on the panel of the model;
- a key in the path of a modulated signal allows to connect and disconnect the modulator output from the input of a communication channel;
- a noise generator, producing realization of white noise with the Gaussian probability distribution, the value of the power spectral density $N_0$ is set on the panel of the model;
- an attenuator for noise attenuation; it is possible to set attenuation from 0 to 10 dB with a step 1 dB or to turn off noise on the panel of the model;
- a communication channel which forms the sum of a signal and noise, the communication channel bandwidth $F_{ch}$, Hz is set on the panel of the virtual model;
- a meter of an average power, connected with a communication channel output;
- a demodulator for demodulation of the BASK, BFSK, BPSK, BDPSK and QPSK signals provides optimal demodulation of a signal which is in a communication channel. The algorithm of demodulation changes properly with a change of set modulation type in the modulator;
- a comparator of a bit at the modulator input and a bit at the demodulator output, if these bits differ, then a signal about the error at demodulation is formed;
- a counter of error bits $N_{err}$;
- a counter of transmitted bits $N_{all}$;
- indicators of measured average power at the communication channel output, the number of the of error bits $N_{err}$ and the number of the transmitted bits $N_{all}$.

The program is stopped when the desired value of the transmitted bit number $N_{all} = 10000$ is completed. If the number of error bits is insufficient for the calculation of error probability, the restart of the program is executed, and the corresponding values $N_{err}$ and $N_{all}$ are added to calculate the error probability.

5.2 **The calibration of SNR:**

1) Measure the average power of a signal. It is necessary to set:

– a digital signal rate $R$ from the interval 1000–10000 bits/s;

– the modulation type is arbitrary;

– a signal is "On", noise is "Off";

– the communication channel bandwidth must satisfy the condition $F_{ch} \geq F_s$, the signal bandwidth $F_s$ ;

– the measuring of "Average power".

Run the program and write down the measured value of the signal average power $P_s$.

2) Calculate and set the power spectral density of noise $N_0$, at which SNR $h_b^2 = 1$, i.e. 0 dB. Calculate $N_0 = P_s/R$ using the definition $h_b^2 = \dfrac{E_b}{N_0} = \dfrac{P_s T_b}{N_0} = \dfrac{P_s}{N_0 \cdot R} = 1$. Use the symbol $m$ for $10^{-3}$ and $u$ for $10^{-6}$ at setting the value $N_0$.

For example, BPSK is set, $R$ = 10 kbit/s, $F_{ch}$ = 12 kHz; measured $P_s$ = 1 V$^2$; calculated $N_0 = 10^{-4}$ V$^2$/Hz. It is necessary to set $N_0 = 0.1m$.

Turn off a signal, set attenuation in the noise path 0 dB, run the program and write down the measured value of the average noise power $P_n$. Make sure that $P_n = F_{ch} \cdot N_0$. In the further measurements of error probability do not change $R$ and $N_0$ settings, and the SNR change by setting of the proper attenuation in the noise path: the attenuation of noise, represented in decibels, sets the same SNR value $h_b^2$ in decibels.

5.3 **The measuring of error probability** (coefficient of error). To do it set:

- measure of "Coefficient of errors";
- a signal "On";
- a modulation type is set by your teacher (BASK, BFSK, BPSK, BDPSK or QPSK);
- the communication channel bandwidth is set from condition $F_{ch} \geq F_s$, and the signal bandwidth $F_s$ .

Make a table according to the sample table 2. Set the attenuation of noise from 3 to 9 dB (for BASK, BFSK) and from 2 to 6 dB (for BPSK, BDPSK and QPSK) with a step 1 dB and run the program, fill in the columns of your table: the modulation type, the SNR $h_b^2$, the bit error number $N_{err}$, the number of the transmitted bits $N_{all}$. Complete execution of the program provides $N_{all}$ = 10000.

**Table 2** – Results of error probability measuring

| Modulation type | Communication channel bandwidth $F_{ch}$, Hz | SNR $h_b^2$, dB | Number of error bit, $N_{err}$ | Number of the transmitted bits, $N_{all}$ | Error probability (error ratio) $p$ |
|---|---|---|---|---|---|
| BASK | | 3 | | | |
| | | 4 | | | |
| | | ⋮ | | | |
| BFSK | | 3 | | | |
| | | ⋮ | | | |

Having measured the results, calculate the error ratio; if the number of the transmitted bits $N_{all}$ is large enough, take the calculated value of an error ratio as the error probability.

Build the graphs of dependences $p = f(h_b^2)$ to measure the results and calculations of all modulation types. Build the graphs in the picture, where the results of your homework are presented.

5.4 **The measure of an error probability at the changed communication channel bandwidth**. To set the communication channel bandwidth, multiply it 1,5...2 times, and repeat the measurements of error probability for one of the modulation types. Make sure that error probability does not depend on the communication channel bandwidth and the value of the noise average power at the demodulator input; it depends on the noise power spectral density $N_0$.

**6 Conclusions**

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

## LW 4.1 THE STUDY OF BLOCK ERROR-CONTROL CODES

**1 Objectives**

1.1 The study of error-control code principles.

1.2 The study of Hamming systematic code (7, 4) codec structure.

1.3 The experimental study of a block code encoder and decoder operation postulates.

**2 Main postulates**

See textbook [2, section 5].

**3 Questions**

3.1 Give the definition of error-control codes, systematic error-control codes.

3.2 Explain the purpose of error-control code encoder and decoder.

3.3 What is a code distance?

3.4 What is redundancy and a code rate?

3.5 Write down expressions for determining the code control ability with a given code distance.

3.6 What codes are named cyclic?

3.7 How do you write codewords as polynomials?

3.8 What is a generator matrix? How do you build a generator matrix for a systematic code?

3.9 What is a check matrix?

3.10 What is a syndrome?

3.11 Explain the principle of Hamming code decoder construction.

3.12 Explain cyclic codes encoding and decoding principles.

**4 Homework**

4.1 A code (7, 4) is set by a generator matrix

$$\mathbf{G} = \begin{pmatrix} 1 & 0 & 0 & 0 & 0 & 1 & 1 \\ 0 & 1 & 0 & 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 0 & 1 & 1 & 0 \\ 0 & 0 & 0 & 1 & 1 & 1 & 1 \end{pmatrix}.$$

Write down the team number $N$ in binary numbers. Consider these four digits as an information word at the encoder input, calculate the codeword at the coder output. Create the given code check matrix and a code syndromes table. Put the first order error into the symbol $b_N$ of a formed codeword; calculate a syndrome for the codeword at the decoder input. Make sure a syndrome corresponds to the error symbol $\hat{b}_N$.

4.2 Write down the team number $N + 8$ in binary numbers. Consider this number as an information word, the length $k = 5$ at the encoder input. Form a cyclic code (10, 5) codeword, using the generated polynomial $g(x) = x^5 + x^4 + x^2 + 1$.

4.3 For three error words $e_1(x)$, $e_2(x)$ and $e_3(x)$, given in table 1, calculate syndromes, and then, using table 2, define the decoders work result. If the calculated syndrome is written in table 2, a decoder inverts a codeword symbol which is considered as an error. If the calculated syndrome isn't in table 2, a decoder does not change codeword symbols which are decoded. The codeword at the decoder output is the first $k$ symbols of the codeword at the decoder input.

**Table 1** – Polynomials of errors for the homework

| Team number $N$ | $e_1(x)$ – single error | $e_2(x)$ – doubled error | $e_3(x)$ – triple error |
|---|---|---|---|
| 1 | $x^9$ | $x^9 + 1$ | $x^7 + x^6 + 1$ |
| 2 | $x^8$ | $x^6 + x^8$ | $x^9 + x^8 + x^4$ |
| 3 | $x^7$ | $x^6 + x$ | $x^9 + x^8 + x^2$ |
| 4 | $x^6$ | $x^9 + x$ | $x^7 + x^6 + x$ |
| 5 | $x^5$ | $x^7 + x^6$ | $x^8 + x^7 + x^2$ |
| 6 | $x^4$ | $x^7 + x^4$ | $x^8 + x^7 + x^3$ |
| 7 | $x^3$ | $x^9 + x^2$ | $x^8 + x^7 + x$ |
| 8 | $x^2$ | $x^6 + x^3$ | $x^8 + x^6 + x$ |
| 9 | $x$ | $x^7 + x^2$ | $x^9 + x^6 + x^4$ |
| 10 | 1 | $x^8 + x$ | $x^8 + x^6 + x^4$ |

**Table 2 –** Syndromes for first order errors

| Error polynomial $e(x)$ | $x^9$ | $x^8$ | $x^7$ | $x^6$ | $x^5$ | $x^4$ | $x^3$ | $x^2$ | $x$ | 1 |
|---|---|---|---|---|---|---|---|---|---|---|
| Syndrome $s(x)$ | $x^4 + x^3 + 1$ | $x^4 + x^2 + x$ | $x^3 + x + 1$ | $x^4 + x^3 + x^2 + x + 1$ | $x^4 + x^2 + 1$ | $x^4$ | $x^3$ | $x^2$ | $x$ | 1 |

4.4 Be ready to answer the questions.

**5 Laboratory work**

5.1 A laboratory model of the Hamming systematic code (7, 4) is executed on the personal computer. Start program **4.1a** on a computer. A code (7, 4) is described by a generator matrix. The control of a model is done by the left mouse button. Study the method of input data setting. The errors are entered by setting in «1» digit position (positions) of an error block $e_1$, $e_2$, $e_3$, $e_4$, $e_5$, $e_6$, $e_7$ (which must contain an error).

A model produces the conversion of a codeword at the encoder output to a codeword at the decoder input by the rule $\hat{b}_i = b_i \oplus e_i$, for $i = 1, 2, \ldots, 7$.

5.2 **Enter the information bits** obtained in the homework, after coding make sure the results of your homework are correct.

5.3 **Put in an error** into $b_1$, $b_2$, $b_3$, $b_4$, $b_5$, $b_6$, $b_7$ symbols in turns, make sure the decoding and a syndrome table calculated in the homework are correct.

5.4 **Put in a double error** into arbitrary two symbols, make sure, that a decoder tries to correct errors in accordance with a syndrome and puts the third error. Repeat the experiment for the next two-three other double errors.

5.5 **Put in a triple error** into symbols $b_1$, $b_2$, $b_3$. Make sure that a syndrome is equal to zero – it shows that $d_{\min} = 3$. Repeat the experiment putting a triple error into symbols $b_1$, $b_4$, $b_5$. Use a check matrix to define what other triple errors result in the permitted words.

5.6 **Start the program 4.1b on a computer.** It is a virtual model of a cyclic codec, the block diagram of which is presented in figure 1. A model is intended for cyclic codes: (7, 4), (10, 5), (10, 6), (11, 7), (12, 8), (13, 9), (14, 10), (15, 11), encoding and decoding processes research. A model allows to consistently research encoding, transmission over a telecommunication channel and decoding. It contains the red color options which a student should use. The blue color windows are used to indicate model work results.

A coder forms the permitted codeword by the $r = n - k$ checking symbols calculation and adds information symbols to them. The obtained codeword is indicated at the encoder output. It is necessary to enter error words, length $n$, consisting of 1 and 0 for the transmission over a telecommunication channel. The symbol 1 is set in those positions, where an error at the transmission by a telecommunication channel must be. In a telecommunication channel a codeword from an encoder is added with a combination of errors by XOR. The obtained codeword is indicated at the telecommunication channel output.

A decoder divides a codeword into a generate polynomial at its input. A window under a decoder shows the codeword syndrome (in a binary presentation); a left window shows the decoder decision about the error symbol number (all codes studied in the model allow to correct single errors). If a syndrome is zero, a decoder gives the message "No errors". If a syndrome is not zero, a decoder determines the error symbol number and "Error in $x^p$ " message, where $p$ is an error symbol number in the table of syndromes. The table of syndromes contains only $n$ syndromes which correspond to $n$ symbols of the decoded codeword. The possible syndrome number is equal to $2^{n-k}$. If $n < 2^{n-k} - 1$, a syndrome which is absent in the syndromes table can appear in a decoder. In this case a decoder gives out the message "Unknown error". If a decoder defines the error symbol number, it gives out the message "Error in $x^p$". It corrects this symbol and takes away the last $n - k$ symbols. If the syndrome is zero or it is absent in the table of syndromes, a decoder only takes away the last $n - k$ symbols. The information codeword appears at the decoder output.

5.7 **The research into encoding process.** The cyclic code (10, 5) from homework is studied. For this purpose:

- in the menu "What do we study?" choose "Encoding";
- choose a code (10, 5) and set a proper generate polynomial;
- enter an information codeword you received in your homework.

Run the program and compare the codeword at the encoder output with the calculated one in the homework.

5.8 **The study of transmission process over a communication channel.** For this purpose:

- in the menu "What do we study?" choose "Transmission by a channel";
- enter a zero error word, length $n$.

Run the program using the same settings (from 5.6). Make sure that in case of a zero error codeword at the output, "Telecommunication channel" coincides with a codeword at the input.

Set an error word, which corresponds to the single error $e_1(x)$ from Table 2 for your variant. Run the program, compare codewords at the telecommunication channel input and output and make sure the correct process is going on.

5.9 **Research into a decoding process.** For this purpose:

- in the menu "What do we study?" choose "Decoding";
- enter a zero error word.

Run the program using the same settings. Make sure that a syndrome is zero, and a codeword at the decoder output coincides with a codeword at the encoder input.

Set an error word which corresponds to the single error $e_1(x)$ from your homework. Run the program using the same settings in an encoder. Create a table with the decoding results according to the sample Table 3.

**Table 3 –** Cyclic code (10, 5) for the codeword $N + 8 = 22$ (encoder input – 10110, encoder output - 1011001101) the studied results

| Error word $e(x)$ | Encoder input $\hat{b}(x)$ | Encoder output $\hat{a}(x)$ | Syndrome $s(x)$ | Error symbol number, which decoder defines |
|---|---|---|---|---|
| $x^9$ | $\underline{0}011001101$ | 10110 | $x^4 + x^3 + 1$ | $x^9$ |
| … | … | … | … | … |

Repeat the study of a decoding at double $e_2(x)$ and triple $e_3(x)$ errors from your homework. Compare the obtained results with the calculations in your homework.

Repeat the study of a decoding at the arbitrary fourfold error $e_4(x)$. For certain error words, for example, $e_4(x) = x^6 + x^5 + x^3 + x$ syndrome is zeroes, that confirms that a code (10, 5) has a code distance $d_{\min} = 4$.

5.10 **Research into another cyclic code (*n*, *k*).**

The teacher gives a cyclic code ($n$, $k$). Repeat the study described in 5.9.

## 6 Conclusions

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

# LW 4.2 RESEARCH INTO NOISE IMMUNITY OF BLOCK ERROR-CONTROL CODES

## 1 Objectives

1.1 The study of decoding algorithms of ($n$, $k$) block error-control codes with the errors correction.

1.2 The experimental study of ($n$, $k$) block error-control codes noise immunity with the errors correction.

1.3 The experimental study of a coding gain (CG).

## 2 Main postulates

See textbook [2, subsection 5.6].

**Reference data.** Table 1 shows the parameters of ECC: $n$ is a codeword length, $k$ is the number of information bits, $q_{\text{cor}}$ is the number of corrected errors, $C_n^q$ is the number of combinations of $n$ by $q$, $R_{\text{code}}$ is a code rate.

**Table 1** – Parameters of certain cyclic codes

| $n$ | $k$ | $q_{\text{cor}}$ | $C_n^{q_{\text{cor}}+1}$ | $R_{\text{code}}$ |
|---|---|---|---|---|
| 7 | 4 | 1 | 21 | 0,57 |
| 15 | 11 | 1 | 105 | 0,73 |
| | 7 | 2 | 455 | 0,47 |
| 31 | 26 | 1 | 465 | 0,84 |
| | 21 | 2 | 4495 | 0,68 |
| | 16 | 3 | 31465 | 0,52 |
| | 11 | 4 | 169911 | 0,35 |
| 63 | 57 | 1 | 1953 | 0,90 |
| | 51 | 2 | 39711 | 0,81 |
| | 45 | 3 | 595665 | 0,71 |
| | 39 | 4 | 7028847 | 0,62 |
| | 36 | 5 | 67945521 | 0,57 |
| 127 | 120 | 1 | 8001 | 0,95 |
| | 113 | 2 | 333375 | 0,89 |
| | 106 | 3 | 10334625 | 0,84 |
| | 99 | 4 | $2{,}54\cdot10^8$ | 0,78 |
| | 92 | 5 | $5{,}17\cdot10^9$ | 0,97 |
| 255 | 239 | 2 | 2731135 | 0,94 |
| | 231 | 3 | $1{,}72\cdot10^8$ | 0,91 |
| | 223 | 4 | $8{,}64\cdot10^9$ | 0,87 |
| | 215 | 5 | $3{,}6\cdot10^{11}$ | 0,84 |
| | 207 | 6 | $1{,}28\cdot10^{13}$ | 0,81 |
| 511 | 493 | 2 | $2{,}21\cdot10^7$ | 0,96 |
| | 484 | 3 | $2{,}81\cdot10^9$ | 0,95 |

## 3 Questions

3.1 What is a syndrome of a code combination and what is it used for?

3.2 What are the advantages of the error-control Reed-Solomon code compared to BCH code?

3.3 What parameters of error-control code determine the dependence of symbol error probability at the decoder output from the error probability at its input?

3.4 How is an error corrected in the code combinations of an error-control code?

3.5 What determines the CG of an error-control code?

3.6 What parameters of an error-control code does CG depend on?

## 4 Homework

4.1 Calculate the dependence of symbol error probability at the decoder output from the error probability at its input for values $p_{\text{in}}=10^{-1}\ldots10^{-6}$ with a code

– teams №№ 1 and 7 – (255, 239);
– teams №№ 2 and 8 – (127, 106);
– teams №№ 3 and 9 – (63, 51);
– teams №№ 4 and 10 – (31, 21);
– teams №№ 5 and 11 – (255, 231);
– teams №№ 6 and 12 – (127, 113).

The number of combinations $C_n^{q_{\text{cor}}+1}$ can be obtained from Table 1. Draw the corresponding graph dependence [2, figure 5.11].

4.2 Calculate the error probability at the decoder input for the allowable error probability at its output $p_{\text{all}}=2N\cdot10^{-4}$, where $N$ is your team number.

4.3 Be ready to answer the questions.

## 5 Laboratory work

**5.1 The study of a decoder computer model.** Run MatLab, clicking the icon on the desktop, and then write “block_encoding” in the command row and press “Enter”.

The model is created in the MatLab. The model allows you to change the settings of all blocks of a transmission system. The message length $L$ is unlimited. The codec cannot be used or chosen from three options: Hamming code, BCH code or Reed-Solomon code. The interleaver cannot be used either. The block matrix interleaver (X,Y) either cannot be chosen. In the model you can change the modulation method and levels of the modulated signal. The communication channel can be used with additive white Gaussian noise (AWGN) and constant parameters or fading. Signal/noise ratio varies from 2 dB to 9 dB.

The model allows you to build graphic of symbol error probability dependence at the decoder output $p_{\text{out}}$ from the error probability at its input $p_{\text{in}}$, i.e. $p_{\text{out}} = f(p_{\text{in}})$, and symbol error probability dependence at the decoder output from signal/noise ratio $p_{\text{out}} = f(E_b/N_0)$. Using markers one can calculate the coding gain at an allowable error probability at the decoder output $p_{\text{all}}$. $N$ is the number of error symbols at the decoder output.

**5.2 The charting functions $p_{\text{out}} = f(p_{\text{in}})$ and $p_{\text{out}} = f(E_b/N_0)$ without ECC.** To do this set the transmission system mode «without a code». Other settings should be the following: «Without interleaving», «BPSK» modulation, «AWGN» channel. Change the signal/noise ratio from 2 to 9 dB with step 1 dB, each time click «To transmit» and at the same time add the results of each measurement to «Archive of measurements». Dependencies $p_{\text{out}} = f(p_{\text{in}})$ and $p_{\text{out}} = f(E_b/N_0)$ will be automatically displayed on the scoreboard. The message length $L$ must be set within $10^5\ldots10^7$.

**5.3 The experimental research into the Hamming decoder correcting ability.** Choose the type of a «Hamming» code. Set the length of the codewords you got in your homework. Change option «Series of measurements» in «Simulation result» and «Archive of measurements» boxes. Then measure again as in 5.2.

Set $p_{all}$ according to the data obtained in your homework. Using markers measure signal/noise ratio at the transmission system «Without a code» and with a «Hamming» code. Write down the value of CG.

**5.4 The experimental research into BCH decoder correcting ability.** Choose the type of a «BCH» code and according to the data obtained in your homework set the parameters of a $(n,k)$ code. Change the option «Series of measurements» in «Simulation result» and «Archive of measurements» boxes. Then measure again as in 5.2.

Set $p_{all}$ according to the data obtained in your homework. Using markers measure signal/noise ratio at the transmission system «Without a code» and with a «BCH» code. Write the value of CG. Compare the dependence with that one calculated in your homework. Make conclusion about the correct results of your calculations at home. Draw the dependences of the symbol error probability at the decoder output $p_{out}$ from the error probability at its input $p_{in}$, i.e. $p_{out} = f(p_{in})$, and the dependence of the symbol error probability at the decoder output from signal/noise ratio $p_{out} = f(E_b/N_0)$ in the same graphs. Each graph should be placed in one page of your workbook.

**5.5 The research into correcting codes properties.** Compare the values of CG in 5.3 and 5.4, make the conclusions. Change the parameter $p_{all}$ and conclude on the value of coding gain changes. Change the code parameters without changing the code rate. Make the conclusions. Change the code rate by changing one of the code parameters. Measure CG for the four code rates. According to the results of measurements build a dependence of CG on the code rate for the error-control BCH code.

**6 Conclusions**

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, etc.).

## LW 4.3 THE STUDY OF CODING AND DECODING BY CONVOLUTIONAL ECC

**1 Objectives**

1.1 The study of the structure codec convolutional code (7, 5).
1.2 Learn the algorithms for decoding convolutional codes.

**2 Main postulates**

See textbook [2, subsections 6.1 – 6.4].

**3 Questions**

3.1 Define convolutional codes (CC).
3.2 What is the free distance convolutional code and what does it describe?
3.3 What is the length of the CC code restrictions?
3.4 Define metrics: branch, path and state.
3.5 How can I describe the convolutional encoder code work?
3.6 How do you build a CC trellis diagram encoder?

3.7 List decoding algorithms of convolutional codes.
3.8 Explain the principle of the Viterbi decoder to decode CC.
3.9 What are the advantages and disadvantages of the Viterbi algorithm.
3.10 What is a soft decoding?
3.11 What determines the survived path on the grid and how do you find it?
3.12 How are decoder convolutional code errors corrected?

**4 Homework**

4.1 CC with a generating polynomial ***g*(*i*)** = (7, 5) is set. Write down the number (12·$N$ + 900) in a binary notation, where $N$ is the number of your team. Encode the binary sequence CC (7, 5), construct the CC encoder trellis diagram and mark the path coding on it. For the trellis diagram [2, section 6.2] determine the free distance convolutional code (7, 5) and the multiplicity of errors that are corrected by this code.
4.2 Be ready to answer the questions.

**5 Laboratory work**

**5.1 Introduction to the virtual model.** Start the program **4.3**, use the icon **TT(English)** on a desktop. Study the laboratory model scheme on the computer display. Learn the input information symbols, find the encoder and decoder input errors. Later study the sequence of information bits (characters) in your homework.

A convolutional code is given by a generated polynomial ***g*(*i*)** = (7, 5).

Enter mistakes using the symbols in the “Approved”. To repeat the encoding process it is recommended to erase the memory encoder by pressing “Clear” on “CC decoder” panel. Repeat the same for the decoder clicking “Clear” on “CC decoder” panel. Click “Clear All” on the code parameters panel before a new study.

**5.2 The research into coding.** You must enter information bits **a**, obtained in your homework in the encoder panel. Then click “Step by Step” encoding process (until the button will no longer be active). The report should contain the sequence of information bits, the contents of the register and the encoder output sequence of code symbols at each step. Make sure data obtained in your homework is correct.

**5.3 The research into decoding process with no errors.** This is the testing of a decoder feeding “Decoder” sequence of code symbols received during encoding (see 5.2). To do this click “Step by Step” on “CC decoder” panel until it is active. Then click “Decision”, which may lead to the survived path. Compare it with the means of coding and make conclusion on the disability decoder.

**5.4 The research into the presence of decoding errors.** First we need to clear the memory register by clicking “Clear” on “CC decoder” panel. Then enter a single error in one of the first six adopted code symbols by pressing the left mouse button on the code symbols in the order you want to enter an error. Repeat the procedure described in 5.3. Redraw a fragment of the obtained trellis diagram for the first four steps of a decoding ($t_0$ - $t_4$) in your report. Redraw all possible paths, the survived path and recovered sequence of information bits. Make conclusions about correcting errors.

**5.5 The research into correcting ability of the decoder.** First enter two consecutive errors scattered in any adopted code symbols and repeat the instructions in

5.4, i.e. decode. Then enter three consecutive errors at decoding. Make conclusion what errors of multiplicity the decoder corrects or doesn't correct.

The results of the study can be fixed as shown below. Error code symbols and source information bits are underlined.

| | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Accepted code symbols | 11 | 01 | 0<u>0</u> | 01 | 01 | 0<u>0</u> | <u>1</u>0 | 00 | 10 | 11 | 00 | 11 | 00 |
| Decoded bits | 1 | 0 | 1 | 0 | 0 | <u>1</u> | <u>1</u> | 1 | 0 | 0 | 1 | 1 | 0 |

**6 Conclusions**

Make a conclusion about each item of the laboratory work, analyse the obtained results (coincidence of theoretical and experimental data, CC (7, 5) correcting ability, etc.).

## LW 4.4 THE STUDY OF NOISE IMMUNITY OF CONVOLUTIONAL ERROR-CONTROL CODES

**1 Objectives**

1.1 The experimental research into noise immunity of convolutional error-control codes.

1.2 The research into coding gain.

1.3 The research into decoding complexity.

**2 Main postulates**

See textbook [2, subsections 6.4 – 6.6].

**3 Questions**

3.1 What error-control code parameters are defined by the dependence of the error symbols probability at the decoder output and the error probability at its input?

3.2 What determines the ACG of an error-control code?

3.3 What is the difference between CG and ACG of a convolutional code?

3.4 What gain does a soft decoding provide compared with the hard decoding?

3.5 What is the complexity of a decoding? How do you define it?

3.6 What is the spectrum of convolutional code weights?

**4 Homework**

4.1 Calculate the dependence of the error symbols probability at the output of the decoder signal / noise ratio for the convolutional code

- teams №№ 1 and 7 – (15, 17);
- teams №№ 2 and 8 – (13, 15);
- teams №№ 3 and 9 – (13, 7);
- teams №№ 4 and 10 – (17, 13);
- teams №№ 5 and 11 – (15, 13);
- teams №№ 6 and 12 – (7, 13).

We can use formulas [2, subsection 6.5] to calculate.

Draw the corresponding graph dependence.

4.2 Be ready to answer the questions.

## 5 Laboratory work

5.1 **The study of the computer model of a decoder.** Run the program "CC codec". Study methods how to correct errors by convolutional code, i.e. the introduction of input data, running the program, reading the results.

The model is in the programming language C++. The model allows you to change the signal / noise channel, the message length and parameters of an error-control convolutional code. The message length is unlimited. You can choose the parameters of a code from four options: (17,15), (15,13), (17,13), (13,7). The model uses a communication channel with a white additive Gaussian noise (AWGN) with constant parameters. SNR varies from 2 dB to 6 dB.

The model allows us to construct graphs of symbol error probability at the output of the decoder signal / noise ratio $p_d = f(E_b/N_0)$.

5.2 The plotting function $p_d = f(E_b/N_0)$. Set the operating mode "Codes researching". Select "To plot noise immunity" and "Value of SNR". Enter the value 6 dB. Run button «Generation». Dependence $p_d = f(E_b/N_0)$ will be displayed automatically. Redraw the resulting graph. Compare the simulation results with the results of your homework. Make conclusions.

5.3 CG calculation. Calculate CG for each code. Compare them with each other and with the corresponding ACG values. Make conclusions about the impact of allowable bit error probability at the output of the decoder with the value of CG.

## 6 Conclusions

Make conclusion about each item of the work, analyse the obtained results (coincidence of theoretical and experimental data, CG calculation, etc.).

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Education publication

*Ivaschenko Petro Vasilyovych*
*Rozenvasser Denis Mikhailovych*

# EDUCATION MANUAL ON TELECOMMUNICATION THEORY AND THEORY OF INFORMATION AND CODING

**Laboratory Works**

Signed for printing 17.02.17
Format 60/88/16 Order № 5982
Edition of 100 copies. Amount of quires 7,75.
Printed by the publishing RISO company equipment
Printed at the Editorial and Publishing Center of O. S. Popov ONAT