

UDC 53:004

RESEARCH OF INFLUENCE OF SPEECH CODECS ON TRANSMISSION QUALITY OF THE DIGITAL SIGNAL

Zh.K. Ishchanova, G.P. Amochaeva, A.K. Tussupbekova

Karaganda state University named after E.A. Buketov, Universitetskaia St. 28, Karaganda, 100026, Kazakhstan,
Karaganda_07@mail.ru

The Kazakhstan telephone system of the general using contains both analog and digital segments. For a long time analog segments served for an information exchange between subscribers, however introduction of digital segments has led qualitative information transfer for long distances because digital technologies possess a number of advantages.

Keywords: analog segments, telephone system, signal constellations, characteristics of errors, telephone ports.

Digital segments are less subjected to distortion and interference when comparing with analog ones. As binary digital ports give a significant signal only at work in one of two states that is 'switched on' and 'switched off', disturbance should be great enough to transfer a working point of the port from one state into another one. Presence of only two states facilitates restoration of a signal and, hence, prevents accumulation in the course of noise transmission or other disturbances. While using digital technologies very low frequency of occurrence of errors plus application the procedures of revealing and correction of errors make possible high frequent accuracy of a signal. There are also other important advantages of digital communication. Digital links are more reliable and can be produced at lower prices, than analog ones. Besides, the digital software admits more flexible realization, than analog ones [1].

Despite achievements in the field of high-speed digital transmission systems, modems for commutated telephone ports keep their appeal. Because for introduction of high-speed technologies, such as xDSL, Ethernet, data transmission in networks of mobile connection of the 2nd and 3rd generations, modernization of a communication network and installation of the expensive equipment are required. Therefore transmission of digital data by analog ports by means of the modem that is technology dial-up is used in telephony. It is explained by a wide circulation and availability of such ports where the primary appointment was speech transmission. If a subscriber can use only analog port then only the modem can solve its problems on data transmission. Modem access remains especially claimed at departmental communication networks (for example, at power engineering networks, oil industry workers), and also in the remote and sparsely populated areas of Kazakhstan. [2]

The dial-up technology is less modernized and noise immuned in comparison with xDSL, Ethernet, an IP-telephony, data transmission in networks of mobile connection of the 2nd and 3rd generations, however it is simple in the realization and doesn't demand the great expenses. Therefore this technology is the most comprehensible in telephone systems as at present the cardinal telephony change is impossible. The dial-up technology provides gradual transition to use of digital segments, joining them to analog segments.

Data transmission by analog ports can be carried out using both one-dimensional symbols of pulse-amplitude modulation (PAM-technology), and two-dimensional symbols quarter amplitude modulation (KAM- technology). For data transmission by telephone ports it is recommended to use only two-dimensional symbols of KAM- technology because while coding these signals are set by two parameters that is accepted to consider more exact reproduction of the information. Therefore in given article the method of quarter amplitude modulation is considered as this mode of modulation is recommended by ISES for use in telephone ports. [3]

On joints of digital and analog segments of a network standard codecs ICM are used. Standard codecs possess a number of advantages: they are less sensitive to digital errors, bypassing and bring an insignificant delay of a signal. Use of these codecs allows to have a considerable quantity of consecutive transitions on a network without essential drop of quality [2].

Analog data is represented by digital signals and transferred by telephone ports by means of codecs (coder/decoder) which approximate signals by means of bit streams. While analog-digital transformation of a linear signal of the codec the former is exposed to specific distortions that lead to increase of error probability. Digital signals are subjected to influence of noise of coders which are impossible to simulate and count manually as noise is a random variable.

A lot of works are devoted to problems of digital signals transmission by analog ports. However a number of problems of speech codecs concerning influence, namely logarithmic and nonlinear ones, on quality of QAM-signals transmission in the scientific literature are regarded insufficiently. Therefore the theme of the research of speech codecs influence on digital signals is actual.

In the given article influence of speech codecs on quality of digital information transmission by telephone ports will be investigated. Namely influence of the analog-digital converter on QAM-signals transmission as these signals is exposed to influence of specific distortions which can't be simulated by additive Gaussian noise that is connected with nonlinearity of the used quantizator. In this connection points of constellations of QAM-signals distant from the center are subjected to stronger influence of distortions, than near ones. Accordingly the object of the article is speech codecs.

The research objective is investigation of speech codecs influence on qualities of digital signals transmission of QAM-technology, the estimation of influence degree is made by computer modeling.

For achievement of the specified purpose following problems are solved:

1. The comparative analysis of existing methods of analog-digital transformation of speech signals;
2. The analysis of properties of signals of quantizator amplitude modulation;
3. The theoretical analysis of influence of codecs ICM on quality of QAM – signals transmission;
4. Creation of imitating model of influence of analog-digital transformation on the QAM-signal;
5. Investigation the received model.

While carrying out the investigation methods of the theory of digital processing of signals, theories of chains and signals, theories of electric connection, the theory of the information and casual processes, methods of mathematical statistics and engine modeling have been used.

Scientific novelty of the given work issues that by means of computer modulation the imitating model of quantizator ICM-codec influence on KAM-signals by means of which it is possible to estimate influence as own noise of system – additive noise, and other noises. Moreover by means of the created program it is possible to state an estimation of the minimum achievable probability of an error by drive of digital KAM-signals by the telephone port equipped with codecs ICM.

There is a set of methods of coding for transformation of an analog signal into the digital form - time, frequency, parametrical. Methods of frequency coding of the information issue in modulation by information signal on so-called 'carrying frequencies'. Methods of time coding issue in the exact description and reproduction of the form of fluctuations in time, i.e. we code directly the signal form. They include: ICM with a uniform and non-uniform scale of quantization, DICM, delta-modulation, ADKM, coding with splitting on subgamuts. Parametrical coding is based on use of voice coders which use the analysis for transformation and representation in a short form and synthesis for restoration of a source speech signal on the basis of its short representation.

Referring to stated above it is possible to make a conclusion that it is considered to be the best methods of time coding because they are noise defended most of all. From all methods of coding in time area the greatest distribution in digital telephony methods ICM and ADICM were accepted since their realization is a simple and at the same time useful signal that is easier to identify while hindrances, distortions and noise of converters.

ICM issues in representation of a continuous analog signal in the form of sequence equidistant from each other impulses where their amplitude is presented by a binary code (quantization on level), and coding as well. Similar transformation allows raising essentially reliability of signal transmission and storage. ADICM is a modernized version of DICM which issues in the following: prediction the current value of reference point on the basis of previous M reference point. Its modernization issues that here quantizator and the predictor adapt to the changing environment of an entrance signal.

Data transmission by analog ports can be carried out using both one-dimensional symbols of pulse-amplitude modulation and two-dimensional symbols quarter amplitude modulation. ISES recommends using only two-dimensional symbols of QAM-technology for data transmission by TF ports. Two-dimensional signals as information transfer sources by the telephone port are more reliable, economic and profitable, because two information parameters are already a pledge of noise defended and better transmission. Therefore in the thesis two-dimensional signals, namely QAM-signals are considered. QAM-signals turn out by simultaneous change of inphase (I) and quarter (Q) amplitude a component of the carrying harmonious fluctuation, shifted on a phase from each other on $\pi/2$. The resultant signal Z is formed as a result of summation of these fluctuations.

QAM-modulation is formed by simultaneous coding of amplitude and a phase of carrying fluctuation. For the given modulation it is characteristic that while modulation of inphase and quarter components of carrying fluctuation the same value of a change pace of amplitude is used. Therefore the terminations of vectors of the modulated fluctuation form a rectangular grid on a phase plane valid - $\text{Re}\{Z\}$ and imaginary - $\text{Im}\{Z\}$ components of a vector of the modulated signal. The number of knots of this grid is defined by type of used algorithm QAM. The scheme of knots arrangement on a phase plane modulated QAM fluctuations is accepted to name 'constellation'.

In work the analysis of recommendations ISES for the modems working on telephone ports, using the given kind of modulation is carried out. [4]

Since the introduction of digital segments, equipment ICM (reliable for transformation of the analog information into digital information) has great influences on quality of QAM-signals transmission. ICM the codec in a telephone system possesses the certain nonlinearity providing increase of quantizator efficiency for speech signals. This nonlinearity reduces productivity of modems, such decline of productivity in directly proportional to speed increase of data transmission. In the second chapter of the thesis the attention to influence of logarithmic and nonlinear quantizator on quality of QAM-signals transmission is focused

While carrying out the analysis of influence of the speech codec on the QAM-signal the following conclusions have been made:

1. The codec includes a logarithmic or nonlinear compander; hence, an error of each referring point is regular distributed interval random variable that is proportional to the value of the referring point.
2. The transmitted signal is generated by two kinds of noise that are multiplicative and additive ones where first makes more impact on value of error probability.
3. With growth of amplitude of referring point the noise value grows as well.
4. Noise has the ellipse form; however an ellipse axis doesn't coincide with a radial axis of a point of constellation.
5. The size of constellation makes less impact on quantization noise than its form.

To estimate influence of additive and multiplicative noise on points of constellation of QAM-signals the imitating model of influence of the ICM-codec on the digital signal transferred by

telephone ports has been created. Because the theoretical estimation of influence of noise takes a lot of time and doesn't show an evident picture of signal constellations of QAM-signals. The program consists of two modules. The first module is used for construction of points of constellation of the QAM-signal and composing schedules of noise distribution around the constellation points. This module creates the evident picture of signal constellation subjected to influence of multiplicative and additive noise. The second module is used for calculation of noise immunity of a signal and construction of the schedule of dependence of error factor on noise immunities by additive and multiplicative noises.

The program is realized in a programming language C++. Information input and output in the program is realized through the operating system console.

To write the program of imitating model algorithms of their work have been initially created. By the block diagram the code of work of the module which is programmed in the program 'far' in language C++ has been created.

In Figure 1 the window of already created program of the first module is resulted where the initial data for calculation of error quantity, error factor and value AQD of noise are requested. A number of iterations, the maximum pressure of a constellation point on an one-dimensional axis and noise immunity on additive and multiplicative noises are accordingly related.

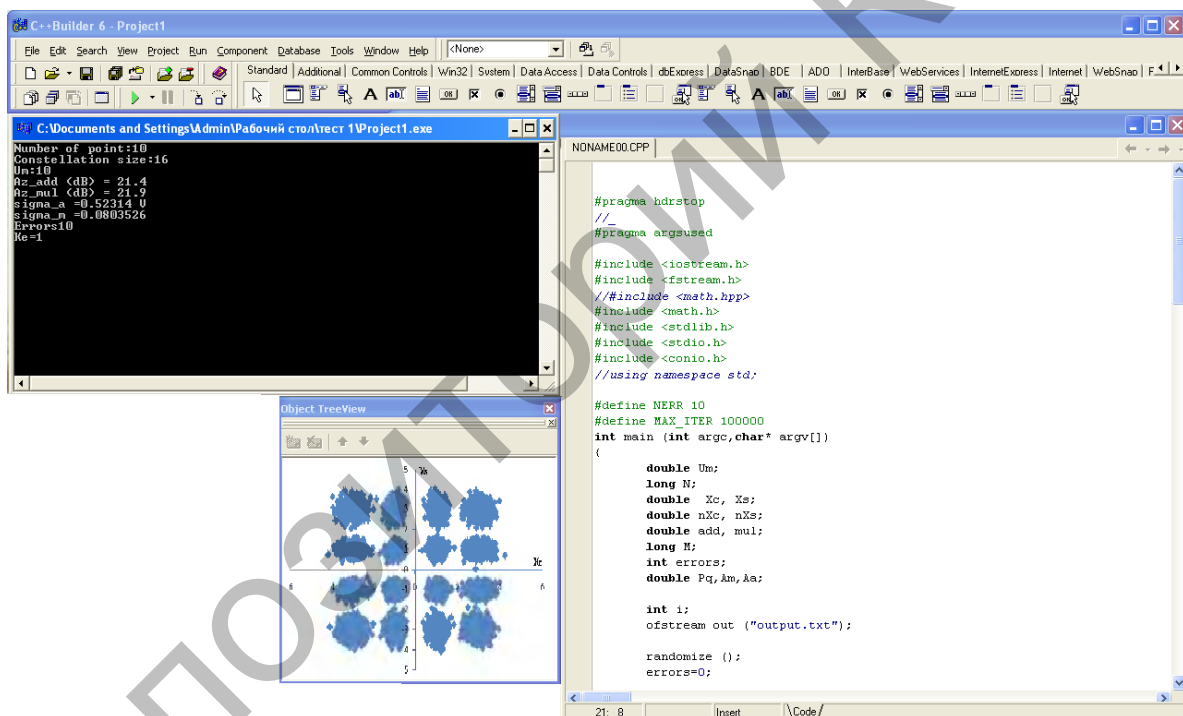


Fig. 1. Window of input-output calculations of the first module

This model has a number of advantages because it gives out results for all types of QAM-signals. It is enough to enter the variables of investigated type and depending on them corresponding results will be resulted.

The first module deduces the quantity of errors, error factor and value of noise AQD. This module proves that the transmitted signal is generated by two kinds of noise - multiply and additive and with growth of amplitude of readout the noise size grows also.

Further the signal constellation depending on number QAM is constructed. The program shows an evident picture of distribution of noise by vectorgram where ratios

between multiplicative and additive noise and error probability are considered. By means of the received data it is evident that noise has the ellipse form and the ellipse axis doesn't coincide with a radial axis of constellation point.

For an example in article results of calculations of the program for three constellations are resulted: QAM-16, QAM-64, QAM-268. The kind of signal constellation depends on a ratio between the signal and the noise.

For QAM -16 signal constellations on an input of the solving device with a ratio between a signal and noise equal 17.5 dB and 20 dB for error probability 10^{-3} look as follows (fig. 2):

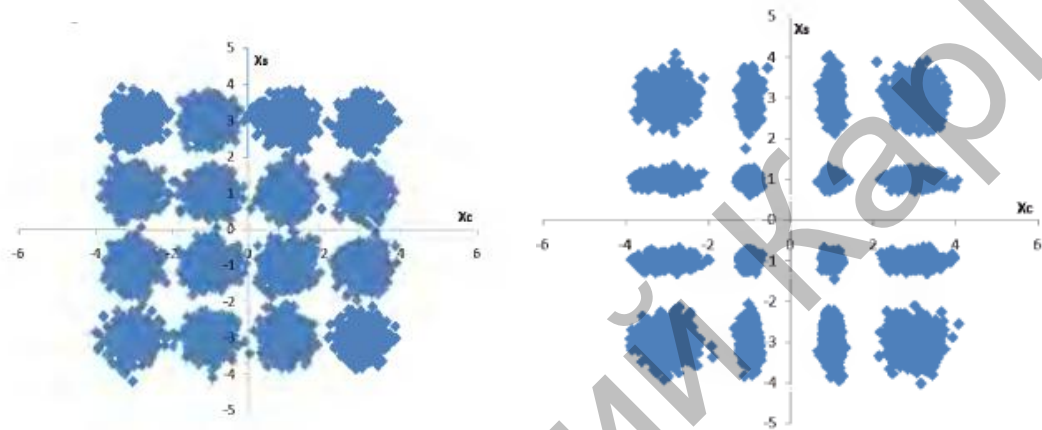


Fig. 2. Signal constellations QAM-16 having $A_{3u} = 17.5$ dB, $A_{3u} = 20$ dB

Having analysed the received images it is possible to draw a conclusion that with ratio growth between a signal and noise of constellation points lose its ellipse form.

By results for QAM-64 and QAM-268 signal constellations on an input of the solving device at ratios between the signal and the noise equal 23 dB and 30 dB accordingly for error probability 10^{-3} look like (fig. 3):

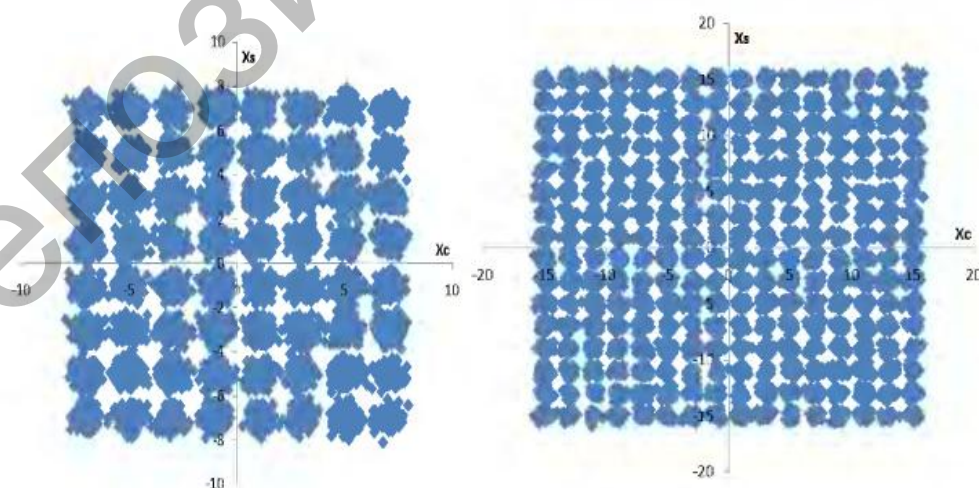


Fig. 3. Signal constellations QAM-64 at $A_{3u} = 23$ dB and QAM-268 at $A_{3u} = 30$ dB

Their difference issues in a number of constellation points.

The second module of the program is intended for calculation of noise immunity of a signal and construction of the schedule of error factor dependence on noise immunities on additive and multiplicative noise.

In Figure 4 calculating windows of the second module is presented. Such initial data of a signal are requested as QAM type, the maximum pressure of a point of constellation on an one-dimensional axis and area on which it is necessary to investigate noise immunity, that is the maximum and minimum values and it is necessary to choose a type of investigation concerning additive or multiplicative noise.

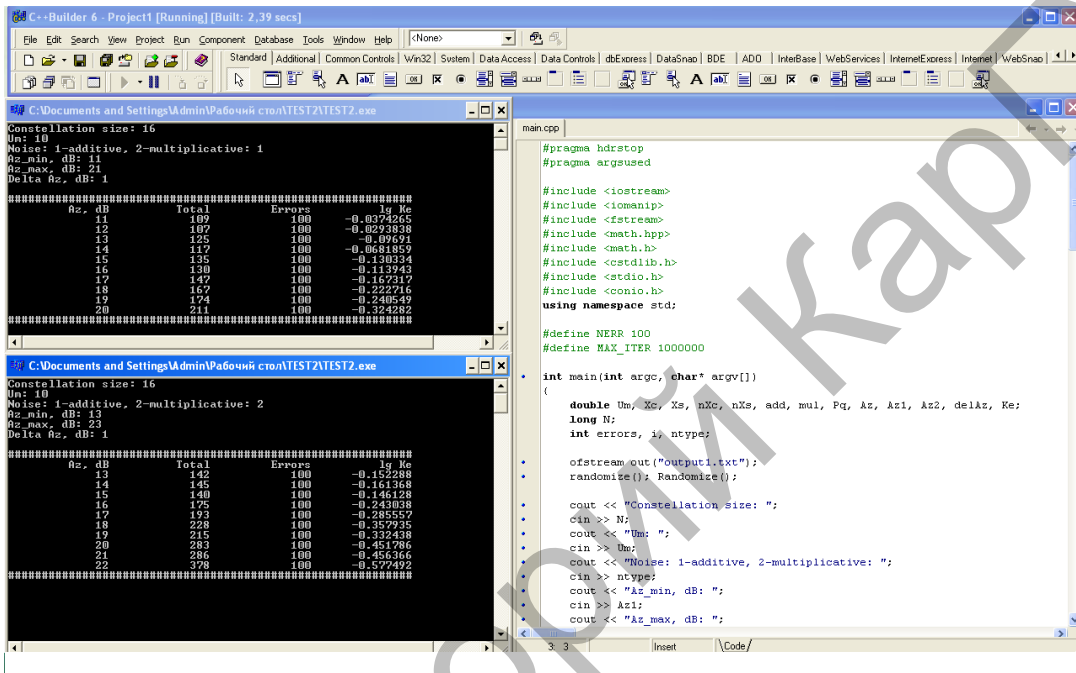


Fig. 4. The window of input-output of the second module for QAM-16.

Characteristics of errors (fig. 5) are received by means of the program 'curves'.

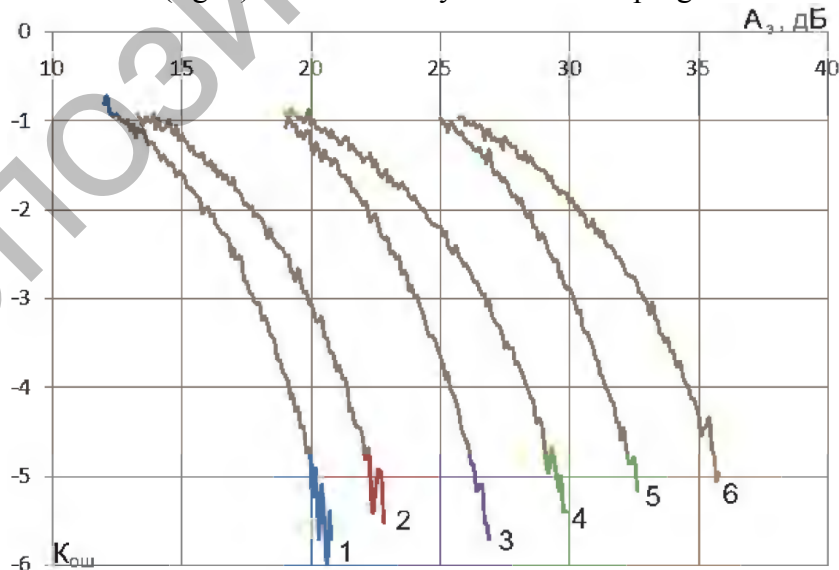


Fig. 5. Characteristics of errors

Curves 1 and 2 correspond to constellation QAM-16, the curve 1 describes dependence of error factor on noise immunity on additive noise, a curve 2 - on noise immunity on multiplicative noise.

Curves 3 and 4 correspond to constellation QAM-64 and also describe dependence of error factor on noise immunity on additive and multiplicative noise.

Curves 5 and 6 correspond to constellation QAM-256.

As while transmission QAM signal by the telephone port to system there are both additive, and multiplicative noise components, the real curve will lie between the curves constructed for cases of pure additive and pure multiplicative noise.

While carrying out the investigation of speech codecs influence on quality of digital signals transmission by means of computer modeling, the following conclusions have been formulated:

1. The program which states the estimation of influence degree of speech codecs on quality of QAM-signals transmission by computer modulation is developed.

2. From the received results of imitating model of the first module it is evident that with ratio increase between the signal and noise, constellation points lose their elliptic shape that is confirmed by theoretical calculations.

3. The second module of the program shows that with increase of the number of the QAM-signal its noise immunity increases as well, multiplicative noise makes more essential impact on value of error probability and, accordingly, demands more noise immunity.

4. The regarded program is possible to recommend for experts in the field of working out of telecommunication equipment and in educational process.

Apparently from the resulted schedules, the multiplicative noise makes more essential impact on value of error probability and, accordingly, it demands more noise immunity (2-3 dB more) for maintenance the demanded error probability.

The results received in article can be used in the further theoretical researches when designing the transmission systems using telephone ports, and in the course of training while consideration the modem reports and processes of digital information transmission by telephone ports.

References:

1. Bellami Dj. Digital telephony. – M: Eko-Trends, 2004.- 640 p.
2. Kuricyn S.A. Base of construction telecommunication system of transmission. – SPb: Information centre of voice, 2004.
3. Rabiner L.R.,Shafer R.V. Digital processing of speech signal. – M: Radio and connection, 1981. – 463p.
4. Kruhmalev V.V., Gordienko V.N., Mochenov A.D. Digital system of transmission // Hot line. – Telecom,2007. – 350p.