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STUDY OF SOFTWARE FILTERING OF A DIGITAL SIGNAL BASED ON THE STANDART PSK-31

ДОСЛІДЖЕННЯ ПРОГРАМНОЇ ФІЛЬТРАЦІЇ ЦИФРОВОГО СИГНАЛУ НА БАЗІ СТАНДАРТУ PSK-31

The basis of modern systems for transmitting digital information through radio channels is the coding of each symbol using a reference oscillation segment with certain values of the basic parameters — amplitude, frequency, initial phase. According to the changed parameter, the main families are distinguished-amplitude manipulation (AM, ASK — Amplitude Shift Keying), frequency manipulation (FM, FSK — Frequency Shift Keying) and phase manipulation (FM, PSK — Phase Shift Keying). In addition to the basic types of manipulation, combinational ones are also widely used — for example, quadrature amplitude shift keying (KAM, QASK — Quadrature Amplitude Shift Keying).

With the development of computing technology, software and integrated microcircuits, new types of communication appeared. One of them is PSK-31. Initially, operation in the PSK-31 mode was based on the use of expensive digital signal processing (DSP) integrated circuits. The rapid spread of the PSK-31 mode began in 1998, when the English radio amateur Peter Martinez created the PSK31 SBW computer program, which implements digital signal processing.

A distinctive feature of digital communication systems (DCS) is that for a finite period of time they send a signal consisting of a finite set of elementary signals (in contrast to analog communication systems, where the signal consists of an infinite set of elementary signals). In DCS systems, the task of the receiver is not to accurately reproduce the transmitted signal, but to determine, based on the noise-distorted signal, which signal from the final set was sent by the transmitter. An important criterion of the performance of the DCS system is the probability of error (PE).

***Keywords**: amplitude manipulation, PSK-31, DCS, pulse width modulation.*

Системи цифрового зв'язку стають усе більше привабливими внаслідок постійно зростаючого попиту і через те, що цифрова передача пропонує можливості обробки інформації, не доступні при використанні аналогової передачі.

В основі сучасних систем передачі цифрової інформації через радіоканали лежить кодування кожного символу за допомогою сегмента опорного коливання з певними значеннями базових параметрів — амплітуди, частоти, початкової фази. Відповідно до змінюваного параметра розрізняються основні сімейства — амплітудна маніпуляція (АМ, ASK — Amplitude Shift Keying), частотна маніпуляція (ЧМ, FSK — Frequency Shift Keying) і фазова маніпуляція (ФМ, PSK — Phase Shift Keying). Крім базових видів маніпуляції також широко застосовуються комбінаційні — наприклад, квадратурно амплітудна (КАМ, QASK — Quadrature Amplitude Shift Keying).

З розвитком обчислювальної техніки, програмного забезпечення і інтегральних мікросхем, з'явилися нові види зв'язку. Одним з них є PSK-31. Спочатку робота в режимі PSK-31 були засновані на використанні дорогих інтегральних мікросхем цифрової обробки сигналу (DSP). Початок стрімкого поширення режиму PSK-31 почалося в 1998 р., коли англійський радіоаматор Пітер Мартінез створив комп'ютерну програму PSK31 SBW, що реалізує обробку цифрових сигналів.

Відмінною рисою систем цифрового зв'язку (digital communication system — DCS) є те, що за кінцевий проміжок часу вони посилають сигнал, що складається з кінцевого набору елементарних сигналів (на відміну від систем аналогового зв'язку, де сигнал складається з нескінченної безлічі елементарних сигналів). У системах DCS завданням приймача є не точне відтворення переданого сигналу, а визначення на основі перекрученого шумами сигналу, який саме сигнал з кінцевого набору був посланий передавачем. Важливим критерієм продуктивності системи DCS є ймовірність помилки (PE).

Розроблена та реалізована у вигляді лабораторного макету практична схема приймача PSK-31. Проведені дослідження основних параметрів і характеристик розробленого зразка. Додатково проведено комп'ютерне моделювання стандарту цифрового зв'язку PSK-31.

***Ключові слова**: амплітудна маніпуляція, PSK-31, DCS, широтна-імпульсна модуляція.*

Problem's formulation

Currently, among the possible options for communication, digital ones are the most common, however, due to the fact that most equipment is also digital and the presence of pulsed power sources actively affect the immunity of the receiver. Therefore, an urgent technical task is the development of a digital filter that will improve the quality of the received signal and reduce the number of errors.

Analysis of recent research and publications

Digital software allows for more flexible implementation than analog (for example, microprocessors, digital switches, and large-scale integrated circuits (LSI). The use of digital signals and time-division multiplexing (TDM) makes it easier to use analog signals and frequency-division multiplexing (FDM). When transmitting and switching, different types of digital signals (data, telegraph, telephone, television) can be considered identical: after all, a bit is a bit. In addition, for ease of switching and processing, digital messages can be grouped into autonomous units called packets. Digital technologies naturally incorporate functions that protect against interference and signal suppression and provide encryption or secrecy.

Data exchange is mainly between two computers, or between a computer and digital devices, or a terminal. Such digital end devices are better (and of course) served by digital communication channels. Digital systems require more intensive processing than analog systems. For digital systems, it is necessary to allocate a significant part of resources for synchronization at various levels.

Another disadvantage of digital communication systems is that quality degradation is marginal. If the signal-to-noise ratio drops below some threshold, the quality of service can jump from very good to very bad. In analog systems, quality deterioration occurs more smoothly.

DSP allows you to create a filter with almost any frequency response, quickly change its type and parameters, ensure the stability of characteristics, correlation processing, which are unavailable to analog filters. The type of PSK31 communication began with the use of DSP microcircuits, but the "heyday" of PSK31 occurred only after the use of computer "sound cards" in combination with software, which significantly reduced the cost of hardware. At this time, DSP processors are used quite widely and every year their "field of activity" increases, penetrating all areas of technology. Along with DSP microcircuits, there are programs in which the role of the DSP processor is performed by a program that simulates the algorithm of the DSP processor, together with the "sound card" of the computer.

There are several programs that implement the DSP algorithm programmatically, for example: DSPhil, it should also be noted that the program is free. A good impression is made by the SR5 Spectrum Analyzer program, which contains a rich toolkit and makes it possible to make any filters "on the fly" with the help of a computer "mouse". Decent results can be obtained when using the ChromaSaund DSP program — which has a beautiful interface and allows you to make "preparations" of user filters. Another DSP filter program VE3AGM DSP filter. All the listed programs work under Windows® management, there are other programs as well. In addition, there are programs for the MSDOS® environment. The most popular of them is DSP Blaster (DB). The program is designed for the DOS environment (it is written in Assembler), and it turns out that it works quickly, which is a key point for programs that process signals in real time. There is, however, one limitation — the program works only with Creative Labs® "sound cards"[1—3].

Band-pass and rejection filters with a variable passband will "cut out" interfering signals, or on the contrary, will leave the area of the band in which the communication is carried out. Reducing the noise level — will allow communication with stations that have a weak signal level. Automatic gain control(AGC) — will maintain the signal at the given level. Adaptive filters will adjust themselves to the necessary parameters (hence the name — adaptive), depending on the nature of the signal. All this is in software DSPs[4].

Formulation of the study purpose

Since the accumulation of noise is inextricably linked to analog signals, as a result, they cannot be reproduced perfectly. When using digital technologies, a very low error rate plus the use of error detection and correction procedures enable high signal accuracy. It remains to be noted that similar procedures are not available with analog technologies.

PSK (Phase Shift Keying) translates as phase "keying" (manipulation). A more narrow-band and interference-resistant phase manipulation is used here. Together with the application of special measures (smoothed cosine signal) it was possible to obtain a radiation band of 31 Hz at a transmission speed of 31,25 baud. All transmitted characters are encoded with a certain code. Character encoding is similar to RTTY(Radio Teletype) with logical 0s and 1s, but unlike RTTY, the codes are of variable length. This means that symbols that occur frequently are assigned shorter code combinations, which is why a higher transmission speed is achieved in RZK-31 mode compared to RTTY.

Presenting main material

As you know, the wider the bandwidth of the receiver, the worse the signal-to-noise ratio. In this case, we can use the minimum possible band (for comparison, remember that in RTTY mode it should be 300...500 Hz, and in SSB mode it should be about 3000 Hz), and it turns out to improve the signal-to-noise ratio many times. This, in turn, will allow to reduce the power of the transmitter with unchanged probability of transmitted information.

Working with such narrow strips has its own peculiarities. On long routes, especially polar ones, on days of solar disturbance, deterioration of communication on PSK-31 is possible. This is due to the fact that the disturbed ionosphere creates the so-called "shaking fading". As a result, the spectrum of signals expands, and the advantages of PSK-31 compared to other broadband types of communication are lost in this case.

Since computer signal processing takes place in PSK-31 mode, it became possible to install a second, auxiliary (software) filter in the processing program in addition to the narrow-band filter of the receiver, which further improves the quality of reception. Because a sound signal is fed to the computer from the output of the LF receiver, when using a wide-band filter (3000 Hz), the program allows you to observe a kind of panorama of received signals (as in panoramic set-top boxes) and, moreover, to receive dozens of stations at once.

In the transceivers of the past, there were parametric reference generators, and not synthesizers, as now, and the frequency stability left much to be desired. Technically, this type of communication, with a band of 31 Hz, could not be implemented in wide practice without the use of special measures. At this time, transceivers are different, so most programs for PSK-31 mode provide additional opportunities for automatic frequency adjustment by software. Software noise suppressors are also used.

If the frequency of the transceiver is unstable, work on PSK-31 (and especially in QPSK mode) will be difficult, and if the transmission path is not linear, the signal bandwidth will be much wider. Tuning to a station operating in PSK-31 mode has its own peculiarities, and various aids are provided to facilitate it software options and indicators.

The disadvantage common to all DSP programs is the delay of the signal during processing for about 0,5 s (depends on the performance of the computer, the bandwidth and the Q-factor of the filter). Therefore, when tuning, especially with a narrow-band filter, delay effects are possible, which you need to get used to. Or tune in with the DSP turned off, and then, having decided on the received station, turn it on.

A modern amateur radio "receiver" is a digital signal processing program with digital modulation from the input of a computer's sound card.

The amateur radio signal on the air is transmitted in the range of 3—30 MHz. The received signal from the antenna in the frequency converter of the single-band transceiver is transferred to the area of audio frequencies. The output of the converter is an analog band signal. This signal is fed to the input of the sound card of the computer. In the analog-to-digital converter (ADC) of the sound card, the signal is sampled and quantized. The processing of this digital signal is done by the program. To date, the signal transmission method named PSK-31 (author Martynets P.) has become the de facto "standard". It uses digital relative binary phase modulation (PSK) or quadrature phase modulation (QPSK) in combination with convolutional interference-resistant coding with a code rate of 1/2. The frequency band of the signal occupies 31Hz. Signal formation in this band is performed by an interpolating digital filter, with root (sqrt) cosine smoothing amplitude-frequency response (frequency response) and linear phase response. A digital Nyquist filter matched to the transmitted signal is used in the signal processing program. If the communication channel does not introduce frequency distortions, the cascade connection of the receiver and transmitter filters will form a Nyquist filter with cosine frequency response smoothing.

In the path of the first IF, to reduce the passage of the signal of the first local oscillator to the amplifier, the second IF frequency bandwidth at frequencies above 10 MHz, therefore, the first IF should not be higher than the standard value of 10,7 MHz. The transmission coefficient of the first mixer is 18dB, the second mixer is 21 dB. When using a chip with an external smooth-range generator (for example, with a frequency synthesizer), the signal from it is fed to pins 21 and/or 22. The PSK-31 station signal is received by the assembled receiver operating in one sideband mode. The low-frequency part of the signal is fed to the line input of the computer's sound card. Further processing of the signal is performed by software.

The sound frequency amplifier is assembled on the LM386 microcircuit, in standard operation it has an amplification (40 dB). DSP program SR5 (upper part of the figure, above the horizontal yellow line) with several PSK31 signals. In the SR5 program, two band filters of 150—200 Hz are formed at the frequencies of two working PSK31 stations.

The psk31ab model on Simulink of the PSK-31 "standard" digital transmitting and receiving part with an acoustic channel is shown in Fig. 1.

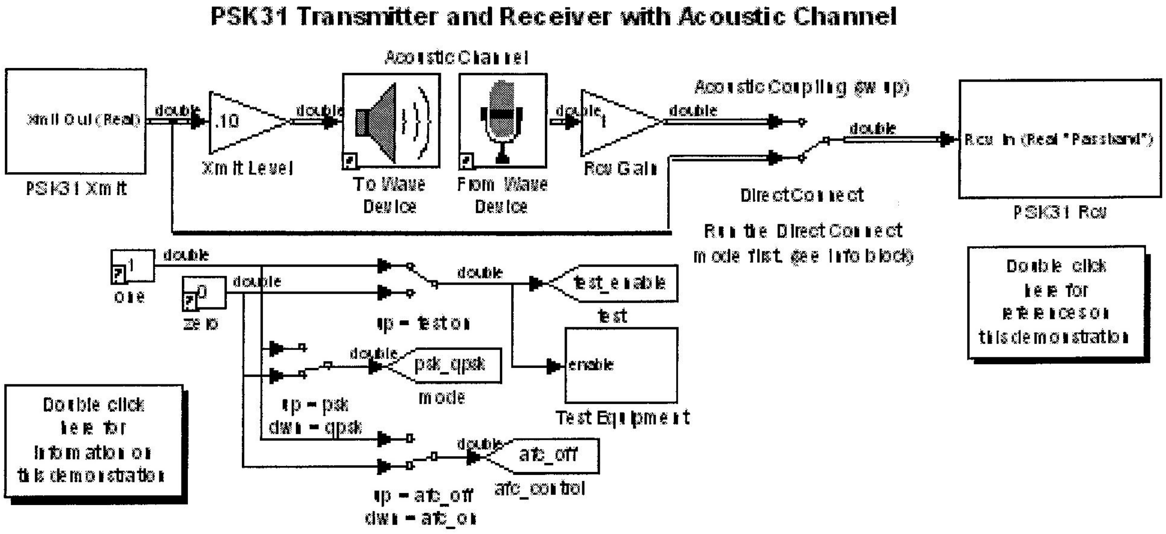

Fig. 1. A real-time model with an acoustic communication channel

The message symbols are converted into a phase-shifted signal at a frequency of 810 Hz on the line output of the sound board. The acoustic signal from the sound reproduction equipment of the computer is received by the microphone. The acoustic channel introduces both noise and phase distortions into the transmitted signal (Fig. 2). The signal from the microphone input is sampled at a frequency of 8 kHz and processed by a signal processing program. The processing program includes frequency tuning based on the FFT algorithm, phase synchronization based on the Costas algorithm.

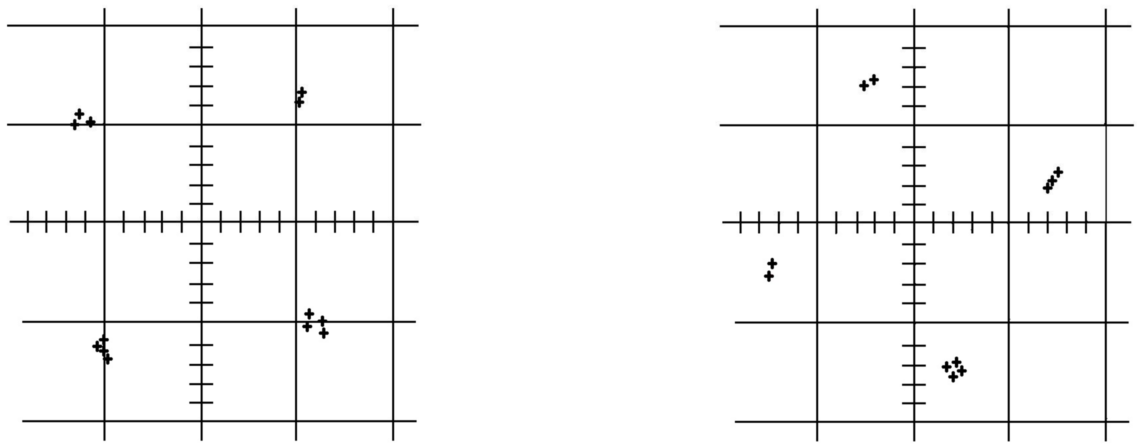

Fig. 2. Amplitude-phase constellations of the received signal with quadrature phase modulation at different times. The signal is transmitted through an acoustic channel

Due to a lack of computer resources, the model may not function in real time. At the same time, the received text does not correspond to the transmitted text. The testl model allows you to check the correctness of the model's functioning in real time. For this, the model generates an unmodulated signal at a frequency of 800 Hz. This signal can be listened to and simultaneously viewed at the microphone input in the time and frequency domains. The presence of interruptions when listening instead of a "pure tone" indicates insufficient computer resources for real-time signal processing. In this case, the model must be precompiled. The easiest way to do this is to enable the Accelerator mode instead of the Normal mode before starting the model.

The method of modulation, when the parameters of the carrier oscillation change in leaps and bounds, is called manipulation. Depending on which parameters are changed, amplitude, phase, frequency and quadrature manipulation are distinguished.

The package of Communications and Signal Processing MATLAB contains a number of functions that implement analog and digital modulation and demodulation of signals, giving a result in the form of a set of readings of a valid modulating signal or its complex envelope [3,5—8].

Modeling of phase manipulation, amplitude, quadrature and frequency manipulation in the MATLAB environment was carried out (Fig. 3—6).

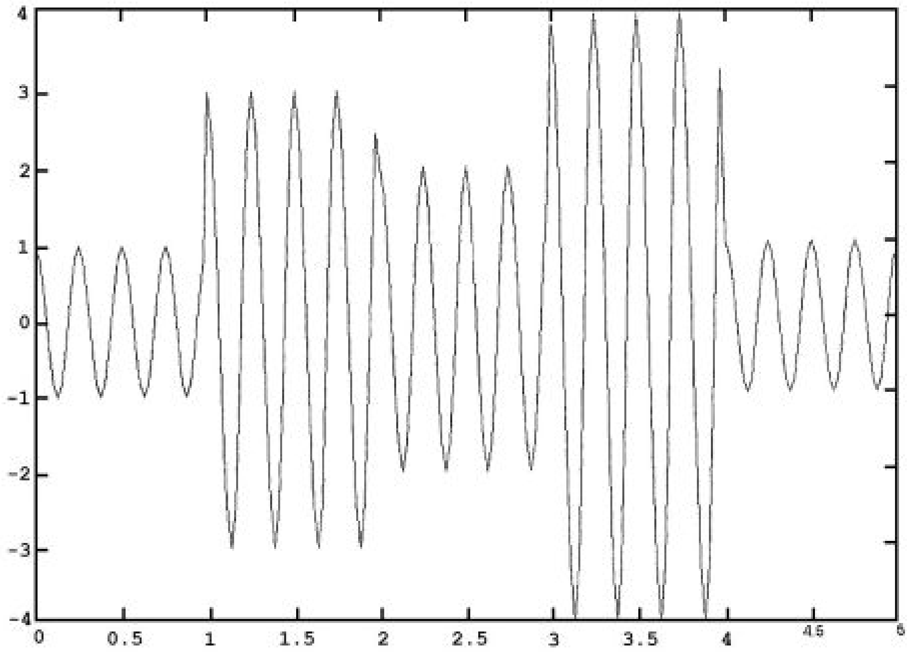

Fig. 3. Signal with amplitude manipulation

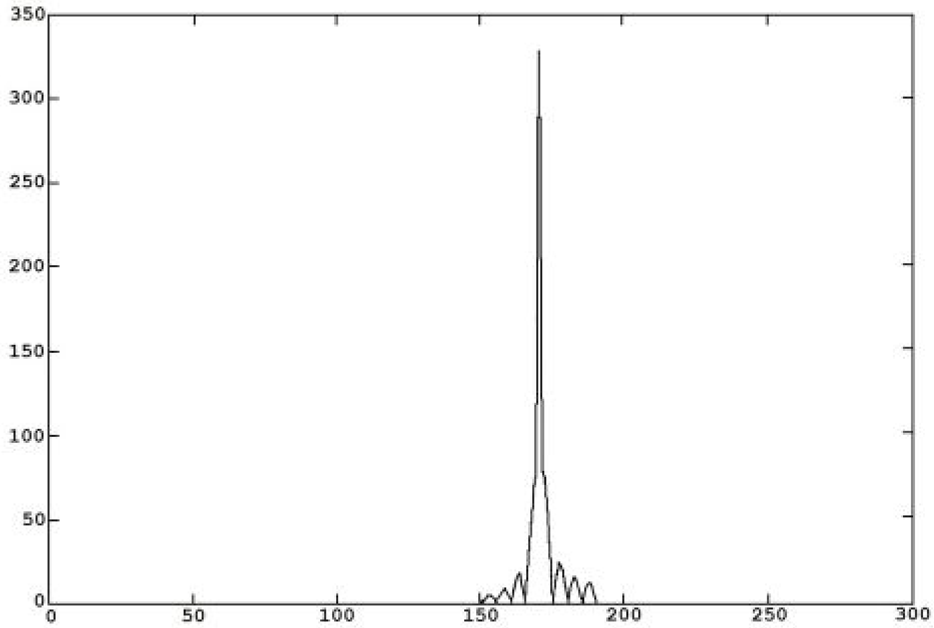

Fig. 4. Signal spectrum with amplitude manipulation

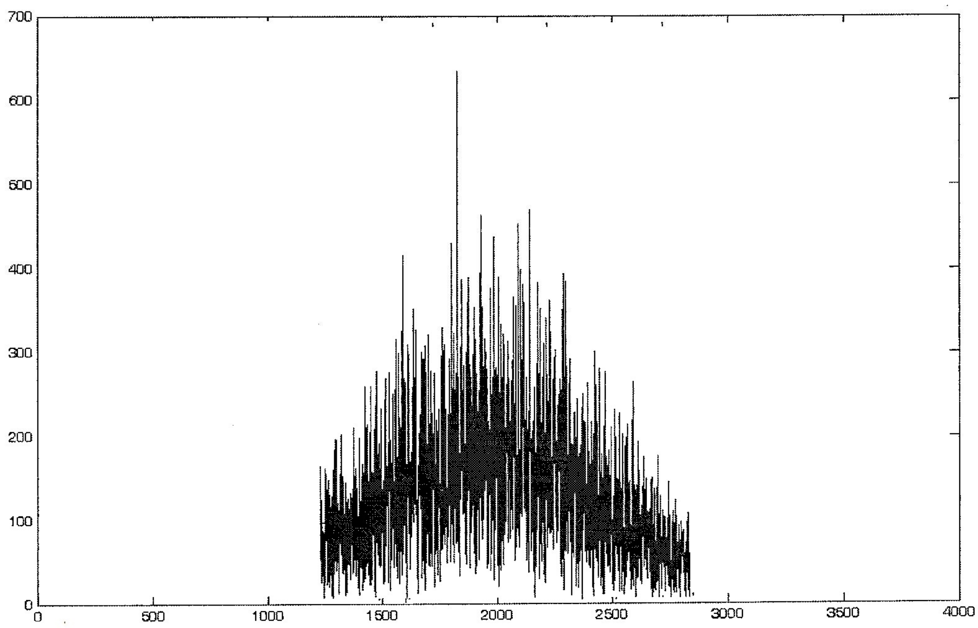

Fig. 5. Signal spectrum with 16-position quadrature manipulation

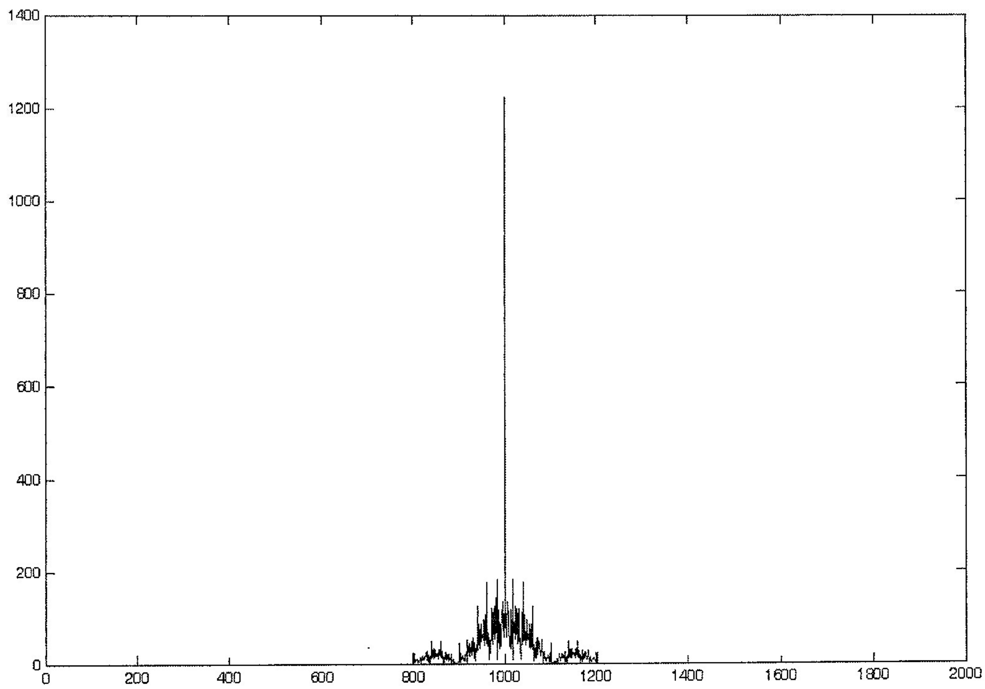

Fig. 6. Signal spectrum with frequency manipulation

The temporal dynamics of the manipulated signals is a sequence of rectangular pulses (parcels) and has the following general form

ASK , PSK, QASK:

$$u(t) = \sum_{n=-\infty}^{\infty} A_n u_0(t - nT)\cos(2\pi f_0 t + \varphi_n);$$

FSK:

$$u(t) = A_0 \sum_{n=-\infty}^{\infty} u_0(t - nT)\cos(2\pi f_0 t + \varphi_0),$$

where A_n, φ_n — is the amplitude and the initial phase of the n — th parcel, f_n — the carrier frequency, $u_0(t)$ — the envelope of the parcel (rectangular pulse of unit amplitude).

Conclusions

In this work, the PSK-31 digital system was investigated, as a result, a computer model of this system was obtained. Studies have shown that this system is fully operational and is in good agreement with theoretical representations. Comparative modeling of various digital modulation options (PSK, FSK, ASK, APK) was conducted. Based on the simulation results, the used type of modulation turned out to be one of the best. Software noise reduction and software gain control were also applied. The results of the use of software DSP filters improve the dynamic characteristics of the PSK-31 receiving path.

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