COMPLEX SIGNALS BASED ON SHORT CHIRP SIGNALS AND PHASE CODING

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The main problem associated with detecting and locating of objects by means of radiolocation and hydrolocation systems is the appropriate selection of transmitting signals. These signals need to be difficult to detect by the watched object and need to be resistant to disturbances. The most often used ones are signals with linear frequency modulation (chirp signals) and signals with phase coding. New possibilities of the realization of complex signals appeared along with development of the digital technique. The usage of signal composite from short signals with linear frequency modulation with use of phase coding was analysed in this paper. Short noise-like sequences (Barker's codes) were used for coding the initial phase of chirp signals. This article presents results of research obtained for a signal of this type, at its identification by means of matching filtration. These results were compared with results obtained for single signal with linear frequency modulation with the same duration and bandwidth.

INTRODUCTION

Noise-like codes and wide-band signals have been more and more often used to the different researches in recent years. A short noise-like code allows increasing speed and reliability location systems. Algorithms of their identification allow to make regular and simple structures, very easy transformed to structures based on neural networks and programmable logic device. New possibilities of the realization of complex signals appeared along with development of the digital technique. It is advisable to use the wide band signals with linear frequency modulation and different initial phase, which so called chirp signal, as carriers for nose-like codes. Due to wide-band character of these signals, they show larger resistance to different disturbances, and also to Doppler's effect [1,2]. Such connection of these two techniques permits to create highly reliable location system.

The purpose of this paper is to increase efficiency of digital location systems through the application of complex signals based on short chirp signals and phase coding.

The correlation or matching filtration are most often applied to detect and identify these types of signals. The chance of detecting and identifying the signal exists even in the presence of considerable disturbances [3]. Digital matching filtration can be realized in time domain, where it is basing on convolution operation, or in frequency domain, where fast convolutions are being utilized. The realization of matching filtration by means of fast convolutions may lead to the limitation of resolution and speed of working of devices based on this algorithm in comparison with the algorithm based on convolution in time domain.

1. MATCHING FILTRATION IN TIME DOMAIN

Convolution is the principle of matching digital filtration in time domain and it can be shows as:

$$y_n = \sum_{m=0}^{N-1} x_{n-m} h_m \, , \tag{1}$$

where $\{x_n\}$ - input time series to filtration, $\{h_n\}$- weight factors of impulse response of filter, $\{y_n\}$ – result of filtration, N - number of sampling signal or length of code series. For binary series, which are concerned with the sign series, algorithm (1) shows as follows:

$$y_n^{(c)} = \sum_{m=0}^{N-1} E_{n-m}^{(x)} E_m^{(h)} \tag{2}$$

where $\forall E \in \{-1,1\}$, $\{E_i^{(x)}\}$ - input series, $\{E_i^{(h)}\}$ - filter factors, $y_n^{(c)}$ - result of convolution function for code (sign) series. Algorithm (1) of work of a specialized processor to digital matching filtration of one-bit noise-like codes can be written in the following way:

$$y_n = E^{(x)} \cdot E^{(\mathbf{h})} \, , \tag{3}$$

where:

$$E^{(x)} = \left\| E_n^{(x)} \cdots E_{n-N+1}^{(x)} \right\|, \; E^{(h)} = \left\| \begin{matrix} E_0^{(h)} \\ \vdots \\ E_{N-1}^{(h)} \end{matrix} \right\| \tag{4}$$

Chirp-signal with constant amplitude and linear frequency modulation are studied in this paper. This signal is defined on basis of time series and gives as follows:

$$x(t) = A cos\left[2\pi\left(at + f_1\right)t + \varphi_0\right] \tag{5}$$

where $a=\Delta f/\tau_i$, $\Delta f=f_2-f_1$ - deviation of the frequency, f_1- initial frequency, f_2- final frequency, τ_i - during of a radio pulse, φ_0-an initial phase.

For realization of the digital matching filtration the signal (5) was represented in a form of time series $\{x_n\}$ with a sampling rate $f_d \geq 2f_2$, whence the number of samples is equal $n = \overline{1, N}$, where $N = \mathrm{ENT}(\tau_i f_d)$, ENT - whole from bottom part of number. The pulse response of the matching filter without a smoothing window is a mirror display of an input signal (5):

$$h_n = x_{N-n}, \ n = \overline{1,N}. \tag{6}$$

For reduction of the Gibb's oscillations occurrence at filtration of chirp-signals smoothing windows are applied. In time domain the use of windows occurs by multiplication of the appropriate weight factors of the pulse response and a window $\{h_n w_n\}$. Then algorithm (1) shows as follows:

$$y_n = \mathbf{X} \cdot \mathbf{H}_w, \ \mathbf{H}_w = \left\| \begin{matrix} h_0 w_0 \\ \vdots \\ h_{N-1} w_{N-1} \end{matrix} \right\| \tag{7}$$

Smoothing window permit to increase a ratio between main and side lobes of an output signal of the filter, but simultaneously cause width increase of the main lobe. Therefore choice of definite window depends on the requirements and parameters of signal. On the basis of the analysis of matching filtration in time domain, it is possible to create a location system based on signals with linear frequency modulation and noise-like codes at same time.

2. COMPLEX SIGNALS

The modulation of the transmitted pulse eliminates few important drawbacks of single unmodulated pulse [1,2], such as weak range accuracy, weak resistance to interferences and eavesdropping.

Linear frequency modulation is often used because of its ease of application. The bandwidth of the transmitted pulse is then given by the frequency deviation produced by the modulation. Pulse with linear frequency modulation (chirp) belongs to the group of noise pulses but it has advantage over them. It is relying for possibility to control their parameters.

Another type of modulation used often is phase coding, which, in its simplest version, consists of breaking down the basic pulse into consecutive subpulses, multiplied by a sequence and selected in order to obtain the desired ambiguity function. This kind of modulation enables the generation of an ambiguity function with a rather narrow main peak, at the cost of sidelobes that are difficult to reduce in the Doppler mode.

For manipulating phase of subpulses are often used noise-like codes with special group of codes named Barker's codes. Barker coding makes it possible to produce coded sequences with maximum autocorrelation value and minimum sidelobes level [1]. These codes exist for length 3,4,5,7,11 and 13 bits.

In this paper, pulse consisting of short subpulses with linear frequency modulation about the various initial phase is proposed. An example of such pulse corresponding with Barkers code 11101, figure 1a shows. Figure 1b shows result of filtration this signal in linear scale and figure 1c in logarithmic scale. The space between main lobe and side lobe is described in logarithmic scale as $20\lg(A_m/|A_s|)$, where A_m - mainlobe amplitude, A_s - sidelobe amplitude.

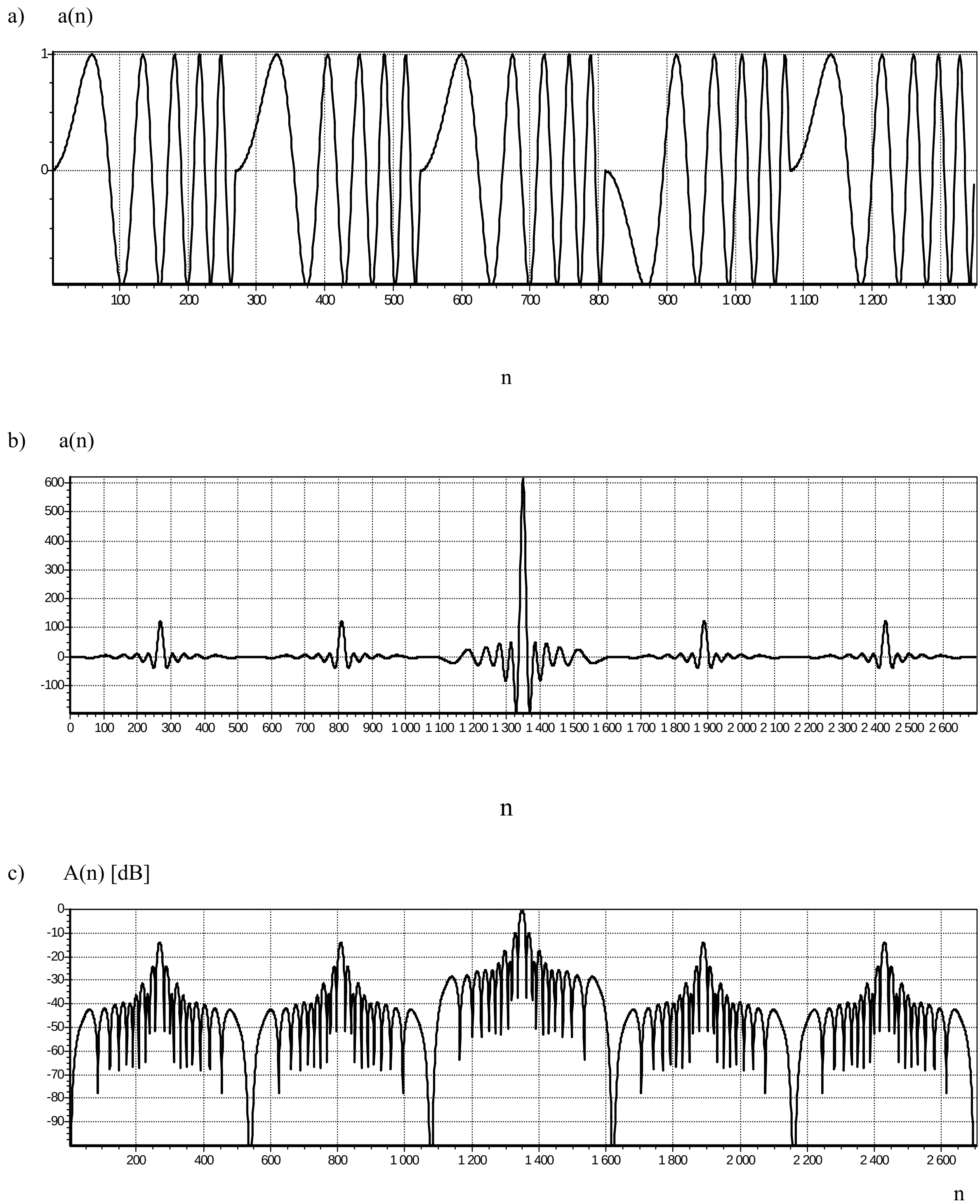

Fig.1 a) Signal consisting of short subpulses with linear frequency modulation about the various initial phase corresponding with Barkers code 11101, b) signal on output of matching filter, c) signal on output of matching filter in dB

Figure 2 shows relationship between mainlobe to sidelobe ratio and sampling frequency f_S (normalized with respect to Nyquist frequency f_N) for various Barker's codes. This figure confirms, that the mainlobe to sidelobe ratio is not depending on sampling frequency but it is depending on length of code only, and for example it equals 13dB for five bits code.

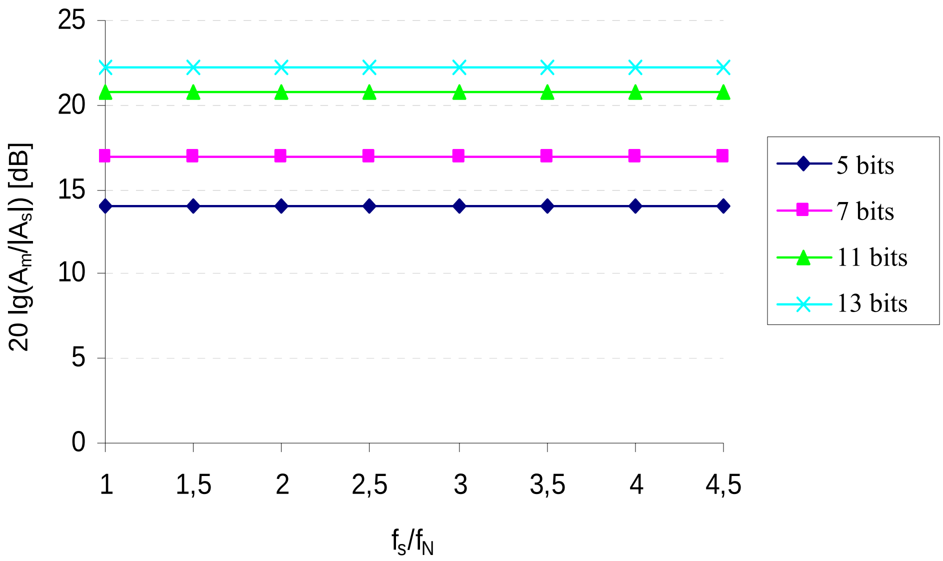

Fig.2 Relationship between mainlobe to sidelobe ratio and sampling frequency for various Barker's codes

3. FILTRATION IN PRESENCE OF DISTURBATIONS

Research in presence of disturbations were provided at usage of simulation methods. The block diagram for this research was presented in figure 3.

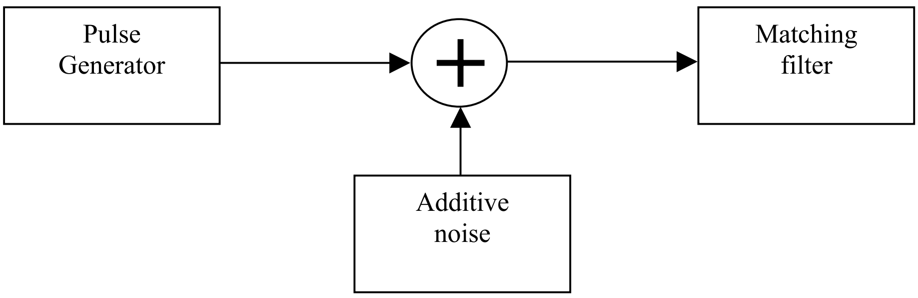

Fig.3 Block diagram for research

The white noise and the Weibull's model of interferences were utilized in research. The formula for the probability density function of the standard Weibull distribution is [4,5]:

$$, \quad p(X) = \frac{a}{b}\left(\frac{X}{b}\right)^{\alpha-1} \exp\left[-\left(\frac{X}{b}\right)^{\alpha}\right], \quad X \geq 0 \qquad (8)$$

where α - shape parameter, b – scale parameter. This model is often using to describe of disturbances from earth surface (shape parameter in range from 0.3 to 0.6) and atmospherics disturbances (shape parameter in range from 1.2 to 2.0). The signal to noise ratio (SNR) is described in logarithmic scale as $10\lg(P_x/P_\xi)$, where P_x – power of signal, P_ξ - power of noise.

Results of research were presented for matching filter with rectangular window. Input signal was build of short chirp signals with parameters: middle frequency f_0=70 MHz, band B=5 MHz, duration 2μs and initial phase 0° or 180°. Initial phase was selected in dependence on five-bits Barker's code (11101). In this case, duration of all sequence equals 10μs (BT=50). Figure 4 shows relationship between mainlobe to sidelobe ratio and signal to noise ratio (SNR) for various frequency sampling. Frequency sampling equals 290MHz corresponding with lower-band sampling and other corresponding with band sampling.

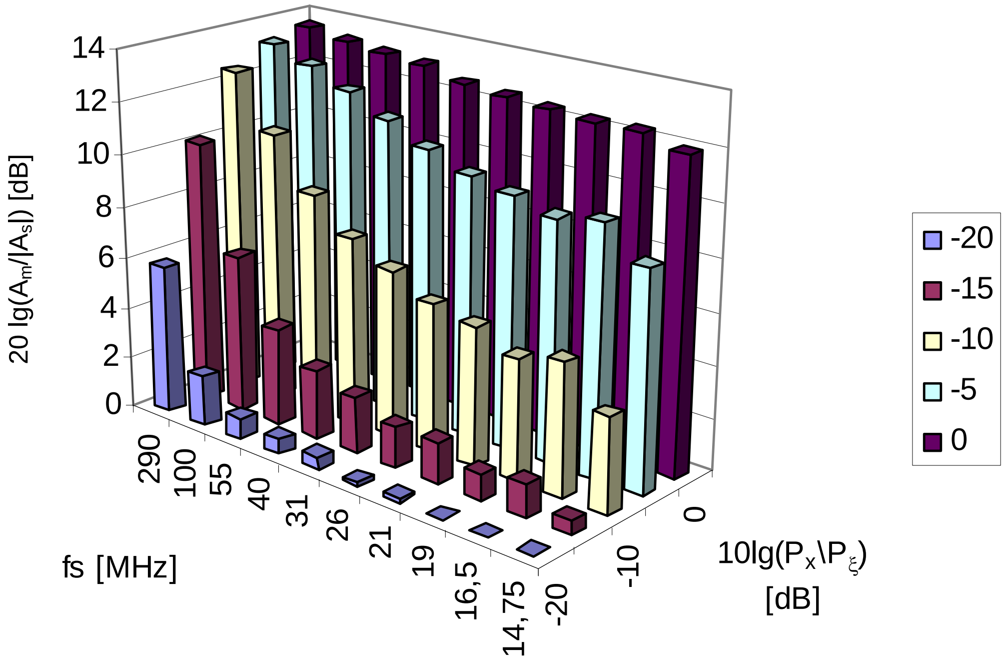

Fig.4 Relationship between mainlobe to sidelobe ratio and signal to noise ratio on input of matching filter with Hamming's window for various sampling frequency

Obtained results show, the choice of low frequencies of sampling has small influence at the power of the noise comparable with the power of the signal. If the signal to noise ratio on input of the filter equals –20dB the best results are obtained for traditional sampling. Results of filtration of single chirp signal with Hamming's window and filtration of signal with linear frequency modulation and phase manipulation (LFM_C1) with rectangular window are compared on figure 5. These results were obtained at frequency sampling equal 16,5MHz.

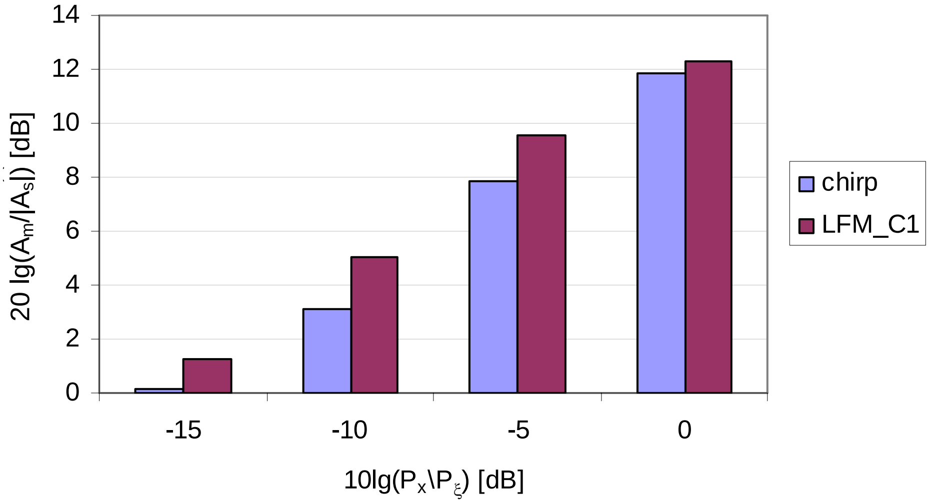

Fig.5 Relationship between mainlobe to sidelobe ratio and signal to noise ratio on input of filter with Hamming's window for single chirp signal and signal with linear frequency modulation and phase manipulation (LFM_C1) at sampling frequency 16,5 MHz

Filtration of the LFM_C1 signal, about the same BT product like single chirp, is ensuring the bigger relation of the mainlobe to sidelobe (about 2dB) at low frequencies sampling.

The same comparison was performed for results, obtained at the frequency of sampling 290 MHz (lower-band sampling). The result of this comparison figure 6 shows.

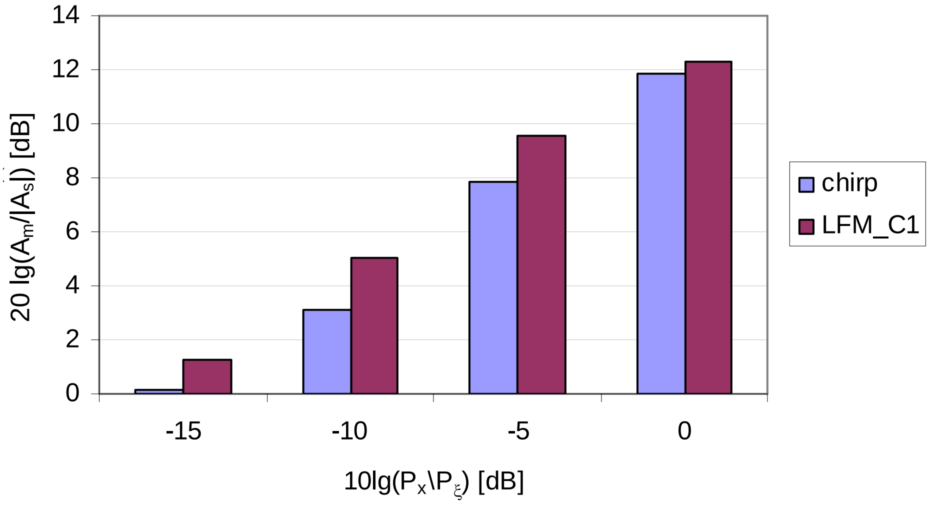

Fig.6 Relationship between mainlobe to sidelobe ratio and signal to noise ratio on input of filter with Hamming's window for single chirp signal and signal with linear frequency modulation and phase manipulation (LFM_C1) at sampling frequency 290 MHz

Obtained results confirm, that also at lower-band sampling with the frequency equal to the double Nyquist's frequency, at the signal to noise ratio of the on the input of the filter smaller from -10 dB, effects of filtration of the LFM_C1 signal are better than at filtration of the single chirp signal.

Influence of the signal to the noise ratio on the level of the mainlobe and sidelobes was shown on figure 7 for three signals about the same duration, treat put together from the various number of short chirp signals, dependent on the length of manipulating sequence.

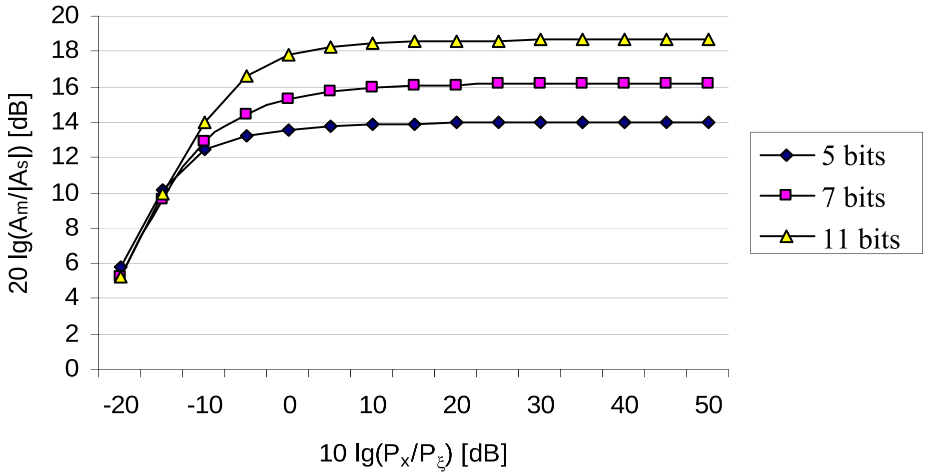

Fig.7 Relationship between mainlobe to sidelobe ratio and signal to noise ratio on input of filter for various Barker's code

How research shown, the length of manipulation sequence hasn't influence on mainlobe to sidelobe ratio at the high level of disturbations. The result of filtration is depending on the length of code at the low level of disturbations.

4. CONCLUSION

In proposed complex signals space between the main lobe and sidelobes is the same as for the manipulating noise-like sequence, but the designed signal is showing bigger resistance to disturbances in comparison with single chirp signal with the same duration and the width of the band.

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