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STUDYING THE EXACTITUDE OF DIGITAL MATCHING FILTRATION OF WIDEPASS LOCATION SIGNALS*

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The resolution of matching digital filtration of widepass location signals in time domain is studied in this work. The advantages of this kind of processing in comparison to the processing on fast convolutions in frequency domain are demonstrated. A choice of parameters of the long and short chirp signals and smoothing window have been studied to increase the ratio of the main lobe to side lobes of convolutions. This choice has carried out with the use of computer simulation and result has shown in this paper.

INTRODUCTION

Widepass signals with the use of linear modulation of frequency (the so-called chirped signals) is used in the hydrolocation, in particular in the chirp sonars [1]. In digital location, when measuring time shifts, the matching filtration for output compression these pulses are often used.

Lately, for chirp-signals compression the specialized processors on "fast convolution" [2,3] have been used. That algorithm provides for the use of FFT to define Fourier's image input signal on the basis of $N \cdot lbN$ multiplications and summations, where $lbN = log_2 N$, and N complex multiplications to define N products, and $N \cdot lbN$ multiplications and summations for the IFFT of these products. This algorithm corresponding convolution in time domain and PCM format

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

where $\{y_n\}$ – filtration result; $\{x_n\}$ – input signal samples; $\{h_n\}$ – weight factors of impulse response, N – number of weight factors and signal input samples

Algorithm (1) total number of multiplications in such MF in PCM format and time domain is N^2. However, if MF consist of N many bit multipliers working in PCM format, all

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multiplications are made by $2N-1$ intervals of one multiplication. It is about lbN times faster than the algorithm which ensures MF with fast convolutions in frequency domain.

The purpose of this work is to study the influence of parameters of chirped signals as well as impulse response of matching digital filters on exactitude of matching filtration in time domain used for digital location.

1. MATCHING DIGITAL FILTRATION IN TIME DOMAIN

In order to decrease the influence of Gibbs' oscillations the smoothing window $\{w_n\}$ is used, which is put on in time domain by means of the use of the multiplication of corresponding weight factors of impulse response and window: $\{h_n\, w_n\}$ [4].

The smoothing window allows to increase the ratio of main lobe to side lobes of the output compressed pulses, but at the same time to increase the width of the main lobe. In this work we studied the influence of next windows on compression of pulses: rectangular, Hamming's, Blacman's, Blacman-Harris'-3, Gauss'.

The work-out algorithms are checked with the use of computer simulations. The cosine signals with changing frequency are used to study this

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

where $a = \Delta f / 2\tau_i$, $\Delta f = f_2 - f_1$ - deviation of the frequency, f_1 - initial frequency, f_2 - final frequency, τ_i - pulse duration, φ_0 .- initial phase.

For the realization of matching digital filtration we shall present signal (2) in the form of time series $\{x_n\}$, by selecting the sampling rate f_s according to the condition $f_s \geq 2f_2$. Then $n = \overline{1,N}$, where $N = ENT(\tau_i f_s)$. Impulse response (IR) matching filter with weight factors $\{h_n = x_{N-n}\}$ is the mirror reflection of signal (2).

The results of computer simulations of algorithm (1) with use of Hamming's window are presented in Fig. 1.

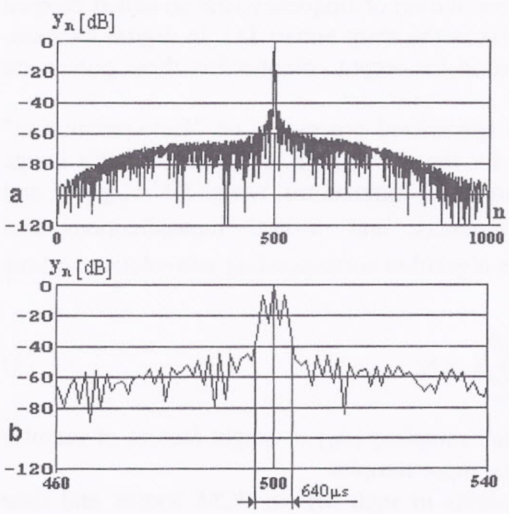

Fig. 1. Result of computer simulations of matching filtration with the use of Hamming's window.

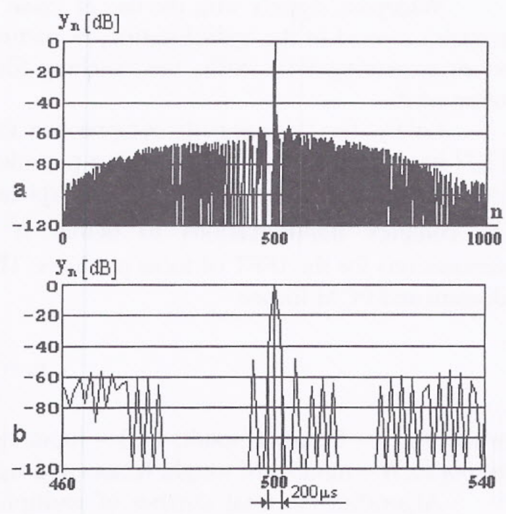

Fig.2. Result of computer simulations of matching filtration with the use of Hamming's window for positive values of convolution only.

We studied the time series corresponding to the chirp-signal (2) with the parameters: $f_1 = 0$, $\Delta f = 6kHz$, $\tau_i = 40ms$, $\varphi_0 = \pi/2$. The frequency of signal (2) linear change, and product $\Delta f \cdot \tau_i$ equal 240. The sampling rate and the number of the samples of the input signal are found with the use of the condition $f_s \geq 2f_2 = 12kHz$ and $N = 500 > f_{s.min} \cdot \tau_i = 480$. The result of matching filtration in logarithmical scale is shown in Fig. 1a. The ratio of main lobe to side lobe of output compressed pulses outside of main lobe limits exceed $60dB$ for indicated parameters of signal by the use Hamming's window. This signal is shown in Fig. 1b in the wide scale. In this event it allow to study a width (duration) of main lobe which equal $640\mu s$ on level $-40dB$.

We can improve the obtained parameters of compressed pulses if we use some peculiarity of digital systems. For example we can easily remove the all negative values of convolution. The results of computer simulations of matching filtration when we use only positive values of convolution $\{y_n\}$ for indicated parameters of signal are shown in Fig.2a. These results in the wide scale are shown in Fig.2b.

According to Fig.2 the rejection of convolution's negative values lead to the increase of SNR in a neighbourhood of main lobe and to the essentially narrowing of it. For this example, the width of main lobe is $200\mu s$ on level $-40dB$. The absence of the smooth passages from main lobe to side lobes allow to improve a resolution of these pulses recognition and is another advantage of this method.

Computer simulations showed that this ratio depends essentially on the initial phase and number of samples N of the chirp-signal. Therefore we can essential improve the obtained results if we will assign the corresponding N and φ_0. For example the 2-D dependence of the ratio SNR of maim lobe to maximum side lobe from initial phase φ_0 and number samples N of chirp-signal is shown in Fig.3 by parameters of chirp-signal: $\tau_i = 10ms$, $f_1 = 0$, $\Delta f = 15kHZ$, $\tau_i \cdot \Delta f = 150$.

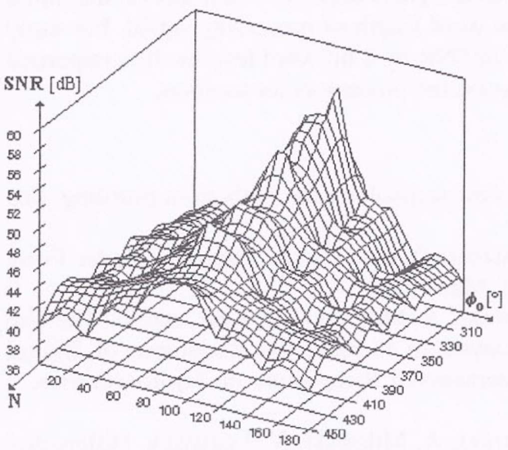

Fig.3. Dependence of the SNR from initial phase φ_0 of chirp-signal and number N for only positive values of convolution.

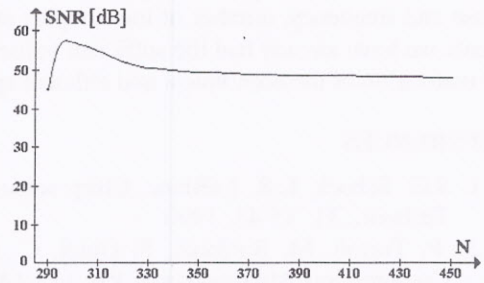

Fig.4. Average dependence of the SNR from number samples N by $\varphi_0 = 100°$ for only positive values of convolution.

The Nyquist's frequency is $f_N = f_s = 2f_2 = 2\Delta f$ if $f_1 = 0$ and then number of samples is $N = \tau_i \cdot f_s = 300$. According to Fig.3 the best values of the SNR for the above mention parameters of signal are for a sampling rate within f_N by initial phase φ_0 change in limits from $80°$ to $100°$. When $f_s > f_N$ (number of samples $N > N_{min}$), the SNR is decreased by initial phase $\varphi_0 = 5\pi/9 = 100°$.

Maximum values of SNR achieve $57dB$ for this event. The SNR values are better visible on 1-D graph (Fig.4).

The found dependence of the ratio of main lobe to side lobes from initial phase and initial frequency of chirp-signal are shown by different values of the products $\{\Delta f \cdot \tau_i\}$ of this signal. However, when the value $\Delta f \cdot \tau_i$ is being decreased this influence is less visible.

The main difficult of practical realization of specialized processors for digital matching filtration is caused the big word lengths of processing dates. The computer simulations of matching filtration by different smoothing windows and different parameters of chirp-signals showed the SNR hasn't already improved when word length surpass the 12 bit. However for many events this word length may be essential less. For example we showed in Fig.5 the family of dependences of SNR from word length by the use of rectangular window and different initial phases for chirp-signal by parameters: $\tau_i = 2.5ms$, $f_1 = 0.3kHz$, $\Delta f = 15kHZ$, $\tau_i \cdot \Delta f = 37.5$, $N = 76$. According to Fig.5 the maximum SNR by initial phase $\varphi_1 = 1.7rad$, is decreased less than $3dB$ even by 6 bit words. It is very useful for realization of the fast-acting matching digital filters for location.

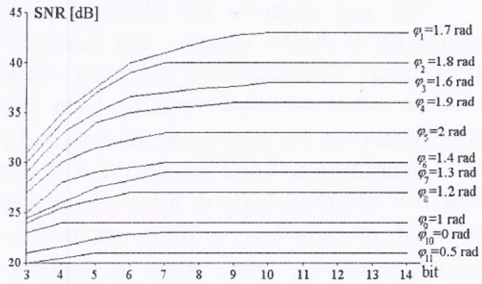

Fig.5. The SNR as function of coding word length by the usr of rectangular window.

2. CONCLUSION

The N-channels matching filters in time domain are more fast-acting (about $lb\ N$ times) than the filters on fast convolutions in frequency domain and have identical resolution. The authors found out that pulse compression of chirped signals depends essentially on the initial phase and frequency, number of the samples and word length of processing signal. For many events we have already had the sufficient values of SNR by 6 bit word lengths. It is important for realization of the economical and efficient specialized processors for location.

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