SIMULATION OF THE TIME – SPATIAL FILTRATION ALGORITHM BY 2D FFT EXPANDED METHOD AND ITS PROPERTIES

TADEUSZ JANOWSKI, ANDRZEJ KOTŁOWSKI,
WŁADYSŁAW MĘCIŃSKI, MARCIN OKUNIEWSKI

OBR Centrum Techniki Morskiej
ul. Dickmana 62, 81-109 Gdynia, Poland
tjanow@ctm.gdynia.pl

The article presents the simulation results of the time-spatial matched filtration algorithm in detail described in [1]. The features of the algorithm are presented on figures of the realized simulations. Firstly, the simulation results in case of far field are described. Successively the realizability of different filtrations types within algorithms is presented. Next the possibility of the algorithm adaptation to near field is shown. The last of the presented properties are possibilities of the directions of arrival choice.

INTRODUCTION

The meaningful role in active sonar has the signal processing in order to proper underwater targets localization. The wideband sonar gives the new quality due to simultaneously increase of the SNR and range resolution, but it requires the high computing power. One method of the time-spatial filtration for wideband signal is presented in [1]. Those method decreases necessary computing power. Presented method has features worth presentation. These properties are presented on simulation examples.

1. SIMULATION

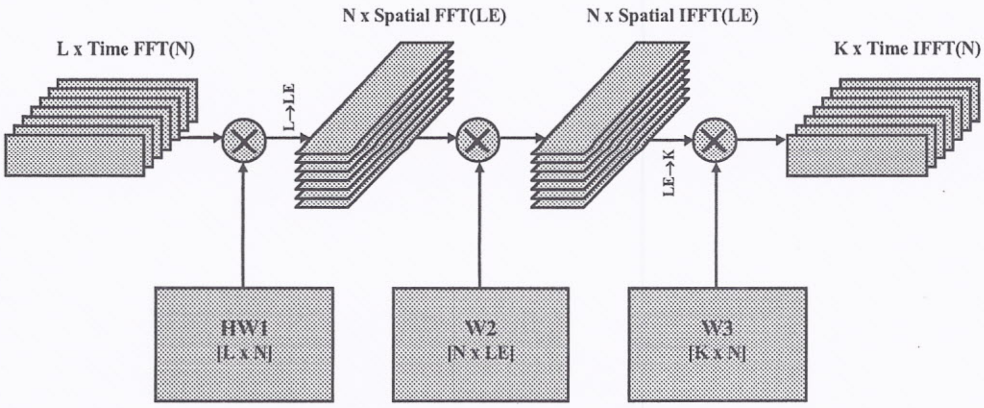

Fig. 1 Algorithm block diagram.

The best verification of the developed algorithm is its implementation. This requires simulation of the echo signal reflected from "point target". Prepared computer simulator make possible selection of the following impulse parameters:

- length,
- central frequency,
- frequency band,
- type of modulation: CW, LFM and HFM.

Next, the independent sonar reception channel has been simulated. The linear transducer of the uniformly distributed elements has been modeled. The signals from transducer are sent to I/Q detectors. The I/Q detection consist in high frequency wideband signal transformation into low frequency complex signal. I/Q detectors consume output signals from each transducer element and they are steered by single generator. At the I/Q detectors block output, the vectors of complex data are obtained at uniform rate. These signals are successively processed by extended 2D FFT algorithm.

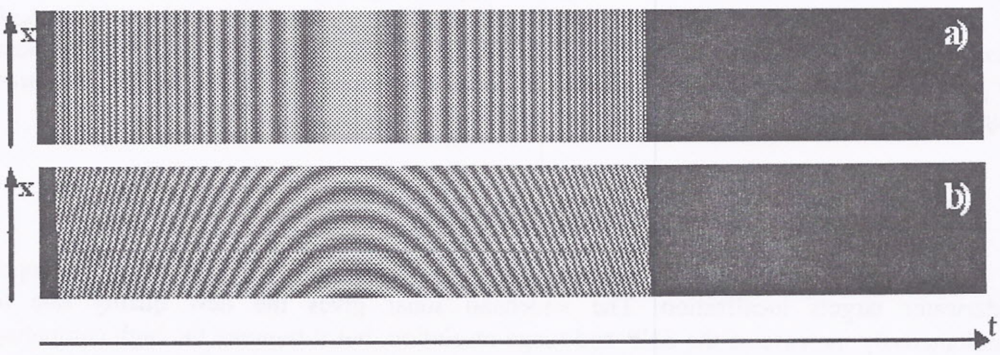

Fig. 2 Phase of the input signal for Extended 2D FFT
a) DOA=0°, b) DOA=10°.

2. THE MATCHED FILTRATION

On the method derivation phase the decomposition of the signal pattern into two components was realized. One component depends only on transmitted impulse form. The second is the function of the wave delays at the transducer. This component depends on: distance from transducer central element (x), target bearing (Θ) and for near field case – the target distance from transducer centre. However, such decomposition is not necessary. Fundamental is the fact of those components independence. As the result of this, the algorithm can be used for any type of transmitted impulse modulation. The results for different impulse modulation are similar as it is shown on the *Fig. 3* a) and b). Of course, if the frequency band decreases – in extreme case to single frequency- the possibilities of the pulse compression reduces too – *Fig. 3*) c.

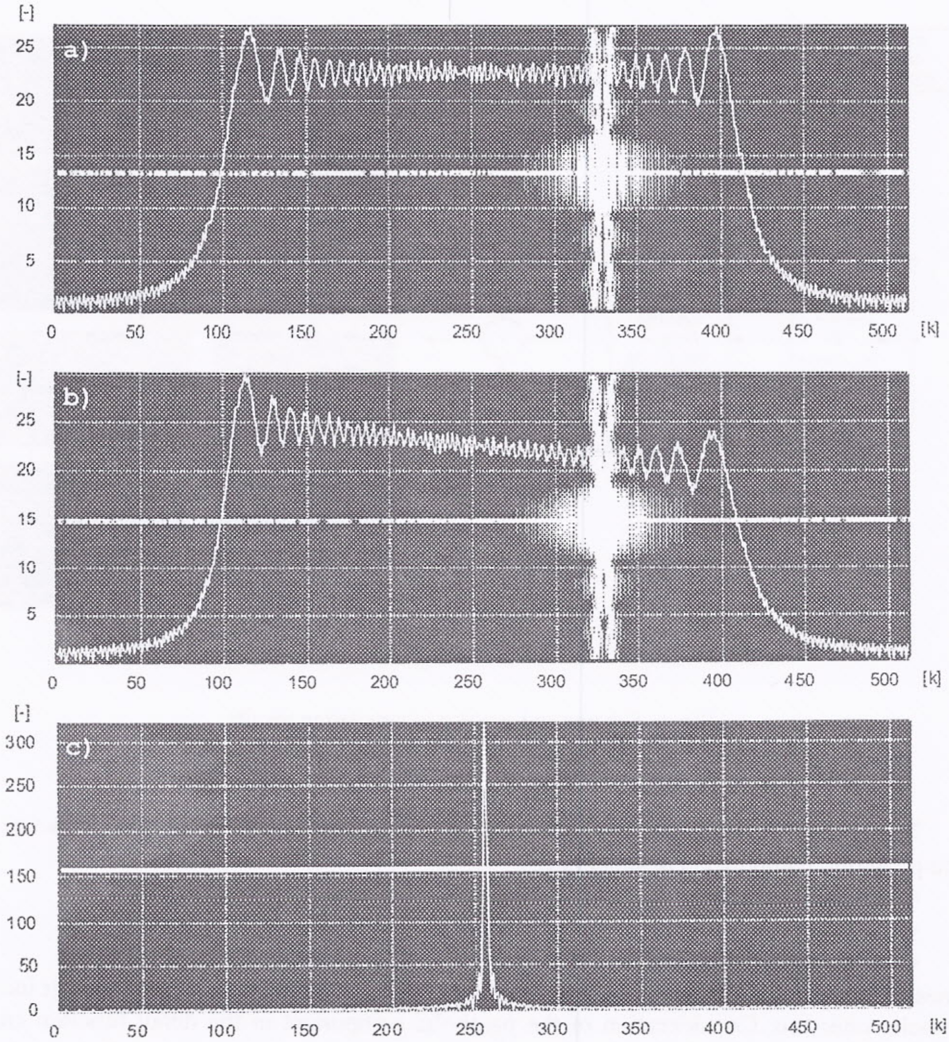

Fig. 3 Spectrum of the echo signal and segment Extended 2D FFT output for modulations: a) LFM, b) HFM, c) CW.

It is noticeable that changes are realized without any modification of the algorithm structure. The change of the modulation pattern is realized only by change one of the W1, W2 or W3 weighting coefficients. Filtrations different from matched one can be included into algorithm structure as well.

<u>Example:</u> Low-pass filters permitting signal decimation are introduced at the beamformer output. Their simplest case - walking average at beamformer output of P length can be decomposed into P matched filters of M length shifted by one sample, which outputs are summed. Successively, such configuration can be transformed into single filter of M+P-1 length. It is equipollent to matched filter. The algorithm output without and with walking average of 6 length is illustrated on the figure 4.

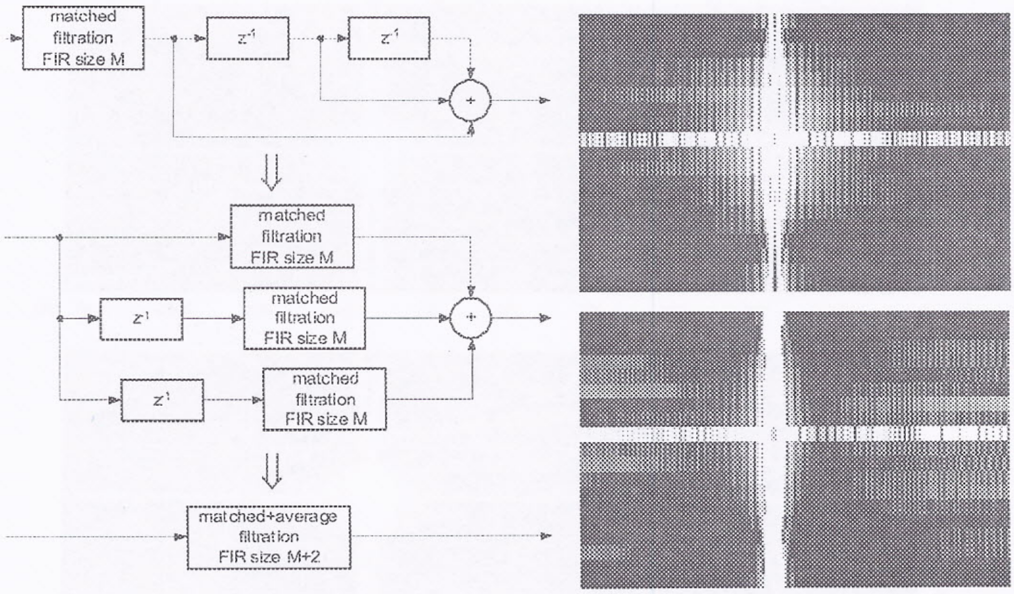

Fig. 4 Scheme and graphic of simulation results.

3. THE WEIGHTING MODIFICATION - FOCUSING

The crux of the Bluestein method in far field is decomposition of the delay function into three parabolic components dependent on:

- distance from transducer center,
- sum of the distance from transducer center and direction of arrival,
- direction of arrival.

In near field case the delay can be expanded into series. Except linear component, it includes parabolic ones too. Consideration of the parabolic component in the delay function enables focusing. Of course this method is not precise. The remainder of the series components causes error. Some modification done on the series minimize integral of error in the sonar observation sector. It provides to quite well results as shown on the figure 5.

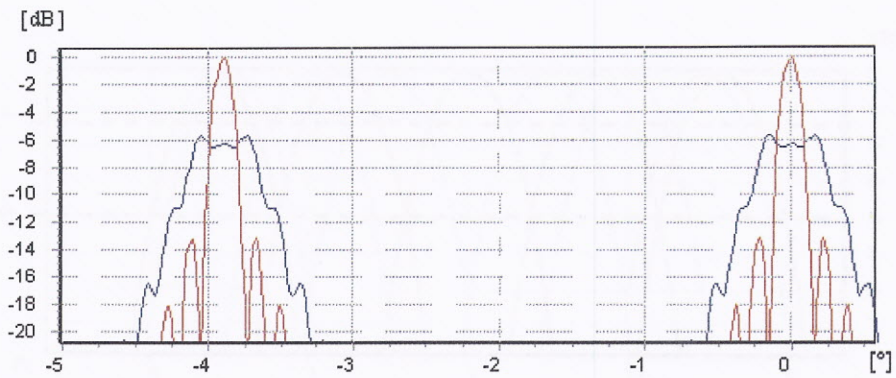

Fig. 5 Beampatterns for two directions with (red) and without (blue) focusing.

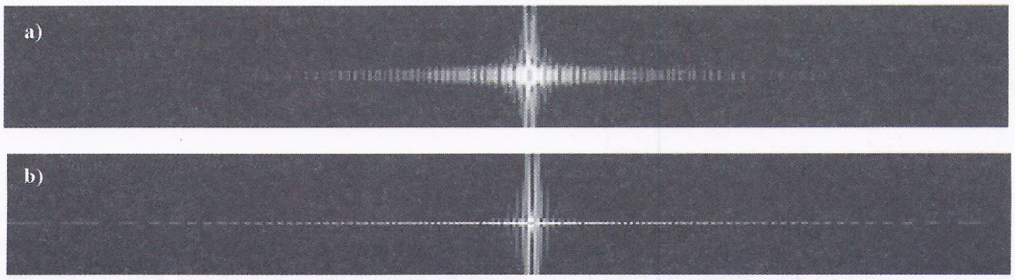

Fig. 6 Output segment: a) without, b) with focusing.

4. DENSITY OF DIRECTIONS

Possibility of angular resolution arbitrary selection is the next property of the proposed method. The directions (for calculation by discrete Fourier transform) are distributed in accordance with following formula:

$$\Theta_r = \operatorname{asin}\left(\frac{r \cdot \lambda}{L}\right) \tag{1}$$

Extended 2D FFT applies the Fractional Discrete Fourier Transform to direction calculation. In accordance with that unrestricted density of directions can be chosen. This method is adapted into hydroacoustics purpose. So, the antenna can be virtually rebuilt into enlarged one. This method requires increase of the internal segment size to LE (power of 2). LE-L+1 output directions are obtained. Hence, the all outputs can be allocated into observation range. The example of the beams overlap on chosen level is presented on the fig. 7.

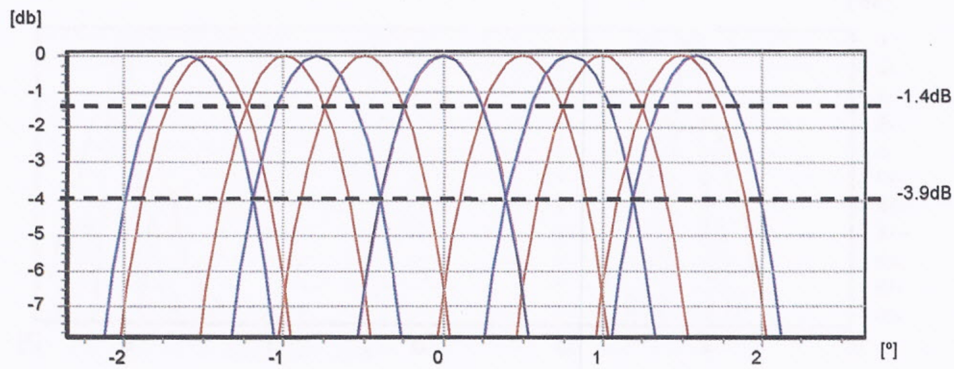

Fig. 7 Beampatterns for antenna lengths: L=50 (blue) and L=80 (red).

5. COMPUTING POWER

Computing power, required to realize all available direction by this method is:

$$\frac{N \cdot LE \cdot \left(\frac{1}{2} \cdot \log_2(N) + 2 + \log_2(LE)\right)}{N - M + 1 - Slant} \quad \text{complex multiplications} \qquad (2)$$

where: *Slant* depends on maximal DOA, M is impulse length and N is segment size in time. For comparison, directly realized phase-delay beamformer need about four times more computing power.

6. CONCLUSIONS

The simulations proved the developed method correctness. Properties coming from theory have been confirmed by experiments. Comparatively lower computing power consumption leads to usage this method in practical implementation of the sonar systems.

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