Methods and Techniques for Improving Processing Efficiency of Broadband Signals

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Abstract

Background/Objectives: The study analyzes methods and techniques for improving the effective speed of computing units to solve the problems of digital processing of broadband signals by introducing special vector instructions. **Methods**: Uniquely effective methods of digital matched filtering (MF) have been suggested for broadband signal processing: sub-aperture method and the coded moving total method. **Findings**: Performance evaluation of the matched filtering schemes and algorithms showed that, in a number of MF broadband signal problems, the coded moving total method proved to be 3-20 times more effective than the direct convolution method, and two times more effective than the fast convolution method based on fast Fourier transform (FFT) algorithms; the method of sub-apertures proved to be 2-6 times more effective than the direct convolution method. **Applications/Improvements:** Two equipment instruction manuals for moving summation in EL-core IP-cores have been developed, improving their effective performance of filtration by several times. The algorithms are used in ORWELL-R (OJSC R&D Center ELVEES) modern security radar station.

Keywords: Computing Units, Digital Signal Processing, Matched Filtering, Radio Detection and Ranging

1. Introduction

One of the most important trends in developing modern instrumentation systems is to employ the fast growing digital technologies together with new algorithms for signal processing, making the best use of computing resources.

Today, there is a great number of ways to improve the efficiency and speed of computing units (CUs). Two principal methods for improving CU efficiency are distinguished: extensive (quantitative)¹ and intensive (qualitative)²⁻³. Intensive methods imply faster speed of CUs operations while their number remains the same; quantitative methods imply increasing the number of CUs operating in parallel. One of the qualitative methods for improving CU efficiency preserving the fixed number of commands is to introduce new commands that perform more manipulations with the data and substitute several

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simple instructions. This approach affects the size of the software code and the command performance.

Elaborating digital signal processing (DSP) algorithms makes for more intensive development of instrumentations and information-measuring radio-electronic systems aimed at solving the problems of data compression, spectral analysis, adaptive correlation processing, etc. One of the subject areas of digital signal processing is represented by digital filtering. In modern radar stations (RS), to improve the range resolution at the same impulse rate, the signals of quite a long duration with intra pulse modulation are widely applied. Matched filtering (MF) ensures temporary compression of such impulse and is one of the most crucial and resource-consuming stages in digital signal processing¹⁻⁴. The quality and the efficiency of this operation affect further results of processing. In information-measuring systems (IMS) with active transducer (AT), these results are represented by target detection and recognition. Based on the outcomes of the analysis of major DSP algorithms in several multifunctional IMS with AT, upon averaging the values across different operational modes, the specific weight of the MF arithmetical operations amounted to circa 61% (Table 1). Therefore, this study suggests the new methods of MF for broadband signals and the ways to improve its computing efficiency in CUs applying special instructions.

2. Concept Headings

The main objective of this study is to show the efficiency of the developed MF methods for linear frequency-modulated signals (sub-aperture method) and for the signals with phase code manipulations (the coded moving total method) as compared to MF algorithms based on a fast Fourier transform (FFT) and to the method of direct convolution; and also to show their computing efficiency in Multicore (R&D Center ELVEES) signal controllers series⁵, that have been successfully employed by modern security radar systems (ORWELL-R)⁶.

3. Results

The algorithm for realizing the matched filter (MF) depends on the characteristics of a signal, on the number of counts of the processed arrays and on the tasks set for the digital signal processing system⁷⁻⁸. Several modifications of MF could be distinguished:

- Finite impulse response filter (FIR-filter) in its direct form;
- FIR-filter based on fast algorithms of high-frequency convolution;

DSP algorithms	Specific weight of arithmetical operations, %
Matched filtering	61
Recursive and non-recursive filtering	10
Element-by-element operations with arrays (addition, multiplication, comparison, etc.)	20
Threshold processing	4
Other	5

Table 1.Specific weight of arithmetical operations inprincipal DSP algorithms

- FIR-filter based on algorithms of convolution with short delays;
- Multi-dimensional FIR-filter based on relatively prime dividers;
- FIR-filter based on polynomial algorithms.

Each type of MF has both advantages and disadvantages of its own. Investigating methods and algorithms of MF for effective and for complex arbitrary length signals is an important area for further developing DSP, insofar as the conventional methods based on fast algorithms have serious constraints, namely, the processed vector length has to be divisible by the power of figure two. In a number of applications, it becomes necessary to apply filtering to the signals with different length⁹, therefore, a signal is either cut down, and some useful information is lost, or it is supplemented with zeroes, which makes the signalnoise ratio lower at the output of the system. To solve this type of tasks, new approaches should be developed.

An important parameter of MF is the period of delay in obtaining processing results upon receiving a signal. This delay depends not only on the architecture of the units¹⁰, but also on the arrangement of computing conveyor and DSP algorithms. The delay in signal processing has to be minimized to cut down the time of the system's response to changing environment. The delay in optimum processing is understood as the time between the moment of receiving signal counts at the input of the filter and the moment of their coming to the output. In the algorithms for optimum digital processing, the minimum delay should not exceed the time for receiving the signal counts equal to the number of counts of the impulse characteristic of the filter and should not depend on the number of counts of the return signal¹¹.

In IMS with AT, linear frequency-modulated (LFM) signals and the signals with phase code manipulations (PCM) are mostly used as broadband signals. For matched filtering in frequency domain, a fast Fourier transform algorithm is widely applied¹², which, however, has an obvious disadvantage of considerable conveying delay in producing the processing result. A shorter delay makes it possible to cut down the total time of response of the information-measuring system to changing interference-target environment, particularly, to ensure tracking the high-speed and maneuvering targets. Given the set problem, the study proposes MF methods for radio detection and ranging broadband signals with minimum delays. Digital matched filter is described mathematically according to the formula as follows (1):

$$y(n) = \sum_{k=0}^{N_h - 1} h(k) x(n - k)$$
(1)

Where h(k) is the impulse characteristic of the filter, N_h is the length of the impulse characteristic of the filter; x(k), y(n) are noise-contaminated input and output of the matched filter, respectively. Formula (1) shows that, for the purposes of matched filtering, the multiple operations of addition, multiplication, subtraction should be performed taking into account the shift that realizes the signal delay for one sampling period. The abovementioned operations are principal (basic) operations. Analysis of their number, given the architectural features and the system of commands in a microelectronic unit, will constitute the main criterion for evaluating the efficiency of matched filter algorithms.

MF in its direct form is written as convolution transformation (1), whose obvious advantage is that the results of filtering are obtained with the delays equal to the length of the impulse characteristic of a filter; however, the computing complexity of the method makes $O(N^2)$ and it is not always applicable in real systems¹³. The principal advantage of this method is represented by independent signal processing and by the possibility to obtain the results of convolution "on-line" without referring to the following counts. The methods with such characteristics are sometimes prerequisite for some types of instrumentations units in information-measuring systems with active transducers. Such systems can include multifunctional IMS with AT, the systems with high spatial resolution and others, i.e., the systems where the results of optimal processing have to be provided with minimum delays, often even before receiving all signal counts. Therefore, modification of this method, aimed at narrowing down the amount of arithmetical operations, will be useful for realizing MF in IMS with AT of this type.

For MF of broadband signals with linear frequency modulation, it is suggested that the method of subapertures should be applied. In the methods of signal processing (filtering), aperture is understood as the number of counts of the impulse characteristic (order) of the filter, at which the filter is operating at some particular moment. In sub-aperture method, the aperture (impulse characteristic) of the matched filter for signals with linear frequency modulation (LFM) is equally split in linear components in the time domain (given the properties of LFM-signal, this operation is equivalent to splitting the band of the signal in *Q* equal intervals in frequency domain), and these components are called sub-apertures.

Filtering is performed by means of moving the window (aperture) of the filter consisting of Q sub-apertures along the counts of the signal. At each location of the aperture, the uniform set of operations is performed, that predetermines the so-called response of the filter (Figure 1). The counts of the signal are consequently fed to each scheme of the relevant sub-aperture starting with number i = Q - 1. Then, according to the impulse characteristic of the filter h(n), the phase of the signal shifts to the phase of sub-aperture e_i and comes to be accumulated in the N_a counting collector Ss_i . At the final stage, the values accumulated in the summation unit are totaled taking into account the initial phases of sub-apertures c_i .

One of the basic characteristics of the method is the number of sub-apertures Q of the filter that directly depends on the admissible phase error. At phase error $\frac{\pi}{4} \left(\pm \frac{\pi}{8}\right)$ the number of MF sub-apertures is determined by the formula as follows (2)¹⁴:

$$Q = \sqrt{N_h} \tag{2}$$

Where $N_{\rm h}$ is the length of the filter impulse characteristics.

The advantages of sub-aperture method are that the results of MF are obtained with the delay equal to the length of the impulse characteristic, and the number of arithmetical operations is $2N_a/7$ times lower than in the direct convolution method. Minimum interference



Figure 1. Flowchart for calculating MF applying subaperture method.

between signal counts improves the quality characteristics of the filter, especially when integer arithmetic is applied.

The suggested scheme, however, has a number of disadvantages; one of them is represented by the double line of delays that affects the amount of memory required for storing the data during this method implementation. Given the abovementioned disadvantage, to improve MF efficiency of LFM-signals, it has been suggested that the sub-aperture method should be modified according to the signal parameters. The number of sub-aperture counts $N_{\rm a}$ and the number of sub-apertures Q are selected in such a way that the phase of the signal ϕ within one sub-aperture is changed by the angle dividable by π . The dependency of the signal phase on the length of the sub-aperture is described by equation (3). It should be noted that phase error $E(\phi)$ does not depend on the selected point of approximation within sub-aperture (4), therefore, zero is selected as the initial value of the phase (Figure 2).

$$\varphi = \frac{\pi \Delta F t^2}{T} = \frac{\pi \Delta F T t^2}{T^2} = \frac{\pi}{N_a^2} \left(\frac{N_h t}{T}\right)^2 = \frac{\pi}{N_a^2} i^2 \qquad (3)$$

$$E(\varphi) = \pi (i^2 + i + 0.5) - \pi (0.5 + i)^2 = \frac{\pi}{4}$$
(4)

Where *i* is the number of sub-apertures, ΔF is the frequency band of the signal; *i* = 0,..., $N_{\rm h}$ – 1; *t* = 0,..., *T*.

Given the specified peculiar features of signal splitting in sub-apertures and of the selection of the point of approximation in the phase within the sub-aperture, relevant MF flow chart is proposed in Figure 3. In the scheme of matched filter algorithm, the impulse characteristic is split in eight sub-apertures with the parameters shown in Table 2. The line of delay *z* makes it possible, in the scheme of each sub-aperture, to store the preceding $N_a =$ 20 values of the signal counts. As the signal is multiplied cyclically by the factor that accounts for the sub-aperture phase, and the results of multiplication are not stored (there is no second line of delay by contrast to the scheme



Figure 2. Phase approximation within sub-aperture.



Figure 3. Modified flowchart for calculating MF applying sub-aperture method.

Table 2.	Parameters of sub-aperture method for
linear freq	uency-modulated signal in ORWELL-R
security ra	adar station

Parameter	Value
Bandwidth-duration product, B	64
Length of filter support function, N_h	160
Admissible phase error	$\pm \frac{\pi}{8}$
Sub-aperture number, $oldsymbol{Q}$	8
Number of sub-aperture counts, N_a	20
Signal duration, <i>T</i>	0.76 microsecond
LFM-signal deviation, ΔF	83.7 MHz
Sub-aperture deviation	10.5 MHz
Duration of received signal, <i>N</i>	1200

shown in Figure 1), and the phase within the sub-aperture is changed by π , in order to compensate the additional cyclical accumulation of phase at π , the signal counts are multiplied by factor $e^{-j\pi}$, which is equivalent to multiplying by "-1". Thus, taking into account the additional phase shift at the input of the sub-aperture processing system, it proved possible to eliminate extra line of delay in the filter; and the phase difference between sub-apertures is compensated through multiplication by $c_i = \pm 1$ at the output of each sub-aperture.

The modified sub-aperture method improves the speed of the matched filtering by 85% as compared to the method of direct convolution and by 17% as compared to MF based on FFT, thereat deteriorating the quality characteristics by just 0.2 dB.

Further decrease in the number of sub-apertures with simultaneous increase in phase error is not of much interest for realizing in real systems, as the levels of side-lobes increase drastically. Thus, for example, with the same initial data, in the systems with active transducer with half as much sub-apertures and admitting phase error π , the level of side-lobes will already amount to 11.6 dB, that is by 2.6 dB higher as compared to the direct convolution method.

To solve the problem of MF with minimum delays in phase code modulation (PCM), the method of coded moving total has been developed. In PCM-signal, the phase of the carrier wave $\phi(t)$ takes N_h pseudo-random values (5), and the equation of direct convolution (1) is transformed into the equation as follows (6):

$$s(t) = A\cos(2\pi f t + \varphi(t)) \tag{5}$$

$$y(n) = \sum_{i=0}^{I-1} \operatorname{sign}(h_i) \sum_{k=0}^{K_i - 1} x(n-k)$$
(6)

Where *I* is the number of areas in a pseudo-random array, where the supporting function of the filter maintains sign; K_i is the length of *i* area.

The flow chart for calculating MF applying the method of coded moving total is shown in Figure 4. The signal counts come in sequence to each *i* area of length K_i , starting with number i = I - 1. Then, taking into account the sign of the area $(-1)^i$, it is accumulated in collector Ss_i . At the final stage, all values of the areas are summed.

The advantages of the method of coded moving total are represented by the facts that the results of matched filtering are obtained with minimum delays, the number of arithmetical operations is $\sqrt{N_h}$ times lower than it is with the method of direct convolution. There is no multiplication, which improves the quality characteristics of the filter, especially when integer arithmetic is applied.



Figure 4. Flowchart for MF of PCM-signals applying the method of coded moving total.

4. Discussion

The suggested methods for realizing matched filtering: sub-aperture method for LFM-signal and the method of coded moving total for PCM-signal proved to be more efficient as compared to the method of direct convolution. These methods employ the operation of moving summation (totaling) that improves the efficiency of the methods by several times.

To evaluate the characteristics of the developed method of coded moving total and of the method of sub-apertures, a comparative analysis of MF methods for broadband signals has been carried out (Figure 5). For the purposes of evaluation, the methods ensuring complex MF with approximately similar level of precision have been selected. Along with the developed methods, the comparison included the method of direct convolution and the method of fast convolution based on FFT; FFT and RFFT, these are the algorithms of direct and reverse fast Fourier transform. Below, the calculations of the number of arithmetical operations are given for different MF methods taking into account the peculiar features of the architecture of the multi-core processors.

The number of operations for MF applying the method of direct convolution is as follows:

$$N_{MDC} = (\underbrace{4+4}_{\text{complex multiplication and adding}})N_h = 8N_h$$

The number of operations for MF based on FFT amounts to:

$$N_{FFT} = \frac{\underbrace{5*\log_{2}(2N_{h})*2N_{h}}_{FFT} + \underbrace{6*2N_{h}}_{complex multiplication} + \underbrace{5*\log_{2}(2N_{h})*2N_{h}}_{RFFT}}_{N_{h}} = 20\log_{2}(2N_{h}) + 12$$

The number of operations for MF applying sub-aperture method makes:



Figure 5. Comparative analysis of methods for MF radar signals.

$$N_{SA} = \underbrace{ \begin{array}{c} 2 \\ \underline{\text{adding Re/Im}} \\ moving total \end{array}}_{\text{moving total}} + \underbrace{ (\underbrace{4} \\ \underline{\text{multiplying}} \\ \underline{\text{adding}}) \sqrt{N_h} \\ \underline{\text{heterodyne+coherent addition of sub-appertures}} = 12\sqrt{N_h} \\ . \end{array}$$

The number of operations for MF applying the method of coded moving total amounts to:

$$N_{CMT} = \frac{2}{\text{Re/Im}} * 2\sqrt{N_h} = 4\sqrt{N_h}$$

According to Figure 5, the method of sub-apertures is efficient in terms of the number of arithmetical operations for MF of the signals, whose bandwidth duration product does not exceed $N_{\rm h} = 256$; the method of coded moving total is efficient for the signals with the bandwidth duration product not higher than $N_{\rm h} = 4096$.

Digital Signal Processors (DSP) are widely applied in both civil and military computing and controlling systems. Their areas of application include processing audio- and video signals, navigation, communications, hydro-acoustics, radio detection and ranging, digital television, remote Earth probing systems. Developing new generations of such systems calls for higher efficiency of signal processing, therefore, the investigations aimed at improving the functional characteristics of signal processors by means of improving their architecture are of current importance. By contrast to the previous stage of micro-processor technological development, when the productivity growth was usually achieved through improving the technology of production and by adopting ever shorter periods for designing integrated circuits, over recent years, marked by the transition from single-core to multi-core architecture, it is exactly the optimum architectural solutions that have been gaining ever growing significance.

Based on the undertaken investigations and earlier developments on matched filter methods, efficient in terms of precision and speed, it is suggested that the operations of moving summation should be included in the list of basic instructions of the processors or of the processor cores dedicated to digital signal processing. This could make one of the areas of improvements in the systems of instructions and in the process of adapting calculating units to filtering algorithms (not just matched filtering) that will increase the efficiency of the methods two times at the very least (depending on how the instructions are implemented).

EL-core-xx processor cores are aimed at realizing, on their bases, a wide range of applications for signal processing with considerably varying requirements in terms of efficiency, throughput capacity of data buses, storing capacity and other characteristics. Major principles for developing EL-core-xx DSP-cores are scalability, adjustability and the ensured capability of effective interaction with other cores in the multi-core system on a chip.

EL-core-xx processor cores are meant for building, on their bases, the computing and controlling systems meeting the wide range of requirements to efficiency, to the formats of the processed data, to memory capacity and to other characteristics. Such a wide spectrum of applications is ensured by reconfiguration capabilities integrated at different architectural levels of DSP-cores under study.

1) Reconfigurable Harvard architecture

In EL-core-xx DSP-cores, reconfigurable Harvard architecture has been realized enabling the redistribution of the data and of the total amount of memory between the programs.

2) Reconfigurable register file

In EL-core-xx DSP-cores, the operations can be performed with scalar or vector type data represented in formats of 16/32/64/128 bit with both fixed and floating point. The above mentioned capabilities are realized through the reconfigurable register file and through the reconfigurable data processing path. 128-digit register file used in EL-core-30M DSP-core possesses the widest range of capabilities.

3) Reconfigurable data processing path

VLIW-type system of instructions employed by EL-core-xx DSP-cores makes it possible to perform simultaneously up to two computing commands and up to two transfer instructions within one instruction. The scheme of the computing path in EL-core-xx IP-cores is built in such a way that it allows introducing new instructions and functional blocks that considerably improve the efficiency of DSP algorithms.

The algorithm of moving summation is described mathematically by the equation as follows (5):

$$y(n) = y(n-1) - x(n-k) + x(n)$$
(5)

where N = 0,..., N - 1, y(n < 0) = 0, x(n < 0) = 0, k is the number of the elements of vector x, along which the summation is performed; x(n) is the input vector; y(n) is the output vector.

In Multicore (MCF-0428, NVcom-01) processors two operations of moving summation have been introduced into the basic system of instructions of EL-core IP-cores¹⁵.

- RA8 instruction making it possible to perform 8 integer moving totals in 16-digit arithmetic;
- RA4 instruction making it possible to perform 4 integer moving totals in 16-digit arithmetic.

The developed instructions enable narrowing down the number of arithmetic operations 16 times (RA8) and 8 times (RA4) applying the method of coded moving total. The moving summation instructions can reduce the number of arithmetic operations by several times in such algorithms as median filtering, in different frequency filters and in smoothing algorithms.

5. Conclusion

Based on the undertaken investigations, the quantity of arithmetical operations required for realizing matched filtering with the existing algorithms has been evaluated. It has been demonstrated that to solve a problem of obtaining "on-line" signal in MF with minimum delays, the existing solutions are not always applicable to instrumentation complexes of information-measuring systems with active transducer.

The study suggests two methods of matched filtering for broadband signals with minimum delays: the method of coded moving total (for PCM-signal) and the method of sub-apertures (for LFM-signal). The sub-aperture method is efficient in terms of the number of arithmetical operations for MF of the signal, whose bandwidth duration product does not exceed $N_{\rm h} = 256$, while the method of coded moving total is efficient for the signal with the bandwidth duration product not higher than $N_{\rm h} = 4096$. Delays are only predetermined by the length of the supporting function of the filter, and this fact improves the precision of MF calculations in integer arithmetic.

Evaluating the efficiency of the schemes and algorithms for matched filtering showed that in a number of problems of MF of broadband signals, the method of coded moving total proved 3-20 times more effective than the method of direct convolution, and two times more effective than the method of fast convolution based on FFT algorithms; the methods of sub-apertures proved 2-6 times more effective than the direct convolution method (depending on the number of signal counts). The above mentioned approaches to reducing the number of arithmetical operations can be applied to other filtering algorithms.

Efficient structure of reconfigurable multi-core computing and controlling systems-on-chips based

on EL-core-xx DSP-cores is ensured by the complex solutions including the following: reconfigurable Harvard architecture, reconfigurable register file and data processing path, capability to reconfigure the flows of data and to control them at inter-core level. Such architectural solutions make it possible to introduce new instructions and functional blocks that considerably improve the efficiency of DSP algorithms.

Making use of these reconfigurable possibilities of EL-core-xx DSP-cores helps reducing hardware and energy losses, improving reliability of the newly created computing and controlling systems. Based on the results of the undertaken investigations, two hardware moving summation instructions have been developed for EL-core IP-cores, that improve their effective speed in solving the tasks of filtering by several times. The developed algorithms are used in modern radar station ORWELL-R (OJSC R&D Center ELVEES).

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