Bull. Nov. Comp. Center, Math. Model. in Geoph., 18 (2015), 69–80
© 2015 NCC Publisher

On statistical adaptation of the order filters for the signal form and its noise specificity

Vladimir Znak

Abstract. This paper is aimed to the improvement of the quality of the data of vibro-seismic research under the condition of the wave form preservation of a sounding signal. For this purpose, it is offered to use the order statistics filters. As seismic data are mostly harmonic or frequencymodulated (or sweep) signals in the frequency range 1 to 10 Hz, these filters must be adapted so as to suppress noise but not unduly distort a signal because such filters possess specific features. To this end, it is reasonable to attract statistical adaptation of such filters for the form of a signal and specificity of noise To illustrate the proposed methodology of order statistics filters adaptation, models of a sweep signal from the frequency band of 72–82 Hz were processed and the quality of the signal enhanced considerably over the quality of the signal without processing.

Keywords: weighted order statistics filters, periodic signals processing, numerical study

Introduction

Generally, vibro-seismic data are mostly records of harmonic or frequencymodulated (or sweep) signals under the condition of some digitization frequency. Such records [1] are commonly a narrow band within 1 to 10 Hz range. Most often, methods of such signals analysis and their processing are in the field of harmonic analysis. In the given paper, the weighted order statistics (WOS) filters are offered for the preprocessing or the processing of frequency-modulated (FM) and harmonic signals corrupted by noise and for their analysis.

The problem of noise signals processing with the WOS filters is not new, since these filters possess a number of advantages in comparison with other filters [2], namely: 1) a remarkable ability of the impulse noise removal, 2) noise robustness, 3) preservation of steps for a signal in the form of a telegraphic sequence (with such specific features as the first derivative discontinuity). Duncan and Beresford [3] studied the behavior of the median filter diagonally applied across seismic traces for producing lateral smoothing and found that the median filters preserved steps as compared to the low-pass and the linear filters. At the same time, according to [2], the length of a median filter should surpass the length of a linear filter in one and half times for the equal decrease of signal noise. On the other hand, such filters are nonlinear ones and their response tends to zero while the filter length (n) approaches an integer number of signal periods. This circumstance demands a special attention when processing periodic signals and solving the problem of the corresponding adaptation of the WOS filters.

Among the papers dealing with the solution of the above problem, we will note some of them [4–6], as they exploit different approaches. In [5], the problem is solved by making use of the hybrid (the WOS and the linear finite impulse response — FIR) filters. In this case, the corresponding task is formulated as a demand "... that the filter should be able to discard those samples in the window which are not in phase with the center of the window" [5, p. 1621]. The approach of [4] admits negative weights in the median filter performance. At the same time, the approach of [5], as [4], requires a preliminary training (learning) of filters under the mean absolute error criterion. Finally, in [6], zero weights are added to an appropriate set.

However, owing to nonlinearity, the analytical estimation of their behavior is a very complicated process. The behavior of the WOS filters appreciably depends, on the one hand, on the filter project (values of scales, the size of the aperture or a window of the analysis, a sequence of operations in the multistage process of filtration) and, on the other hand, it depends on the form of a signal and specificity of noise. So, the results of the research [3] show the importance of the filter design for attaining high attenuation levels of noise without causing a considerable signal distortion. At the same time, the analytical estimation of the quality of signal processing by the WOS filters is very complicated owing to nonlinearity of such filters. Thus, it can be supposed that the response of the WOS filters is a casual event in the general case. Under these conditions, it seems appropriate to use a numerical method for solving mathematical problems by means of modeling random variables, i.e., a method of statistical trials to select the most efficient project of the WOS filter.

In this paper, the task of interest is the processing of periodic signals and monitoring of the quality of the restored signals depending on using different projects of filters and their parameters. The basic points of this approach were discussed at the EAGE Conference [7]. At the same time, a specialized computer system [8] was utilized in the course of the above trials.

1. The basic definitions of the WOS filters

Before considering the research into selecting projects of the WOS filters, it is worthwhile to give the basic definitions of such objects.

Let a periodic signal be one-dimensional time series $X = \{x_1, \dots, x_N\}$ recorded at discrete instants of time $t_1, \dots, t_N$, $t_i - t_{i-1} = \Delta t = \text{const}$, $i = 2, \dots, N$.

Let a sequence

$$Y = \{y_i : \ i = 1, \ldots, n\} \tag{1}$$

present some samples of a signal which includes n quantities of numerical data.

The term "order filters" comes from the notion "a variational row" of mathematical statistics, where numerical values of a data row are arranged in increasing (or decreasing) order. Here, the rth-order statistics $\tilde{y}_r$ is defined as the rth quantity in size, $r = 1, \ldots, n$. Generally, it is possible to set the locus of any term $\tilde{y}_r$ of a variational row by means of a relative number α of preceding elements (smaller or equal in size) and a relative number β of subsequent ones (equal to or greater than it): $\alpha = (r-1)/(n-1)$, $\beta = 1 - \alpha$. In this case, one can speak about the percentile form of the filter. It is denoted as $\tilde{y}_{\alpha,n}$ if there exists the number $n_\alpha = (n-1)\alpha$ of $\breve{y}_i$ quantities and the number $n_\beta = (n-1)\beta$ of $\widehat{y}_i$ quantities provided

$$\breve{y}_i \le \tilde{y}_{\alpha,n}, \quad \widehat{y}_i \ge \tilde{y}_{\alpha,n}, \quad \tilde{y}_{\alpha,n} \bigcup_{i=1}^{n_\alpha} \breve{y}_i \bigcup_{i=1}^{n_\beta} \widehat{y}_i = Y.$$

Now, the formal definition of the order filtration procedure can be presented as a sequence of the following operations:

i) $Y \subset X$ is a sampling of $n = 2v + 1$ signal values ($v \in Z_+$, n is odd).

ii) $\tilde{Y} = \{\tilde{y}_1 \le \ldots \le \tilde{y}_{(n-1)/2} \le \ldots \le \tilde{y}_n\}$ is the sampling Y, whose value are ordered in the increasing order.

iii) $\mathrm{Rank}_\alpha(y_1, \ldots, y_{(n-1)/2}, \ldots, y_n)$ maps the central value $y_{(n-1)/2} \in Y$ to $\tilde{y}_{\alpha,n} = \tilde{y}_r \in \tilde{Y}$, $r = (n-1)\alpha$.

The last formula is the definition of the standard order statistic filter.

In the special case of $\alpha = 0.5$, the filter Rank_α is the median one $\mathrm{Med}(y_1, \ldots, y_{(n-1)/2}, \ldots, y_n) = \tilde{y}_{(n-1)/2}$.

Let we have a set W of integer quantities w_i ($i = 1, \ldots, n$), where every quantity w_i is associated with the corresponding sample $y_i \in Y$. This w_i is called a weight and can be treated as the number of copies of the sample y_i. For example, [9] defines the weighted median value of a sequence of numbers as a simple median of the extended sequence formed by repeating each term w_i times. At the same time, weights are usually set symmetric with respect to the central element (CE) of sequence (1). Weights are introduced for emphasizing some elements of a sequence [2, 10, 11]. The extended sequence of samples y_i thereby gains a new quality as a set with the number of elements $N(W) = N(w_1, \ldots, w_n) = \sum_{i=1}^{n} w_i$. At the same time, $N(W)$ is also odd. In this case, the formula

$$\tilde{y}_{\alpha,N(W)} = \mathrm{Rank}_{\alpha,W}(y_1, \dots, y_{(n-1)/2}, \dots, y_n) \qquad (2)$$

is the definition of the WOS filter in the general case. This generalization allows a filter to keep properties of the median one [12], but changes the result of the filtering, i.e., $\tilde{y}_{0.5,n} \neq \tilde{y}_{0.5,N(W)}$ [11].

In the general case, a filter as the whole can include some ordered set (a sequence) of separate filters of signal processing. Let such singled filter be called a filter node. At the same time, such node includes some "filter terms" sequence of length n.

Passing to the question of processing the periodic signals by the WOS filters, we can equalize to zero the weights of all the terms of the filter input sequence except the weight of the central element (CE) and the weights of such terms which are apart from one another by the length of a period (the idea formulated in [6]). The use of zero weights is equivalent to decimation (sieving). An analogous technique is widely exploited in the field of images processing. Thus, the offered approach demands the knowledge of the frequency band of a signal under processing only unlike approaches proposed by [4, 5]. This allows us to project the so-called co-phased WOS

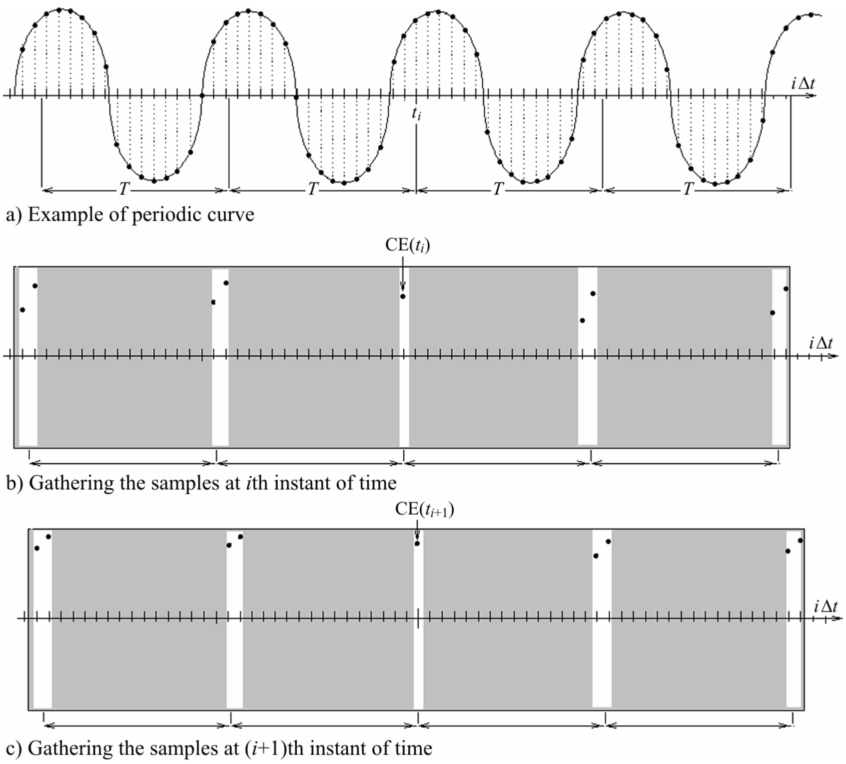

Figure 1. Sampling of periodic signal values closest to the CE phase at different instants of time

(CoPhWOS) filters. At the same time, an important condition of such a filter is providing the filter length equalizing to m periods of a signal, where m is even ($m = 2, 4$, etc.). The above informal definition of such filters is illustrated in Figure 1.

According to the concept of weights, the corresponding set W of the CoPhWOS filter includes a subset $W^* = \{w^* \in W :\ w^* = 0\}$.

Taking into account the symmetry of CoPhWOS filters, we re-index elements of the sampling Y as $\{y_{-v}, \dots, y_0, \dots, y_v\}$, so that the central element has zero index. The procedure for calculating the weights of the CoPhWOS filter is as follows [6]:

1. Compute the length of the filter as $n = 2v + 1$, $v = \lceil RT/\Delta t\rceil$;
2. Initialize $w_0 := 1$, $w_{\pm i} := 0$, $i = 1, \dots, v$;
3. In cycle by $j = 1, \dots, R$, update weights according to the rules:

$$\begin{aligned} &V_j := jT/\Delta t;\\ &\text{if } V_j = \lfloor V_j\rfloor = \lceil V_j\rceil,\ \text{then } w_{\pm V_j} := 1;\ \text{otherwise}\\ &w_{\pm\lfloor V_j\rfloor} := \lceil V_j\rceil - V_j,\ w_{\pm\lceil V_j\rceil} := V_j - \lfloor V_j\rfloor. \end{aligned} \tag{3}$$

Here T is the period of a corresponding frequency and $w_0 = 1$ is the weight of the CE. The frequency which defines values of the filter weights will be called the working frequency of the filter.

According to values of nonzero terms assigned in procedure (3), we introduce functions

$$\phi_0(y) = y_0, \quad \phi_{\pm j}(y) = \begin{cases} w_{\pm V_j}\, y_{\pm V_j} & \text{if } V_j = \lfloor V_j\rfloor = \lceil V_j\rceil,\\ w_{\pm\lfloor V_j\rfloor}\, y_{\pm\lfloor V_j\rfloor} + w_{\pm\lceil V_j\rceil}\, y_{\pm\lceil V_j\rceil} & \text{otherwise.}\end{cases}$$

As defined in procedure (3), the index $j = 1, \dots, r$ of the corresponding function is nothing but the periods quantity between the CE and this filter term under the condition of symmetry of the node terms concerning the CE.

Let us use the notation CoPhWOS^f_R to designate such a filter, where R is a radius of the filter and f is the working frequency of the filter. Now, the CoPhWOS^f_R filter of radius R is

$$\tilde{y}^f_{\alpha,N(k_0\phi_0,k_1\phi_{\pm1},\dots,k_R\phi_{\pm R})} = \mathrm{Rank}_{\alpha,K}(\phi_0(y), \phi_{\pm1}(y), \dots, \phi_{\pm R}(y)), \tag{4}$$

where $K = \{k_0, \dots, k_R\}$ is a sequence of weights and $k_0 \neq 0$.

Similarly, the definition of the standard WOS filter of radius R will be used in the notation $\tilde{y}_{\alpha,N(w_0,w_{\pm1},\dots,w_{\pm R})}$. Here some terms can also include zero weights, however values of the corresponding filter data do not depend on a signal frequency.

The above algorithm is readily adapted to the case where the weight of the central element exceeds the weights of other components not less than by unit – for emphasizing its value. The algorithm in this case ensures the symmetry of weights as well. However, here, the weight w_0 of CE should be odd. The non-integer weights can be transformed to integers with some loss of accuracy, by introduction an appropriate factor. Schemes for treatment with non-integer weights were offered by [13, 14].

2. The basic procedures of the statistical trials method of the WOS filters projects

It is possible to note the following parameters of the WOS and CoPhWOS filters which influence the quality of signal processing:

- values of the weights $W = \{w_1, w_2, \ldots\}$ of the WOS filters as well as the distribution of zero weights which are the functions of frequencies in the case of the CoPhWOS filters;
- the size of the aperture or the window of analysis;
- a sequence of operations in the multistage filtration process;
- the operations type;
- percentiles of the WOS and the CoPhWOS filters.

The CoPhWOS filter satisfies the requirement of discarding those samples in the window which are not in phase with the center of the window [5] if a signal is a harmonic one on the whole length of its existence. However, in the course of the FM signal processing, the above condition is not satisfied. According to [6], such a filter of the two-period length of a FM signal will save the width $\Delta f(\text{CoPhWOS}) \in 1.0 \div 1.5$ Hz in its frequency band. That is, if $\Delta f(\text{FM}) > \Delta f(\text{CoPhWOS})$, it is needed to attract a special technique for restoration of a source signal in the whole frequency band. Such a case was studied in [15], however, it is not considered here.

The algorithm of the corresponding signals processing providing a variation of the above data and parameters in the course of signal processing is presented in Figure 2.

The basic denotations of the above algorithm are the following:

- $\text{filter}_i = \{\text{node}_{k(i)} :\ k(i) = 0, \ldots, K(i) - 1\}$ is a sequence of data of consequent steps of the ith filter of the corresponding set;
- $\{\text{filter}_i :\ i = 1, \ldots, I\}$ is a filter set,
- filter(beg)/filter(end) is the first/last filters used from a filters bank;
- a frequencies set is $\{f_l :\ l = 1, \ldots, L\}$;
- $f(\text{beg})/f(\text{end})$ is the first/last using work frequency;

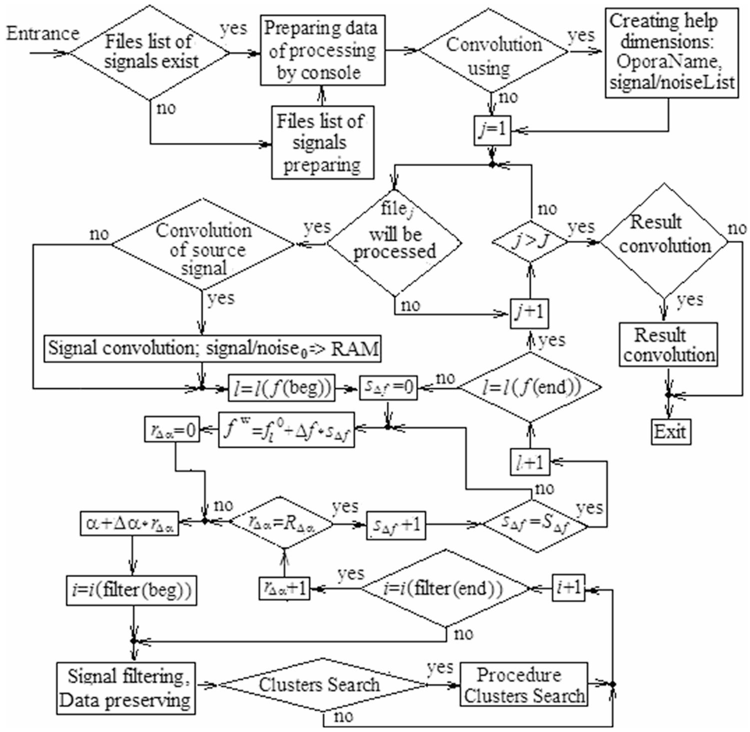

Figure 2. The algorithm of monitoring the filters projects

- Δf is the step size of frequency updating;
- $S_{\Delta f}$ is the number of steps of frequency updating, $s = 0, \ldots, S_{\Delta f} - 1$;
- f^w is the current working frequency;
- $\Delta\alpha$ is the step size of procentile updating;
- $R_{\Delta\alpha}$ is the number of steps of procentile updating, $r = 0, \ldots, R_{\Delta\alpha} - 1$.

Other notations are used in the course of the cluster analysis [15] and will not be discussed here.

In the general case, a filter includes a sequence of nodes of the whole processing a signal. Some filter nodes can include a coupled action of the same type, but with different data. In this case, data for the second filter are attracted and the results of both actions are composed.

A specialized interactive computer system based on the graphical interface was utilized for execution of the above algorithm [8]. Within the frame of the above interface, $\boldsymbol{N}_1, \boldsymbol{N}_2, \boldsymbol{N}_3 \in \{0, 1, \ldots, 101\}$ denote working files numbers. Since in the general case, the process of the signal processing as a whole can include some sequence of singled steps, each step of action which is connected with a corresponding filter node, can be presented by the scheme:

processing data of a working file $\boldsymbol{N}_1$ with consequent transfer of results to a working file $\boldsymbol{N}_3$ in the course of data processing. Thus, the traffic of data in the course of signal filtering can be presented by an oriented graph with a single source node and a single finish node. The processing operations sequence and corresponding nodes data are defined using a computer console in the interactive mode [8].

The new working files are created in the course of running the filtering procedure. The working file of the source node of each filter is associated with the number "0". At the same time, the working file of the last filter node is associated with the number "101" that designates termination of signal processing.

The list of available filter operations is listed in Table 1. Here, the symbol $\oplus$ denotes the operation of data superposition.

Thus, the user can employ both nonlinear and linear operations. The efficiency of such an approach was demonstrated in [5].

Table 1. The list of available filter operations

Operation name	Operation content
Transfer	data $\in \boldsymbol{N}_1 \Rightarrow \boldsymbol{N}_3$
Data composition	(data $\in \boldsymbol{N}_1 \oplus$ data $\in \boldsymbol{N}_2) \Rightarrow \boldsymbol{N}_3$
The CoPhWOS filter	$\tilde{y}^f_{\alpha,N(k_0\phi_0,k_1\phi_{\pm1},\dots k_R\phi_{\pm R})}$
The coupled CoPhWOS filter	$\tilde{y}^f_{\alpha,N(k_0^{(1)}\phi_0,k_1^{(1)}\phi_{\pm1},\dots k_{R_2}^{(1)}\phi_{\pm R_1})} \oplus$ $\tilde{y}^f_{1-\alpha,N(k_0^{(2)}\phi_0,k_1^{(2)}\phi_{\pm1},\dots k_{R_2}^{(2)}\phi_{\pm R_2})}$
The co-phased average	$\Big(\sum\limits_{j=-R}^{R} k_{\lvert j\rvert}\phi_j(y)\Big) \Big/ \Big(\sum\limits_{j=-R}^{R} k_{\lvert j\rvert}\Big)$
The coupled co-phased average	$\Big(\sum\limits_{j=-R_1}^{R_1} k^{(1)}_{\lvert j\rvert}\phi_j(y)\Big) \Big/ \Big(\sum\limits_{j=-R_1}^{R_1} k^{(1)}_{\lvert j\rvert}\Big) \oplus$ $\Big(\sum\limits_{j=-R_2}^{R_2} k^{(2)}_{\lvert j\rvert}\phi_j(y)\Big) \Big/ \Big(\sum\limits_{j=-R_2}^{R_2} k^{(2)}_{\lvert j\rvert}\Big)$
The standard WOS filter	$\tilde{y}_{\alpha,N(w_0,w_{\pm1},\dots,w_{\pm R})}$
The coupled standard WOS filter	$\tilde{y}^{(1)}_{\alpha_1,N(w_0^{(1)},w_{\pm1}^{(1)},\dots,w_{\pm R_1}^{(1)})} \oplus \tilde{y}^{(2)}_{\alpha_2,N(w_0^{(2)},w_{\pm1}^{(2)},\dots,w_{\pm R_2}^{(2)})}$
The standard average	$\Big(\sum\limits_{j=-R}^{R} w_j y_j\Big) \Big/ \Big(\sum\limits_{j=-R}^{R} w_j\Big)$
The Coupled standard average	$\Big(\sum\limits_{j=-R_1}^{R_1} w_j^{(1)} y_j\Big) \Big/ \Big(\sum\limits_{j=-R_1}^{R_1} w_j^{(1)}\Big) \oplus$ $\Big(\sum\limits_{j=-R_2}^{R_2} w_j^{(2)} y_j\Big) \Big/ \Big(\sum\limits_{j=-R_2}^{R_2} w_j^{(2)}\Big)$

3. Some results of signal processing

The methodology in question was investigated on the models of linear frequency-modulated signals using numerical modeling. Parameters and characteristics of the corresponding signals are the following: the start time of a sounding signal is 0 s; the arrival time of the sounding signal in a noise signal is 4 s (i.e., the sounding signal is recorded when removing from a vibration source). At the same time, the bandwidth of the sounding signal is $7.2 \div 8.2$ Hz, the digitization frequency being $\Delta t = 0.08$ s as for a sounding signal and for a noised signal. The white noise with zero average Gaussian distribution was used for obtaining a noise signal model.

The results of the experiments are shown in Table 2, where $\hat{s}/\xi_0$ is the signal-to-noise ratio obtained by means of convolution of a noise signal with a sounding signal. At the same time, s/n is the estimation of the ratio of the mean square deviation of the sounding signal and the model of noise signal data before the processing.

Table 2. Results of convolution of a filtered noise signal with a sounding signal

$\hat{s}/\xi_0 = 99.4$, $s/n = 0.2$		$\hat{s}/\xi_0 = 87.6$, $s/n = 0.1$		$\hat{s}/\xi_0 = 10.1$, $s/n = 0.01$	
f, Hz	$\hat{s}/\xi$	f, Hz	$\hat{s}/\xi$	f, Hz	$\hat{s}/\xi$
7.750	125.6	7.70	93.0	7.65	14.4
7.765	127.0	7.71	96.8	7.67	15.9
7.770	147.5	7.72	94.1	7.68	10.2
7.775	137.8	7.73	90.4	7.69	10.1
7.785	145.7	7.74	91.0		

The results shown in Table 2 are obtained by means of the filter structure presented in Table 3. The filter structure used in others experiments is shown in Table 4. The research results for this case are shown in Table 5 for $\hat{s}/\xi = 13.374$, $s/n = 0.066$.

Table 3

$\boldsymbol{N}_1$	$\boldsymbol{N}_2$	Operation	$\boldsymbol{N}_3$
		Signal $\Rightarrow$ Working file 0	
0		$3\phi_0(y) + 2\phi_{\pm1}(y) + 1\phi_{\pm2}(y))/5$	1
1		$\tilde{y}^f_{0.5,9(3\phi_0,2\phi_{\pm1},1\phi_{\pm2})}$	2
1		$\tilde{y}^f_{0.5,7(1\phi_0,1\phi_{\pm1},1\phi_{\pm2},1\phi_{\pm3})}$	3
2	3	$\oplus$	4
0		$\tilde{y}^f_{0.75,9(3\phi_0,2\phi_{\pm1},1\phi_{\pm2})} \oplus \tilde{y}^f_{0.25,9(3\phi_0,2\phi_{\pm1},1\phi_{\pm2})}$	5
0		$\tilde{y}^f_{0.7,9(3\phi_0,2\phi_{\pm1},1\phi_{\pm2})} \oplus \tilde{y}^f_{0.3,9(3\phi_0,0\phi_{\pm1},2\phi_{\pm2},1\phi_{\pm3})}$	6
5	6	$\oplus$	7
4	7	$\oplus$	8
8		$\tilde{y}^f_{0.5,121(11\phi_0,10\phi_{\pm1},9\phi_{\pm2},\dots,1\phi_{\pm10})}$	9
8		$\tilde{y}^f_{0.7,9(3\phi_0,2\phi_{\pm1},1\phi_{\pm2})} \oplus \tilde{y}^f_{0.3,9(3\phi_0,0\phi_{\pm1},2\phi_{\pm2},1\phi_{\pm3})}$	10
9	10	$\oplus$	11
8	11	$\oplus$	101
		Work file 101 $\Rightarrow$ Preservation file	

Table 4

N_1	N_2	Operation	N_3
		Signal $\Rightarrow$ Working file 0	
0		$\tilde{y}_{0.5,9(3_0,2_{\pm1},1_{\pm2})}$	1
0		$\tilde{y}^f_{0.75,9(3\phi_0,2\phi_{\pm1},0\phi_{\pm2},1\phi_{\pm3})} \oplus \tilde{y}^f_{0.25,9(3\phi_0,0\phi_{\pm1},2\phi_{\pm2},1\phi_{\pm3})}$	2
1	2	$\oplus$	101
		Work file 101 $\Rightarrow$ Preservation file	

Table 5. Results of convolution of a filtered noise signal with a sounding signal

Working frequency, Hz	7.975	8.000	8.025	8.050	8.075	8.100	8.125
Signal-to-noise ratio ($\hat{s}/\xi$)	11.402	12.855	12.463	14.281	14.493	13.513	12.418

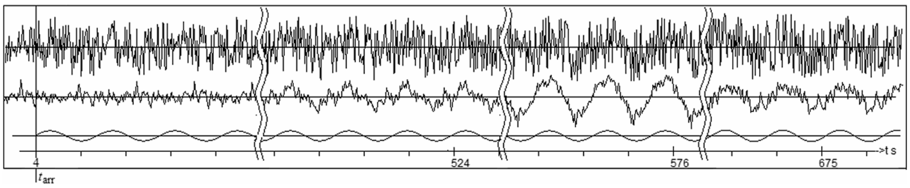

Figure 3. The model of a noised signal, the results of processing, and the sounding signal (from top to bottom)

The results of processing the model of a noise signal are shown in Figure 3, where a noise signal is the result of composition of a sounding signal and additive white noise with the value of $s/n = 0.2$. At the same time, the period of the signals recording is 1100 s and the arrival time of a sounding signal in a noised signal is $t_{\text{arr}} = 4$ s. Figure 3 allows us to note the dynamics of preserving the waveform of a FM signal by the co-phased filters as a function of time.

4. Conclusion and discussion

In this paper, the approach to the statistical adaptation of the WOS filters for processing frequency-modulated signals is proposed. The basic points of this approach are attracting a filters bank and varying filters data and processing parameters. The corresponding algorithm of filtering the FM signals under the conditions of statistical trials of the WOS filters projects was developed. The results of the experiments conducted demonstrate the dynamics of a considerable dependence of the signals quality processing from the value of the work frequency of the CoPhWOS filter and its structure including values of the corresponding data. It can be supposed that the results obtained demonstrate the possibility of the proposed approach to be rather effective even in the case of $s/n \ll 1$.

In the considered methodology, selecting the qualitative projects of the WOS filters is assigned to the user. The use of such methods as dynamic programming, sequential approaches, games theory or the maximum likelihood are of some interest with respect to selecting appropriate results. In the case of attracting any of the above methods, there can be used two different ways of research: 1) selecting an appropriate filter from the existing filters bank with different data of signal processing, 2) consequent construction of some filter by means of the gradual complication of the filter structure. In the both ways of research, it is rather desirable to provide computer construction of different WOS filters. However, meanwhile any proposals in this field are absent because the problem becomes complicated owing to nonlinearity of the WOS filters. This circumstance is valid for any selecting strategy of the qualitative projects of the WOS filters. To all the other above-mentioned notations, procedures of the statistical trials as well as the procedures of signal processing by the WOS filters demand high computer costs, i.e., the corresponding realization of the approach proposed demands the use of graphical processors or supercomputers. The importance of an increase in computation efficiency is also important for the improvement of the quality of the noise FM signals. Really, the width of a frequency zone, in which the waveform of a periodic signal remains the dependence on the length of the CoPhWOS filter. The quality of a periodic signal is preserved in the frequency band of the width $1.0 \div 1.5$ Hz if a length of CoPhWOS filter does not exceed two periods of a FM signal [6]. That is, increasing the filter nodes lengths will decrease a frequency zone, where the sounding signal quality is preserved as demonstrated by the above results of the signal processing. Hence, it is required to increase the quantity of the frequency zones which are subjects of filtration for the qualitative processing of a signal. The last circumstance, in turn, will entail an increase of the computer time expenses.

Acknowledgements. The author wishes to express his gratitude to the PACE EAGE Fund for the award PACE Delegate Support for attending “The 6th Saint Petersburg International Conference & Exhibition—Geosciences—Investing in the Future”, which allowed the author to discuss the matter of research with a wide range of colleagues.

References

[1] Nikolaev A.V. A capability of vibrosounding of the Earth // Proc. Acad. Sci., USSR. Series: Physics of the Earth. — 1975. — No. 4. — P. 10–21.

[2] Justusson B.I. Median filtering: statistical properties // Two-Dimensional Digital Signal Processing II: Transforms and Median Filters / T.S. Huang, ed. — Berlin: Springer, 1981. — P. 161–196.

[3] Duncan G., Beresford G. Median filter behaviour with seismic data // Geophysical Prospecting. — 1995. — Vol. 43. — P. 329–345.

[4] Arce G.R. A general weighted median filters structure admitting negative weights // IEEE Transactions on signal processing. — 1998. — Vol. 46, No. 12. — P. 3195–3205.

[5] Yin L., Neuvo Y. Fast adaptation and performance characteristics of FIR-WOS hybrid filters // IEEE Transactions on signal processing. — 1994. — Vol. 42, No. 7. — P. 1610–1628.

[6] Znak V.I. Co-phased median filters, some peculiarities of sweep signal processing // Mathematical Geology. — 2005. — Vol. 37, No. 2. — P. 207–221.

[7] Znak V.I. Some questions of the adapting the order filters to the signal form and character of noise // Extended Abstracts "6th S.-Petersburg Int. Conf. & Exhibition — Geosciences — Investing in the Future". — 2014. — DOI:10.3997/2214-4609.20140191. — http://www.earthdoc.org.

[8] Znak V. On statistical adaptation of the order filters for processing periodic signals using graphical interface // Proc. "11th International Conference on Pattern Recognitions and Image Analysis. New Information Technologies (PRIA-11-2013)". — Samara, 2013. — Vol. II. — P. 483–486.

[9] Claerbout J.F., Moir F. Robust modeling with erratic data // Geophysics. — 1973. — Vol. 38. — P. 826–844.

[10] Ko S.-J., Lee Y.H. Center weighted median filters and their applications to image enhancement // IEEE Trans. Circuits, Syst. — 1991. — Vol. 38. — P. 984–993.

[11] Znak, V.I. Weighted median filter and VSP result processing // Problemno-orientirovannye vychislitel'nye kompleksy / Sib. Branch of Russ. Acad. of Sci. — Novosibirsk: Computing Center, 1992. — P. 78–85 (In Russian).

[12] Znak V.I. Some models of noise signals and heuristic search for weighted-order statistics // Math. Comput. Modeling. — 1993. — Vol. 18, No. 7. — P. 1–7.

[13] Znak V.I. Synphase-weighted median filter: evaluation of weights and processing of a harmonic signal // Sib. J. Numer. Mathematics / Sib. Branch of Russ. Acad. of Sci. — Novosibirsk, 2001. — Vol. 4, No. 1. — P. 31–40 (In Russian).

[14] Znak V.I. Percentile filtering: some generalizations; the experience of geophysical data processing // Bull. Novosibirsk Computing Center. Series: Math. Model. in Geoph. — Novosibirsk, 1998. — Iss. 1. — P. 111–116.

[15] Znak V. Something about processing, analysis and restoration of periodic signals // Proc. "Ninth IASTED International Conference on Signal Processing, Pattern Recognition and Applications (SPPRA 2012)". — 2012. — P. 778–781.