Sidelobe Suppression of LFM Signal Using Variability Index (VI)-Wavelet Threshold Method

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Abstract In the pulse compression radar using linear frequency modulation (LFM), the sidelobe suppression signal processing is very important to overcome an effect that may mask smaller targets or maybe mistaken as separate targets. This paper considers a novel framework of the sidelobe suppression for linear frequency modulation signal (LFM) at a receiver when existing internal receive noise. In pulse compression for LFM signal, there exist not only range mainlobe components but also range sidelobe components in the matched filter output. It is necessary to effectively suppress the sidelobe components at the receiver of radar.

This paper presents variability index (VI)-threshold operation based on the wavelet transform to further reduce these range sidelobes, and then analyzes it compared to the conventional window method. The results show that the proposed method is very effective in suppression of sidelobes components for LFM signal. It also indicates that this method gives better a peak to sidelobe ratio (PSR) performance than other methods for the sidelobe suppression.

Keywords LFM, pulse compression, sidelobe suppression, Wavelet transform, variability index

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1 Introduction

In modern radars pulse compression is being widely used to achieve long-range target detection and a high range resolution simultaneously [1, 2]. Pulse compression technique is used to enhance radar performance in terms of more efficient use of high-power transmitters and increasing the system resolving capability, and the LFM signal is widely used in radar because it can be generated easily. A standard approach in pulse compression is to correlate the received signal with a reference waveform, i.e. a matched filter (MF). The MF output of LFM signal can be approximated as $\sin(x)/x$ or sinc(x), shaped autocorrelation function (ACF). But the pulse compression of LFM signal by MF produces sidelobes which is objectionable in a way that they may mask smaller targets in vicinity [1-8]. These sidelobes are undesirable. Range sidelobes are inherent part of the pulse compression mechanism and these are occurring due to abrupt rise in the signal spectrum of rectangular pulse. The conventional method is used to suppress these ambiguous sidelobes by modelling the rectangular shape of the chirp spectrum using amplitude weighing and it can be suppressed to the required level by using a suitable window function [1-9]. In radar systems, weighing technique in the time or frequency domain is mostly used to reduce these range sidelobes with broadening and a loss in the mainlobe. Time domain weighing is preferred to frequency domain weighing, because it produces lower sidelobe output [2]. Radar pulse compression effect using different operations for the sidelobe suppression is given as results in terms of a peak to sidelobe ratio (PSR). Various methods for the sidelobe suppression have been proposed in the literatures [1-5, 9, 12-14, 16].

Typically, to reduce these sidelobes convolutional windows have been applied as weighted function for radar pulse compression which are more sensitive to Doppler shift as compared to conventional windows [2]. To reduce the sidelobes for LFM waveform, there is an interesting approach related to the design of efficient NLFM waveforms namely, a temporal predistortioning method of LFM signals by suitable nonlinear frequency laws [3, 15]. By this method an average sidelobe reduction of 6 dB is also obtained when no internal receiver noise.

Also, adaptive approach was proposed as a new technique to adaptively shape the spectral response of target returns, thereby controlling their range sidelobes adaptively. This method had proposed to reduce processing loss of pulse compression methods that adopt spectral shaping to achieve range sidelobe suppression [4]. A sidelobe suppression based on Sparse Regularization was used for a reconstruction method for SAR imaging [5].

It compared with traditional MF method, and the convex optimization method outperforms in high resolution and sidelobe suppression and greatly improves the imaging quality. The pre-condition is that the signal is sparse in some domain, including time, frequency, space etc.

Preceding Literatures lack neither a mathematical model of the sidelobe suppression filter response nor a simulation model for the effect of noise on the PSR performance of LFM radars, when existing internal receiver noise environment, Generally, the pulse compression and sidelobe suppression in LFM radar are the processing for the receiving signals in the state which is existed internal receiver noise.

For this, this paper presents a variability index (VI) adaptive operation based on wavelet transform for the sidelobe suppression on pulse compression when existing the band-limited internal receiver noise, and analyzes in contrast with conventional method. The rest of this paper is organized as follows. In Sect. 2, the LFM signal processing method, which is based on the pulse compression and sidelobe suppression by window weighting is defined. In Sect. 3, the signal processing method for sidelobe suppression of LFM waveform when existing the internal receiver noises is mathematically analyzed. Finally, in Sect. 4, the sidelobe suppression performance of this method is studied through simulation.

2 Preliminary

Let consider a radar system employing LFM waveforms. The continuous-time model of the transmitted LFM waveform can be written as

$$s_t(t) = \sum_{k=0}^{K-1} u(t-kT) \tag{1}$$

where

$$u(t) = A_0 \cdot rect(t/\tau) \cdot \cos\left[2\pi\left(f_0 t + \frac{1}{2}\mu t^2\right)\right] \tag{2}$$

is the LFM pulse having rectangular envelope, k is the number of pulses, T is the pulse period, $\mu = B/\tau$ is the sweep rate measured in Hertz per second, B is the LFM waveform bandwidth, τ is the uncompressed pulse width, f_0 is a carrier frequency of transmitted signal, and $rect(t/\tau)$ is given by

$$rect(t/\tau) = \begin{cases} 1, & |t| \le \tau/2 \\ 0, & |t| > \tau/2 \end{cases} \tag{3}$$

The complex envelope of (2) can be written as

$$\begin{aligned} u(t) &= A_0 \cdot rect(t/\tau) \cdot \exp\left[j\left(2\pi(f_0 t + \frac{1}{2}\mu t^2)\right)\right] \\ &= A_0 \cdot rect(t/\tau) \cdot \exp\left(j\pi\mu t^2\right) \cdot \exp(j\omega_0 t) = v(t)\exp(j\omega_0 t) \end{aligned} \tag{4}$$

where $v(t)$ is the complex envelop of a single LFM at baseband and is as follows;

$$v(t) = A_0 \cdot rect(t/\tau)\exp\left(j\pi\mu t^2\right) \tag{5}$$

A conventional approach in pulse compression is to correlate the received signal with a reference waveform. The impulse response of the matched filter for the complex envelop of the LFM signal of (5) can be expressed as:

$$h(t) = v^*(\tau - t) = \frac{1}{A_0} rect\left(\frac{\tau - t}{\tau}\right) \exp\left[-j\pi\mu(\tau - t)^2\right] \tag{6}$$

The matched filter output magnitude response due to the LFM signal can be calculated by performing a convolution process between the signal $V(t)$ and the matched filter impulse response $h(t)$ as follows [8]:

$$y(t) = h(t) * v(t) \tag{7}$$

$$|y(t)| = \text{sinc}[\pi B(t - \tau)] \tag{8}$$

For a point scatterer in white Gaussian noise, it is well-known that correlating the received signal with a delayed copy of the reference waveform, i.e. a matched filter (MF), maximizes the output signal-to-noise ratio (SNR). Unfortunately, the MF output has a low peak to sidelobe ratio (PSR). When the received signal contains both strong and weak echoes, the presence of weak targets can be masked by the range sidelobes of a sufficiently stronger echo if they are not suppressed adequately. To achieve range sidelobe suppression for radar systems with LFM waveform, a spectral window is often utilized to shape the spectral response of the reference waveform, leading to a mismatched filter [2, 17]. Fig. 1 shows the diagram of LFM pulse compression using the spectral window.

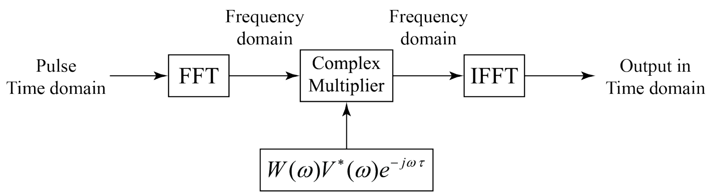

Fig. 1 The diagram of LFM pulse compression using the spectral window

To enhance the PSR and thereby allowing the detection of weak targets in the presence of stronger echoes, a spectral window $W(\omega)$ is often used to shape the spectral response of the reference waveform. This leads to a mismatched filter with the resulting reference waveform $h(t)$ now being given by

$$h(t) = \mathcal{F}^{-1}\left[W(\omega)V^*(\omega)e^{-j\omega\tau}\right] \tag{9}$$

where $\mathcal{F}^{-1}$ denotes the inverse Fourier transform operations and $V(\omega)$ is the

Fourier transform of $v(t)$. The windowed weighing function for the sidelobe suppression is given by

$$W(f) = K + (1-K)\cos^n\left(\frac{\pi f}{B}\right) \tag{10}$$

where K and n is coefficients related with a window shape. If $K = 0.08$, $n = 2$, it is Hamming weight function. In this case, the sidelobe is reduced to a level of –40dB and the mainlobe loss is about 2 or 3dB. This is obtained as the results of pulse compression in the case of no internal receiver noise. However, in the presence of internal receiver noise, the efficiency of pulse compression and sidelobe suppression by the aforementioned algorithms is reduced, and the smaller the input SNR of the radar system, the more the sidelobe component is increased, and the more the PSR is decreased. In this paper we propose a new thresholding technique to the wavelet transform by variability index (VI) of target returns in the state which is existed internal receiver noise.

3 Wavelet Threshold Processing by Variability Index (VI) Estimation

3.1 Threshold method in Wavelet Shrinkage by Variability Index (VI)

Based on the study and analysis of different sidelobe suppression algorithms of LFM signals in the literatures, we propose a variability index (VI)-wavelet adaptive shrinkage method to suppress the range sidelobes due to LFM pulse compression of target returns in the presence of internal receiver noise, and prove its accuracy by simulation. Fig. 2 shows the wavelet shrinkage adaptive processing diagram using VI estimating operations.

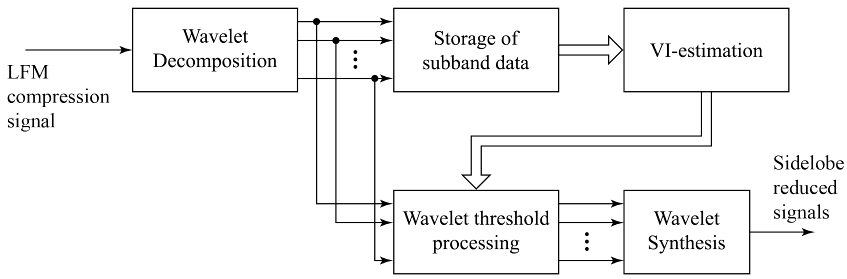

Fig. 2 VI-wavelet shrinkage adaptive processing diagram

The wavelet subband decomposition is used to divide into multiple bands based on the multi-scale wavelet coefficients according to the dyadic-scale, and then the wavelet threshold processing is performed on each subband outputs.

The signal detection using wavelet shrinkage and threshold method relies on the basic idea that the energy of a signal will often be concentrated on a few coefficients while the energy of noise is spread among all coefficients in wavelet multi-domain [11]. If the mainlobe width obtained after pulse compression is defined as τ_0, this width is defined as a range bin on a pulse repetition interval (PRI) T, and $M = T/\tau_0$ bins on the PRI will be existed. For the i th range bin ($0 \le i \le M-1$) of a PRI, the coefficients of the wavelet multiband are obtained by multi-resolution analysis (MRA), and M range bins of all are processed by MRA.

N data of i th range bin from the past $n-N$ to current n th PRI are used for VI calculation, and finally determine whether the target signal exists or not.

In radar signal processing, N is the number of the processing pulses, and it is limited by the number of target echoes received by the radar and related with the pulse repetition interval (PRI), the width of antenna beam and the rotation speed of antenna.

The VI-wavelet adaptive processing can be classified into the following steps:

Step 1. Calculate the multi-scale wavelet coefficients with respect to the dyadic scales $a = 2^j$, $1 \le j \le J$ for the received signal of i th range bin. The multi-scale wavelet coefficients are the approximation components $cA_{j,k}$ and the detail components $cd_{j,k}$ respectively. In this paper, we assume that the approximation and the detail coefficients at the higher scale than *J*=3 are not meaningful for the processing.

Step 2. Store the multiscale wavelet coefficients, $cA_{j,k}$ and $cd_{j,k}$ in the unit of range bin and then calculate VI of the N data stored in advance with respect to $cA_{j,k}$ and $cd_{j,k}$ respectively. The VI is used for the threshold processing of the signals decomposed into multiscale wavelet coefficients.

Step 3. For each wavelet coefficient, perform the threshold operation by the parameters determined to the VI estimation and then find the output after wavelet synthesis. These steps make sidelobe suppression of LFM signal in the presence of bandlimited noise. For an LFM signals, the output $x(n)$ obtained after sidelobe reduction by pulse compression and window function can be divided into wavelet subbands by MRA.

In the wavelet decomposition step, the basic functions such as Daubench, Morlet, and Shannon bases can be used. The Shannon function is the simple example of an orthonormal wavelet, and the Scaling function that belongs to the Shannon wavelet is the impulse responses of the ideal lowpass filter and it is given by

$$\phi(t) = \frac{\sin \pi t}{\pi t} \tag{11}$$

We may assume that the subspaces V_m contain lowpass signals and that the bandwidth of the signals contained in V_m reduces with increasing *m*. Because of the scaling property, the subspaces V_m are spanned by scaled and time-shifted versions of the scaling function $\phi(t)$:

$$V_m = span[\phi(2^{-m}t - k)], \quad m,\ k \in \mathbf{Z} \tag{12}$$

Also, if the functions $\varphi(2^{-m}t - k),\ k \in \mathbf{Z}$ span the subspace W_m containing bandpass signals, the Shannon Wavelet with impulse response of ideal bandpass filter is given by

$$\phi(t) = \frac{\sin \pi t/2}{\pi t/2} \cos\frac{3\pi}{2} t \tag{13}$$

When $V_0 = V_1 \oplus W_1$, the function $\phi_{0k}(t) = \phi(t-k) \in V_0$ can be written as linear combinations of the base functions for the spaces V_1 and W_1. The approach for wavelet analysis by Multi-rate filtering may be given by

$$\phi_{0k}(t) = \sum_p [h_0(2p-k)\phi_{ip}(t) + h_1(2p-k)\varphi_{ip}(t)] \tag{14}$$

where $\varphi_{1k}(t) = \varphi(t-k) \in W_0$ is the orthogonal base for W_1. The coefficients $h_0(p)$ and $h_1(p)$ as the impulse responses of low-pass and high-pass filter can be expressed as follows:

$$h_0(2p-k) = <\phi_{0k},\ \phi_{1k}>, \quad h_1(2p-k) = <\phi_{0k},\ \varphi_{1k}> \tag{15}$$

3.2 Variability Index (VI) Estimation

The VI is a second-order statistic that is closely related to an estimate of the shape parameter. Its value is a function of the estimated population mean μ and population variance σ^2 [10]. The VI estimating design partitions the reference windows in the channels of all wavelet subbands. For any channel input *x* of VI estimation in Fig. 2, The VI is then calculated for each split window using:

$$VI = 1 + \frac{\hat{\sigma}^2}{\hat{\mu}^2} = 1 + \frac{1}{N}\sum_{n=0}^{N-1} \frac{[x_n - E(x)]^2}{[E(x)]^2}, \tag{16}$$

where $E(x)$ is the arithmetic mean of the *N* cells in the reference window. *VI* is independent of the noise power in a homogeneous environment, but changes considerably in the presence of interfering targets within the reference window. The VI is compared with a threshold K_{VI}, to decide if the output cells of wavelet subband with which the VI is computed are from a homogeneous (non-variable) environment or from a non- homogeneous (variable) environment using the following hypothesis test:

$$\begin{cases} VI \geq K_{vi}, & \text{invariable(only noise)} \\ VI < K_{vi}, & \text{variable(noise plus target)} \end{cases} \tag{17}$$

If input noise in radar system is bandlimited white Gaussian noise, the output noise in the envelope detector of LFM pulse compression has Rayleigh distribution characteristics. When the variance of Gaussian noise is σ_0^2 , the envelope noise obtained after envelope detection is a random noise with Rayleigh distribution characteristics, its mean and mean square value are given by

$$E(x) = \sqrt{\frac{\pi}{2}}\sigma_0, \quad E(x^2) = 2\sigma_0^2 \tag{18}$$

where $E(x)$ and $E(x^2)$ are satisfied when N is infinite. In this case, the expected value of VI statistic is as follows:

$$E[VI] = 1 + \frac{E[(x - E(x))^2]}{[E(x)]^2} = 1 + \frac{E(x^2) - [E(x)]^2}{[E(x)]^2} \tag{19}$$

$$E[VI] = \frac{E(x^2)}{[E(x)]^2} = \frac{4}{\pi} = 1.2732 \tag{20}$$

If the input signal only is in the state of noise environment, the expected value of the VI statistic is 1.2732, but the expected value of the VI statistic differs from 1.2732 in the case with the state of noise plus target. According to this principle, after pulse compression of the LFM signal, the mainlobe component and noise are identified by VI . The signal of i th range bin is stored at intervals of pulse repetition period (PRI) as transmitting parameter of radar, and VI of i th range bin is computed by (16) using the stored data of i th range bin. For implementation purposes, it is possible to reduce the computational requirements associated with generating the VI statistic using an alternative definition. In this case, the simplified statistic $V\hat{I}$ is obtained by using the biased, maximum likelihood estimate of the population variance.

$$V\hat{I} = 1 + \frac{Va\hat{r}(x)}{\hat{\mu}^2} = 1 + \frac{\frac{1}{N}\sum_{n=0}^{N-1}\left(x_n - \frac{1}{N}\sum_{n=0}^{N-1}x_n\right)^2}{\left(\frac{1}{N}\sum_{n=0}^{N-1}x_n\right)^2}$$

$$\hat{VI} = N \frac{\sum_{n=0}^{N-1} x_n^2}{\left(\sum_{n=0}^{N-1} x_n\right)^2} \tag{21}$$

The relation between the expected value of the VI statistic and $\hat{VI}$ is $E[VI] = \lim_{N\to\infty} \hat{VI}$.

In practice a finite number N of cells used to calculate $\hat{VI}$ is determined by the number of received target pulse in radar. When a finite number of cells are used for estimation of VI, the analytic expression for probability distribution of VI is unavailable.

In lieu of analytical expression for the probability distribution of the VI statistic, it is possible to use Monte Carlo simulation to generate independent and identically distributed (IID) Rayleigh random numbers for the cells used compute the VI for a large number of trials. For the i th trial, a sample VI_i is calculated from the random numbers. The resulting values of VI_i are used to estimate the VI probability density function and to estimate required values of K_v to yield various values of the hypothesis test error α_0.

$$\alpha_0 = P\left[VI < K_v \mid \text{only noise}\right] \tag{22}$$

The hypothesis test error α_0 of (22) means the probability that might be wrongly identified as a target signal because the estimated variability index $\hat{VI}$ is smaller than K_v, though a certain sample data x is a Rayleigh distributed noise.

Table. 1 summarizes the relationship between α_0 and K_v for a number of representative windows sizes N based on the simulation results for 1, 000, 000 trials, and it is based on using Monte Carlo operation.

Table.1 Simulation results for K_v as a function of α_0 and N

error probability	K_v		
	$N = 32$	$N = 64$	$N = 128$
$\alpha_0 = 0.1$	1.1841	1.2109	1.2310
$\alpha_0 = 0.01$	1.1358	1.1748	1.1966
$\alpha_0 = 0.001$	1.1069	1.1447	1.1724

3.3 Determination of Wavelet Threshold Value

As input in the diagram of Fig. 2, the LFM compression signals are decomposed of the multi-scale wavelet coefficients CA_{ik} and Cd_{jk}, and the threshold function based on $V\hat{I}$ is defined as follows:

$$\eta_N(z,\ V\hat{I}) = \begin{cases} 0, & V\hat{I} > \hat{K}_v \\ z - V\hat{I}, & V\hat{I} < \hat{K}_v \end{cases} \tag{23}$$

Where z is the signal corresponding to multi-scale wavelet coefficients CA_{ik} and Cd_{jk} ($1 \le j \le 2$), $\eta_L(z,\ V\hat{I})$ is the output signal after threshold comparison.

As a final step, the output signals after threshold comparison are synthesized by wavelet synthesis by multi-rate filtering, and the coefficients of the synthesis filters with the dyadic scales are given by

$$\left.\begin{aligned} g_0(k) = h_0^*(-k) &\ \leftrightarrow\ G_0(z) = \widetilde{H}_0(-z) \\ g_1(k) = h_1^*(-k) &\ \leftrightarrow\ G_1(z) = \widetilde{H}_1(-z) \end{aligned}\right\} \tag{24}$$

where $h_0(k)$ and $h_1(k)$ are the coefficients of wavelet decomposition filter given from (15).

Fig. 3 shows the system diagram for VI-wavelet threshold processing for the sidelobe suppression of LFM signal.

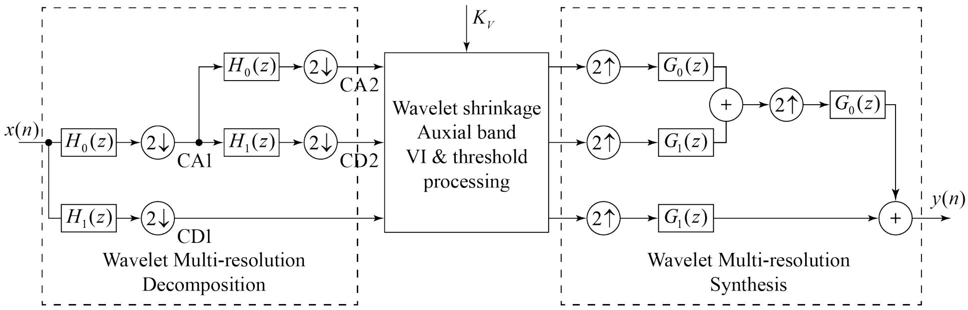

Fig. 3 Diagram of the VI-wavelet threshold processing for the sidelobe reduction

4 Simulation Results

Here LFM signal having duration, center frequency, pulse repetition interval (PRI) and bandwidth of $24\mu s$, $60MHz$, $100\mu s$ and $2.5MHz$ respectively is used for simulation study.

In Hamming window weighting of LFM waveform, PSR after pulse compression is about 35 dB, and the width of mainlobe increases about 2 times than rectangular window [2].

However, in the presence of internal receiver noise, the sidelobe level incre ases and depends on the signal-to-noise ratio (SNR) of the received signal.

In the above condition, the output waveform after pulse compression of LFM receive signal for SNR = -10 dB is shown in Fig. 4 (a), and here sidelobe occurs at a level of -13 dB compared to the peak of the mainlobe.

This paper was shown the sidelobe suppression that occurs in the process of pulse compression for LFM signal when existing the bandlimited white noise at a receiver.

In the proposed method of this paper, the largest discrete wavelet transform lever J=3 is selected and then the coefficients $h_0(k)$ and $h_1(k)$ for the wavelet analysis filter are obtained from (15), and the coefficients $g_0(k)$ and $g_1(k)$ for the wavelet synthesis filter are determined from (24).

Since the pulse width obtained after pulse compression is about $0.5\mu s$, the number of all range bins given as a unit of $0.5\mu s$ is 200, and then the range bins of 200 are processed by the proposed method in parallel.

From LFM parameters of a radar, windows size of $V\hat{I}$ is determined as N=32.

Here, in the case of $N = 32$ and $\alpha_0 = 0.01$, the threshold $\hat{K}_v$ is 1.1358 from Table 1.

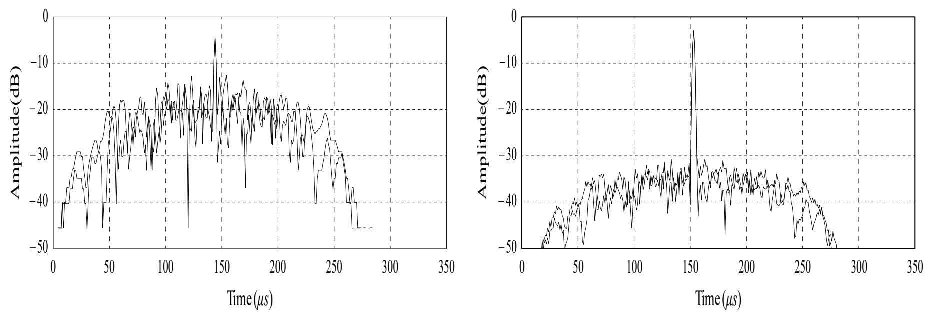

Fig. 4 PSR characteristics of LFM in the case of SNR=–10dB(a) PSR by classic method, (b) PSR by VI estimation method

This paper compared and estimated the PSR according to SNR using the window weighting method and VI-adaptive Wavelet processing method respectively.

As shown in Fig. 4(a), when SNR = −10dB, maximum PSR is 5dB after pulse compression, and it is very difficult to detect target by CFAR (Constant False Alarm Rate). Using the new method in this paper under the same conditions the PSR increases to 45dB, which is 35dB larger than PSR by window weighting method when existing bandlimited noises (bandwidth is 2.5MHz), Fig. 4 (b) was shown.

Convolutional windows are giving better results in terms of PSR than the conventional windows in the case absence of internal receiver noise, and at higher Doppler shifts the convolutional windows give better PSR values, i.e., about 4~5dB [2]. But convolutional window is not giving better results than the window weighting when existing internal receiver noise. The new method proposed in this paper are giving better results than previous methods (window weighting and convolutional window).

As shown in Table 2 the proposed method improves PSR by 33dB than previous methods, only when SNR >–15dB.

Table 2. Comparison of PSR for two methods

Input SNR (dB)	PSR (dB)		
	Previous method	New method	Gain(dB)
0	17	55	38
–5	11	49	38
–10	5	41	36
–15	0	33	33

5 Conclusion

This paper deals with the application of VI-wavelet threshold method to an important problem in signal processing, that is, a sidelobe suppression of LFM waveform in the presence of internal receiver noise. To improve the reliability and efficiency of traditional sidelobe suppression methods [1-5], a new method has been presented for sidelobe suppression in the LFM pulse compression when existing bandlimited white noise of receiver, and it has been proved by simulation. First, the wavelet subband decomposition by multi-scale wavelet coefficients is used to divide into multiple bands for LFM receive signals. Second, the outputs in wavelet subbands is obtained by VI threshold processing, and it determines in terms of the estimation of each subband data. Finally, wavelet synthesis is used to form to the output that sidelobe components is reduced. Simulation results show that the proposed method can provide better sidelobe suppression performance than previous methods.

References

1. M. Archana, M. Gnana priya, Low Power LFM Pulse Compression RADAR with Sidelobe Suppression. International Journal of Advanced

Research in Electrical, Electronics and Instrumentation Engineering. **3**, 10672–10679 (2014)

2. Sahoo, Ajith Kumar and Panda, Ganapati, Sidelobe reduction Of LFM signal using convolutional windows. [Online] Jan - 2011. [Cited: July 07, 2014.] http://dspace.nitrkl.ac.in/dspace/handle/2080/1351 (2014)
3. Iulian-Constantin Vizitiu, Some Aspects of sidelobe Reduction in Pulse Compression Radars Using NLFM Signal Processing. Progress In Electromagnetics Research. 47, 119–129 (2014)
4. Van Khanh Nguyen and Mike D. E. Turley, A Range Sidelobe Suppression Technique based on Adaptive Spectral Shaping for LFM Waveforms, National Security and ISR Division, Defence Science and Technology P.O. Box 1500, Edinburgh 5111, AUSTRALIA (2017)
5. Xiaoxiang Zhu, Guanghu Jin, Feng He, Zhen Dong. Radar Imaging of Sidelobe Suppression Based on Sparse Regularization. Journal of Computer and Communications. 4, 108–115 (2016)
6. Ahmed Ramadan, Mustafa El - Hoshy, Alaa El-Din Sayed Hafez, A New Biphasic Coding LFM for Pulse Compression Radar. International Journal of Scientific & Engineering Research. 5, 732–737 (2014)
7. Ahmed Abu El-Fadl, Performance Analysis of Linear Frequency Modulated Pulse Compression Radars under Pulsed Noise Jamming. International Journal of Engineering Research & Technology. 2, 1286–1290 (2013)
8. Arya V.J, Matched Filtering Algorithm for Pulse Compression Radar. The National Symposium on Antenna Signal Processing & Interdisciplinary in Electronics. 78–85 (2017)
9. Vinod Kumar, Ajit Kumar Sahoo, Grating lobe and Sidelobe Suppression using Multi-Objective Optimization Techniques. International Conference on Communication and Signal Processing.246–251 (2015)
10. Ajay Kumar Yadav, Laxmi Kant, Moving Target Detection Using VI-CFAR Algorithm on MATLAB Platform. International Journal of Advanced Research in Computer Science and Software Engineering. 3, 915–918 (2013)
11. D. L. Donoho, De-noising by soft-thresholding. IEEE Trans. Inform. Theory. Vol 41, No 3, 613–627(1995)
12. Varshney, L. R. and D. Thomas, Sidelobe reduction for matched fllter range processing. Proceedings of IEEE Radar Conference, 446-451 (2003)
13. De Witte, E. and H. D. Gri–ths, Improved ultra-low range sidelobe pulse compression waveform. design IET Electronics Letters, Vol. 40, No. 22, 1448-1450 (2004)
14. Vizitiu, I. C., L. Anton, G. Iubu, and F. Popescu, Sidelobes reduction using frequency predistortioning techniques on LFM signals. *Proceedings of IEEE ISETC Conference*, 381-384 (2012)

15. Doerry, A. W., Generating precision nonlinear FM chirp waveforms. SPIE Proceedings, Radar Sensor Technology XI, Vol. 6547 (2007)
16. Boukefia, S., Y. Jiang, and T. Jiang, Sidelobe reduction with nonlinear frequency modulated waveforms. Proceedings of IEEE CSPA Conference, 399-403 (2011)
17. L. R. Varshney and D. Thomas, Sidelobe reduction for matched filter range processing," *IEEE Radar Conference*, Huntsville, AL, USA, 446-451 (2003)