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Sidelobe reduction in Synthetic Aperture Radar (SAR) using Multi Stages of Linear Frequency Modulated (MS-LFM) pulse

[1]Z. M. Fawzy, [2]A. A. Shaalan, [3]M. M. Fouad, [4]*K. F.A. Hussein*
[1,2,3]*Elecrtronic and Communication Engineering Department , Zagazig University, Egypt.*
[4]*Electronics Research Institute, Cairo, Egypt.*

Abstract:
A commonly used Synthetic Aperture Radar (SAR) pulse is the Linear Frequency Modulated (LFM) pulse. It has the advantage of greater bandwidth while keeping the pulse duration short and the envelope constant. However, the matched filter output of this signal contains range sidelobes with the first sidelobe at a level of -13.2 dB to the peak of the main lobe. In this paper, the linear frequency modulation (LFM) waveform and matched filter response are introduced. The principle of Multi Stages of Linear Frequency Modulation (MS-LFM) is also discussed. Simulation results of the proposed MS-LFM signal are presented, where sidelobe level reduction of more than -20 dB can be achieved.
Keywords: *Pulse Compression, Matched Filter, SideLobe Reduction, Synthetic Aperture Radar (SAR), Linear Frequency Modulation (LFM), Multi Stages of Linear Frequency Modulation (MS-LFM).*

I. Introduction

The matched filter and pulse compression concepts are the basic of synthetic aperture radar (SAR) processing algorithms [1]. High range resolution demands a small value of pulse width, but good detection range performance demands high transmitted pulse energy, and hence a large value of pulse width. So instead of a short pulse, a long pulse of carrier wave is transmitted to get the required energy for long range target detection. A wide band-width which is associated with short pulse is obtained by modulating the carrier. In the receiver a matched filter is used to compress the energy into short pulse to get the required range resolution [2-4].

One of the simplest and most widely used form of carrier modulation is the Linear Frequency Modulation (LFM) or chirp waveform. An LFM signal is a kind of signal in which the frequency of the transmitted signal is varied over a pulse duration of T. This variation of the frequency from low to high or vice versa is known as “chirping”. Changing the frequency from low to high is called “up-chirp”. Similarly, changing the frequency from high to low is called “down-chirp”. However, the LFM has large sidelobes with respect to the mainlobe. Reducing the sidelobes can be achieved by many ways such as linear filtering the output, applying window function or data tapering [4-8].

In this research domain, another technique of Multi Stages-LFM signals with changing the frequency from high to low represents an important class of continuous phase modulation waveforms with applicability inside pulse compression radar systems. They have been claimed to provide a sidelobe reduction and hence a high-range resolution.

The rest of the paper is organized as follows. A brief description about LFM signal is described in section 2. The procedure for reducing sidelobe using MS-LFM is presented in section 3. Simulation results and discussion are presented in section 4. Finally the conclusions of the investigation are provided.

II. Linear Frequency Modulation (LFM) waveform

The utility of chirp waveform in imaging radar comes about because the duration of this signal can be long compared to that of continuous wave burst pulse and yet the result is the same effective bandwidth. LFM signals have the characteristic that the instantaneous frequency increases or decreases linearly over the duration of the signal. Thus the chirp waveform can be described by

$$s(t) = cos\left[2\pi\left(f_c t + k\frac{t^2}{2}\right)\right] \qquad -\frac{T}{2} \le t \le +\frac{T}{2}$$

(1)

Where fc is the starting frequency(Hz) and k is the chirp rate (Hz/s). The instantaneous frequency is given by

$$f_i(t) = f_c + kt$$

(2)

The bandwidth of this signal is given by

$$B = kt$$

(3)

Although the LFM signal has a duration of T, it can behave like a pulse with duration equivalent to the inverse of its bandwidth, i.e, $T_{eq} = 1/B$. The signal processing that allows this to happen is known as pulse compression. The amount of this compression is given by $T/T_{eq} = TB$ which is the time bandwidth product of the waveform [8]. The matched filter of the LFM has an impulse response, h(t), that is time inverse of the signal at the receiver input and it is given by

$$h(t) = K' cos\left[2\pi\left(f_c t - k\frac{t^2}{2}\right)\right], \qquad -\frac{T}{2} \le t \le \frac{T}{2}$$

(4)

where K$'$ is a factor that result from target backscattering. The output of a matched filter can be obtained either by the convolution integral or the cross correlation integral [9]. For the convolution integral, let $y(t)$ be the output of the matched filter. The output $y(t)$ is then given by

$$y(t) = h(t) * s(t) = \int_{-\infty}^{\infty} h(t-\lambda)s(\lambda)d\lambda$$

(5)

The closed form solution of the above equation can be obtained through a considerable amount of trigonometric and algebraic manipulation [9-10]. The result of this calculation is in the form of $\frac{\sin x}{x} = sinc\ x$ with the first sidelobe at a level of approximated 13.2 dB to the peak of the main lobe. Fig.1.shows the plot of LFM signal, the modulation signal and its auto-correlation function. The LFM signal is generated based on the specification of Time Bandwidth product of 200.

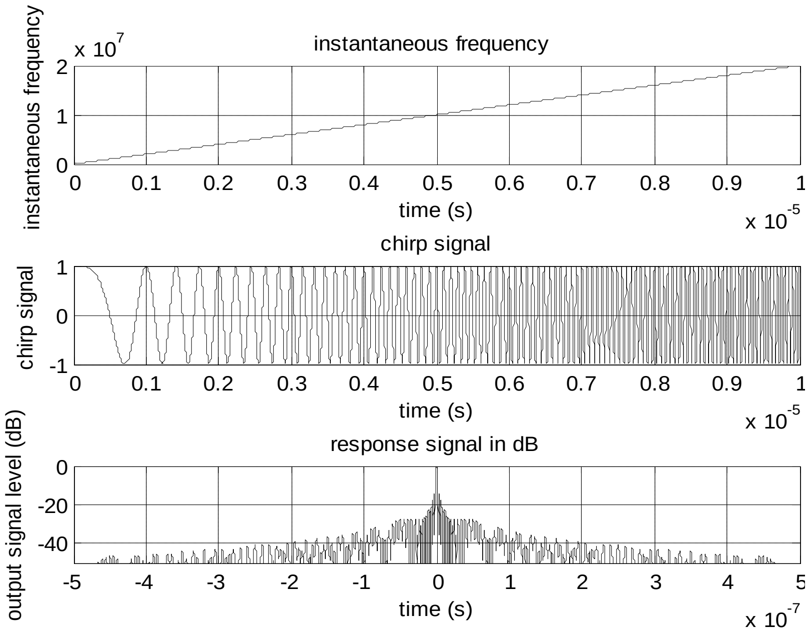

Fig.1. Linear FM waveform.

III. Multi Stages-Linear Frequency Modulation (MS-LFM) waveform

LFM function with changing the frequency from high to low is introduced for pulse compression as shown in Figure 2 where its instantaneous frequency can be written as

$$f_i(t) = B - K\,t \qquad 0 \le t \le T$$

(6)

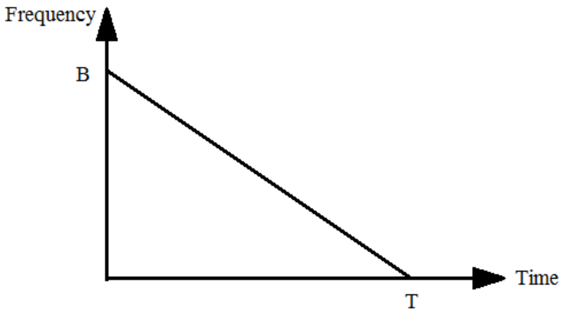

Fig.2. Instantaneous frequency for LFM waveform

Thus, the chirp waveform can be described by

$$s(t) = \cos\left[2\pi\left(B\,t - k\frac{t^2}{2}\right)\right] \qquad 0 \le t \le T \tag{7}$$

The stages of LFM used in this paper are

(1) Two Stages – LFM

$$f_i(t) = \begin{cases} B - \frac{B_1}{T_1}t & 0 \le t \le T_1 \\ \frac{B_2}{T_2}(T - t) & T_1 \le t \le T \end{cases} \tag{8}$$

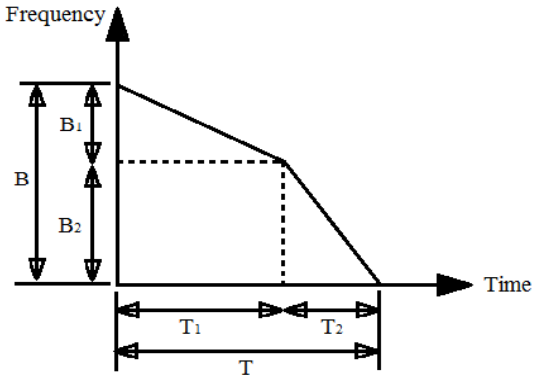

Fig.3. Instantaneous frequency for Two Stages-LFM waveform

(2) Three Stages – LFM

$$f_i(t) = \begin{cases} B - \frac{B_1}{T_1}t & 0 \le t \le T_1 \\ B_3 + \frac{B_2}{T_2}(T_1 + T_2 - t) & T_1 \le t \le T_1 + T_2 \\ \frac{B_3}{T_3}(T - t) & (T_1 + T_2) \le t \le T \end{cases} \tag{9}$$

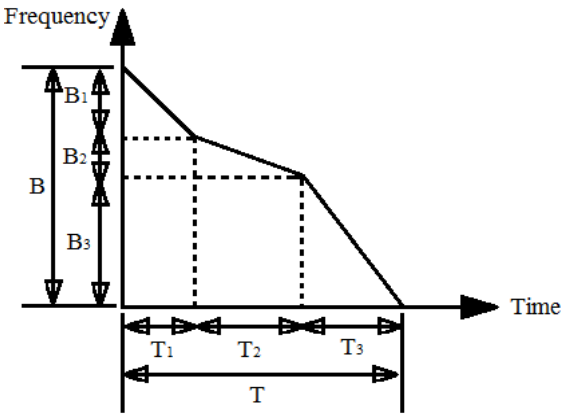

Fig.4. Instantaneous frequency for 3-Stages-LFM waveform

(3) Four Stages – LFM

$$f_i(t) = \begin{cases} B - \frac{B_1}{T_1}t & 0 \le t \le T_1 \\ (B - B_1) - \frac{B_2}{T_2}(t - T_1) & T_1 \le t \le T' \\ B_4 - \frac{B_3}{T_3}(t - T + T_4) & T' \le t \le T'' \\ \frac{B_4}{T_4}(T - t) & T'' \le t \le T \end{cases} \quad (10)$$

Where $T' = T_1 + T_2$ and $T'' = T_1 + T_2 + T_3$

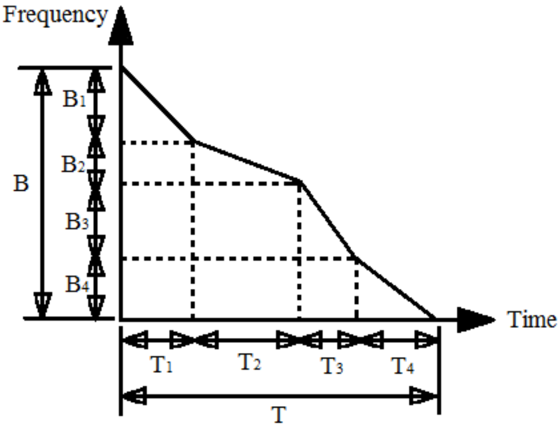

Fig.5. Instantaneous frequency for 4-Stages-LFM waveform

(4) Five Stages – LFM

$$f_i(t) = \begin{cases} B - \frac{B_1}{T_1}t & 0 \le t \le T_1 \\ (B - B_1) - \frac{B_2}{T_2}(t - T_1) & T_1 \le t \le T' \\ (B - B') - \frac{B_3}{T_3}(t - T') & T' \le t \le T'' \\ B_5 - \frac{B_4}{T_4}(t - T + T_5) & T'' \le t \le T''' \\ \frac{B_5}{T_5}(T - t) & T''' \le t \le T \end{cases} \quad (11)$$

Where $T' = T_1 + T_2$, $T'' = T_1 + T_2 + T_3$, $T''' = T_1 + T_2 + T_3 + T_4$ and $B' = B_1 + B_2$.

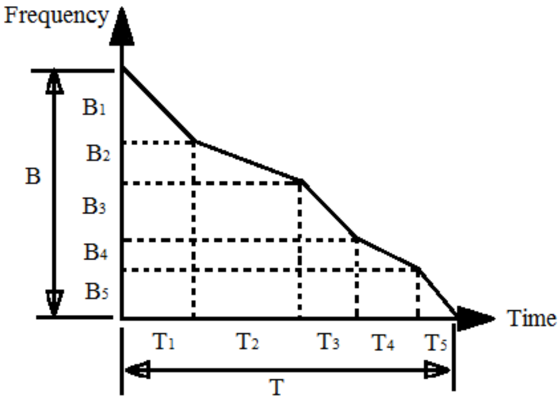

Fig.6. Instantaneous frequency for 5-Stages-LFM waveform

IV. Results and discussion

The LFM signal with duration and bandwidth of 10µs and 20MHz respectively is used for simulation study. A few simulations have been carried out for different values of T1, T2, T3, T4, T5, B1, B2, B3, B4 and B5. Figure 7 to figure 10 show the highest sidelobe level for each case. The relation between number of stages N and the highest side lobe level is given in table1.

Table 1: relation between number of stages N and the highest side lobe level

N	1	2	3	4	5
SideLobe Level(dB)	-13.2	-19.65	-20.15	-17.6	-16.4

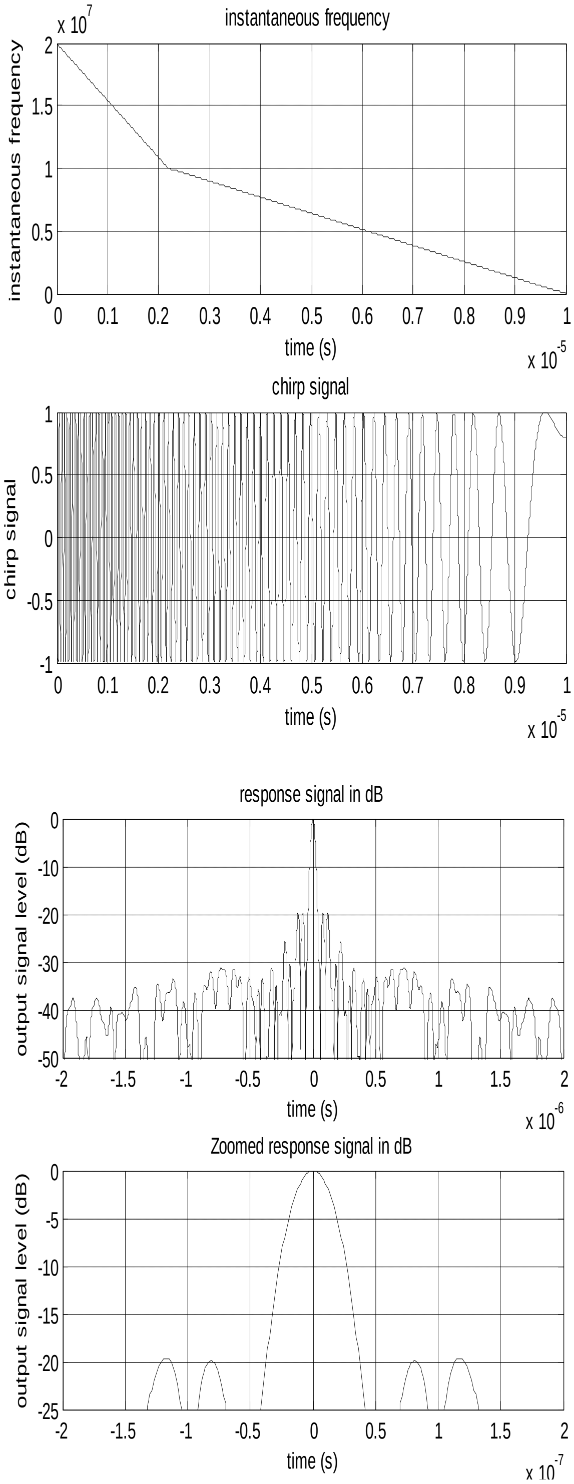

Fig.7. Modulation signal for Two Stages-LFM waveform with a sidelobe level of -19.65 dB.

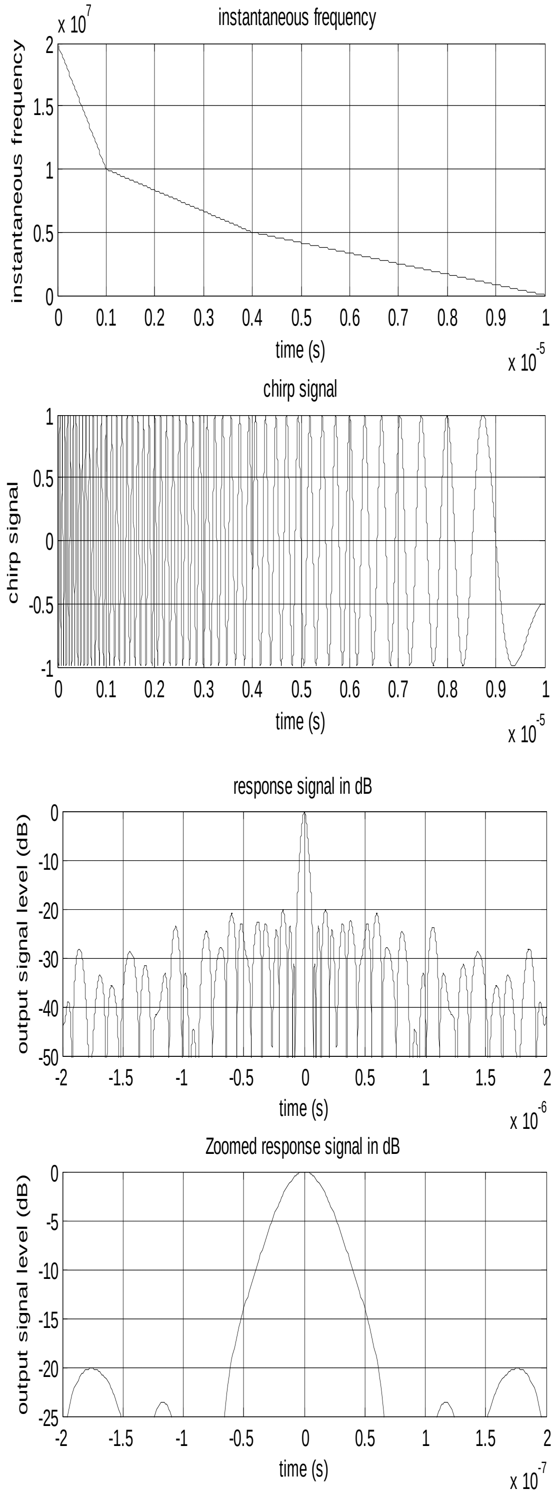

Fig.8. Modulation signal for Three Stages-LFM waveform with a sidelobe level of -20.15dB.

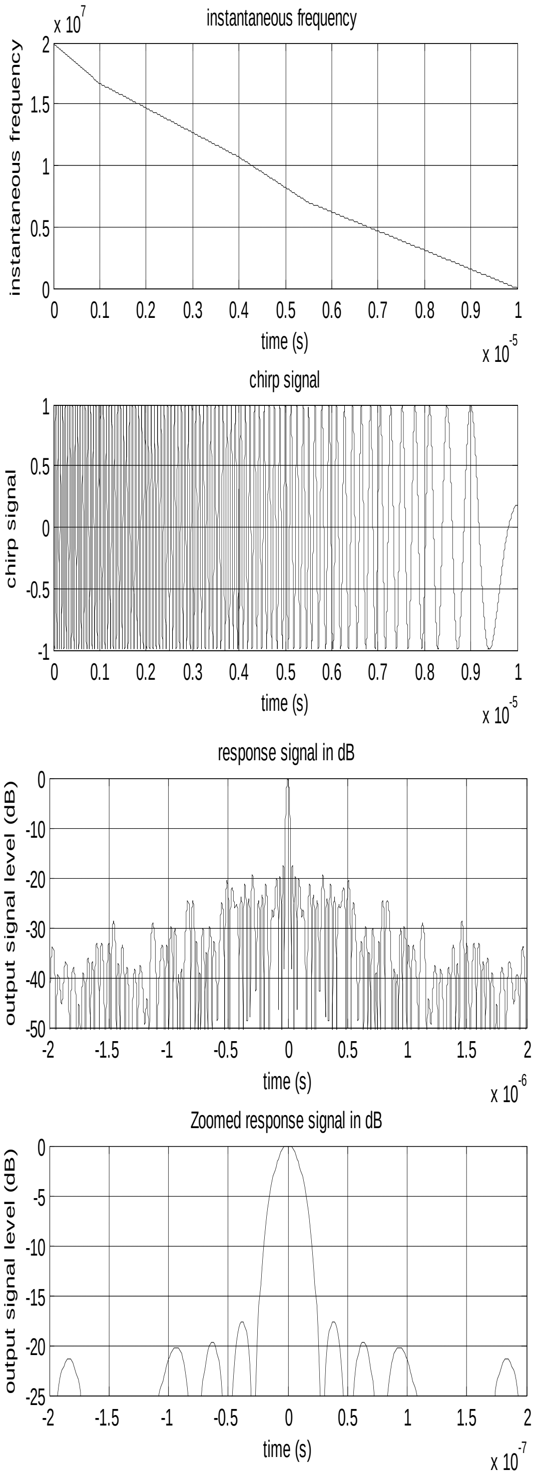

Fig.9. Modulation signal for Four Stages-LFM waveform with a sidelobe level of -17.6dB.

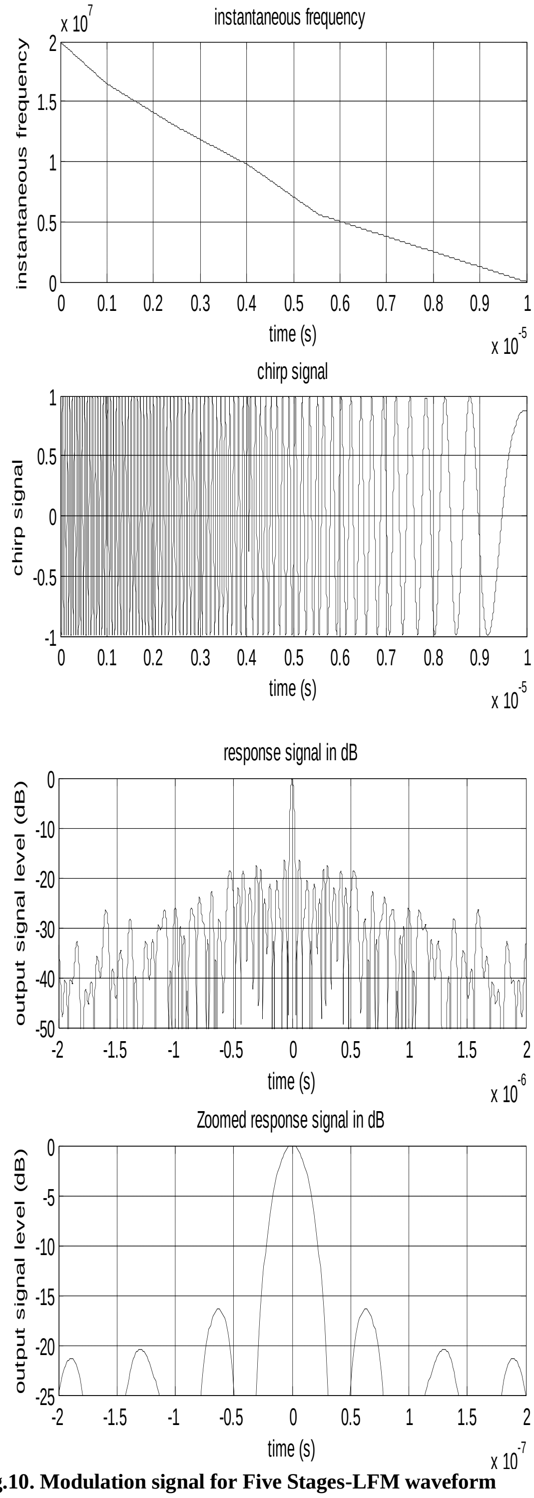

Fig.10. Modulation signal for Five Stages-LFM waveform with a sidelobe level of -16.4dB.

V. Conclusion

This paper presents the principle of LFM signal in term of matched filter response and pulse compression. The autocorrelation function of LFM waveform can be approximated by a sinc function with -13.2 dB sidelobes. The principle of a chirped pulse with MS-LFM is introduced as a way for achieving fine SAR imaging resolution and good signal to noise ratio. The simulation results show an improvement of 3.2 dB to 6.95 dB depends on the number of stages applied. It has been shown that the Three-Stages LFM signal capable of achieving the highest sidelobe reduction with a level of -20.15 dB. In summary, the LFM has been demonstrated to be an effective technique for sidelobes suppression for its simple implementation scheme without applying any window function or data tapering.

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