

Reduction in sidelobe and SNR improves by using Digital Pulse Compression Technique

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ABSTRACT

Digital pulse compression is the technique to convert short pulse to long pulse because energy content of long pulse with low peak power would be the same with short pulse having high peak power. In this paper pulse compression is performed by Matched Filter in signal processing environment. At some finite interval of time the high peak amplitude of signal with narrow bandwidth gives the energy of the transmitted pulse from the radar. Matched filter gives the signal to noise ratio at the receiver output. Sidelobe reduces to -63 db. Signal to noise ratio (SNR) improves to 12dB. Non linear frequency modulated (NLFM) pulse compression is used. They have been claimed to provide a high range resolution, improved SNR, low cost, good interference mitigation and spectrum weighting function. High range side lobe can cause poor performance in both target and weather detection. The results have been validated using experimental data.

Keywords: Digital pulse compression, FFT, IFFT, peak side lobe level, Linear frequency modulation (LFM), Non linear frequency modulation(NLFM), matched filter, side lobe suppression filter.

I. INTRODUCTION

The Pulse compression is the technique in which a long duration pulse with high energy is modulated. Linear frequency modulated or phase modulated increases the bandwidth of the transmitted signal. Now these modulated pulse is transmitted through parabolic reflector in radar. These long duration pulse strikes with the different number of targets, echo returned to the receiving antenna. These signal passed through matched filter and noise present in the signal is compressed by the matched filter processor. Pure matched filter gives a low side lobes. The NLFM signals also assure better detection rate characteristics and they are more accurate in range determination.

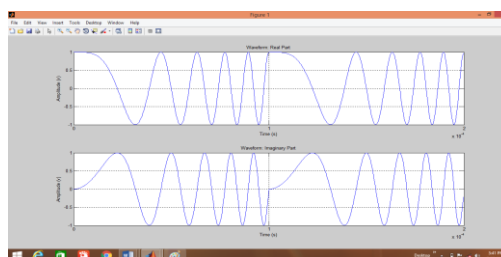
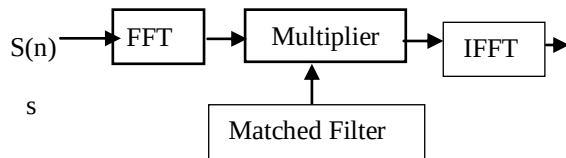
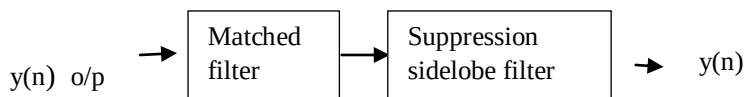


Fig1. The Non linear frequency modulated signal

In the pulse compression NLFM pulse resolving closely placed multiple targets known as range resolution. Another advantage it reduces sidelobe to many possible preferable values. The mathematical expressions is discussed to achieve the desired value. The NLFM signal is given to the matched filter. The output of the matched filter is given to the sidelobe suppression filter.



$S(n)$ is the transmitted signal. $y(n)$ is the output signal. A compressed output received in which sidelobe reduces by suppression filter. The description is given below



Sidelobe suppression to -63 dB.

II. PULSE COMPRESSION METHOD

This section includes some of the discussions of journal papers on RADAR pulse compression method.

Shinriki, et al.,[8] have given a new idea of pulse compression method for normal pulse. In this proposed method a filter whose impulse response in frequency domain is the ratio between the desired waveform and input signal in frequency domain. This method has the advantage of compressing to arbitrary pulse width given by the desired waveform and this method yields low peak sidelobe level.

Justin Sagayaraj M, Manisha Sanal, have given few sample optimum biphas codes with low sidelobes. They have Optimized FIR filters for digital pulse compression waveform. They have achieved low sidelobe level of -35 dB to -40 dB.

M. Archana, M. Gnanapriya have given the idea of low power pulse compression RADAR with Sidelobe suppression. This method discuss the LFM pulse compression method in frequency domain. The higher sidelobe level is suppressed by using different windows method.

III. EVALUATION OF PARAMETERS

The parameters are peak sidelobe ratio (PSR) is given by

$$PSR = 10 \log_{10} \frac{\text{Peak sidelobe pulse power}}{\text{main lobe pulse power}}$$

Pulse compression ratio is given by

$$PCR = \frac{\text{Duration of pulse before compression}}{\text{Duration of compressed pulse}} =$$

Where c is the speed of light, τ is the pulse width.

$$PCR = \frac{c \cdot \tau}{\tau} (1)$$

The instantaneous frequency modulation function of tangent based NLFM is given by

$$f(t) = \frac{\beta \tan \frac{2\gamma t}{\tau}}{2 \tan \gamma}, \quad -\frac{\tau}{2} < t < \frac{\tau}{2} \quad (2)$$

where $\gamma = \tan^{-1} \alpha$, $\alpha > 0$ is sidelobe control factor. β is the bandwidth.

Hybrid Non Linear Frequency Modulation can be implemented. The instantaneous frequency of NLFM is given by

$$f(t) = \frac{t}{\tau} \left[\beta_1 + \beta_2 \frac{1}{\sqrt{1 - 4\frac{t^2}{\tau^2}}} \right], \quad -\frac{\tau}{2} < t < \frac{\tau}{2} \quad (3)$$

The parameter $\frac{\beta_1 t}{\tau}$ represents Linear Frequency Modulation term. While the parameter β_c is chebyshev shaped spectrum i.e. constant sidelobe level.

The Phase of NLFM is obtained by integrating the instantaneous frequency. LFM signal has the higher peak sidelobe level. Non linear frequency modulated signal has reduced the sidelobe level to greater extent.

III. DESIGN OF MATCHED FILTER

Matched filter improves the SNR at the output of the receiver. Matched filter is a system used in the initial stage of digital system receiver. Matched filter is used to decrease Probability error by increasing Signal to Noise ratio (SNR).

The matched filter has an adaptability to the signals of same waveform despite different amplitude and time delay. Signal $s(t)$ matches to the signal $as(t-\tau)$. The general method is to use MATLAB tools. To generating ideal chirp signal, filter use this chirp signal gives the spectrum and compress the result because echo signal is not ideal, the noise and clutter are not identical. So SNR at the output of the receiver.

$$\left(\frac{S}{N}\right) = \frac{|s(t)|^2}{No} \quad (4)$$

$$\text{Where,} \quad S(t) = \int_{-\infty}^{+\infty} S_i(f) \cdot H(f) e^{-i\omega t} df$$

$$No(t) = \int_{-\infty}^{+\infty} \frac{No}{2} |H(f)|^2 df$$

$No(t)$ is the output noise at receiver. $S(t)$ is the signal received at the output. $S_i(f)$ is the fourier transform of the input signal. $H(f)$ is the matched filter response.

Now by using Schwartz inequality

$$|\int_{-\infty}^{+\infty} X(f) Y(f) df|^2 = \int_{-\infty}^{+\infty} |X(f)|^2 df \int_{-\infty}^{+\infty} |Y(f)|^2 df$$

$$H(f) = S_i(f) e^{-i\omega t} \quad (5)$$

This is our required matched filter expression.

$$\frac{S}{N} = \frac{2E_o}{No} \quad (6)$$

Where E_o is the output energy and N_o is the output noise. Signal to noise ratio is inversely proportional to noise and directly proportional to energy. The time delay signal or time lead signal will be matched filter by the transmitted signal. Basically the amplitude is increased at some time interval and pulse width of noise is compressed. SNR improves at the receiver of the pulsed radar.

IV. COMPARISION OF WINDOW TECHNIQUES

HAMMING WINDOW: It is used to find out leakage factor, relative sidelobe attenuation, main lobe width (-3dB).

$$w(n) = 0.54 - 0.46 \cos\left(2\pi \frac{n}{N}\right), \quad 0 \leq n \leq N \quad (7)$$

When $N = 140$

Leakage factor = 0.04%

Relative sidelobe attenuation = -42.68 dB

Main lobe width (-3dB) = 0.018555

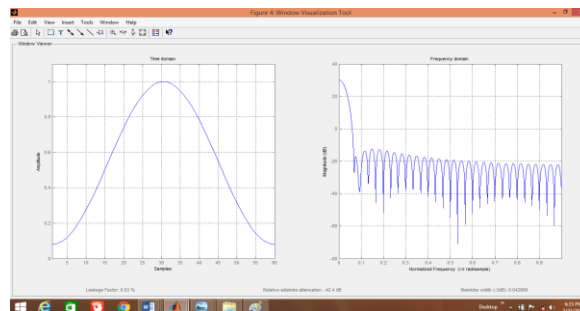


Fig2.Sidelobe attenuation curves, $N = 140$

When $N = 124$

Leakage factor = 0.03. %

Relative sidelobe attenuation = -42.6 dB

Main lobe width (-3dB) = 0.019531

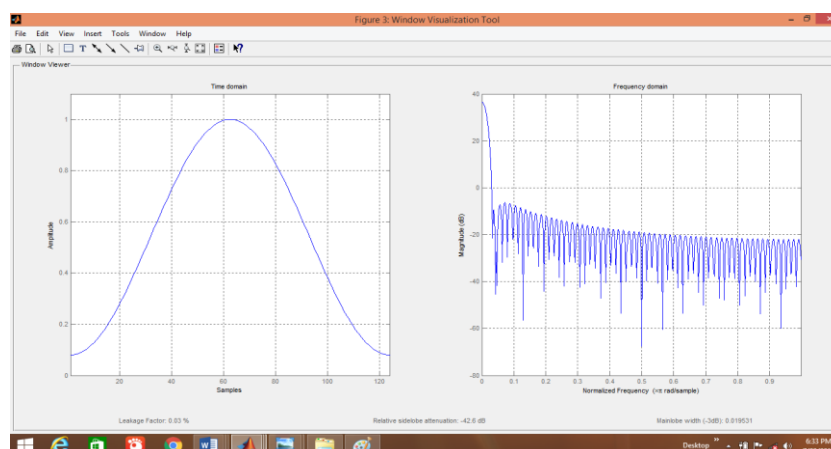


Fig3. Sidelobe attenuation curves, $N = 124$

$$\beta = \begin{cases} 0.1102(\alpha - 8.7), & \alpha > 50 \\ 0.5842(\alpha - 21)^{0.4} + 0.07886(\alpha - 21), & 50 \geq \alpha \geq 21 \\ 0, & \alpha < 21 \end{cases} \quad (8)$$

When $\beta = 0.5$ and $\alpha = 140$

Leakage factor = 8.6 %

Relative sidelobe attenuation = -13.6 dB

Main lobe width (-3dB) = 0.012695

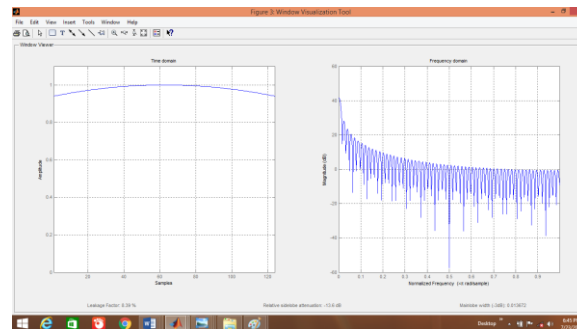


Fig4. Sidelobe attenuation curves, $\alpha = 140$

When $\beta = 0.5$ and $\alpha = 124$

Leakage factor = 8.39 %

Relative sidelobe attenuation = -13.6 dB

Main lobe width (-3dB) = 0.013672

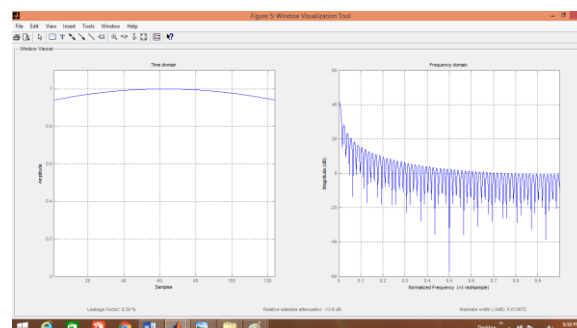


Fig5. Sidelobe attenuation curves, $\alpha = 124$

it is - Hence for above comparison hamming window technique provides better result. Leakage factor is always less in hamming window compared to Kaiser window for finding out the sidelobe attenuation. In hamming window technique sidelobe attenuation provides better result as -42.69 dB. In Kaiser window technique 13.6 dB.

V. RESULT OF MATLAB SIMULATION

The compressed output pulse is obtained in matlab. The long duration pulse is compressed and at some time interval say 't' is having a high peak amplitude pulse waveform as shown in fig.

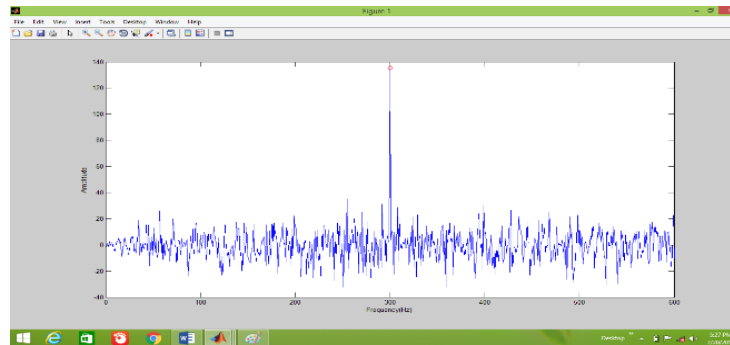


Fig6. Theinitilpulse compression.

Sidelobe values lies between -20dB to 20dB.

The reduction in the sidelobe is the major work in this paper. For this reduction signal waveform analyser tool is used to get desired result. The ratio of Peak sidelobe power to the main lobe power is find out.

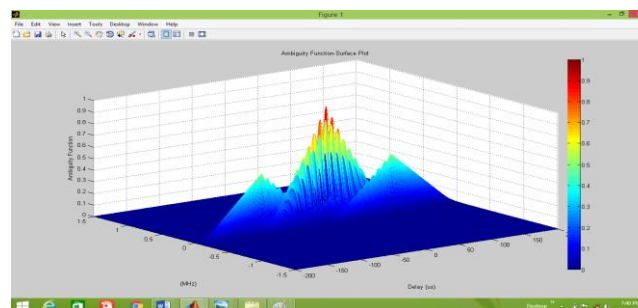


Fig7.The shape of ambiguity surface function.

The reduction in the sidelobe for mismatching filter is a hecting process for some window techniques. For this mismatching filter the Kaiser window gives the appropriate reduction in the sidelobe.

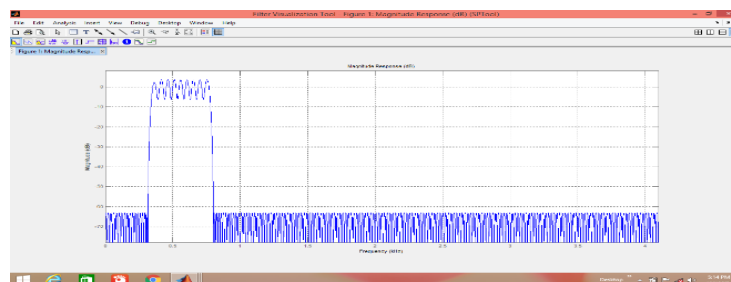


Fig8. The sidelobe representation waveform.

The sidelobe peak value is approx. 62dB.

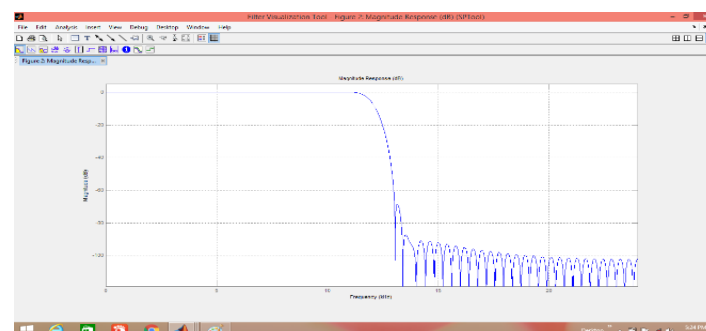


Fig 9. Window filter waveform.

The amplitude of sidelobe lies -62dB to – 70dB.

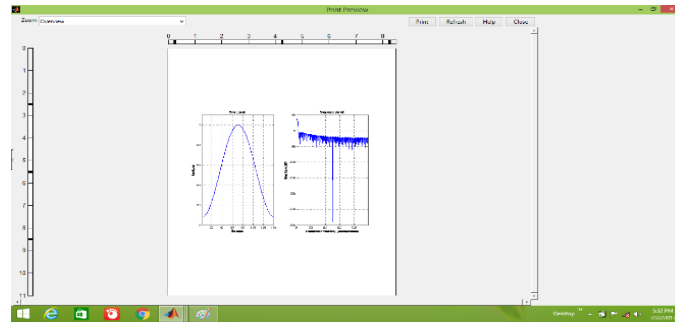


Fig10. The sidelobe reduces pulse.

It reduces to – 42 dB in MATLAB environment.

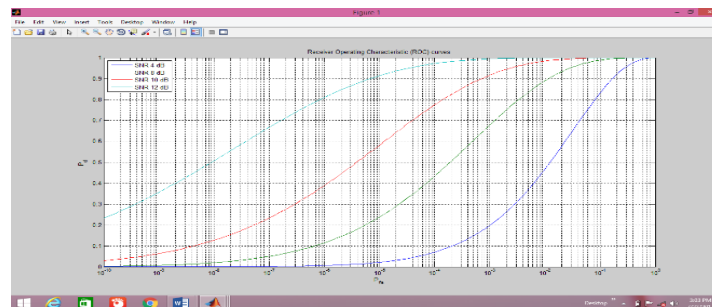


Fig11. ROC curves evaluate SNR = 12dB.

12 dB represented by above red curves which is having some initial probability of pulse detection (P_d). Probability of false alarm is very low value(P_{fa}).

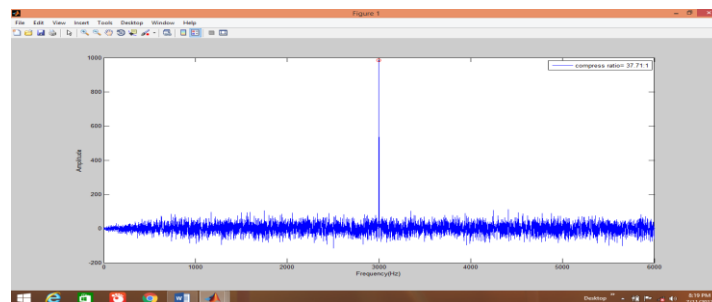


Fig 12. The digital pulse compression output.

Pulse compression ratio = 37.71

VI. CONCLUSION

The real time generation of LFM and NLFM waveform is used precisely that gives a proper range sidelobe levels. The NLFM processing algorithm has the advantage to improve the shape of the compression pulse waveform. The effective sidelobe reduction technique NLFM algorithm is used. It has a wide applicability in the pulse compression technique. The higher sidelobe level reduces by using different window techniques. The parameters peak sidelobe level, main lobe width, compressed pulse width of hamming and Kaiser window is calculated. The window are used according to their requirement to suppress the sidelobes. This paper examines the real time digital pulse compression by matched filter in frequency domain. Pulse compression for LFM radar signal has been simulated and implemented using MATLAB. Digital pulse compression by matched filter perform in matlab software with the help of programming. Pulse compression is implemented to improve the



radar range resolution long pulse for long range detection. Both matlab simulation and implementation results show a very good agreement with the theory and concepts of pulse compression. Pulse compression with the reduced sidelobes is the main work area.

In Future work Costas Pulses can be used that follow the frequency hopping sequence gives sidelobe levels which is $1/N$ of the mainlobe peak for any delay and Doppler shift. Work can also be extended to add some Doppler effect and constant false alarm rate algorithm for finding speed, eliminating clutter and echoes disturbances.

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