RADAR PULSE COMPRESSION USING FREQUENCY MODULATED SIGNAL

A Thesis Submitted in Partial Fulfillment
Of the Requirements for the Degree of

Bachelor of Technology
In
Electronics and Instrumentation Engineering

By

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Department of Electronic & Communication Engineering

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NATIONAL INSTITUTE OF TECHNOLOGY

ROURKELA

C E R T I F I C A T E

This is to certify that the thesis entitled "***RADAR PULSE COMPRESSION USING FREQUENCY MODULATED SIGNAL***", submitted by ***SWAGAT DAS (109EI0321)*** and ***SWASTIK JENA (109EI0325)*** for the award of Bachelor of Technology Degree in '***ELECTRONICS & INSTRUMENTATION***' Engineering at the National Institute of Technology (NIT), Rourkela is them under my supervision.

DATE: Prof. Ajit Kumar Sahoo

PLACE: Department of Electronics and Communication

NIT Rourkela

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We would like to thank all my friends, faculty members and staff of the Department of Electronics and Communication Engineering, N.I.T. Rourkela for their extreme help throughout course.

Swastik Jena (109EI0325)

Swagat Das (109EI0321)

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ABSTRACT

Range resolution for given radar can be improved by using very short pulses. But utilizing short pulses decreases the average transmitted power. To solve these problems pulse compression technique is used. It consists of two types of correlation process: matched filtering and stretch processing. Generally, we use matched filtering for narrowband signal and stretch processing for wideband signals. LFM signal is used in both the process as its bandwidth is independent of its pulse width. In this thesis we have analyzed two pulse compression technique and effect of time bandwidth product, Doppler shift on LFM signal passed through different windows. In radar masking effect is observed due to hiding of the far target's weak echo by near target's strong echo. Its removal is done by subtracting a replica of nearby target echo from the received signal. Matched filtering and stretch processing methods are used for this purpose.

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1.1. ***Pulse compression***:

Range compression is achieved by using Linear Frequency Modulated waveform and applying pulse compression, to achieve a good range resolution. Here, the average transmitted power of a relatively long pulse is generated, while obtaining the range resolution corresponding to a short pulse.

1.1.1. ***Matched filter***:

It is a linear filter used to maximize the SNR of the received signal in presence of noise. Here, the received signal, with noise, is tested with a replica of transmitted signal to maximize the signal energy over the additive noise.

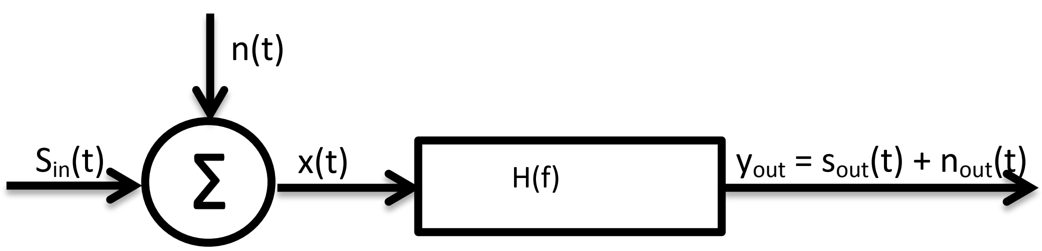


Fig.1.1: The matched filter receiver

1.1.2. ***Ideal Ambiguity function***:

It provides perfect resolution between two very close neighboring targets. It is a 2D impulse (or Dirac delta) function located at (t=0, f D=0).

1.1.3. ***Range compression***:

LFM or chirp signal is used to compress the range signal. In Range compression matched filtering is used to increase the signal-to –noise ratio (SNR). Then pulse compression technique is used to obtain range compressed wave form

1.1.4. ***Azimuth compression***:

Long synthetic antenna is used to obtain compression along the azimuth or along the cross range. Long synthetic antenna is formed by the radar motion.

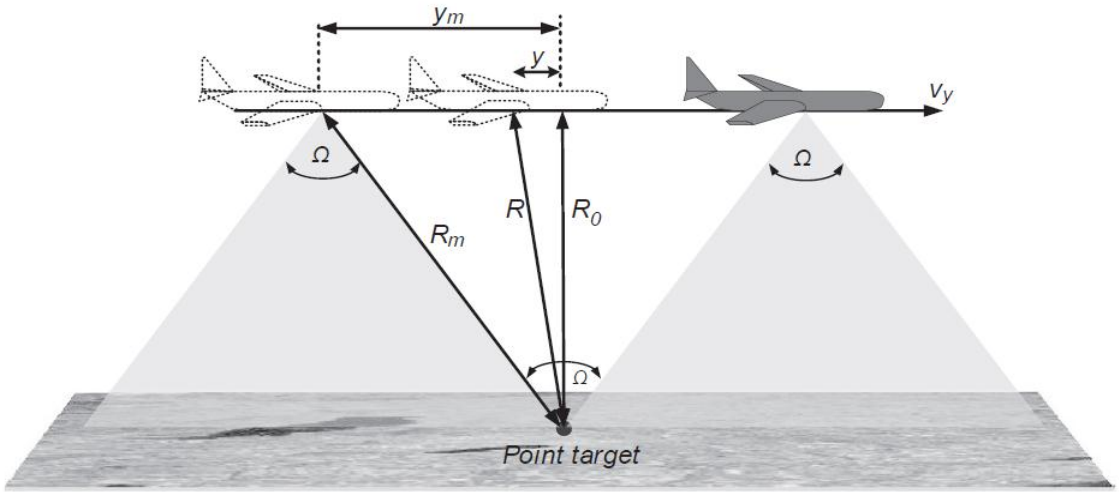


Fig.1.2: Geometry for synthetic aperture radar (SAR).

1.2. ***Synthetic Aperture Radar***

Synthetic-aperture radar (SAR) is a form of radar. SAR uses relative motion between antenna and the target to give distinct long-term coherent-signal variations. These are then used to obtain spatial resolution of the image. It is a remote sensing technique used for imaging remote targets on a land scape. In 1951, Carl Wiley observed that Doppler spectrum of the received signal can be utilized to synthesize a long aperture so that very close targets can be resolved.

1.2.1. ***Types of SAR***

There are three modes of SAR operation:

i) side looking SAR or strip-map SAR ,

ii) spotlight SAR and

iii) Scan SAR.

1.2.2. ***SAR system design:***

First, waveform generator is used to generate a Linear Frequency Modulated (LFM) pulse. Then, the signal is transmitted and received signal is collected by the receiver. After that the raw signal is passed through analog to digital converter and signal processing unit. To correct the error due to platform motion, range migration and range walk SAR system uses Inertial Measurement Units (IMU) and Global Positioning System (GPS) for recording history of flight.

1.2.3. ***Resolution of SAR:***

There are two types of resolution system in SAR i.e. range (slant range) and cross range (transverse range, azimuth track). Frequency modulated (chirp signal), a wideband transmitted signal, is used for generating high resolution in range in SAR system. For resolution in cross range dimension, coherent processing of target's electromagnetic scattering is done.

SAR gives both amplitude and phase information of the received signal. This helps in generating interferometric SAR (IFSAR).

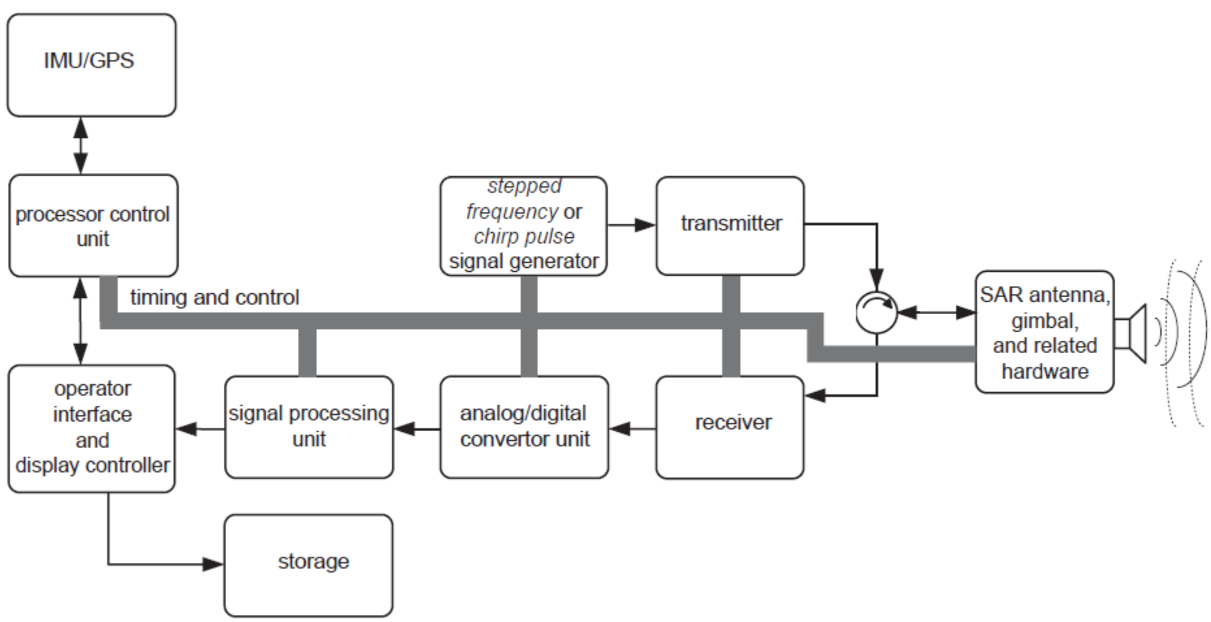


Fig.1.3: Block diagram of a SAR system.

Range resolution of SAR system:

$$\Delta r = \frac{c}{2B} \tag{1.1}$$

For finer resolution wide frequency bandwidth B is taken. With wide frequency waveform, pulse with short time duration is generated. Due to the short time duration enough energy could not supplied to the pulse. So very small portion of transmitted power is scattered back to radar and it's become difficult to sense received signal above the noise floor. To tackle this problem, an LFM or chirp waveform is utilized.

$$f_i(t) = f_o + B\frac{t}{T_o} \quad ; \qquad -\frac{T_o}{2} \leq t \leq \frac{T_o}{2} \tag{1.2}$$

Cross range resolution for SAR system:

$$\Delta y = \frac{R\lambda}{2D} \tag{1.3}$$

If a synthetic array of antenna is formed by moving a single antenna along a synthetic length D_{SA}

$$\Delta y = \frac{R\lambda}{2D_{SA}} \tag{1.4}$$

SAR image formation: The range compression and azimuth compression are applied independently to get the final SAR image from the 2D multi-frequency, multi-aspect scattered field data.

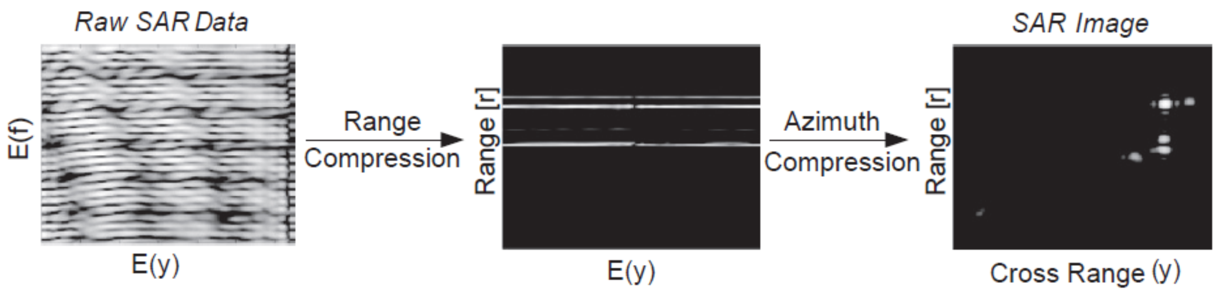


Fig.1.4. SAR image is formed by applying range compression and azimuth compression algorithms independently.

1.3. Objective of the thesis

- Signal processing through pulse compression using different correlation processors and analyzing their response to wideband and narrowband signals.
- To analyze the LFM signal taking in account time bandwidth product, Doppler Effect and effect of windows.
- Masking effect removal using matched filter and stretch processor.

1.4. ***Structure and chapter wise contribution of the thesis***:

- **Chapter 1**:- Synthetic aperture radar (SAR), SAR types, SAR design; SAR resolution is briefly explained in this chapter. The concept of range compression, pulse compression, matched filter, ambiguity function and azimuth compression are introduced in this chapter. The objective of the thesis and structure of the thesis chapter wise are outlined.
- **Chapter 2**:-Concepts about pulse compression, range resolution, matched filtering and stretch processing is discussed. Simulation results of matched filtering and stretch processor are generated. The outputs of these simulations are analyzed.

- **Chapter 3**:- Concept of LFM signal is outlined. Effect of windows, Doppler shift and time band width product is discussed. Simulations are results are analyzed and conclusion is out lined.
- **Chapter 4**:- Masking effect on radar signal is discussed. Removal of masking effect using matched filtering and stretch processor is explained and the simulation results are discussed.

1.4. Conclusion

Synthetic aperture radar (SAR), SAR types, SAR design, SAR resolution is briefly explained in this chapter. The concept of range compression, pulse compression, matched filter, ambiguity function and azimuth compression are introduced in this chapter. The objective of the thesis and structure of the thesis chapter wise are outlined. Objective of the thesis and structure of the thesis is outlined.

2.1 Introduction

There are two types of resolution system in SAR i.e. range (slant range) and cross range (transverse range, azimuth track). Frequency modulated (chirp signal), a wideband transmitted signal, is used for generating high resolution in range in SAR system. For resolution in cross range dimension, coherent processing of target's electromagnetic scattering is done.

Very short pulses are required for generating good resolution in a SAR. But the average power transmitting through the signal decreases with the use of short pulses, This in turn, affect the SAR's normal modes of operation drastically, mostly when we use multi-function and surveillance SARs. The average transmitted power is directly proportional to the receiver signal to noise ratio. It is often desirable to increase the pulse width is therefore increased (i.e., increase the average transmitted power) while, maintaining adequate range resolution at the same time. Pulse compression techniques are considered for implementing this process. Pulse compression generally helps us to generate the average transmitted power of a comparatively long pulse, while generating the range resolution with respect to a short pulse. Normally, there are two types of pulse compression technique: analyze analog and digital pulse compression techniques. Two analog pulse compression techniques are described in this chapter. One is correlation processing. This technique is mainly used for narrow band and some medium band radar operations. The other one is "stretch processing". This is normally used for very wide band radar operations. Digital pulse compression is also discussed briefly in this chapter.

2.2 Time bandwidth product (TBp)

Let us take into consideration a SAR system that employs a matched filter receiver. And we considered the matched filter receiver bandwidth be ***B***. Now, the noise power available inside the matched filter bandwidth will be given by

$$N_i = 2\frac{N_o}{2}B \qquad (2.1)$$

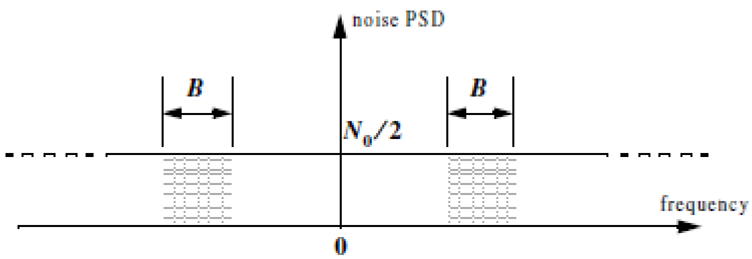


Fig 2.1 Input noise power

Here, the factor of 2 is used to account for both negative and positive frequency bands, as shown in the above figure. Now, the average input signal power over a pulseτ' duration is

$$S_i = \frac{E}{\tau'} \qquad (2.2)$$

Where, E = signal energy

Matched filter input SNR is given by

$$(SNR)_i = \frac{S_i}{N_i} \qquad (2.3)$$

Now the ratio between output peaks instantaneous SNR to the input SNR is:

$$\frac{SNR(t_o)}{(SNR)_i} = 2B\tau' \qquad (2.4)$$

Where, $B\tau'$ = time bandwidth product

Matched filter gain (compression gain) is proportional to the time bandwidth product. It is the factor, time band width product, by which the output SNR is increased over the input.

Radar equation for pulse radar:

$$(SNR)_{\tau_c} = \frac{P_t \tau_c G^2 \lambda^2 \sigma}{(4\pi)^3 R^4 k T_e F L} \qquad (2.5)$$

Where, P_i is the peak power, τ' is the pulse width, G is antenna gain, α is the target RCS, R is the range and k is the Boltzmann's constant.

In pulse compression process, first, modulated long pulses are transmitted. The SAR echoes are then compressed into very short pulses. In a transmitted pulse a large no of very small/short sub pulses are present. All these short pulses have the anticipated compressed pulse width. If we are not changing the radar parameters and operating with the same transmitted signal then SNR will also remain constant independent of the change in signal bandwidth. So while using pulse compression we are able achieve very high quality resolution, without changing the pulse width, just by increasing the bandwidth. Range resolution and signal bandwidth depends inversely on each other.

2.3. Analog pulse compression

It consist of mainly two types i.e. correlation processor and stretch processor.

2.3.1 Correlation processor

Here, pulse compression is achieved by applying frequency modulation to a long pulse at transmission. Then the received signal is compressed using a matched filter receiver. Here large compression ratio is achieved by utilizing long pulses and wideband LFM modulation.

2.4 **Matched filter**

It is a linear invariant system. Its output is mainly described mathematically. Matched filter output is the convolution between input and the impulse response.

$$Y\ (T) = s\ (T) * h\ (T) \qquad (2.6)$$

Where, s (T) is the input signal, h (T) is matched filter impulse response and $*$ is the symbolic representation of convolution operator.

Now, by using Fourier transform property:

$$FFT\{s\ (T) * h\ (T) = S\ (f) \cdot H\ (f) \qquad (2.7)$$

After the proper sampling of both the signal y (t), the compressed signal is can be obtained from

$$y = FFT^{-1}\{S.H\} \qquad (2.8)$$

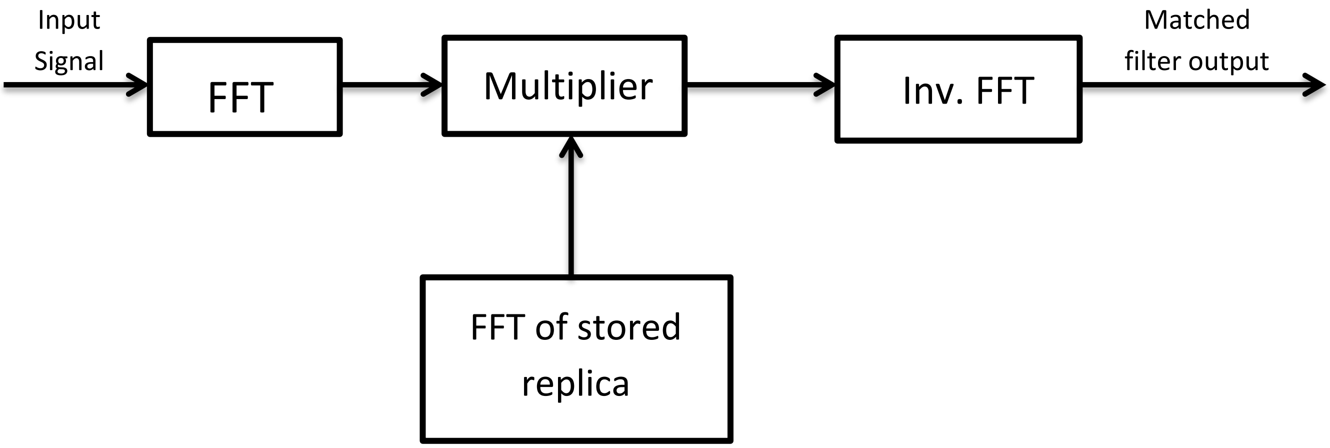


Fig 2.2: Computation of matched filter output using an FFT

While using pulse compression we always try to achieve maximum pulse compression ratio and highly reduced side lobe levels. Noise and jammers are located at the side lobe, interfere and disturb the received signal in the main lobe. So, highly reduced side lobe levels are avoided. Weighting functions (windows) are used eliminate side lobe levels. But this process leads to loss in resolution of main lobe and decrease in SNR (i.e. loss in peak value).

Table 2.1: Simulation parameters

Bandwidth	1e8 Hz
Scattered range(In meters)	10, 75 ,120
Range cross section	1 , 2 , 1
Time period	5 μs

2.5 Simulation Results

The following simulations by using mat lab 7.10.0(R2010a) for 3 targets using matched filter

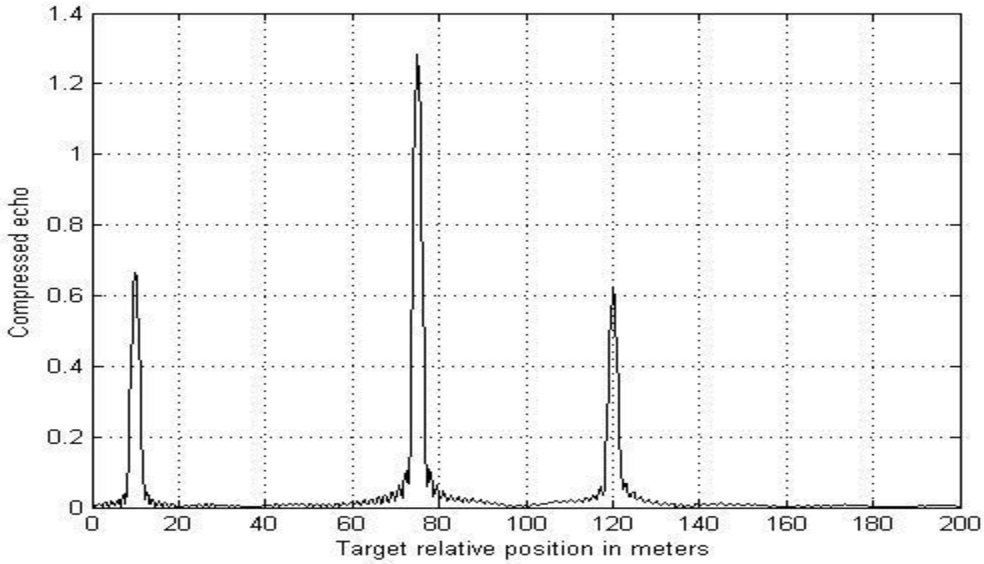


Fig 2.3: Match filter output using Kaiser Window.

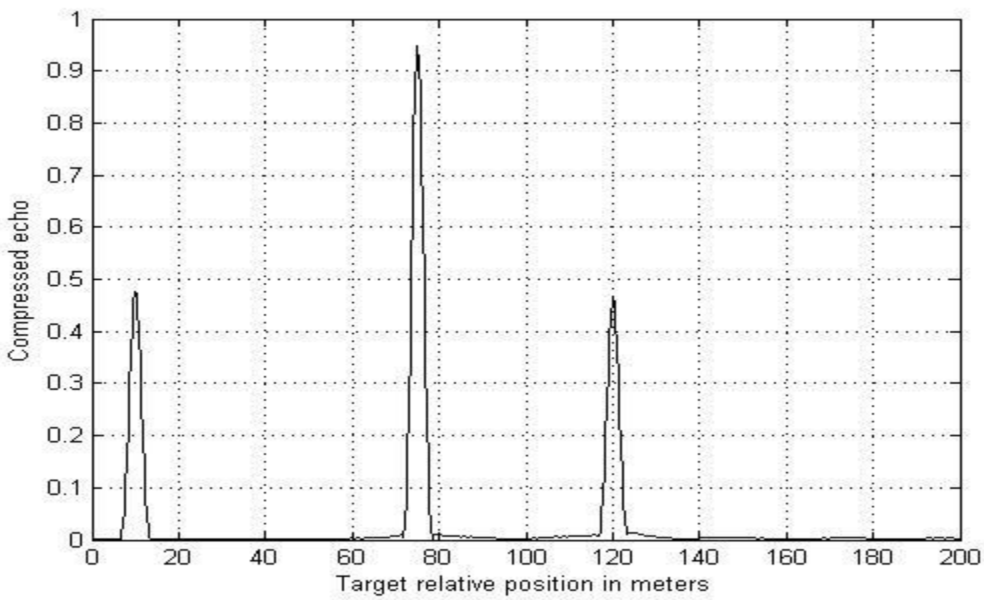


Fig 2.4: Match filter output using Chebyshev window

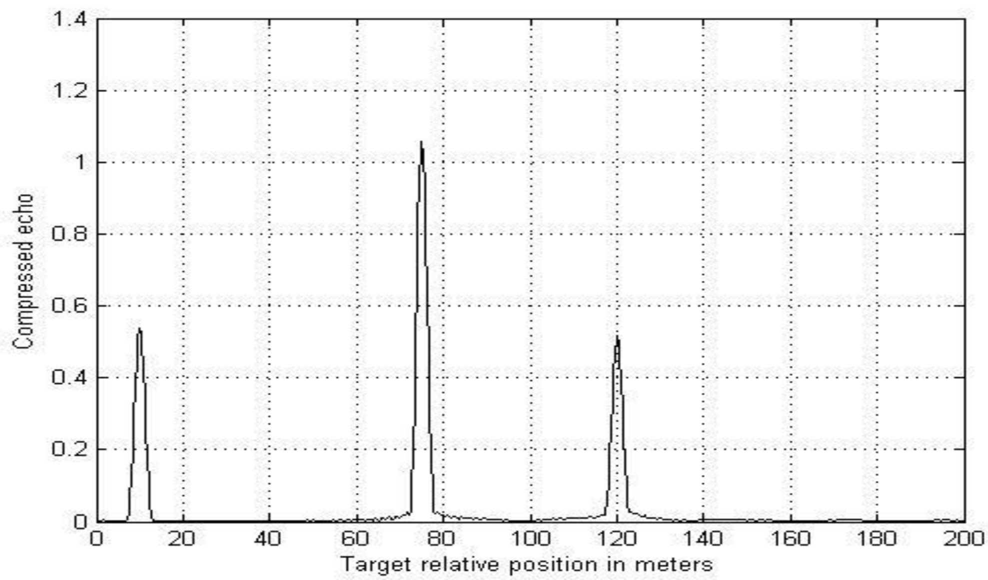


Fig 2.5 Match filtering using hamming window

2.6 Stretch Processing

Stretch processing is generally used to process extremely high bandwidth LFM waveforms. It is also known as active correlation this processing technique consists of the following steps: in the beginning the returns from the radar are mixed with a replica (reference signal) of the transmitted waveform. This is then followed by Low Pass Filtering (LPF) and coherent detection. Then we do Analog to Digital (A/D) conversion and in the end, we use Narrow Band Filters for extracting tones proportional to target range, as stretch processing effectively converts time delay into frequency all returns from the same range bin produce the same constant frequency. A block diagram for a stretch processing receiver is shown in fig. 2.2. The replica which is taken is a LFM waveform which has the same LFM slope as the transmitted LFM signal. It exist over the duration of the radar “receive-window,” This Receive window of the radar is computed by taking the difference between the maximum and minimum range of the radar. The radar transmitted signal is given by the following equation

$$s(t) = \cos\left(2\pi\left(f_0 t + \frac{B}{2\tau'} t^2\right)\right) \qquad 0 \leq t \leq \tau' \tag{2.10}$$

Where $f_0$start frequency of the LFM signal. The reference signal is given by

$$s_{ref} t = 2\cos\left(2\pi\left(f_0 t + \frac{B}{2\tau'} t^2\right)\right) \qquad 0 \leq t \leq T_{rec} \tag{2.11}$$

Where T_{rec} is the received window and is given by

$$T_{rec} = \frac{2(R_{max} - R_{min})}{c} \tag{2.12}$$

Here we have assumed that there is a point scattered at a range rather received signal is given by

$$s_r(t) = \mathrm{a}\cos[(2\pi(f_0(t - \Delta\tau) + \frac{\mu}{2}(t - \Delta\tau)^2] \tag{2.13}$$

Where ‘a’ is proportional to target range cross section, antenna gain, and range attenuation.

Here, $\frac{2R}{C}$ is the time delay.

The output of the mixer is the multiplication of the received signal and the reference signal. Then low pass filtering of the signal is done and the output is given and as we know that $\tau' \ll \frac{2R}{C}$ so the above equation is approximated, hence proper sampling of LFM signal and then taking FFT of sampled sequence results in a peak at some frequency which indicates the presence of a target at a range

$$R_0 = \frac{f_1 c\tau\prime}{2B}$$

Table 2.2 Simulation parameters

Bandwidth	1e10 Hz
Scattered range	10, 75 ,120
Range cross section	1 , 2 , 1
Time period	50 ns

2.7 Simulation Results

The following simulations by using mat lab 7.10.0(R2010a) for 3 targets using stretch processing

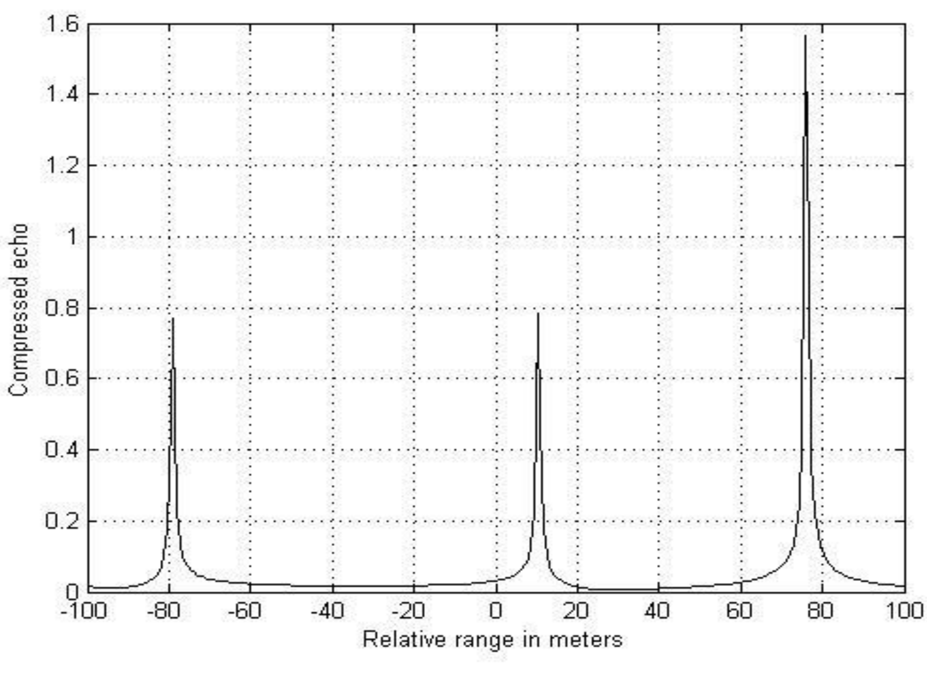


Fig 2.6 Stretch processing output

2.8 Conclusion

In Pulse compression of radar signal to get the resolved range coming from three targets, two of its techniques were analyzed, Matched Filtering and Stretched Processing. It is found that matched filter improves the SNR (signal to noise ratio) and attenuates the side lobes. While in Stretch processing complete elimination of side lobes can be seen. Matched filter is mainly suitable for narrow band signal and shows complexity while used for wideband signals. On the other hand stretch processing is suitable for wideband signals. Wideband signals are easily processed by stretched processing technique than matched filtering technique, which takes a lot of time for processing wideband signals and even not compatible for wideband signals beyond a certain limit.

3.1 Introduction:

Pulsed Radar has a range resolution of

$$\Delta r = \frac{C}{2B} \tag{3.1}$$

Where, B=bandwidth of the pulse.

C=velocity of light in vacuum.

The Fourier theory says, the frequency bandwidth, Bob the signal is inversely proportional to pulse duration in the following manner as

$$B = \frac{1}{\tau} \tag{3.2}$$

This shows that the range resolution is directly promotional to its pulse duration in the following manner

$$\Delta r = \frac{\tau C}{2} \tag{3.3}$$

Hence for a good range resolution we need to keep the duration of a pulse small. But the problem with short pulses is that it will not be possible to put enough energy on this pulse. Sufficiently wide pulse cannot achieve a wide bandwidth because if we are using an un-modulated pulse having constant frequency, its time duration will be very small and will not be possible to enough energy on it. This problem can be solved by using a modulated pulse of sufficient duration so that it will provide the required bandwidth for the operation of radar. The most common waveform used for this purpose is the LFM (Linear frequency modulated) pulse, also known as the chirp pulse. .This waveform repeats itself in every interval called PRI (pulse repetition interval) or also known as *pulse repetition period* .The LFM signal taken over here is given by:

$$s(t) = e^{-i2\pi\left(\frac{B}{T}\right)t^2} \qquad -\frac{T}{2} \le t \le \frac{T}{2} \tag{3.4}$$

Where, B=bandwidth of the signal and T=time period

3.2 Effect of windows on side lobe reduction of LFM signal

Window functions are mathematical expressions which are having a particular value in an interval and have no value outside it. One example of a window is a rectangular window that is constant inside the interval and zero elsewhere .It describes the shape of its graphical representation. When the window is multiplied by another function or waveform, their product is also zero-valued outside the interval. And all that the part where they overlap, only exists.

The LFM signal was introduced to following 8 windows:

3.2.1 Rectangular window

The rectangular window has one value over its length. It has the following equation.

$$w(n) = 1.0 \text{ for } n = 0, 1, 2, \ldots, N-1 \tag{3.5}$$

Where N is the length of the window and w is the window value.

A rectangular window when applied just limits the signal to within a finite time interval. Therefore it is equivalent to not using any window.

3.2.2 Hanning window

The Henning window resembles half a cycle of a cosine wave. The Henning window has the following equation

$$w(n = 0.5 - 0.5cos\frac{2\pi n}{N} \qquad (3.6)$$

For n = 0, 1, 2, N – 1

Where N is the length of the window and w is the window value. `

3.2.3. Hamming window

It is a modified version of the Henning window. It has a shape like a cosine wave. Hamming window is described by the following window.

$$w(n) = .54 - .4cos\frac{2\pi n}{N} \qquad (3.7)$$

For n = 0, 1, 2, N – 1

Where N is the length of the window and w is the window value.

3.2.4 Kaiser window

The Kaiser window is a flexible smoothing window whose shape can be modified by adjusting the beta (β) parameter. w = Kaiser (L, beta) returns an L-point Kaiser window in the column vector w. beta is the Kaiser Window β parameter that affects the side lobe attenuation of the Fourier transform of the window.

3.2.5. Blackmannharris window

It does the window sampling by using *'sflag'*. This can either be 'periodic' or 'symmetric' (the default). The 'periodic' flag is generally used for Fourier transform, so that spectral analysis can be done. Its equation is given by:

$$a_0 - a_1 \cos\left(\frac{2\pi n}{N}\right) + a_2 \cos\left(\frac{2\pi 2n}{N}\right) - a_3 \cos\left(\frac{2\pi 3n}{N}\right) \qquad (3.9)$$

Where $\frac{-N}{2} \le n \le \frac{N}{2}$ and the window length is given by L=N+1

3.2.6 Flattop window

Flat Top window has very low pass band ripple (< 0.01 dB) they are generally used for calibration purposes. They have a bandwidth equivalent to approximately 2.5 times wider than a Han window. Its equation is given by:

$$w(n) = a_0 - a_1 \cos\left(\frac{2\pi n}{N}\right) + a_2 \cos\left(\frac{4\pi n}{N}\right) - a_3 \cos\left(\frac{6\pi n}{N}\right) + a_4\left(\frac{8\pi n}{N}\right) \qquad (3.10)$$

Where $0 \le n \le N$ and w (n) =0elsewhere and the window length is L=N+1.

3.2.7 Nuttall window

$$\omega(n) = a_o - a_1 \cos\left(\frac{2\pi n}{N-1}\right) + a_2 \cos\left(\frac{4\pi n}{N-1}\right) - a_3 \cos\left(\frac{6\pi n}{N-1}\right) \qquad (3.11)$$

3.2.8 Triangular window

$$w(n) = 1 - \left|\frac{n - \frac{N-1}{2}}{\frac{N+1}{2}}\right| \qquad (3.12)$$

3.2.9 PSR

It is called peak to side lobe ratio (PSR). It is given by

$$PSR = 10log_{10} \frac{Peak\ sidelobe\ Power}{mainlobe\ power} \quad (3.13)$$

3.2.10 Simulations results

The LFM signal was taken for two different time bandwidth product (TBp.=50 and TBp=500)and was passed through 8 windows for each time bandwidth product and mat lab simulation for each time bandwidth product with 8 different windows was taken and shown below

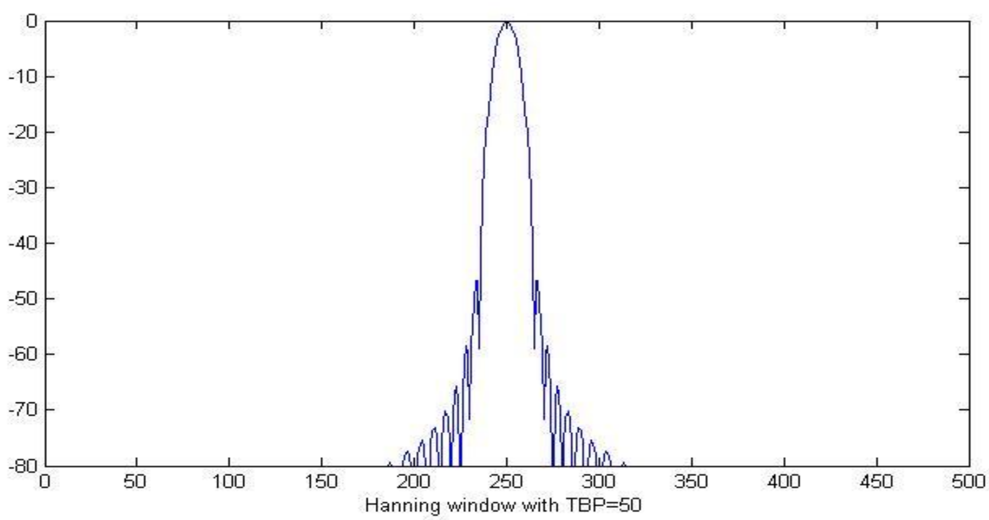


Fig 3.1: LFM signal with TBp=50 and passed through hanning window

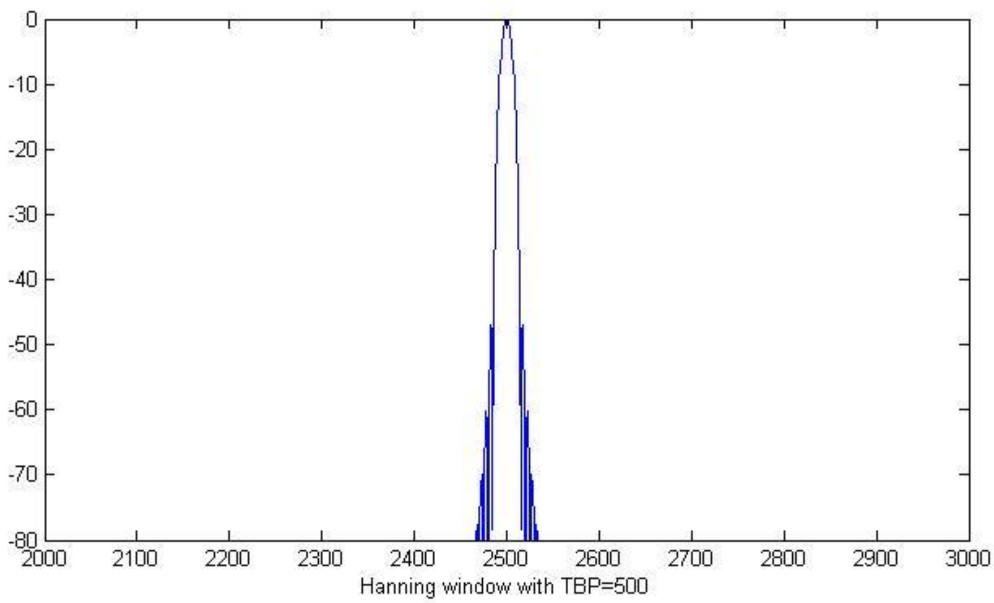


Fig 3.2: LFM signal with TBp=500 and passed through hanning window

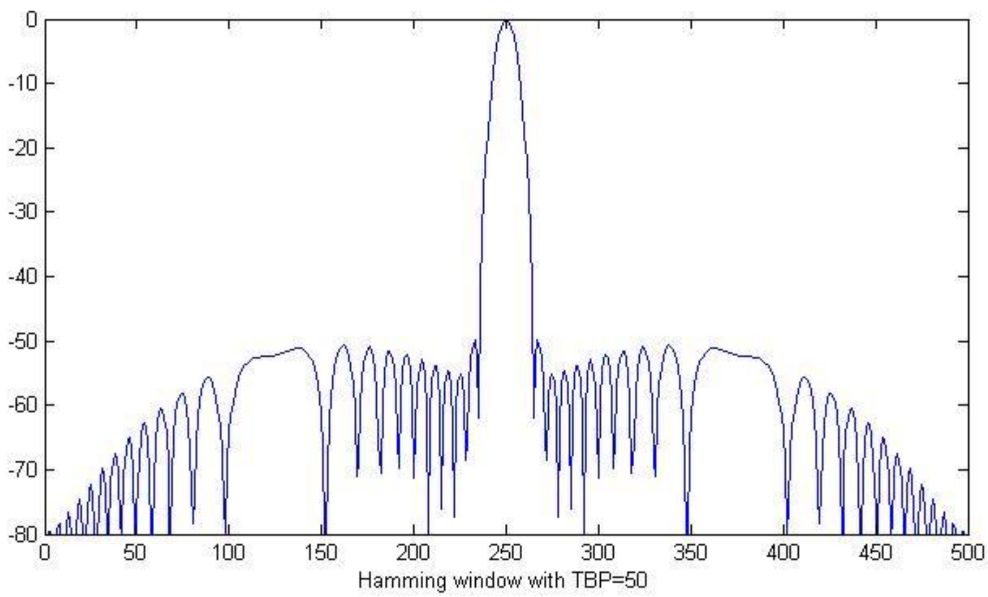


Fig. 3.3: LFM signal with TBp=50 and passed through hamming window

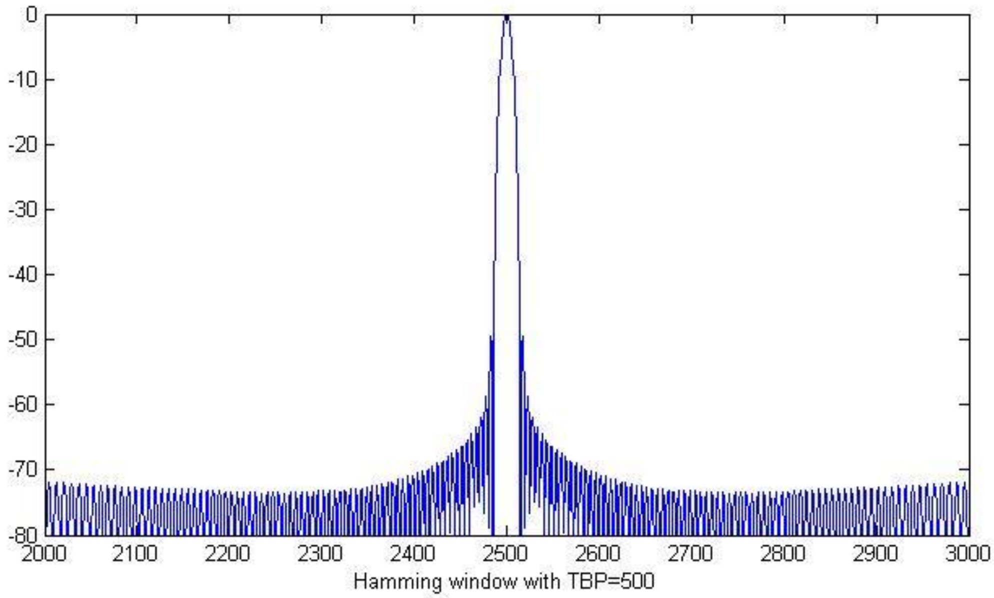


Fig 3.4 LFM signal with TBp=500 and passed through hamming window

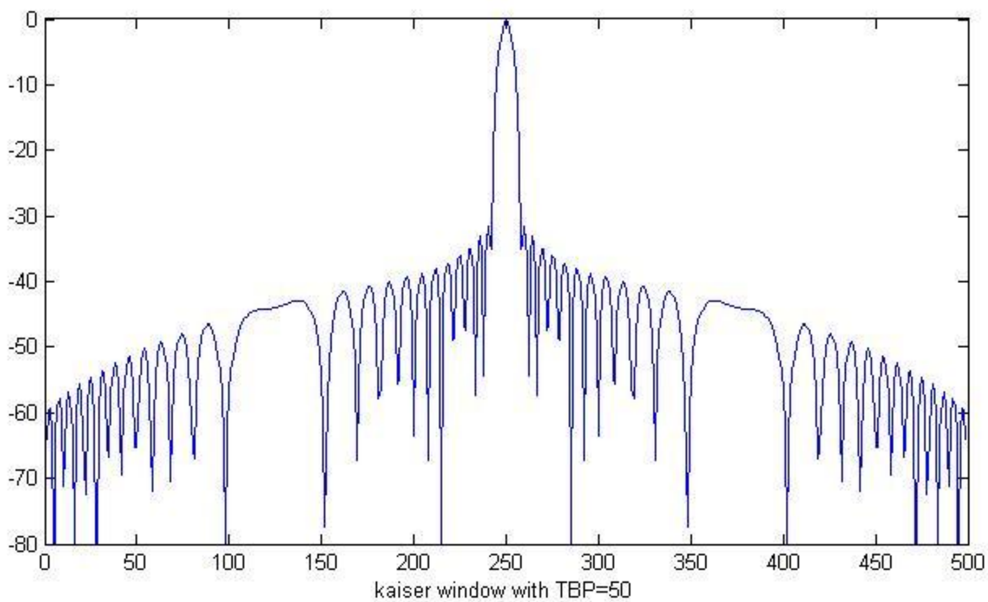


Fig 3.5 LFM signal with TBp=50 and passed through kaiser window

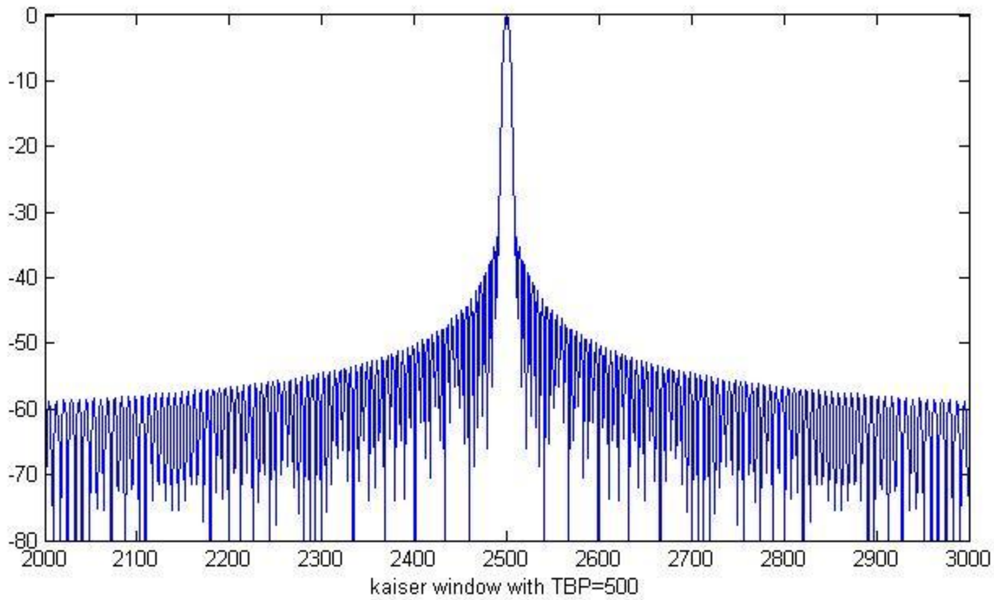


Fig 3.6: LFM signal with TBp. =500 and passed through Kaiser Window

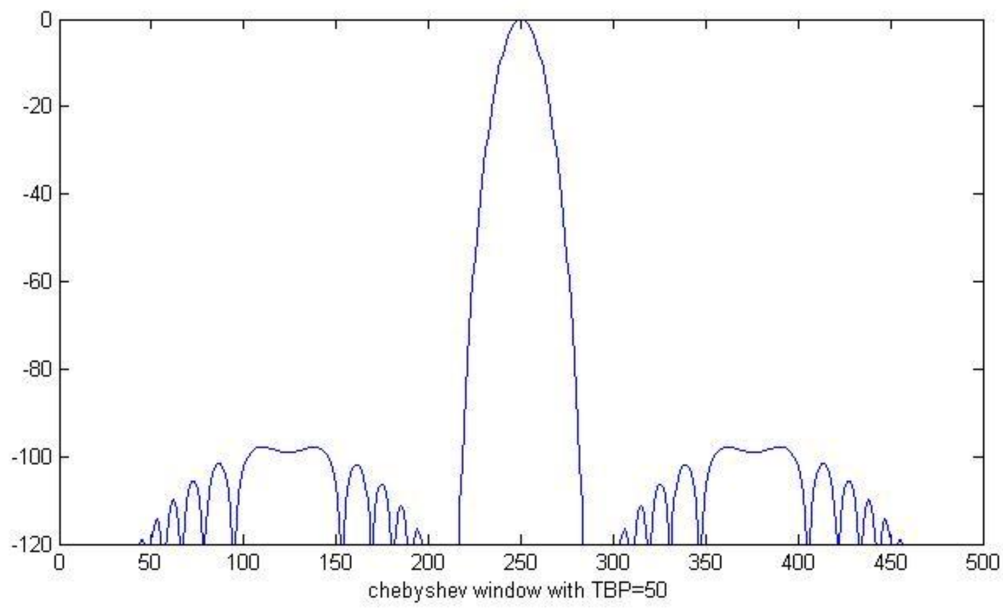


Fig 3.7: LFM signal with TBp=50 and passed through Chebyshev window

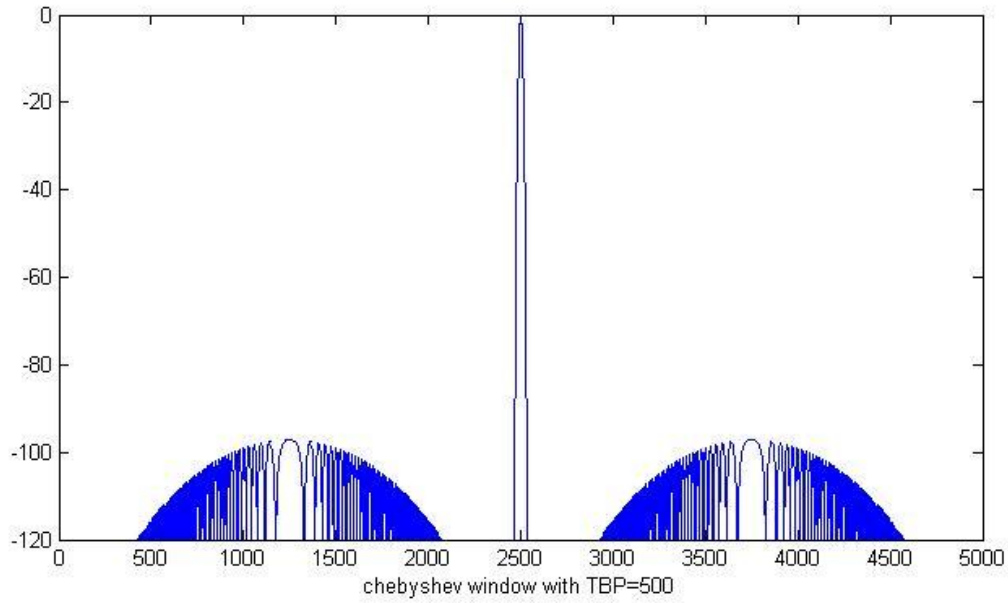


Fig 3.8: LFM signal with TBp=500 and passed through Chebyshev window

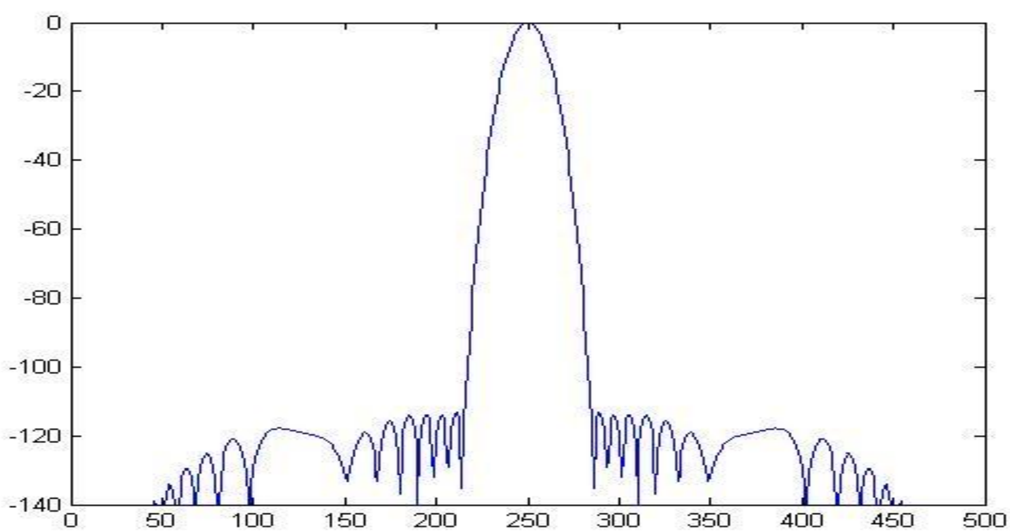


Fig 3.9: LFM signal with TBp=50 and passed through Blackmanharris window

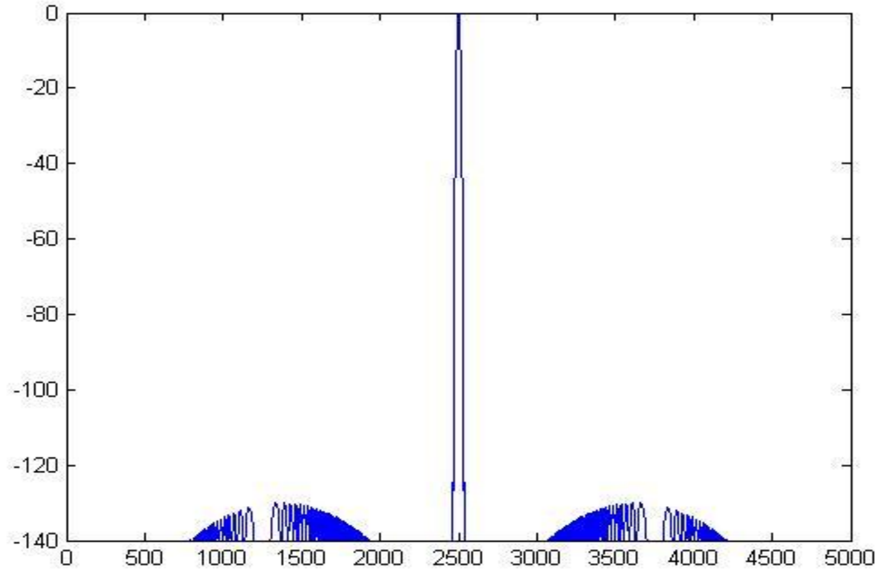


Fig 3.10: LFM signal with TBp=500 and passed through Blackmanharris window

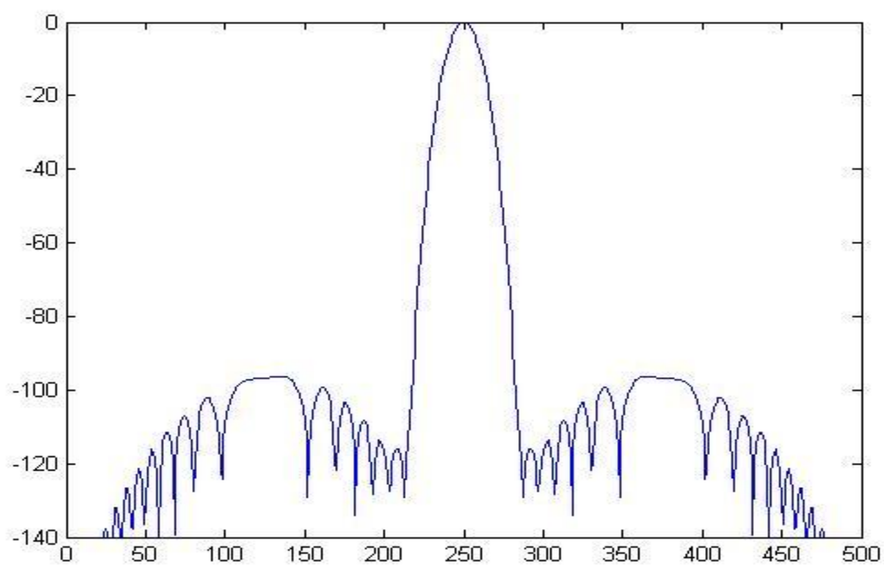


Fig 3.11: LFM signal with TBp=50 and passed through Nuttal window

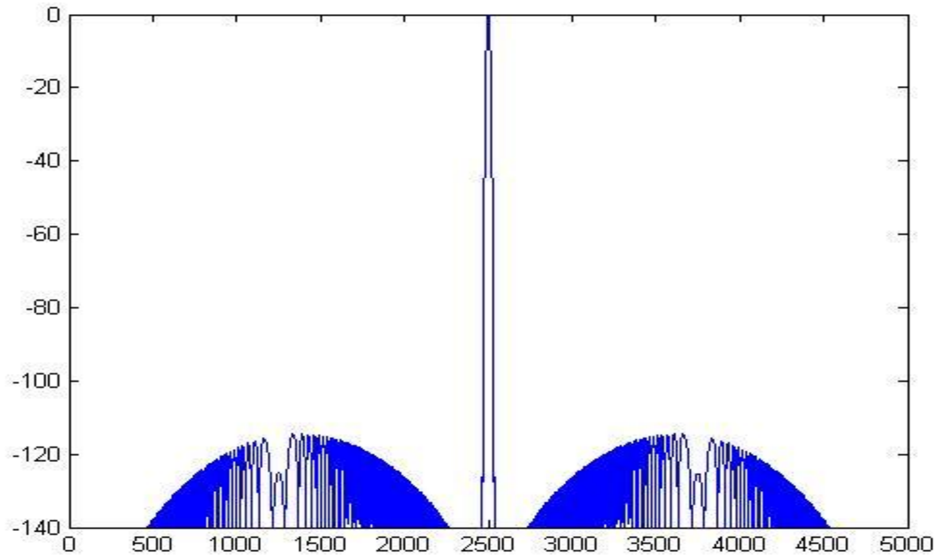
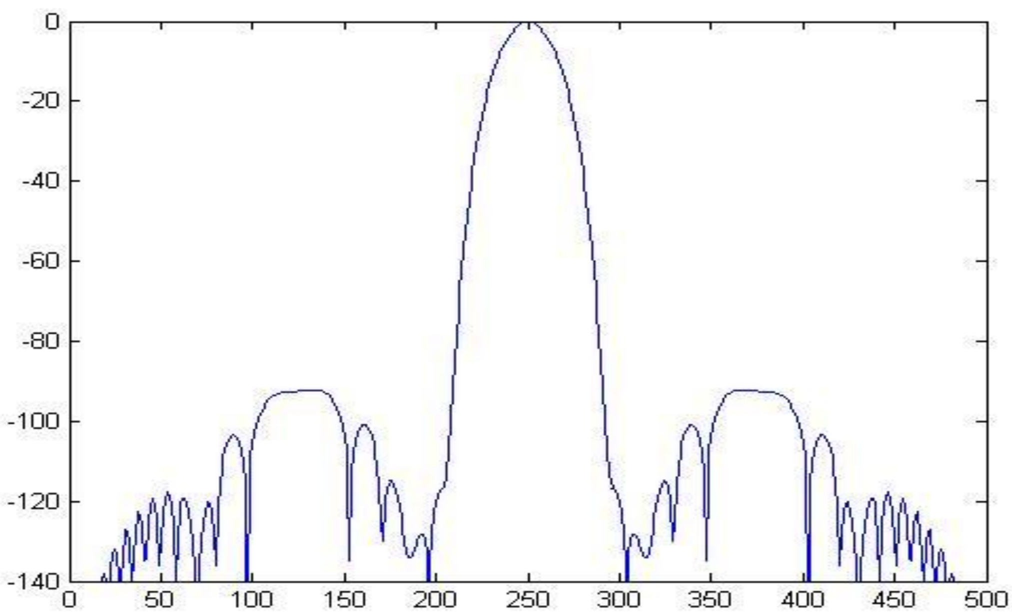


Fig 3.12: LFM signal with TBp=500 and passed through Nuttal window



. Fig 3.13 LFM signal with TBp=50 and passed through Flattop window

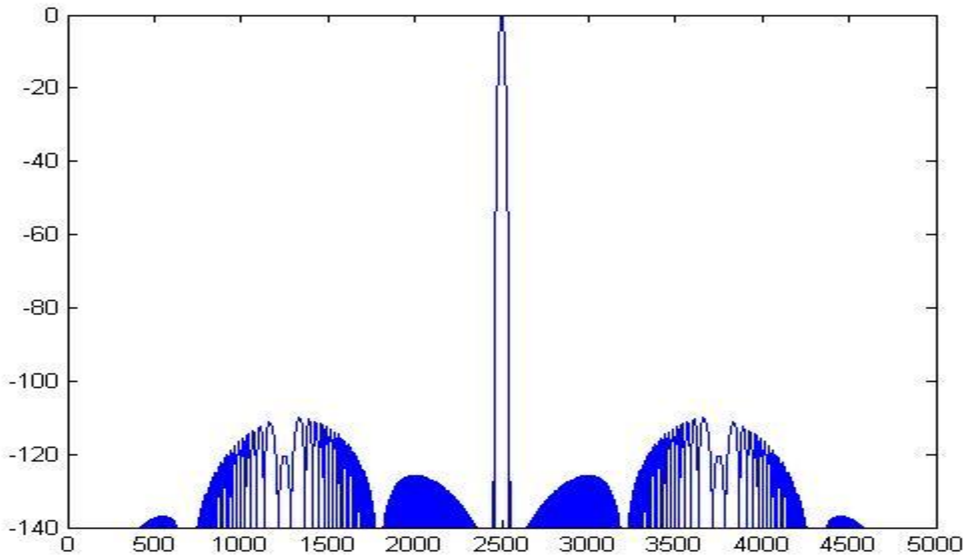


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Fig 3.14: LFM signal with TBp=500 and passed through Flattop window

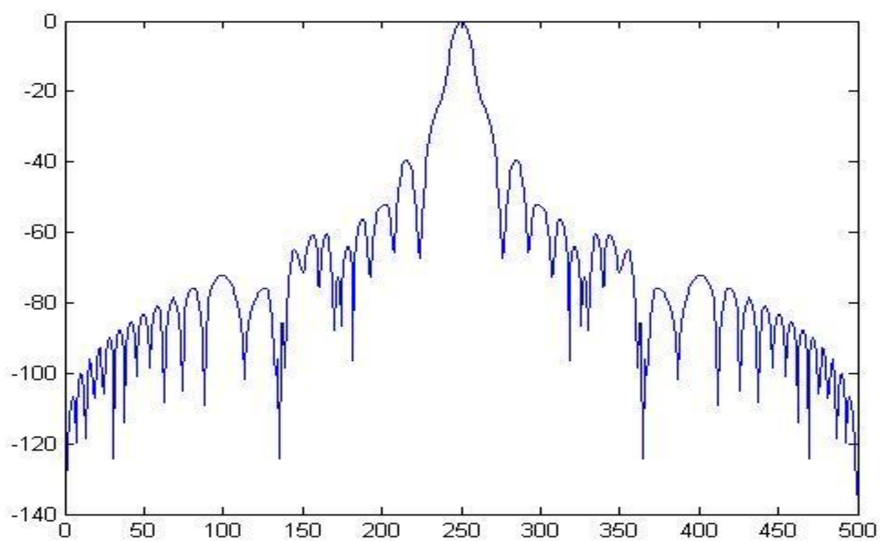
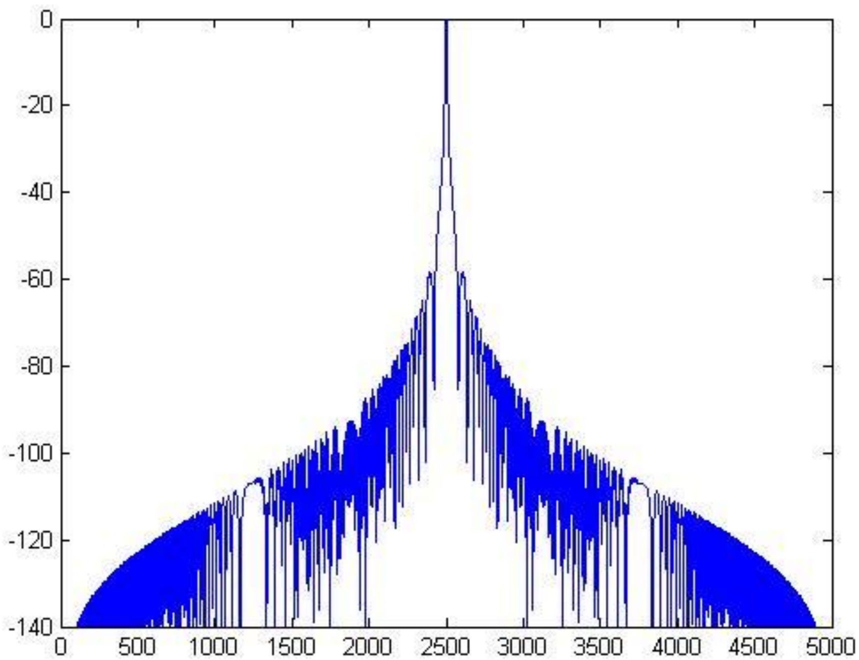


Fig 3.15: LFM signal with TBp = 50 and passed through Triangular window



. Fig 3.16: LFM signal with TBp=500 and passed through Triangular window

Table 3.1 PSR value w.r.t time bandwidth product in different windows

WINDOW	**OUTPUT WITH TBP=50(PSR)**	**OUTPUT WITH TBP=500**
Henning	-46	-46.5
hamming	-50	-50.5
Kaiser	-33	-34
Blackmanharris	-118	-130
Nuttall	-98	-118
Flattop	-95	-110
Triangle	-40	-60

3.2.11 Result

From the above we draw the following conclusions:-

1. Blackmanharris window has the best Side lobe reduction where rectangular window has the worst Side lobe reduction.
2. The Side lobe reduction is least affected by time bandwidth product. In some windows, they are with a little bit of variation in generalized hamming window. Whereas in higher order generalized cosine windows like Blackmanharris, Nuttall and flattop window PSR is large with the change in bandwidth time product.
3. In Kaiser Window the 'β' parameter provides a relation between reduction in side-lobe level and main-lobe width. Larger values of β gives lower side-lobe levels, but at the same time it widens the main lobe. But widening the main lobe results in reduction in frequency resolution when the window is used for spectrum analysis.

3.3 Doppler effects on LFM signals

RADAR signals encounter Doppler Effect when there is a moving target. Radars use the Doppler frequency to extract target radial velocity. Here we will observe the effect of Doppler frequency on LFM signal. FM signal without adding Doppler frequency is given below.

$$s_t = e^{(2i\pi((\frac{b}{taup})(t^2)))} \tag{3.14}$$

LFM signal with A Doppler shift is given by

$$s_{t_2} = e^{(2i\pi((\frac{b}{taup})t^2+(f_d)t))} \tag{3.15}$$

Where, f_d =Doppler frequency

Table 3.2 Results of Doppler shift effect without window is given below

$PSR\ with\ \frac{f_d}{B} = 0.1$	$PSR\ with\ \frac{f_d}{B} = 0.2$	$PSR\ with\ \frac{f_d}{B} = 0.3$
-13.8	-13.2	-13.2

3.3.1 Simulation Results

The LFM signal was added with varying Doppler shift (f_d) and its effect on passing the LFM signal with four windows was studied

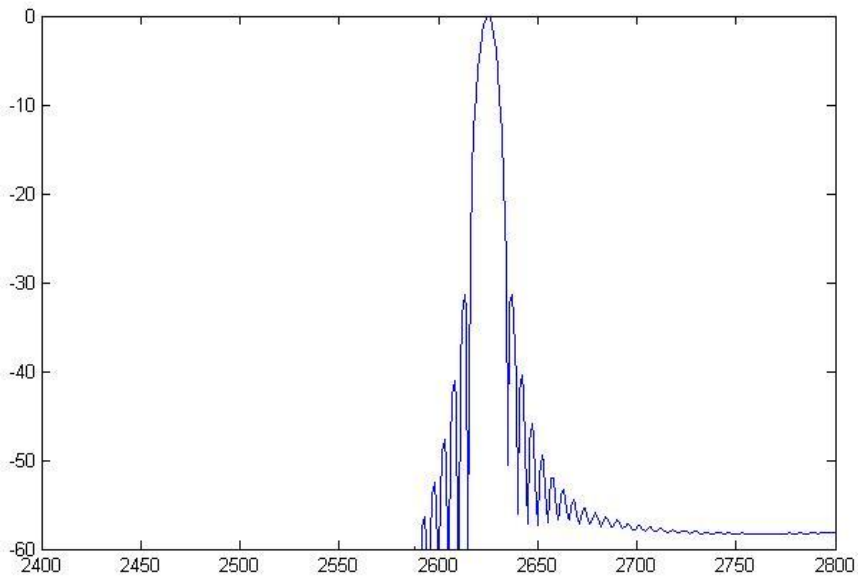


Fig3.17. LFM signal with fd/B=0.1 passed though Hanning window

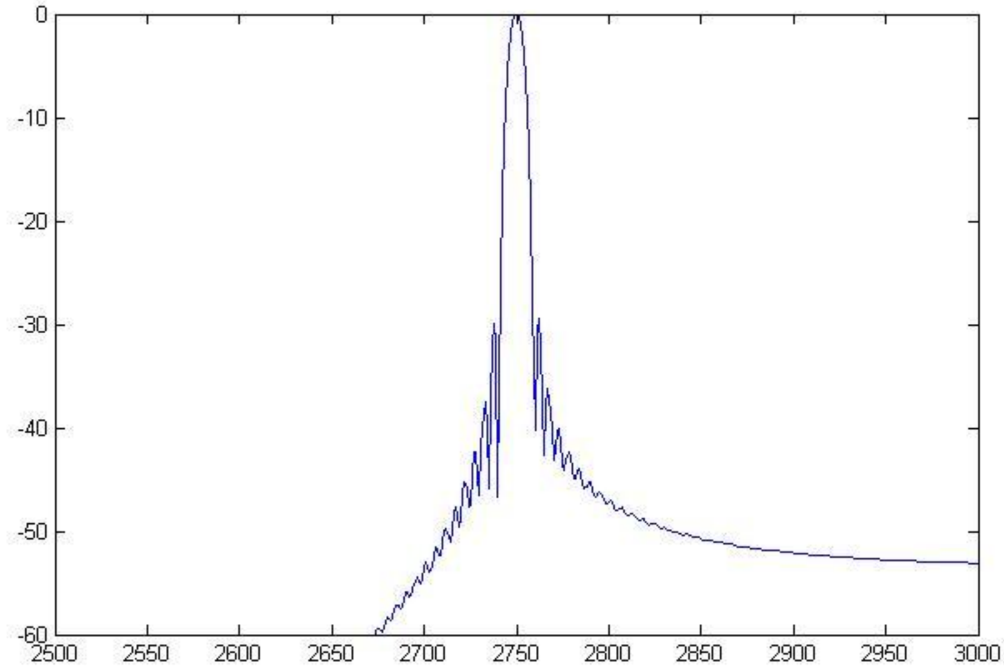


Fig3.18. LFM signal with fd/B=0.2 passed though Hanning window

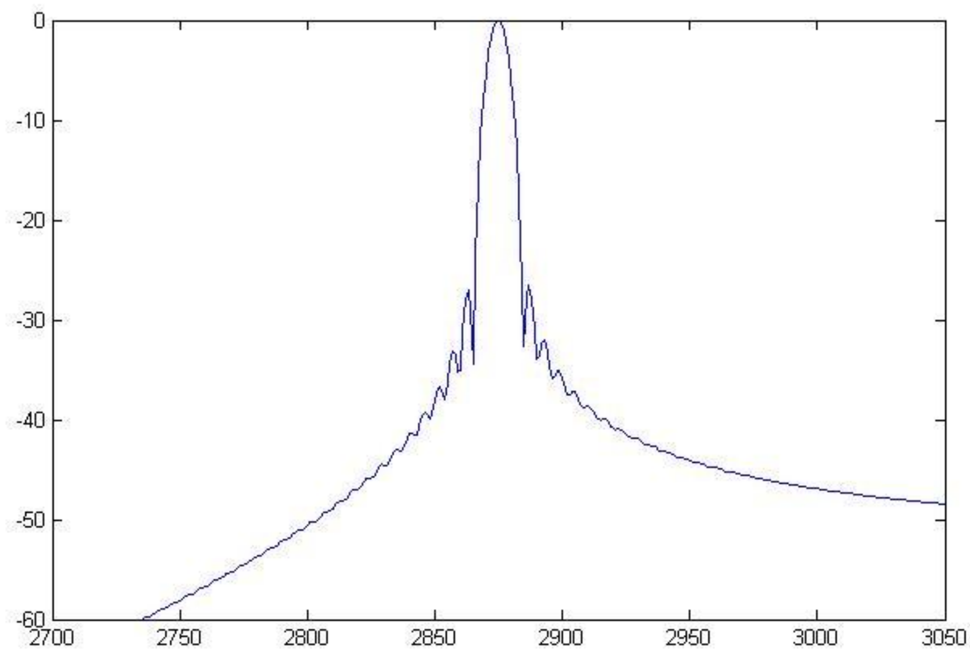


Fig3.19. LFM signal with fd/B=0.3 passed though Hanning window

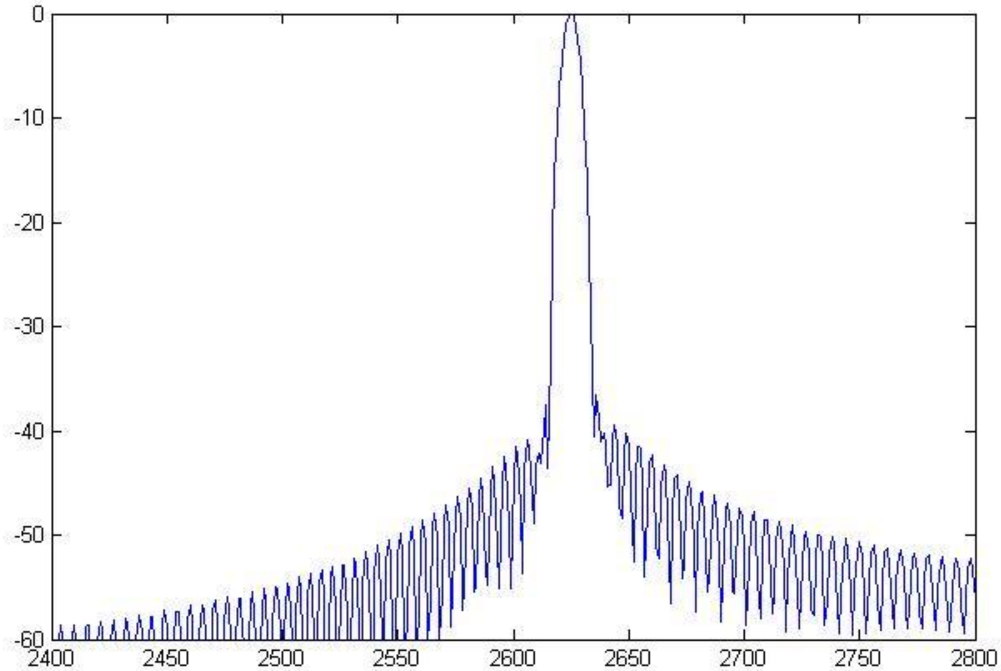


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Fig3.20. LFM signal with fd/B=0.1 passed though Hamming window

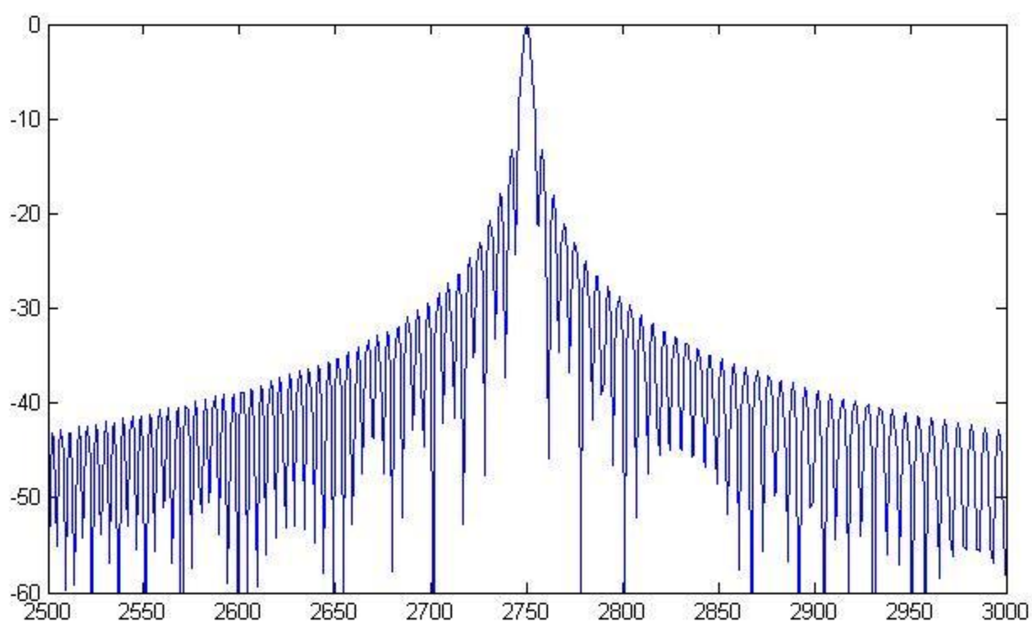

Fig 3.21: LFM signal with fd/B=0.2 passed though Hamming Window

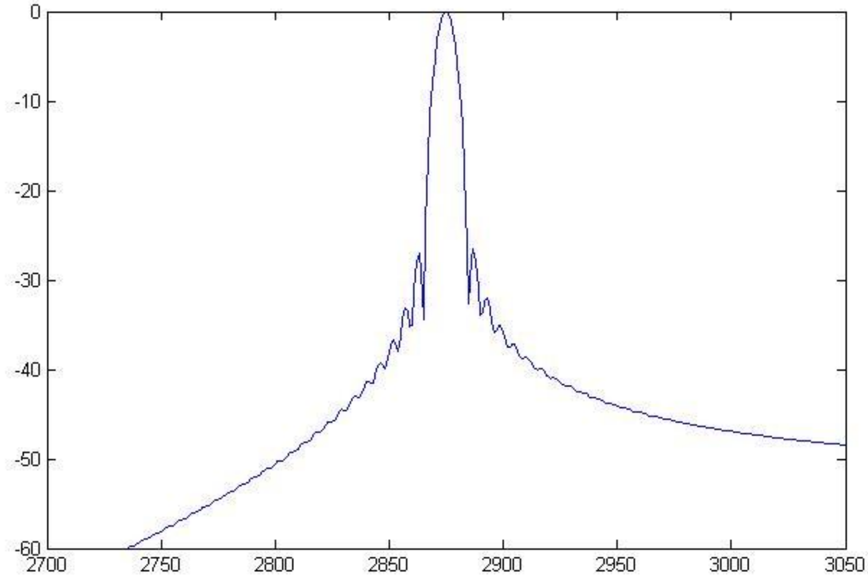

Fig 3.22: LFM signal with fd/B=.3 passed though hamming window

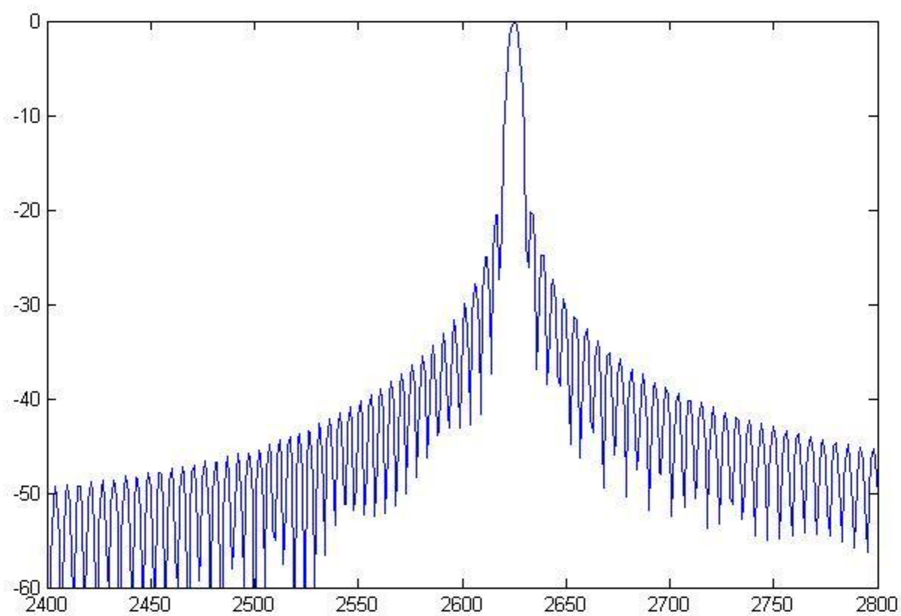

Fig 3.23: LFM signal with fd/B=0.1 passed though Kaiser Window

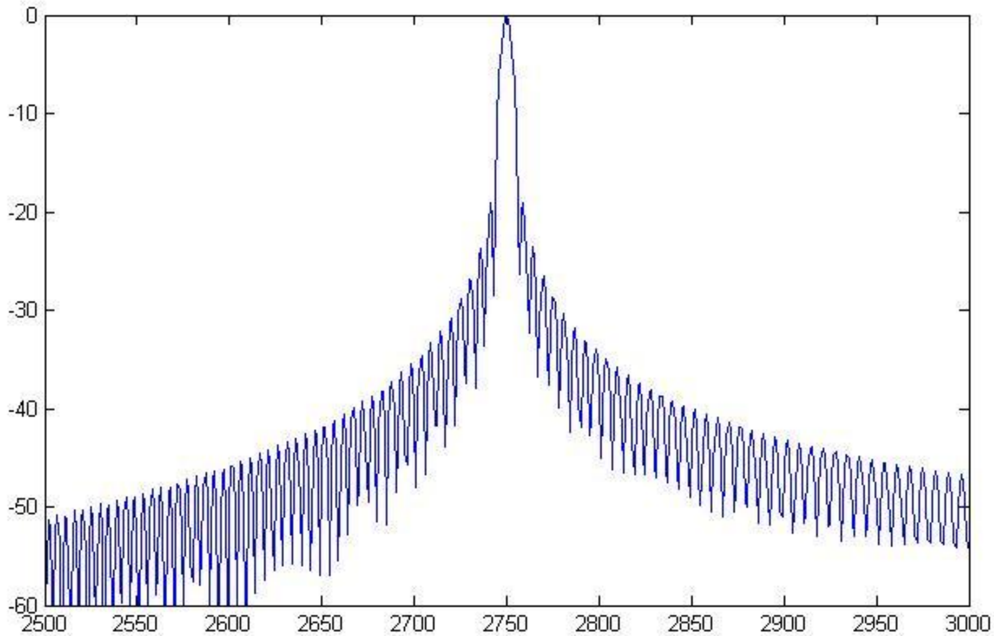

Fig 3.24: LFM signal with fd/B=0.2 passed though Kaiser Window

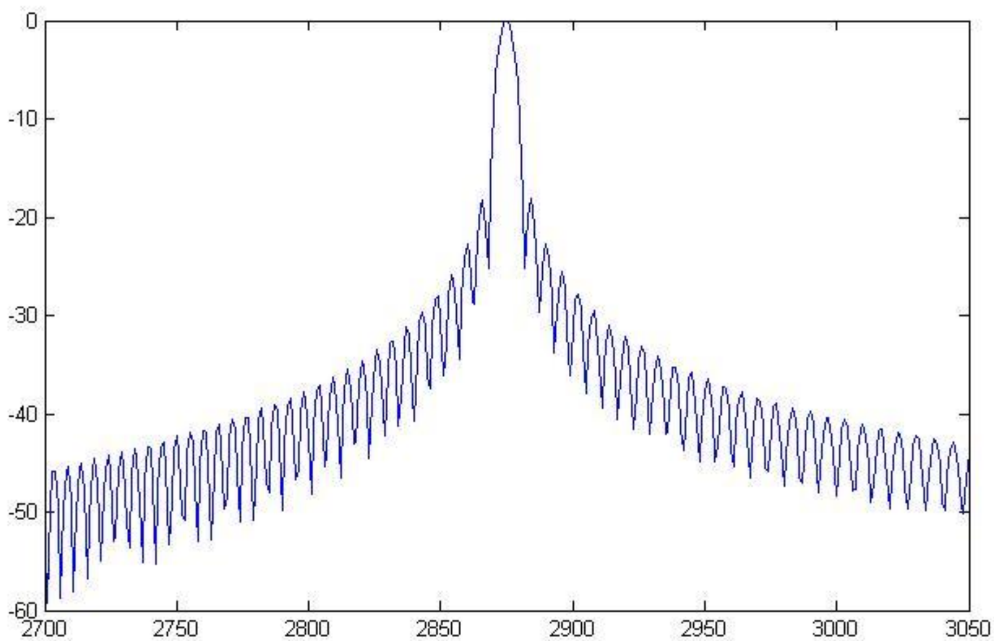

Fig 3.25: LFM signal with fd/B=0.3 passed though Kaiser Window

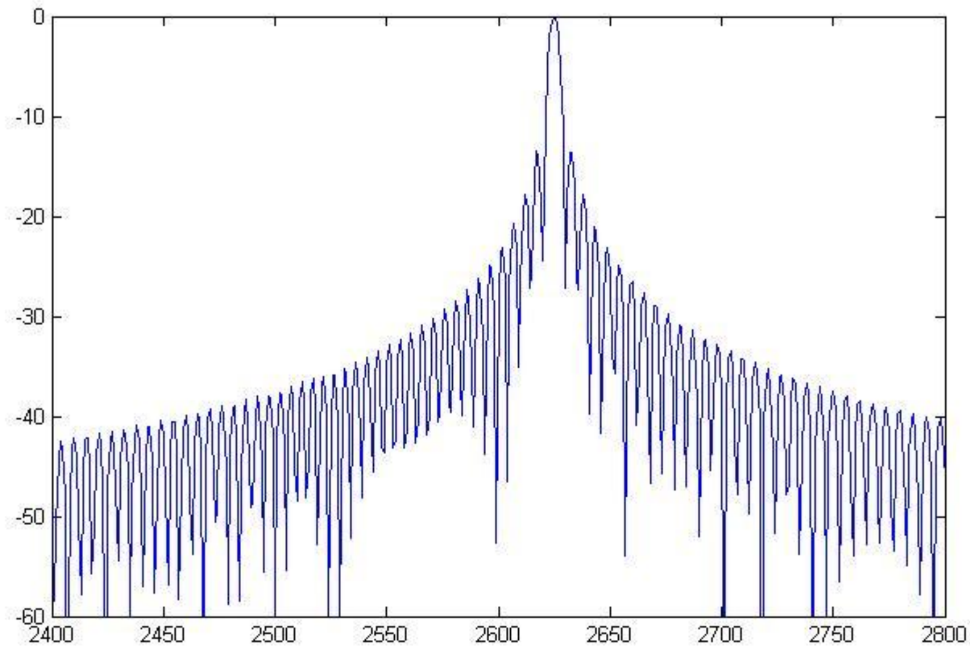

Fig 3.26: LFM signal with fd/B=0.1 passed though Rectangular window

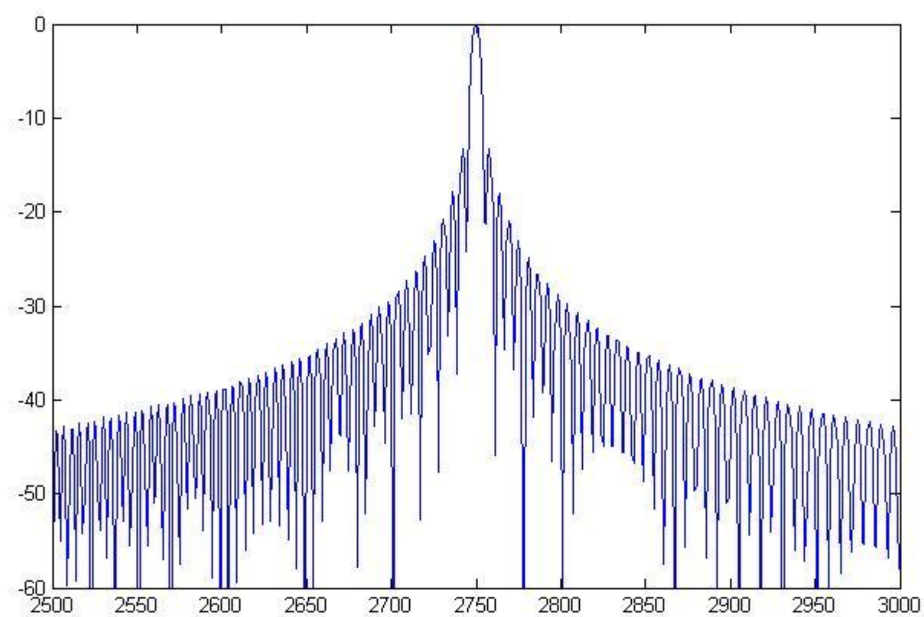

Fig3.27 LFM signal with fd/B=0.2 passed though Rectangular window

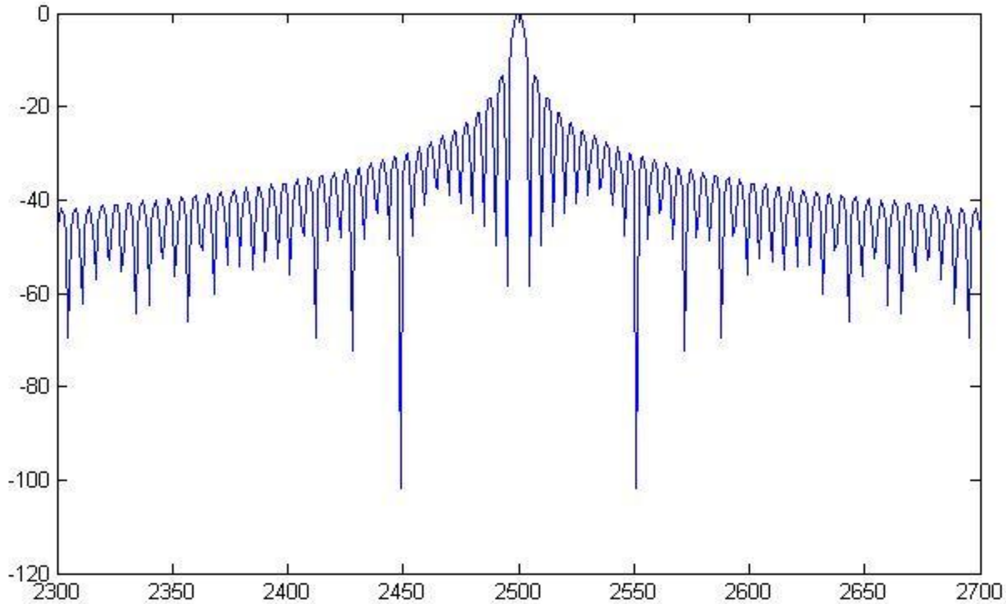

Fig 3.28: LFM signal with fd/B=0.3 passed though Rectangular window

Table 3.3 Results of Doppler shift on LFM signal when passed through four different windows

window	PSR with $\frac{f_d}{B} = 0.1$	PSR with $\frac{f_d}{B} = 0.2$	PSR with $\frac{f_d}{B} = .3$
Hanning	-32	-30	-26
Hamming	-36	-32	-28
Kaiser	-22	-19	-18
Rectangular	-13	-12	-10

3.4 Result

1. The Doppler shift effect is not felt so much in the absence of window and the PSR for three different values of fd/B(where B=bandwidth).The PSR remains almost the same for the three values.

2. The Doppler shift effect is then checked by passing the signal through 4 windows (Hanning, Hamming, Kaiser, and Rectangular) and we found out that PSR decreased with increase in Doppler Effect.

3.5 Conclusion

LFM (Linear Frequency Modulated) signals are the most popularly used signals for radar signal processing. In this chapter LFM signal was analyzed taking into account time bandwidth product and Doppler effect For analyzing effect of time bandwidth product on LFM signal, it was passed with nine windows and time bandwidth product was taken 50 for 1st case and time bandwidth product was taken 500 for 2nd case. It is observed that the PSR (Peak to Signal Ratio) increases with increase in time bandwidth product. Next LFM signal was Doppler shifted and it is observed that the PSR decreases with increase in Doppler shift. It is also observed that the LFM signal, when passed through Blackmanharris window, the Side lobe suppression is maximum.

4.1 Introduction

Signal processing in noise radar is based on the calculation of the correlation between transmitted and received signals. Received signals collected from nearby targets produce very high side lobes in the correlation function. While the signals received from far targets are weak .These weak echoes of far targets are masked by the strong echoes, this effect is called as masking effect. Masking effect can be removed by methods. Considering only the Doppler shift of target echoes for removal of masking effect is one of the old methods. But, this method does not remove the strong target echo completely when targets migrate between range resolution cells in the integration time. The signal stretch processing based method, gives an improved result over the existing methods. The detection of weak targets in the presence of the fast and strong ones is also possible by this method.

The effect of masking weak (far) target echoes by strong (near) ones is a major obstacle for construction of any synthetic aperture radar. This challenge can be easily tackled with range gain control in pulsed radar. The time separation between near and far echoes, helps in solving the problem. FMCW radars Analogue filters are used in frequency modulated continuous radar (single or double null at zero frequency), benefiting from the frequency separation of echoes. This separation is absent in continuous wave noise radars. For this reason, weak target’s echoes are masked by the side lobes originated from range compression blocks. Radar sensitivity and detection range decrease by this masking effect.

Following are some of the methods for solving the masking effect:

- Removal of crosstalk signal and ground clutter from the received signal .An adaptive lattice filter is used for this propose.
- Removal of nonzero Doppler clutter .Model of the target echoes, having non-zero Doppler frequency is produced and is subtracted from the received signal.
- Target echo modeling is done by considering the time and Doppler shift of the transmitted signal. It is used for relatively slow moving targets (range displacement during the integration time must be much smaller than the size of the range gated.
- In medium- and high-resolution radars, the range gate size is relatively small .The fast moving target echoes can easily migrate between the range resolution cells. The range migration of the fast echoes degrades the effectiveness of the cancellation procedures, as the modeled signal does not fit the received target echoes. A sophisticated target echo model must be considered, for efficient implementation target removal procedure.

SAR uses Range –Doppler algorithm for detection of target.

4.2. Range Doppler algorithm (RDA)

Matched filtering is performed separately in the Fourier transformed range and azimuthal range in RDA. The Fourier transforms are calculated via fast Fourier transfer (FFT) .Range cell migration correction (RCMC) is performed in the range time and azimuthal frequency domain. The 2D raw signal is analyzed as a series range time signal for each azimuth bin. Each range time signal undergoes matched filtering in the range frequency/azimuth time domain through range FFTs applied to the range time signal. Each signal is then transferred to time domain producing the range compressed signal. These signals are then composed into a series of signal with respect to azimuth time at different range bin. Then the signal is passed through FFT and RCMC. After passing it again through IFFT, the final target image is generated.

Here range-Doppler correlation function is considered for target detection.

Correlation of received baseband signal done with the transmitted signal. Before considering the transmitted signal replica for correlation it is shifted and modulated. The 2D correlation function is calculated. Then targets are declared at local maxima. Range of the target echoes:

If the received signal comes from a single target then optimal (in the mean square sense) detection is done. But in real conditions antenna receives reflected signals from multiple points. The ground clutter and weather clutter echoes are also present in the collected echoes. The echoes are reflected from various ranges of scatterers. FMCW radar separates these in time or frequency domain. Interference of target and clutter echoes takes place. Effect produced due to these overlapping leads to the masking effect. The far targets have weak and the nearby targets have strong echoes. These nearby targets and ground clutter reflection mask the far target reflected signal. So, the previously mentioned correlation detection formula can't be used here (more than one target echoes).

The received signal containing many point echoes

$$s_r(t)=a_1 \exp\left(j2\pi\left(f_o(t-\tau_1)+\frac{\mu}{2}\,(t-\tau_1)^2\right)\right) \qquad (4.1)$$

Where, $\tau_1=2R_1/c$

Signal envelop stretch is used in radar receiver processing. Stretch processing is done in the case of a linear target movement model $R(t)=R_0+v(t)$. For pulse radar, range migration processing can be done following pulse compression by coherent integration over several range cells. In case of an FMCW, stretch processing is implemented effectively with a group of sub-band filters, which are also known as frequency-dependent delay lines. A noise radar receiver analyzed here is basically a generally correlation receiver is used in SAR. And this has a strong echo adaptive cancellation scheme. In the derivations of both the correlation receiver and the strong echo canceller a template is used. A template the expected target echo generated using the known transmitted signal as a base. Modeled echo can also be generated without using stretch. Here the baseband template is modulated with a target Doppler frequency to match the Doppler shift in the received signal. In stretch processing, stretching of the template is done first.

In practice, all the processing is done by digital means. The transmitted signal template is normally generated by sampling an analogue noise. Another way of generating this template is by storing the digitally generated pseudo-noise signal. This is the same pseudo noise signal that we use to modulate the carrier. For stretching of the reference signal the sample rate of the signal have to be changed according to the equation

$$x_{ref}(\mathrm{n,v}) = x_T(\mathrm{n}\ t_s.(1+\frac{2v}{c})) \tag{4.2}$$

Where, x_T (t) is the continuous representation of a discrete-time transmitted signal template x_T(n) and ts is the sampling interval. The velocity v can be positive or negative .The time scale has to be dilated or compressed. With practical values of velocity v, the stretch factor (ratio of sampling rates, (1 + 2v/c) differs from 1 by a very tiny fraction, e.g. for (v + 150) m/s the stretch factor is equal to 1+1026. For this reason, effective resampling is a very difficult task to do. For rescaling by a ratio of 2 relatively small integers image processing is a very useful method.

Nearest-neighbor interpolation is a simple resampling method under image processing. This is a very fast method. But it has very low accuracy. It is a method that uses multivariate interpolation single or multiple dimensions. Its algorithm is such that, it only chose the value of particular nearest point. Values of the neighboring points ignored. As a result, a piecewise constant interpolant is obtained. For stretch factor very close to 1, periodic removal of one out of c/2v samples is done.

4.3 Various Methods in Stretch Processing to remove Masking Effect

4.3.1: Linear interpolation method

Here linear combinations of adjacent original samples are considered for calculating the new samples. It can be considered as a method of curve fitting using the linear polynomials.

$$x_{ref}(\mathrm{n,v}) = \alpha x_T(\lfloor nt_s.(1+\frac{2v}{c})\rfloor) + (1-\alpha)(\lfloor nt_s.(1+\frac{2v}{c})\rfloor) \tag{4.3}$$

4.3.2 Poly-phase Method

Here upsampling by an integer factor N and then downsampling by another factor M are done. Polyphase filtering method use low-pass filters to remove aliasing. Aliasing is the effect that causes various signals to become indistinguishable when sampled. In a standard application, a rational M/N stretch factor is generated by linear filter with M coefficient sets, changing for each output sample. In the stretch processing as the M/N is equal to a number very close to 1. The number of coefficient sets in stretch processing used to be very large. So it is tried to achieve a smaller number of coefficient sets. Here a polyphase resampler is used to evaluate sample values for time instants from a set of points. These points are oversampled by a certain integer factor.

Then the most exact value at the desired point between the available samples is sorted out by applying linear interpolation method.

4.3.3 Chirp-z transforms

It is an efficient method to compute the z-transform on a set of points that are defined as a where n =N0 ... N1. We first set,

$$a = \exp\ (j2p\ (1+2v/c)/N \qquad (4.4)$$

And then calculation of the chirp-z transform of an N-point FFT is done. After that we effectively obtain the signal resampled with an accurate required rate. But chirp-z transform is a very complex method than the FFT. Its efficient implementation is very difficult to achieve.

Table 4.1 Simulation Parameters

Bandwidth	1e8 Hz
Scatterer range(in meters)	20, 100
Range cross section	2 , 0.05
Time period	5 μs

4.3.4 Simulation Results

Signal from two targets, one from near target and one from far off target were received and processed using matched filter and stretched processor. In 1st and 3rd figures masking effect is clearly visible whereas, in 2nd and 4th figures masking effect is removed.

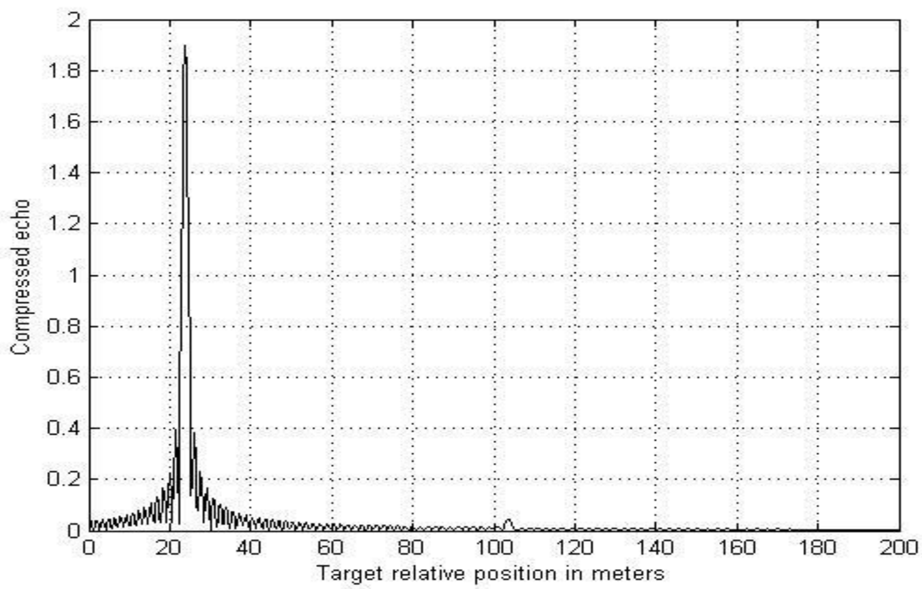

Fig4.1 Masked output using matched filter

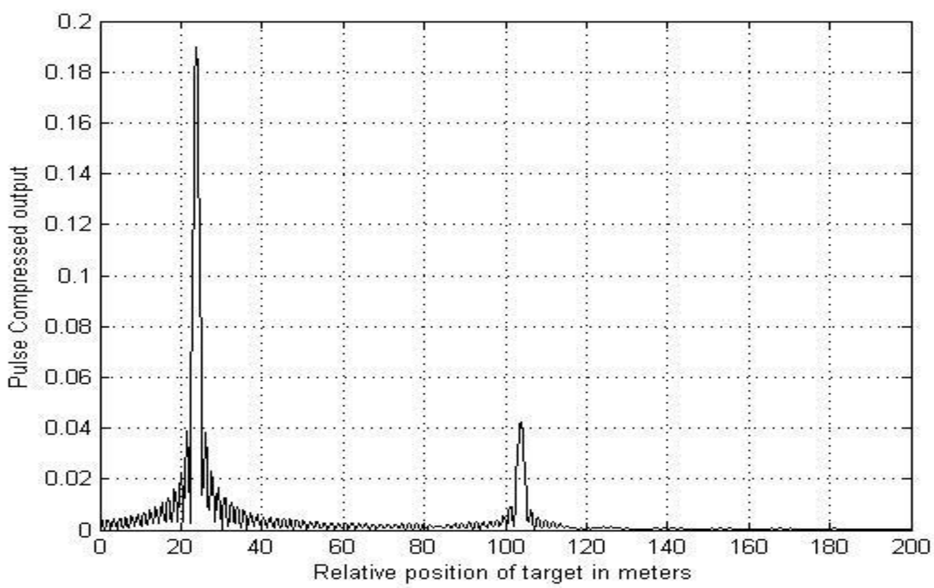

Fig4.2 Unmasked output using matched filter

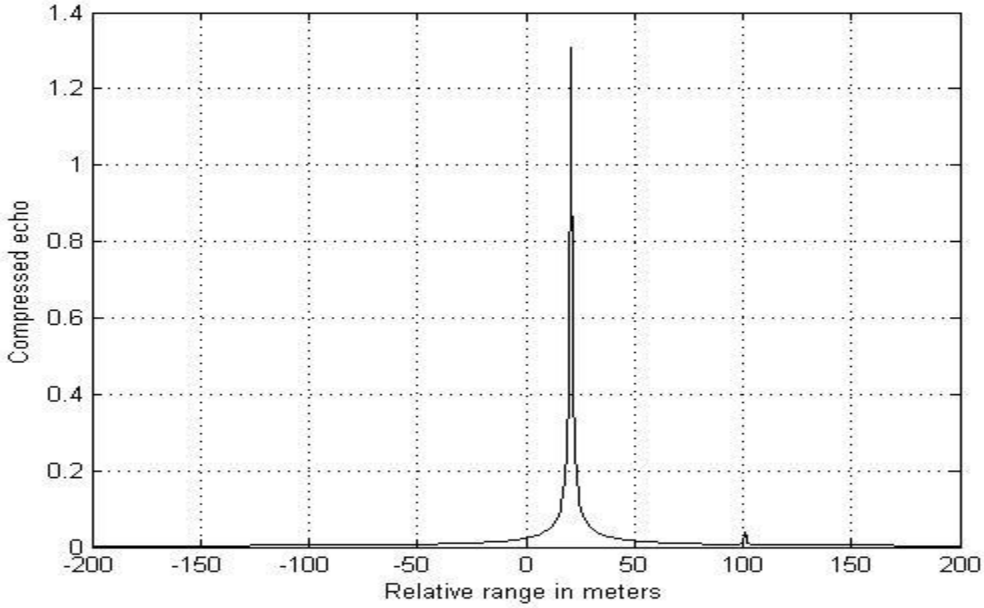

Fig4.3 Masked output using stretch processing

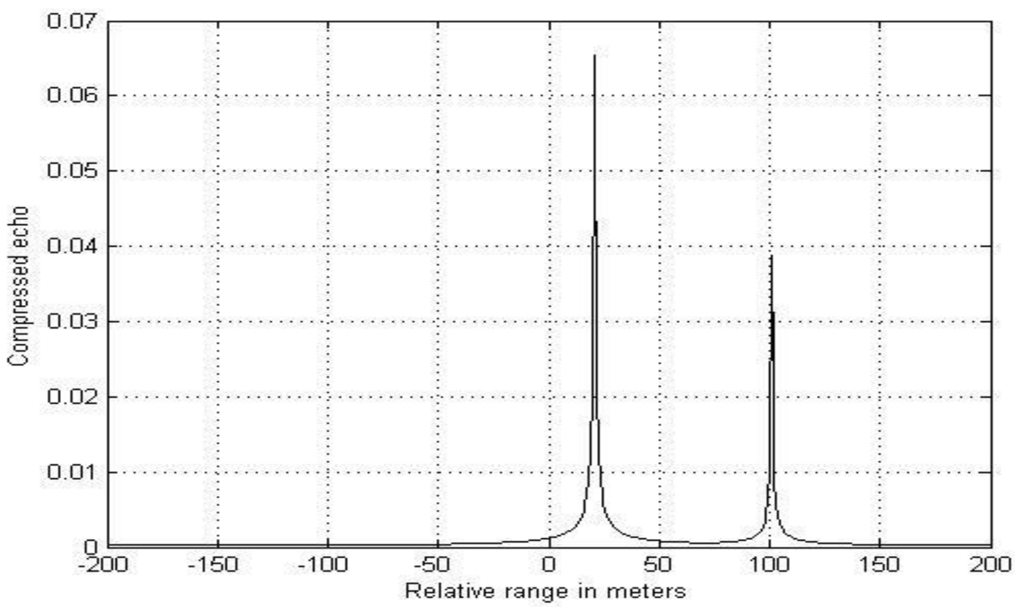

Fig4.4 Unmasked output using stretch processing

4.4 Conclusion

Masking effect is a common problem radar signal processing in which a nearby target masks or hides echos coming from a far target. In this chapter we have tried to remove the masking effect on far off target by simply by subtracting echo of nearby target from the received signal. The two methods described in chapter 2:-.Matched filtering and Stretch processing were used to process the signals. It is found out that stretch processing has better masking effect removal than match filtering process

In pulse compression for correlation of received signal with replica of transmitted signal matched filtering preferred while dealing with narrowband signals and stretch process is preferred while dealing with wideband signal. It is also concluded that stretch processing is a better technique when range resolution is taken into account i.e. Stretch processing method gives better range resolution than match filtering technique. Stretch processor also has better side lobe cancellation than match filter method and it does quicker processing than matched filtering.

LFM signals when passed through 8 different windows:

1. Hanning window
2. Hamming window
3. Kaiser window
4. Rectangular window
5. Blackmanharris window
6. Nuttal window
7. Flattop window
8. Triangular window

Best side lobe cancellation effect is shown by Blackmanharris window. PSR of LFM signals using different windows, increases with increase in time band width product and decreases with increase in Doppler shift.

Masking effect is a common problem radar signal processing in which a nearby target masks or hides echoes coming from a far target. We have tried to remove the masking effect on far off target due to a nearby target simply by subtracting echo of nearby target from the received signal. The two methods used were. Matched filtering and Stretch processing .It was found out that stretch processing has better masking effect removal than match filtering process

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