SIDELOBE LEVEL REDUCTION IN PULSE COMPRESSION WITH CLASSICAL ORTHOGONAL POLYNOMIALS AND WOO FILTER

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DECLARATION

I hereby declare that the work reported in the M-Tech dissertation entitled **"Sidelobe Level Reduction in Pulse Compression with Classical Orthogonal Polynomials and Woo Filter"** submitted at **Jaypee University of Information Technology, Waknaghat India,** is an authentic record of my work carried out under the supervision of **Mr. Salman Raju Talluri**. I have not submitted this work elsewhere for any other degree or diploma.

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CERTIFICATE

This is to certify that the work reported in the M-Tech. dissertation entitled "**Sidelobe Level Reduction in Pulse Compression with Classical Orthogonal Polynomials and Woo Filter**", submitted by **Ankur Thakur** at **Jaypee University of Information Technology, Waknaghat**, **India,** is a bonafide record of his original work carried out under my supervision. This work has not been submitted elsewhere for any other degree or diploma.

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ABSTRACT

The present dissertation aims to make an in-depth study on the Radar pulse compression. Pulse compression is used in the radar systems to avail the benefits for large range detection and high range resolution capability by long and short duration pulses respectively. Usually, matched filter (MF) and auto-correlation function (ACF) are used to analyse the pulse compression output. The output of the pulse compression of a modulated signal is associated with range side-lobes along with the main-lobe. These side-lobes are undesirable because they may contain the information associated to the target which is nearer to a stronger or desired target due to this the performance of the radar detection system is affected. In this dissertation, to improve the performance of radar system, few investigations have been made to reduce the side-lobes in the pulse compression output. Firstly, a fundamental comparative analysis on time-bandwidth product of a short duration exponential signal is analysed. The amplitude variation has been varied with an exponent as an integer and non-integer are observed.

In order to improve time-bandwidth product in the pulse compression, different Classical Orthogonal polynomials of different orders are analysed. Then Chebyshev polynomials are used to generate the sequence which has low sidelobe levels in the output of the compressed signal. In this dissertation using P4 code of length 1000, merit factors such as peak sidelobe level, integrated sidelobe level and relative mainlobe width are calculated for the proposed model. The results are compared with other techniques that use the Woo filter concepts.

LIST OF ABBREVIATIONS

AF	Ambiguity Function
ACF	Auto Correlation Function
BW	Band-Width
CW	Continuous Waveform
FM	Frequency Modulation
FFT	Fast Fourier Transform
IFFT	Inverse Fast Fourier Transform
ISL	Integrated Side-lobe Level
LFM	Linear Frequency Modulation
MF	Matched Filter
MMSE	Minimum Mean Square Error
MSE	Mean Square Error
NLFM	Non Linear Frequency Modulation
PM	Phase Modulation
PCR	Pulse Compression Ratio
PRI	Pulse Repetition Interval
PSL	Peak Side-lobe Level
RADAR	Radio Detection And Ranging

SNR Signal to Noise Ratio

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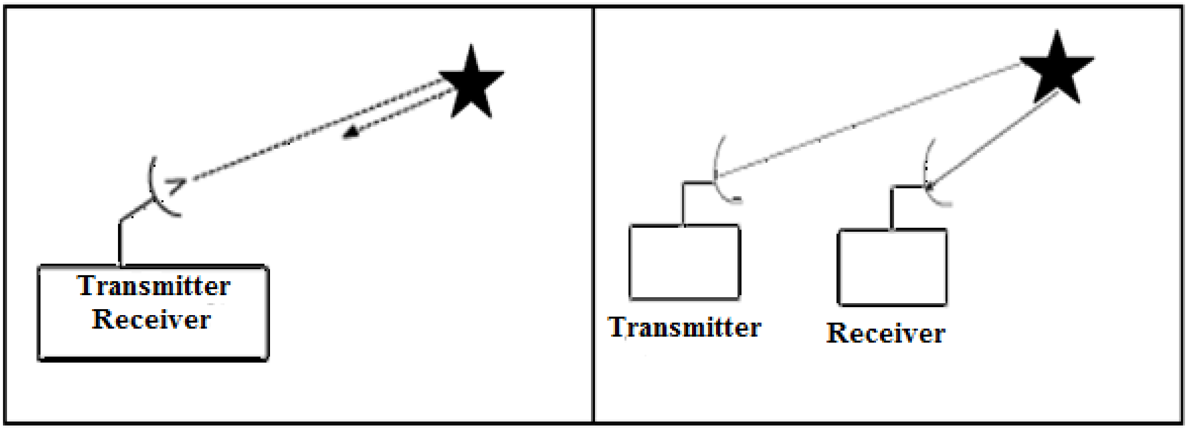
CHAPTER-1

INTRODUCTION

1.1 Radar Principle

Radar is an acronym of Radio Detection And Ranging. Radar basically transmitted an electromagnetic signal in to space by the transmitting antenna. Some portion of the transmitted signal incident on the target and gets reflected in many directions, some of the reflected signal or echoes are collected by the radar's receiving antenna. These echoes are used to extract the information about the target such as range, velocity and other identifying characteristics in all weather conditions [1, 2].

There are two types of radars monostatic and bistatic. In the monostatic, transmitter and receiver are at the same location, where as in bistatic transmitter and receiver are separated by some distance as shown in Figure 1.1.



(a) **(b)**

Figure 1.1: Radar transmitter and receiver (a) Monostatic and (b) Bistatic radar

Active and passive types of radars are exists. In active radars both the transmitter and receiver works simultaneously, where as in passive radar system only one work at a time. In the continuous radar, signals are transmitted continuously while receiving target echoes on a separate antenna. Usually continuous waveform (CW) is used for the measurement of unambiguous Doppler shift through which speed of the target can be determined. However, due to continuous nature of the waveform the target range measurement is entirely ambiguous. On the other hand, for accurate range measurement pulse waveform based radar system are used. The primary advantage of pulsed radar is that due to pulsating nature of radar waveform, transmitter and receiver can share the same antenna. Range resolution and maximum range detection are the two important factors to be considered for radar waveform design. For range resolution duration of the pulse should be small. But, if the pulse width is decreased, the amount of energy in the pulse is decreased and hence maximum range detection gets reduced because signal strength at the receiver is very low, which is not sufficient for target detection. To overcome this problem pulse compression techniques are used in the radar systems.

1.2 Range and Velocity Resolution

In order to determine the range of the target, the delay between the transmitted pulse and reflected pulse have to be measured and by knowing the propagation speed of the electromagnetic wave in the space or different media [4]. For real time applications more pulses at some repetitive interval will be transmitted rather than transmitting a single pulse, whose period is called as pulse repetition interval (PRI).

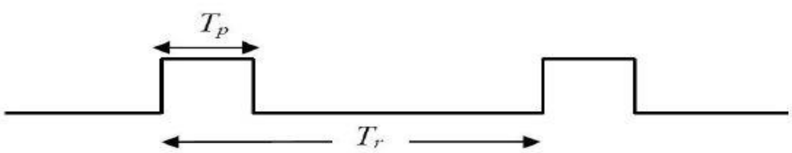


Figure 1.2: Pulsed radar waveform

T_p and T_r are the pulse duration and pulse repetition interval respectively as shown in the Figure 1.2. The minimum distance between two nearby targets can be resolved is called as range resolution given by

$$R = cT_p/2 \qquad (1.1)$$

Where R is the range resolution, T_p is the pulse duration and c is the velocity of electromagnetic wave. The value of R should be small as much as possible because it can differentiate two nearby targets in term of small distance.

In order to determine the velocity of the object or to find the velocity resolution, the relative frequency shift between the transmitted signal and the echo signal has to be measured and this frequency shift is used to find the radial velocity of the object using [1, 3]

$$f_d = 2v_r/\lambda \qquad (1.2)$$

Where the Doppler frequency shift is f_d and v_r is the radial velocity of the target. Velocity resolution is defined as, if two targets are moving with two dissimilar velocities which are close by, even then the radar system must identify them as two different targets. In general a radar system has to resolve the targets in both range and velocity without any ambiguities. This puts a constraint on the transmitted pulse duration and the power that has to be transmitted in pulse mode. If the spectral spreads are more, then that pulse is more suitable for the better resolution in frequency domain [5, 6] it means the resolution in frequency can be attained as the reciprocal of the time duration. Maximum unambiguous range R_u that can be measured by pulsed radar [6] can be calculated as

$$R_u = \frac{cT_r}{2} \tag{1.3}$$

It means it can distinguish two nearby target in terms of smaller distance. Generally range resolution depends on the bandwidth of the transmitted signal rather than the duration of the pulse. It means the waveforms which have higher bandwidth is preferred for good range resolution, that's by in pulse compression long duration pulse is transmitted with angle modulation, that will increase the overall bandwidth and this bandwidth responsible for range determination. On the other hand, long duration pulse which is transmitted is responsible for good detection in radar system

1.3 Pulse Compression

Pulse compression is used in the radar system to get the benefits of long and short duration pulses by a single pulse. To detect a target, reflected pulse should have more strength, for this purpose transmitted pulse should have more energy for long distance transmission. The energy contents depend on the peak power as well as the duration of the transmitted pulse. The product of peak power and total pulse duration estimate the energy of the signal [3]. In pulse compression short duration pulses are used for range resolution. Practically, the pulse duration cannot be decreased indefinitely [5], because a very short pulse requires high peak power to get adequate energy for large distance transmission. To generate high peak power, overall radar equipment become heavier, bigger and by which cost of the system increases. That is by a pulse having low peak power and longer duration is required at the transmitter for long range detection. Large bandwidth implies narrow effective duration [5, 6]. The waveform with small effective duration is produced when the long duration waveform with angle modulation transmitted, and then the received signal is passed through the MF to observe the pulse compression output. This concept is called pulse compression [5].

So here pulse compression ratio (PCR) or time-bandwidth product concept comes, its value is always greater than unity [4, 6]. PCR is the main parameter in pulse compression it should be more for good detection as well as for good range resolution. The pulse compression ratio is defined as

$$PCR = \frac{width\ of\ the\ pulse\ before\ compression}{width\ of\ the\ pulse\ after\ compression}$$

Using pulse compression, simultaneously system can obtain good detection and highly accurate range measurements.

1.4 Ambiguity Function

Ambiguity function (AF) is an analytical tool for designing the waveform and it analyse the waveform behaviour paired with its matched filter response. It is useful to examine resolution, sidelobe behaviour and ambiguities in both range and Doppler. It is often necessary to examine a waveform and understand its resolution and ambiguity in both range and speed domains. The range and speed is measured using the delay and Doppler shift respectively. In order to measure range and speed of an object ambiguity function can be used, that is represented as

$$X(t_d, f_d) = \int_{t=-\infty}^{\infty} x(t)x^*(t - t_d)\, e^{j2\pi f_d t} dt \qquad (1.4)$$

$X(t_d, f_d)$ is ambiguity function, t_d and $f_d$are the time Delay and Doppler shift respectively. $x(t)$ is the transmitted signal. AF depends on the two parameters, time delay (t_d) and Doppler shift (f_d). Three properties of AF are of immediate interest. First states that total area under AF is always constant. This generally holds the conservation of energy statement implies that one cannot remove energy from AF surface until this portion is placed somewhere on the surface of AF.

$$\int_{-\infty}^{+\infty} \int_{-\infty}^{+\infty} |X(t_d, f_d)|^2\, dt_d df_d = E^2 \qquad (1.5)$$

The second property states that when the filter matched Doppler to the echo and it is samples exactly at a delay corresponding to the target range then it gives energy of the waveform.

$$|X(t_d, f_d)| \leq |X(0,0)| = E \qquad (1.6)$$

The third property states that AF response is always symmetrical around the delay Doppler plane.

$$|X(t_d, f_d)| = |X(-t_d, -f_d)| \qquad (1.7)$$

1.5 Literature Survey

In order to start the study regarding pulse compression, the first step is to study the research papers that have been published by other researchers. The papers that are related to this title are chosen and studied. With the help of this literature review, it gives more clear understanding related to the pulse compression. So many techniques for pulse compression have been developed such as linear frequency modulation, non-linear frequency modulation, Barker codes. There are many several types of pulse compression waveforms useful in Radar. Firstly linear frequency modulation (LFM) is developed; this is one of the most useful waveform in order to compress the pulse. In Costas frequency modulation (FM), frequency are chosen such that the resulting waveform has an uncertainty function that rapidly decrease from its maximum in both delay and Doppler frequency co-ordinates, and has very low side-lobe levels over most of the delay-Doppler plane.

In [10], the authors have proposed a technique in which auto-correlation function (ACF) for modified two and tri-stage non-linear frequency modulation (NLFM) signal are analysed and -19dB side-lobe suppression is achieved without disturbing the relative main-lobe width. Peak sidelobe levels (PSL) and integrated sidelobe level (ISL) are reduced by sidelobe canceller, developed by Woo and Griffiths [11]. In [19], the application of least-mean-squares approximated by developing inverse filtering techniques to radar range sidelobe reduction has been discussed. The performance of the least mean square inverse filter is compared with the matched filter. A filter which completely suppresses the range side-lobes of a 13-element Barker sequence is only 0.2 dB worse than a matched filter in noise. By using two different Barker codes, a neural fuzzy network is developed [20] which gives significant advantages for range and velocity resolution and also sidelobe are suppressed.

Baghel and Panda [21], have proposed a hybrid model for the phase coded waveforms in which MF output is modulated by the output of radial function for different Barker codes. In addition to this, the hardware requirement is also significantly less to implement this hybrid model without any training iterations as in the neural network. Vizitiu [22] has produced a technique to overcome the problems of LFM signal that is stretching of the main lobe width which disturb the range resolution by using nonlinear laws. The advantage of P4 codes are, it can be derived for any length sequence and these are cyclic shifted codes which gives better sidelobe reduction than other polyphase codes [26]. The weighting in time and frequency domain is applied in order to suppress the side-lobes as discussed in [27]. Rihaczek [29] has proposed a sidelobe suppression technique in which only few distinct tap weight line are used to reduce the side-lobes that reduces the complexity of digital processor. The Barker codes and multistage Barker codes are used for higher pulse compression factors. Indranil and Adly [30], have developed a model in which a mismatched filter, comprised of a matched filter is cascaded with a parameterized multiplicative finite-duration impulse response filter but this technique uses fewer multipliers and adders. . In [31], the author realized a filter that reduces the side-lobes of the quadrat phase-coded waveform by applying the bi-phase to quadrat phase transformation to the filter designs that reduce the side-lobes of the prototype bi-phase code. In [32], the author demonstrated that two sample sliding window sub tractor in the output of a digital Frank or Pl code compressor can limit the compressed pulse range and side-lobes to those of the Barker codes with unlimited pulse compression ratios. Its significant sidelobe reduction is attained at the cost of 1dB loss in the signal-to-noise ratio. In other approach, ACF for LFM signal is analysed in sinc function having -13dB sidelobe, it also represents modified two and tri-stage NLFM [33] by which -19dB sidelobe suppression is achieved. It is observed that NLFM are good for sidelobe reduction without disturbing signal to noise ratio (SNR).

Lee [34] has proposed a simple technique for polyphase codes to synthesize amplitude weighting correlators at the cost of minimal range resolution loss. The side-lobes generated by these weighted codes are significantly smaller and uniformly flat over all time delays than those achieved in the Barker codes Felhauer [35], derived a new class of polyphase codes by step approximation of the phase function of a NLFM chirp signal with a favourable energy density spectrum. The significant advantages of these codes over the conventional polyphase codes are lower auto-correlation side-lobes and an improved tolerance of low Doppler shifts.

A hyper chaotic coding scheme and corresponding optimal selection method are proposed by authors [36], to obtain the phase code signal, which exhibits great performance for sidelobe reduction. When Gaussian-noise variance is less than 0.01, reduction in the side lobe of -55dB exists, but the signal-to-noise ratio loss is less than 0.05dB.In [37], the authors have presents a technique for the design of mismatched received finite impulse response filters based on the minimization of Lp-norms of the side-lobes. The target is to reduce PSL for the convolution of the transmitted pulse and the received pulse after some delay. A closed-form solution is derived for the least-squares case and an expression for the optimization of the higher order norms is developed

1.5 Problem Formulation

In the pulse compression output, side lobes are undesired. In the literature there are many techniques based on LFM and NLFM for the sidelobe level reduction. So to reduce these side lobes investigations has been done regarding some transmitted waveforms. To analyse the time-bandwidth product, exponents signal of integer and non-integer order has been observed. Then different Classical Orthogonal polynomials are incorporated to observe the time-bandwidth product. Chebyshev polynomial based sequence is generated that increase the length of the overall sequence and it gives better peak sidelobe level reduction. Woo filter based model is proposed which uses polyphase code, gives better reduction in the PSL, ISL and relative mainlobe width.

1.6 Dissertation Organization

This dissertation includes six chapters. An outline of each chapter is given below:

Chapter 1^{st} gives an introduction of radar system, range and velocity resolution, pulse compression, ambiguity function, literature survey and problem formulation.

Chapter 2^{nd} presents the comparative analysis on exponential form of pulse for better range and velocity resolution.

Chapter 3^{rd} presents a study on different Classical Orthogonal polynomials in order to improve time-bandwidth product.

Chapter 4^{th} presents the Chebyshev polynomial based sequence generation in order to reduce the side lobes of matched filter response.

Chapter 5^{th} presents the Woo filter concept of sidelobe reduction, in which proposed techniques results are compared with other Woo filter sidelobe reduction techniques.

Chapter 6^{th} concludes this dissertation, summarizing the major results of all chapters and offering suggestions for future work on this topic.

CHAPTER-2

COMPARATIVE ANALYSIS ON AN EXPONENTIAL FORM OF PULSE WITH BETTER RESOLUTION IN RANGE AND VELOCITY

2.1 Introduction

There has been lot of research has been carried out on pulse compression, this chapter presents the pulse compression with the basic pulse in exponential form with its comparison in different expressions which serves as an extension on pulse compression with non-integer exponents. This analysis gives a new form of pulses that can be used in real time pulse compression techniques for practical applications. Here comparative analyses on an exponential form of pulse with an integer and non-integer exponent are done in order to improve resolution in range and velocity, which are used in pulse compression. The time-bandwidth product is the figure of merit for the pulse compression and hence the frequency spectrum of the transmitted pulse is obtained with the help of fast Fourier transform (FFT) [7]. To make a comparative analysis, different pulses have been considered and time-bandwidth products have been tabulated. It is often necessary to examine a waveform and understand its resolution and ambiguity in both range and speed domains. The range is measured using delay and speed is measured using the Doppler shift. In radar systems, the transmitted signal is represented with $x(t)$ and the received signal can be represented as $r(t)$. The received signal can be approximated in terms of transmitted signal with some attenuation, delay and frequency shift in the frequency domain as

$$r(t) = \alpha(t)\, x(t - T)e^{j2\pi f_d t} \qquad (2.1)$$

Where $\alpha(t)$ can be taken as attenuation of the signal as a function of time. Usually, this can be considered as a constant. T represents the delay between the transmitted signal and received signal while the complex exponential multiplication represents the variation in the frequency of the received signal due to relative motion between target the radar system. In radar systems to maximize the peak signal to noise ratio, a matched filter (MF) is used which acts like a correlator and its impulse response can be given as [5, 6].

$$h(t) = r^*(\tau - t) \qquad (2.2)$$

Where τ is the parameter used for maximizing the output of the filter at predefined time.

The output of the MF can be used to find the peak side lobe levels in the ACF which gives an idea about the range resolution and velocity resolution. PSL which measures the ratio of maximum side lobe magnitude to the in phase value of the ACF. This can be calculated as

$$PSL = 20log_{10}\frac{max_{1\leq l\leq N}|C(l)|}{|C(0)|} \qquad (2.3)$$

N is number of side lobes, $C(l)$ is output of MF.

AF behaves like a MF when there is no Doppler-shift exists and its response is always symmetrical around the origin, and total area under this function is always constant. Generally AF gives single central peak at the origin while the remaining energy will be spread uniformly in the delay-Doppler plane only for the ideal case and this narrow peak which lie at the origin implies good resolution in both range and velocity [3]. At the receiving end, when filter is matched in Doppler to the reflected signal and is sampled at a delay corresponding to the actual range of target, then the response of this filter will be maximum at the origin which gives good resolution property. If this function is sampled at some different delay, then the response of the filter will be less than to the maximum and there is possibility of ambiguity in range and velocity.

An exponential pulse of the form

$$x(t) = \alpha \cdot e^{f(t)}.rect(t/\tau) \qquad (2.4)$$

has been analysed where $f(t)$ is of the form $\beta \cdot t^n$ where β is a constant and exponent n is integer and non-integer as well. Along with this simple pulse, this pulse has been modified with differentiation and multiplication with signum function, and modulating these pulses with a sinusoidal carrier signal.

2.2 Simulation Results and Discussion

In the first place a simple exponential pulse has been analysed. Mathematically this pulse can be represented as

$$x_1(t) = \alpha \cdot \left(e^{t^n}.u(-t) + e^{-t^n}.u(t)\right) \qquad (2.5)$$

Here n, α are the order and amplitude of the signal. Exponential pulses having non-integer order of half and integer order of one, two, three, and five has been analysed for every case. Time domain and frequency spectrum analyses for exponent signal are shown in Figure 2.1. Frequency spectrum of the exponential pulses has been observed with the FFT.

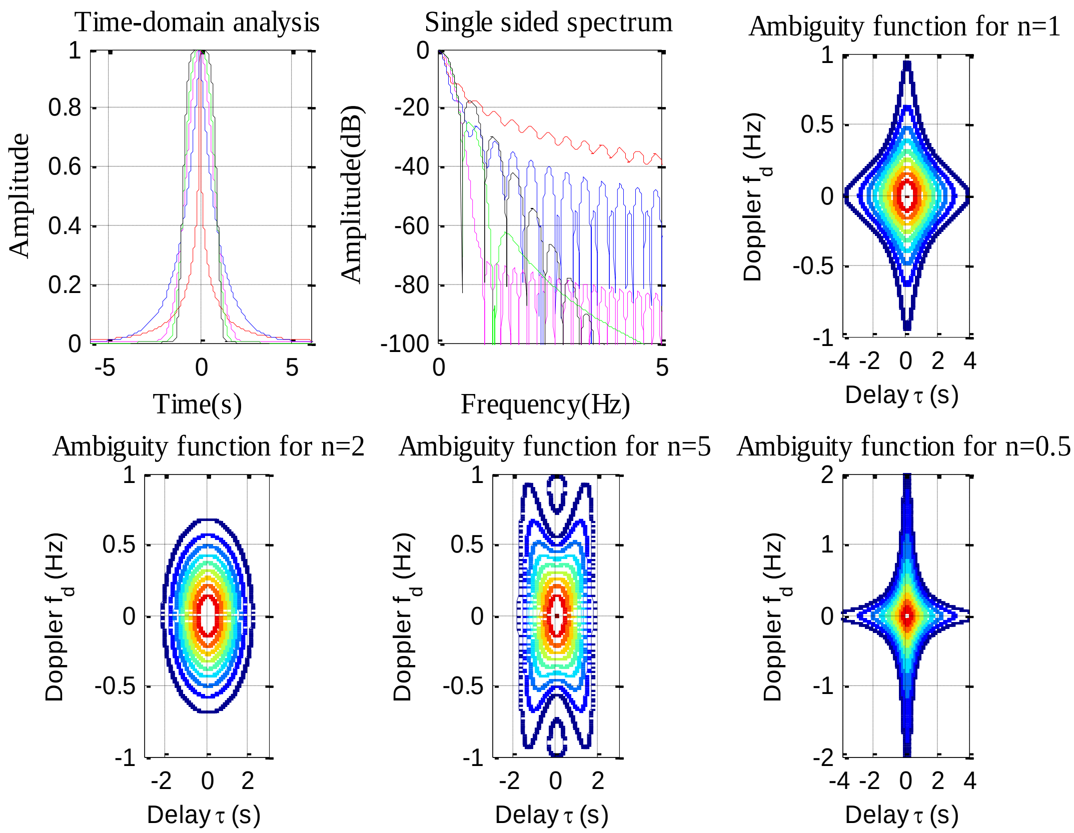


Figure 2.1: Time domain, frequency domain and ambiguity functions for the first exponential signal with different exponents

Red trace is for n=0.5, blue trace is for n=1, magenta is for n=2, green is for n=3 and black is for n=5. The colour of the traces has been preserved in all the succeeding figures for comparison. From Figure 2.1 it is observed that for order two its spectrum is flat than other orders, but MF outputs are not compressed. After observing this, the polarity of the pulse has been taken as positive half and negative half for the entire pulse. This is like multiplying the given pulse with signum function. Mathematically bi-phase exponential signal can be represented as

$$x_2(t) = \alpha \cdot \left(e^{t^n}.u(-t) - e^{-t^n}.u(t)\right) \quad (2.6)$$

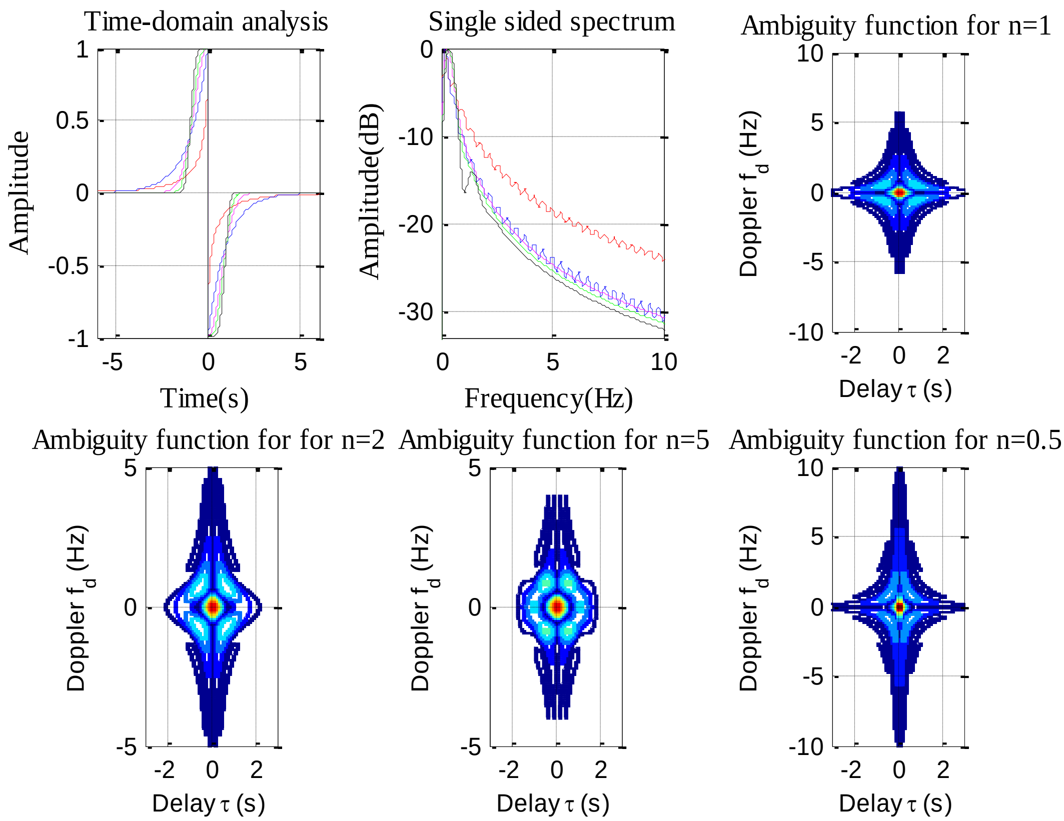


Figure 2.2: Time domain, frequency domain and ambiguity functions for the second exponential signal with different exponents

A simple mathematical operation is done on the original exponential pulse by differentiation. Differentiated exponential pulses can be represented as

$$x_3(t) = \alpha \cdot \left(n.t^{n-1}.e^{t^n}.u(-t) - n.t^{n-1}.e^{-t^n}.u(t)\right) \quad (2.7)$$

Table 2.1: Represents the PSL, 3-dB beam-width, main lobe width for the simple exponential pulses

Sr. No.	Order(n)	Peak sidelobe level(dB)	3dB Beam width	Mainlobe width
1	0.5	-Nil-	0.14	0.70
2	1	-Nil-	0.13	0.76
3	2	-Nil-	0.19	2
4	3	-Nil-	0.21	1.12
5	5	-Nil-	0.23	1.08

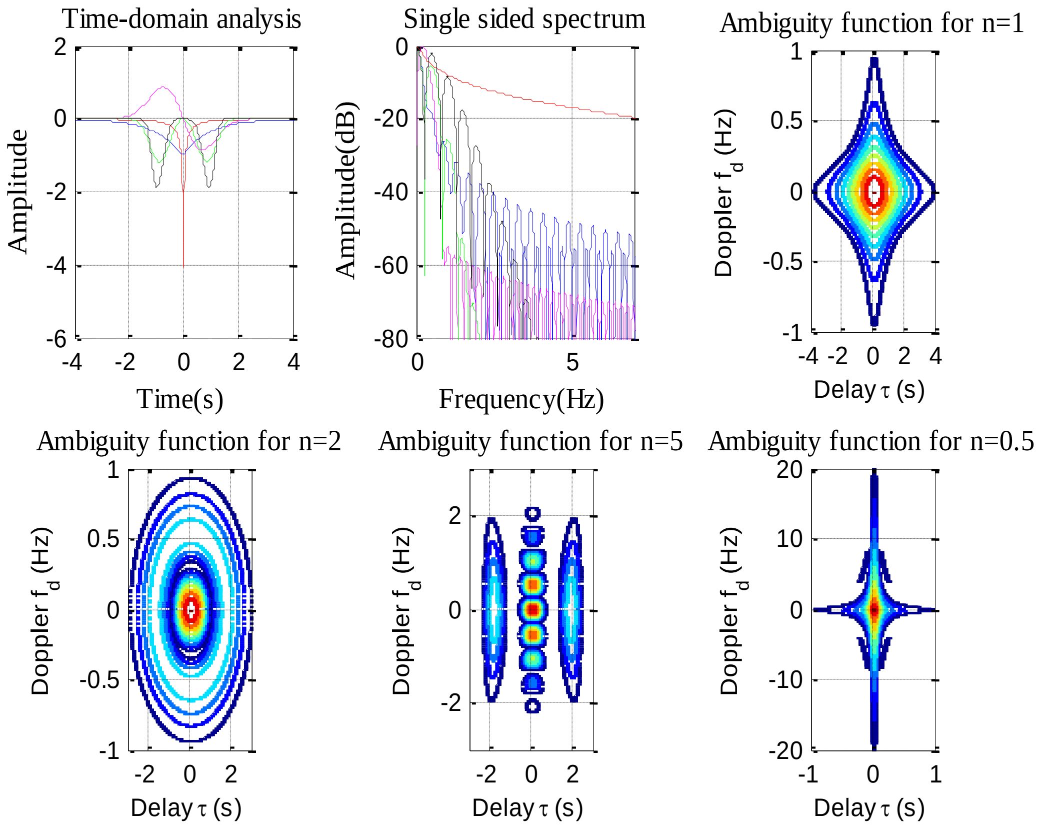


Figure 2.3: Time domain, frequency domain and ambiguity functions for the third exponential signal with different exponents

But usually for the transmission purposes, modulation of the input signal is required. Now these exponential pulses are to be multiplied by sinusoidal signal. Here cosine signal is incorporated having frequency 4Hz. In this chapter, this frequency is taken for convenience; by altering this frequency, conclusion will not altered, so this frequency can be up scaled or down scaled as per requirements. By multiplying with cosine signal to the simple exponential pulses, there time analysis (auto correlation and matched filter response) and frequency spectrum are analysed.

Mathematically cosine exponential pulses can be represented as

$$x_4(t) = \alpha \cdot \cos(2\pi f t)\left(e^{t^n}.u(-t) + e^{-t^n}.u(t)\right) \qquad (2.8)$$

Now multiply the bi-phase exponents to the cosine signal as shown below

$$x_5(t) = \alpha \cdot \cos(2\pi f t)\left(e^{t^n}.u(-t) - e^{-t^n}.u(t)\right) \qquad (2.9)$$

Now multiply the differentiated exponents to the cosine signal as represented below

$$x_6(t) = \alpha \cdot \cos(2\pi f t)(n.t^{n-1}.e^{t^n}.u(-t) - n.t^{n-1}.e^{-t^n}.u(t)) \quad (2.10)$$

Table 2.2: Represents the PSL, 3-dB beam-width, main lobe width for the Bi-phase exponential pulses

Sr. No.	Order(n)	Peak sidelobe level(dB)	3dB Beam width	Mainlobe width
1	0.5	-16.19	0.34	0.86
2	1	-13.82	0.34	0.92
3	2	-11.56	0.53	--
4	3	-9.32	0.57	--
5	5	-7.69	0.59	2.2

By observing this table pulse having order half gives better PSL and order five have worst PSL.

Table 2.3: Represents the PSL, 3-dB beam-width, main lobe width for the differentiated exponential pulses

Sr. No.	Order(n)	Peak sidelobe level(dB)	3dB beam width	Mainlobe width
1	0.5	-Nil-	0.26	--
2	1	-Nil-	0.13	0.80
3	2	-6.94	0.10	1.86
4	3	-6.03	0.14	0.56
5	5	-6.10	0.14	0.54

2.3 Discussions on Ambiguity Function

In ambiguity analysis [2], x-axis represents the delay and y-axis represents the Doppler shift. If the same pattern is repeating in x-axis and in y-axis then range and velocity resolution can be determined respectively. In this chapter the ambiguity function has been analysed for different exponents of order one, two, five and half. For simple exponential pulse ambiguity functions analyses gives not any range as well as not any speed resolution as shown on Figure 2.1, because there is not any repetition of the pattern. Moreover order half has good ambiguity function result because pattern is more concentrated towards the origin as compared to the other exponents.

For Bi-phase exponential pulse ambiguity functions analyses gives not any range as well as not any speed resolution as shown in Figure 2.2 because of no repetition in pattern. But fractional order have good ambiguity results because pattern is more concentrated at origin and this analysis are good than simple exponential pulses because in half order bi-phase exponent pattern come closer to origin. Figure 2.3 represents the ambiguity analysis of differentiated exponential pulses, it is observed that for order one, two and half, resolution cannot be determined but for order five both resolutions exist, by which speed and range of an object can be measured. For order five there is more ambiguity for speed resolution. It is observed that for order half ambiguity result is good as compared to other exponents because pattern is more concentric at the origin. So non-integer exponents gives good result for pulse compression as well as better ambiguity analysis.

Now by multiply these pulses to the sinusoidal signal of frequency 4Hz, for simple cosine exponential pulses Ambiguity analysis gives range resolution as well as Doppler shift, it is observed that when frequency is increasing then delay patterns come closer to each other and Doppler shift is increasing which is repeating after twice of the applied frequency . If in the delay axis pattern come closer its range resolution is good because pulse become narrower. Here applied frequency is 4Hz hence at 8Hz, 12Hz and an integer multiples of 4Hz s, the same pattern is repeating on the y-axis which gives the velocity ambiguity and same pattern is repeating on the x-axis which gives the ambiguity in range simultaneously. For bi-phase cosine exponential pulses, Ambiguity analyses give range resolution as well as Doppler shift, by which range and speed resolution can be determined.

Finally, ACF which is the outputs of the MF is represented in Figure 2.4. The best ACF should have the narrow beam and very small relative side lobe levels. It is observed that there output of the MF for non-integer value is giving better result compared to the other forms of the signals. ACF response is better for non-integer order that is half. It is observed that pulse is narrower and it has no side lobe which is desired. Order one also have not any side lobe but the MF response is wider which is not desired. So results are good for non-integer order pulse.

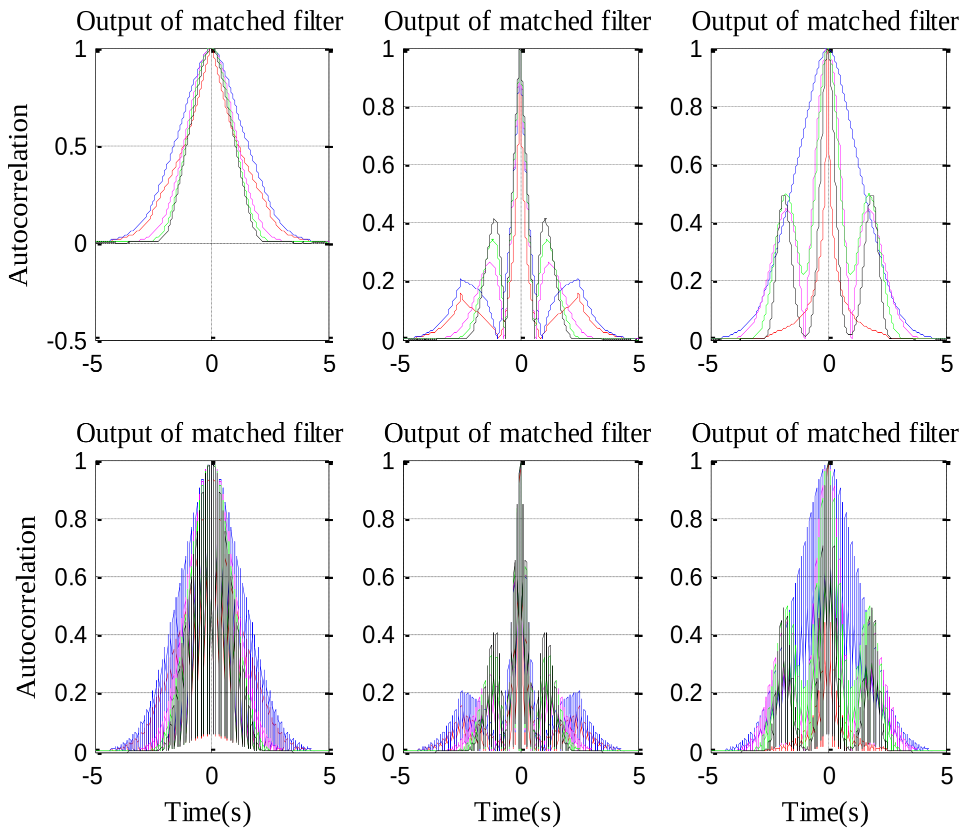


Figure 2.4: ACF for all the signals in sequence

2.4 Conclusion

After the detailed analysis on the quantitative parameters of the pulse compression is carried out with an exponential kind of pulse in different mathematical forms. The exponents used in this chapter are of an integer and non-integer. From the analysis, it is observed that the differentiated exponential with non-integer index is giving slightly better requirements with reference to pulse compression which can be used in practical applications. From the simulation it is concluded that ambiguity analysis are also good for non-integer form of pulses because delay-Doppler plane is more concentrated at the origin than other exponents ambiguity function.

CHAPTER-3

ANALYSIS ON TIME-BANDWIDTH PRODUCT WITH DIFFERENT CLASSICAL ORTHOGONAL POLYNOMIALS

3.1 Introduction

The theme of this chapter is to analyse and compare the pulse compression with Classical Orthogonal polynomials namely, Chebyshev, Laguerre, Legendre and Hermite polynomials for different orders [13, 14]. Pulse compression is used in radar systems to improve the range resolution by increasing the time-bandwidth product [1] of the transmitted pulse. Pulse compression is done by modulating the instantaneous angle of the transmitted carrier pulse of fixed duration [3]. The instantaneous angle modulations considered in this chapter are of three types. In first case, angle is varied in proportional to the Classical Orthogonal polynomials. In second case, angle is proportional to integral of Classical Orthogonal polynomials and lastly, angle is proportional to derivative of the Classical Orthogonal polynomial. The main purpose of this analysis is to use the best of all these Classical Orthogonal polynomials in pulse compression and to compare the quantitative parameters of pulse compression such as time-bandwidth product. For all these polynomials, the optimal value of time-bandwidth product are tabulated and compared with each other. For a better range resolution, τ must be as small as possible. A rectangular pulse with duration τ has resolution bandwidth (BW) as $1/\tau$. Hence the range resolution can be expressed in terms of bandwidth as

$$R = c/(2BW) \tag{3.1}$$

For better resolution in range, the bandwidth of the pulse has to be very large which indicates a shorter pulse. This shorter pulse makes difficulty in decision of the target. Hence the BW of the pulse has to be increased as much as possible while maintaining the duration of the pulse fixed [4, 5]. To satisfy this constraint, modulation can be applied on the pulse with the equation

$$x(t) = A\cos\big(2\pi f t + \varphi(t)\big) \times Rect(t/\tau) \tag{3.2}$$

Where $Rect(t/\tau)$ a rectangular pulse of duration is τ. Here τ has been fixed and the search has to be conducted for the best possible function $\varphi(t)$, such that the spectrum of $x(t)$

has to spread flatly over the large band of frequencies. There are many functions possible for $\varphi(t)$, but in this chapter, the functions are confined to the Classical polynomials due to the wide area applications of these Classical polynomials in engineering domain [16]. This chapter presents the detailed analysis of the pulse compression with respect to Classical orthogonal polynomials. In order to observe time-bandwidth product, spectrum of different Classical Orthogonal polynomials has been obtained. Depending on the values of α (optimizing factor), the spectrum is expanding smoothly up to a certain value after that spectrum get distorted. Hence this variable α has to be selected such that the time-bandwidth can be improved without any distortion in the signal spectrum.

3.2 Classical Orthogonal Polynomials Implementation

The transmitted pulse can be expressed in mathematical form as $x(t)$. This precisely represents a carrier of duration τ seconds whose angle is varied in accordance with $\varphi(t)$. Variations in the function $\varphi(t)$ give the modulation in the transmitted pulse. Here three types of variations are considered. Firstly, $\varphi(t)$ is varied in proportion to the Classical Orthogonal polynomial $p_n(t)$. Here $p_n(t)$ can be any Orthogonal polynomial of order n. Secondly, $\varphi(t)$ is varied in proportion to the integral of the Classical Orthogonal polynomial and finally $\varphi(t)$ is varied in proportion to the derivative of the Classical Orthogonal polynomial. First case is considered as the phase modulation of the carrier with $\varphi(t)$ and second case can be considered as frequency modulation. Third case can be treated as general angle modulation of carrier with $\varphi(t)$. The argument of the cosine function has been taken as $\theta(t) = 2\pi f t + \alpha\, p_n(t)$ and $\theta(t) = 2\pi f t + \alpha \int p_n(t)dt$ and $\theta(t) = 2\pi f t + \alpha\ dp_n(t)/dt$. The maximum variations in the instantaneous frequency and the maximum phase deviation the transmitted pulse can be controlled with the parameter α and after the analysis is carried out, it is possible to come up with the maximization of time-bandwidth product. Hence α is considered as optimizing parameter for maximum bandwidth.

In the simulations, the above mentioned three variations are considered with first four order polynomials of all Classical Orthogonal polynomials. The duration of the pulse has been fixed constant for all simulations and the carrier frequency f has been taken as 61 Hz. This is arbitrary and the conclusions are not going to change because of this choice, as the frequency can be scaled up according to the requirements in practical applications. All four types of the Classical polynomials are given in Table 3.1.

Table 3.1: Recursive equations for Classical Orthogonal Polynomials of order *n*

Chebyshev(T_n)	$T_{n+1} = 2xT_n - T_{n-1}$
Legendre(P_n)	$P_{n+1} = \frac{1}{n+1}((2n+1)xP_n - nP_{n-1})$
Laguerre(L_n)	$L_{n+1} = \frac{1}{n+1}((2n+1-x)L_n - nL_{n-1})$
Hermite(H_n)	$H_{n+1} = 2xH_n - 2nH_{n-1}$

3.3 Simulation Results and Discussion

Figure 3.1 represents the transmitted signals with different polynomials for the same optimizing parameter α with $\varphi(t)$ proportional to $p_n(t)$. With the help of fast Fourier transform, the spectrum for the transmitted pulse is obtained.

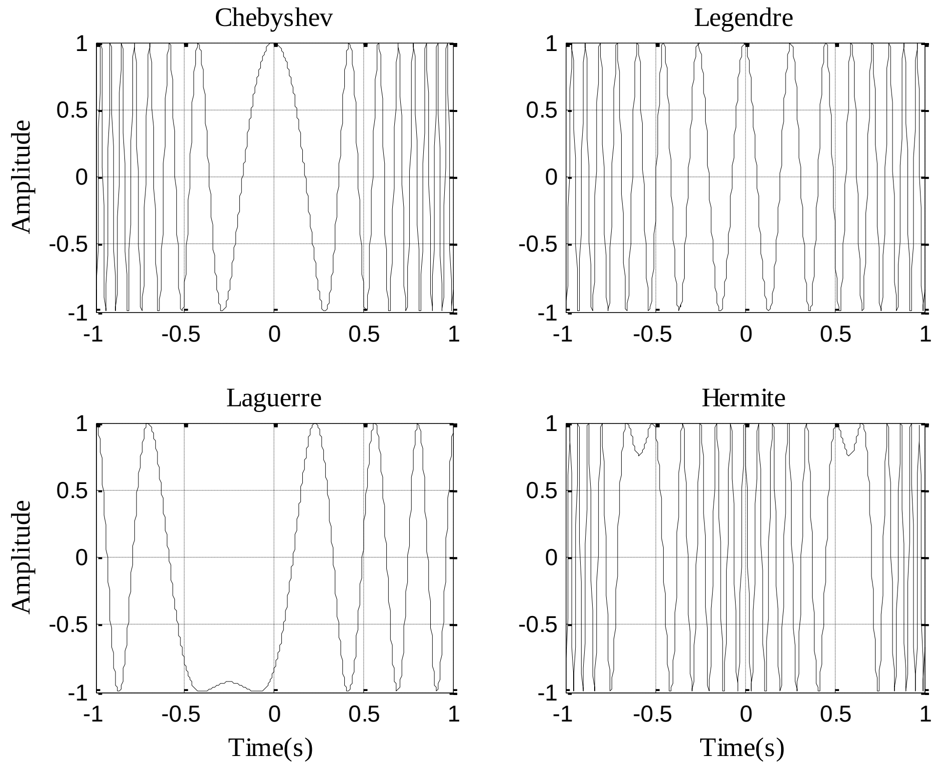


Figure 3.1: Transmitted signal with different Classical Orthogonal Polynomials

Figure 3.2 represents the frequency spectrum for different optimizing parameters for all four types of polynomials. First, second, third and fourth columns are for Chebyshev, Legendre, Laguerre and Hermite Polynomials respectively.

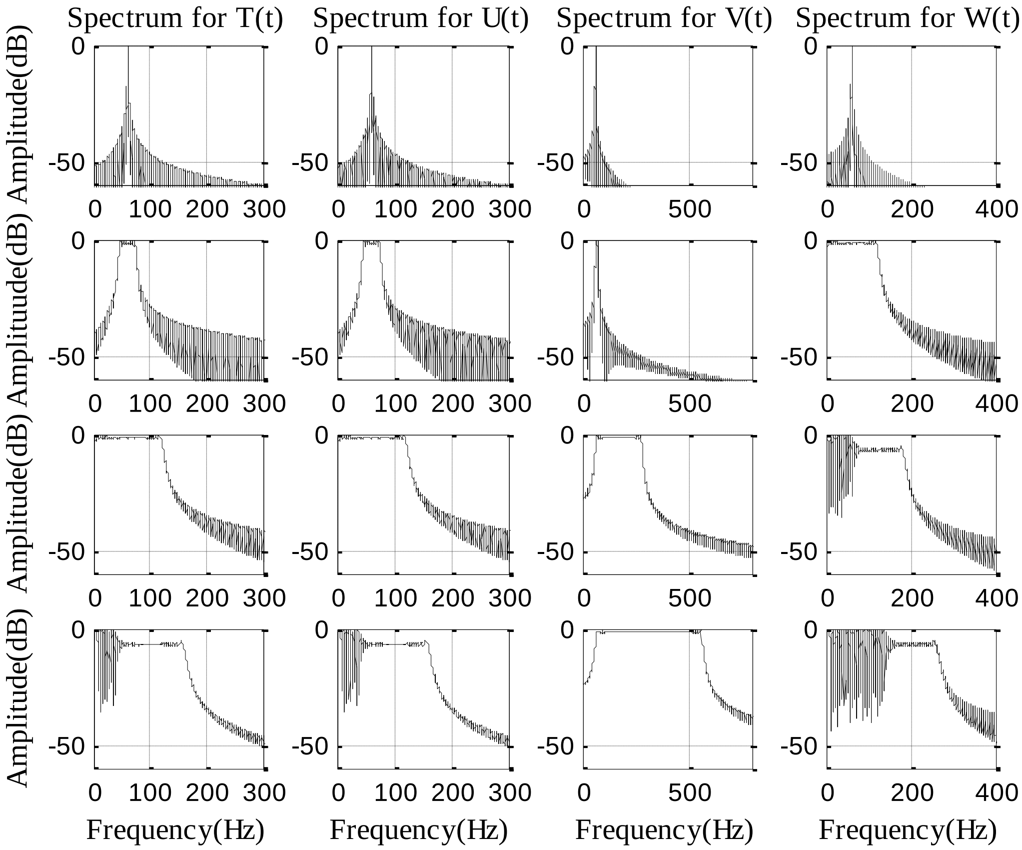


Figure 3.2: Spectrum for Classical Orthogonal Polynomials having order one and different values of α

From Figure 3.2 it is observed that by increasing the α from small number to large number, the spectrum of the signal is spreading smoothly from narrow band to large band and then there is a distortion in the spectrum distribution. This can be observed in the column-wise plots. This indicates that there is an optimal value of α which maximizes the time-bandwidth product which is the main requirement in pulse compression. This has been calculated for all types of polynomials and tabulated. The optimal value is possible for all the Classical polynomials except for the Laguerre polynomials which are deviated from the rest of the polynomials. This can be attributed to the fact that Laguerre polynomials are monotonic in the entire time duration and the variations in the arguments are too fast (very high frequency). If there is a practical device which can support such a huge frequency variations with high accuracy in a short period of time (restriction on the physical reliability of the source), the Laguerre polynomials are a better choice as far as the pulse compression is require.

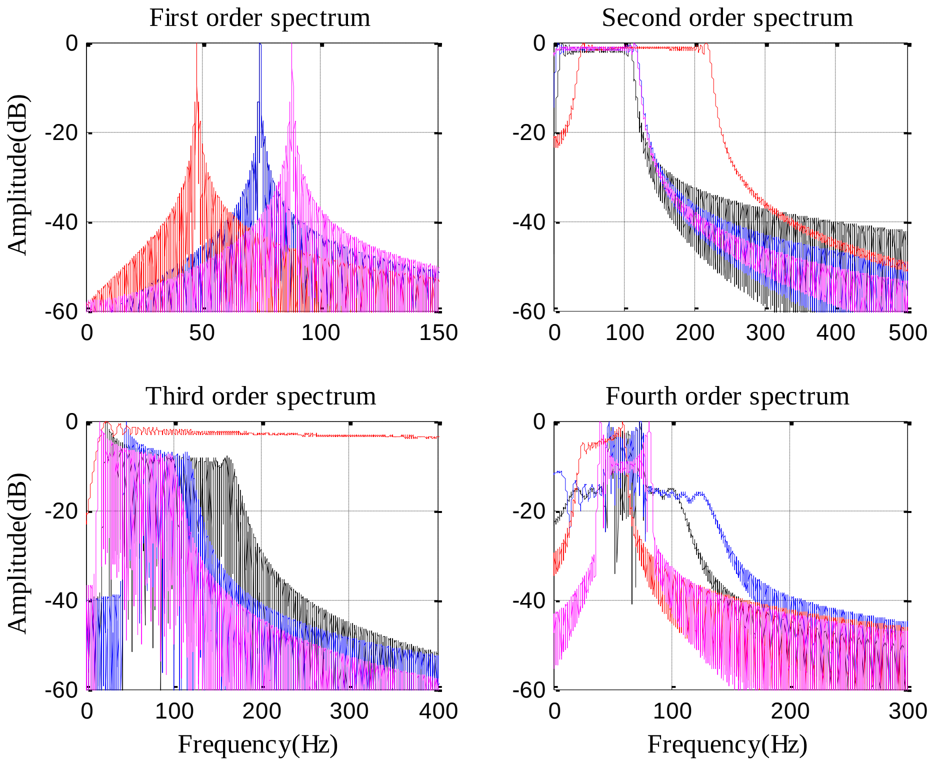


Figure 3.3: Spectrum for Classical Orthogonal Polynomials having different order in Phase Modulation

A comparative analysis on all the spectral properties of these transmitted pulses have been carried out with the three above mentioned cases. Figure 3.3 to Figure 3.5 represents the spectral variations of different order Classical polynomials in phase modulation (first case), frequency modulation (second case) and finally the general case of angle modulation. In which Black, Blue, Red and Magenta represents Chebyshev, Legendre, Laguerre and Hermite polynomials respectively. The colour of the traces has been preserved in all the succeeding figures for comparison.

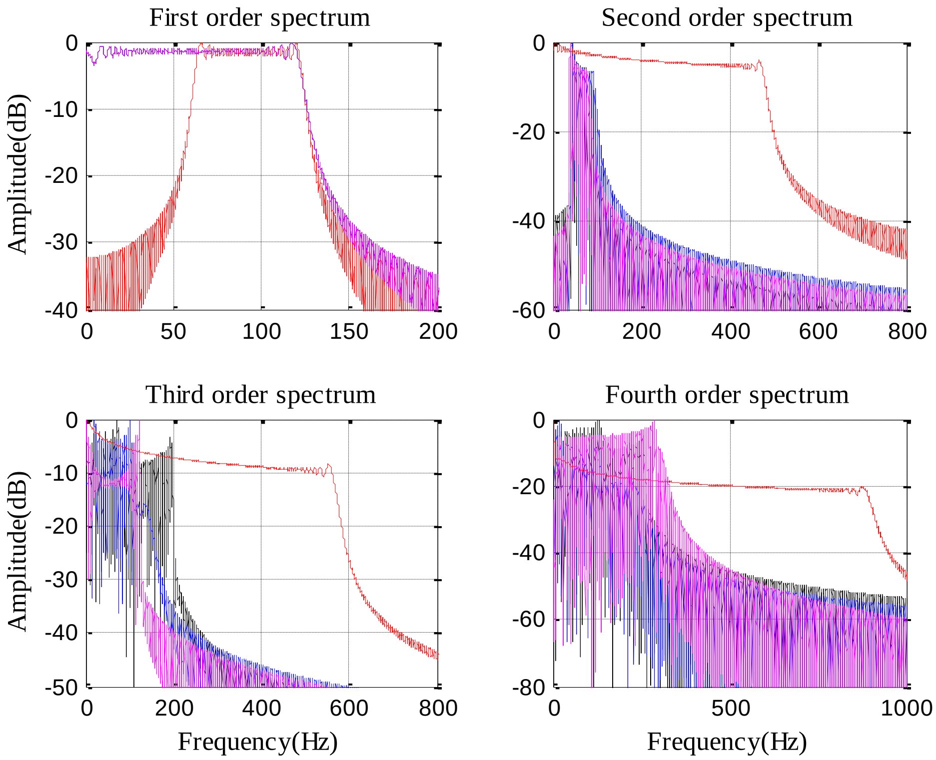


Figure 3.4: Spectrum for Classical Orthogonal Polynomials having different order in Frequency Modulation

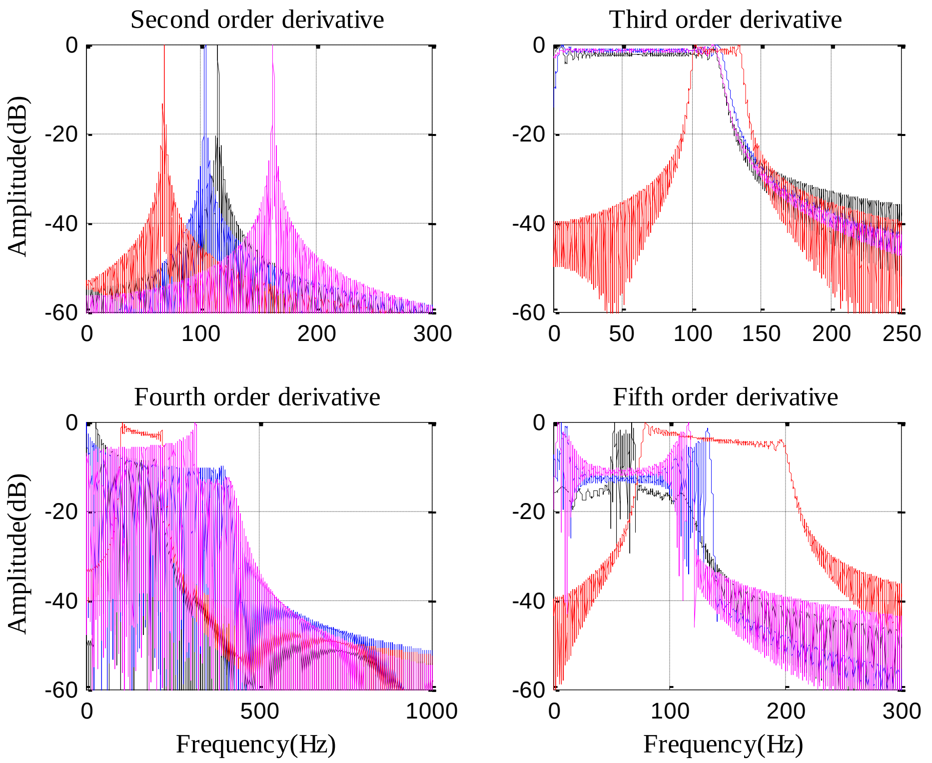


Figure 3.5: Spectrum for differentiated Classical Orthogonal Polynomials having different order

The spectrum is observed for order of two, three and four for proportional, integral and derivative variations and the results are tabulated in the Table 3.2, Table 3.3 and Table 3.4 for phase, frequency and general angle modulations.

Table 3.2: Optimal values (α) and bandwidth for Classical Orthogonal Polynomials in Phase modulation

Polynomials	2nd order		4th order	
	α	B.W.	α	B.W.
Chebyshev	85	110	21	82*
Legendre	130	117	50	116*
Laguerre	600**	178**	21	35
Hermite	48	117	3	50*

* Represents averaging the spectrum is required.

** Laguerre polynomials whose spectrum is increasing up to any optimising value.

Table 3.3: Optimal values (α) and bandwidth for Classical Orthogonal Polynomials in Frequency modulation

Polynomials	1st order		2nd order		3rd order		4th order	
	α	B.W.	α	B.W.	α	B.W.	α	B.W.
Chebyshev	400	120	--	--	--	--	--	--
Legendre	400	120	--	--	--	--	--	--
Laguerre	200**	55**	760	471	570	560	610	880
Hermite	200	120	--	--	70	120	--	--

From the tables and graphs, it is observed that, when the order of the polynomial is increased, the spectral distribution is not smooth and the first order polynomials are giving better properties in case of frequency modulation compared to all other polynomials. At the same time, it is also observed that the frequency modulation with first order is similar to phase modulation with second order which is a known fact of frequency modulation and phase modulations are inter-related.

Table 3.4: Optimal values (α) and bandwidth for differentiated Classical Orthogonal Polynomials

Polynomials	3^{nd} order derivative		5^{th} order derivative	
	α	B.W.	α	B.W.
Chebyshev	16	118	2	115^{*}
Legendre	27	118	5	135^{*}
Laguerre	120^{**}	35^{**}	40^{**}	120^{**}
Hermite	8	118	0.8	120^{*}

Even though they appear to be almost same in spectrum, there is a slight difference in the spectrum due to the difference in the functions in the arguments. Out of these two options, frequency modulation is better than the phase modulation. In frequency modulation, by comparing time-bandwidth product in polynomials to polynomials, Except Laguerre, all other polynomials have good bandwidth in first order. Second ordered Laguerre polynomial has good bandwidth that is 500Hz at α is equal to 760. Polynomials having order three, Laguerre and Hermite gives flat spectrum but Laguerre have better result that is bandwidth of 560Hz at α is equal to 570. Laguerre polynomial having order four gives better bandwidth than other order polynomials that is 880Hz at α is equal to 610. So Laguerre polynomials are better in order to improve time-bandwidth product than other polynomials. It is also observed that when α's value is increased then spectrum of the signal is spreading but there are some ripples in the response along with attenuation in the spectrum. These ripples can be averaged in order to make the response smooth.

In phase modulation, polynomials having order one have not any optimal bandwidth. For order two all the polynomials have good bandwidth with different α values. Polynomials having order four, Hermite have better bandwidth that is 50Hz at α is equal to 3.Finally, the spectral contents are observed for higher order polynomials and they are not suitable for the pulse compression as the spectral distribution is highly non-uniform as represented in Figure 3.6. Here the order of the polynomials (for Chebyshev) is taken as 2 and 31.

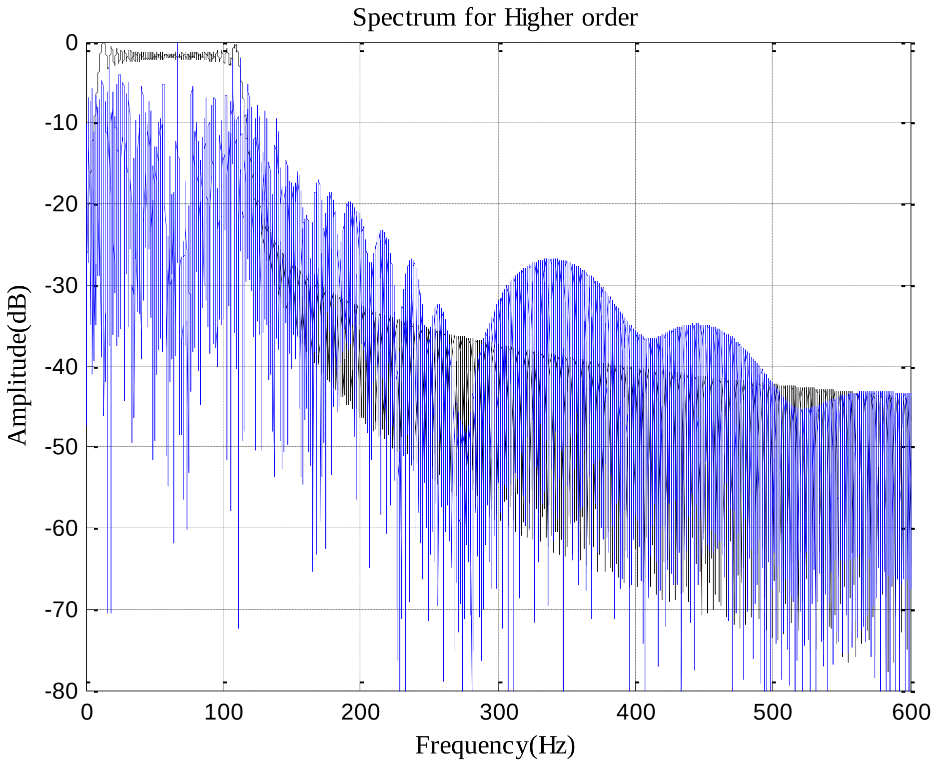


Figure 3.6: Spectrum for Chebyshev Polynomial of *n*=2 (Black) and *n*=31 (Blue)

3.4 Conclusion

The complete detailed analysis on pulse compression with the Classical Orthogonal polynomials is carried out and it is concluded that the Laguerre polynomials are a better choice for pulse compression if there is no restrictions on the physical implementation of the source. After Laguerre polynomials, Legendre polynomials are giving better time-bandwidth product for frequency modulation with order one. After the analysis, it is observed that the Laguerre polynomials are better than others with respect to pulse compression and frequency modulation gives better time-bandwidth product for these polynomials than phase modulation.

CHAPTER-4

SEQUENCE GENERATION WITH CHEBYSHEV POLYNOMIAL HAVING SMALL RELATIVE SIDELOBE LEVEL

4.1 Introduction

This chapter describes the improvement in PSL, time-bandwidth product and AF with Chebyshev polynomial of different orders. This new signal is obtained by changing the polarities of the pulses depending on the zero-crossing [13, 14]. In maximizing the output of MF, PSL and relative main lobe width are the main parameters to indicate the suitability of the pulse to avoid the ambiguities in range and velocity at the radar receiver. Here two cases has been considered to analyse the PSL, time-bandwidth product and AF behaviour for Chebyshev polynomial of different orders [15, 16], second is modification in the cycles of Chebyshev polynomial of different order is incorporated. Also time-bandwidth product [2, 3] is analysed for the modified polynomials. A binary sequence with bi-polarity has been multiplied these pulses to get reduce further in the side lobe level. After this the smallest duration of the pulse has been used in determining the optimal duration of binary sequence to have the smallest MSE [17, 18] between the number of pulses incorporated and original sequence. This is giving a much larger sequence with less PSL by reducing the search domain considerably. It is observed that the polynomial which is derived with half cycle modification have better results for PSL at the output of ACF. By using this approach higher order polynomials PSL can be observed with the help of low order polynomials that saves the simulation time.

The transmitted signal in radar system [1] can be represented as

$$x(t) = \alpha(t)\, cos(2\pi f t + \varphi(t)) \qquad (4.1)$$

Where $\alpha(t)$ represents the AM whose amplitude varies with respect to the time and $\varphi(t)$ represents the angle modulation. The received signal can be represented as

$$r(t) = cos(2\pi f t + \ \Omega(t) - T_d) \qquad (4.2)$$

Where $\Omega(t)$ can be taken as the phase differences coming from relative velocity between the target and the radar system, while T_d can be taken as the round trip time delay [3]. This received signal is passed through the MF in order to improve the signal to noise ratio [4, 5].

In summary, during the process of pulse compression side-lobes are undesired. In two ways these side-lobes can be suppressed [1, 3, 4] in one matched filter's output can be passed through some weighting filters but by this technique SNR of the system is reduced to some extent. In the second method, the search for the transmitting waveform which has low side-lobes in the MF response can be used during angle modulation. In this chapter second case has been considered to reduce the side-lobes, so the search has to be conducted for the best possible function $\varphi(t)$ as represented in equation 4.1, such that the spectrum of $x(t)$ has to spread flatly over the large band of frequencies. There are many functions possible for $\varphi(t)$, but here this function is confined to the Chebyshev polynomial due to the wide applications of this polynomial in engineering domain [13, 16]. This chapter presents the detailed analysis on side-lobes reduction in pulse compression with respect to modified Chebyshev polynomial and also their AF are observed, which tells about the range and velocity resolution.

4.2 Generation of the Sequence for Chebyshev Polynomial

Chebyshev polynomial introduces a trigonometric mapping method which produces a real valued sequence having better auto-correlation property [14, 15]. Here two cases has been considered to check the PSL behaviour, in the first case Chebyshev polynomial are analysed, after that cycle of the Chebyshev polynomials are modified by multiplying the Chebyshev polynomials cycle with some +1 or -1 polarity which are generated by polynomial zero-crossing, basically two possibilities exist to change the cycle of polynomial i.e. half cycle and full cycle can be changed. Here both modifications have been done and their MF responses are analysed. Depending upon the order of the polynomials there are many possibility to change the cycle of polynomials, here out of these possibilities best case (gives better PSL) has been considered, after that MF response is analysed for these modified polynomials. In order to generate full cycle modification, only odd number ordered polynomial has to be considered, if even number order polynomials are incorporated for full cycle modification then some part of the signal is truncated which alter the desired results. Whereas half cycle modification are possible for both (even and odd) ordered polynomials.

Recursive equation for Chebyshev polynomial of order n can be calculated [13] as represented in Table 4.1.

Table 4.1: Chebyshev Polynomials of order *n*

Order(n)	Chebyshev polynomial(T_n)
0	*1*
1	x
2	$2x^2 - 1$
3	$4x^3 - 3x$
n	$T_{n+1} = 2xT_n - T_{n-1}$

Steps to generate sequence:

Step 1: Take a Chebyshev polynomial.

Step 2: Modify cycles of the polynomial in the following manner.

- Find the zero crossings of the polynomial. Count the number of positive and negative pulses in the region -1 to 1.
- Generate all the possible binary sequences according to the number of pulses obtained above.
- Multiply this binary sequence with Chebyshev polynomial.
- Out of these possibilities, the smallest PSL is stored and designate this as best PSL.

Step 3: Find the optimal duration (τ_{opt}) for the modified half cycle polynomial.

- Calculate least duration of the pulse by zero-crossing and divide by it with overall length of the polynomial.
- Get the fractional duration of all the pulses.
- Round-off this duration to its nearest integer value.

Step 4: Calculate mean square between rounded and actual duration.

- Take the duration which has least MSE.

- Calculate numbers of cycles that can be added to the modified half cycle polynomial.
- For the above obtained durations, expand the each pulse by integer number of τ_{opt} duration.

Step 5: Analysed their MF response.

4.3 Simulation Results and Discussion

ACF for *n*=27 is analysed as represented in Figure 4.1, first row just gives time-domain representation of Chebyshev polynomial and its MF response. It is observed that PSL level are not suppressed, which may lead to the unwanted target detection. In order to suppress the PSL, modifications in the cycle of Chebyshev polynomial is incorporated, here in Figure 4.1 full cycle modification is done (Row two) for this particular polynomial, red signal is used for polarity generation that will multiply with the above Chebyshev polynomial in order to change the cycles of the polynomial and this modified full cycle polynomial MF response is analysed.

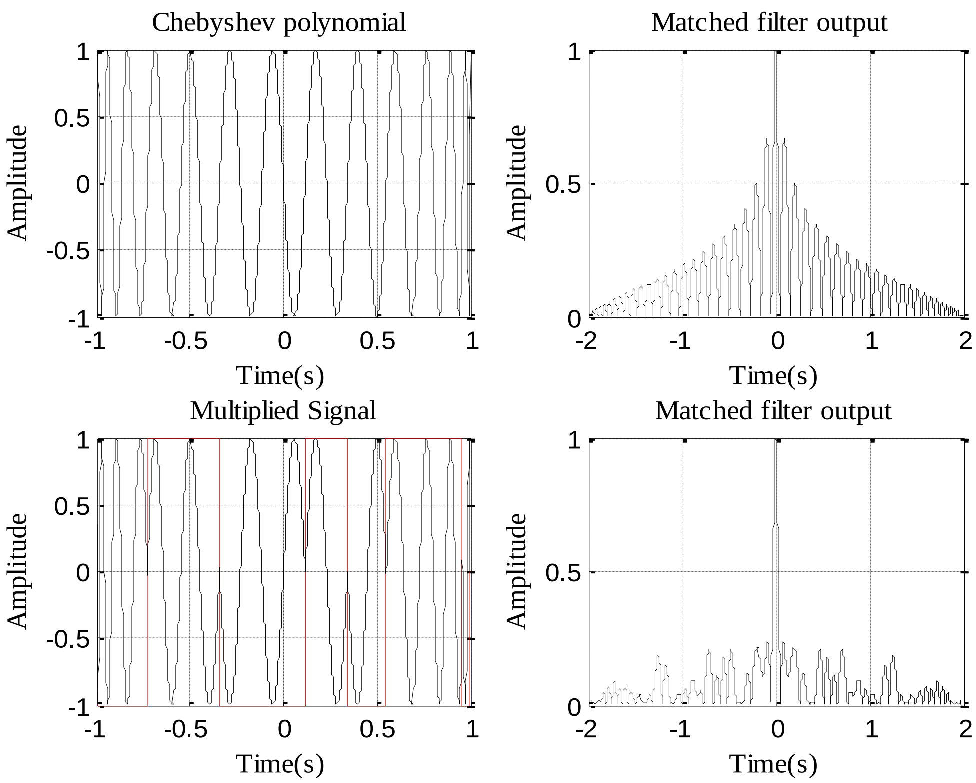


Figure 4.1: Time-domain representation and MF output of Chebyshev Polynomial for *n*=27

It is observed that it have better PSL than simple Chebyshev polynomial. Here *n*=27 is taken just for convenience, by altering order of these polynomials overall conclusion does not changed. Figure 4.1 represents the time-domain representation and MF response for Chebyshev polynomial (First Row) and Full cycle modified polynomial (Second Row) for *n*=27. In the same manner, for *n*=15 Chebyshev polynomial response is observed which have no better PSL, so modification in the cycle is incorporated to reduce PSL, here half cycle are modified for the particular polynomial and its MF response is analysed, it is observed that this half cycle modified Chebyshev polynomial have better PSL as represented Figure 4.2 to all other observed cases, first row corresponds to the Chebyshev polynomials and second row corresponds to the half cycle modified Chebyshev polynomial.

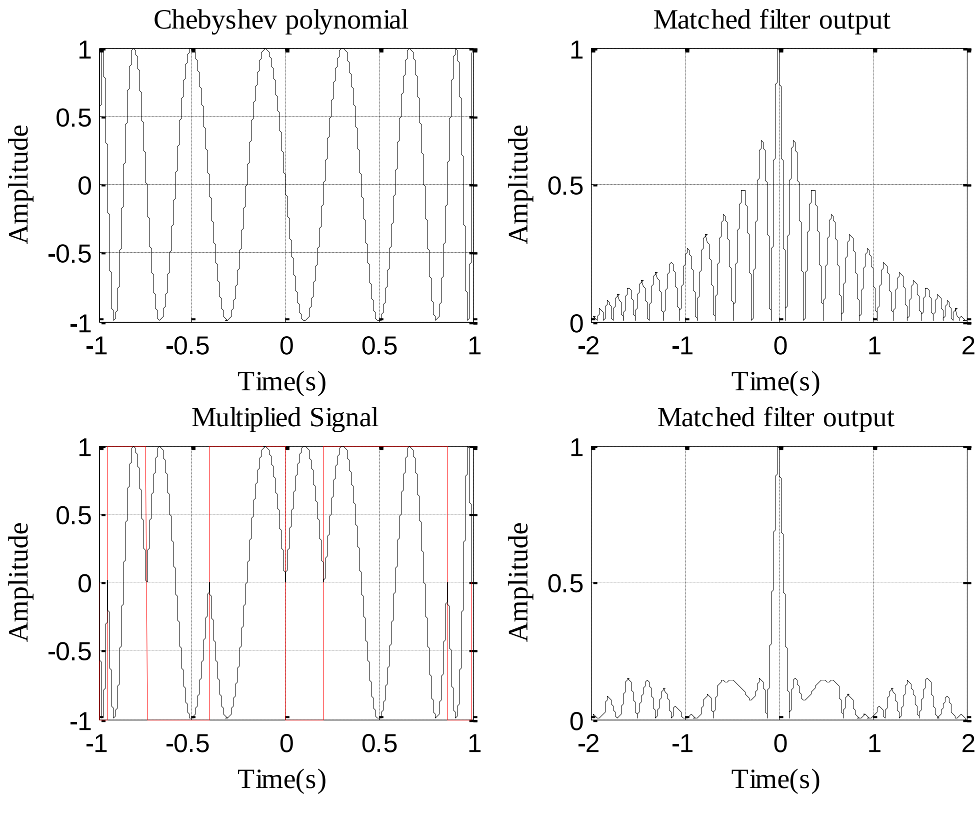


Figure 4.2: Time-domain representation and MF output of Chebyshev Polynomial for *n*=15

Using this step by step approach, PSL of Chebyshev polynomial, full and half cycle modified polynomial are observed and represented in the Figure 4.3. This represents the best PSL level out of all the possibility of the different Chebyshev polynomials (Black : Chebyshev polynomial, Blue : Full cycle modified polynomial and Red : Half cycle modified Chebyshev polynomials) in the MF response and it is observed that half cycle modified polynomial gives better PSL than other two cases and Chebyshev polynomial gives worst PSL.

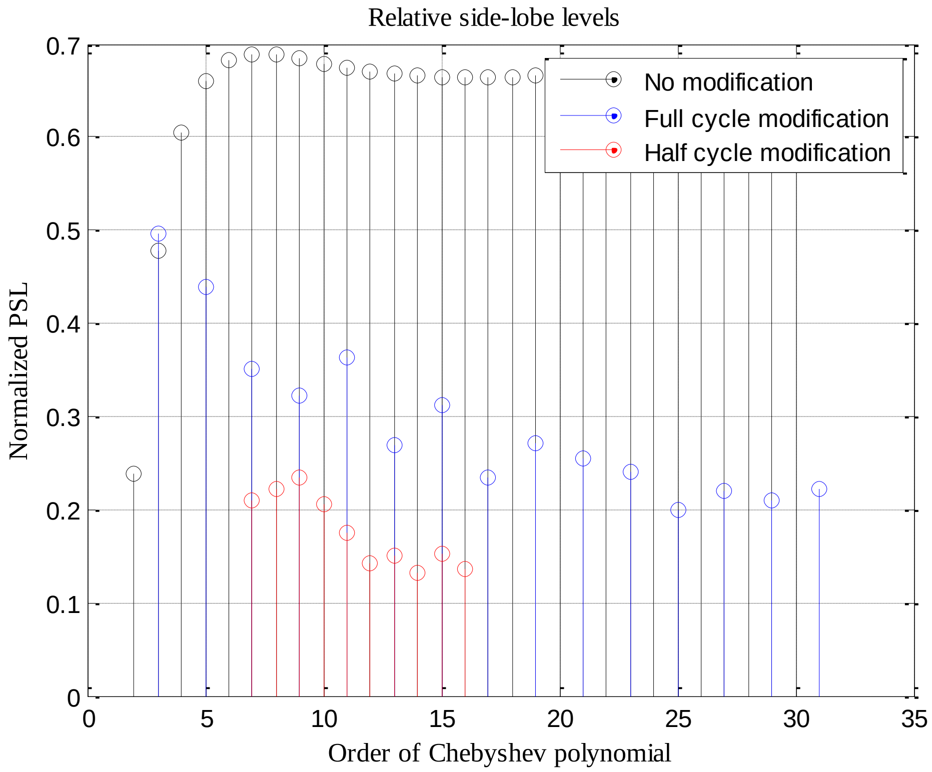


Figure 4.3: Relative side-lobe levels for different Chebyshev Polynomial

As time-bandwidth product is the main parameter in pulse compression it should be more for good detection as well as for good resolutions. If bandwidth of the transmitting signal is high, then the pulse duration is small and this gives a good range resolution. Usually duration of the pulse cannot be reduced too much because Fourier theory says that a signal having bandwidth *B* have always duration greater than *1/B* i.e. its time-bandwidth product is always greater than unity. This time-bandwidth product usually describes in term of PCR [22, 23] which is the ratio of width of the pulse after compression to the width of the pulse before compression. If its value is greater than one then pulse compression exists. After pulse compression overall bandwidth of the transmitted pulse has been increased which responsible for range resolution and the long duration pulse which is transmitted in time-domain that responsible for detection of the object. So in Figure 4.4 spectral contents of Chebyshev polynomials are compared and it is observed that the modified half cycle polynomial have considerably wider frequency response than the simple Chebyshev polynomial, these frequency spectrums are obtained with the help of FFT [7].

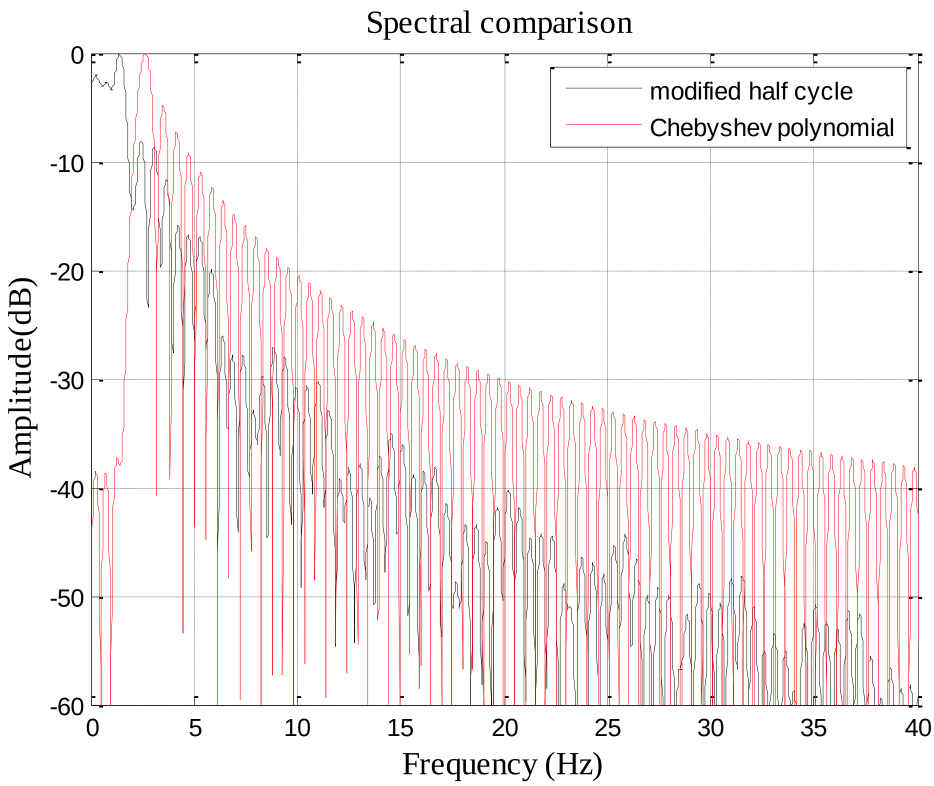
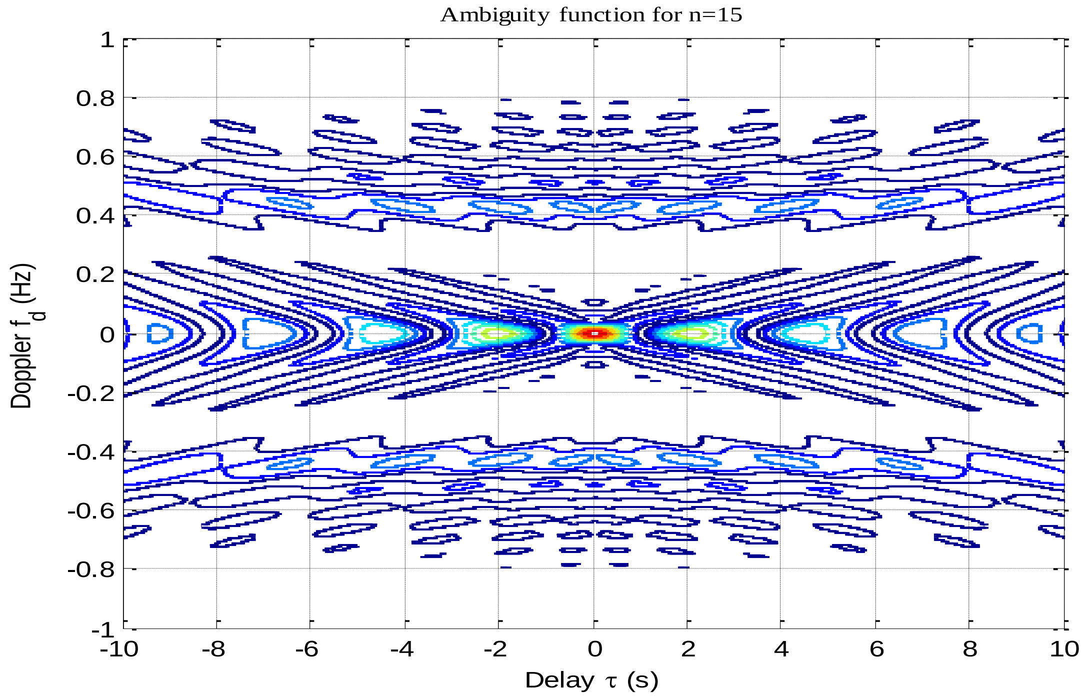
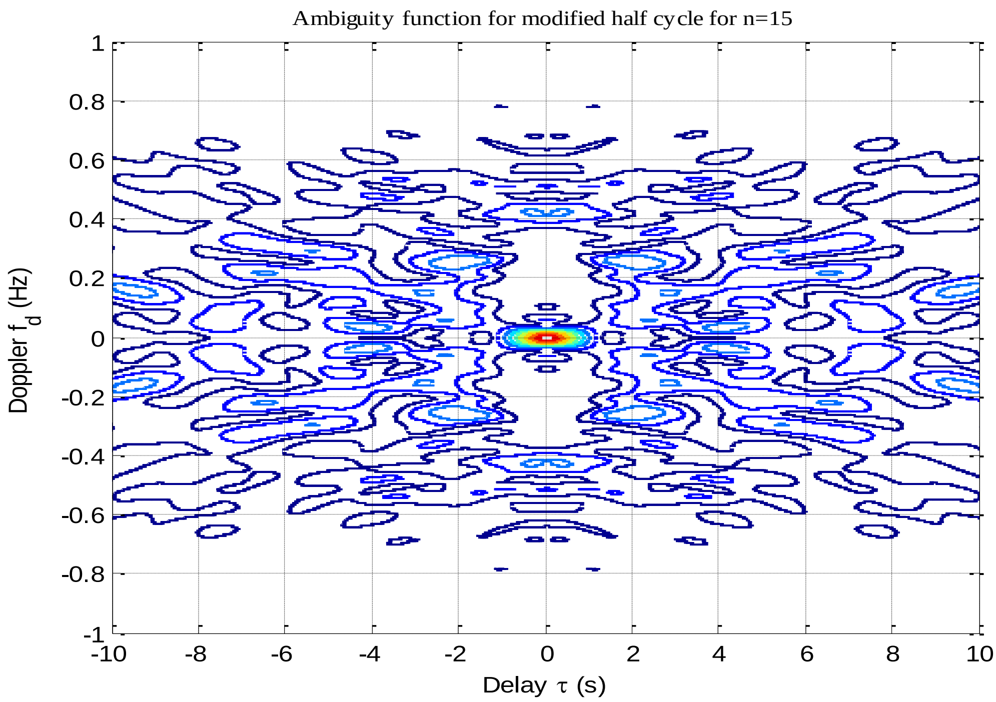


Figure 4.4: Spectral comparison for *n*=15

In order to observe the range and velocity resolution; AF for Chebyshev polynomials (Figure 4.5(a)) and modified half cycle polynomials (Figure 4.5(b)) are analysed. From the simulation it is observed that for simple Chebyshev polynomials delay-Doppler plane have not uniform energy spreading which may lead to the unwanted target detection and its resolution is not good for range and velocity determination, it means these polynomials cannot be used for resolution purposes. Now for half cycle modified Chebyshev polynomial's ambiguity analysis [2, 4] are carried for these modified polynomial delay-Doppler plane is more concentrated at the origin which tells about the target characteristics without any ambiguity in range and velocity. Similarly for full cycle modified polynomials AF for order 27 is analysed in Figure 4.6 and from the simulation, it is observed that half cycle modified Chebyshev polynomials are good for range as well as velocity resolution because first one AF is more concentrated around the origin in delay-Doppler plane than latter one.

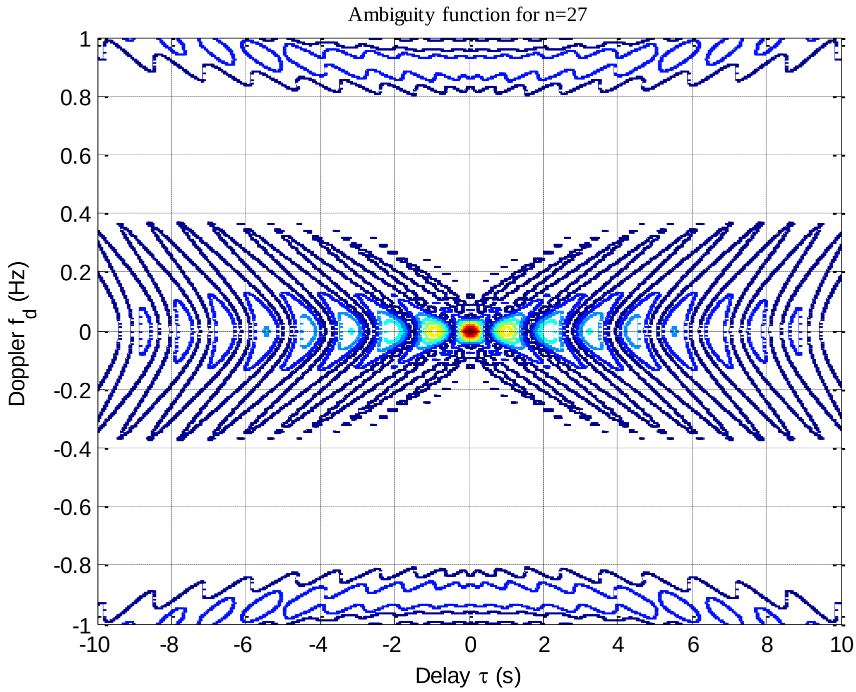


(a)

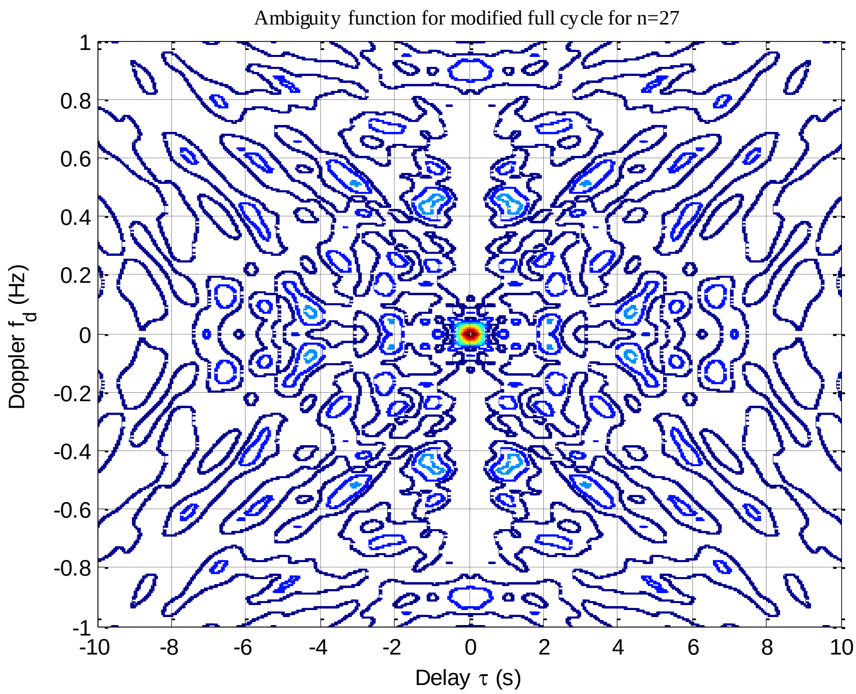


(b)

Figure 4.5: AF for *n*=15 (a) Chebyshev Polynomial and (b) modified full cycle Polynomial



(a)



(b)

Figure 4.6: AF for *n*=27 (a) Chebyshev Polynomial and (b) modified full cycle Polynomial

Once this half cycle modification in the polynomial is done then calculate the least duration of the cycles, here it named as optimal duration (τ_{opt}). After this divide all cycle duration with this τ_{opt} and rounding these durations to its nearest integer value. Now all the cycle duration is an integer multiple to the optimal one. Depending on this duration, numbers of cycles are calculated that are added into modified polynomial. MSE is calculated between original and rounded duration of the cycles after division with the least duration pulse. Then only that duration of the modified half cycle polynomial is taken which has least MSE. After this appends the number of cycles having duration of the τ_{opt} in the modified half cycle polynomial and finally their MF response are analysed.

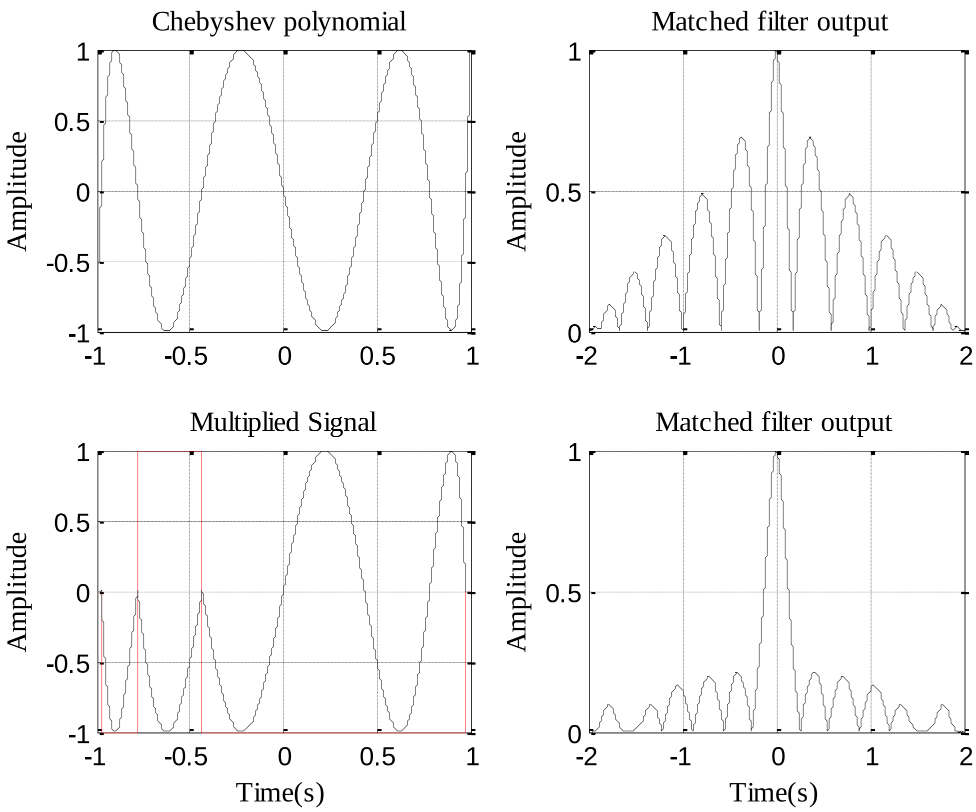


Figure 4.7: Time-domain representation and MF output of Chebyshev Polynomial for n=7

Here in Figure 4.7 order of the polynomial is taken as seven and it is plotted in time-domain which consists of three complete cycles, it means for odd number ordered polynomials $\frac{n-1}{2}$ complete cycles will exist, ACF for this polynomial is observed and it is concluded that output of the MF have some high PSL which are undesired, because these side-lobes may contains the information associated with some nearby targets. In order to suppress these side-lobes some modification in the cycles of the polynomial is incorporated. Half cycle modification has been considered because it is giving better PSL than full cycle modified polynomial.

All cycles in the polynomial are not of equal duration as observed in the Figure 4.7. Let τ_{opt} is the optimal duration of modified half cycle polynomial which can be calculated by zero-crossing of cycles and divide this duration by total length of the polynomial. This τ_{opt} is calculated only for that modified polynomial which has better PSL in the MF response and then divide all the half cycle duration of modified polynomial with τ_{opt}. It is clear that after dividing, duration may be in the form of fraction, so this fractional duration is rounded to its nearest integer value. It means number of pulses which are integer multiple to the optimal duration can be added into the modified half cycle polynomial, each added cycle have duration equal to the optimal one.

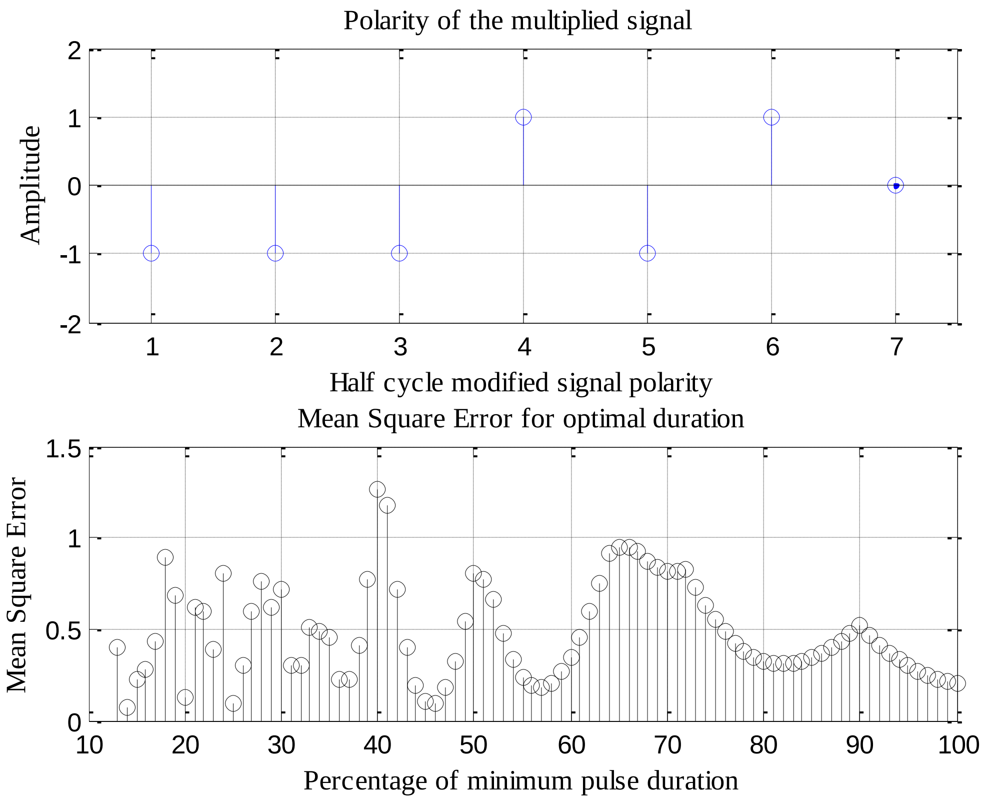


Figure 4.8: Multiplied signal polarity and optimal duration MSE for *n*=7

During rounding the fractional duration to its nearest integer number, some part of the signal may be discarded that alter the performance of the system. So only those modified polynomials are considered which have MMSE. $e(t)$ is the error deviation [18] in the actual $(x(t))$ and the rounded duration $(\hat{x}(t))$.

$$e(t) = x(t) - \hat{x}(t) \qquad (4.3)$$

So MSE can be calculated as

$$MSE = \frac{1}{T}\int_0^T e(t)^2 dt \qquad (4.4)$$

T is the overall duration of the polynomial.

Here Figure 4.8 represents the polarities of the multiplied signal and percentage of error present in rounded duration with respect to the original duration after division for order seven. Only those modified polynomial sequence whose MSE is small and contain maximum duration of the optimal one, is considered here. First three best cases have been analysed for order seven and eight. It is observed that order seven has low MSE at 82 % (first optimal), at 57% (second optimal) and at 46 % (third optimal) of the minimum duration pulse, depending on these particular points, modification in the cycles is incorporated by adding more number of cycles, for first optimal (5,9,12,12,9,5) cycles can be added, it means five cycles for first duration having polarity -1 , nine cycles for second duration having polarity -1, twelve cycles for third duration having polarity -1 and so on, can be added in seventh order modified half cycle polynomial. Each added cycle has duration equal to the optimal one. Here polarities of these added cycles are same as that of the modified polynomial. Now modified polynomial contains total 52 half cycles, whose ACF is observed as shown in Figure 4.8 and it has better PSL. Similarly for the second optimal case, numbers of cycles that can be added to the modified polynomial are of total length 22 cycles that are of (2,4,5,5,4,2), for first duration two cycles, second duration four cycles, third duration five cycles and so on can be added. In the same manner for third optimal, number of cycles are calculated and there PSL are observed.

From the simulation it can be concluded that by appending the cycles in to modified half cycle polynomial, level has been suppressed and overall numbers of cycles have been increased, this corresponds to the higher order polynomial. So using low order polynomials, PSL behaviour of higher orders can be observed by which simulation time can be saved to run the high order polynomials. It is also observed that by adding Barker codes in to the modified polynomial have not suppressed PSL in the MF response. No doubt side-lobes have been suppressed but main-lobe width is increased slightly that disturb the range resolution. It happens in pulse compression if side-lobes are suppressed then its main-lobe width is increased slightly as observed in Figure 4.8 and vice versa. Here Figure 4.9 represents the polarities of the multiplied signal and percentage of error present in rounded duration with respect to the original duration after division for order eight.

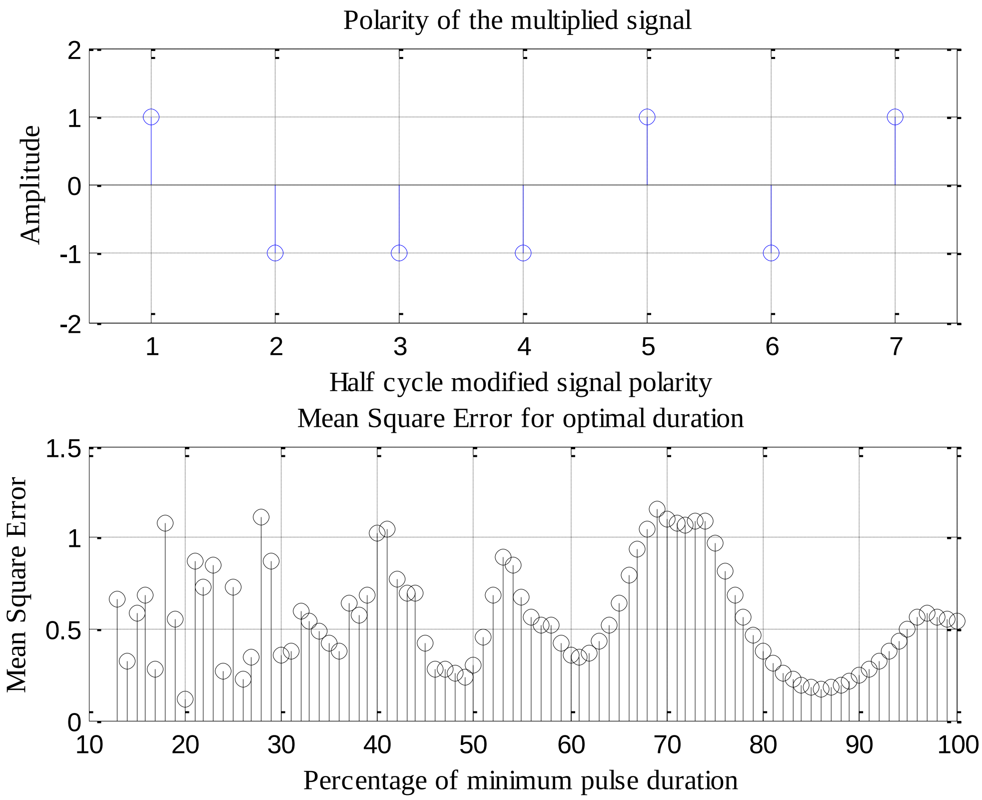


Figure 4.9: Multiplied signal polarity and optimal duration MSE for *n*=8

Order eight have multiplied signal polarities are [+1 -1 -1 -1 +1 -1 +1]. In the same way, first three optimal duration having least MSE are considered and numbers of cycles are calculated which can be added into the modified half cycle polynomial. First best exist at 86 % of the optimal duration, have number of cycles that can be added are (7,12,16,17,16,12,7), second optimal exist at 61% of the optimal has (3,5,6,7,6,5,3) cycles and third optimal exist at 49 % of the minimum duration pulse. Append numbers of cycles into modified polynomial depending on these optimal cases. Now ACF are analysed for these modified polynomials having order eight as shown in Figure 4.10 and it have some improvement in PSL. So these half cycles modified polynomial can be used in pulse compression in order to suppress the PSL in place of $\varphi(t)$ in equation 4.1.

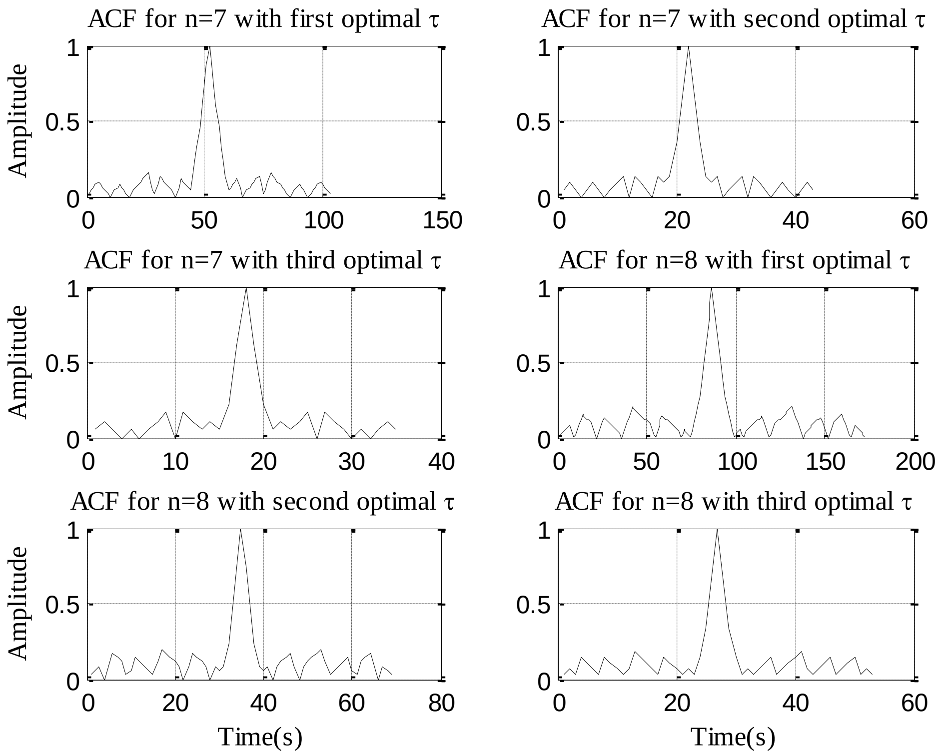


Figure 4.10: ACF for *n*= 7 and 8 having different optimal duration

4.4 Conclusion

After the detailed study, it is observed that half cycle modified Chebyshev polynomial has better PSL in the ACF. Along with this, there is slightly improvement in the time-bandwidth product exists. Multiplying sequences which generate the full cycle and half cycle modifications have the same auto-correlation response. Simulated results shows that AF is more concentrated towards the centre, that means good range and velocity resolution exists. By adding more number of cycles which have duration equal to the optimal one and have least mean square error into the modified half cycle polynomial, gives better results for PSL. Drawback of this approach is to increase the main-lobe width of ACF response that affects the range resolution. It is also concluded that by using low order polynomial, ACF for higher order polynomial can be observed which saves the simulation time. So these modified polynomials can be used for transmission in radar system to suppress the PSL.

CHAPTER-5

SIDE-LOBE LEVEL REDUCTION USING WOO FILTER

5.1 Introduction

In this chapter a technique for the reduction of side-lobes in pulse compression is presented. The peak side-lobe level, integrated side-lobe level and relative main-lobe width are computed using P4 polyphase codes, and compared the proposed technique results with other pulse compression techniques such as Woo filter as well as modified Woo filter [24, 25]. This new pulse compression technique is implemented by shifting the input P4 polyphase codes and multiplied it with reference signal in the frequency domain, after that the side-lobe behaviour is analysed by converting it into the time domain. It is observed the method which is introduced in this chapter gives good results for sidelobe reduction but mainlobe width is increased slightly that affects the range resolution. Woo filter is the combination of MF for P4 codes [25]. The two correlation filters are combined together to produce a single discrete filter called Woo filter. First correlator is the ACF of the original P4 code and second correlator is generated by correlating the original code with the conjugate signal of itself but shifted by one bit.

This chapter describes a new form of pulse compression filter for polyphase codes are presented which generates a flat uniform sidelobe pattern. A uniform sidelobe level pattern represents an optimum performance criterion [26] in the design of pulse compression waveforms. The side-lobes levels are decided solely by the length of the phase codes, which can be set arbitrarily. The use of P4 codes involves a small loss and degradation in range resolution, but excellent Doppler tolerance [27, 28]. Phase coded waveforms tend to be favoured when low side-lobes levels are desired. Barker codes are known to give good performance [29, 30]; they achieve unit peak side-lobes level throughout the entire sidelobe regions that is why they are known as perfect codes. However, their limited code length and high Doppler shift sensitivity restrict their applications. In radars systems, pulse compression is used to increase the range resolution. Range resolution is the capability of the system to separate two nearby targets in term of distance as two different targets [1]. Generally in radar, short and long duration pulses are required for good range and velocity resolution respectively. In pulse compression long duration pulses are transmitted by angle modulation to achieve the larger time-bandwidth. It means long duration pulse which is transmitted is responsible for detection and bandwidth which is increased after modulation is responsible for range resolution determination. For a particular waveform MF and ACF can be used to observe pulse compression response [3]; for good pulse compression main peak should be narrow and it contain low side-lobes level. The mainlobe of the filter respond to the main target but due to the presence of small PSL in the MF output that may corresponds to some nearby targets. These side lobes decreases pulse compression ratio. To reduce the PSL many pulse compression techniques has been developed. In order to suppress these side lobes in this chapter a new pulse compression technique has been developed that is based on Woo filter concepts for P4 codes.

5.2 Polyphase Codes

The codes that use harmonically related phases based on a certain fundamental phase increment are called Polyphase codes [31, 32] and by frequency modulation pulse compression waveform with either a local oscillator at the band edge of the waveform these codes are derived conceptually coherently detecting. These codes are derived by dividing the waveform into sub codes of equal duration, and using phase value for each sub code that best matches the overall phase trajectory of the underlying waveform.

Frank and P1 are well known for low range side lobes to derive these codes step frequency is used. By LFM P3 and P4 polyphase codes can be derived [33, 34] which can be used for pulse compression. The advantage of P4 codes are, it can be derived for any length sequence and these are cyclic shifted codes which gives better sidelobe reduction than other polyphase codes [35, 36]. So in this chapter P4 code is considered to analyse the response of newly developed Woo filter. Phases of the P4 codes [24] are calculated using

$$\emptyset(i) = \frac{\pi}{N}(i - 1 - N)(i - 1) \qquad (5.1)$$

Where N is total length of polyphase code.

P4 code elements are calculated by $x(i) = exp[j\emptyset(i)]$.

5.3 Performance Measures

Based on these parameters performance of this proposed model is compared with other pulse compression technique that uses woo filter concepts for P4 codes [24, 25]. These are explained as:

PSL for a code of length N measures the ratio of maximum sidelobe magnitude to the main lobe amplitude. Its value should be low as much as possible and it can be calculated as

$$PSL(dB) = 20log_{10}\frac{maxi_{1\le l<N}|M(l)|}{|M(0)|} \quad (5.2)$$

Where $M(l)$ is amplitude of lth sample number of compressed pulse.

Ratio of energy in the side lobes to the mainlobe of the compressed pulse is called as ISL. For better pulse compression its value should be low and it can be calculated by

$$ISL(dB) = 10log_{10}\frac{2\sum_{l=1}^{N-1}|M(l)|^2}{|M(0)|^2} \quad (5.3)$$

Relative mainlobe width stretched a little bit in time domain after pulse compression that disturbs the range resolution. For perfect case it should remain narrow and it is defined as

$$Relative\ mainlobe\ width = \frac{\frac{mainlobe\ width\ of\ technique\ in\ its\ PSL(dB) level}{PSL(dB) of\ technique}}{\frac{mainlobe\ width\ of\ P4\ code\ in\ its\ PSL(dB) level}{PSL(dB) of\ P4\ code}}$$

5.4 New Pulse Compression Technique

Lewis [32] argued that the adjacent cells within the P3 and P4 autocorrelation outputs differ in magnitude by no more than the magnitude of one element cell. This assertion was made intuitively on the ground that an additional input into the matched filter cannot cause variations of more than its own magnitude at the output. The time range side-lobes reduction scheme introduced by Lewis can be interpreted as a pulse compression on P codes by use of combination of two separate correlation filters. Thus a new type of compression filter is needed to achieve this optimal uniform side-lobe which is explained here.

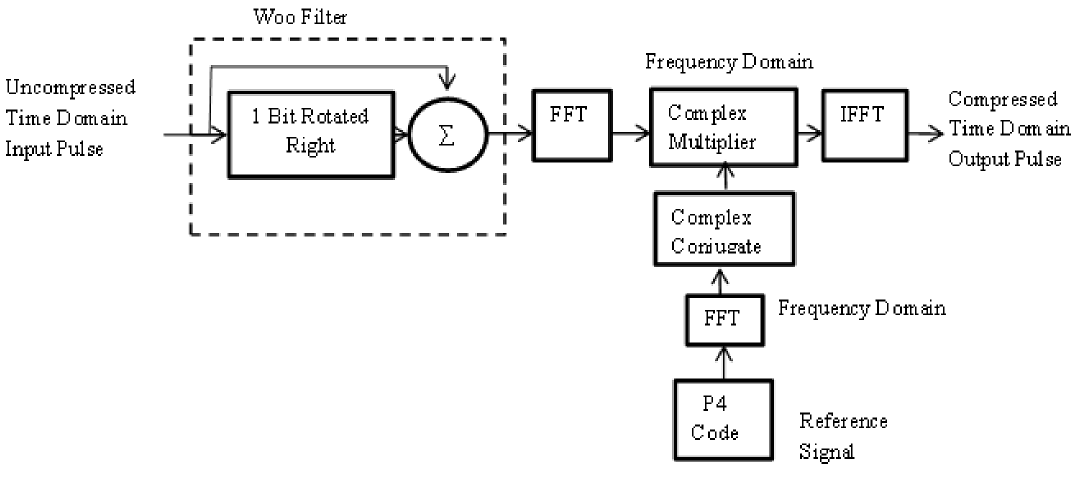
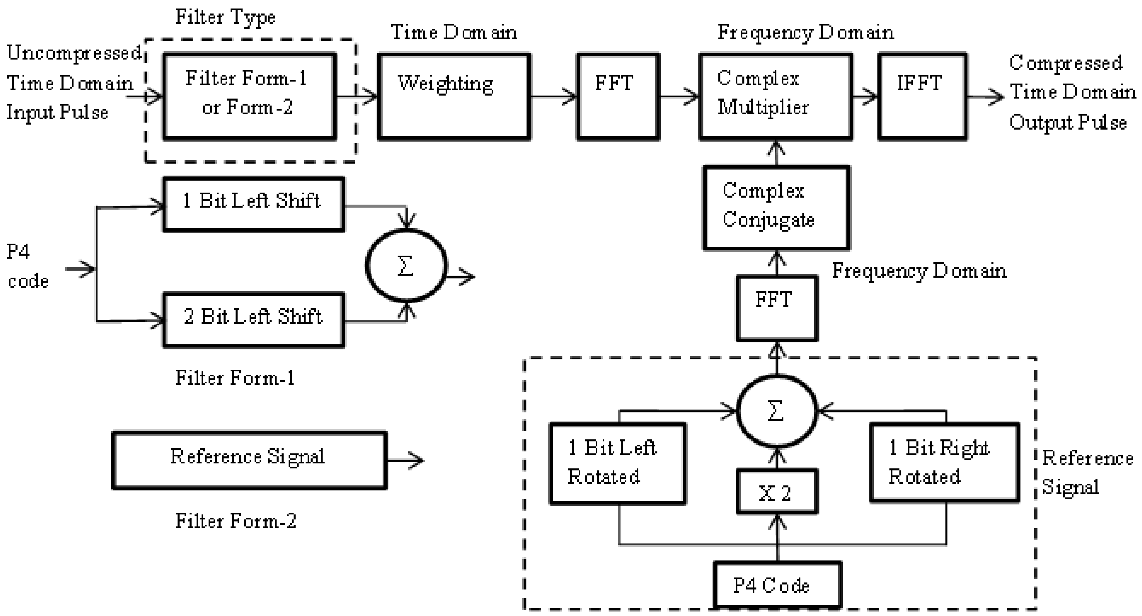


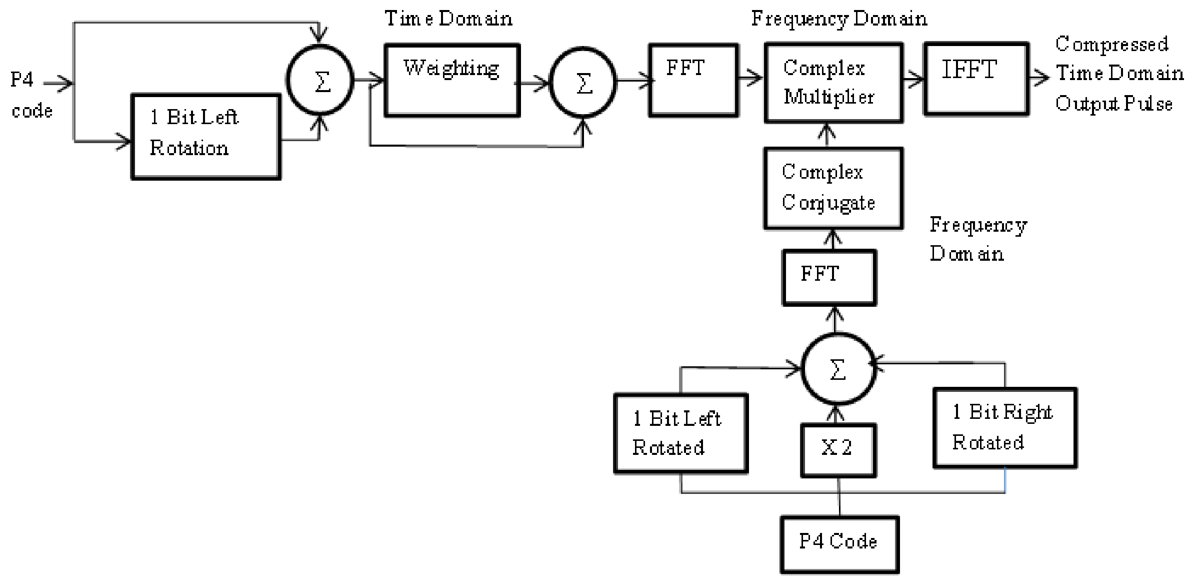
Figure 5.1: A block diagram of Woo filter based on FFT and IFFT

Figure 5.1 represents the Woo filter model which is based on FFT and IFFT. It is the combination of uncompressed input signal as well as the reference signal. Input signal is the summation of coming P4 code and one bit rotation of the code that is passed through the FFT in order to get its frequency domain. The reference signal (P4 code) is converted in to frequency domain and then complex conjugate is taken which further multiplied with the coming input signal in frequency domain. To observe pulse compression output this combined signal is converted in to time domain by IFFT.

In the proposed filter for pulse compression, reference signal is the combination of one bit left rotation and one bit right rotation and multiply the P4 code by factor two which is converted in to frequency domain. Conjugate version of this signal is multiplied with the coming input signal. This proposed filter is divided in to two forms. In first form the input signal is the summation of one and two bit left rotation of P4 codes as shown in Figure 5.2 (a). Weighting technique is also incorporated here in order to suppress the side lobes further [33]. There are many window techniques some of these are explained in this chapter. In the proposed filter form-2 the reference signal is same as for form-1, but applied input signal is also same as of reference signal as shown on Figure 5.2 (a).



(a)



(b)

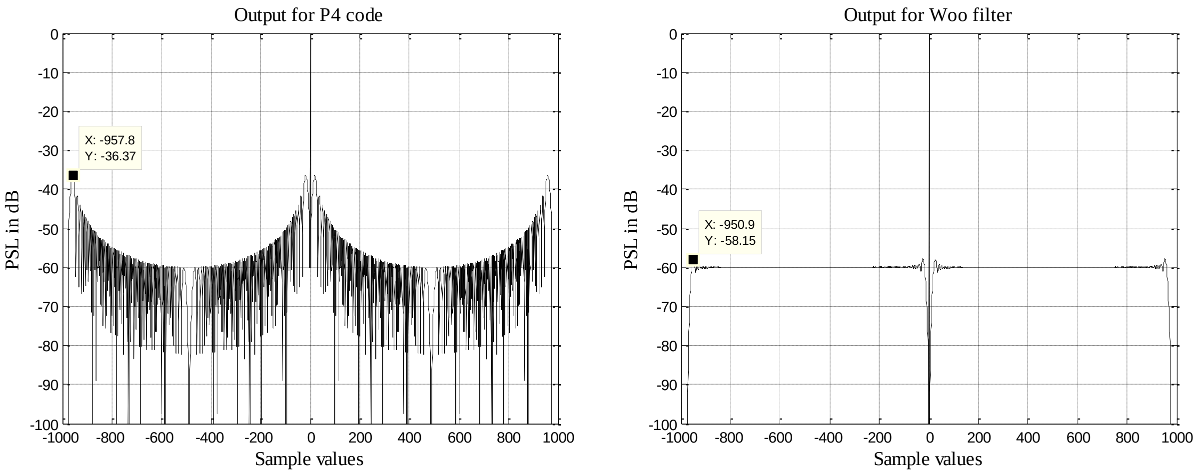
Figure 5.2: A block diagram for pulse compression (a) proposed filter-1 and (b) filter-2

In proposed filter-2, raised weighting is applied rather than simple weighting to the coming input signal in time domain because it suppress the PSL more and remaining structure is same as of filter-1.

5.5 Simulation Results and Discussion

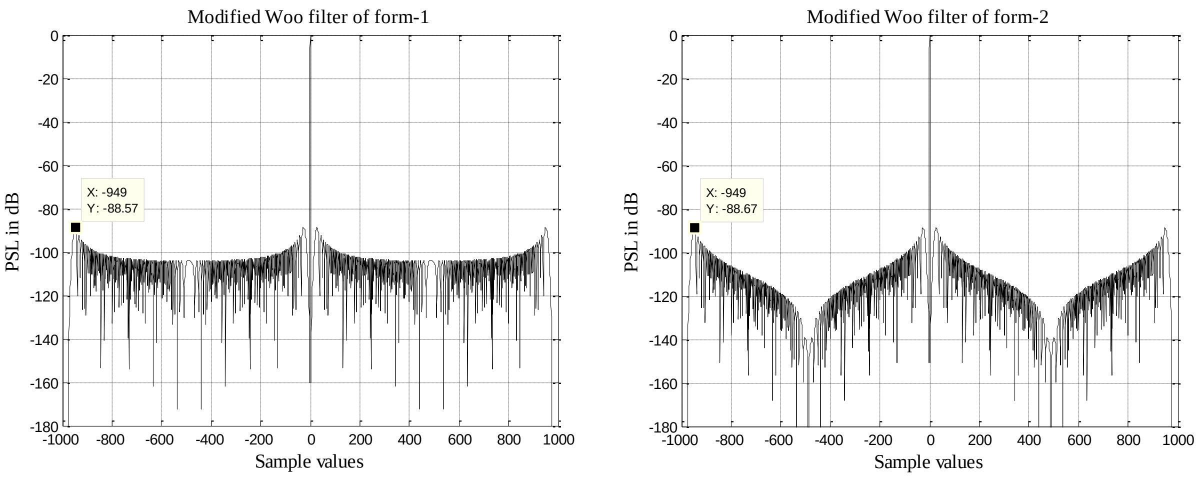
Here P4 code of length 1000 is taken, performance of this new technique is compared with other pulse compression techniques related to the Woo filter in order to suppress the side lobes. The authors [24], developed a modified Woo filter technique in which uncompressed input signal (P4) is summed by one bit rotation of the same input signal. These rotations of the bits may be right side and left side, named as modified Woo filter-1 and filter-2 respectively. Here the reference signal is same as in our proposed technique. Starting with p4 code to proposed model output for pulse compression is analysed. In Figure 5.3 (a), a P4 code output is observed having PSL value of -36.37dB to the main peak. Figure 5.3 (b), output of Woo filter [24] is observed for the Figure 5.1, it have PSL at -58.15dB. This Woo filter is further classified in to two forms [25]. In form-1, input signal is shifted toward right by one bit and added with coming P4 code. The reference signal is passed to the same Woo filter as of input combination and the remaining structure is same. In the form-2 structures is almost same but shift the input P4 code to the left side. There pulse compression output is observed and shown in the Figure 5.4 (a) and 5.4 (b) respectively. From the simulation it is observed that PSL exists at -88.57dB and -88.67dB for the modified form-1 and form-2.

Figure 5.5 (a) shows the pulse compression results for modified Woo filter-1 [24] having -104dB PSL. Figure 5.5 (b) shows the output for filter-2 having -107.6dB PSL. It is observed that this technique is good for side lobe reduction than other modified Woo filters.



(a) **(b)**

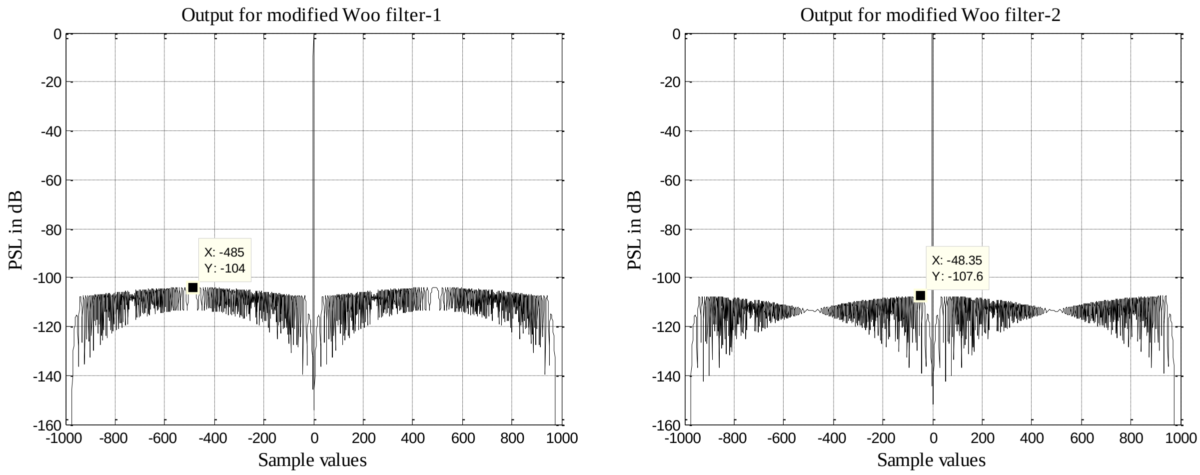
Figure 5.3: Pulse compression output generated by (a) P4 code and (b) Woo filter



(a) **(b)**

Figure 5.4: Pulse compression output generated by modified (a) Woo filter of form-1 and (b) Woo filter of form-2

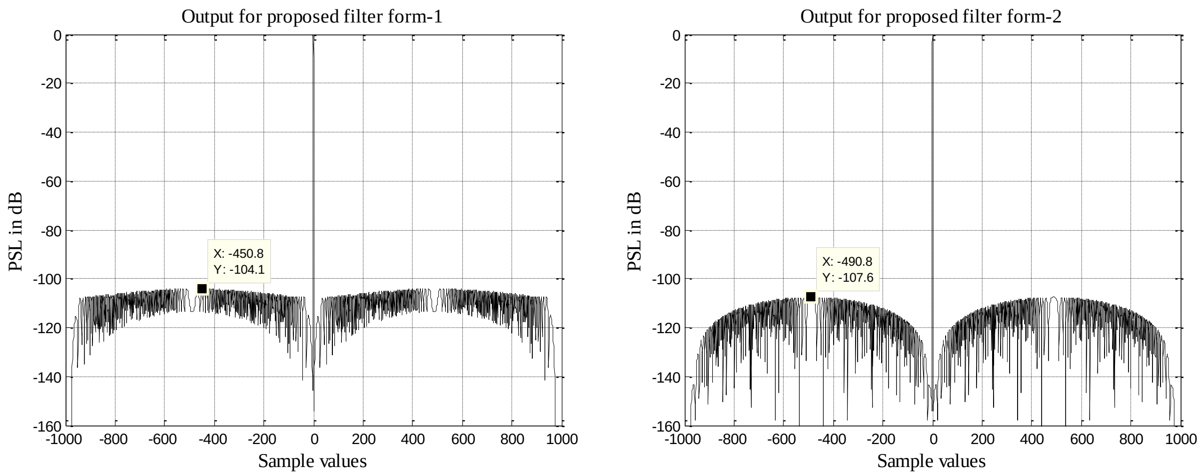
Figures 5.6 (a) and (b) represent the output of proposed filter form-1 and form-2 without weighting. For proposed filter form-1 in which applying input signal is the combination of one and two bits left rotations that is applied to the FFT as shown Figure 5.2 (a). It is observed that PSL level exists at -104.1dB, ISL is -75.34dB and relative mainlobe width is 2.51. By comparing it to the modified Woo filter form-1 without weighting it has 0.1dB more reduction in side lobes and ISL is also improved a little bit, whereas -46.29dB reduction in PSL is achieved than Woo filter, as shown in Table 5.1.



(a) **(b)**

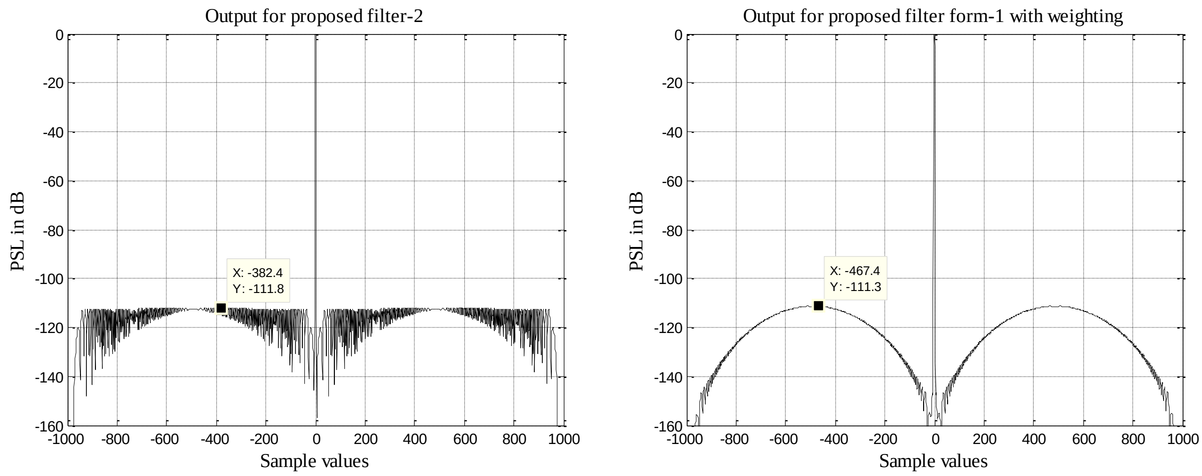
Figure 5.5: Pulse compression output generated by modified (a) Woo filter-1 and (b) Woo filter-2

Investigation has been done regarding the rotation of bits, it is observed if second and third shifts are combined to make input signal then there is no improvement in PSL. It is also concluded that left rotation of bits are more suitable for pulse compression because these has more reduction in side lobes than other combination of shifts. Here also in all the proposed methods left rotation is done for input signal. Performance measuring parameters for P4, Woo filter, modified Woo filter [24] and proposed methods are calculated and represented in Table 5.1. In proposed filter form-2 has the same input signal which is applied at the reference section. Its performance parameters are calculated and it is observed that there is little improvement in ISL than recently developed Woo filter.



(a) **(b)**

Figure 5.6: Pulse compression output generated by proposed filter without weighting (a) form-1 and (b) form-2



(a) **(b)**

Figure 5.7: Pulse compression output generated by proposed (a) filter-2 and (b) filter form-1 with Blackman window

Table 5.1: PSL, ISL and relative mainlobe width comparison for various sidelobe reduction techniques

Pulse Compression Techniques	Peak Side-lobes Level(dB)	Integrated Side-lobes Level (dB)	Relative Main-lobe Width
P4 code	-36.37	-16.99	1
Woo filter	-58.15	-27.08	2.03
Modified Woo filter - 1	-104	-75.18	2.51
Modified Woo filter - 2	-107.6	-79.11	2.47
Modified Woo filter-1 with Hamming window	-110.8	-82.18	3.20
Modified Woo filter-2 with Hamming window	-112.1	-83.10	3.37
Proposed filter form-1 Without Windowing	-104.1	-75.34	2.51
Proposed filter from-2 Without Windowing	-107.6	-80.43	2.97
Proposed filter -2	-111.8	-81.44	4.77

In proposed filter-2 as shown in Figure 5.2 (b), pulse compression output is shown in Figure 5.7 (a), have -42dB side lobe reduction than Woo filter. By comparing the results of this proposed method with the modified Woo filter [25], there is around -19dB PSL gain and -8dB ISL gain exists. It is observed that raised window is giving more reduction in side lobes than directly applying weighting to the signal as shown in filter-2. All these simulations have been done in MATLAB with 4096 FFT points. No doubt weighting can be used for further reduction in side lobes [29, 34] but on the loss of signal to noise ratio or relative main lobe width that affect the range resolution. From the results it can be observed that by applying different window relative mainlobe width has been increased slightly.

Figure 5.7 (b) shows the pulse compression output for proposed filter form-1 with Blackman weighting, by which -7dB gain is achieved. In the similar way, other weighting techniques are applied for the proposed models; results for PSL and ISL are tabulated in Table 5.2.

Table 5.2: PSL and ISL comparison for proposed method with various windows

Window Techniques	Proposed filter form-1		Proposed filter form-2		Proposed filter-2	
	PSL(dB)	ISL(dB)	PSL(dB)	ISL(dB)	PSL(dB)	ISL(dB)
Hamming	-110.9	-83.02	-111.5	-83.95	-111.6	-81.21
Hanning	-112	-83.46	-112	84.03	-111.8	-81.40
Blackman	-111.3	-83.41	-111.3	-83.99	-111.8	-81.44
Nuttall	-110.5	-83.44	-110.5	-83.86	-111.7	-81.46

5.6 Conclusion

The analysis and simulation in this chapter have demonstrated a new pulse compression technique which gives significant improvement in PSL and ISL but mainlobe width is increased slightly that affects the range resolution. Further it is observed that if more numbers of shifts are incorporated for the input P4 code then results deviate from the optimal PSL. By applying amplitude weighting in time domain more PSL and ISL reduction is achieved. The proposed model enhances the performance for the pulse compression in radar system but at the sacrifice in range resolution.

CHAPTER-6

CONCLUSION AND FUTURE SCOPE

6.1 Conclusion

The dissertation emphases on the sidelobe level reduction in the pulse compression. In the present work, investigation has been made on developing the efficient pulse compression techniques for better range and velocity resolution. From the analysis, it is observed that the differentiated exponential with non-integer index is giving slightly better requirements with reference to pulse compression. The complete detailed analysis on pulse compression with the Classical Orthogonal polynomials is carried out and it is concluded that the Laguerre polynomials are a better choice for pulse compression if there is no restrictions on the physical implementation of the source. It is observed that frequency modulation gives better time-bandwidth product for these polynomials than phase modulation. Simulated results shows that better PSL as well as better time-bandwidth product exist for modified half cycle Chebyshev polynomials, and its ambiguity function is more concentrated towards the centre, so these polynomials can be applied for the practical applications in radar system. It is also concluded that by using low order polynomial, ACF for higher order polynomial can be observed which saves the simulation time.

In another section, a new pulse compression technique is developed which gives significant improvement in PSL and ISL but mainlobe width is increased slightly that affects the range resolution. This technique uses the Woo filter concepts. Further it is observed that if more numbers of shifts are incorporated for the input P4 code then results deviate from the optimal PSL. By applying amplitude weighting in time domain more PSL and ISL reduction is achieved.

6.2 Future Scope

The research work presented in this dissertation can be extended. The ACF for Classical Orthogonal polynomials has been observed, to analyse the range and velocity resolution AF can be analysed. Sequence generated by Chebyshev polynomial can be used to analyse the pulse compression output for better ambiguity in range and velocity. To improve the PSL, ISL and relative mainlobe width the sidelobe cancellation technique can be investigated that reduces the side-lobes. There is a scope of designing the polyphase codes for Woo filter which has lower side-lobes in the pulse compression output.

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