

USING SPREAD SPECTRUM CODED PINGs IN ACTIVE SONAR TECHNOLOGY

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ABSTRACT

USING SPREAD SPECTRUM CODED PINGS IN ACTIVE SONAR TECHNOLOGY

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Performance of coded signals in active Sound Navigation and Ranging (SONAR) technology is studied in this work. In this work, the possibility of having covertness and environment friendliness in SONAR systems is investigated. Spread spectrum ping signal is considered to achieve low probability of detection and interception, while maintaining good performance. Direct Sequence Spread Spectrum (DSSS) coded transmitted signals having a sequence type of maximal with lengths from 7 chips to 127 chips and tone burst pulse having single length as a reference signal are used as spread spectrum ping signal. The length of the tone burst pulse can not be increased indefinitely because of the multipath propagation. The problem of detection and localization of the targets and the cross correlation properties of the maximal length sequences are investigated as well. The Signal to Noise Ratio (SNR) and Signal to Interference and Distortion ratio (SINAD) are important parameters in detection and localization of the targets. It is found that as the number of chip length increases, the SNR and SINAD increase but the improvement of the SINAD is comparatively less because of the cross correlation properties of the maximal length sequences.

Keywords: Target Detection and Localization, Covertness, Direct Sequence Spread Spectrum Modulation (DSSS), Maximal Length Sequences, Tone Burst Pulse, SNR, SINAD.

ÖZET

AKTİF SONAR SİSTEMLERİNDE YAYILI İZGE TEKNİĞİ İLE KODLANMIŞ İŞARETLERİN KULLANIMI

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Bu tezde, kodlanmış işaretlerin aktif sonar sistemlerindeki performansı ile örtülü ve çevreye duyarlı sonar sistemlerine sahip olabilme olasılığı incelenmektedir. Düşük tespit edilme olasılığı ve iyi bir performans elde edebilmek için yayılı izge işaretleri kullanılmıştır. Uzunluğu 7 çip dizi uzunluğundan 127 çip dizi uzunluğuna kadar değişen, çip türü maksimal olan, doğrudan dizili yayılı izge modülasyon (DDYİ) yöntemi ile kodlanmış işaretler ile referans sinyali olarak kullanılan tek ton darbeler yayılı izge işaretleri olarak kullanılmıştır. Tek ton darbelerin uzunluğu çok yollu yayılımdan dolayı sınırsız bir şekilde artırılmaz. Hedefin tespiti, yerinin saptanması problemi ve çip türü maksimal olan kodlanmış sinyallerin birbirleriyle olan benzerlikleri de incelenmiştir. Hedefin tespiti ve yerinin saptanmasında, işaret gürültü oranı ve kodlanarak farklı sektörlere gönderilen işaretlerin birbirleriyle etkileşimi önemli parametrelerdir. Çip dizi uzunluğu artırıldıkça, işaret gürültü oranı ve işaret etkileşim oranı artmaktadır. İşaret etkileşim oranındaki değişim, kullanılan maksimal çip türünün korelasyon özelliğinden dolayı, çok az ölçülmektedir.

Anahtar sözcükler: Hedef Tespiti ve Yerinin Saptanması, Örtülü Sistem, Doğrudan Dizili Yayılı İzge (DDYİ), Maksimal Uzunluklu Diziler, Tek Ton Darbe, İşaret Gürültü Oranı, İşaret etkileşim oranı.

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Chapter 1

INTRODUCTION

Sound waves in underwater acoustic environment is called underwater acoustic waves. The propagation of sound in underwater environment is a mechanical phenomena and depends on the mechanical properties of the medium.

Underwater acoustic systems are used to detect and localize the targets by using sound propagation. Some of them are trying to sense the radiated acoustic energy from the target without transmission which are called passive sonar systems and some of them generate and emit a pulse of sound, called a ping, and then receive the radiated acoustic energy from the target which are called active sonar systems. Active sonar systems can be used in both transmission and reception processes. Throughout this work, active sonar concept is studied. Active sonar system has a signal generator, power amplifier, sonar array and a beamformer. Sonar array is made up of transducers. Transducers are energy conversion elements which convert the electrical energy at their electrical inputs to the acoustical energy at their acoustical outputs and vice versa. The transmission and reception efficiency of the transducers are important parameters in detecting the target [1,11]. Also, Beamforming technique is a signal processing technique and is used in order to obtain directional transmission and reception and concentrate the acoustic energy into the beam [2]. Because the total flight time and sound speed are known parameters, the distance of the target can be found.

In conventional active sonar systems, a single pulse is transmitted from the array and echoes are received and beamformed at the receiver part of the sonar. The performance of this type of sonar is poor because it is not possible to detect and localize greater number of targets and the probability to be detected

by the enemies is very high. In this work, different DSSS coded signals having a sequence type of maximal with different lengths from single chip to 127 chips are transmitted through different sectors from the different elements of the array. Then, the echoes come from different sectors are received and beamformed individually. Basically, beamforming technique is utilized in both transmission and reception processes. Using coded signals have a number of advantages such as increased robustness to interference, increased tolerance to multipath, ability to detect greater number of targets, lower probability of detection and interception and increased range resolution [12]. In this work, for simplicity, an idealized deepwater medium is considered, that is, the temperature, salinity, atmospheric pressure, chemical composition and weather conditions are constant over time.

Detailed information about beamforming technique is given in Chapter 2. The principles of conventional active sonar technology is discussed in Chapter 3. The performance of Spread Spectrum pings as a new ping technology in active sonar systems, covert use of active sonar technology, and simulation results are discussed in Chapter 4.

Chapter 2

BEAMFORMING

2.1 Angular Response of a Line Hydrophone

To describe and understand some basic topics, Line hydrophone is used and shown in the Figure 2.1. The angle measured between the sound waves direction and a plane perpendicular to the x-axis is denoted as φ . Signal at the origin is represented by $s(t)$. Thus, the signal at any point on the line hydrophone is time shifted version of the signal at the origin. The value of this time shifting is $x \sin(\varphi)/c$.

Thus, the signal at any point on the line hydrophone can be represented as,

$$s(t, x) = s\left(t + \frac{x \sin(\varphi)}{c}\right) \quad (2.1)$$

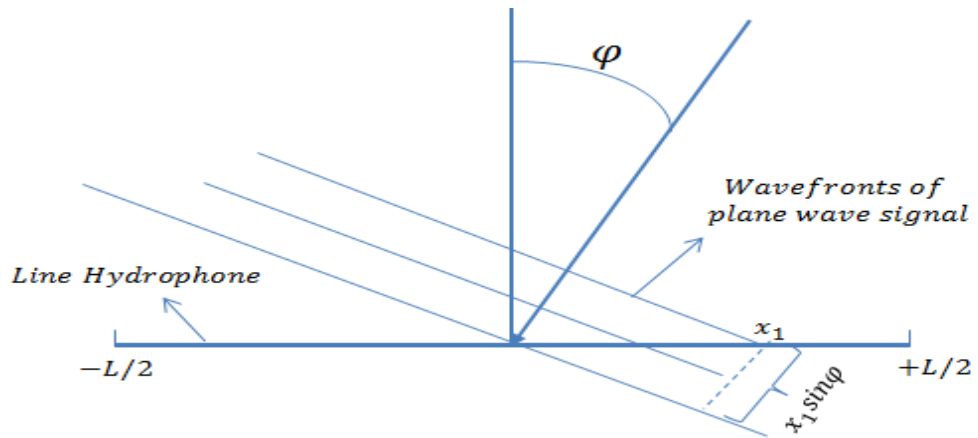


Figure 2.1: Plane wave signal incident on a line hydrophone.

By integration of the hydrophone response to the signal along its length, the line hydrophone output can be found. $g(x) dx$ is the line hydrophone response to the unit signal at x . $S_o(t, \varphi)$ is the output of the line hydrophone which is written as a function of time and angle φ [2,22]. Thus, the total output of the line hydrophone resulting from a plane wave at angle φ is,

$$S_o(t, \varphi) = \int_{-\infty}^{\infty} g(x) s\left(t + \frac{x \sin(\varphi)}{c}\right) dx \quad (2.2)$$

The hydrophone response is called the aperture function. Assume that $s(t)$ has a frequency domain Fourier Transform, $S(f)$. After rearranging the (2.2), we observe,

$$S_o(t, \varphi) = \int_{-\infty}^{\infty} S(f) \left[\int_{-\infty}^{\infty} g(x) \exp\left(\frac{j2\pi f x \sin(\varphi)}{c}\right) dx \right] \exp(j2\pi f t) df \quad (2.3)$$

The inner integral is the Fourier transform of the aperture function $g(x)$ from the x domain to a domain represented by the variable $u = \sin(\varphi) / \lambda$. The transform of the aperture function is called Pattern Function and it is defined by,

$$G(u) = G(f, \varphi) = \int_{-\infty}^{\infty} g(x) \exp(j2\pi x u) dx \quad (2.4)$$

Substitution of this equation into the equation (2.3) gives,

$$S_o(t, \varphi) = \int_{-\infty}^{\infty} S(f) G(f, \varphi) \exp(j2\pi f t) df \quad (2.5)$$

2.2 Array Shading

For simplicity, aperture function can be assumed to have a unit and constant amplitude over the active aperture and the amplitude is zero elsewhere which is referred to as Uniform Aperture Function. Uniform aperture function is suitable in case of coming single plane wave at an angle φ in isotropic noise.

In the case of presence of interfering plane waves at angles outside the main lobe of the pattern function, side lobe levels should be below those obtained with uniform aperture function. By modifying the shape of the aperture function, desirable side lobe levels can be achieved. Modification of the shape of the aperture function is called Array Shading. Two different aperture function is defined below.

- (1) Uniform Aperture Function
- (2) Taylor Weighted Aperture Function

Choosing different aperture function provides low level of side lobes; but results in increase in the main lobe width. Taylor aperture weighting technique provides more uniform low side lobe levels with very moderate main lobe spreading than the uniform aperture weighting technique. Figure 2.2 shows the Uniform distribution and 25-30-35 dB Taylor distribution [2,3].

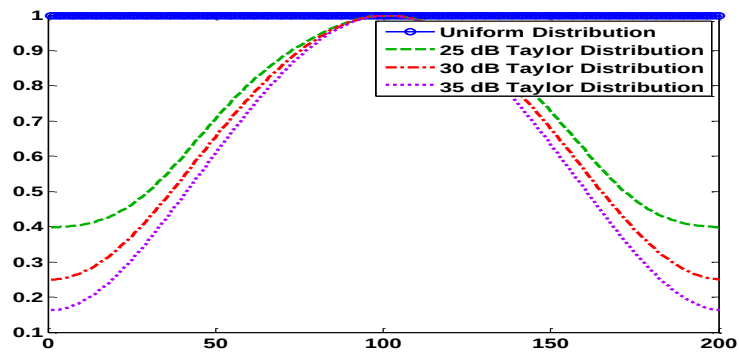


Figure 2.2: Uniform vs Taylor Weighting.

2.3 General One Dimensional Formulation Without Electronic Scanning

One dimensional array which consists of M elements is shown in the Figure 2.3. The distance between the centers of the elements is d . Thus, the overall length of the array, L , is $(M - 1) d$. Each element in the array has a complex voltage denoted as a_m . Consider that a signal is coming to the array from the direction θ . This signal is captured by each of the array elements and is then summed together coherently for the composite signal. The expression for the coherent sum of voltages can be represented as,

$$AF = \sum_{m=1}^M A_m \exp\left(j \frac{2\pi}{\lambda} x_m \sin(\theta)\right) \quad (2.6)$$

where x_m is the position of each element and represented as,

$$x_m = (m - 0.5 (M + 1))d \quad (2.7)$$

AF is the array factor and is the function of aperture distribution and frequency. In the case of without electronic scanning, the AF in the equation has a maximum value when $\theta=0^\circ$. Regardless of the dimension of the array, maximum value of the array factor is always equal to the number of elements in the array.

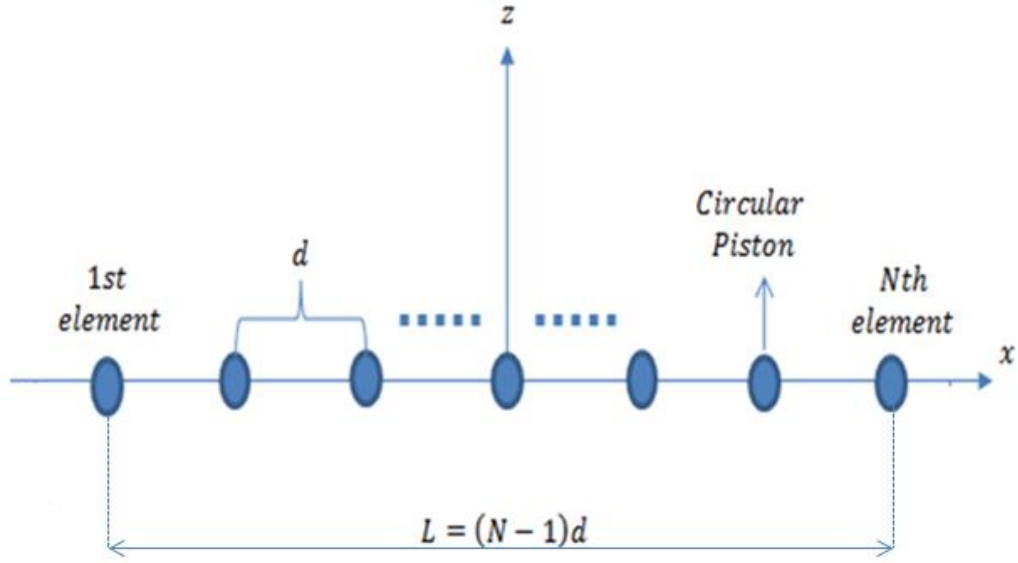


Figure 2.3: One Dimensional array

Circular piston has a wide variety application areas so throughout this work, circular piston is used as an array element with an element pattern (EP) shown below [4].

$$EP = 2 \frac{J_1(k \sin(\theta))}{k \sin(\theta)} \quad (2.8)$$

Array Factor and Element pattern describes the spatial response of the array together [4]. Thus, the total pattern representation can be found by the multiplication of the element pattern (EP) with array function (AF) as shown in below.

$$F(\theta) = EP \times AF = 2 \frac{J_1(k \sin(\theta))}{k \sin(\theta)} \sum_{m=1}^M A_m \exp\left(j \frac{2\pi}{\lambda} x_m \sin(\theta)\right) \quad (2.9)$$

2.4 General One Dimensional Formulation With Electronic Scanning

The direction that the array beam pattern is directed to is referred to as scan angle which is denoted as θ_0 . Adjusting the time or phase delay of each element provides the ability of scanning the beam [1,2,3]. By rewriting the (2.6) and expanding the complex voltage at each element gives

$$AF = \sum_{m=1}^M a_m \exp(j \theta_m) \exp\left(j \frac{2\pi}{\lambda} x_m \sin(\theta)\right) \quad (2.10)$$

The AF has a maximum value at θ_0 when $\theta_m = -\frac{2\pi}{\lambda} x_m \sin(\theta_0)$. Substituting this expression into the above equation, we can obtain,

$$AF = \sum_{m=1}^M a_m \exp j \left(\frac{2\pi}{\lambda} x_m \sin(\theta) - \frac{2\pi}{\lambda} x_m \sin(\theta_0) \right) \quad (2.11)$$

This equation states that applying the appropriate phase at each element, the ESA beam can be moved spatially without physically moving the entire array. The overall pattern function can now be expressed as,

$$F(\theta) = 2 \frac{J_1(k a \sin(\theta))}{k a \sin(\theta)} \sum_{m=1}^M A M \exp j \left(\frac{2\pi}{\lambda} x_m \sin(\theta) - \frac{2\pi}{\lambda} x_m \sin(\theta_0) \right) \quad (2.12)$$

where $J_1(k a \sin(\theta))$ is the Bessel function of the first kind with argument $k a \sin(\theta)$, k is the wave number, a is the radius of each element in the array and θ is the angle of incidence [4,21].

2.5 Element Pattern , Array Factor , Total Pattern and Electronic Scanning

In the experiments, frequency is 30 kHz and 30 elements are used. Element spacing and radius of each element are defined as a function of λ . The distance between the center of each element (d) is half of the λ and radius of each element is quarter of λ . Array Factor (AF), Element Pattern (EP) and Total Pattern (PAT) without electronic scanning are shown in Figure 2.4.

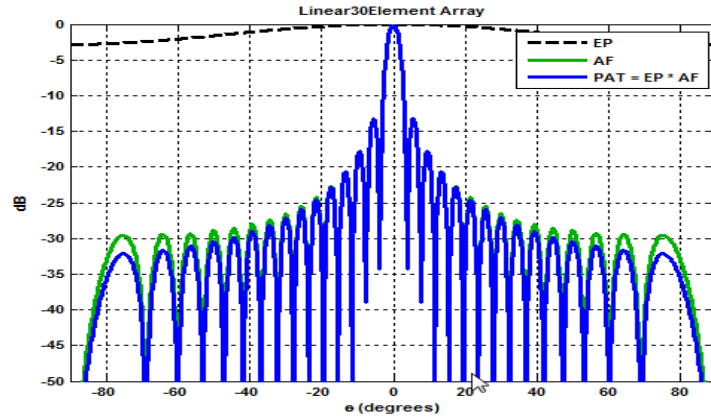


Figure 2.4: Array Factor (AF), Element Pattern (EP) and Total Pattern (PAT).

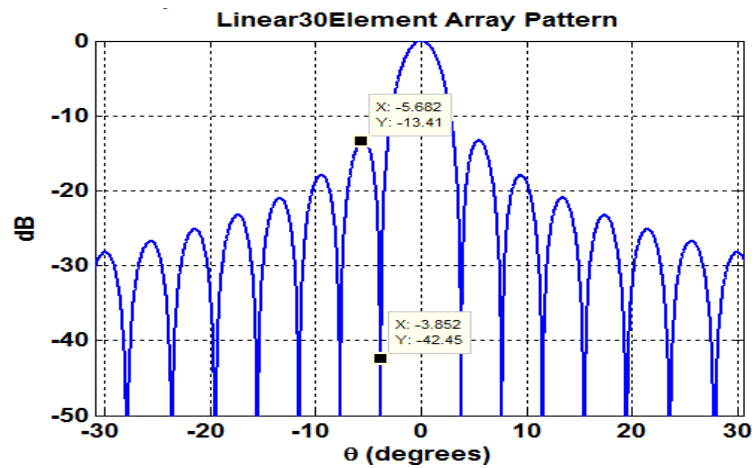


Figure 2.5: Total Pattern (PAT) of the array.

A closer look of the array pattern is shown in Figure 2.5. In this case there is no electronic scanning such that array is directed to $\theta = 0^\circ$. It means array pattern has a maximum at $\theta = 0^\circ$. There are several nulls in the array pattern. The minor lobes, commonly called sidelobes, have an amplitude decreasing with the order. An important parameter which is called sidelobe suppression is used to define the difference between the major lobe and the most notable lobe. Also sidelobe suppression means how well an array radiates acoustic power to the desired direction rather than unwanted direction [3,4,5]. The most notable lobe is the first one in this case. Thus, the sidelobe suppression is about 13 dB.

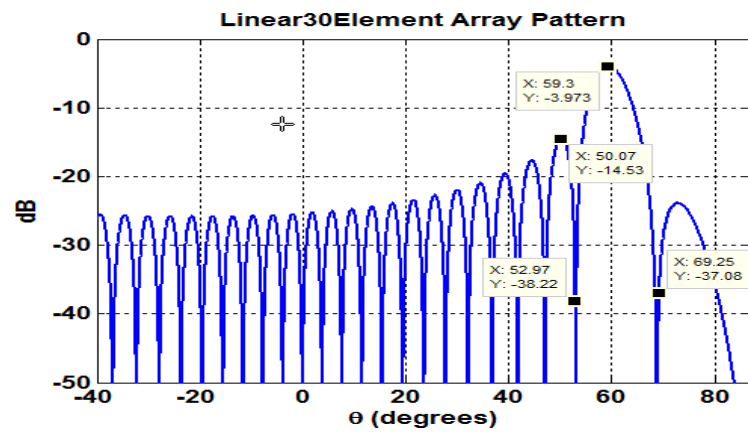


Figure 2.6: Affect of Electronic Scanning

By giving different amplitude or phase to each element of the array, the array can be directed to desired direction. As seen from Figure 2.6, the array is directed to $\theta = 60^\circ$. It means that the array pattern has a maximum at $\theta = 60^\circ$. Also the minor lobe is about 10 dB below the main lobe as opposed to 13 dB in the without electronic scanning case.

The other impact of the electronic scanning is broadening of the main beam. In Figure 2.5, the angular extent of the main beam is about 8° and as the beam is scanned as shown in Figure 2.6, the main beam broadens and the angular extent of the main beam is about 17° .

2.6 Beamwidth

The beamwidth is the angular extent of the main beam where the power decreases to a certain value. The half power point or 3 dB point of an array beam is the angle at which the power drops 3 dB below the peak and it is also referred to as half power beamwidth [6].

$$\theta_{BW} = \frac{k \lambda}{L \cos(\theta_0)} \quad (2.13)$$

where k is the beamwidth factor and λ is the wave length. Beamwidth of the array is proportional with λ and is inverse proportional with L and cosine θ_0 . Figure 2.7, Figure 2.8, Figure 2.9 show the array beamwidth as a function of scan angle for $k=1,2,3$ and for 20,25,30,35 kHz.

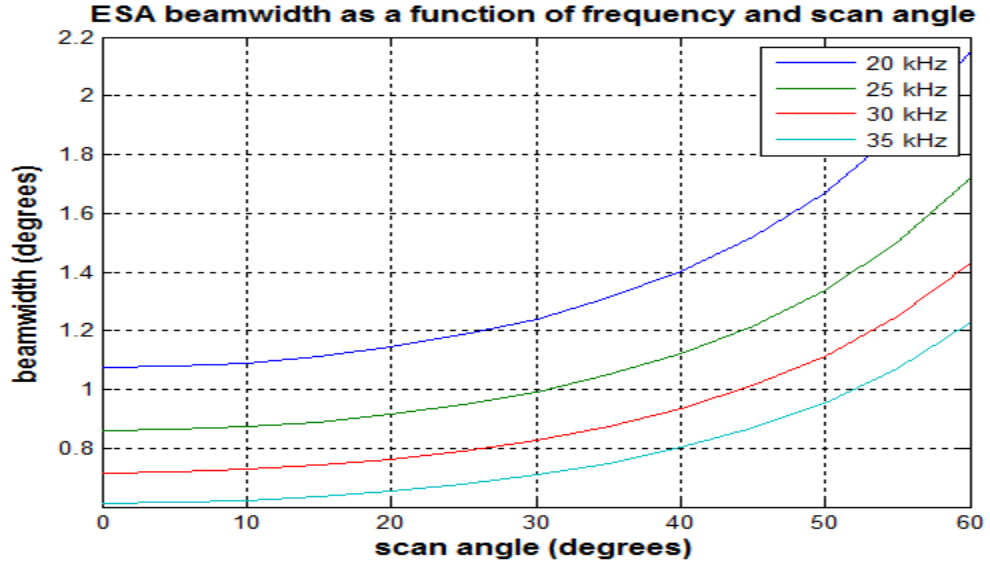


Figure 2.7: Array beamwidth as a function of θ_0 for $k=1$.

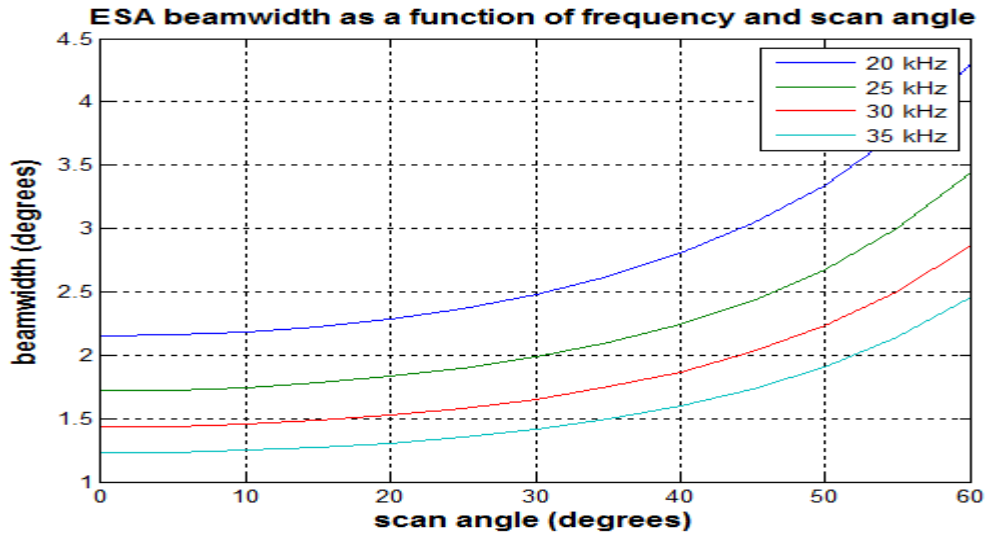


Figure 2.8: Array beamwidth as a function of θ_0 for $k=2$.

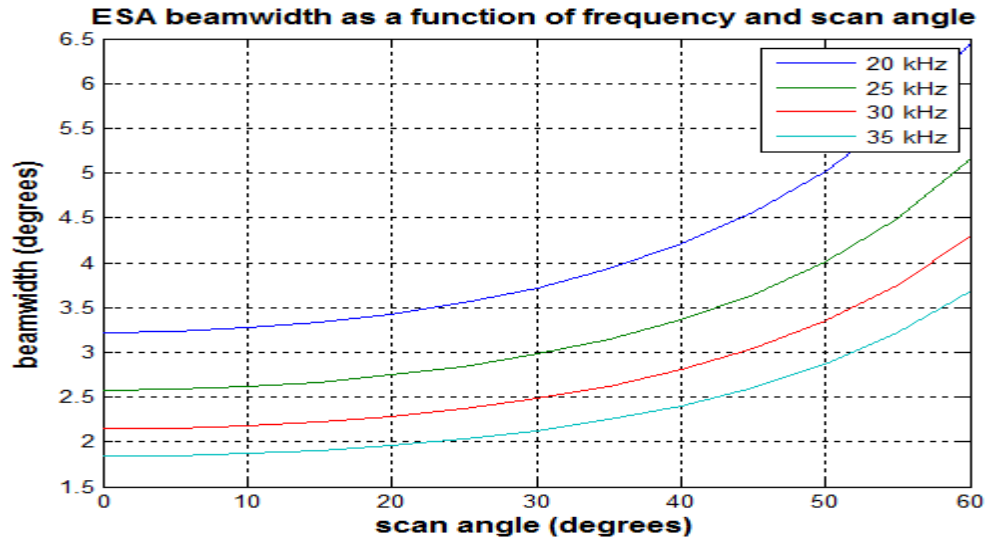


Figure 2.9: Array beamwidth as a function of θ_0 for $k=3$.

It can be clearly seen from the figures that at a certain scan angle, as the frequency increases the beamwidth of the main beam decreases. Also at a certain frequency, as the scan angle increases, the beamwidth of the main beam increases as well. These are the impacts of the electronic scanning on the radiation pattern.

2.7 Grating Lobes

In signal processing, in order to reconstruct a signal from its samples $x[n] = x(nTs)$, samples should be taken properly which is called Shannon Sampling Theorem [7]. Similar to signal processing theory, in the sonar arrays, the element spacing should be sampled properly as well. The element spacing should be equal to or less than the half of the wavelength, $\lambda/2$. If we chose the element spacing greater than $\lambda/2$, the grating lobes which are the replicas of the main lobe will occur (See Figure 2.10, Figure 2.11, Figure 2.12, Figure 2.13).

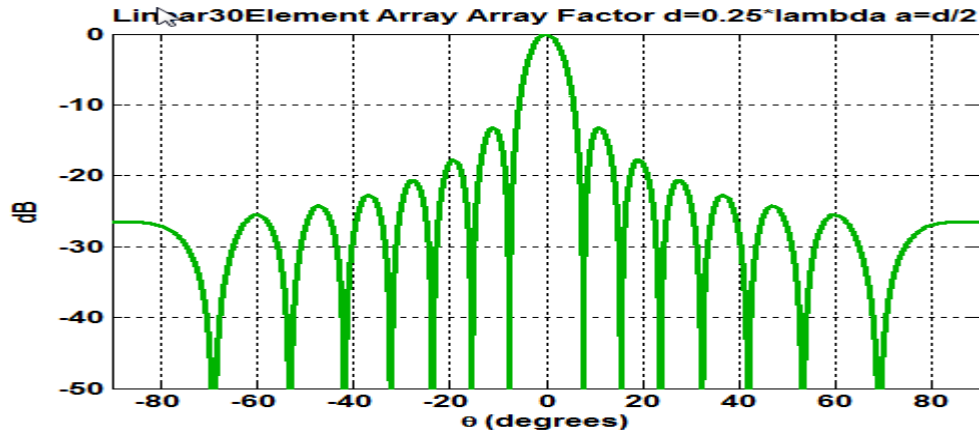


Figure 2.10: Plot of Array Factor for $d = \lambda/4$ and $a = d/2$.

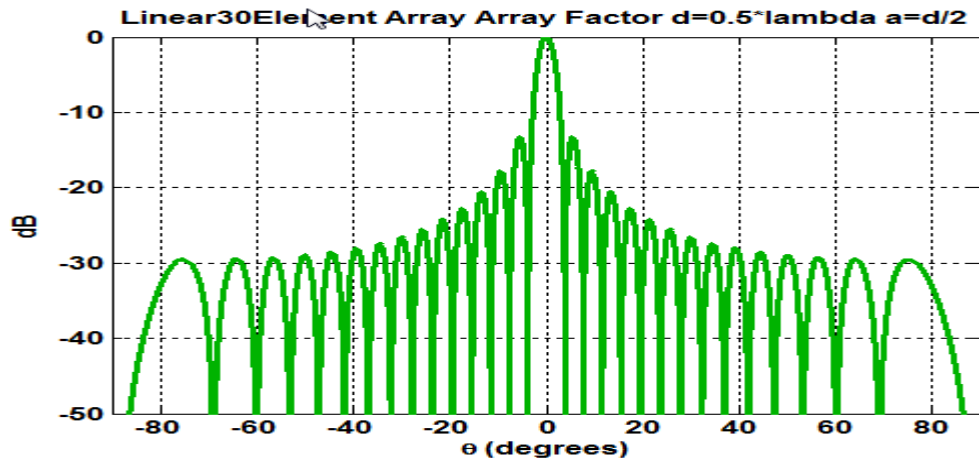


Figure 2.11: Plot of Array Factor for $d = \lambda/2$ and $a = d/2$.

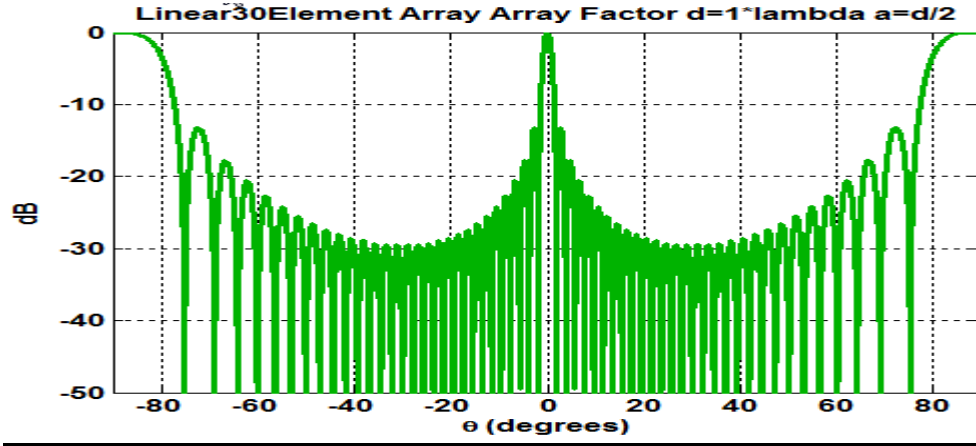


Figure 2.12: Plot of Array Factor for $d = \lambda$ and $a = d/2$.

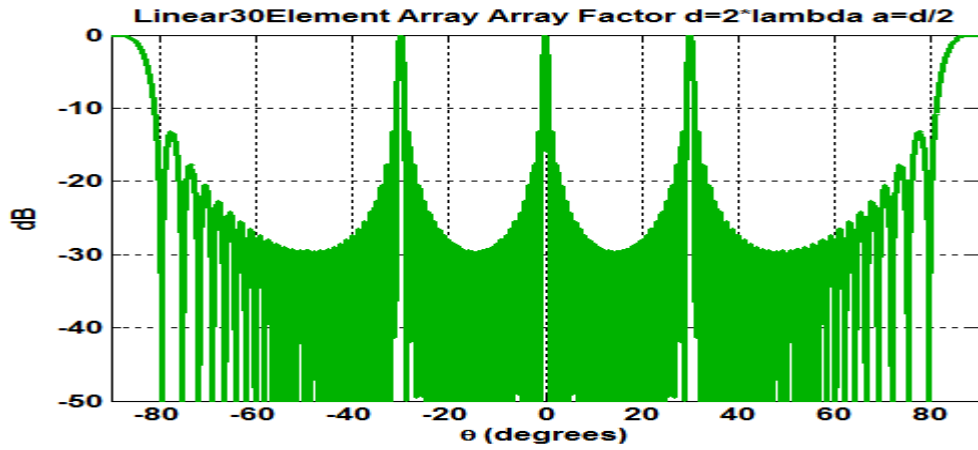


Figure 2.13: Plot of Array Factor for $d = 2\lambda$ and $a = d/2$.

An Array Factor is a periodic function and its alternative expression is defined below.

$$AF = \frac{\sin M\pi d \left(\frac{\sin \theta_0}{\lambda_0} - \frac{\sin \theta}{\lambda} \right)}{M\pi d \left(\frac{\sin \theta_0}{\lambda_0} - \frac{\sin \theta}{\lambda} \right)} \quad (2.14)$$

According to this expression, the maximum values of the array factor can be achieved when

$$\pi d \left(\frac{\sin \theta_0}{\lambda_0} - \frac{\sin \theta}{\lambda} \right) = \mp K \pi \quad (2.15)$$

where $K=0,1,2,3,\dots$

After some rearrangements, expression (2.15) reduces to

$$\sin(\theta_{GL}) = \lambda \frac{\sin \theta_0}{\lambda_0} \mp \frac{K}{d} \lambda \quad (2.16)$$

In this expression, the first term of the right hand side shows the location of the main beam and the second term of the right hand side shows the location of grating lobes. These figures are obtained without electronic scanning such that $\theta_0 = 0^\circ$. Thus, main beam occurs at 0° . It can be clearly seen from the figures that the location of the grating lobes depends on the frequency and the element spacing [3,5].

2.8 Effect of Some Fundamental Topics On One Dimensional Pattern

Detailed information is given here about the effects of some parameters such as element amplitude distribution, frequency, number of elements, and scan angle on 1D pattern.

2.8.1 Effect of The Element Amplitude Distribution

Voltage at each element of the array is represented as $A_m = a_M \exp(j \theta_m)$ where a_M is the amplitude and θ_m is the phase. Changing the phase (θ_m) of each element and changing the element amplitude distribution allows us to scan the beam spatially and to decrease side lobe levels respectively.

Figure 2.14 and Figure 2.15 are shown to compare the side lobe levels which are achieved with Uniform and Taylor distribution. In Figure 2.14, the first side lobes are ≈ 13 dB below relative to the peak of the main beam. In Figure 2.15, the first side lobes are ≈ 30 dB below relative to the peak of the main beam.

Decreasing the side lobe level relative to the main beam is not possible without spreading the main beam width. It is important to select an aperture distribution that provides low level side lobes with smallest spreading of the main lobe.

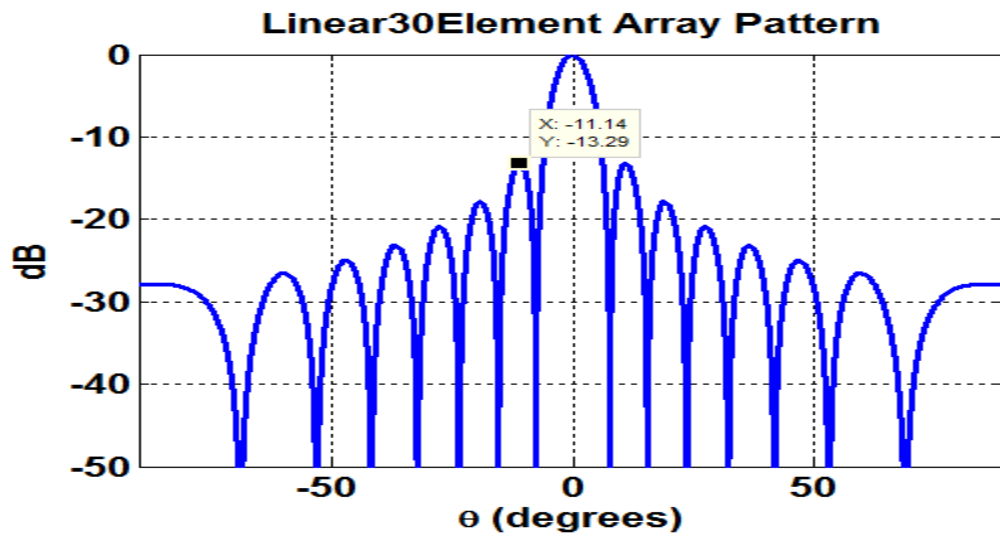


Figure 2.14: Total pattern which is plotted with Uniform Distribution.

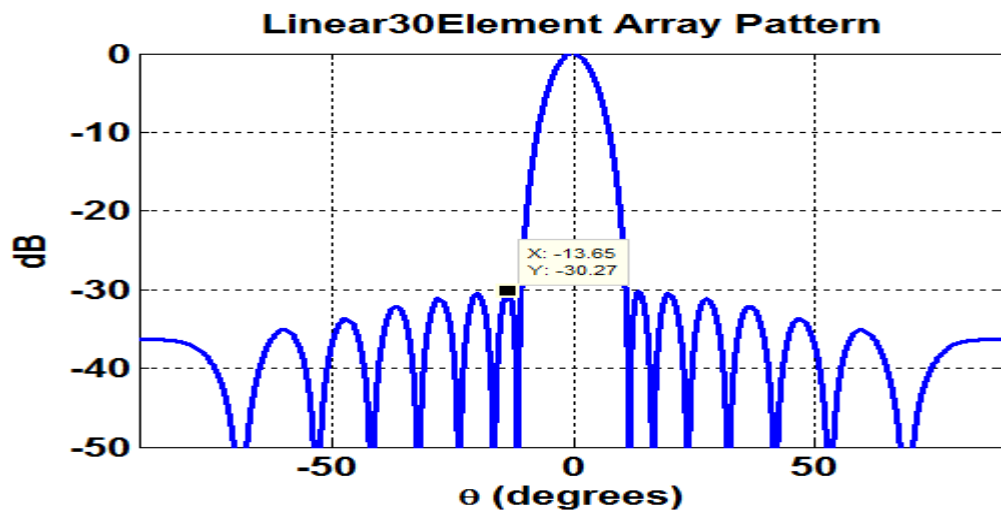


Figure 2.15: Total pattern which is plotted with Taylor Distribution

In Figure 2.15, the difference between sound pressure level in the desired direction and that at the unwanted direction, commonly called minor lobe suppression, is about 30 dB and also the main beam width spreads 1.53 times that obtained with uniform distribution. This spreading value is very moderate. Thus, Taylor Weighting Distribution is effective than the Uniform Distribution. In this work, the main aim is not to find most effective amplitude distribution so Uniform Distribution is used in this work.

2.8.2 Effect of the Frequency

To find the effect of frequency, two similar sized arrays are used and the properties of the arrays are shown in Table 2.1.

Table 2.1: Properties of the arrays made up of 30 elements

	ARRAY 1	ARRAY 2
f	30 kHz	60 kHz
λ	0.05 m	0.025 m
d	0.0125 m ($d = \lambda/4$)	0.0125 m ($d = \lambda/2$)
a	0.0063 m ($a = \lambda/8$)	0.0063 m ($a = \lambda/4$)

To interpret the effect of frequency on the total pattern, the radius of the circular piston (a) and the distance between the centers (d) as a function of lambda (λ) should be taken into account. As a function of lambda, d and a have larger values in 60 kHz case. Thus, the array operating at 60 kHz behaves more directive which has narrower main lobe (See Figure 2.16, Figure 2.17).

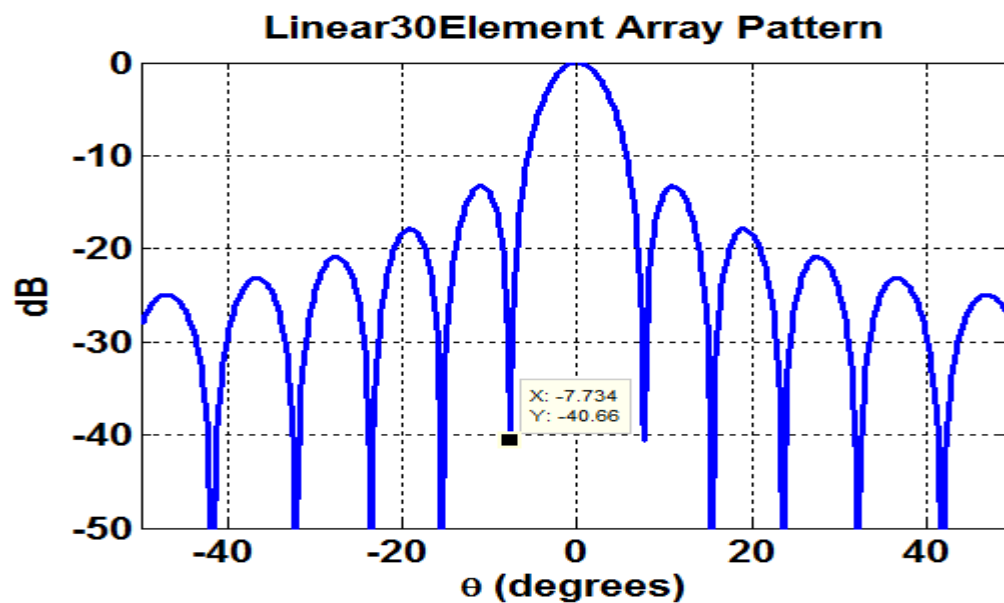


Figure 2.16: Plot of Total Pattern with operating frequency at 30 kHz.

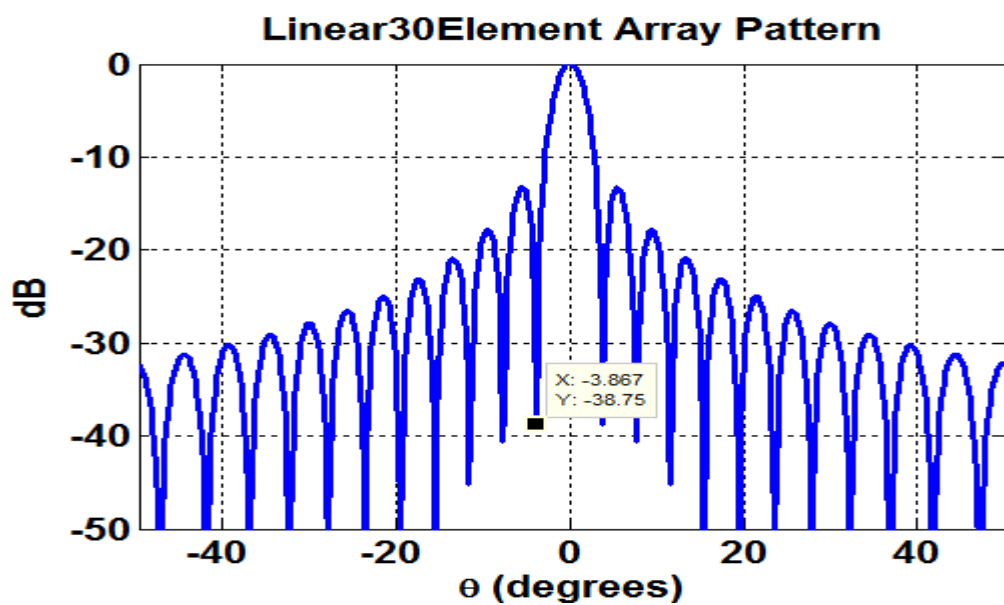


Figure 2.17: Plot of Total Pattern with operating frequency at 60 kHz.

2.8.3 Effect of the Number of Elements

To find the effect of number of elements, two similar sized (except number of elements) arrays that both operating at 50 kHz are used.

Figure 2.18 and Figure 2.19 shows the total pattern of the array with 11 elements and 21 elements respectively. Beamwidth is inversely proportional with the number of elements so larger amount of elements means narrower beam width.

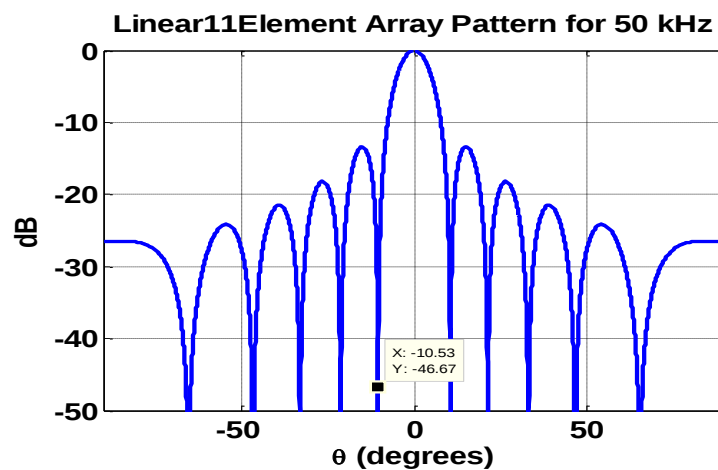


Figure 2.18: Total Pattern of the array which consists 11 elements.

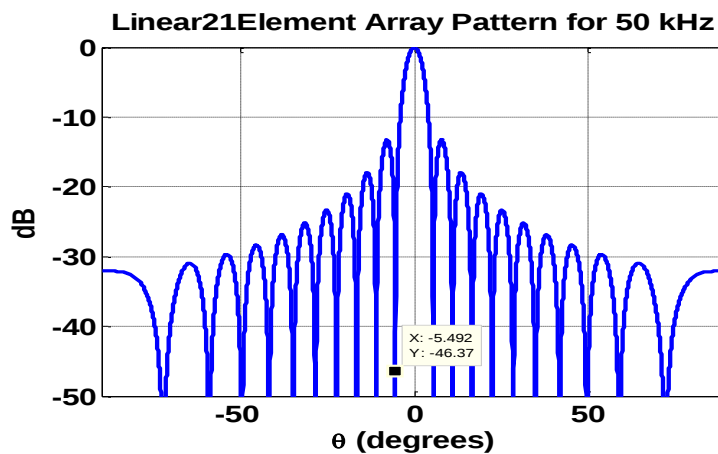


Figure 2.19: Total Pattern of the array which consists 21 elements.

2.8.4 Effect of the Scanning Angle

The array should have an ability to scan the beam spatially. With the ability of scanning, the array can emit acoustic waves to the different sectors. While increasing the scanning angle, you can see that the beamwidth of the main lobe widens.

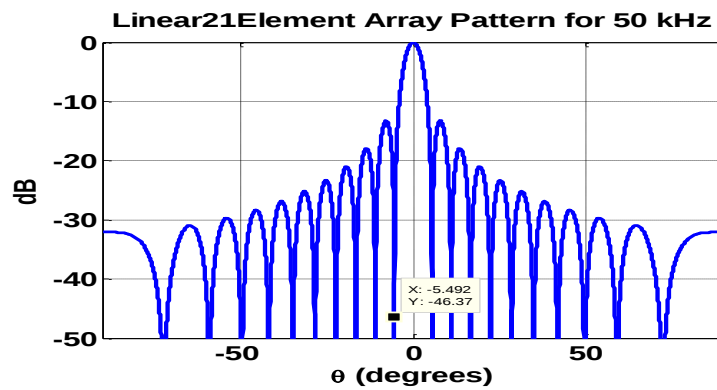


Figure 2.20: Total Pattern of the array where $\theta_0 = 0^\circ$.

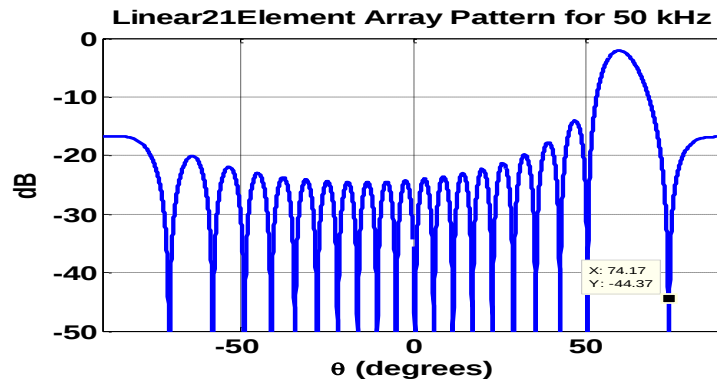


Figure 2.21: Total Pattern of the array where $\theta_0 = 60^\circ$.

The main beam broadening occurs with a rate of $1/\cos(\theta)$ (See Figure 2.20 and Figure 2.21). In Figure 2.21, scanning angle is 60° so two times broadening of the main lobe occurs.

2.9 General Pressure Formulation of the One Dimensional Array

To find the farfield pressure of a circular piston, it can be thought that a circular piston is made up of infinitesimal vibrating ring pistons. Thus, ring radius and thickness becomes variable. After integrating the ring piston pressure expression over these variables, 1D array farfield pressure of circular piston can be expressed as follows

$$P(r, \theta, t) = j \frac{P_o R_o}{r} EP AF \exp(j (wt - kr)) \quad (2.17)$$

P_o is the pressure amplitude but it is not the exact value of the pressure amplitude at the piston face. It is an imaginary quantity that represents the pressure amplitude at the piston face if the piston diameter is infinite. It can be represented in terms of sound velocity, density and particle velocity at the piston face as below

$$P_o = \rho c u_o \quad (2.18)$$

R_o is the another quantity of interest. It is called rayleigh distance. In general rayleigh distance is

$$R_o = S/\lambda \quad (2.19)$$

where S is the piston area and λ is wavelength [4]. For circular piston Rayleigh distance is

$$R_o = \frac{k a^2}{2} \quad (2.20)$$

2.9.1 Farfield Pressure of the One Dimensional Array

Farfield region is the region beyond which the directional characteristics of an array does not change with the distance and the sound pressure is inversely proportional with the distance. Here, farfield pressure is found by using two similar sized arrays (See Table 2.1). According to these experiments, we can also see the effects of the frequency on the pressure. The results are shown in Figure 2.22 and Figure 2.23, respectively. Increasing frequency affects the main beam width as mentioned before. It is known that main beam width is inversely proportional with the operating frequency. Also, the farfield pressure is proportional with the total pattern of the array which is shown in (2.17), so the farfield pressure is affected with the frequency. In Figure 2.22, the main beam width is about 21° and in Figure 2.23, the main beam width is about 10° . Here, θ represents the angle of incidence or it is the angle the incoming wave makes with the normal of the array surface.

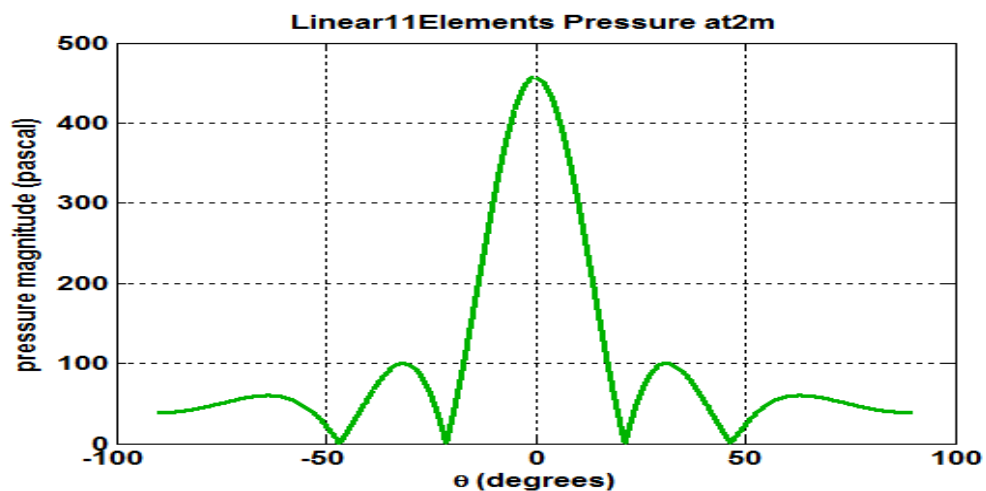


Figure 2.22: Pressure of the array where $f=30\text{kHz}$.

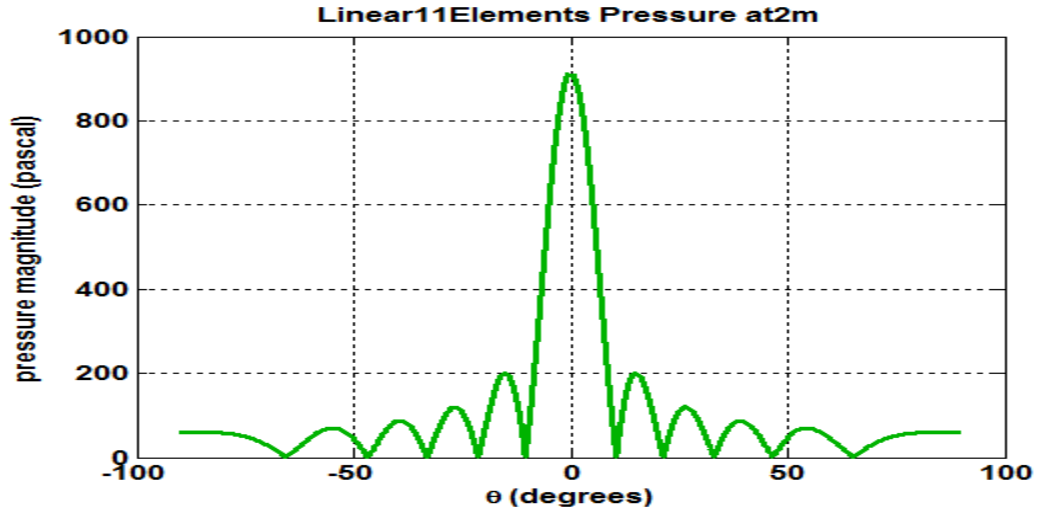


Figure 2.23: Pressure of the array where $f=60\text{kHz}$.

The two farfield pressure for 30 and 60 kHz is obtained under the same particle velocity. The power emitted at the circular piston face can be calculated as follows.

$$W_{\text{CIRCULAR PISTON}} = S u_{rms}^2 \text{Re}(Zp) \quad (2.21)$$

where S is the circular piston surface area, u_{rms} is the root mean square of the particle velocity and Zp is the radiation impedance presented to the circular piston by the medium. The real part of the impedance determines the emitted power and it can be represented as follows [4,8].

$$\text{Re}(Zp) = \rho_o c_o R_1(2ka) \text{ where } R_1(2ka) = 1 - \frac{2 J_1(2ka)}{2ka} \quad (2.22)$$

$J_1(2ka)$ and $R_1(2ka)$ are plotted as a function of their arguments, $2ka$, in Figure 2.24 and Figure 2.25 respectively.

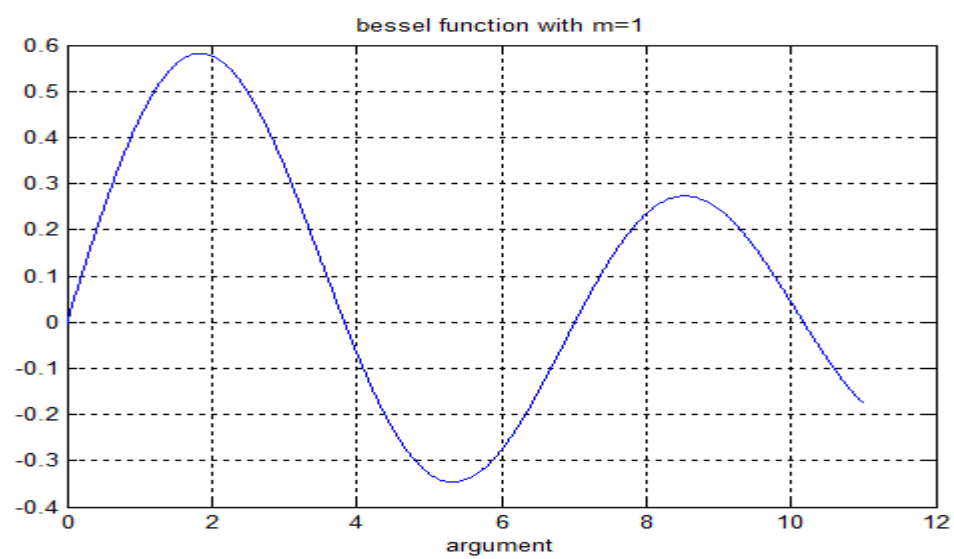


Figure 2.24: Bessel function of first kind with $m=1$.

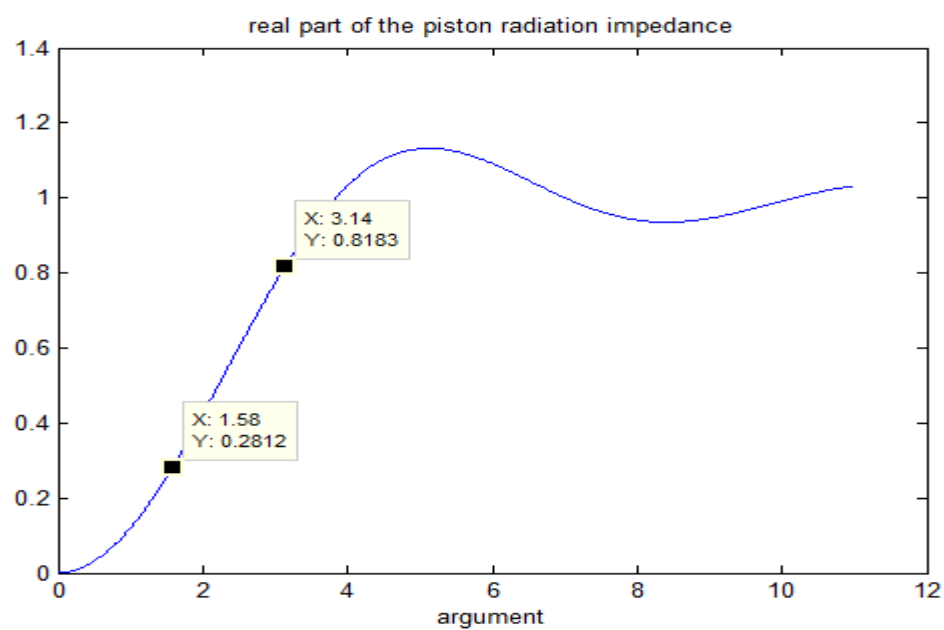


Figure 2.25: Real part of piston radiation impedance

After some rearranging, the power emitted at the piston face can be simplified to

$$W = \frac{\pi a^2}{2\rho_o c_o} P_o^2 R_1(2ka) \quad (2.23)$$

where k is the wavelength for 30 and 60 kHz, a is the radius of circular piston and ρ_o is the density of the medium. $R_1(2ka)$ values for 30 and 60 KHz are shown in Figure 2.25.

$$R_1(2ka) = R_1(1.5708) \cong 0.2812 \text{ for } 30 \text{ kHz}$$

$$R_1(2ka) = R_1(3.1416) \cong 0.8183 \text{ for } 60 \text{ kHz}$$

Except the real part of the piston radiation impedances, $R_1(2ka)$, the other terms are the same for 30 and 60 kHz frequencies. The ratio between the powers emitted at the pistons surface is shown below.

$$\frac{W_{60 \text{ kHz}}}{W_{30 \text{ kHz}}} \cong \frac{0.8183}{0.2812} = 2.92$$

The main lobe width gets narrower in 60 kHz frequency case but simultaneously the number of side lobes increased as compared with 30 kHz frequency case. As a result, the power is delievered into the main lobe and the side lobes. The relation between pressure pattern obtained in Figure 2.22 and Figure 2.23 for 30 and 60 kHz frequencies and the power is as follows.

$$\frac{\int_{-\pi/2}^{\pi/2} P_{30kHz}^2 d\theta}{\int_{-\pi/2}^{\pi/2} P_{60kHz}^2 d\theta} \cong \frac{W_{60 \text{ kHz}}}{W_{30 \text{ kHz}}} \cong 2.92$$

The square of pressure is taken in the above equation because the power is proportional with the square of pressure [1,2,4,9]. Thus, the peak value of pressure in 60 kHz frequency case is greater than the 30 kHz frequency case.

2.10 General Two Dimensional (2D) Array Formulation

2D arrays have elements in both x and y direction. The circular pistons are settled in $x - y$ plane and the radiation is assumed through $+z$ direction which is shown in Figure 2.26. Total number of elements is represented by MN where M and N are the number of elements in x and y direction respectively. Also element spacing in x and y direction is represented by dx and dy , respectively. The position of each element according to the x direction is,

$$x_m = (m - 0.5 (M + 1)) dx \text{ for } m = 1, 2, \dots, M \quad (2.24)$$

and the position of each element according to the y direction is,

$$y_n = (n - 0.5 (N + 1)) dy \text{ for } n = 1, 2, \dots, N \quad (2.25)$$

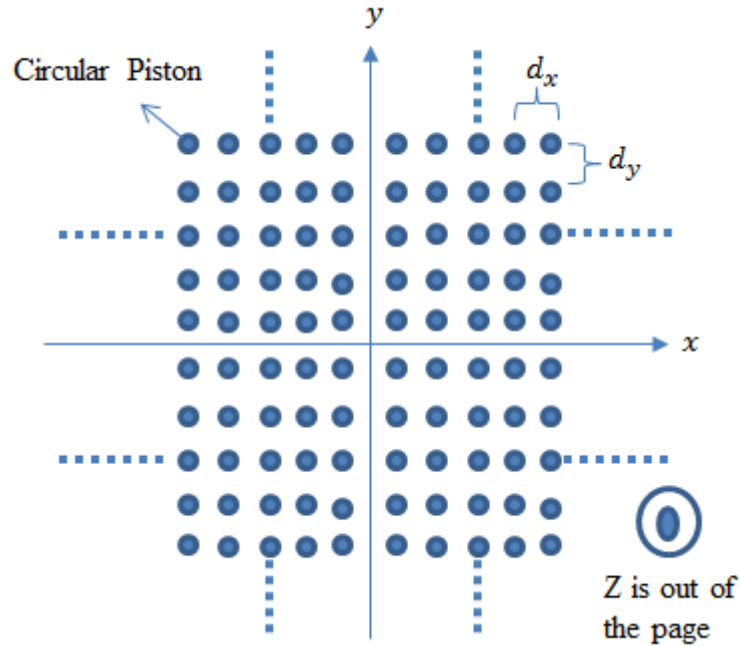


Figure 2.26: Two Dimensional array

By including the additional elements in y direction, fundamental concepts and general expressions for $1D$ arrays can be expanded for the $2D$ arrays [3].

2.10.1 Array Factor (AF), Element Pattern (EP) and Total Pattern (PAT) of the Two Dimensional Array

Array factor for 2D arrays is generally defined as follows

$$AF = \sum_{l=1}^{MN} C_l \exp\left(j \frac{2\pi}{\lambda} x_l \sin(\theta) \cos(\varphi)\right) \exp\left(j \frac{2\pi}{\lambda} y_l \sin(\theta) \sin(\varphi)\right) \quad (2.26)$$

where C_l is the complex voltage of the each element in the array and it can be written as $C_l = c_l \exp(-j \theta_l)$. θ_l can be written as a function of θ_0 and φ_0 .

$$\theta_l = \frac{2\pi}{\lambda} x_l \sin(\theta_0) \cos(\theta_0) + \frac{2\pi}{\lambda} y_l \sin(\theta_0) \sin(\theta_0) \quad (2.27)$$

Substituting this equation into the (2.26), gives the two dimensional array factor.

$$AF = \sum_{l=1}^{MN} c_l \frac{\exp(j \frac{2\pi}{\lambda} x_l \sin(\theta) \cos(\varphi) + j \frac{2\pi}{\lambda} y_l \sin(\theta) \sin(\varphi))}{\exp(j \frac{2\pi}{\lambda} x_l \sin(\theta_0) \cos(\varphi_0) + j \frac{2\pi}{\lambda} y_l \sin(\theta_0) \sin(\varphi_0))} \quad (2.28)$$

2.10.2 Array Coordinate System Description

When studying 2D arrays, it is important to specify which coordinate system is used. By using these coordinate systems, a point in the space can be defined with corresponding angle pairs. Figure 2.27 shows a two dimensional array in the three dimensional space. Array is on the $x - y$ plane and is assumed to radiate in the $+z$ direction. R is defined as a point in the space. P_{xy} , P_{xz} , P_{yz} are the projections of the point R onto the planes. This figure is a basic figure and provides us to understand some fundamental topics of the coordinate systems. The coordinate systems are:

- (1) Antenna Coordinate System
- (2) Radar Coordinate System
- (3) Antenna Cone Angle Coordinate System

According to the applications, Antenna Coordinate System has more advantageous than the others. Therefore, throughout this work, antenna coordinate system is used.

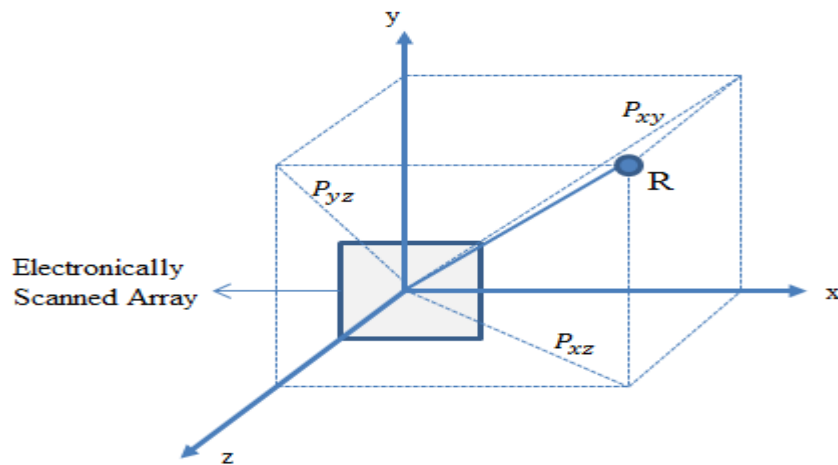


Figure 2.27: Two Dimensional array in Three Dimensional space

2.10.3 Antenna Coordinate System

An Antenna coordinate system enables us to represent a point in space as a corresponding angle pair which is (θ_z, φ) as shown in Figure 2.28. θ_z is the angle which is measured from the z axis to the point. φ is the angle which is measured from the x axis to the projection of the point onto $x - y$ plane. Also when the magnitude of R is set to unity the position of R can be defined as,

$$R = \sin(\theta_z) \cos(\varphi), \sin(\theta_z) \sin(\varphi), \cos(\theta_z) \quad (2.29)$$

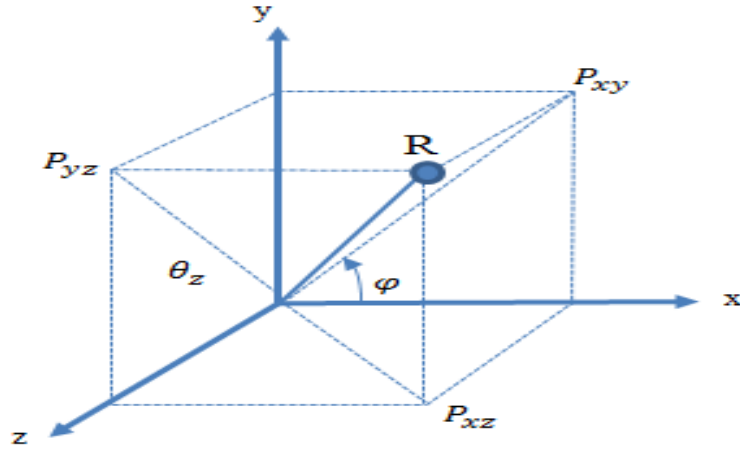


Figure 2.28: Antenna Coordinate System

2.10.4 Matlab Calculations for Array Factor (AF), Total Pattern (PAT) and Electronic Scanning

In order to obtain array factor and total pattern, sinespace representation which is basically projection of three dimensional space onto two dimensional space is used. As shown in the figures below, x and y axes are represented by U and V

respectively which are the variables of sinespace representation. U and V can be evaluated in terms of antenna coordinate angle pairs as follows

$$U = \sin(\theta_z) \cos(\varphi) \text{ and } V = \sin(\theta_z) \sin(\varphi) \quad (2.30)$$

Two and three dimensional views of array factor are shown in Figure 2.29 and Figure 2.30 with 30 kHz operating frequency and 400 elements (20×20 array).

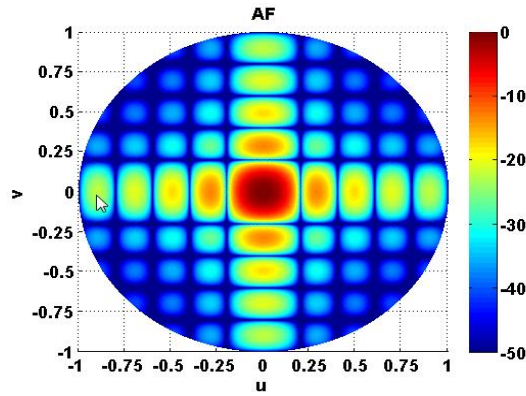


Figure 2.29: Two dimensional view of the AF with Uniform distribution

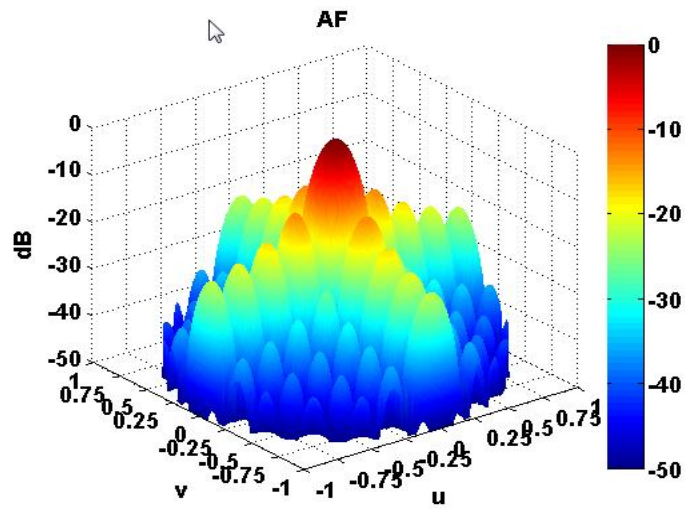


Figure 2.30: Three dimensional view of the AF with Uniform distribution

In Figure 2.29 and Figure 2.30, the normalized Array Factor of the 20×20 array in which each element is modeled as a circular piston, is plotted in dB with uniform distribution. Colorbar shows the magnitudes of the array factor in dB with respect to colors. Here there is no electronic scanning, in other words there is no phase shifts given to each array element so array has a maximum value at $u = 0$ and $v = 0$.

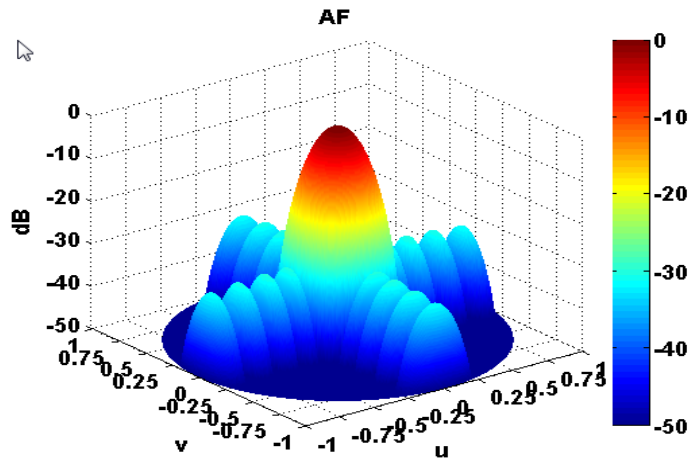


Figure 2.31: Three dimensional view of the AF with Taylor distribution

In order to find the effect of amplitude distribution, only three dimensional view of the normalized Array Factor is plotted in dB with Taylor distribution. It can be clearly seen from Figure 2.31, side lobe levels reduces to 30 dB below the main beam with moderate main beam broadening.

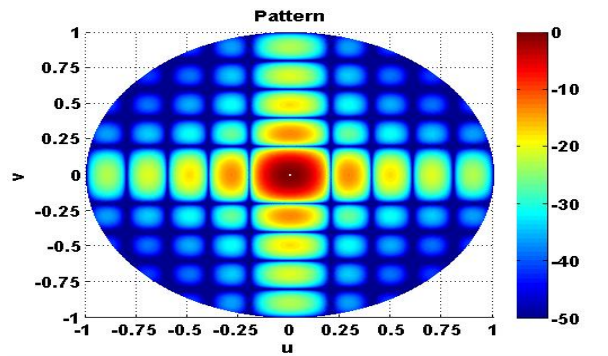


Figure 2.32: Two dimensional view of the Total Pattern with Uniform distribution

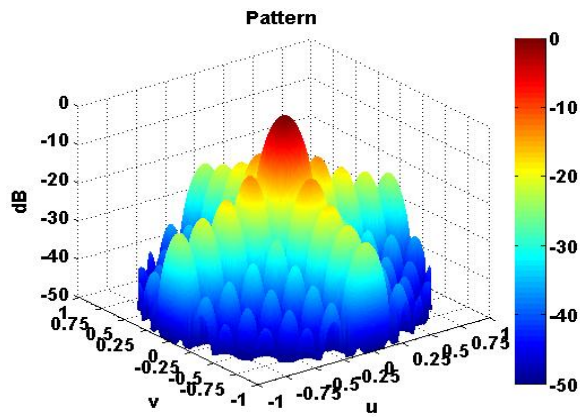


Figure 2.33: Three dimensional view of the Total Pattern with Uniform distribution

In Figure 2.32 and Figure 2.33, the normalized Total Pattern which is the multiplication of Element Pattern and Array Factor is plotted in dB with uniform distribution and without electronic scanning. As in the previous case Total Pattern has a maximum value at $u = 0$ and $v = 0$.

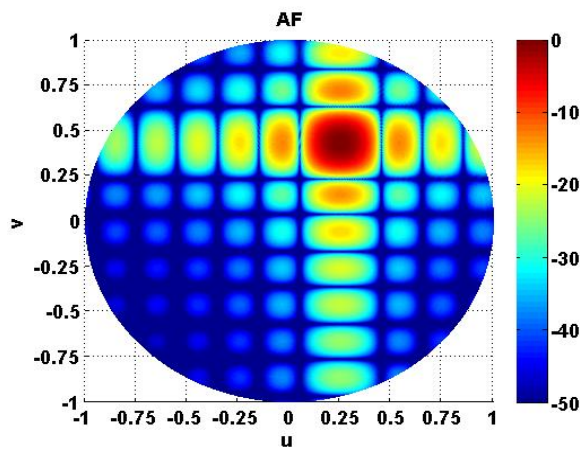


Figure 2.34: Two dimensional view of the Array Factor with Uniform distribution and electronic scanning

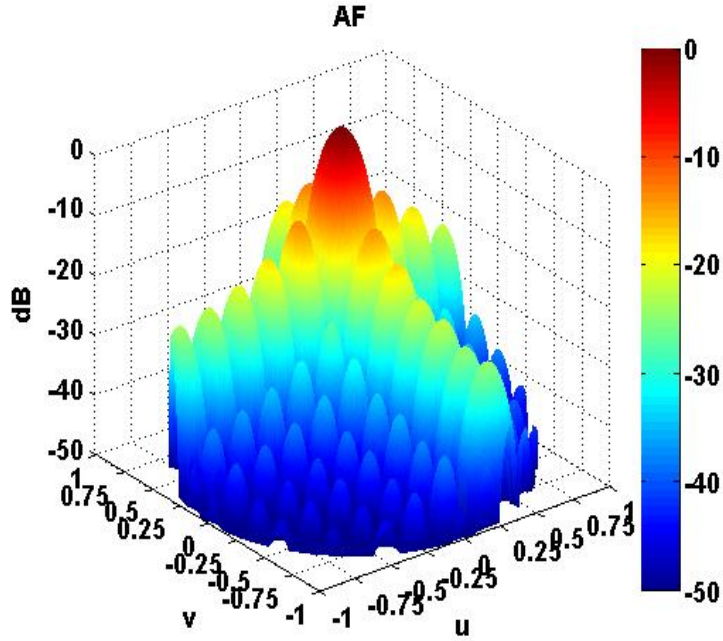


Figure 2.35: Three dimensional view of the Array Factor with Uniform distribution and electronic scanning.

In Figure 2.34 and Figure 2.35, Normalized Array factor is plotted in dB with Uniform distribution and electronic scanning. Scanning angle in the θ direction, θ_o , is 30° and in the φ direction, φ_o , is 60° . Thus, the sinespace representation variables, u and v , are calculated as follows.

$$u = \sin(\theta_z) \cos(\varphi) = \sin(30) \cos(60) = 0.25$$

$$v = \sin(\theta_z) \sin(\varphi) = \sin(30) \sin(60) \cong 0.433$$

This means, as can be seen from Figure 2.34 and Figure 2.35, Array Factor has a maximum value at $u = 0.25$ and $v = 0.433$.

2.10.4 Farfield Pressure of the Two Dimensional Array

2D array farfield pressure becomes more complicated due to additional elements in y direction but it can be expressed as follows

$$P(r, \theta, t) = j \frac{P_0 \times R_0}{r} \times EP \times AF \times \exp(j \times (wt - kr)) \quad (2.31)$$

where $P_0 = \rho_0 \times c_0 \times u_0$

To find the pressure at the farfield, $u(t)$ is taken as $u_0 \cos(w_0 t)$. Figure 2.36 and Figure 2.37 are obtained without electronic scanning, in other words, $\theta_0 = 0^\circ$, $\phi_0 = 0^\circ$ and they show the three dimensional view of the pressure for the array having properties defined in Table 2.1.

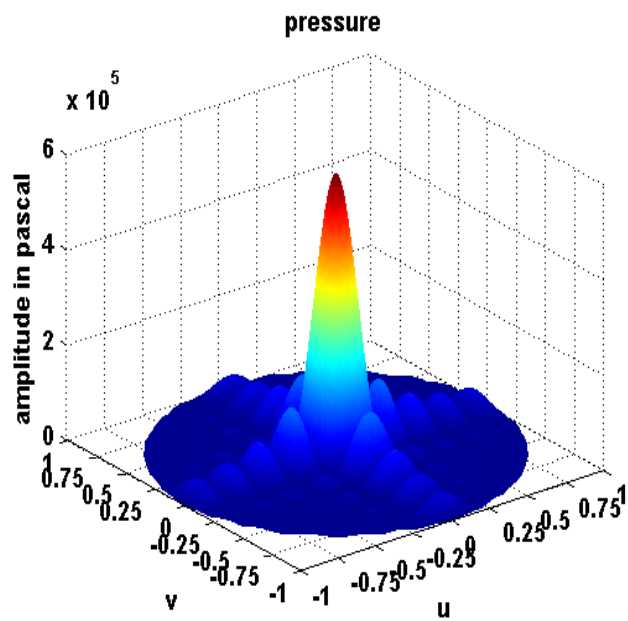


Figure 2.36: Three dimensional view of the Pressure for 30 kHz.

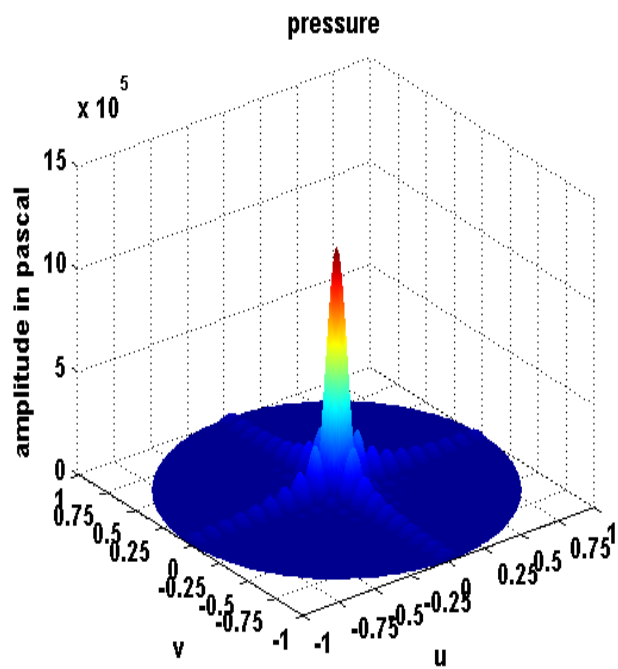


Figure 2.37: Three dimensional view of the Pressure for 60 kHz.

First array ($d = \lambda/4 = 0.0125 \text{ m}$ and $a = \lambda/8 = 0.0063 \text{ m}$) is operated under 30 kHz operating frequency and second array ($d = \lambda/2 = 0.0125 \text{ m}$ and $a = \lambda/4 = 0.0063 \text{ m}$) is operated under 60 kHz operating frequency. Array elements operating with two different frequencies, have same particle velocity. It can be seen from the Figure 2.36 and Figure 2.37, as the frequency increases, the farfield pressure of the array gets directive.

There is a parameter called Directivity Index (DI), which is a measure of how an available acoustic power can be concentrated in a desired direction and can be represented as

$$DI = 10 \log_{10} D_i \quad (2.32)$$

where D_i is the intensity directivity factor which can be defined as the ratio between the maximum intensity and the average intensity [10]. The other parameter is Source Level (SL) which is a measure of the far field pressure on the axis of an array referred to $1 \mu\text{Pa}$ and a point 1 m away from the acoustic center of the element. SL can be represented as follows.

$$SL = 10 \log_{10} W + DI + 170.8 \text{ dB referred to } 1 \mu\text{Pa and } 1 \text{ m} \quad (2.33)$$

Array operating with 60 kHz frequency has a bigger radiation resistance and narrower main lobe compared to 30 kHz. Thus, the radiated power and directivity index is greater than obtained in 30 kHz case. Therefore, the above equation shows us that in higher frequencies, the source level increases so peak value of the pressure increases as well.

Chapter 3

PRINCIPLES OF CONVENTIONAL ACTIVE SONAR TECHNOLOGY

Sonar systems are made up of arrays having generally 36 stacks and each stack often has 8 transducers. Sonar systems are used in a wide variety underwater applications with frequency extends from 1 hz to 1 Mhz. Applications over this wide frequency range requires different Sonar designs such as planar, conformal and spherical [11,23]. In this work, 3×3 planar array is used for simplicity and each element (transducer) is modeled as circular piston in order to make performance analysis of spread spectrum ping in active sonar technologies. Moreover, the transfer response of the transducer is omitted in this work. Beamforming technique is performed at one frequency but active sonars utilize pulses. So, the effect of each frequency component of the pulse has to be taken into account. Here, $u(t)$ is equal to $u_0 f(t)$ where $f(t)$ is defined as below.

$$f(t) = \begin{cases} 0, & t < 0 \text{ and } t \geq T_p \\ \sin(w_0 t), & \text{otherwise} \end{cases} \quad (3.1)$$

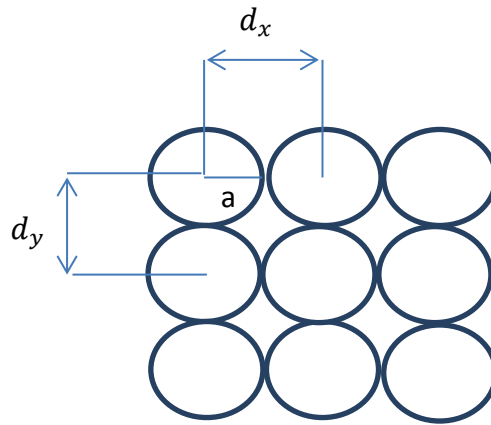


Figure 3.1: 3×3 active sonar array having $d_x = d_y = \lambda/2$ and $a = \lambda/4$

Here, the 3×3 active sonar array made up of circular pistons as elements is used. Each piston has a radius of a and elements are positioned as in Figure 3.1. The frequency, f_0 , is taken as 50 kHz. The pulse duration, $T_p = 20 \times \pi / \omega_0$, is about $200 \mu\text{sec}$ and the total duration of the pulse is about $2000 \mu\text{sec}$. Figure 3.2 shows this pulse in time domain.

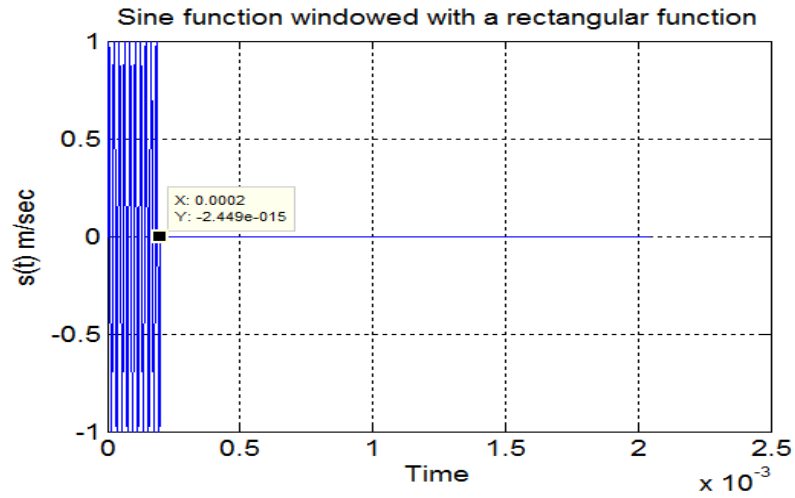


Figure 3.2: Pulse in time domain.

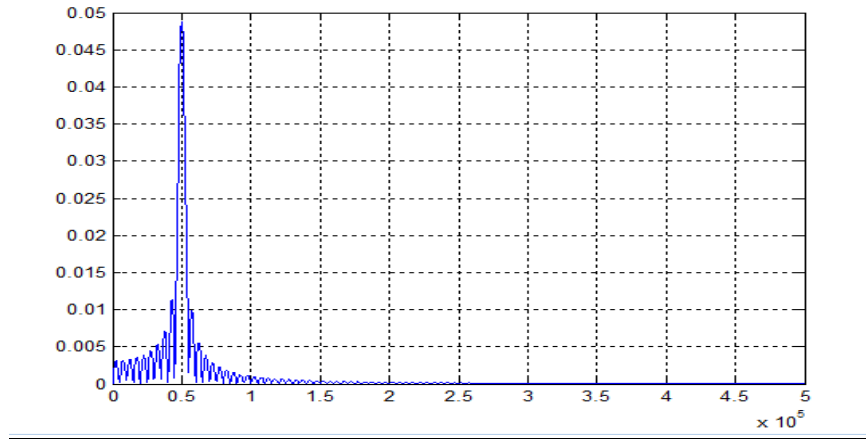


Figure 3.3: Pulse in frequency domain.

The fourier transform of a sinusoidal function is an impulse and fourier transform of a rectangular function is a sinc function. The multiplication of the signal with rectangular function in time domain equals to convolution of their frequency domains. Figure 3.3 shows the frequency domain of the rectangular

windowed pulse. Using rectangular window has a drawback that is the side lobes are relatively very high [7].

Pressure matrix in frequency domain is obtained by taking the frequency range and the theta angle vector into account as shown in the equation below.

$$P(f, \theta) = j \frac{P_0 R_0}{r} AF(f, \theta) EP(f, \theta) S(f) e^{-jkR} \quad (3.2)$$

where r is the distance between the array and the farfield point. $S(f)$ is the frequency domain representation of the pulse. In order to find the farfield pressure as a function of angle of incidence (θ) and frequency, the frequency domain representation of the pulse is multiplied with the beamformer transfer function as in the equation above. The term, e^{-jkR} , represents the time harmonic radiation and minus sign means that the radiation occurs in the forward hemisphere.

After generating the frequency domain pressure in the farfield region, by taking inverse fourier transform of this frequency domain pressure at one angle and at all frequency range, we can get the time domain pressure at this angle. Figure 3.4 and Figure 3.5 show the time domain pressure at $\theta = 0^\circ$ and $\theta = 30^\circ$, respectively. It must be noted that, in Figure 3.4 and Figure 3.5 transients occur due to the rectangular windowing.

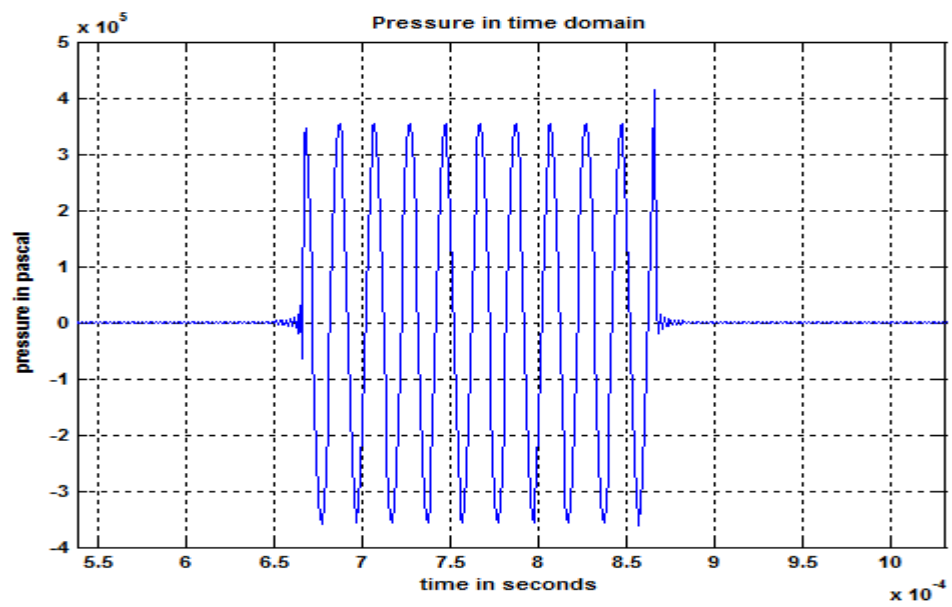


Figure 3.4: Time domain pressure at $\theta = 0^\circ$.

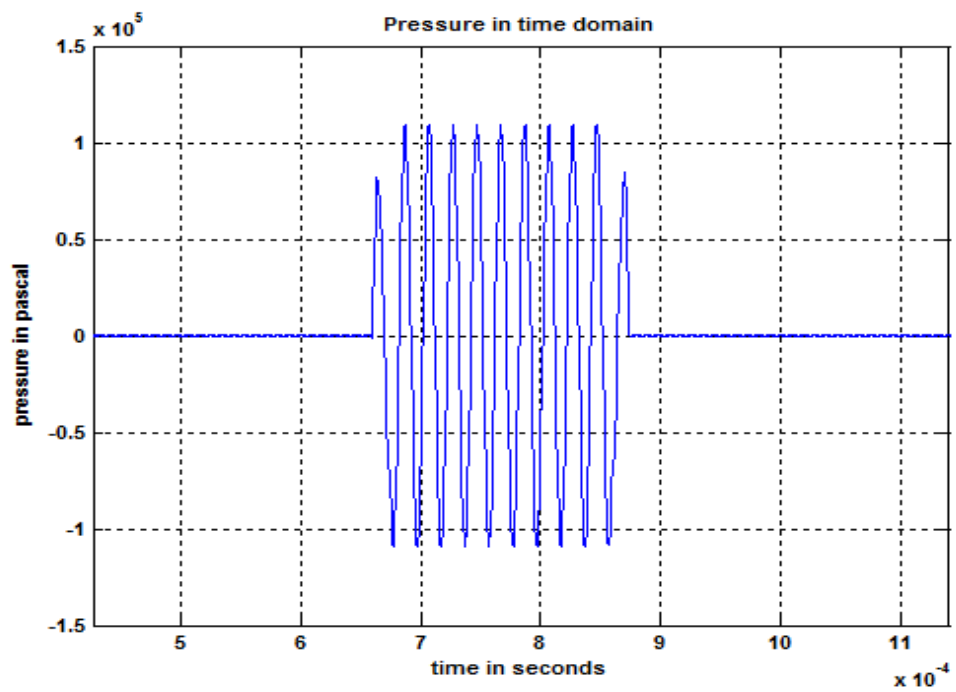


Figure 3.5: Time domain pressure at $\theta = 30^\circ$.

The frequency spectrum depends on which time windowing is used. There are variety types of time windowing techniques such as rectangular, raised cosine, hamming, etc... Here, instead of using a pulse which is shown in Figure 3.2, a pulse windowed with raised cosine is used. The expression of the raised cosine can be expressed as follows,

$$r(t) = 0.5 - 0.5 \times \cos \left[\left(2 \times \frac{\pi}{T_p} \right) \times t_{p_{range}} \right] \quad (3.3)$$

The Raised Cosine signal obtained from equation (3.3), pulse windowed with Raised Cosine and the frequency domain representation are shown in the Figure 3.6, Figure 3.7 and Figure 3.8 respectively.

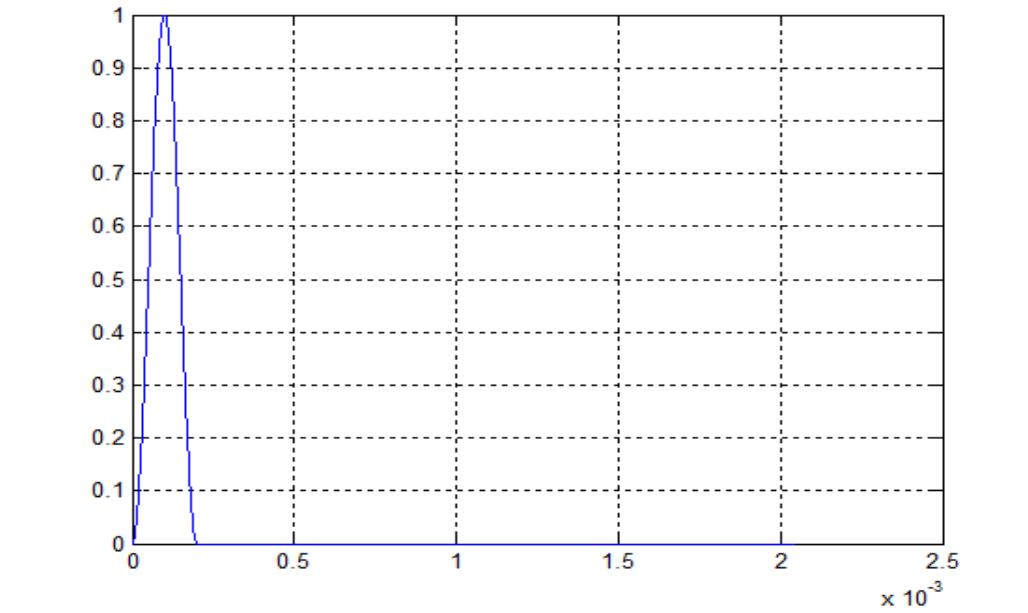


Figure 3.6: Raised Cosine in time domain.

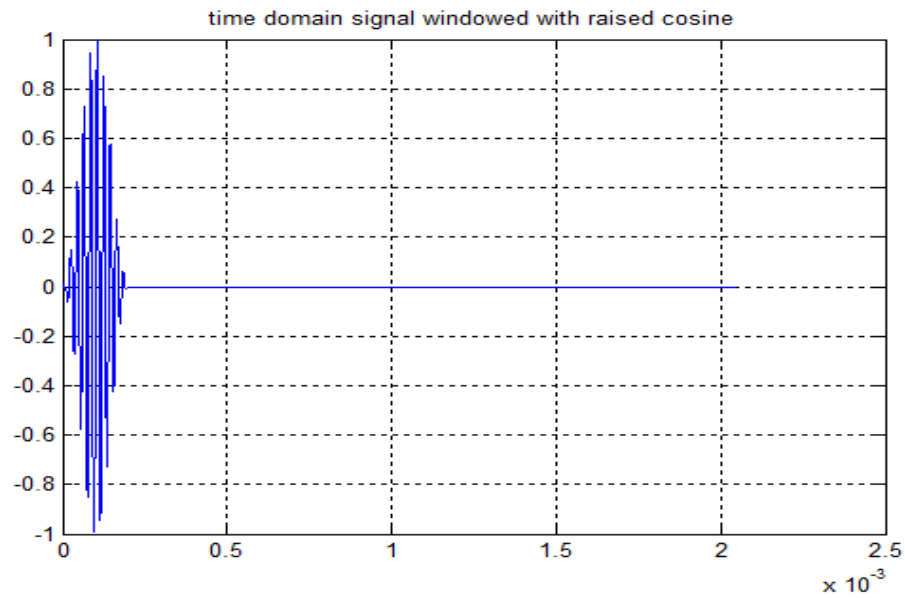


Figure 3.7: Pulse windowed with raised cosine in time domain.

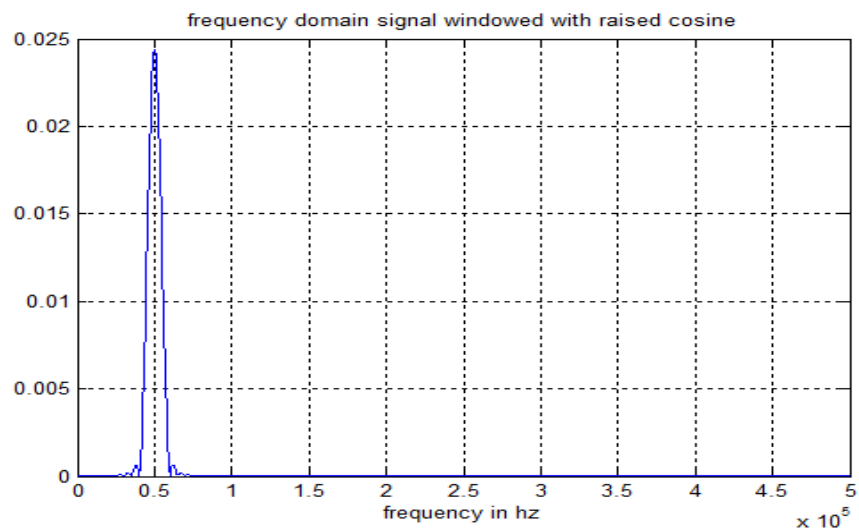


Figure 3.8: Frequency domain representation of windowed signal.

The main side lobes problem of rectangular windowing can be alleviated by using raised cosine windowing. As can be seen from the Figure 3.8, the side lobes level is very low compared those obtained in rectangular windowing and also they are not visible after some frequency values.

The time domain pressure obtained by using raised coseine windowing at $\theta = 0^\circ$ and $\theta = 30^\circ$ is shown in the Figure 3.9 and Figure 3.10, respectively.

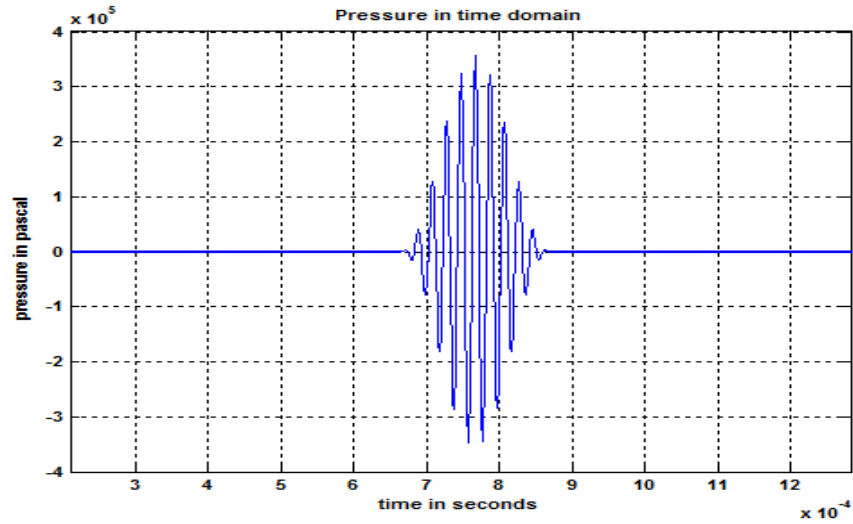


Figure 3.9: Pressure in time domain at $\theta = 0^\circ$.

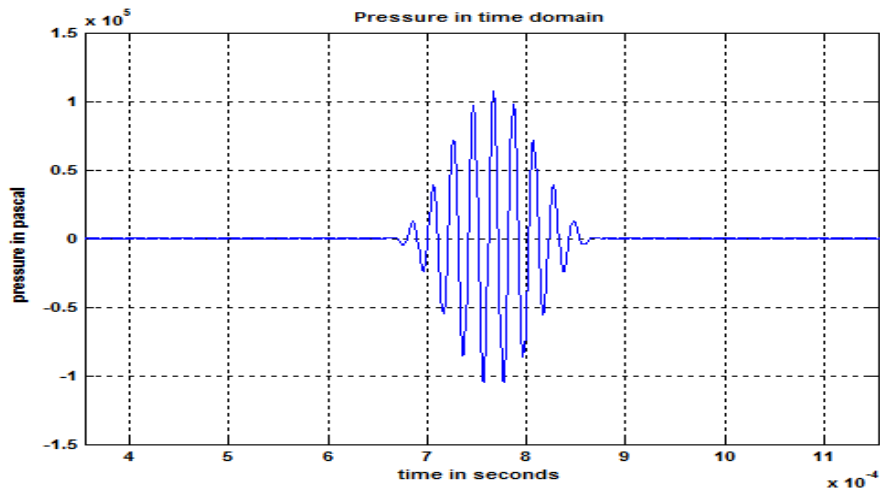


Figure 3.10: Pressure in time domain at $\theta = 30^\circ$.

One of the major drawback of rectangular windowing is that the transients occur. By using raised cosine windowing, the transients do not occur. Moreover, by using raised cosine windowing, the band of the transducer can be used more efficient and the acoustic power can be concentrated into the main beam easily [4,7,11].

Chapter 4

COVERT USE OF ACTIVE SONAR SYSTEMS BY MEANS OF SPREAD SPECTRUM PINGS

In conventional active sonar systems, a pulse of sound called ping is transmitted from the array without beamforming and the echoes coming from the target received by the array and beamformed. So, the target detection and localization are performed. Beamforming can be possible in transmission mode, only when the coded signals are transmitted [12,22]. In this work, Beamforming technique is used in both transmission and reception mode. The coded direct sequence spread spectrum signals with length 7, 15, 31, 63 127 chips generated for the wireless communication systems are used. These coded signals are transmitted to the different sectors and echoes received from different sectors by beamforming technique. In order to increase the performance, correlation receiver is also used in reception mode. It is found that, the coded signals generated for wireless communication applications are insufficient so new ping technology must be found for active sonar systems.

4.1 Direct Sequence Spread Spectrum

In communication systems, sharing the same band of frequencies comes into prominence in recent years owing to scarce available bandwidth. Some techniques are developed to share the same band of frequencies efficiently such as Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA). FDMA is multiple access technique in which each user is assigned to a frequency band. TDMA is also multiple access technique in which

each user is allowed to use all frequency bands in different time slots. There is an another multiple access technique which is called Code Division Multiple Access (CDMA) in which Spread Spectrum techniques are employed and each user can send an information simultaneously so that users can share the total frequency band at the same time without interference [13].

Spread spectrum technique is a modulation technique in which the information signal is spread over a much wider bandwidth before transmission. Spread spectrum technique is divided into two parts: Frequency Hopping (FH) and Direct Sequence (DS). In this work, Direct Sequence Spread Spectrum (DSSS) technique is used. A transmitted DSSS signal can be defined as follows

$$x(t) = A \times d(t) \times q(t) \times \cos(2 \times \pi \times fc \times t + \theta) \quad (4.1)$$

where A is the amplitude, $d(t)$ is the information, $q(t)$ is the spreading code waveform, fc is carrier frequency and θ is the phase. Spreading process is shown in Figure 4 .1 and can be achieved by multiplying the information with a spreading code which changes at every chip duration, T_c . Since chip duration, T_c , is much smaller than the pulse duration of the information signal, T_b , the bandwidth of the spread spectrum signal is much larger than the bandwidth of the information signal [13,14,15].

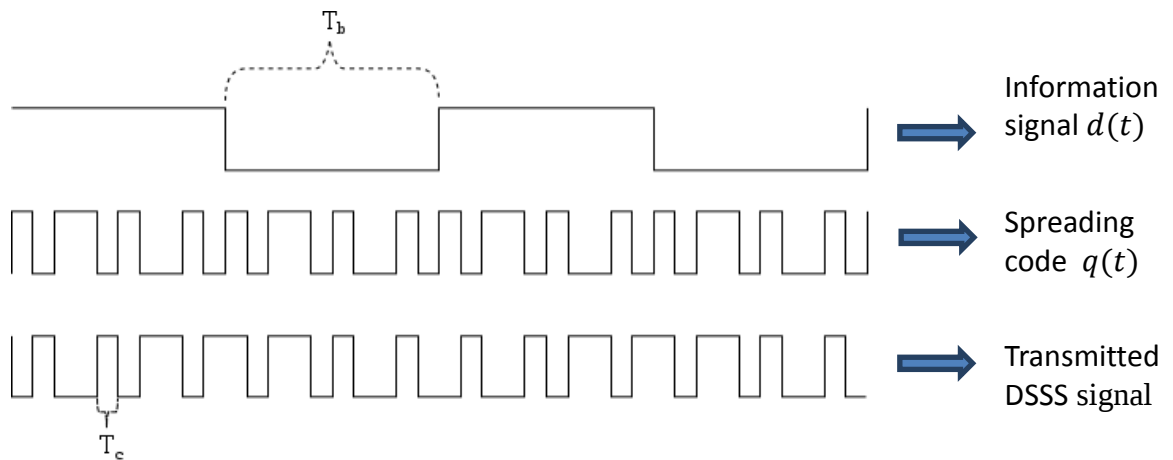


Figure 4 .1: Basic schema of how to generate the Direct Sequence Spread Spectrum signal

In Figure 4.2, a transmitted 15 chip DSSS signal with $T_c = 100 \mu s$ and one bit information which has $T_b = 15T_c = 1500 \mu s$ could be seen. At some certain times, 180° phase shifts exist. In digital domain this DSSS signal can be represented as 000100110101111.

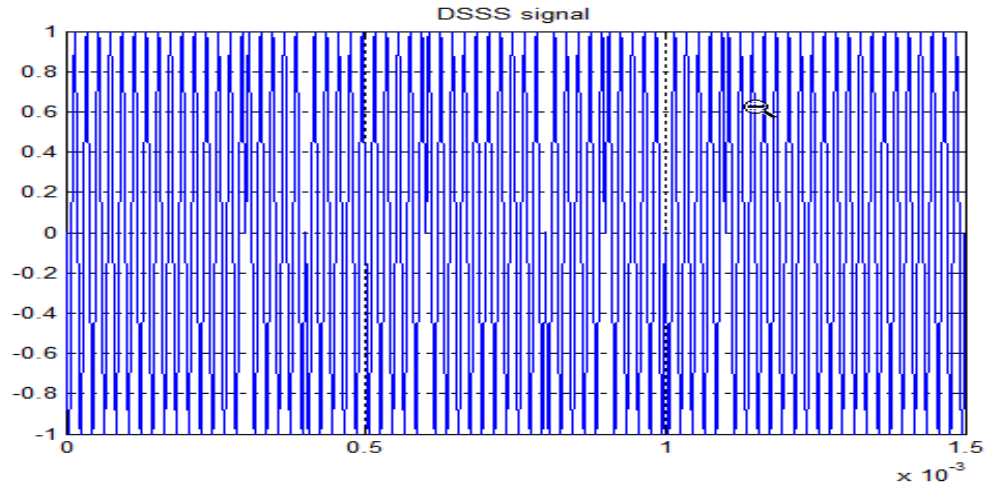


Figure 4.2: 15 Chip DSSS signal

The frequency spectrum of 15 chip DSSS signal is illustrated in Figure 4.3. By using DSSS modulating technique, one bit information which has a duration of $T_b = 15T_c = 1500 \mu s$ and null to null bandwidth of $2/T_b \approx 1.333 \text{ kHz}$ is spread over a much wider null to null bandwidth of 20 kHz [14].

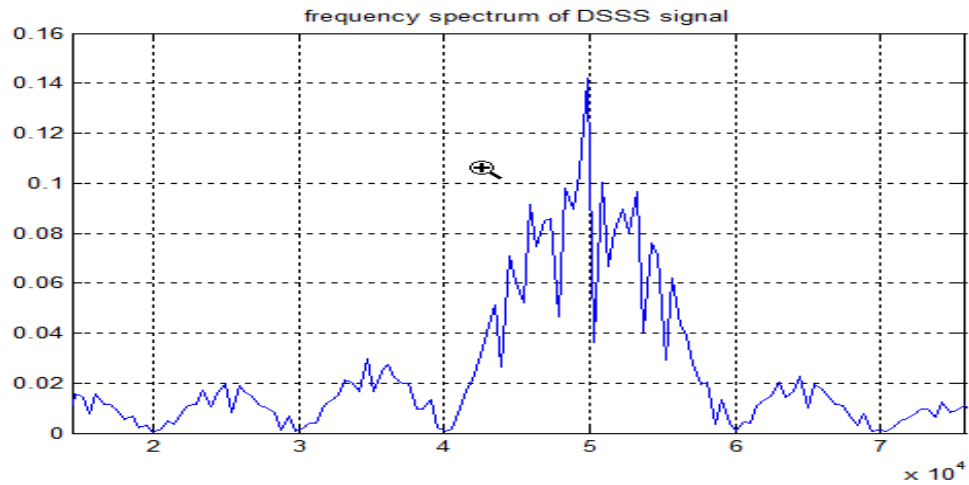


Figure 4.3: Frequency spectrum of 15 Chip DSSS signal

DSSS modulation technique provides a communication which has a low probability of interception and detection. Even if unfriendly listeners or enemies listen the environment, they can not detect the communication [14,16]. Also, better range resolution and increased robustness to multipath and interference can be achieved by using DSSS technique.

4.2 Maximal Length Sequences

Maximal Length (ML) Sequences which are also called m-sequences or pseudo noise sequences are commonly used in CDMA systems for wireless communication applications due to their properties. Maximal Length sequences can be generated by Linear Feedback Shift Registers (LFSR). LFSR consists of n storage states, feedback logic circuit and modulo-2 adders. Initial state and feedback logic circuit completely specifies the consecutive states so that ML sequences are deterministic. Initial state is shifted through the feedback shift register in response to clock signal and the first state is determined by the feedback logic and the last state determines the output. In order to prevent the output to remain zero all the time, initial state can not be chosen as zeros. Since there are $2^n - 1$ different arrangements of the states, any sequence has a length of $2^n - 1$. This also leads $2^n - 1$ different possible ML sequences [15,16,17].

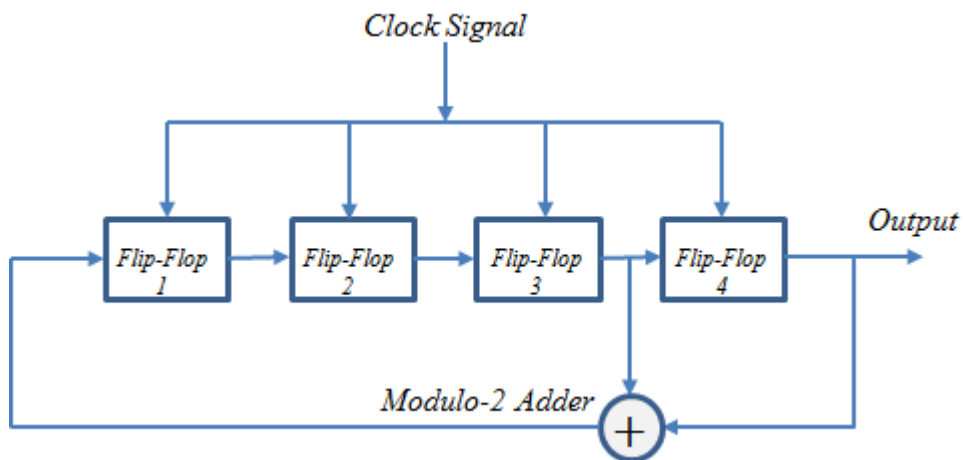


Figure 4.4: Linear Feedback Shift Register with 4 stages

Four stage Linear feedback shift register which produces sequence of length 15 is illustrated in Figure 4.4. Let the initial state of each flip-flop be (0, 0, 0, 1). After shifting, modulo-2 adding and feeding processes, LFSR produces a sequence (0 0 0 1 0 0 1 1 0 1 0 1 1 1 1) which is shown in Table 4.1 as code output. It can be noticed that from the table, initial state of the flip flop is repeated at every 15 cycles.

Table 4.1: Initial states of the Linear Feedback Shift Register

Clock Signal	Flip Flop 1	Flip Flop 2	Flip Flop 3	Flip Flop 4	Code Output
Initial State	0	0	0	1	1
↑	1	0	0	0	0
↑	0	1	0	0	0
↑	0	0	1	0	0
↑	1	0	0	1	1
↑	1	1	0	0	0
↑	0	1	1	0	0
↑	1	0	1	1	1
↑	0	1	0	1	1
↑	1	0	1	0	0
↑	1	1	0	1	1
↑	1	1	1	1	0
↑	1	1	1	1	1
↑	0	1	1	1	1
↑	0	0	1	1	1
↑	0	0	0	1	1
↑	1	0	0	0	0

Maximal Length sequences have many properties. The number of 1's always exceeds the number of 0's by 1 which is called BALANCE property. The 15 possible ML sequences are demonstrated in Table 4.2. It can be noticed that any left or right cycle shift of a sequence is also a ML sequence which is called SHIFT property. The modulo-2 summation of any two ML sequence is also a ML sequence which is called ADD property [17].

Table 4.2: Maximal Length sequences generated by the Linear Feedback Shift Register

0	0	0	1	0	0	1	1	0	1	0	1	1	1	1
0	0	1	0	0	1	1	0	1	0	1	1	1	1	0
0	1	0	0	1	1	0	1	0	1	1	1	1	0	0
1	0	0	1	1	0	1	0	1	1	1	1	0	0	0
0	0	1	1	0	1	0	1	1	1	1	0	0	0	1
0	1	1	0	1	0	1	1	1	1	0	0	0	1	0
1	1	0	1	0	1	1	1	1	0	0	0	1	0	0
1	0	1	0	1	1	1	1	0	0	0	1	0	0	1
0	1	0	1	1	1	1	0	0	0	1	0	0	1	1
1	0	1	1	1	1	0	0	0	1	0	0	1	1	0
0	1	1	1	1	0	0	0	1	0	0	1	1	0	1
1	1	1	1	0	0	0	1	0	0	1	1	0	1	0
1	1	1	0	0	0	1	0	0	1	1	0	1	0	1
1	1	0	0	0	1	0	0	1	1	0	1	0	1	1
1	0	0	0	1	0	0	1	1	0	1	0	1	1	1

Maximal Length sequences have been used in a wide variety areas because of their ideal autocorrelation properties. The autocorrelation function of ML sequence with length N is as follows

$$R(i) = \begin{cases} 1 & \text{for } i = 0 \\ -1/N & \text{for } i \neq 0 \end{cases} \quad (4.2)$$

This autocorrelation function means that if two possible different sequence is compared bit by bit, the number of disagreements is always one more than the number of agreements.

Correlation results of two different codes with 7 chips and 15 chips are shown in Figure 4.5 and Figure 4.6. It can be stated that as N increases, the cross correlation peak values decreases but the decreasing rate or improvement differs slightly.

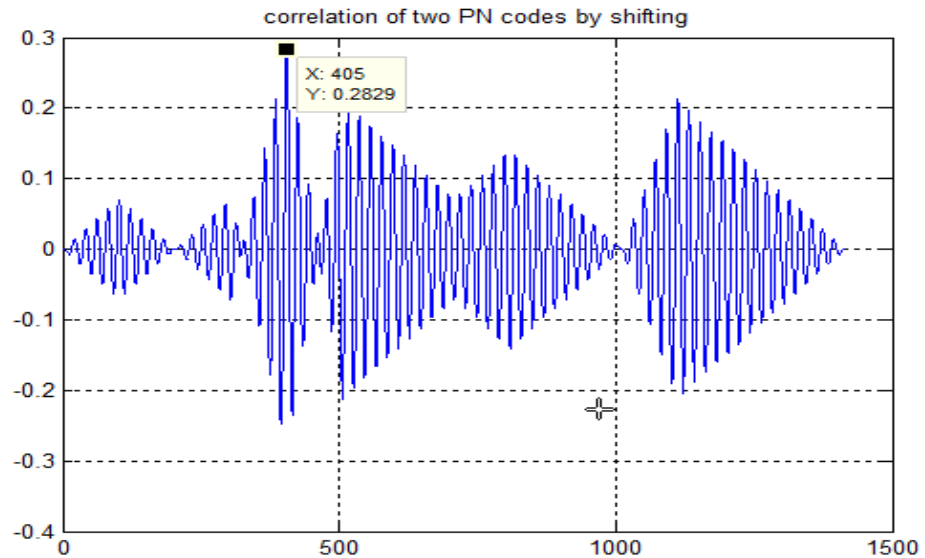


Figure 4.5: Correlation result of two different 7 chip PN codes

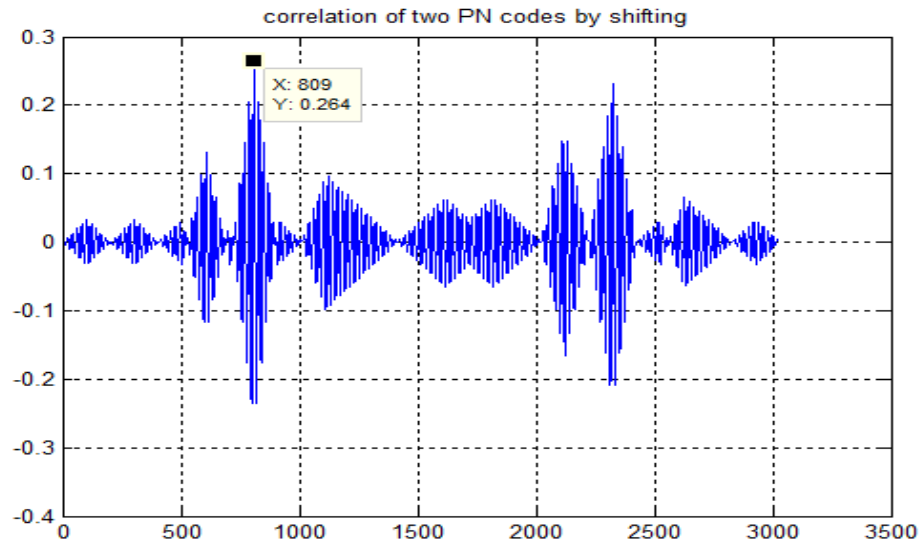


Figure 4.6: Correlation result of two different 15 chip PN codes

4.3 Acoustic Properties and Signatures of Underwater Targets

Sound navigation and ranging systems produce and transmit a pulse of sound, often called a ping, then detect the reflected energy from the target. This type of systems are called Active Sonar Systems. Some systems only try to intercept the radiated energy from the target. This type of systems are called Passive Sonar Systems. Active sonar systems are of interest throughout this work. Active sonars basically consist of a transmitter and a receiver. When the transmitter and receiver are in the same place, the operation is monostatic while they are in different places, the operation is multistatic operation.

In active sonar systems, target strength is an acoustic signature and determines the intensity of the wave which is reflected from the target [2]. Monostatic, bistatic operations and target strength phenomena is illustrated in Figure 4.7. The source intensity, I_0 is measured at 1m away from the source.

Considering spherical spreading, the intensity at the target can be represented as follows

$$I_i = I_0/r_1^2 \quad (4.3)$$

I_{r1} is the intensity reflected from the target and depends on the physical characteristics of the target and the incoming wave such as frequency and angle. It can be represented as follows

$$I_{r1} = t_{s1} \times I_i \quad (4.4)$$

Because of the spherical spreading, the intensity measured at 1 m from the receiver is,

$$I_R(\text{Monostatic}) = I_{r1}/r_1^2 = t_{s1} \times I_i/r_1^2 = t_{s1} \times I_0/r_1^4 \quad (4.5)$$

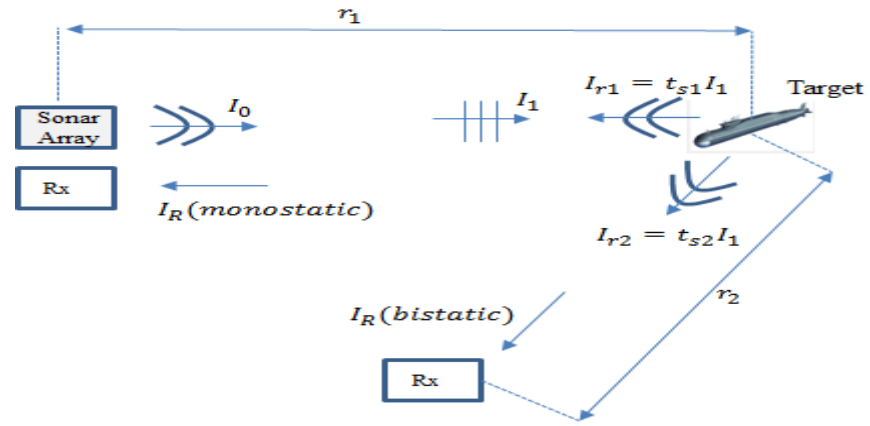


Figure 4.7: Monostatic and Bistatic operation

t_{s1} and t_{s2} are different because of the shape of the target. For the bistatic case, the intensity at the receiver can be represented as

$$I_R(\text{Bistatic}) = I_{r2}/r_2^2 = t_{s2} \times I_i/r_2^2 = t_{s2} \times I_0/(r_1^2 \times r_2^2) \quad (4.6)$$

In the figures below, by taking the target strength as 0 dB, the correlation results of reference and received signals are shown for a sinusoidal signal windowed with rectangular function and DSSS signals with different number of chips (7,15) under the noiseless environment.

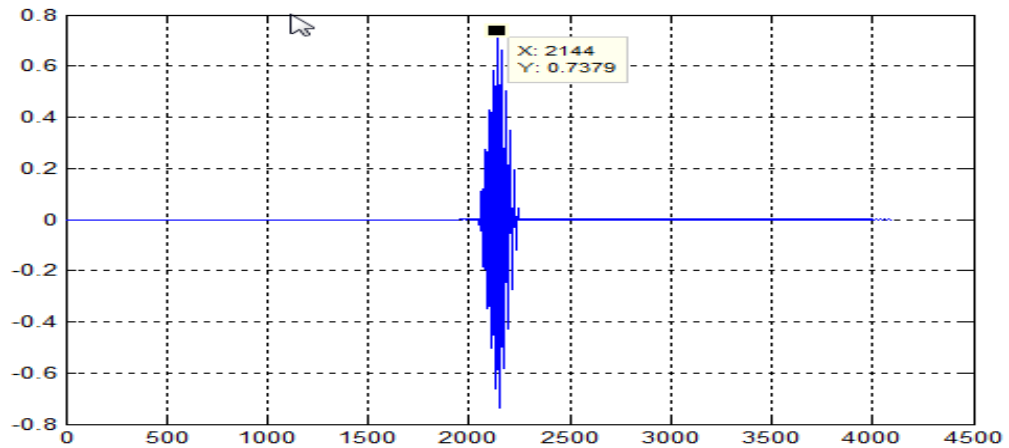


Figure 4.8: Correlation result of transmitted and received signal for the sinusoidal signal (TS = 0 dB).

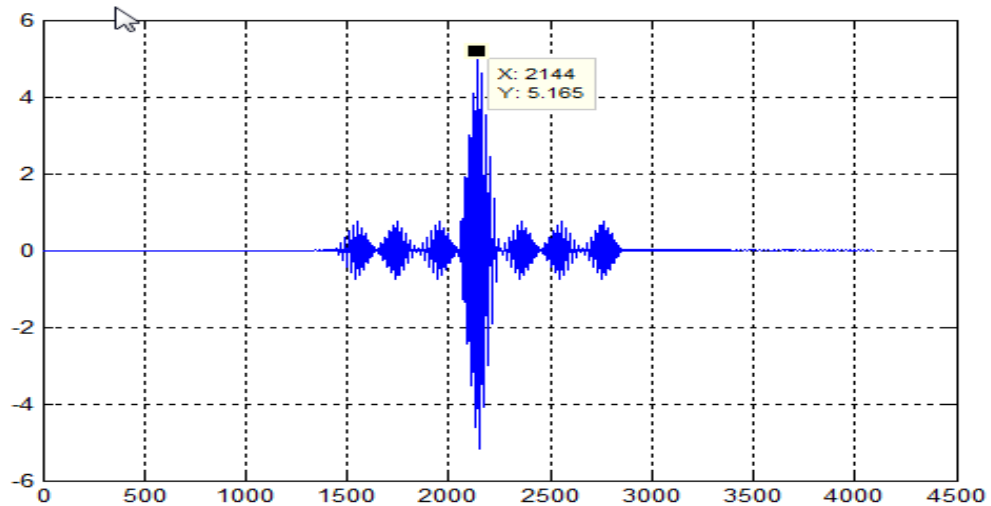


Figure 4.9: Correlation result of transmitted and received signal for the 7 chip DSSS signal (TS = 0 dB).

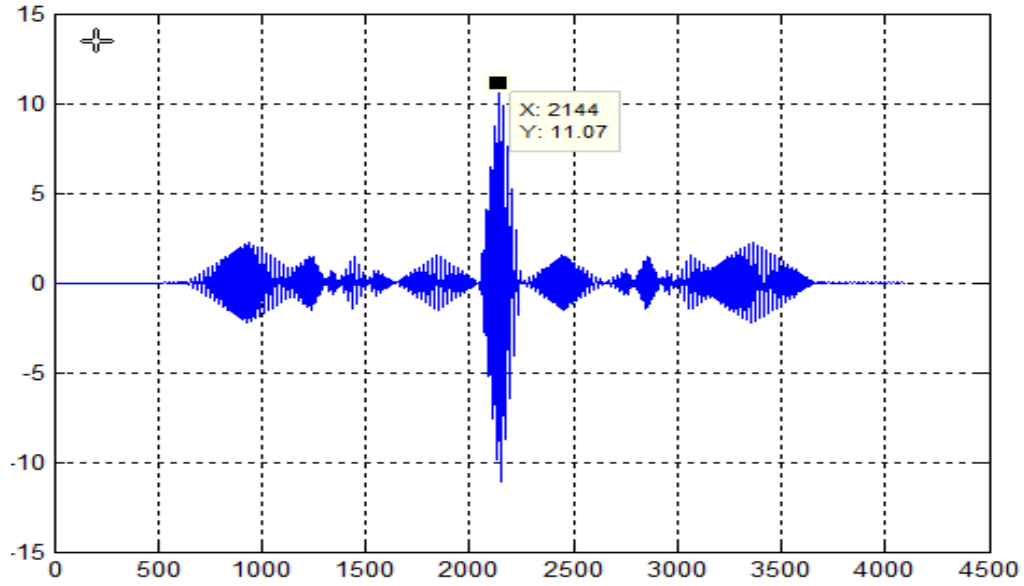


Figure 4.10: Correlation result of transmitted and received signal for the 15 chip DSSS signal (TS = 0 dB).

The active sonar systems take the correlation between the reference signal and received signal also correlation between reference signal and noise. In the figures above, the environment is assumed as noiseless so only the correlation between reference signal and received signal is considered. Actually what we have done is as follows.

$$\int_{-\infty}^{+\infty} s(t) y(t - \tau) d(t) \quad (4.7)$$

where $s(t)$ is the reference signal, $y(t)$ is the received signal and τ is the delay. This figures show that, when the number of chip is increased, the peak value of the correlation is increased so that the detection probability of the underwater target is increased.

4.4 Underwater Ambient Noise and its Characteristics

Ambient noise which is commonly known as background noise, is the received signal by the omnidirectional hydrophone located at a point in the sea when all the identifiable noise sources and self noise sources are eliminated. As the name implies, ambient noise surrounds the hydrophone from all sides. Ambient noise sources can be divided into two parts, natural sources and artificial (human made) sources such as distant shipping, seismic turbulences, agitation of the sea surface by the wind, thermal agitation of the sea water molecules, etc...[1,2,18]

In order to find the spectrum level of the ambient noise in dB relative to $1\mu Pa$ in a 1 Hz frequency band, measurements are performed in a broad frequency range extending from 1 Hz up to 100 kHz. The frequency range is divided into five frequency regions in which characteristics of ambient noise change with different slopes as shown in Figure 4.11 [18].

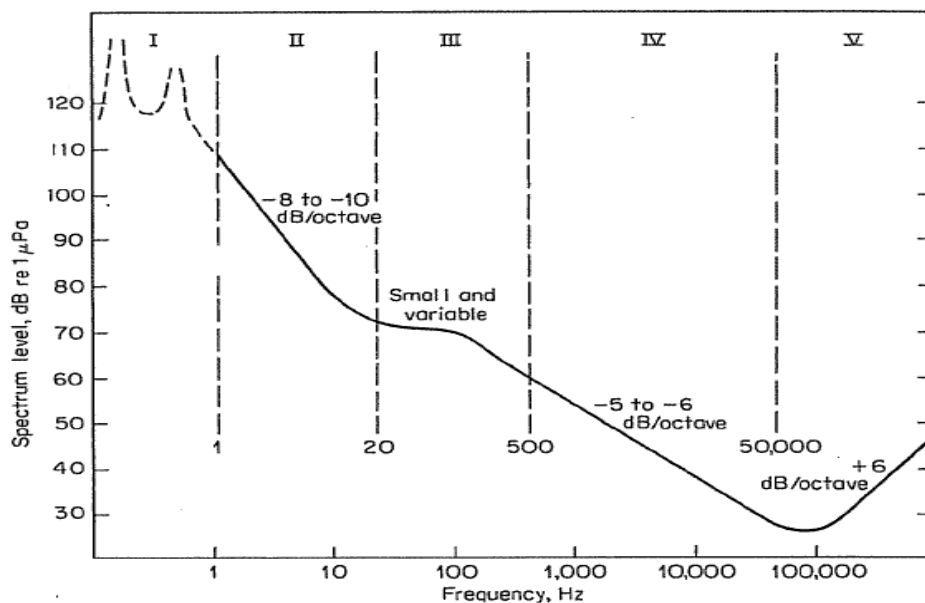


Figure 4.11: Logarithmic scale of the Spectrum level ($dB//1\mu Pa/Hz$) of ambient noise in different conditions.

Ambient noise as a parameter of the sonar systems is correlated with the reference signal at the receiver part of the sonar. In our work, frequency range between 35 kHz and 75 kHz stores the %95 of the total energy of transmitted signal. Thus, the spectrum level of the ambient noise in region IV having a slope of -5 to -6 dB/octave is utilized.

Logarithmic scale of the region IV is converted to the linear scale with unit Pa/Hz as shown in Figure 4.12.

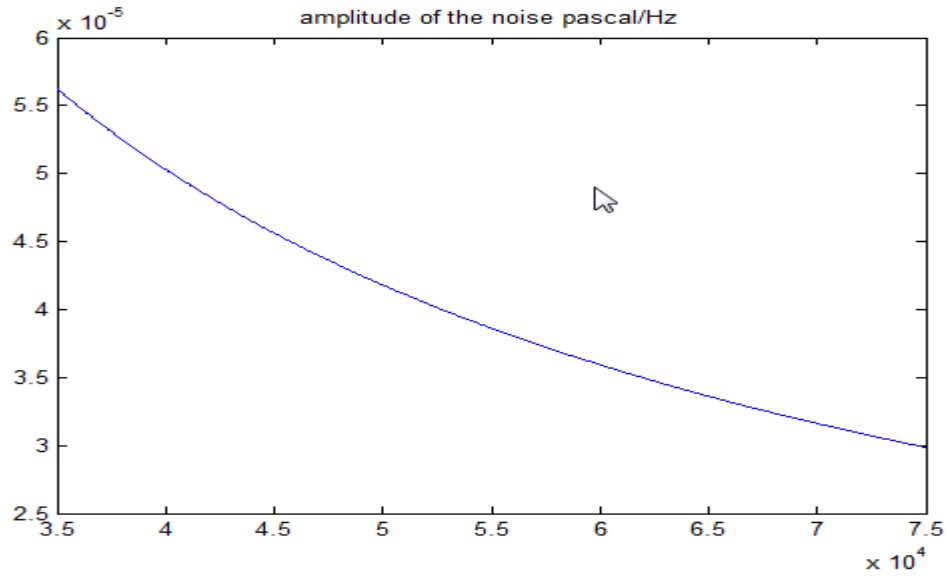


Figure 4.12: Linear scale of ambient noise in **Pa/Hz**.

Sound intensity is defined as the sound power per unit area or time average of the energy flow through a unit area [1,2,4]. Sound intensity can be described as follows,

$$\vec{I} = p \times \vec{v} \quad (4.8)$$

where $\vec{I}$ is the sound intensity, p is the sound pressure and $\vec{v}$ is the particle velocity. Both sound intensity and particle velocity are vector quantities which means that the direction of the sound intensity is the direction in which

the energy flows. Also sound intensity as a function of pressure and characteristic impedance can be written as follows,

$$I = p_{rms}^2 / \rho_0 c_0 \quad (4.9)$$

Sound power is the sound energy per unit time and sound power measured in watts passing through a surface can be computed as sound intensity times surface.

$$W = I \times S \quad (4.10)$$

Considering the ambient noise spectrum level defined above, the noise power can be computed by

$$W_n = I_n \times S = \left(\int_{f_1}^{f_2} p_n^2 / df \right) S / \rho_0 c_0 \quad (4.11)$$

In sonar systems, success in detecting the target can be improved by increasing the signal to noise ratio (SNR) which is often expressed in decibels. SNR compares the level of desired signal to the level of background noise. In other words, the SNR is the ratio between the signal power and noise power. It is obvious that, in order to calculate signal to noise ratio, the equation below should be calculated.

$$\frac{\int_{f_1}^{f_2} p_s^2 / df}{\int_{f_1}^{f_2} p_n^2 / df} \quad (4.12)$$

In Figure 4.13, the power spectral density of the noise is illustrated in 35 kHz and 75 kHz frequency range. The noise power can be also obtained by using the noise power spectral density with a unit of

$$(watts / m^2) \times (\rho \times c) \quad (4.13)$$

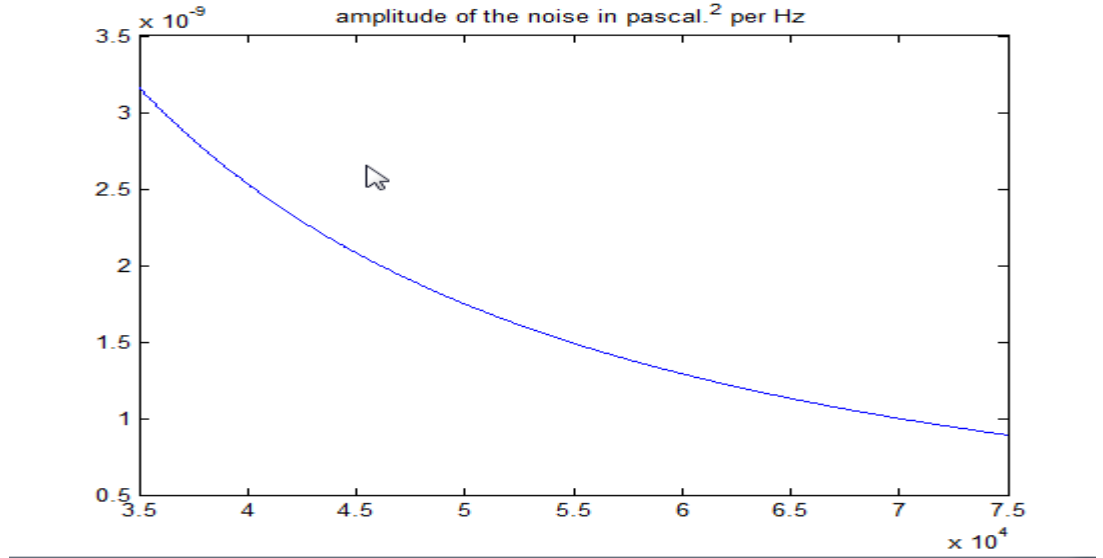


Figure 4.13: Linear scale of ambient noise in Pa^2/Hz

In noise limited communication systems, maximum achievable Signal to Noise ratio (SNR) must be obtained in order to increase detection and noise performance of the system. It can be thought that, because of the spreaded nature of DSSS signal, the noise and detection performances of DSSS modulation technique is poor which is needed a wide band receive filter passing more noise into the system. By using matched filtering, the waveform and receive filter response stay in a matched condition, thus, SNR, detection and noise performances of the system is increased [9,14,19].

The SNR at the output of the matched filter is as follows

$$SNR_{out} = \frac{E_{signal}}{PSD_{noise}} \quad (4.14)$$

The SNR at the matched filter output is proportional to the signal in the received echo and is inversely proportional to the power spectral density of the noise. It does not depend on the total noise power at the filter input and the bandwidth of the filter. It can be stated that, two waveforms having the same energy generate the same SNR each is processed by its own matched filter [20].

4.5 Signal to Noise ratio (SNR) and Signal to Interference and Distortion Ratio (SINAD) Results

In this work, two different scenarios are considered to make the performance analysis of spread spectrum pings in active sonar technology. In the both scenarios, the performance of spread spectrum pings with different lengths change from 7 to 127 chips is compared with the performance of Tone burst pings which are used as a reference. In the first scenario, the spread spectrum coded ping is used in conventional active sonar system and is transmitted through the entire sector. The description of the first scenario is illustrated in Figure 4.14.

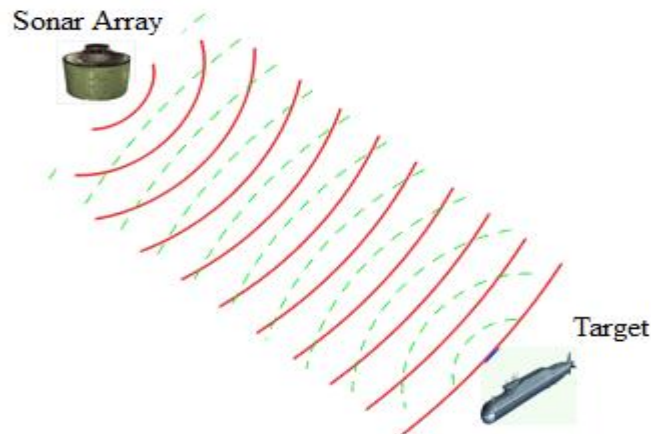


Figure 4.14: The description of the first scenario.

The results of first scenario are illustrated in Table 4.3. It shows the SNR values with respect to number of chips. As the number of chips is increased, the SNR values are increased as well. Therefore, the success in detecting the target is increased. It can be concluded that, using spread spectrum pings in conventional active sonar technology performs very well.

Table 4.3: Signal to Noise Ratio with respect to number of chips

Number of chips	SNR in dB
Single chip	0
7 chip	9.6
15 chip	14
31 chip	15.2
63 chip	18.3
127 chip	22.1

The main aim is to achieve high covertness and high probability of detection. Thus, in the final scenario, different spread spectrum coded pings are used in active sonar technology and transmitted through the directions that the target might be located. Using different spread spectrum coded pings in active sonar systems enables covertness and robustness to the interference. In this scenario, a reference signal coded by C1 is transmitted through a sector, another signal coded by C2 is transmitted another sector and so on. The description of the second scenario is illustrated in Figure 4.15.

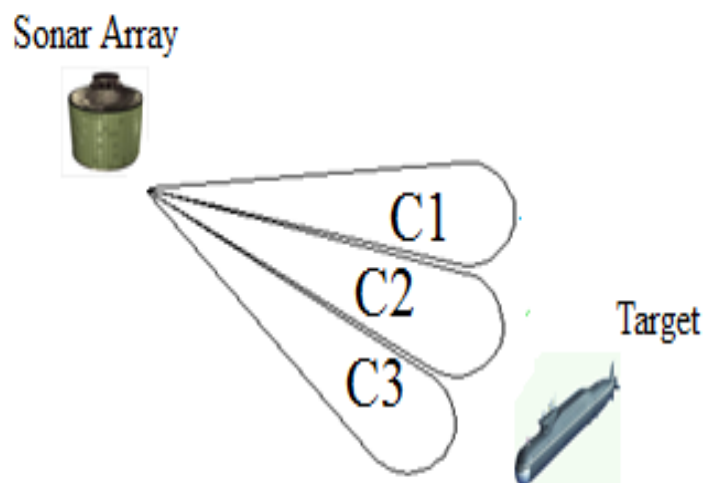


Figure 4.15: The description of the second scenario

An interaction or interference between each transmitted coded signal will occur in this scenario. Interference should be minimized in order to get a reliable detection and localization of the target. SINAD values with respect to number of chips is shown in Table 4.4.

Table 4.4:Signal to Interference Ratio with respect to number of chips

Number of chips	SINAD in dB
Single chip	0
7 chip	6.7
15 chip	8.2
31 chip	8.7
63 chip	9
127 chip	9.1

As the number of chips is increased, SINAD values are increased as well but the increasing rate or improvement is comparatively less because of the cross correlation properties of the maximal length sequences. The maximal length sequences are generated for wireless communication systems so they are not suitable for active sonar technologies. The cross correlation property of the maximal length sequences is insufficient for this application. It can be concluded that using different spread spectrum coded pings requires to research and develop a new spread spectrum coded ping signal for better signal to interference ratio and perfect cross correlation properties.

Chapter 5

DISCUSSION & CONCLUSION

The performance analysis of spread spectrum pings as a new ping technology in active sonar systems is investigated in this work. As a spread spectrum ping, maximal length sequences generated for wireless communication purposes are used. In this work, first, in order to understand beamforming technique and some fundamental topics of sonar systems, 1D (one dimensional) and by adding some elements in y direction, 2D (two dimensional) arrays are modeled and the effects of frequency, number of elements in the array, element spacing, the radius of element in the radiation pattern and farfield pressure are discussed. Then, in order to make performance analysis, the conventional ping technology is compared with spread spectrum ping technology in active sonar systems by considering two different scenarios defined below.

The length of the conventional ping can not be increased indefinitely due to the multipath propagation but length of the spread spectrum coded pings can be increased almost as much as the range permits. In this work, the length is increased from 7 chips to 127 chips. In conventional active sonar systems, beamforming technique can be used only in reception mode. By using spread spectrum coded pings, beamforming can be performed in both transmission and reception modes.

In this work, two different scenarios are considered in order to make performance analysis of spread spectrum coded pings in active sonar technology. In both scenario the performance of spread spectrum technique is compared with tone burst technique and tone burst technique is taken as reference. In the first scenario, spread spectrum coded ping is used in conventional sonar system and is transmitted through all sector. By using maximal length sequences as a spread spectrum ping enables an improvement in

signal to noise ratio (SNR). In other words, using spread spectrum coded ping in conventional active sonar technology performs very well and it is found that the success in detecting the target is increased. In the last scenario, in order to achieve high covertness, different spread spectrum coded pings are used in active sonar technology. Different spread spectrum coded signals are transmitted to different sectors and the echoes coming from different sectors are received and beamformed. Finally, It is found that signal to interference and distortion ratio (SINAD) between different spread spectrum coded pings is increased but the improvement is comparatively less. As it is mentioned before, the maximal length sequences are generated for wireless communication purposes, i.e, they are not generated for active sonar technologies [13,16]. So, the cross correlation properties of maximal length sequences are insufficient for active sonar technologies. In order to minimize the interference between each sector and increase the success in detecting and localization of the target, a sequence having perfect cross correlation properties must be generated for active sonar applications.

Some assumptions are made during this work such as the particle velocity of each transducer is set to a certain value and the response of transducer is omitted. In practical applications, the voltage given to each element is controlled. The particle velocity at the face of each transducer can not be set to a certain value due to the mutual impedance which occurs because of the interaction between each array elements. In other words, particle velocity is not a controllable parameter and changes very dramatically because of the mutual impedance. Even if, the same voltage is applied to each array element, the particle velocity at the face of each element will be different. If the mutual impedance is omitted, the array will be directed to undesired direction so target can not be detected.

In future work, beamforming by means of electrical parameters must be studied. Furthermore, the coded beam sonar requires super linear and low noise power amplifier. Power levels are so low in spread spectrum technique as

compared with tone burst technique so the output power limitation is irrelevant. Transducers having wide bandwidth and high receive sensitivity must be fabricated. Also the transmission power output is irrelevant as well. Moreover, simulation studies must be done to include the entire wet end.

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