Design of Blind and Non-Blind Adaptive filter for interference cancellation in SONAR

By

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(Muhammad Ahsan)

Dedicated to my parents

Abstract

In adaptive beam forming, the beam produced by single sensor is cumulative result of all elements in SONAR. The side lobes produced by any sensor is cancelled by the neighboring sensors. When any sensor is defective then the interference is not cancelled and the beam pattern produces major degradation in beam pattern by producing the side lobes. Due to this factor, the detection of target becomes difficult. In this research, we have applied two types of filters blind and non-blind adaptive filters that cancel the interference caused due to defective sensors in SONAR. We have used adaptive filter when the desired beam pattern is known at the receiver or it is not known at the receiver side. It has been observed that even if up to 50% of the sensors are degraded, proposed system works at its optimum level. Results show that adaptive filter is computationally efficient, works in presence of noise and operates in the environment when the actual beam pattern is known or not known at the receiver side. It has been noted that filter produces output that is very near to its optimum value. The results show that it is feasible to construct a real time acoustic beam former with realistic parameters using parallel processing architecture.

بسم الله الرحمن الرحيم

CHAPTER 1

1. INTRODUCTION

1.1. INTRODUCTION

What is sonar? - Sound Navigation and Ranging. In a sense it is comparable with radar except in this case we are using acoustic waves propagating through water as opposed to EM radiation in air.The prime role of sonar, be it active or passive, is to DETECT, CLASSIFY and to TRACK a contact, be it hostile or friendly. There are however also some important secondary roles, namely NAVIGATION and COMMUNICATION. The list can be further expanded as below:

a. Search / Attack - specific role sonar systems.
b. Classification - exploits acoustic information to identify contacts.
c. Navigation.
d. Underwater Communication.
e. Noise Making - used for trials and exercises.
f. Bottom Search - mine hunting and hydrography.
g. Oceanography.
h. Active Intercept.
i. Torpedo Warning.
j. Self Noise Monitoring.

The prime role can be divided into two modes, ACTIVE or PASSIVE. With active sonar, sound is transmitted into the sea, and if contact is made, it will be reflected back to the transmitting platform as an echo, providing both bearing and range information. A major disadvantage is that the energy may be intercepted, thereby betraying our own position. Active Intercept sonar is currently only deployed by submarines, however surface ship torpedo defence systems utilise active interceptions, and are an emerging technology. With passive sonar we are purely listening to sound being produced by another party, thus remaining covert. However, only bearing information is provided.

1.2. PROPERTIES OF SOUND

When we hear a sound it means that a moving or vibrating body has transmitted a vibration. The ear combined with the brain has great power of classifying sounds and recognizing the source. The sound source may be the vibrating cone of a loudspeaker, the vibrating string of a violin or the vibrating column of air in an organ pipe but in all cases the origin or sound is a moving source. Sometimes the vibrations are so slow that they can be seen but usually they are so small and fast the individual vibrations are indistinguishable.

For sound to travel from source to receiver there must be a medium to carry the vibrations. On the moon, normal speaking is impossible due to the absence of an atmosphere. If the spacesuits however are connected together by some suitable medium, direct communication can take place. Electromagnetic waves (light, radio waves and radiated heat) are unlike sound waves in this respect as they can be transmitted through empty space. A particular sound can be uniquely defined by its:

a. Frequency - number of vibrations per second.

b. Amplitude - "loudness".

c. Tone - presence of harmonics or multiples of the basic frequency.

The ear is sensitive to sounds in the approximate range 30 Hz to 15,000Hz, this being the Sonic Region. Vibrations above 15 kHz are in the Ultrasonic Region. Sonar uses sound waves in both the sonic and the ultrasonic regions.

1.3. ACOUSTIC FREQUENCY BANDS

The Acoustic Frequency Bands have been defined as follows:

Table 2.1 Acoustic Frequency Bands

Band	Frequency
ULF	0.1 Hz - 1 Hz
ELF	1 Hz - 10 Hz
VLF	10 Hz - 100 Hz
LF	100 Hz - 1 kHz
MF	1 kHz - 10 kHz
HF	10 kHz - 100 kHz
VHF	100 kHz - 1 MHz

Although this definition was established by civilian bodies it has been adopted by the military as a standard. When using frequency bands in a whole ship context care should be taken to avoid confusion with EM radiation and communication frequency bands. The suffix (A), for acoustic, is therefore used e.g. VLF(A) transmissions refer to Very Low Frequency sonar transmissions and not VLF radio communications

1.4. SPEED OF SOUND

Sound travels from its source at a speed which is dependant on the medium through which it is traveling. For example, in air the speed is approximately 330 m/s at normal temperatures, and less at lower temperatures. In water, sound travels at approximately 1500 m/s but this speed is affected by changes in **temperature**, **salinity** and **pressure**. Light waves and radio waves travel at 3×10^8 m/s (186,000 miles/sec) and this comparative slowness of sound is one reason why sonar problems can be very different to those of radar.

1.5. TRANSMISSION OF SOUND

Sound requires a medium through which to travel so let us examine how the medium behaves whilst the sound is being transmitted. Suppose a plate is vibrating in water at some frequency. As the plate moves to the right, the adjacent layers of water on the right are compressed, ie the pressure increases. This high pressure water then starts to compress the next layers of water. Similarly as the plate moves to the left, the water on the right is rarefied (its pressure is reduced below its static pressure) and this starts to rarefy the adjacent layer.

Consequently a series of pressure and rarefaction waves move away from the plate at some speed, ie the 'Speed of Sound'. If, as is usual, the plate is vibrating sinusoidally, the pressure will vary sinusoidally, i.e. if a graph is drawn of pressure against distance from source for some fixed time, the result is sine wave. Also, if a graph is drawn for pressure against time at some constant distance from the source, once again a sine wave appears.

This can be expressed by the formula:

$$P_d = P_{max} \sin(\omega t + \phi)$$

Where P_d is the pressure at distance d from the source.

$\omega = 2\pi f$ (f = frequency of vibration), t = time.

ϕ represents the phase change which occurs when d is altered, measured in radians.

The distance between the pressure peaks (or troughs) is called the wavelength of the sound, this will depend on the frequency. Wavelength can be expressed in the following equation:

λ = speed of sound / frequency = c / f

If ϕ is increased by 2π radians (i.e. 360°) which occurs when d is increased by λ, the value of $\sin(\omega t + \phi)$ is unchanged. This means the pressure is unchanged. If it is assumed that when d = 0, $\phi = 0$, then d/λ is the fraction of 2π that the phase has changed for a point distance d away compared with a point at source. Therefore:

$\phi = 2\pi d/\lambda$

The formula $P_d = P_{max} \sin(\omega t + 2\pi d/\lambda)$ shows pressure varies sinusoidally with time and distance. Pressures at points of distance apart are the same simultaneously (i.e. in phase).The pressure associated with a sound wave is one way of describing its 'strength' and can be compared to voltage in electricity since both are a measure of 'drive'. The unit of pressure most commonly used is the micro Pascal µPa.:

$1\mu Pa = 10^{-5}$dyne/cm = 10^{5} bar

1.6. PARTICLE VELOCITY

Instead of using pressure to measure sound strength, the amount of movement that the layers of the medium undergo as they vibrate backwards and forwards can be used. The velocity of the particles in the medium gives a measure of this movement. Particle velocity is not generally used but it helps to complete the electrical analogy as it is a movement caused by pressure changes. This is equivalent to current in an electrical circuit. The units are cm / sec.

1.7. INTENSITY

Having found an equivalent to electrical voltage and current, it would seem reasonable to have a measure of acoustic power. Since a sound medium such as an ocean is three-dimensional, the power must be measured as it passes through some specified area and the name for power per unit area is 'Intensity'.

In electrics: Voltage x Current = Power

In acoustics: Pressure x Particle Velocity = Intensity

The lowest intensity in air the human ear can detect is about 10^{-16} watts / cm while the intensity near to the face of a sonar transducer can be 1 watt / cm. Intensity is the most usual way of describing the amount of sound present in the sea but pressure could just as easily be used.

1.8. ACOUSTIC IMPEDANCE

In the analogy with electricity, an equivalent has been found for voltage, current and power. There remains only one basic parameter; namely resistance or more generally impedance to be compared. In general impedance is force / motion:

In electrics: Voltage / Current = Impedance

In acoustics: Pressure / Particle Velocity = Acoustic Impedance.

When applied to a medium such as air or water the impedance is called the 'Specific Acoustic Impedance' and is **the pressure required to produce a particle velocity of 1 cm / sec**. To appreciate what this means, imagine holding a large metal sheet in air and pushing it backwards and forwards as fast as possible. The maximum speed will certainly be greater than 1 cm / sec and the force required will be fairly small, ie a small specific acoustic impedance. But try and do this in water and the force required to get a reasonable movement will be extremely large; in fact canoe paddles and rowing oars rely on this principle. This indicates that water has a high specific acoustic impedance. The units used are Acoustic Ohms when pressure is in bars and particle velocity is in cm / sec.

1.9. SPECIFIC ACOUSTIC IMPEDANCE

There is another expression for specific acoustic impedance which Newton derived mathematically from the basic wave propagation equations:

Specific Acoustic Impedance = ρc

where ρ is the density of the medium, and c is the velocity of sound in the medium.

The specific acoustic impedance obtained by ρc and pressure / particle velocity are precisely the same. Although ρc is usually used for acoustic impedance, pressure /

particle velocity has been given here to show that impedance in acoustics has a similar meaning to electrical impedance.

The following table details the specific acoustic impedances of various materials. The values demonstrate the direct relationship between density and impedance.

Table 2.2 Specific Acoustic Impedance

Medium	Specific Acoustic Impedance
Steel	47,000,000 Ac Ω
Nickel	43,500,000 Ac Ω
Quartz	16,500,000 Ac Ω
Rubber	1,550,000 Ac Ω
Sea Water	1,540,000 Ac Ω
Air	420 Ac Ω

The intensities of sounds encountered in underwater acoustics can vary by several orders of magnitude (watts / cm to 10^{-9} watts / cm). Such a large range of numbers makes calculations unwieldy and consequently a logarithmic or decibel notation is universally used.

It so happens that our ears measure intensities on a logarithmic scale quite naturally; for if we hear a sound that is increasing in strength in a number of steps of apparently equal amounts, it is found that the intensities are multiplied by a constant amount to give each step.

The decibel notation gives a method of showing a ratio of intensities. dBs are only defined for power ratios (or intensity ratios) and dBs for voltage or pressure ratios are derived from the power definition.

If we require to express an intensity I in dB, then we must first define a reference intensity I_{ref} to give the comparison or ratio. For sonar the reference intensity is that related to an acoustic pressure of 1 μPa in seawater. Therefore the intensity I in dB is 10 $\log_{10} I/ I_{ref.}$ Originally the bel was defined as $\log_{10} I/ I_{ref}$ but these units were found to be too small numerically so the decibel was born.

SONAR (Sound Navigation and Ranging), has numerous applications in industries, medical and intricate operations performed in Navy. Almost all over the world in Navies, the primary role of sonar is to detect and track enemy targets [1]. However, sonar has got certain limitations which are usually governed by the propagation pattern of underwater sound wave and the temperature variations in the water with respect to its depth. These thermal changes results in the formation of a layer in which the sound energy travels and detects the targets.

The changes in the acoustic wave pattern due to temperature and velocity gradients can be overcome through various techniques [2]. The most common and widely used technique is to change the length of sensor elements of the sonar, thus makes the sonar able to receive the signal from various layers. Such types of sonars are usually called variable depth sonar (VDS) and towed array (TA) sonars. The primary aim of these sonars is to overcome the above mentioned limitations. Thus the ability to change depth or length of the sensor elements provides the sonar to detect and track targets below the layer.

1.10. Motivations and background

SONAR (Sound Navigation and Ranging), is primarily used in navy but also has wide range of applications in industry, medical and robotics. In navy it is used to detect, navigate and track targets [1]. Normally these targets are the hostile targets. The propagation pattern of underwater sound wave and the temperature variations in the water with respect to its depth impose limitations on the working of SONAR. These thermal changes result in the formation of a layer in which the sound energy travels and detects the targets.

Changes in the acoustic wave pattern due to temperature and velocity gradients can be overcome through various techniques [2]. The most common and widely used technique is to change the length of sensor elements of the SONAR in order to make the SONAR to receive the signal from various layers. They are usually called Variable Depth SONAR (VDS) and Towed Array SONAR S(TAS). Change in depth or length of the sensor elements enhances the ability of SONAR to detect and track targets below the layer. Different types of platforms are used in naval equipment such as submarines, aircraft and surface ships. Efficiency and performance of sensors fitted on these platforms lead to successful operation. Some special type of platforms like submarines and midgets heavily depend on the performance of SONAR. SONAR not only serves as an eye for the submarines but also provides data for target acquisition, tracking, self-noise monitoring and bathy (temperature, salinity with respect to depth conditions [3].

Classical adaptive beam forming is being used for source signal estimation in field of signal processing [3]. These algorithms are sensitive to localization errors which limit the reliable signal estimation unless better adaptive filter is implemented. Later on two algorithms were developed to solve this problem [4,5]. But these algorithms have another problem which is signal distortion and does not suppress the noise effectively. The algorithm discussed by Affes, et.al [6] is very useful for adaptive beam generation but it is too complex as it is based upon Maximum Likelihood Detection (MLD). The adaptive beam forming is obtained by applying some adaptive filter [7]. The most generally used adaptive algorithms are Least Mean Square (LMS), Recursive Least Square (RLS), Square Root RLS, Fast RLS and Kalman Filtering algorithm[8].The selection for the most appropriate algorithm depends on the efficiency measuring parameters. These parameters are based on robustness, rate of convergence, maladjustments, tracking, computational requirements, structure and the numerical properties [9]. The adaptive algorithms without depending on its type have some negative as well as positive points. Since it is not possible to have an ideal adaptive filter therefore, a trade off has to be made [10].The adaptive algorithm can be complex or real depending on the nature of the input/output data and tap weights. This research work has proposed a model that is based upon Steepest Decent Algorithm (SDA) that is useful in stationary environment. SDA has been applied with the classic beam forming algorithm to get the effective results.

Any degradation in the performance of SONAR at any instant during the operation at sea will jeopardize the safety of the submarine. To ensure its optimum performance, beam formation plays an important role in SONAR [11, 12]. Different number of elements are normally used at the sensor output to maintain low side lobe levels [13,14]. When some of these elements get defective or inoperative over time, it impairs the beam patterns, directivity and various other parameters of SONAR which results in non-optimal performance.

If all the elements are working then SONAR will have its optimum performance and the beam pattern produce will be similar to the one perceived at the design time. However, if the elements get defective then the beam pattern will be changed and will deteriorate the performance of the SONAR. The aim here is to calculate these degradations/errors and apply correction iteratively on the received signal at the output of a beam former. This will help the SONAR to maintain an optimum performance at all the time without getting any input from the user.

The adaptive algorithm for the compensation of a degraded beam can be implemented in the adaptive beam forming portion as shown in the Figure 1 [15]. This portion is located after the beam former in a general type of a SONAR system [16]. The portion, Array of Sensors can have the sensors in various configurations [17,18]. These configurations could be a linear, planer, cylindrical, conformal and spherical having its own advantages and disadvantages and can be used depending upon the application of a SONAR [19]. TAS applies the concept of line array because of its ability to change operating depth and efficiently detects below layer targets.TAS is very commonly used now a days.

To ensure optimum performance of SONAR, beam formation plays an important role in SONAR. Different number of elements is normally used at the sensor output to maintain low side lobe levels. When some of these elements get defective or inoperative over time, it impairs the beam patterns, directivity and various other parameters of SONAR thus results, in non-optimal performance.

1.11. Thesis outline

This thesis consists of five chapters. Chapter 1 gives the introduction and problem definition with thesis layout. The important properties of sound affecting the underwater propagation, basic fundamental of sonar along with the development of active and passive sonar equations are discussed in Chapter 2. Chapter 3 discusses the fundamental of beam formation and election of suitable adaptive algorithm on the bases of their trade off and its implementation as an adaptive compensator for the optimization of beam formation in sonars along with basic calculations and simulation. Chapter 4 discusses the system model, sonar model results and analyses the results to determine the efficiency of the adaptive algorithm that how close it optimizes the sonar beam in case of defective elements. The thesis summary and suggested future research directions are given in the chapter 5.

1.12. Thesis Contributions

The contributions of this thesis include:

I. Generation of beam pattern using adaptive algorithm, when the body is moving and all sensors are working.
II. Calculation of the defective beam pattern when number of sensors is not operational in array of sensors.
III. Compensation of beam patterns which are degraded due to defective sensors using adaptive algorithms.
IV. Simulation and comparisons of results to show the accuracy of this technique.

Chapter 2

2. Sonar

2.1. Introduction

Fundamentals and background knowledge required to understand operation of sonar are discussed in this chapter. Definition of sonar, its primary and secondary roles, properties of sound, acoustic frequency bands, speed and transmission of sound are reviewed. A short note on sound path prediction, deep sea temperature profile and active/passive sonar equation is provided.

2.2. Sonar Basics

SONAR which is an abbreviation of Sound Navigation and Ranging plays an important role in case of submarines and midgets because when these platforms are submerged nothing can be seen visually and has to depend totally on this equipment. There are two types of sonars, active sonar and passive sonars. As a broader view sonar has two roles to play i.e. primary role and the secondary role [3]. The prime role for both the types of sonars is to detect, classify and to track a contact, weather it is hostile or friendly. However, the secondary roles include navigation, underwater communication, self noise monitoring, bottom search, noise making, active intercept and torpedo warning [4]. The active sonars are mainly used by surface ships for searching the submerged platforms. Like in radars the EM wave is transmitted and the received reflection are then processed and analyzed for a potential target [5]. Similarly, in case of sonar instead of EM wave sound wave is transmitted in the water and the reflected signal/echo is analyzed for the potential under water target with its range and bearing. However, in case of submarines active sonars are rarely used as it gives away the position of the transmitting platform and submarine looses its element of surprise. Therefore, in order to keep the element of surprise the submarines almost all over the world use passive sonar during covert operations and continue listening to the sounds which are being produced by other parties. In this way, the submarine gets the information regarding range and bearing of other platforms at sea without revealing its own position.

2.3. Design Issues

The longitudinal or compression waves in the range of 2 to 20 KHz comprises of sound waves, which are audible to human beings [3]. These waves are generated usually by some vibrating source. The detection of these waves depends upon the frequency with which the source is moving and the medium in which these waves are traveling [2]. The frequency will make the sound more conspicuous whereas, the speed of sound will govern the factor that how quickly the signal is picked. The frequency bands which are normally used in case of underwater sound propagation are given in table 2.1.

The speed of sound waves depends on the compressibility and inertia of the medium through which they are traveling. Therefore, the speed of sound is much higher in solids/liquids then in gases. Therefore, the speed of sound in water is much higher then in air. When the sound wave propagates through water it will be subjected to various losses. These can be split into following categories:

Table 2.1 Acoustic Frequency Bands

BAND	FREQUENCY
ULF	0.1 Hz - 1 Hz
ELF	1 Hz - 10 Hz
VLF	10 Hz - 100 Hz
LF	100 Hz - 1 kHz
MF	1 kHz - 10 kHz
HF	10 kHz - 100 kHz
VHF	100 kHz - 1 MHz

2.4. Spreading

A sound wave radiating in an infinite body of water will suffer spreading losses of intensity proportional to the inverse square of range [3]:

Intensity $\propto$ 1/R2 (NB for an echo, Intensity $\propto$ 1/R4) (2.1)

This is known as 'free-field', or spherical spreading, and is generally the case for mid-ocean, deep water transmissions. If the body of water has containing boundaries (e.g. shallow water or a surface duct) the sound intensity will suffer spreading losses known as cylindrical spreading proportional to the inverse of range:

Intensity $\propto$ 1/R (2.2)

Spreading may follow various laws, but the above types are most typical. A special kind of spreading occurs when the signal from a pulsed source is spread out in time. The pulse becomes elongated by multi-path propagation effects and is 'smeared' in time as range increases.

2.5. Absorption

Unlike the spreading effect on sound intensity, absorption involves a process of conversion of acoustic energy into heat, and thus represents a true loss of energy to the medium [3]. Absorption of sound in seawater is about 30 times greater than that in pure water in the frequency range 5-50 kHz. The increase is attributed to a chemical reaction involving one of the minor dissolved salts in the sea (MgSO4 - effect dominant below 100 kHz). Absorption varies with frequency, and has a degree of temperature dependence as shown in figure 2.1. The fact that higher frequencies are severely attenuated is of great significance in the design and use of military sonars.

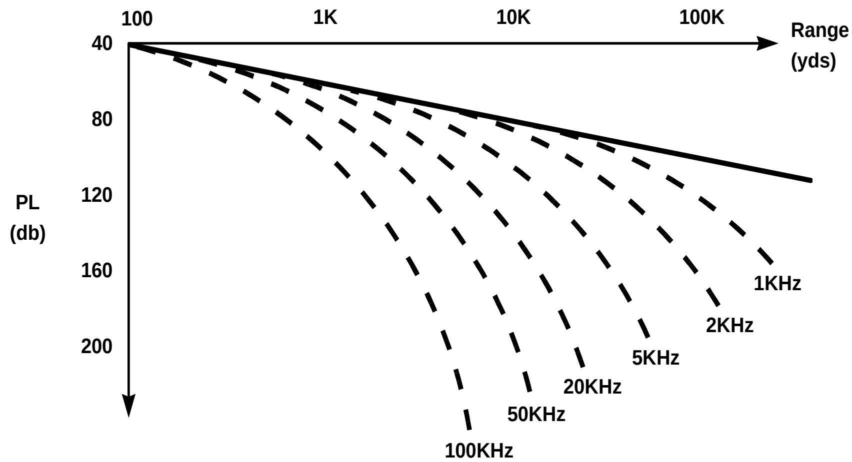


Figure 2.1 Propagation Range vs. Range (w.r.t. frequency)

2.6. Propagation Loss

When propagation measurements are made at sea, it is found that spherical spreading plus absorption provides a reasonable fit to measured data under a wide variety of conditions. Thus, it is possible to provide a handy 'working rule' for estimation of propagation loss (in dBs) for a given range, R:

$$PL = 20 \log10 R + \lambda R \quad (2.3)$$

2.7. Scattering and Reflection

If the transmitted sound energy strikes any underwater object, be it a bubble or an empty can, it will be reflected and scattered. The larger the object, compared to the wavelength, the more sound will be reflected/scattered. A particularly significant source of such energy loss is caused by fish and sea life, such as plankton. In many areas, much of this marine life exists in what is known as the Deep Scattering Layer. The depth of this layer varies between 100 and 1000 meters.

2.8. Sea-Bed and Surface Effects

The energy in a sound wave will be further dissipated when transmissions encounter the seabed or surface. The exact amount of reflection/absorption will depend upon the nature of the interface. With a high sea-state (>4) not only will the increased 'roughness' of the surface cause high reflection losses, but also the bubbles that are formed are usually resonant at typical sonar frequencies and consequently cause significant energy losses near the surface[4]. At the seabed, reflection and absorption tend to complement each other; i.e. rock - high scatter, little absorption, mud - low scatter, high absorption. However, in some areas, the nature of the seabed is such that the 'bottom bounce' technique may be used to good effect.

2.9. Velocity of Sound in the Sea

The velocity of sound in seawater is an oceanographic variable that determines many of the peculiarities of sound transmission in the medium. It varies with depth, the seasons, geographical location and time at a fixed location. Since 1827, experimenters have attempted to measure sound velocity in a natural body of water and also to relate measured values to the parameters of temperature, depth and salinity. Typically [3]

$$C = 1{,}449.2 + 8.623T - 0.0546T + 1.391\,(S - 35) \qquad (2.4)$$

where:

C - Velocity in meters / sec (at zero depth, or atmospheric pressure)

T - Temperature in °C

S - Salinity in parts per thousand

From the above equation, the relative effects of changes in temperature pressure and salinity can be determined by differentiation, yielding the coefficients:

Temp Coefficient $\Delta C/\Delta T = +3.6$ m/s / °C

Salinity Coefficient $\Delta C/\Delta S = +1.4$ m/s / %o

Pressure Coefficient (depth) $\Delta C/\Delta D = +1.7$ m/s / 100m

2.10. Sound Path Prediction

Sound paths in the ocean can be very complex and difficult to predict. The environment affects the passage of sound in two major ways, both of which are significant. Firstly, the sound rays bend or refract as they pass through the medium, and secondly the energy is dissipated (and attenuated) thus weakening the strength of the signal. However, at times sound can become trapped in ducts allowing long range detections to occur. It is possible to mathematically describe the propagation of sound in an elastic medium such as sea water by the use of Ray Theory [1]. This enables representation of propagation by the use of ray diagrams obtained from Snell's Law which states that in a medium consisting of layers of constant velocity, a ray will be refracted between layers by an amount proportional to the layer velocities.

$\cos\theta 1 / C1 = \cos\theta 2 / C2 = \cos\theta 3 / C3$ = constant for any one ray. (2.5)

Sound is refracted toward a region of lower velocity.

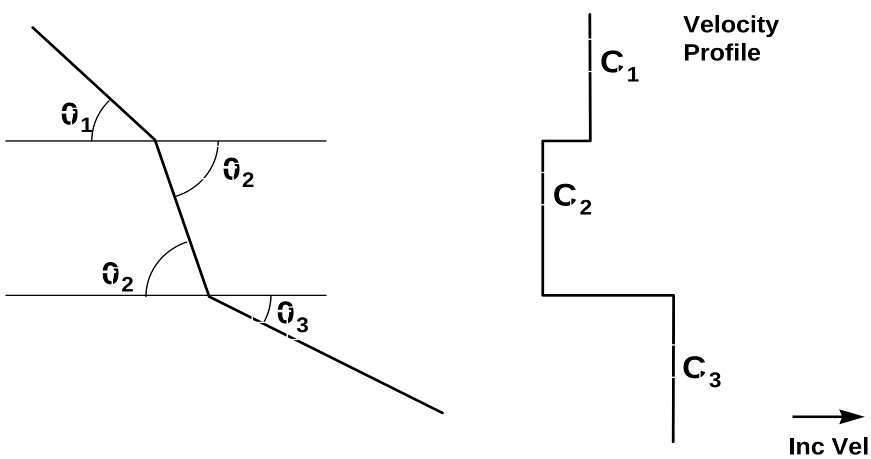


Figure 2.2 Snell's Law

2.11. Velocity Gradients

In practice clearly defined boundaries between regions of differing velocity, as discussed above, seldom occur [3]. If successive layers become thinner and thinner then we approach the situation of a velocity gradient and it can be shown that the ray path approaches the arc of a circle. It should be apparent that in a positive velocity gradient (velocity increasing with depth) rays are curved upwards and vice-versa. This is illustrated in the figure 2.3

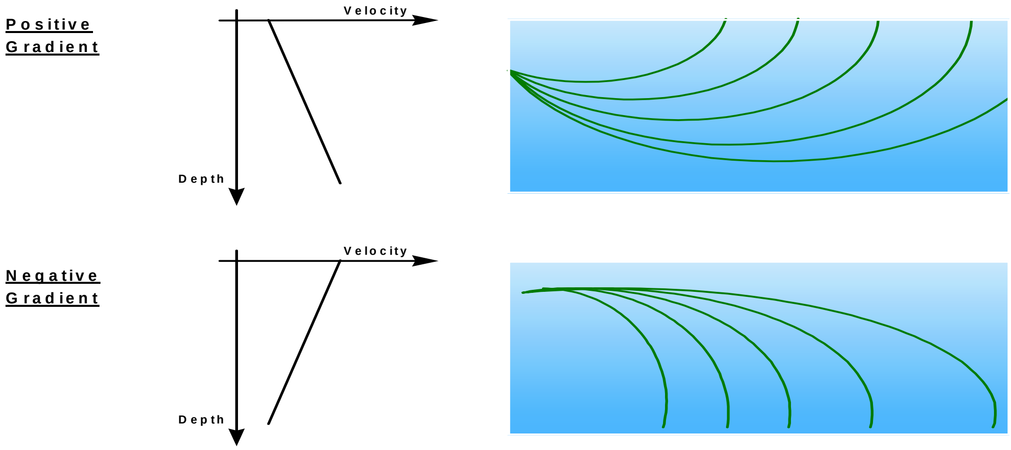


Figure 2.3 Velocity Gradients

2.12. Typical Deep-Sea Temperature Profile

Before looking at some of the typical propagation paths encountered when operating sonars, it is of use to look at a typical temperature profile, for the deep sea case. In general terms the temperature of seawater decreases from the surface, and practically all the changes occur in the first 1000 m, the most marked being in the first 300m [3]. A typical profile is shown in the figure 2.8. The region is discernible into four different layers and may merge together in certain circumstances [1, 3].

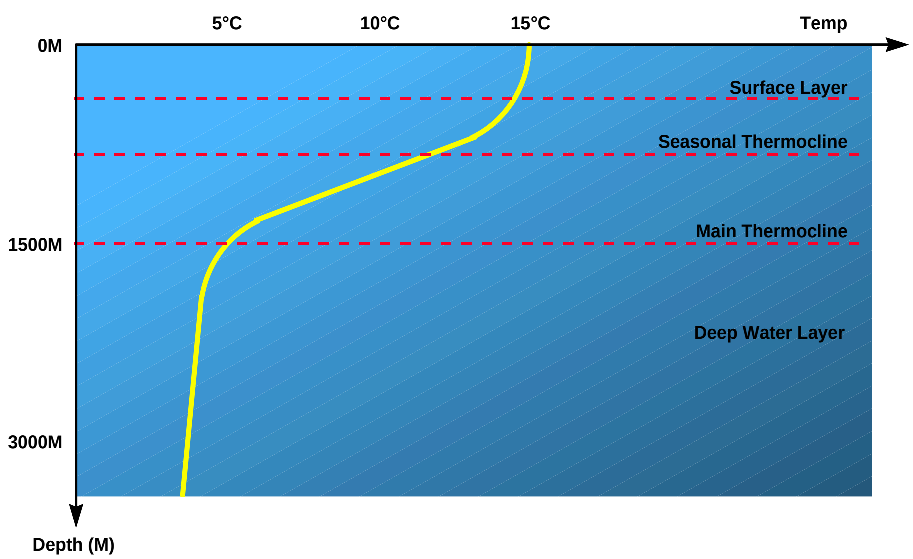


Figure 2.4 Thermal Structure of the Ocean

2.13. Surface Layer

Here the water temperature is strongly influenced by atmospheric conditions and generally appears as a mixed, isothermal layer as a consequence of surface agitation. A strong negative gradient may occur if surface heating is strong (e.g. calm, hot afternoon) or a positive gradient may occur in the case of night time heat loss due to radiation. The depth of this layer may vary from a few meters to as much as hundreds of meters.

2.14. Seasonal Thermo cline

In addition to diurnal temperature variations, seasonal variations in temperature, and hence sound velocity, also occur. During the summer the ambient surface temperature is high, and thus the surface layer sits above a sharp negative thermo cline. In winter conditions however, the colder surface temperatures may be such that there is very little evidence of the thermo cline. The nature of the seasonal thermo cline will also depend upon the latitude.

2.15. Main Thermo cline

In this layer, the main temperature drop occurs, and the gradient is generally independent of surface, or seasonal effects.

2.16. Deep Water Layer

At depths below ~ 1500 m, the water temperature is effectively constant at approximately 2 - 3°C.

2.17. Common Sound Propagation Paths

The number of possible velocity/depth profiles that may be encountered is, of course, infinite, but generally speaking, it is possible to identify several distinct modes of propagation for sound waves, and the nature and significance of these will be outlined as under [3].

2.18. Surface Duct

In the Surface Layer, where temperature is constant, the consequent positive velocity gradient (velocity increasing with depth) causes upward refraction. Sound is trapped in the layer. The amount of sound trapped will be proportional to the depth of the layer, the gradient and the angle of transmission. Sound energy may propagate to long ranges by repeated reflections from the sea surface, but losses do occur, particularly if the sea surface is disturbed and is a poor reflector. Note that rays leaving the source at an angle greater than the critical angle, will actually penetrate the layer, and then be refracted downward by the negative gradient. This splitting of rays gives rise to a shadow zone as shown in figure 2.5. This shadow zone cannot be insonified, regardless of the transmission angle. The onset of the shadow zone will be determined by the duct depth, and may be such as to provide a distinct advantage to an attacking submarine.

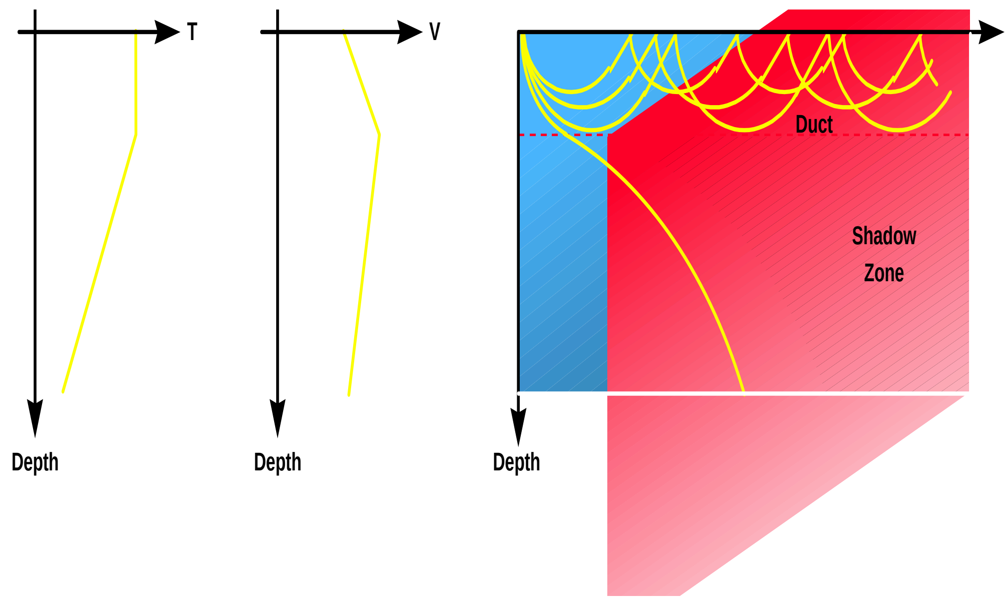


Figure 2.5 Surface Duct

2.19. Convergence Zone

Sound that has been refracted out of the surface duct will eventually encounter the positive velocity gradient associated with the deep water isothermal layer, and thus be bent toward the surface, to re-emerge at the surface some 20-30 kms away. If surface conditions are good then this process may be repeated several times. Convergence zones as shown in figure 2.6, where an increase of 10-15 dB of sound level is experienced, may be anything from 25 kms (Mediterranean) to 50 kms (Atlantic) and are generally circular in shape from 5-10 kms in diameter. Although detection of targets by such a propagation path does not reveal particularly accurate information, it is a very useful means of providing initial detection, if the circumstances are suitable

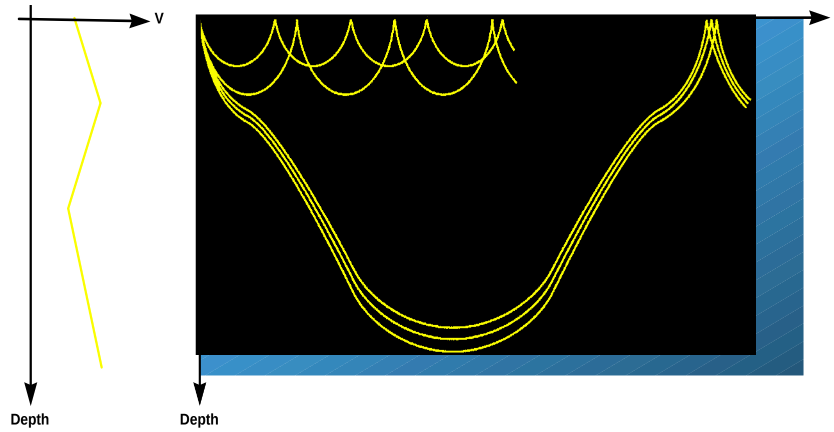


Figure 2.6 Convergence Zone

2.20. Deep Sound Channel

In the deep ocean, a velocity minimum is encountered. For any sound source transmitting in the vicinity of this velocity medium, the effect of the positive velocity gradient and negative velocity gradient will be, to cause the sound energy to be refracted up/down and effectively focus the energy along the so-called channel-axis as shown in figure 2.7. As no reflections are involved, very long propagation ranges are possible, particularly if the source lies exactly on the point at which the velocity minimum occurs. The actual depth at which the velocity minimum occurs will vary with the local conditions and figure 2.8 shows a typical variation of depth of sound channel with latitude.

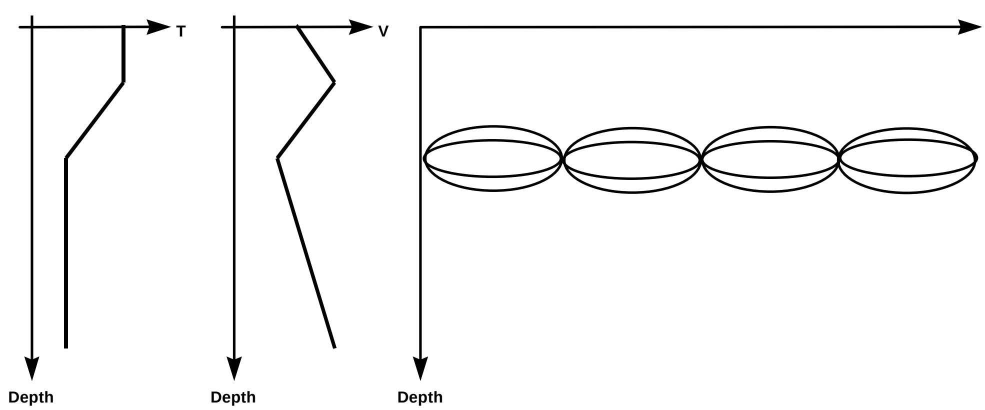


Figure 2.7 Deep Sound Channel

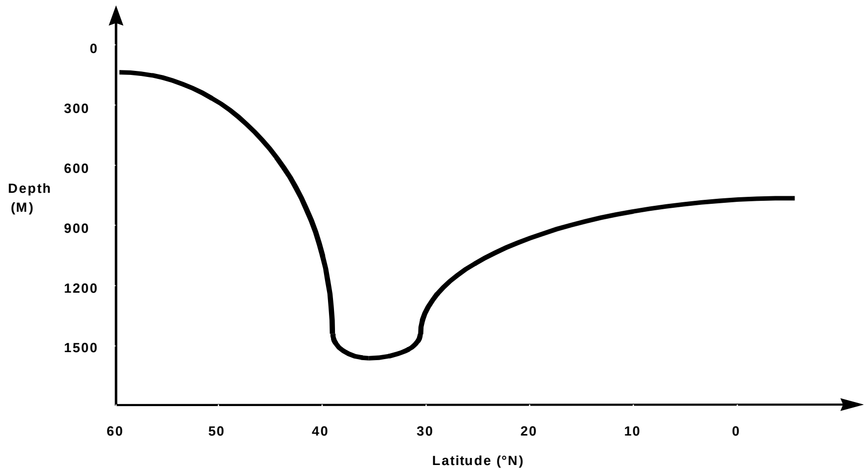


Figure 2.8 Deep Sound Channel Depth Variations

2.21. Bottom Bounce

The three propagation modes discussed above result in shadow zones, where it is virtually impossible for sound energy at normal transmitted frequencies to penetrate. A possible means of obtaining insonification of these areas is to bounce the sonar beam off the seabed as shown in figure 2.9. This of course depends upon the availability of a highly reflective bottom, but given good conditions, reasonable success can be achieved. With angles of depression >20° the effects of refraction are considerably reduced, and can be disregarded. However, for maximum reflectivity, grazing angles near to the horizontal are preferred. Consequently a compromise solution has to be reached. The technique has been employed on trials with some success.

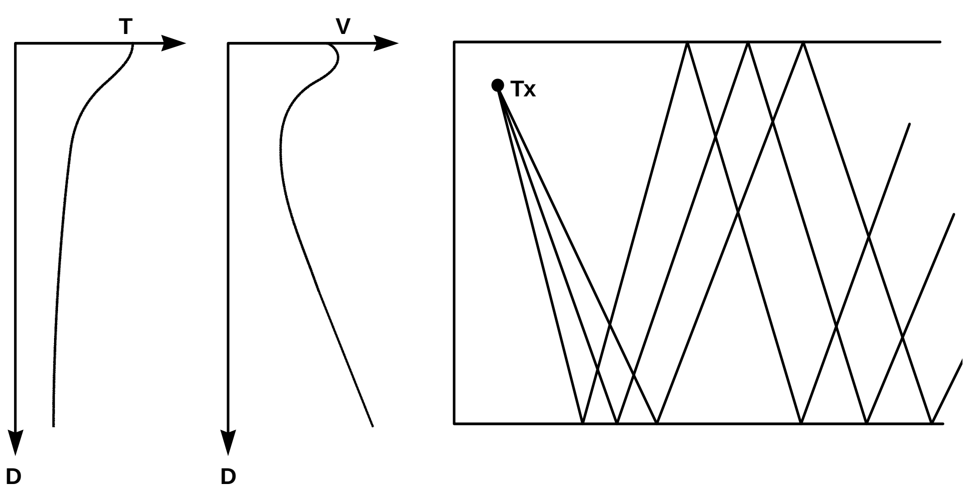


Figure 2.9 Bottom Bounce

2.22. Afternoon Effect

This is an effect of considerable practical importance, especially in Mediterranean and Persian Gulf areas. Surface heating reaches a maximum in the afternoon and causes a very strong negative thermo cline as shown in figure 2.10. Consequently, the sonar transmissions are severely refracted downward with an associated dramatic range reduction. This effect is extremely important in the military environment.

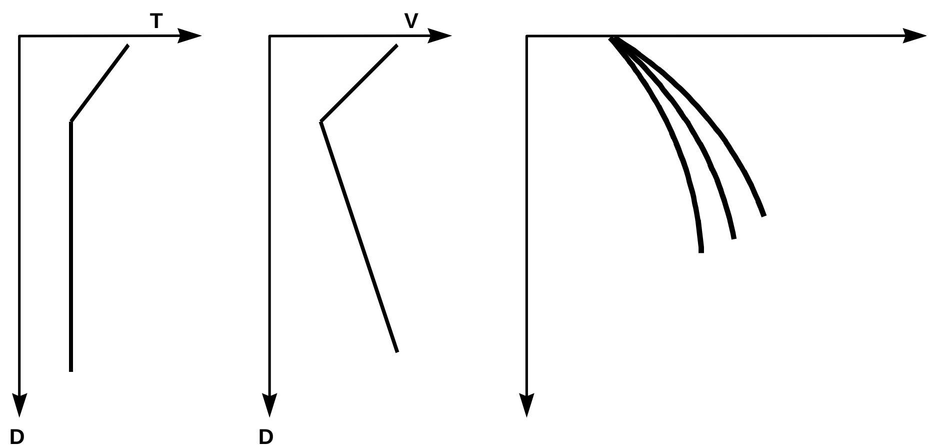


Figure 2.10 Afternoon Effect

2.23. Sonar Equation

There are numerous factors that control the behavior of underwater sound propagation; which when considered altogether is known as sonar parameters. These parameters are inter-related to each other and the expression which gives the relation of these parameters is known as sonar equation [4]. The sonar equation mainly deals with the nature and effects of medium, target and the equipment itself. The main purpose of underwater acoustics is to detect the wanted signal i.e. target from that of the background noises caused due to unwanted energy reflection, ambient noise and the self noise of the platform. The target detection can only take place when the signal to noise ration is greater then the threshold level. The ability of the sonar to detect the target depends on the characteristics of the sea water, type of target and the noise generated by the sonar itself. In order to monitor the performance of sonar, these parameter are noted and related with the help of a sonar equation. Depending on the nature of the sonar equipment the equations can be of two types i.e. Active and Passive sonar equations [2, 3].

2.23.1. Development Of Active Sonar Equation

The performance and efficiency of active sonar depends on various factors. Few of these factors are common and will be used in the development of passive sonar equation as well. The development of active sonar equation requires a brief introduction of few important factors that plays an important role for both active and passive sonars[3].

2.23.1.1. Background Noise (Bn)

Ambient noise together with self noise degrades the ability of the sonar to extract the target information from the received signal.

2.23.1.2. Directivity Index (Di)

Noises and signals arrive at the receiver from all the directions. In case of directional sonars the beam former cancels out the signal which arrives from out side the beam of interest. This amount of signal cancellation is the measure of the directional property of the transducer and is known as the Directivity Index.

2.23.1.3. Target Strength (Ts)

The target strength may be defined as the difference in reflecting ability between the actual target and the reference target (Reference target is the target having target strength of 0 dB from a perfect reflecting sphere of 1m radius). Therefore, the target strength is the echo level of the signal that is reflected back towards the source and it depends on the target size, shape, aspect and construction of the target.

2.23.1.4. Propagation Loss (Pl)

The attenuation in the level of a sound signal once measured at a distance of 1m from the sensor. It depends on the expansion of the wave front, scattering and absorption of acoustic energy in the medium.

2.23.1.5. Source Level (Sl)

Source Level is a measure of the intensity of the sound energy emitted by the transducer. In the case of active sonar the transducer introduces the ping or sound signal into the water. The effective source level of this signal is the signal that exists one metre from the transducer face. Values of active SL vary from about +215 dBs to over 240 dBs dependent on the power capability of the sonar transmitter, as well as type, size and directivity of the transducer array.

2.23.1.6. Detection Threshold (Dt)

Detection threshold may be defined as the minimum signal to noise ratio required at the receiver in order to have 50% probability of detection and a 0.01% probability of false alarm.

2.23.1.7. Bandwidth Correction Factor (BW)

In case of active SONAR the band width of the receiver is always set greater to that of the bandwidth of its transmitter. Therefore, the bandwidth correction factor must be applied. The active sonar equation can now be developed by considering the above mentioned factors and can be given as:

The energy returned to the receiver after traveling the distance twice will be

$$\text{Signal} = SL - 2PL + TS \qquad (2.6)$$

The noise of the receiver will be

$$\text{Noise} = BN - DI - BW \qquad (2.7)$$

So SNR at the transducer will be given as

$$SNR = (SL - 2PL + TS) - (BN - DI + BW) \qquad (2.8)$$

2.23.2. Passive Sonar Equation

In case of passive sonar, most of the factors are same as that of active sonar. The factor which is additional is the target noises (TN) which may be define as the measure of the level of acoustic noise at a distance of 1m from the acoustic centre of the target [2].

The signal that reaches to the to the receiver is given as

$$\text{Signal} = TN - PL \qquad (2.9)$$

Noise seen at the receiver will be

$$\text{Noise} = BN - DI \qquad (2.10)$$

$$SNR = (TN - PL) - (BN - DI) \qquad (2.11)$$

2.24. Conclusion

In this chapter we have discussed the theoretical knowledge pertaining to sonars. It covers a brief introduction about the properties of sound, which is important for underwater propagation during various climatical conditions. The explanation of various terms used for the development of the active and passive sonar equation are also discussed. It gives the fundamental working of sonar which will help to understand its various impediments and their solutions discussed in the subsequent chapters.

CHAPTER 3

3. INTRODUCTION SONAR ARRAYS

3.1. INTRODUCTION

Individual radiating and receive transducer elements are generally grouped together in large arrays so that the required beam pattern can be formed. The size and shape of complete transducer arrays are governed by a number of factors:

a. Function of the set - whether narrow or wide beams are necessary.

b. Frequency of operation - lower frequencies require larger arrays so that acoustic apertures of a significant number of wavelengths can be utilised.

c. Space available - space on the hull of submarines and surface ships is often limited such that larger arrays are often deployed as towed systems.

d. Bearing accuracy - to obtain good bearing accuracy, narrow beams will be required and therefore large arrays relative to wavelength.

e. Power output - high power sonars often have large individual radiating elements. The resulting array will often be large. Therefore, each sonar role will require a different array format, some of which are

  a. Line array.

  b. Planar array.

  c. Cylindrical array.

  d. Conformal array.

  e. Spherical array.

3.2. LINE ARRAY

This is the simplest form of transducer array, which must be used for low frequency sonars where an effective acoustic aperture of hundreds of metres is often required.

The line arrays in Submarines and General Service consist of a series of transducer elements adjacent to each other at fixed intervals. This type of array can be hull mounted as in Sonar 2007, or made up into a towed array as in Sonar 2031 or 2046. A line array is highly directive when broadside to a target, but as the look-angle moves away from broadside the directivity reduces. This point is illustrated in Figure 3.1 where the effective array length varies depending on the look-angle.

As the angle off broadside increases, the effective array length becomes smaller and the resulting beams widen. The total beam pattern consists of beams about 4° wide on the beam but up to 20°-30° along the array axis 'end fire' beams in the case of a 32 array. Bearing accuracy for such a system is therefore variable depending on target relative bearing.

7. The effect on the beam pattern of a line array can be seen in Figure 2.2:

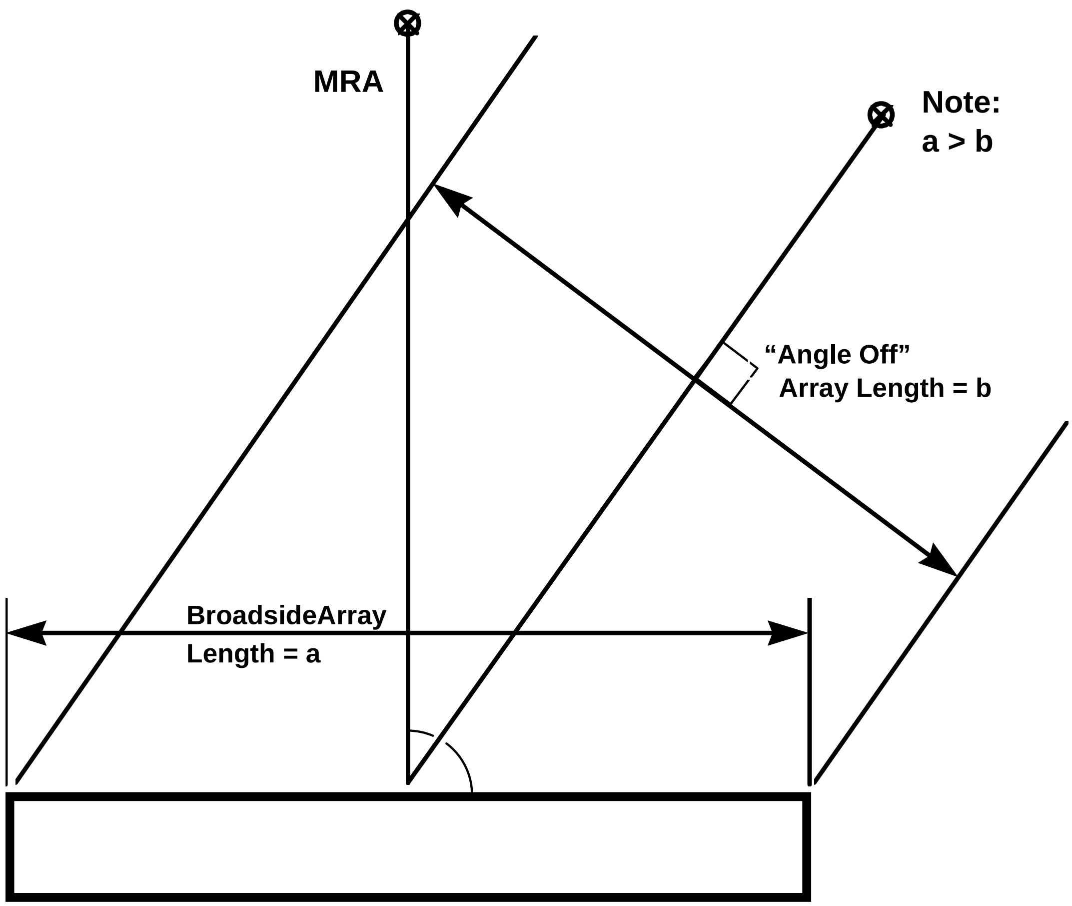


Fig 3.1 - Effective Array Aperture

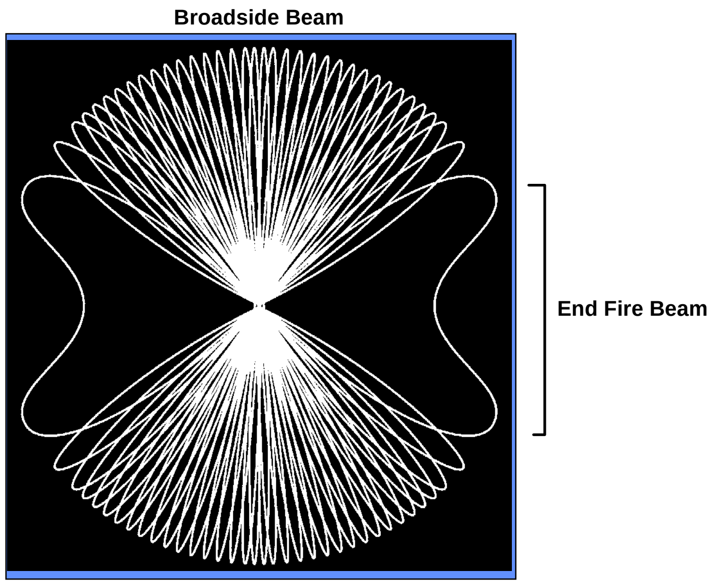


Fig 3.2- Typical Line Array Beam Pattern

Another disadvantage of a line array is inherent in its one dimensional construction. Whilst it is possible to form beams easily in the horizontal plane, vertical beamforming is not possible as the array has no 'height'. The resulting beams extend through a full 360° in the vertical plane, such that Figure 3.2 can only be regarded as a horizontal slice through the ‘donut’ beam pattern. This means that a considerable amount of spurious noise enters the system through the wide vertical beams. In addition, beams 1-32 on the starboard side in Figure 3.2 are the same as beams 1-32 on the port side. Initially, with such a system, the platform will be unable to determine on which side of the array the target lies - this is known as bearing ambiguity.

3.3. PLANAR ARRAYS

An extension of the line array, planar arrays group transducers in a series of lines on the flat face of a fixed or rotating plate. The narrow beam transducer of Sonar 2008 used for underwater communications is an example of this type of array (Figure 3.3):

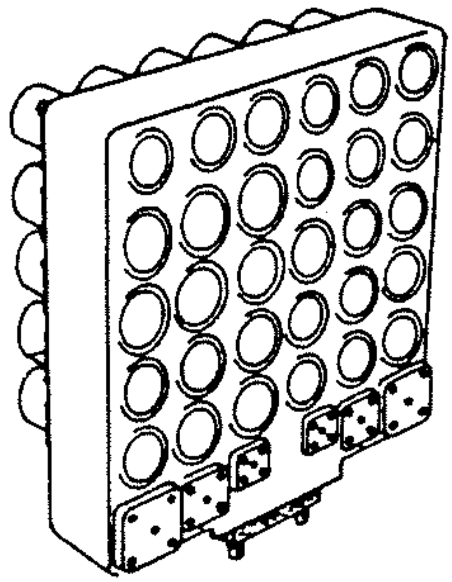


Fig 3.3 - Sonar 2008 Planar Array

The beam pattern produced by this and similar arrays is normally fairly narrow and directional. It is steered by either physically moving the transducer to the correct bearing, or electronic beam steering in the receiver and transmitter beamformers.

For minor sonar applications such as the underwater telephone where speed of transducer response is not important, physical movement of the transducer is adequate for our purposes. For search sets, which have to achieve rapid 360° coverage, the sweep rate problem becomes insurmountable given the speed of modern submarines.

Planar arrays, like line arrays, have a directivity which alters depending on the angle from broadside. Beams towards end fire become distorted and much larger. Therefore, with a planar array it is only possible to steer a beam electronically by about 60° before the beam distortion causes an unacceptable loss of bearing accuracy.A major disadvantage of a rotating transducer array is that training noise can be generated which reduces the performance of the equipment.

3.4. CYLINDRICAL ARRAYS

Cylindrical Arrays are used in most General Service search sonars which require all-round, horizontal coverage and limited vertical coverage.

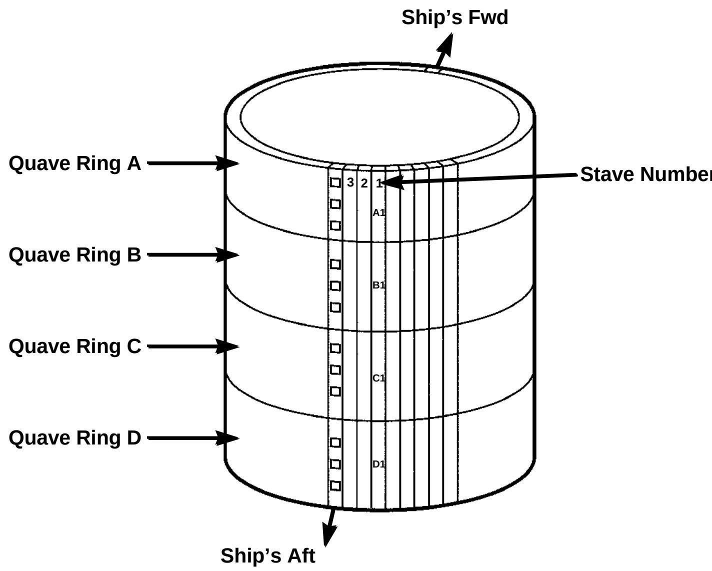


Fig 3.4 - Cylindrical Array

The Sonar 2016 array, as shown above is made up of 768 electrostrictive elements mounted in 64 vertical staves giving 12 elements per stave. Each 12 element stave is divided into 4 horizontal sections known as quaves; each quave having 3 elements wired in parallel.The major advantage of a cylindrical array is that the beam directivity no longer varies with aspect in the horizontal plane so that an undistorted set of horizontal beam patterns may be formed by treating each of the 64 staves as an individual radiating or receiving element.

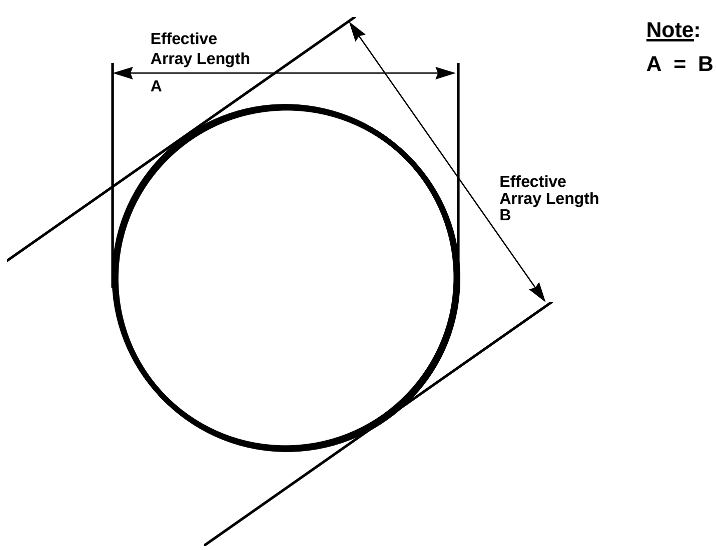


Fig 3.5 - Change of Effective Array Length

In the vertical plane the four quaves of each stave will allow a vertical beam to be formed. With only four effective radiating elements, the beam will be quite wide and distorted as it moves towards end fire in the vertical plane. The vertical directivity will be aspect dependent as the array is linear in that plane.

By phase delaying the signals to each horizontal section of 3 elements, the vertical beam can be depressed up to an angle of 7° to push energy below the surface duct to exploit a convergence zone. Most modern General Service sonars have a CZ mode in which the energy is depressed 7°.

In order to transmit sound energy through a 360° arc, the sonar transducer must protrude clear of the ships hull to give it a clear field of view. The hull outfit is the supporting equipment which holds the transducer below the hull and provides a streamlined cover over the whole assembly. In the case of retractable hull outfits it includes the lifting gear necessary to raise the transducer into the hull.

3.5. CONFORMAL ARRAY

A conformal array is one which follows the contour of the platform which is carrying it. All submarines are fitted with conformal bow arrays. In the case of 2020 / 2054, the sonar transducer array is built around and conforms to the 'bow' shape of the SSN / SSBN. The array is static and tilts backward from the vertical by 20° as shown in Fig 3.6. A GRP dome is fitted to the front of the array thereby providing still water in front of the element faces. This reduces flow noise, thus improving detection ranges. Specialist coatings are provided in front of the array to provide an acoustic cladding similar to tiling, whilst allowing two way propagation of acoustic data.

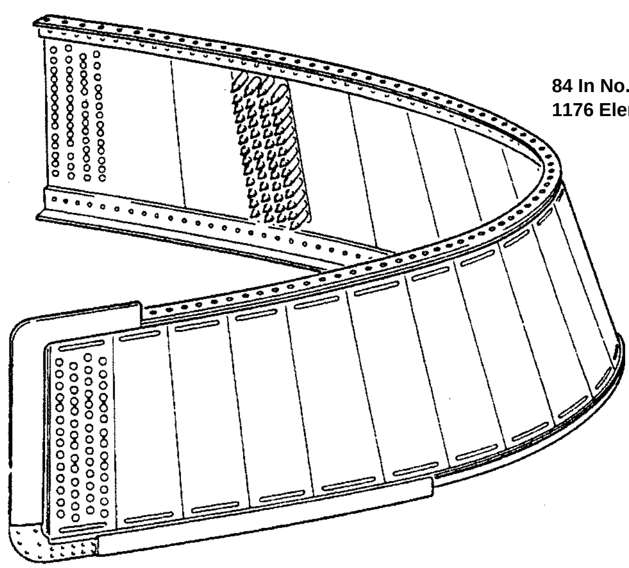


Fig 3.6 - Conformal Array

The number of elements used for beam forming is determined by the angle at which the beam is being formed and its position relative to the array axis. This type of array gives high angular resolution due to the effective array length around the bow. However, because the array is not cylindrical for all aspects, as the look angle varies so the effective array length alters, making the array's directivity variable.

3.6. SPHERICAL ARRAYS

Spherical arrays are employed when continuous coverage in both the vertical and horizontal planes is required. Apart from the array fixing point the vertical and horizontal beams formed will be undistorted.

3.7. SONAR DOMES

All hull mounted arrays in service with the PN are sighted in free flood sonar domes. The purpose of the sonar dome is to ensure that the array sits in still water, to ensure maximum detection performance. However, the manufacture of the dome itself can have major performance implications. The dome must be manufactured from a material which provides a good acoustic impedance match with sea water. If this is not the case the dome would be effectively opaque to the acoustic energy. Domes are therefore made out of special GRP materials, which provide the required impedance matching, and also require specialized paints to coat. In order to reduce external flow noise to a minimum, any sonar dome must be streamlined to ensure laminar, rather than turbulent flow across the acoustic aperture. In addition to streamlining, baffle plates can be fitted externally to domes to direct turbulent flow away from the acoustic aperture.

CHAPTER 4

4. TOWED ARRAYS

4.1. INTRODUCTION

Towed arrays are essential for the detection of modern, quiet submarines and torpedoes. A typical towed array will comprise several hundred hydrophones, together with electronic circuits to preamplify, sample and digitize their outputs. The hydrophones are enclosed in a plastic hose, liquid filled to achieve neutral buoyancy in the sea. The array is towed by a cable of length decided by the speed of the tow.

4.2. TOWED ARRAYS CONSTRUCTION

The basic construction of Surface Ship and Submarine Towed Arrays is identical as shown in the two diagrams, at Fig 4.1 and 4.2

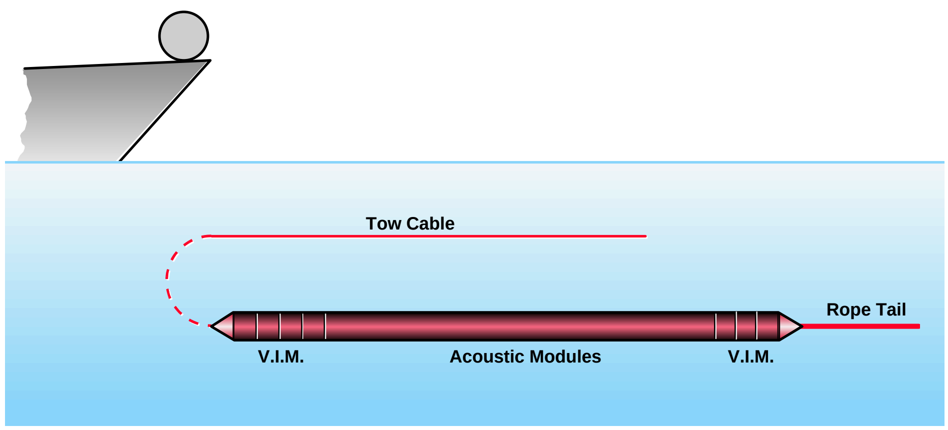


Figure 4.1 - Surface Ship Towed Array

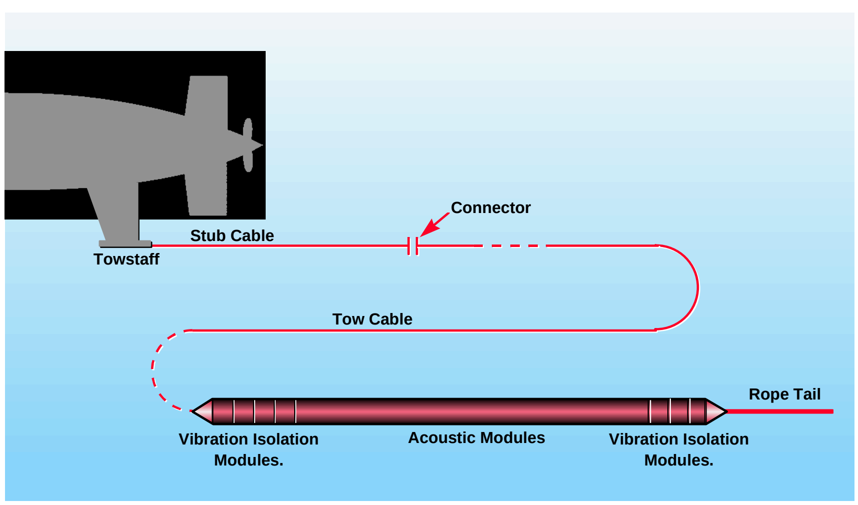


Figure 4.2 - Submarine Towed Array

Towed Arrays are commonly made up with the following constituent elements:

4.2.1. Tow Cable

Contains the signal wires which connect the array and inboard processors, and strength members. The number of wires will determine the number of channels available for use, which will typically be between 32 and 128. Multiplex techniques will greatly reduce the number of wires required (2087 / 2076), thus many more channels can be produced, with a narrower tow cable. The strength member is usually kevlar or corvite (stranded kevlar).

4.2.2. Acoustic Modules

Contain a number of hydrophones which are kept neutrally buoyant by filling them with Kerosene (negatively buoyant). This also stops the elements from collapsing at depth and improves self noise performance. Current technology suggests that filling the arrays with special to type foam would further increase performance.

4.2.3. Vibration Isolation Modules (VIMs)

Used to isolate the acoustic modules from any vibrations set up by the tow cable, eg strumming of the cable itself, or turbulence. At either (or both) ends one VIM will be instrumented to ascertain non-acoustic data, eg heading, depth and temperature. These are also used to determine whether the array is stable following a course alteration.

4.2.4. Rope Tail

Rope tail is used to stop the array end from flailing.

4.3. OPERATIONAL CONSTRAINTS

Following constraints are important in Towed Array operations:

a. Can be towed at high speeds but must be used at low speed, to reduce flow noise and avoid array snaking.

b. Flow noise limited (Refer to Sonar Equations) - a bulk vibration is set up in the array tube.

c. Depth of the array is controlled by the length of stay and the speed of the towing vessel.

d. During a sonar search, the vessel must maintain slow speed in one direction.

e. The array can be a hazard to navigation.

f. Information is lost in the turn because once the array starts to move, bearing information becomes inaccurate as there is no heading reference. Time to steady curves are available for all arrays, for given towing speed, heading change, and length of stay. 2031 has only 1 heading sensor, whereas 2046 / 2054 has 2, but they are still of limited use as the array will curve as it turns.

4.4. TOWED ARRAY OCTAVES

Towed Arrays are described as being a so many ‘octave’ array. The term octave relates to the frequency band coverage of the array. For instance a five octave array will have the following operating frequency bands:

~ 0 - 55 Hz
55 - 110 Hz
110 - 220 Hz
220 - 440 Hz
440 - 880 Hz

Towed arrays cover the frequency spectrum 0 - 880 Hz, thus if a reduced number of octaves are provided the lowest octave will cover the frequency spectrum 0 - 110 Hz or 0 - 220 Hz.

The maximum frequency of the lowest octave is reduced by increasing the inter element spacing. This increases the number of octaves available by splitting the frequency band into ever smaller bandwidths. In general surface ship arrays can be longer than submarine arrays due to towing capability. Current in service surface ship arrays with modern navies have five octaves, whilst submarine towed arrays are processed in three (or four) octaves of 0 - 220 Hz, (0 - 110 Hz, 0 - 220 Hz), 220 - 440 Hz and 440 - 880 Hz [2].

As the inter element spacing doubles for every octave reduction it would be possible to manufacture a towed array where separate modules for each octave are connected consecutively. However, this would lead to an unacceptably long array, with far more hydrophone cables than can be fitted at present. In order to maintain the overall array length to a minimum, the hydrophones for successive octaves are nested, such that overall length is now governed by the length required for the hydrophones in the lowest octave.

Considering the five octaves, a surface ship towed array is made up of the following hydrophone combinations:

880 - 440 Hz 32 elements connected - each element used.

440 - 220 Hz 64 elements connected in pairs - double spacing used.

110 - 220 Hz 128 elements connected in quads - 4 times spacing used.

55 - 110 Hz 256 elements connected in eights- 8 times spacing used.

0 - 55 Hz Should logically be 512 elements but this would make the array too long for present capabilities, so actually 384 elements are connected in 12s, with 12 times spacing used.

The centre of each successive processing octave is fixed at the overall centre of the array.

In practice the array will have a significant curvature, which would affect the beam shapes unless corrected in the beam former. Heading sensors are placed at intervals within the array and their readings used to correct for array curvature; there are at least three sensors (front, rear and middle) and possibly more in very long arrays.

Towed arrays can have diameters as small as, say, 50mm(thin arrays) and as large as, say, 150mm. The length of a towed array is determined by its frequency of operation and the desired DI. When used as a receive array in an active system. The length might be from 10 to 50 m, whereas for a passive array the length might be from 100 to 1000 m.

Passive towed arrays operate over a frequency range of several octaves. The spacing between the elements is maintained about at about $\lambda/2$ at the centre frequency of each octave by a suitable choice among the available elements.

4.5. BEARING AMBIGUITY

A single line array is omni directional in the vertical plane and therefore when horizontal beam are formed, they exhibit a left/right ambiguity. There are several methods to resolve this ambiguity [1].

4.6. Course Change

If the tow vessel, and therefore the towed array, changes its heading, it is possible to resolve the ambiguity. The array heading change is not instantaneous and particularly for a very long array can take a considerable time. Nor is the target stationary. Nevertheless, the true target bearing is often quickly resolved, but may need confirming by making some assumptions about the target's motion.

4.6.1. Twin Arrays

Parallel twin towed arrays use the time delay between the signals arriving at the two arrays to resolve the left/right ambiguity. Maintaining the spacing between two flexible arrays is a practical problem, particularly for very long arrays and during a change of course. Precision, however, is not necessary provided some horizontal spacing survives and the arrays do not cross over. If both arrays are also used to form beams, an improvement in DI of up to 3 dB will result over a limited frequency range.

4.6.2. Triplets

In a triple array, each element now comprises three hydrophones – a triplet – in the vertical plane. Because time delays are measured between all three pairs, the left/right ambiguity can be resolved regardless of any rotation of the array. The method needs a fairly large array diameter to house the triplets and to produce measurable time delays.

4.6.3. SELF-NOISE

Towed arrays are well separated from tow vessels and therefore the vessel-radiated noise is significantly reduced, and except for the ahead bearing of the towed array, this is further reduced by the main lobe to side lobe ratio of the beams. The hydrodynamic noise of the towed array is made negligible at normal tow speeds of up to about 12 knots, and therefore the remaining and dominant noise is the ambient noise of the sea.

Passive towed arrays operate over a frequency range of several octaves. The spacing between the elements is maintained about at about $\lambda/2$ at the centre frequency of each octave by a suitable choice among the available elements.

4.6.4. CONCLUSION

Towed array sonar is specifically discussed in this chapter along with some of its limitations. Major components used in the construction of a standard towed array discussed. The concept of octaves in the towed array is discussed and how we exploit it to our benefit.

CHAPTER 5

5. FUNDAMENTALS OF BEAM FORMATION

5.1. INTRODUCTION

The beam pattern of an array is one of the most important factors in Sonar design and performance. For long range detection the beam must be narrow so that the transmitted power is concentrated to give a high intensity of transmission (Source Level). Also, a narrow beam will reject noise, as noise is in general arriving at the transducer from all directions (Directivity Index). This gives significant improvement in signal / noise ratio. A narrow beam additionally results in improved bearing resolution and accuracy.

The response of a transducer array varies with direction relative to the array. This results from the fact that sinusoidal signals arriving from a particular direction tend to be in phase at all elements of the array, whereas the noise is out of phase. The direction of maximum 'in phase' condition is known as the beam pattern or Directivity Function, D(θ).

5.2. THE SIMPLE 2-ELEMENT ARRAY

Let us consider, two point transmitters, situated some distance apart on an array, transmitting simultaneously. The two transducer elements fed with the same electrical signal will produce individual wide beams of sound which react with each other. The resultant is an interference pattern.

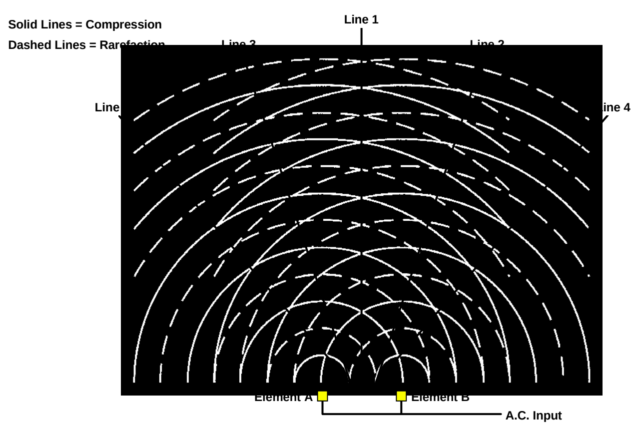


Figure 5.1 - Interference Pattern

Along Line 1 it can be seen that wherever there is a pressure maximum due to element A there is maximum due to element B. Similarly, whenever there is a pressure minimum due to element A there is a pressure minimum due to element B. Each element has the same effect, and combine to produce an increased amplitude. This effect is known as constructive interference.

Along Lines 2 and 3, the opposite situation occurs. wherever there is a pressure maximum due to one of the elements there is a pressure minimum due to the other. Along these lines the two waves are exactly opposite each other, ie 180° out of phase. The combined effect in this case is that the two waves cancel each other with the result that there is zero amplitude along Lines 2 and 3. This effect is known as destructive interference.

When describing the pattern of transmission amplitude against bearing it is normal convention to draw the resulting beam pattern in the form of a Polar Plot as in Fig 5.2. The distribution of transmitted power about the centre line is sometimes also shown on a Cartesian Plot as in Fig 5.3.

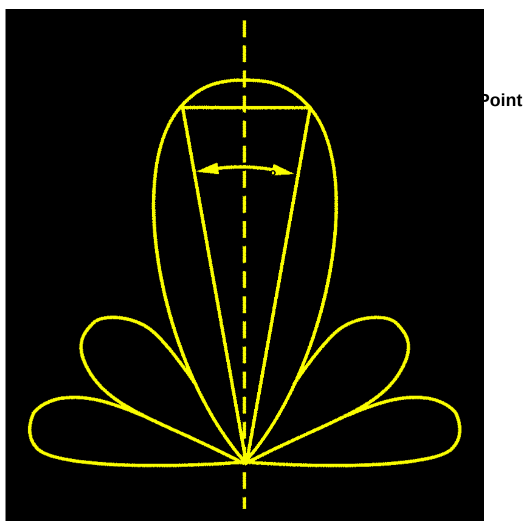


Figure 5.2 - Transmit Beam Pattern - Polar

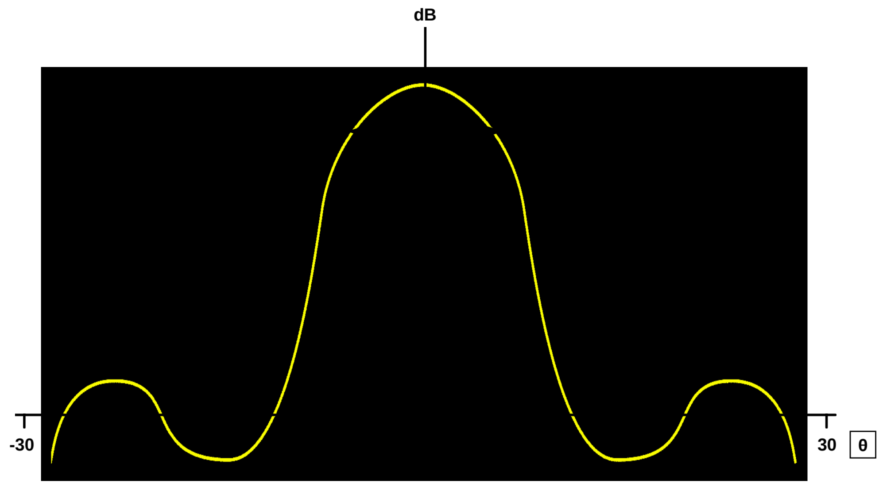


Figure 5.3 - Transmit Beam Pattern - Cartesian

5.3. LARGER TRANSDUCER ARRAYS

In practice, an array has many more than two elements and the Directivity Function D(θ) needs to be developed for such an array. The derivation of such a function is not required, but the result for a linear point array is [1] [2]:

$$D(\theta) = \frac{\sin [n\pi d/\lambda \sin \theta]}{n [\sin \pi d/\lambda \sin \theta]}$$

From the above equation:

a. as the number of elements (n) is changed, or

b. the spacing (d) altered, or

c. the operating frequency ($\lambda = c/f$) changed,

the interference pattern created will change. The number of sidelobes will vary, and also Diffraction Secondaries may be introduced. These are both unwanted features which need to be removed or at least reduced in magnitude. These effects and the methods to achieve this will be discussed shortly.

5.4. FORMATION OF BEAMS

A single transducer element will transmit sound over a wide beam angle, perhaps as much as 180°, which is too great for most sonar applications. It is not practical to narrow the beam by using reflectors, as a light beam is produced in a torch, instead we rely on the interference effect between two or more radiating elements.

For both reception and transmission, transducers must be grouped together in order to produce beams with directional properties superior to those of a single element. The simplest form of grouping is a linear array. A horizontal linear array produces a beam in the horizontal plane but is no more directional in the vertical plane than a single transducer. The opposite is true of a linear array in the vertical plane.

5.5. THE FORMATION OF NARROW RECEIVE BEAMS

Sound signals, whether echoes, reverberations, or noise, returning to a multi-element transducer will cause electrical signals to be produced in each element. This is true not only for signals from sources in the area of the main transmitted beam (eg Line 1 in Fig 5.1) but also for sources in the areas to each side (eg Lines 2 and 3).

Suppose that a signal is received by the two-element transducer of Fig 5.1 from a source on the transducer centre line or Maximum Response Axis (Line 1). Both elements will pick up this signal, and since the source must be equidistant from each element, the electrical signals produced will be in phase. If the two electrical signals are now added to each other a stronger signal will result.

Another signal, received from a source on Line 2, will also produce electrical signals in both elements A and B. However, since any point on Line 2 is further by ½λ from A than B, the two electrical signals will be out of phase by ½λ. If these two electrical signals are now added to each other they will cancel each other and no signal will result.

By adding the electrical signals arriving at the transducer elements, signals from sources in the main beam area will be retained and signals from sources in the areas of Lines 2 and 3 will be eliminated. Thus, the reception beam pattern is identical to the transmission beam pattern.

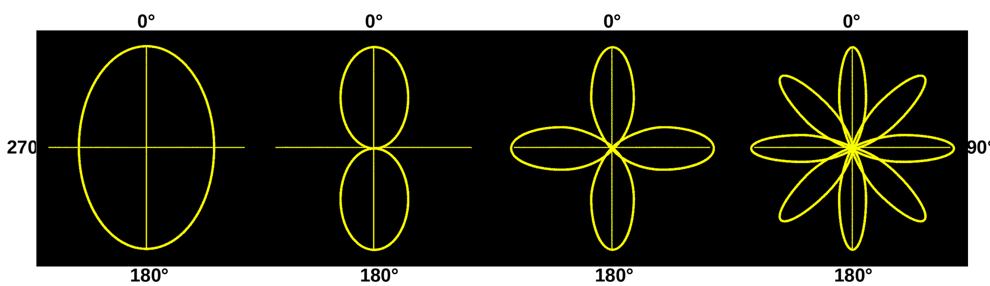
The main reception beam pattern is important when tracking a target because we require the maximum signal to noise ratio achievable. Noise radiating from sources outside the main beam will be eliminated as described above.

5.6. DIFFRACTION SECONDARIES

These are unwanted repetitions of the main beam. It is of paramount importance that these are not allowed to occur in front of the array, as a target held in a Diffraction Secondary cannot be distinguished from one in the main beam. This can be achieved by restricting the element spacing with respect to the wavelength to d ≤ 0.7 [1].

By restricting element spacing to below this figure the diffraction secondaries will be displaced behind the array, where no actual reception is made. If d = λ the first diffraction secondaries will occur at ± 90° to the direction of the array. However, to allow flexibility to electronically beamsteer the array a reduced upper limit of 0.7λ is normally selected. For a towed array, using omni-directional hydrophones, element spacing must be further restricted to 0.5λ or less.

Considering once again the two element array it can be demonstrated how the beam pattern will vary if d, the spacing between the elements is changed as in Fig 5.8.



d=λ/4 d=λ/2 d=λ d=2λ

Fig 5.4 - Beam Pattern vs. Element Spacing

5.7. SIDELOBES

Side lobes are unwanted features created by reduced constructive interference outside the main beam (caused by non-adjacent elements). Whilst they are of smaller amplitude than the main beam (~ -13dBs) they waste power and can lead to bearing ambiguity (a strong target in a side lobe could appear as a weak target in the main lobe). Careful design and construction of transducers can help to minimize them, however it is usual to introduce shading techniques to achieve a reduction (~ -40 dBs).

It can be shown that the number of sidelobes expected is n-2[1] [2], where n is the number of array elements. These sidelobes will appear:

a. Half either side of the Main Beam over a π radians (180°) spread. Which equates to between the Main Beam and its ‘behind array’ repetition. or

b. Between the Main Beam and the first Diffraction Secondary.

5.8. SHADING

Shading is a technique used to reduce the effect of side lobes for both the transmit and receive case. Taking an array of n elements, instead of the same electrical signal being applied to all elements it is 'weighted' such that the sensitivity of the array is altered, similarly, in the receive sense the electrical signals produced by some of the elements will be reduced. The effect is to reduce the magnitude of the sidelobes though the width of the main beam is increased slightly. The following list shows some examples of shading on a 6 element array and the effect is shown in Fig 8.5.

End Weighting 1, 0, 0, 0, 0, 1

- results in a narrow main beam but introduces diffraction secondaries.

Centre Weighting 0, 0, 1, 1, 0, 0

- results in a single, broad beam and no side lobes.

Binomial Taper 0.1, 0.5, 1, 1, 0.5, 0.1

- a compromise between the first 2 examples and produces a fairly wide main beam and no sidelobes.

Dolph-Chebyschev 0.3, 0.69, 1, 1, 0.69, 0.3

- significant sidelobe reduction is achieved. This is the most common function used [3].

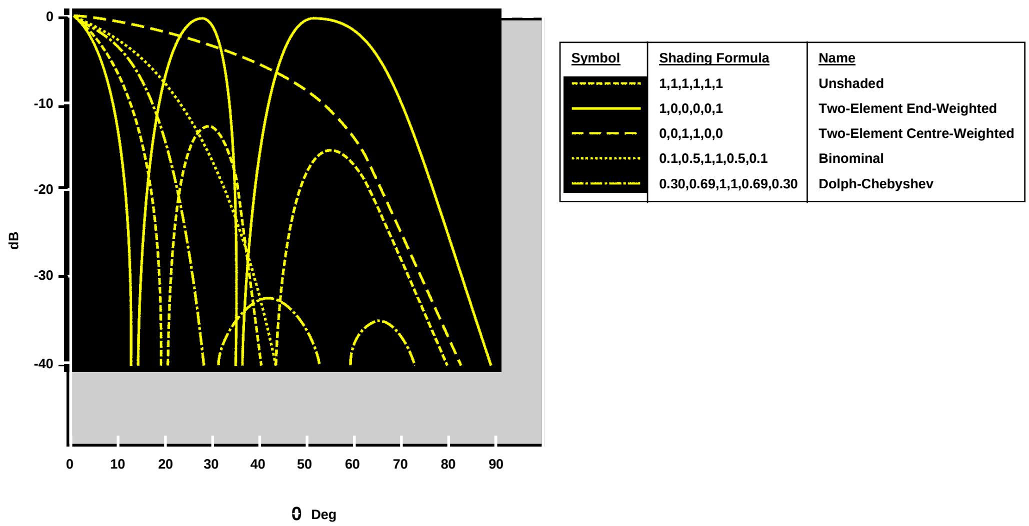


Fig 5.5 - Shading Examples

5.9. BEAMWIDTH

This is normally measured as the angle between the half power (-3 dB) points on the main beam and is a function of frequency (hence wavelength), the number of elements and the spacing between them.

The beamwidth can be derived from the directivity function, making the assumption that the beamwidth is measured at a point equidistant between the maxima and first null, however it is considered sufficient to state the equation and not to derive it.

$$\text{BW (radians)} = \frac{\lambda}{nd} \text{ (for small angles)}$$

It follows that:

a. For a given frequency a longer array will result in a narrower beamwidth.

b. For a given array size, increasing frequency will result in a narrower beamwidth.

5.10. SUMMARY

Fig 5.6 shows 2 beam patterns:

a. Left hand beam pattern, shows an 8 element array (8 - 2 = 6 sidelobes), whose inter-element spacing does not exceed 0.5λ (TA) or 0.7λ (BA).

b. Right hand beam pattern, shows a 5 element array (3 sidelobes between the Main Beam and the first Diffraction Secondary), but inter-element spacing is now 2λ (2nd Diffraction Secondaries appearing from behind the array).

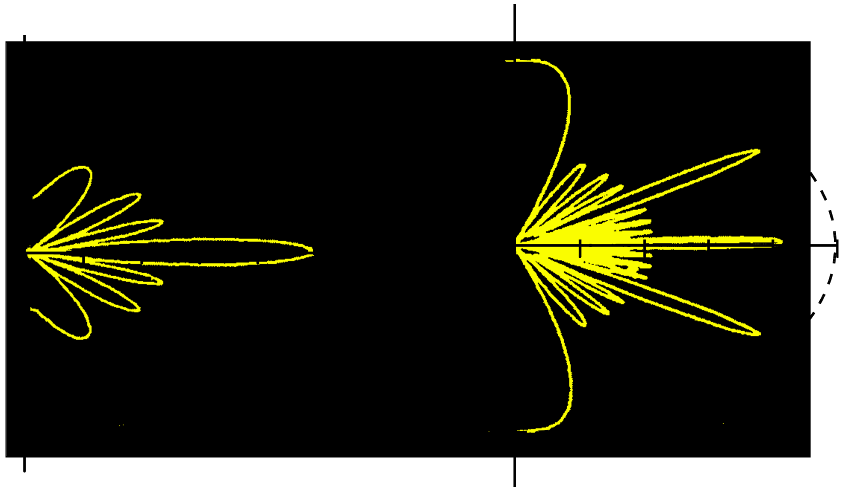


Fig 5.6 - Beam Pattern Summary

It should be noted that, for a given frequency, there will be physical constraints on increasing the size of the array ie size and weight. As one strives to detect low frequencies with any degree of accuracy it should be readily apparent that a move to a towed array is a logical step[3].

5.11. CONCLUSION

This chapter presented the basics of beam formation which will be used to understand the beam formation techniques both in time and frequency domain in the coming chapter.

CHAPTER 6

6. BEAM FORMATION TECHNIQUES

6.1. INTRODUCTION

A beam former is a spatial filter that operates on the output of an array of sensors in order to determine from which direction a sound is coming. It enhances the amplitude of a coherent wavefront relative to background noise and directional interference. The "pointing direction" is called the Maximum Response Angle (MRA), and can be arbitrarily chosen for the beams [4]. The goal of beam forming is to sum multiple elements to achieve a narrower response in a desired direction (the MRA). That way when we hear a sound in a given beam, we know which direction it came from.

6.2. TIME DOMAIN SONAR BEAMFORMING

Time-domain beam forming is realized by delaying and summing the shaded outputs of an array of transducers. Shading is simply amplitude weighting which is done to improve the spatial response characteristics of the beams. The beam forming delays are matched to the anticipated propagation delays of a pressure field incident from a specific direction [5].

Conceptually, time-domain beam forming in a single dimension is quite simple. For M sensors (transducers) each receiving a signal xm(t), the output of a single beam can be calculated by [6]

$$b(t) = \sum_{m=1}^{M} \alpha_m x_m(t - \tau_m)$$

where m is the shading coefficient for the mth sensor, and m is the required time delay applied to the output of the mth sensor. Fig 6.1 shows the block diagram for calculating a single beam using strictly analog methods. Note that analog multiplication, time delay, and summation is required.

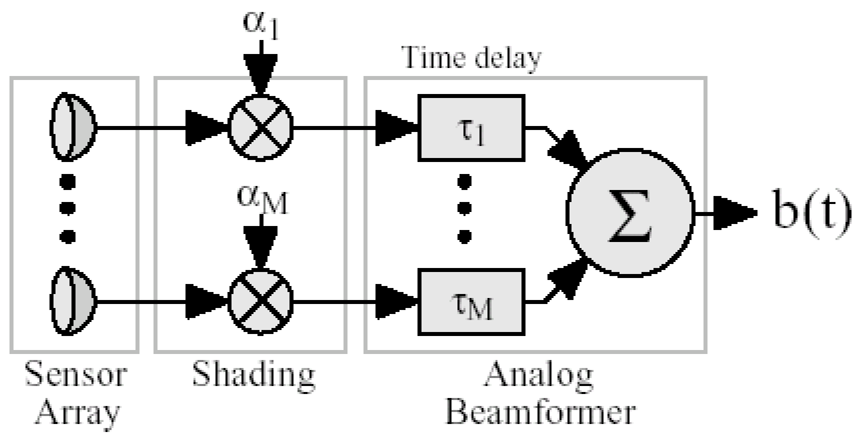


Figure 6.1: Digital Beam former With Digitizing Sensor Array

The beam forming time delays are determined by geometrically projecting the elements of the sensor array onto a line that is perpendicular to the Maximum Response Angle for the desired beam. The distance from each physical element location to the perpendicular line(divided by the speed of sound) is the necessary time delay for the corresponding element.

Consider an array of hydrophones receiving signals from an acoustic source in the far field as in Fig 6.2a. If the outputs from the hydrophones are simply added together then, when the source is broadside to the array, the hydrophone outputs are in phase and will add up coherently.

As the source is moved around the array (or the array rotated), then hydrophones across the array receive signals with differential time delays, so the hydrophones outputs no longer add coherently and the summer output drops as in Fig 6.2b.

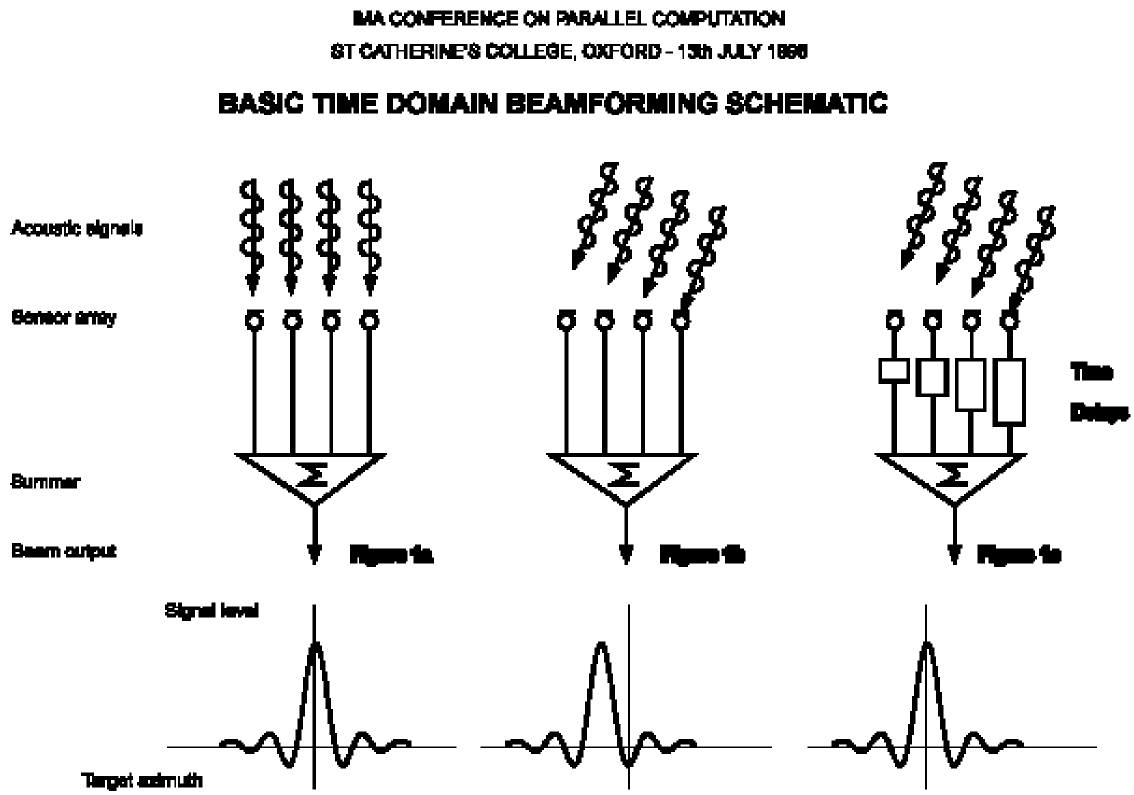


Figure 6.2: Basic Time Domain Beam forming Schematic

If we plot the signal level as the array is rotated, we get the array beam pattern: in this case, where all ‘phone outputs are weighted uniformly, a classical sinc function. In order to reduce the effects of the spatial side-lobes of the beam-pattern, an array shading function is applied across the array.

In radar beam forming systems, this weighted element summation is often all that is needed. The antenna can then be rotated to scan the narrow beam formed by adding together the array elements. However, in sonar we can not usually use this method. Firstly, sonar arrays are often pretty big and heavy so must be mounted on the hull of a ship or submarine. Secondly, even if we could rotate them, as the array was turned, the movement of water across the array face would generate flow-noise. This would swamp out the received signals we were trying to detect. Consequently, sonar arrays are usually mounted in a fixed position on a platform (or towed behind it) and scanned electronically.

Electronic beam-steering can be achieved by introducing a time delay network between the individual ‘phone outputs and the beam summer, so that signals from the required ‘look direction’ are brought into phase and can be added together coherently as in Fig 6.2c.

Conceptually, we could systematically vary these time delays to electronically scan a single beam around the platform, but again we have problems. If we consider an active system (where acoustic energy is transmitted from the platform and targets located by detecting echoes), it takes around 2 minutes for the sound energy to get from the transmitter, out to say 50 kYards, and for any echoes to propagate back (speed of sound in water is around 1500 metres/second). During this time we need to maintain the receive beam looking in the same direction, so as not to miss any potential echoes. So for a typical beamwidth of say around 1 degree, we would need to step the receive beam around in 1 degree, dwelling for two seconds every time we transmit to receive any echoes. To completely search 360 degrees around the ship would take around 12 minutes!! This obviously is not sensible (a torpedo attack can be all done in around 35 seconds), so a number of delay/summation networks are used in parallel to form a fan of beams with respect to the array in Fig 6.3.

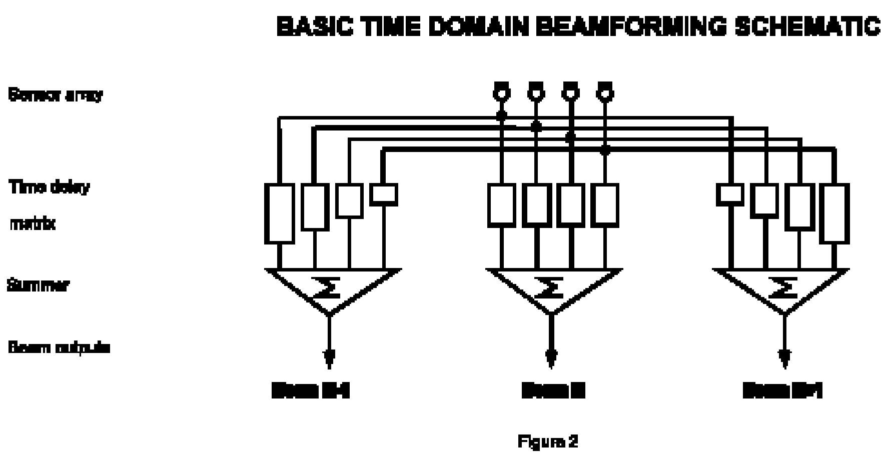


Figure 6.3: Basic Time Domain Beamforming

Although the delay networks are shown separately in Figure 6.3, in practice they usually use a common random access memory store [7]. This store is organised to hold a running time history of the acoustic data received by the array. There is a one-to-one mapping between the element position in the array and where that data is stored in the memory. The store is updated, usually by sampling data from all elements simultaneously at several times the Nyquist rate. In between write updates, samples of the required output beams are generated sequentially. Adding together data accessed by addressing planes across the RAM space-time matrix forms these. For example, address plane 1 provides a systematic time delay along the array and forms a beam to endfire. Address plane 2, a delay down the array to form a downward looking beam. Address plane 3, equal delay in all channels to form a beam normal to the array in both azimuth and elevation.

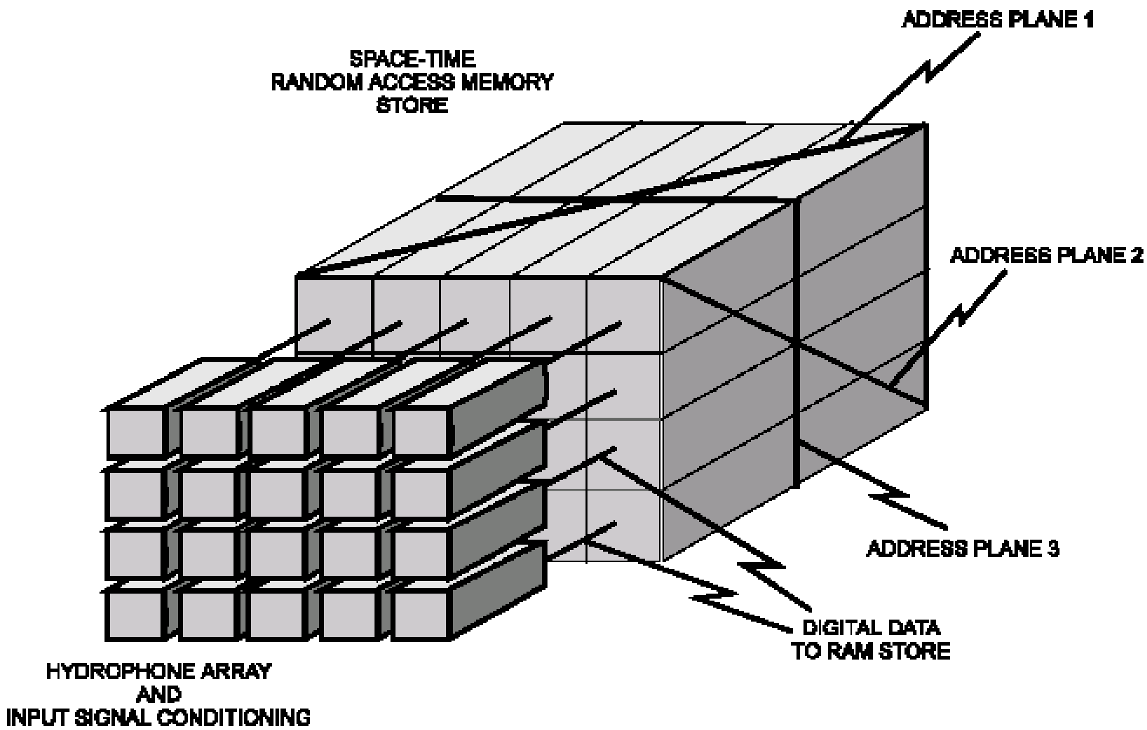


Figure 6.4: RAM Store for Beam forming

It is an easy step from here to stabilize the beams in space by compensating for the effects of platform motion. If the array motion is monitored, using for example a set of tri-axial accelerometers, this motion data can be used to correct the read address planes to compensate for the movement and to inertially stabilise the beams in space. It is also an easy step to generalise the space-time beamformer to handle other array geometries, for example to handle line, cylindrical or conformal arrays, by simply mapping the planar addresses onto the more complex array geometry.

In summary time domain beamforming based on space-time RAM stores is very flexible and widely used. The main problem lies in the overall amount of hardware needed to use this type of system. In practice, to maintain good side-lobe levels, the time resolution used to form beams must be equivalent to sampling the ‘phone data at around 10 times the Nyquist rate. This can be achieved either by heavily oversampling the ‘phone data or by sampling at a lower rate (maybe down to close to Nyquist) and then interpolating the data,often using FIR interpolators, to improve the time resolution of element data into the beam summer. This interpolation can be carried out either before or after the delay/storage operation: in practice a combination of pre-store and post-store interpolation is used. Either way, the need to oversample or to interpolate increases processing load and for large systems, frequency domain realisations are often used to minimise system size and cost. The cost in the case of frequency domain beamformers is in the added complexity of the algorithms.

6.3. FREQUENCY DOMAIN SONAR BEAMFORMING

The main aim of using frequency domain techniques for sonar beamforming is to reduce the amount of hardware needed and hence minimise cost. The time-domain system is very flexible and can work with non-equi-spaced array geometries. It is very efficient with arrays with small numbers of channels, say up to 128 phones, but as it is essentially an $O(N^2)$ process it becomes unwieldy with large arrays.

Many sonar systems need to use spectral data: for example, in active pulse-compression systems, the correlation processing is often conveniently carried out using fast frequency domain techniques and for passive systems data is usually displayed as a spectrum versus time LOFARgram plot for a number of look directions. In these types of system, it may be convenient to use frequency domain beamforming to avoid some of the time-frequency, frequency-time transformations that would be needed if time domain beam forming were used. There are several classes of frequency domain beam forming.

‘Conventional’ beam forming, where array element data is essentially time delayed and added to form beams, equivalent to a spatial FIR filter,

Adaptive beam forming, where more complicated matrix arithmetic is used to suppress interfering signals and to obtain better estimates of wanted targets.

High resolution beam forming, where in a very general sense target signal-to-noise is traded to obtain better array spatial resolution.

Only conventional beam forming will be discussed here. The conventional beam forming algorithms can be again sub-divided into three classes, narrow-band, band-pass and broad-band systems.

6.4. CONVENTIONAL BEAMFORMING

Conventional Beam forming (CBF) attempts to minimize the noise gain while maintaining distortionless response. In CBF,the noise matrix (**N**), is reduced to the identity matrix since it assumes white, Gaussian noise. The resulting array gain (AG) is equal to M, the number of array elements. This makes sense since if you have M copies of the signal plus noise, then there are M copies of the signal to add up. Also, the M copies of the uncorrelated noise measurements do not add up.

However, there are alot of assumptions that go into getting this ideal solution and they all contribute to make the actual array gain less than the ideal value. Several of these assumptions have already been alluded to and include errors in the propagation model, non- ideal characteristics of the spatial filter, and the noise structure not being white and gaussian. Despite these problems, CBF has been the standard beam forming technique employed by passive acoustic linear arrays for many years.

6.4.1. Narrow Band Systems

If the beam former is required to operate at a single frequency, then the time delay steering system outlined above can be replaced by a phase delay approach. For example, the time domain beam former output can be written as

$$A_r(t) = \sum_{k=0}^{k=N-1} W_k . f_k .(t + \tau_{k,r})$$

where N is number of hydrophones in the array

Wk is the array shading function

fk(t) is the time domain kth element data and

If f(t) is generated by a narrow band process, then we can write

f(t) = cos (ωt)

f(t)= Re [exp{-jωt)}]

If we take a snap-shot of data across the array at some time $t=T_0$ when the source is at some angle ϕ with respect to broad-side (assume an equi-spaced line array), then we have

$$f_k(T_o)=\cos(\omega T_o+k\phi)$$
$$= \mathrm{Re}\,[\exp\{-j(\omega T_o+k\phi)\}]$$

where ϕ is the differential phase across successive elements in the array due to the relative bearing ϕ. We can bring the element data into phase by correcting for this $k\phi$ term, when ϕ the array beam sum output is steered to look towards the source at bearing. If we form our beam sum output as:-

$$A_r(T_0) = \sum_{k=0}^{k=N-1} W_k.\, f_k(T_0).\, \exp\{-jk\phi\}$$

$$= N.\cos(\omega T_0)$$

In practice, we know we want to form a fan of say M beams from the array, so we can write

$$A_r(T_0) = \sum_{k=0}^{k=N-1} W_k.\, f_k(T_0).\, \exp\{-jkr\theta\} \quad \ldots\ \text{for } -M/2 <= r <= M/2-1$$

If we choose θ so that $\theta.M/2$ is equal to the differential phase between elements when the source is at end-fire, then we have formed a fan of M beams covering +/-90 degrees about broad-side. This process is repeated on successive snap-shots of array data, with snapshots gathered at some rate faster then Nyquist for the frequency of interest. If we compare this beam equation with that for the discrete Fourier transform (DFT) of a block of time series data B_k given by

$$A_s = \sum_{t=0}^{t=N-1} W_t.\, B_t.\, \exp\{-j2\pi/N.st\} \quad \ldots\ \text{for } -N/2 <= s <= N/2-1$$

then it can be seen that the two are similar and we can re-write the beam forming algorithm as:-

$$A_r(T_0) = \sum_{k=0}^{k=N-1} W_k . f_k(T_0). \exp\{-j2\pi/N.kr\alpha\} \text{ ... for } -N/2 <= r <= N/2-1$$

This is identical to the DFT equation, except for the α term in the complex exponent: α is usually between zero and one for beam forming.

In the DFT case, the coefficients are based on the integral roots of unity, exp{-j2π/N} whereas, in the beam forming case, the coefficients use the fractional roots of unity, exp{-j2π/N.α}.Hence the beam forming equation in a fractional discrete Fourier transform (FDFT) [8] rather than a DFT.The introduction of this factor α removes the symmetry of the basic DFT and at first sight it would appear that an order N^2 process is needed to realize the algorithm. However, there are several techniques that allow the FFT to be used to approximate the required FDFT, particularly for the narrow-band signal case outlined here.

Firstly, one can pad the input data sequence (the snap-shot of data across the array) by appending zeroes to the data block – this is effectively an interpolation process that allows the FFT exponents used for transforming the input data to approach the required FDFT exponents. The second approach is to use the nearest larger convenient FFT block size and to select the transform outputs closest to those that would have been generated by the exact FDFT algorithm. Both of these approaches are used in practice and result in narrow-band beamformers with order N.logN processes. However, there are fast algorithms for the FDFT, similar to the chirp-Z transform, and in many applications, particularly broad-band systems, these are more useful.

6.4.2. Wide Band Frequency Domain Beam forming [9]

The frequency domain systems outlined above rely on the source being narrow band. This is not usually the case in sonar, although it is often a reasonable approximation in radar and some communication systems.

One obvious way to extend the narrow band implementations is to gather a block of data from each sensor in the array (rather than snapshot across the array) and to use an FFT to convert each block into the frequency domain to generate narrow band components. The narrow band beam forming algorithms outlined above can then be applied sequentially to each of the frequency components in turn. This approach, a 2D FFT beam former, is shown schematically in Figure 6.5 below. The system complexity of this type of system is much less than the corresponding time domain implementation for large arrays but there are some problems.

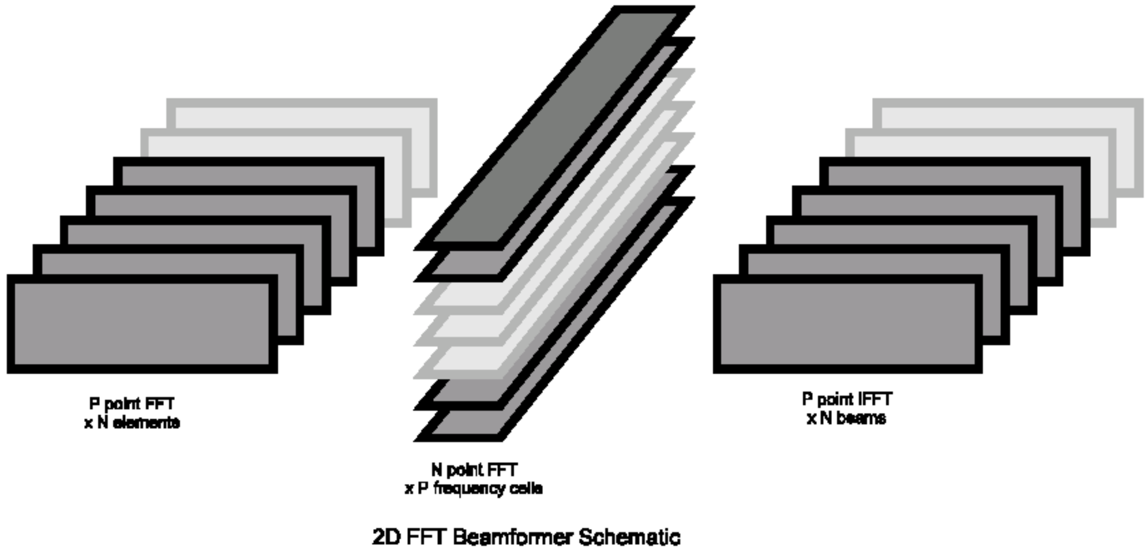


Figure 6.5: 2D FFT Beam former Schematic

Blocks of time domain element data are first transformed into the F-domain using P point FFTs. N point FFTs are performed across the array for each frequency cell of the P point FFTs to form N ‘beams’ each with P frequency cells. The P frequency cells per ‘beam’ are then transfromed back to time domain using P point IFFTs. If either of the simplistic approximations to the FDFT are used for the beamforming part of the process then, because the required value of a changes linearly with frequency, the effective maximum response axis (MRA) of any beam from the system is also frequency dependent. This effect is shown in Figure 6.6.

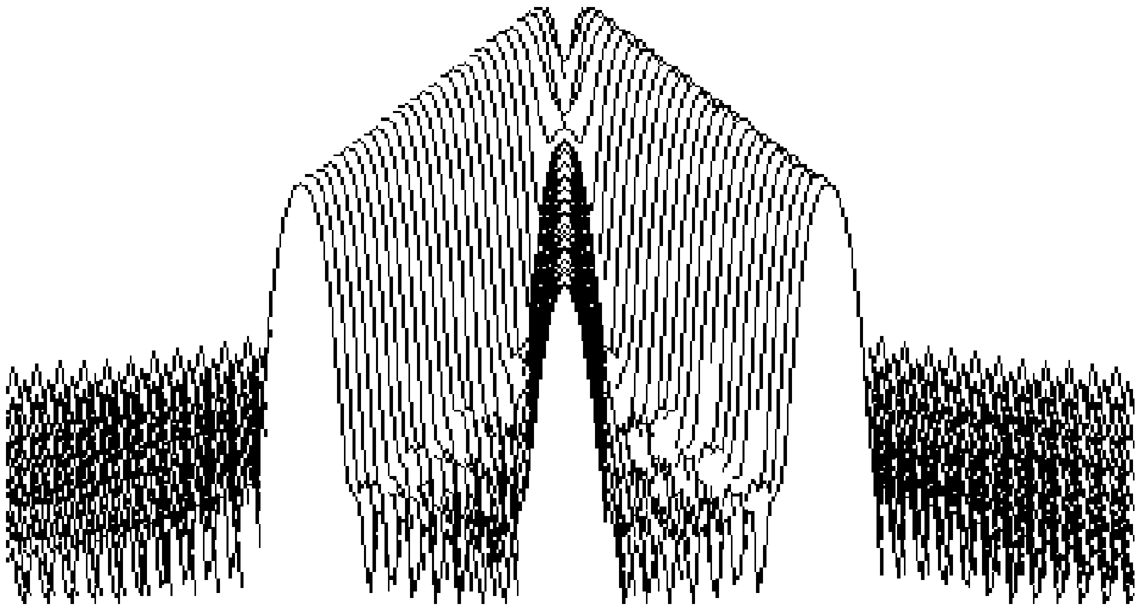


Figure 6.6 2D FFT Beam former Output

X axis corresponds to bearing, Y axis signal level and the Z axis frequency. High frequencies are at the front of the plot with low frequencies at the back. Note that received bearings change with frequency. This shows the bearing/frequency distribution for a 2D FFT beam former receiving three broad-band contacts at +45, 0 and –45 degrees wrt array broad-side. It can be seen that the beam MRAs for targets off broad-side vary linearly with frequency. This creates considerable problems in down-stream processing: systems usually require that beam pointing directions are independent of frequency. (The reason for this frequency/bearing variation is due to the fact that the 2D FFT beam former actually transforms element data into wave-number/frequency space rather than into bearing/frequency space).

A number of techniques have been used in the past to correct for this MRA variation. If the received data is relatively narrow-band, it is often ignored. With broader-band signals, an interpolation process can be used to interpolate the 2D FFT output into beams. The problem with this approach is that the interpolation process usually requires 2D FIRs and the interpolation scheme is often more complex than the time domain beam forming process that it is trying to replace!

The only real way to make the process broad-band is to tackle the problem of finding a fast algorithm for the FDFT. Then the beam forming process can be made exact, with the value of a changed exactly for each frequency component in the broad-band signal, thus producing beams with frequency invariant MRAs.

6.5. IMPLEMENTATION IN TOWED ARRAY

In a multi octave uniform towed array the inter element spacing for every octave is less than or equal to .5lambda and each octave covers a particular frequency range. By doubling the inter-element spacing, the frequency coverage is halved (upper and lower limit). This process is repeated many times to achieve a very wide frequency range. Early towed arrays were single octave. Now multiple octave arrays are in use.

The implementation of a multiple octave array is to have various sections with different inter element spacing, d joined together. There are reasons against this approach i.e. a multi-element array would be very long and sections would have to be produced with different inter element spacing. This would raise production costs.

An alternate method is to nest the array, one inside each other. The high octave array with inter element spacing, d. The Middle octave with inter element spacing twice of that i.e. 2d. Finally the Low octave has twice the spacing again, giving 4d. In the high octave every element is used. In the medium octave every second element is used and in the low octave every four elements.

Each octave is processed by a beam former which can have identical characteristics as the spacing d is changed to ensure optimum frequency coverage. There are however, different band pass filters, normally referred to as anti-aliasing filters in the input stage of each beam former.

6.6. CONCLUSION

This chapter covered the major techniques used for beam formation both in time and frequency domain. Towed array mostly used the delay and sum beam formation method i.e. uses the phase delay technique especially in the narrow band case which we are considering. We will use the same technique to develop our algorithm in the next chapter.

Chapter 7

7. Beam forming and Adaptive Filters

7.1. Introduction

Fundamentals of beam formation are discussed in this Chapter. The importance and need of beam formation is highlighted. The description of analogue and digital beam former along with creation of narrow receive beams, diffraction secondaries, side lobes and shading are also discussed. Further, this chapter also emphasizes the requirement of an adaptive beam former, which is the core issue of this dissertation.

7.2. Essentials of Beam forming

Beam formers are complex networks used to precisely control the amplitude of the acoustic energy passing through them [6, 7]. Beam former configurations vary widely from just few basics building blocks to tens of thousands of them depending on system performance and requirements. However, there are two basic ways to shape an acoustic beam i.e. passive elements and active elements [8] [9]. The simplest beam forming method uses passive reflectors to affect the near field region surrounding the radiating element. Conductive surfaces shaped into carefully planned geometric shapes are used to create a pattern of constructive and destructive interference in the vicinity of the radiating element. The result is a precisely shaped beam of acoustic energy focused in the desired direction with a shape and extent tailored to the application. Using passive elements, beam formers have the advantage of simplicity and low cost

The passive elements beam former has the disadvantage that due to massive structure involved arrays based on passive reflectors cannot move as rapidly as may required in systems that misses track of rapidly moving objects(like torpedoes) or must track many objects "simultaneously" by quickly moving the beam from one target to another. This can be overcome by substituting actively radiating elements with passive elements to create the desired field. Precisely controlled signals/thresholds of the elements create the desired beam shape from the controlled constructive and destructive interference patterns established in the near-field region of the array. The main advantage of using active element is that there is no need to physically move the array to focus the beam on various targets. The beam can be moved arbitrarily. Moreover, the shape or the resulting beam can be dynamically altered from a broad "floodlight" illumination of a region to a "spotlight" small beam focused closely on a target of interest. However, the beam formers work by carefully controlling the acoustic signal energy conveyed to the radiating elements of an array. Therefore, maintaining a precise phase and amplitude relationship through an arbitrary number of active stages requires the employment of complicated algorithms and make the circuitry complex and expensive.

As mentioned above, beam formers work by carefully controlling the amplitude and phase of acoustic energy conveyed to the radiating elements of an array [10]. Manipulating the amplitude and phase of the acoustic energy can be accomplished at various points in the path between acoustic signal generation and its ultimate radiation. However, maintaining a precise phase and amplitude relationship through an arbitrary number of active stages is difficult when system requirements include a broad range of frequencies as well the technical challenges mount.

7.3. Importance/Need of Beam forming

A beam former is a spatial filter that operates on the output of an array of sensors in order to enhance the amplitude of a coherent wave front relative to background noise and directional interference [11][12]. The figure 7.1 shows a curved array of hydrophone sensors, or staves. Each sensor (red circle) is located at an (x,y) coordinate as shown. These sensors are pointed in known direction (blue arrows), and the beam formed points in a chosen direction (green arrows). The "pointing direction" is called the Maximum Response Angle (MRA), and can be arbitrarily chosen for the beams.

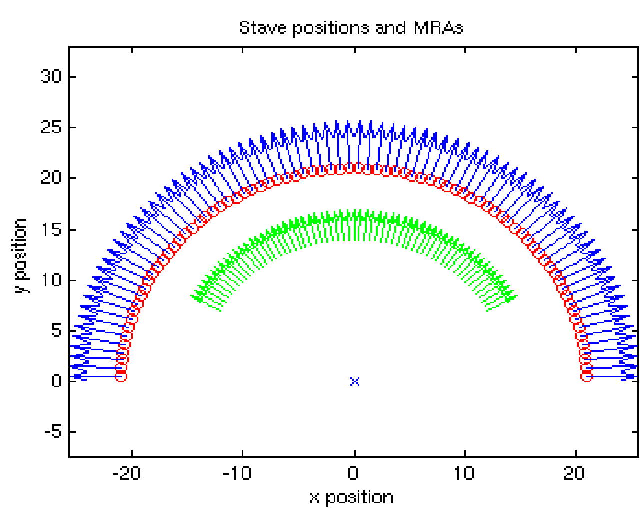


Figure 7.1 Stave Position and MRAs

The response of a given element is plotted on a polar graph as shown in figure 7.2. The angle is the offset from the MRA and the radius is the magnitude response (dB) in that direction. Element responses (determined by the 3dB down point) are very wide and the width is about 90 degrees.

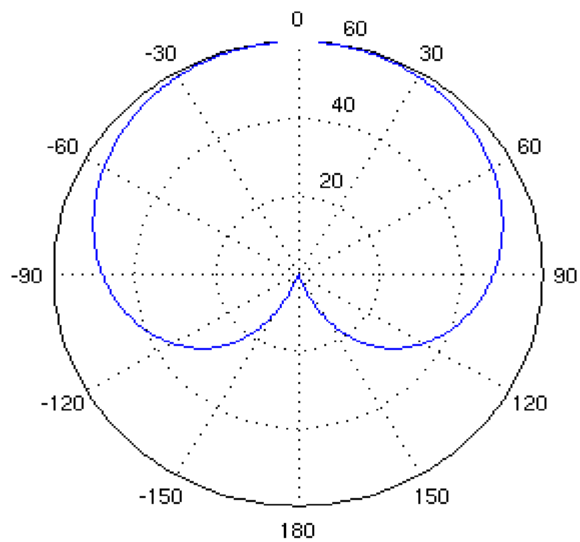


Figure 7.2 Polar Representation of Element Response

The goal of beam forming is to sum multiple elements to achieve a narrower response in a desired direction (the MRA) as shown in figure 7.3 [13]. In this way, when a sound is received from a given beam, its direction of arrival can immediately be known. Real implementations introduce certain undesirable features such as nulls and side lobes, which will not be discussed here.

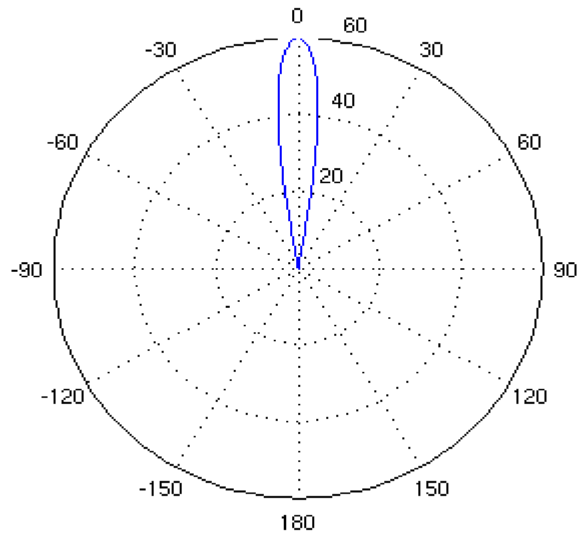


Figure 7.3 Polar Representation of MRA

7.3.1. Analogue Beamformer

Time-domain beam forming is done by delaying and adding shaded outputs from an array of transducers [14]. The (optional) shading of the sensor outputs is done to improve the spatial response characteristics of the beam and is roughly equivalent to "windowing" in DSP theory. Each beam is formed by delaying and summing sensor elements [15]. The block diagram shown in figure 7.4 indicates that how a single beam is formed from N transducers in an analog beam former.

The delay used for each sensor element is determined by array geometry and the desired MRA. Projecting the elements onto a line which is perpendicular to the beam's MRA gives a distance for each element. This distance (divided by the speed of sound) gives the delay required to form the beam at the desired MRA as shown in figure 7.5.

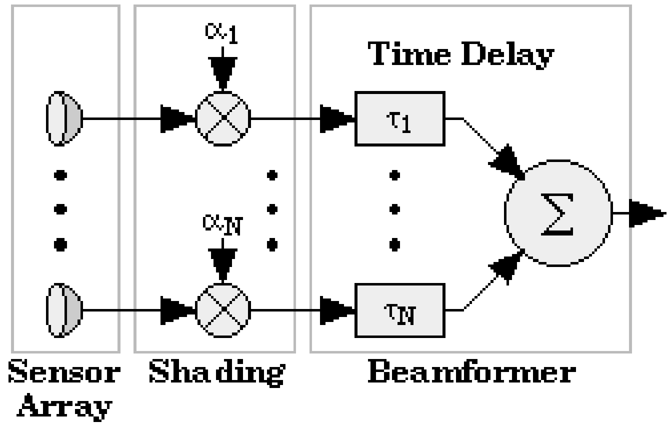


Figure 7.4 Analog Beam former

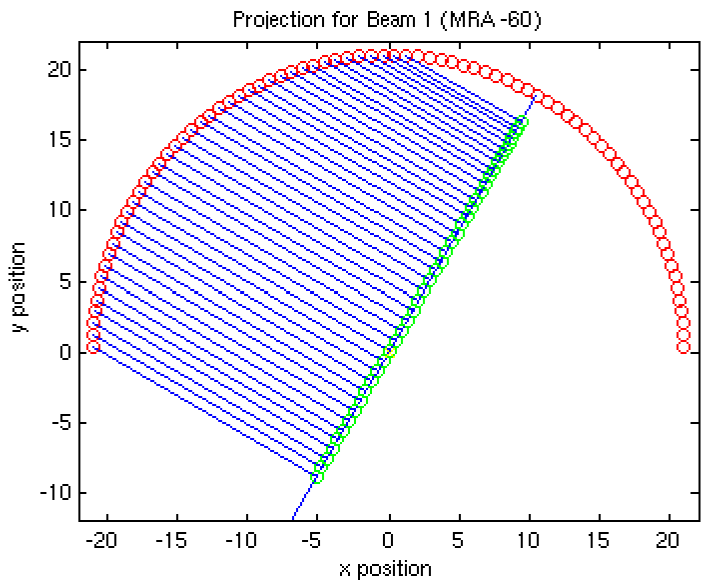


Figure 7.5 Beam Projection MRA 60°

Since our array is curved, each sensor contributes to each beam differently. The plot shown in figure 7.6 indicates high responses with white and low responses with black. If the element and the beam point are in the same direction, the response is high.

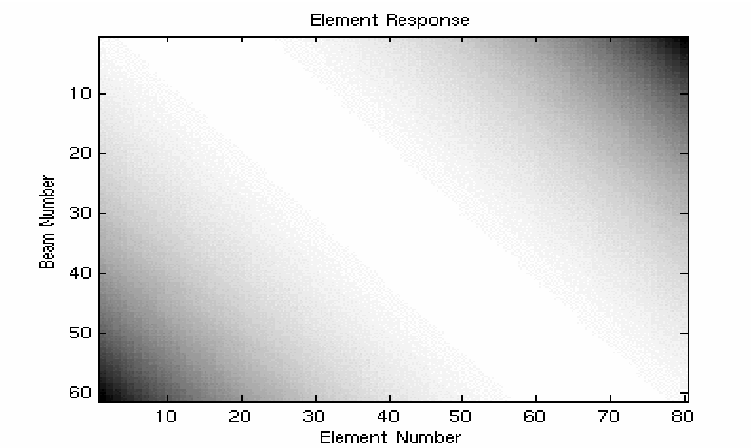


Figure 7.6 High and low Responses

It doesn't make sense to use elements which point in the wrong direction. In the plot shown in figure 7.7, any element which is at some fixed threshold below its maximum is zeroed out. Now by using non-zero elements, each beam is formed. This step saves on processing, with minimal beam degradation.

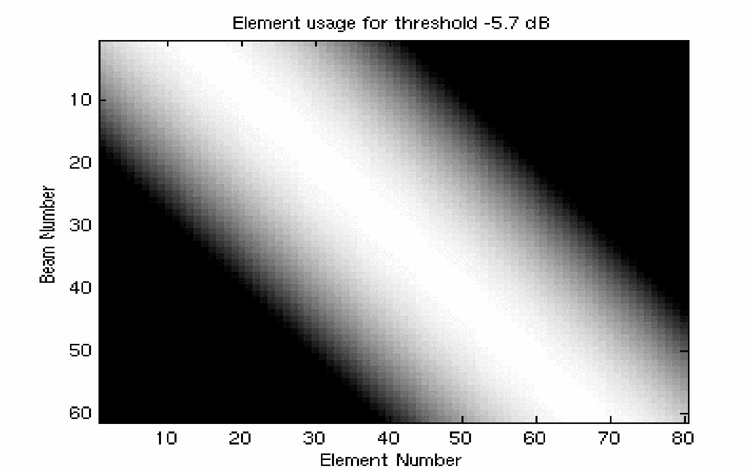


Figure 7.7 Non-zero elements Response

7.3.2. Digital Interpolation of Beam forming

In a digital implementation as shown in figure 7.8, the elements are sampled at a rate just above the Nyquist criterion. Although, this preserves the frequency content of the signal, but does not give enough steering delay resolution [16] [17]. Digital interpolation is performed by increasing the steering-delay resolution by a factor of L. Now time delays are quantized to integer sample delays.

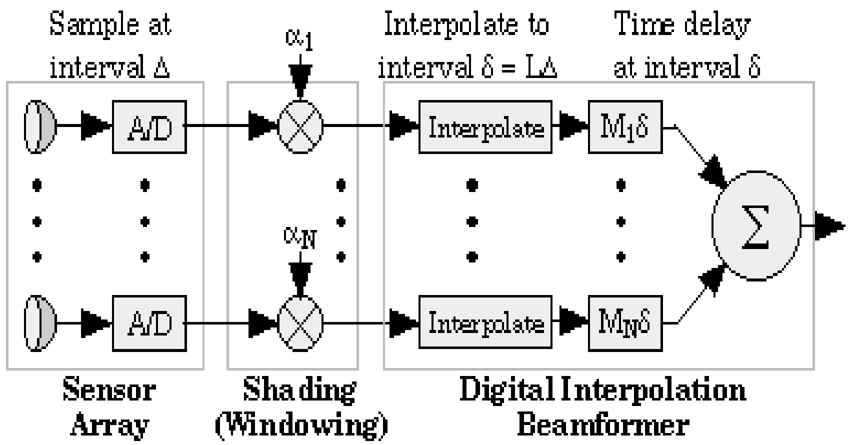


Figure 7.8 Digital Beam former

In this example we use unity shading, and simply interpolate across two samples. As a result, all coefficient values are between zero and one. These values are plotted in figure 7.9 where white is one and black is zero. Non-zero coefficients are extremely sparse, allowing efficient implementation. Note that each "picture" contains the values required to calculate one sample of one beam output.

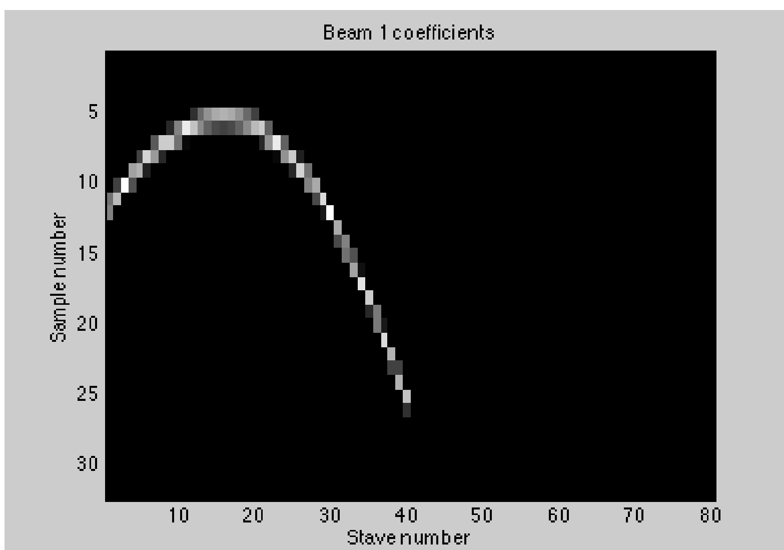


Figure 7.9 plot of shaded interpolation between two values

7.4. Beam forming as a Sparse Fir Filter

Modeling the beam former as a FIR filter allows for a simple, concise organization of the algorithm [13]. For our model we use the following parameters, with values from the example in parentheses

T - the total number of elements in the array (80)

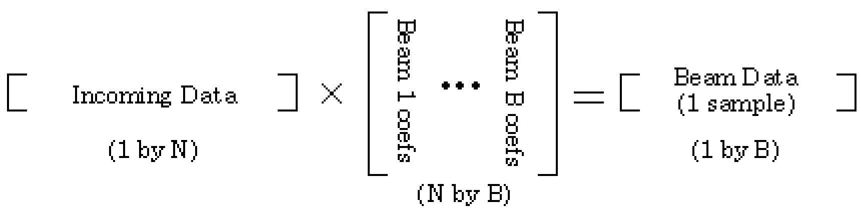
D - the maximum sample delay due to array geometry (31)

L - the length of the interpolation filter (2)

B - the number of beams calculated (61)

S - the number of staves (elements) used to calculate a beam (50)

If multiple samples of the entire sensor array are stored contiguously in memory, then each beam's coefficients can be represented by an FIR filter of length N = (D+L-1)T. Thus, the entire beam forming operation (for one sample of B beams) can be represented by a single operation:



The FIR filter length, N, can be extremely long -- in our example it is 2560. However, the number of non-zero coefficients is only 100 for a sparsity of 96%. As a result, 6100 multiply-accumulates (MACs) are required per sample. At high-frequency sonar sample rates, we are approaching one billion MACs per second.

Coefficient filter length $N = (D + L - 1) \times T$ (7.1)

Non-zero coefficients $C = L \times S$ (7.2)

$$\text{Sparsity} = 1 - \frac{C}{N} \tag{7.3}$$

$$\text{MACs per sample} = BC \tag{7.4}$$

The beam forming is mainly of two types i.e. Switched Beam forming and Adaptive Beam forming. As a broader view the Switched Beam forming is defined as a technique in which the weights of the sensor is selected and predefined in the library. In order to form a beam, then these weights will be calculated from library. This technique does not work when the environment is changing rapidly [18]. For a rapidly changing environment, adaptive beam forming is used; which will be discussed in the ensuing paragraph as well as in the following chapters.

7.5. Adaptive Beam forming

Adaptive Beam forming is a technique in which an array of sensors is exploited to achieve maximum reception in a specified direction by estimating the signal arriving from a desired direction while signals of the same frequency from other directions are rejected. This is achieved by varying the weights of each of the sensors used in the array. It basically uses the idea that, though the signals emanating from different transmitters occupy the same frequency channel, they still arrive from different directions. This spatial separation is exploited to separate the desired signal from the interfering signals. In adaptive Beam forming the optimum weights are iteratively computed using complex algorithms based upon different criteria. There are various Adaptive algorithms which can be used for this purpose. However, few algorithms which are very commonly used are LMS, RLS, QRLS and Kalman [19].

7.6. Basics of Adaptive Algorithm

A wide variety of adaptive filters are available to a system designer. The choice of an adaptive filter for a specific application must depend primarily on cost effectiveness and then there are three important issues that require attention i.e. computational cost, performance and robustness. Practical applications of adaptive filtering are highly diverse, with each application having peculiarities of its own. Thus, the solution for one application may not be suitable for another. Nevertheless, to be successful, the physical understanding of the environment in which the filter has to operate must be developed and its relation must be established with the realities of the applications of interest. Therefore, there is no unique solution to the linear adaptive filtering problem. Rather, a “kit of tools” represented by a variety of recursive algorithms is available, each of which offers a desirable features of its own. However, the challenge of using the adaptive filtering is, first, to understand the capabilities and limitations of various adaptive filtering algorithms and, second, to use this understanding in the selection of the appropriate algorithm for the application at hand.

7.7. General Types of Adaptive Algorithms

The diversity in the adaptive filters provides the user a multiplicity in the choice of appropriate algorithm based on the requirement and the type of its application. However, the most commonly used adaptive algorithms are Least Mean Square(LMS), Recursive Least Square(RLS), Square Root RLS, Fast RLS and Kalman Filtering algorithm [21]. The selection for the most appropriate algorithm will depend on the efficiency measuring parameters. These parameters are based on robustness, rate of convergence, misadjustments, tracking, computational requirements, structure and the numerical properties [19] [20]. According to these parameters the analysis of the aforementioned algorithms can be done to establish the best suited algorithm.

7.7.1. Kalman Filtering Algorithm

A distinctive feature of Kalman filter is that its mathematical formulation is described in term of state-space concepts. Its solution is computed recursively and is applicable with out modification to stationery as well as non-stationery environment. Numerical instability and excessive computational complexity are the most important problems encountered during the application of Kalman filters. The problem of instability is solved in Square- root variant of Kalman filters

7.7.2. Standard RLS Algorithm

It uses the transversal filter similar to linear adaptive filter. The derivation relies on the basic result in linear algebra known as the matrix inversion lemma. Most importantly, the algorithm enjoys the same virtues and suffers from the same limitations as the standard Kalman filtering algorithm. The limitation includes lack of numerical robustness and excessive computational complexity. It is because of these two limitations that prompted the development of two categories of RLS algorithm.

7.7.3. Square Root RLS Algorithm

It is based on QR-decomposition of the incoming data matrix. The technique used for de-composition is the givens rotation or the householder transformation. Whatever technique is used, the important thing is that this algorithm is numerical stable and robust. However, it suffers from excessive computational complexity and high cost.

7.7.4. Fast RLS Algorithm

The standard RLS algorithm and square-root RLS algorithm have a computational complexity that increases as the square of M, where M is the number of adjustable weights (i.e. the number of degrees of freedom) in the algorithm. Such algorithms are often referred to as O(M2) algorithms where O(•) indicates the order of the filter. When M is large the computational complexity of O(M2) algorithm may become objectionable from a hardware implementation point of view. Therefore, RLS algorithm was modified in such a way that the computational complexity assumes an O(M) form. The resulting algorithm is known as fast RLS algorithm which combines the characteristic of recursive linear least-square estimation with an O(M) computational complexity. However, fast transversal filters suffer from a numerical stability problem, which considerably limits their practical use.

7.7.5. Least Mean Square (LMS) Algorithm

The LMS algorithm is simple and capable of achieving satisfactory performance under certain conditions. Although LMS filter is simple and gives satisfactory performance, but its major limitations are a relatively slow rate of convergence and sensitivity to variations in the condition number of the tap inputs. The condition number of a Hermitian matrix is defined as the ratio of its largest eigen value to its smallest eigen value. Nevertheless, the LMS algorithm is highly popular and widely used in a variety of applications.

7.8. Selection of an Adaptive Algorithm

The aforementioned properties of adaptive algorithms depicts that the adaptive algorithm without depending on its type have some negative as well as positive points. RLS algorithm works on blocks of information in the input and has a faster rate of convergence due to which it whitens the input data by using the inverse correlation matrix of the data. This advantage of RLS is achieved at the expense of an increased computational complexity, numerical instability and high implementation cost. Recursive Least-squares (RLS) estimation may be viewed as a special case of Kalman filtering, therefore Kalman filter also suffers from the problem of numerical instability, computational complexity and high cost for hardware implementation. Since it is not possible to have an ideal adaptive filter, therefore a trade off has to be made. The LMS algorithm if considered in terms of computational complexity, hardware implementation cost and storage requirements; the algorithm is considered the most efficient. Further, it does not suffer from the numerical instability problem inherent in the other two algorithms. For these reasons, the LMS algorithm has become the algorithm of first choice in this application. The LMS algorithm can be complex or real depending on the nature of the input/output data and tap weights. Therefore, real LMS algorithm will be applied for the sake of simplicity.

Further, the LMS algorithm is simple and yet capable of achieving satisfactory performance under the right conditions. Moreover, it provides a practical frame of reference for assessing any further improvement that may be attained through the use of more sophisticated adaptive filtering algorithms. LMS adaptive filter is a transversal filter with a single tap weight (and tap weight computation) at each tap in the filter.

7.9. The Structure and Operation of Least Mean Square Algorithm

The LMS algorithm is linear adaptive filtering algorithm, which consists of two basic processes i.e. filtering process, which involves computing the output of the signal in response to an input signal and generating an estimation error by comparing out put with the desired response and the adaptive process, which involve automatic adjustment of parameter of filter in accordance with the estimation error [22].

The combination of these two constitutes a feedback loop shown in figure 5.1. First, the LMS algorithm is built around transversal filter which is responsible for filtering process. Secondly, the portion of adaptive weight control mechanism, which performs the adaptive weight control process on transversal filter.

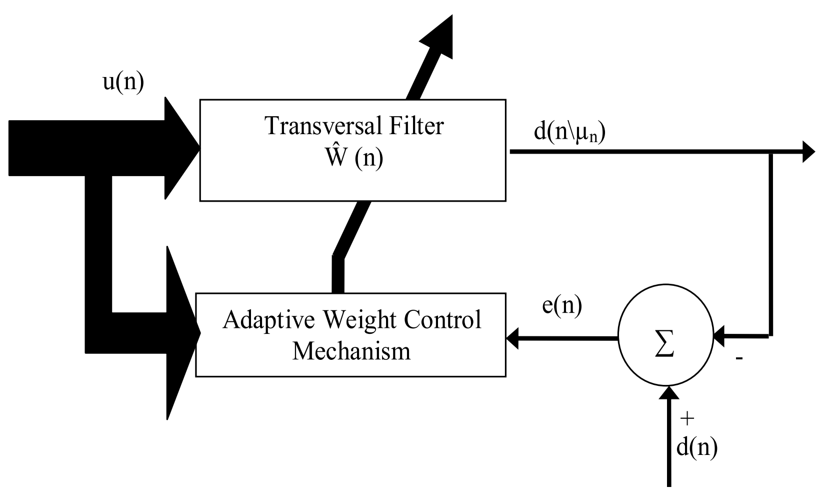


Figure 7.10 Block Diagram of Adaptive Transversal Filter

7.10. Least Mean Square (Lms) Algorithm

Parameters: M = number of taps

μ = step size parameter

$$0 < \mu < \frac{2}{\text{tap} - \text{inputpower}} \tag{7.5}$$

Initialization: If prior knowledge on the top weight vector $\hat{w}(n)$ is available value for $\hat{w}(0)$. Otherwise, set $\hat{w}(0) = 0$

Data:

Given: $u(n)$ = M-by-1 tap-input vector at time n

$d(n)$ = desired response at time n

To be computed:

$\hat{w}H(n + 1)$ = estimate of tap-weight vector at time n + 1

Computation: For n = 0,1,2,3,…,compute

$$e(n) = d(n) - \hat{w}H(n)u(n) \tag{7.6}$$

$$\hat{w}H(n + 1) = \hat{w}(n) + \mu u(n)e^*(n) \tag{7.7}$$

7.11. Conclusion

This chapter has highlighted and discussed the basic theoretical knowledge involved in the operation of a beam former and adaptive algorithms. It also covers the importance and requirement of a beam former along with broad ways to shape an acoustic beam. The basic working of an analogue and digital beam former is also discussed. The advantages and disadvantages of adaptive filters are also discussed in this chapter.

Chapter 8

8. System Model and Analysis

8.1. Introduction

This chapter explains the system model and derivation of the equation then describe the flow chart, pseudo code, simulations and analysis carried out with the help of adaptive beamformer software developed for this report. Change in sonar performance due to change in octave, step size and number of defective elements have been simulated and analyzed. The calculation of errors due to changing beam pattern and compensation of these errors automatically, to ascertain the effectiveness of the adaptive algorithm has been evaluated in this chapter.

8.2. Assortment of Adaptive Algorithm

In this project recursive algorithm used for the adaptive filter is Least Mean Square (LMS). Despite of being simple in implementation, this algorithm provides a practical frame of reference for assessing any further improvement that may be attained through the use of more sophisticated adaptive filtering algorithms in future. This quality of LMS makes the algorithm highly popular and widely used in a variety of applications. Consequently, the implementation of LMS required the information about input signal "u", output signal "y", and the desired signal "d" in order to calculate the error signal "e". This error signal is then applied automatically to adjust the filter parameters in accordance to the error signal to bring the value of output signal close to the desired signal. The required information obtained from a beamformer was processed in the LMS algorithm with the help of software developed for this dissertation.

8.3. System Model

SONAR array has different types of configuration, but widely used type of SONAR is towed SONAR. This paper uses octave SONAR to generate different types of frequencies. There are three main categories of octave SONAR [19].

a) High octave 440-880 H_Z

b) Medium octave 220-440 H_Z

c) Low octave 110-220 H_Z

The configuration of octave SONAR is based upon the nature of octave. For high octave each SONAR works independently. Let distance is d between two sensors. In medium octave, sensors work in group of twos so distance is 2d and in low octave distance is 4d because sensors work in group of fours. The basic system model for initial beam formation to track a target is given in Figure 1. Beam produced by sensor O which is taken as reference antenna is also shown.

The sensor output will be base-band signal, individually weighted and summed to produce overall output. The beam former will have steering capability and also cancellation of interference. The input to the sensor is target multiplied with the steering vector and noise.

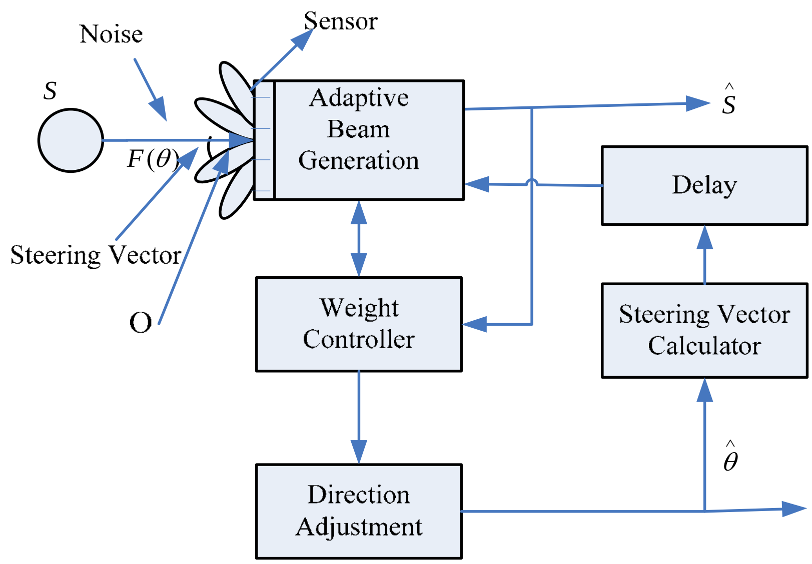


Figure 1: Block Diagram of Basic System Model

Following assumptions are made for this system.

- Noise is Additive White Gaussian Noise (AWGN) with zero mean and unit variance. If it is not true then a white filter is used to convert the color noise to white noise.
- Signal from target S, steering vector F(Θ) and noise N are statistically independent.

Input to the sensor will be.

$$X = S.\,F(\Theta) + N \qquad (8.1)$$

Where S is signal, F(Θ) is steering vector and N is AWGN noise. This input will be given to the adaptive beam former which will produce the estimated value of Signal $\hat{S}$ using adaptive algorithm.

If we estimate the initial position by any localization technique, then the initial weight can be formed [7].

$$W_t = \frac{F(\theta)_{t-1}}{F(\theta)_{t-1} F^H(\theta)_{t-1}} \qquad (8.2)$$

Where W_t is the weight of adaptive filter at time t, $F(\theta)$ is the steering vector, θ is the angle in which beam is produced and H represents the Hermition (complex conjugate and transpose). So these weights will be used by adaptive filter initially. Adaptive filter is tap filter. Output of adaptive filter is obtained as given below.

$$Y = X.W^t \qquad (8.3)$$

Where W is the weight of the filter, X is the input to the filter and Y is the out put of the filter.

For optimization, we shall use the Minimum Mean Square Algorithm.

$$e = d - y \qquad (8.4)$$

Where e is the error, d is the desired response and y is the output of the filter.

For minimization of error [20].

$$e^2 = E[(d - y)(d - y)^H] \qquad (8.5)$$

For minimized value:

$$\partial(e^2)/\partial w = 0 \qquad (8.6)$$

$$\frac{\partial}{\partial w}(e^2) = \frac{\partial}{\partial w}E[d^2] - \frac{\partial}{\partial w}E[yd] - \frac{\partial}{\partial w}E[dy^H] + \frac{\partial}{\partial w}E[yy^H] \qquad (8.7)$$

$$\frac{\partial}{\partial w}(e^2) = \frac{\partial}{\partial w}E\left[(d^2 - yd - dy^H + yy^H)\right] \qquad (8.8)$$

$$\Rightarrow -2E[d.X] + 2E[XX^H]W = 0 \qquad (8.9)$$

Where E is the expectation (Estimated).

So the value of w can be found through recursive relationship.

$$W_{t+1} = W_t + \mu(X_t - G_{t-1}\hat{S}_t)\hat{S}_t^H \qquad (8.10)$$

Where μ is the step size, $\hat{S}_t$ is the estimated value of signal at time t and G is the steering vector.

This filter will adopt the weights to minimize the error. it is also used for adaptive steering vector calculation which will adaptively adjust the angle of to optimum level. This will adopt the steering vector.

$$G_t = G_{t-1} + \mu(X_t - G_{t-1}\hat{S}_t)\hat{S}_t^H \qquad (8.11)$$

After the successful adaptive beam forming, the beam former will track the target and this value will be saved and can be used for any further calculations.

If all sensors are operational, the filter will work and beam former will produce the beam. If any sensor is defective then it will not detect the target as the overall output will be the sum of individual sensors.

When all sensors are not fully functional the matrix which is the sum of all the sensors is not able to produce the beam that can track the target. It results to interference and this interference should be decreased that is produced due to the defective sensors. Adaptive filter has been applied for interference calculations. Two types of adaptive filters can be used to cancel the interference.

- Non-Blind Adaptive Filter
- Blind Adaptive Filter

In non-blind adaptive filter, the desired beam matrix is known at the receiver side and the filter will be trained on the basis of this desired response. In Blind Adaptive Filter, the desired beam matrix is not known at the receiver side and filter will be trained without this matrix. As the input signal is normally fix and the power of the signal is known at the receiver and transmitter so we use this power to adapt the filter.

Let us consider the case when desired response is known at receiver. The input to the filter is the defective beam matrix that is represented as X(n).There are n tap delays that is equal to the length of the beam matrix. Each input of the beam matrix is multiplied by the weights. These weights are updated by weight controller on the basis of difference between the desired matrix and the product of input matrix with the weights. The block diagram of the filter is shown in Figure 2.

This error will be minimized according to the minimum mean square estimation.

$$e = b(n) - o(n) \qquad (8.12)$$

Where e is the error, $b(n)$ is the desired response and $o(n) = w^H x(n)$

(13)

Where w and x are matrices of $1 \times n$ order.

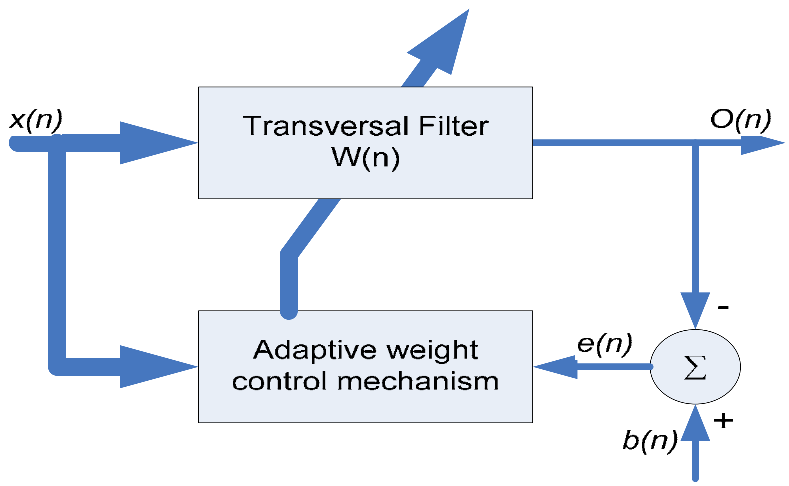


Figure 2: Block Diagram of Adaptive Filter

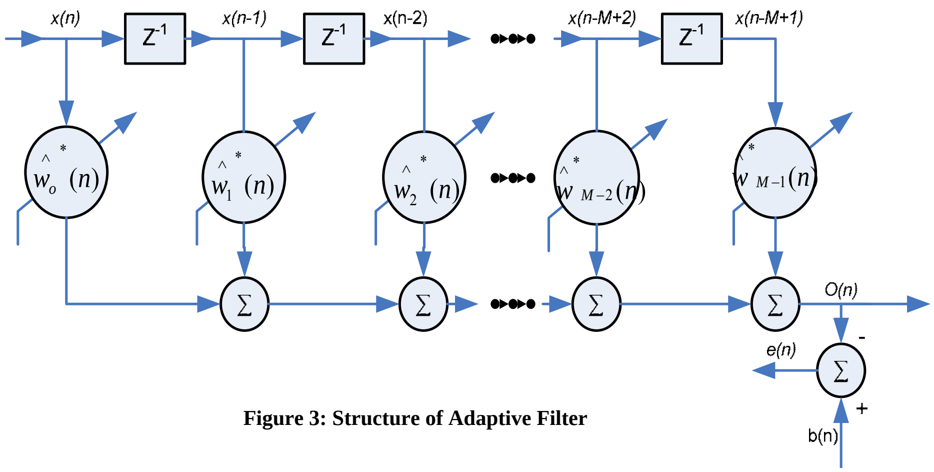


Figure 3: Structure of Adaptive Filter

$$e^2 = [b(n) - o(n)][b(n) - o(n)]^H \qquad (8.14)$$

$$e^2(n) = b(n)b^H(n) - b(n)o^H(n) - o(n)b^H(n) + o(n)o^H(n) \qquad (8.15)$$

This value of error will have minimum value with respect to w .If we differentiate e with respect to w and put it equal to zero.

$$\frac{\partial e^2}{\partial w} = \frac{\partial}{\partial w}\left[b(n)b^H(n) - b(n)o^H(n) - o(n)b^H(n) + o(n)o^H(n)\right] \quad (8.16)$$

$$0 = \frac{\partial}{\partial w}\left[b(n)b^H(n) - b(n)o^H(n) - o(n)b^H(n) + o(n)o^H(n)\right] \quad (8.17)$$

After simplifying the above equation, the updated weights will be equal to:

$$w_{n+1} = w(n) + \mu[b(n) - o(n)w^H(n)] \qquad (8.18)$$

Now we take the blind adaptive filter case where the desired response (beam matrix where all the sensors are healthy) is not known at the receiver .As power of the sensors seems to be the fix, the total power can be obtained by:

$$P = \frac{1}{N}\int_{-\infty}^{\infty} |S(t)|^2\, dt \qquad (8.19)$$

Where P is the power, N is the total no of sensors and S(t) is the signal that is used to produce beam.

Now we shall design a transversal filter to take beam matrix that is produced due to defective sensors. This matrix will be multiplied by the weights and output will be produced as given below.

$$Y = W^{H} X(t) \qquad (8.20)$$

Power of the output signal can be calculated as:

$$P_o = \frac{1}{N}\int_{-\infty}^{\infty}\left|Y(t)^2\right| = \frac{1}{N}\int_{-\infty}^{\infty}\left|w^{H} X(t)\right|^2 \qquad (8.21)$$

The error will be difference between P and P_o Now we shall again apply the minimum mean square estimation to minimize the error and update the weights.

$$e = \left(P - P_o\right) \qquad (8.22)$$

$$e = \frac{1}{N}\int_{-\infty}^{\infty}\left|S(t)\right|^2 - \frac{1}{N}\int_{-\infty}^{\infty}\left|w^{H} X(t)\right|^2 \qquad (8.23)$$

$$e^2 = \frac{1}{N^2}\left[\int_{-\infty}^{\infty}\left|S(t)\right|^2 - ww^{H}\int_{-\infty}^{\infty}\left|X(t)\right|^2\right]\left[\int_{-\infty}^{\infty}\left|S(t)\right|^2 - ww^{H}\int_{-\infty}^{\infty}\left|X(t)\right|^2\right]^{H} \qquad (8.24)$$

Taking the derivative on both the sides and putting the $\partial e^2 / \partial w = 0$, will give updated weights to get minimum error.

After simplification, the updated weights equation becomes as shown below.

$$w_{n+1} = w_n + \frac{\mu}{N}\left[\left(\int_{-\infty}^{\infty}\left|S(t)\right|^2 - ww^{H}\int_{-\infty}^{\infty}\left|X(t)\right|^2\right)dt\right] \qquad (8.25)$$

For our case of sonar the required weighted coefficients are calculated for a specified array in case weighting is used. Otherwise, same amplitude signal is fed to all elements. The specified beam number helps to locate the required look angle [25]. Total array is divided in 32 beams covering from 0 to 180 degree thus each beam covering about 5.425 degrees. A loop matrix is generated for the specified octave, so that appropriate signals can be fed to the individual elements of the array. In high octave selection each individual element is fed a separate signal. In medium, two elements are fed the same signal and in low octave, each four consecutive elements are fed the same signal. The information about the operational status of elements is also fed to the beam former which is being prompted by the user in this case. The angle is converted in to radians and removes -90 from the end of the statement as theta ranges from 0 to 180. The loop is defined for the complete length of the angles in radians and the angle of arrival is computed for the calculation of electrical angles and steering vectors.

$$\text{Theta} = \frac{-90:1:90}{180} \times \pi \tag{8.26}$$

$$\text{Stang} = 180 \times \frac{\text{BeamNumber}}{32} - 90 \tag{8.27}$$

$$\phi = \text{stang} \times \frac{\text{Theta}}{180} \times \pi \tag{8.28}$$

$$\theta = \pi \times \sin\phi \times 32 \tag{8.29}$$

$$S(\theta) = \cos(\theta) + j\sin(\theta) \tag{8.30}$$

$$\text{Steer Vector} = 32 \times S(\theta)$$

Radiation Pattern high octave =

$$\sum_{\theta=o}^{18o} S(\theta) \times \frac{-90:1:90}{180} \times \pi \times ej*32*S(\theta) \times \frac{2\pi}{\lambda} \times \frac{\lambda}{2} \times \sin(\theta) \tag{8.31}$$

The beam former calculates the steering vectors and steering angles normalize its absolute value and plots the radiation pattern when all the elements are operational and it also plot the radiation pattern when the number of elements are defective as specified by the user. Therefore the degradation in the beam can be observed by comparing the defective beam pattern (due to defective elements) with that of the beam pattern when all the elements are working.

The adaptive algorithm is a linear adaptive filtering algorithm, which consists of two basic processes [25].

Filtering Process, which involves computing the output of the signal in response to an input signal and generating an estimation error by comparing output with the desired response.

Adaptive Process, which involve automatic adjustment of parameter of filter in accordance with the estimation error.

The combination of these two constitutes a feedback loop shown in block diagram. First, the algorithm is built around Transversal filter, which is responsible for filtering process. Secondly, the portion of Adaptive weight control mechanism, which performs the adaptive weight control process on Transversal filter [24].

In this research these three relations of Adaptive algorithm are applied by considering variable y as the output of the beamformer when the elements were marked defective. The variable d; also the desired response, is the output of the beamformer when all the elements in the array are operational. The step size is denoted by x and is set to a very small value of 1*e-8 (3.3 * 10-4) so that the algorithm should converge as quickly as possible. However, input signal to the beamformer is denoted by u.

8.4. Selection of an Adaptive Algorithm

The aforementioned properties of adaptive algorithms depicts that the adaptive algorithm without depending on its type have some negative as well as positive points. RLS algorithm works on blocks of information in the input and has a faster rate of convergence due to which it whitens the input data by using the inverse correlation matrix of the data. This advantage of RLS is achieved at the expense of an increased computational complexity, numerical instability and high implementation cost. Recursive Least-squares (RLS) estimation may be viewed as a special case of Kalman filtering, therefore Kalman filter also suffers from the problem of numerical instability, computational complexity and high cost for hardware implementation. Since it is not possible to have an ideal adaptive filter, therefore a trade off has to be made. The LMS algorithm if considered in terms of computational complexity, hardware implementation cost and storage requirements; the algorithm is considered the most efficient. Further, it does not suffer from the numerical instability problem inherent in the other two algorithms. For these reasons, the LMS algorithm has become the algorithm of first choice in this application. The LMS algorithm can be complex or real depending on the nature of the input/output data and tap weights. Therefore, real LMS algorithm will be applied for the sake of simplicity.

Further, the LMS algorithm is simple and yet capable of achieving satisfactory performance under the right conditions. Moreover, it provides a practical frame of reference for assessing any further improvement that may be attained through the use of more sophisticated adaptive filtering algorithms. LMS adaptive filter is a transversal filter with a single tap weight (and tap weight computation) at each tap in the filter.

8.5. Implementation

The adaptive algorithm in case for the compensation of a degraded beam can be implemented in the adaptive beam forming portion, which is located after the beam former as shown in the figure 5.2 in a general type of a sonar system. The portion array of sensors can have the sensors in various configurations. These configurations could be a linear, planer, cylindrical, conformal and spherical having their own advantages and disadvantages and can be used depending upon the application of a sonar. On board submarines towed array which applies the concept of line array because of its ability to change operating depth and efficiently detects below layer targets.

8.6. Sensor Array

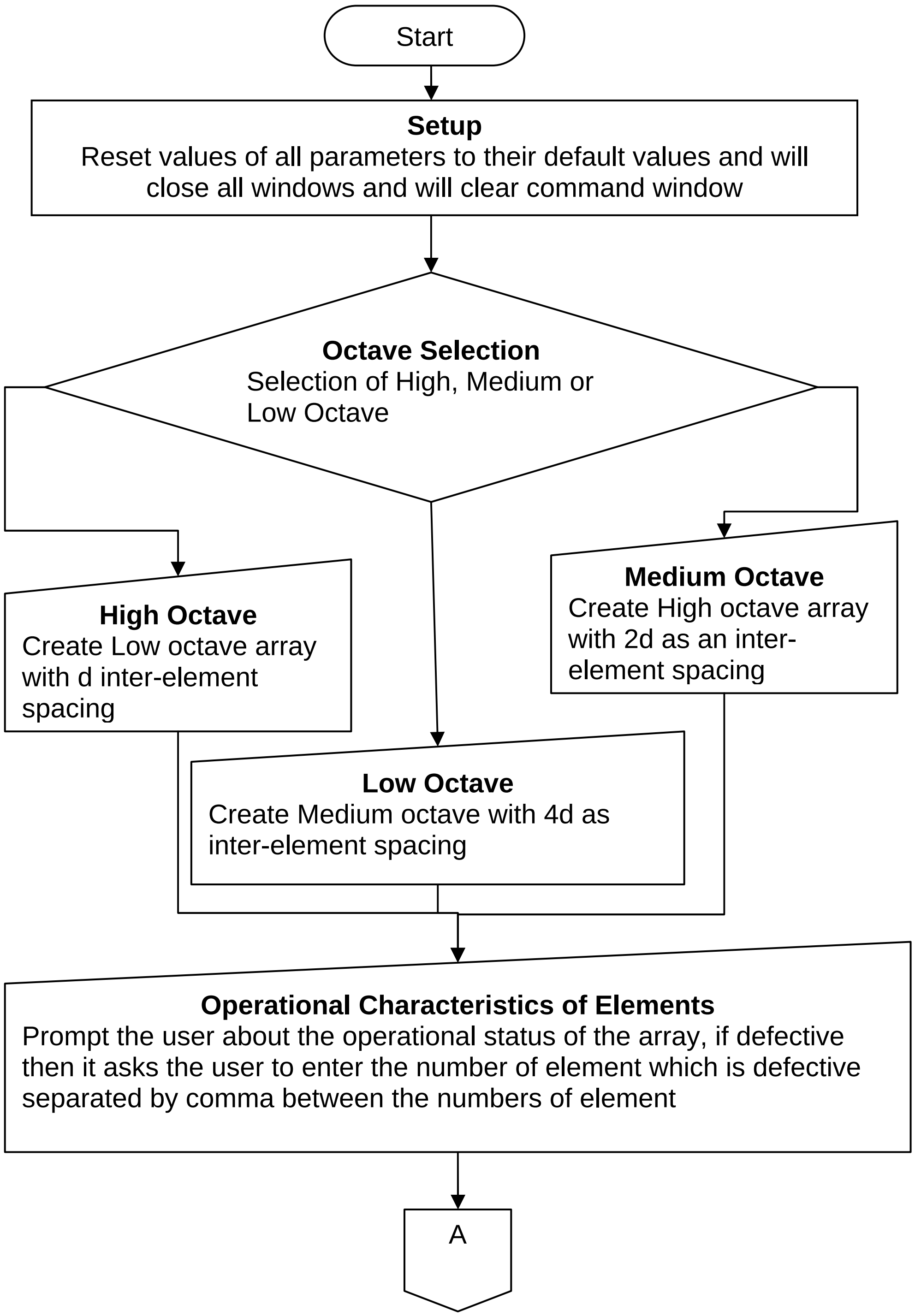
In this project the concept of octave is used in the array sensors similar to the one which is used in towed array fitted on board submarines. The term octave relates to the frequency band coverage of an array. The towed array covers the frequency spectrum of 0 – 880 Hz [1]. Therefore the same spectrum is used in this project as well. In a typical five octave array sonar the operating frequency bands are from 0 – 55 Hz, 55 – 110 Hz, 110 – 220 Hz, 220 – 440 Hz and 440 – 880 Hz. However, for easy development of understanding and simplicity; three octaves sonar is modeled for this project. The octaves are named as High, Medium and Low having frequencies 880Hz, 440Hz and 220Hz respectively. The frequency of 880Hz has been selected as the default High frequency and the subsequent medium and low frequencies have been derived from this High frequency by dividing it by 2 and 4 respectively. The wavelength (Lamda) has been calculated by dividing the speed of sound in water (1500m/s) with that of selected frequency. In all the octaves the inter-element spacing is obtained by dividing the value of lamda by a factor of 2 in order to avoid the diffraction secondaries (Un-wanted repetition of main beam). The total number of elements selected is 32 in high octave to match with the design of actual sonar. However, to keep the inter-element spacing constant, the number of elements is increased to 64 and 128 in the medium and low octave. This increase in the number of elements is automatically done in the sonar model by multiplying it by 2 and 4 respectively.

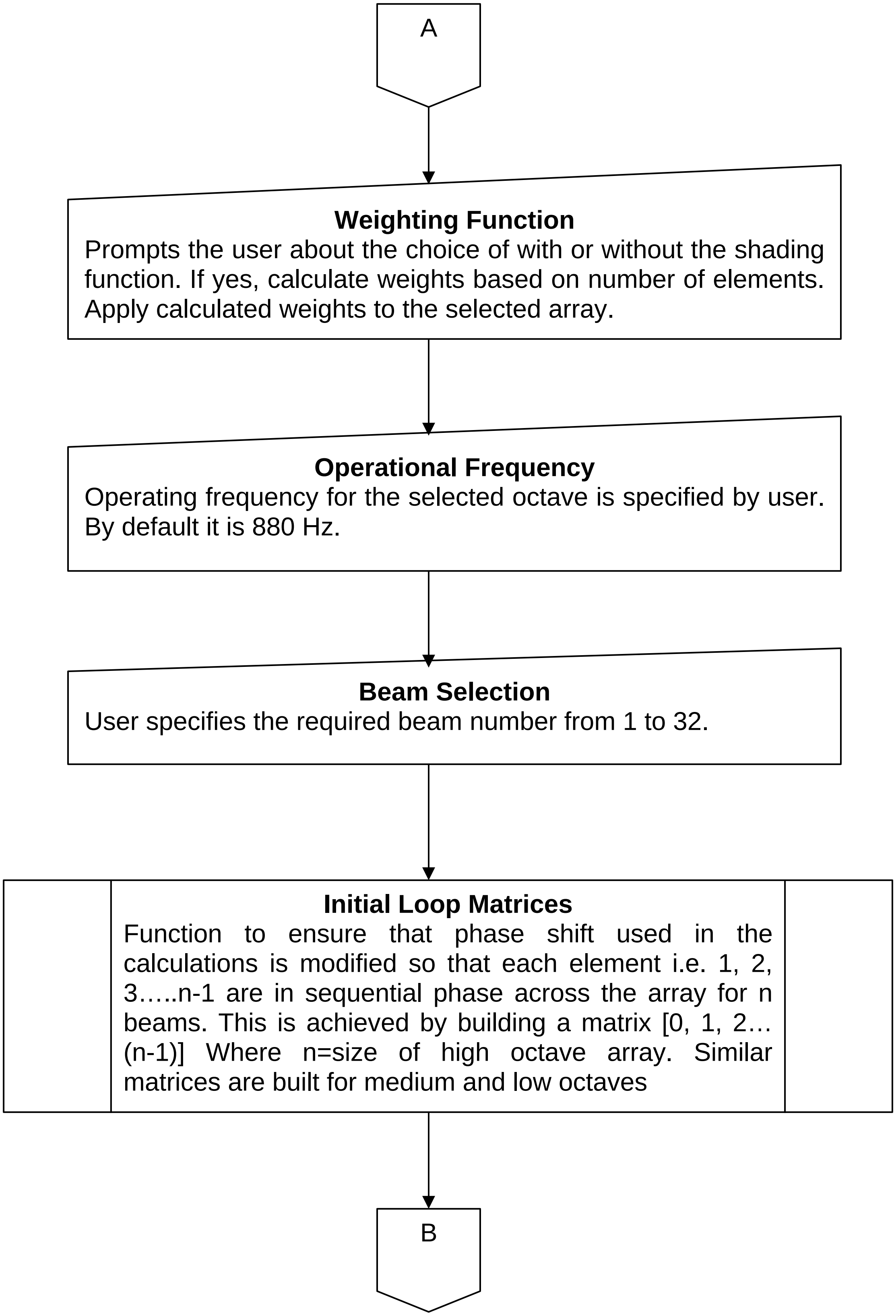
8.7. Beam former

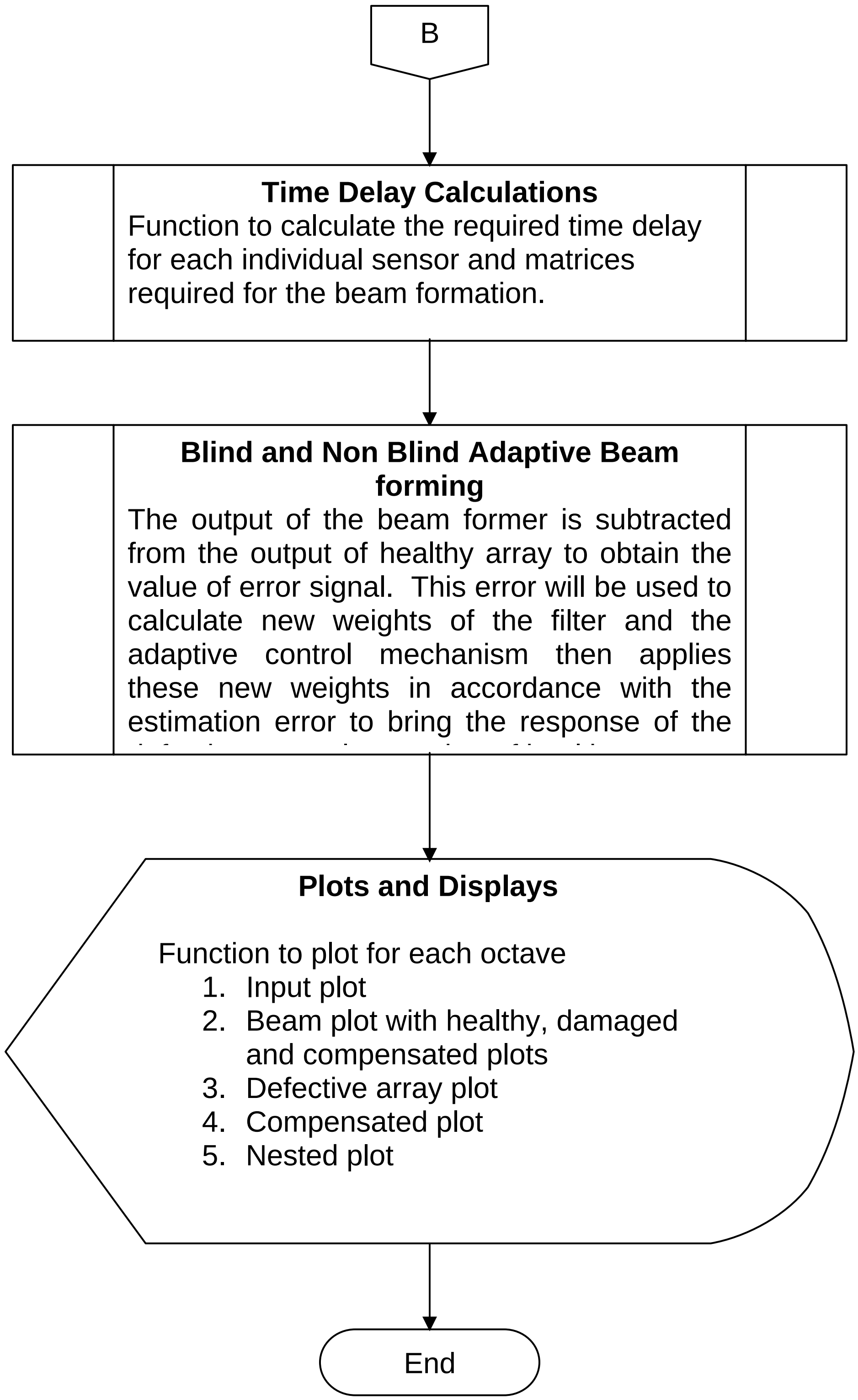
The second stage consists of a beam former. A Beam former is a spatial filter that operates on the output of an array of sensors in order to determine from which direction a sound is coming[18] [19]. It enhances the amplitude of a coherent wave front relative to background noise and directional interference. The "pointing direction" is called the Maximum Response Angle (MRA), and can be arbitrarily chosen for the beams. The goal of beam forming is to sum multiple elements to achieve a narrower response in a desired direction (the MRA). Thus, when a sound is received from a given beam, it gives its direction of arrival. In case of time-domain, beam forming is realized by delaying and summing the shaded outputs of an array of transducers. Shading is simply amplitude weighting which is done to improve the spatial response characteristics of the beams. The beam forming delays are matched to the anticipated propagation delays of a pressure field incident from a specific direction.

In this project, sum and delay beam former is used for this purpose in which the octave selection generates an array of specific number of elements. The required weighted coefficients are calculated for a specified array in case weighting is used. Otherwise, same amplitude signal is fed to all elements. The specified beam number helps to locate the required look angle. Total array is divided in 32 beams covering from 0 to 180 degree. Thus, each beam is covering about 5.625 degrees. A loop matrix is generated for the specified octave, so that appropriate signals can be fed to the individual elements of the array. In high octave selection each individual element is fed a separate signal. In medium, two elements are fed the same signal and in low octave each four consecutive elements are fed the same signal. The information about the operational status of elements is also fed to the beam former, which is being prompted by the user in this case. The beam former then calculates the steering vectors and steering angles and plots the radiation pattern when all the elements are operational and it also plots the radiation pattern when the numbers of elements are defective as specified by the user. Therefore, the degradation in the beam can be observed by comparing the defective beam pattern (due to defective elements) with that of the beam pattern when all the elements are working.

Flow Chart for Beam Formation Software




A
Weighting Function
Prompts the user about the choice of with or without the shading function. If yes, calculate weights based on number of elements. Apply calculated weights to the selected array.
Operational Frequency
Operating frequency for the selected octave is specified by user. By default it is 880 Hz.
Beam Selection
User specifies the required beam number from 1 to 32.
Initial Loop Matrices
Function to ensure that phase shift used in the calculations is modified so that each element i.e. 1, 2, 3…..n-1 are in sequential phase across the array for n beams. This is achieved by building a matrix [0, 1, 2… (n-1)] Where n=size of high octave array. Similar matrices are built for medium and low octaves
B


B
Time Delay Calculations
Function to calculate the required time delay for each individual sensor and matrices required for the beam formation.
Blind and Non Blind Adaptive Beam forming
The output of the beam former is subtracted from the output of healthy array to obtain the value of error signal. This error will be used to calculate new weights of the filter and the adaptive control mechanism then applies these new weights in accordance with the estimation error to bring the response of the
Plots and Displays
Function to plot for each octave
1. Input plot
2. Beam plot with healthy, damaged and compensated plots
3. Defective array plot
4. Compensated plot
5. Nested plot
End

8.8. Analysis and Simulation

The adaptive filters discussed above have been implemented in Matlab. In this simulation, the octave SONAR is taken and the different types of octave SONARS are analyzed. Array of sensors have 32 elements and the operated frequency is 880 Hz. The beam pattern of beam No 16 is analyzed so that the both sides beam pattern can be seen.

We have analyzed the beam pattern when the different number of sensors is defective. It has been observed that when 50% or fewer sensors are defective, it can minimize the error and error approaches to zero.

8.8.1. Non-Blind Adaptive Filter:

The following results were obtained using non-blind adaptive filter. We have analyzed different results but a case has been noted when 11 sensors numbered as 1,3,4,5,6,7,12,14,21,25 and 31 are defective, we have applied Dolph Shelby function as input. The beam patterns (when all elements are operational, when elements are defective and after the compensation using non-blind filter)are shown in Figure 8. Combined beam patterns for the above case are also analyzed for high, medium and low octave.

Individual and combined beam patterns for high octave are shown in Figure 4 and Figure 5. It has been observed that side lobes are 20 db down to the main lobe and they are at ±25 degree and ± 70 degree. So the new detection is impossible. It has also been observed that main lobe is 5 db down for defective elements. When adaptive filter is applied, the error is approximately 2db which is negligible as compared to previous one.

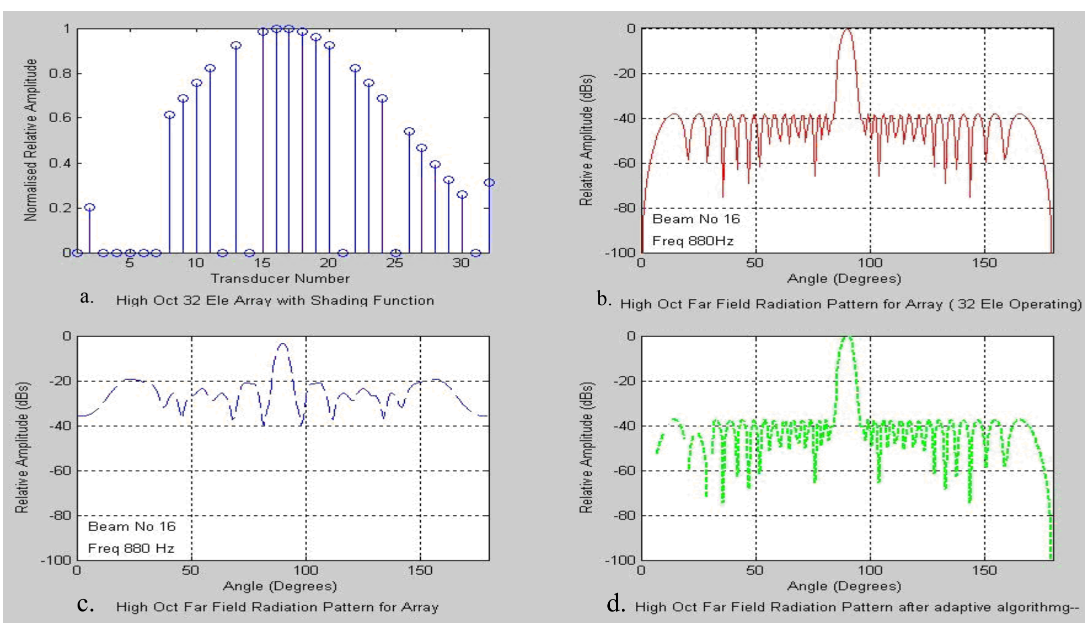


Figure 4: High Octave Beam Patterns

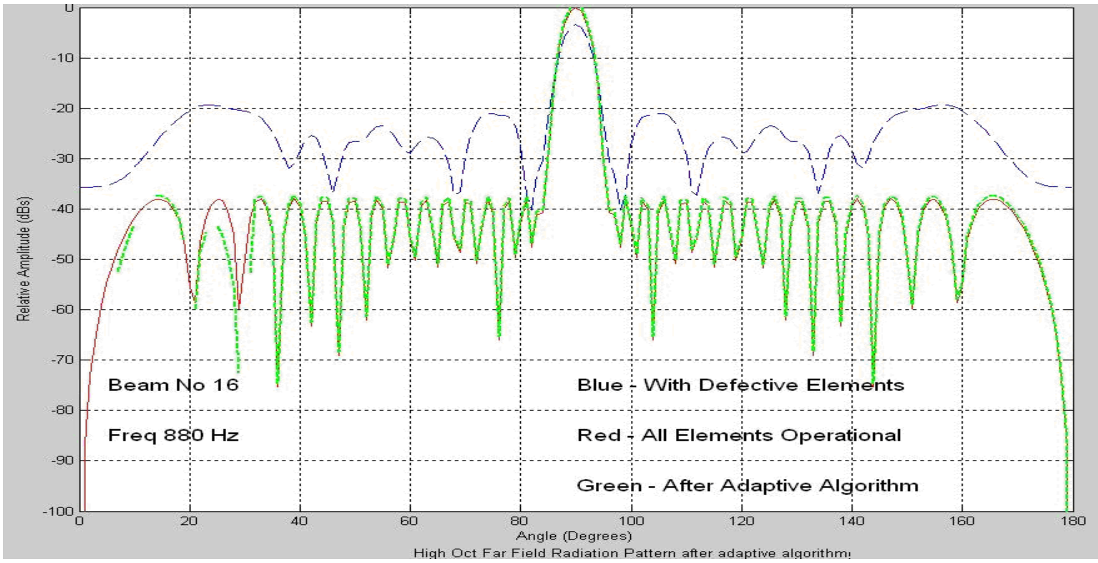


Figure 5: High Octave Combined Beam Patterns

Medium octave analysis is shown in Figure 6 and Figure 7.side lobes are 18 db down to the main lobe and they are at ±20 degree and ± 50 degree. Detection is almost impossible due to interference. Main Lobe is 5 db down for defective elements causing degradation in beam. When adaptive filter is applied, the error is approximately 1.5 db which is negligible as compared to previous one.

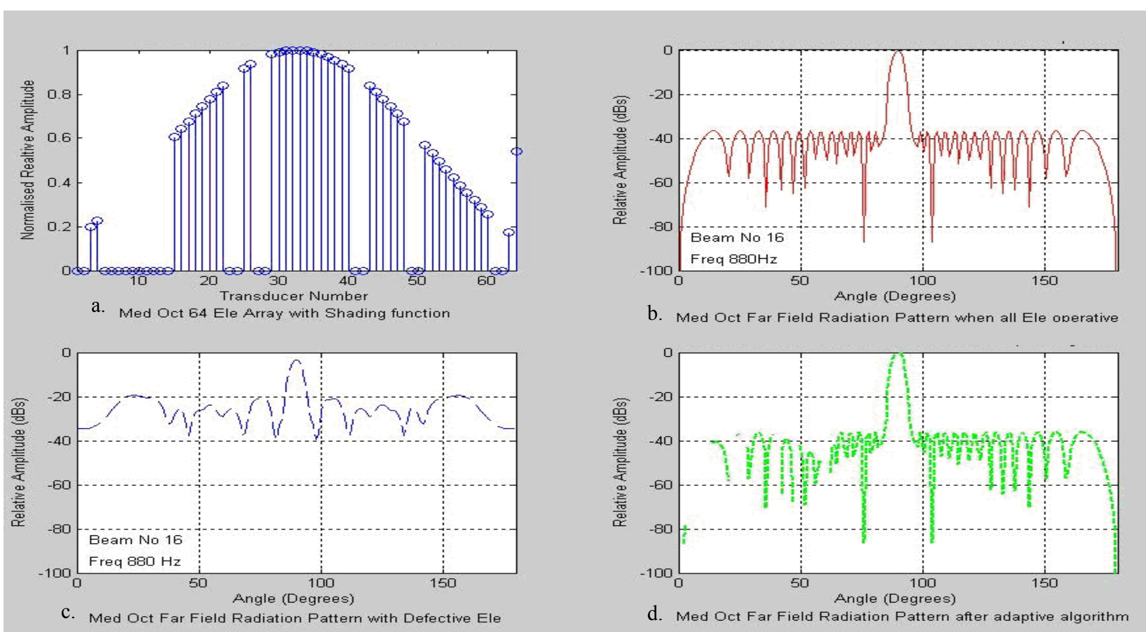


Figure 6: Medium Octave Beam Patterns

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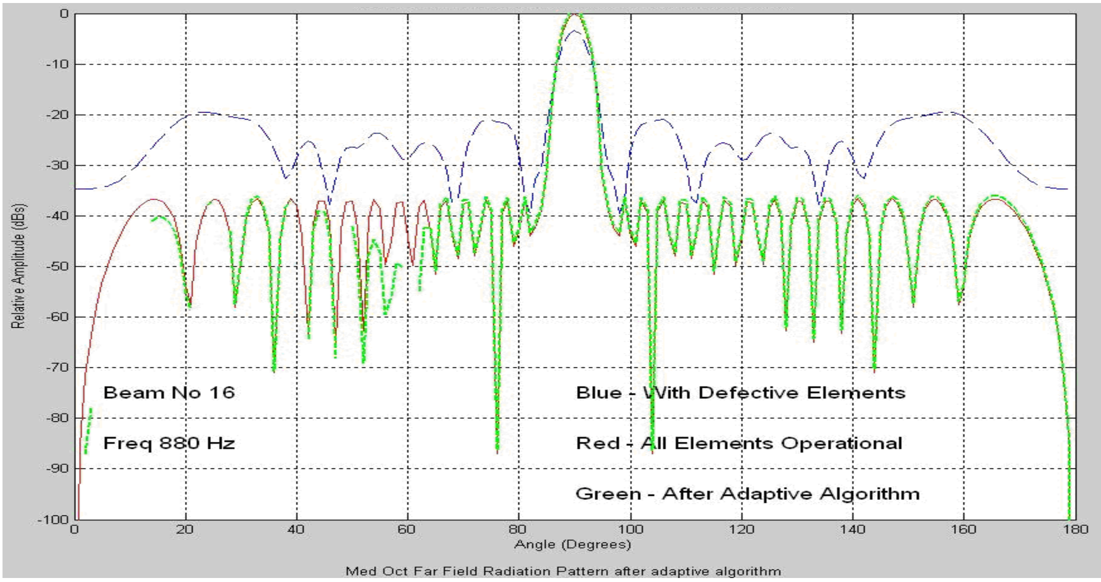

Figure 7: Medium Octave Combined Beam Patterns

Low octave analysis as individual and combined is shown in Figure 8 and Figure 9. This analysis shows that side lobes are 22 db down to the main lobe and they are at ±20 degree, ±40 degree and ± 70 degree. So the new detection is almost impossible. It has been observed that main lobe is 3 db down for defective elements. When adaptive filter is applied, the error is approximately 3 db which is negligible as compared to previous one.

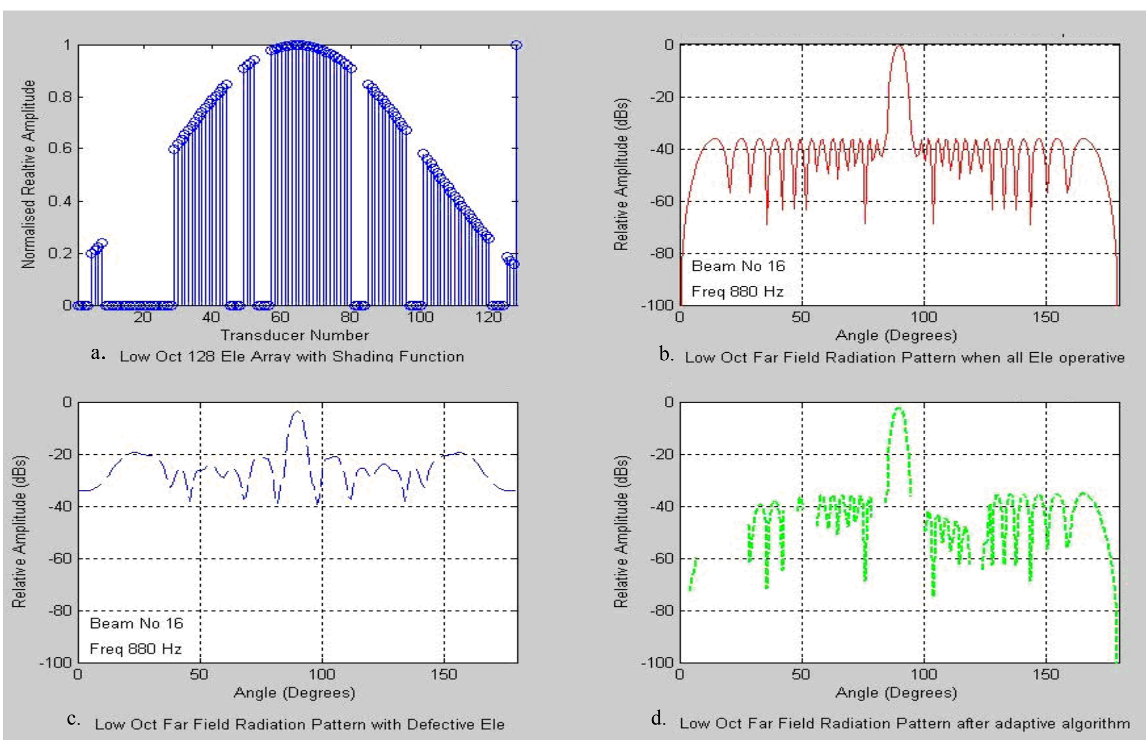

Figure 8: Low Octave Beam Patterns

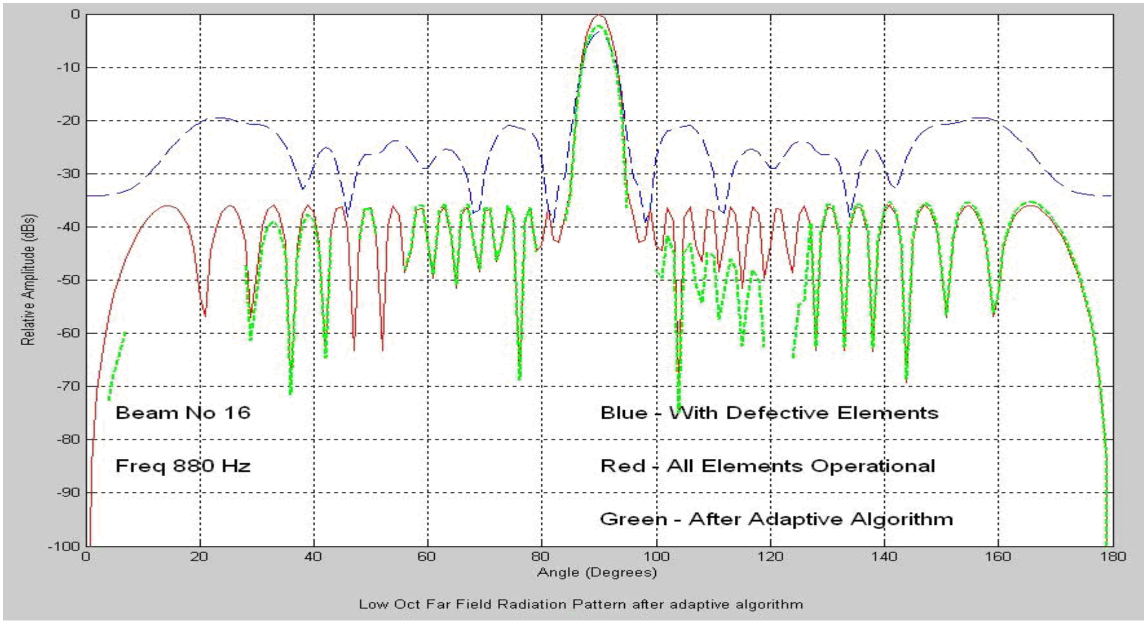

Figure 9: Low Octave Combined Beam Pattern

8.8.2. Blind Adaptive Filter:

The following results have been obtained using Blind adaptive filter. We have analyzed the same case as we discussed for non-blind adaptive algorithm.

Individual and combined beam patterns for high octave are shown in Figure 10 and Figure 11. It is observed that side lobes are 30 db down to the main lobe and they are at ±10 degree It is also observed that main lobe is 1 db down for defective elements. When adaptive filter is applied the error is approximately 1 db which is negligible as compared to previous one

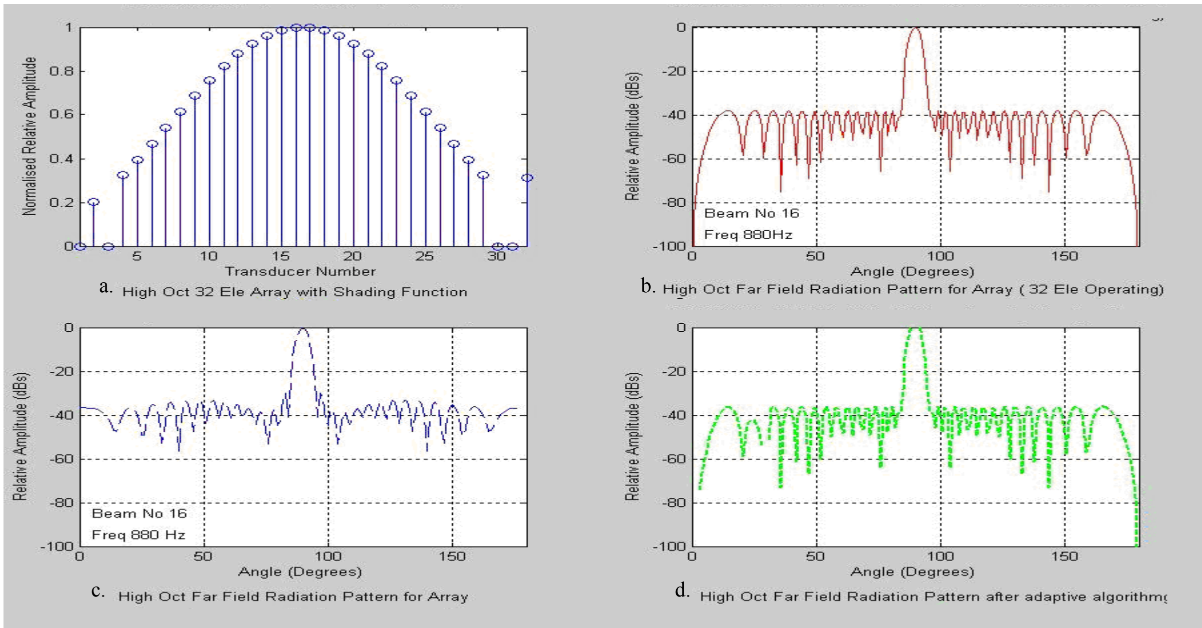

Figure 10: High Octave Beam Patterns

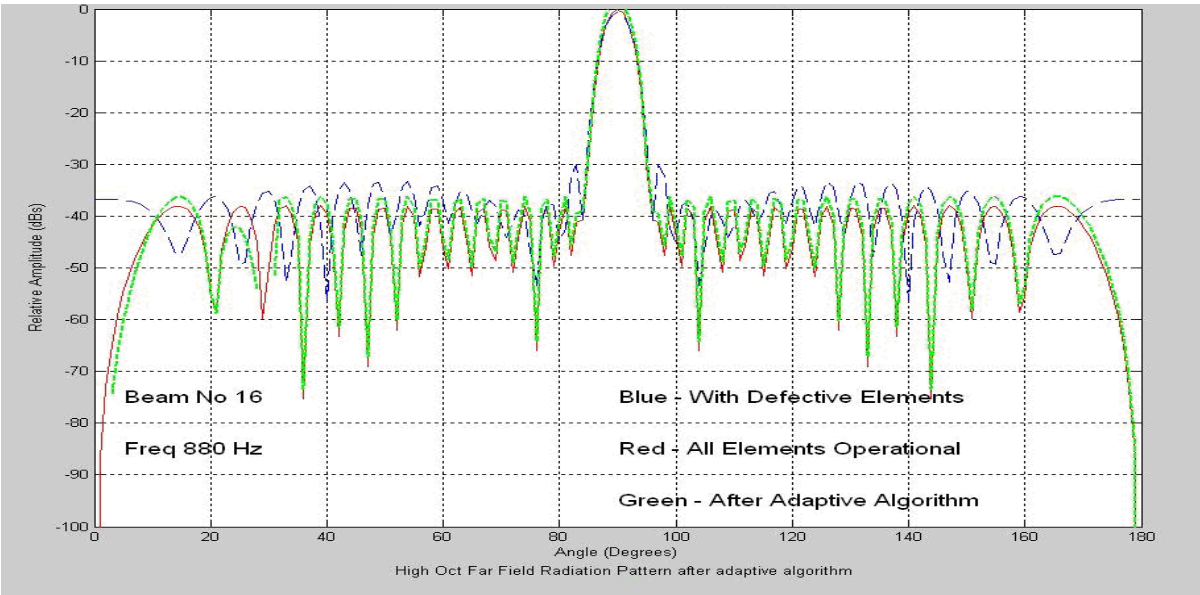

Figure 11: High Octave Combined Beam Patterns

Individual and combined beam patterns for medium octave are shown in Figure 12 and Figure 13.This analysis shows that side lobes are 18 db down to the main lobe and they are at ±20 degree and ± 50 degree. So, the new detection is almost impossible. Main Lobe is 6 db down for defective elements. When adaptive filter is applied the error is approximately 2 db which is negligible.

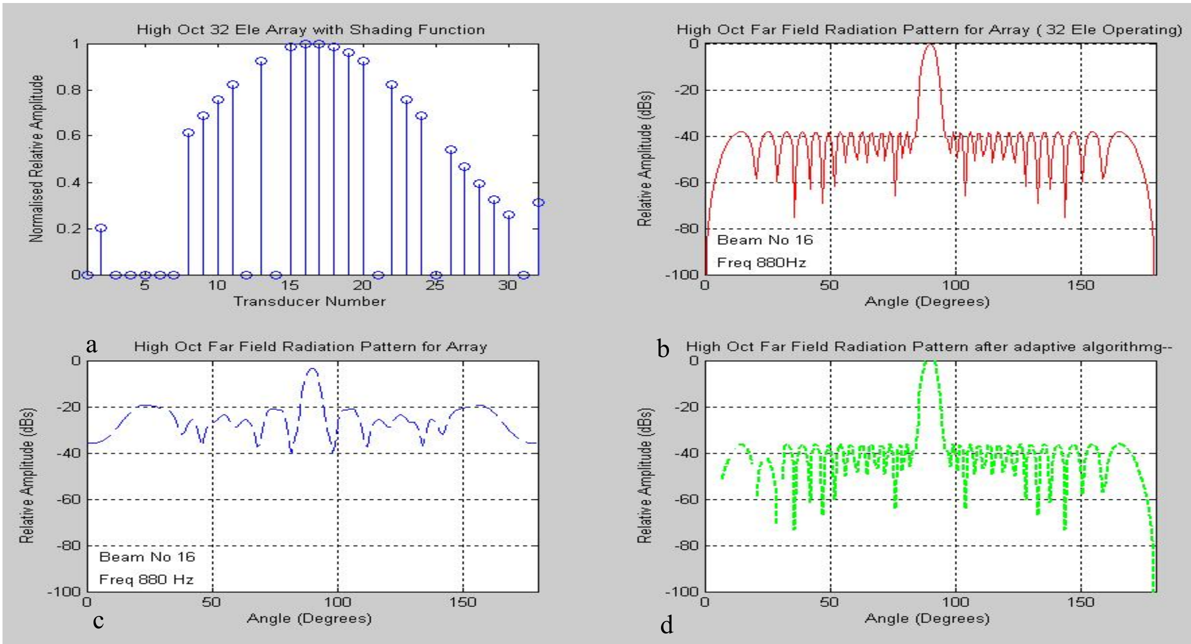

Figure 12: Medium Octave Beam Patterns

Figure 13: Medium Octave Combined Beam Patterns

Low octave results are shown in Figure 14 and Figure 15. It has been observed that side lobes are 22 db down to the main lobe and they are at ±20 degree, ±40 degree and ± 70 degree. So the new detection is almost impossible. it is also clear that main Lobe is 3 db down for defective elements. When adaptive filter is applied the error is approximately 2 db which is negligible.

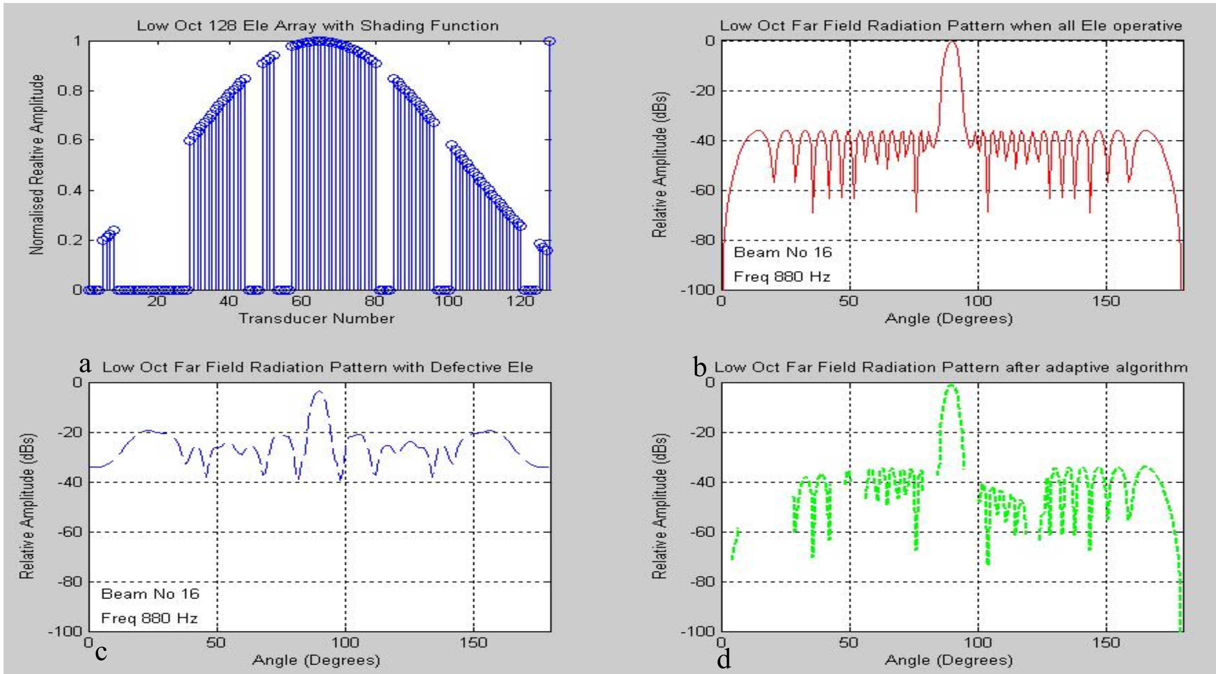

Figure 14: Low Octave Beam Patterns

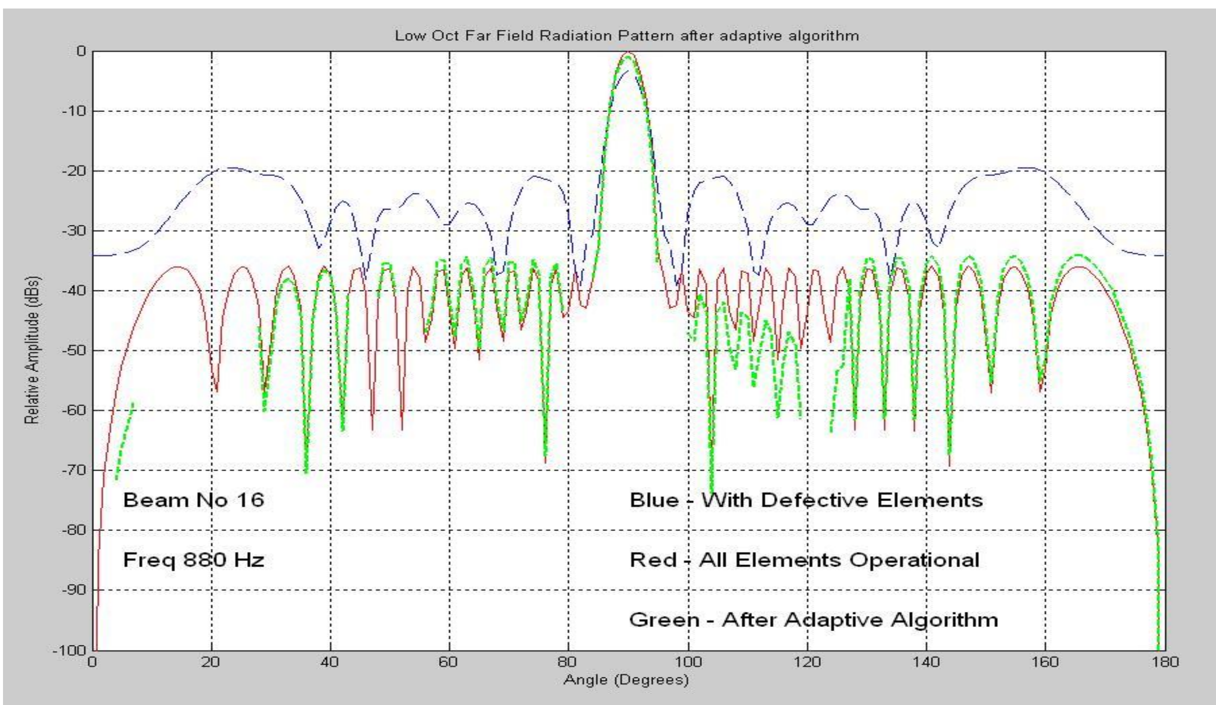

Figure 15: Low Octave Combined Beam Patterns

The maladjustments of the adaptive filter depend upon the step size. If step size is a small, the adaptive process behaves slowly. Various values of step size are taken and the error has been noted against each along with settling time to select the most optimum value of step size. High octave with 32 elements is selected in which element number 5 is considered defective. A weighted input signal is applied and beam number 14 is analyzed for observing the variation in compensated beam error along with the variations in the settling time. These variations of error and settling time are plotted against different values of step size as shown in Figure 16.The graph clearly shows that the error increases initially from 0.0624 to 0.1845. The settling time reduces from 8.8 milliseconds to 2.94 milliseconds from e-1 to e-5 and its value becomes 2.3 milliseconds at e-6 which is the minimum possible achieved by the algorithm. After e-6, the error remains constant but settling time starts increasing. So, the most optimum value for step size selected is e^{-6} to ensure the filter stability in case of varying number of defective sensors.

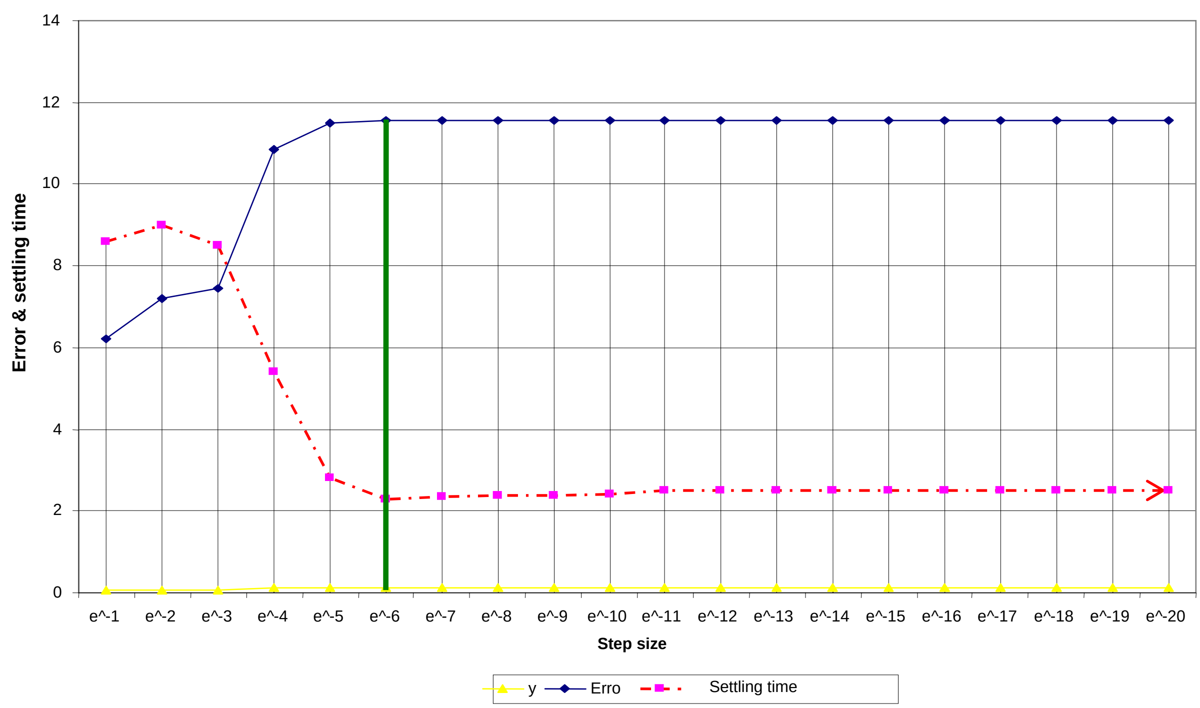

Figure 17: Effect of Step size on error and settling time for Blind Adaptive Filter

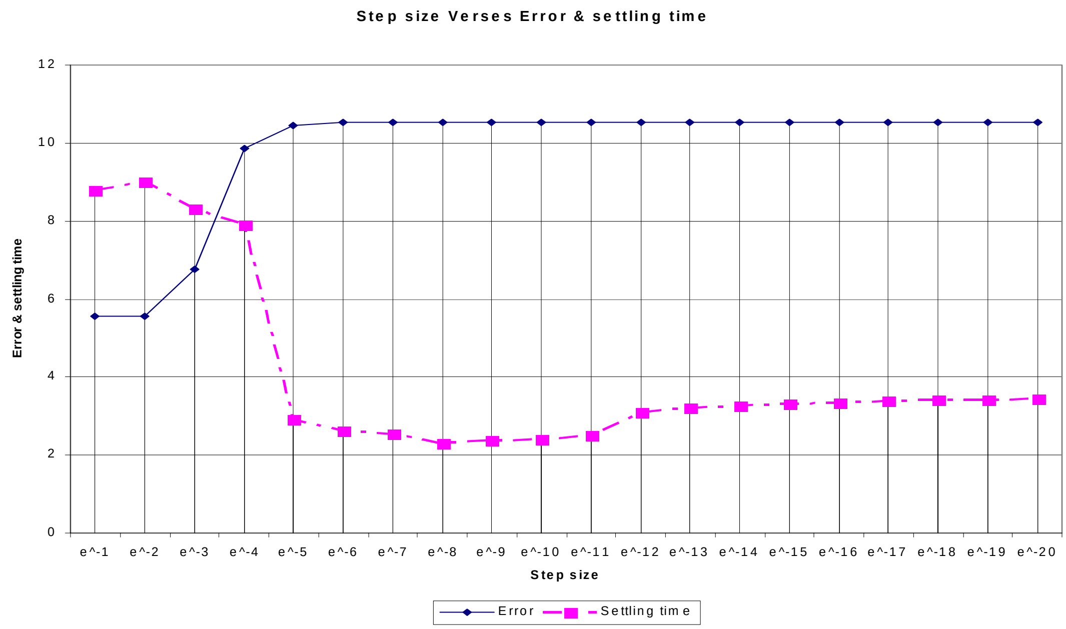

Figure 17: Effect of Step size on error and settling time for Non-Blind Adaptive Filter

A weighted input signal is applied and beam number 14 is analyzed for observing the variation in compensated beam error along with the variations in the settling time. These variations of error and settling time are plotted against different values of step size as shown in Figure 16.The graph clearly shows that the error increases initially from .0574 to 0.1045. For graphical representation (error * 1000), as the step size is reduced from e-1 to e^{-6} and then it remains constant after e-6.

However, the settling time reduces from 8.8 milliseconds to 2.94 milliseconds from e^{-1} to e^{-6} and its value becomes 2.32 milliseconds at e^{-8} which is the minimum possible achieved by the algorithm. After e^{-8} the error remains constant but settling time starts increasing. So the most optimum value for step size selected is e^{-8} to ensure the filter stability in case of varying number of defective sensors.

8.13 Conclusions

In this chapter we described the problem definition, system model, pseudo code to write the code, simulation results and comparisons of these results. From this it is clear that our technique can correct the errors caused due to defective sensors in sonars.

Chapter 9

Conclusions

The sonars used on board submarines normally operate below thermal layer. In most cases it uses a nested multi octave array of sensors having the inter element spacing less than half wavelength of operating frequency. This inter element spacing is vulnerable to change in case of defective elements or degradation in their performance over a period of time. This unaccounted change in inter element spacing causes the change in the beam pattern and thus greatly reduces the performance of the sonar at sea.

In this research work these degradations have been calculated in the form of error and are applied iteratively with the help of an adaptive algorithm in order to reduce it to minimum. The algorithm when applied automatically on the output of the defective array beam pattern, cancel out the errors and bring the beam pattern very close to the beam pattern of a healthy array which is also a desired beam pattern when all the elements are working. In this way it allows the sonar to work at its optimum performance (in terms of error).

This objective has been achieved by implementing a sum and delay beamformer to produce beam patterns generated by a uniform multi octave linear array. The damaged array beam pattern has been simulated by specifying the number of defective elements in the array. Later the output of a healthy array and the output of damaged array are subtracted in order to estimate the difference between the two beam patterns in terms of error signal. This error signal is then used to calculate the weights of the filter and applied in the adaptive process to bring the error close to zero. A comparison between healthy array, damaged array and the compensated beam allows to ascertaining the effect of damaged array on operational performance and the effectiveness of the adaptive algorithm used in the sonar model.

The main effect of defective/degraded elements on array is to increase inter element spacing at some points and array does not remain uniform any more. Increased inter element spacing causes a number of undesired effects like diffraction secondaries, increased side lobe levels and a broadened main beam. Effects on sonar performance due to increase in inter element spacing depends on location and number of defective elements. Beam number or look angle is important, because if defective elements lie in same direction as desired look angle then beam pattern will be affected most. When look angle is changed in a direction away from defective elements the effect starts to decrease.

An algorithm was developed in MATLAB to implement delay and sum beamformer. Algorithm calculates delay required for each individual element based on total number of elements, octave in operation and any tapering function. These delayed signals are then summed up to form beam patterns. The error signal is then calculated and used to establish the new weights of the filter. The weights of the filter are then adjusted to their new values with the help of an adaptive algorithm so that the filter should converge to its optimum solution and brings the error signal very close to zero.

This research has attained the primary purpose of implementing the adaptive filter after the beamformer in order to eliminate the requirement of manual training of sonar array to obtain its optimum performance. In order to accomplish this aim a sum and hold beamformer is used along with an adaptive filter. The objectives, which were achieved, are modeling of sonar and its beam formation in MATLAB, simulation of damaged, healthy and compensated array, calculation of errors due to changing beam pattern and compensation of these errors automatically to make the system work at its optimum level without any human involvement and lastly the effectiveness of the adaptive algorithm.

5.1 Contributions of the thesis

The contributions of this thesis include:

I. Generation of beam pattern using adaptive algorithm, when the body is moving and all sensors are working.

II. When number of sensors are not operational in array of sensors calculation the defective beam pattern.

III. Compensation of beam patterns which are degraded due to defective sensors using adaptive algorithms.

IV. Simulation and comparisons of results to show the accuracy of this technique.

5.2 Future Work

The sonar model is based on a beam former and an adaptive filter. As far as the beam former is concerned a delayed and sum beam former is used in this model, however, other beam formers can be employed for the same purpose for comparative performance analysis. The present system uses a non-blind adaptive algorithm based on robustness, numerical stability; low implementation cost and computational complexity. Whereas, the factors of low convergence rate, gradient noise and maladjustments are considered as the trade off while selecting the adaptive filter. This trade off results in the graceful degradation of bandwidth capacity and system efficiency. Therefore, further expansion is possible in the existing sonar model by enhancing the bandwidth and system efficiency. This could be achieved by some expert system, modifying the algorithm or implementing the blind adaptive algorithms to make the system more efficient and reliable.

Chapter 10

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