Single Sensor Source Localization

Major Project

Submitted in Partial Fulfillment of the requirements

for the Degree of

Master of Technology

in

Electronics and Communication Engineering

(Communication Engineering)

By

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Ahmedabad 382 481

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Declaration

This is to certify that

a. The thesis comprises my original work towards the degree of Master of Technology in Communication Engineering at Nirma university and has not been submitted elsewhere for a degree

b. Due acknowledgement has been made in the text to all other material used.

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Disclaimer

"The content of this thesis does not represent the technology, opinions, beleifs or positions of NSTL (DRDO), its employees, vendors, customers or its associates. The work done in the thesis solely belongs to the author."

CERTIFICATE

This is to certify that the Major Project Report entitled **"Single Sensor Source Localization"** submitted by **Ms. Chitturi Pavani (13MECC04)**, towards the partial fulfillment of the requirements for the award of degree in **Master of Technology** in **Electronics and Communication Engineering (Communication Engineering)** of Nirma University is the record of work carried out by her under our supervision and guidance. The work submitted has reached a level required for being accepted for examination. The results embodied in this major project to the best of my knowledge have not been submitted to any other University or Institution for award of any degree or diploma.

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Acknowledgements

I take immense pleasure in thanking of my mentor Shri. G. V. Krishna Kumar, Scientist-F,HOD,NSTL(DRDO), Vishakapatnam for having permitted me to carry out this project work. I also thank Mr. Padmanabham M. (Scientist-C) for helping me throughout with my project work.

I wish to express my deep sense of gratitude to my guide Prof. Mehul R Naik, Assistant Professor, Institute of technology, Nirma University for being a source of inspiration and for timely guidance during the project.

I would like to express my gratitude and sincere thanks to Dr. P. N. Tekwani, Head of the Department of Electrical Engineering; and Dr. D. K. Kothari, Section Head of Electronics and Communication Engineering for allowing me to undertake this thesis work and for their guidance during the review process. I would like to thank to our Director Dr. K. Kotecha for allowing me to undertake this thesis work.

I would like to express my gratitude and sincere thanks to Ms. M. Radha ,Senior Administrative officer(Grade-II),NSTL, Vishakapatnam, to allow me to carry out my Final year thesis work at NSTL (DRDO). I would like to thank all other team members of Acoustic test center for supporting and guiding me during the thesis.

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Abstract

Passive source localization using single hydrophone has received considerable attention in the last decade particularly in marine biological studies. If proven successful, it should find tremendous applications, both in military and civilian. Underwater vehicles such as gliders, AUVs, ROVs, torpedoes, mines, etc. seldom have more than one sensor fitted on them for carrying out reconnaissance/ threat neutralization operations. Therefore, this paper investigates the usage of a method based on multipath time delays to estimate the slant range between the source and receiver. Time delays estimated theoretically were used to establish the usefulness/ limitations of this method. Surface ships radiated noise signature was subjected to cepstral analysis to estimate time delays between the direct path and the surface reflected path. The time delays thus obtained were used for estimating the slant range between the source and receiver. The comparison between the measured and the estimated time delays was found to be satisfactory. An error analysis was carried out to study the uncertainities added by the ocean environment to the measurements. LS method was also proposed to find out the motion parameters. Simulations were carried for this method using data series obtained from bellhop model.

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Acronyms

NSTL .... Naval Science Technological Laboratories

DRDO .... Defence Research and Development Organisation

SONAR .... Sound NAvigation and Ranging

OAT .... Ocean Acoustic Tomography

AUV .... Autonomous Underwater Vehicle

ROV .... Remotely Operated Underwater Vehicle

HEAT .... Harmonic Extraction and Analysis Tool

EM .... Electromagnetic

IDE .... Integrated Development Environment

MATLAB .... MATrix LABoratory

SNR .... Signal to Noise Ratio

CPA .... Closest Position of Approach

IFFT .... Inverse Fast Fourier Transform

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Chapter 1

Introduction

Naval Science Technological Laboratory (NSTL) was established on 20th August 1969 in vizag under Defence Research and Development Organization (DRDO) for design, development, testing and evaluation of major naval systems such as torpedoes, mines, decoys, targets, launchers and simulators. NSTL has world class facilities such as high speed towing tank, cavitation tunnel, wind tunnel, shock test facility, floating shock platform and various other vibration and noise facilities. NSTL is presently headed by Shri. C. D. Malleswara Rao (Director of NSTL). NSTL also designs stealth technologies for naval platforms. An Acoustic Test Centre was established at NSTL under the warfare technology division for noise evaluation and qualification of various types of equipment, machinery, torpedoes, etc. The aim of this department is to study the acoustic signatures of underwater weapons or marine vessels. Acoustic Test Center has facilities such as anechoic chamber, reverberation chamber and acoustic tank for testing and experimentation purpose. This thesis work has been carried out at the acoustic test center under the guidance of G. V. Krishna Kumar (Scientist-'F'), HOD of Noise Studies division.

1.1 Background

This section discusses in brief about the field of work, its practical applications and various other aspects of this field.

1.1.1 Underwater acoustics

Acoustic waves are a type of longitudinal waves (pressure waves) that propagate by means of compression and decompression of the medium. These waves have the same direction of vibration as their direction of travel. Underwater Acoustics is the study of propagation of acoustic waves in water and the interaction of the mechanical waves that constitute sound with the water and its boundaries. The water may be in the ocean, sea, lake or tank. Underwater acoustics finds its application in the field of underwater communication, SONARs, civil and navy.

As quoted in the book by porter[1] "underwater acoustics is the science of sound in the sea and encompasses not only the study of sound propagation but also the masking of sound by different interfering acoustic phenomena." In underwater acoustics where the medium is bounded by complex interfaces and is inhomogenous, ray theory or normal mode theory has been an indispensible tool for understanding and studying sound propagation. But mostly nowadays ray theory is only used owing to its simplicity and less computations requirement. Research in the field of underwater acoustics requires extensive and costly equipments with atleast one ship and often an assortment of at-sea platforms equipped with sound projectors, receiving arrays and sensors to measure the ocean environment.

1.1.2 Underwater acoustic modelling

In order to characterize the channel one needs to solve the acoustic wave equation, which is a partial differential equation describing the motion of a wave in the medium. The propagation of sound in an elastic medium is described mathematically by solving acoustic wave equation with appropriate boundary conditions. Acoustic propagation whether in shallow or deep waters, can be broadly divided two different scenarios, range independent and range dependent. Typically range independent propagation models assume that the sound speed varies only with the depth and it varies very little or no variations in the horizontal distance. On the other hand the range dependent propagation models take even the horizontal variation of sound speed also in to consideration while estimating the propagation losses between the source and

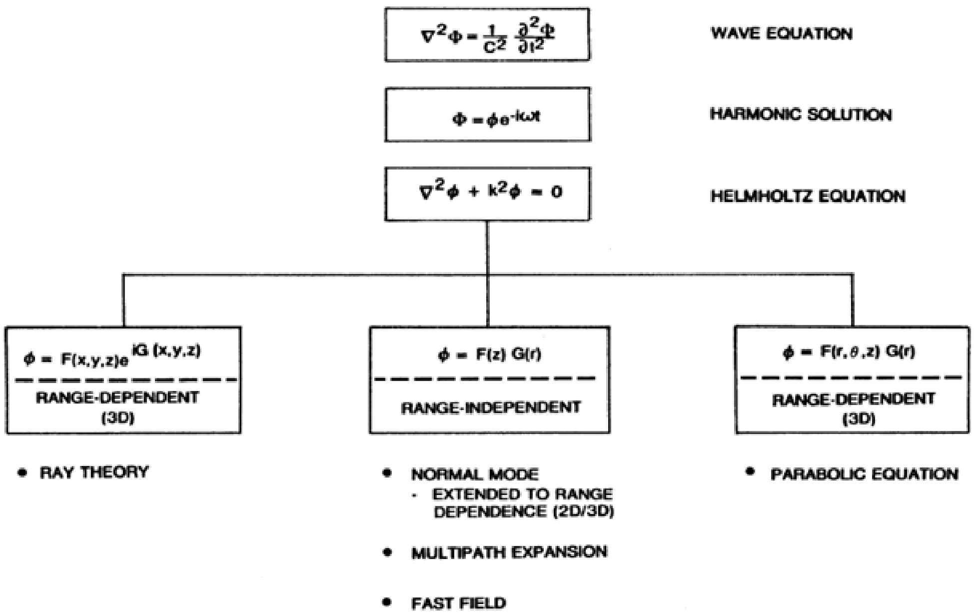

Figure 1.1: Summary of relationship among theoretical approaches for propagation modelling (Jensen and Krol, 1975).

receivers. Summary of relationships among theoretical approaches for propagation modelling is presented at Figure.4.1 (Jensen and Krol, 1975). Currently underwater acoustic systems such as sonars or communication modems uses higher frequencies ranging from couple of kHz to tens of kHz. Therefore in the current thesis the analysis was restricted to only higher frequencies 20 kHz. In this frequency range of interest models based on ray theory are more appropriate in terms of accuracy as well as computational time. Therefore the details of propagation modelling are only restricted to only ray theory.

1.1.3 Behaviour of acoustic waves in ocean environment

Acoustic waves exhibit the phenomenon of refraction, reflection and diffraction. The airocean surface interface is influenced largely by the wind and can range from flat (sea state 0) to the mountaineous seas caused by severe storms (sea state 6). The surface is extremely dynamic and contributes significantly to the variability of the marine environment. The other interface, the ocean seafloor, is relatively stable over time. Its movement depends on local water currents and occasional seismic activities. There are exceptions to this such as undersea volcanoes and mud slumps, attributed to the built of seafloor material into unstable slopes, similar to terrestrial rock slides. The surface interface usually consists of air and water (and, in some places, ice and water). The bottom interface consists of water and materials that range from mud of all densities to knife-edged hard coral and volcanic rock. The speed of acoustic waves in water is 1500 m/s and in air is 340 m/s. It varies from 1450m/s to 1550m/s depending on the salinity, pressure and temperature of the water.

Effect of temperature on acoustic waves propagating in ocean environment

The variation of speed of sound most importantly varies due to change in temperature. Sound travels faster in warm water than in cold water and is very influential in some parts of the ocean. For change in 1 degree of temperature, the velocity of the acoustic waves in water increases/decreases by 4.6m/s. Temperature has both spatial and temporal effect on the sound speed profiles. In the sound speed profile, the mid layer of ocean is dependent on temperature. The temperature of this layer decreases from surface to the bottom. Thus, the velocity of this layer decreases as the depth increases.

Effect of salinity on acoustic waves propagating in ocean environment

After temperature, salinity plays a major role in the ocean environment. For change in 1 ppt (parts per thousand) of salinity, the velocity of the acoustic waves in water increases/decreases by 1.3m/s. Sound speed in the oceans that is dependent on salinity varies seasonally and spatially. Salinity has both temporal and spatial variations.

Effect of pressure on acoustic waves propagating in ocean environment

Pressure that can be derived from depth does not vary significantly from season to season or from place to place; it changes only with depth. The effect of pressure however is much less as compared to temperature and salinity. For change in 1 meter of depth, the velocity of the acoustic waves in water increases/decreases by 0.016m/s. There are many equations for calculating the sound speed profiles based on these three parameters. The equation required can be chosen on the basis of accuracy required

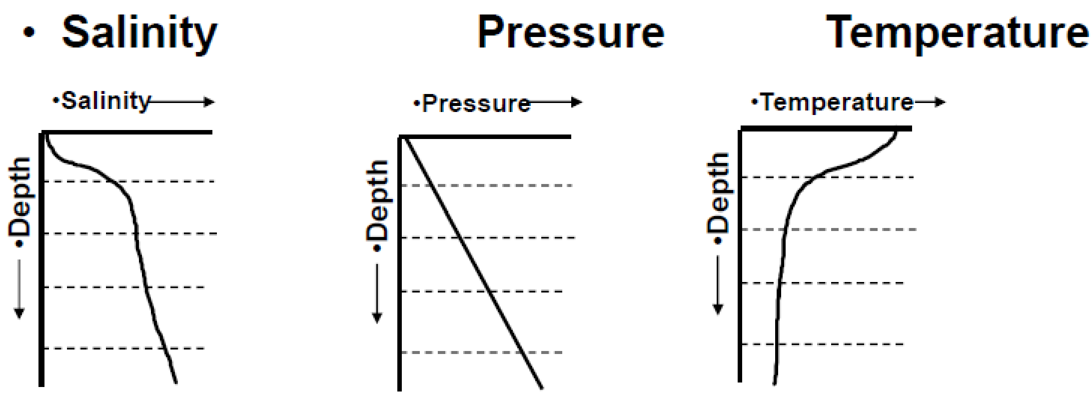

Figure 1.2: Effect of salinity, temperature and pressure on sound speed profile with respect to depth

for calculating the sound speed profile. One of these standard equation has been described. The sound velocity in the ocean can be approximated with the following empirical formula[1]:

$C = 1448.96 + 4.591T - 0.05304T^2 + 0.0002374T^3 + 0.0160Z + (1.340 - 0.0102T)(S - 35) + 1.675 \times 10^{-7z} - 7.139 \times 10^{-13} \times TZ^3$

C= Speed of sound (m/sec).

T=Temperature (Degree Celsius).

S=Salinity (parts per thousand).

Z=Depth (meters).

Depending on these parameters an ocean's sound speed profiles are obtained. These sound speed profiles affect the acoustic transmission paths to a large extent. Also depending on shallow or deep waters, the sound speed profiles are different.

1.2 Sound speed profile

The sound speed profile is made up of several characteristics parts depending on the variability of temperature with depth (Figure 1.2). They are briefly explained as follows:

- Surface channel: Generally, the ocean temperatures are constant for the top most layer of the profile. Such layer depths can be ranging from few meters in shallow waters to tens of meters in deep waters. This essentially means that the sound speed is supposed to be constant in these layers. However the sound increases with depth. Therefore, although the temperature is constant as the water depth is increasing the sound speed also increases thereby forming a layer of positive sound speed gradient which is called surface channel. Such channels in the top layers are good for the acoustic propagation and the systems work well if both source and receivers are placed in this channel.

- Thermocline layer: This is a layer having monotonous variation of temperature with depth. The temperature of this layer decreases with depth. Therefore, sound speed also decreases in this layer as change temperature dominates the change in depth.

- Deep channel: This layer is present between thermocline and deep isothermal layer. This happens only in deep waters of large ocean basins with depths averaging anywhere between few hundred meters to few kilometres.

- Isothermal layer: As the name suggests, the temperature of this layer remains constant throughout the layer. The deeper layers in the ocean are approximately isothermal. Sound speed in this layer increases linearly with depth, because of hydrostatic pressure.

1.3 Sound Propagation in Ocean Waters

For analysing the propagation of acoustic waves in ocean environment, it is required to understand the ocean behaviour. In simulation, ocean can basically be modelled in two ways:(1) Unbounded or ideal ocean and (2) Bounded or practical ocean.
In unbounded ocean, an acoustic wave gets attenuated only due to geometrical spreading. Rest all ocean parameters are considered to be at constant state and no other factors are affecting the transmission of acoustic waves in this case.

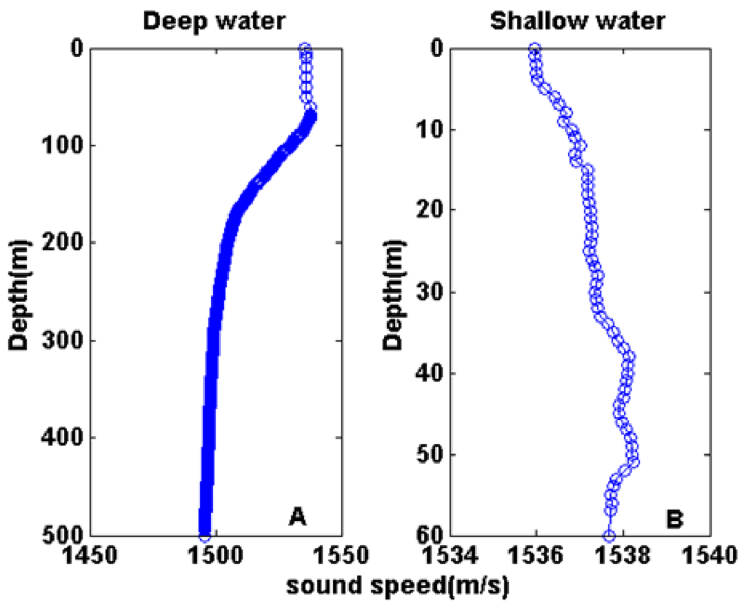

Figure 1.3: Sound speed profiles for deep and shallow water environments off vizag coast

In bounded ocean, these acoustic waves get further weakened due to factors such as transmission loss(due to absorption, scattering, refraction and reflection), multipath effect and doppler shift if the source is moving and reverberation.

1.3.1 Sound propagation in shallow waters

The average depth of shallow waters for modelling the channels is considered to be 200 meters. The principle characteristics of shallow water propagation are that the sound speed profile is downward refracting of nearly constant over depth, meaning that long range propagation takes place exclusively via bottom interacting paths. In shallow water, surface, volume, and bottom properties are all important. They are spatially varying and the oceanographic parameters are also temporally varying. These parameters are generally not known in sufficient detail and with enough accuracy to permit long-range predictions in a satisfactory manner. Since the seafloor is a lossy boundary, propagation in shallow water is dominated by bottom reflection loss at low and intermediate frequencies ($<$ 1 kHz) and scattering losses at high frequencies.

There is a critical frequency below which the shallow-water channel ceases to act as a waveguide, causing energy radiated by the source to propagate directly into the bottom. In the case of shallow water, the increased penetration of sound into a lossy seabed with decreasing frequency causes the overall attenuation of waterborne sound to increase with decreasing frequency. From experiments carried out in shallow water, we can observe high attenuation at both high and low frequencies, while intermediate frequencies have the lowest attenuation.

1.3.2 Sound propagation in deep waters

The average depth of deep waters for modelling the channels is considered to be 2000 meters. In deep water propagation, there is an upward refracting sound speed profile that allows long range propagation without much of bottom interaction. The ray paths obtained are either refracted- refracted or refracted surface- reflected.

There are two ranges for propagation in deep waters. At shorter ranges, there is a nearly straight-line path between the source and the receiver. At longer ranges, propagation occurs in the mixed layer, involving repeated surface reflections and leakage out of the channel to a receiver below. At still greater ranges, convergence zones are seen along with straight line path and refracted paths.

1.4 Factors affecting sound propagation in ocean environment

Geometric spreading is used to describe the decrease in intensity and apparent weakening of the signal due to the spreading of the energy as it gets farther from the source. This occurs because the increase in the area of the surface is proportional to the square of the distance from the source, every time the distance is doubled, the area increases fourfold and the intensity decreases fourfold (same energy, larger area). Depeding on the depth of the ocean and intersinty of source level, this spreading can be cylindrical or spherical loss. The equation for geometrical spreading losses for cylindrical and spherical losses can be given as[2]

$$TL = 20log\left(\frac{R}{R_{1m}}\right)$$
$$TL = 20logR$$

Absorption of sound waves in ocean water because it is not a perfect fluid and due to conversion of energy of these molecules into heat energy, they suffer losses weakening the sound energy.

Reflections of acoustic waves are due to sea surface and sea bed. These reflections may not be specular. The sea bed remains almost constant but the sea surface keeps on changing due to different currents causing the surface waves.

Refraction also play a role in attenuation of sound. The speed of sound in ocean is dependent on temperature, pressure and salinity. If all three parameters are constant, then the sound travels in a straight path. If any these or all of these parameters change, then the speed of sound will change causing bending of the sound. The change in the direction of the sound due to changes in sound velocity is called refraction. In case these parameters vary, the sound takes a curved path. In real ocean environment, these parameters vary throughout the depth of the ocean.

Scattering is a physical process where sound is forced to deviate from a straight trajectory by one or more paths due to localised non uniformities in the medium through which they pass. In real ocean environment, the sea surface is not smooth but has rough surfaces. Due to irregularities at the sea surface and sea bed, scattering of sound waves occur causing formation of one or more paths.

Reverberation also distorts the acoustic waves. When sound strikes a surface at an angle, it becomes spread out due to the geometry of the sound and the angle and roughness of the surface.

1.4.1 Underwater acoustics applications

Military applications

Most of the research and industrialisation effort in underwater acoustics was long linked to military applications. These systems are therefore mostly aimed at detecting, locating and identifying two types of target: submarines and mines. Sonar is detecting, locating, classifying and tracking targets radiating low level signals in the presence of noise.

SONARS

Soanrs are one of the most important applications of the underwater acoustics and amongst the most widely used equipments. Military sonars are classified into three main categories, depending on their mode of operation, one is active, second is passive and third is mine hunting active sonar.

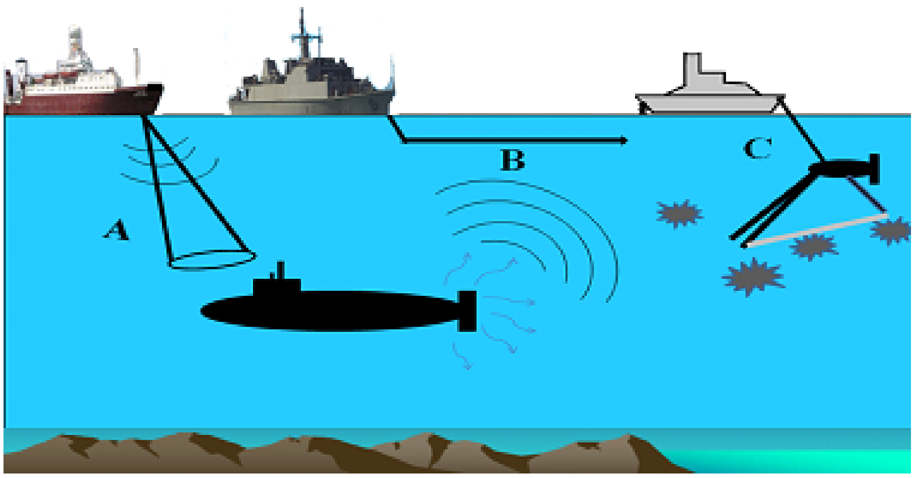

Figure 1.4: Schematic depiction of three types of sonars

- Active sonar: These sonars transmit and receive echoes from a target. The measured time delay or the Doppler imposed by the target or own platform is used to estimate the distance between sonar and its target there for wave forms which are suitable and sensitive to Doppler are being used in these applications. Mine hunting is a type of active sonars with very high resolution. They are used to detect and identify the mines laid the wave forms used in this case are desired to be Doppler insensitive

- Passive sonar: Passive sonars are used to detect the noise radiated by a target vessel. Their main advantage of this system is total stealth. They can be on submarines as well as on the ships hunting them. They work at very low frequencies, between a few tens of Hertz and a few kilohertz. Passive sonars are used as surveillance networks, airborne systems, mines and Torpedoes [2].

These sonars transmit and receive echoes from a target. The measured time delay or the Doppler imposed by the target or own platform is used to estimate the distance between sonar and its target there for wave forms which are suitable and sensitive to Doppler are being used in these applications. Mine hunting is a type of active sonars with very high resolution. They are used to detect and identify the mines laid the wave forms used in this case are desired to be Doppler insensitive

Civilian applications

The following are the subdivisions of various civilian applications

- **Navigation**
  Navigation safety has contributed in a major way to the initial development of underwater acoustics, and has been the starting points of many systems. Acoustic beacons, single-beam bathymmetric sounders are used in navigation and mapping.

- **Fisheries acoustics**
  Marine fisheries make extensive use of underwater acoustics techniques. Modern vessels for industrial and small scale fishing are always heavily equipped with sonar systems. The first function of these acoustics systems is to detect and locate fish schools and to aid in their capture.

- **Marine geology and seafloor mapping**
  Marine geosciences have been largely secondary to the development of tools dedicated to acoustic investigation. Depending on their structure and processing

principles these tools can provide sonar images of seafloor features .Three types of acoustic system are used in marine geosciences; sidescan snares, multibeam sounders and sediment profilers.

- **Physical oceanography**

  Physical oceanography uses low-frequency acoustics techniques to measure the physical characteristics of water masses at medium scale. This method, known as Ocean Acoustic Tomography (OAT). OAT measures the propagation time of multiple paths between a transmitter and a receiver on each side of the oceanic area to study. The major interest of acoustic ocean tomography is that allows quasi-instantaneous monitoring of large area (involving depth-profiling probes deployed from a ship).

1.5 Motivation

Underwater acoustics is the study of propagation of sound waves by compressions and rarefaction of the medium in this case water. This field is not yet completely exploited and has much scope in the future. A difficult aspect of this field is to model the real life scenario of ocean and its properties into an experimental setup for the purpose of research. This field finds its main applications in navy and military.

The topic of single sensor passive source localization in underwater acoustic signal processing is receiving enormous attention among the world scientific community as underwater vehicles are getting miniaturized. Thus, a single hydrophone should be capable of capturing the radiated noise of these underwater vehicles and should perform the same function as complex passive sonar systems. This radiated noise data can be used to estimate the parameters of the underwater vehicles such as its range, slant range,bearing and velocity of a moving vehicle.Single sensor source localization has various potential applications in marine,navy and civil.

1.6 Statement of purpose

Passive source localization using single hydrophone has received considerable attention in the last decade particularly in marine biological studies. If proven successful, it has tremendous applications in both military and civilian applications. Underwater vehicles such as gliders, AUVs, ROVs, torpedoes, mines, etc., seldom have more than one sensor fitted on them for carrying out reconnaissance / threat neutralization operations. Therefore, this thesis investigates the usage of a method based on multipath time delays to estimate the slant ranges between the source and receiver. Time delays estimated theoretically were used to establish the usefulness /limitations of this method. Surface ships radiated noise signature was subjected to cepstral analysis to estimate time delays between the direct path and surface reflected path. The time delays thus obtained were used for estimating the slant range between the source and receiver. The comparison between the measured and estimated time delays was found to be satisfactory.Also error analysis was carried out to find the propagation of error from time delays to slant ranges. Also least squares method has been proposed to find out the motion parameters such as slant range, velocity and time delay at the closest position of approach. Simulations performed have proved the validity of LS method to estimate the motion parameters.

1.7 Author's contribution

Experiment was conducted where ship was transiting above a single hydrophone. The hydrophone captured radiated noise of ship. Thus cepstral analysis was performed on this radiated noise to obtain multipath time delays. A method was formulated to calculate the slant range using these time delays. Ray based model was used to theoretically estimate time delays in the same conditions and using these time delays also slant range was estimated. Thus, measured slant ranges were compared with estimated slant ranges and necessary conclusions were drawn. An error analysis was carried to find out the propagation of errors from time delays to slant ranges.

1.8 Overview

Chapter 1 is basically an introduction to the overall thesis briefly describing all the concepts theoretically. It briefly describes about the area of research.

Chapter 2 Gives a brief description of important concepts such as cepstrum which forms the basis of this thesis. Using these concepts, research work has been carried out.

Chapter 3 is the significant part of thesis. A real time radiated noise of ship is being used to estimate the range between the vessel and the ship using power cepstrum and has been verified using theoretically estimated time delays from bellhop model.

Chapter 4 Shows error analysis of the slant range estimations in order to get the accuracy of TDE using cepstral technique. Also shows a new technique based on LS method for obtaining motion parameters such as time delay at CPA, Range at CPA and velocity of moving source.

Chapter 5 gives necessary conclusions about the work done by author. It also gives an insight about the future scope of the work.

1.9 Summary

The following chapter gives an overall description of the field of research. Underwater acoustics is the field based on which this thesis is carried out. Acoustic waves are clearly preferred over EM waves due to less absorption of these signals in waters. Acoustic waves are affected by temperature, pressure and salinity in ocean environment. This effect can be shown by an equation given in previous section. It can be understood how difficult it is to model the real ocean environment. This is the reason due to which this field is not yet completely explored. Underwater acoustics plays and integral role in military and navy.The aim of single hydrophone source localization is to provide range /depth using the information contained in the time series received by the hydrophone. The first step of source localization is estimation of the time difference of arrivals between various multipath arrivals from the received data of a hydrophone. Various methods have been proposed for time delays estimation (TDE) such as autocorrelation method, MFP (match field processing), Lloyd mirror effect and cepstral analysis. The second step is to formulate various algorithms to obtain the motion parameters such as range, depth, velocity, bearing etc. Radiated noise of a surface ship has been obtained from a single hydrophone. Cepstral analysis has been performed on radiated noise to obtain measured time delays. These time delays have been used to estimate slant range between ship and sensor. These slant ranges have been verified estimated time delays of the ray model. They have found to be similar.

Chapter 2

Literature survey

2.1 Introduction

The process of determining the location of an acoustic source with respect to some reference frame passively is known as passive acoustic source localization.Source localization has wide range of applications from remote sensing to global positioning systems. In underwater acoustics, source localization finds its applications in (i) marine sciences: localization and tracking of marine mammals (ii) threat neutralization operations, SONARS and on-board surveillance in naval systems (iii) localization of air-crafts. Various schemes have been proposed for source localization.For near field sources, localization ban be achieved using the information of time difference of arrivals (TDOA) of the acoustics signal. The same method called as time delay estimation method (TDE) has been used in the present thesis.It can be understood later that accurate TDE is significant for motion parameters estimations. Basically source localization using TDE method can be done in 2 steps (i) TDE from the received time series data (ii) Implement algorithm for estimating motion parameters using the multipath time delays obtained from TDE. In order to understand the Time delay estimation method, the concept of multipath is required to be understood clearly.

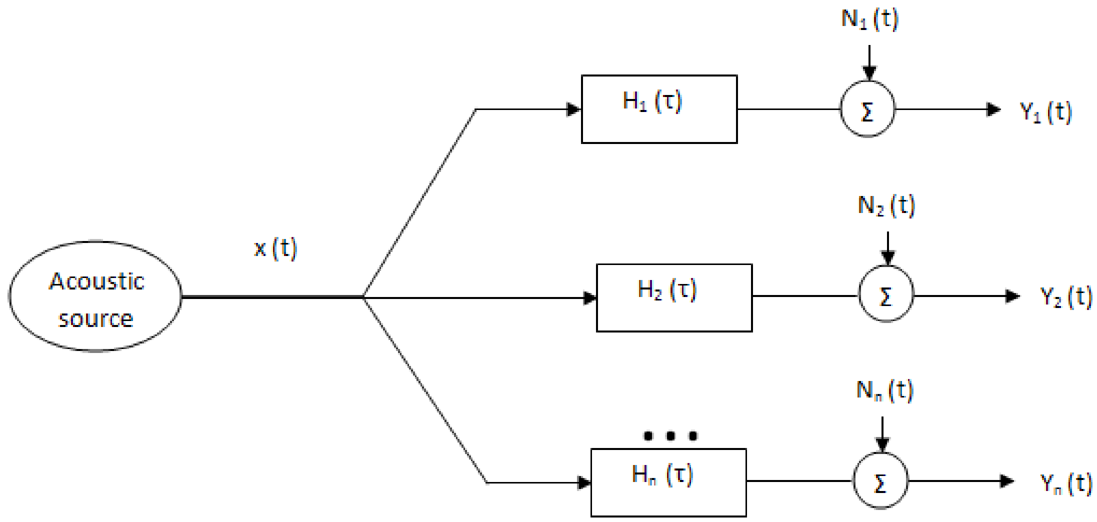

Figure 2.1: Multipath model of communication channel

2.1.1 Multipath effect

when an omnidirectional source emits a signal, it travels through various paths and gets convolved with the channel impulse response and also some noise gets added to the signal. Thus at the receiver, a multiple delayed versions of the same signal is obtained.The following equation shows signal obtained at the receiver.

$$S(t) = \sum_{k=1}^{N} H_k(t) * X_k(t - \tau_k) + N_k(t)$$

where S(t): time series obtained at the receiver

$H_k(t)$: Channel impulse response of each delayed version of signal

$X_k(t - \tau_k)$: Delayed signal $N_k(t)$:uncorrelated noise added to each delayed version of signal Unlike in air, ocean is bounded by seabed at the bottom and sea surface at the top. Thus acoustic waves while traversing across ocean suffers multiple reflections from seabed and sea surface resulting in the multipath effect. Thus, when a signal is received at the receiver, multipath signals having different time delays is received. The following figure gives a clear idea about multipath effect. Here, DP is direct path; S:surface reflected path; B:bottom reflected path; BS: Bottom surface reflected path; SB: surface bottom reflected path. For a large range between the transmitter

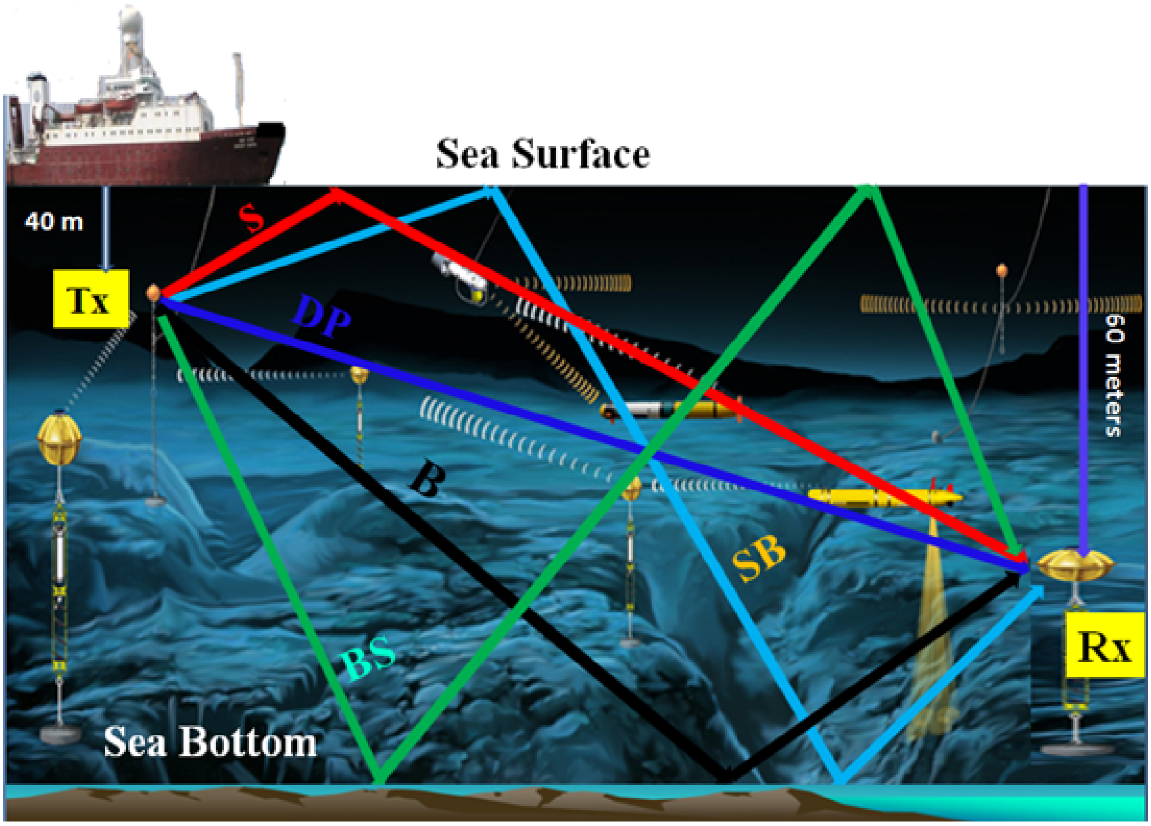

Figure 2.2: Multipath model of the underwater communication channel

and receiver, the transmitted signal propagates to the receiver via various paths. The delay associated with the each path depends on its geometry and prevailing ocean conditions. The signals, while propagating, undergo successive reflections at the interfaces and the propagation paths are determined by the sound speed characteristics prevailed in the channel. Due to these processes, a given signal can therefore propagate from a source to a receiver along several distinct paths corresponding to different directions and durations causing the phenomena called as multipath effect. When a hydrophone receives time series data, the following figure shows the details of direct path and reflected paths.

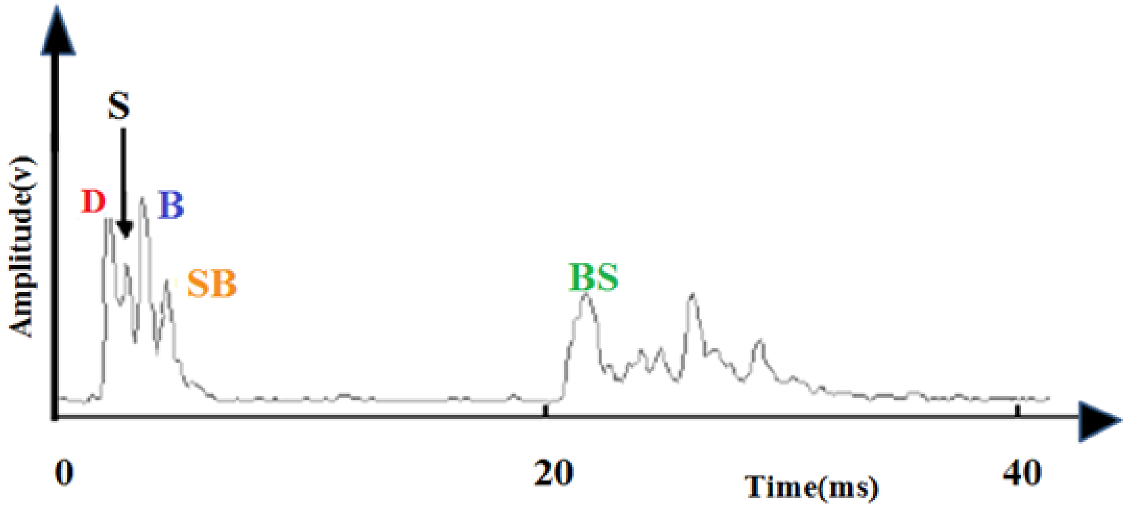

Figure 2.3: Received time series due to multipath effect

Multipath effect due to type of frequencies

At high frequencies, for short signals, multipaths are observable in time domain. Hence, we can resolve each of the paths such as direct path, bottom reflected path etc.

At low frequencies, for longer signals, the different paths are of longer duration such as direct path and surface reflected path. Thus different paths get overlapped and form interference fringes. Thus the different multipaths are not resolvable at low frequencies.

Multipath effect due to depth of ocean

The average depth of shallow waters for modelling the channels is considered to be 200 meters. Ocean profiles in shallow waters are very unpredictable and varying. In shallow water, the surface, volume, and bottom properties are all important, are spatially varying the oceanographic parameters are also temporally varying and the parameters are generally not known in sufficient detail and with enough accuracy to permit long-range predictions in a satisfactory way. In shallow waters, the important ray paths are either refracted bottom-reflected or surface-reflected bottom reflected. The effect of scattering is observed more in shallow waters due to surface waves. This results in echoes consisting of multiple delayed multipath replications instead of point like multipath time delays.

The average depth of deep waters for modelling the channels is considered to be 2000 meters. In deep waters, mostly surface reflected path and direct paths are observed at the receiver. Apart from multipath, shadow zone and convergence zones are also present in deep waters.

In the present thesis, only shallow waters are assumed.

Lloyd mirror effect

In calm sea waters, where sea surface is smooth, the direct path (having no phase change) and the surface reflected path (having phase different from direct path) come closer at a far off distance and interfere each other constructively and destructively to obtain interference fringes. This phenomena is called lloyd mirror effect. This effect is seen more at longer ranges. Lloyd mirror effect is prominent in deep waters due to less number of bottom reflected paths.

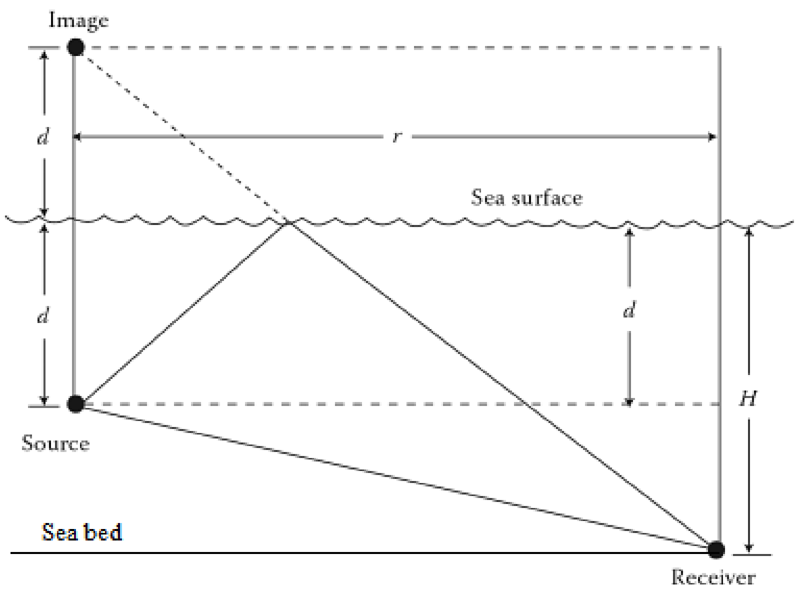

Figure 2.4: Geometry of image interference

2.2 Passive acoustic source localization

source localization is the method of locating a sound source given measurements of the sound field. Localization of a target can be achieved by estimating its motion parameters such as range, depth, velocity, bearing etc in a passive manner from measurements such as sound field or pressure.This thesis deals with source localization of a heavy surface acoustic vessel emitting a broadband signal in shallow waters.Broadband source localization is difficult in shallow waters due to low SNRs and greater interferences, reverberation etc. Time domain source localization was first carried out by Brienzo by extracting the time series data received on a vertical array in deep waters. Most widely used technique for source localization is time delays estimation.

2.3 Review of past work on time delay estimation

Various schemes have been proposed for TDE, each of them having their own merits and demerits.Following schemes have been proposed in literature so far.

Matched field processing (MFP) originally proposed by clay, suggested a modelling of all ocean parameters for accurate TDE. Matched field operates in the frequency domain.MFP performance is highly sensitive to correct knowledge of water columns and environmental parameters. MFP has been used to estimate source range/depth but less accurately either due to coarse resolution or insufficient knowledge of water columns (bathymetry data, sound speed profile). It may not work well in the case of unpredictable ocean environments and broadband signals.

Lloyd mirror effect was later on used for the time delays estimation. It is basically an image processing method where the striation patterns generated due to the constructive and destructive interference between multipaths are used to estimate the time delays.

Autocorrelation method was proposed for TDE. This technique had high SNR gains, simple and also easy to implement. But this technique was dependent on the frequency range selection. However this technique fails to work properly in scenarios such as reverberation, strong transients and interferences. Owing to chich cepstral based technique was introduced later on. This technique is widely used in SONARs.

Cepstral analysis was used for TDE. It was proposed by Wu in 2000, where he used direct path and surface reflected path time delays to estimate the range and also used direct path, surface reflected path and bottom reflected path to estimate range and depth of source. Cepstral analysis doesn't require ocean environment parameters or frequency band of interest for TDE. It only requires the source receiver geometry for TDE. This technique has been implemented to localize surface acoustic vessel in this thesis. These are few of the many proposed schemes for TDE which have been extensively implemented and studied by the researchers in depth.

2.4 Cepstral analysis for TDE

2.4.1 Introduction

Cepstrum was first proposed in 1963 by Bogert. Cepstrum was first defined simply as "The power spectrum of the logarithm power spectrum", The word came up by just reversing the first four alphabets of the word spectrum. Cepstrum is basically the result of inverse fourier transform of logarithm of estimated power spectrum.The real cepstrum mathematical equation is given as

$$c(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} ln \left| X(\omega) \right| e^{j\omega n} d\omega$$

where c(n): cepstra in time domain and X(ω): Spectrum in frequency domain
If the phase information is also taken into account of cepstrum, then we obtain complex cepstrum. Cepstrum graph is (time delays) Quefrency versus time. This means both quefrency and time series are unit of time. Quefrency is not time but time difference between two signals. Cepstral analysis is a nonlinear signal processing technique

with a various applications in fields such as speech and image processing. It can be shown by a simple block diagram.

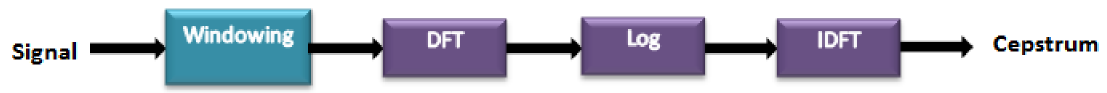

Figure 2.5: Cepstrum basic block diagram

Cepstrum can be Real cepstrum, Complex cepstrum, Power cepstrum and Phase Cepstrum analysis finds various applications in many fields such as speech processing, seismic studies, image processing such as restoration of old recordings, radars and sonars, underwater acoustics signal processing etc. In underwater acoustics, this technique has been applied for whales body length estimation using sperm whale sonar clicks, for jet aircrafts motion parameter estimation, for target identification, for ship noise extraction and for range/depth estimation of acoustic vessel. Power cepstral analysis is found to be suitable for TDE whereas complex cepstrum is found to be useful in wavelet detection. Cepstral analysis for TDE of a continuous broadband signal is difficult to achieve as compared to a pulsed signal such as dolphin clicks.

2.4.2 Derivation for estimating time delays from 2 paths using cepstral analysis

Here, we consider a simple case of estimating time delay using cepstral analysis between direct path (DP) and bottom reflected path in air. Here the surface is assumed to be perfectly reflected. This is an exercise to basically understand the TDE from cepstral analysis.

From the following figure, we can define

l1: length of direct path

l2: length of bottom reflected path

y(t): received signal

x(t): transmitted signal

$h(t-\tau)$: channel impulse response function

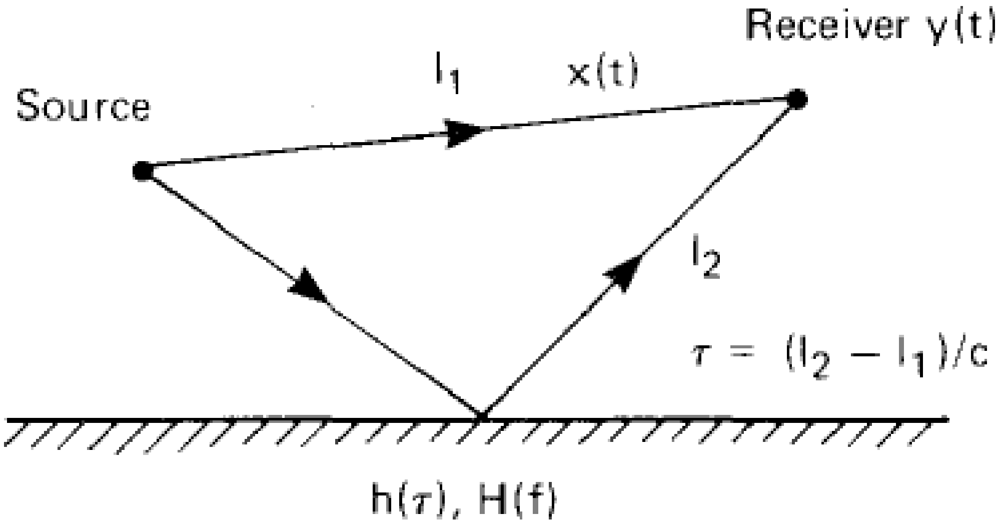

Figure 2.6: signal with improper reflection

τ: time delay between DP and BRP

Thus in time domain,

$y(t) = x(t) + \frac{l_1}{l_2} x(t) * h(t-\tau)$

Transforming this by fourier transform we get,

$Y(f) = X(f)[1 + \frac{l_1}{l_2} H(f) e^{-j2\pi f\tau}]$

Power spectrum can be obtained as the modulus squared

$|Y(f)|^2 = |X(f)|^2 [1 + \frac{l_1}{l_2} H(f) e^{-j2\pi f\tau}][1 + \frac{l_1}{l_2} H(f) e^{-j2\pi f\tau}]^*$ Taking the logarithm of power spectrum

$log\,|Y|^2 = log\,|X|^2 + log\left|1 + \frac{l_1}{l_2} H(f) e^{-j2\pi f\tau}\right| + log\left|1 + \frac{l_1}{l_2} H^*(f) e^{-j2\pi f\tau}\right|$

Using the log series expansion we get

$log(1+x) = x - \frac{x^2}{2} + \frac{x^3}{3}....$

Taking inverse fourier transform gives the power cepstrum as

$C_x(t) = C_y(t) + \frac{l_1}{l_2} h(t-\tau) - (\frac{l_1}{l_2}^2)\frac{1}{2} h(t-\tau) * h(t-\tau) + ... + \frac{l_1}{l_2}\frac{1}{2} h(-t-\tau) - (\frac{l_1}{l_2}^2)\frac{1}{2} h(-t-\tau) * h(-t-\tau)$

$\frac{l_1}{l_2}$ is the attenuation factor and τ is the time delay

It can be seen from the derivation that the cepstrum gets delayed by τ and gets attenuated by a factor of $\frac{l_1}{l_2}$. The power spectrum is an even function and logarithm is also an even function thus the power cepstrum is also an even function. For one-sided power spectrum, the cepstrum is as follows

$C(t) = F^{-1}S(f)$

$S(f) = 2log(S(f))forf > 0$

$S(f) = log(S(f))forf = 0$

$S(f) = 0forf < 0$

Chapter 3

Slant range estimation using a single hydrophone

3.1 Introduction

Localisation and monitoring of marine vessels are significant for monitoring of harbor traffic, port security and security of marine reserved areas where hunting, fishing or collecting marine organisms are prohibited. Thus passive source localisation of marine traffic is done using their acoustic signatures as acoustic waves can travel longer distances. Currently, the topic of passive source localisation is receiving massive attention among the world scientific community as underwater vehicles are getting miniaturized. So, a single hydrophone attached on to underwater systems with size constraints such as gliders, AUV, ROV etc has to perform the same function as passive sonar system. Therefore, this chapter investigates the usage of method based on multipath time delays used to estimate the slant range between the source (marine vessel) and receiver(a hydrophone). Time delays estimated from bellhop ray model was used to estimate the slant ranges theoretically. These slant ranges were used validate the slant range obtained from experimental measurements.

3.2 Work structure

Time delays were theoretically simulated using a bellhop ray tracing model for verifying the reliability of the proposed technique under a particular ocean environment. Slant ranges were calculated from simulated time delays. It was observed that time delays estimation was realizable using the proposed technique in real time ocean environment. Thus, an experiment was conducted in shallow waters with a ship to move over a single hydrophone at different speeds and data was recorded using a single hydrophone. Cepstrum analysis was applied to the pre-processed data using matlab to obtain multipath time delays. Slant ranges were calculated from experimentally obtained multipath time delays. Both the slant ranges were compared. Necessary conclusions were drawn and limitations of this method were identified.

3.3 Contributions made by author

- Time delays were theoretically simulated using a bellhop ray tracing model for verifying the reliability of the proposed model under a particular ocean environment.
- Slant ranges were calculated from simulated time delays
- An experiment was conducted in shallow waters with a ship to move over a single hydrophone at different speeds and data was recorded
- Cepstrum analysis was applied to the pre-processed data using Matlab to obtain multipath time delays.
- Slant ranges were calculated from experimentally obtained multipath time delays. Both the slant ranges were compared.

3.4 Experimental details

An experiment was conducted by the scientists of NSTL in shallow waters off Indian west coast in Arabian Sea. A ship was made several runs over a bottom mounted

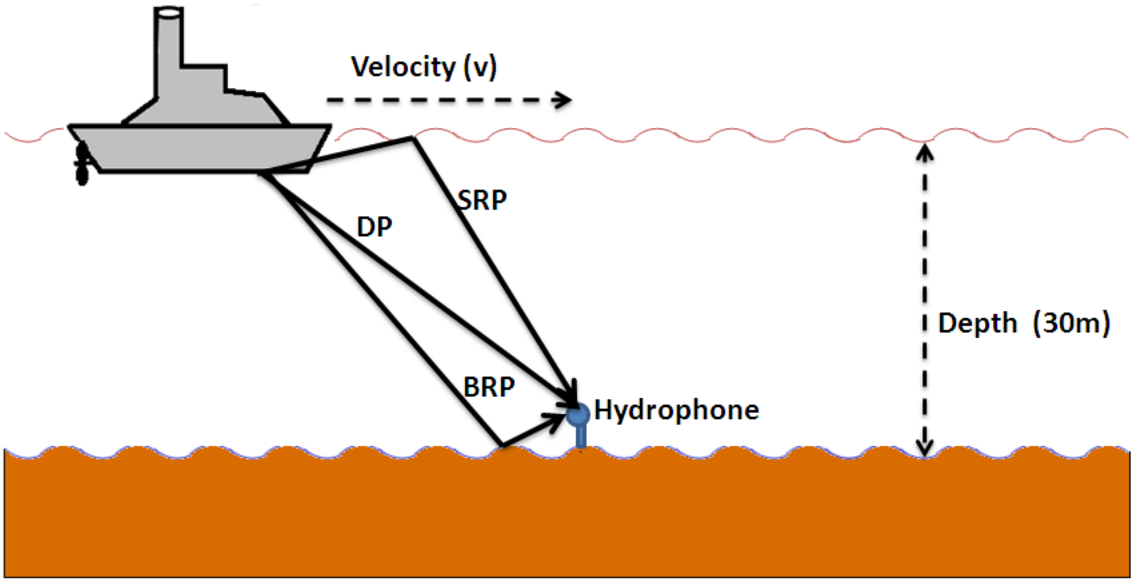

Figure 3.1: Experimental setup

hydrophone with velocities ranging from 10 to 19 knots As the waters were very shallow of around 30m depth, the sound speed profile was uniform across the depth with speeds of 1540 m/s. The following table provides other required details. A schematic diagram depicting time delays received by a hydrophone placed on the sea bottom would receive was shown at Figure 4.1. As the waters are very shallow the sound speed profile was mostly uniform across the depth with speeds of 1540 m/s. The analysis presented herewith corresponds to a ship speed of 18 knots (9 m/s) only.

Parameters	Values
Ship's velocity	9 m/s
station depth	30 m (seabed=0 m)
Hydrophone depth	1 m (seabed=0m)
average sound speed in ocean	1540 m/s

Table 3.1: Experimental details

3.4.1 Preprocessing of raw hydrophone data

Due to the presence of boundaries of ocean, a broadband noise generated from a ship or a submarine travels through several paths. A hydrophone placed above the sea bed will receive noise from all these paths. As the number of bounces from the sea bed and sea surface increases, the intensity of the broadband noise decreases. The prominent ones are only direct path (DP), surface relected path (SP) and bottom reflected (BP) path. Due to effect of these paths, lloyd mirror pattern can be observed in shallow waters. The time difference DP and other reflected paths is called multipath effect. Hydrophone received radiated noise data of the ship. This data was sampled at 256 kHz for pre-designated distances. Then this sampled data was subjected to normalized short time fourier transform. Then the spectrogram was plotted in figure, where lloyd mirror pattern is clearly visible. The presence of striations visible in the power spectrum confirms the presence of strong multipath in shallow waters. The following figure shows the radiated noise signature of the ship obtained at the hydrophone.

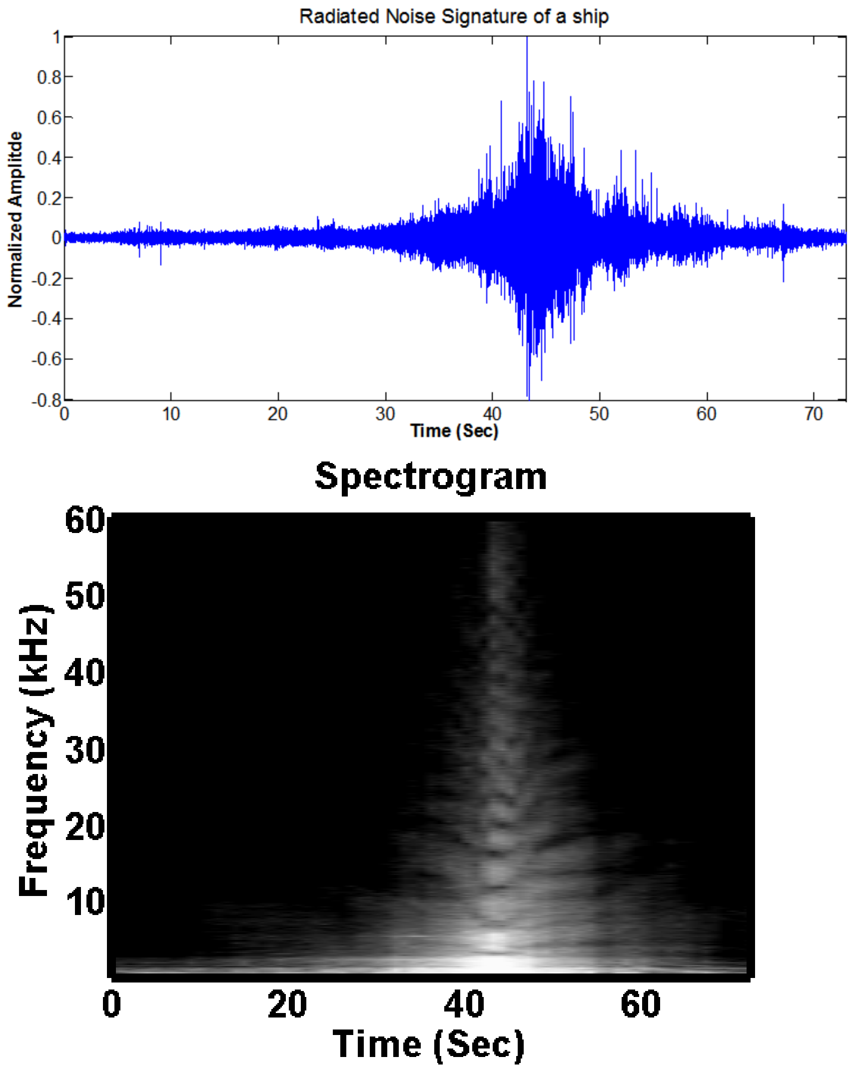

Figure 3.2: Power spectrum showing lloyd mirror pattern

This radiated noise signature is raw hydrophone data of a real merchant ship transiting across the hydrophone. The amplitudes are normalised pressure data measured in micro pascal and normalized with minimum sensitivity measured by hydrophone. This data is total of 92 seconds but truncated to show the RN data clearly. The figure is 256 kHz sampled hydrophone data.

3.5 TDE using power cepstrum

The measured radiated noise sampled at 256 kHz was used for estimating the multipath time delays using cepstral analysis. In this process, initially, the data was partitioned into non overlapping samples of 0.5 seconds each. Then cepstral analysis was performed. The following flowchart clearly explains the complete process. The following figure 4.2 shows the cepstrogram with clear view of surface reflected path but only till CPA.

3.5.1 Derivation of power cepstrum for multipath

The time delayed and attenuated signal can be shown as

$x(n) = s(n) + \sum a_m s(n - n_m) + g(n)$ (1)

Taking fourier transform for this signal (2)

$X(\omega) = S(\omega)(1 + \sum_{m=1}^{M} a_m e^{-j\omega m}) + G(\omega)$

The power Spectrum can be shown as

$P = |S|^2 Q(1 + y)$

Let $SNR =$

From the equation (3) taking the logarithm power spectrum we get

$\hat{P} = Log(|S|^2) + log\,|Q| + log(1 + y)$

Polynomial expansion of $log(1 + y) = \sum_{i=1}^{\infty}(-1)^{i+1}\frac{y^i}{i}$

In the polynomial expansion the amplitudes of higher order harmonics or higher order terms is

$A_{m,i} = (-1)^{i+1} \frac{a_m^i}{iQ^i}$

Thus the order of the higher order harmonics decreases exponentially and hyperbolically.

3.5.2 Results of cepstral analysis

The following the cepstrum output from measurements is compared with theoretically estimated time delays and seen to be matching with SRP time delays. From the figure, the lower part showing higher amplitudes are basically the DP undulations.

3.6 Theoretical time delays estimation

Although, the presence of multipath propagation was revealed by the measurements, the order of multipath time delays was not known. Therefore, a ray based acoustic propagation model called as Bellhop was used for the given experimental geometry to estimate the delay arrival between the DP and SRP. This was done to verify whether the measured time delays were indeed precise or not. For calculating the time delays, the source depth is kept at 6.4m the receiver depth at 29m with sea depth being 30m for a range of 320m on either side of the hydrophone (total range 640m). The time delays were very less when the source was at farthest distance and increased with decreasing distance between the source and receiver. The maximum time delay was observed over the hydrophone which is at 8 ms. For proper understanding of the model used to create these time delays, some theoretical background of ray theory needs to be understood which is briefly explained in the next section.

3.6.1 Ray theory

Ray theory is a theoretical approach to solving the wave equation. The main tenets of ray theory are the existence of a wave front along which the phase of the solution is constant, and the existence of rays that describe the spatial location of acoustical energy radiating from the source in a manner analogous to optical ray theory. Ray tracing programs are commonly used in underwater acoustics for modelling the propagation of high-frequency acoustic waves as a function of time. Ray tracing traditionally involves integrating a set of differential equations called the ray equations, which describe the rays trajectory. The amplitude of a ray is determined by the cross-section of the ray tube bounded by adjacent rays. The main drawback associated with ray tracing is the existence of shadow zones, which are zones through which no rays pass, resulting in a pressure field of zero everywhere within them. In actuality, there is always some diffraction of sound into the areas given as shadow zones in ray tracing, resulting in a discrepancy between the exact solution and that predicted by ray tracing methods. Ray tracing is a useful tool for modelling the propagation of high-frequency sound. Ray tracing is intuitive and radiated sound is readily visualized and calculated [4]. The following schematic diagram (Figure 3.5) shows five types of 2D range independent propagation paths in the ocean. The broken line is indicative of sound speed profile in the deep and shallow ocean, although it should be borne in mind that the detailed shapes of the profiles vary from place to place and time to time. The purpose of (2D) ocean-acoustic propagation modelling is to compute the intensity and phase of the field as a function of frequency and range for given source and receiver depth. Thus, it is desirable that the models should be capable of handling the different types of propagation paths shown in Figure 3.5

3.6.2 Ray model

Ray tracing is particularly useful for deep water problems, where generally only a few rays are significant. In shallow water, far from the source, many rays are incident on the receiver, whereas only a few modes are present, the obvious implication being that in such circumstances mode models are preferable to tracing rays. As a rough guide, ray tracing is satisfactory when the wavelength is very much less than any of the length scales in the problem. As well as water depth, these lengths scales include bottom and surface roughness, the size of focal regions, and the distance over which appreciable changes in sound speed occur. Ray-tracing models are fast to compute, providing a pictorial representation, in the form of ray diagrams, of the field in the channel. Further advantages of ray tracing are that the directionality of the source and receiver can be fairly easily accommodated, by introducing appropriate launch-and arrival-angle weighting factors; and rays can be traced through range-dependent sound speed profiles and over complicated bathymetry. On the other hand, there are difficulties in keeping track of phase at bottom reflections; many rays have to be traced; and the computations must be performed at all ranges till the receiver. Perhaps the most serious disadvantages, however, is that wave effects such as diffraction band caustics cannot be handled satisfactorily by ray tracing, which limits its usefulness for investigating bottom interactions and low frequency propagation. Indeed, a danger of ray tracing is that it may generate false caustics and produce shadow zones whose boundaries are unphysically sharp. Shear waves in an elastic bottom are beyond the capabilities of ray tracing models. Most popular in this class of model is BELLHOP developed by porter MB (1987).

3.6.3 Bellhop model

Ray-tracing models are fast to compute, providing a pictorial representation, in the form of ray diagrams, of the field in the channel. Based on different algorithms, many ray tracing models have been designed, The most famous ray tracing model is bellhop ray tracing acoustic propagation model. It was developed in 1987 by Porter Bucker at the Space and Naval Warfare Systems Centre in San Diego. Bellhop uses Gaussian beam tracing to produce the ray-trace and impulse response plots. Gaussian beam tracing uses standard ray tracing to describe a central ray, about which a beam is formed whose intensity decays on a normal from the ray following a Gaussian distribution.

Input environment file

The main input file to Bellhop is called the environment file. The environment file is a simple text file given in a structured format specifying the details of the environment, the source and receiver characteristics, and the type of analysis to be performed. Environmental file given as input to estimate theoretical time delays can be as shown below. There are many types of output files in bellhop. Based on the parameters given to the environment file of bellhop model, we can get three types of output files as ray file, arrival file and transmission loss file.

The following figure was obtained for the theoretical time delay estimation for a particular experimental setup using bellhop ray model.

Output impulse response plot

In this model, arrival file is represented with 'A' in the environment file which gives the impulse response of the input signal. Arrival file plots impulse response with Amplitude versus time delay of each received signal at the receiver. The impulse response plot shows the time dispersion of initial pulse at the source into multiple impulses at the receiver. Time dispersion is caused due to different path lengths and the travel times associated with each ray. The contents of the arrival file give the details of each impulse of the received signal. The difference between time of these impulses gives the time delays. The bellhop model is executed using the following command in command window to get impulse response plot. !bellhop.exe file name Plotarr ('file name. arr', number of receivers, number of sources, 1) Let us see an example of arrival file output

Amplitude	phase	delay	takeoff angle(src)	takeoff angle(rcr)	surface	bottom
5.00 +e-4	0.0	1.33	0.5777573	0.5777575	0	0

Table 3.2: Example of arrival file of single impulse.

Print file

The extension of this file is file name. prt. This file shows the summary of the bellhop execution of environment file and output files (Arrival file and impulse response plot). This file is used to observe the bugs and errors during execution. This file shows the complete summary of all errors, output detaills etc.

3.7 Slant range calculation

The accuracy of slant range estimation using single hydrophone essentially depends on accurate estimation of time delays from the measurements which are indeed very difficult to obtain.

3.7.1 Source receiver geometry

Cepstrum analysis are used for extraction of time delays. BRP, DP and SRP are the most prominent ones. In this case, Only the DP undulation and SRP. Cepstrum only exploits the source receiver geoemtry for TDE and doesnt require any actual environment parameters. The following figure shows in general the source receiver geometry consisting of SISO and only 3 paths showing the DP, SRP and BRP. The Mirror image of the source are also depicted to show the effect of the lloyd mirror pattern.

3.7.2 Calculations

The following figure shows the source-receiver. Simple calculations were don to derive the slant range using the hyperbolic equation and source-receiver geometry.

The following equations were used to calculate the slant ranges from source depth, receiver depth and surface reflected time delays[7].

H_h: horizontal distance τ: Difference of time delays between DP and SRP
c: Sound speed in ocean
H_s: depth of source with respect to sea surface
H_r: Depth of receiver with respect to sea surface
R_d: Slant range between source and receiver

$$H_h = \sqrt{(\tfrac{4H_s^2}{c^2\tau^2} - 1)(H_r^2 - \tfrac{c^2\tau^2}{4})}$$

$$R_d = \sqrt{H_h^2 + (H_r - H_t)^2}$$

3.7.3 Results and conclusion

An experiment aimed at estimating the slant range between the surface ship and a hydrophone placed above the sea floor was conducted in shallow waters of Indian west coast. The station depth was 30m and the hydrophone was 1m above the sea bed. The radiated noise signature thus collected was subjected to cepstral analysis to estimate the time delays between the DP and SRP. Mean time the time delays were also estimated using Bellhop model to verify the measured time delays accuracies. It was found that the measured and modeled time delays compared well. However the time delays could not be resolved when the sea was rough. Although, there are some limitations to this method, overall usefulness of this technique outweighs the limitations, as slant range between source and receiver can be estimated using single hydrophone. It can be seen from figure that the time delays estimated using Bellhop model and the measured time delays are in close agreement which implies that the estimated slant ranges are also in close agreement. The measured time delay curve was continuous till the CPA and thereafter it was seen only intermittently. This occured due to the fact that sea started picking up and it has eventually became rough (sea state was greater than 2). The subsequent measurements with different ship speeds did not show any clear paths, suggesting that the method based on multipath time delays for estimation of slant range is likely to fail in rough seas and works well in calm waters only. More experimental data is required to arrive at any conclusion on this aspect.

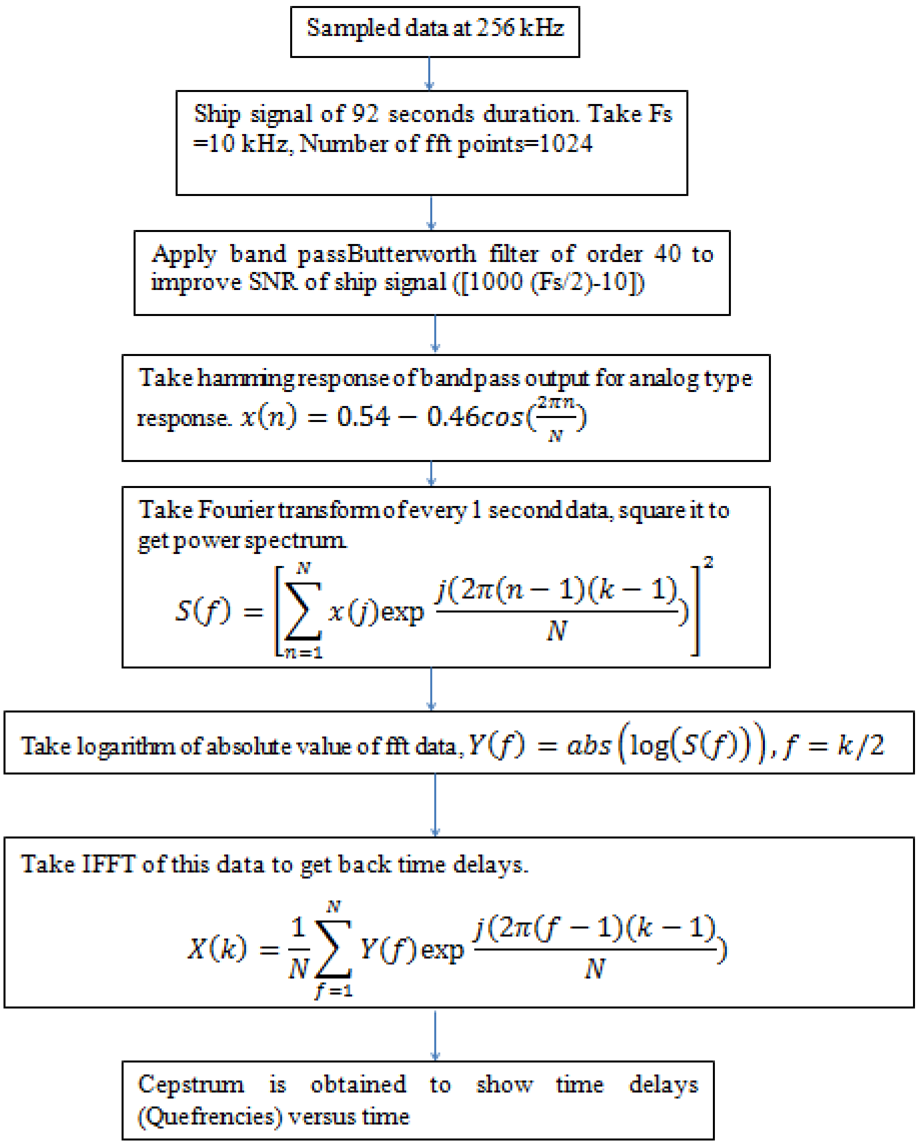

Figure 3.3: Block diagram showing TDE from power cepstrum

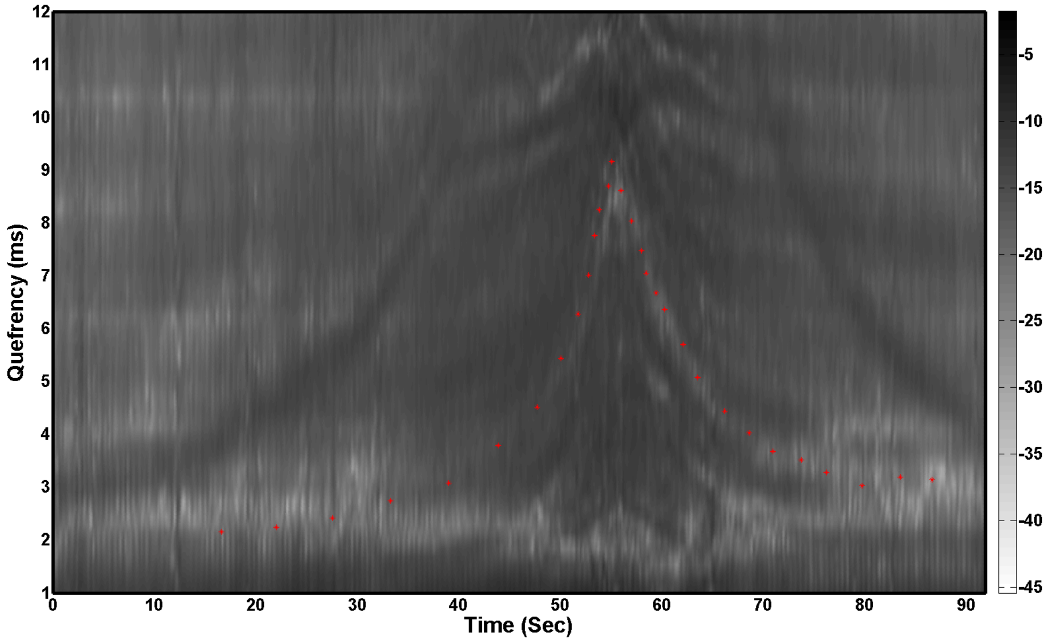

Figure 3.4: Power cepstrum output

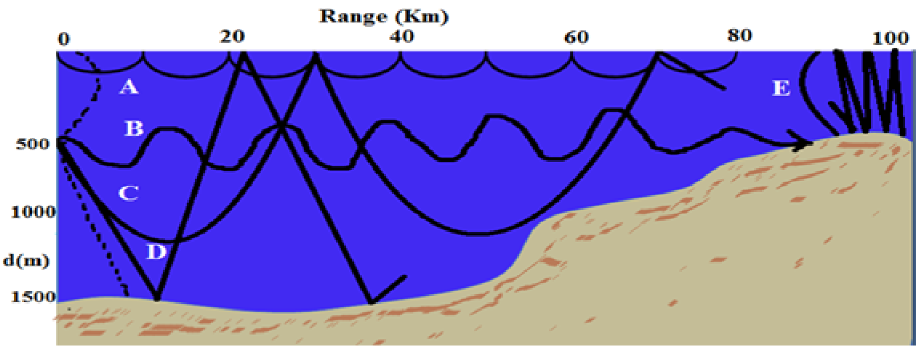

Figure 3.5: Schematic of ray paths in the ocean.

```
'ISO'                          ! TITLE
200.0                          ! FREQ (Hz)
1                              ! NMEDIA
'CVW'                          ! SSPOPT (Analytic or C-linear interpolation)
0 0.0   300.0                  ! DEPTH of bottom (m)
0.000   1500.0 /
30.000 1500.0 /
'A' 0.0                        ! Type of file
30.0 1500.00 1.0 1.7 /         ! Sea bottom composition specification
1                              ! NSD
6.5 /                          ! SD (1: NSD) (m)
1                              ! NRD
29.0 /                           ! RD (1: NRD) (m)
1                              ! NR
329 /                          ! R (1: NR) (km)
'A'                            !'R/C/I/S'
0                              ! NBeams
-89.0 89.0 /                   ! ALPHA1,2 (degrees)
0.0 30.1 0.4                   ! STEP (m), ZBOX (m), RBOX (km)
```

Figure 3.6: Bellhop model: Input Environment file snapshot

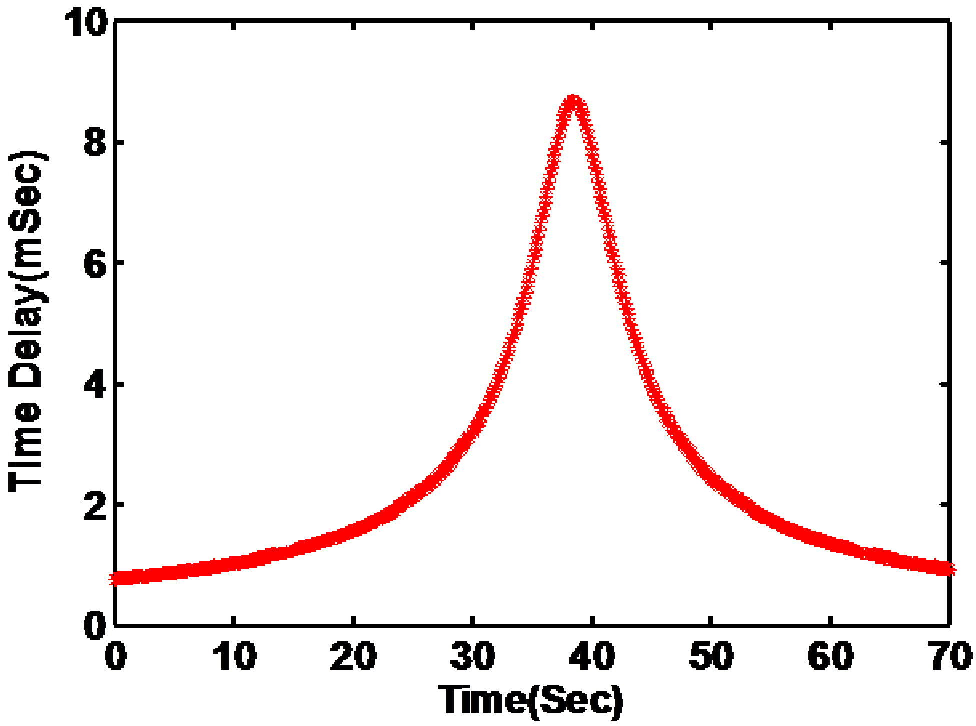

Figure 3.7: Time delay estimates from bellhop ray model

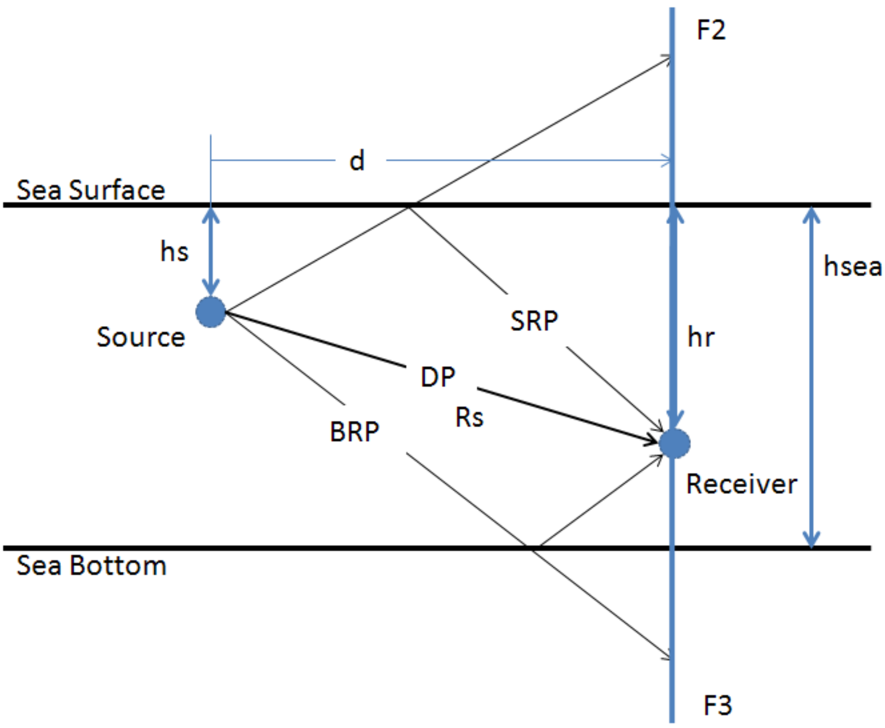

Figure 3.8: Source receiver geometry

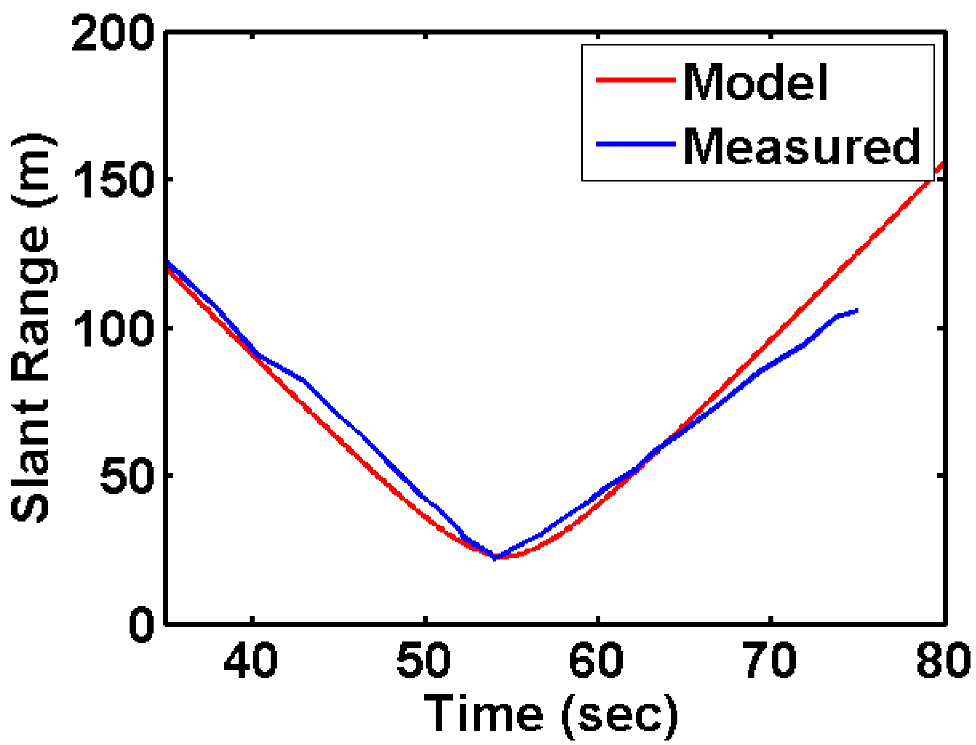

Figure 3.9: Comparison of slant ranges obtained from modelled and estimated time delays

Chapter 4

Error analysis and LS method for parameter estimation

4.1 Error propagation

Imperfection of measurements, instrumentation and various other human factors always lead to minor variations in the results from an experiment. Thus they are in most of the cases not accurate, i. e. some error is assiated to each result. For a most of the systems, the practical and theoretical parameters do not match due to some random errors added by external factors in the case of practical measurements. For calculation of the parameters from the measurements, there will be some propagation of errors from measured to the calculated values. There are various equations for calculation of propagation of errors from measurements to parameters. In this thesis, the estimation precision needs to be determined for error propagation from estimated time delays to slant ranges. This was basically achieved in two steps for better understanding.

- Simulations were performed where slant ranges were calculated for time delays with different standard deviation. This was done to understand the standard deviations of the time delays from the true values.
- The error propagation equation was applied to calculate the range of slant range errors.

4.2 Range deviation

The following method has been carried out for the performance analysis of the localization in order the find out the estimation precision of the TDE techniques. The following steps were followed to calculate the range deviations based on time delays with different standard deviations.

- For a particular range, the time delay of SRP was obtained using bellhop model where the parameters of the experimental setup was given as input to the model
- A random data set was generated for time delay with standard deviations ranging from 1 to 10us.
- Slant ranges were calculated for these time delays
- deviation of slant ranges were calculated and was plotted.
- Similarly for other ranges, time delays were obtained and slant range deviations were calculated.

4.3 Error propagation from time delays to slant ranges

The general equation for error propagation is shown below. But as the propagaton is seen only from time delays to slant ranges, the formula was modified Slant ranges are calculated using the following equations:

The following figure (no.)shows the slant range deviations for ranges :100, 300, 500, 800 and 1000m. It can be seen from the figure that at low standard deviations , the slant range errors are less for all ranges, but as the standard deviations increase the the slant range errors are more even for small ranges.

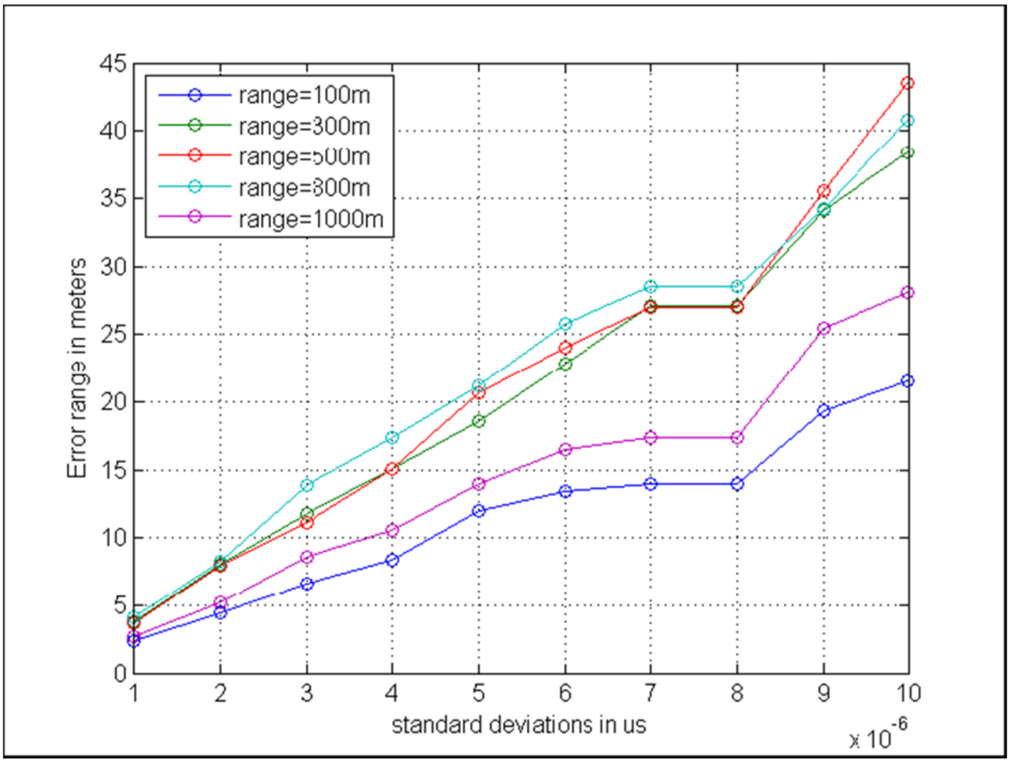

Figure 4.1: Slant range deviations

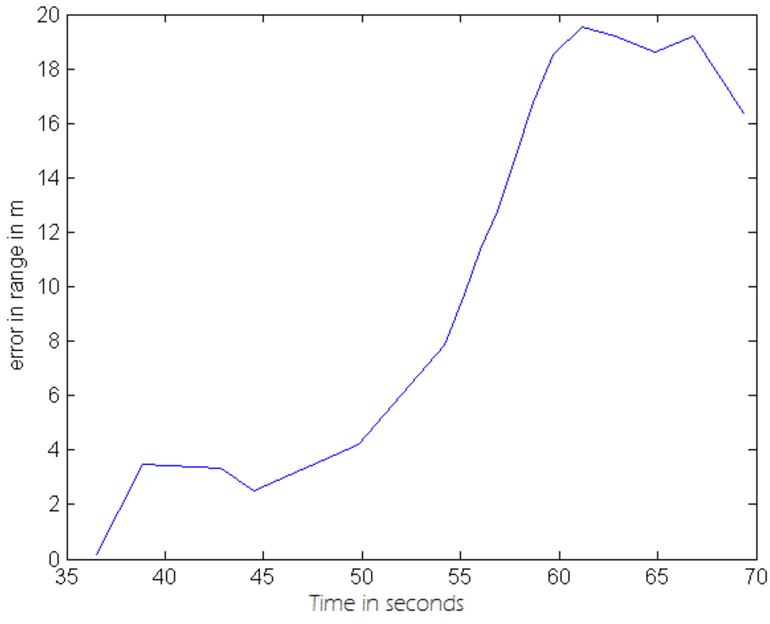

Figure 4.2: Slant range errors for SRP time delays

4.4 Least squares estimation method

Least squares method is basically the method where the square of the difference of the true values and measurements are taken and used to estimate the parameters. Here LS method is used based on the equations of time delays used to calculate motion parameters such as velocity, range etc. Using time vector and time delays, these motion paramters are calculated. This method can also be referred to as equation error method. Some work has been carried out using this method in order to estimate the motion parameters such as time delay at CPA, range at CPA and velocity. Some inital estimate is required for the analysis in this method. The proposed method has been only simulated and no real data has been used to verify this method.

4.5 System model:Algorithm

The three parameters Range at CPA, velocity and time delay at CPA are unknown, where as only time delays are known. Based on basic equations obtained from the source-receiver geometry, a model can be formulated to estimate the three unknown parameters. as We know for any observation vector.
$y(t) = h(t)x + e(t)$
where the y: observation vector
h: channel impulse response
x: input signal
e:error in the observation vector
thus we get three simple equations
$x_1 = v^2$
$x_2 = \tau_c$
$x_3 = \tau_c^2 v^2 + R_c$
and $h = [t^2, -2t, 1]$ where v:velocity of the source
h: time vector
R_c:Range at CPA
τ_c: time delays at CPA

The main aim is to estimate x from y.

4.6 Algorithm

Thus we have to estimate the vector x and obtain an algorithm such that it minimizes the error cost function. Thus for this two estimates are taken:(i) unweighted error estimate and (ii) weighted error estimates.

So the cost function is $J = (y - Hx)^T W(y - Hx)$

thus the estimate x is then

$\hat{x} = (H^T WH)^{-1} H^T Wy$

where the W: weigthed matrix For unweighted error estimates W is taken as identity matrix and for for weighted error estimates W was taken as R_e^{-1}

where R_e is diagonal matrix whose diagonal elements equals the variance of errors.

4.7 Simulations and results

Simulations were carried out where the parameters of experiments setup were taken and time delays were calculated. Then modelled time delays were given different standard deviations of 10us and 50us seconds. These time delays were used to estimate the three parameters and obtain the mean variance and standard deviation of each of these parameters The following 2 figures show the three estimated parameters from simulations for a large data set of 1000 observations. For better results a larger data set is preferrable. The following figures show deviations from true values due to the low snrs.

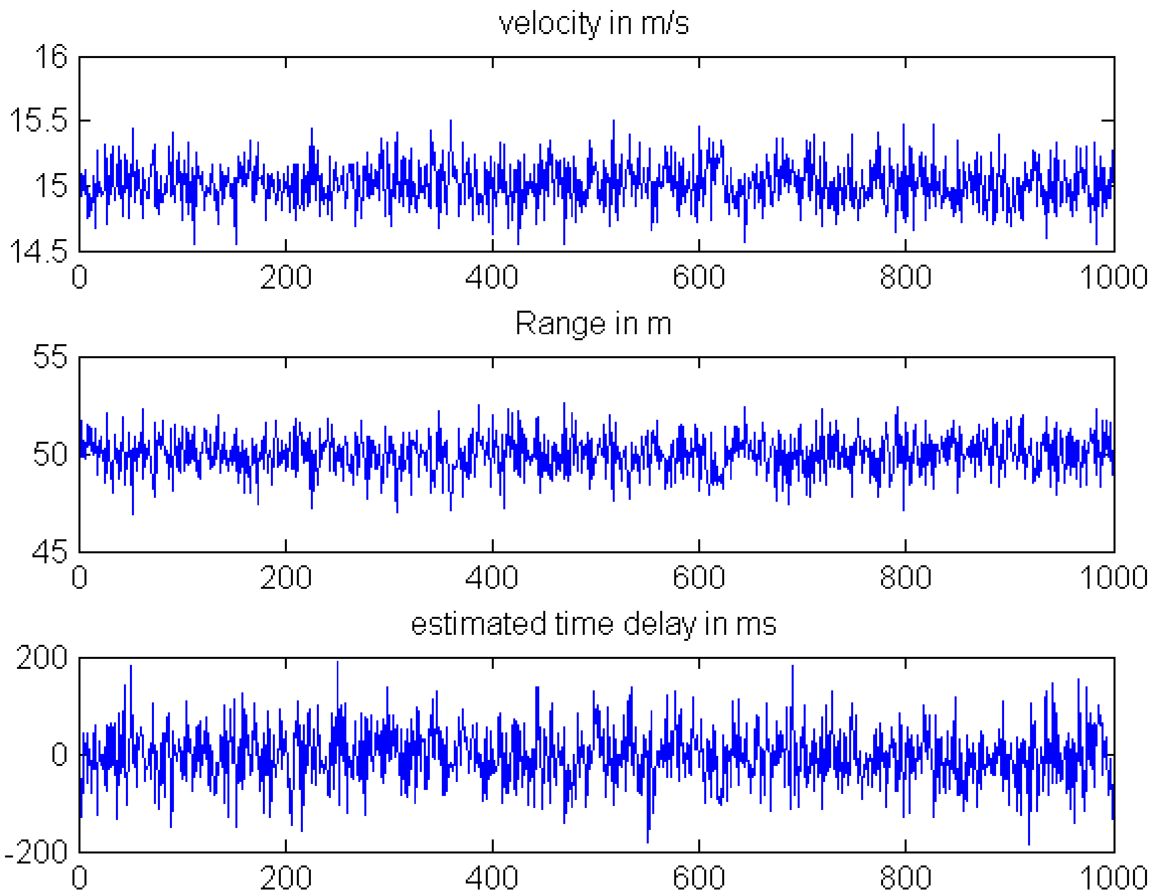

Figure 4.3: Simulation results with 10 us standard deviation

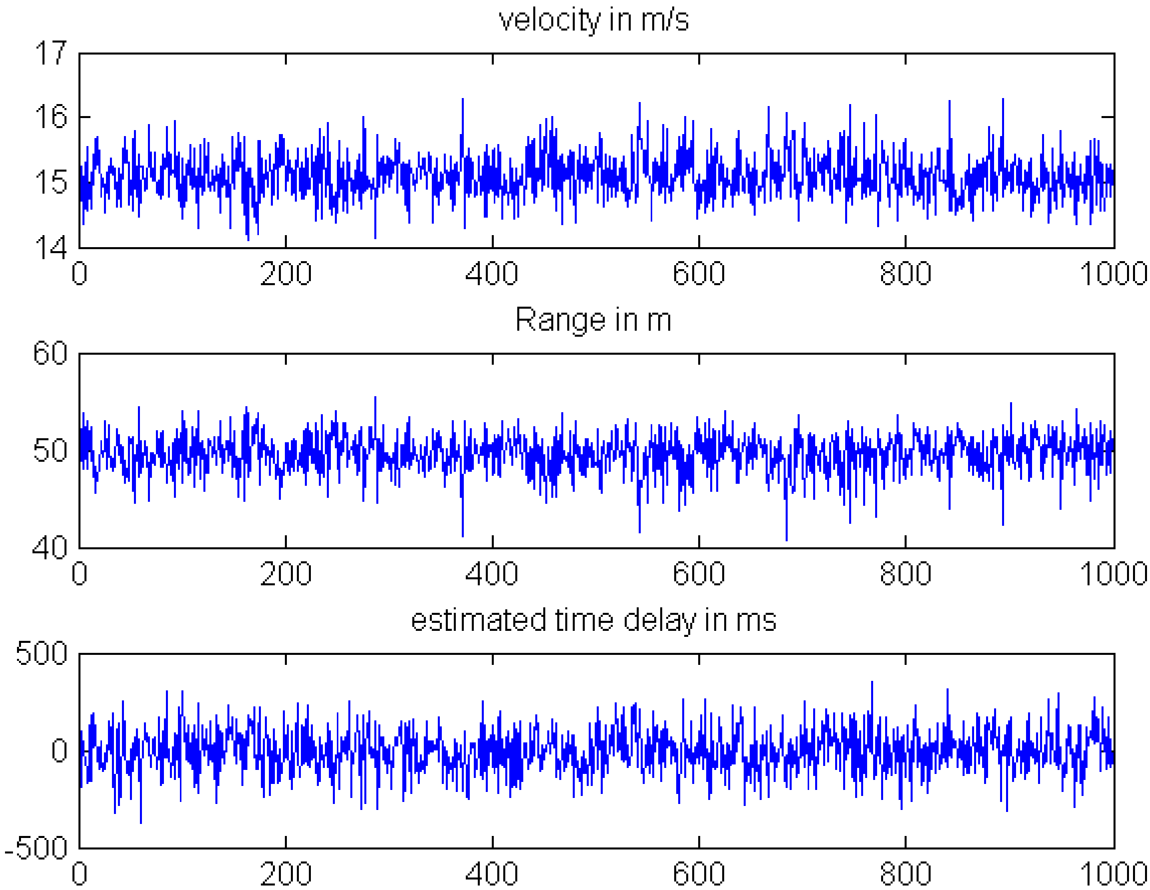

Figure 4.4: Simulation results with 50 us standard deviation

STD=10us		unweighted estimate	Weighted estimate
V(m/s)	bias	0.31	-0.067
	STD	0.178	0.12
	RMSE	0.181	0.135
τc	bias	0.001	0.001
	STD	0.09	0.053
	RMSE	0.091	0.053
Rc	bias	0.048	0.05
	STD	0.72	0.34
	RMSE	0.75	0.34

Figure 4.5: simulation results in tabular form

STD=10us		unweighted estimate	Weighted estimate
V(m/s)	bias	0.31	-0.067
	STD	0.178	0.12
	RMSE	0.181	0.135
τc	bias	0.001	0.001
	STD	0.09	0.053
	RMSE	0.091	0.053
Rc	bias	0.048	0.05
	STD	0.72	0.34
	RMSE	0.75	0.34

Figure 4.6: simulation results in tabular form

Chapter 5

Conclusions and future scope

5.1 Conclusions

THe following work of slant range estimation has varied potential application in the coming future and is an unexploited field of research. The world scientific community has started gaining interest in this field and finds many applications. The slant range estimation of a surface acoustic vessels in shallow waters is a difficult task. The following thesis uses the power cepstrum method for TDE. Accurate time delay estimation is necessary for the proper estimation of the slant ranges. The underlying assumptions made in this method is that the sea state is calm with negligible or no surface waves for perfect reflection. The sea bottom is assumed to be flat. The power spectrum of radiated noise signature of the ship confirmed the presence of multipath by showing lloyd mirror effect. Then the cepstrum was applied and the quefrency v/s time scale figure was obtained showing the time delays. However the time delays could not be resolved when the sea was rough. Although, there are some limitations to this method, overall usefulness of this technique outweighs the limitations, as slant range between source and receiver can be estimated using a single hydrophone. The time delays compared well with the modelled time delays obtained from bellhop ray model. The time delays were then used to calculate slant ranges. Both the modelled and the estimated slant ranges were also compared and was satisfactorily matching from 30 s to 80 s of the ship's transit. The error analysis of the slant ranges were performed and tolerances of range deviations was from 5 to 20m.

5.2 Future scope

In slant range parameter estimation,work is going in the improvement of the SNRs in cepstrum to obtain precise time delays.We have tried liftering process, auto correlation method and moving average till now and some improvement in SNR has been observed.Also using the estimated time delays, depth, velocity of a moving marine vessel can be estimated.

This technique has only been used in shallow waters and needs to be investigated in the case of deep waters. Also the bottom reflections paths are not clear. Hardware implementation can be done for this method and can be later tested and verified.

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Appendix A

Important terms

Acoustic signature is a collection of all fundamental frequencies and harmonics of a particular ship or a vessel. It is unique for every vessel. It is produced due to the machinery of the vessel.

Ambient noise is the background noise present in the ocean.Ambient noise is background noise in the sea due to either natural or man-made causes, and may be divided into four general categories: hydrodynamic, seismic, ocean traffic, and biological.At lower frequencies the main source of this noise is ocean turbulence and microseisms.At high frequencies, the main source of this noise is thermal noise due to water molecules.

Attenuation of sound energy in seawater arises principally through the action of two independent factors, absorption and scattering, with an additional contribution from bottom loss.Attenuation is mainly function of frequency of the signal.

Closest position of approach is the distance when the hydrophone or sonar is closest to the vessel. This means that the vessel lies just above the sensor.

Hydrophone Marine equivalent for aerial microphones.Underwater microphone (Receiver) converts acoustic pressure variations at its surface to electrical output signal.With every hydrophone an important parameter of its sensitivity is related.

Kalman filter phase equations Prediction phase

$\hat{\Gamma_k^-} = F\hat{\Gamma}_k - 1$

$\hat{P_k^-} = F\hat{P}_{k-1}F^T + Q$

Correction phase

$K_k = P_k^- H^T (HP_k^- H^T + R)^{-1}$

$\hat{\Gamma}_k = \hat{\Gamma}_k^- + K_k(z_k - H\hat{\Gamma}_k^-)$

$P_k^{=}(1 - K_k H)P_k^-$

Kalman Gain is a weighting factor used to minimize the error associated with the difference between the measurements and predicted values.

Optimal Estimator for linear systems described by functional differential equations is constructed. The problem is solved by deriving equations for the unbiased estimation error as well as for its covariance. A functional of the estimation error at the terminal time is minimized by an optimal choice of the gain matrices of the estimator.

Sonar is accronym for Sound Navigation and Ranging. SONAR is a marine equivalent for RADAR.There are two types of sonars: active and passive sonars.

Spectrogram is a visual representation of the spectrum of frequencies in a sound or other signal as they vary with time or some other variable.The spectrogram used here shows a range of frequencies with respect to time.

System model (state space model) is the conceptual model that describes and represents a system. It consists of the inputs and the procedure.It can be considered as an algorithm for computing a model.

Transmission loss ,Consider a source of sound located in the sea. The intensity of the sound can be measured at any point in the sea, near to or far from the source. For purposes of measuring intensity at the source, the intensity measurement is generally taken at one unit distance from the source and labeled Io. The intensity can then be measured at any distant point where a hydrophone is located and denoted I. It is operationally significant to compare the two values.

One way to do this is to form the ration Io/I. Note that if the ratio, denoted n, is greater than unity, the intensity at the source is greater than at the receiver, as would be expected. If n = Io/I, then

$10logn = 10logIo - 10logI$

Appendix B

List of publications

- Paper presented in International symposium on underwater technology, 2015 sponsored by IEEE and OES, Japan held at NIOT, Chennai,23-25 February. Paper titled "Transiting Ships Slant Range Estimation using Single Hydrophone in Shallow Waters," by G V Krishnakumar, C Pavani, Padmanabham M, B Sudhakar and Mehul Naik

- Paper accepted and reviewed for August,2015 issue of "Indian journal of marine sciences". Paper titled "Slant range estimation of ship using cepstral analysis," by C Pavani, G. V. Krishnakumar and Mehul Naik.