Design of an Active Acoustic Sensor System for an Autonomous Underwater Vehicle

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Letter of Transmittal

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Dear Professor Mark Bush,

It is with great honour that I submit this thesis, entitled "*Design of an Active Acoustic Sensor System for an Autonomous Underwater Vehicle*" as partial fulfilment of the requirements of the degree of Bachelor of Engineering with Honours.

Yours faithfully,

Minh Tu Nguyen

Abstract

Unstructured oceanic environments present great challenges to AUV navigation. However, with continual improvements in sensor technology, new methods of navigating hazardous underwater terrain are far more effective than ever before.

To date, much research has focused on maximising the functionality of AUVs at the expense of cost. In contrast, this thesis aims to develop an active acoustic sensor system that determines the distance an obstacle or landmark is from an AUV, while optimising cost efficiency. Although this has been accomplished successfully on land-based autonomous vehicles, these systems have not been implemented on AUVs. The focus is to design a system that consists of four distance sensors directed to the port, starboard, bow and downward side of the AUV.

The sensor system is custom-made using low cost components comprising the commercially available Navman Depth 2100 transducer and the LM1812 ultrasonic transceiver chip. The processing of sensor data will be accomplished by an Eyebot (Motorola 68332) microcontroller.

Outcomes of the project included the successful design of a prototype sensor for the active acoustic sensor system and successful testing and verification to demonstrate correct sensor functioning, which provides the basis for further research in sensor development for the University's AUV, called the *Mako*. Final comments include a proposal for a control system for the sensor application of wall-following, as well as recommendations for future improvements and research.

Acknowledgements

Successfully completing this project would not have been possible without the tremendous amount of time, assistance, and understanding of the people in my university life and my personal life.

I would like to acknowledge my Honours supervisor, Associate Professor Thomas Bräunl, whose guidance and expertise was pivotal to the project.

A very big thank you must also go to my family, especially my parents, for the infinite ways in which they have supported my studies this year. A special thankyou must go to my sister, brother-in-law and Terry O'Neill for all their proofreading. An extra special thank you to my girlfriend, Clara Ng, to all my friends and fellow AUV team members, and to all the foreign exchange students, for their support, humour and knowledge that kept me going through the year.

Finally, thanks are also due to the members in the electronic workshop of the EE department, especially Jonathon Brant and Ivan Neubronner, for their knowledge in hardware that was crucial to the project.

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Nomenclature

Abbreviations

Below is the list of abbreviations used in this thesis:

ADC – Analogue to Digital Converter
AUV – Autonomous Underwater Vehicle
AWGN – Additive White Gaussian Noise
CPU – Central Processor Unit
DOA – Direction of Arrival
DSP – Digital Signal Processor
EKF – Extended Kalman Filter
FET – Field Effect Transistor
GPS – Global Positioning System
IRQ – Interrupt Request
ITC – Input Transition Counter
PAC – Pin Action Control
RAM – Random Access Memory
LC –Inductor-Capacitor
LMS – Least mean square
RF – Radio Frequency
RMS – Root Mean Square
ROBIOS – Robot Basic Input Output System
SNR – Signal to noise ratio
TBS – Time Base Select
TDE – Time Delay Estimation
TPU – Time Processor Unit

Variables

Below is the list of variables used in this thesis:

x – *x coordinate of AUV*
y – *y coordinate of AUV*
θ – *angle coordinate of AUV*
v – *velocity of AUV(m/s)*
ω – *angular velocity (rads/s)*
d_{offx} – *x distance from centre of AUV(m)*
α – *thruster distance from centre of AUV(m)*
φ – *half beamwidth of echo sounder*
$U(k)$ – *input matrix for AUV control equations*
$\xi(k)$ – *measurement error of echo sounder*
w – *white Gaussian noise in process*
$z(k)$ – *the distance measured by echo sounder*

x_w – *x coordinate of wall*
y_w – *y coordinate of the wall*
γ – *angle of the wall*
d – *distance of AUV from wall(m)*
r – *distance of sensor from wall(m)*
d_{offy} – *y distance from centre of AUV(m)*
Ω_{max} – *maximum thruster speed(m/s)*
$X(k)$ – *position state matrix of AUV*
$E_x(k)$ – *process error*
F – *the function matrix in the control equations*
v – *white Gaussian noise in measurement*
H – *the function matrix for z with respect to X*

A(k) – Jacobian matrix of partial derivatives of F with respect to X

J(k) – Jacobian matrix of partial derivatives of H with respect to X

Q(k) – process error covariance matrix

K – Kalman filter gain

σ_y^2 *- error covariance of the y coordinate*

v_{des} *– the desired velocity of the AUV*

μ *– proportionality factor*

e_i *– error in parameter i*

$\tilde{i}$ *- approximate value of parameter i*

W(k) – Jacobian matrix of partial derivatives of F with respect to w

V(k) – Jacobian matrix of partial derivatives of H with respect to v

R(k) – measurement error covariance matrix

σ_x^2 *- error covariance of the x coordinate*

σ_θ^2 *- error covariance of the angle coordinate*

ω_{des} *– desired angular velocity of AUV*

β_i *–* i^{th} *experimental parameter*

$\hat{i}$ *- estimate of parameter i*

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CHAPTER 1

Introduction

In the realm of automation and robotics, the surrounding environment is a challenge for any autonomous vehicle to navigate due to the presence of stationary and moving obstacles. To effectively negotiate the environment, a vehicle of this nature must be able to sense the presence of obstacles and determine a path around them so as to avoid collision.

1.1 Sensor Systems

Just as there are many different types of robotics and autonomous vehicles, there are a variety of different ways in which these vehicles can sense their surrounding environment. This is particularly true of the underwater environment. Many surface autonomous vehicles use light as a way of sensing. However, in the sub-sea environment, light becomes attenuated over shorter distances meaning that vision becomes more difficult and RF communication and GPS (global positioning system) become impossible. Therefore, an alternative means of sensing in the water environment is required.

Although sound has limitations above water due to its short range, it travels very well in water making it the medium of choice for underwater sensor systems. The use of sound for sensing is commonly termed sonar, which is an acronym for SOund Navigation And Ranging. Sonar is becoming increasingly more applicable to many fields including surveying, commercial fishing and defence.

Sonar can be used in both an active and a passive mode. Active sonar requires a sonar transducer to emit a signal which is then received after it has been reflected from a distant surface. Passive sonar, however, only requires a receiver, called a hydrophone, to pick up signals emitted from a sound source. Active sonar allows the user to gather range information as well as direction information, whereas passive sonar can only deliver direction information.

The development of a good sensory system is imperative for an autonomous underwater vehicle to perform its required tasks efficiently. Although much of the research into sonar has been conducted by defence and commercial groups, several universities have devoted research towards creating more efficient sensory systems to improve AUV task completion.

1.2 Passive and Active Sonar Systems

Passive sonar has many applications especially in the military. It is used to identify and monitor foreign vessels by tracking their characteristic sounds. In the commercial industry, passive sonar is used to locate lost vessels that are submerged. Using an underwater acoustic receiver, an acoustic beacon attached to the vessel can be detected and used to locate its position.

Improving technology is, however, allowing submarines to become quieter and therefore more difficult to detect with passive sonar. Thus, passive sonar is no longer adequate for all underwater sensing scenarios. In addition, it is very difficult to range correctly with passive sonar because it is only capable of giving the direction of arrival. The implication of this is that two estimates of direction of arrival, from two distinct locations are required to approximate the range of the sound source. In contrast, active sonar does not require a sound to be emitted for an object to be detected and, the complete round trip time can be used to calculate the range of the object. Due to the ‘beam-like’ nature of active sonar transducers, they are also able to determine the direction of arrival of a signal.

1.3 Algorithms for Navigation with Sonar

Navigation underwater requires consideration of some key aspects. A mobile robot that navigates in an unknown or changing environment needs to maintain a dynamic model of its environment in order to update existing environmental knowledge. However, it is not possible to create a dynamic map of the environment unless the AUV can detect objects that already exist on an internal map, and any new objects that need to be added. Its sensors not only give the AUV the ability to prevent collision of the AUV with any of the obstacles that lie within an environment, but also give the AUV the ability to use the obstacles as landmarks from which to navigate towards a target. A good sensor system is useless, though, unless an appropriate adaptive algorithm for interpreting the sonar data is created.

1.4 Project Motivations

The completion of a sonar sensor system is of vital importance to the AUV because for an AUV to become completely autonomous, it needs to be able to visualise the environment around it.

Cost efficiency is an important goal of the project. The AUV may be used to research commercially viable options for underwater robotics.

The project is, therefore, to develop a cost effective sonar system that can perform active sonar. This means that a system must be developed that will not only work, but will also be inexpensive to build.

1.5 Outline of the Thesis

In this paper, the problem of designing an acoustic sensor system for an AUV is addressed. The paper is organised as follows.

Chapter 2 contains background information about underwater sound transmission and navigation theory necessary to accomplish the design of a sensor system and a navigation unit.

Chapter 3 outlines the requirements of general autonomy. This is incorporated with the cost requirements for the system to provide a complete set of requirements for the sensor system.

Chapter 4 explains the shortcomings of commercially available echo sounders in achieving the requirements outlined in chapter 3 and describes an improved prototype design for a new echo sounder circuit.

Chapter 5 discusses the actual design of the interface between the prototype and the Eyebot, including hardware decisions and software decisions that are needed to be made to achieve a seamless interface.

Chapter 6 provides the verification and testing that is completed on the echo sounder circuit and its interface with the Eyebot. Some key problems are identified and recommendations are made regarding possible solutions.

Chapter 7 discusses the development of navigational equations that will be the basis of controlling the AUV using the acoustic sensor system. The extended Kalman filter (EKF) is applied to the theoretical situation of wall-following and simultaneous location and mapping (SLAM) with the AUV. The EKF is a set of equations, which provide an estimate of state from the use of redundant sensor information. The SLAM uses the EKF to estimate its position and the position of obstacles.

Chapter 8 discusses some potential future developments as well as further research areas.

Chapter 9 provides some concluding comments and a discussion of the project's achievements.

CHAPTER 2

Underwater Sound Transmission and Navigation

There are many aspects to the detection of sound underwater and the consequent navigation of an AUV using sensors. What a sonar system attempts to do is determine the nature of the environment using only sound so that the information can be used for navigational purposes. Understanding the background in underwater sound transmission and navigation is crucial to proceeding with any part of this project.

2.1 Passive Direction of Arrival Detection

After gaining an understanding of how different universities have developed their systems, it is necessary to research further to determine how to better develop these ideas.

The Journals of the Acoustical Society of America proved to be a good source of information on the field of signal processing and underwater acoustics. Amongst the literature, key articles were found which addressed the subject of time delay estimation (TDE) and direction of arrival (DOA) calculations.

2.1.1 Time Delay Estimation

Chern and Lin [1] came up with a simple and efficient method of TDE that involved a direct computation combined with an adaptive least means squares TDE algorithm. This was advantageous in accuracy and execution time; however it required that use of a continual stream of data, which was computationally expensive. They also investigated the capability of

TDE in a multipath environment. A multipath environment is one where the original signal reflects off different surfaces before entering the receiver, resulting in delayed versions of the signal at the receiver. This confuses the receiver as it does not know exactly which the true signal is.

A completely different way of calculating the TDE, called the window-correlation technique, was brought forward by Callison et al [2]. The method was built on the principles of the cross correlation for TDE, which is briefly discussed briefly in [3]. They demonstrated that by "windowing" the data record, events could be easily identified in a noisy environment. They showed that this technique could be used for signals with a signal-to-noise ratio (SNR) of down to 0 dB and this would prove very valuable in combination with low priced transducers.

2.1.2 Direction of Arrival Calculations

Once the TDE is calculated, the DOA can be calculated using the TDE's. Berdugo et al [4] investigated DOA calculations based on time delays. Their approach was different to the usual maximum likelihood DOA estimators in that the DOA is extracted directly from the delay times of the receivers and the geometry of the receivers. They also showed that the azimuth and elevation estimator achieves the Cramer-Rao Lower Bound if the TDE achieves the CRLB as well. This is one of the simplest methods of DOA calculations using the TDE's.

Eigenstructure methods involve a projection of the signal onto a noise subspace, as a function of direction, and finding a value that minimises this quantity. The value that minimises this amount in the noise subspace will maximise the amount in signal subspace value, as the noise and signal subspace are orthogonal. Cornell University used the MUSIC algorithm [5] to calculate the DOA in 2002, which is an eigenstructure method. However, Paulraj and Kailath [6] developed a similar method that could work on wave fronts with only partial spatial coherence. The MUSIC algorithm assumes perfect spatial coherence.

This leads to the awareness that models are not perfect. Models that can better characterise realistic systems in terms of relaxed assumptions and greater error will have a superior performance in practice. Research needs to be completed in the area of errors in practical situations. Daku, Salt and McIntyre [7] investigate locating the source in a multipath environment.

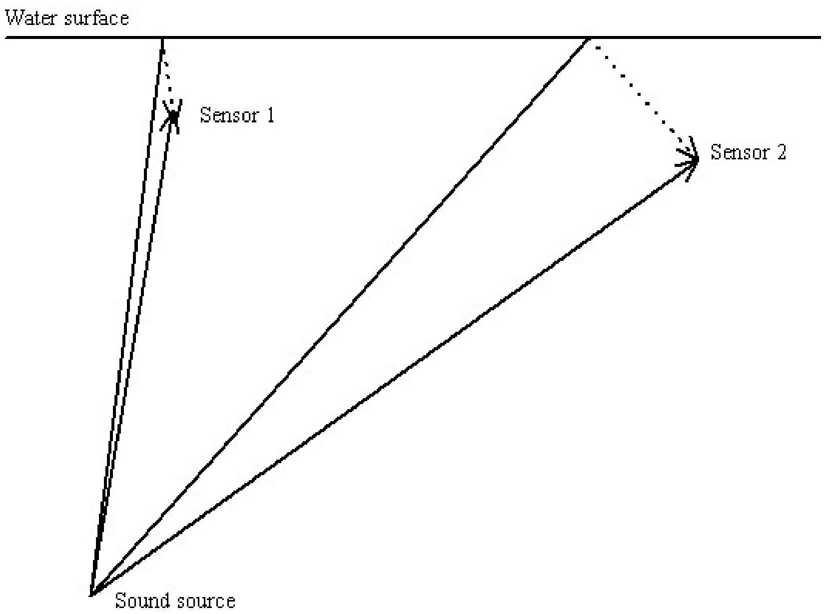


Figure 2.1: Multipath environments cause ambiguity at the receivers

They investigate the effect that reverberations, or unwanted echoes, in the multipath environment have on the ability of a system to locate a source. This is an important result as the AUV will always be in these environments. They are able to obtain the variance of localisation error and also demonstrate that in regions where the time delays of the multipath signals matched the direct signal, large source localisation errors occur.

2.2 Active Acoustic Sensor Systems

Passive sonar does not allow an AUV to avoid obstacles. Therefore, in order to navigate effectively around obstacles underwater, a well designed active sonar system needs to be designed. This involves the proper design of a hardware system and a software system to complement it. The hardware component must be designed to suit the nature of the navigation that the AUV is supposed to complete. More complex navigation on larger AUVs require the use of more sophisticated sonar systems than AUVs that only need to navigate simple obstacle courses, but these come at a much greater cost. The software and navigation algorithms work in unison with the hardware to allow for reliable navigation to be executed by the AUV.

Many different approaches have been suggested as to how to implement a sensory system on an AUV. A common approach is to create some form of imaging sonar. Williams et al. [8] describe a system with ready-made imaging sonar that provides 360 degrees of sonar vision for the AUV. This device can be directly linked to a navigational computer for analysis. The imaging sonar provides a full graphical display of the sonar returns that the device collects. By using this data, the AUV has a continuous 360 degree view of its environment to allow it to easily determine the locality of obstacles and itself within unstructured terrain.

In contrast, Ip and Rad [9] explore the use of discrete sonar sensors that are mounted at specific intervals around the robot so that it can sense the proximity of obstacles that are around it. This design is different from the one that is implemented by S. B. Williams et al. (2000) in that the AUV does not have the full view of the surrounding environment, only discrete distances at specific intervals around the robot as in Figure 2.2.

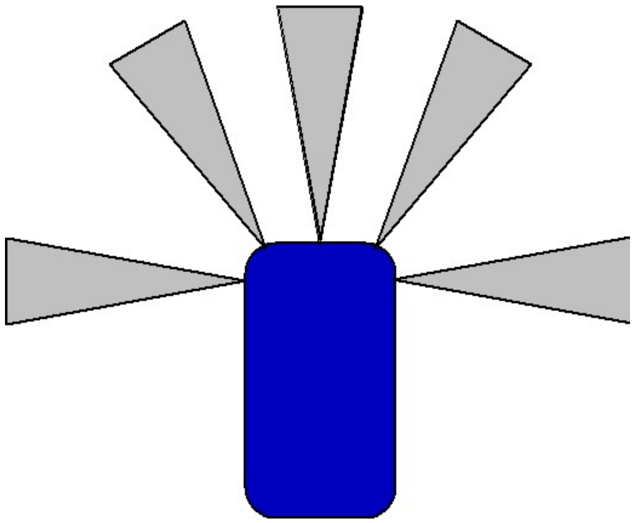


Figure 2.2: Circular array demonstrating the discrete nature of data

This design also has not been implemented on an AUV, but carries the same principle that could be used with the AUV.

Yet another different approach is discussed by Ruiz et al [10]. The entire system for their AUV consists of a Doppler velocity log, a tri-axial compass and side scan sonar. Side scan sonar is similar to the 360 degree imaging sonar except that it can only see in a narrow field of vision.

2.3 Fundamentals of Sonar Sensing

Sound transmission is a key part of underwater navigation as other forms of sensing, such as light, do not have the range capabilities that sound has underwater. In order to successfully use sonar to determine range of objects, one must understand how echo sounding works and what may affect the detection of a sonar signal.

2.3.1 Echo Sounding Principles

The echo sounder works on a very simple principle. The operation of the transmitter and receiver is similar to a basic AM transmitter and receiver.

A microcontroller, such as the Eyebot's Motorola 68332 microcontroller, will send the sensor a pulse signal. The sensor then uses this pulse signal to switch a carrier signal of particular frequency on and off, similar to an AM transmitter switching between a maximum and zero amplitude signal. This modulated pulse signal is then driven using a transformer to increase the amplitude of the signal enough for the signal to travel a significant distance. The transducer then converts the signal from electrical to acoustic vibrational energy through its piezo-electric material. Figure 2.3 shows the operation of the echo sounder.

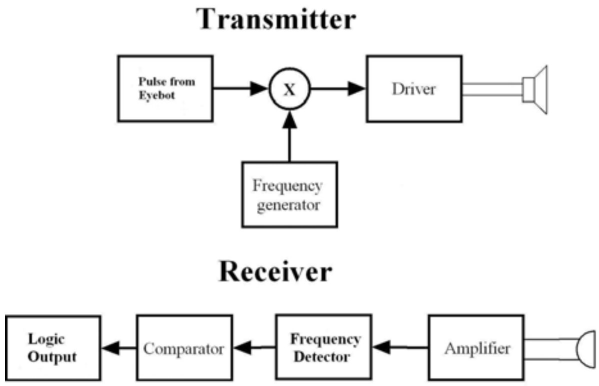


Figure 2.3: Echo sounder operation flow chart

On receiving the signal, the transducer converts the acoustic signal back to an electrical signal and then the signal is amplified to increase the signal strength. It is then passed through a frequency detector to determine if the right frequency has been received. On confirmation of a correct signal, the logic output to the microcontroller is then driven low to indicate a successful reception of the pulse signal.

2.3.2 The Sonar Equation

For active sonar where reverberations, or echoes, have a strong presence, the signal strength needs to satisfy the following equation, in order for the acoustic system to be able to identify the signal transmitted [11]:

$$SL - 2TL + TS - RL + DI \geq DT \qquad (2.1)$$

- *SL* is the sound level of the transmitter, or intensity at the transmitter

- *TL* is the loss of sound level during transmission of the signal in one direction.
- *TS* is the target strength and is dependent on the reflectivity of the given surface.
- *RL* is the level of reverberation, or echoes, encountered at the receiver.
- *DI* is the directivity index of the transducer, with a higher *DI* indicating more sound is transmitted in the desired direction.
- *DT* is the detection threshold and is defined as the signal level required allowing detection of the signal for 50 percent of the time.

All components are in decibels.

The equation is very similar for the passive sonar case, except the target strength is no longer applicable because the signal is not reflected of a target surface, and the transmission loss is only in one direction meaning that the multiplier for the *TL* is removed. The equation is as follows:

$$SL - TL - RL + DI \geq DT \tag{2.2}$$

2.4 Transmission Losses in Acoustic Systems

For sound to travel in a medium, the acoustic media must be compressible. A local disturbance cannot instantaneously travel from one point to another, but must take a finite amount of time to be transmitted, depending on the compressibility and the density of the medium [12].

In most cases in any form of media, the intensity of sound waves continuously decreases over the distance that they propagate. This can be usually accounted for by geometrical spreading of a wave as well as the absorption and scattering of energy from the wave by the medium involved. Then there are the frequency dependent components of loss and also reverberations that may cause the degradation of the signal.

2.4.1 Transmission Intensity Losses of Acoustic Systems

The intensity at two radially different points in space would have the intensities I_1 and I_2. For spherical waves, the surface area is related to the square of the radius of the wave front, thus:

$$I_2 = I_1 \left(\frac{r_1}{r_2} \right)^2 \tag{2.3}$$

However, usually the source provides some directionality for its sound waves and the intensity rule cannot apply. More often, the equation is usually modified to:

$$I_2 = I_1 \left(\frac{r_1}{r_2} \right)^n \tag{2.4}$$

The n in the equation is dependent on the directionality of the wave emitted from the source and is non-integral and less than two.

2.4.2 Scattering and Absorption Losses of Acoustic Systems

Scattering and absorption of the medium provide another common loss of intensity of the sound wave. The percentage loss over a particular distance is constant and can thus be written as an exponential term with an absorption coefficient, a. This gives rise to the equation:

$$I_2 = I_1 \left(\frac{r_1}{r_2} \right)^n \exp\left[2a(r_2 - r_1)\right] \tag{2.5}$$

By taking the logarithms of both sides of the equation, and then rearranging the equation, the propagation loss, N, or the difference of intensity can be termed as:

$$N = 10n \log_{10}\left(\frac{r_1}{r_2} \right) + \alpha(r_2 - r_1) + H \text{ dB} \tag{2.6}$$

The H term accounts for the differences between the theoretical losses and that which are observed, due to diffraction, reflection and refraction.

2.4.3 Frequency Dependent Losses in Acoustic Media

Knowledge of the frequency dependent losses is important as it assists in determining what frequency is ideal for echo sounding.

The losses are associated with dissipation of heat due to frictional forms of energy. There are generally two losses of this form that are dominant in sound propagation; these are bulk viscosity and absorption due to relaxation [13]. It is usually the bulk viscosity that has the greatest effect on the attenuation of sound waves and is given by the following equation [2]:

$$\alpha_v = \frac{2v\omega^2}{3c^3} \tag{2.7}$$

In the equation, υ is the kinematic viscosity and ω is the angular frequency of the sound wave. This loss has a square dependence on frequency.

The second loss of excess absorption due to relaxation can be explained by the absorption and retransmission of energy from the medium. The time it takes for a particle to return to a relaxed state after it has absorbed energy from a sound wave is called the relaxation time, τ. If the energy is returned to the sound wave in phase, then the energy is added constructively. However, if the energy is returned to the sound wave out of phase, then it causes destructive interference. If the period of the wave is comparable to the relaxation time, then high attenuation occurs. The equation for this phenomenon is given by [12]:

$$\alpha = 2\alpha_m \frac{f^2 f_r}{f^2 + f_r^2} \tag{2.8}$$

The α_m term is the maximum attenuation coefficient at the relaxation frequency, $1/2\pi\tau$, and *f* is the frequency of the sound wave.

It can be seen from the above losses, that the any acoustic system is limited in range. The frequency of the sound wave propagating through a medium has a significant effect on the attenuation of the sound wave in the medium and thus on the effective range of the underwater acoustic system.

2.4.4 Reverberation Considerations for Acoustic Systems

Acoustic signals, especially in an environment that is relatively enclosed, such as a swimming pool, are backscattered off the many surfaces, meaning that the signal obtained by the receiver will contain components from many different ray paths, masking the direct path signal. These components are called reverberations. There is an upper limit to the source level that can be transmitted. This is due to the fact that higher source levels represent higher reverberation levels. This limit will be where the increase in signal to noise ratio (SNR) from the source level will be non-beneficial for the receiver as it is countered by the decrease in SNR from reverberation level, using equation 2.1. Thus the echo sounder must be designed to have just enough signal level to transmit over a desired range. Any more signal level will contribute unnecessarily to reverberation.

2.5 Characteristics of Acoustic Transducers

An acoustic transducer is a physical device that converts electrical energy to vibrational energy and vibrational energy to electrical energy. This section discusses the characteristics of an acoustic transducer. This includes the resonant frequency of an acoustic transducer and the directivity of a transducer.

2.5.1 The Resonant Frequency of an Acoustic Transducer

All materials have a resonant frequency, including the membrane of an acoustic transducer. This property of the material can be used to great advantage in the transmission and reception of an underwater acoustic signal. Resonance is an increase in the oscillatory energy absorbed by a material when the frequency of the oscillations matches the material's natural frequency of vibration. This property is very useful because it means that by sending the right frequency signal through the transducer, a higher percentage of power can be converted efficiently into the transmission of the signal. The resonance of the transducer also has another valuable property. Since the transducer can convert electrical signals to vibrational signals and vice versa, the same effect can occur when the transducer is receiving a signal. This is advantageous because the transducer material, therefore, naturally filters out noise from other frequencies, resulting in a better signal to noise ratio when the signal is received. Transducer materials can be made with the precise physical properties in order for it to resonate at a desired frequency. Figure 2.4 shows that a transducer has a resonant and anti-resonant frequency, shown as the minimum and maximum in the magnitude plot, respectively. A transmitter should transmit at the resonant frequency and the receiver should receive at the anti-resonant frequency. In a dual transducer echo sounder system, the transmitter transducer's resonant frequency is tuned to the receiver's anti-resonant frequency. In a single transducer system, the two frequencies are made as close as possible, but the system will be transmitting and receiving at a frequency that is not optimal, but between the two frequencies.

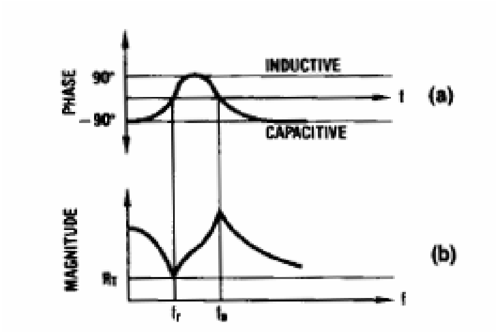


Figure 2.4: The resonant and anti-resonant frequency of a transducer [14]

The resonant and anti-resonant frequencies used for many boat depth sounders lie close 200 kHz. The resonant frequency of air transducers, such as the ones used in Polaroid's sonar ranging system operate at a resonant frequency of 40 kHz.

2.5.2 Directivity of Circular Transducers

The directivity of the transducer is an important characteristic of an acoustic system. It is basically how directional the signal is transmitted. A transducer with high directivity will transmit more power in a particular direction than one with a lower directivity. By understanding the geometry and nature of the transducer, the directivity of the transducer can be found.

Firstly assume that the transducer is used for receiving a cosine wave that is sent from a source that is far away, such that the distance between the source and the transducer is very large in comparison to the length of the transducer. This allows the incoming waves to be considered parallel to each other as in Figure 2.5.

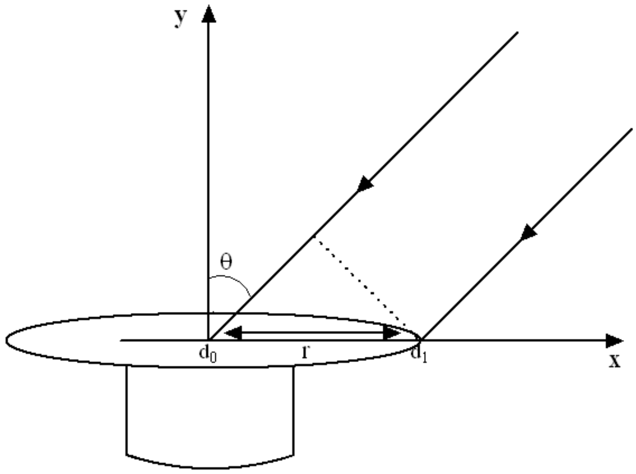


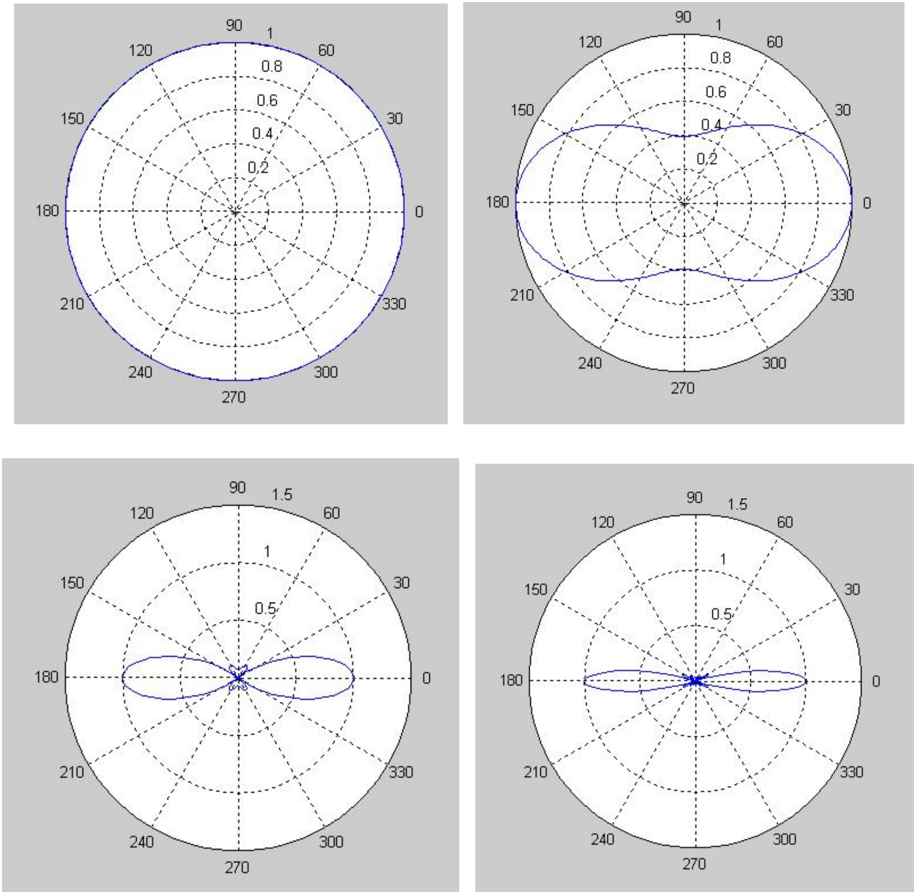
Figure 2.5: Sound waves being received by a transducer

The wave that is received at the centre of the transducer is $\cos(\omega t)$. At a point that is *r* away from the centre, the wave received is $\cos(\omega t - kr)$, where *k* is $\frac{2\pi}{\lambda}\sin\theta$ and λ is the wavelength in the medium. The sensitivity is not uniform overall the surface, since the circular surface is wider in the middle than on the sides.

This directivity function for a circular transducer is derived to be a first order Bessel function of the first kind [12] as given below:

$$D(K) = 2\frac{J_1\left(\frac{Kl}{2}\right)}{\frac{Kl}{2}} \tag{2.9}$$

Polar plots from Matlab using equation 2.9 demonstrate the directivity of the transducer. As can be seen by the plots in Figure 2.6, a higher frequency transducer produces a more directed signal resulting in a greater range and finer angular resolution or that better reception of the signal can be achieved. All the plots have the same maximum intensity of 1.



Figures 2.6 a), b), c) & d): Directivity for transducers of frequencies 5Hz, 40 kHz, 100 KHz and 200 kHz, respectively

Using the same principles in reverse, it is easy to see why the directivity function would be the same for a transmitter of the same type of transducer.

Therefore, if a greater directivity for the transducer is required at the receiver, then a higher frequency transducer will be necessary. However, this will come at a cost of more losses that are presence with higher frequency signals, reducing the signal to noise ratio.

There is clearly an optimal choice for the frequency of the transducer, depending on the application. For a better angular resolution, the frequency must be high. This results in greater losses, and thus a smaller range. However, if the angular resolution is not important, then choosing a lower frequency will result in a greater range for the sensor.

2.6 The Extended Kalman Filter

The Kalman filter is a set of mathematical equations that provides an efficient method to estimate the state of a system that minimises the mean square error of the estimates. The

Kalman filter has been used to providing estimates of past, present and future states of a system, meaning that the filter is a very powerful tool [15].

The Kalman filter is used to provide estimates of a state of a process or a system that is governed by linear control equations. However, if a non-linear set of control equations are used to define a system, another method of filtering is required.

The extended Kalman filter is set of equations that are used to solve non-linear control equations. This works on a principle similar to that of a Taylor series, where the estimation is linearised around the current estimate of the system, which may not have a linear relationship.

2.6.1 The Development of Filter Equations

Welch and Bishop [15] give an introduction to the Kalman filter. Control equations are developed and then used to create the EKF equations.

The control equations for a non-linear system are as follows:

$$\begin{aligned} X(k) &= F(X(k-1), V(k)) + w(k-1) \\ z(k) &= H(X(k)) + v(k) \end{aligned} \tag{2.10}$$

where the w and v terms are the random process and measurement noise variables, with zero mean and a covariance of Q and V respectively. Now, using a process similar to a Taylor series, the following equations have been developed to provide a consequent estimate based on the current estimate of the state:

$$\begin{aligned} X(k) &\approx \tilde{X}(k) + A\left[X(k-1) - \hat{X}(k-1)\right] + Ww(k-1) \\ z(k) &\approx \tilde{z}(k) + J\left[X(k) - \tilde{X}(k)\right] + Vv(k) \end{aligned} \tag{2.11}$$

where $\tilde{X}(k)$ and $\tilde{z}(k)$ are approximate state measurements, $\hat{X}(k-1)$ is the previous estimate of the state.

The A, J, W and V matrices are therefore the Jacobian matrices of partial derivatives from equation 2.10.

- A is the matrix of partial derivatives of function F with respect to X
- W is the matrix of partial derivatives of F with respect to w
- J is the matrix of partial derivatives of H with respect to X
- V is the matrix of partial derivatives of H with respect to v

$$
\begin{aligned}
A_{[i,j]} &= \frac{\partial F_{[i]}}{\partial X_{[j]}} \\
W_{[i,j]} &= \frac{\partial F_{[i]}}{\partial w_{[j]}} \\
J_{[i,j]} &= \frac{\partial H_{[i]}}{\partial X_{[j]}} \\
V_{[i,j]} &= \frac{\partial F_{[i]}}{\partial v_{[j]}}
\end{aligned}
\tag{2.12}
$$

New errors can be given as:

$$
\begin{aligned}
\tilde{e}_{x_k} &= A\left[X(k-1) - \hat{X}(k-1)\right] + \xi_k \\
\tilde{e}_{z_k} &= J\left[\tilde{e}_{x_k}\right] + \eta_k
\end{aligned}
\tag{2.13}
$$

where the two errors are:

$$
\begin{aligned}
\tilde{e}_{x_k} &= x_k - \tilde{x}_k \\
\tilde{e}_{z_k} &= z_k - \tilde{z}_k
\end{aligned}
$$

However, the equations in (2.13) appear very similar to the standard linear control equations that an ordinary Kalman filter can estimate for. This leads to the use of the Kalman filter to provide an estimate of the new error from (2.13), which will be called $\hat{e}_k$. Now according to the ordinary Kalman filter equations:

$$
\hat{e}_k = \hat{e}_k^- + K\left(\tilde{e}_{z_k} - L\hat{e}_k^-\right) \tag{2.14}
$$

Assuming that the predicted error is equal to zero, then equation 2.14 simplifies to:

$$
\hat{e}_k = K\tilde{e}_{z_k} \tag{2.15}
$$

Since $\hat{e}_k$ represents $\tilde{e}_{x_k}$, the following can be written:

$$
\begin{aligned}
\hat{e}_k &\equiv \tilde{e}_{x_k} \\
\hat{e}_k &= x_k - \tilde{x}_k \\
x_k &= \tilde{x}_k + \hat{e}_k
\end{aligned}
\tag{2.16}
$$

However, since value of the position state cannot be known, it must be replaced with an estimate of the position state.

$$
\begin{aligned}
\hat{x}_k &= \tilde{x}_k + \hat{e}_k \\
&= \tilde{x}_k + K\tilde{e}_{z_k} \\
\hat{x}_k &= \tilde{x}_k + K\left(z_k - \tilde{z}_k\right)
\end{aligned}
\tag{2.17}
$$

2.6.2 The Extended Kalman Filter Equations

Deviating slightly from the equations given by Welch and Bishop [15], the W and the V matrices are assumed to be identity matrices. This is done by assuming that the measurement noise and process noise are both white and additive and applying this to equation 2.12. This simplifies the filter equations.

The Kalman filter equations for the prediction stage are given as:

$$\begin{aligned} \hat{X}(k)^- &= F\left(\hat{X}(k-1), V(k)\right) \\ P(k)^- &= A(k)P(k-1)A(k)^T + Q(k-1)^T \end{aligned} \tag{2.18}$$

The filter equations for the correction stage, using equation 2.17 are:

$$\begin{aligned} K(k) &= P(k)^- J(k)^T \left[J(k)P(k)^- J(k)^T + R(k)\right]^{-1} \\ \hat{X}(k) &= \hat{X}(k)^- + K(k)\left[z(k) - H\left(\hat{X}(k)^-\right)\right] \\ P(k) &= (I - K(k)J(k))P(k)^- \end{aligned} \tag{2.19}$$

To proceed with these equations, the first estimates from the prediction stage are set by the user. The filter then corrects these estimates, using equation 2.19, before proceeding back to the prediction stage and this process is continued recursively, back and forth through the prediction and correction stages.

2.6.3 Wall Following and the Extended Kalman Filter

One common task for an autonomous vehicle is to follow the surface of a wall. This requires the measurements of the compass and the speedometer of the vehicle to be very accurate. However, with any measuring device, there will always be errors in their measurements, which will affect the accuracy of any positioning that these devices estimate.

One way to decrease the errors in the estimations of positioning is to use other sensors to more accurately assist with the localisation of the vehicle. The sonar sensors can be used to assist with the correct positioning of the vehicle whilst measuring the distance from the edge of the swimming pool. By fusing the sonar data with the speed and direction readings, a more accurate position of the AUV can be found.

A Kalman filter allows for the fusion of sensor readings. However, for a control system for a wall following task, the control equations are non-linear. This means that the ordinary Kalman filter cannot be used. Instead, a variation on the Kalman filter, known as an extended Kalman filter, which attempts to develop a linear set of equations from the non-linear control equations, is used.

2.6.4 Simultaneous Location and Mapping (SLAM)

Wallner and Dillmann [16] describe a method of map refinement that uses a local model of a map stored in a database and readings from the sonar sensors to produce a model of the robot's perceivable environment. Their method of mapping the robot's environment can be broken down into two parts. The first part is the maintenance and refinement of the local map that is stored within the robot's memory. The second part is a grid based modelling of new obstacles found by the robot and integrated into the current map.

The robot basically uses its sensor reading to discover obstacles. When it has located them, it checks to determine if the obstacle is on the robot's internal map. If it is not, then a new obstacle is mapped. Otherwise, the new sensor reading is used to improve the robot's idea of where the obstacle is and increase the probability of the obstacle's existence. This process of increasing probability is continued until the object has presented enough information for it to become 'known' to the robot. This, too, must use the EKF for more accurate localisation of the obstacles and the robot.

CHAPTER 3

Project Requirements

The main motivation for designing the AUV is to provide the basis for future research in underwater control systems. Thus, the AUV needs to be designed so that it can be easily adapted to suite many fields of research. However, the AUV must meet financial constraints, so feasibility and functional analysis are significant elements in this project. The main goals for the design of the AUV are therefore to create a system that is adaptable, functional and cost effective.

3.1 General Requirements for Autonomy

A set of requirements needs to be developed for the AUV so that the AUV can be universally autonomous. The meaning of universal autonomy is the control of a vessel, without external communication, to navigate in any number of environments.

All that the AUV can use for navigational aids are its onboard camera and its complement of sonar sensors. The camera is very limited in its ability to assist with navigation as it is pointed downwards with no ability to pan left or right, up or down. The purpose of the camera is to detect navigational markers or key objects on the pool floor that will assist the AUV in orientating itself in the environment. Thus it is mainly up to the sonar system to determine the proximity of and avoid collisions with obstacles and localise the AUV within the environment. The sonar system will also assist with determining the depth of the AUV in the pool.

The following basic requirements were derived for the sonar system to ensure the AUV's successful navigation of the environment using only sound:

- To be able to avoid an obstacle that may hinder the path of the AUV
- To determine the range an obstacle is from the AUV
- To detect landmarks that may assist the orientation of the AUV
- To map new landmarks to assist in the localisation of the AUV

3.1.1 Competition Use

The best means of testing the AUV in different environments is to place the AUV into competitions that have a changing set of tasks to accomplish each year. Not only does this provide the AUV with a continually different environment to test its capabilities, but the competitions allow respective AUV groups to have their systems judged against other groups. This is important in the development of ideas as different groups become aware of new approaches to solving problems associated with sensing and controlling an AUV.

As previously mentioned in chapter one, the AUVSI [17] annually hold competitions for all types of autonomous vehicles including underwater vehicles. For the past few years, the AUVSI has set similar tasks for the acoustic component of the competition mission. The goal of these missions is to demonstrate vehicle autonomy by being able to sense acoustic cues in the water and determine the direction of the cues.

A new competition will be introduced in the year 2005 that will enable university groups and possibly industry groups in the Asia Pacific region to compete in an event with similar aspirations to the underwater competition based in North America, but with different competition missions and tasks. Possible tasks for a competition like this could include [18]:

- Wall following
- Pipeline following
- Target Finding
- Obstacle mapping

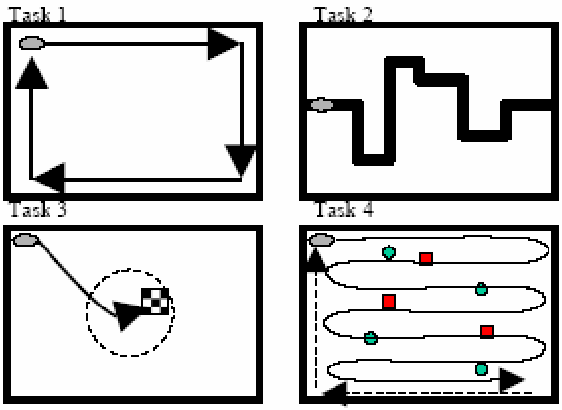


Figure 3.1: Possible competition tasks including wall following, pipeline following, target finding and obstacle mapping [18]

Though the AUV is not designed primarily for competition use, a competitive environment will nurture the improvement of the design standards for the AUV.

3.2 Software Requirements for Sonar Navigation

Low level software needs to be written for the successful integration of sensors to the AUV and to provide navigation control for the AUV.

Since most sensors relay a signal from their output, the AUV needs to be able to interpret these signals and convert them to real data. This will often require low level programming of the interface with the sensor.

Once this has been achieved, the sensor can be used to assist with the navigation of the AUV. This will usually mean using some navigational algorithm to control the AUV to perform certain tasks. The navigational control must be robust as the environment is never completely predictable. The algorithms must be able to adapt to these environments.

3.3 Cost Requirements for the AUV

Financial constraints for the project are a limiting factor in the development of sonar system for the AUV. Some requirements may need to be compromised or even removed so that more important objectives can be completed.

Due to the limiting nature of these constraints, not all options for the design of a system are feasible. Systems will need to able complete the tasks required of them at the cheapest possible cost. Therefore, whilst some options may be able to perform better at completing a task, such as having a greater detection range, they may not be as feasible when compared with a system that may perform slightly worse but is functionally adequate for what is required and significantly decreases the cost of the system.

With this in mind, the design requirements of the sonar system need to be specified as the minimal possible requirements that still allow the AUV to complete a set task at the least expensive cost.

3.4 The Complete Requirements for the Sonar System

Incorporating the above requirements into the one set, the following are the requirements for the sonar system of the AUV:

- Resolution for ranging of at least 5cm
- Maximum detection range of 5-10 metres
- Data rate of at least 10Hz
- Data must be easily transferred to the CPU or Eyebot
- System must be less than $1000

The requirements for range, resolution and data rate stem from the need for precision navigation.

The data needs to be updated at a fast enough rate to enable the AUV to attain a more continuous view of the environment. It becomes very difficult to steer the AUV when there are relatively long intervals between the data. The data rate is the most important of the three. If the data rate is insufficient, then the AUV may be blind at crucial times.

The range of the AUV is needed to enable the AUV to gain a larger view of the environment around it. However, data rate is dependent on the range of the echo sounders. To have a longer range, the signal must be stronger, so that the signal can travel further in the water. However, this means that it will take longer for the signal to attenuate to a level that is undetectable for the echo sounder when located in a highly reverberant environment. Only at this level can the sensor begin to sound again. Thus, there is a trade off between the data rate and the range of the echo sounder. The range of 5 metres has been chosen for the design of this project.

It will not be possible to accurately gauge how far an object or surface is from the AUV without a good resolution. This will be most obvious when the AUV is required to follow the pool wall. A worse resolution will result in the AUV oscillating about a mean distance from the wall. However, considering the size of the AUV and the pool environment in which the AUV is in, a resolution of at most 5cm will suffice.

By having as many sensors as possible pointing in as many degrees as possible, the AUV is better able to see all the obstacles around it and can therefore judge its position amongst the obstacles better.

The achievement of these requirements for designing the sonar system of the AUV will fulfil both objectives for the design of the AUV, which are functionality and cost effectiveness.

CHAPTER 4

Hardware Design

The hardware design of the active sonar system is crucial to the AUV successfully determining its distance from an obstacle. It is imperative the system be able to communicate this information accurately to the CPU so that informed decisions can be made while the AUV navigates the surrounding environment.

Although imaging sonar systems are preferable, due to financial constraint, a discrete sonar system is employed. Consequently, the active sonar system is comprised of four individual echo sounding units. Three facing the port, starboard, and bow sides for detection of lateral obstacles and one facing down to provide depth measurement.

4.1 Commercially Available Echo Sounders

Although not of high quality, commercially available depth sounders are relatively inexpensive, and are adequate for simple echo sounding tasks. It was necessary, however, to determine the suitability of using depth sounders as proximity sensors on the AUV. One such depth sounder was tested, and the results are described in the following sections.

4.1.1 The Navman Depth2100 Echo Sounder

Initially, the Navman echo sounder was purchased to provide depth readings for the AUV. The idea came from the fact the boats could achieve depth readings from commercially available echo sounders with reasonable accuracy at a far lesser cost than purchasing a digital altimeter.

From this idea of depth measurements stemmed the idea that the echo sounder did not necessarily only have to measure depth, but could also measure proximity of objects and surfaces in the lateral directions. Thus, although the echo sounders were initially only intended for depth measurements, it was determined that an array of these sensors could function as a complete acoustic sensor system.

However, an acoustic sensor system based on echo sounders requires a device of far superior performance than the Navman Depth2100, which was consequently no longer suitable for the project.

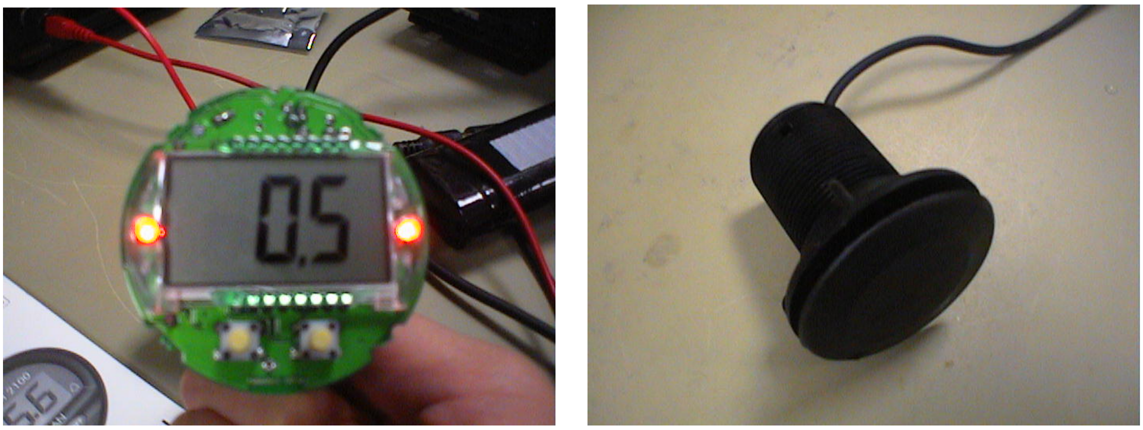


Figure 4.1a) & b): the Navman echo sounder circuit board and transducer

4.1.2 Shortcomings of the Navman Depth 2100 Echo Sounder

There are several key problems with the Navman echo sounder that make the device a poor choice when it comes to use on the AUV. These problems are:

- An inaccurate resolution of 10cm
- A very slow data rate of 1 Hz
- A serial data line
- Lack of programmability

Having a resolution of 10cm means that the AUV will not be able to accurately maintain specific distances from objects or maintain a specified depth. Since the echo sounder will be used mainly for navigation and collision detection, the resolution needs to be better than 5cm, as stated in chapter 3.

The data rate of 1 Hz is also very slow. This means that the AUV will not be able to adjust quickly enough to any immediate changes to the environment, and must rely on the fact that changes are gradual and the environment is predictable. However, the environment is never completely predictable, thus the data rate of the echo sounder needs to be much faster to accommodate for uncertainties within the environment.

The echo sounder is not programmable which means that the echo sounder parameters are not controllable. The only aspect of the echo sounder that is controllable is turning the echo sounder on or off. Thus, if data is required, the echo sounder is turned on, if the data is not required, the sounder is turned off. This means that the sounder is not very practical for the AUV.

The serial data line results in the following problems:

- Insufficient serial ports available on the Eyebot
- The need for development of a protocol to distinguish between each of the individual echo sounders, which are operated simultaneously.
- Unacceptable delays in sound detection due to potentially longer than four second read cycles.

Taking these factors into consideration, the Navman echo sounder is not functionally adequate for use with the AUV. Furthermore, only by significantly changing the hardware, and possibly the software, can the Navman echo sounder perform the tasks required.

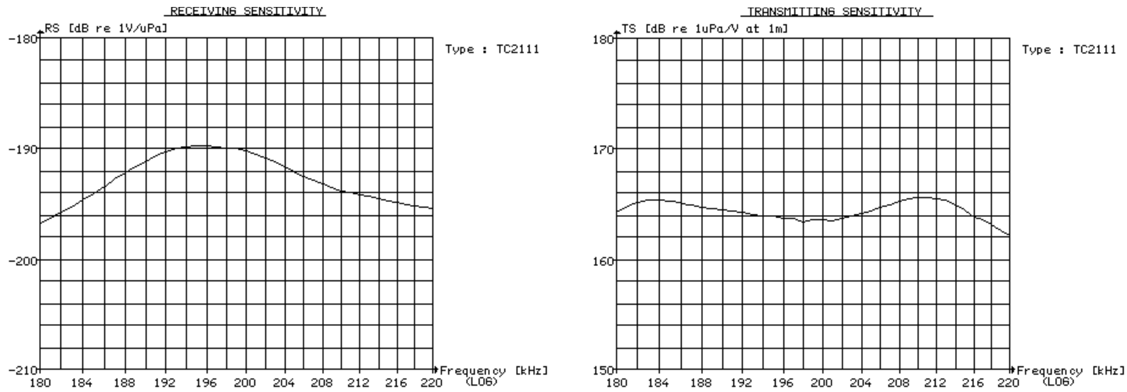
4.2 Need for a New Design

A better option is to design a new echo sounder circuit board with the functionality required to achieve the goals of the AUV. Since cost is such a limiting factor on the design of the acoustic sensor system, there must be a significant amount of bias towards designing a cheaper system. Also, as the AUV is completely operated on batteries, there must be a consideration towards the power consumption of the system, but not at the expense of cost benefits.

4.2.1 Number of Transducers

The number of transducers is an important decision concerning the design of an acoustic system. The options are a one or two transducer echo sounder.

The main advantage of a two transducer system is that the resonant frequency of the transmitter transducer can be matched to the anti-resonant frequency of the receiver, as discussed in chapter 2. This will enable maximum power emission efficiency for the transmitter and maximum power reception efficiency at the receiver. At other frequencies, the transducer has some reactance, meaning power is not transferred as efficiently. A single transducer attempts to minimise the gap between the two frequencies, but as can be seen in figure 4.2, they do not exactly match. The plots demonstrate the receiving anti-resonant frequency as a peak in figure 4.2a), and the transmitting resonant frequency as a local minimum in figure 4.2b). The optimal frequency for a single transducer will be close to both the anti-resonant and resonant frequency.



Figures 4.2 a) & b): Receiver and transmitter sensitivities of a single transducer [19]

The problem is that the two transducer system is more expensive due to the need to purchase two transducers. As there is a greater priority placed on cost than there is on power conversion efficiency, the single transducer design is selected. The transducer will still resonate, but not to the same extent as a two transducer system.

4.2.2 Choice of Transducer

There are many types of transducers available on the market, varying in both price and quality. Obviously, the more expensive the transducer, the better the quality is. Thus, the choice of transducer must hinge heavily on the type of application the transducer will be used for. It must be investigated whether a less expensive transducer can fulfil the requirements for the acoustic sonar system.

The type of application that the echo sounder is to be used for is basic echo sounding. This application requires the use of a transducer that can effectively transmit and detect an acoustic ping in water. Since the range for the echo sounder will be in the order of 10 metres, the sensitivity of the transducer is not a very significant concern.

Another factor that is used to determine the type of transducer is the power consumption of the transducer and the echo sounder as a whole, as stated earlier in this section. This can be accomplished by choosing a transducer that outputs less power. However, the power consumption is also closely linked to the range of the echo sounder. If low power consumption is required, a smaller power acoustic ping can be transmitted, meaning that the range of the echo sounder is reduced.

More expensive transducers only have slightly better physical characteristics than the much cheaper transducers found on fish finders and boat depth sounders. The difference between the two comes down to the fact that the more expensive transducer is developed closer to specification than the cheaper transducer, and thus is more reliable and better specified.

However, for the purposes of the project, the cheaper Navman transducer is capable of achieving the tasks at a lesser expense than the Reson TC 2111 and consumes less power [19], [20]. Thus, since a greater weighting has been placed on producing the circuit at the least cost, the choice of the inexpensive transducer over the more specified transducer has been made.

4.2.3 Choosing a Design for a New Circuit

Much time was spent on investigating ways in which to approach the task of designing a new echo sounder circuit that could fulfil the requirements set in Chapter 3. There is little information on the design of underwater echo sounder circuits that is readily available both on the Internet and in literature. The initial time spent researching methods focussed mainly on adapting ultrasonic switches or AM transmitters. Figure 4.3 shows an example of an ultrasonic switch that was tested for suitability [21]. Figure 4.4 shows the circuit layout on a prototyping board.

The transmitter for this design was flawed, with respect to underwater echo sounding, as there is less than 9V supplied to the transducer, with this voltage not being stepped up before transmission. This meant that the system could only work within a range of less than 1m due to a lack of signal intensity, which is not sufficient when the system is converted to an underwater system.

The receiver was a very basic circuit of only a few parts. The first was a two transistor amplifier. The second was an envelope detector consisting of a rectifier followed by a low pass RC filter. The comparator at the end just sets the threshold for the logic. The simplicity of this design meant that it was very limited in its ability to detect signals.

Other problems with these types of design are that they cannot be easily ported to a single transducer system and that there are many components that will reduce the SNR of the system, decreasing the chances of detecting an echo. A circuit that caters for a single transducer and has fewer components on the circuit board is needed for this project.

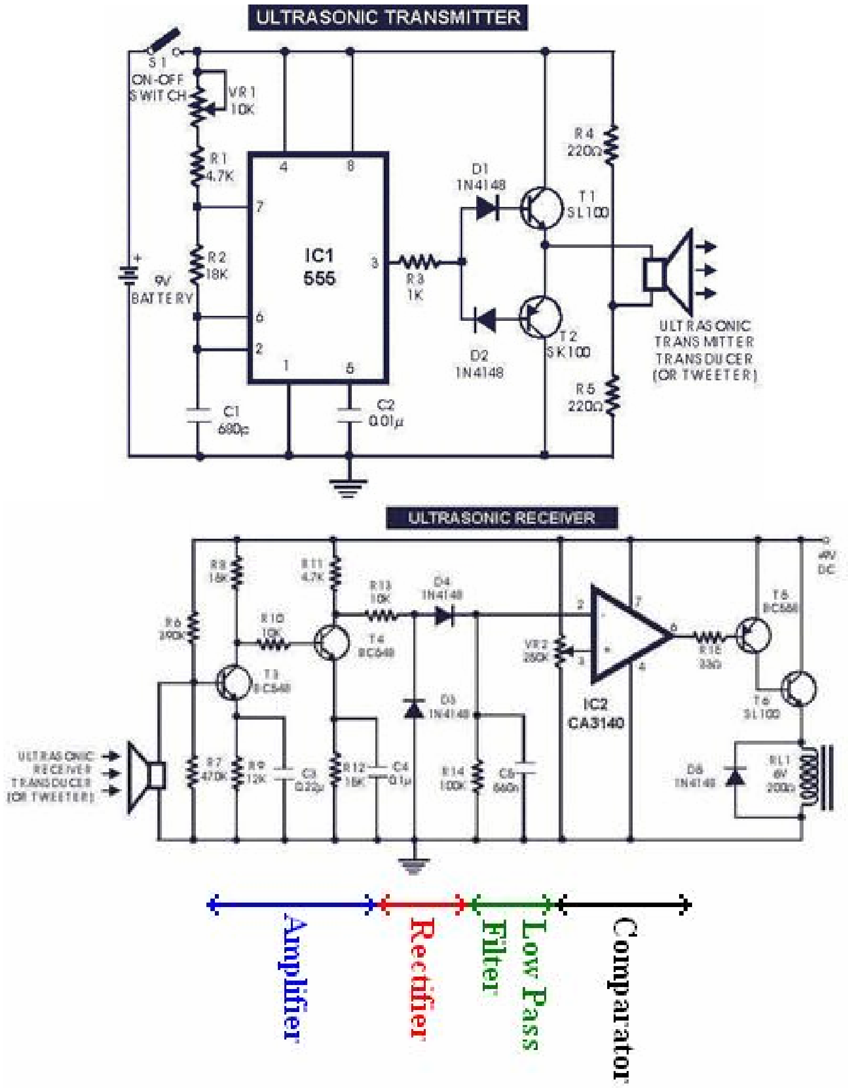


Figure 4.3: Design for an ultrasonic switch [21]

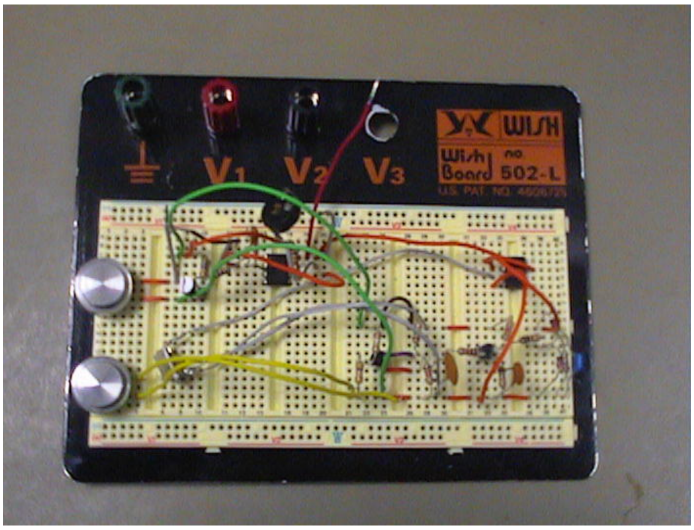


Figure 4.4: Circuit layout of an ultrasonic switch

4.2.4 Design of a Circuit Prototype

A design that utilised a National Semiconductor LM1812 ultrasonic transceiver chip was used for development of the echo sounder circuit. Though the line of LM1812 chips is no longer in production, they are still available from vendors internationally. This design was determined to be the best to design a prototype with.

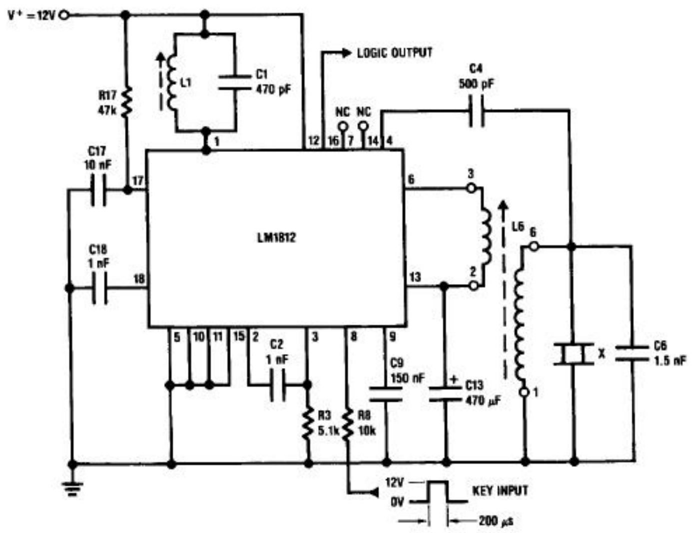


Figure 4.5: Circuit diagram of the echo sounder circuit design [14]

Obtaining a copy of the datasheet from National Semiconductors [14], it was found that the chip could be used in a circuit (Figure 4.5), similar to one designed in the datasheet, which could fulfil the task of underwater echo sounding.

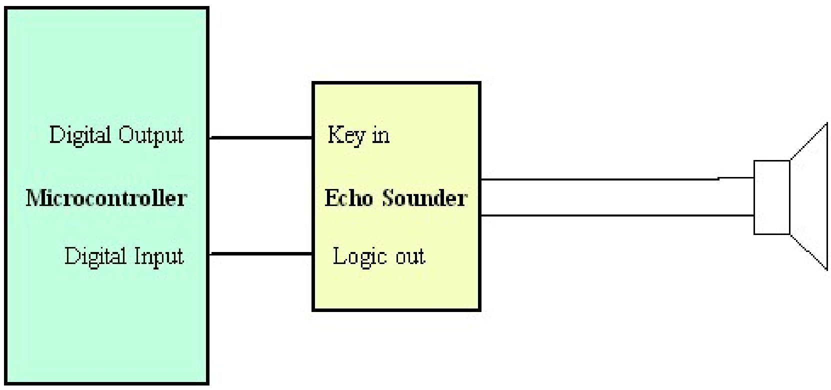


Figure 4.6: Echo sounder communication lines with a microcontroller

Figure 4.6 demonstrates how the microcontroller communicates with the echo sounder circuit. Basically, the circuit is a device that links to a microcontroller and only has provisions for sending and receiving a 200 kHz frequency from the transducer. The pulse, which is modulated to the 200 kHz carrier frequency, must be supplied by the microcontroller. Also, any timing must be accomplished by the microcontroller.

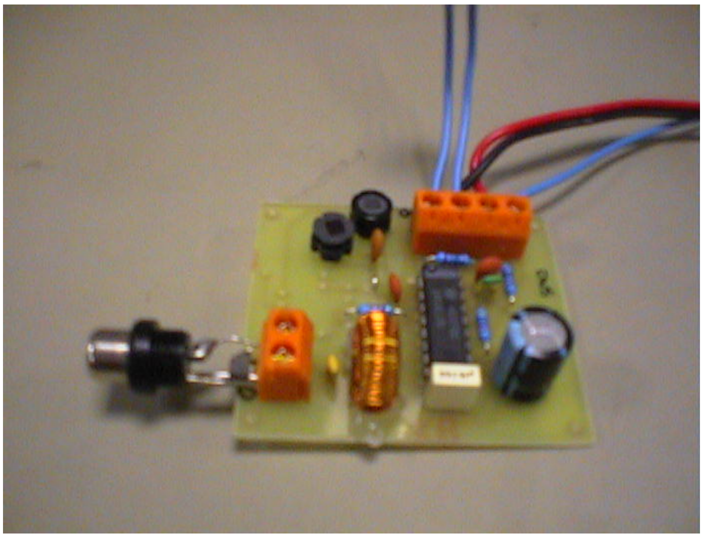


Figure 4.7: Circuit board of prototype

4.2.5 Advantage of the Prototype Design

It may seem like the whole system will be limited in ability due to the circuit's simplicity, but it is in fact quite beneficial. Since most of the echo processing is completed by the microcontroller, the system becomes quite flexible, allowing it to closely match the design requirements for the project.

Because the timing is handled by the microcontroller, the echo sounding system is no longer limited by a set program, preset in a microcontroller of the echo sounder, like the Navman echo sounder. In the Navman echo sounder, the echo sounding program, including the transmission, reception and timing of the ultrasonic pulse was already preset in the echo sounder's microcontroller. The problem with this was that it is virtually impossible to reprogram the microcontroller with more efficient code, as the code set for the microcontroller is unknown. Thus the only output possible from this echo sounder is the serial data in the NMEA 0183 protocol format, which only provides serial data at a rate of 1Hz. Thus the key parameters of range, resolution and data rate can be adjusted on the new circuit prototype.

Since the input and output of the echo sounder are only logic and not in serial form, they can be easily interfaced to any microcontroller's digital output and input (IO) with no need to provide a protocol for transferring data. Also, with the basic input and output of the circuit, it is not difficult to interface more than one device to the digital IO, as microcontrollers usually have an abundance of IO ports. Thus, interfacing with the Eyebot is much simpler.

CHAPTER 5

The Eyebot Interface

The interfacing of the echo sounder to the Eyebot is an important design step because without proper interfacing, both in hardware and software, the sensor will not be able to communicate information to the Eyebot, rendering the sensor useless. Design choices such as the type of interface, method of connection and implementation of software are discussed in this chapter.

5.1 Choosing the Interface

The echo sounder circuit is very basic in its technique for transmitting and receiving a signal. The circuit provides a modulated signal that needs to be switched on and off for an acoustic pulse to be transmitted. The return signal is then detected by the circuit and represented by a low on the circuit's logic output. There is no provision for the processing of distance measurements by the echo sounder. All this must be completed on a different circuit or microcontroller. Before the echo sounding circuit prototype can be used with the Eyebot, an interface with the Eyebot needs to be established.

5.1.1 Eyebot versus Onboard Transmission and Detection

There are two ways to establish a connection between the echo sounder and the Eyebot, each with there own benefits and disadvantages, which need to be assessed so that the most effective solution can be found.

The method of performing the signal processing on the Eyebot's Motorola 68332 microcontroller is preferred, as opposed to performing it on a separate microcontroller. The reasons for this are:

- There is no need to set up a transfer protocol between the Eyebot's microcontroller and the other microcontrollers. If the data was processed separate from the Eyebot microcontroller, then the data would need to be transferred to the Eyebot. The only solution is to send data via a serial protocol. This would consume much time, and it would be difficult to multiplex multiple sounders to the one Eyebot. It is more efficient to process the data on the Eyebot.
- There is no need to purchase other microcontrollers and learn the code set for them.
- The Motorola chip has a theoretical resolution capability of up to 238.4ns which is more than sufficient for echo sounding.

The only disadvantage of this solution is that the Eyebot will not be able to process other data during the detection and processing of the data from the echo sounder, which may have an effect on the timing of other processes. However, as long as this is taken into consideration whilst designing the interface, then the problem can be minimised or even eliminated.

Thus the choice is clear that the Motorola MC68332 CPU (Central Processing Unit) on the Eyebot will be used for the signal detection of the echo sounder. The Motorola microcontroller has a number of features allowing for flexibility in the development of an interface between the Eyebot and the echo sounder.

5.1.2 Choosing a Direct Connection to the Eyebot

The problem lies in how to utilise the functions available on the CPU to effectively create an interface. The solution to the problem requires choosing the most effective method of providing the input to the echo sounder and of receiving the output returned from the echo sounder.

The echo sounder has two pins, which connect to a microcontroller. One pin is the key input where the CPU sends the required pulse signal that will be modulated with the 200 kHz carrier frequency and transmitted through the transducer. The second pin that connects to the CPU is the logic output of the echo sounder. This output will be pulled low only when there is a 200 kHz signal received on the transducer line. This may occur several times for every pulse that is transmitted because the echoes from different surfaces will be just delayed versions of the signal. Therefore, the Eyebot interface is basically a pulse signal through the key input pin, followed by the delayed reception of a number of pulse echoes.

The digital output of the Eyebot is adequate to transmit the sonar pulse as the CPU can be told to switch on the digital IO, then wait for precisely the right amount of time and then finally switch the digital output off. This is seen in Figure 5.1.

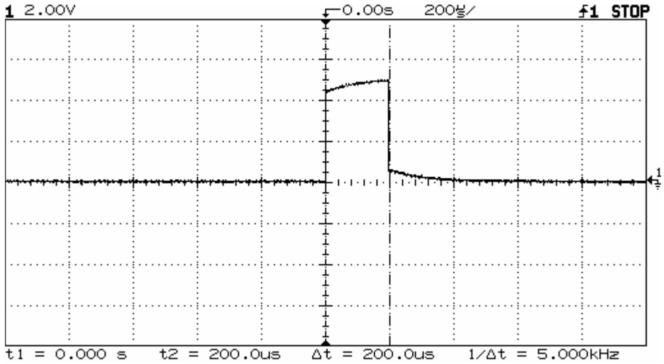


Figure 5.1: the 200µs pulse sent from the digital output of the Eyebot

The best way to accomplish the detection of a signal on the logic output of the echo sounder is to provide some way for the echo sounder to interrupt the CPU when the logic output goes low. This ensures that the exact times are recorded for processing. The interrupt system of the CPU allows an external device or one of the microcontroller's modules to interrupt the processor execution. The best approach to interrupting the CPU by the echo sounder is through programming one of the microcontroller's modules. Since the recording of the transmission and reception times requires a timer, using the Time Processor Unit (TPU) is the optimal way to interface the logic output of the echo sounder.

5.2 Sounding Flow Chart

To establish exactly how the echo sounder unit will function with the Eyebot, a complete echo sounding flow chart needs to be created and the functions required for the echo sounding process need to be identified.

5.2.1 Mode of Operation

The first step before creating the flow chart is to determine when the echo sounder will be on. The options for data collection are to have the echo sounder continuously collecting the data and returning it to the Eyebot, or to only collect the data when the Eyebot requires it.

The latter approach is chosen over the former for two main reasons:

- The sounding process is not time triggered but event triggered, removing the need to implement an interrupt for this process. If a problem arises with the use of CPU timed interrupts, it may cause timing errors in the rest of the timed events on the CPU interrupt vector table. It is wiser to implement the commencement of sounding as an event triggered process.
- The data is guaranteed to be current. If the echo sounder were to continually record data, there would be a period of time when the data does not refresh in the cycle. If the Eyebot were to read the data at the end of this time, then the data would no longer be current. The better approach is to commence sounding when data is required, ensuring the data is up to date.

5.2.2 Triggered Events and Subsequent Actions

The flow diagram has to be carefully designed to remove all time interrupts that may be necessary, such as the reading of the data and the timing out of the echo sounder when no pulse is returned. These may cause timing errors if used incorrectly.

The two timed events in the echo sounder's operations are:

- Reading of the data variables
- Time out of the TPU channels if there is no returned signal

These events require the use of timers to ensure the correct timing of these events. However, it is desirable to remove the use of interrupts for these events.

Instead of using a time interrupt, a wait is used to determine when to read the data, since the wait time will be very small. For a 5m range, the time of flight is 6.69ms in fresh water. The end of the wait time marks the time out of the echo sounder if the pulse is not received, meaning that an interrupt is not required for this task either.

If the channels time out, the CPU turns the channels off, raises the new data flag and moves the maximum value for distance into the distance variable of the C program. This indicates that no return pulse was registered for the echo sounding cycle and thus the distance is assumed beyond the range of the sonar.

The reading of data requires only the returning of the distance variable in the C program.

The reception of the echoes must rely on the TPU capabilities to accomplish the interrupt which means that new code must be created for these operations using assembly language.

Care must be taken to ensure that the interrupt requires little time to process to make sure there are minimal timing problems.

When the CPU is interrupted by the reception of the returned signal, the CPU immediately turns the channel off, and then checks the new data flag. If it is low, it raises the flag and then stores the data from the channels' registers. If it is high, then the interrupt does not do anything. This is because the channel has timed out.

The echo sounding process is not time triggered but event triggered for reasons stated earlier in section 5.2.1.

On the triggering of the transmission stage, the CPU must clear the new data flag and commence the operation of all TPU channels involved with echo sounding. Following this, one of the digital output pins of the Eyebot is switched on, and then the switched off after a 200µs wait.

The reason for switching the channel on before the pulse is transmitted is because the TPU channels have the capability to record the transition time of two events. By commencing the channels before the pulse, the TPU channels can record the time of the transmitted pulse as well as record the time of the received pulse. Thus the TPU channels carry the two times needed for distance calculations in its registers, meaning that the CPU does not need to obtain the transmit time by other means.

The CPU must then wait 10ms for the possible interrupt, triggered by the reception of the echo, to occur. As explained earlier in this section, if the interrupt does not occur, then the channel times out.

5.2.3 The Flow Chart

Figure 5.2 is a diagram for the operation of the echo sounder. The 10ms wait will be sufficient time for the received pulse to trigger an interrupt on the TPU channel or for the TPU channel to time out, as long as the reflector is within the 5-10m maximum range that is require of the sensor.

It is assumed that the echoes will attenuate much quicker than the rate at which data is required from the Eyebot. Though this will be the case for most instances, it may be possible to create an error by calling a new echo sounding process before the reverberations, or secondary echoes, have time to dissipate, especially when more power is used. This must be taken into account when programming the control system that uses the echo sounder.

By observing the flow chart, it can be seen that four C functions need to be written for the cycle of the echo sounder; one for the transmission of the acoustic pulse, one for the reception interrupt, one for the timing out of a TPU channel and one for the reading of the data. This requires setting the low level parameters of the TPU.

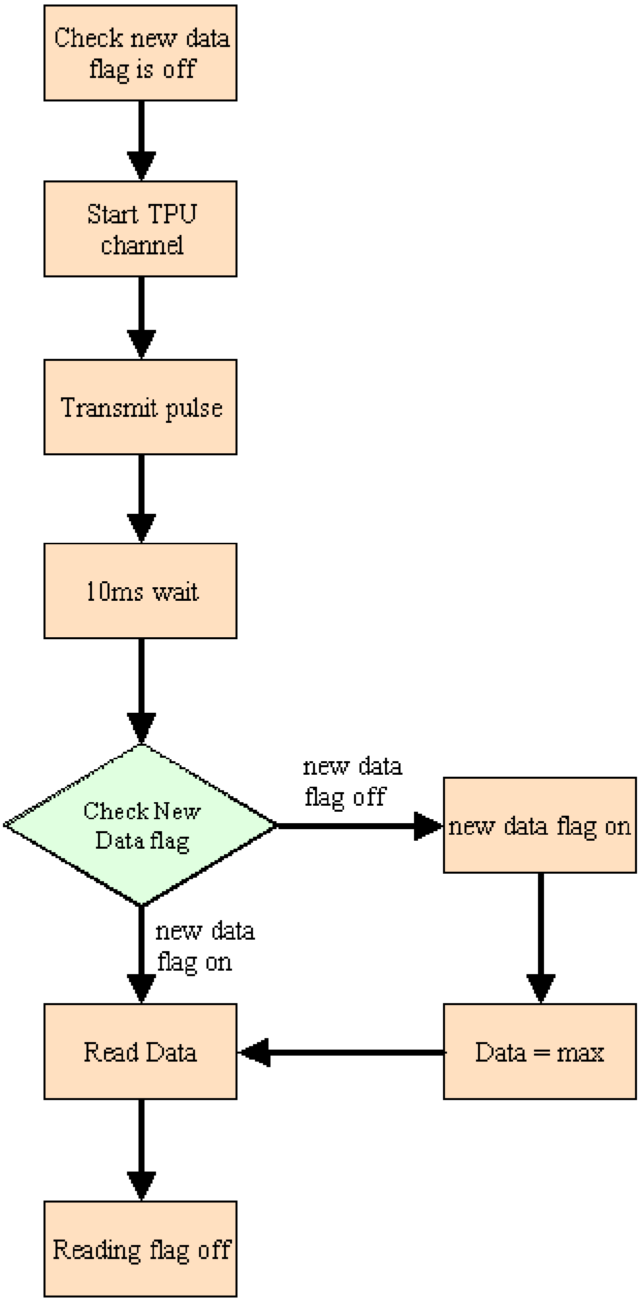


Figure 5.2: The flow chart for the echo sounding process

5.3 The Time Processor Unit

The TPU module is a very powerful and very flexible module of the Motorola microcontroller. The main advantage of the TPU module is that its timing is completely independent of the CPU instruction timing once it has been initialised [22].

The TPU module solves the problem of multiple interrupts to the CPU because the TPU can be used to count the number of pulses and only interrupt the CPU when the correct number of pulses has been registered. The TPU channel also solves the problem of interfacing multiple echo sounders to the Eyebot. Since the entire signal processing is accomplished with the CPU, the echo sounder need only send the return pulses in the form of logic outputs. These can be directly connected to the TPU channels and, since there are up to 16 available channels, there is no difficulty interfacing a small number of echo sounders to the same CPU.

5.3.1 The Input Transition Counter

With the TPU channels, there are specific functions that will allow for the detection of pulses. The best option for the reception of the echo sounder’s logic output is the Input Transition Counter (ITC) function of the TPU channels. The ITC mode allows a specific number of input transitions to be registered.

5.3.2 ITC Parameter RAM

For the correct operation of the ITC function, the parameter RAM of each channel must be set. The channels’ parameter RAM stores all the information about the operation of the channel at any one time and also contains the data registers that can store data from a particular operation, such as times or number of transitions.

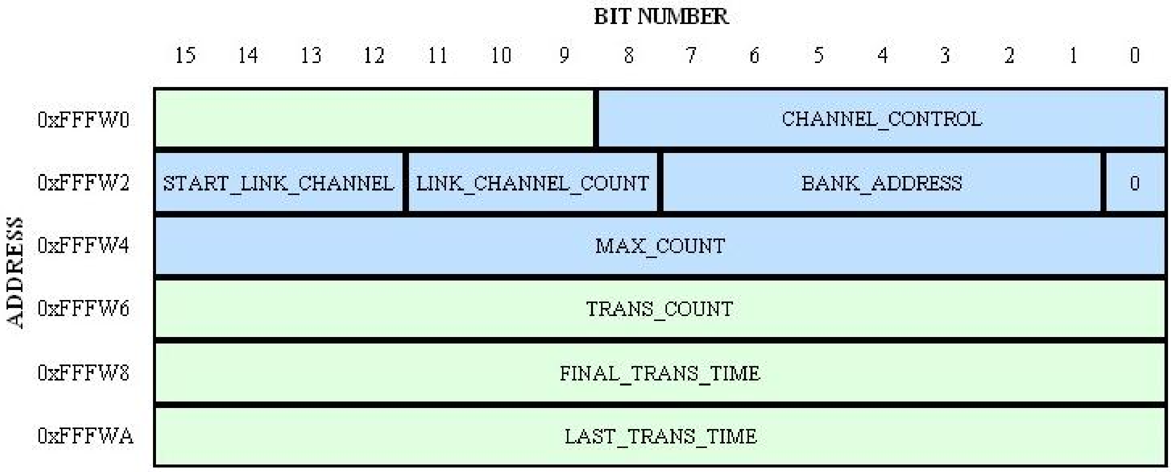


Figure 5.3: The ITC parameter RAM [22]

In Figure 5.3, the different parameters for the ITC function can be seen.

- The CHANNEL CONTROL identifies the type of event to trigger the channel and selects between timer one and two for use with the channel. The event could be a falling edge, a rising edge or both. Because the logic output of the echo sounder will change from a high state to a low state on reception of a transducer pulse, the trigger event is set to a falling edge. The timer reference is timer one.
- MAX COUNT is set as the number of transitions that need to be registered before the TPU performs an interrupt. As stated before, this means that the CPU will only be interrupted once with data from the echo sounder. Since the TPU will see both the transmitted and received pulse as a falling edge, this parameter is set to two to account for both pulses.
- FINAL TRANSITION TIME is the time of the event that actually triggers the interrupt for the TPU channel. This register will store the time when final pulse arrives, which is the received pulse.
- LAST TRANSITION TIME is the time of the event before the interrupt event. This register will store the time of the penultimate pulse, which is the transmitted pulse.

By taking the difference between the two counter values, FINAL TRANSITION TIME and LAST TRASNITION TIME, the time of flight can be calculated and the distance measured.

5.3.3 TPU Initialisation

To initialise the TPU and its channels, a number of other registers need to be set in assembly language. A full listing of the parameters that need to be set, as well as how they should be set, is given in Appendix A.

This initialisation must be run in order for the TPU and its channels to function correctly. This includes setting the parameter RAM of the TPU channels.

5.3.4 Range and Resolution

The range and resolution of the echo sounder are important parameters that can be determined by the TPU. There is a trade-off between these parameters because the counters on the TPU are 16 bit, which increment on every TPU clock pulse. This means that when the counter reaches a value of 65535, it resets to 0 and then continues counting. What this means is that there is a maximum range of 65535 counts before there is ambiguity in the counter value. For example, a count of 1 would be the same as 65537 according to the TPU. The clock pulses to the counters can be set to different frequencies meaning that the effective range can be increased, but this comes at the expense of the resolution. By having a lower frequency, the range would increase because each clock pulse is longer, but the resolution is worse as it can only resolve times in lengths of the clock pulse.

Herein lies the problem, which is to achieve the best possible resolution whilst still achieving a good range. A good range for the echo sounder in a pool environment is 5m. The AUV only needs to know what is immediately in its proximity. To convert the distance range into a time range, the following equation is used, as well as the speed of sound in water of 1481m/s [23].

$$\begin{aligned}\text{time range} &= \frac{2\times \text{distance range}}{\text{speed of sound}} \\ &= \frac{2\times 5}{1481} \\ \text{time range} &= 6.75ms\end{aligned} \tag{5.1}$$

Thus, the effective range of the echo sounder needs to be approximately 6.75ms. Comparing these amongst the values in Appendix B, it is observed that the resolution possible is actually the best possible resolution, which is 238.4ns. This equates to a theoretical distance resolution of approximately 17.8μm, which is more than sufficient for the purposes of echo sounding. Thus, for the project, a time resolution of 238.4ns and a range of 15.6ms are chosen.

CHAPTER 6

Hardware Verification and Experimental Results

The echo sounder needs to be properly tested and verified before it is implemented as a sensor system for the AUV. The interface must work, so that the echo sounder can communicate information to the Eyebot. The characteristics of the echo sounder need to be tested to uncover any flaws in the system and to ensure an accurate model of the sensor system.

6.1 Testing the Interface

It is vital to individually test both the interface to the Eyebot and the interface to the echo sounder so as to avoid any mismatch in voltage between the two components. A voltage mismatch could potentially cause serious circuitry damage. In addition, diagnosing errors in either component is more difficult once the two units are connected.

6.1.1 Testing the Eyebot Interface

Before connecting the echo sounder, the TPU channels on the Eyebot are tested to ensure that they are functioning correctly. To proceed with the testing, a simple experiment is set up. A connection from a digital output pin to one of the TPU channels is made using a 1kΩ resistor. Figure 6.1 is a picture of how the experiment was set up. The resistor is to ensure that if there is a voltage mismatch between the pins then only a small amount of current will flow between them, preventing damage to the circuitry.

The experiment simulates a connection between the echo sounder and the Eybot digital output and TPU channels. This experiment tests to see if the Eyebot digital output is correctly sending to the echo sounder and if the TPU can correctly recognise the two falling edges that the echo sounder returns.

The test code for the experiment consists of starting the TPU channels and transmitting two 200µs pulses delayed by a wait of approximately 10ms. This is then followed by a wait of one second. The Eyebot LCD screen is used to display the output.

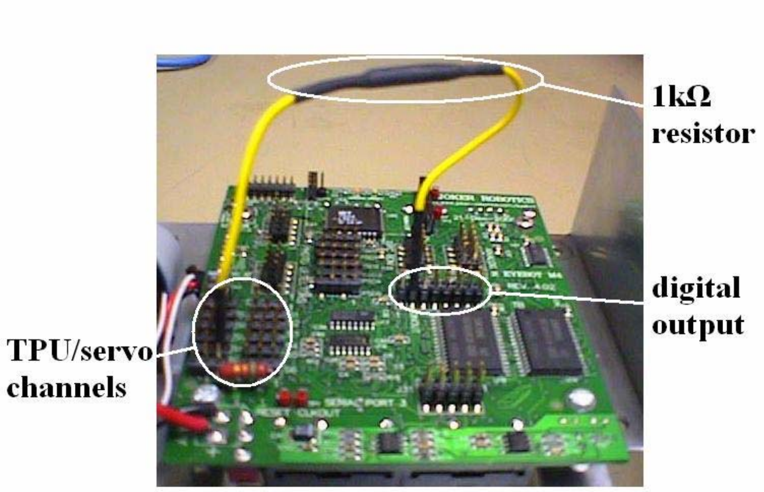


Figure 6.1: Experimental Setup for testing the Eyebot interface

Figure 6.2 is the signal that was passed from the digital output to the TPU channel, measured through an oscilloscope. Whilst the signal is not the same as the logic output of the echo sounder, it does provide two falling edges.

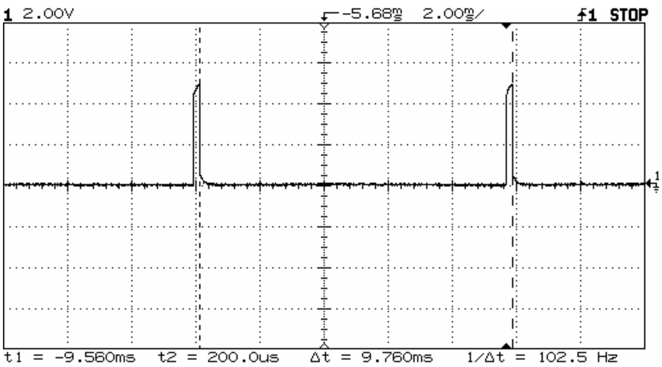


Figure 6.2: Two pulses separated by approximately 10ms

The time that is registered by the TPU is 40943 counts, which converts to:

$$\begin{aligned} \text{time} &= T_{c1} \times \text{number of counts} \\ &= 238.4\left(10^{-9}\right) \times 40943 \\ \text{time} &= 9.76 \times 10^{-3} \text{ sec} \end{aligned}$$

This result is the same as the time measured from the oscilloscope in Figure 6.2.

It is therefore verified that the TPU channels can be triggered by a falling signal edge and the digital output is functioning correctly. The echo sounder should therefore interface properly with the Eyebot. This also demonstrates the accuracy of the TPU channels at detecting the edges.

6.1.2 Testing the Echo Sounder Interface

The echo sounder interface also needs to be tested to ensure that the correct voltages and signals are received and transmitted, before connection to the Eyebot. The experimental set up involves connecting the echo sounder's key input to the digital output of the Eyebot, through a 1kΩ resistor. The logic output of the echo sounder is then measured using an oscilloscope to analyse the signal and the voltage output.

The experiment is to determine if the signal at the logic output is compatible with the TPU and to see if the digital output is compatible with the key input.

The test code is just the transmission of the 200μs pulse from the digital output to the key input of the echo sounder.

Figure 6.3 demonstrates the experimental set up.

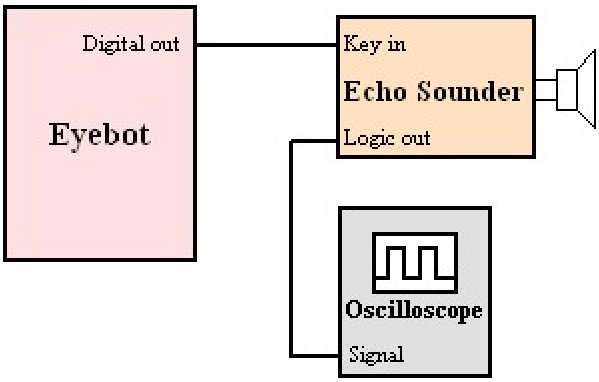


Figure 6.3: The experimental setup for the echo sounder interface

Figure 6.4 shows the signal when no pulse is sent from the Eyebot.

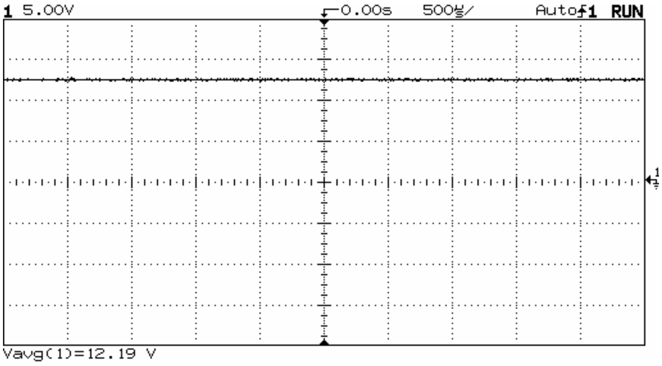


Figure 6.4: No signal through the echo sounder

What is learnt from Figure 6.4 is that there is a 12.19V signal that is transmitted through the logic output of the echo sounder. The reason for this is that the signal does not have a reference. This is unacceptable because the TPU channels can only accept a voltage of 5V at its pin. To solve this problem, a pull down resistor needs to be used. This is to be placed between the logic output and the ground of the echo sounder. This properly grounds the signal and converts it to one that is compatible with the TPU channel.

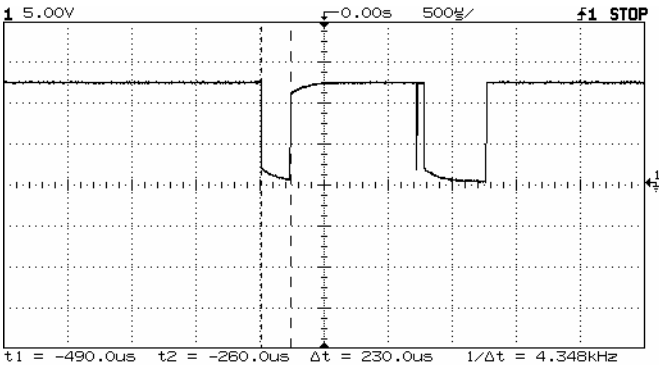


Figure 6.5: 200µs pulse sent via the digital output, and the transducer is placed in water

Figure 6.5 demonstrates the presence of echoes that are returned to the echo sounder, as conjectured during the design stage of the echo sounder. These need to be handled by the TPU channel. The setting of the MAX_COUNT parameter for the TPU as stated in section 5.3.2 should result in only one echo registering with the TPU. The proper reception of pulses indicates that the digital output is at a sufficient voltage to key the echo sounder circuit.

The echo sounder has been modified according to the tests performed on the echo sounder interface. In Figure 6.6, a 10kΩ trimpot resistor is used as a pull down resistor. Figure 6.7 shows that the voltage of the logic output of the echo sounder has changed to 4.63V due to the resistor. This makes it compatible with the TPU channel.

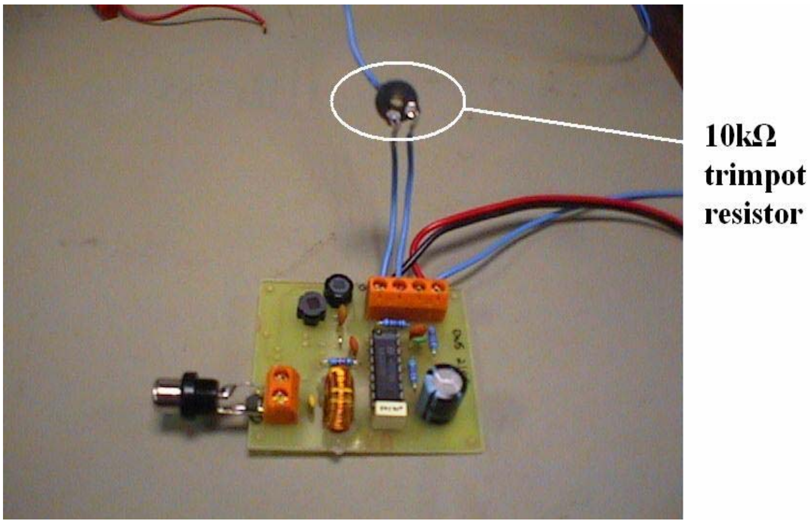


Figure 6.6: the pull down resistor placed between the logic output and the ground

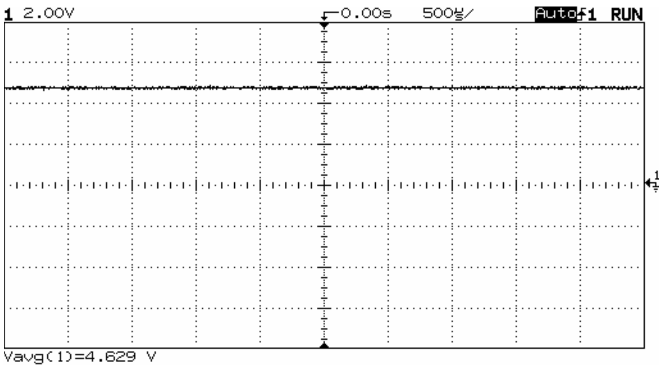


Figure 6.7: Signal when pull down resistor is used

6.1.3 Testing the Combined Interface

The experimental set up for the combined interface is exactly the same as the one used for the echo sounder interface, except that the logic output of the echo sounder is now connected to the TPU channels.

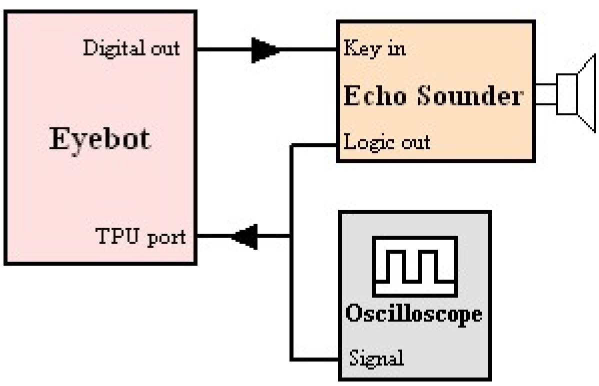


Figure 6.8: The experimental setup for the combined interface

An experiment is conducted to verify that the echo sounder and the Eyebot are correctly interfaced together.

The test code implemented for this experiment starts the TPU channel and then transmits a 200µs pulse. A key deactivated wait then precedes a display of the time of flight for the signal on the Eyebot LCD screen.

To test for the functionality of the echo sounder interface, two experiments test for the two cases which the echo sounder should detect.

The first case is when the echo sounder does not receive any returned pulse. This is simulated by removing the transducer from the water.

The test verified, as in figure 6.9, that the TPU channel does not receive a second falling edge meaning that the CPU is not interrupted. A time out needs to be called for this situation and a maximum time difference recorded by the Eyebot to signify that a real value cannot be calculated because the surface is either out of range or the transducer is not in the water.

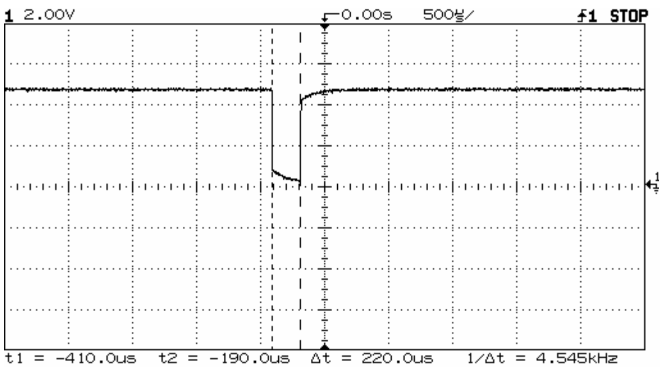


Figure 6.9: Signal at the TPU channel when the transducer is not in the water

The second experiment is to represent a successful transmission of the signal. This is accomplished by placing the echo sounder in a small container of water. This will result in significant reverberation, or echoing, which tests to see if the echo sounder can account for this and only detect the first echo (the first falling edge on the logic output from the echo sounder).

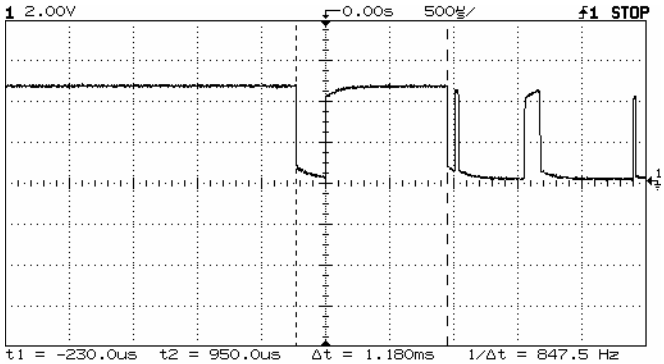


Figure 6.10: Signal that is received at the TPU channel when the transducer is in the water

Figure 6.10 shows the signal transmitted to the Eyebot from the logic output of the echo sounder. Notice the multiple secondary echoes that the TPU channel must handle. The measured time difference from the oscilloscope is compared with the actual time difference recorded by the Eyebot and displayed on the LCD screen. The time difference as measured using the oscilloscope is 1.180ms. The time difference recorded by the Eyebot is 4945 counts. Now each count represents a time of 238.4ns since the smallest resolution is being used (refer to Appendix B), so the actual time is:

$$\begin{aligned} \text{time} &= T_{c1} \times \text{number of counts} \\ &= 238.4\left(10^{-9}\right) \times 4945 \\ \text{time} &= 1.179 \times 10^{-3}\ \text{sec} \end{aligned}$$

This is approximately the same value as the value attained by measuring the time on the oscilloscope.

The interface can be seen to be functioning according to the specification required. The signals have been properly matched to suit the voltage requirements of the TPU channels. The transmitted and received pulses are correctly identified by the TPU channels and the time of flight can be properly calculated. Thus the interface has been verified to function correctly.

6.2 Determining the Characteristics of Echo Sounding

Once the interface between the Eyebot and the echo sounder is established, it is necessary to determine the exact physical characteristics of the echo sounder sensor. For these experiments, the timing out of the channel is not necessary and thus has not been included in the code set for testing.

6.2.1 The Linearity of Returns

To verify that the C functions programmed for the echo sounder allowed the Eyebot to determine the time of flight of the signal, the echo sounder was tested in a long channel of water in the Hydraulics Laboratory within the university.

The interface was connected, as outlined in Section 6.1.3. To ensure that that there is a constant voltage supply to both the Eyebot and the echo sounder, both power supplies are voltage regulated power supplies.

The test script for the Eyebot to test the echo sounder is as follows:

- Initialise the required TPU channel

Then cycle through a loop consisting of the following:

- Start the TPU channel
- Transmit a 200μs pulse from the Eyebot
- Wait until a key is pressed to ensure that an interrupt has occurred
- Display the time of flight on the LCD screen

The loop is to obtain different values for the same location. The results are collected over a distance from 0.18m to 3.5m and then the values for each distance point is then averaged to give an average time of flight for the distance point.

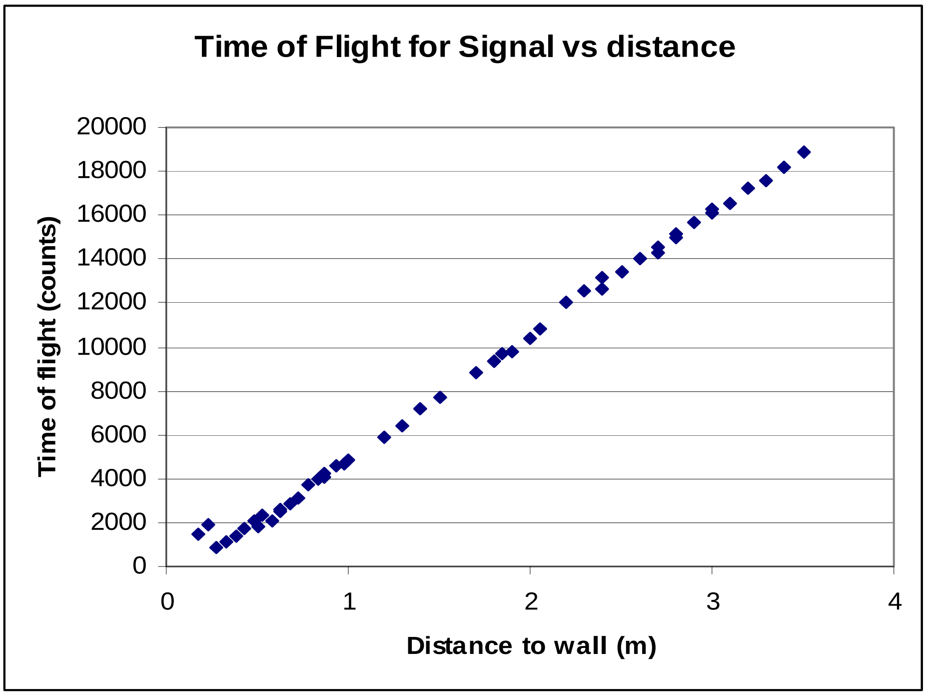


Figure 6.11: Calculated time of flight over a range of distances for the echo sounder

Figure 6.11 is a plot of the experiment, which demonstrates a linear relationship between the time of flight and the distance to the surface. A statistical analysis of the data shows that gradient of the function is 5.653x10^3 counts/metre.

6.2.2 Calibrating the Echo Sounder

For different types of water and for different temperatures, the speed of sound will vary. It is vital for the accuracy of the echo sounder that the system be correctly calibrated for the type of environment in which it will be used.

Given a set of data, like the set in Figure 6.11, it is possible to obtain the speed of sound in the test medium. Now each data point represents a particular time of flight for a given distance. Assuming that all the data is linear, then the change in time per unit distance between any two points equates to the instantaneous rate of change of time with respect to distance.

$$\frac{dt}{ds} = \frac{\Delta t}{\Delta s} \tag{6.1}$$

When this gradient is inverted, it becomes a rate of change of distance with respect to time. This is the speed of sound in the medium.

$$c_m = \frac{ds}{dt} \tag{6.2}$$

So by finding the gradient for the line of regression of the data points, it is possible to determine the speed of sound in the medium. Not only is this result useful for calibrating the echo sounder, but it also has applications where the speed characteristic of a particular environment is required.

To test this theory, the data set in Figure 6.11 is used. The line of regression for the data has a gradient of:

$$\frac{dt_c}{ds_w} = 5.653 \times 10^3 \, counts / metre \tag{6.3}$$

where t_c is the time in counts and s_w is the distance to the surface.

This value must be halved to account for the fact that the sound wave must travel the distance twice, once away from the echo sounder and once returning to the echo sounder.

$$\frac{dt_c}{ds_r} = 2.826 \times 10^3 \, counts / metre$$

where s_r represents the round trip distance.

Converting the counts to time in seconds using the set resolution of 238.4ns results in the following:

$$\frac{dt}{ds_r} = 2.826\left(10^3\right) \times 238.4\left(10^{-9}\right)$$
$$\frac{dt}{ds_r} = 6.738 \times 10^{-4} \sec/ metre$$

Inverting this result will give the speed of sound in the medium.

$$\begin{aligned} c_m &= \frac{ds_r}{dt} \\ &= \frac{1}{6.738 \times 10^{-4}} \\ c_m &= 1.484 \times 10^3 \, m/s \end{aligned} \tag{6.4}$$

This result is closely approximates the speed of sound in fresh water, which is 1481m/s.

6.2.3 Mean Square Error for the Echo Sounder

Whilst the data appears to be linear, there is some error, even in the averaged data for the plot.

The root mean square (RMS) error for the data from the line of best fit is calculated to be 167.2 counts. The RMS error is calculated by subtracting the predicted value from the observed value to result in an error, which is then squared. The average of the squared values is taken to form the mean square value. The RMS value is just the square root of the mean square value.

The 167.2 counts RMS will result in an error of margin of:

$$\begin{aligned} \text{distance error} &= \frac{\text{count error} \times \text{resolution}}{2} \times \text{speed of sound} \\ &= \frac{167.2 \times 238.4\left(10^{-9}\right)}{2} \times 1484 \\ \text{distance error} &= 2.96 \times 10^{-2} \, m \end{aligned} \tag{6.5}$$

Assuming that the speed of sound in the water is 1484m/s as in equation 6.5.

The error is, in part, due to the difficulty in precisely measuring the distance from the transducer to the surface of the wall.

This error is significant but not large in comparison to the size of the AUV itself and the pool environment for which it will be competing in. As the minimum requirement for the resolution of the echo sounder is 5cm, as stated in chapter 3, this result means that the current echo sounder is sufficient for the resolution requirement.

This result will be important for the EKF equations that will be discussed in Chapter 7. The error covariance is required to complete the correction cycle of the EKF. If the error, or noise, is assumed to be additive white Gaussian noise (AWGN), then the covariance of this noise is the mean square value for the error. For this instance, the covariance is 8.76×10^{-4} m^2.

This means that the measurement noise for the sensor will have a normal distribution of:

$$p(w) \sim N\left(0, 8.76 \times 10^{-4}\right) \tag{6.6}$$

The result is used by the EKF to better model the sensor readings, allowing for more accurate estimations of state, as will be discussed in Chapter 7.

This error covariance will not be the same for all environments and will need to be calculated for each environment.

6.2.4 Minimum Detectable Static Distance

What can also be noted from Figure 6.11 is that there is a minimum detectable distance for the echo sounder. This range can be observed to be approximately 0.25m. The theoretical value of this distance can be calculated from the time of the ringing in the transducer, which is 220µs, as in Figure 6.9:

$$\begin{aligned} \text{min distance} &= \frac{\text{min time} \times \text{speed of sound}}{2} \\ &= \frac{220\left(10^{-6}\right) \times 1484}{2} \\ \text{min distance} &= 0.163m \end{aligned} \tag{6.7}$$

There is a discrepancy between the measured data and the real data. This is due to the signal having to traverse the length of the transducer cable twice, resulting in a greater time between the pulses, and consequently a slightly greater minimum distance. Thus the minimum detectable static distance in fresh water is 0.163m.

6.2.5 Maximum Detectable Static Distance

For active sonar, according to the formula given in Chapter 2, the detection threshold is the signal level at which the arriving signal can be detected 50% of the time.

The echo sounder is therefore tested to determine the distance that the device can no longer detect the arriving signal for 50% of the time. To test for the maximum range of the echo sounder, the same set up for determining the linearity of the sonar returns was employed.

Starting at a distance of 2m from the surface of the reflecting wall, 500 measurements are taken to determine whether the signal has been detected for over 50% of the time. Table 6.1 shows the results of the experiment.

Distance (m)	Sample 1	Sample 2	Sample 3	Sample 4	Sample 5	Detection Percentage
2	100	100	97	98	100	99
2.2	100	100	99	98	100	99.4
2.4	100	100	98	98	100	99.2
2.6	99	100	98	98	99	98.8
2.8	99	99	97	98	98	98.2
3	99	97	97	100	97	98
3.2	90	82	87	88	82	85.8
3.4	95	98	99	98	99	97.8
3.6	87	97	90	89	88	90.2
3.8	13	18	18	14	14	15.4
4	14	19	14	19	15	16.2
4.2	1	2	1	1	2	1.4
4.4	0	0	0	0	0	0

Table 6.1: Samples for sonar returns for distances from 2.4m to 3.5m

Table 6.1 shows that the detection of the return signal begins to significantly drop at 3.8m. At distances longer than 3.8m, the detection begins to drop to 0% making it extremely difficult for the AUV to detect obstacles.

Figure 6.12 shows the result of Table 6.1 in a graphical form.

The reason for the increasing lack of detection after 3.8m is that the signal becomes attenuated to a level that is undetectable by the echo sounder. Thus the maximum detectable static distance is 3.8m in fresh water.

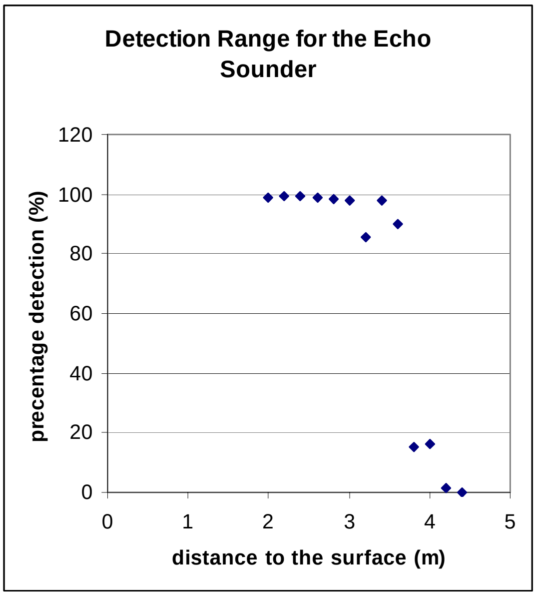


Figure 6.12: Detection of an echo for different distances

There are a number of factors that affect the attenuation of the signal in the water. These have been discussed thoroughly in Chapter 2. The key factors that result in highly attenuated signals at the receiver are:

- The low intensity of the transmitted signal
- The signal's susceptibility, at 200 kHz, to losses due to bulk viscosity and absorption due to relaxation
- The wall's reflectivity, which will result in only some of the signal being reflected by the wall whilst the rest is transmitted through the wall
- The natural attenuation of the signal in the medium due to absorption and scattering

This clearly is not the distance that the transducer is capable of achieving and does not meet the requirement of 5m, as stated in chapter 3. The Navman echo sounder circuit that the transducer was taken from can achieve a distance of up to 184m. The LM1812 ultrasonic transceiver chip is capable of depth sounding up to 30m. This clearly indicates that not enough power is being converted to acoustic energy at the transmitter stage.

6.2.6 Investigation of Signal Transmission Intensity

There were three possible reasons for the lack of signal transmission intensity. These were:

- The key input voltage to the LM1812 chip in not high enough
- The LC resonator has not been correctly tuned to 200 kHz

- There is a problem with stepping the voltage up to the required level for the transducer.

One concern was that the 5V pulse from the Eyebot did not have enough voltage to meet the 12V suggested in the LM1812 datasheet [14]. A simple test was conducted to determine if this may have resulted in a weaker signal transmitted by the echo sounder. The following circuit is connected between the digital output of the Eyebot and the echo sounder.

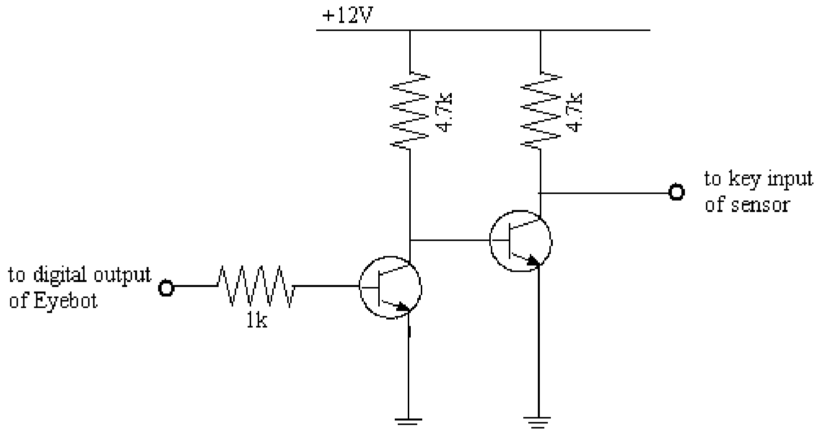


Figure 6.13: Driver circuit to increase the voltage at the input of echo sounder

Figure 6.14 is the display of the oscilloscope when no driver is used. Figure 6.15 is the display when a driver is used. The signals have been inverted so that they can be easily captured by the oscilloscope. If the amplitude was increased, the transducer should ring much longer, resulting in the logic output being low for longer. As can be seen in the figures, the increase in voltage has not resulted in a wider logic output pulse. This means the key input voltage can be discounted as a factor for the lack of signal intensity.

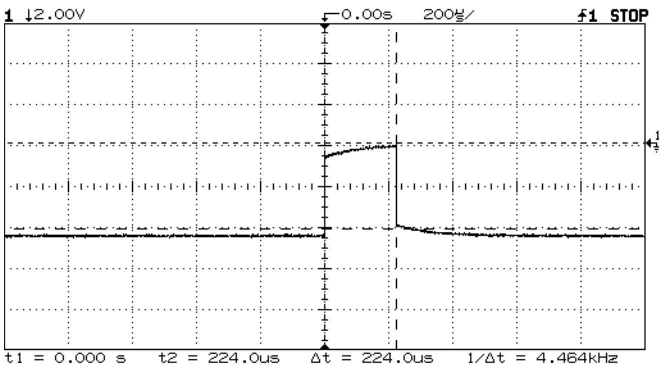


Figure 6.14: Oscilloscope display for when no driver is used

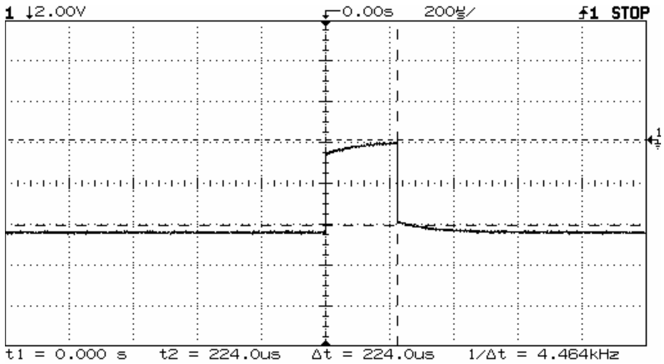


Figure 6.15: Oscilloscope display for when driver is used

Another possible problem could have been that the LC resonator was not correctly tuned. Since the resonant frequency of the transducer is 200 kHz, this frequency must be exactly attained by the LC resonator. To conduct a test for this, the oscilloscope was connected to pin 6 of the LM1812 chip, which is shown in Figure 6.16. Pin 6 is the output signal to the transducer, from the chip. A pulse was then keyed from the Eyebot.

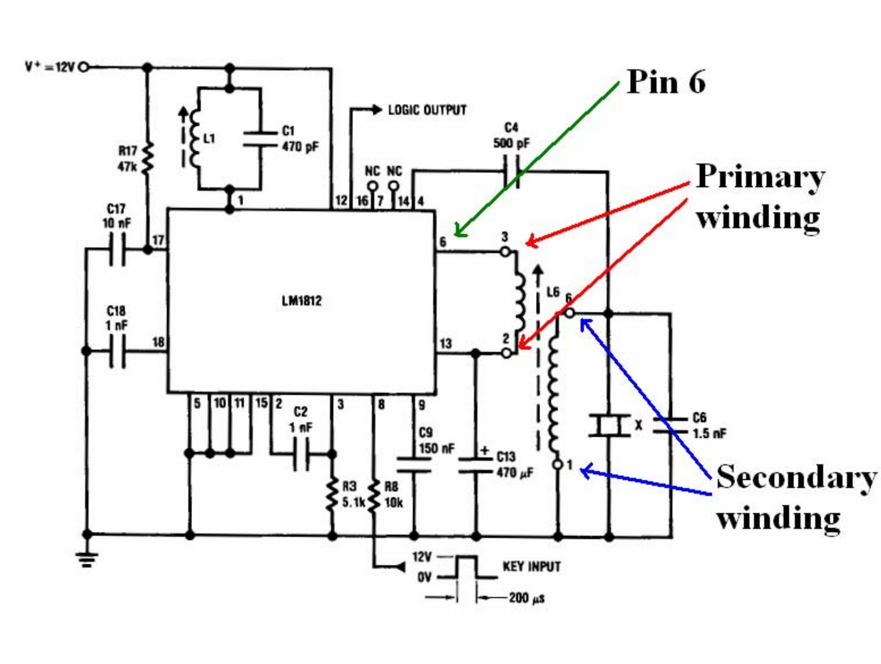


Figure 6.16: Location of pin 6 and transformer windings on the circuit diagram [14]

Figure 6.17 shows the result of the experiment. The measured frequency of the output from the LM1812 chip is exactly 200 kHz, which indicates that the LC resonator has been tuned correctly.

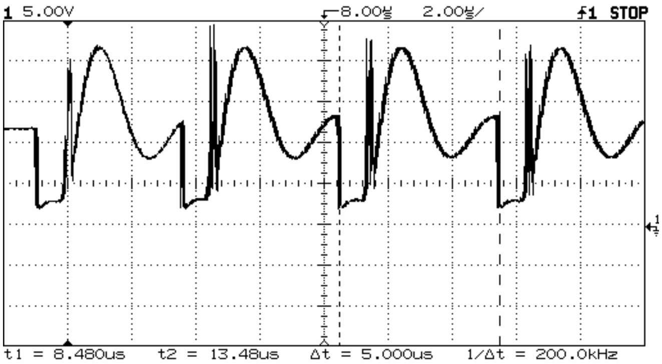


Figure 6.17: Frequency of LM1812 output

The final possible cause for the lack of signal intensity is that the voltage signal to the transformer, which drives the transducer, is not stepped up to the right amplitude. If there is a problem at the step-up transformer stage, then there will not be significant lack of signal intensity for transmission. The test was conducted using the following procedure.

- Connect the oscilloscope across the primary windings of the transformer, shown in Figure 6.16, and key a pulse from the Eyebot to determine the peak-peak voltage (Vp-p) of the input to the transformer. Repeat this, connecting the oscilloscope across the secondary windings of the transformer to determine the Vp-p of the output of the transformer.

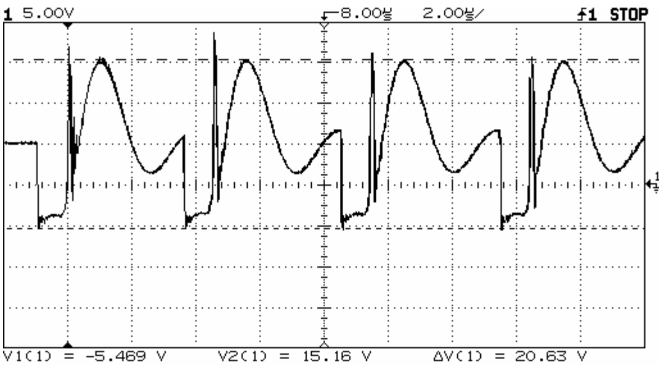


Figure 6.18: Peak-peak voltage of transformer input

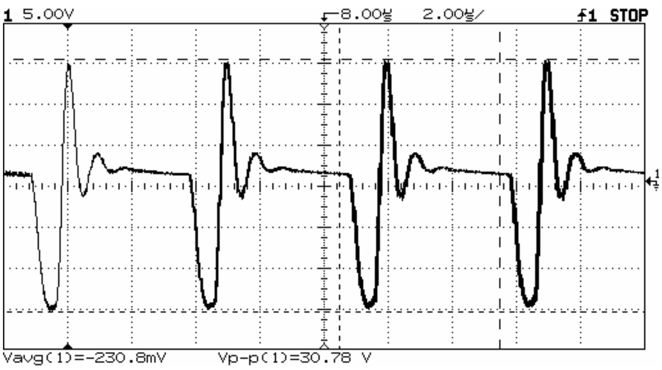


Figure 6.19: Peak-peak voltage of transformer output

Figures 6.18 and 6.19 show the observations of this experiment. As can be seen from the figures, there voltage has only been stepped up from 20.63Vp-p to 30.78Vp-p. However, the transformer has been designed with a turns ratio of 4.5:1. This means that the voltage should be stepped up by a factor of 4.5. This clearly is not the case.

The problem may be that the transformer does not properly match the source impedance to the load impedance of the transducer at 200 kHz. When these are matched, then there is maximum power transfer to the transducer. If the load and source impedance are given as:

$$R_L = \frac{V_2}{I_2} \qquad R_S = \frac{V_1}{I_1} \tag{6.8}$$

And the transformer equations are:

$$V_1 = \frac{N_1}{N_2} V_2 \qquad I_1 = \frac{N_2}{N_1} I_2 \tag{6.9}$$

where N_1 and N_2 are primary and secondary windings respectively, then the impedances can be matched by adjusting the windings to suit the following equation:

$$R_S = \left(\frac{N_1}{N_2}\right)^2 R_L \tag{6.10}$$

To solve this impedance mismatch problem, the transformer's windings need to be adjusted so that equation 6.10 is correctly met.

There are two other solutions possible that target the intensity of the signal, one implemented at the transmitter and one implemented at the receiver. The first is to develop a pulse amplifier that will increase the intensity of the signal at the transmitter. The second solution is to have a larger gain at the receiver to detect the smaller intensity signals. This can be achieved through a time variable attenuator. Both of these will be discussed in Chapter 8.

6.2.7 Time Redundancies for a Simple Fault Tolerant System

For the data seen in Table 6.1, it is clear that a fault tolerant system needs to be implemented to ensure that there is minimal probability of erroneous readings from the sensor.

A simple system is implemented for the echo sounder based on the mid-value select method for choosing data points. The mid-value select method selects the median value from three readings to attempt to eliminate the chance of returning an outlying data point. In this case, the three values will be consecutive readings from the echo sounder. This will utilise the time redundancies to improve the estimate for distance.

The problem with a mid value select method is that the data returned by the echo sounder will have outliers lower than the true value. A much better approach is to choose the maximum from a set of three data points. This, too, is flawed in that it still may be possible to achieve an outlier above the true value.

A new method for fault tolerance is proposed. This involves the use of a pre-filter to remove most of the lower outliers. The pre-filter is just the maximum value from a number, selected by the user, of consecutive readings from the echo sounder. This is performed three times to obtain three maximum values. The median value of these three values is then selected to ensure that any higher outliers are not selected.

Distance (m)	Samples					Detection Percentage
2	10232	10242	10241	10231	10228	100
	10243	10240	10245	10245	10228	
2.2	11447	11449	11448	11449	11465	100
	11449	11449	11449	11450	11449	
2.4	12744	12747	12801	12745	12746	100
	12745	12799	12746	12746	12746	
2.6	13807	13828	13824	13810	13822	100
	13809	13810	13822	13808	13809	
2.8	14968	14967	14967	14968	14968	100
	14966	14968	14968	14968	14966	
3	16104	16106	16116	16103	16103	100
	16102	16103	16102	16102	16102	
3.2	17251	17254	17250	17251	17253	100
	17256	17250	17249	17250	17251	
3.4	18352	18353	18356	18352	18350	100
	18352	18352	18354	18351	18349	
3.6	19457	19424	19424	19423	19423	100
	19423	19422	19437	19412	19423	
3.8	20579	20571	20581	20578	20580	100
	20580	20578	20579	20576	20575	
4	21606	21610	21608	21586	21620	60
	1724	1724	21607	1699	1991	
4.2	22791	22763	11442	22742	22756	90
	22767	6906	22742	11668	22743	
4.4	1661	1672	1671	1689	1676	0
	1656	1724	1664	1676	1674	

Table 6.2: Samples of sonar returns using extended mid-value select method

Table 6.2 shows the percentage detection over the range of distances measured for Table 6.1 using a pre-filter of size 3. This means that a maximum is chosen from three consecutive readings. It can be seen that, whilst the fault tolerance implemented does not increase the detectable range by much, it allows a virtually flawless detection rate over the entire detectable range.

This method of fault tolerance may take approximately nine times longer to process than just obtaining one reading, but it allows sonar returns to be more continuous, which is essential for navigational purposes. The pre-filter can be shortened if less time for processing is necessary. The form of fault tolerance implementation only has significant merit in the static cases, as is discussed in the next section.

6.3 Performance of the Sensor in a Dynamic Environment

The final part of this chapter deals with the performance of the echo sounder at tracking objects that move, or when the echo sounder is moving. Up until now, the tests performed on the echo sounder involved static experiments where the echo sounder was placed a certain distance from a wall and the time of flight measured.

This final experiment deals with a fixed transducer position in the water, with the transducer pointed into free space. A moving wooden rod is placed in from of the transducer and is moved back and forth in front of the transducer. The distance measurement is recorded and stored in an array, ready for uploading to a PC upon completion of the measurement stage. The points are then plotted on to a graph.

A new code set has been written for this experiment with the timing out of the channels, as is seen in the flow chart in Chapter 5, accounted for in assembly and C functions. Figures 6.20 and 6.21 show the graphs of the experiment. The data set is collected using the modified mid-value select method.

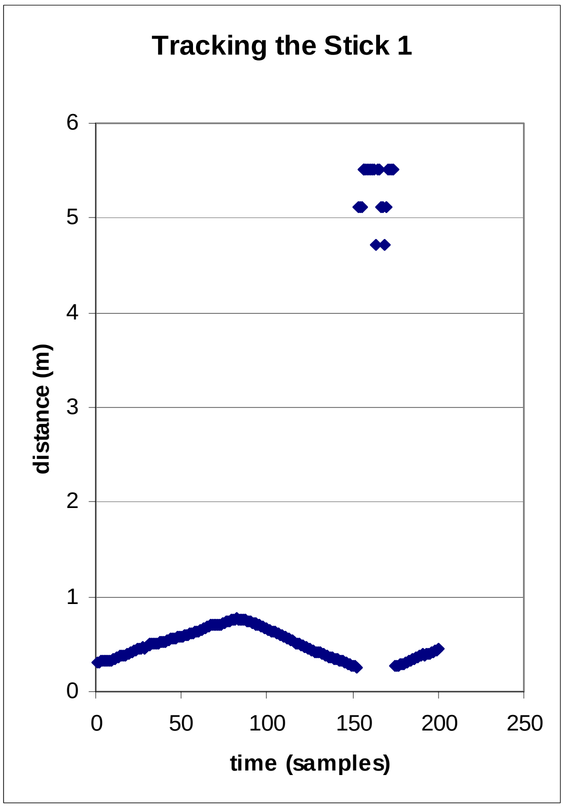


Figure 6.20: Distance wooden rod is from sensor over time

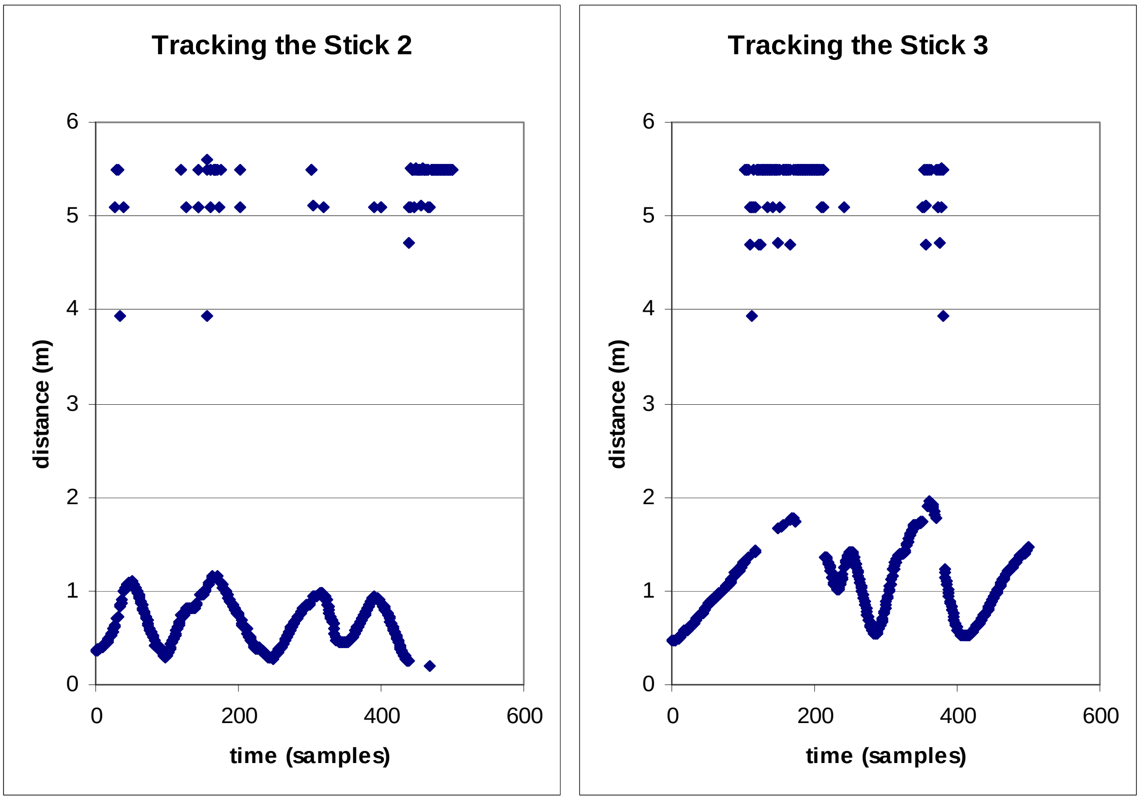


Figure 6.21 a) & b): Measuring distance from the rod over 500 time samples

What can be seen from these graphs is that even with the implementation of a simple fault tolerant system, it is not possible to achieve zero errors in measurement.

The reason for this is that the obstacle is moving creating turbulence in the water. This turbulence will distort the echo's frequency by adding random noise to the signal. As the circuit is tuned to 200 kHz, it will not be able to detect a significant distortion in frequency. Thus for the AUV to detect a wall or obstacle, it must be travelling at low speeds. This is ensured on the current AUV as the thrusters only allow low velocities. At the speeds the AUV will be travelling at, there should be minimal turbulence in the water. However, care must be taken to ensure that the sensors are not placed near areas of significant turbulence on the AUV, such as near the thrusters.

There is also another reason for the erroneous readings. The echo signal is subject to a phenomenon called the Doppler Effect. The Doppler Effect is when the observed frequency of the signal changes from the source frequency because the object is moving away or towards the source. The frequency of the observed signal becomes:

$$\frac{\Delta f}{f} = \frac{v_{source}}{v_{sound}} \tag{6.11}$$

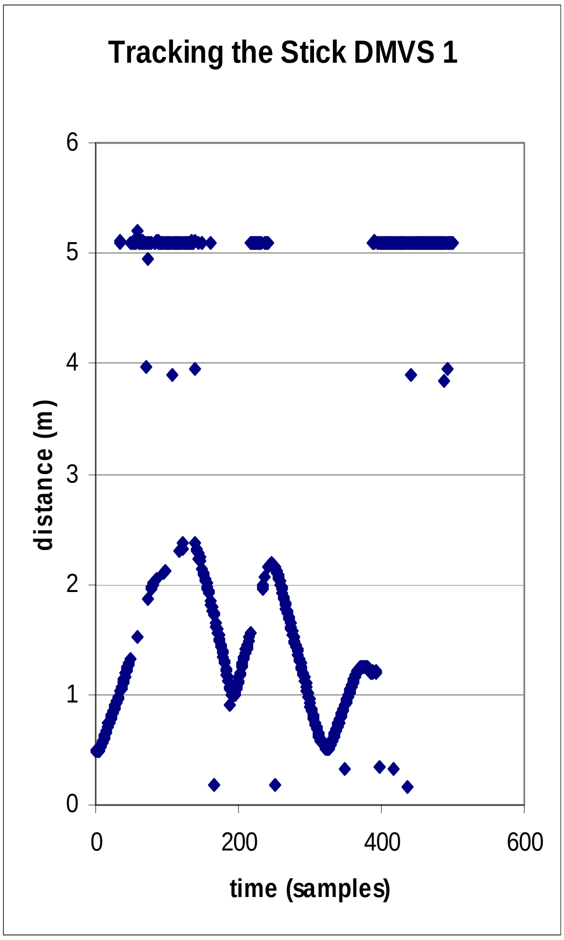
Since the Doppler Effect occurs both towards the rod and back to the sensor the formula can be written as:

$$\frac{\Delta f}{f} = \frac{2v_{rod}}{v_{sound}} \tag{6.12}$$

However, since the speed of the AUV will be very small with respect to the speed of sound, this effect is minimal in comparison to the effect that the turbulence has on the amount of erroneous readings.

At slow speeds, like in Figure 6.20, the sensor can track the rod well, with errors only when the rod is closer than the minimum detectable distance. This is compared with Figure 6.21 where the rod was moved much quicker. More errors are introduced as the sensor cannot detect the frequency properly.

An attempt is made to reduce the number of higher outliers by adjusting the modified mid-value select method. For this experiment, 10 samples are taken and the 7th sample in ascending order is taken as the estimated value. A) and b) of Figure 6.22 demonstrate the results of such an experiment.



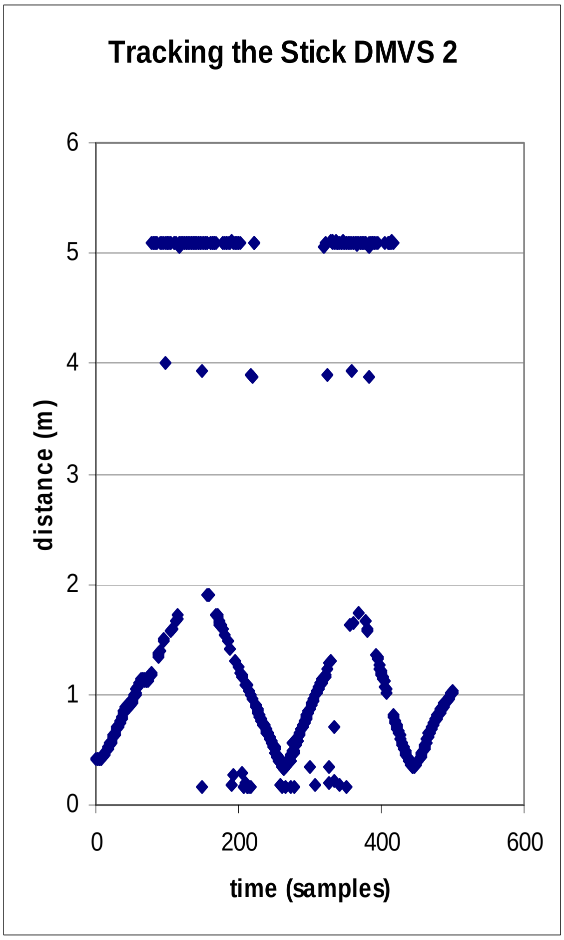


Figure 6.22: 7th value select method on the tracking of a wooden rod

As can be seen from the plots, outliers from above the true value are removed, but are then replaced by outliers from below the true value. What this means is that it is impossible to remove all the outliers from being detected using this method of fault tolerance. However, using the earlier modified fault tolerance method, the percentage error is 20%, as opposed to 40%, for the latter, to 20%.

The errors may be reduced partially by investigating the population of data points to determine where the data, which is within 10% of the true value, lies in relation to the ordered data set. For example, if the investigation found that this data lies in the 80th percentile, then 10 sample points can be taken and sorted in ascending order, and the 8th data point will be the estimate of the data. This is an area for future investigation.

Obviously, the more data points collected, the better the robustness of the system, but this will result in a slower data rate. 10 sample points will result in a data rate of approximately 10Hz, which is the minimum requirement for this project, and thus should not be exceeded.

6.4 Summary of Experimentation

Some important results have been obtained through experimentation with the echo sounder unit. These results are:

- Verification of the Eyebot interface to the echo sounder to be completely functional
- Verification of the linearity of the sonar returns as a function of distance
- Establishment of a method of calibrating the echo sounder to the speed characteristics of the medium
- Determination that the mean square error for static measurements of distance in fresh water is 2.96cm. This falls within the required resolution of 5cm, as stated in Chapter 3
- Determination that the minimum detectable distance in the test environment is 0.25m.
- Determination that the maximum detectable distance using the current circuit in fresh water is 3.8m. This does not meet the minimum 5m requirement. However, the recommendations made should allow the sensor to achieve this target
- Knowledge that the lack of detection range is due to the transformer mismatching the transducer and source impedance, resulting in a lack of transmitted signal intensity
- Implementation of a simple fault tolerant system using time redundancies to improve the robustness of the sensor for static distance measurements
- Knowledge of sensor degradation when dynamically measuring distances
- Knowledge that errors in dynamic distance measurements cannot be completely removed using time redundant fault tolerance but can be decreased by modifying the mid-value select method
- Knowledge that at low speeds, turbulence and the Doppler Effect are minimised and thus the sensor is capable of tracking obstacles effectively

CHAPTER 7

Navigation Using the Sonar Sensors

Navigation is an important part of an AUV's autonomy. In order to be completely autonomous, the AUV must successfully navigate a path to a target or goal, using only its sensors. Since sound is the best form of sensing in water over distances, it is the medium of choice for many applications. Since an echo sounder has been implemented, it is possible to perform such tasks as wall following and obstacle avoidance. This section will discuss the development of extended Kalman filter equations and control equations that can be used with the AUV in conjunction with the sonar sensors.

7.1 Active Sonar Application

For an AUV, the wall following problem is a key aspect of its navigation. In an unknown environment, when a new wall is found, a search algorithm is usually implemented so that the autonomous vehicle can collect information about the nature of the wall, such as orientation, position and length. The only method of detecting a wall is by active sonar.

The wall following problem first needs to be formulated before a set of control equations can be derived for the AUV. These equations can then be used to create the extended Kalman filter equations.

7.1.1 The Wall Following Problem

The AUV is required to follow the wall of a pool as it navigates a full circle about the pool. This means that the AUV needs to remain a constant distance from the pool's edge whilst maintaining a constant velocity. The AUV has a compass, a velocity sensor and an echo sounder with which to complete the task.

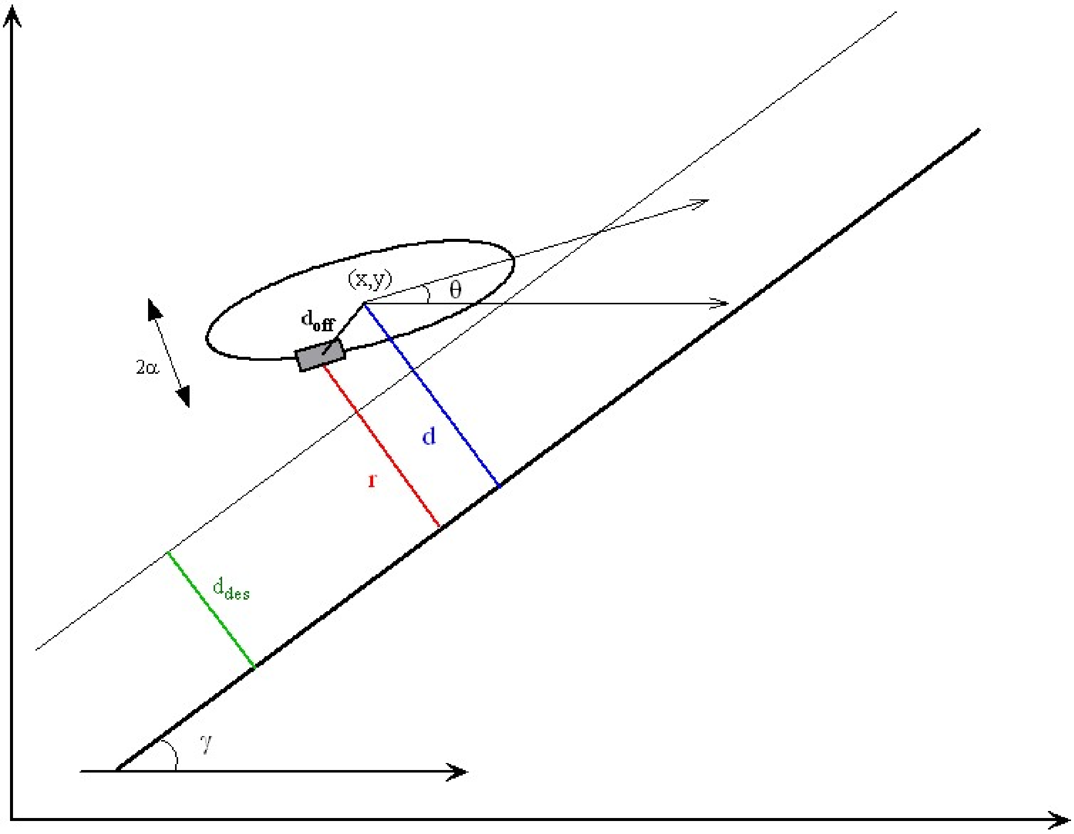


Figure 7.1: The wall following problem

Using the problem formulation from Bemporad et al [24] and Figure 7.1, a set of equations for the wall following task is defined.

From the information from these sensors, the position state of the AUV can be determined using the following equations:

$$\begin{aligned} \dot{x} &= v\cos\theta \\ \dot{y} &= v\sin\theta \\ \dot{\theta} &= \omega \end{aligned} \tag{7.1}$$

where v is the speed of the AUV whilst ω is the angular velocity of the AUV.

What is required is the design of the feedback controller that will allow the AUV to move at a constant velocity at a constant distance from the wall of the pool.

Assuming that the wall is considered straight and infinite, the wall can be determined by a line passing through a point.

$$\frac{dy}{dx} = \tan\gamma$$
$$\frac{\Delta y}{\Delta x} = \frac{dy}{dx}$$
$$= \tan\gamma$$
$$\frac{\Delta y}{\Delta x} = \frac{\sin\gamma}{\cos\gamma}$$
$$(y - y_w)\cos\gamma = (x - x_w)\sin\gamma$$

where (x_w, y_w) is a point on the wall. The equation for the wall can be written as:

$$(y - y_w)\cos\gamma - (x - x_w)\sin\gamma = 0 \tag{7.2}$$

The zero represents the distance from the wall, thus all points satisfying this equation is on the wall.

The equation for the distance to the wall can thus be written as:

$$d = (y - y_w)\cos\gamma - (x - x_w)\sin\gamma \tag{7.3}$$

The distance that the echo sounder's transducer will be from the wall is calculated as:

$$r = (y + d_{offy} - y_w)\cos\gamma - (x + d_{offx} - x_w)\sin\gamma \tag{7.4}$$

where (d_{offx}, d_{offy}) is the position of the transducer with reference to the centre of the AUV.

7.1.2 Constraints of the System

Now, the AUV is not capable of high speeds, which means that there is a speed constraint on the controller that will be used. The benefit of this is that low velocities enable a control system that is more robust. This places the constraint on the motors of the AUV:

$$|v \pm \alpha\omega| \leq \Omega_{\max} \tag{7.5}$$

where 2α is the distance between the forward thrusters, Ω_{max} is the maximum speed attained by each of the thrusters.

There is also a constraint on the orientation of the AUV from the surface of the wall. Because the echo sounder has a finite beamwidth in which it can detect an echo return, in theory, AUV cannot tilt more than half the beamwidth of the echo sounder, as can be seen in Figure 7.2. This leads to the constraint that:

$$|\theta - \gamma| \leq \phi \tag{7.6}$$

where φ is the half beamwidth of the echo sounder.

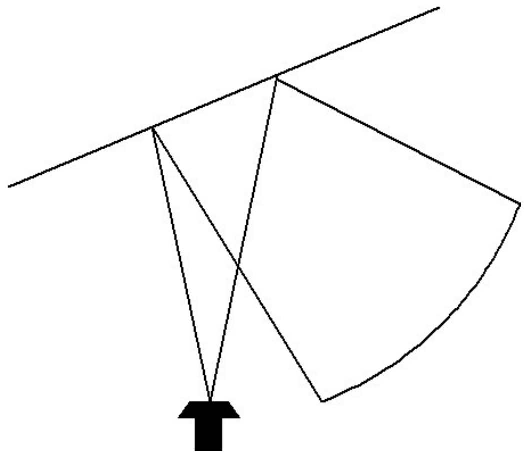


Figure 7.2: No returned echo at an angle greater than half the beamwidth

7.2 The Extended Kalman Filter

The sensors on the AUV are always prone to measurement error. This means that in practice, the actual coordinates of the AUV will never be accurately known. The only way to localise the AUV is to use the noise corrupted sensors to provide a distorted view of the environment.

The magnetic compass on the AUV is the most classical absolute orientation sensor. However the precision that can be obtained is usually low and it can be strongly affected by external disturbances. When an extra magnetic field, for example motors and electrical equipment, is present, the precision of the compass can be very bad. This is the case with the AUV, where the hard disk for the onboard computer is in close proximity to the compass.

The velocity meter is, at best, tolerable in determining the speed of the AUV. It is based on a paddle wheel with two magnets on the circumference to trigger a pulse. The velocity is calculated by counting the number of pulses per unit time. The resolution for this device is poor at low velocities as the resolution is large with respect to lower velocities due to quantisation.

Because information from sensors is incomplete, erroneous and uncertain, it is essential that the system fuse redundant information from multiple sensors. This is the acquisition of data that, in part, measures the same quantities as the other data, and combines these measurements to determine a more accurate view of the environment.

The extended Kalman Filter (EKF) is a method of accomplishing the fusion of data from different sensors. The EKF is used to obtain an estimate of the position state from an estimate of the distance from the wall. The sensor will use the error in distance reading to correct the predicted position state.

7.2.1 The Control Equations

Firstly, a set of control equations is required to give an indication of the system. The position state is the variable to be estimated by the EKF. For the position state:

$$X(k) = \begin{bmatrix} x(k) \\ y(k) \\ \theta(k) \end{bmatrix} \tag{7.7}$$

The input of the system is:

$$U(k) = \begin{bmatrix} v(k)\cos\theta \\ v(k)\sin\theta \\ \omega(k) \end{bmatrix} \tag{7.8}$$

The control equation is therefore given as:

$$X(k) = \begin{bmatrix} x(k-1) + v(k)T_c\cos\theta \\ y(k-1) + v(k)T_c\sin\theta \\ \theta(k-1) + \omega(k)T_c \end{bmatrix} + E_X(k) \tag{7.9}$$

$$z(k) = [y(k)\cos\gamma - x(k)\sin\gamma] + (d_{offy} - y_w)\cos\gamma + (d_{offx} - x_w)\sin\gamma + \xi(k) \tag{7.10}$$

It is clear why the EKF needs to be applied here. The control equations for this system are non-linear and thus cannot be filtered using the standard Kalman filter.

As defined in Chapter 2, the Jacobian matrices of partial derivatives A and H need to be calculated in order to proceed with the filtering process. These matrices are calculated as follows:

$$A = \begin{bmatrix} 1 & 0 & -v(k)T_c\sin\theta \\ 0 & 1 & v(k)T_c\cos\theta \\ 0 & 0 & 1 \end{bmatrix} \tag{7.11}$$

$$H = \begin{bmatrix} -\sin\gamma & \cos\gamma & 0 \end{bmatrix} \tag{7.12}$$

$$W = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix} \tag{7.13}$$

$$V = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix} \tag{7.14}$$

where W and V are identity matrices because it is assumed the noise is white and additive.

7.2.2 Filter Predict and Correct Cycle

The filtering process then proceeds with the time update equations, which predict the position state variable and the error covariance matrix, using the compass and the velocity meter. These are based on the equations in Section 2.6.

$$\hat{X}(k)^- = \begin{bmatrix} \hat{x}(k-1) + v(k)T_c\cos\theta \\ \hat{y}(k-1) + v(k)T_c\sin\theta \\ \hat{\theta}(k-1) + \omega(k)T_c \end{bmatrix} \tag{7.15}$$

$$P(k)^- = \begin{bmatrix} 1 & 0 & -v(k)T_c\sin\theta \\ 0 & 1 & v(k)T_c\cos\theta \\ 0 & 0 & 1 \end{bmatrix} P(k-1) \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ -v(k)T_c\sin\theta & v(k)T_c\cos\theta & 1 \end{bmatrix} + \begin{bmatrix} \sigma_x^2 & 0 & 0 \\ 0 & \sigma_y^2 & 0 \\ 0 & 0 & \sigma_\theta^2 \end{bmatrix} \tag{7.16}$$

Once these have been calculated, the two variables can then be corrected using the returns of the echo sounder. This must be preceded by the adjustment of the Kalman gain. In equation 7.17, the error covariance, R(k), is the mean square error calculated in Chapter 6.

$$K(k) = P(k)^- \begin{bmatrix} -\sin\gamma \\ \cos\gamma \\ 0 \end{bmatrix} \left(\begin{bmatrix} -\sin\gamma & \cos\gamma & 0 \end{bmatrix} P(k)^- \begin{bmatrix} -\sin\gamma \\ \cos\gamma \\ 0 \end{bmatrix} + R(k) \right)^{-1} \tag{7.17}$$

$$\hat{X}(k) = \hat{X}(k)^{-} + K(k)\left[z(k) - \left(y + d_{offy} - y_w\right)\cos\gamma + \left(x + d_{offx} - x_w\right)\sin\gamma\right] \tag{7.18}$$

$$P(k) = \left(I - K(k)\left[-\sin\gamma \quad \cos\gamma \quad 0\right]\right)P(k)^{-} \tag{7.19}$$

The filtering then cycles back to the prediction stage and the filtering begins again.

7.3 The Feedback Controller

When the position state has been calculated, then the values can be fed back into the controller for the system. Using the control laws provided by Bemporad et al [24], a feedback controller is made. These are:

$$\begin{aligned} v &= \mu\hat{v}_{des} \\ \omega &= -\mu\hat{\omega}_{des} \end{aligned} \tag{7.20}$$

The μ is used to ensure that the speeds for the thrusters do not exceed their maximum speed. As long as maximum speed of the thrusters is never exceeded, this value can be assumed unity.

The desired velocity is a constant value and does not need to be calculated. The desired angular velocity, however, can be calculated using the following [24]:

$$\hat{\omega}_{des} = -\frac{\beta_0\left(d - d_{des}\right)}{v_{des}} - \left(\beta_1 + \beta_2\left|\hat{d} - d_{des}\right|\right)\tan\left(\hat{\theta} - \gamma\right) \tag{7.21}$$

This controller is basically a modified proportional controller which takes into account the error for distance to the wall and the orientation error relative to the wall simultaneously. The beta values are set by the user experimentally to achieve the best results for the particular application.

CHAPTER 8

Future Work

The work that has been completed in this thesis provides the basis for future research and development of sonar systems. There are a number of key areas that can be developed in further work within the Mobile Robotics Lab. These key areas are:

- Developing a pulse amplifier that will correctly reach the desired range of the sonar system
- Developing a time variable attenuator to assist in better detection with the echo sounders
- Research into the feasibility of using a receiver based on the current echo sounder to be the base of a passive sonar array
- Implementing a control system in practice that will utilise the EKF to allow the AUV to perform the tasks of wall following and investigate the possibility of performing simultaneous location and mapping with the AUV
- Further investigate the parameters of the echo sounder to provide a more realistic model of the sonar system.

8.1 Improving the Echo Sounder Design

Two key design changes to the current echo sounder circuit are suggested for future work to improve the performance of the system over the current one.

8.1.1 The Pulse Amplifier

As discussed in Chapter 6, the power output from the transducer is not sufficient to perform echo sounding over the required minimum of 5m.

One recommendation to increase the amount of power that is outputted from the transducer is to increase the voltage across the transformer of the circuit. Figure 8.1 depicts a pulse amplifier [14]. This pulse amplifier attempts to increase the signal strength using a transistor combined with a transformer. The transistor allows for most of the supply voltage, V^+, to drop across the primary of the transformer when it is switched on. V^+ can be increased to allow a much higher voltage to fall across the primary windings of the transformer. Since the transformer is a step-up transformer, the voltage across the secondary winding is increased by the turns ratio of the transformer. This will allow for a greater amplitude signal that will be able to travel further.

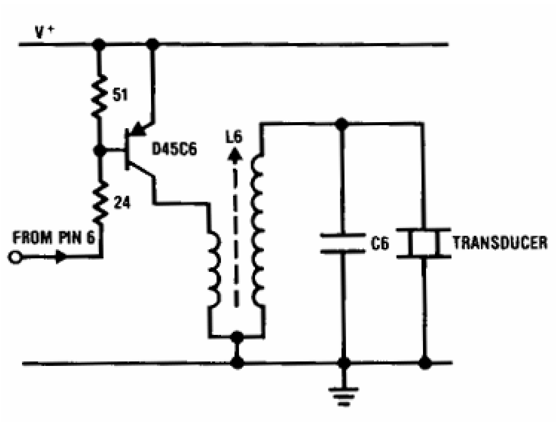


Figure 8.1: The pulse amplifier [14]

However, care must be taken to design a pulse amplifier that only increases the range of the sensor to between 5 metres and 10 metres. This is just to enable the signal to attenuate fast enough to allow consecutive readings to occur rapidly. The current system has a range of less than 5 metres and has a sounding period of less than 6.67ms. Thus, the current time out for the echo sounder waiting for a sonar return is 10ms which can easily be accomplished by the ROBIOS function, OSWait(1). This allows approximately 100 measurements to occur each second for virtually continuous measurements.

8.1.2 The Time Variable Attenuator

A second design change involves the use of a time variable attenuator to improve the performance of the current system.

With the use of a single transducer, as in the current system, the echo sounder is more susceptible to ringing from the transducer. While the transducer is ringing, the system cannot detect any echoes, limiting the minimum detectable distance.

Currently, it is not possible to decrease the minimum detectable distance because the transducer's ringing saturates the receiver. This is not a problem with the existing system because the sensors will be offset by approximately the minimum detectable distance from the extremity of the AUV.

If the minimum distance is found to be too great, it can be decreased using a circuit provided by National [14]. The circuit is called a time variable attenuator.

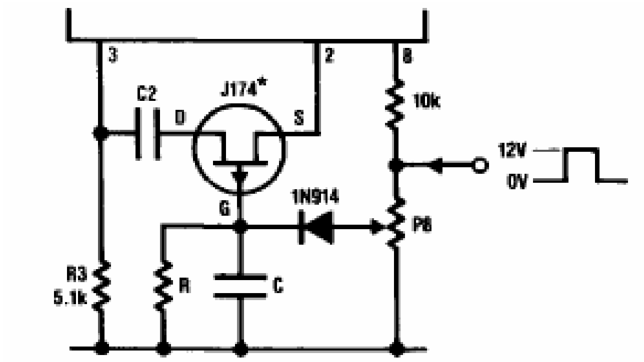


Figure 8.2: Time variable attenuator [14]

The time variable attenuator works by changing the gain of the receiver from a minimum to a maximum over a small period of time. When the transmission pulse comes from the microcontroller, it turns the field effect transistor off, causing the gain of the receiver to become very small. This is beneficial because, at this time, the transducer's ringing is so intense that it saturates the receiver, preventing it from detecting echoes. Since the close echoes are much stronger in intensity than the ringing, they can still be detected even when the gain is near zero. Over time, the gate voltage becomes higher and the attenuation of the receiver signal becomes much smaller. This is also beneficial as the far-range echoes become significantly less intense and require a much larger amplification than the close range echoes. These far-range echoes take longer to return and are thus amplified greater with the time variable attenuator. Thus by implementing a time variable attenuator, the detectable range is increased to include much smaller and larger values.

8.2 Future Research

Some key areas are identified for possible future research in the AUV field. These areas include the development of a passive sonar system, the testing of control systems using simulations of the AUV, and researching the sonar system to provide a more realistic model of the system for use in simulations and calculations.

8.2.1 Designing a Passive Sonar Array

There are times when a passive sonar system is required. For example, in competition, the passive sonar has been used to locate a recovery zone marked by an acoustic beacon. In general underwater applications, an acoustic beacon can be used on a lost submerged vessel that the AUV may be required to find. Methods of developing a passive sonar array need to be investigated for such situations.

One simple method of designing a passive sonar unit is to use the receiver of an active sonar unit attached to an omni-directional hydrophone, which is basically an underwater microphone, instead of a directional transducer. The feasibility of using a device similar to the current unit for passive sonar detection can be investigated in future. If the device is feasible, research can be undertaken into passive sonar detection, such as in Appendix C.

8.2.2 Testing the Control System

The EKF is an extremely powerful tool in the navigation of the AUV using sensors. As is stated in Chapter 7, the EKF fuses redundant sensor readings to determine a better localisation of the AUV in its environment.

The EKF equations presented in this thesis can be used as the basis for testing the effectiveness of the EKF in navigating an AUV around the edge of a pool environment or an unstructured underwater environment.

These tests can be performed in software before implementing them in hardware. With the onset of a simulation for the AUV available for use within the university, it would be possible to test complex algorithms on realistic models of the AUV and its complement of sensors to see if it is possible to use a particular type of control system for the AUV, before committing any finance towards a hardware system. Models can be created to simulate any hardware that is available on the market, so that comparisons can be made to determine the optimum type of hardware system.

8.2.3 Developing a Better Model for the Sensor System

The parameters discovered in this thesis provide some insight as to how the acoustic sensor system behaves on the AUV. A model using this information can be built into a simulation to determine the extent of its effectiveness as a sensor system on the AUV.

Further research into the modelling of the sensor system will allow a more realistic model to be achieved, meaning that the simulation software will be able to provide a more accurate representation of how the AUV will behave in a real environment.

CHAPTER 9

Conclusion

The design of an active acoustic sensor system is presented in this thesis. There has been much work accomplished in the design phase of this project.

Much time was spent researching different approaches to solving the problem of designing a new echo sounder circuit that could fulfil the requirements set out in Chapter 3 and ensure the design's best chance of success. After a design was chosen, it needed to be quickly converted into a physical design to initiate functional testing of the echo sounder and establish the suitability of the new circuit as a part of an active sonar system.

9.1 Outcomes of the Project

There are a number of contributions that were made in this project, outlined in the sections below.

9.1.1 Designing the Echo Sounder Circuit

Many alternatives were considered in the design stage of prototyping a new echo sounder circuit. This included AM transceivers and ultrasonic switches. All these devices had a similar form of operation to an actual echo sounder. However, these designs did not meet the requirements for the project as most of these contained many components which resulted in poor SNR performance. They also used a dual transducer system, rather than a single transducer system. The chosen design was a circuit that contained a dedicated ultrasonic transceiver chip that could perform the entire signal transmission and detection. This allowed for the number of components to decrease on the circuit board and meant that a single transducer system could be implemented. The main benefit of the prototype was that it incurred a cost of less than $150 including the transducer. With the system consisting of four sensors, the total is less than $600. Thus the requirement of the system being less than $1000 was met.

9.1.2 Interfacing to the Eyebot

To allow for proper functioning of the echo sounder, some design choices needed to be made to ensure a correct interface between the Eyebot and the echo sounder.

Decisions made regarding design were:

- Making the pulse to the echo sounder come from the digital output of the Eyebot to ensure that a correctly timed pulse could be transmitted to the echo sounder. The echo sounder required a 200µs pulse.
- Making the logic output from the Eyebot would connect directly to the TPU to enable to the Eyebot to measure the time of flight of the signal, as the TPU accommodates for timing events.

Once the physical connections had been established, the focus turned to interfacing the two units. Using a flow chart for guidance, some C functions were created to properly transmit, receive, time out and read data from the echo sounder. Using a suitable time out of 10ms, the data rate was guaranteed to be approximately 100Hz, exceeding the 1Hz data rate required.

9.1.3 Hardware Verification and Experimental Results

During this stage, the echo sounder was verified as correctly interfaced to the Eyebot.

Experimentation with the echo sounder obtained some key data, such as the minimum and maximum detectable range for the current system and the speed of sound in the test medium. The error margin achieved fell within the 5cm resolution required of the sensor.

Some suggestions for improving the design of the circuit were made and a simple fault tolerant system using time redundancies was incorporated into the software to make the data from the echo sounder more robust to errors in sonar readings. Whilst not achieving the goal of a maximum detection range between 5m and 10m, the suggestions made will enable the development of a sensor that will meet this requirement.

9.1.4 The Wall Following Algorithm

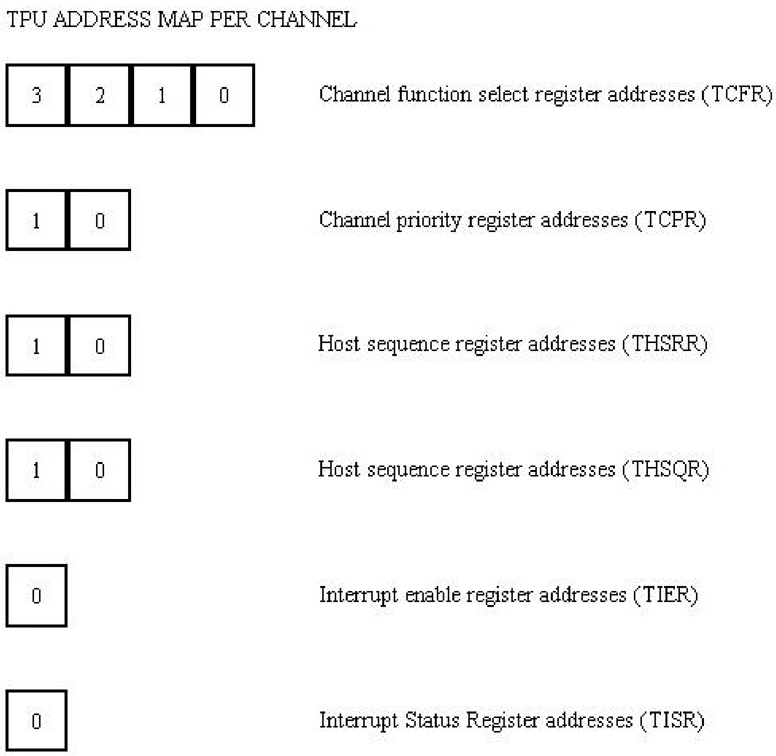
The EKF is used with a control system for a mobile robot designed by Bemporad et al (2000) to design a set of EKF equations and control equations for the AUV. These can be used, once the control system of the AUV is fully developed, to navigate the AUV around the edge of a pool.

9.2 Final Word

The goal of this project was to design a low active acoustic sensor system for an AUV. The design was accomplished, with effective system functioning and establishment of control equations to use with the system. Further research needs to be conducted to improve the range and modelling of the echo sounder system. However, this project has provided a platform on which future research can build on improving AUV performance in a number of different environments.

Appendix A: The Time Processor Unit

A.1 Channel Control



Function code	Host service request code	Host Sequence code
0xA	1 = Initialise channel	0 = No link, single capture 1 = No link, continuous 2 = Link channels, single capture 4 = Link channels, continuous

Table A.1: TPU code options [22]

- The interrupt enable must be set to 1 if the TPU channel is to interrupt the CPU
- The interrupt status register indicates when an interrupt has occurred for the channel and does not need to be set
- The channel priority register is the register that begins operations for the channel and must be set last. The code for it determines the priority at which it is serviced by the TPU. This ranges from 0 to 4. If the register is zero, this means the channel is not on.

Apart from the interrupt status register, all the registers need to be set before the channel can operate properly.

A.2 Parameter RAM

The following is what is required in the parameter RAM in order to allow the ITC function to operate properly.

Parameter	Value	Meaning
Inputs:		
CHANNEL_CONTROL		
Bits[1:0]	{11}	Input channel
Bits[4:2]	{000}	Do not detect transition
	{001}	Trigger on rising edge
	{010}	Trigger on falling edge
	{011}	Trigger on either edge
Bits[8:5]	{00xx}	Timer on input channel
	{000x}	Reference timer 1
	{001x}	Reference timer 2
MAX_COUNT	$0 < N \leq 0xFFFF$	Number of events to count

Table A.2: Parameter RAM for ITC [22]

Appendix B: The Timer Frequencies

The table defines the possible clock frequencies that are available to the timer used for the TPU [Harman, 1991]. This is based on a 16.78MHz system clock. The resolution is the clock signal's period used for clocking the timer. As the distances for the echo sounder range exceed the 5m to 10m range boundary set for the project, the time range may also need to increase. This will mean that the resolution will not be as fine. The 31.25ms range will be necessary when the distance range is between 11.7m and 23.4m.

Frequency	Resolution	Range
4.19MHz	238.4ns	15.6ms
2.097MHz	476.8ns	31.25ms
1.049MHz	953.7ns	62.5ms
0.524MHz	1.91µs	125ms
262.1kHz	3.81µs	0.25s
131.1kHz	7.63µs	0.5s
65.53kHz	15.3µs	1.0s

Table B.1: List of TPU timer frequencies, and corresponding resolutions and ranges [22]

Appendix C: Passive Sonar Application

The possibility of being able the convert the active sonar unit to a passive sonar unit, by blanking out the transmitter, enables the development of passive sonar applications. The main use for passive sonar is the detection of a sound source underwater. The advantages for this approach are:

- The sensor can detect a sound source from twice the distance that an active sonar unit can, due to the fact that the signal must be sent out and reflected before it is detected, reducing the range of the active sonar. This is assuming the same signal intensity is transmitted from both.
- The location of the sensor cannot be discovered when using passive sonar, because the sensor does not emit a sound. When a sensor emits a sound, it is possible to locate the sensor using an array of passive sonar. This is why passive sonar is used in the military.

It is, therefore, beneficial to research passive sonar detection.

C.1 Time Delay Estimations

The main method of detecting the DOA of a signal is by using the time delays between sensors and the geometry of the sensors to triangulate the DOA. With the use of a system, implemented with sensors similar to the ones for this project, it is easy to determine an approximate time delay between sensors.

In order to triangulate the DOA of a signal in 3 dimensions, 3 time delays are required. This will mean that at least 3 sensors will need to be placed in the array. However, the arrangement of 3 sensors means that the sensors will have to be timing continually as there is no way of determining when a signal will be approaching.

A much more effective method is to use four sensors. This system will rely on a trigger system to activate the sonar array. Initially, also the sensors will be set to detect a signal of the correct frequency. This will not require any CPU processing time as the TPU is a separate module from the CPU on the Motorola 68332 microcontroller. Once a signal impinges on the first sensor in the array, the three other sensors' timers are triggered, using the linking capabilities of the TPU channels, to determine how long it takes for the signal to arrive at each individual sensor. This is shown in Figure C.1. The red sensor is the sensor that detects the signal, and the other three green ones must time how long it takes for the signal to arrive.

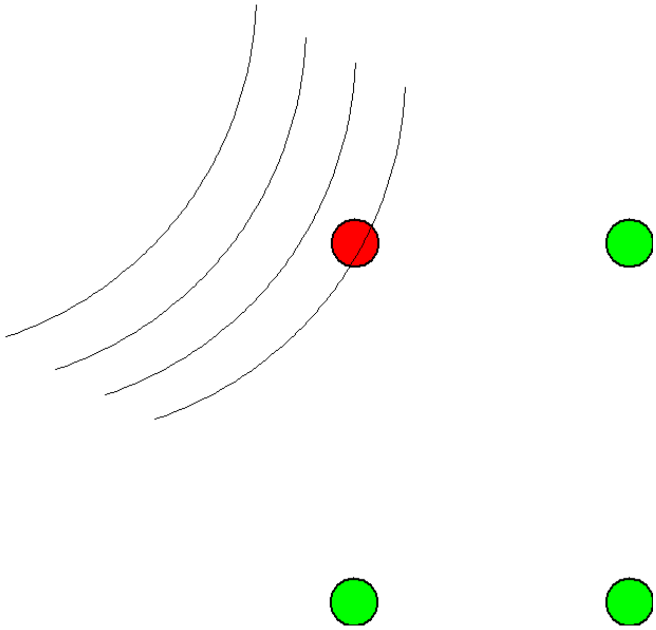


Figure C.1: Signal impinging on the first sensor of passive array

For a much more robust method of detecting the time delays between sensors, a digital signal processor (DSP) is required. The method behind this is to digitalise the signal that is being received by the sensors. It is accomplished by using an ADC.

The digitalised signal is then sent to the central processor where the signal is cross correlated with a neighbouring sensor's signal to determine the time delay between the two sensors. A cross correlation is a function that reaches its maximum value when the phase or time delay between two signals is matched. To perform a cross correlation on two signal sequences, the best approach is to perform a fast Fourier transform (FFT) on both of them, multiply the sequences together and then inverse FFT the resultant sequence [2], as this is computationally less expensive.

$$
\begin{aligned}
R_{xy}(\lambda) &= \frac{1}{N}\sum_{k=0}^{N-|\lambda|-1} x(k)y(k+\lambda) \\
&= \frac{1}{N}F^{-1}\left[X*(f)Y(f)\right] \\
R_{xy}(\lambda) &= \frac{1}{N}F^{-1}\left[F[x(k)]F[y(k)]\right]
\end{aligned}
\tag{C.1}
$$

By analysing the resultant cross correlation sequence for the maximum value, the time delay between the two sensors is discovered.

This method is far more robust then the first method, that uses the echo sounder as a receiver to detect time delays, because this can be used when the SNR at the sensors is not very high. The cross correlation give the best estimate of the time delay in a noisy environment, whereas the echo sounder receiver will not be able to detect the signal and thus cannot provide a TDE for noisy environments. This does come at a cost as the DSP systems are much more expensive and difficult to build.

C.2 Direction of Arrival Calculation

Once the TDEs are determined for the sensor array, then the DOA can be calculated, using the TDEs and the geometry of the sensors.

The time delay direction finding (TDDF) algorithm by Berdugo et al [4] is the easiest method for determining the DOA.

Now the sensor positions can be represented by:

$$\vec{r}_i = \begin{bmatrix} x_i & y_i & z_i \end{bmatrix} \tag{C.2}$$

The differential time delays can be placed in a vector:

$$\vec{\tau} = \begin{bmatrix} \tau_{12} \\ \tau_{13} \\ \tau_{14} \end{bmatrix}; \tau_{ij} \equiv \tau_j - \tau_i \tag{C.3}$$

And the DOA vector is:

$$\vec{k} = \begin{bmatrix} k_x \\ k_y \\ k_z \end{bmatrix} = \begin{bmatrix} \sin\theta\cos\phi \\ \sin\theta\sin\phi \\ \cos\theta \end{bmatrix} \tag{C.4}$$

The time delay between two sensors can be determined as the projection of the distance vector between the two sensors onto the incoming wave vector, k, divided by the speed of sound, c.

$$\vec{\tau} = -\frac{R\vec{k}}{c}; R \equiv \begin{bmatrix} \vec{r}_2 - \vec{r}_1 \\ \vec{r}_3 - \vec{r}_1 \\ \vec{r}_4 - \vec{r}_1 \end{bmatrix} \tag{C.5}$$

The main approach to this method is to attempt to find the DOA vector that will minimise the error, ε, between the estimated time difference and the predicted time difference. This can be accomplished via a least squares error method.

$$\vec{\varepsilon} = \frac{\hat{R} \times \vec{k}}{c} + \hat{\vec{\tau}} \tag{C.6}$$

Where $\hat{R}$ and $\hat{\vec{\tau}}$ are the estimated values of the distance vector and the time difference vector, respectively.

From the DOA vector, it is simple to determine the azimuthal angle, φ, and the elevation angle, θ.

$$\begin{aligned} k_x &= \sin\theta\cos\phi \\ k_y &= \sin\theta\sin\phi \\ \frac{k_y}{k_x} &= \frac{\sin\phi}{\cos\phi} = \tan\phi \\ \hat{\phi} &= \tan^{-1}\left(\frac{\hat{k}_y}{\hat{k}_x}\right) \end{aligned} \tag{C.7}$$

$$\begin{aligned} k_z &= \cos\theta \\ \hat{\theta} &= \cos^{-1}(k_z) \end{aligned} \tag{C.8}$$

Appendix D: The Contents of the CD

D.1 The Thesis

Copies of the thesis have been stored as Word and PDF documents

D.2 The Pictures and Drawings

All the pictures used specifically for the sonar component of the AUV is included in this file

D.3 The Code Set

The code on the CD has been sorted out into three lots. The different sections represent the three different stages of coding in this project. They are:

- Testing the Navman Depth 2100 echo sounder
- Testing and verification of the prototype echo sounder hardware
- Final code of the prototype echo sounder and dynamic testing

D.4 The Datasheets and Research

Datasheets, including one for the LM1812 ultrasonic transceiver chip, have been included as well as research documents that may be useful for future research. These are stored in PDF format.

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