Borrowing the Bat's Ear for Automation – Ultrasonic Measurements in an Industrial Environment

Gunnar Lindstedt

Department of
Industrial Electrical Engineering and
Automation (IEA)
Lund Institute of Technology (LTH)
Box 118
S-221 00 LUND
Sweden

ISBN 91-88934-02-0
CODEN: LUTEDX/(TEIE-1012)/1-122/(1996)

Abstract

This work focuses on measurement principles based on ultrasound for applications in industrial automation. The measurements made are of geometric nature. The methods used are when possible based on experience and/or inspiration from the study of bats. A platform for ultrasonic measurements, that allows a VME-based host computer to transmit and receive signals, with a considerable flexibility has been built. Sensor units utilizing both piezoelectric and electrostatic transducers are used. The dimensions of the units are about 10x10x7 centimeters. Frequencies in the range 40-200 kHz are used. The presented measurements include distance, flat surface spatial angle, object localization, object orientation, and object recognition. To make the methods robust matched filters are used in several of the measurements. Frequency sweeps are used to make the ultrasonic measurements more robust to various environmental parameters. Methods that utilize the new information provided by the frequency sweeps are also suggested.

Contents

Preface

The View of Ultrasound
Ultrasound and ultrasonic techniques are perceived by man as abstract subjects since they are based on physical phenomena that we can use but not feel with our senses. In this group we find many related subjects like X-ray, electromagnetic fields or nuclear radiation. However, the way we look at ultrasound is probably even more confused by the fact that we, for very good reasons, call it sound and immediately associate it with the audible frequencies. The confusion is caused by the different use of the sound. Man uses sound primarily as a communication medium and secondarily for an approximate localization. The latter use has a much lower priority and resolution. On the other hand the artificial use of ultrasound primarily focuses on the determination of different geometric measures. This different purpose further increases the confusion and often makes us associate the technique with hearing instead of seeing, that might be closer to the actual use. To directly compare ultrasonic measurements with optical ones is however not adequate. If the comparison is made using the optical conditions as a reference the resolution for ultrasonics is certainly inferior. This is because we treat with special favor the two dimensions that the optical methods are good at measuring. If we instead consider how the bat hunts and catches a prey in a time interval of fractions of a second, using only ultrasonic echolocation, the comparison result is somewhat different.

It is my intention with the following chapters to try to explain some of the ultrasonic measurement techniques and in that way also reduce the confusion mentioned above. There are all reasons for a humble approach to this area. The presented measurements seem both primitive and clumsy when compared to the measurements performed by bats. This makes a continuous comparison to the bat techniques a valuable tool. The size of a bats brain approximately equals the size of a pearl but in this area he is a superior specialist.

Previous Work and Subprojects
The work presented in this thesis is partly based on work earlier presented in my Licentiate Thesis [1]. To make the entire material conveniently accessible to the reader and to update the previous material when needed, it has been brought together in the following chapters. Material from the publications [2]-[6] is also included. During the work a number of guest researchers and students working on their Master's Theses have contributed to the achieved results by studying special problems. Their works are here listed in a chronological order:

In 1991 Martin Urban made some early measurements of the functions of piezoelectric elements, [7], as a visiting scholar. Later the same year Kirill Kowalew made his Master's Thesis work primarily focused on the detection of the beginning of an echo using piezoelectric elements. Main results from this work is found in [8]. In 1993-1994 Sven Rogdahl also made his Master's Thesis work. This mainly concerned the localization problem and related filtering [9]. Last year, 1995, Luis Sobral made a Master's Thesis work at the Department of Automatic Control under supervision of Dr. Rolf Johansson. During this work an object recognition method for translated objects was developed [10].

Acknowledgements
This work has been financially supported by The Swedish National Board for Industrial and Technical Development (NUTEK), under contract ”Moving Autonomous Systems”.

The presented work was carried out at the Department of Industrial Electrical Engineering and Automation, IEA. I would like to express my gratitude to my supervisor at the Department, Professor Gustaf Olsson, for his experienced and enthusiastic guiding through the work and for always trying to make some time for a discussion, even if his schedule had three parallel activities already. I would also like to thank everybody at the Department for all help I have received during the work and for the creation of an open, friendly and creative atmosphere to work in. Furthermore I express my sincere thanks to my colleagues from other departments working on the contract ”Moving Autonomous Systems” for many stimulating discussions.

For giving me the opportunity to do this work I also want to thank my family, my wife Mariette and our two daughters Ingrid and Kristina. You have shown a tremendous patience with the family member that has been in exile at the computer and not been a part of the social life of the family for the recent months.

Gunnar Lindstedt

References for the Preface in order of appearance.
The complete reference information is found in the main reference list later in the thesis.

[1] Lindstedt, G. (1993) A Versatile Ultrasonic Sensor System for Industrial Robot Applications.
[2] Lindstedt, G., G. Olsson (1991) An Ultrasonic System for Robot Sensor Feedback.
[3] Lindstedt, G., G. Olsson (1992) Using Low Cost Piezoelectric Ultrasonic Sensors for 3D Measurements in Robots.
[4] Lindstedt, G. (1992) Computation of Spatial Angle from Ultrasonic Measurements.
[5] Lindstedt, G., G. Olsson (1993) Using Ultrasonics for Sensing in a Robotic Environment.
[6] Lindstedt, G. (1995) Design Catalogue of Modules in Ultrasonic Measurement System.

[7] Urban, M. (1991) Ultrasonic Transducers and their Application on Robot Control.
[8] Lindstedt, G., K. Kowalew (1991) Distance Measurement in Robotics Using Ultrasonic Sensors.
[9] Rogdahl, S. (1994) Objektlokalisering med ultraljudssensorer i robotmiljö.
[10] Sobral, L. (1995) System Identification Applied to Ultrasonic Detection Problems

Ultrasonic Imaging 1

This chapter makes an approach towards the topic of imaging for feedback purposes in general. It is emphasized that the purpose of the imaging defines the required information and thus defines the set of appropriate techniques. The chapter gives an overview of applications where different forms of imaging is used for various purposes. The information provided by ultrasonic measurement systems in different media is discussed. The ultimate state of the art in ultrasonic imaging in air, namely the bat with its highly skilled navigation system, is described. Finally an overview of the thesis is given.

1.1 Introduction

This thesis describes several ways of using ultrasonic echo-signals to perform different kinds of imaging in order to identify the environment. The work is primarily inspired by feedback problems in the robotic area [Lindstedt, 1993] but is applicable on a wide variety of automation problems. The main contributions are found in an area between simple measurements, like one dimensional position measurements, and methods for detailed description of a topology like the ones found in [Knoll, 1991] and [Watanabe and Yoneyama, 1992]. The latter is maybe the group that comes to mind when talking about the topic of imaging. However, all images contains only a subset of the original information (e.g. a black and white photo contains much detail and view information but very little color and depth information). Consequently, there is no absolute limit for what can be considered as an imaging process and the term will here be used in a very wide meaning.

Humans often try to design systems that execute actions that could have been executed by humans. This is a large share of the automation field. As a consequence, the sensors and the actuators are of the same nature that a human would have used. This is often not the most suitable solution.

The topic for this thesis is about sensory systems not typical for humans. Roughly speaking, it could be called ”variations on seeing with sound”.

1.2 Purpose of Imaging Measurements

There are many operations in industrial processes that require some kind of feedback from measurement systems. The measurement information can be used in one or more of the following purposes:

- part of a feedback loop;
- error detection and diagnosis;
- personal safety;
- quality control or inspection.

At the design stage of an industrial process it is, however, not indisputable that these tasks should be solved with a measurement systems. Alternative solutions like redesign of the executional part of the process step may often make the measurement system superfluous. These solutions are often, for good reasons, preferred since measurement systems may be costly or not sufficiently dependable. It is, however, important to note that a redesign of an operation usually makes it more expensive. A balance between more elaborate design and adequate measurement systems is thus established. Consequently there is always a market for measurement systems with a good (performance + robustness) / cost ratio.

To introduce a sensor for a single scalar variable in a mechanical process is usually a rather simple task. The costs are possible to foresee, the uncertainty is known and the transducer signal can readily be connected to the control system. Consequently, the considerations in these cases are straightforward.

When the measurement have to include information about more unstructured relations, like the shape of a complicated object or the correct assembly of a workpiece with many included components, more elaborate considerations have to be made. This is the area where the emphasis of this thesis is found. The methods and experiments are applied on geometrical measurements, so the following discussion is focused exclusively on this category of measurements.

Vision Systems
One of the most successful measurement principles concerning geometry is the use of vision systems. A video camera and an appropriate computer with a frame grabber facility can often serve as a powerful tool. The fact that these components drastically have become cheaper during recent years has led to a commercial breakthrough in industrial automation applications. During 1995 about 10 000 systems were sold in Europe and probably even more systems will be sold in years to come.

In many of the applications where vision is used, there may be other technically preferable solutions. Nevertheless, there are two important reasons why vision is still used:

- Video cameras and video image processing equipment have become highly standardized components that are produced in large series. Because of this the costs are in their favor. This makes them attractive compared to systems based on more rarely used principles.
- The preference for human like solutions, that were mentioned earlier, attracts most designers. The ”hands on”-feeling when using a picture image makes the programming less abstract than e.g. a pattern recognition method on an ultrasonic echo signal.

There are several examples where vision systems have been used for tasks where they are not appropriate. This usually has lead to a cumbersome programming phase.

The vision solution together with tactile information is close to a human approach to geometrical measurement problems. Tactile sensors to give information similar to the one provided by a human hand have however not been applied to industrial processes to any large extent. Problems with the sensor design and lack of a comprehensive information structure are typical factors that limit their use. Thus, a vision measurement system in a multi-purpose industrial use is very much inferior compared to the human system.

Other Systems for Geometrical Measurements
To make the industrial measurement systems for geometry more skilled the perspective must be wider than to look at the measurement principles used by the human. From applied physics and measurement technology we know many principles that can be useful e.g.

- laser triangulation; [Hirzinger, 1987]
- transition time methods using microwaves, light, or sound; [Jarvis, 1983]
- interferometric methods; [Idesawa, 1987]

The list can be made long. For specific measurement purposes it is almost always possible to find a suitable measurement principle. In a multi purpose approach (i.e. looking for a measurement method on the same specification level as vision) the list is much shorter.

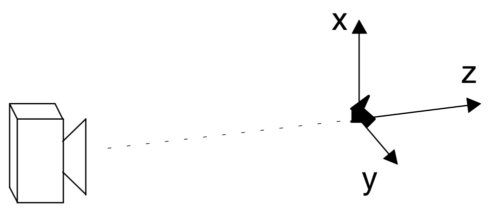

Figure 1.1 The geometrical measurement situation

A simple classification of the measurement principles can be made according to Figure 1.1. If the measurement system is in the observer position then vision-like systems provide geometrical information in the x- and y-directions while triangulation and transition time (pulse echo) systems primarily provide z-direction information. Variations in the measurement system setup like stereo vision or scanning triangulation may of course add information dimensions but are often associated with limitations.

When selecting measurement principles for industrial use it is not sufficient to analyze the physical characteristics. To work in a wide range of industrial environments the measurement principle must be robust. The effect on the measurements of inevitable disturbances should be possible to compensate for.

The system cost is also essential. Vision systems have shown that a high cost system may turn into a low- or medium-cost system due to both the numbers produced and the technical development. However, it is always an advantage if the measurement principle implies low costs because this usually increases the interest from the industrial users at an earlier stage.

If the measurement problem is focused on the z-axis information or if z-axis information has to be added to another measurement system various pulse-echo methods are of great interest. The use of microwaves or light implies that the time-delays becomes very short for most adequate distances in manufacturing. Consequently, the systems become expensive. In these cases ultrasonic measurements can provide a feasible compromise between resolution and time-delay. Some of the main advantages with ultrasonic measurements are:

- no risk of hurting people in the environment,
- low cost sensors and systems,
- insensitive to light, smoke, ”dirt” , EMI.

When a pulse-echo ultrasonic measurement system is to be used for some kind of imaging in the z-axis direction the excellent two-dimensional images in medical applications are easy to remember. However, these pictures are rather complicated to synthesize from the echo signals. The goal is to produce a picture (vision-like) of the different layers of tissue, bone, liquid and so on. This is done from a signal containing echoes from the changes in acoustic impedance between layers. The directions for the echoes also have to be determined to produce the two-dimensional image. In industrial measurements the purpose often doesn't require such a descriptive image. Since produced parts have a more defined structure than humans pattern recognition methods are more applicable. This can also be performed in a much shorter time.

1.3 Current Applications of Ultrasonic Imaging

The first practical use of ultrasonics is dated somewhere around 1915. At this time the applications were underwater measurements and this is significant for the further use of ultrasonics. Most use and development has concerned ultrasonics in liquids or in solid materials. However, improvements in sensor technology and signal processing make the use of ultrasonics in air and other gases a feasible solution to many industrial measurement problems.

One-dimensional Measurements
For pure non-tactile distance measurements it is possible to buy a complete system based on pulse-echo ultrasonics and built into a industry standard transducer housing . This unit is highly reliable and can be used in environments where for instance optical methods can't be used. A simpler version of this measurement system is the handheld distance meter available at a cost of about $30-$40. The accuracy is of the order 1%, the resolution is 1 cm and the measurement range is from 0.5 meter up to about 10 meters if the object is large, like a large wall.

Two-dimensional Measurements
An application where ultrasonic transducers actually are used to produce a two-dimensional image is the mobile robot obstacle avoidance systems. [Buchberger, 1993] Various designs have been presented but most of them consist of a number of fixed sensors placed around the robot so that obstacles in different directions can be detected. This system often covers all directions around the robot (Figure 1.2) but sometimes only the most interesting areas (forward and backward direction) are scanned.

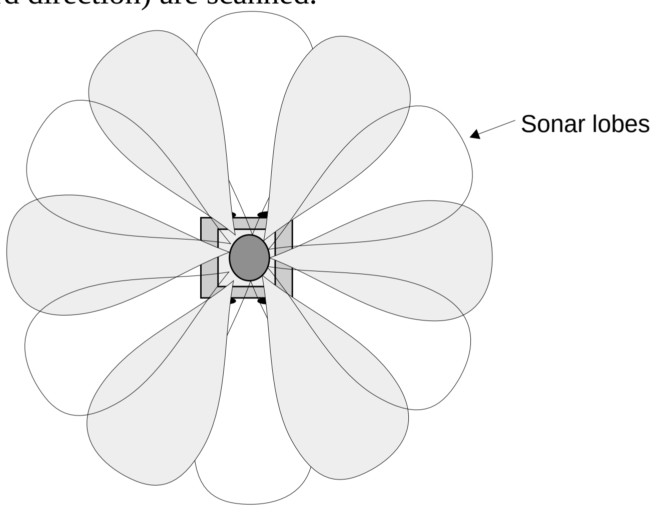

Figure 1.2 Ultrasonic obstacle detection system in a mobile robot

At a first glance these configurations seem to be as simple to use as the one-dimensional transducer but in practical use many problems arise due to crosstalk and the directivity of the sensors. Mobile robots often move rapidly and moving obstacles may turn up within fractions of a second. Consequently, the system have to detect an obstacle very fast. This is difficult because of the transducer crosstalk. The transducers can't transmit pulses simultaneously if not special precautions (e.g. synchronization to generate a circular wavefront) have been taken. In large rooms echoes from distant large walls also reduce the possible repetition rate for the measurement.

Medical Imaging

Although this thesis emphasizes ultrasound in air it is almost impossible to avoid the area of ultrasonics in medical imaging [Jacobson, 1975]. Large efforts have been spent to develop techniques for different types of imaging of the human body. The goal for most imaging in this area is to produce a comprehensive image that corresponds as much as possible to the physical layout of the body interior. This is crucial since the interpreting (diagnosis) is done manually.

First of all it is important to note that the physical conditions for an ultrasonic signal transmitted and measured in liquid are quite different compared to the conditions in air and in gases. The significantly higher speed of sound in liquid and soft tissue (≈1500 m/s) compared to sound (≈340 m/s) makes the wavelength much longer. This calls for the use of higher frequencies. For medical diagnostics are usually frequencies in the range 0.5-10 MHz used. These high frequencies would be very difficult to use in air for several reasons. Some of the typical conditions for ultrasound in air are:

- the attenuation is high at high frequency,
- the transmitted lobe is narrow,
- reflections are beam-like and thus only detectable in one direction,
- the transducer design is more difficult (smaller dimensions and thinner layers)

The medical imaging principles are usually divided into three groups that are called A-, B-, and C-scanning. The measurement principles are indicated in Figure 1.3.

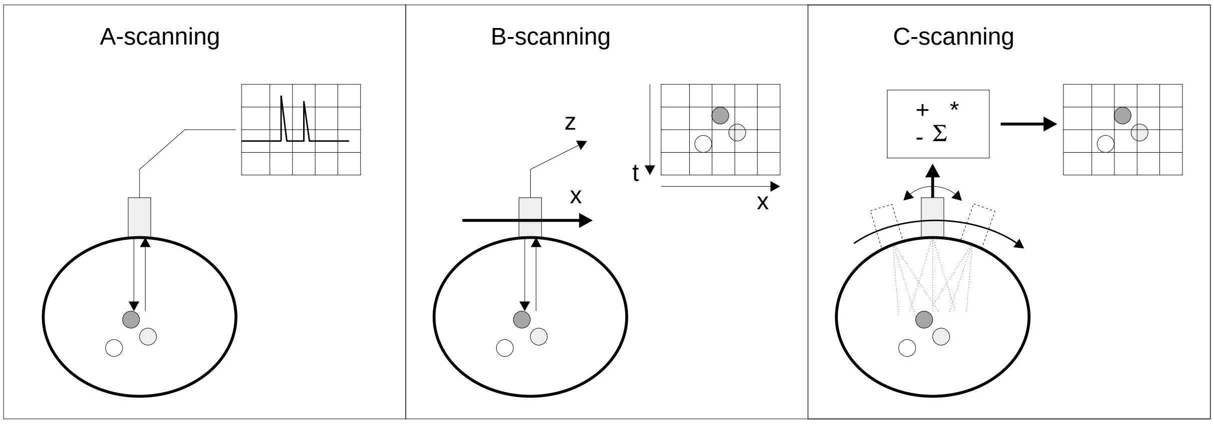

Figure 1.3 Scanning principles

The **A-scanning** is almost equivalent to the ultrasonic material testing that is done in the industry to detect internal cracks in solid materials. A single transducer transmits a pulse into the body and the reflected echoes are detected. Reflections are caused by all layer boundaries that the signal passes. When the signal meets a surface where it proceeds from a medium with one acoustic impedance to another with a different impedance a part of it is reflected. The amplitude of the echoes and their delay time from the transmission gives information about the type of media and the position. This method is for instance a powerful tool in situations where it is crucial to detect if cavities in the body are filled with liquid or empty.

In the **B-scanning** the transducer unit from the A-scanning is moved sideways so that the measurement information is coming from a slice of the body. The resulting image is usually presented as a two-dimensional area where one dimension represents the position of the transducer. The other dimension represents the (delay-) time. Finally the intensity or color in the individual point represents the echo energy. In this way is a cross-section of the body shown as an image (tomographic image). This method has got many problems connected with it. The two major ones are:

- A high reflection amplitude requires that the layer change plane is close to perpendicular to the transmission direction. Other planes give very vague contours.
- If different layers with different speeds of sound are passed during an echo the distance may be difficult to calculate. The time-delay and the distance are no longer proportional with the same coefficient during the entire ”time of flight”. The image is produced with several parallel measurements. Because of this the calculated position of distant layers may not be comparable over the image due to different configurations in closer layers.

The **C-scanning** is also called compound scanning. This is the most complicated method but also the method that produces the most comprehensible images. Measurements are made with transmissions in different directions from each transmission point. The transmission point is then moved along a circular path around the examined area. In this way the layers in the body are exposed from different directions so that contours that are invisible when the B-scanning is used now can be detected. This method requires more signal processing to produce an image but the result is significantly better.

Ultrasonic measurements for medical diagnosis is today a standard method in clinical use. It can detect tissues and liquids that can’t be examined with x-ray although the resolution is inferior. Ultrasound for diagnostic purposes is absolutely harmless and the examinations usually require almost no preparations.

1.4 The Bats Ear - The Potential of Ultrasonic Measurements

When human attempts to use ultrasound in air for imaging purposes are compared to the bats superior ultrasonic system the human is bound to be inferior. The bats real-time performance, accuracy and resolution are quite impressive. But is the bat actually solving the same problems that have to be solved in automation applications? The answer to this is undoubtedly yes! The bats navigation system is very versatile and can handle most of the problems caused by disturbances, noise, interference and physical phenomena. Consequently they can serve as our ideal model. The basic principles for the bats sonar systems are described in detail in [Griffin, 1958].

It is easy to be impressed by ”the real sonar experts”, the bats. Their measurement rate is very high. During normal navigation they transmit about 10 pulses per second but can, while hunting a prey, increase the rate to about 200 pulses per second. It is also interesting to note that they direct the transmission towards the most interesting direction, for instance a prey. [Simmons, 1989], [Kuc and Barshan, 1992].

The bats transmit pulses in the frequency range 20-100 kHz depending on activity and species. A noticeable difference compared to existing industrial systems is that most bats transmit a pulse that consists of a frequency sweep. This sweep also contains harmonics. The repetition rate and the length of the pulse is adapted to the activity so that an adequate information flow and resolution can be achieved for instance during the final approach to a prey that will be captured.

When several bats simultaneously fly in a dark cave the ”ultrasonic pollution” must be massive. Still each bat is able to navigate using its own echoes. Research in this field has shown that the bat recognizes its own ”voice” and thus separates its own echoes. The voice characteristics are, like for humans, formed by a specific frequency profile. This profile puts a tag on the speech. This tag, however, should not be mixed up with the main frequency level. This fact is easily understood if one thinks of a human singer (like Leonard Cohen). The singer is as easy to recognize whether he sings a C or an A. Moreover, a singer can add a specific amplitude variation so that the stress is put at a specific time interval of the tone. Together with the super- and sub-harmonics a complicated tag is formed. Accordingly, also bats can change the frequency of the transmitted pulse within the range 50-100 kHz, maintaining the tag. Complex signals with varying frequency can also be used by the bats but always with a limited pulse length so that the pulse-echo principle is used.

It would be presumptuous to think that we in the near future could be as skilled as the bats in using ultrasonic imaging. Still, we can learn a lot from the ”bat principles” and make artificial implementations of them. The basic bat lesson is, that it is crucial which signal is produced by the transmitter. It is obvious that some frequencies are more suitable than others for detection of a specific object. Moreover, the repetition rate and the amplitude are other important parameters. In other words, *no advanced signal processing can replace a creatively chosen signal*.

When various ultrasonic measurement methods meet the demands from a real industrial measurement task usually the uncomplicated methods are preferred. This is due to both time consumption and lack of robustness associated with the more sophisticated methods. The adequate question is then: How complicated is the bat signal processing and is it possible to do something like it in an artificial unit? A lot of research has been done in the field of describing the bats sonar processing [Suga, 1990].

However, the main conclusion is that it is crucial not to think about all imaging as if the goal is to produce a two- or three-dimensional picture of the environment. The processing that the bat performs is strictly focused on the purpose and so should an industrial system be. If the bat is hunting a prey, then a special process is done upon the information concerning the prey movement. This has to be a high speed process to allow a sufficiently fast feedback. Information about a fixed object that should be avoided don’t have to be processed this rapidly. Accordingly this ”ordinary” navigation process can take place on a lower priority-level and be processed in parallel. This is the same thing that a human can perceive as a natural defense function. If something moves rapidly at the edge of the eyes vision lobe it is noticed with a high priority and catches our attention. This happens although it is almost impossible to get any detailed information when we try to look at something so far from the eyes center-axis. The high priority signal is also used to perform an involuntary turn towards the rapid movement. This purpose driven signal processing is a field where we can learn a lot from the bats.

1.5 Topic for the Thesis

As indicated in the previous sections a lot of valuable work can be made in the area of creating ultrasonic measurement principles that can be used in a wide variety of industrial situations. The measurements have to be robust in industrial environments which most often leads to straight-forward solutions. It is essential to understand that many industrial measurement needs can be satisfied without creating images that are easy to interpret for a human. It is also important that a measuring device has to have limited dimensions. If the bats can – we should at least try.

On the basis of these considerations this thesis describes the design of and measurements with an experimental platform for ultrasonic measurements in industrial applications. The design involves both the use of piezoelectric and electrostatic sensor elements. The measurements include distance, spatial angle, object recognition and object localization. Experiments with frequency sweeps are also described.

1.6 Essential Results

The work covered by this thesis shows that a sensor unit with a feasible size can be used for a wide variety of measurements in automation situations. The measurements include distance, spatial angle, object recognition, object localization and object orientation measurements. A direct industrial application of the object recognition methods seems quite feasible. The utilization of transmitted signals with varying frequency makes many of the measurements more reliable and extends the usability of the measurement principle.

A flexible modular system has been designed, built and tested by the measurements. All subsystems in the measurement sequence are designed at component level. Consequently, hardware solutions to all measurement problems are included. This also gives an impression of the favorably low price of a commercial version of the system.

1.7 Overview of the Thesis

Chapter 2 of this thesis briefly describes the principles for ultrasonic sensors for use in air. The data and features of the sensor types are evaluated. The physics for ultrasonic waves in air and some of the phenomena that must be considered are also described.

In chapter 3 the design of the measurement system hardware is found. Both the sensor units (piezoelectric and electrostatic) are described together with the necessary electronics. This includes transmission pulse generation, driving the transducers, sensor amplification, sampling and data storing.

Chapter 4 describes distance measurements. The front detection of an echo is a big problem, especially when piezoelectric sensors are used. A number of solutions to this problem is presented here.

Chapter 5 is dealing with flat surface angular measurements and the closely related object localization problem. High accuracy spatial angular measurements can be made when the difference in phase angles is measured When the surface is too small an angular measurement can’t be done. If objects with smaller dimensions are used the measurement becomes an object localization. This problem requires a classification because the interpretation of the position of an object depends upon the relation between the distance to the object and the size of the object. A small object is easy to define the position for. However, from an echolocation point of view the position of a large object is the position of the closest reflecting part. This may not be the same position for all sensing elements. This is discussed in this chapter and some measurement results are presented.

Chapter 6 describes some powerful and directly industrially applicable object recognition methods. Recognition with both one and three sensors are presented with measurement results. In the three sensor case an object orientation measurement is made.

In chapter 7 are the limitations of single frequency measurements discussed. The echolocation principles of the bat is described. Finally are the bat principles and the principles that are possible to apply in an ”electronic bat” compared.

Chapter 8 discusses the application of multifrequency measurements, especially the frequency sweep. Experiments where frequency sweeps are used in some of the previously presented measurements are shown.

Chapter 9 finally summarizes the achieved results, analyzes the industrial relevance, and indicates possible further research and development concerning the use of measurements using airborne ultrasound in industrial applications.

Basic Mechanisms of Ultrasonics 2

This chapter explains important parts of the physical background for ultrasonic measurements in air. The functional principles for the most commonly used sensor elements, the piezoelectric and the electrostatic elements, are described. Characteristics and typical data for the sensor types are compared. During practical measurements some phenomena and special situations may show up. These are described and discussed in the last part in the chapter.

2.1 Ultrasonic Waves in Air

General Considerations
Ultrasound is usually defined as all sound with a frequency above 20 kHz. The only practical reason for this limit is that ultrasound then becomes equal to sound with a frequency too high for humans to hear. If we consider the physical behavior of sound with various frequencies there is no natural sharp limit. Some parameters are continuously affected when the frequency is changed while many stay the same. This means that much of the theory that is valid for audible sound can also be used for ultrasound.

For all the theory that will be presented it is crucial to assume that the medium for the ultrasonic waves should not be changed after that the wave has passed it. A major condition for this to be true is that the amplitude of the wave is sufficiently low.

The Speed of Sound
Sound propagates in free gases and liquids (without bounding surfaces) primarily as a longitudinal wave. For a wave that propagates in the direction of the x-axis the wave equation can be written as

$$\frac{\partial^2 \xi}{\partial t^2} = c^2 \cdot \frac{\partial^2 \xi}{\partial x^2} \tag{2.1}$$

where ξ is the particle displacement, t the time and c the propagation velocity. From this expression and what is known about the relations between pressure, density, volume and temperature in gases it is possible to give an expression for the propagation velocity in a gas. In [Blitz, 1967] and [Kinzler and Frey, 1982] it is shown that

$$c = \sqrt{\frac{\gamma \cdot P}{\rho}} = \sqrt{\frac{\gamma \cdot R \cdot T}{M}} \tag{2.2}$$

where γ is the ratio between the specific heat at constant pressure and that at constant volume (c_p / c_v), and P is the pressure, ρ the density, R the universal gas constant, T the absolute temperature and M the molecular weight. From the first part of expression (2.2) it is possible to calculate an approximate numeric value for the velocity of sound in air. At normal atmospheric pressure and a temperature of 0°C (ρ depends on the temperature) the velocity is about 331 m/s.

From the second part of expression (2.2) it is obvious that the velocity is proportional to the square root of the absolute temperature

$$c \sim \sqrt{T} \tag{2.3}$$

From this correspondence we get an expression that makes it possible to compensate for a temperature deviation when the velocity is known at a specific temperature T_0,

$$c_1 = c_0 \cdot \sqrt{1 + \frac{T_1 - T_0}{T_0}} \tag{2.4}$$

where c_0 and c_1 are the velocities at T_0 and T_1 respectively. From (2.4) it is found that the sensitivity at 0°C is about 0.6 m/s / °C .When the speed of sound in air is to be measured there is another problem than the temperature. Air is actually a mixture of gases. The primary components are, of course, oxygen and nitrogen. In practice the air also consists of various concentrations of water vapour and carbon dioxide. These concentrations affect the velocity. The influence of this is, however, usually not large compared to the influence of temperature changes. Nevertheless, it is important to keep these effects in mind when measurements are made in environments where large concentrations of other gases are present. More about the speed of sound in air can be found in [Wong, 1986].

Wave Models

In the previous section it was stated that sound propagates in a gas as a longitudinal wave. However, a sound-wave also has to be defined by its shape (wave-front propagation). There are two types of waves that are easy to handle and therefore often referred to. The first one is the plane parallel wave, Figure 2.1. In this case the wavefront moves like a geometric plane in a direction perpendicular to the plane. The pressure in all points in an arbitrary plane parallel with, but behind, this one is uniform.

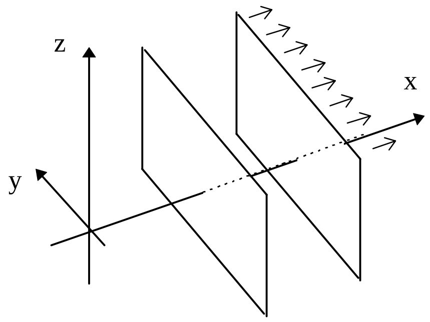

Figure 2.1 Plane parallel wave model

The second one is the spherical wave, Figure 2.2. In this case the origin of the wave is assumed to be a point source transmitting in all directions. The pressure in an arbitrary sphere is of course uniform but there is a major difference compared to the plane wave. The pressure amplitude decreases with the distance to the source.

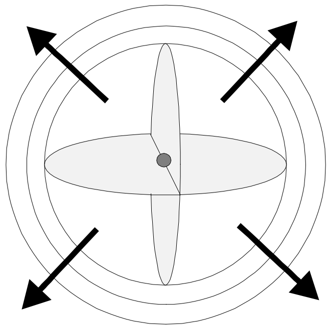

Figure 2.2 Spherical wave from a point source

Usually neither of these two models is appropriate for practical ultrasonic sources. A transmitter can usually be replaced with a number of point sources if the wave is studied from within a bounded area. Another common way of finding an equivalent source is to use the model of a plane parallel wave, with the wavelength λ, that passes through a circular opening with the diameter D in an infinite baffle, called the piston model. This diffraction results in a sound lobe in the far field, the Fraunhofer region. The intensity in this lobe reaches it first minimum, according to [Nordling and Österman, 1980] , at the angle θ_0 where

$$\sin(\theta_0) = 1.22 \cdot \frac{\lambda}{D} \tag{2.5}$$

This is a very useful model for many practical transducers. Such a transducer with the near zone (Fresnel region) and the far zone (Fraunhofer region) is shown in Figure 2.3. A further discussion of this type of model is found in [Barshan and Kuc, 1990].

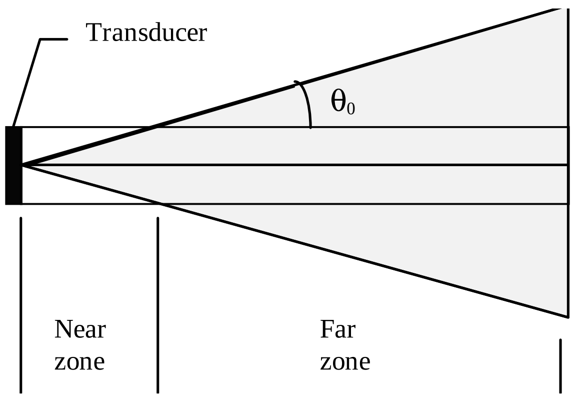

Figure 2.3 The piston model

Frequencies
There is no absolute upper limit defined for the frequency of ultrasound. Two practical reasons, however, limit the frequency range that can be used for measurements in air.

Primarily the attenuation per meter is higher at higher frequencies. This is no major problem for frequencies below about 100 kHz. As a rule of thumb the amplitude is decreased to 50 % after about 2 meters at 100 kHz [Lindström et al 1982]. This assumes a plane wave. For frequencies in the MHz-region, however, the attenuation problem becomes highly significant.

Secondarily the lobe is much narrower at higher frequencies according to (2.5), since the wavelength is decreased. The effective diameter of transducers for higher frequencies is generally often smaller than for low frequencies. This makes the lobe beam-like. As a consequence only one point of a reflecting object is ”illuminated” by the beam. Following the law of reflection the angle of reflection is equal to the angle of incidence and an echo from the object is only produced in one direction.

The frequency that is chosen for a measurement has a large influence of the accuracy that can be achieved. Since the wavelength in air at 20 kHz is about 17 mm a higher frequency might be preferred for precision measurements. It should be noted, however, that the resolution for distance measurement using the pulse echo method is not limited to the magnitude of the wavelength. A sophisticated method for finding the beginning of the echo-pulse may give a resolution and an accuracy within fractions of a wavelength.

It is very difficult to make categorical statements about the amount of noise and disturbances at different frequencies since various sources may produce undesired signals at quite different frequency ranges. A measurement in every special application may therefore be needed. In general the amount of disturbances and noise is lower at higher frequencies, due to the higher damping for higher frequencies.

Attenuation

There are two major reasons for wave amplitude attenuation in air. The dominating one is usually diffraction while the other one, absorption, gives a smaller contribution.

Diffraction: In most practical situations we deal with waves that propagate in a lobe with a significant angle. This also means that the area covered by the wave-front increases with the distance from the source. Consequently, the pressure amplitude decreases.

Absorption: The wave propagation in a gas means that local pressure maxima and minima have to be produced. The movement of particles requires a force and this operation must require a certain amount of energy that the wave loses. A more detailed and less intuitive discussion is found in [Blitz, 1967] . This loss is logarithmic and can be expressed on the form,

$$P_x = P_0 \cdot e^{-\mu \cdot x} \qquad (2.6)$$

where P_0 is the pressure amplitude at the distance x=0 and P_x the pressure amplitude at the distance x, and μ an absorption coefficient.

As indicated earlier he absorption coefficient depends on the frequency. A higher frequency implies a higher attenuation. The amount of water vapour in the air has a large impact on the absorption coefficient. Further details can be found in [Lindström et al, 1982].

Impedance and Reflection

The acoustic characteristic impedance R_a for a medium is defined as

$$R_a = \rho \cdot c \tag{2.7}$$

This is inspired from electrical impedance in the sense that it describes the relation between the acoustic pressure and the particle velocity. This is, however, not the only correspondence we can find between electric circuits and acoustics. The signal reflection in a transmission line impedance mismatching is almost the same as the reflection of an acoustic wave meeting a medium with another acoustic impedance.

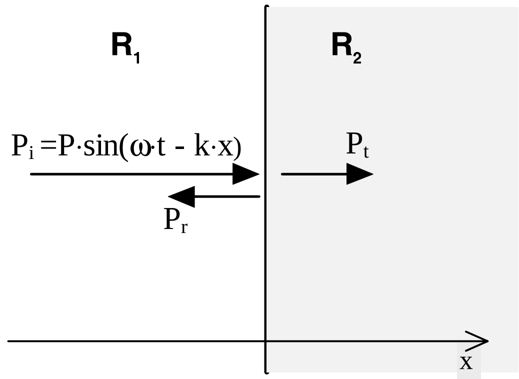

Figure 2.4 Reflection at a plane boundary between two media.

In Figure 2.4 a parallel plane acoustic wave with pressure amplitude P_i propagates in the normal direction to a boundary. When the wave meets the boundary it is divided into one reflected (P_r) and one transmitted wave (P_t).

Phase: If the acoustic impedance R_2 is larger than R_1, (like waves in air reflected at solid objects), then the reflected wave, P_r, has the same phase angle as the incoming, P_i . This means that the reflected wave can be considered as a continuation of the incoming one but with the opposite direction of propagation. Contrary, if $R_1 > R_2$ the reflected wave, P_r is phase shifted 180° compared to P_i. The transmitted wave, P_t is always a continuation of the incoming one, P_i.

Amplitude: The pressure amplitude for the reflected and transmitted waves can be calculated using the coefficients α_r and α_t . They divide the incoming wave in two parts under the assumption that the pressure on both sides of the bounding should be equal. The particle velocities on both sides should also be equal to maintain the media contact. Consequently will the values of the coefficients are [Blitz, 1967],

$$\alpha_r = \left(\frac{R_2 - R_1}{R_1 + R_2} \right)^2 \tag{2.8}$$

$$\alpha_t = \frac{4 \cdot R_1 \cdot R_2}{\left(R_1 + R_2\right)^2} \tag{2.9}$$

If the incident wave meets the boundary from an angle other than the normal direction the resulting waves become different regarding both directions and amplitudes. This is also described in [Blitz, 1967].

When echo measurements are made in air the difference between the impedance of the air and the reflecting objects is large. It is often a reasonable approximation to consider that only the reflected wave is produced, so the surface that the wave meets may be considered as a mirror. This means that the there is no loss in the reflection and that the angle of reflection is equal to the angle of incidence. This is a reasonable approximation for plane parallel waves that meet large flat surfaces.

Reflecting Objects
When measurements are made in an imaging purpose, as described in Chapter 1, the reflection of the waves at boundaries is an important process. The reflection mechanism at ideal conditions was described in the previous section, but in practice the situation is generally much more complicated. The boundaries belong to what we usually call reflecting objects. These objects may have many reflecting surfaces with arbitrary size, orientation and position. Consequently, the description model for these objects is a key problem in the imaging measurement.

Ultrasonic measurements can be made using either a continuously transmitted signal or a pulsed signal. The *continuous method* is primarily used for interferometric measurements. For imaging measurements the method is less suitable but may be used if the causality problem is solved. This means that the correspondence between a transmitted wave and the received echo must be clear. This can be done if for example the transmitted signal is modulated and a correlation algorithm provides the distance information in the echo. Otherwise, if a simple periodic signal is transmitted the interpretation of the echo soon becomes ambiguous.

The *pulse-echo* measurement has the advantage of providing a finite amount of data. The causality problem is much simpler since a distinct wavefront propagates and thus produces new echoes. However, the length of the transmitted pulse is crucial. Very often echoes from different surfaces interfere. Then it is not obvious to tell by looking at the echo which surfaces that produced it.

A pulse-echo measurement in practice is often performed in the following steps:

- an ultrasonic signal is sent from a transducer with a specific lobe;
- it is reflected at an object (or several objects) with specific geometry ;
- it is received in a point (not necessarily the same as the transmission point).

It is easily understood from the previous sections that the signal that reaches the receiver soon becomes a complicated product of several components. This is actually the main reason why the interpretation of ultrasonic echoes is difficult.

It was stated in Chapter 1 that the purpose of the imaging defines what approach should be made at the interpretation of the ultrasonic echo. In some applications a comparison of two objects is quite sufficient. Another application may require information about what surfaces produced the echo. This information may include distance, direction as well as the size or the surface. Many interpretation methods exist but below follows three important ones.

1. The radar (Synthetic Aperture Radar) method. An approach inspired by radar techniques is to consider the reflecting object as a set of point scatterers. A point scatterer reflects an incoming wave in all directions without any effect on the pulse-shape. The object echoes are recorded from several points around the object (SAR) or echoes are recorded from one point while the object is slightly rotated between the echoes (ISAR). A grid of possible point-scatterers is then set up and through correlation methods the points in this grid can be given values that correspond to the amount of reflections from them. More about these methods can be found in [Jonsson, 1985] and [Mensa, 1991].

2. Pattern recognition method. Another approach is to perform some kind of pattern recognition on the incoming echo. If many circumstances around the measurements are fixed and the main purpose is to distinguish a limited number of objects from each other this may be sufficient.

3. Transfer function method. If the resulting echo is assumed to be the product of a number of transfer functions operating on the initially transmitted signal it is possible to identify an impulse response for the reflection at the object. We still have many of the problems that were present in the pattern recognition case. These include:

- from what angle is the object illuminated by the transmitter,
- from what angle is the object observed by the receiver,
- what is the orientation of the object and
- what effect does the distance to the object have.

On the other hand the varying part is isolated and thus easier to handle minor variations in. More about these last two methods is found in chapter 6.

The main conclusion concerning the description and model of the reflecting objects is that this is the key problem when measuring with ultrasound.

2.2 Transducer Elements

There are many physical methods to generate and detect ultrasonic signals. Several of the principles that are used in the audible frequency range can't be used for ultrasonics. On the other hand some new possibilities become apparent. A physical approach to several principles is found in [Blitz, 1967]. Several of the methods are most suitable for ultrasonic transmitting and receiving in solid materials or liquids and practically unfeasible in air or other gases. The reason for these compatibility problems is often differences in specific acoustic impedance. Since air and most gases have got a low acoustic impedance the signal propagation between air and a solid material, or vice versa, causes large losses in signal amplitude as described in the previous sections. The two most important transducer principles for ultrasound in air are the electrostatic and the piezoelectric.

The electrostatic transducer is a popular sensor family, developed mainly by Polaroid® during the recent decades. Consequently it is often referred to as the Polaroid transducer. It was mainly meant to be used in the distance measuring system in cameras. The basic principle is much older than the Polaroid system and has been used in other applications previously. Nevertheless, the transducer made by Polaroid is a well designed and successful commercial product that has been important for the industrial use of range measurement based on ultrasonics. For further information on the Polaroid sensors see the product information from Polaroid in the reference list.

The basic design of the electrostatic transducer can be compared to a capacitor. One of the poles consists of a thin (low weight) membrane coated with a conducting material. The other pole, made of conducting material, is usually grooved and mounted in parallel with the membrane and as close to it as possible without galvanic contact with the coating. Key parameters for the capacitor formed in this way is the dielectric constant and thickness of the foil. A force can be attached to the membrane by connecting a voltage source between the poles. This force varies with the voltage and an ultrasonic transmitter is obtained. The voltage required is usually rather high, a few hundred volts. To make it possible to transmit normal linear bipolar signals a DC bias is required.

The same design can also be used to receive ultrasonic signals. With a high AC impedance bias source the transducer works as a charged capacitor. The capacitance, C, is proportional to the inverse of the distance between the charged surfaces. When a sound pressure is applied to the membrane this distance varies. The equation

$$Q = C \cdot U \tag{2.10}$$

where Q is the charge in the capacitor and U is the voltage over it must still be satisfied. Consequently the voltage over the transducer varies with the sound pressure since the charge, from an AC point of view, remains constant. Consequently the same element can be used as both receiver and transmitter.

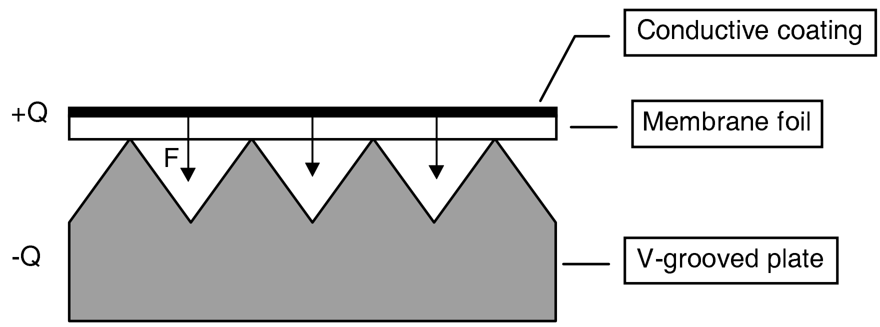

Figure 2.5 Functional principle for the electrostatic transducer

In the Polaroid design the membrane is a special foil consisting of a Kapton® film coated with gold. The foil is stretched over a V-grooved plate, Figure 2.5. The transducer is protected by a stainless steel housing. A few models with different housings are available. The sizes of these transducers are the main drawback since they are from 30 up to 45 mm in diameter. A further analysis of this structure and its dynamic features is found in [Mattila et. al., 1995].

The piezoelectric transducer is based on a piezoelectric crystal that either is forced into mechanic oscillation by an external voltage or generates a voltage when exposed to mechanic oscillation. Consequently both sending and receiving functions can, also here, be performed by the same element. One major difference between the electrostatic and the piezoelectric elements is that the oscillation in the latter one is initiated in a solid material. This means that the acoustic impedance difference between the crystal and the surrounding material determines how much of the wave energy that will propagate out of the crystal. A large difference in impedance, as in the case of surrounding air, implies that only a small part of the energy will leave the crystal. This problem has to be solved to produce a transducer system that can be used in air. A common way of solving this is by using a quarter-wavelength resonance, as described in [Blitz, 1967].

As a consequence of the physical design of a piezoelectric element the coupling between the voltage over the device and the ultrasonic pressure-wave in the air is very complex. The crystal itself and the acoustic impedance adaptation are resonant systems that the signal has to pass on its way. This causes several problems in most measurement applications.

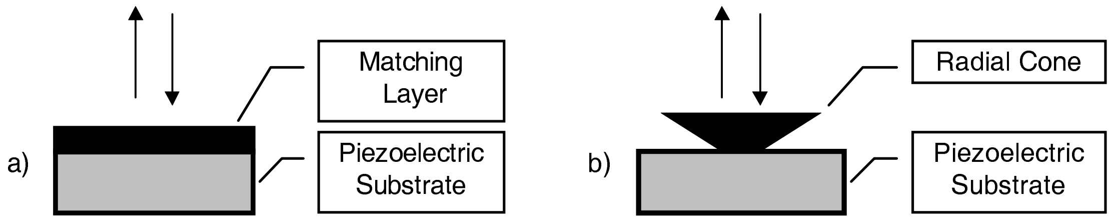

Figure 2.6 Piezoelectric transducer with ***a)*** *matching $\lambda/4$ -layer and* ***b)*** *horn like design*

Several manufacturers produce piezoelectric transducers. Among them Matsushita- and Murata- elements are often found in low cost applications. Their most common transducer is a 40 kHz one, based on amplification and impedance matching with a mechanical horn-like design. It is capsuled in an open housing that allows air flow through one side. Water proof transducers are also available (Murata) but are usually somewhat less sensitive. Elements for precision measurements are often based on thickness expansion mode of the piezoelectric material in combination with impedance matching layers. This design can easily be hermetically sealed but is primarily used for higher ultrasonic frequencies, larger than 100 kHz. In the work [Freire Bastos et al, 1994] the internal design of the transducers is shown more in detail together with a temperature test of a few different transducer types.

The features of these two groups of elements can be summarized and compared as shown in Table 2.1. (Advantage + /Disadvantage -)

	Piezoelectric	Electrostatic
Size	ϕ < 20 mm	ϕ > 30 mm
Frequency Range	40kHz - 2 MHz	40 kHz - 100 kHz
Bandwidth (per element)	--	++
Impulse response damping	--	+
Sensitivity	+	+
Noise immunity	--	+
Simple interface	+	-
Low price	++	+

Table 2.1 Ultrasonic transducer type comparison

This comparison does not tell the whole truth but emphasizes some of the differences. These feature differences might need some comments and explanations.
The *sizes* of the piezo-electric elements become much smaller for higher frequency designs (≥100kHz). The electrostatic elements have to be rather large not to lose sensitivity.

- The *frequency range* for the piezoelectric elements can be extended up to a few MHz with high precision crystal processing and assembly. The electrostatic elements are difficult to use at these high frequencies because of the amplitude decrease. Please note that frequency range is interpreted as: the range in which elements can be manufactured. Not as the range for one element.
- The *bandwidth* of the piezo-electric element is determined by its physical design as a resonator. It has got a high sensitivity for signals close to the resonance frequency but has got a small bandwidth. The electrostatic element, on the other hand, has got a wide bandwidth. If the definition is extended a little to allow amplitude drops just below 3 dB the entire frequency range can be covered with a single element.
- The big problem with the piezoelectric elements is the *impulse response damping*. The signal is highly affected by the element dynamics. An impulse makes the transmitting element oscillate for several periods. The same thing happens with the receiver. This complicates the signal analysis because all echoes have got a long tail.
- The *sensitivity* for a signal of the resonance frequency is somewhat higher for the piezoelectric element than for the electrostatic one. However, this is no major advantage since the piezoelectric crystal starts oscillating at this frequency due to many disturbances sources. This implies a low *noise immunity*.
- To design the *interface* circuits for the electrostatic element is a cumbersome task. The high voltage bias eliminates the direct use of standard components like OP-amps. To handle the combination of high voltage and high bandwidth can be a quite difficult task.
- Finally the *prices* of the sensor elements are quite equal. Real low cost versions of the piezoelectric elements are available. These ones costs less than $10 in small quantities while electrostatic ones costs about $40. High quality piezoelectric elements or elements for higher frequencies are usually somewhat more expensive and cost up to $100. However, ultrasonic sensors are still comparatively inexpensive sensors.

The different shapes of the ultrasonic echoes using theses two type of transducers are illustrated in Figures 2.7 and 2.8.

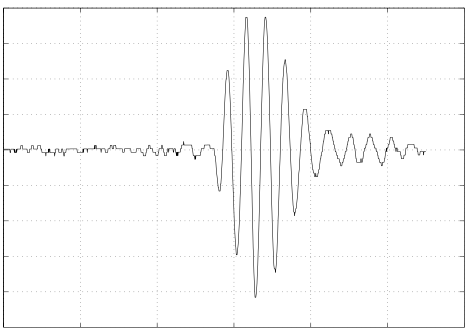

Figure 2.7 Ultrasonic echo from a flat surface using an electrostatic transducer.

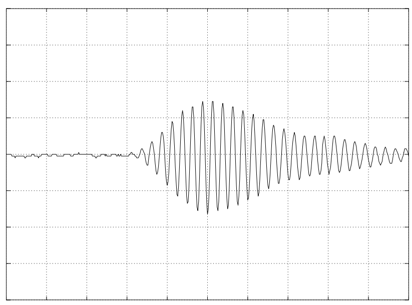

Figure 2.8 Ultrasonic echo from a flat surface using a piezoelectric transducer.

2.3 Ultrasound Phenomena

After experimenting with ultrasonic signals for a while some phenomena, that are not obvious for the novice, becomes apparent. These phenomena can, of course, be explained with the physical theory but they often generate much confusion the first time they are encountered.

Fatal Interference

The radar-like behavior of ultrasonic pulse-echo systems makes it easy to think about the transmitted pulse as a dirac or at least as an oscillation with an extremely short wavelength compared to the measurement object size. This is often not true since several periods of a limited frequency have to be transmitted to receive an echo with sufficient amplitude. The wavelength of a 60 kHz oscillation in air is about 6 mm. This makes it necessary to analyze what can happen if interference within one period of the oscillation occurs.

Let's try to design a wave-trap that as soon as possible eliminates the returned echo. Figure 2.9 shows a reflecting surface that is absolutely flat and below that a surface with a profile step of a quarter of a wavelength ($\lambda/4$). The latter configuration makes the returned echo from the more distant surface part to be delayed half a wavelength ($\lambda/2$). A pulse has to travel both towards the surface and from it before it interferes. If the surfaces are orthogonal to the transmission axis and the surface step intersects the axis a symmetrical situation is created where the sum of the reflected signals is theoretically zero. This happens because two equally strong signals with opposite phase are summed. For causality reasons the first half period of the echo is not affected but this is usually a part of an echo sequence that has a low amplitude.

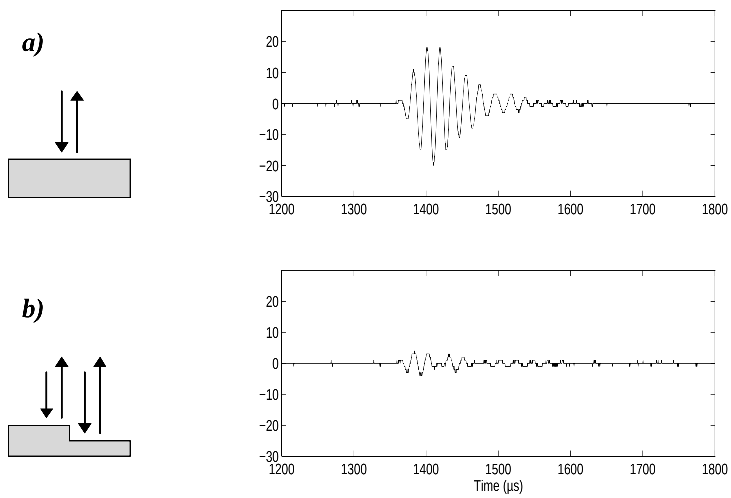

Figure 2.9 The interference problem. The transmitted signal is 4 periods at 60 kHz in both figures. In figure a) the signal is reflected at a flat surface. In b) half of the reflecting surface is about 1.5 mm (≈ $\lambda/4$) further away from the transmitter.

This is of course a specially designed example of how this interference can reduce the echo amplitude drastically. However, the problem occurs frequently even if the phase shift is not always $(2n+1)\cdot\pi$ radians and the echo signal is completely wiped out. If the phenomena is kept in mind it may be easier to understand why unexpected reduction of amplitude sometimes occurs in echoes.

Lobe Angles and Reflections

A commonly used approximation of a practical transmitter lobe is that the lobe consists of a set of beams with an amplitude distribution that corresponds to the deviation angle from the center axis. Using this model it is possible to define a subset of beams that will illuminate an object. We then know what sound pressure is put on the different parts of the object, the phase of the wave at different locations and the direction for the incoming wave. With this set of information at hand it seems to be a simple task to compute the resulting echo field. This is however, not the case.

It was earlier stated that the laws of specular reflection were applicable on the reflection of plane parallel ultrasonic waves that meet flat surfaces. It would then be tempting to assume that these laws also could be applied to our beams that meets the object. Unfortunately, it is not that simple. The primary reason for this is that the object surfaces often are so small that the Fraunhofer diffraction, Equation (2.5), is considerable. The current case is a special variant of the experimental conditions that is used to explain this diffraction. Because of this the main direction of the echo is defined by the laws of specular reflection – but the Fraunhofer diffraction defines the lobe of the reflection. The reflecting surface that is treated may then have a contour that is not flat and consequently, the computation of the resulting echo usually is a very complicated task.

An important conclusion is that the amplitude of an echo in directions other than the one that follows specular reflection is highly dependent on the shape of the reflecting surface.

Corners and Edges

An edge can usually be approximated with a line of point scatterers. This means that the echo from an edge is transmitted in a wide lobe. Consequently, edges and corners of objects are good scatterers that generate echoes even if the surfaces of the object lead the echo waves in directions where there is no receiver. Usually the problem is that the amplitude of the echo from the corner is small. A sharp edge produces a negligible echo while a rounded edge is easier to detect. The scattering from an edge is especially important when ultrasonic waves propagate through tubes or other bounded spaces. In these spaces lateral waves are usually generated. When these waves meet the opening edges the edges scatter the waves. A lobe that is much wider than the one obtained from the Fraunhofer diffraction may then be the result.

Transmitted versus Received Signal Amplitude

The main reason for signal amplitude decrease is usually diffraction. This attenuation is depending on the distance. Thus, in a pulse echo system, signals close to the transmitter have a much higher magnitude than echoes from distant objects. The ratio may very well be of the order 100 to 1. Precaution must therefore be taken to avoid paths that may lead the very strong transmitted waves directly into the receiver if the receiver is sensitive for saturation. For instance this is the case when using many of the piezoelectric receivers. If such a receiver is exposed to a high amplitude signal the crystal may continue to oscillate for several milliseconds. Measurements during this interval will then be impossible.

Measurement System Design 3

This chapter describes the considerations for and the design of a system for ultrasonic measurements. The development of an experimental platform for sensor units equipped with both electrostatic and/or piezoelectric transducers is described. Focus is put on interesting details like the sensor unit transducer configuration, transducer interface and the data acquisition system. Possible designs to improve the real time performance of the system is also discussed at the end of the chapter.

3.1 Design Specifications

The goal for the measurement system is to form a flexible platform for miscellaneous experiments using ultrasonic sensors. On this platform configurations and solutions applicable to industrial systems like industrial robots and NC-machines have to be possible to test and evaluate. Important parameters in these industrially applicable systems are the size of the measurement unit together with the number and placement of the sensors. The latter is extremely significant since it defines what measurements can be made with the measurement unit.

The system has to allow advanced signal processing by a computer system software so that algorithms can be implemented and tested easily. Since both electrostatic and piezoelectric transducers, in various configurations are going to be used the system has to be built on a modular basis. Details concerning the modules, like electrical drawings, are found in the design catalogue [Lindstedt, 1995]. This chapter primarily describes the design mainlines. When the electrical circuits illustrate the basic ideas of the design they are included here.

A coarse division of the measurement system can be made in the parts:

- computerbased hardware functions,
- software routines and
- application specific hardware.

The first part serves as a bridge between the host computer system and the second part. An overview is shown in Figure 3.1.

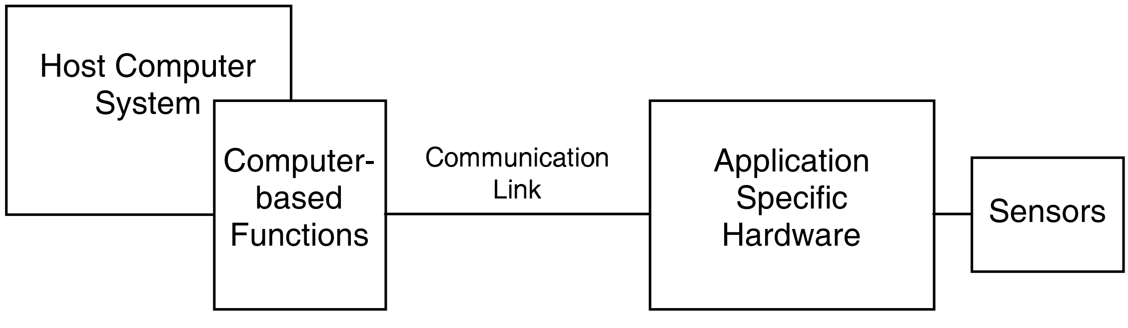

Figure 3.1 Coarse overview of the measurement system

Sensors

The system has to be able to handle both electrostatic and piezoelectric sensors in various configurations. For practical reasons solutions where all sensors are contained in one sensor unit are preferred. The industrial situations very often limits the space where measurement devices can be positioned. Consequently the unit has to be small and the practical disadvantage of using several units in the measurement scenario is unacceptable. The available space in for example a robot application where the sensor unit is placed close to the end-effector of the robot is approximately somewhere about 1 dm^3. This is clearly a less favorable situation than those presented in the descriptive measurements of [Knoll, 1991], [Watanabe, Yoneyama, 1992] or the SAR (Synthetic aperture radar) methods described by [Mensa, 1991]. The approach to the sensor configuration problem thus is:

- one unit,
- moderate size,
- extract as much information as possible from the measurements.

This will give a system that is not optimal for the information acquisition but will form a reasonable approach towards a general purpose measurement system that can find a wide variety of applications in industrial processes.

Possible Measurements

For a multi purpose measurement system of the current type some clearly definable measurement tasks can be identified. The ***distance measurement*** is obvious but if more than one frequency can be used the resolution and accuracy can be improved. However, the sensor configuration doesn't affect these measurements to any large extent. On the other hand the distance measurement can be used for obstacle detection. In this case distance measurements from more than one point can provide direction information and thus make an ***object localization***. In the three-dimensional case transmitter/receiver pairs should be placed along two independent lines to provide this information. Figure 3.2 shows the two-dimensional scenario.

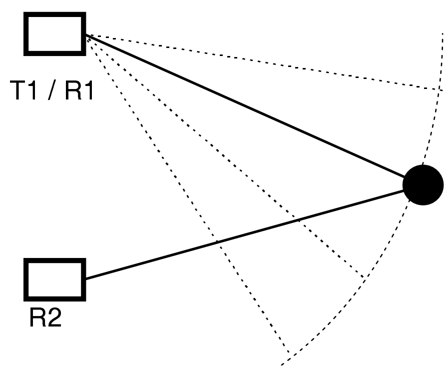

Figure 3.2 Localization in two dimensions using a combined transmitter/receiver (T1/R1) and an extra receiver (R2).

It should be noted that the reflecting object has to be comparably small if this method is to be used. Lobe angles and shape of the reflecting surface also limits the use of this type of localization. Still it can be proven useful in many situations. More about this method follows in Chapter 5.

Another method that can be useful is the ***angular measurement*** as shown in Figure 3.3. The difference between the echo delay T1-R1 and T1-R2 is recorded. The shortest paths between the transducers and the surface is thus measured. If the surface is large and flat the inclination can be computed with a high resolution. The use of one more sensor implies that this measurement can be done in three dimensions (two inclination angles) in a manner similar to the localization. More about these measurements is found in Chapter 5.

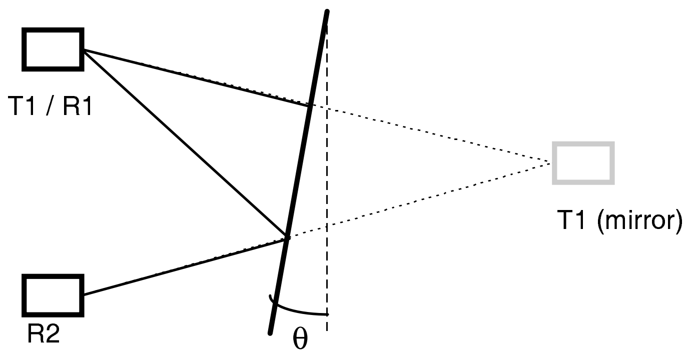

Figure 3.3 Measurement of the inclination of a flat surface.

When the measurements concern ***recognition and object identification*** the requirements for the sensor placement are not that clear. To compensate for object rotations and other operations it is desirable to make measurements in as many directions as possible. However, the physical limitations for a unit of the current type reduces the possibilities to do this. Instead effort has to be put in the area of signal processing on the available signals.

Application Specific Hardware

This part of the design has to be flexible and easy to adapt to new transducer configurations or other modifications of the measurement system. Because of this it should be built on a modular basis. A couple of common functions can easily be identified but their implementation may vary between different kind of sensors, sensor configurations or purposes of the measurements.

The ***transmission pulse generations*** for the electrostatic and the piezoelectric elements are significantly different. The electrostatic element is designed to work at a bias-level of about 150 V and the modulation range is ±150 V around this. This element type can produce signals in a wide frequency spectrum and consequently many modulation types can be used. To utilize this freedom the system should allow arbitrary design of the transmitted signal shape for the electrostatic transducers. The piezoelectric elements are resonant with a high Q-value and can’t be modulated to any large extent. The problem that has to be solved for these elements is the energy transfer to start the oscillation in the transducer and if possible force the oscillation to stop after the intended transmission. It should of course be possible to control the number of periods in the transmitted signal, and this can be done. This makes it possible to adapt the signal to the distance of the measured objects since several periods let the piezo-material oscillate at full amplitude thus providing a strong transmitted signal. Short distance measurements, on the other hand, should use few periods since this gives more precise echo-delays from multiple objects, in other words more like an impulse response.

Most transducer elements can be used as both receiver and transmitter. This is an advantage since the physical dimensions of the measurement unit are critical. Furthermore, the use of an arbitrary element as the transmitting one means that the ”illumination” of the measurement area can be done from several different angles. This requires a ***transmitter/receiver selection*** function to be included. This function is not critical in a microsecond perspective but a switch between transmitter/receiver configurations should be able to take place between two measurements.

The data acquisition in the system requires special attention. It is important for the following signal processing that the signals are sampled simultaneously in all channels at periodic and precise time instants. This should be done by a unit for ***sampling and data storage.*** The sampling rate has to be sufficiently high to give a good representation of the highest frequency used. It should also be adjustable so that when measuring at lower frequencies the amount of data produced does not become unnecessarily large. Since the highest usable frequency for the current measurements is somewhere around 200 kHz and it is reasonable to sample about 10 times per period an appropriate sampling frequency would be in the region of 2 MHz. Furthermore, the collected data must be stored in an intermediate storage buffer since a sampling frequency of this magnitude requires a unit that works autonomously to the computer system. The size of this buffer must allow storage of the longest echo sequence that has to be analyzed. A reasonable value is to store about 1 meter at maximum sampling rate.

Computer-based Functions

The specifications for the computer-based functions can be approached in two ways. Either the demands from the measurement side defines a set of suitable computer system platforms or a specific computer that for some reason has to be used specifies the performance that is possible to implement. The areas where limitations may be set up by the computer platforms primarily concerns data throughput and possibilities for front-end processing. Ultrasonic signal measurements that are made for feedback purposes require a high data throughput if the signal processing will be done by the computer software. Datablocks of 50-100 kBytes is produced in each measurement and the physical limit for the repetition rate at reasonable measurement distances can be as high as 100 Hz. Consequently, data may be produced at a rate of 10 Mbytes/s. This throughput can be difficult to achieve on many computer platforms. However, fortunately this rate is not common in practical measurements since other limitations than the pure physical time-of-flight[1] limit are present. If the entire dataset has to be processed the time for the processing is much longer than the previously indicated rates allow. Usually the interesting part of the echo can be localized and only a subset of the data needs to be studied carefully. This is done by reducing the sampling rate and then selecting a time window.

A conclusion of this analysis is that a high throughput-rate is required. No absolute requirement can be set up since the rate depends on the used data processing algorithm. The upper limit of 10 Mbytes/s doesn’t have to be achieved but the rate should at least be about 1 MBytes/s.

[1] The time it takes for a transmitted pulse to travel from the transmitter to the reflecting object and the back to the receiver is here called ”time-of-flight”.

3.2 A Feasible Approach to the Sensor Configuration Problem

The layout and design of the sensor unit is, as earlier stated, a key problem since it means a compromise between the practically acceptable dimensions and the possibilities to get sufficient information from the measurements.

Sensor Layout
A sensor unit that will be possible to place on a robot arm must have a highly limited size. The same is true in other situations where the executive process is flexible and parts or tools are swept through a large space. To make the intended measurements we need a matrix-configuration of sensors that must occupy a certain area. A reasonable assumption for the size of the available area is that it ought to be somewhere around 1 dm^2. This leaves us with the question how to place the transducers in the area to make the best use of them.

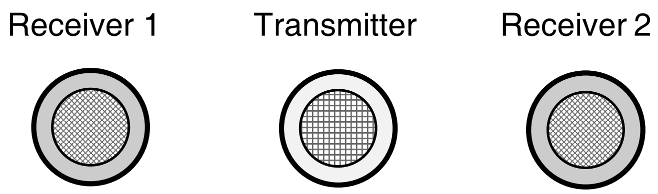

Figure 3.4 Symmetric transducer placement for two-dimensional measurements.

Most measurements reported in the literature, e.g. [Kuc and Barshan, 1992] and [Klöör et al, 1988] , that deal with the problems of angular measurements or object localization assumes the configuration indicated in Figure 3.4. The symmetric design makes the calculations easy. Furthermore, the symmetrical lobes of the transducers make the amplitudes equal when the signal is reflected at a surface parallel with the sensor plane. This is in other words a nice and preferable configuration.

In our case with an extremely limited area for the transducer placement the symmetrical model is unacceptable. Each piezoelectric transducer occupies about 20 mm for the mounting and each electrostatic about 35 mm. Piezoelectric receivers are also sensitive to the direct wave from the transmitter since the damping of this high amplitude wave consumes considerable time. The only cure for this is acoustical shielding *and* (not *or*) increased distance between the elements. In both the piezoelectric and the electrostatic cases a configuration of three cooperating elements in a row is too large to fit into the sensor unit.

An important fact is that most transducer elements can act as both transmitter and receiver. This double function can normally be used within the same pulse-echo cycle, with the exception of low frequency piezoelectric elements. These elements continues to oscillate after the transmission so that incoming echoes can not be detected for a long period of time. In all other cases the double function can be used.

To avoid placing transducers close to each other a configuration where the elements are placed close to the outer contour of the measurement unit seems favorable. If a rectangular shape is assumed the elements should be placed in the corners. To maintain a symmetry the shape may also be assumed quadratic. The transducer configuration using this principle is shown in Figure 3.5.

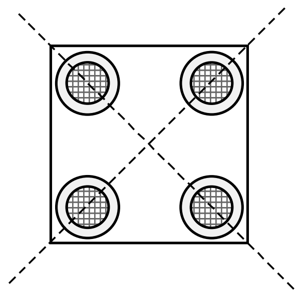

Figure 3.5 Minimal transducer configuration in a minimized area

In the following chapters it will be shown how this geometry can be used for three-dimensional (spatial) angle measurements and object localization. Two-dimensional angle measurements (referring to Figure 3.4) may also be made along the axes indicated in Figure 3.5. In this case the transmitter has a slight displacement but the conditions are symmetrical in the measurement direction. An advantage of the sensor layout described above is that redundant information can be obtained when measuring spatial angles through alternation of the transmitting element (four possibilities). There is also a long distance between the elements (higher angular accuracy) while the overall dimensions are kept small.

Using the described placement principle two sensor units have been designed. One unit for piezoelectric transducers shown in Figure 3.6 and one unit for electrostatic transducers shown in Figure 3.7.

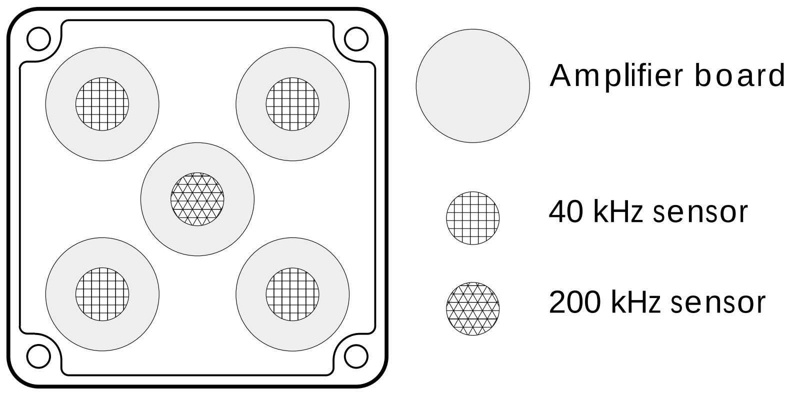

Figure 3.6 The piezoelectric sensor unit. The dimensions of the unit are 100x100x90 mm.

Piezoelectric Unit
The sensor unit contains four 40 kHz transducers symmetrically mounted around one 200 kHz transducer. Any of the four 40 kHz transducers can be chosen for transmitting the signal, while the other three are used as receivers. This flexibility allows a very versatile measurement system. The 200 kHz transducer is used both as a transmitter and a receiver.

The 40 kHz elements are used for the distance and angle measurements at long distances while the 200 kHz element is used for more accurate distance measurements close to an object. Adding the short distance sensor, 200 kHz, improves the performance of the unit. It is also placed where it does not increase the overall sensor unit dimensions. Furthermore it is well centered in the unit which is suitable for measurements at short range.

The 40 kHz transducers in this final configuration are manufactured by Murata (type MA40A5) while the 200 kHz transducer is designed by Hans Persson at the Department of Electrical Measurements, Lund Institute of Technology, Sweden. This is a thickness expansion mode piezoelectric transducer that has got several advantages compared to the ones commercially available. For example it has a wider lobe and a higher sensitivity. In spite of these excellent features the cost in an industrial application would probably be quite reasonable.

Since the signals from the sensors have got a very low amplitude they have to be amplified about 1000-5000 times. Therefore it is essential, that some of the signal processing is performed as close as possible to the primary sensor elements. The original sensor signal is easily disturbed and must be amplified before it proceeds through any long cable. Because of this, preamplifiers are mounted directly on the sensor elements. These amplifiers have to be small which is achieved by using SMD (surface mount device) techniques for the electronics. The amplification is about 1000 times. By using third order low pass filters as part of the amplifiers, the noise components above the sensor frequency are highly reduced.

By using SMD technique the relay multiplexer can also be included in the sensor unit. Since the wire carrying the activating pulse is a large disturbance source it has to be separately shielded. Having the relay multiplexer in the sensor unit makes it possible to use only two cables to connect the unit to the other equipment. One cable contains the activating pulse while the other contains both the power supply, the control signals and the measured signals.

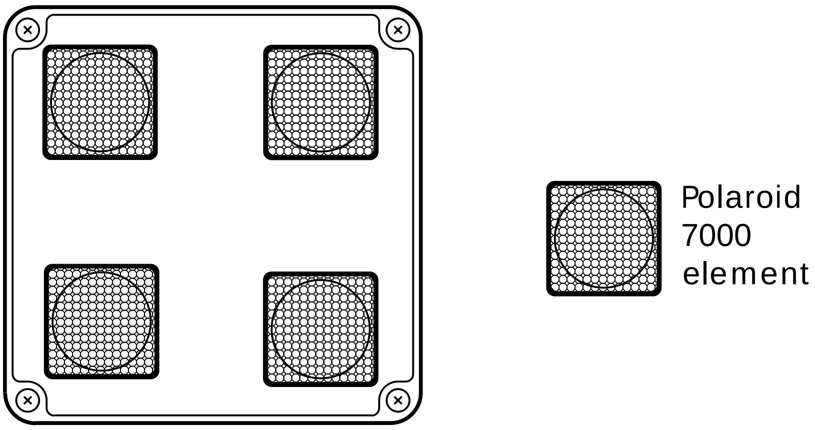

Figure 3.7 *The electrostatic sensor unit. The dimensions of the unit are 110x110x60 mm*

The SMD techniques and the localization of the pre-amplifiers has solved many problems, that were significant in previous versions of the sensor. For example, the noise introduced in the electrical connections created a lot of problems, and the solutions to this were much more cumbersome than in the final design.

Electrostatic Unit

Based on the experience from the piezoelectric version a unit equipped with electrostatic transducers has been built. The Polaroid 7000 element has been used in the four transducer positions. This element has a bandwidth from 50-60 kHz (3dB limit) but can handle signals in the range 40-70 kHz with an amplitude that is sufficiently large to be used in most situations. The considerations and possibilities that were made and described for the four piezoelectric elements in the previous section are valid also for this configuration. The flexibility of selecting the transmitting element is utilized here as well.

The electrostatic elements can be placed much closer to each other without the problem of large signal crosstalk that destroyed the measurements for the piezo elements. There is a crosstalk also in this configuration. However, the signal from the transducer element returns to normal almost immediately after the sound pressure has disappeared. Consequently, the actual measurements will not be affected. Although this makes the placement of the elements more free they are placed close to the corners anyway. This is to maintain a good accuracy in angular or localization measurements.

The principle of early amplification is used also in this unit. Each element is connected to a preamplifier in the sensor unit. This amplifier also solves the bias problem. The unit is connected to a 150 V DC source. This voltage is stabilized and fed to each transducer element. The signal from each element is then filtered so that the high voltage component is eliminated from the amplified signal. A second conditioning amplifier in the unit makes the signal sufficiently strong to be transmitted to the measurement system.

Piezoelectric Sensor Mounting Precautions

It was mentioned in the design of the piezoelectric unit that special problems may occur when piezoelectric transducers are mounted too close. During the development of the unit several approaches have been made to find a proper mounting of the elements. The two major problems were:

- transmission of the sound in the mounting material;
- ultrasonic leakage through the air directly from the transmitter to the receiver.

The *first* problem is rather easy to handle since the speed of sound is lower in air than in most solid materials. Therefore by a causality consideration it is possible to discriminate all information that comes to the receiver too early. Nevertheless this is not sufficient since,

- reflections inside the mounting material can delay the pulse,
- the signal from the receiver may be affected for a long period of time due to the element dynamics mentioned in previous paragraphs, and
- the pulse may be transmitted into the air from another part of the mounting material.

These problems can be solved if a material with a high damping is used. However such materials are often very soft and can't be used for the mechanical mounting. Cork has proven to be the best compromise and so far solves the noticeable part of this problem.

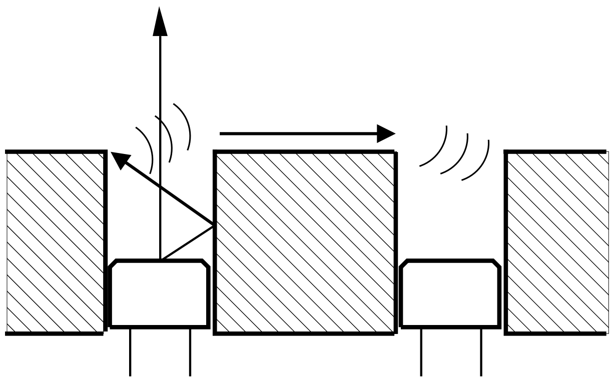

Figure 3.8 Leakage due to transversal waves

The *second* problem is much worse to solve since ultrasound with a frequency of 40 kHz seems to "leak in all possible directions". The theoretic lobe angle from a transmitter can be computed using expression (2.5) as shown in Chapter 2. However, this expression gives the angle at which the first minimum in the lobe is found. Amplitude for larger angles is usually neglected since it is rapidly decreasing in Fraunhofer diffraction. This is based upon the assumption that diffraction occurs when a wave front is sent towards a plane with a circular opening.

Since the sides of the transmitting element work as membranes the element must be buried in the mounting material, as indicated in Figure 3.8. In this case the situation becomes slightly different since the element is placed at the end of a tube ending in free air. This configuration introduces transversal waves that makes the intensity a much more complex function of the angle . One important effect is that the intensity in the wave following the mounting material surface is no more negligible.

The "leakage" problem has not been solved completely but is highly reduced with the mounting shown in Figure 3.9. The transversal waves are attenuated by the mineral wool and leakage in the gap between the cork and the wool is prevented by the plastic tube.

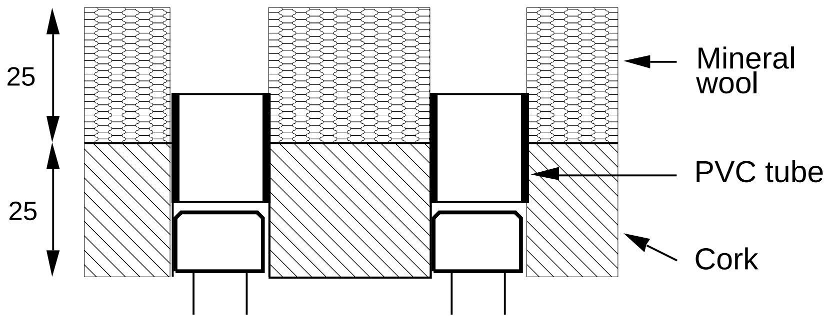

Figure 3.9. The preferred mounting principle (size indications in mm).

The thickness of the mineral wool has been optimized since a too long tube made of wool attenuates the desired longitudinal signal too much. In many applications the soft front surface is not acceptable. Because of this most experiments have been done using only the cork front.

The leakage problem can be further reduced by increasing the distance between the elements but as mentioned earlier this is not possible in the current design. Furthermore, in an angular measurement the required size of the flat reflecting area (measurement object) is increased if the distance between the sensor elements is increased.

Experiments with walls, perpendicular to the wool surface, of different materials between the elements were made but this did not reduce the leakage significantly.

3.3 Driving the Transmitting Element

To achieve good measurement conditions the control of the transmitted signal is of great importance. This concerns not only parameters like timing of the wavefront, amplitude of the signal, and transmitted frequency. Also the entire shape of the transmitted wave including tails caused by the transmitter element resonance is of interest. Some methods like matched filters are based on a detailed knowledge of the transmitted wave. Process identification methods can also take advantage of a good knowledge of this part of the system.

There is a significant difference between the control capabilities of the piezoelectric and the electrostatic elements. The electrostatic one is much simpler to control since there is a good correspondence between the transmitted sound wave and the applied. On the contrary, the piezoelectric one has a complex dynamic system that is difficult to control.

Piezoelectric Element Driver Circuit

Since the piezoelectric elements are resonant systems they need some "acceleration-time" to reach full signal amplitude and some "retardation-time" to stop oscillating. In pulse-echo systems where the measurement is based on the beginning of the echo this is a problem. If the beginning of the echo has a low amplitude it is difficult to detect. On the other hand, if the same element is to be used both as receiver and transmitter the "retardation-time" can cause problems. These problems are apparent at short distances due to short time delays between the pulse and the echo.

Piezoelectric elements are usually activated by one of the following two methods:

(1) A pseudo dirac pulse, as the one shown in Figure 3.10, is used to activate a piezoelectric transducer with the highest possible energy in the beginning of the transmitted ultrasonic pulse. This pseudo dirac consists of a high voltage pulse with a short duration. The voltage peak is usually above 100 V and the duration significantly less than one half period of the resonance frequency. The main drawback of this method is that the maximum amplitude for the transducer is never reached. Consequently, measurements at long distances and approximate measurements using low level echoes are more difficult to perform.

(2) A pulse sequence of a sine- or symmetric square-wave close to the resonance frequency, as shown in Figure 3.11, is used to activate the transmitting element. The amplitude of this activating pulse sequence is for most elements about 10-20 volts. After a few periods using this sequence the transmitted ultrasonic pulse is close to the maximum amplitude.

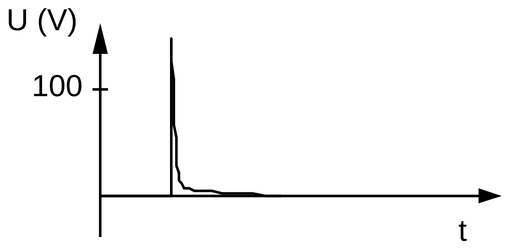

Figure 3.10. Single impulse (pseudo dirac) - high voltage. (Method 1)

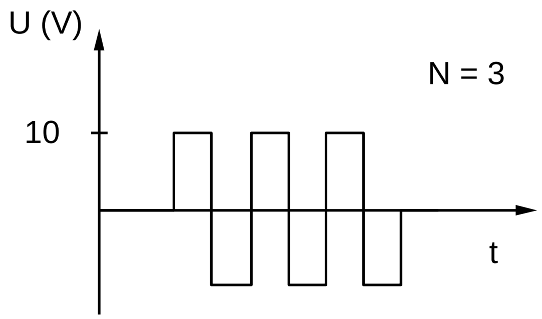

Figure 3.11. Repeated pulse - low voltage. (Method 2)

Both these methods have been implemented in the pulse generator unit for piezoelectric elements. To generate the high voltage peak required in the first method a high voltage cascade is used. This is based on a capacitor-diode ladder supplied with a 150 kHz square wave. The output voltage from the last ladder stage is about 130 V. This voltage is used in the circuit shown in Figure 3.12 to generate the pseudo dirac. Note that the circuit shown generates a negative impulse.

Method 2 is somewhat simpler to implement since normal signal levels are used. The output stage shown in Figure 3.13 is capable of driving most piezoelectric elements. The square wave is generated by alternately turning the transistors on. The control signals for the transistors are generated by a programmable state machine. This machine consists of PLD-circuits and a synchronous counter. The number of pulses generated and the frequency can be externally altered via a simple data bus. This design makes it possible for the measuring program to vary the pulse sequence depending on the measuring task. The idle state, i.e. when no transistor is in the on-state, also simplifies the use of the same element for both transmitting and receiving since this is a high impedance state.

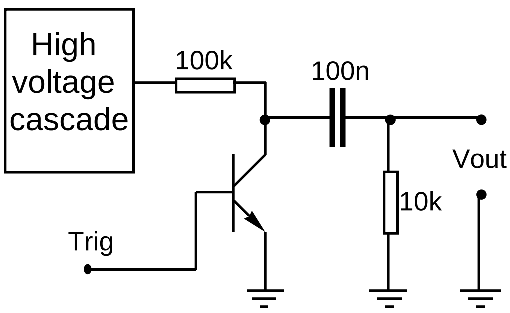

Figure 3.12 Simple pseudo dirac circuit. This circuit generates a negative dirac.

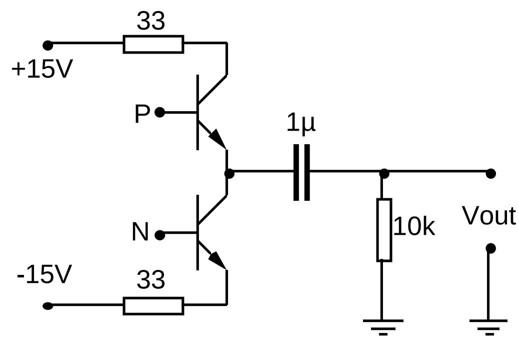

Figure 3.13 Push pull circuit to generate the square wave. Inputs P and N are individually controlled by the state machine mentioned in the text.

The previously mentioned problem to stop the transmitting element from oscillating after the transmission pulse prevents the element from being used as a receiver for a considerable time. If an active damping could reduce this time the measuring range would be improved. It is apparent from Figure 3.13 that pulsing of the element can be made in an arbitrary way. Since the transistors are controlled from a state machine the implementation is quite simple.

Several attempts with opposite phase pulsing and other signals that seem likely to damp the oscillation have been made. *All of them were without any success!* Every new active pulse obviously generates new oscillation modes and although the phase angle of the oscillation may be affected the overall amplitude is not reduced.

Since no active damping seems to work a passive method has been tested. If the oscillating transducer element is connected in parallel with a matched impedance, energy ought to be consumed in the resistive parts. The simplest, but probably not the most efficient, impedance has been used, a pure resistance. This resistance must not be connected during the transmission pulse since this would make the total too large during the transmission. On the other hand mechanical switches are too slow to be used in this application.

A possible solution is to use an analogue switch. The main drawback of an analogue switch is the on-resistance. A switch with a low resistance, 150 Ω, has been used. Since these switches are based on a FET-transistor technique they can usually be connected in parallel. Consequently, to reduce the on-resistance four switches are connected in parallel, resulting in a resistance of less than 40 Ω. The effect of such a switch connecting the transducer to a grounding resistor has been tested. The best result for all transducers were achieved when the grounding resistor was shorted to ground. This configuration with the analogue switch directly grounding the transducer with only the internal resistance gave a reduction of the unusable time period of less than 5%. Because of the rather poor improvement no further testing with complex impedance were done. Nevertheless, this configuration has been included in the pulse generator design.

To make the pulse generator flexible and versatile important parameters are controllable via the data bus mentioned above. An external trigger input allows the controlling program to send a pulse. During manual operation a free running mode with adjustable pulse repetition rate is also included. This simplifies measurement set-ups with echo views on an oscilloscope.

The pulse generator is a single output device. The output from the generator is transferred in a separate coaxial cable to the sensor unit. In the sensor unit a relay multiplexer switches the signal to the element that is going to operate as transmitter. This design minimizes the amount of wires needed between the sensor unit and the other equipment.

As a conclusion for the transmission using piezoelectric elements it can be stated that the problem of full control of the transmitted wave is far from its solution. For each element it will probably be possible to make a model that can predict the resulting wave after a certain input signal. However, no attempts in this direction has been done during the presented work. The reason for this is that the electrostatic elements offer much better capabilities in this area. Still, the piezoelectric elements are definitely the best choice in many applications. Among their advantages are the simple driver circuits presented here. More design details about the pulse-generator for piezoelectric elements, PLS1, are found in the design catalogue [Lindstedt, 1995].

Electrostatic Element Driver Circuit
Driving the electrostatic elements is in principle rather simple. A varying voltage has to be applied over the element and the membrane pressure is proportional to the square of the applied voltage. Electrically, the equivalent circuit of the element can be approximated with a capacitor of about 300-700 pF. The design of the element makes it work only with attraction forces. Consequently, to be able to move in both directions around the normal position a bias voltage has to be applied. Nevertheless there are two major problems left:

- The voltage range is 0-300 V with a bias of 150 V, and
- the flexibility and frequency response makes it desirable to be able to generate arbitrary functions as input signal to the element.

The first problem can be divided into two parts. First a high-voltage source that can deliver voltages of about 300 V DC has to be designed. Secondly , an amplifier that can work with these high voltages with sufficiently high cut off frequency has to be designed.

The high voltage source has to be safe to work with. This can be achieved if two conditions are met:
- the short circuit power should be kept as low as possible;
- the voltage should be floating and only refer to low-voltage supply potential.

Use of only normal low-voltage supply for the generation is convenient. A low loss voltage adjustment possibility is also an advantage.

Using these specifications a circuit board DC/DC-converter was designed. The first version is found in the earlier mentioned design catalogue as the board ”HVGEN”. A simplified scheme is found in Figure 3.14. The principle is that a normal PC-board transformer is used as a down-to-up transformer. The low-voltage has a middle-tap that is connected to the positive supply. By grounding the outer connections of the low-voltage side through the controlled transistor switches an alternating flux is generated in the transformer. The length of the grounding periods is controlled by the PWM-logic (Pulse Width Modulation) but the switches are controlled so that a symmetrical flux is produced. The switching frequency is constant.

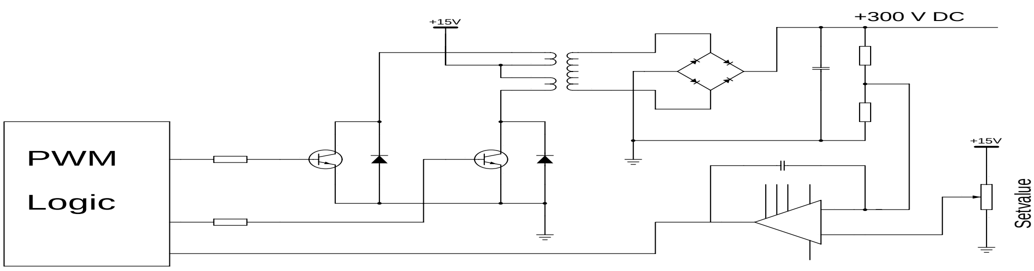

Figure 3.14 The high voltage generation circuit

The high voltage side of the transformer is connected to a full-wave rectifier bridge and filtered so that a DC high voltage is produced. A feedback amplifier with an adjustable voltage reference controls the PWM-logic so that the voltage is kept constant. This adjustable high voltage source can be used for both the bias voltage of the receiving elements and for the supply of the transmitter amplifier.

To design a high voltage amplifier with the current specifications is not a simple but a feasible problem. However, a survey for components that could be suitable for the application showed that a few manufacturers have some operational amplifiers that are made for high voltage applications. To use these is probably a better and most certainly a time-saving choice. The device PA41 made by APEX was chosen. With this amplifier the design of the amplifier stage becomes a normal OP-amp design task. The rather straight forward design is found in the design catalogue as the board ”HVAMP”. After some improvements the combination of the high voltage generation and the amplifier was integrated on the board ”HVDRV”.

Through the board ”HVDRV” the physical interface to the electrostatic element is implemented. Still, a tool for the logic design of the transmitted function has to be made. In simple applications the PLS1-board, previously described, can be used but to utilize the full potential of the electrostatic elements there is a call for a more flexible device. Analog circuits consisting of burst-generators, VCO:s and other components can produce almost arbitrary signals but they must be configured at the hardware design stage. In this case we need something that can synthesize signals arbitrarily chosen from experiment to experiment. This is only achievable by digitally controlled systems.

The basic idea for the function generator is that the signal is stored as digital words in a memory. After that a trigger condition is met the stored sequence is fed to a D/A-converter at a specified rate. When the sequence is ended the generator waits for a new trigger-signal or a new loading of data. A block-scheme of the unit is shown in Figure 3.15.

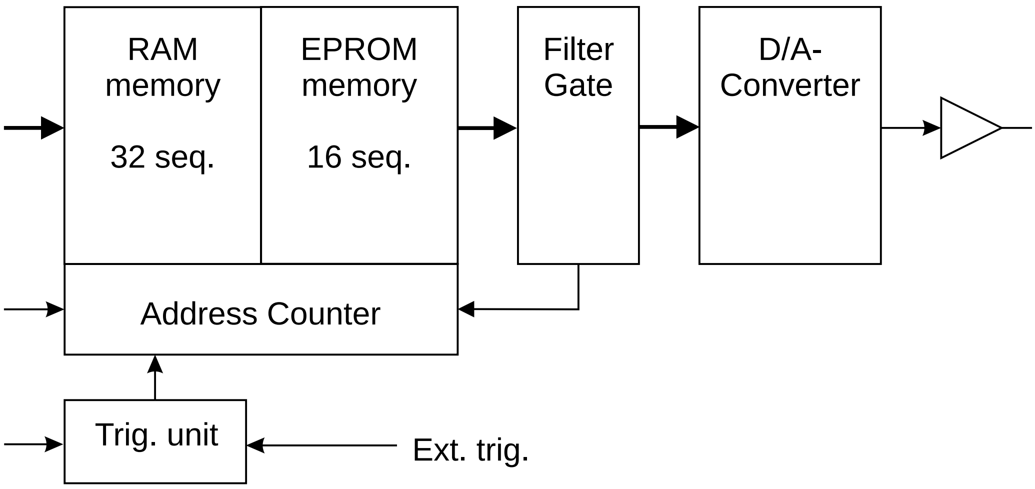

Figure 3.15   The function generator

This unit is realized as the board ”FGEN” and is found in the design catalogue. It consists of a state-machine, an address counter, a RAM-memory, an EPROM, and a D/A-converter. To minimize the data exchange during the loading of a sequence a word length of 8 bits is used. A higher resolution in the transmitted signals is not needed. To be able to load sequences of a variable length one bit-pattern, all ones, is used as a stop token. This is by a gate circuit interpreted as the zero-level to the D/A-converter but stops the address counter. The EPROM can be loaded with frequently used sequences that don’t have to be loaded. 16 sequences can be loaded in the EPROM and 32 sequences in the RAM. The address-counter rate can be either 1.25 or 10 MHz.

The unit ”FGEN” can be compared to a modern commercial function generator with a programmable function shape. These commercial units are, however, not suitable for use in a system like this one. The data transfer to the function memory is usually made via a slow link like e.g. GPIB-bus. Furthermore, the number of functions that may be stored in the memory is usually limited to one or two. This together with the difficulties to physically and logically adapt such a unit into the measurement system made the design of the unit ”FGEN” an obvious choice.

3.4 Data Sampling

According to the specifications presented earlier in this chapter there is a call for an autonomous sampling unit. This unit is autonomous in the sense that it starts the sampling at a specific trigger signal and then collects a preset amount of data at a preset rate. The unit doesn’t need any interaction with the system computer until the sampling sequence is finished. A ready-signal tells the computer that the data is available.

The sampling unit is implemented as the boards ”SMPA” and ”SMPB”, found in the design catalogue. The design is optimized for a fast data transfer. No address other than the start address is specified by the computer. Consecutive readings gives consecutive samples. A handshaking strobe is used to automatically increment the sampling unit address counter so that data is transferred as quickly as the host computer can read it.

The unit samples 8 bit data simultaneously in 4 channels. A maximum sampling rate of 2.5 MHz can be achieved. Each channel has a memory of 8 kbytes but the design is prepared for larger data storage by changing the memory circuits. A block scheme of the sampling unit is shown in Figure 3.16.

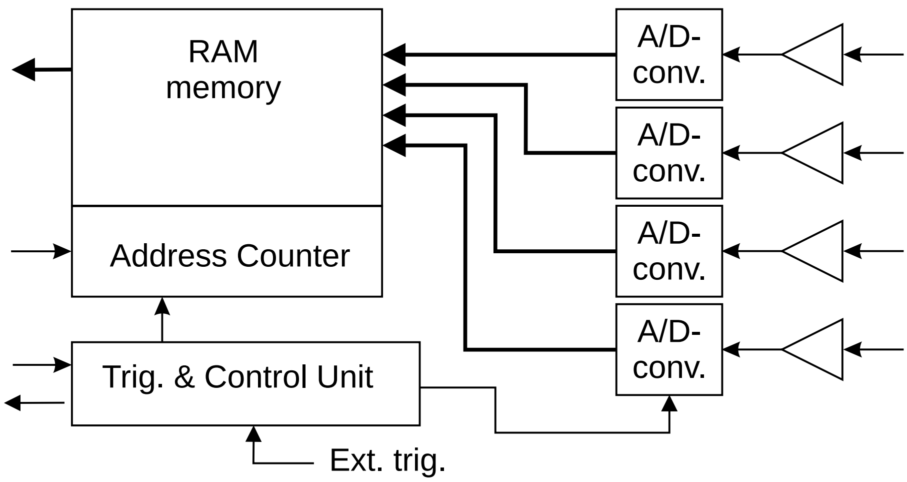

Figure 3.16 The sampling unit.

3.5 System Overview

With the overview of the measurement system from Figure 3.1 as a skeleton and the descriptions of the sub-units in this chapter and in the design catalogue [Lindstedt, 1995] it is possible to make a more detailed system description. In Figure 3.17 the names of the sub-units have been added to clarify their physical localization.

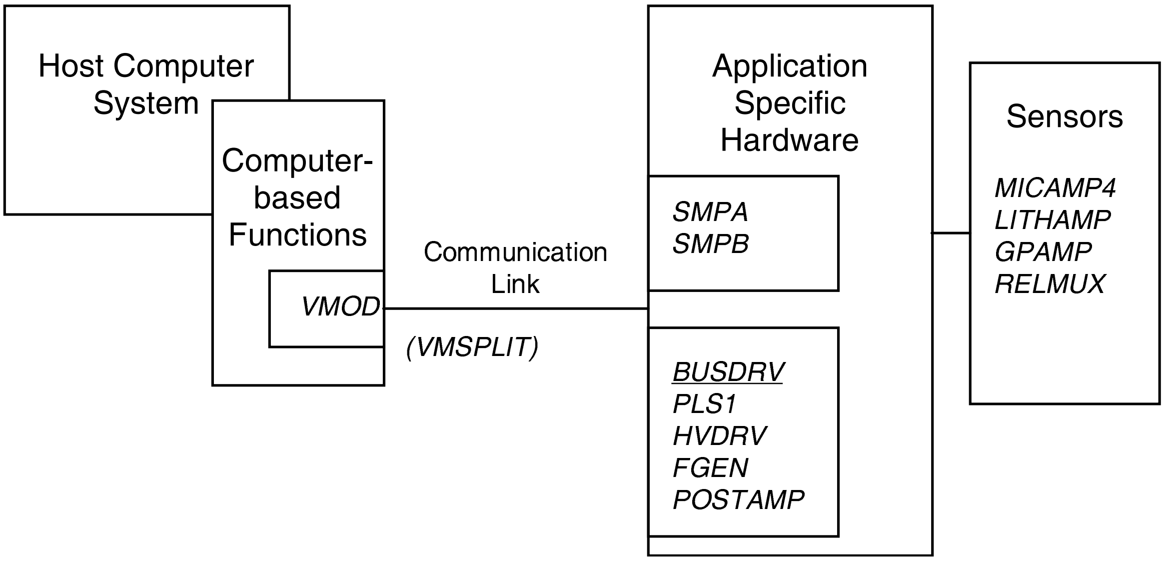

Figure 3.17   System overview with names of sub-units.

The computer based functions consist of a ”piggyback” adapter to the commercial VME-board VMOD made by PEP Modular Computers. The adapter, that also is called ”VMOD”, is designed to form the necessary ports for the two parallel buses that make up the connection between the computer and the application specific hardware. An interconnection board ”VMSPLIT” is used to connect the two bus cables to the connector of the VMOD VME-board.

One of the two buses is reserved for the sampling unit consisting of ”SMPA” and ”SMPB”. The electrical layout of this bus allows the fast data transfer. The other bus is more general purpose in its design. Through a board called ”BUSDRV” it forms a backplane bus that is used for the control of the other boards and their parameter selections. Among these boards are :

- ”PLS1” that is used for the transmitter driving of piezoelectric elements,
- ”HVDRV” and ”FGEN” that are used for the transmitter driving of electrostatic elements, and
- ”POSTAMP” the input amplifier that adjusts the signal levels from the sensor unit before the sampling and also provides the multiplexer signals for the transmitter selection.

These application specific hardware modules are connected to the sensor unit with two cables. One that contains the transmitter voltage pulse (separately shielded) and one with the sensor signals, the multiplexer data and the sensor unit power supply.

In the sensor unit the sensors, amplifier boards (”MICAMP4”, ”LITHAMP”, ”GPAMP”) and the relay multiplexer (”RELMUX”) are situated.

3.6 Optimization for Real Time Applications

The main guideline for the described design has been full flexibility. This means that the total amount of raw data is transferred to a host computer. In a system working with developed and tested algorithms a lot of data processing can be done much earlier in the sequence. A separate computer may very well be located at the sampling unit with direct access to the sampled data via e.g. a shared dual ported memory area. This structure also means that the data processing can start before the sampling is finished. Digital signal processing using DSP:s can also be done after the A/D-conversion before the data is stored in the memory buffer. Since design efforts in the direction towards early data processing reduces the amount of data that has to be presented to the host computer the communication link performance becomes less important. The host computer is also supplied with information on a higher level and will get more time for decisions of strategic nature.

Distance Measurements 4

This chapter describes principles and methods for distance and angle measurements. Piezoelectric and electrostatic transducers are, when necessary, handled separately. The handled angular measurements includes both two-dimensional (one angle) and three-dimensional (two angles) cases. Last in the chapter some practical experiments are presented to give an impression of possible accuracy.

4.1 Distance Measurement Introduction

Outside the area of pure physics the principle of distance measurement is one of the first presented applications for ultrasound. The development of the radar pulse technique had a large impact on the development of ultrasonics during the late 1940s. The coupling between radar and ultrasonic methodology has been maintained over the years and the relationship is still obvious in many applications like underwater sonar [Urick, 1983]. Both techniques are based on time measurement for the propagation of an impulse with a known velocity, c, in the current medium. If the measured delay-time, t_d, is measured the total length can be expressed as:

$$l = t_d \cdot c \tag{4.1}$$

The measurement may be done like a

transmitter$\rightarrow$ receiver (TR) or

transmitter$\rightarrow$ reflector $\rightarrow$ receiver (TRR)

measurement as indicated in Figure 4.1. If the purpose of the measurement is to produce an image based on a principle like the synthetic aperture radar (SAR) several measurements from different positions or angles has to be done. Still, the basic principle used is the distance measurement.

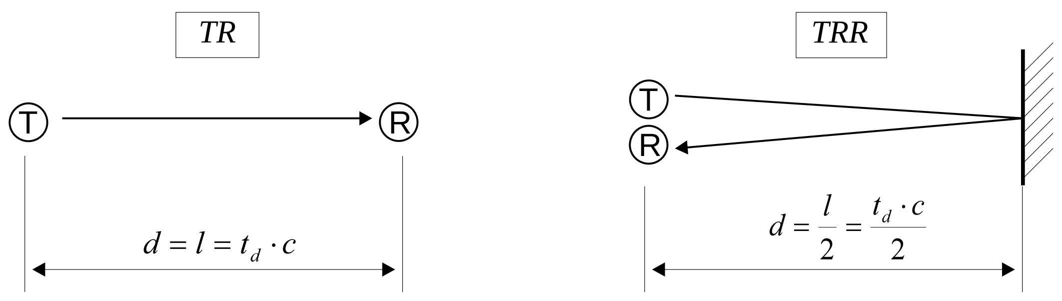

Figure 4.1 Distance measurement using the transmitter to receiver delay (TR) and the pulse echo delay (TRR).

In the *TR*-case the measurement usually is a single scalar distance measurement but in the *TRR*-case multiple echoes from different distances may also be of interest. This is actually the basis for most image-oriented measurements. The reason for using the *TR*-principle is often that a long distance has to measured or that interfering objects are close to the shortest path between the two points. In TR-measurements the signal is sent only in one direction while the TRR-measurement requires an echo that doubles the signal propagation distance. To separate the correct echo from echoes produced by interfering objects can in some cases be an almost impossible problem to solve.

In commercially available pulse echo distance measurement units (*TRR*-type) the most common task is to measure the distance to the closest reflecting object. This may seem to be a simple task but several factors can be difficult to control. Among them are:

- amplitude variations due to distance and reflecting object shape,
- measurement noise and disturbances,
- varying shape of the received echo-pulse,
- difficulties to determine the beginning of the received echo-pulse, and
- temperature variations that affect the speed of sound.

Some of these factors are sometimes taken care of in commercially available distance measurement devices but many of them are usually not handled. The reasons why some of the problems are left out can be many. In some cases the intended application implies that the problem can't occur because of the nature of the measurement situation. In other cases the problems lead to restrictions for use, limited accuracy or even limited reliability.

Since the distance measurements in this thesis are the basis for the following image-oriented measurements as many problems as possible will be treated.

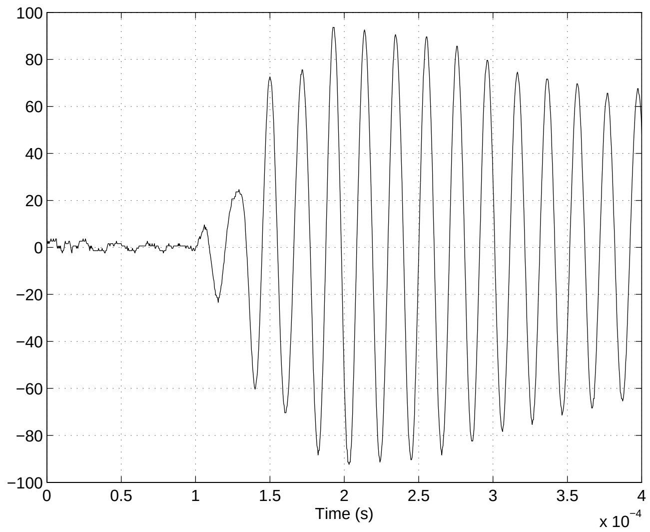

Figure 4.2 The beginning of a measured echo pulse using electrostatic elements. (A commercial handheld distance meter is used as signal source.)[2]

Practical measurements of distance requires that the time between the beginning of the transmission of a pulse to the start of the receiving of the same pulse can be measured with

[2] The measurement of the shown pulse is not a trivial task since the moment for pulse transmissions can't be controlled from outside the unit. Because of this two sensor elements were placed in sequence (0.5 m distance) along the pulse propagation path. The signal from the first one was used to form a digital trig-pulse that was connected to the sampling unit. The signal from the second sensor was then sampled. In this way the pre-trig problem was solved.

sufficient accuracy. The main problem in this measurement is to determine the instant at which the received pulse begins. A significant increase in the amplitude is easy to detect when the pulse has arrived but the amplitude at the pulse start is almost not noticeable. A typical shape is shown in Figure 4.2. Furthermore, if the measurement is made in a noisy environment the problem turns out to be practically impossible to solve.

The described problem is encountered using both electrostatic and piezoelectric sensors, although it is more difficult to solve in the piezoelectric case. Several time-constants in systems connected in sequence result in an pulse envelope with a derivative of about zero at the pulse beginning. The only action that can be taken to improve the behavior is to select a transmitted pulse shape with as much energy as possible at the beginning. Furthermore, the electronic system connected to the transducers should of course not introduce any significant new time constants.

4.2 Methods and Tools

Level-detection with Assumed Offset

The beginning of the pulse is difficult to detect, but the continuing part can be easily found. Because of this many systems make the detection with a threshold that is crossed by the pulse continuation. This gives a time instant that definitely is too late. However, by experience and/or transducer models the timing error can be estimated. The principle for this measurement is shown in Figure 4.3. This estimation could be very good if it only depended on the sensors and the transmitted pulse shape. In real applications unknown factors are usually present, namely the position and shape of the reflecting object. Compensation for the distance to the object is easily implemented since this can be a function of the elapsed time. Worse is the shape of the reflecting object. The fatal interference described in Chapter 2 and shown in Figure 2.9 may serve as an extreme example. It is obvious that such an object would create a lot of problems using this detection principle even if sufficient signal amplitude was reached. The error is, because of the nature of the method, distributed around steps of one wavelength (or half a wavelength if a symmetric threshold is used). The wavelength at 50 kHz in air is about 7 mm which in an echo measurement (forth and back) corresponds to a measured length of 3.5 mm.

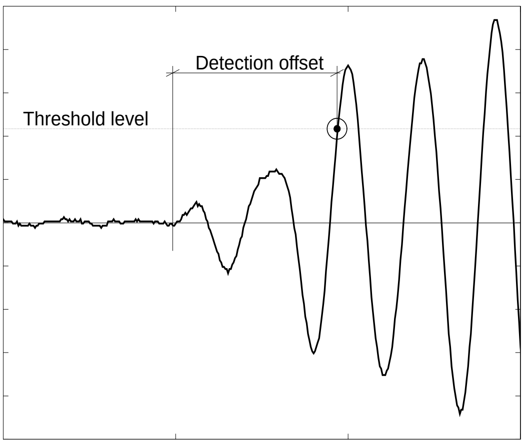

Figure 4.3 Measurement using a threshold level and a delay time estimate. The same pulse as in 4.2 is used.

This measurement principle is used primarily in limited accuracy commercial systems. The specification of the maximum error is usually about one centimeter. The principle should be applied in measurement situations where large reflecting objects are used. Although this method seems a bit primitive it is probably the most widely used. The simple algorithm can be easily implemented in electronic hardware. Consequently, if the measurement conditions concerning accuracy and object size can be accepted, it is a quite good method for practical measurements if it is combined with a gain control to compensate for attenuation at long distances.

An extension to the method is to extrapolate the beginning from the maxima of the received pulse. This has been tried using piezoelectric elements but doesn’t seem to improve the reliability of the method. The method is still sensitive to the reflecting object shape.

Backtracking the Zero-Crossing

The backtracking method is primarily intended to be used for piezoelectric sensors. They have a strong resonance in combination with a rather inert behavior. This make the estimation of the time from the pulse beginning to the threshold value described above difficult. The variations can be very large. This calls for a better use of the data in this crucial time interval.

If a pulse is detected using a threshold value it is possible to localize a minimum or a maximum close to the threshold crossing. It is then a high probability that we will find a previous extreme value about half a period before the current extreme value. To find this new value we define a time-window starting slightly before, and ending slightly after, the time instant half a period before the current extreme value. We then try to find a new extreme value in this window and, if found, use this as the new current extreme value. This is repeated until no extreme value can be found or a discrimination limit is passed.

This method is a phase-locking detection with a capture range that is defined by the time window. It works well on piezoelectric sensors because of their inert behavior. If a signal of considerable energy is activating the resonance no quick phase changes can take place. This means that it is highly probable that the start of the oscillation due to the received pulse can be found in a rather narrow time window. Thus, the very first oscillation can often be localized at an amplitude comparable with the background noise if we know where to look for it. One problem using this method is that the background noise and/or previously received signals makes the piezo-material oscillate at a constant but low amplitude. If this ”background”-oscillation is locked in the capture window the backtracking can proceed far behind the echo pulse. A discrimination based on the amplitude sequence and the distance from the threshold crossing can reduce this problem.

An implementation of this method is described in [Lindstedt, Kowalew, 1991]. The method is considerably better compared to threshold-methods when piezoelectric sensors are used. Weak echoes produce large estimation errors when thresholds are used while the backtracking often is successful finding the correct first oscillation. A condition that is important for the success is that the constant ”background”-oscillation is small. This can be achieved if the acoustic shielding between the transmitter and the receiving element is good (e.g. Figure 3.9) and the environment noise level is low.

Phase-Angle Observation

A common way of representing frequency components is as a complex number describing signal amplitude and phase angle compared to a reference. This can also be applied to a received ultrasonic echo. Many other applications use this method e.g. radar [Mensa, 1991 pp. 66-68], [Skolnik, 1962 pp. 437-438]. The method is called coherent demodulation or I/Q-demodulation (Inphase/Quadrature). An overview of the method is shown in Figure 4.4. A primary reason for the use of this method is often to reduce the need for fast sampling of the signals. Even if this is not needed when we are dealing with the frequency region of airborne ultrasound the method is an excellent interpretation aid (although it actually doesn't add any new information). The basic principle for this interpretation is that we know what frequency has been transmitted. Consequently, the features of the echo should be possible to express as the amplitude and the phase angle of a signal with this frequency. If a frequency reference deviation is present this will show up as a constant increase or decrease (rotation) of the phase angle. The signal can be expressed as:

$$S = A \cdot e^{j\varphi} = A \cdot \left[\cos(\varphi) + j\sin(\varphi)\right] \tag{4.2}$$

The input signal is mixed (multiplied) with a frequency equal to the transmitted. In one branch a sine is used and in the other branch a cosine (orthogonal) is used. The products then consist of one component with the double transmission frequency and one component that is sine or cosine for the phase angle. The products are filtered in lowpass-filters to eliminate the high frequency parts and maintain the phase-angle components. These components may then form a complex representation of the wave that is easy to convert to amplitude and phase. Figure 4.5 shows the phase and amplitude curves together with the original input echo-signal. It is obvious that the phase-shift information makes the interpretation much easier.

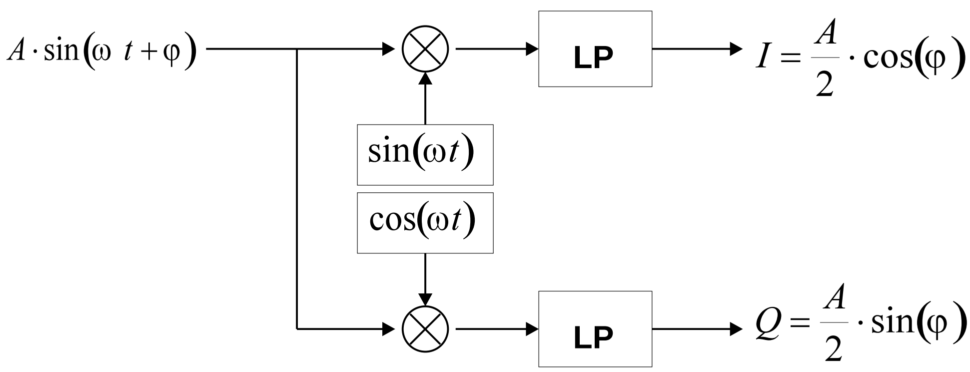

Figure 4.4 The I/Q-demodulator.

Subtraction of Constant Oscillation in Piezoelectric Sensors

It has earlier been mentioned that piezoelectric materials often are oscillating at their resonance frequency with a low but constant amplitude before the actual echo signal arrives. This oscillation is sometimes caused by remaining energy from a direct wave from the transmitter but may also be caused by small disturbances like slightly movements in the air. Since the elements have an inert behavior the oscillation usually is stable over a time of several periods. An assumption that this oscillation is superimposed on the actual echo and thus can be subtracted from it would simplify the search for the echo start. Such an experiment is shown in Figure 4.6. First of all a point where it is certain that the echo has not started is fixed. Then a time corresponding to about five periods of the resonance frequency is observed. This time is from the fixed point and backwards. A fitting of amplitude and phase of a sine-function is done to the signal in this time interval. The designed sine is the subtracted from the whole echo sequence in the interesting interval.

This method has been tested on signals before the backtracking method is used. Especially when the echo suffers from contamination from a strong transmission-signal the improvement is significant.

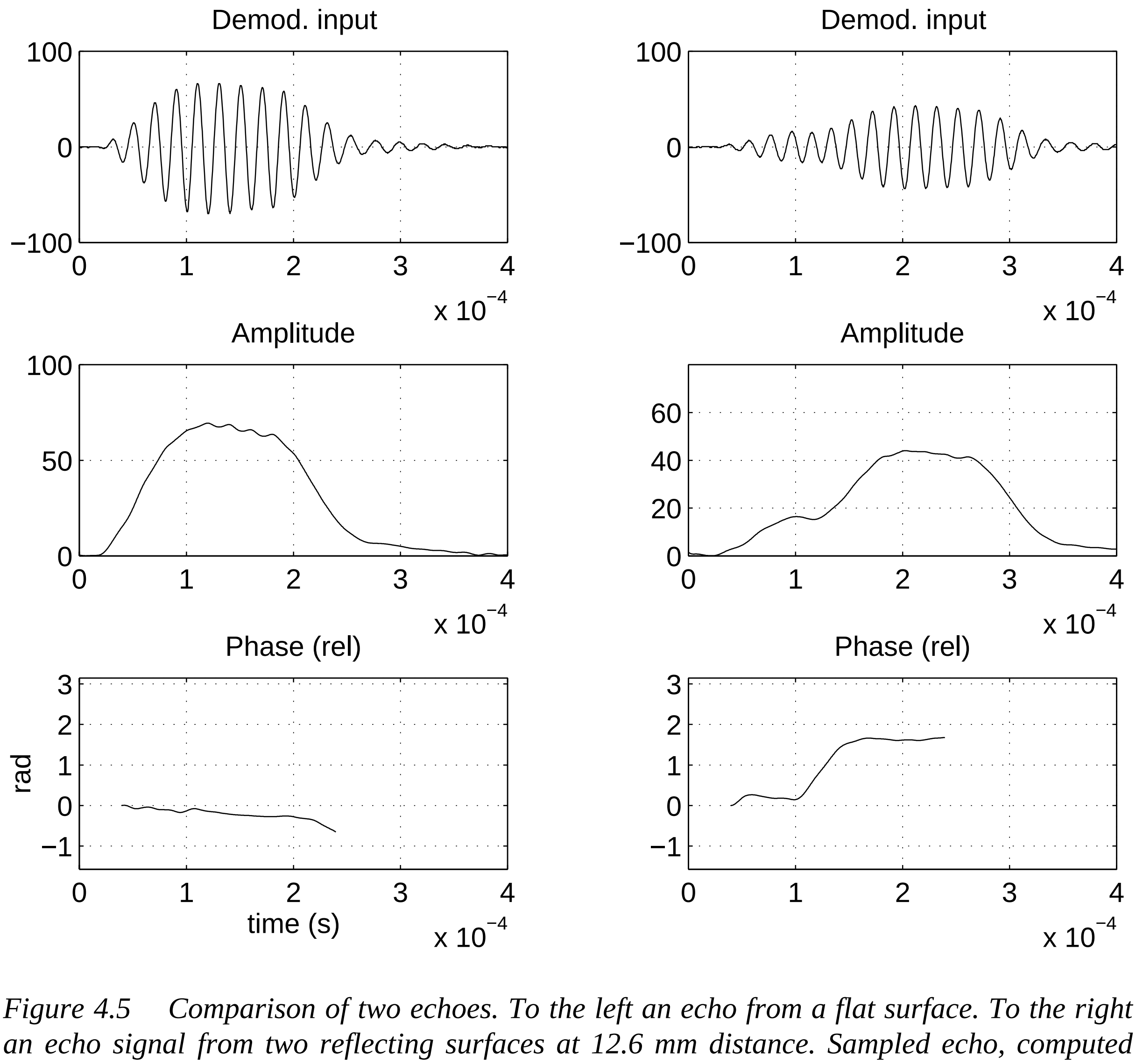

Figure 4.5 Comparison of two echoes. To the left an echo from a flat surface. To the right an echo signal from two reflecting surfaces at 12.6 mm distance. Sampled echo, computed amplitude, and phase curves are shown. The two surfaces give a significant shift in the phase curve. The theoretical phase difference for the two surfaces is 23.3 or (8π-1.88) radians. This means that the echo from the second surface arrives that much later. It is thus recognized as a signal 1.88 radians before the current one. The reference for the phase curves is the phase in the beginning of the echoes. (Phase curves are not significant at amplitudes close to zero.)

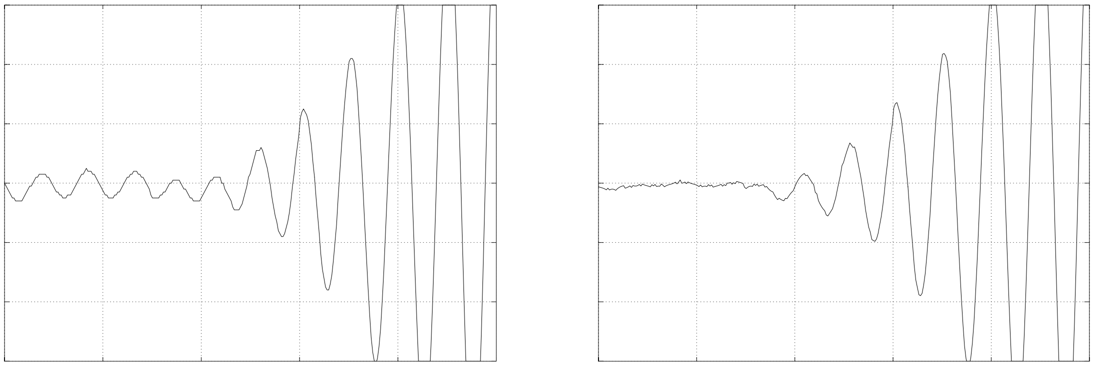

Figure 4.6 Echo signal contaminated with a constant oscillation to the left. Resulting signal after subtraction of estimated oscillation to the right.

Filters

The piezoelectric signals are found in a narrow frequency span because of the resonance of the piezoelectric material. Because of this sharp selective bandpass filters can be used to eliminate a lot of disturbances and noise. Also when the electrostatic sensors are used filters that select only the frequency content of the transmitted pulse can eliminate many problems. The wide band characteristics of the electrostatic elements make them react on many unwanted signals. This increases the need for bandpass filtering. However, filters must be used with great care and the following points are important to note when they are designed:

- the filter must not introduce any significant delay to the signal since this has a direct impact on the measurement result;
- moving reflectors cause Doppler effects that may cause the signal to change the frequency so that it is outside the passband of a highly selective filter;
- transient episodes in the signal sequence must contain frequency components outside the used sine signal frequency. There is a significant risk that these signal shapes are altered.

The first one of these three points can often be taken care of since the filtering is done on a batch of data from one echo sequence. This means that non-causal filters can be used. Consequently, forward/backward filtering is applicable. Filter types for various purposes may then be designed without the introduction of any time delays.

Doppler effects in scenarios where the velocities are less than 1 m/s are often of less importance from a distance measurement method point of view. At higher speeds the distance measurements are usually not of importance for the action since their repetition rate is too slow. More about the Doppler effects in ultrasonic measurements is found in Chapter 7.

The transient behavior is usually only a problem when narrow passbands are used or the filters are of very high order. The easiest way to check this is to observe a number of typical echoes and compare the raw data to the filtered. Unacceptable behavior is usually detected after a few experiments.

Some practical filter experiments using the piezoelectric sensor unit elements described in Chapter 3 is found in the master thesis by [Rogdahl, 1994].

Matched Filters

The knowledge of the transmitted pulse shape is a very useful piece of information. The previously described methods don't make full use of it. This information is fully utilized in the matched filter which gives a sometimes quite astonishing result. The basic principle for the matched filter is that the signal we receive is assumed to be the same signal as we transmitted, *p(t)*, with noise, *n(t)*, added to it. Under this assumption it must be possible to design a filter that is extremely good at detecting the transmitted signal shape from the noise. Under the assumption that the added noise is white it can be shown that the filter that gives the optimal signal to noise ratio has an impulse response, *h(t)*, that is the transmitted signal run backward in time. A simplified overview is shown in Figure 4.7.

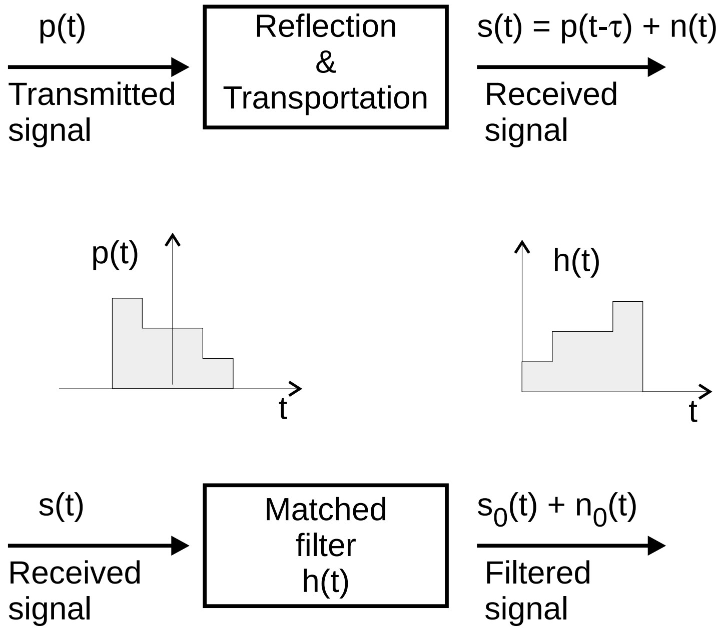

Figure 4.7 Matched filter principle

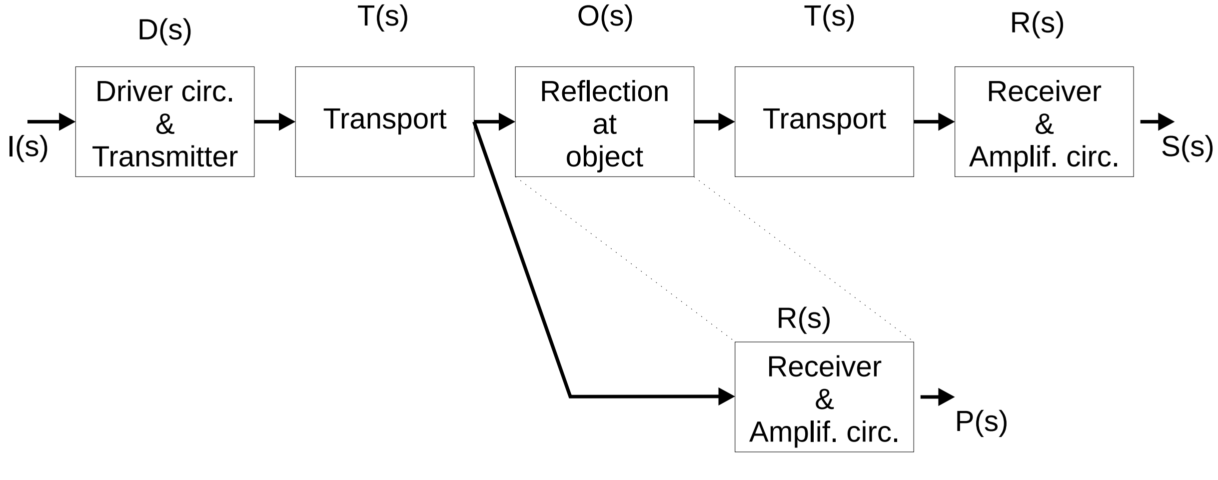

Figure 4.8 Measurement chain for ultrasonic measurements. An extra measurement device is added to give a better estimation of the transmitted pulse.

It is important to note that the matched filter does *not* try to reproduce the transmitted signal shape from the received signal *s(t).* It can rather be considered as a detection function for the given pattern. Under the theoretical assumptions above the filtered output is the autocorrelation function of the input with the noise added. This nature of the output means that it points out the instant when the transmitted pulse has the highest probability to be found in the received sequence. This is of course an excellent tool for distance measurement. The requirement that the noise should be white is, as very often, of more theoretical than practical importance. In most cases all uncorrelated noise is efficiently suppressed by the filter.

The implementation of the matched filter can sometimes be difficult. For causality reasons it is not possible to point out the most probable position of the pulse at the time the pulse begins. This means that in some applications there is always a delay in the information produced. If the filter is to be implemented using Fourier- or Laplace-transforms there are sometimes a difficulty in finding the transform for *h(t).* More about this and other practical implementation problems can be found in [Skolnik (1962) pp. 408-418].

The scenario that was described when we introduced the matched filter is highly idealized if we compare it to ultrasonic measurements. A more realistic view is shown in Figure 4.8. Here is *I(s)* the intended pulse to be transmitted. This pulse is modified by the transfer functions of the electronic driver circuits and the transmitter, *D(s).* If the path from the transmitter to the reflecting object is free and no major sources of disturbance are present then *T(s)* may be described as a linear attenuation, i.e. the pulse shape is retained. In the figure the receiver is assumed to be in the same position as the transmitter. Consequently the same transfer function, *T(s),* is used for the return of the echo. The shape and position of the reflecting object certainly affects the pulse. This is represented by O(s). Finally the receiving element and the amplifier is represented by R(s). Thus the received signal is:

$$S(s) = I(s) \cdot D(s) \cdot T(s) \cdot O(s) \cdot T(s) \cdot R(s) \tag{4.3}$$

The challenge is to make a reliable reconstruction of *I(s),* given *S(s).* This is certainly a non-trivial task since all of the transfer functions in (4.3) contribute to transform *I(s)* so much, that *S(s)=I(s)+noise* becomes an unrealistic approach. Even if the transport is ignored a basic knowledge about the ultrasonic element dynamics - included in *D(s)* and *R(s)* - (see e.g. [Mattila, 1995]) tells that the actual pulse must be estimated in a better way. To do this the extra receiver that is indicated in Figure 4.8 was introduced. This receiver is positioned at the reflecting object and will thus measure the incoming pulse before reflection. Since the receiver has the same transfer function as the echo-receiver the measured pulse is:

$$P(s) = I(s) \cdot D(s) \cdot T(s) \cdot R(s) \tag{4.4}$$

If we now assume that the transport transfer functions were possible to ignore the only difference compared to *S(s)* is the object transfer function *O(s).* There will of course be a noise component in *P(s)* but since the measurement is made on a direct transmission to the receiver the signal is strong and consequently the signal-to-noise ratio is high. This measurement is made at very favorable conditions. Consequently, the beginning of the transmitted signal is readily detected.

If *P(s)* is used as the signal that we are looking for in the echo sequence the ideal conditions are met if the object is a point scatterer. This is of course a highly limiting condition but since many objects can be considered as a set of point scatterers the condition may be slightly modified. If the object can be described by a small number of scatterers separated in time segments of the echo the information produced by the filter will still be useful for single measurements.

In most pulsed ultrasonic applications it is possible to process data in a batch or at least semi-batch manner. This means that causality is usually not a problem. Because of this the filter implementation can be done in a straightforward way. The filter is a convolution of two functions in time. Since the data are sampled the time is discrete.

$$p(k) \quad \{0 \le k \le (n-1)\} \tag{4.5}$$

$$h(k) = p(n-1-k) \quad \{0 \le k \le (n-1)\} \tag{4.6}$$

$$s(k) \quad \{0 \le k \le (m-1)\} \tag{4.7}$$

$$s_0(k) = \sum_{\nu=0}^{n-1} s(k+n-1-\nu) \cdot h(\nu) \tag{4.8}$$

The sequence h is referring to Figure 4.7 while the sequences p and s are referring to Figure 4.8. The sequence s_0 is the filter output.

The filtering expression (4.8) is a normal convolution except for the time shift *(n-1)* to synchronize it to the echo sequence in index. In the Figure 4.9 a measured transmitted pulse *p(t)* is shown. This is a pulse that can be measured once and then stored as a reference. In the Figures 4.10 and 4.11 the echo from a 1.2 mm thick wire at a distance of 0.4 meters is measured and detected using a matched filter. The electrical signal sent to the transmitter is four periods of a 50 kHz sine. Since the echo is weak a high receiver amplification is used. This has resulted in considerable noise. In Figure 4.10 the noise consist of thermal noise from the electronics and background sound e.g. a computer fan. In the Figure 4.11 pressurized air is leaking in a way that has been noticed to have ultrasonic frequency components detectable with the current configuration. In both cases the noise reduction is remarkable.

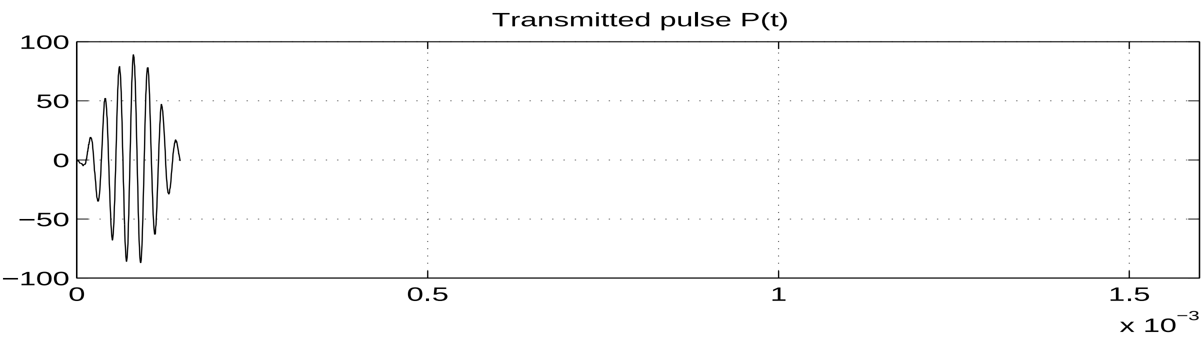

Figure 4.9 Measurement of the transmitted pulse

It has to be emphasized that the measurement of *p(t)* can be made in a separate off-line experiment, so that *p(t)* is known a priori. The shape of *p(t)* is not affected by the speed of sound which makes the method insensitive to temperature variations. The reason is that the signal *p(t)* is recorded as a direct signal and not as a reflection. The practical consequence is that there is no need to locate a transducer close to the measured object in the real time application. Instead *p(t)* is, once and for all, stored as a data sequence in the computer. Of course, each used signal shape, *i(t)*, have to have a corresponding *p(t).* However, this is usually not a problem since most applications use only a few different shapes for the transmitted signal.

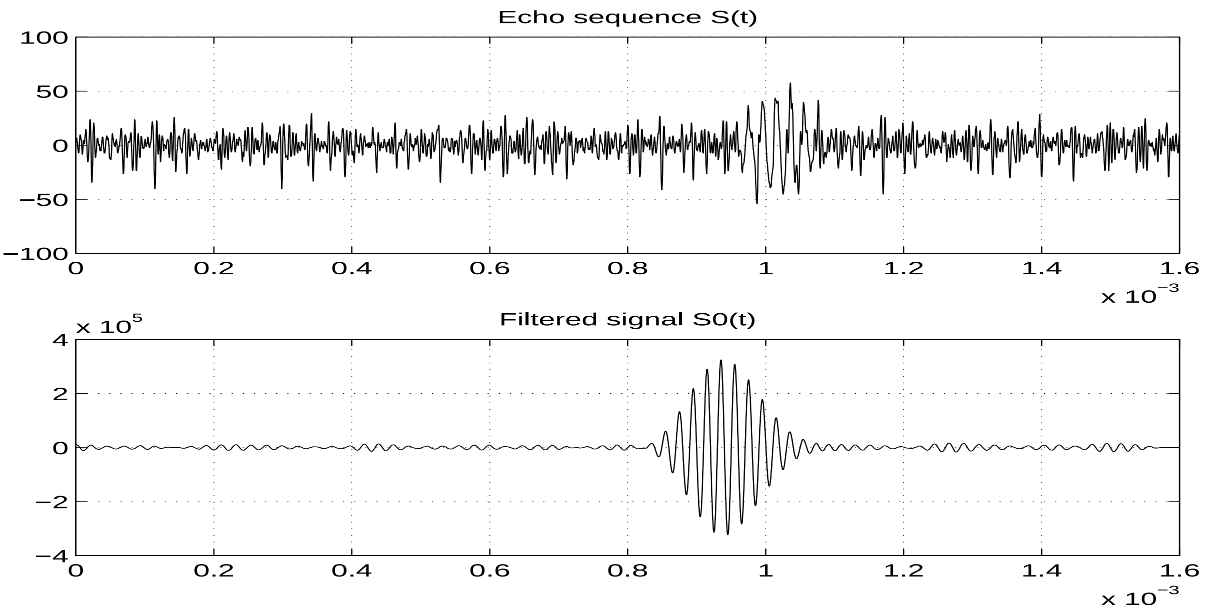

Figure 4.10 Echo from a 1.2 mm wire at a distance of 0.4 m. The peak output of the filter shows the echo start.

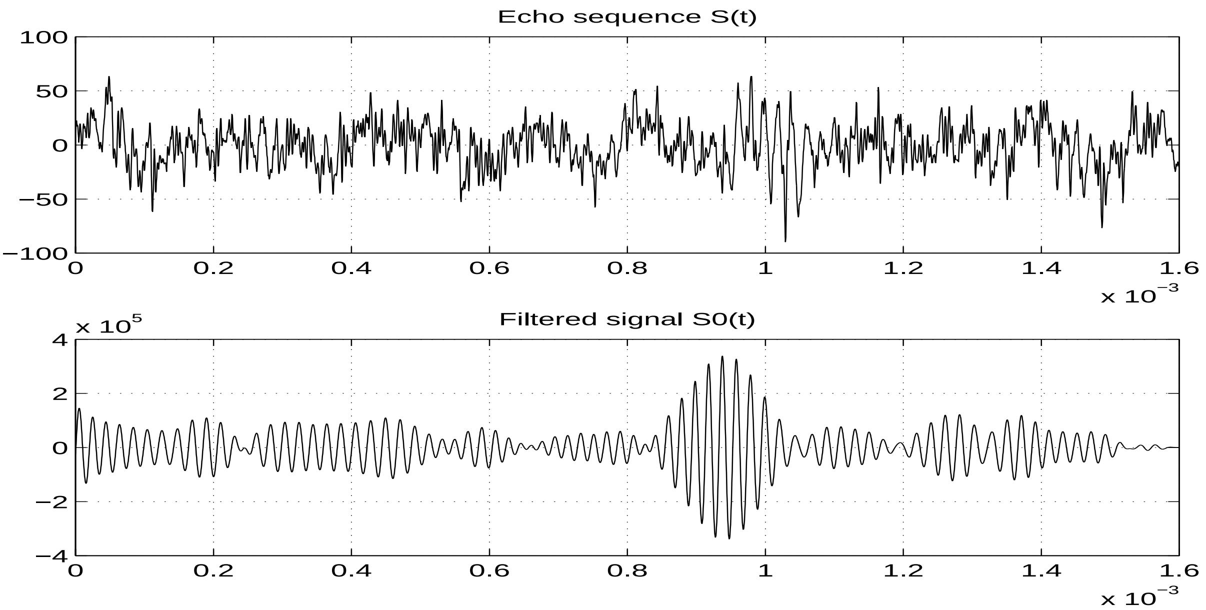

Figure 4.11 Same as 4.10 but with an added noise source that was known to give correlated frequency components in the measurement pulse frequency band.

Transmitted Signal Selection
When the transmitted signal is to be selected a number of principles is at hand. Most of them are also used in radar technology. The main groups are:

- continuous Wave (CW) single frequency,
- pulsed single frequency,
- CW stepped frequency,
- pulsed stepped frequency, and
- swept frequency (FM).

In the previously described measurement methods we have more or less assumed the single pulse principle. This is, however, not necessarily the only usable principle in many of the methods. Of the described signal groups it is only the CW-signals that requires totally different methods. These signal are usually used together with Doppler, angular or interferometric methods i.e. not in distance measurements. Consequently, most of the other signals can be used directly or with small modifications to improve the measurements using the described methods. More about multi-frequency methods is described in Chapter 8.

When the Polaroid Corporation developed their electrostatic transducer system (primarily intended for ranging in cameras) problems like fatal interference and long distance attenuation probably became apparent very soon. These problems were described in general in Chapter 2. In the ranging system developed by Polaroid solutions to these problems are presented [Polaroid Corporation, 1980]. Since the system is intended for use also in non-computer-based applications the solutions are implemented in hardware.

In the Polaroid system the transmitted pulses consist of the following four sequential parts:

- 8 cycles at 60 kHz,
- 8 cycles at 57 kHz,
- 16 cycles at 53 kHz, and
- 24 cycles at 50 kHz.

This gives a total transmission of 56 cycles during a time period of about 1 ms. The sequence will, of course, make it possible to avoid the problem with fatal interference. Some part of the echo has to return a detectable signal. However, the solution of this problem was probably not the main intention of the designers at Polaroid. Their main concern was to create a device that could give reliable measurements over a wide distance range i.e. up to 10 m. This requires a good handling of the echo amplitude at various distances.

The elapsed time since the transmission tells what distance we can expect echoes from. Because of this it is not difficult to compute the appropriate gain each moment. The gain is in the Polaroid system programmable to one of sixteen levels. A counter started at the pulse transmission selects the current level. However, the gain is not the only parameter controlled by these levels. The bandpass characteristics of the amplifier is also affected. Each higher level corresponds to a higher Q-value. At the final levels the filter selects almost only 50 kHz components. Two things are achieved with this method. First, the amplitude is further increased at the later parts of the echo. This can be necessary since the amplitude dynamics is about 100 dB over the current range. The more powerful transmission (24 cycles at 50 kHz) is also selected. Secondly the operating frequency range is narrow at the end of the echo. This is an advantage since otherwise the risk for disturbances is much higher when the gain is increased. The later parts of the transmitted pulse are used for the measurements at long distance. This introduces a delay in the signal. A linear adjustment to reduce this is possible but it is also true that the relative error at longer distances isn't that sensitive to small offsets. The Polaroid unit has a special threshold detection. The signal is filtered through a bandpass filter and is then rectified. The rectifier output is then integrated. This becomes a kind of energy function that is approaching and crossing the threshold.

Some of the sonar gain control and detection functions have been implemented in IC:s by Texas Instruments. Their TL851 and TL852 can be used for electrostatic microphones while TL853 can replace TL851 if piezoelectric microphones are used [Texas Instruments, 1989].

The example from Polaroid shows that a creative selection of the transmitted signal combined with a feasible amount of signal processing can improve the measurements significantly. Further examples in the direction of multi frequency measurements will be discussed in Chapter 8.

4.3 Experimental Results

The range over which measurements have been done has not been extended to more than a couple of meters. This is because the focus of the measurements has been put on describing the workspace in an automation situation e.g. an assembly robot workspace. Consequently, the accuracy of the used distance measurement methods have primarily been tested for shorter distances. However, most of the used methods may also be applied in scenarios where longer distances are of interest. for instance like mobile robots.

The accuracy of all distance measurements depends a lot on the shape of the reflecting surface. Objects with surfaces that are small and not perpendicular to the acoustic axis can be almost impossible to detect. This gives no information about the possibilities of the methods. Because of this the described accuracy refers, when nothing else is stated, to measurements towards flat surfaces, perpendicular to the acoustic axis, and of sufficient size to produce a satisfactory echo signal.

For the 40 kHz transducers it has been possible to achieve an absolute precision of better than 1-2 mm. The corresponding values for the 200 kHz device is about 0.5 mm. The 200 kHz transducer gives reliable distance information up to at least 350 mm, while the 40 kHz transducers are useful down to significantly shorter distances. Consequently there is a favorable overlap of validity ranges going from the long distance 40 kHz measurements down to the short distance 200 kHz measurements. The measurements have been made using the level-detection combined with the backtracking methods.

The electrostatic elements measure distance with an accuracy that is comparable to the 40 kHz piezoelectric ones. However, their different shape of the echo makes the detection, using the same method as for the piezoelectric ones, more certain. The consequence of this is that measurements can be made towards smaller objects and in worse signal-to-noise situations.

Angle Measurements and Object Localization

5

This chapter makes an approach towards the increased possibilities using more than one receiving element. Measurements of inclination angle at the reflecting surface, under the assumption that the surface is sufficiently large, is described. This is done for both the two- and three-dimensional cases. Measurement of an object position is done under the assumption that the size/distance relations are favorable enough to consider the object to be a point scatterer.

5.1 Introduction

Information Obtained by Various Numbers and Positions of Transducers

The distance measurements described in the previous chapter is a one-dimensional operation. The direction to (position of) the reflecting object can't be measured since the echo delay depends only on the distance to the object. The only estimation that can be done is based on the amplitude that is affected by the angle between the acoustic axis and the object direction. Then the amplitude of the echo when the object is on the acoustic axis must be well known.

The introduction of a second receiving point dramatically changes the possibilities for conclusions about the object location in two dimensions. The third receiver can, if it is appropriately placed, make a three-dimensional measurement. More receivers give additional information and further increase the confidence of the geometry in the object situation. This sounds very straight forward, but the reality is somewhat more complicated. Everything would be simple if the world was built from well separated point scatterers. However, real objects consist of multiple surfaces of various dimensions and shape. We observe them from various distances and various angles. Many objects may be present at the same time. It is consequently easy to understand that ultrasonic measurements using transducers with lobes like the one indicated in Figure 2.3 and with mainly specular reflections will met a lot of ambiguities and amplitude problems.

The measurement unit configuration that is used plays an important role in defining what information can be extracted. In radar-oriented approaches a vector of equally spaced transducer elements is often assumed. By transmitting from one element at a time and making a coherent summation of the received signals in the vector a synthetic aperture, SAR, may be created. This can produce a two-dimensional reasonably detailed map over the point-scatterer density on the map. Methods including beam forming by transmitting from more than one element at a time may increase the accuracy of the measurements. General radar-methods in this direction can be found in [Mensa, 1991] while a specific beam forming approach using ultrasound is described in [Webb and Wykes, 1996].

The ideal for the focused SAR-methods is an array of sensors placed densely on a spherical shell around the scattering object. With this configuration a three-dimensional high resolution image of the contour of the object can be created. This ideal configuration gives an idea of the correspondence between the number of observation points, the distance between the points, the angle between the points, and the resulting measurement accuracy.

Relevant Solvable Problems

When a versatile measurement device for industrial automation use is to be designed the large array-configurations of high resolution SAR-methods can't be used for space consumption reasons[3]. The sensor unit configurations described in Chapter 3 can consequently only have a limited use of the SAR-methods. Because of this our measurements approach starts at the other end of the problem:

> –Given the sensor unit array, defined by the available space, what measurements, possibly utilizing a priori information, can be done?

This question forces us to look at some special cases where the geometry is possible to handle. From these special cases it will then be possible to define limits for allowable approximations to make the special solutions applicable. Two special cases can be formulated:

- A flat surface is in front of the sensor unit. The dimensions and the position of the surface is sufficiently large and centered to let all paths from the transmitter to the respective receivers go via the surface. These paths must be the natural specular paths. The problem if the natural reflection point is outside the surface is further discussed in Section 5.5 and is illustrated in Figure 5.8.
- A point scatterer is somewhere in the space in front of the sensor unit. The scatterer is covered by the transmitter lobe and will thus reflect an echo that will reach the receivers.

In these two highly idealized situations it is possible to measure spatial angle and position respectively. The measurements will be described in the following sections 5.2, 5.3 and 5.4. A discussion of limits and approximations then follows in section 5.5.

5.2 Two-dimensional Flat Surface Angular Measurements

An angular measurement is based on the fact that if an ultrasonic wave is sent towards a flat surface it is reflected in a specular reflection i.e. the angle of incidence equals the angle of reflection. This means that we can look upon the reflecting surface as a mirror. The transmitter lobe can be modeled as a number of beams of various intensity. In this model the beam that will hit the receiver can be constructed by searching for a reflection point that satisfies the law of specular reflection. The beam found in this way also describes the shortest path from transmitter to receiver via the surface. Because of this there can't be any problem introduced by diffraction causing a new wavefront that arrives earlier.

Consequently, a configuration as the one shown in Figure 5.1 with one transmitting element and two receiving ones can be used to measure the inclination of a surface. By measuring the time from transmission of a pulse to reception at transducers *a* and *b* the distance r_0 and the angle α can be calculated.

[3] The case when the measurement unit is placed in an industrial robot and the imaging process is allowed to control the robot is, of course, an exception. This movable sensor unit configuration can produce the necessary conditions for SAR-methods to be used. This is, however, not treated here but the methods described in [Mensa, 1991] may then be applied directly.

If the phase angle between signals from two transducers are measured when the sensor plane is in parallel with a base plane surface a calibration value can be stored. Using this value small deviations can be measured with high accuracy since the phase angle difference is easily detected. In a pulsed system this phase difference may cover several cycles of the signal. The measurement can, however, also be made with a continuous wave (CW) ultrasonic signal. An application of this technique in assembly of fog lamps for cars is found in [Klöör et al, 1988].

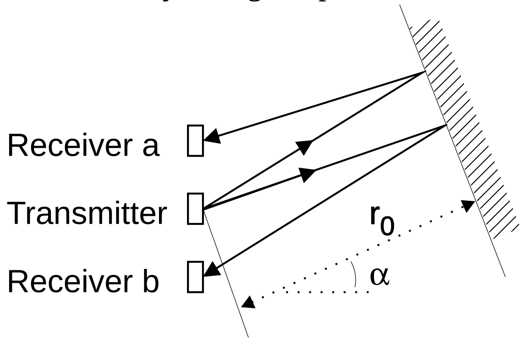

Figure 5.1. Angle measurement.

Before the detailed description of the measurements it is important to note that angular measurements are dependent on two major restrictions:

- The surface must have a measurement area that is large enough to reflect the beams indicated in Figure 5.1. Consequently it has to measure at least half the distance between the receivers. Furthermore, this minimum measure will only be sufficient provided that the object is placed exactly in the correct position.
- The measurement area must be within the transmitter lobe angle so that both beam reflection points are sufficiently "illuminated" by the transmitter.

These restrictions might seem a bit limiting at the first glance but will usually not cause any trouble. The usable lobe of transducer can often be approximated with expression (2.5). This means that for most elements in the 40-60 kHz band the lobe is about 30-50°. In the case of large signal amplitudes, significantly larger angles can be used. In the sensor units, described in Chapter 3, the distance between the sensors is less than 85 mm between the diagonal elements. This requires a surface with 43 mm diameter. If this is a too large dimension for the measurement task a sensor configuration with the elements located more closely is possible.

The simplest configuration to analyze is the two-dimensional case with symmetric receiver locations. In this case we assume that the receivers are placed at a distance of $d/2$ on each side of the transmitter as indicated in Figure 5.2. The transmitter is situated in the origin and the reflecting plane is described by ($y=-r_0$). The distance to the reflecting plane from the transmitter is r_0 and the angle between the sensor plane and the reflecting plane is α. The known (measured) variables are thus: a, b and d.

To reduce the number of unknown we first define:

$$p = x_a = -x_b$$

$$q = y_b = -y_a$$

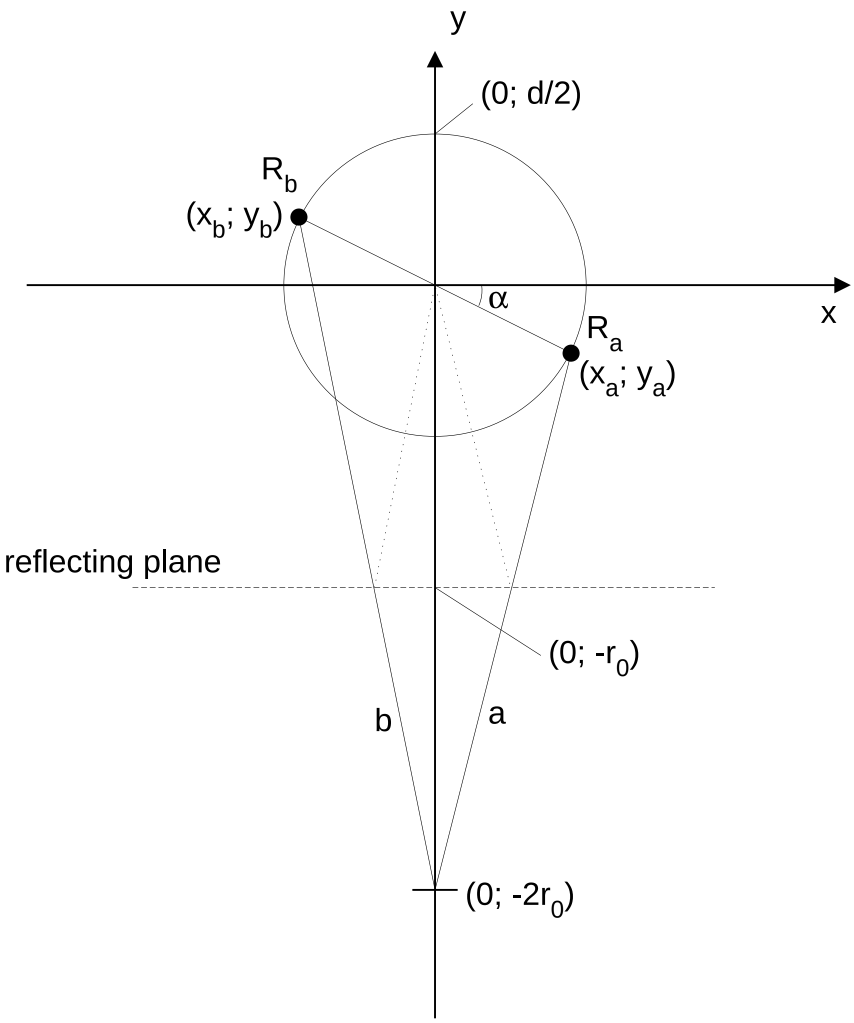

Figure 5.2 The transmitter is located in (0,0), while the receivers (R_a and R_b) are placed at a distance d/2 from the transmitter. The distances from the receivers to the point (0;-2 r_0) are called a and b respectively.

This makes it possible to use three orthogonal triangles to formulate the necessary equations:

$$p^2 + q^2 = \frac{d^2}{4} \tag{5.1a}$$

$$(2 \cdot r_0 - q)^2 + p^2 = a^2 \tag{5.2a}$$

$$(2 \cdot r_0 + q)^2 + p^2 = b^2 \tag{5.3a}$$

From the first expression it is possible to extract p^2 :

$$p^2 = \frac{d^2}{4} - q^2$$

If this is inserted in (5.2a) and (5.3a) the system is reduced to:

$$(2 \cdot r_0 - q)^2 + \frac{d^2}{4} - q^2 = a^2 \tag{5.2b}$$

$$(2 \cdot r_0 + q)^2 + \frac{d^2}{4} - q^2 = b^2 \tag{5.3b}$$

or

$$4 \cdot r_0^2 - 4 \cdot r_0 \cdot q + q^2 + \frac{d^2}{4} - q^2 = 4 \cdot r_0^2 - 4 \cdot r_0 \cdot q + \frac{d^2}{4} = a^2 \tag{5.2c}$$

$$4 \cdot r_0^2 + 4 \cdot r_0 \cdot q + q^2 + \frac{d^2}{4} - q^2 = 4 \cdot r_0^2 + 4 \cdot r_0 \cdot q + \frac{d^2}{4} = b^2 \tag{5.3c}$$

By subtracting and adding the equations, respectively, simpler expressions can be obtained.

$$8 \cdot r_0 \cdot q = b^2 - a^2 \qquad \text{(5.3c) - (5.2c)}$$

$$8 \cdot r_0^2 + \frac{d^2}{2} = a^2 + b^2 \qquad \text{(5.2c) + (5.3c)}$$

It is now possible to calculate the distance r_0 from (5.2c) + (5.3c).

$$r_0 = \sqrt{\frac{a^2 + b^2 - \frac{d^2}{2}}{8}} \tag{5.4}$$

Now that r_0 is known q can be calculated from (5.3c) - (5.2c). The angle α can then be expressed by looking in Figure 5.2.

$$q = \frac{b^2 - a^2}{8 \cdot r_0}$$

$$\alpha = \arcsin\left(\frac{2 \cdot q}{d}\right) \tag{5.5}$$

With the expressions (5.4) and (5.5) it is thus possible to calculate both the distance, r_0, and the angle, α from the distances a and b. A corresponding analysis but with asymmetric receiver locations is found in [Lindstedt, 1992].

5.3 Three-dimensional Flat Surface Angular Measurements

Real problems often require measurements of spatial angles rather than the two dimensional ones that were treated in the previous section. This is a more complicated measurement task, but many of the ideas from the previous section can be generalized.

Spatial angle measurements are described in various papers e.g. [Holmberg, 1992a] and [Klöör et al, 1988]. The measurements are usually based on a given sensor layout and contain lots of equal distances and symmetry axes. These conditions can however not always be satisfied due to demands for a compact measurement device. If the ultrasonic elements are used as both receivers and transmitters, the transmitter might be selected for each measurement and accordingly the sensor layout will change. This is the case with the configurations described in Chapter 3.

The computation method described here has no requirement for special symmetry or distances and is applicable to all cases where the transmitter and receivers can be placed in a plane. Of course all elements may not be placed along a line since such a configuration is only capable of measuring one angle. The assumed sensor layout is shown in Figure 5.3. The transmitting element is assumed to be placed in the origin and in the zero angle case the receivers are placed in the *x*-*z*-plane. The zero angle case is defined as a situation where the sensor plane and the reflecting plane are parallel.

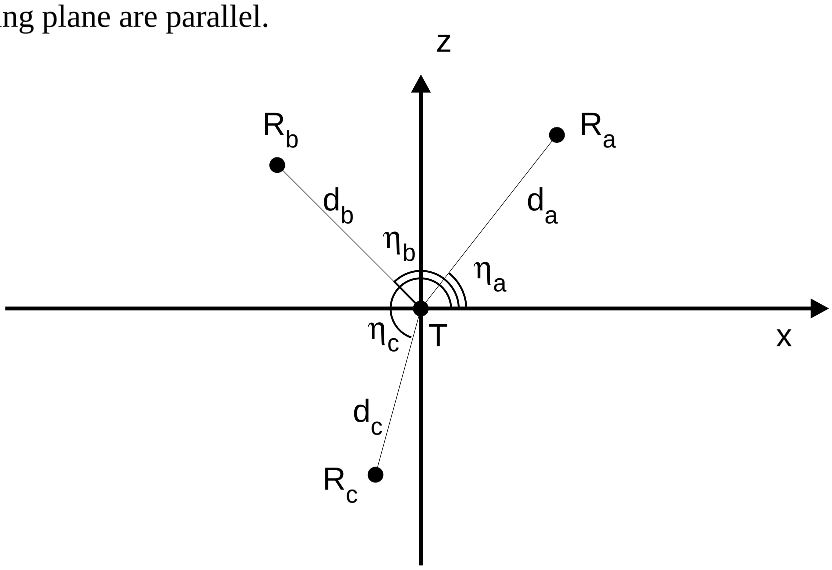

Figure 5.3 The transmitter and receiver positions in the x-z-plane. (Zero angle case is shown)

The rotation in this three-dimensional case is, like in the previous ones, made by working in a frame where the reflecting object is fixed and the transmitter is placed in the origin. The rotations can then be made around the *x*- and *z*-axes. We assume that the sensor plane (the *x*-*z*-plane) first is rotated a positive angle γ around the *x*-axis and then a positive angle α around the *z*-axis. The movements of the receivers from Figure 5.3 in a case where both γ and α are positive are shown in Figure 5.4. The reflecting plane, $y=-r_0$ is also indicated in the figure.

The locations of the receivers R_a, R_b and R_c in the zero angle case is described by the points (like in [Craig, 1989] *sine* and *cosine* are abbreviated to *s* and *c*):

$$P_{a0} = (c\eta_a;\quad 0;\quad s\eta_a)\cdot d_a = (x_{a0};\quad 0;\quad z_{a0})$$

$$P_{b0} = (c\eta_b;\quad 0;\quad s\eta_b)\cdot d_b = (x_{b0};\quad 0;\quad z_{b0}) \tag{5.6}$$

$$P_{c0} = (c\eta_c;\quad 0;\quad s\eta_c)\cdot d_c = (x_{c0};\quad 0;\quad z_{c0})$$

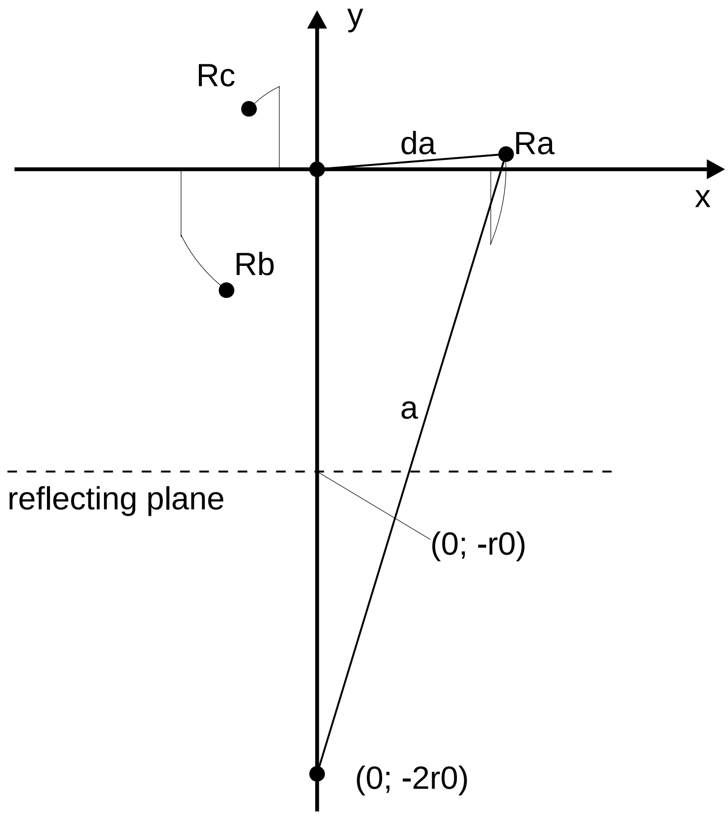

Figure 5.4 The measurement situation after two rotations

Two rotation matrices can now be used to perform the coordinate transformation of the receiver locations. After this transformation the positions for the receivers can be expressed as the points P_a, P_b and P_c. For the notation we refer to [Craig, 1989].

$$R_x(\gamma) = \begin{pmatrix} 1 & 0 & 0 \\ 0 & c\gamma & -s\gamma \\ 0 & s\gamma & c\gamma \end{pmatrix}; \qquad R_z(\alpha) = \begin{pmatrix} c\alpha & -s\alpha & 0 \\ s\alpha & c\alpha & 0 \\ 0 & 0 & 1 \end{pmatrix}$$

$$R_{xz}(\gamma,\alpha) = R_z(\alpha)\cdot R_x(\gamma) = \begin{pmatrix} c\alpha & -s\alpha c\gamma & s\alpha s\gamma \\ s\alpha & c\alpha c\gamma & -c\alpha s\gamma \\ 0 & s\gamma & c\gamma \end{pmatrix}$$

$$\begin{aligned} P_a &= R_{xz}(\gamma,\alpha)\cdot P_{a0} = \left(x_{a0}c\alpha + z_{a0}s\alpha s\gamma;\quad x_{a0}s\alpha - z_{a0}c\alpha s\gamma;\quad z_{a0}c\gamma\right) = \left(x_a;\quad y_a;\quad z_a\right) \\ P_b &= R_{xz}(\gamma,\alpha)\cdot P_{b0} = \left(x_{b0}c\alpha + z_{b0}s\alpha s\gamma;\quad x_{b0}s\alpha - z_{b0}c\alpha s\gamma;\quad z_{b0}c\gamma\right) = \left(x_b;\quad y_b;\quad z_b\right) \\ P_c &= R_{xz}(\gamma,\alpha)\cdot P_{c0} = \left(x_{c0}c\alpha + z_{c0}s\alpha s\gamma;\quad x_{c0}s\alpha - z_{c0}c\alpha s\gamma;\quad z_{c0}c\gamma\right) = \left(x_c;\quad y_c;\quad z_c\right) \end{aligned} \tag{5.7}$$

Since all rotations are made around the axes of the frame the receiver locations are always placed in a spherical orbit and the distances to the origin must remain the same. This means that the following is true:

$$\begin{aligned} x_a^2 + y_a^2 + z_a^2 &= d_a^2 \\ x_b^2 + y_b^2 + z_b^2 &= d_b^2 \\ x_c^2 + y_c^2 + z_c^2 &= d_c^2 \end{aligned} \tag{5.8}$$

Like in the two dimensional case another orthogonal triangle may be expressed for each point (*a*, *b* and *c* are as previously the length equivalent for the time from transmission to reception at each point),

$$\begin{aligned} \left(2\cdot r_0 + y_a\right)^2 + x_a^2 + z_a^2 &= a^2 \\ \left(2\cdot r_0 + y_b\right)^2 + x_b^2 + z_b^2 &= b^2 \\ \left(2\cdot r_0 + y_c\right)^2 + x_c^2 + z_c^2 &= c^2 \end{aligned} \tag{5.9}$$

From (5.8) we know that

$$x_n^2 + z_n^2 = d_n^2 - y_n^2$$

where *n* can denote one of the three indexes *a*, *b*, or *c*. If we insert this in (5.9) we can eliminate the *x*- and *z*-coordinates,

$$\begin{aligned} 4\cdot r_0^2 + 4\cdot r_0\cdot y_a + d_a^2 &= a^2 \\ 4\cdot r_0^2 + 4\cdot r_0\cdot y_b + d_b^2 &= b^2 \\ 4\cdot r_0^2 + 4\cdot r_0\cdot y_c + d_c^2 &= c^2 \end{aligned} \tag{5.10}$$

Using the translation between the zero angle coordinates and the actual ones we have previously expressed the y-coordinates as:

$$y_n = x_{n0}\cdot v - z_{n0}\cdot w \tag{5.11}$$

where

$$v = s\alpha$$
$$w = c\alpha s\gamma$$

If this is inserted in the equation system (5.10) we get:

$$\begin{aligned} 4r_0^2 + 4r_0\left(x_{a0}v - z_{a0}w\right) + d_a^2 - a^2 &= 0 \\ 4r_0^2 + 4r_0\left(x_{b0}v - z_{b0}w\right) + d_b^2 - b^2 &= 0 \\ 4r_0^2 + 4r_0\left(x_{c0}v - z_{c0}w\right) + d_c^2 - c^2 &= 0 \end{aligned} \tag{5.12}$$

In this system *a*, *b* and *c* are measured while r_0, *v* and *w* are unknown. All other parameters are constant. Since the system is not linear an analytic solution is not obvious. Alternatively this can be solved by iterative methods. The Newton-Raphson method is known to be both robust and efficient. To use this method we make some vector and matrix definitions:

$$R = \begin{bmatrix} r_0 \\ v \\ w \end{bmatrix} \tag{5.13}$$

$$A(R) = \begin{bmatrix} 4r_0^2 + 4r_0\left(x_{a0}v - z_{a0}w\right) + d_a^2 - a^2 \\ 4r_0^2 + 4r_0\left(x_{b0}v - z_{b0}w\right) + d_b^2 - b^2 \\ 4r_0^2 + 4r_0\left(x_{c0}v - z_{c0}w\right) + d_c^2 - c^2 \end{bmatrix} = \begin{bmatrix} f_1 \\ f_2 \\ f_3 \end{bmatrix} \tag{5.14}$$

$$\frac{\partial A}{\partial R} = \begin{bmatrix} \frac{\partial f_1}{\partial r_0} & \frac{\partial f_1}{\partial v} & \frac{\partial f_1}{\partial w} \\ \frac{\partial f_2}{\partial r_0} & \frac{\partial f_2}{\partial v} & \frac{\partial f_2}{\partial w} \\ \frac{\partial f_3}{\partial r_0} & \frac{\partial f_3}{\partial v} & \frac{\partial f_3}{\partial w} \end{bmatrix} = \begin{bmatrix} 8r_0 + 4x_{a0}v - 4z_{a0}w & 4r_0x_{a0} & -4r_0z_{a0} \\ 8r_0 + 4x_{b0}v - 4z_{b0}w & 4r_0x_{b0} & -4r_0z_{b0} \\ 8r_0 + 4x_{c0}v - 4z_{c0}w & 4r_0x_{c0} & -4r_0z_{c0} \end{bmatrix} \tag{5.15}$$

If we let R_n denote the *n:th* guess of the vector *R* a Newton-Raphson formulation of the system will be:

$$R_{n+1} = R_n - \left(\frac{\partial A}{\partial R_n}\right)^{-1} \cdot A\left(R_n\right) \tag{5.16}$$

The matrix inverse is a pure numerical operation and can be made with e.g. a L-U factorization method.

The main drawback when using iterative methods is often difficulties in finding good initial values. This is crucial since the number of iterations to get the desired accuracy depends completely on the initial value guess. A method for finding good initial values is therefore most valuable.

Referring to Figure 5.4 we consider the triangle a-d_a-$2\ r_0$. If the angles are small and $2\ r_0$ is much larger than d_a we can assume the angle between $2r_0$ and d_a to be orthogonal. Then r_0 can be estimated to:

$$\tilde{r}_0 = \frac{1}{2}\sqrt{a^2 - d_a^2} \tag{5.17}$$

This initial estimation can be done for each of the receiver locations. The error in the total estimation may be smaller if all points are included. Some practical experiments have shown that a geometrical mean of the estimations gives a good value,

$$\tilde{r}_0 = \frac{1}{2}\sqrt{\frac{a^2 + b^2 + c^2 - d_a^2 - d_b^2 - d_c^2}{3}} \tag{5.18}$$

This estimation can be compared with the 2D r_0 expression (5.4). It is more difficult to make an initial estimate of the angles α and γ. It is easier to make an indirect estimation of the angles by first estimating the receiver locations. Under the assumptions made for the r_0 estimation we can approximate the x- and z-coordinates to remain unchanged from the zero angle case to the actual one. Assuming that r_0 is much larger than the d_n-values we use an approximation of the displacement in the y-direction:

$$\begin{aligned} \tilde{P}_a &= \begin{pmatrix} x_{a0} & a - 2\tilde{r}_0 & z_{a0} \end{pmatrix} \\ \tilde{P}_b &= \begin{pmatrix} x_{b0} & b - 2\tilde{r}_0 & z_{b0} \end{pmatrix} \\ \tilde{P}_c &= \begin{pmatrix} x_{c0} & c - 2\tilde{r}_0 & z_{c0} \end{pmatrix} \end{aligned} \tag{5.19}$$

These points describe the estimated receiver plane. To obtain a vector that is orthogonal to this
plane we first define two vectors V_1 and V_2 in the plane and then compute the vector product of them,

$$\tilde{V}_1 = \tilde{P}_a - \tilde{P}_b = \begin{pmatrix} x_{a0} - x_{b0} & a - b & z_{a0} - z_{b0} \end{pmatrix}$$

$$\tilde{V}_2 = \tilde{P}_a - \tilde{P}_c = \begin{pmatrix} x_{a0} - x_{c0} & a - c & z_{a0} - z_{c0} \end{pmatrix}$$

$$\tilde{V}_1 \times \tilde{V}_2 = \begin{pmatrix} (a-b)(z_{a0} - z_{c0}) - (a-c)(z_{a0} - z_{b0}) \\ -(x_{a0} - x_{b0})(z_{a0} - z_{c0}) + (z_{a0} - z_{b0})(x_{a0} - x_{c0}) \\ (a-c)(x_{a0} - x_{b0}) - (a-b)(x_{a0} - x_{c0}) \end{pmatrix}^T = \begin{pmatrix} \tilde{n}_x & \tilde{n}_y & \tilde{n}_z \end{pmatrix} \tag{5.20}$$

Since this vector product is parallel with the normal vector of the plane estimates of α and γ can be calculated from the following expressions:

$$\tan\tilde{\alpha} = \frac{\tilde{n}_x}{-\tilde{n}_y} \tag{5.21}$$

$$\tan\tilde{\gamma} = \frac{-\tilde{n}_z}{\sqrt{\tilde{n}_x^2 + \tilde{n}_y^2}} \tag{5.22}$$

This can be summed up in the following algorithm:

Knowing the distances *a*, *b*, *c* and the receiver coordinates in the zero angle case it is possible to use (5.18), (5.20), (5.21) and (5.22) to compute initial estimates for r_0, α and γ.

Iterative use of (5.16) then gives an arbitrary precision. The variables in (5.16) are defined in (5.13), (5.14) and (5.15), while *v* and *w* are defined in (5.11).

5.4 Localization of Small Objects

The contrary situation to the spatial angle measurement is the object localization. Here the preconditions are quite opposite i.e. a small object in a limited area is required. The purpose is to determine the position of the object. In the ideal case the object is a point scatterer and no other objects are placed in the area. The localization situation is shown in Figure 5.5. To localize an object in space at least three independent receivers are needed.

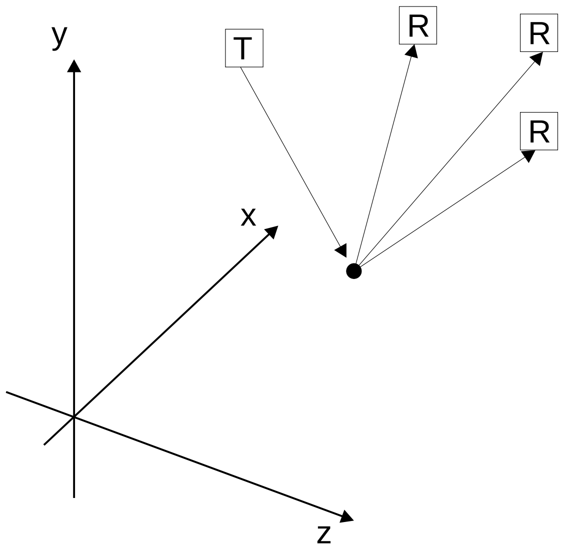

Figure 5.5 Localization principles

The model for solving the angle problem was constructed by an imaginary transmitter behind the reflecting surface. In the localization case we do not need any imaginary transmitter since all signal paths are very simple. From the transmitter a wave is sent towards the scatterer. Then the (point-)scatterer reflects the wave by sending beams in all directions. The beams aimed at the receivers goes directly to them. Because of this no other points than the elements and the scatterer are involved in the measurement. Consequently the problem can be formulated using only the following coordinates:

Point scatterer: $P = (x_p , y_p , z_p)$

Receivers: $A=(x_a , y_a , z_a)$
$B=(x_b , y_b , z_b)$
$C=(x_c , y_c , z_c)$

Transmitter: $T = (x_t , y_t , z_t)$

The three distances measured with the ultrasonic transducers, transmitter to receiver, are then called:

Distances: a, b, and c

It is then possible to express the relationships in a vector formulation as:

$$\begin{aligned} \left|\vec{T}-\vec{P}\right| + \left|\vec{P}-\vec{A}\right| &= a \\ \left|\vec{T}-\vec{P}\right| + \left|\vec{P}-\vec{B}\right| &= b \\ \left|\vec{T}-\vec{P}\right| + \left|\vec{P}-\vec{C}\right| &= c \end{aligned} \tag{5.23}$$

or in coordinates:

$$\begin{aligned} \sqrt{\left(x_t-x_p\right)^2+\left(y_t-y_p\right)^2+\left(z_t-z_p\right)^2}+\sqrt{\left(x_p-x_a\right)^2+\left(y_p-y_a\right)^2+\left(z_p-z_a\right)^2} &= a \\ \sqrt{\left(x_t-x_p\right)^2+\left(y_t-y_p\right)^2+\left(z_t-z_p\right)^2}+\sqrt{\left(x_p-x_b\right)^2+\left(y_p-y_b\right)^2+\left(z_p-z_b\right)^2} &= b \\ \sqrt{\left(x_t-x_p\right)^2+\left(y_t-y_p\right)^2+\left(z_t-z_p\right)^2}+\sqrt{\left(x_p-x_c\right)^2+\left(y_p-y_c\right)^2+\left(z_p-z_c\right)^2} &= c \end{aligned} \tag{5.24}$$

From this a coordinate formulation can be done but the resulting equation system is not readily solved. The unknown vector is P. Since P is needed in both the length-computations that include nonlinear operations a complicated equation system is produced. Because of this the system ought to be simplified if possible.

The first simplification is the same that was done in the angular measurements, to place the transducer in the origin. This is not a limitation but rather a selection of a reference.

The physical layout of the sensor units is very structured and does not require the general formulation in expression (5.23) and (5.24). All cooperating sensors are placed in quadratic patterns. This means that the sensor description in Figure 5.6 can be used. The sensor matrix is placed in the x-z-plane. The receiving elements are placed so that one is on the x-axis and one on the z-axis. The only parameter that has to be defined for the sensor units is the distance, *d*, between the elements. With the references used in Figure 5.6 the measurements are made with the objects in the space (y<0). With these assumptions the new coordinates for the transmitter and the receivers become:

Receivers: $A=(d,0,0)$
$B=(d,0,d)$
$C=(0,0,d)$

Transmitter $T = (0,0,0)$

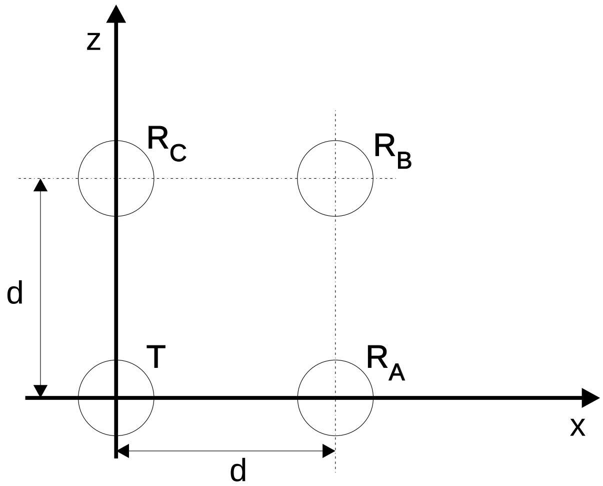

Figure 5.6 Specific sensor configuration for the used sensor units. Sensor matrix is viewed from the object side.

With the new coordinates expression 5.24 become much easier to handle:

$$\begin{aligned}
\sqrt{x_p{}^2+y_p{}^2+z_p{}^2}+\sqrt{\left(x_p-d\right)^2+y_p{}^2+z_p{}^2} &= a\\
\sqrt{x_p{}^2+y_p{}^2+z_p{}^2}+\sqrt{\left(x_p-d\right)^2+y_p{}^2+\left(z_p-d\right)^2} &= b\\
\sqrt{x_p{}^2+y_p{}^2+z_p{}^2}+\sqrt{x_p{}^2+y_p{}^2+\left(z_p-d\right)^2} &= c
\end{aligned} \tag{5.25}$$

This is not an equation system that is trivial to solve but with a combination of the use of the program packet MAPLE® for algebraic manipulation and hand calculations it can be shown that:

$$x_p=\frac{1}{2d}\cdot\left[d^2+\frac{a(b-c)(b+c-a)}{(b-a-c)}\right] \tag{5.26}$$

$$z_p=\frac{1}{2d}\cdot\left[d^2+\frac{c(b-a)(a+b-c)}{(b-a-c)}\right] \tag{5.27}$$

The expression for y_p is somewhat more complicated but can be expressed using x_p and z_p as:

$$y_p = -\frac{1}{2a} \cdot \sqrt{\left(d^2 - a^2\right) \cdot \left[4 \cdot x_p \cdot \left(x_p - d\right) + \left(d^2 - a^2\right)\right] - \left(2a \cdot z_p\right)^2} \quad (5.28)$$

This is one of two possible solutions but since we know the transmission direction this is the relevant one. With these expressions, (5.26-28), it is a straight-forward operation to compute a position from the three measured delay-times that are translated into distances.

5.5 Ambiguities due to Object Geometry

The measurement situations in the previous sections are all highly idealized. Concerning the angle measurements it is feasible that flat surfaces may occur in practical measurement situations. These surfaces are, however, not infinite but of limited size. This makes their position crucial if the method still will be applicable under ideal conditions. It is therefore of great importance to determine the limits concerning size and position of the reflecting surface.

Contrary, in the object localization situation the ideal conditions are never met since the method requires a point scatterer. The approximation that is used has to fulfill two conditions:
1) The shape of the object must scatter the incoming wave over a wide reflection angle although the reflection is specular.
2) The size of the object must be so small that the reflected beams can be considered originating from one point.

In both cases there is a need for limits and or error estimation when the methods are to be used in practice.

Angle Measurement Limitations
The basic condition for the dimensions of the reflecting flat surface in an angle measurement is that all ”natural reflection points” must be on the surface. The use of an imaginary transmitter behind the reflecting surface makes it easier to find these points. As previously described the transmitter is placed on a line that goes through the transmitter and is orthogonal to the surface. The distance between the imaginary transmitter and the plane is the same as between the real transmitter and the plane. The scenario is shown in Figure 5.7. The reflection points are defined by the intersection of each line from the imaginary transmitter to each receiver. These points have to be on the real surface. As can be seen the in Figure 5.7 a projection of the receiver array can be done on the surface. The view-point is the imaginary transmitter.

In the two-dimensional case described in Figure 5.2 the exact distance between the reflection points can be expressed as:

$$d_{refl} = \frac{2 \cdot r_0^2 \cdot d \cdot \cos(\alpha)}{4 \cdot r_0^2 - \frac{d^2}{4} \cdot \sin(\alpha)} \approx \frac{d}{2} \cdot \cos(\alpha) \quad (5.29)$$

Two very likely assumptions make the approximation possible:
- the measuring distance, r_0, is significantly larger than the transducer element distances, d;

- the measurement is done at small angles.

Under these assumptions the expression tells us that the distance between the reflection points is slightly less than half the distance between the receivers. This calculation can also be extended to the three-dimensional case but a more practical approach is to make the geometric construction in Figure 5.7 and the knowledge that the reflecting surface is about half the size of the receiver matrix.

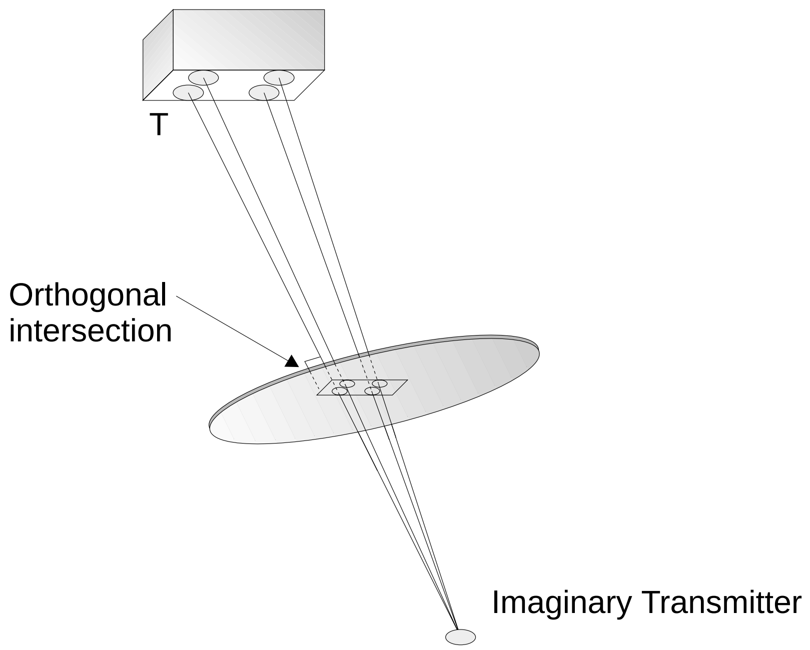

Figure 5.7 The angle measurement scenario. The reflection points can be constructed by a projection of the receivers on the reflecting surface.

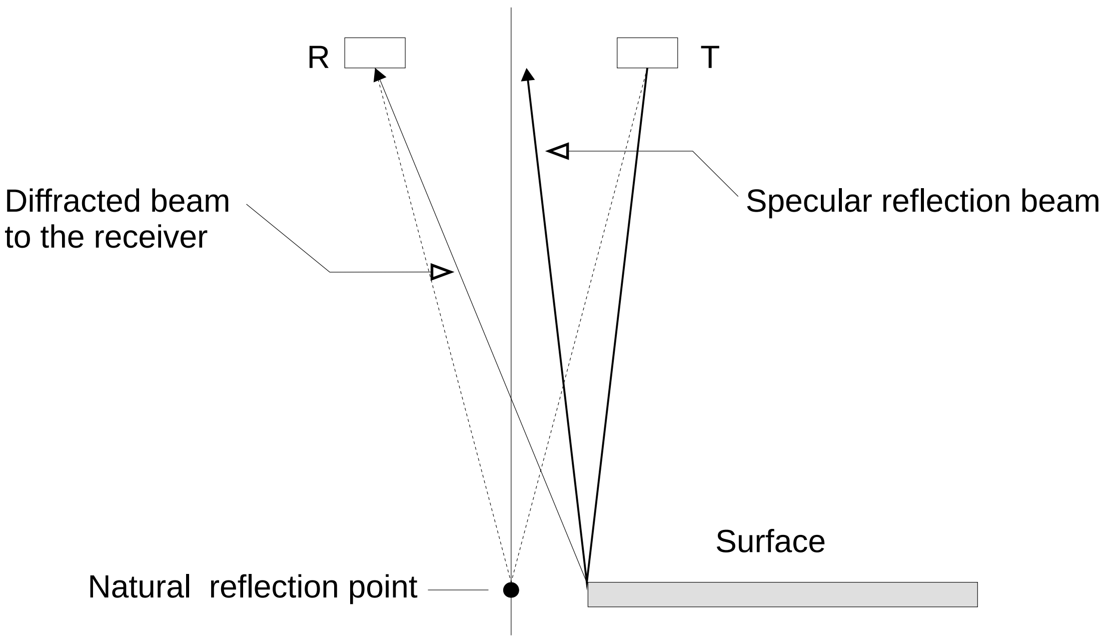

Figure 5.8 The natural reflection point is outside the surface. The reflection takes place at the edge.

What will then happen if the surface is too small? The measurements will of course not work properly but a result is usually produced if the natural reflection point is just slightly outside the surface. As can be seen in Figure 5.8 a reflection is made on the surface just close to the edge. This will not be a specular reflection from the receiver point of view. A sufficient wave is usually produced anyway due to diffraction. Since the new reflection point is not on the shortest path between the transmitter and the receiver the pulse is delayed. This delay makes the angle measurement algorithms to think that this part of the surface is more distant and this affects the angles in a corresponding way.

Object Localization Limitations

The object localization method is actually a point scatterer localization method. Because of this all practical application of this method is an approximation. The best object to approximate a point scatterer is usually a spherical object with a hard surface. The size has to be highly limited since the object is approximated with a point. This, on the other hand, means that the incoming wave is reflected at a very small surface. The reflected wave is, due to the convex shape, widely diffracted. As a consequence, the received signal amplitude is extremely low. Accordingly, the measurement conditions, concerning noise and disturbances, are not very favorable from the start. This limits the use of the measurement from a practical point of view.

To give an analytical expression that describes the error introduced by the size of the object is a difficult task. The computations of the position depend on many parameters and it is very difficult to give simple expressions for their individual influence. To get an impression of error magnitude it is probably more adequate to examine a practical example numerically.

We assume that the object that will be localized is a rod with a circular cross-section. The circular side is heading towards the sensor unit and will thus form a circular reflecting surface. The diffraction from an object of this type can be improved if the edge is slightly rounded. We assume that this fulfill the wide scattering condition. For the geometrical setup we use the electrostatic unit and place the scatterer at a distance of half a meter centered to the sensor unit. The diameter of the rod is now assumed to be 0, 2, 5, 10, and 20 mm. The experiment seen in the y-axis direction is shown in Figure 5.9.

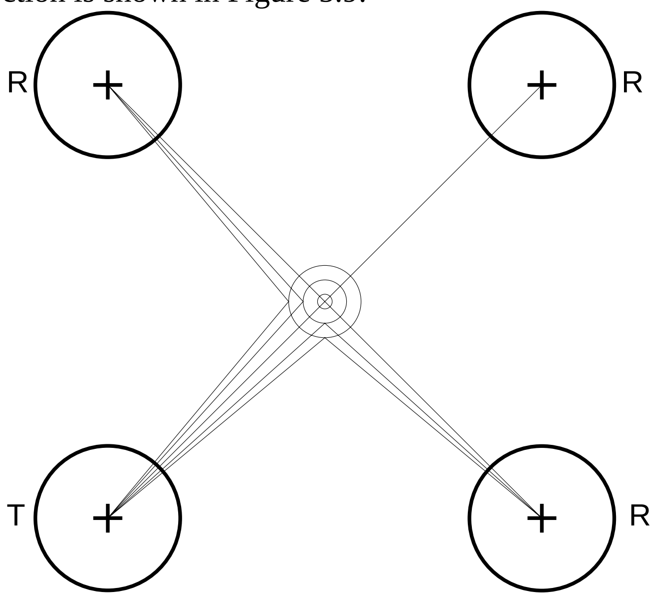

Figure 5.9 Different paths (shortest) from the transmitter to the receivers as the scatter size is increased

Rod size	Reflection point a-beam	Reflection point b-beam	Reflection point c-beam	Error in x and z	Error in y
0	(30,-500,30)	(30,-500,30)	(30,-500,30)	0	0
2	(30,-500,29)	(30,-500,30)	(29,-500,30)	-1.0	0.0
5	(30,-500,27.5)	(30,-500,30)	(27.5,-500,30)	-2.4	0.0
10	(30,-500,25)	(30,-500,30)	(25,-500,30)	-4.6	0.0
20	(30,-500,20)	(30,-500,30)	(20,-500,30)	-8.3	0.1

Table 5.1 Numeric calculation of position estimation error. (All values in millimeters)

In this calculation it is assumed that there is no problem finding the echo signal front. This is of course not true. The problem was discussed in Chapter 5 and several methods were described. In the current case we are dealing with low level signals with lots of noise. This makes methods like matched filters suitable. It is then important to note the difference between different kind of reflections. A spherical scatterer usually doesn't destroy the transmitted pulse shape although the amplitude is drastically reduced. When a scatterer like the suggested circular rod-end is used it is quite feasible that large variations in signal amplitude and shape can be created in various directions. This makes an accurate detection of the echo front somewhat more complicated. This may result in significantly larger position estimation errors than indicated in the table.

5.6 Experimental Results

For most measurements in this chapter it is difficult to express absolute accuracy since very much depends on the measurement conditions. In general it can be stated that angular measurements with sufficiently large reflecting surfaces can be made extremely accurate. With piezoelectric transducers accuracy better than 1° is easily achieved. The corresponding value for electrostatic ones is about 0.5° (the phase difference in the signals correspond to a few samples). The difference in accuracy is due to the harmonics of the piezoelectric elements and that the electrostatic ones have less individual variations. Before the measurements the units should be calibrated with a surface perpendicular to the acoustic axis of the elements. This makes it possible to store a phase offset that can be used in the following measurements. This offset is crucial if a good absolute accuracy is required. Initial phase offsets are easily created in the sensor mounting or in the electronic circuits since the measured phase differences correspond to fractions of millimeters. An offset compensation is an effective way of dealing with this problem.

The usable measurement angle is a bit different between the sensor units. For the piezoelectric unit angles up 30° can be measured while the electrostatic elements can be used up to about 25°. The larger element diameter resulting in a more narrow lobe reduces the measurement range.

Localization experiments have been done in the master thesis work [Rogdahl, 1994]. In these experiments the piezoelectric unit was used but unfortunately the localization was not very successful. A number of mistakes made in these experiments have later been discovered but the problems with the sensors were so big that no further attempts have been made using this unit.

More successful localization measurements have been done with the electrostatic unit using small spherical objects (10-15 mm diameter) at a distance of about 40 cm. A transmitted pulse with 4 cycles of a 50 kHz sine is used. The echo amplitude in these measurements is extremely low. Consequently the signal-to-noise ratio is poor and the only usable echo detection method that can give accurate distance information is matched filter peak detection. Using these tools together with the expressions (5.26-5.28) it has, however, been possible to localize the sphere with an error of less than 10 mm (typically 5 mm). The errors are primarily found in the x- and z-coordinates while the error in the y-coordinate seldom exceeds 1 mm. Measurement data is shown in Figure 5.10. The sphere (15 mm diameter) is placed in the position (20,-392,50). Using the shown data the measured position is (26,-391,47).

The localization measurement has two major problems. The first one is the low-level signal that reduces the accuracy. The second is the limited measurement area. Since the signals are weak the only usable area is approximately the one that is ***covered by the 3dB-lobe of all the elements***. Since this lobe for the used elements is up to about 10° from the acoustic axis of the element this area is small. At the distance 40 cm it is about the same size as the sensor matrix i.e. a little less than 1 dm^2.

The experience from these localization experiments is that such measurements are possible to do but seem to have a limited accuracy and are apparently sensitive to high levels of noise and disturbances. The measurement area is small which reduces the number of possible applications. Lots of improvements, especially concerning the used transmission signal, can probably make the accuracy and reliability better.

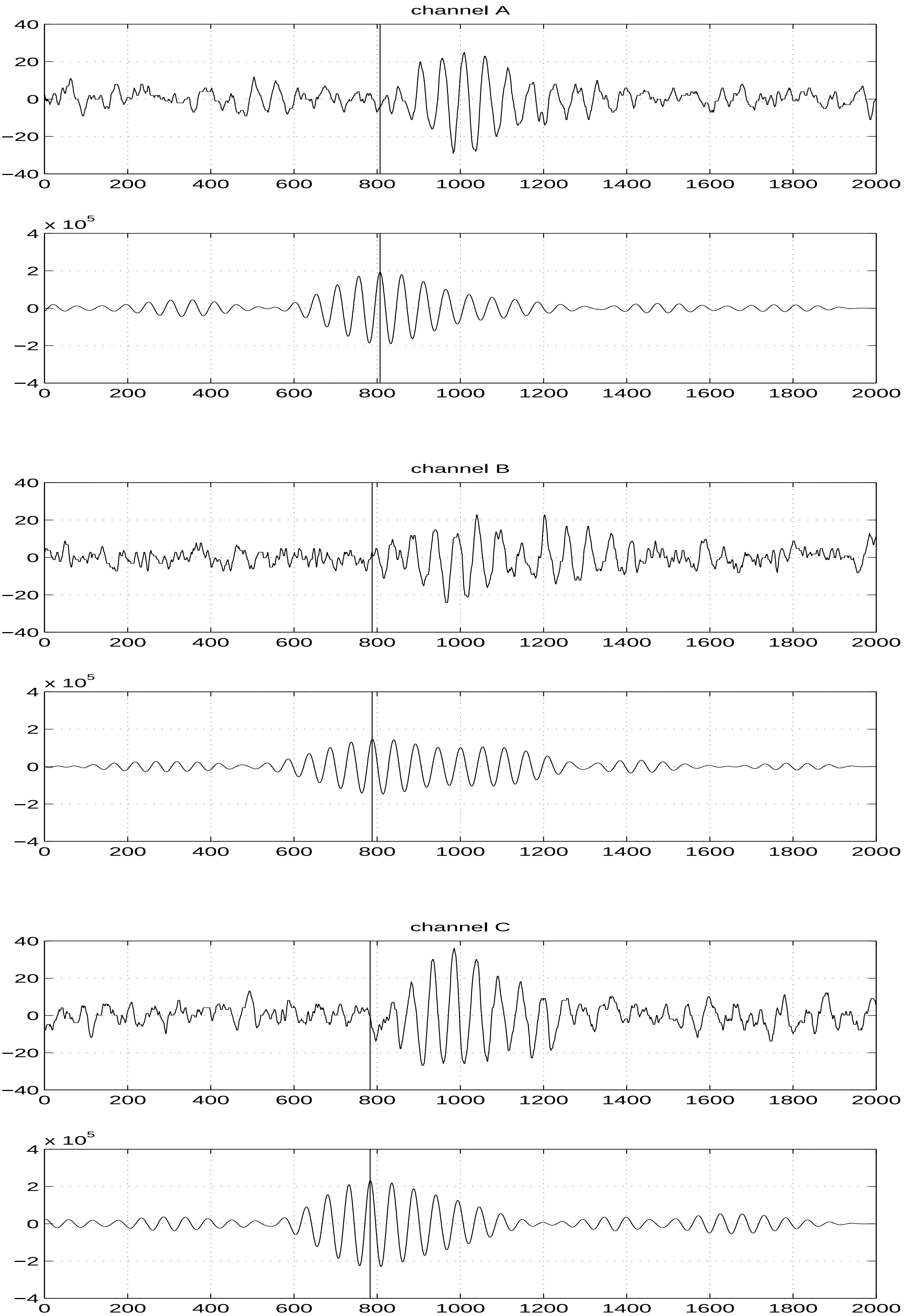

Figure 5.10 Sampled and filtered data from a localization measurement. A matched filter is used. The maximum peaks in the filtered outputs, i.e. the beginning of the echoes, are marked.

Object Recognition 6

This chapter describes the possibilities and limitations when object recognition is used as a tool in industrial automation applications. Various scenarios concerning a priori knowledge of the object position and orientation are discussed together with methods based on different amount of sensor information. Interpretation methods based on the information from the current sensor units are presented

6.1 Introduction

In the previously described measurements the distance information is the important value that is extracted from the echo signals. Many reflecting objects are complex and consist of several scattering surfaces at different distances with various sizes and orientations. For distance measuring purposes this is considered as a disadvantage since it makes the echo shape complex and the interpretation more complicated. On the contrary, this echo information can be used as a signature of the reflecting object. A comparison of signatures may give sufficient information to separate one object from another or even from a large amount of different objects. This is the basic principle for the object recognition measurements. The idea of this type of recognition is shown in Figure 6.1.

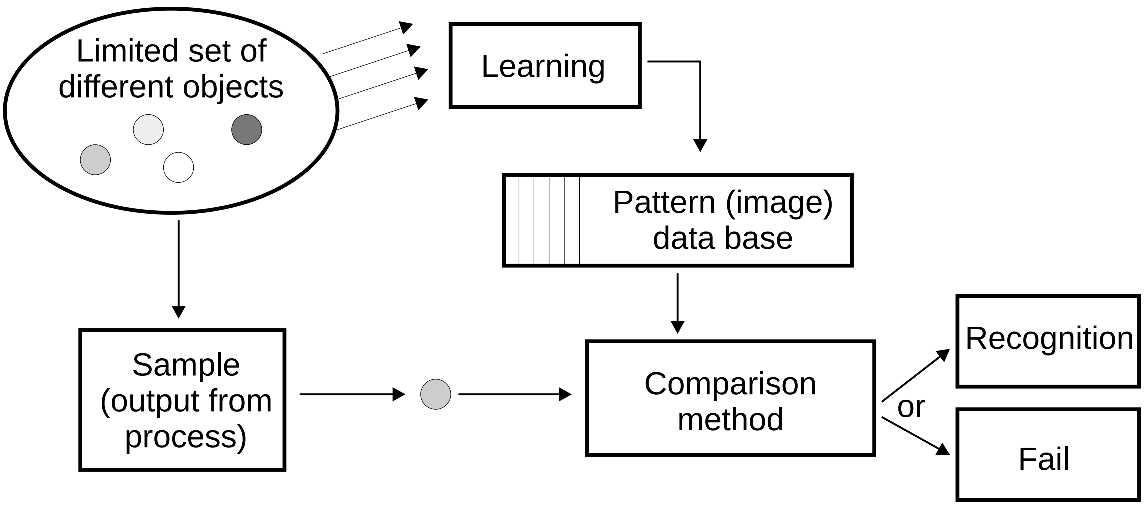

Figure 6.1 Recognition principles. First operation: Create database from known objects. Second operation: Recognize the current object or fail.

A large number of possible applications for the recognition in industrial automation is easy to foresee. The examined object in this measurement can be complex. This means that it may very well consist of sections of large work-pieces, machine parts, or tools. Consequently, supervision, quality control and final inspection are examples of operations where the recognition can play an important role.

Related work in this area can be found in a wide range of complexity. An example of the most uncomplicated single sensor methods is described by [Regtien and Hakkesteegt, 1988]. [Watanabe and Yoneyama, 1992] presents a sensor-array based method with a neural network for the signal interpretation. The theoretic background for recognition of corners and edges in three dimensions is described by [Hong and Kleeman, 1992]. Furthermore many of the patterns obtained when images of radar echoes are created is applicable also on ultrasonic signals. This is described in chapter four of [Mensa, 1991]. When the methods described in the literature are compared it is apparent that their initial amount of sensor-data is very different. The simpler methods[4] use only one sensor in one position while the more complicated require measurements from sensor matrixes or even from sensors rotated around the object. This has a large impact on the created pattern or object image. The simpler pattern methods are only able to recognize the object from one direction while the more complicated ones can produce a two- or three-dimensional image. The produced images of the objects differ a lot between the methods. The simpler methods are oriented towards comparison of patterns that are difficult to relate to the geometry of the object. This is often not the case for the more complex methods. They often produce object images that have an obvious relationship to the object geometry.

The sensor units described in Chapter 3 are in this sense able to make low- to medium-complexity measurements. The 200 kHz sensor in the piezoelectric unit is suitable for high resolution measurements because of its small wavelength. However, this is a single sensor; thus it will only be able to perform the simple type of measurements. The four element arrays in both types of sensor units can be used with a slightly higher degree of freedom since more measurement points are available. This will be further discussed in the measurement method descriptions.

The simplest form of recognition measurement, the one sensor case, is with a straight forward pattern recognition method forced to fail if the object is rotated or translated in a way that changes the echo from the object. In many applications this is not a problem since the object position and orientation are known and only the shape is of interest. However, in other applications small deviations in position may occur, e.g. after transportation by a conveyor belt. In some of these cases the orientation may also be of interest for the next operation e.g. grasping by a industrial robot manipulator. Consequently, a successful handling of these situations makes the recognition a much more valuable tool. In the following sections these problems will be discussed.

6.2 Recognition of Objects in a Fixed Position

Recognition using 200 kHz piezoelectric sensor

When small objects are to be recognized a shorter wavelength is an advantage since this increases the depth-resolution. The disadvantage is that the measurement has to be made at a short distance. With the used sensor the measurements can be made from 15 up to 30 cm. The size of the recognized object area is also limited by the transducer lobe. Centered objects with diameters up to 5-10 cm, depending on the distance, can be recognized. The significance of different reflecting surfaces is, of course, depending on the surface inclination and size. However, the basic rule is that the closer the surface is to the acoustic axis the more significant it is for the recognition.

[4] Simple in this case means that the practical measurement arrangements are simple. The used interpretation method may very well be highly sophisticated.

Initially the sensor is positioned over the object that is to be recognized and the height over the working area is constant during the data acquisition phase. Consequently the position of the sensor vs. the object will be about the same in the learning phase and in the recognition phase. Furthermore, the size of the object should allow illumination by the sensor lobe. All object parts that are to be used for the recognition have to generate echoes that can be measured by the sensor.

The 200 kHz sensor is used both for transmitting and receiving. All the signal amplitudes, *y(kh)*, are sampled and stored. For a recognition analysis it is unsuitable to use the whole data set if we know in what part of the sequence the object echo is located. Consequently the recognition is done on a subset of the sampled data. Further reduction of the amount of data can of course be done by grouping the samples. From a physical point of view there are natural ways of grouping the data. Because of this we first study the following steps in the data treatment.

An ultrasonic echo is never 100% reproducible. Spikes and disturbances interfere with the signal during the sampling. The position of the sensor vs. the object is never exactly the same. It is obvious that statistical methods have to be used for the data comparison. Since a sample-by-sample comparison is not meaningful due to these problems we have chosen to make a *shape* comparison. An envelope detection like in [Regtien and Hakkesteegt, 1988] could be a possibility but since a noise eliminating peak detection requires interpolation this is computationally inefficient

The chosen method is an *energy* oriented one. An *rms*-value of the incoming samples can easily be calculated. This means that the influence of large signal amplitudes (real echoes) is greater than that of small ones (often noise). The *rms*-value has to have a low sensitivity for the phase angle in the ultrasonic signal. Therefore the number of samples in the *rms* calculation must be chosen with care. If the number of samples of the transducer signal is close to a multiple of one half period the mentioned sensitivity becomes low.

The *rms*-value can be viewed as a quadratic moving average (MA) filter. By simply filtering the transducer signal through the MA filter a too long data vector is produced. In order to decrease the size of the vector that consists of filtered data it is decimated.

One filtered value r_p is obtained by calculating the *rms*-value over one period of the transducer signal, (one period for the 200 kHz transducer is 5 µs) which we define as the *rms* value of n_m samples. Typically n_m is of the order 12. For the period with index p this means

$$r_p = \sqrt{\sum_{k=p \cdot n_m}^{n_m \cdot (p+1)-1} [y(kh)]^2} \tag{6.1}$$

We can now define a *signature* vector S, consisting of n values,

$$S = [r_0, r_1, \ldots r_{n-1}] \tag{6.2}$$

Such a definition of S has proven to give good results. The S vector is a compact description of the object signature in terms of energy vs. time in the ultrasonic echo. For a sequence corresponding to about 100 mm using the 200 kHz signal a vector of about 60 (=n) values is produced. In Figure 6.2 some objects are shown together with their echoes and their calculated S-vectors.

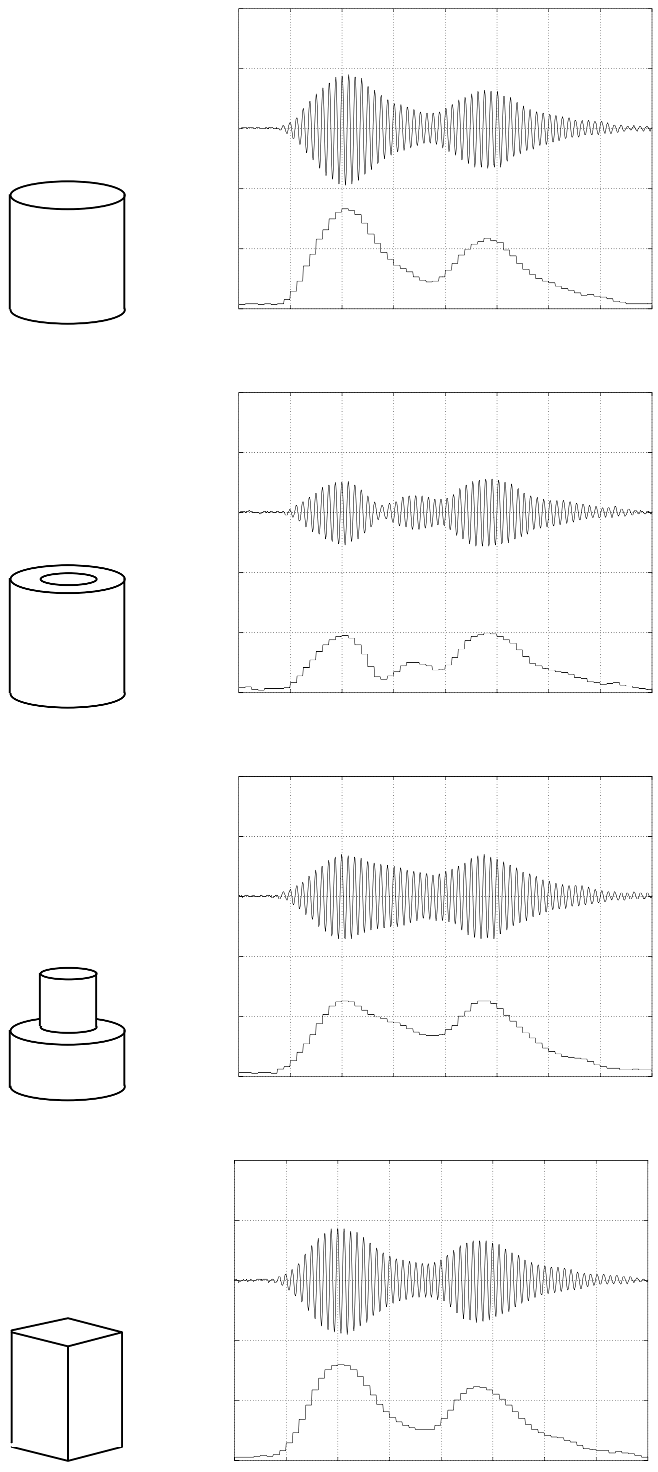

Figure 6.2. The figure shows four objects that are recognized viewed from above. The upper curves in the diagrams are the sampled values, y(kh). The lower ones are the calculated signatures, S. The first and the last objects have the same area at the same height towards the sensor. Consequently their echoes look very similar in the diagrams. However, a numerical comparison of their S-vectors makes them separable.

A comparison of two echo signatures, S_a and S_b, can be made by calculating the sum:

$$Q_{ab} = \sum_{p=0}^{n-1} (r_{pa} - r_{pb})^2 \tag{6.3}$$

where r_{pa} is the element r_p in vector S_a and r_{pb} is the element r_p in vector S_b. The lower the sum the better the fitting. The computer can store the signature vectors for a limited number of objects. From this data-base an unknown object can be recognized as the one with the smallest Q-sum. After some experience an absolute limit for acceptance can be drawn. In Figure 6.3 the energy functions from an recognition experiment with three objects in the data-base are shown.

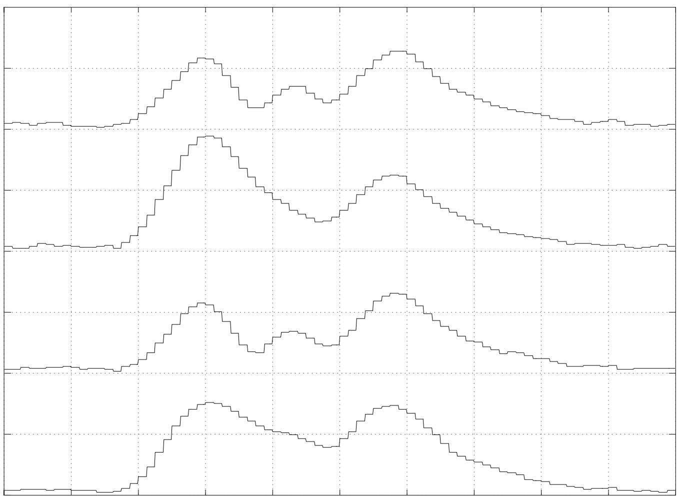

Figure 6.3. Comparison of different signature curves, S, for object recognition. (Offset added to separate the curves.) The upper curve shows an unknown signature that can be fitted to one of the three lower ones from the database.

About 20 arbitrarily chosen objects, mainly normal machine parts, were included in the experiments. Many of them are geometrically quite similar. Some have surfaces at the same height, others have surfaces of the same size at different heights (height differences $\approx$ 1 mm). For objects that are obviously different the recognition procedure has a very high confidence. If the object is displaced about 2-4 *mm* the object recognition is still successful.

Consequently the resolution of this method is so good that it can be used in a dynamic operation. Recognition of objects with large geometrical differences can be made even if the location is not known precisely. For accurate recognition of similar objects the position has to be exactly known. (As an example of the precision measurements it has been shown a statistically significant difference between the signatures from the two sides of a coin.)

All presented measurements are made in a laboratory environment with a favorable disturbance environment. To test the method additional disturbances have been purposefully added. The results have been encouraging.

Recognition Using Electrostatic Sensors

The recognition principle described for the 200 kHz sensor is also applicable for the other types of sensors. The piezoelectric 40 kHz sensors are difficult to use with high accuracy at shorter distances but can be used for long range recognition. However, they have in this aspect no major advantage compared to electrostatic sensors.

When the electrostatic elements are used, instead of the 200 kHz piezoelectric one, in a recognition purpose the comparable distance and object size ought to be larger. This is a natural consequence since the operating frequency is lower. Such experiments have been done with equally good or, even better, recognition probability. The signal quality is much better for these elements. Since the repeatability and reproducibility are also better a more interesting challenge is to try to solve the same measurement problems as with the 200 kHz element. The immediate consequence of this is that longer wavelength makes changes in the signal to appear differently. Using a lower frequency (like 40-60 kHz), i.e. a longer wavelength, makes the signal to behave differently. Using the higher frequency an echo from a new reflecting surface would appear in a following cycle. In the lower frequency case this echo may appear in the same cycle as the first. These echoes must now be detected within the current cycle if the precision shall be maintained. It is difficult to theoretically predict what this will do to the recognition method.

Because of this, experiments with the electrostatic unit were made. Samples from the resulting echoes and energy functions are shown in Figure 6.4. The objects are the same as in the measurement shown in Figure 6.2. They are, however, not placed on a reflecting surface but on a fixture that is invisible to the sensors. This is done to eliminate the echo from the surface that would have been too large compared to the object echoes. The reason for this is that we are now using elements with wider lobes in both transmission and reception. The influence on the actual recognition process is small.

The measurement was made at 55 kHz with 4 cycles transmitted. This frequency was selected as the one that gave the best recognition results in the interval 40-60 kHz. Actually, the recognition results were *better than those with the 200 kHz-sensor*. This means that the probability for correct recognition of echoes from very similar objects like e.g. the cylinder and the cube increased. Among the possible reasons for this there are two that seem likely:

- The direct response of the electrostatic element makes it more sensitive to small changes in the reflecting surface. The element is ”faster”.
- The frequency is optimal for this kind of objects. A corresponding matching can’t be done around 200 kHz since the element has an extremely narrow bandwidth.

Probably the truth is a combination of these reasons. Further experiments on a larger variety of objects may tell if the result is valid in a wider sense. It is, however, too early to state that the electrostatic element should be generally better for recognition purposes. The higher frequency might have a larger impact on the result when other objects are used.

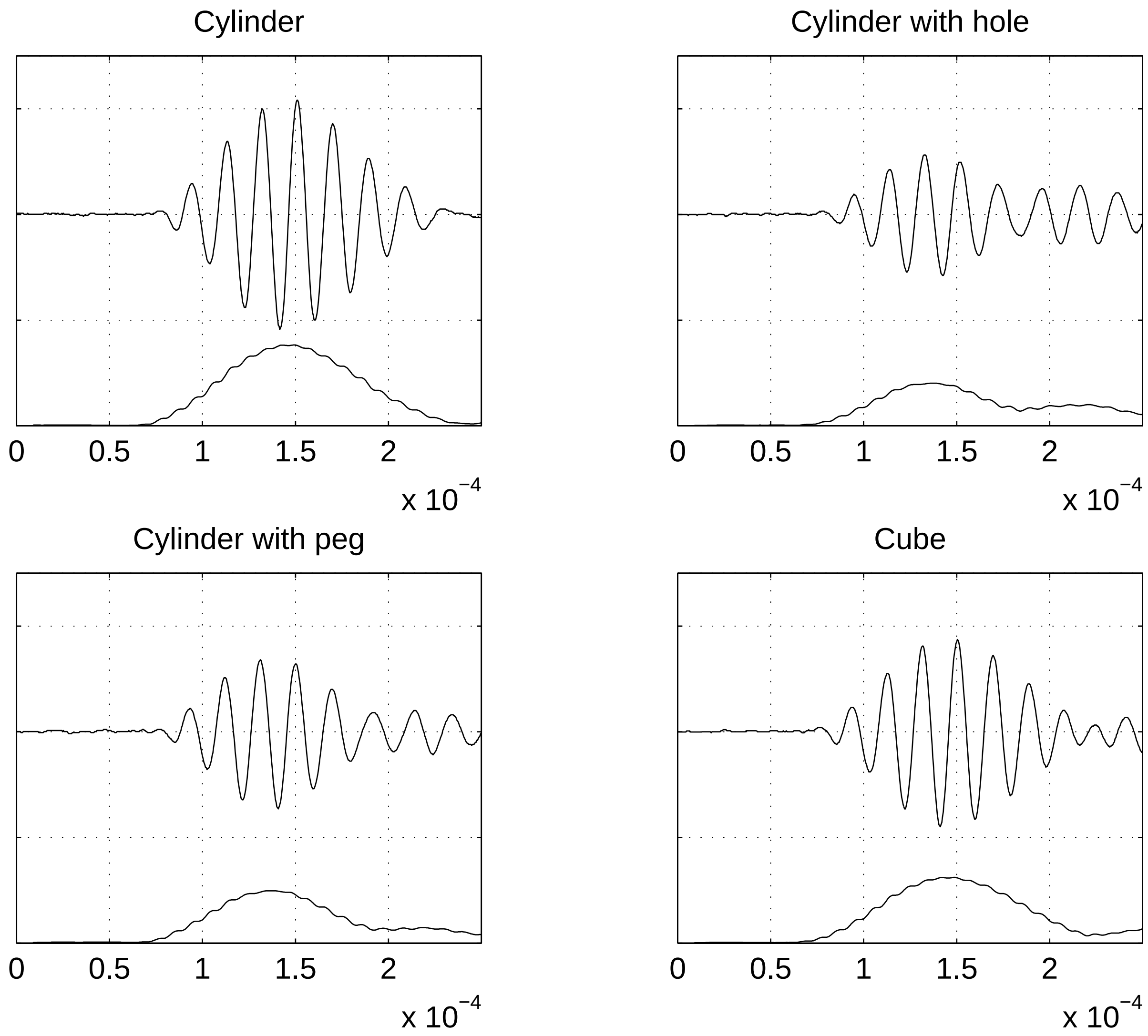

Figure 6.4 Recognition of the same objects as in Fig 6.2. Electrostatic elements are used. The frequency is 55 kHz and 4 pulses are transmitted.

Limitations of the ”rms-method”

The previously described recognition method can be denoted as the ”rms-method” since the recognition is based on a function that is created from the mean of the squares of the echo sequence. This is a principle that seems quite feasible when the idea of recognition based on ultrasonic echoes first comes to mind. The shape of the echo envelope is utilized and the impact of spurious disturbances is reduced. However, after some experiments it is soon obvious that some of the most valuable information in the echo sequence is lost, i. e. the phase of the signal. The most sensitive way to detect the contribution from a new reflecting surface in the signal is to study the phase curve.

The risk of fatal interference is always present when monofrequency measurements are made. This results in zero-level amplitude and consequently are these objects difficult to recognize. The improvement of the recognition result in the case with electrostatic elements and a frequency change shows how sensitive the result can be to such changes.

Recognition problems often shows up when the recognition is done using old reference data that was collected at a previous measurement. The reason for these problems is usually that the temperature has changed. This affects the speed of sound and consequently the phase angle of the echo-signals that are summed from different surfaces. The effect on the ”rms-function” can be highly significant.

Even if the ”rms-method” is a comparably robust method these limitations show that improvements are needed. Some of the problems will be treated in the following sections and chapters.

Applying an I/Q-demodulator to the recognition

One of the limitations of the ”rms-method” is that it doesn’t use the phase information. A tool that extract this information in a convenient way is the I/Q-demodulation, described in Chapter 4. The output from the demodulator is an amplitude- and a phase-function. The amplitude function can be used instead of the ”rms”-sequence. The phase function then gives a new opportunity to also compare the phase shift in the signal.

There are a number of restrictions that has to be obeyed when this method is used. One of them is that the phase information is not valid when the signal is close to the noise level. This means that a reference phase must not be fixed until the real echo has begun. If silent periods exist in the echo sequence more than one reference may be necessary.

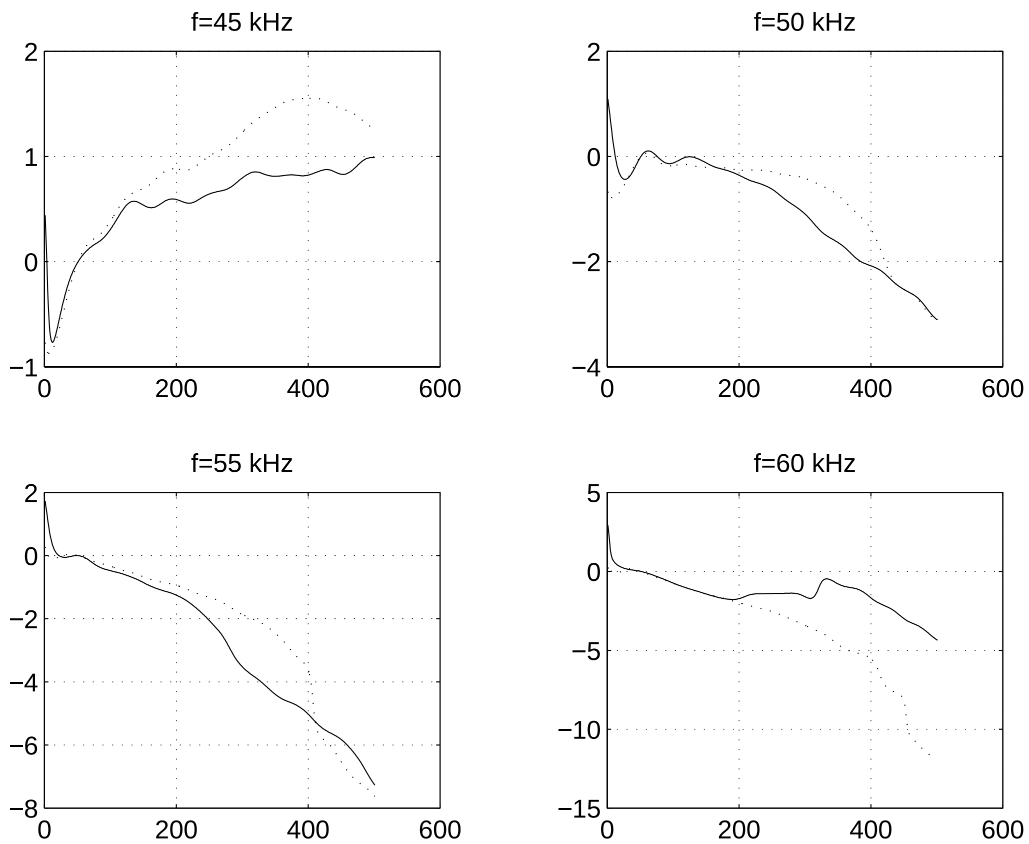

Figure 6.5 The phase curves for the cylinder with a hole and the solid cylinder (dotted) are shown. The diagram time interval starts at the beginning of the echo pulse. Four different transmission frequencies are used as indicated. (Units are samples and radians respectively.)

It is also important to notice that the demodulator is based on a low-pass filter. This introduces a short but often noticeable delay in the signal. However, the main delay problems when the phase is used for recognition is not in this area. The reason why phase changes are slow and often difficult to detect is found in the composition of the echo signal. First of all the transducer elements are not ideal. Immediate response is not available in neither the transmission nor the reception. This gives the transmitted signal an envelope that is increasing during the first cycles. Secondly a summing of the echo signal is made when echoes from different surfaces interfere. In this summing parts with different delay of the transmitted signal are included. These parts are softly taking over the main role in the echo since the envelope of the transmitted signal is not a rectangular function. This means that the phase shifts are soft and not easy to make a sharp recognition of. In Figure 6.5 the phase curves for two objects are shown at four different frequencies. The objects are the same that were used previously in this chapter.

Few cycles are often used in the transmitted signal to avoid making the echoes from different surfaces ”melt together”. However, this also means that all echo sequences can be considered transient. Since the I/Q-demodulation assumes that the initial transient should have disappeared and only the stationary oscillations remain, then the interpretation of an I/Q-demodulation in our case may be misleading. Different shapes of the objects can then cause any phase function. Because of this the phase function is almost never constant in any interval.

After the description of these problems one could expect that the phase function is almost useless for recognition purposes. This is not true. The problem is mainly a psychological one since the phase curve doesn’t give the results one could expect at a first glance. Considering the involved mechanisms the result is quite feasible. If the curve is used for a least square comparison like the one applied to the amplitude function it certainly adds some information. The general improvement of the recognition result should, however, not be overestimated. In cases where small distance changes within objects with large reflecting surfaces are to be done the method is very good. In these cases longer transmission sequences can be applied.

6.3 Translation in a Plane

The methods described in section 6.2 are usable also for successful recognition even if the object is slightly moved in the plane perpendicular to the acoustic axis. However, the recognition probability decreases rapidly with the displacement length.[5] Since applications exist where the position of the object is not exactly known, there is a need for recognition methods applicable in this scenario.

A first approach might be to store recognition data for a grid of possible object positions. The grid density can be chosen so that an object displaced from one gridpoint is successfully recognized until the object is closer to an adjacent gridpoint. This principle would certainly work but the amount of stored data have to be huge when the covered placement area is large. This makes the method less interesting to use. Even if the amount of data would be feasible to collect, store, and handle the comparison would probably be too time consuming.

A thinkable solution is to use the object localization from Chapter 5 to determine what position the object have. Then, only patterns from objects in this position have to be compared with the current one. However, the problems connected with object localization

[5] Displacement in the direction of the acoustic axis is not treated. This can of course be compensated for with a time offset. Large displacements can also affect the echo shape. This then becomes a comparable problem with the displacement in the plane.

concerning accuracy, signal/noise ratio, and limitations for object shapes highly reduces the benefit with this method combination.

Another and more successful approach have been evaluated and tested using the 200 kHz sensor of the equipment. This has been done in a master thesis work by Luis Sobral from Coimbra University, Portugal. The experiments were guided by the author while the analysis work was supervised by Dr. Rolf Johansson, dept. of Automatic Control, Lund. The basic idea behind the work is to identify the echo from an object exposed to a certain transmission pulse *with a system that has an impulse response equal to the echo*. This is done using a system identification method that produces a system on state space form to represent the echo. The identification method is described in [Johansson and Lindstedt, 1995] where a continuous-time state space identification is done.

In the first attempt to identify the echo an identification of the oscillating echo signal was made. This did not work out well since the dominance of the resonance- or carrier- frequency made comparisons difficult to perform. The more successful method was based on a Hilbert transform to generate the envelope of the echo. This envelope was then considered to be the impulse response of a system and identification of this system on state space form was done. The transfer function data, of this system, was then used for the identification. A linear regression model was made from measurements of known objects in known positions. The object to be recognized was then compared with the known ones using the linear regression and a least squares fit.

Bode diagrams for the transfer functions of the systems created by different objects in different positions can be plotted as found in [Sobral et al, 1996]. A good measurement of the success in separating objects rather than positions is to study the features of the curves. It is obvious that the curves for the same object are close while the shape of the curves for different objects are different. It is also shown in the presented measurements that the recognition is successful up to a displacement of about 9 mm. A thorough presentation of this work is found in the master thesis [Sobral, 1995].

6.4 Rotation Around the Acoustic Axis

The previously described object recognition methods use only one transducer or a pair consisting of one transmitter and one receiver. An interesting challenge in object recognition is to also be able to recognize and determine object rotations. This type of measurement requires information from more sensors. The configuration of the sensor-units described in Chapter 4 ought to be well suited for this application. Consequently measurements have been made to discover the possibilities of this method. However, only a limited experience has been gained. Therefore the study still is considered a feasibility study rather than a fully developed method. Basic principles and some measurement results from the electrostatic unit are presented here.

The sensor configuration used in the sensor units have been described in many of the previous measurements. The Figure 5.6 may serve as an example. From this Figure we can see that the measurement configuration typically is that one element is used for transmission and the other three for reception. If we apply the ”rms-method” to the three received echoes we get three signature vectors. For a natural object, like a machine part, with asymmetrical surfaces with different orientations these vectors will be specific for the current object having the current orientation.

A feature of the sensor unit is that any of the four elements can be used as the transmitter. This means that, without physically moving anything, the object can be illuminated from four different directions. The four possible selections are shown in Figure 6.6. Consequently the second selection can be used to illuminate the object in a way that is equivalent to an object rotation of +90° around the rotation axis. The third equivalent to +180° and so on. This means that twelve signatures can be recorded. They belong to three sensor groups, relative to the transmitter, namely the left hand, the opposite, and the right hand positioned receivers Since each signature represents an object orientation the groups may be represented in three diagrams as shown in Figure 6.7.

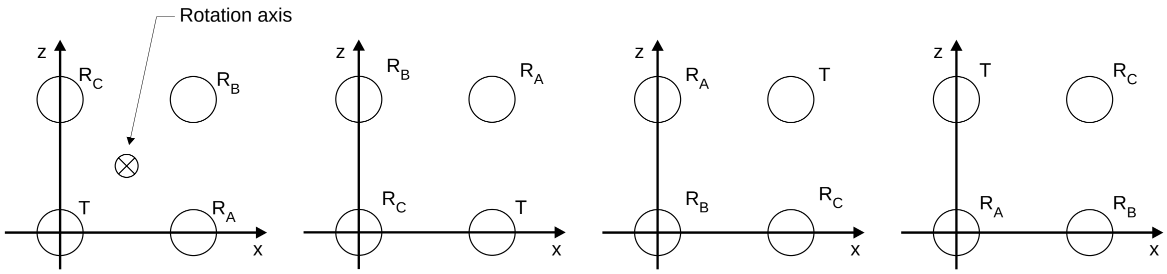

Figure 6.6 The four possible functional rotations without moving the object or the sensor unit. (The sensor function is rotated counter clockwise. This "rotates" the object clockwise.)

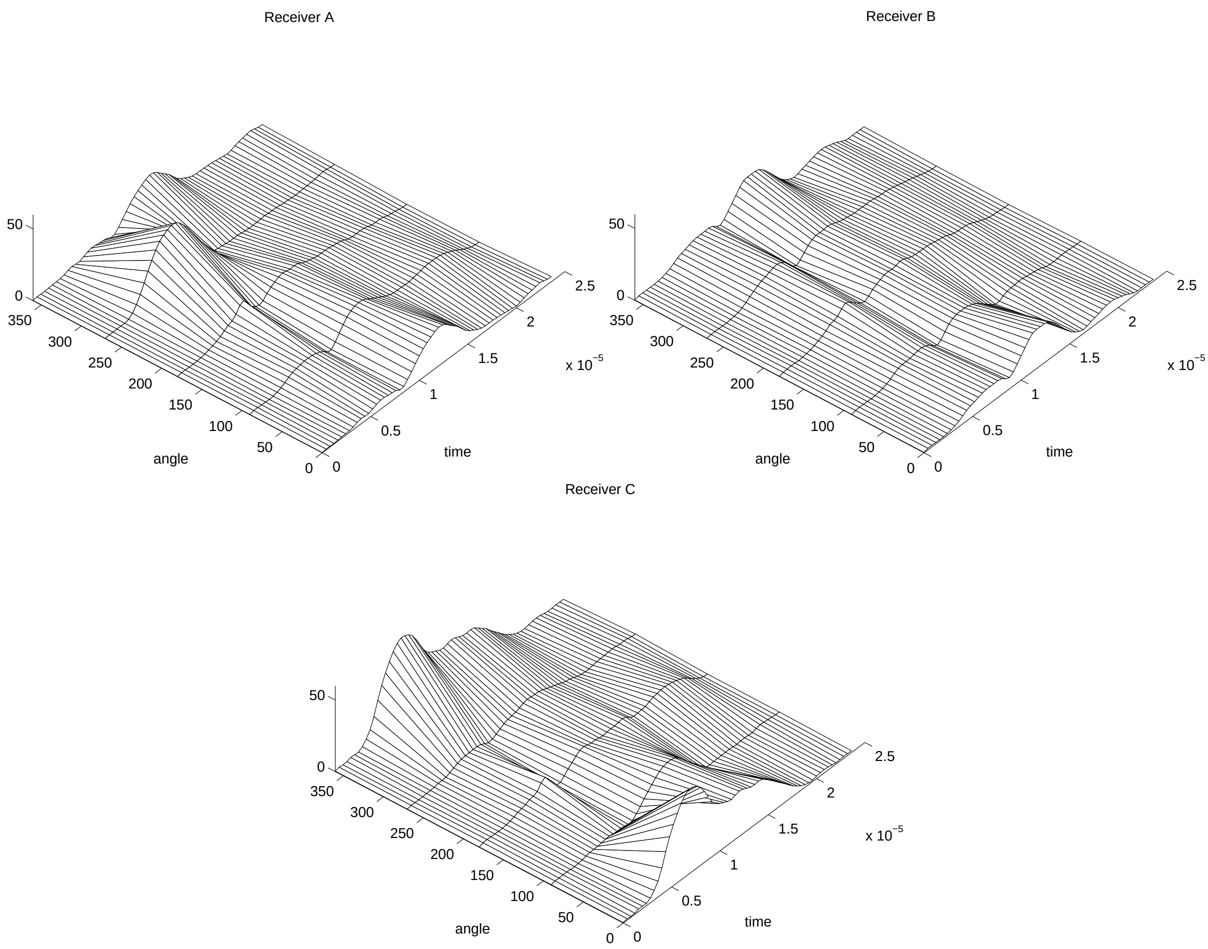

Figure 6.7 Each diagram shows the echo signatures for one receiver relative to transmitter position and the four possible rotations. (Zero angle case is shown also at 360°.)

The diagrams in Figure 6.7 are computed from a measurement of echoes from a industrial 6-pole connector with a housing and a cable outlet. This object has cavities, many surfaces with different inclination, and is comparably asymmetrical. 4 cycles of a 55 kHz signal is transmitted.[6]

Earlier measurements were based on one single sensor position and one single object orientation. The key question now is if the collected amount of data is usable to determine the orientation of the object. As in many ultrasonic measurements the answer is: It depends on the object. Well conditioned objects may be measured using only these data. Cavities in objects are often only possible to measure in a small sector. This requires a measurement sequence with smaller angles increments. However, it seems that arbitrary objects, like machine parts, are not possible to measure using only the immediately available four directions. If a measurement can be made in a scenario where the object or the sensor unit is rotated 45° the situation is much better. In Figure 6.8 the diagrams from the previous figure are modified with the new information. It is easy to see that several local maxima where hidden in the previous graph. This is obvious for sensors A and C.

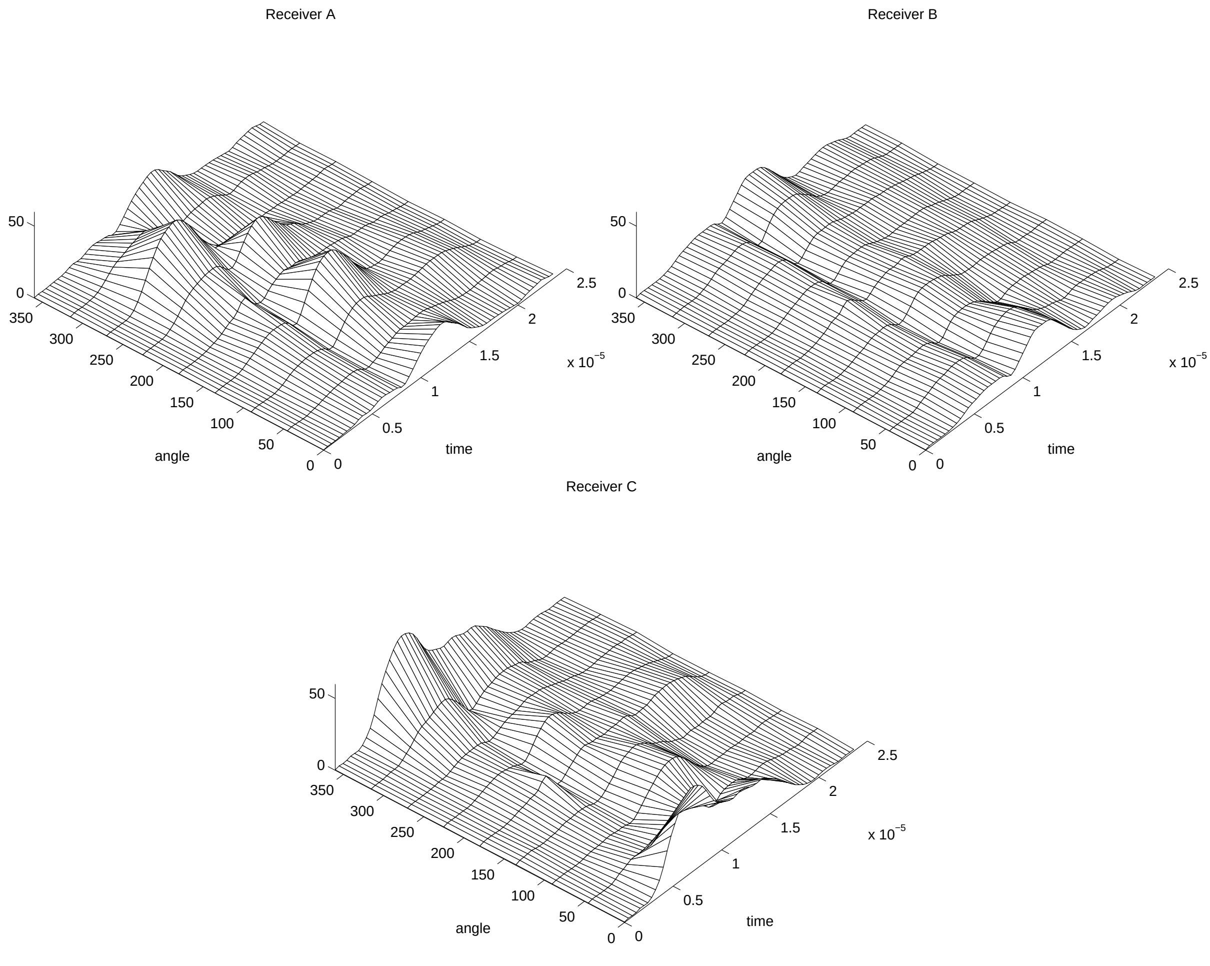

Figure 6.8 Echo information from two measurement situation where the object relatively the sensor unit are rotated 45°.

[6] The measurements in the current section is made by rotation of the object and not by changing the transmitting element. For well balanced receivers this will however not change the results.

This angular resolution can of course be increased further and this might be necessary for objects that are including narrow deep cavities. However, most objects can be measured with this resolution. The first problem that usually shows up is caused by some symmetry in the object that makes it difficult to chose one of two orientations.

The measurement of the current object orientation is made by finding the best fitting of the current data to the recorded reference. To get values for angles between the measured ones interpolation has to be made. It is possible that a weighting function should improve the result. This function should work in such a way that measured signature deviations are less important where the difference between the reference data for two adjacent angles is large. This would reduce the impact from uncertain reference data. It has however not been tested.

The three reference surfaces after interpolation are denoted by:

$$S_A(k,\theta); \quad S_B(k,\theta); \quad S_C(k,\theta);$$

where k is the time index for the signature of length n and θ is the angle of the object. The measured data is:

$$S_{XA}(k); \quad S_{XB}(k); \quad S_{XC}(k);$$

for the three receivers respectively. The fitting is then made by minimizing the function:

$$Q(\theta) = \sum_{k=0}^{n-1} \left[\left(S_{XA}(k) - S_A(k,\theta)\right)^2 + \left(S_{XB}(k) - S_B(k,\theta)\right)^2 + \left(S_{XC}(k) - S_C(k,\theta)\right)^2 \right]$$

A measurement of the 6-pole connector object rotated an angle of +20° has been made. The Q-function for this case is shown in Figure 6.9. Two minima are easy to identify. The lowest and correct one is about 18° while the other one is caused by a symmetry in the object. The object orientation can here be measured with this single measurement. However, this is not always a dependable result. Often this first measurement doesn’t give such a distinct minimum. To utilize all available information it is wise to use all four possibilities in Figure 6.6. The four resulting Q-functions for the current object are shown in Figure 6.10. The most reliable result from this method is usually achieved by adding the four Q-functions. This is shown in Figure 6.11.

The plotted Q-functions don’t have to be computed in a measurement situation. Much faster methods to find the local minima and compare them can be applied. In this case the amount of data processing is quite feasible and the measurement can be made fast.

A drawback of the proposed method is that the four possible measurements in a fixed measurement situation don’t seem to be sufficient to ensure safe orientation measurement. In cases where the sensor unit is placed in a robot arm or if a separate measurement of the object can be made this is of less importance.

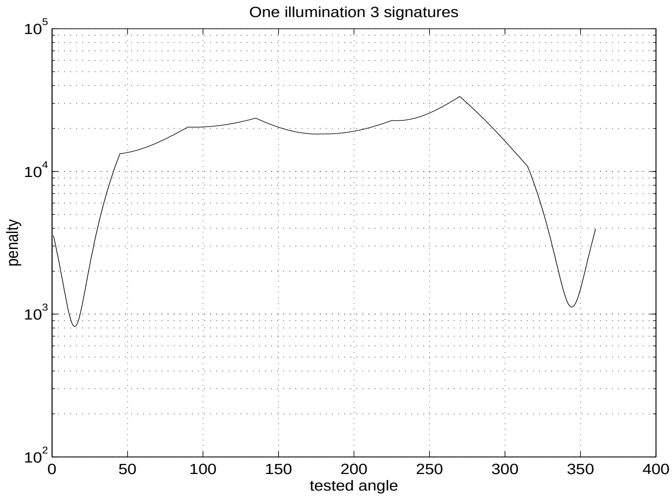

Figure 6.9 The Q-function from one measurement. The minimal value is obtained at about 18°.

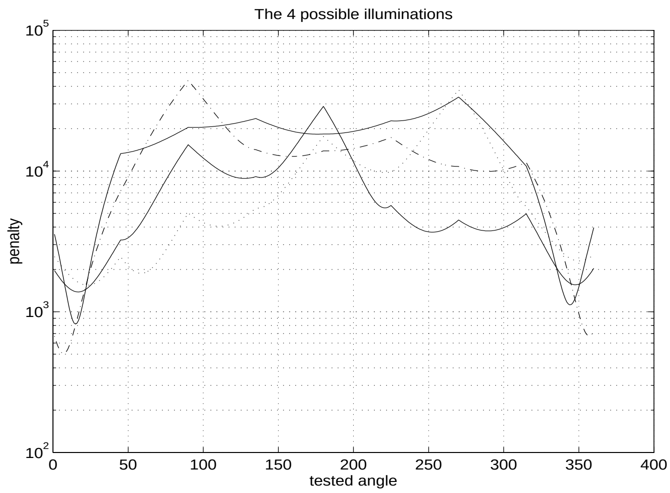

Figure 6.10 Q-functions from the four possible measurements.

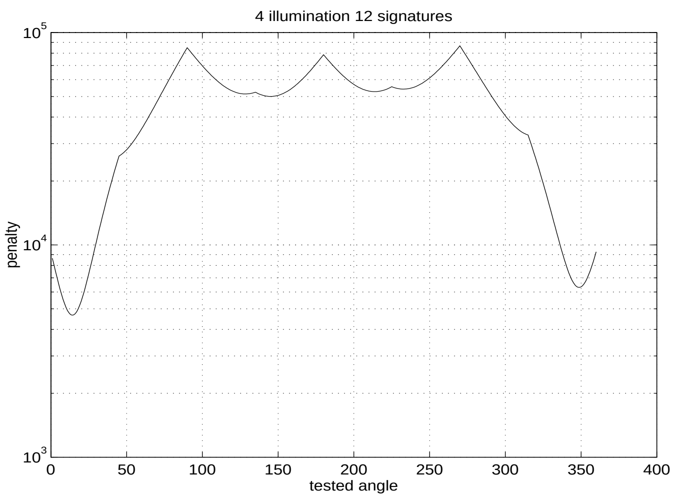

Figure 6.11 The sum of the Q-functions gives the most reliable result.

Extensions and Improvements — The Bat Lesson 7

In this chapter problems connected to many of the previously presented measurements are discussed. Use of several frequencies in the measurements is suggested as a solution to some of the problems. The echolocation principles of the bat have been studied to get an impression of what can be achieved in a well designed system. Finally, a comparison of the physical differences in instrumentation (sensors) between the bat and the used sensor units is done.

7.1 Limitations of Single Frequency Measurements

A very basic limitation of ultrasonic measurements is the length relation between interesting contour variations in the axial direction and the wavelength. Even if the wavelength is sufficiently short, problems can be caused by the length of the transmitted signal sequence. These limitations are caused by the comparatively low speed of sound. For instance, in most radar applications the wavelength compared to the dimensions of the measured objects is more favorable. This makes it possible to consider the received echo as a number of impulses, each coming form a reflecting surface. This is definitely not the case in ultrasonic measurements. The waves coming from different surfaces often interfere and make the interpretation a difficult task.

A higher frequency can, of course, solve some of these problems. In the paper [Sundström, 1991] a 1 MHz transducer is used for a high resolution measurement of surface structures. Furthermore, the used transducer is focused with a focal length of about 25 mm. This is truly a high resolution measurement. The axial resolution is about 0.03 mm and the lateral resolution about 1 mm. Compared to the results achieved with our sensor configuration this is quite an outstanding performance. However, the 1 MHz measurement is designed to solve a highly specialized task. Since the high frequency sensor is focused this means that relevant information is acquired in a volume of a few cubic millimeters. To collect a complete surface information the surface has to be scanned. This means that the sensor has to be moved over the entire surface that is measured. Consequently, this measurement is far from the intended applications of our sensor unit.

Higher frequency measurements are, as discussed in Chapter 2, primarily limited by two factors, the amplitude attenuation and the narrow lobe. This makes the use of high frequency ultrasound impossible for multi purpose applications at distances in the meter-range. Consequently, we have to deal with the problem of the comparably long wavelength.

The problems that show up in the interpretation of the echo sequences are caused by the interfering echoes from different surfaces. The information that we want is usually carried by the echo sequence, but is difficult to extract. In the previous chapters we have been using methods that can help us with this interpretation. The studies of the phase angle in Section 4.2 is one method that can be used. However, it is often to slow for the application. A usually better method is the matched filter, that is also described in Section 4.2. This filtering uses almost all information in the signal and are in many aspects a favorable choice. The major problem using this method is that the peak in the filter output has a tendency to obtain side lobes of significant amplitude. This complicates the separation of echoes that arrive closely in time. It is shown by [Audenaert et al, 1992] that this problem can be reduced if a carefully chosen pattern is used for the transmitted signal. The signal that was used by Audenaert et al was composed by a binary codeword where every bit corresponds to 4 cycles of the carrier frequency. The bits in the codeword then controlled the polarity of the transmitted signal. The codeword was then chosen so that the autocorrelation-function of the signal had the lowest possible sidelobes. Consequently, the sidelobes of the filtered echo signal get low amplitude sidelobes, thus making it easier to detect close echoes.

However, this is not the entire problem. The echo mixing can give effects like the fatal interference, described in Chapter 2. In this case the returned signal level becomes small. This may cause the echo to be almost impossible to interpret. Consequently, there is a need for further improvements.

A problem that often shows up when ultrasonic measurements are to be reproduced is that the temperature affects the echo interference. Because of the change in the speed of sound the phase angle under which the waves are summed is thus changed if the distances for the echoes differ. This can however be compensated by an adjustment of the frequency. This frequency variation is actually a key to the solution of many problems in single frequency measurements.

7.2 Stepped Frequency Systems

As described in Chapter 4 the Polaroid® ranging system was designed to use four frequencies for each measurement. This was based on the knowledge of the problems connected to single frequency measurements. In the basic form a measurement of this type can produce information that otherwise would have required four different measurements to acquire. Such a stepped frequency system is certainly a more robust system. The echo signal processing from stepped frequency systems can be done in many ways. One approach is to design sharp bandpass filters and detect the arrival of each frequency package. Thus four measurements of the delay time are obtained. Again, the matched filter method is a more sophisticated approach. To be able to compare the result we first study some known functions and their autocorrelation functions (ACF:s). In Figure 7.1 the ACF of four periods of a sine wave is shown. The side-lobes are strong and their amplitudes are slowly decreasing. The envelope has a typical triangular shape. This is an example of a function that will be difficult to study.

In Figure 7.2 the received pulse is used instead. The pulse is recorded as described in the matched filter section in Chapter 4. This example shows that it is very difficult to make estimates of the effects on the ACF of the low pass characteristics of transducers and other units that affect the signal shape. This is probably the main reason why the result in the work by Audenaert et al doesn't suppress the sidelobes as much as indicated by the initial codeword's ACF. This function will be even more difficult to study than the original sine wave.

Let us now finally study the ACF of the sequence suggested by Polaroid. The function is much more complicated with local maxima and a specific shape. However, although a signal composed by four frequencies with different number of periods is used, there is almost no improvement of the ACF. This indicates that the transmitted signal has to be carefully chosen. The addition of silent periods of variable length between the frequency packages can improve the ACF but the improvement is minimal compared to the code-word generated function.

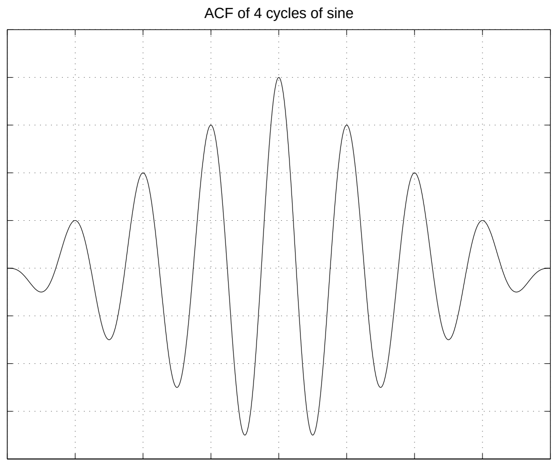

Figure 7.1 The ACF of four cycles of sine-wave

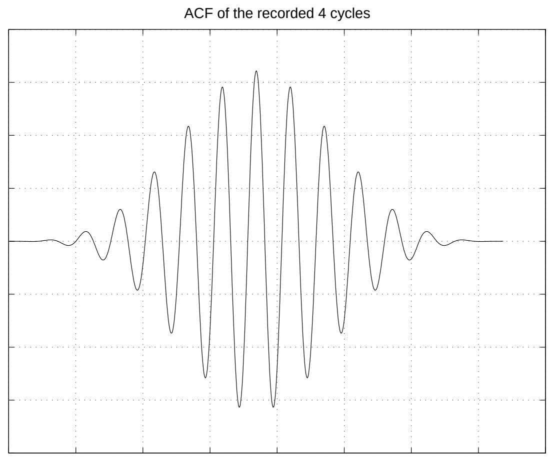

Figure 7.2 The ACF of the recorded ”real” pulse caused by the intended four cycles

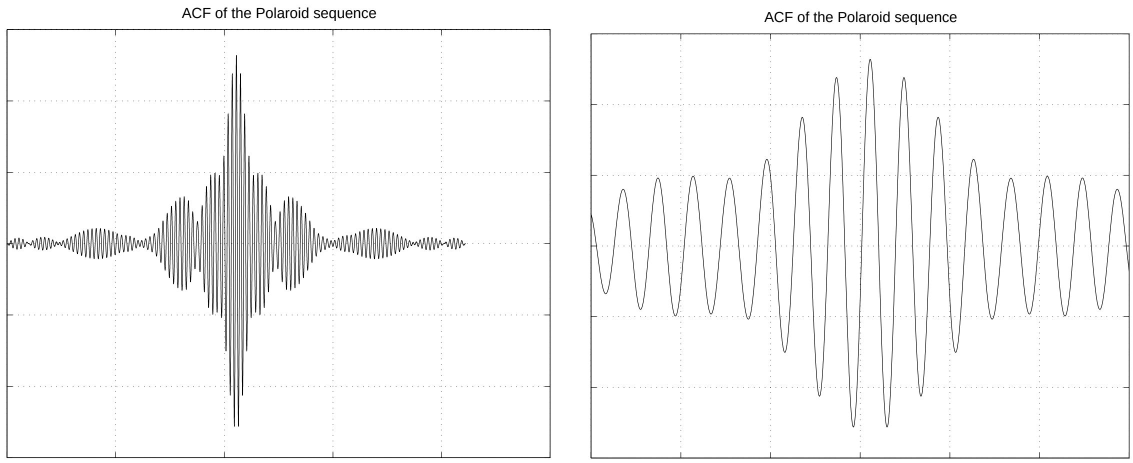

Figure 7.3 The ACF of the sequence suggested by Polaroid. To the left the whole sequence is shown. To the right the area around the peak is enlarged.

7.3 The Bat — a Master of Ultrasonic Imaging

When better solutions in ultrasonic imaging are required it would be a large mistake not to study the ultrasonic expert in nature – the bat. Or maybe we should say the bat family since more than 800 species exist. All of them use the principle of ultrasonic echolocation. The bats ”instrumentation” consists of one ultrasonic transmitter in the region of their throat, and two ultrasonic receivers namely the ears. So far the bat species are all similar but from here many variations can be found. The shape of their ears are different depending on if the bats primary food consists of insects or if it is more the vegetarian style. A bat that has to be a good hunter has to have large ears. The used ultrasonic signals also varies a lot. In dense vegetation compared to open air different signals are suitable for echolocation and prey hunting.

Furthermore, the ability to interpret the echo signals is different among the species. This is maybe the most interesting function but also the one that is most difficult to understand. Of course, size, shape and behavior in other aspects is also varied but this goes beyond the topic for this thesis

The ”sensor configuration” of the bat is rather easy to understand since it is very similar to the ”acoustic sensor configuration” of man. However, the differences shows up very quickly since we can’t close our eyes, shout, and tell that a person is standing at a distance of x centimeters in a certain direction. This is something a bat can. To find out how they do it we first take a look at the transmitted signal. Bats can transmit signals of constant frequency, CF, or frequency modulated, FM, or a combination of these, CF-FM. The FM sequence is always a frequency sweep, chirp, from higher to lower frequencies. Typically the used second harmonic of this chirp goes from about 100 kHz down to 50 kHz. If the CF-FM combination is used it starts with the CF-part and ends with the FM-part. The used signals are typical for the species. When bats transmit frequencies the harmonics play an important role. Actually, for many species the most important frequency is the second harmonic. This is also where the energy is concentrated.

To understand that the bat can measure distance with the ultrasonic pulses is not that very difficult to understand if we don’t go into the signal processing. What may be more surprising is that an important measurement for a bat is the Doppler shift. When the bat approaches a prey the echo frequency is higher than the transmitted one. This is detected by the bat. It is sometimes compensated for through an adjustment in the transmitted frequency. Consequently the received frequency is returned to the range where the bat’s ear is most sensitive. As a further consequence the transmitted pulse is not in this range. This can be compared to the problem with leakage described in Chapter 4. Through the Doppler frequency shift the bat can measure the relative speed to the prey. It has also been shown that some species can detect the flutter from the wings of the prey using the Doppler shift. This, and the other Doppler measurements are made using the CF-pulse.

The adaptation of the transmitted pulse seems to be a key parameter in the bat echolocation. The rate of the pulse transmission is controlled. During normal navigation a bat transmits about 10 pulses per second. During the last phase of hunting a prey this can be increased up to 200 pulses per second for some species. The shape of the transmitted pulse is also adapted to the situation. When the target is far away pulses with a lower frequency are used. These pulses are attenuated less than the high frequency ones. The amplitudes can also be varied for some species.

The CF pulse is very usable for the Doppler measurements but it is not sufficient for precise localization and precision measurements. For these measurements the bat uses the FM signal. The broad frequency content of this signal returns much more information than the CF-pulse.

The distance and the relative speed measurements that the bat do has already been mentioned. This is however not sufficient for prey hunting. The identification of a prey is a complex process. A first classification is made based on the echo amplitude compared to the delay time that gives the distance. From this information the approximate size of the possible prey can be done. The following neural process that probably is, to a large extent, based on experience is more complicated. The echo information from the FM-signal together with flutter from the wings is probably used.

Furthermore, the localization of a prey is not only a question of distance. The direction in terms of azimuthal and elevation angles also have to be measured. The azimuthal angle measurement is easy to understand since the two ears are placed in this plane. The echo delay difference between these two receivers combined with distance information gives this angle. The elevation, however, is less intuitive. This measurement is, exactly as for man, made by the delays caused by the structure of the outer ear. This correlation oriented measurement maybe seems a bit strange but since man has no problem to tell whether a sound comes from above or below it obviously works.

Another quite astonishing function in the bat echolocation system is that a bat is able to recognize it's own signals in environments where lots of bats navigate using echolocation simultaneously. How can this be possible? It seems that correlation, also in this case, plays an important role. When a pulse is transmitted it was earlier mentioned that the second harmonic usually is the most important frequency. Actually, the first harmonic is so weak that other bats can't hear it. Instead the first harmonic propagates through the bat's tissue and stimulates the reception neurons and make them sensitive to the incoming signal that is delayed and possibly Doppler shifted. Another explanation to the selection ability that can be an alternative and/or a complement is that the frequency content of the transmitted signal is specific like the speech of man. We have no problem recognizing hundreds of different voices and associate the appropriate person with them.

Much more can be said about the bats and their echolocation system. One of the first complete presentations that were made of the bat echolocation is [Griffin, 1958]. A lot more research has been done during the following years. The understanding of neural functions within the bats has been significantly improved. A comprehensible description can be found in [Suga, 1990]. In the paper by [Simmons, 1989] a more external and physical description of the bat echolocation system is presented.

The conclusions from this study of the bats can be summarized as follows:
- use a flexible selection of the transmitted signal;
- many frequency components are needed to get a detailed information;
- correlation between the transmitted and received signals are of uttermost importance;
- sophisticated measurements can be made with a simple measurement setup if the signal interpretation is adequate.

This forms an interesting platform for further improvements.

7.4 The Electronic Bat — a Comparison

One further conclusion that can be drawn from the study of the bat is that if we were able to perform half of the measurements that the bats do, with the same precision, we would consider ourselves as being highly skilled. This calls for an investigation of where the problems are. Is the instrumentation that we use adequate for the task or are our sensors not good enough? Is the neural computations that the bat do something that we can't implement in our interpreting software? Is the problem not in the single echo interpretation but in the assembly of the information and the following decision process? There are many questions and a lot of them go beyond the scope of this thesis. Let us try anyway to do a comparison.

As can be seen from Figure 7.4 the resemblance between the suggested sensor unit and the bat is striking. The transmitter and the two receivers have a direct correspondence. The extra receiver in the electronic unit seems to be redundant. However, this is not the case if we think of the elevation measurement made by the bat. Since the bat uses the outer ear to collect the elevation information the remark maybe should be that we ought to have two extra sensors instead. The influence of this difference might actually make the electronic unit somewhat inferior. However, the difference is not a fundamental one since the elevation information definitely can be obtained, also with the electronic unit.

The selection and assembly of the transmitted pulse together with the repetition rate can with a design like the one described in Chapter 4 be closely imitated. A possible improvement might be a frequency control of the timebase. This would allow minor frequency changes without reloading an entire echo shape. It can easily be accomplished with a phase-locked loop on the time-base of the function generator.

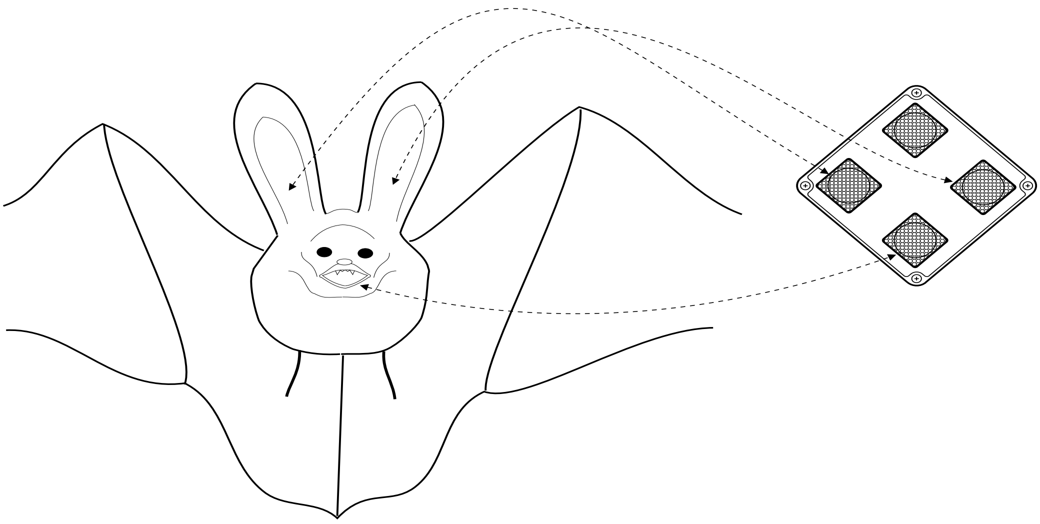

Figure 7.4 The biological bat and the electronic one.

A comparison of the used transducer elements must also be made. Concerning the frequency range the commercial elements used in the current sensor unit is slightly inferior. However, this doesn’t have to be a limitation. Much better wide bandwidth microphones are available for instance from the company Brüel&Kjær. This doesn’t seem to be any limitation anyway.

A much more severe problem is the amplitude range of the elements. This is not a problem in the meaning that artificial elements can’t respond to signals of an extremely high or low amplitude. The problem is the creation of a measurement chain that has sufficient dynamic capability to immediately switch between extremely high- and low-level signals. This is something that both humans and bats can do but it often cause problems when artificial implementations are to be made. An equivalent resolution of more than twenty bits are often required. This can usually not be realized. The solution in the sensor unit must instead be based on a range switch. If a measurement returns an echo that is not possible to interpret in the current range the measurement must be repeated after a range switch. Parallel architectures where it is accepted that some branches are saturated and some are too insensitive can also be used to gain time.

As a summary of the instrumentation part of the comparison it can be stated that solely minor restrictions are found. These ought to cause the ”electronic bat” only a slightly inferior performance.
A comparison of the next step in the measurements chain leads us to the neuron networks in the bat. We can here identify several operations that are executed by neurons concerning frequency, frequency shift, and amplitude detection. Many of these operations can in an artificial device be executed in hardware as well as in software. The hardware solutions are often more bat-like since the processing is parallel. Time is usually also gained. The more complicated operations performed by the neurons concern the delay sensitivity. This is the basis for the correlation relationship that is a prerequisite for many of the bat measurements. Many of these processes that go on in the neuron network have been described in the literature. Still much can be considered as being sub-systems in a process where the total strategy is defined only by assumptions.

This is presumably the point where we lose the possibility to directly compare our electronic bat with the real one. At the point where we lose contact with the well known mechanisms in the bat and goes into the unknown we can however state that the most important function that is utilized is the signal correlation. The selection of signals that are suitable for correlation measurements like frequency sweep are preferred as we have seen in the ”external” bat descriptions. This is also the principle that takes us below the wavelength resolution threshold. The elevation angle measurement is a correlation measurement of the sub-echoes from the outer ear. The list of measurements based on correlation principles is long and will probably be even longer as the understanding of the bat measurement principles is increased.[7]

[7] A powerful demonstration of what correlation related methods can achieve is found in [Holmberg, 1992b]. In this work is a ranging measurement made that is highly insensitive to noise and disturbances.

Applying Varying Frequency Signals 8

In this chapter is the advantages of transmissions including several frequency components illustrated. The applicability for some of the previously described measurements is also discussed.

8.1 Problems Solved by Frequency Variation

When the bats were analyzed in Chapter 7 it was made clear that they use frequency sweeps to acquire the detail information from their target. This corresponds well to our conclusions about the limitations of single frequency systems. With knowledge about the interference phenomena it is easy to understand that a lot of problems can be solved wit the use of several frequencies. This is illustrated by the example in Figure 8.1. A surface that create fatal interference for frequencies around 60 kHz is used.

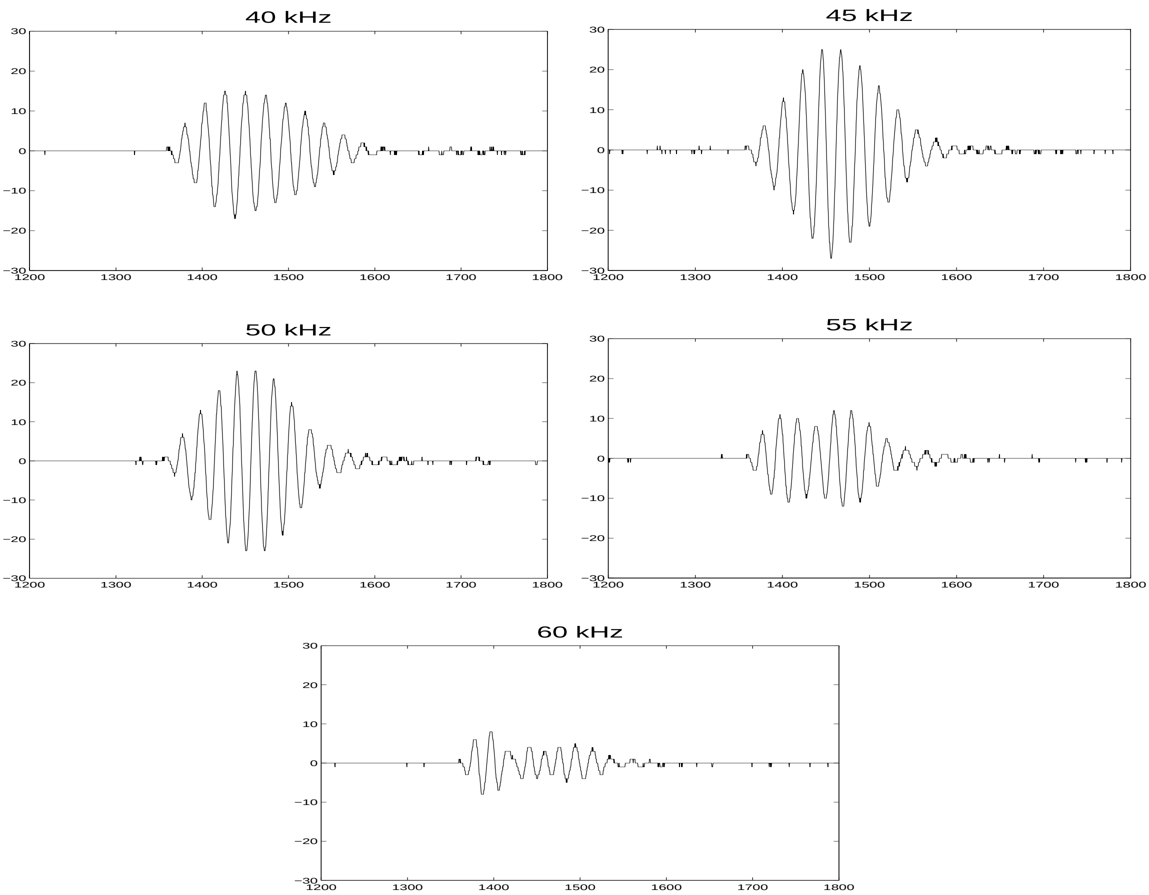

Figure 8.1 Echo shape at five different frequencies. The same reflecting surface is used in all five measurements.

The remaining problem is to condense the information from multifrequency measurements into a comprehensible model. The frequency variation method is in this aspect a central question. Stepped frequency systems in various forms have been discussed previously. They certainly add new measurement information compared to single frequency methods. However, the implementation of the frequency changes often seems unnatural. From many points of view a frequency sweep is a better choice. The approach seems very straight-forward. First the lower and upper frequency limits are chosen. The basis for this choice is that all distance differences between surfaces that are of interest shall give significant echoes within some part of the frequency interval. The more complicated the object shape is, the wider the frequency span has to be. After this the sweep rate is chosen. A sweep that is passing the frequencies too fast will not establish detectable echo relations at each frequency region, thus returning only some transient information that is difficult to interpret. On the other hand the total length of the frequency sweep must be reasonable. This is crucial to allow a feasible repetition rate in the measurements and to keep the amount of echo data sufficiently small for reasonable computation times. A compromise that is acceptable in both aspects can usually be found.

Another quality of the frequency sweep is the nice behavior of the function that is going to be transmitted by the transducers. The frequency components included is almost entirely in the frequency interval of the start and stop frequencies. The functions is continuous and has a continuos derivative except for the beginning and the end. These points may have slight tendencies to start unwanted oscillations in the transmitter system but this is usually negligible. The nice behavior of the sweep means that the correspondence between the intended function and the actually transmitted one can be good. Actually it can be much better than if more nonlinear functions like stepped frequency with pauses or the function suggested by [Audenaert et al, 1992].

A problem related to the interference problem is the temperature dependence of the echo shapes. The interference effects that can be detected at one temperature often appears quite differently at another temperature due to the velocity changes. This is something that frequency variations can compensate for in two ways. Either the transmitted frequency is compensated to produce the same wavelength at both measurements, or a frequency sweep is used at both occasions. In the latter case the frequency variation provides the very same interference information but at different frequencies. This gives new possibilities to make comparisons for various purposes.

To summarize the potentials of frequency variations we can state that the information provided by frequency variation in general is superior to the one provided by single frequency measurements. The frequency sweep is a compact and efficient way of applying varying frequency signals. The only clearly identified measurement where single frequencies have to be used is the Doppler measurement.

8.2 Models to Represent Multi Frequency Echoes

Repeated Single Frequency Measurements

In Figure 8.1 a multifrequency measurement was shown. However, it is not very practical to use a representation consisting of a number of separate curves for each object in applications like object recognition. A more comprehensible representation is the one shown in Figure 8.2. The echoes are here converted to ”rms-functions” and plotted in a three dimensional diagram.

This spectrogram representation gives an impression of how interpolation between the curves can be done. In this way a surface is created.
In this example five measurements were made. When interpolation between the curves is done it still seems to be a risk for local maxima or minima. The main conclusion is that the more frequencies we measure at, the more information we get. This calls for more measurements or another way of solving the problem.

Frequency Sweep Measurements
The use of a frequency sweep is a way that exposes the object to all frequencies in a certain interval. In the design of the sweep we have to specify three parameters:

- the start frequency, f_1,
- the stop frequency, f_2, and
- the sweep rate, r (Hz/s).

The last parameter may also be exchanged with the number of cycles, N, of the sweep. (This will be done below.) If we assume that the sweep is linear and starts at the time $t_1=0$ and stops at t_2 we can express r as:

$$r = \frac{f_2 - f_1}{t_2 - t_1} \tag{8.1}$$

The sine-function we use for the sweep should at each instant be representing an equivalent frequency in the interval. If we express this instantaneous frequency function as:

$$sin(arg) = sin(\omega_i \cdot t) \tag{8.2}$$

we find that we have two time dependencies. The sine is a function in time and the angular frequency is a function in time. Because of this we can't make any linear time-relation for the angular frequency. Instead we must specify the time derivative of the argument since this is what we will notice as the instantaneous angular frequency. The linear sweep in the time interval, { $t_1 < t < t_2$ }, under the assumption that $t_1=0$, then corresponds to:

$$\frac{d\,arg}{dt} = 2 \cdot \pi \cdot \left(f_1 + \frac{f_2 - f_1}{t_2} \cdot t \right) = \omega_1 + \frac{\omega_2 - \omega_1}{t_2} \cdot t \tag{8.3}$$

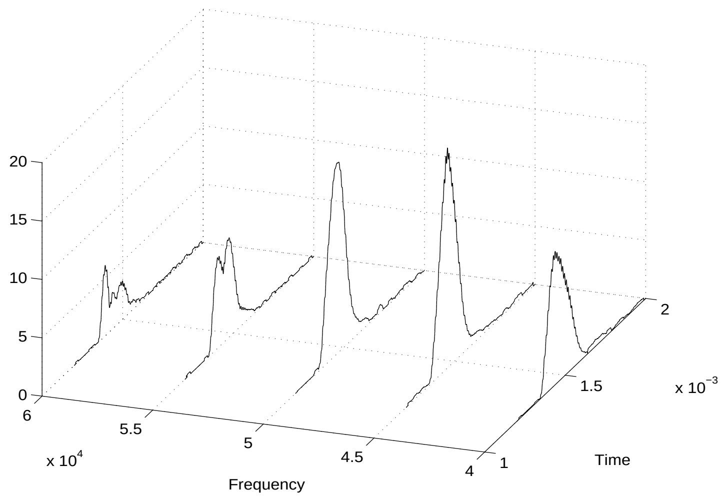

Figure 8.2 "Rms-functions" for the five echoes in Figure 8.1 plotted in a spectrogram.

The argument can then be calculated as:

$$arg = \int \left(\omega_1 + \frac{\omega_2 - \omega_1}{t_2} \cdot t \right) dt \; = \; \omega_1 \cdot t + \frac{\left(\omega_2 - \omega_1\right)}{t_2} \cdot \frac{t^2}{2} + \varphi_0 \qquad (8.4)$$

where φ_0 is set to zero. It is often of practical importance that the sweep starts and stops at the positive and negative zero crossing of the sine wave respectively. This condition makes it convenient to specify N instead of r. This can be done if we solve the argument expression for the time t_2 and the specified number of elapsed cycles:

$$\omega_1 \cdot t_2 + \frac{\left(\omega_2 - \omega_1\right)}{t_2} \cdot \frac{t_2^2}{2} = N \cdot 2 \cdot \pi \qquad \Rightarrow \qquad t_2 = \frac{2 \cdot N}{\left(f_1 + f_2\right)} \qquad (8.5)$$

With the use of a frequency sweep of this kind when measuring the previous object we get a totally different echo shape as shown in Figure 8.3. The frequency is here swept from 60 down to 40 kHz during 32 cycles.

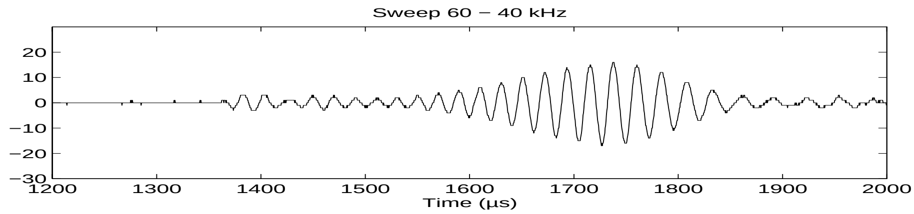

Figure 8.3 Frequency sweep echo from the surface used in Figure 8.1.

This echo contains a lot of information from the different frequency regions. As we can see the echo is very small in the 60 kHz region in the beginning of the echo. However, later in the echo significant amplitudes are produced. The problem that we are facing now is how to make a comprehensive representation of this echo. The spectrogram approach from Figure 8.2 has several advantages since the different variables are easily separated on the axes. A representation of this type implies that we are able to separate the echo signal frequencies. This can be done in several ways. The two most obvious ones are the FFT method and the filter method. In the FFT method the best separation is probably achieved. This is however a method that requires several numeric computations and will be somewhat slow. The filter method on the other hand can to a great extent be implemented in hardware and is thus less time-consuming. The importance of the quality difference between the methods depends much on the purpose of the measurement. If the purpose is a recognition measurement where shapes are to be compared between functions generated by the same method any of the methods can be used. For purposes that requires a precise frequency separation the FFT is probably a better choice. However, in this case it is important to remember that the frequency sweep don't deliver any exact single frequency echoes since the frequency is constantly changed.

An implementation of the filter method has been done in software. The input signal is processed by 21 parallel sixth order Chebyshev bandpass filters. The filters are designed to have a bandwidth of about 1 kHz each. The output from this system is shown as the surface in Figure 8.4. The input is the signal in 8.3.

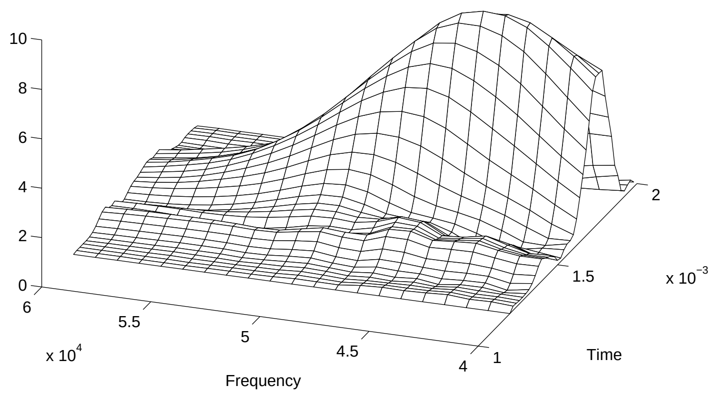

Figure 8.4 A bandpass filter generated surface from the input sequence shown in Figure 8.3.

A comparison of the Figures 8.4 and 8.2 shows that the shapes of the surfaces have similarities but they are not, and should not be expected to be, equal.

8.3 Applicability to Distance Measurements

The use of a set of frequencies to make distance measurements have been discussed several times earlier. For example this has been done in Section 7.2. This method gives obvious advantages compared to single frequency measurements. However, an interesting question that is not previously treated is if the frequency sweep also can improve distance measurements. A survey of the methods described in Chapter 4 forces us to reject most candidates. The beginning of the echo is not easier to detect because of the frequency sweep. Consequently all methods focused on the echo beginning are not improved. The phase oriented methods are difficult to use since the frequency is constantly changed. This leaves us with one remaining candidate and that is the matched filter. This method has proven to be a powerful tool in many of the measurements. It is especially good in measurement situations where significant levels of uncorrelated noise and/or low level signals are present.

The autocorrelation function, ACF, is an extremely important parameter for this method. The ACF is of great interest because it determines the possibilities to detect echoes that arrive with a short time interval between them. If the side-lobes of the ACF are large and slowly decreasing with the distance from the peak it is difficult to find close echoes. The ACF was tested for a number of signals in Section 7.2 but no really good signal was found. This calls for a test of frequency sweeps. In Figure 8.5 the ACF of a computed sweep from 60 to 40 kHz is shown. The largest side-lobes have an amplitude of about 75% of the peak. The amplitude of the following lobes are then rapidly decreasing. Consequently, this seems to be a rather good function to work with. We certainly know that functions with much better selectivity in the ACF can be designed. But the problem usually is that the limited bandwidth of the measurement system have influence on the signal so that the resulting ACF differs much from the theoretical one. In this aspect the frequency sweep ought to be a ”nice” function. This is certainly true. In Figure 8.6 the measured ACF from the system is shown. The sidelobe relation is about 85%. This is not quite as good as the theoretical value but still acceptable. It should be noticed that the correspondence for the rest of the sidelobes is very good.

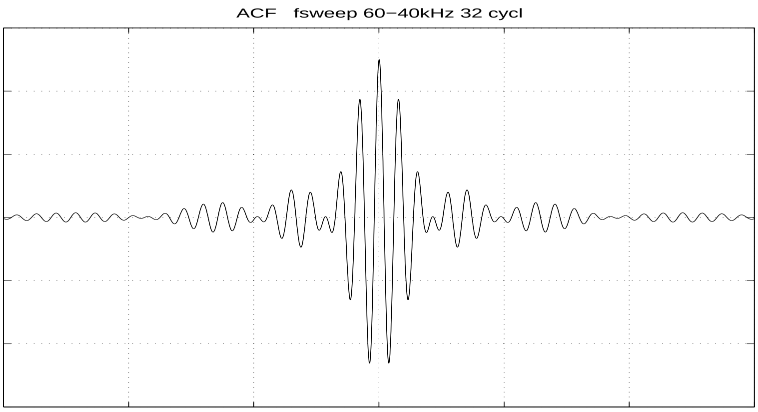

Figure 8.5 The ACF of a computed frequency sweep from 60 to 40 kHz in 32 cycles

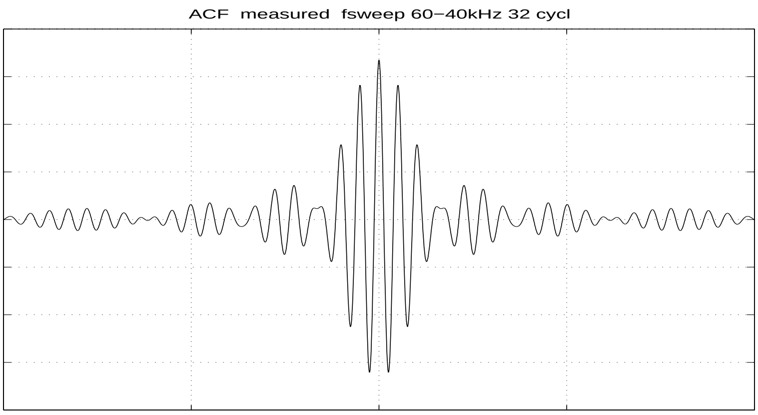

Figure 8.6 The ACF of a measured frequency sweep from 60 to 40 kHz in 32 cycles

A question that immediately has to be stated is whether the ACF is affected by the sweep parameters. The answer to this is definitely yes. An extremely slow sweep contains several adjacent cycles with almost the same cycle time. This will give a low selectivity in the ACF. On the other hand a too short and fast sweep is not good either. The fast frequency change will give problems in the reproduction in the real limited bandwidth system. A short sequence will also reduce the amount of statistical data, thus reducing the noise suppression. Consequently an optimum exists. This optimum seems to be somewhere in the range of 16 to 32 pulses for a ±20% frequency sweep around the center frequency like in the current case.

In this discussion we have only been treating linear frequency sweeps. Although it is not treated here the possibility of using nonlinear sweeps ought to be pointed out. The bats use sweeps that in most cases are almost linear. However in some phases of prey capture they seem to be sweeping slower in the lower frequency range.

The conclusion from this study is that the frequency sweep is a good function in combination with matched filter measurements. The statistical data is recorded during a considerable time and frequency span. This ought to be a good base for the further processing.

The angular measurements and the object localization are primarily based on distance measurements. Consequently advantages gained by applying frequency sweeps to distance measurement will also improve these other measurements.

Let us conclude this discussion of the applicability to distance measurements with a practical experiment. The measurement of the 1.2 mm wire that was made in Chapter 4 is a true challenge for a frequency sweep. Although the measurement was successful using four monofrequency cycles and a matched filter the interpretation margin was limited. It is certainly an interesting question if the previously shown ACF:s can maintain their good shape in a noisy environment.

In Figure 8.7 and 8.8 the echo sequence and the matched filter output are shown. The shape of the filter output is affected by the noise but the interpretation margin is certainly improved. This is especially true if we look at sidelobes beyond the first one.

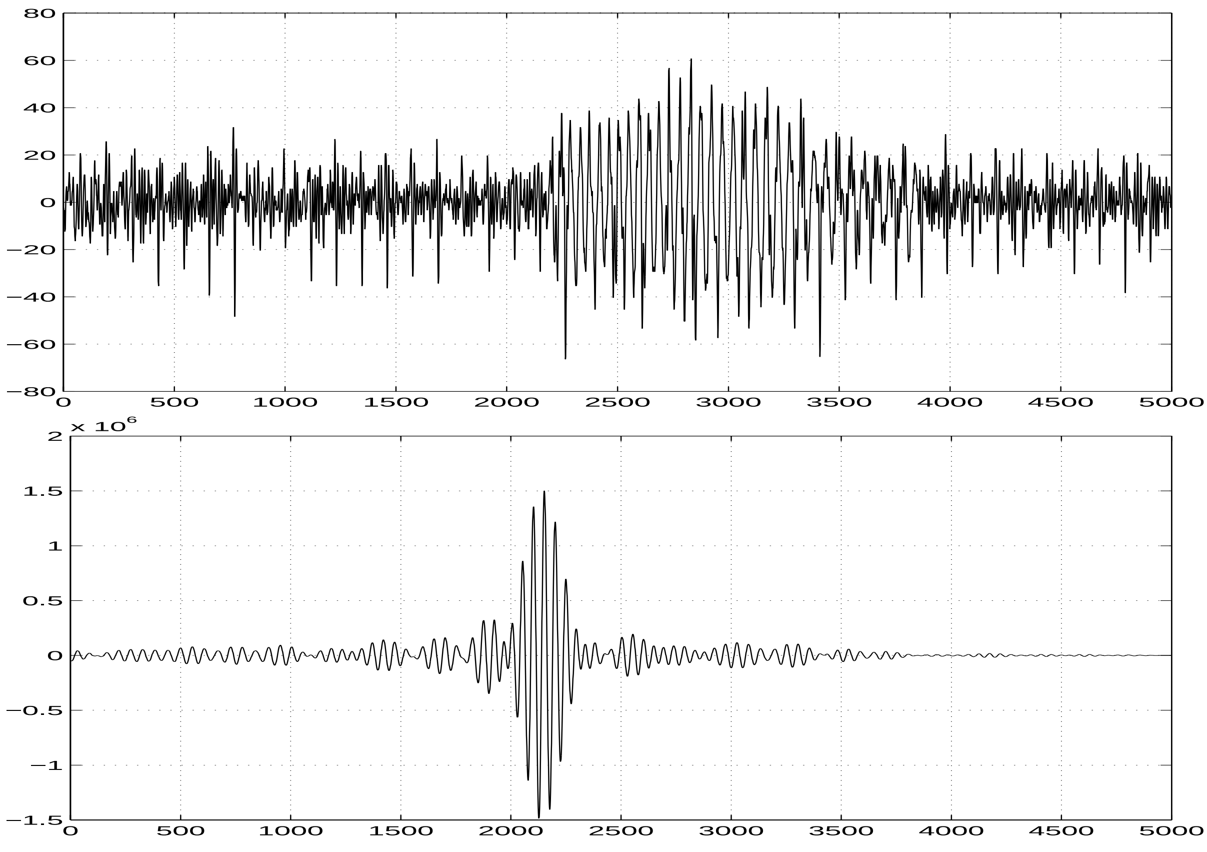

Figure 8.7 At the top the echo sequence from a 32 cycle frequency sweep from 60 to 40 kHz is shown. Below the output from the matched filter is shown.

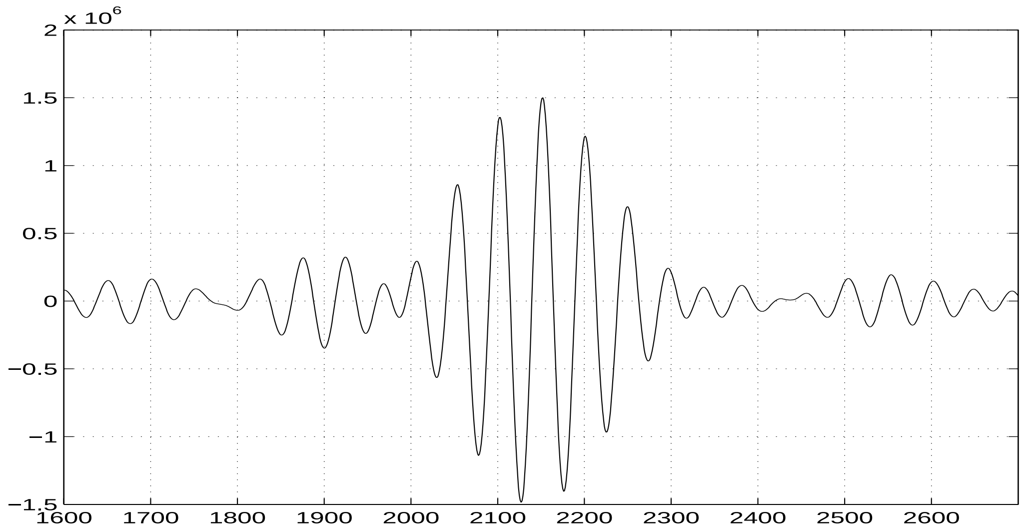

Figure 8.8 An enlarged section around the peak in the filter output from Figure 8.7.

8.4 Applicability to Object Recognition

An obvious application for the described multi-frequency echo model is object recognition. The methods described in Chapter 6 can directly utilize this model. The fitting of curves is then replaced by fitting of planes. It is obvious that the statistical material is much larger but the surface representation provides another advantage as well. We have seen that interference problems can cause trouble in the measurements. Even worse, in the object recognition case the temperature dependent speed of sound affects the interference. We can look upon this as if the wavelength changes. Thus, the summing of the echo waves is made at another phase-angle. A variation in frequency will be able to compensate this wavelength change and consequently cause the same interference situation.

In Figure 8.9 the echoes from an object consisting of several surfaces are shown. The object has been illuminated with the same constant frequency signal at two different temperatures. The rms-method from Chapter 6 is then used on the resulting echo. As illustrated in the figure the probability for recognition with the use of a least squares method is not very large. The signature function is significantly changed. The temperature difference in this example was only 5°C. The pulse acquired at the lower temperature, of course, also arrives later and its time scale is ”stretched”. However, this is something that can be handled with ”ordinary” temperature compensation.

Using a constant frequency a complete compensation for the change in the speed of sound can’t be done in the measured data since the interference is affected. This makes the detection of small deviations impossible and the acceptance level for recognition has to be set very generously. Nevertheless, there is a way to compensate the temperature change. If the transmitted frequency is changed to compensate the temperature it will be possible to reproduce the interference effects. However, this seems to be unnecessarily complicated if the problem can be solved with the use of a frequency sweep.

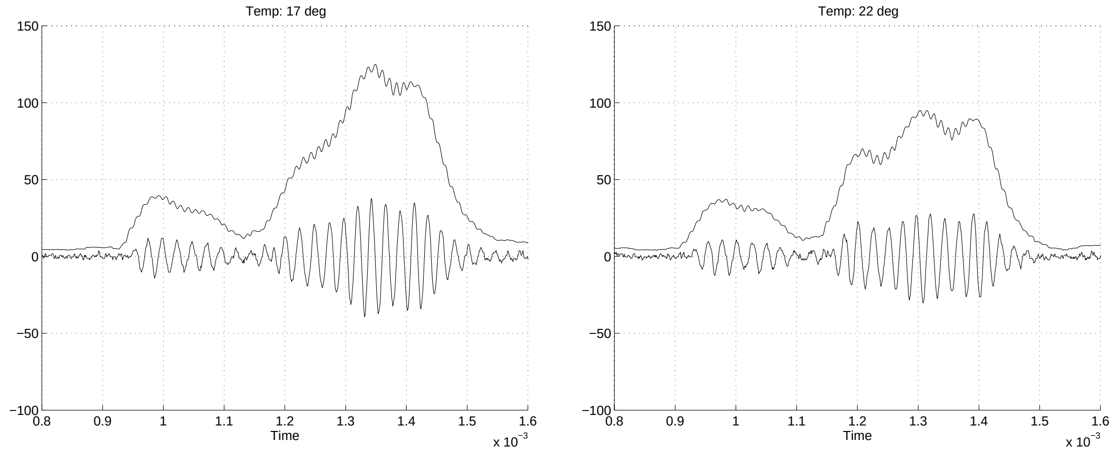

Figure 8.9 The echo signal and the corresponding rms-function for two measurements at two different temperatures using a constant frequency. To the left the measurement is made at 17°C and to the right at 22°C.

Let us illuminate the same object that generated the echoes in Figure 8.9 with a frequency sweep from 60 to 40 kHz with 32 cycles. The resulting echoes from this experiment are shown in Figure 8.10 using the previously described spectrogram model. The surfaces have an obvious relationship but the profiles are differently located in the time-frequency plane. A direct comparison will of course give a significant difference as shown in Figure 8.11. In fact it is in the order of magnitude as the initial amplitudes.

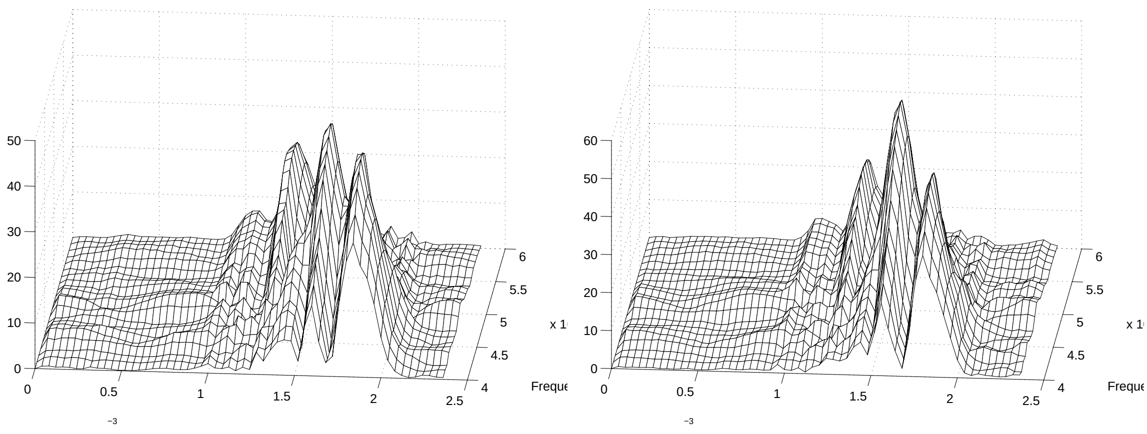

Figure 8.10 Spectrogram representations of the echo rms-functions after transmission of a frequency sweep toward the surface. To the left the measurement is made at 17°C and to the right at 22°C.

It would now be a possibility to make a compensation of the time-scale in one of the diagrams according to Equation (2.4). We have then made a compensation for the delay caused by the temperature change. To compensate for the different interference the surfaces should be compared in such a way that the same wavelengths are compared. This means a translation in the frequency axis direction. However, this translation will cause a change in the time-scale since the different frequency components were not transmitted simultaneously. The sweep rate defined in Equation (8.1) can be used to calculate this offset along the time axis. Together this would theoretically make the compensation.

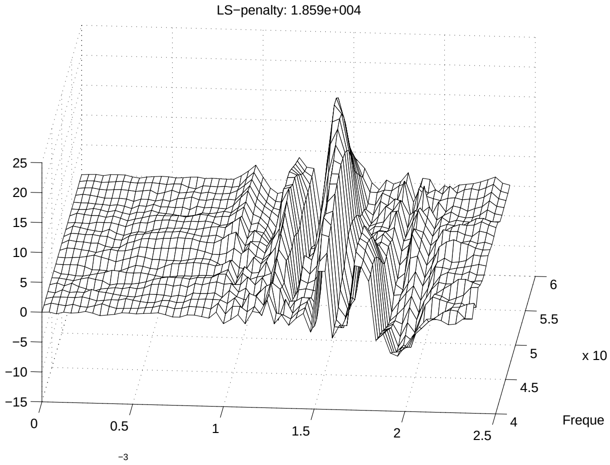

Figure 8.11 The direct difference between the two surfaces in Figure 8.10.

This compensation has been tested but it doesn't seem to give a Sufficiently good result. An optimization has been done where the time and frequency compensations are allowed to vary. This optimization usually produces a time compensation with very good accuracy (the velocity difference accuracy within 10% for significant temperature changes). However, the frequency compensation is usually less than expected. A small compensation, not more than half of the one from the theory, is obtained. The reason for this is probably that the filters are not perfect frequency separators and they are exposed to a transient in the beginning of the echo.

Even if the method doesn't work exactly as expected a great improvement is obtained when object recognition is made at various temperatures. In Figure 8.12 The surface difference after the optimized compensation is shown.

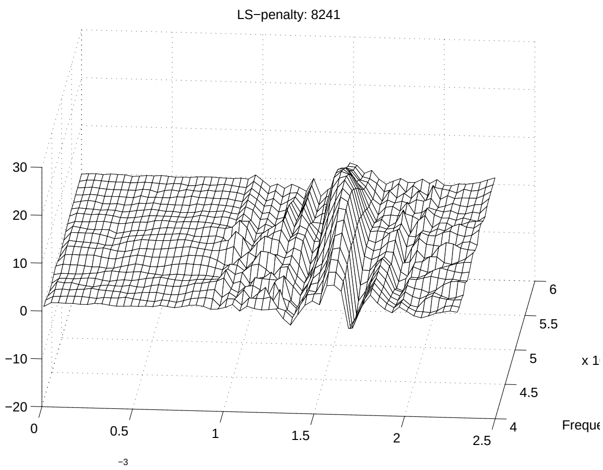

Figure 8.12 Difference between the surfaces after an optimal compensation of time and frequency.

Using a frequency sweep a method has been created that will be a successful platform for object recognition. It opens up a field of research, where objects will be recognized by surface fitting. There remains a lot of research to define suitable recognition algorithms for frequency sweep signals. So far the potential of this kind of object recognition has proven to be very promising.

Conclusions and Topics for Future Research 9

This chapter summarizes the achieved results from the previous chapters and tries to give an overview of their real time applicability. Many of the treated subtopics provide wide possibilities for future evaluation and expansion. Some of the especially interesting possibilities are emphasized.

9.1 Summary of Results

In the preceding chapters ultrasonic measurement principles have been described for the measurement of distance, flat surface spatial angles, object location, object orientation and object recognition. For many of the measurements different variants have been suggested. This variety of approaches also increases the applicability of the methods in harsh environments. Another important aspect has been to only use methods possible to implement in real time applications utilizing reasonable computer structures. Consequently, the suggested methods have to require not more than a feasible amount of computation power.

The distance measurement is also the basis for the angular measurements and the object localization. This means that all improvements concerning distance measurements also can be used by some other measurements. In the previous chapters the most powerful technique used to determine distances is definitely the matched filter. This is not a new technique. However, the suggested adding of an extra receiver in an off-line measurement to calibrate the filter has not been found in the literature. This is an obvious improvement since the pulse shape that is measured after the influence of the transmitter and the receiver transfer functions gives a much better correlation. The use of frequency sweeps then will make a further improvement of the measurement robustness. This technique, that is a product of studying the bats, ensures that usable echoes are obtained from almost any surface shape that has a significant size.

Methods for object recognition using ultrasonic signals have been presented. For situations where the object position is known the methods are directly applicable. The use of frequency sweeps in the transmitted signal is crucial from two points of view:

- the echo signal has to be a product of more than one frequency if contribution from all present surfaces is desired;
- the sweep makes it possible to improve the transformation of signatures from one temperature to another.

These two points represent the solution to two severe problems that have limited the use of object recognition.

For situations where the object position is not known a successful recognition method has been described in the cooperative work publications by Dr. Rolf Johansson, Mr.Luis Sobral, and the author. This work has been done using one 200 kHz piezoelectric sensor and shows very promising results.

In cases where the object orientation is not known a valuable tool has been provided. The object recognition oriented method for measuring the object orientation is an example on how valuable three dimensional information can be produced with the limited size sensor unit. This can also be used as a robust recognition method for objects with a known orientation.

One important results from the current work is the design of a functional system that is the platform for all the measurements. This platform includes the two designed sensor units, one equipped with piezoelectric sensors and the other with electrostatic sensors. In both cases it has been proven that valuable information that goes far beyond the simple distance measurements can be obtained from units of highly limited size. The platform also includes functional circuits for all analog and digital signal processing that is needed to make a host computer able to transmit and receive ultrasonic signals.

9.2 Industrial Relevance

The measurement methods described in this thesis are all focused on different complexity levels of imaging in industrial automation. The current use of ultrasound in industrial processes is quite limited. When used it is solely for measurements on the lowest complexity levels. One of the intentions of the current work was to prove that the ultrasonic techniques can provide lots of valuable information on higher complexity levels. This has been done by demonstrating the variety of measurements mentioned in the previous section.

A crucial parameter for industrial applicability is robustness. The struggle against noise and disturbances is always a topic with a high priority. The traditional methods of repeated measurements and rejection of outliers have been reinforced with the methods using matched filters and frequency sweeps. The matched filters provide an excellent reduction of uncorrelated noise. On the other hand, the frequency sweep makes it possible to perform measurements also in environments where constant or intermittent ultrasonic disturbance sources are present.

The temperature dependence of the speed of sound is no major problem in the distance oriented measurements and is actually compensated for even in the simplest systems available. However, this has been a problem in the methods that are using the echo shape since the interference patterns are changed. By applying a frequency sweep as the transmitted signal the sensitivity is here significantly suppressed.

An important parameter for all measurement systems for industrial use is their cost. In the described system no exclusive and costly components are used. In the measurement system platform the hardware costs for the prototype equipment added to the host computer is about $400-500. In other words, this is a genuine low cost technique.

Another aspect is that many measurements can be done by the same sensor unit. This is a major advantage since only one hardware configuration has to be manufactured. The function is then determined by the software or a simple configurations procedure.

The necessary modifications of the hardware platform for industrial use primarily concerns the data exchange. The parallel bus from the host computer to the measurement system is not suitable in an industrial environment. A more efficient solution would be one or a few serial high speed links, for example following the RS485 standard. Data may then be exchanged using only a few twisted pair connections. This requires the design of a special host adapter for the VME-bus or whatever computer system is used. Such a communication system is currently being developed.

The possible applications for a versatile measurement system of this type are many. For example in robotics, the use of an ultrasonic sensor unit that can measure distance, angles, object locations, object orientations and recognize objects and situations is a very valuable tool. Following the trend of open robot systems this type of versatile sensing devices will probably be a common component in the near future. The reason for this is that the open systems provide a wider flexibility for custom design solutions.

In manufacturing industries there are often several process steps that have a need for supervision. The recognition measurements can in these cases find lots of applications. The consequences of errors are often costly and early detection can save a lot of money. The inspection and quality control are also applications where recognition can be applied.

A major threshold that has to be crossed to achieve a wider industrial use of ultrasonic measurements might not be a technical, but a psychological one. The feeling for what can be ”seen” with sound is a matter of experience. To rely on a method that matches patterns recorded through non-tactile measurements that have no similarities with the human senses, seems questionable at a first glance. Hopefully this threshold will be removed with a widely spread experience from successful ultrasonic measurements.

9.3 Topics for Future Research

The author is well aware of the fact that several lose ends has been dropped on the way since the available time has been limited. This means that the main topic for each measurement type is handled but all further interesting expansions might not have been done. In this section I will try to emphasize the topics that from my point of view are the most interesting ones to continue the research on.

The knowledge that we get from the bat principles gives us a kind of ”right answer list”. By studying the bats we can see how they solve several of the measurement problems that we want to solve. This is not a specific topic for further research but a basic rule when we run into trouble.

The results from frequency sweeps are so far very encouraging. However, a systematic survey over the effects of various sweep parameters ought to be done. The possibilities with nonlinear sweeps and the use of harmonics should also be examined. This might imply the use of elements with a higher bandwidth.

The object recognition measurements provide several topics for future research. The system identification methods applied in the work concerning recognition of an object in several positions show that feature extraction is possible in a more general way than with the direct shape comparison methods. It was stated already in Chapter 2 that the description of the reflecting object is a key problem. These and other identification methods that has the capability of describing objects and echoes in terms of transfer functions can probably be valuable tools.

The description model for multi-frequency echoes is not working exactly as intended when it is used for temperature compensation with a translation in the direction of the frequency axis. Presumably the problem can be found in the selectivity of the frequency separation. A survey over possible methods and filters ought to be done. Special attention should then be paid to the transient behavior in the beginning of the echo. The influence of the sweep rate should also be considered. The potential for this method can be very high if a true temperature compensation that eliminates the interference effect can be achieved.

An application that is known to have severe crosstalk problems is the navigation and obstacle detection sonar in mobile robots. These measurement systems usually consist of a ring of ultrasonic transducers around the robot. Since the adjacent elements have overlapping lobes they can’t be used at the same time. This means that the sequence for one measurement using all elements in the ring often can’t be made sufficiently fast. The use of wisely chosen frequency sweeps combined with matched filters can probably improve the performance and let more elements operate simultaneously.

This last topic once again points out how inferior we still are compared to the ultrasonic experts, the bats. This is a problem where their solution is superior to what ever we may invent. A close study of the great knowledge that has been collected about the bats will probably give some new ideas. So the conclusion once again is: listen to the bat and learn!

References

Audenaert, K., H. Peremans, Y. Kawahara and J. van Campenhout (1992) Accurate Ranging of Multiple Objects using Ultrasonic Sensors, *IEEE Int. Conf. on Robotics and Automation,* Nice, France, pp. 1733-1738.

Barshan, B., R. Kuc (1990) Differentiating Sonar Reflections from Corners and Planes by Employing an Intelligent Sensor*, IEEE Trans. on Pattern Analysis and Machine Intelligence*, Vol 12, No. 6, June, pp. 560-569.

Blitz, J. (1967) *Fundamentals of ultrasonics*, 2nd Ed., Butterworths, London.

Buchberger, M., K-W. Jörg, E. von Puttkamer (1993) Laserradar and Sonar Based World Modeling and Motion Control for Fast Obstacle Avoidance of the Autonomous Mobile Robot MOBOT-IV, *IEEE Int. Conf. on Robotics and Automation,* Atlanta, GA, pp. 534-540.

Craig, J. (1989) *Introduction to Robotics Mechanics and Control,* 2:nd Ed, Addison-Wesley, ISBN 0-201-09528-9.

Freire Bastos, T., J.M. Martín, L. Calderón, R. Ceres (1994) Ultrasonic Signal Variations Covered by Thermal Disturbances, Sensors and Actuators, Vol. A44, No. 2, August, pp. 131-135.

Griffin, D.R. (1958) *Listening in the Dark*, Yale University Press, New Haven. Other issues: Dover Publications, New York (1974), and Cornell University Press, Itacha N.Y. (1986).

Hirzinger, G. (1987) Sensory Feedback in Robotics – State of the Art in Research and Industry, *IFAC World Congress on Automatic Control*, Vol 4, pp. 204-217.

Holmberg, P. (1992a) Applied Signal Processing in Measuring Applications - Towards Sensor Controlled Robots, (Licentiate thesis) Dept. of Physics and Measurement Technology, Linköping University, Sweden.

Holmberg, P. (1992b) Robust Ultrasonic Range Finder, *Meas. Sci. Technol*., Vol. 3, pp.1025-1037.

Hong, M. L., L. Kleeman (1992) Analysis of Ultrasonic Differentiation of Three Dimensional Corners, Edges and Planes, *IEEE Int. Conf. on Robotics and Automation,* Nice, France, pp. 580-584.

Idesawa, M. (1987) Optical Range Finding Methods for Robotics, *Fourth Int. Conf. Robotics Research*, Santa Cruz, pp. 91-98.

Jarvis, R. A. (1983) A Perspective on Range Finding Techniques for Computer Vision, *IEEE Trans. on Pattern Analysis and Machine Intelligence*, Vol PAMI-5, No. 2, March, pp. 122-139

Jacobson, B. (1975) *Medicin och Teknik* (in Swedish), Andra upplagan, Studentlitteratur, Lund, Sweden.

Johansson, R. and G. Lindstedt (1995) An Algorithm for Continous-Time State Space Identification, *IEEE Conf. Decision and Control*, New Orleans, LA, pp. 721-726.

Jonsson, T. (1985) Pulsad Akustisk Bildgenerering: Multipelspridning - Avbildningsdefekter (in Swedish), FOA report D 30385-E.

Kinzler, L.E., A.R. Frey (1982) *Fundamentals of Acoustics*, 2nd Ed, John Wiley, New York.

Klöör, P., T. Jonsson and Å. Wernersson (1988) Monitoring the Assembly Sequence Using Pulsed Coherent Sonar, *9th Int. Conf. Assembly and Automation*, London, March.

Knoll, A. (1991) Ultrasonic Holography Techniques for Localizing and Imaging Solid Objects. *IEEE Transactions on Robotics and Automation,* Vol 7, No 4, August 1991.

Kuc, R., B. Barshan (1992) Bat-Like Sonar for Guiding Mobile Robots. *IEEE Control Systems*, August.

Lindstedt, G. (1991) A Platform for Sensor Feedback Experiments in Robotics, Report TEIE-7036, Dept of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindstedt, G., K. Kowalew (1991) Distance Measurement in Robotics Using Ultrasonic Sensors. Report TEIE-7049, Dept. of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindstedt, G., G. Olsson (1991) An Ultrasonic System for Robot Sensor Feedback, Report TEIE-7049, Dept of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindstedt, G., G. Olsson (1992) Using Low Cost Piezoelectric Ultrasonic Sensors for 3D Measurements in Robots*, IFAC International Symposium on Low Cost Automation,* Vienna.

Lindstedt, G. (1992) Computation of Spatial Angle from Ultrasonic Measurements, Report TEIE-7055, Dept of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindstedt, G., G. Olsson (1993) Using Ultrasonics for Sensing in a Robotic Environment. *IEEE Int. Conf. on Robotics and Automation,* Atlanta, May.

Lindstedt, G. (1993) A Versatile Ultrasonic Sensor System for Industrial Robot Applications, (Licentiate thesis) TEIE-1003, Dept. of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindstedt, G. (1995) Design Catalogue of Modules in Ultrasonic Measurement System, Report TEIE-7084, Dept of Industrial Electrical Engineering and Automation, LTH, Lund, Sweden.

Lindström, K., L. Mauritzson, G. Benoni, P. Svedman and S. Willner (1982), Applications of air-borne ultrasound to biomedical measurements, *Medical & Biological Engineering & Computing,* may, pp. 393-400.

Mattila, P, F. Tsuzuki, H. Väätäjä and K. Sasaki (1995) Electroacoustic Model for Electrostatic Ultrasonic Trasnducers with V-Grooved Backplates*, IEEE Trans. on Ultrasonics, Ferroelectrics, and Frequency Control*, Vol 42, No. 1, January.

Mensa, D. L. (1991) *High Resolution Radar Cross-Section Imaging*, 2nd Ed, Artech House, Norwood, MA, ISBN 0-89006-389-3.

Nordling, C. and J. Österman, (1980) *Physics Handbook*, Studentlitteratur, Lund, Sweden.

Polaroid Corporation, (1980) *Ultrasonic Ranging System*, Polaroid Corp., Norwood Massachusetts.

Regtien, P.P.L. and H.C. Hakkesteegt (1988) A Low-Cost Sonar System for Object Identification, *Int. Symposium on Industrial Robots*, April.

Rogdahl, S. (1994) Objektlokalisering med ultraljudssensorer i robotmiljö, (in Swedish), Report TEIE-5093, Dept. of Industrial Electrical Engineering and Automation, LTH, Lund.

Simmons, James A (1989) A view of the World through a Bats Ear: The Formation of Acoustic Images in Echolocation. *Cognition*, 33, 155-199, 1989

Skolnik, M. I. (1962) *Introduction to Radar Systems*, McGraw-Hill, USA.

Sobral, L. (1995) System Identification Applied to Ultrasonic Detection Problems, (Master Thesis), TFRT-5532, Dept of Automatic Control, LTH, Lund, Sweden.

Sobral, L., R. Johansson, G. Lindstedt, G. Olsson (1996) Ultrasonic Detection in Robotic Environments, *IEEE Int. Conf. on Robotics and Automation,* Minneapolis.

Suga, N. (1990) Biosonar and Neural Computation in Bats, *Scientific American*, vol 262, pp. 60-68, June.

Sundström, N., P.O. Börjesson, N.-G. Holmer, L. Olsson, and H.W. Persson (1991) Registration of Surface Structures Using Airborne Focused Ultrasound, *Ultrasound in Medicin and Biology*, Vol. 17, No. 5, pp. 513-518.

Texas Instruments (1989) *Linear Circuits Data Book, Amplifiers, Comparators, and Special Functions*, Vol 1

Urban, M. (1991) Ultrasonic Transducers and their Application on Robot Control, TEIE-7046, Dept. of Industrial Electrical Engineering and Automation, LTH, Lund.

Urick, R. (1983) *Principles of Underwater Sound*, 3rd Ed., McGraw Hill.

Watanabe, S. and M. Yoneyama (1992) An ultrasonic Visual Sensor for Three-Dimensional Object Recognition Using Neural Networks. *IEEE Transactions on Robotics and Automation,* Vol 8, No 2, April 1992.

Webb, P. and C. Wykes (1996) High-Resolution Beam Forming for Ultrasonic Arrays, *IEEE Transactions on Robotics and Automation,* Vol 12, No 1, February 1996.

Wong, G. (1986) Speed of Sound in Standard Air, *J. Acoustical Soc. Am.*, Vol. 79, No. 5, pp. 1359-1366, May.