Ultrasound Measurements in Moving Multi-phase Suspensions

Johan Carlson

Master of Science Programme
Department of Computer Science and Electrical Engineering
Division of Signal Processing

Abstract

In many industrial applications, non-invasive measurements of different flows play an important role. For flows consisting of only one phase, such as water or gas, there are several methods available. Most of these methods, however, is not applicable to multi-phase flows, such as paper fibre suspensions or iron ore slurry.

The overall goal of this project is to develop a method for measuring multi-phase flows with ultrasound.

In this thesis, as a first step of the project, we use ultrasound to identify some of the physical phenomena occurring in a medium. We have examined the effects of absorption, dispersion, and scattering for different concentrations of solid particles in water.

The results indicate that particle concentration as well as other properties, such as temperature and particle size, significantly affect the received signal.
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Preface

This thesis is the result of my master’s project at the Division of Industrial Electronics, Luleå University of Technology. During the progress of this work I have been confronted with some of the brutal forces of mother nature. With my, quite theoretical, background of signal processing and computer science, the task of designing and operating experimental equipment was a great challenge. For almost every new idea I wanted to try, I ended up either modifying, or rebuilding the test system, and when you have a time limit on your work, this is indeed frustrating. Now, afterwards, I can only say that even if I did not come as far as I had planned, I learned a lot!

This work could, however, not have been done without the help from the people around me. I would like to address my very special thanks to my supervisors Anders Grennberg and Jerker Delsing, for their guidance and encouragement, and to Philippe Symoneaux, for helping me with all the practical matters concerning the measurements. Others who helped with theoretical as well as practical problems are: Torbjörn Löfquist, Jan van Deventer, Lennart Gustafsson, Olov Marklund, Kennet Hartvik. Many of the practical problems could not have been solved without Kurt Nyberg at the Technical Support Service at the University, for building the test systems, and professor Hans Theliander at Chemical Engineering Design, for helping me with the design. I would also like to thank my father for proofreading and for his valuable comments.

Thank you all!

Johan Carlson
Chapter 1

Introduction

1.1 Background

In chemical and process industry, such as paper pulp and mining industry, multi-phase flows play important role. One example of a multi-phase flow is water mixed with air bubbles and some solid particles. The task of measuring the different parameters of such a system is not an easy one. The book by Chaoki et. al [1] gives a good overview of current methods for measuring multi-phase flows. Examples of such methods are optical, radiation, and particle tracking techniques. Sometimes a probe is inserted in the flow.

All of these methods have major drawbacks which makes the search for new techniques interesting. In, for example, iron ore slurry or a paper fibre suspension, the mixture is opaque and, for this reason, optical methods are unsuitable. Tomographic techniques, using for example gamma rays or X-rays, can be hazardous to the operator. These techniques are also quite expensive. The oldest and most straightforward techniques involve either inserting a probe or sampling the flow with valves closing and opening (see fig 1.1). If the flowing medium contains solid particles, inserting a probe may restrict the flow and cause congestion in the system.

Figure 1.1: In the left figure measurements are made by inserting a probe in the flow. In the right figure the flow is measured by taking out samples with the closing and opening of valves.
Ultrasound has the advantage of being both inexpensive and harmless, at least for the frequencies and intensities of interest for this application. Also, if the sound energy is sufficiently low, the measured medium is not affected. Although similar to optical methods, ultrasound has the advantage that it can penetrate even opaque media.

1.2 Previous Work

In 1986, Anders Sixtensson [2] examined the possibilities of determining the concentration of paper-fibres in water by measuring the acoustic attenuation of an ultrasonic echo. The author found that the method probably is applicable in a real-life situation, but that using a continuous wave in a pipe resulted in ringings in the system, which distorted the measurements.

Currently, Torbjörn Löfquist at Luleå University of Technology is working on a project very similar to that of Sixtensson. The main difference is that pulsed sound is used instead of continuous waves. Recently, some results of this study have been presented in [3].

1.3 Goal

The overall objective of the present project is to develop a method for measuring mass fractions in multi-phase flows, using ultrasound. One of the main ideas is to have one transmitter and several receivers, and then study how the received image is affected by the properties of the suspension.

In this thesis, as a first step of the project, we will try to identify how some of the sound characteristics are affected by different physical properties of the suspension. How will, for example, the particle size affect the energy spread of the ultrasonic signal, how will density and distance affect the absorption of energy, and how will the shape of the signal envelope change due to dispersion in a medium.

Next chapter contains a description of the experimental equipment. The planning and execution of experiments is described in chapter 3, followed by the analysis in chapter 4. The thesis ends with a discussion of the results and of what we will do next.

1.4 Basic Ultrasonic Theory

Ultrasound is often defined to be sound with higher frequencies than can be perceived by the human ear. This is, however, a quite wide concept. In this project, we have chosen to use sound with frequencies in the range 500 kHz to 10 MHz, typically around 3 MHz.

How sound propagates in a medium also varies considerably. In solids, there are longitudinal, transversal, and surface waves, but in fluids there are only longitudinal waves. We also assume that the wave front is planar and orthogonal to the direction of the sound.

To avoid standing waves and resonance phenomena, which can occur if continuous waves are used, we use only pulsed sound in the experiments. The time interval between pulses was
chosen much longer than the propagation time of one pulse in the suspension. Figure 1.2 shows the result of measuring 100 pulses after passing through 125 mm of pure water. The variations in shape and arrival time between the received signals carry information about the medium through which the sound waves have passed. For example, signals which have passed through a homogeneous medium with no movement will look much more alike than the pulses in figure 1.2.

![Graph](image)

Figure 1.2: Example of 100 pulses which have passed through 125 mm of pure water.

The choice of frequency of the transmitted pulse is by necessity a compromise between resolution and pulse energy. Pulses with higher frequency are attenuated more than pulses with lower frequency, and high-frequency pulses therefore require higher energy. Increasing the pulse energy could, however, affect the measured medium. On the other hand, if the particles are smaller than the wavelength of the sound, it becomes impossible to distinguish between them, even though they affect the received signal. One example of this is when there is a high concentration of small air bubbles in the suspension. The received signals are strongly attenuated, but there are only minor variations between them.

A frequency of 3 MHz corresponds to a wavelength of about 0.3 mm. This frequency is low enough for the signal to pass through the medium without too much loss of energy. This wavelength is also much shorter than the radius of the solid particles in the medium, which makes it possible to distinguish between them. Figure 1.3 shows the frequency characteristics
of the transmitting transducer.

There are well-established mathematical models of sound propagation both in fluids and in solids, see for example [10]. In the present application, however, we have a mixture of fluid, air bubbles and solid particles, and the derivation of fundamental mathematical models becomes too complicated and beyond the scope of this thesis. In the chapters to follow we will therefore use much simpler models.
Chapter 2

Experimental Setup

This chapter describes the equipment used in the experiments. This can be divided into two main categories, the data acquisition equipment, and the flow equipment.

2.1 Data Acquisition Equipment

With data acquisition equipment we mean the hardware and software required to collect data. Figure 2.1 gives a schematic view of this equipment. The box marked Flow system represents either one of the two systems used. They will be described in detail later.

![Schematic view of the experimental setup](image)

Figure 2.1: Schematic view of the experimental setup. The flow system can be either the flow pipe or the suspension tank

2.1.1 Pulse generator

The pulse generator used in the experiments is a Panametrics pulser receiver, model 5052PR, with adjustable pulse energy and repetition rate. In the flow pipe, the pulse energy was set to
its lowest level, and to level 3 (out of four) in the suspension tank. The reason for increasing
the pulse energy in the suspension tank was that the propagation distance of the sound is
longer, and thus, introducing more noise to the system.

2.1.2 Ultrasonic transducers and pinducers
For transmitting ultrasonic pulses we used a piezoceramic transducer with center frequency
of 3 MHz, and a diameter of 14 mm. It was manufactured by Ceram AB, Lund, Sweden.

The receiver is a point-like ultrasonic microphone, pinducer, with a bandwidth of 10 MHz.
The crystal diameter of the receiver is 1.35 mm. The large bandwidth makes it possible to
use different transducers without having to change the receiver. The pinducer used was a
VP-1093 pinducer with a 10 MHz crystal, manufactured by Valpey-Fisher Corp.

2.1.3 Digitizing oscilloscope
The oscilloscope is a Tektronix TDS 724A digitizing oscilloscope. We used a sampling time
of 2 ns, and an input bandwidth of 20 MHz in all measurements.

2.1.4 Temperature probe
For each registered ultrasonic pulse, we also measured the temperature of the suspension.
For this, we used a Systemteknik S2541 Thermolyzer, with a specified accuracy of $5 \cdot 10^{-3}$
°C. The probe was then connected to the serial port on the computer.

2.1.5 Computer and data acquisition software
The oscilloscope transfers the digitized data to a computer, in this case a 120 MHz Pen-
tium, using a GPIB data collection card. We have written specialized software to store the
ultrasonic signal as well as the temperature.

2.2 Flow Equipment
As mentioned in the introduction, the goal of this project is to develop a method for measur-
ing mass fractions in multi-phase flows. As a first step in this project, we have to determine
which sound characteristics that will change, and which physical phenomena that cause these
changes.

2.2.1 Flow pipe
To examine how flow rate and flow direction affect properties of the received signal, for
example the waveform, we used the flow pipe described in figure 2.2. Unfortunately, this
system had a few major drawbacks which prevented us to measure what we planned. First,
we noticed that the heating of the pump caused the temperature of the flow to increase
during the measurements. Since the speed of sound is strongly temperature-dependent, this made it impossible to observe some of the most interesting properties of the flow. Another problem was that when we mixed solid particles in the flow, these sedimented and got stuck in the container.

To overcome these difficulties and to save time, we designed the suspension tank described in the next section.

![Diagram of flow pipe with labels: pinducer and transducer.](image)

**Figure 2.2:** Flow pipe used in the measurements.

### 2.2.2 Suspension tank

In most of the measurements we used the suspension tank depicted in figure 2.3. The volume of this tank is much less than that of the flow pipe, and this makes the suspension tank much easier to handle. The drawback is that we lose information contained in flow speed and flow direction. The stirring also causes some turbulence that could influence the measurements.
Figure 2.3: Suspension tank used in the measurements.
Chapter 3

Measurements

3.1 Temperature

Since the speed of sound is strongly temperature-dependent, all measurements also included logging of the temperature, before and after recording the ultrasonic pulse. In addition, all experiments were conducted at temperature in the range of 18-25 °C. When possible, the effects of temperature changes were compensated for according to the model described in section 4.2.

3.2 Suspensions

In this section we describe the suspensions we have investigated and how they were prepared.

3.2.1 Water

Since we can not measure the transmitted pulse before it enters the medium, we need a reference signal to compare the received signals with. In all of the suspensions water takes in excess of 95 per cent of the total volume. This, and the fact that water is approximately non-dispersive (see section 4.3) under the current experimental conditions, makes water a suitable medium for recording the reference signal.

For simplicity reasons, we used ordinary tap water.

3.2.2 Water with air bubbles

Whether we like it or not, air bubbles are always present in the suspensions. We first performed these measurements using the flow system, to find out how the amount and size of air bubbles affect the sound waves. We found that small air bubbles resulted only in attenuation of the received energy, but that the signal waveform was not perturbed.
3.2.3 Water with glass spheres

The second type of suspension we studied was a suspension of glass spheres in water. There are certainly air bubbles present, but as we found in previous experiments, this has no significant effect. This is, of course, under the assumption that there are no significant interaction effects between the glass spheres and the air bubbles.

The spheres used have a diameter of 2 mm which should be large enough to cause scattering effects (see section 4.4). The density was determined to be about 2.5 g/cm³, to be compared with the density of water, which for this temperature interval, is about 1.00 g/cm³ [4].

We prepared mixtures with a volume fraction of 1, 2, and 3 per cent, respectively.

3.2.4 Water with plastic beads

The third type of suspension was a mixture of water and cylindrical plastic beads, as depicted in figure 3.1 We examined suspensions with volume fractions of 1 and 2 per cent. All measurements with these plastic beads were made in the suspension tank.
Chapter 4

Analysis

This chapter begins with a description of the calculations we performed on almost all measured data. The following sections then deals with the theory of different acoustical phenomena, and how these can be observed through the measurements.

4.1 Analysis Tools

4.1.1 Estimating time delays

For each transducer position we measure several pulses, for example 100. These pulses were measured under almost identical conditions, and should therefore look very much alike. Figure 4.1 shows the signals recorded for 100 pulse transmitted at the same point. The variation in amplitude contains information about the suspension. There is also a small time delay between the pulses. Some of this comes from triggering of the equipment, some from other mechanical disturbances, but there could also be some information about the properties of the suspension.

No matter where the time delays come from, they must be estimated. This section describes how we did that.

A frequently used method to estimate time delays, $\theta$, between two pulses, $s[k]$ and $r[k]$, with the same shape, is to maximize the cross-correlation function, $R_{sr}[k]$, according to equation (4.1).

$$\theta = \arg \max_k R_{sr}[k] = \arg \max_k \sum_{n=-\infty}^{\infty} s[n] r[n - k].$$  \hspace{1cm} (4.1)

This will, however, only yield an estimate in whole samples. In [6] an alternative method for estimation of subsample time delays is proposed. Given that the delay is small, and the signal is narrowband, this estimator has been shown to perform better than the cross-covariance.

To estimate the subsample time delay, $\theta$, between $r[k]$ and $s[k]$, where $r[k]$ is the reference signal, we used the following algorithm:
Define the vectors
\[
\mathbf{r} = \begin{bmatrix} r[1] & r[2] & \ldots & r[N] \end{bmatrix}^T \\
\mathbf{s} = \begin{bmatrix} s[1] & s[2] & \ldots & s[N] \end{bmatrix}^T
\] 
(4.2) 
(4.3)

\[
\mathbf{b} = -\frac{\tilde{\mathbf{r}}}{\sqrt{\mathbf{r}^T \mathbf{r}}},
\] 
(4.4)

where \(\tilde{\mathbf{r}}\) is the Hilbert transform of the reference signal.

The subsample time delay estimate, \(\hat{\theta}\), is then given by:
\[
\hat{\theta} = \frac{1}{\omega_0} \arcsin \left( \frac{\mathbf{s}^T}{\sqrt{\mathbf{s}^T \mathbf{s}}} \mathbf{b} \right).
\] 
(4.5)

If the angular centre frequency, \(\omega_0\), is unknown, it can be estimated by introducing a known time delay and rewriting (4.5). This gives us:
\[
\tilde{\omega}_0 = \frac{1}{T_s} \arcsin \left( \begin{bmatrix} 0 & \mathbf{r}^T \end{bmatrix} \begin{bmatrix} \mathbf{b} \\ 0 \end{bmatrix} \right).
\] 
(4.6)

A complete derivation of the Hilbert correlator can be found in [6].

For larger delays, we use a combination of these two methods. First, we estimate the integer part of the delay, using a cross-correlator, and then after shifting according to this, we estimate the remaining subsample delay with the Hilbert correlator. The next section describes how to shift a discrete-time signal a fraction of a sample.
4.1.2 Sensitivity to distance changes

In this section, we describe how we investigated the sensitivity of the sound propagation time to changes in distance. The purpose was to see how vibrations and other mechanical disturbances in the suspension tank affect the time delay between recorded signals.

After placing the transducer and pinducer so that their centers were aligned, we measured 1000 pulses in pure water for 6 different rotation speeds, we calculated the time delays $\theta$, measured in samples, relative to the first measured pulse in each series.

From the histogram plot over the 1000 $\theta$:s, as in figure 4.2, we see that they seem approximately normally distributed.

Figure 4.2: Distribution of time delays in pure water with no movement.

In figure 4.2 we see that the average time delay is not equal to zero. If they had come from mechanical disturbances, they should be expected to have an average equal to zeros. This observed bias comes from the fact that the time delays are estimated relative to the first pulse, and not to exact measurement. The interesting information is, however, the interval in which the time delays fall. If we assume that the delays are samples from a normal distribution with mean $\mu$ and standard deviation $\sigma$, a 95 per cent confidence interval can be calculated. Since we do not know the mean and standard deviation, these have to be estimated. This was done as in equation (4.8).

$$\hat{\mu} = \frac{1}{N} \sum_{k=1}^{N} \theta_k$$  \hspace{1cm} (4.7)
\[
\hat{\sigma} = \left[ \frac{1}{N-1} \sum_{k=1}^{N} (\theta_k - \hat{\mu})^2 \right]^{1/2} 
\]

(4.8)

The corresponding precision limits are then given by [9]

\[
P_\theta = t_{N-1} \cdot \hat{\sigma},
\]

(4.9)

\[
P_\mu = t_{N-1} \cdot \frac{\hat{\sigma}}{\sqrt{N}},
\]

(4.10)

where the \( t \)-distribution is used because \( \hat{\sigma} \) is an estimated value.

This means that, with 95 per cent probability, the interval \([-P_\theta - P_\mu, +P_\theta + P_\mu]\) will enclose the time delays, \( \theta_k \).

To see whether or not the time delay corresponding to the precision limit, \( P_\theta + P_\mu \), is likely to come from vibrations, we calculated the change in distance this would require between the transmitting transducer and the receiver. The sampling time was \( 2 \cdot 10^{-9} \)s and the speed of sound is typically around 1480 m/s. For the measurements shown in figure 4.2 the upper interval limit was determined to be 1.39 samples. Equation (4.11) gives the corresponding change in distance, \( ds \).

\[
ds = (P_\theta + P_\mu) \cdot \overline{T} \cdot v = 1.39 \cdot 2 \cdot 10^{-9} \cdot 1480 \approx 4.0 \ \mu m
\]

(4.11)

As we see, the distance corresponding to the time delays is very small in this case, and most certainly come from different mechanical disturbances. In this case, however, we did not have any stirring of the water, and thus, almost no vibration disturbances.

Calculating the precision limits for different rotation speeds, we found that they increase proportionally even if we have only pure water in the system. At the same speed of rotation as in the rest of the measurements, the distance change corresponding to the time delay precision limit is still as low as 17 \( \mu m \). Since the transducer is mounted on a coordinate table below the tank, and only pushed against the surface by a small spring, the time delays could certainly come from small changes in propagation distance.

### 4.1.3 Shifting and averaging pulses

The purpose of measuring a series of pulses at each point was to achieve more reliable values, by averaging the pulses. If the mean of the measured pulses is calculated directly, the time delays will result in a deformation of the signal waveform. To avoid this, the pulses are first shifted according to the estimated delay. Since we were working with sampled signals, shifting a fraction of a sample requires interpolation. For simplicity reasons, we used Lagrange’s interpolation formula [7] in three points. Figure 4.3 describes how this interpolation applies.

\[
s'[n] = s(nT + \theta) \approx \frac{\theta(1-\theta)}{2} \cdot s[n-1] + (1-\theta^2) \cdot s[n] + \frac{\theta(1+\theta)}{2} \cdot s[n+1],
\]

(4.12)
where $s[n]$ is the measured pulse and $\theta$ is the estimated time delay. Noting that the polynomial coefficients depend only on $\theta$ and not on the signal itself, we can rewrite the shifting as a convolution

$$s'[n] = s[n] * h_\theta[n] = \sum_{k=-\infty}^{\infty} s[k]h_\theta[k - n], \quad (4.13)$$

where the filter $h_\theta[n]$ is given by:

$$h_\theta[n] = \begin{cases} 
\theta(1 - \theta)^2/2, & n = -1 \\
(1 - \theta^2), & n = 0 \\
\theta(1 + \theta)/2, & n = 1 \\
0, & \text{otherwise}
\end{cases} \quad (4.14)$$

When the pulses have been shifted using this filter, we calculate an average pulse. Figure 4.4(a) shows the standard deviation between the pulses, taken pointwise, before shifting. We see that the variation is largest at the zero-crossings of the pulses. In figure 4.4(b), the pulses have been shifted, and the remaining deviation is located at the maxima of the pulses. These variations increase when the concentration of solid particles increases, and thus, contain information about the suspension.

### 4.1.4 Envelope detection

One method to examine how the signal waveform changes is to examine the envelope of the signal. Assume that the signal can be modeled as

$$s(t) = e(t) \cdot m(t), \quad (4.15)$$

where $e(t)$ is the low frequency part, and $m(t)$ is the real high frequency part of the signal, for example $\cos(\omega t)$. If $e(t)$ and $m(t)$ have nonoverlapping spectra, we can write the *Hilbert transform* of $s(t)$ as (see [8], theorem 2.6.5 for proof)

$$\tilde{s}(t) = e(t) \cdot \tilde{m}(t), \quad (4.16)$$

where $\tilde{m}(t)$ denotes the *Hilbert transform* of $m(t)$.

Define

$$z(t) = s(t) + j \cdot \tilde{s}(t) = e(t) \cdot m(t) + j \cdot e(t) \cdot \tilde{m}(t). \quad (4.17)$$
Figure 4.4: Measurements in a suspension with water and 1% of glass spheres. (a) - pointwise standard deviation of 100 measured pulses before shifting. (b) - pointwise standard deviation of 100 pulses after shifting.

Let

\[ m(t) = \cos(\omega t). \]  

(4.18)

The *Hilbert transform* of (4.18) is [7]

\[ \tilde{m}(t) = -\sin(\omega t). \]  

(4.19)

Using this, and taking the absolute value of \( z(t) \), we obtain

\[ \text{abs}(z(t)) = |e(t)| \cdot \sqrt{\cos^2(\omega t) + \sin^2(\omega t)} = |e(t)|, \]

(4.20)

which is the envelope we were looking for, if \( e(t) \geq 0 \). As long as the modulating signal, \( m(t) \), is narrowband, \( \text{abs}(z(t)) \) works well as an envelope detector.

Figure 4.5 shows the result of applying this envelope detector to a measured ultrasonic signal. We see that the signal satisfies the assumptions made above.

### 4.1.5 Estimating the signal-to-noise ratio (SNR)

For the different suspensions, the signal-to-noise ratio (SNR) varies, depending on how much energy that passes through. The SNR also differs between measurements made in the flow
Figure 4.5: Result of applying the envelope detector to a measured ultrasonic signal. The pulse is the average of 100 signals measured in pure water.

pipe and measurements made in the suspension tank, mainly due to the longer distance of the latter.

A rough estimate of the average SNR per sample is given by equation (4.21).

$$\hat{SNR} = 10 \cdot \log_{10} \left( \frac{s^T s}{v^T v} - 1 \right),$$ \hfill (4.21)

where $s$ is signal plus noise, and $v$ is noise only. Both vectors are of the same length.

<table>
<thead>
<tr>
<th>Suspension</th>
<th>Average SNR/sample (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pure water</td>
<td>22.6</td>
</tr>
<tr>
<td>1% glass spheres</td>
<td>19.0</td>
</tr>
<tr>
<td>2% glass spheres</td>
<td>16.4</td>
</tr>
<tr>
<td>3% glass spheres</td>
<td>14.2</td>
</tr>
<tr>
<td>1% plastic beads</td>
<td>16.2</td>
</tr>
<tr>
<td>2% plastic beads</td>
<td>13.0</td>
</tr>
</tbody>
</table>

We see that the SNR decrease with concentration, even though the same pulse energy was used. This is expected. Since the variation in signal amplitude increases with concentration,
there will be more strongly attenuated pulses, and thus, the average SNR decreases. This correlation between SNR and concentration could, together with other information about the suspension, provide information about the measured medium.

### 4.2 Temperature Dependencies

#### 4.2.1 Theory

The speed of sound in a medium is highly temperature dependent, and there are accurate values available for pure water [5]. In the flow system (see section 2.2.1), the temperature of the suspension increases during the measurements, due to the heating from the pump. The speed of sound will therefore increase during the measurements, and for the case of pure water, the difference in propagation time, $\Delta t$, can be calculated as follows:

$$\Delta t = \frac{x}{v_2} - \frac{x}{v_1},$$

(4.22)

where $x$ is the distance through the medium and $v_2$ and $v_1$ are the corresponding stop and start velocities respectively. The velocities are determined by interpolating linearly in the table presented in [5]. Dividing the time delay in 4.22 with the sampling time gives a value that can be compared with the delays calculated as in section 4.1.1.

If we have only pure water in the system, equation 4.22 gives us a very accurate estimation of the propagation time. Then we assume that our suspension consists of both pure water and an unknown amount of a solid phase. The propagation time will then differ from the estimated value. This difference contains information about the concentration of the solid phase. One problem is the difficulty of measuring the propagation time. One way of dealing with this is to perform the measurements at two different temperatures, and use the method described in 4.1.1 to estimate the difference in propagation time. Another problem is that temperature changes is not the only factor resulting in time delays; there is also a phenomenon called dispersion (see section 4.3).

#### 4.2.2 Measurements

These measurements were performed in the flow system and the reason for this is that the pump causes a linear increase of the temperature. This causes a linear increase of the time shift between the measured pulses. In figure 4.6, the measured time delay is compared with the time shift estimated by equation (4.22).

We see from figure 4.6 that the estimated change in speed of sound, due to the temperature change, deviates slightly from the measured delays. When the measurements were performed, we observed that there was a fog-like abundance of very small air bubbles present in the system. We have not been able to measure the size and concentration of these bubbles, but the deviation from the values measured for pure water is small enough to make it likely that the bubbles caused it.
Figure 4.6: Time delay due to increasing temperature, estimated by interpolating in the table by Del Grosso and Mader [5], and measured by least-squares fit of a line.

\[ x(t) \rightarrow H(\omega) \rightarrow y(t) \]

Figure 4.7: Linear time invariant system. \( x(t) \) is the input signal, \( H(\omega) \) is the frequency response of the system, and \( y(t) \) is the output signal.

4.3 Dispersion

4.3.1 Theory

A medium is said to be dispersive if different frequencies propagate with different velocities through the medium. This can be modeled as a linear filter, as described in figure 4.7. We know from Fourier transform theory that, if \( H(\omega) = e^{-j\omega\tau} \), the output signal will be a delayed copy of the input signal. We say that the medium is dispersive if \( \arg H(\omega) \) is non-linear.

Equation (4.23) is an example of the frequency function of a dispersive filter.

\[
H(\omega) = A \cdot e^{-j\sqrt{\omega}}, \tag{4.23}
\]

where \( A \) is an arbitrary constant and \( \omega \) denotes the angular frequency.
Figure 4.8(a) shows a broadband signal used as input to the dispersion filter in equation (4.23). Figure 4.8(b) shows the output of the dispersion filter. Even though the amplitude of the filter is equal to one, there is a significant change in waveform of the output signal. This is because the different frequency components of the input signal is associated with a different phase shift.

Figure 4.8: The upper plot is a broadband input signal, and the lower plot is the output of the dispersion filter.

If the dispersion filter is applied to a measured ultrasonic pulse, we obtain the result shown in figure 4.9. The reason there is almost no change in signal envelope is that the signal is very narrowband.

### 4.3.2 Measurements

From the previous section we know that for a narrowband pulse, the change in signal waveform due to dispersion is quite small. From section 4.2 we also know that even very small changes in temperature can cause time delays of several samples. With the current experimental setup we can not control the temperature accurately enough to compensate for this effect. Thus, we can not determine which time delays come from temperature and dispersion,
Figure 4.9: The upper plot shows how dispersion affects the measured pulse. In the lower plot we see that there is very little change in signal envelope.

respectively. At the concentrations of solid particles we have investigated, it is difficult to see any signs of dispersion.

One way of dealing with this could be by also registering the echo from the tank surfaces and use this as a reference pulse. Figure 4.10 describes the principle for obtaining a reference echo, as used in [3].

4.4 Scattering

4.4.1 Theory

Under certain conditions, scattering occurs when sound waves are obstructed by particles. As a rule of thumb (see for example [10], chapter 8) one can assume that scattering occurs if the condition shown in equation (4.24) is satisfied.

\[
\frac{2\pi a}{\lambda} >> 1
\]

(4.24)

where \(\lambda\) is the wavelength of the ultrasound and \(a\) is the radius of the scattering particle.
The theory of scattering is much too complex to be investigated thoroughly in this thesis, and in order to analyze the possible effects of scattering, we have derived a simplified mathematical model. Figure 4.11 illustrates the general idea of the line-of-sight scattering model.

Assume that the plane waves transmitted from the transducer is reflected by a spherical particle as in figure 4.11. The result is that the received signal can be seen as a sum of delayed and attenuated waves. Equation (4.25) is a mathematical representation of this idea

\[ y(t) = \alpha_0 p(t) + \sum_{n=1}^{N-1} \alpha_n p(t - \tau_n) \]  

(4.25)

where \( \{\alpha_0, \alpha_1, \ldots, \alpha_{N-1}\} \) are the attenuation coefficients for the \( N \) scatterers, \( p(t) \) is the transmitted pulse, and \( \tau_n \) is the time delays corresponding to each scatterer. The attenuation coefficients depend on several factors, for example, the position of the scatterer and the relationship between wavelength and particle size.

Assume that a particle will affect the sound if it is located somewhere in the volume given by the transmitting transducer area and the distance \( D \). Figure 4.12 supports this, since we see that most of the received energy is located in that region.

If the particle coordinates \( (x_n, y_n, z_n) \) are uniformly distributed in this volume, the propagation distance can be expressed as

\[ s_n(x_n, y_n, z_n) = \sqrt{x_n^2 + y_n^2 + z_n^2} + \sqrt{x_n^2 + y_n^2 + (D - z_n)^2} \]  

(4.26)

where the coordinates \( x_n \) and \( y_n \) are random and uniformly distributed over the interval \( [-L, \frac{d}{2}] \) and the coordinates \( z_n \) are uniformly distributed over the interval \( [0, D] \). This implies that
we only have sound energy present in a cylindrical volume with the same diameter as the transducer. This is an oversimplification, but it holds as an approximation at this stage.

Using the distances $s_n$ from eq. (4.26), we obtain the corresponding time delay $\tau_n$ as

$$\tau_n = \frac{s_n(x_n, y_n, z_n) - D}{v},$$

where $v$ is the speed of sound and $D$ is the distance of the direct wave. For an equivalent in the sampled case, we divide by the sampling time $T_s$ to get the corresponding delay in samples.

For different relationships between particle size and wavelength, the scattered energy is distributed differently in space. For simplicity we assume that the energy spreads spherically from the scatterer. We also assume that no energy is absorbed by the scatterer. Thus, the attenuation coefficient for each scattered pulse can be approximated to be

$$\alpha_n = \frac{K_2}{s_{n2}^2}, \ n \neq 0$$

where $s_{n2}^2$ is the distance between the $n$:th scatterer and the receiver, and $K_2$ is some constant. Generally, the attenuation coefficients for the scattered pulses are very small compared with the attenuation of the direct wave $\alpha_0$. There are cases, however, when no direct wave is present, because some particles shadow the path. The amount of energy that passes by a particle depends on many things, and considering all details would yield very complex relationships. For simplicity, we therefore assume the received energy to be proportional to the area of the receiver which is not shadowed. This leads to the following expression for $\alpha_0$.

$$\alpha_0 = K_0 \left( 1 - K_1 \left( \frac{A}{A_{tot}} \right) \right)$$

Figure 4.11: Line-of-sight model for single scattering.
where $K_0$ is a constant corresponding to the attenuation of an undisturbed wave, $K_1$ is a constant corresponding to the amount of energy lost by the shadowing, and $\frac{A_{\text{r}}}{A_{\text{s}}}$ is the ratio of the shadow to the receiver area. If there is only one scatterer that shadows the receiver, we have the case illustrated in figure 4.13. The area of the intersection, $A$, as given in figure 4.13 then is

$$A = \begin{cases} \frac{r^2}{2} (\beta - \sin \beta) + \frac{a^2}{2} (\gamma - \sin \gamma), & \min(r, a) < R < r + a \\ \frac{\pi}{2} \left[ \frac{\min(r, a)^2}{2} \right] & R \leq \min(r, a) \\ 0, & R \geq r + a \end{cases}$$

(4.30)

where, by the law of cosines [7],

$$\beta = 2 \cdot \cos^{-1} \left( \frac{r^2 + R^2 - a^2}{2 \cdot r \cdot R} \right)$$

(4.31)

$$\gamma = 2 \cdot \cos^{-1} \left( \frac{a^2 + R^2 - r^2}{2 \cdot a \cdot R} \right)$$

(4.32)

If there are more than one scatterer shadowing the receiver, the expression gets more complicated. For simplicity, we sum the contributions of each scatterer, making sure that the total shadowing effect is not greater than $K_1$. 

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Figure 4.13: Situation where a scatterer shadows the receiver.

Figure 4.14: (a) - input signal. (b) - 10 outputs of the scattering model. \( SNR = 26 \, dB \), \( K_0 = 1 \), \( K_1 = 0.20 \), \( K_2 = 5 \cdot 10^{-8} \).

Figure 4.14(b) shows the result of applying the scattering model in equation (4.25) to 10 ultrasonic pulses. The particle radius in this simulation is 1 mm, which is the same as for the glass spheres used in the experiments. The concentration in this simulation was set to a volume fraction of 1 per cent.

We see in figure 4.14 that there are large variations in the pulse maxima, as for the case showed in figure 4.4(b). This phenomenon occurs when the receiver is shadowed by particles so that no direct wave is present. The effect of other scatterers, causing a time delay, shows mainly in the tail of the received pulse. This effect is small compared to the shadowing of the direct wave.
Another effect of scattering which is likely to occur, is that the total amount of received energy is spread over a larger area. At this stage, the model can only simulate the case when the transmitting transducer and the receiver is centered at the origin.

4.4.2 Measurements

If figure 4.15 we see that when there are glass spheres present, the region around receiver in which 95 per cent of the received energy is located is wider than for pure water. These regions become wider as the concentration increases.

![Graph showing received energy vs distance from origin for different concentrations of glass spheres.](image)

Figure 4.15: The marked intervals show the regions in which 95% of the total energy is located.

4.5 Absorption

4.5.1 Theory

In the previous section, we saw that for a medium that scatters sound waves, the received energy will spread over a wider region. The total amount of received energy will also be smaller, since some of the energy is reflected away from the receiver. It is difficult to determine if the attenuation of the signal comes from scattering or absorption, or possibly from
both. The assumption we have made is that if the received energy is not spread over a wider
area, then the attenuation is due to absorption and not to scattering.

4.5.2 Measurements

When we used the plastic beads (see figure 3.1), no spread of energy at the receiving end
was observed. This is shown in figure 4.16. In order to check whether the energy was

![Graph showing received energy in pure water and a suspension with 1 per cent of plastic beads.](image)

Figure 4.16: Received energy in pure water and a suspension with 1 per cent of plastic beads.

reflected back to the transducer, we arranged the system to measure echoes instead of through
transmission. Since the area of the transducer is quite large, even tiny fraction of the energy
reflected from the beads should be visible. This was, however not observed. The only echoes
we detected were those that could be connected to the reflecting surfaces of the container.
The energy lost could have been absorbed by the plastic beads.
Chapter 5

Conclusions

The goal of this Master’s project was to investigate which factors of acoustic signals are significantly affected by factors related to properties of a multi-phase suspension, and to identify underlying acoustical phenomena in the measured data.

We found that when we increase the concentration of the solid phase, the signal attenuation also increases. Since the suspension is under motion, the attenuation is not constant over a certain time interval, but varies considerably. We have also seen that the variations in signal amplitude increases with concentration.

When the solid particles are large compared to the wavelength of the sound, the signal energy is scattered over a wider area at the receiving surface. Using some \textit{a priori} knowledge about the properties of the solid phase, this knowledge can probably be used for determining the concentration.

Also, it seems that the scattering model derived in section 4.4, to some extent, can be used to explain the variations in the amplitude of the recorded signal.

One problem has been the influence of air bubbles, but if we can reduce the concentration and size of the bubbles by having a closed system, this effect will be negligible. It is, however, likely that air bubbles will be present in all practical cases.

The most important problem so far has been the high temperature dependence of the propagation time. Since we, at this point, have only one receiver, we have not been able to measure dispersion effects. At the frequencies used, water can be regarded as a non-dispersive medium. This means that signals that have passed through pure water could be used as reference signals. Since the temperature varies too much between different measurements, this was, unfortunately, not possible.
Chapter 6

Further Work

During the progress of this work, we have seen that the propagation time of the ultrasound probably can provide a lot of information about the suspension under study. Because the speed of sound in water is strongly temperature dependent, we have not been able to investigate this. The next step will therefore be to conduct measurements with several receivers simultaneously. This will involve modification of the data acquisition software as well as of the experimental setup.

Due to practical problems with the flow pipe system, almost all measurements were conducted in the suspension tank. However, with this equipment, we lose information related to the flow speed and direction. Also, the suspension tank introduces some turbulence that could have effect on the propagation time of the signal. To overcome these problems, we have started on the design of a new flow system which will be built in the near future. The new system will also include a flow meter, to control the actual speed of the flow, and a heat exchanger to reduce the temperature fluctuations caused by the pump.

In order to be able to measure how the phase velocity of a pulse changes in a dispersive medium, we need also to have access either to the transmitted pulse, or to some other reference signal which can be considered to be constant for all measurements. Registering echoes from the surfaces of the flow system may give such a reference signal. To achieve this, we have to modify both the experimental setup and the computer software.

Since various effects are likely to be frequency dependent, we will also perform measurements using different transmitting transducers with different center frequencies.

Varying different parameters of the solid phase, such as particle size and density, will also yield a better understanding of the overall process of sound propagation in suspensions.
Bibliography


