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Automatic Loudspeaker Location Detection for use in Ambisonic Systems

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Abstract

To allow easier configuration of Ambisonic surround sound systems different methods of automatically locating loudspeakers have been considered: first, separation measurements using a single microphone and propagation delays; second, angle measurement using a SoundField microphone and amplitude differences; and third, angle measurement using multiple microphones and propagation delays. Tests on the first two were conducted, and accuracies to less than a sample-interval for propagation delays and to within $\pm 1^\circ$ for angle measurements are shown to be feasible. A utility to allow continuous multi-channel audio input and output within MATLAB has been designed and using this an Ambisonics decoder has been implemented.

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Contents

Glossary of Terms

Abbreviation	**Details**
A/D	Analogue-to-Digital (converter)
ALSA	Advanced Linux Sound Architecture
API	Application Program Interface
ASIO	Audio Stream Input/Output
D/A	Digital-to-Analogue (converter)
FHT	Fast Hadamard Transform
FIR	Finite Impulse Response
GUI	Graphical User Interface
IDE	Integrated Development Environment
IDFT	Inverse Discrete Fourier Transform
IR	Impulse Response
IRS	Inverse Repeated Sequence
MLS	Maximum Length Sequence
MME	(Windows) Multimedia Extension
OATSP	Optimum Aoshima's Time-Stretched Pulse
PC	Personal Computer
PD	Pure Data
RF	Radio Frequency
SDK	Software Development Kit
SNR	Signal-to-Noise Ratio
TSP	Time-Stretched Pulse
VST	Virtual Studio Technology

Chapter 1

Introduction

With the ever increasing popularity of surround sound systems there is always the question 'what will be next?'. One possible answer, although not a new technology, is Ambisonics[1]. This allows for much greater flexibility in the number and arrangement of loudspeakers when compared to the more conventional 5.1 surround sound systems. Additionally it provides better sound localisation behind the listener and the ability to include 'height' information. However, to obtain the best performance the loudspeaker positioning is much more critical than with other systems. Although the standard loudspeaker arrangements are always regular, it is also possible to use irregular arrangements provided the loudspeaker locations are known by the signal decoder. Configuring this can be time consuming and potentially inaccurate, requiring fine tuning to obtain the best performance which is not appealing to those who are far more used to things working straight out the box. This project investigated how to determine the location of loudspeakers in a room automatically, thus enabling much easier system setup.

Chapter 2 provides background information relevant to this project, specifically concentrating on Ambisonics and its differences from other surround sound systems. The aims for the project are then given in chapter 3, considering the com-

[1]Ambisonics is a registered trademark of Nimbus Communications International

plete automated system in three sections: determining the location of the loudspeakers, implementing an Ambisonics decoder, and overall system implementation. In chapter 4 details of the tests used to determine the most suitable computer configuration for the remainder of the project are given, resulting in the requirement to develop a multi-channel audio handling utility for MATLAB. An overview of the significant parts of the development of this utility, from requirements to implementation and testing, are provided in chapters 5 to 7.

In chapter 8, different excitation signals are considered that could be used to determine the location of a loudspeaker, including implementation information for the signals to be used in all subsequent tests. The first set of tests conducted and their results are detailed in chapter 9. These tests used a single microphone and were designed to determine the practical accuracy possible when measuring distances using propagation delays. Chapter 10 then describes the test configuration and results obtained when using a SoundField microphone to measure the angle between loudspeakers. Methods using multiple microphones to determine the location of a loudspeaker are given in chapter 11.

A basic continuous Ambisonics decoder implemented within MATLAB is described in chapter 12 and then the possibility of implementing a complete, automatically configuring, Ambisonics system using wireless communications is discussed in chapter 13. An overview of the project time management is given in chapter 14, potential further work to extend this project is outlined in chapter 15 with chapter 16 bringing together conclusions on the project.

The project was conducted at the Department of Speech, Music and Hearing (TMH), in the School of Computer Science and Communication at the Royal Institute of Technology (KTH), Stockholm. Funding was supplied by both The Swedish Foundation for International Cooperation in Research and Higher Education (STINT) and the UK Socrates-Erasmus Council.

Chapter 2

Background

The technology used in the reproduction of sound is continuously advancing in all areas, ranging from the initial recording through to signal processing, storage, transmission and final output from loudspeakers. This section provides a brief overview of some of these developments that are specifically relevant to this project.

2.1 Surround sound history

Sound reproduction devices have become commonplace in many homes in the western world, appearing in different forms including computers, mobile phones, televisions, MP3 players, stereos and surround sound systems. These can all, however, be grouped into three categories: single (mono), double (stereo) and multiple (surround) channel devices.

Mono recording and playback devices were first created over a century ago and since then various advances have occurred. The most significant of these was the development of the stereo system as we know it today by Alan Blumlein of EMI and Harvey Fletcher of Bell Labs, amongst others, in the 1930s. Although this work also considered the use of many more than 2 channels, it was only in the 1970s that consumer multi-channel systems appeared[31]. These were quadraphonic (or quad) systems using four loudspeakers placed evenly around the listener. This technique was based on the simplified ‘intensity panning’, or ‘pan-pot’, approach already being used for stereo systems: that is, altering the relative amplitude of a signal sent to each loudspeaker to generate phantom images appearing somewhere between them[21]. However, as summarised by Gerzon[15] and Malham[21], there are many other mechanisms besides intensity difference used to provide cues to the location of a sound source. Without taking more of these into account when trying to create phantom images the results can be very poor[7, 21, 29]:

- instability and noticeable elevation[5] of images occurs in addition to a “hole in the middle” effect if the front loudspeakers are separated by more than $60°$, as with quadraphonics;
- there is generally poor localisation of sound sources behind the listener;
- localisation is between poor and nonexistent to the sides of the listener;
- image localisation is sensitive to both misplacement of loudspeakers and head position.

To avoid such problems the surround sound systems that were successful from just after this era, which are still in use in a similar form today, do not attempt to create a full $360°$ soundfield of stable phantom images. Instead, they tend to use 3 front loudspeakers to generate stable front phantom images (especially for off-centre listening), whilst additional loudspeakers around the listener produce a diffuse ‘surround’ sound, such as in the common 5.1 surround sound systems[31].

An alternative approach to surround sound reproduction is to “attempt to reconstruct in the listening space a set of audio waveforms which matches those in the recording space”[22]. To fully achieve this a very large[1] number of channels would

[1]Values quoted include 400,000 channels for a two-metre diameter sphere[15] and 40,000 channels for any radius sphere[22]

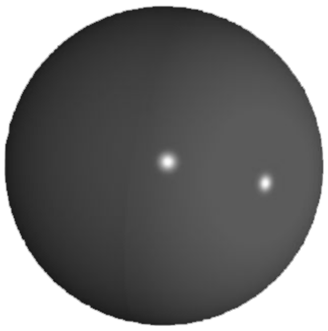
be required so different approaches to significantly reduce this are being developed. These are Wave Field synthesis[3], Holophony[24] and Ambisonics which, until recently, were regarded as competing technologies[22]. For this project only Ambisonics was considered.

2.2 Ambisonics

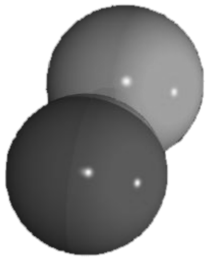
Ambisonic systems differ from other surround systems, such as those using the standard irregularly spaced 5.1 loudspeaker layout, both in the way sounds are recorded and in the way they are played back. For playback there is no predefined loudspeaker configuration—the only requirements are that the decoder being used is compatible with the configuration and the loudspeakers are located accurately. This does not mean that all configurations are as effective as others, but it does allow much greater flexibility in the layout and number of loudspeakers used. For example, the same Ambisonics signal can be decoded for use with 4 or more loudspeakers in a horizontal plane (pantophonic systems) or 6 or more loudspeakers in a 3-dimensional arrangement such as an octahedron or cuboid (periphonic systems)[16].

This is achieved by describing the required soundfield at the central listening location using spherical harmonics. For a first-order pantophonic system this requires 3 channels: W, an omni-directional mono sum; X, a (Front - Back) difference signal; and Y, a (Left - Right) difference signal. Additionally, to achieve full periphony a fourth component, Z, the (Up - Down) difference signal is also required. Combined together these signals, as shown in figure 2.1, are called B-Format and it is this on which the Ambisonic logo, shown in figure 2.2, is based.

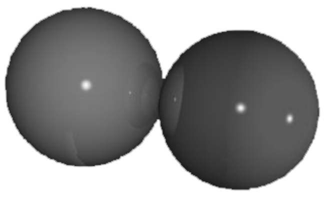
In comparison, the signals received by 5.1 surround sound systems are created specifically for the standard loudspeaker layout and are received as one signal per loudspeaker. Therefore, to ensure the end listener hears what the recording engineer intended, both the engineer's layout and the end listener's layout must be



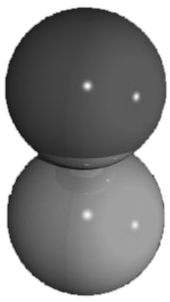
(a) W, an omni-directional sum



(b) X, a (Front - Back) difference



(c) Y, a (Left - Right) difference



(d) Z, a (Up - Down) difference

Figure 2.1: The four signals used to describe a soundfield in a first-order periphonic Ambisonic system where the front is orientated out of the page. Dark-grey represents signals in-phase whilst light-grey represents out-of-phase. Images generated by Víctor Luaña, Quantum Chemistry Group, Departamento de Química Física y Analítica, Universidad de Oviedo, Spain

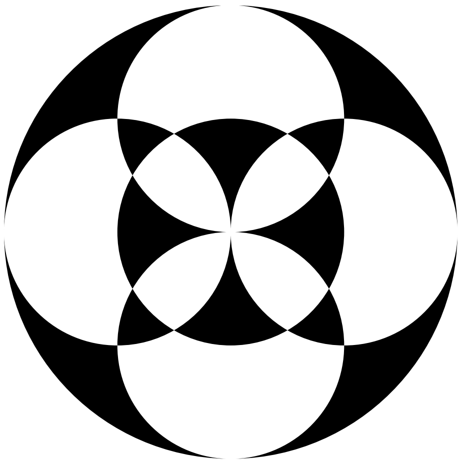


Figure 2.2: The Ambisonics logo includes: a bounding circle to represent the omnidirectional sum, W; two pairs of circles representing the Front-Back and Left-Right figure-of-eight difference components, X and Y; and a central circle to represent the Up-Down component, Z. Taken from [7].

similar—there is little flexibility to be able to use alternative loudspeaker arrangements.

For simple first-order Ambisonic systems the B-Format signals can either be electronically created from monophonic sources, as summarised in [21], or recorded live using a SoundField[2] microphone containing four capsules in a tetrahedral array. Higher order signals are currently predominantly generated electronically although the creation of microphones capable of producing such signals is a subject of current interest[28]. Although these higher order signals allow for improved directionality and a larger sweet spot, that is a larger area in which the listener can be positioned, the project only considered first-order signals.

From these signals, the individual loudspeaker feeds can be calculated using a simple decoding process. For regular layouts, each component of the B-Format signal is shelf filtered with different gains above and below 700 Hz to compensate for the frequency-dependant properties of human hearing. The output to each loudspeaker is then a weighted sum of these filtered signals with the weightings,

[2]See `http://www.soundfield.com/` for more information.

or decoding parameters, calculated using the angle of the loudspeaker relative to due front[16]. When using irregular loudspeaker layouts the decoding of the B-Format signal can be just as simple although determining the optimal decoding parameters is a much more complex process. Despite this, a couple of methods have been proposed to calculate such parameters, including one based on a Tabu search[44]. A more detailed explanation of this approach combined with a very thorough overview of decoding for regular and irregular loudspeaker configurations can be found in [42].

Chapter 3

Project Aims

Compared to other surround sound formats, Ambisonic systems offer many benefits:

- full periphony capability.
- the ability to use more speakers to give “a larger listening area and a more stable sound localisation”[20] as well as reducing “loudspeaker emphasis” (attraction of the phantom images towards the loudspeakers)[16].
- the ability to decode for any loudspeaker arrangement, although some are better than others.

For a system to work effectively the decoder must be configured for the loudspeaker layout used and so either the loudspeakers must be placed where the decoder requires, or the decoder must be told where the loudspeakers are located. Although this is dependant on the decoder, neither are very suitable solutions for most users of such systems—it is unusual to have a room containing all doors, windows and furniture in the correct locations for a perfect regular loudspeaker arrangement yet it is cumbersome and potentially inaccurate to manually measure where the loudspeakers have been located.

The aim of this project was therefore to investigate a means of automatically detecting the arrangement of loudspeakers within a room so that a suitable B-Format Ambisonics decoding algorithm could be determined for each loudspeaker. Additionally, to provide ease of installation and future extendability the project would specifically consider how this could be achieved with signal processing distributed amongst the loudspeakers using wireless communication.

To achieve this the project was divided into three sections: loudspeaker layout characterisation, decoding a B-Format signal, and practical implementation.

3.1 Loudspeaker Layout Characterisation

Within a room there are different approaches that can be used to determine the location of loudspeakers, each with their own advantages and disadvantages including accuracy, cost and practicality. For example, a cheap but impractical and potentially inaccurate approach is for the user to measure the loudspeaker locations and manually input the relevant measurements to the system. Alternatively a method based on radio-frequency (RF) transmitters and receivers in each loudspeaker could be implemented and even used for calibration whilst the system is in use. However, such a system would make all loudspeakers more expensive to produce and would require a minimum number of loudspeakers to successfully use triangulation.

Instead, methods utilising sound to determine the loudspeaker locations were investigated. This was because of an expected increase in accuracy compared to the RF approach, due to lower signal speeds, as well as reduced overall costs in a final system through the re-use of system elements such as amplifiers and loudspeakers. Additionally, this approach could allow the same technique to be used with both standard loudspeakers wired to a central unit implementing all the processing and dedicate loudspeakers using wireless links—something not possible with an RF approach.

When determining the loudspeaker locations, two potentially distinct parts to the problem were realised—the first was determining the loudspeaker locations relative to each other, and the second was 'grounding' this arrangement within the room relative to the listener. However, by using the listener's location when determining the loudspeaker locations these two could be combined.

Using sound, the distance between two points can be measured with propagation time delays. Therefore by placing a single microphone at the listener's location it was expected to be possible to determine the distance to each loudspeaker. However, to fully characterise a particular layout the angle of each loudspeaker relative to the listener would also be important. Therefore, to measure this an investigation of two different methods was planned: the first would use a single SoundField microphone producing a B-Format signal whilst the second would use four separate microphones in a known configuration. The former method would then measure the angle using amplitude differences whilst the latter would use timing differences. Both of these techniques have previously been used—a SoundField microphone in [11] and multiple omni-direction microphones in the Trinnov Optimizer produced by Trinnov Audio[1]—although no direct comparison of the techniques has been performed, including publication of the accuracy achievable under certain circumstances.

A third alternative considered briefly was the inclusion of a microphone within each loudspeaker enclosure. With such an arrangement it has been shown that, provided there are enough loudspeaker-microphone pairs, it is possible to determine all relative locations just using separation distances measured with sound propagation delays[30]. However this would require multiple loudspeakers[2], with each having to be specifically manufactured to include a microphone unless a method of using the loudspeaker itself as a microphone could be found. Additionally this would not locate the listener within the setup, and multiple nodes would be required at

[1]See `http://www.trinnov.com/` for more information.

[2]The number of microphone-loudspeaker pairs required would depend on the location estimation procedure used as well as the accuracy of the time delay measurements and the location accuracy required[30].

the listener's location to correctly rectify this. This idea was therefore rejected.

Although the first two methods could be analysed to determine if they were 'accurate enough' for a particular scenario, such as when using a first-order Ambisonics system, this would not necessarily allow for alternative systems to use the same approach without further testing. For this reason no specific target accuracy was set and additionally, due to time constraints, the required accuracy for any particular system would not be determined.

Thus, the main aims were to evaluate the accuracy achievable using both of the methods mentioned above including investigating the type of sound signals that could to used, how to process the signals received by each microphone and to what accuracy both distance and angle measurement could be achieved.

3.2 Decoding a B-Format signal

Many years of research have already been spent determining the 'best' method of decoding B-Format signals for different loudspeaker configurations, and this project was not aimed at furthering this work. Instead, the main aim was to implement a basic decoder for 3-dimensional regular loudspeaker arrangements based on "virtual microphone" signals calculated as described in [9], and used to determine the loudspeaker signals as in [42]. Although not optimised, provided the loudspeaker locations were approximately regular this could then be used to implement an initial fully working system which could be extended if time allowed. Such possible extensions included implementation of the Tabu search method described in [42] initially for 2-dimensions followed by a further extension to a 3-dimensional system, thus optimising the signals for the exact locations of the loudspeakers. Additional further extensions might also include the effect of different distances between loudspeaker and listener and the implementation of frequency response correction for each loudspeaker, although it was envisaged that there would probably not be enough time to pursue these.

3.3 Practical Implementation

A stand-alone system, including the distribution of processing amongst the loudspeakers via wireless communication, would have been an ideal end to the project. However this was not seen to be feasible due to the large amount of specific hardware that would have to be designed and constructed, including the writing and debugging of all associated firmware. Therefore the aim was to completely implement a working system using software running on a personal computer (PC) containing a multiple input/outut soundcard. By designing the software in a modular format, future hardware and algorithm development could then be achieved more easily. However it was also realised that, compared to dedicated hardware, there are many additional software 'layers' when using a PC that could potentially introduce unpredictable timings.

Due to this, it was necessary first to determine if the required timing accuracies and synchronisation between audio channels could be achieved using a PC. For example, to accurately measure the amplitude differences between the four B-Format signals produced by a SoundField Microphone, all signal recordings must be temporally aligned. Similarly, different timing delays between loudspeaker feeds could, depending on their magnitude, completely destroy the surround effect. Provided the required timings could be achieved, the aim was then to implement the rest of the system including determining the loudspeaker locations, calculating the required decoding parameters and then implementing signal decoding, outputting the required signals derived from files containing the B-Format signals. However due to the potential size of such files, and to model the software more accurately on the operation of the final system, the idea was to implement this decoding in real time: that is, to continuously decode the signals in small blocks of samples but without any gaps in the resulting audio output.

Further to this practical implementation, the requirements of a stand alone system would be considered such as the necessary specification of the wireless links—uni- or bi-directional, data rate, and the data needed to be transmitted—as well as where processing could be distributed and how the required timing accuracies might be achieved.

3.4 Summary

From these different sections of the project, the main aims were to:

- Configure a suitable test system, based around a PC, that could be used to record and output signals on multiple channels, analyse the recorded signals and ideally also allow a continuous Ambisonics decoder to be implemented.
- Determine the accuracy to which microphone-loudspeaker separation can be measured using sound propagation delays, including a comparison of different output signals from the loudspeaker.
- Determine the accuracy of angle measurement using both a SoundField microphone and four omni-directional microphones.
- Implement a basic Ambisonics decoder capable of decoding a B-Format signal into the signals required by each loudspeaker.
- Consider how such a system to automatically locate loudspeakers and then decode B-Format signals could be practically implemented using wireless communication between the different system components.

Chapter 4

Test Environment

To accomplish the aims of this project it was realised that a significant amount of audio data analysis would be required to compare the performance of different approaches to loudspeaker localisation. Such analysis can be achieved using a wide variety of tools ranging from custom written dedicated applications, for example using C or Java[1], to applications offering a high-level scripting or graphical language, such as MATLAB[2]. Comparing these extremes, the former generally allows better integration with other system components and the creation of highly optimised routines. Alternatively the latter provides powerful 'building blocks', allowing faster algorithm development, with easier graph plotting and data manipulation. Due to the need to implement and compare different algorithms without any initial requirements for highly optimised routines, it was decided to investigate the suitability of MATLAB.

[1]A high-level, object-oriented programming language from Sun Microsystems. See `http://java.sun.com/` for more information.

[2]Produced by The MathWorks, Inc. See `http://www.mathworks.com/` for more information.

4.1 Audio within MATLAB

MATLAB is renowned for being a powerful application capable of numerical data processing far more complicated than that required for this project. Therefore when considering its suitability this was not a significant factor—if any required processing blocks did not already exist it would be possible to implement them. However, the standard audio input and output capabilities offered were found to be much less versatile.

4.1.1 Windows Multimedia Extension (MME)

Within MATLAB, running on Windows XP, basic support for wave input and output uses the standard Windows Multimedia Extension (MME) and is provided through various functions as detailed in the MATLAB help[3]. For the playing and recording of mono or stereo signals, where sample accurate synchronisation between output and input is not required, these functions are more than adequate. However, to obtain useful results, this project required accurate synchronisation between input and output channels to measure signal propagation time delays. A simple solution when only requiring a single input is to record in stereo and loop the audio output directly back into the second input. By comparing the two recorded channels the variable time delays within the computer are no longer a factor. When more than one channel needs to be recorded, or more than one output channel is to be used, such a solution may become much more complicated. This is due to two independent limitations: first the MME drivers for the soundcard being used[4] represent the device as multiple stereo pairs, and second the MATLAB functions only support up to two channels[5].

[3]See either `http://www.mathworks.com/access/helpdesk/help/techdoc/matlab.html` or the help documentation supplied with MATLAB.

[4]M-Audio Delta 1010LT

[5]Two channel limitation indicated in MATLAB online help for sound, wavplay, wavrecord, audioplayer and audiorecorder.

To determine the significance of this channel division, a simple loop back test between two stereo output and input pairs was conducted, configured as shown in table 4.1. (Channels 3 to 6 were used because input channels 1 and 2 used a different connector type.)

		Input channel			
		3	4	5	6
Output channel	3	■			
	4			■	
	5		■		
	6				■

Table 4.1: Soundcard loop back configuration used for channel synchronisation tests. Shaded cells mark connections between output and input channels.

By calculating the position of the maximum cross correlation between the output signal and each input signal, the relative time offsets of the audio objects could be found. With such processing any non-repetitive output signal whose autocorrelation had a single distinct peak would be sufficient, and a chirp was one such signal. For this reason, two stereo `audioplayer` objects were used to play a Tukey windowed 0.8 second linear chirp from 20 Hz to 20 kHz whilst two stereo `audiorecorder` objects were used to record the four input signals, all configured to use 16 bit quantisation at 44.1 kHz. Additionally, to try and ensure the complete chirp was always recorded the output signal was zero padded at the start and end by 0.1 s and a recording duration of 1.0 s was used.

To eliminate any variations caused by using a combination of blocking and non-blocking functions only non-blocking functions were used, followed by a polling loop to wait for the test to complete, as shown in listing 4.1.

By running the tests with only one output channel active at any one time the crosstalk between channels was measured. In the worst case, the peak in cross correlation with a crosstalk signal was still 1.3×10^5 times smaller than that with

```
play(player1);
play(player2);
record(recorder1, duration);
record(recorder2, duration);

while isrecording(recorder1) || isrecording(recorder2)
    pause(0.04);
end
```

Listing 4.1: Sequence of commands used to output and record audio using the non-blocking functions in `audioplayer` and `audiorecorder` objects. (`player1` and `recorder1` were configured to use channels 3 and 4 whilst `player2` and `recorder2` were configured to use channels 5 and 6.)

the direct signal. As such it was determined that any crosstalk between the input channels would not have any significant effect on the delays measured.

To ascertain the extent of delay variation, each test was repeated 10 times using the same `audioplayer` and `audiorecorder` objects and this sequence was then repeated 100 times with new object instances each time. To simulate both multiple uses in close succession and longer periods of no use, a short delay of approximately 0.5 s was introduced between consecutive tests and a random delay between 30 s and 90 s was used in between creating new audio objects.

		Loop back connection (output⇒input)			
	Number of occurrences	**3⇒3**	**4⇒5**	**5⇒4**	**6⇒6**
Recording delay (samples)	**-67**	91	0	91	0
	-66	871	0	870	0
	189	7	98	7	98
	190	30	900	31	899
	446	0	1	0	2
	1214	1	1	1	1
	Total number of tests	*1000*	*1000*	*1000*	*1000*

Table 4.2: Variation in loop back delays using Windows MME drivers via `audioplayer` and `audiorecorder` objects in MATLAB.

A summary of the delays measured on each loop back are given in table 4.2. From this it can be seen that the relative delays of the two stereo inputs were different on most occasions whereas there is a strong indication that the two inputs of each pair, and hence the two stereo outputs, were the same almost every time. By comparing the delays for each test it was found that the latter was true on all but one occasion. In this spurious test the difference in delays was 256 samples, which notably was also a common factor in the difference between many of the other delay values measured: -67 and 189; and -66, 189, 446 and 1214. Additionally, the difference in delay between the two stereo inputs was found to always be either 0 or 256 samples.

One explanation for such behaviour is that somewhere within both the input and output signal paths data is handled in blocks of 256 samples, and the start of playing/recording must always align with the start of a block. As such, if both `audioplayer`/`audiorecorder` objects are set to `play`/`record` during the same block they will be perfectly synchronised. However, if a new block is started between the objects then there will be a 256 sample difference with the second object delayed relative to the first. The number of occasions when the objects are synchronised compared to the number of times when there is a time offset is then an indication of the time required to run the relevant function—in this case much longer for `record` than `play`.

As within MATLAB no method could be found to determine when each of these sample blocks was starting, it was concluded that there would be no way to guarantee the relative timing of more than one stereo output or more than one stereo input. Therefore, using this approach, output of more than two channels simultaneously could never be 100% reliable and recording using more than one channel would only be feasible if each signal was paired with a reference signal.

The anomalous delay of 1214 samples was found to have occurred on all channels during the very first run of the test. As MATLAB had been restarted prior to commencing the test, this was most likely due to delays loading the required files. Also, although no explanation for the single sample variations could be found, this was deemed not to be important due to the other more significant variations in relative channel timings rendering this approach unsuitable for the project.

4.1.2 ASIO Protocol support

Within Windows there are two other commonly used ways to communicate with some soundcards: DirectSound and Steinberg's Audio Stream Input/Output (ASIO) protocol[6]. Neither of these are supported directly by MATLAB although both, in addition to MME, can be accessed though the pa_wavplay utility[7]. When using MME this utility reports the soundcard as multiple stereo pairs including one input with the name 'Multi'. When using DirectSound a very similar response is obtained, although in this instance the 'Multi' input is stated as supporting 12 channels. However, because the utility's `pa_wavplay`, `pa_wavrecord` and `pa_wavplayrecord` (combined play and record) functions are all blocking, only a single stereo output could ever be used at once, and so this option was not investigated further.

In contrast to DirectSound and MME, when using the utility with the ASIO protocol the soundcard is reported as a single device with 12 input channels and 10 output channels. Although the utility's functions are still blocking, this allows access to more than two input and output channels simultaneously within one function call.

The ASIO standard has become popular for use with audio applications due to its low latency and its versatility including "variable bit depths and sample rates, multi-channel operation and synchronization" [35]. It was therefore expected that this approach would not encounter the same problems as with the MME drivers. To confirm this an identical test to that above was constructed, using the `pa_playrec` function to play and record on all 4 channels simultaneously.

[6]ASIO is a trademark and software of Steinberg Media Technologies GmbH.

[7]pa_wavplay utility written by Matt Frear and available online at http://www.mathworks.com/matlabcentral/fileexchange/loadFile.do?objectId=4017

		Loop back connection (output⇒input)			
	Number of occurrences	**3⇒3**	**4⇒5**	**5⇒4**	**6⇒6**
Recording delay (samples)	**-584**	1	1	1	1
	-579	746	746	746	746
	-578	253	253	253	253
	Total number of tests	*1000*	*1000*	*1000*	*1000*

Table 4.3: Variation in loop back delays using the ASIO protocol via the pa_wavplay utility in MATLAB.

A summary of the delays measured on each channel is given in table 4.3, showing a much smaller range of delay values than when using the MME drivers. By comparing the results within each test it was found that the measured delay on each of the four channels was always identical. Additionally it was found that all delays of -579 samples occurred initially before changing to a delay of -578 samples. To try and determine the causes of the variations in delays the test run was repeated, producing the results shown in table 4.4. On this occasion a single spurious delay occurred at a different point through the test compared to the first run. However, as with the first run, all channels were affected identically. The suspected cause of this variation was the running of other background tasks within Windows. As this is very difficult to determine exactly it was decided not to investigate it any further. Importantly, however, it was decided that these tests demonstrated satisfactorily that all input channels remain synchronised, and therefore a single reference can be used even with more than one input signal.

4.2 Alternatives to MATLAB on Windows

Although the pa_wavplay utility provided the synchronisation required, it was still not ideal due to the time taken to initialise the ASIO drivers at the start

		Loop back connection (output⇒input)			
	Number of occurrences	**3⇒3**	**4⇒5**	**5⇒4**	**6⇒6**
Recording delay (samples)	**-583**	1	1	1	1
	-578	999	999	999	999
	Total number of tests	*1000*	*1000*	*1000*	*1000*

Table 4.4: Variation in loop back delays using the ASIO protocol via the pa_wavplay utility in MATLAB. Second test run.

of every recording. Therefore alternative approaches were also considered. The first of these was to use Linux instead of Windows. An AGNULA Live CD[8] was used to determine if the soundcard was supported, and if so, how it was represented. Using ALSA (Advanced Linux Sound Architecture) the soundcard was described as a single 12 input, 10 output channel device, identical to that reported by ASIO under Windows. To verify that all channels could be accessed, Pure Data (PD)[9], a "real-time graphical programming environment for audio, video, and graphical processing" was used[27]. Although this could not easily confirm the relative timings of the channels, it did confirm that all channels could be accessed. As this was done using a Linux Boot CD it was not possible to verify operation within MATLAB, as such an application would require Linux to be installed onto the computer, something that could potentially be very time consuming to no avail. Additionally, from the research into using MATLAB on Windows, it was suspected that audio support would still be limited to 2 channels on a device, even if the device drivers supported more.

From further investigation into PD, which could have been used under either Windows or Linux, it was decided that, although suitable for implementing an Ambisonics decoder, it would not be suitable for implementing and comparing different

[8]AGNULA/DeMuDi LIVE CD V1.1.1 - a Debian-based GNU/Linux distribution for audio/video. Available online at `http://www.agnula.org/`

[9]Available online at `http://puredata.info/`

approaches to determining loudspeaker location. This would be best implemented by recording and then post-processing the microphone signals. Therefore, for similar reasons in addition to the limitation in audio handling capabilities[10], Simulink[11] was also deemed to be unsuitable.

As the ASIO drivers provided the required timing accuracies, other applications that utilised these drivers were also considered. These included, very generically, audio multitrack applications where one or more channels would be used to output a pre-created test signal whilst other channels would record the microphone and loop back signals. These signals would then have to be saved and processed in another application, such as MATLAB, making each test cycle relatively laborious even if some parts of the process could be automated. Additionally, this approach would not allow for easy complete system implementation. An alternative to this was to use an application such as AudioMulch[12] as a Virtual Studio Technology (VST) host and then use custom written VST plug-ins to enable automation of more of the above process, such as playing and recording files. However this would still not be appropriate for a complete system implementation, unless the whole system was implemented in the VST plug-in. It would also require very specific plug-ins to be implemented initially that could take a long time to develop and yet be of no use beyond the end of the project.

Therefore, although other applications could have been suitable, it was decided that, due to the ease of use and processing power offered, MATLAB should be used.

[10]Schillebeeckx *et al* found that despite the MME drivers for their soundcard, a Soundscape Mixtreme, supporting more than two channels, Simulink had the same two channel limit as in MATLAB[32, 43].

[11]Produced by The MathWorks, Inc. See `http://www.mathworks.com/` for more information.

[12]See `http://www.audiomulch.com/` for more information.

4.3 Using PortAudio within MATLAB

Following the decision to use MATLAB, the internal operation of the pa_wavplay utility was investigated[13]. This was primarily because, like other applications including AudioMulch, PD and Audacity[14], the utility was based on PortAudio, a "free, cross platform, open-source, audio I/O library" [25], and so could potentially be used as the basis for the underlying complete system by the end of the project.

When the project started there were two versions of PortAudio available, each using a different Application Program Interface (API)—V18 was stable and had been for some time whilst V19, including many enhancements to the API and changes in behaviour, was under development. Of these, V18 was initially used because it offered all the basic functionality that was required, was fully implemented and was likely to contain fewer bugs than V19.

It was found that it is relatively easy to write code in C that can be compiled into a dynamic link library, know as a MEX-file, and then accessed from within MATLAB in the same manner as accessing an m-file (see Appendix A for more information on MEX-files). Also the simplicity of audio handling using PortAudio was discovered (see Appendix B for more information on PortAudio). As a result, it was decided that it would make more sense to fully investigate and develop the audio handling required for the whole project at this point, including continuous synchronous input and output. By doing so problems would not be encountered further into the project when trying to change between the old and new approaches. Additionally, by considering the requirements for the utility and then developing it separately from the rest of the project, the resulting utility could be structured sensibly, avoiding many of the pitfalls found in applications developed over time by adding functionality as and when required. Thus the utility would be useful not only for the rest of this project but also for extensions to the project or for

[13]pa_wavplay utility source code available online at `http://sourceforge.net/projects/pa-wavplay/`

[14]See `http://audacity.sourceforge.net/` for more information.

any other projects with some or all of the same audio handling requirements.

The original timetable had not allowed for developing the utility in this way but instead expected all audio handling routines to be developed as required. However the benefits for both this and future projects, as outlined above, were deemed to be significant enough to adopt this approach despite its potential to reduce the amount of progress that could be made in other parts of the project.

To achieve continuous signal decoding it was clear that the pa_wavplay utility would need to be modified to provide both non-blocking performance and continuous audio output. For this to function correctly, the PortAudio callback would have to work regardless of what else MATLAB was doing and so a basic conceptual test was created using the pa_wavplay utility. This implemented audio output through a non-blocking function and so allowed MATLAB to run other code whilst audio was playing. Various processor and memory intensive operations, such as `primes` and `fft`, were used during test runs but no glitches in the audio, a pure 1 kHz sine wave, could be heard.

This showed that non-blocking audio routines were feasible, and so the exact requirements for a new utility to meet the demands of the project were defined, as in chapter 5. The implementation was then divided into two parts: core MEX-file code that could be used when implementing other MEX-files (see chapter 6), and MEX-file code specific for the utility (see chapter 7). All implementation was conducted using C because, although either C or C++ could have been used, C was the preferred programming language and there were not seen to be any significant advantages in using C++, especially when considering the time required to re-learn the language.

Chapter 5

Audio Handling Utility Requirements

To achieve all the aims of this project a new MATLAB audio handling utility, to be called `playrec`, was required. The main audio specific requirements were as follows, where input (record) and output (play) are always relative to the signal direction at the audio interface on the soundcard:

- To allow simultaneous audio input and output on all channels of a multi-channel soundcard, accessed using ASIO drivers.
- That, within the limits of the hardware, all input channels should remain synchronised with each other and all output channels should remain synchronised with each other. This is to allow a single loop back channel to be used to determine the relative timings of any input and output channels.
- That the ability to record only, play only, or combine play and record simultaneously should be possible. These 'requests' should be queued in order of arrival using non-blocking functions so that they can occur immediately one after the other. There is no requirement that more than one 'request' occurs simultaneously. This is to allow continuous input and/or output to

be spread across multiple ‘requests’ without any glitches in between. For example, future output samples can be calculated whilst previously calculated samples are played.

- That the lengths, in samples, of the input and output in a combined input/output ‘request’ need not necessarily be the same. So for example, the recording of a microphone signal may continue after the end of an output signal without the need to either zero-pad the output signal or use a second record ‘request’ to complete the recording.

- That, within the limits of the hardware, during a single use of the utility whenever the combined input/output ‘request’ is used the relative timings of the inputs and outputs should remain the same. This is to ensure that if a long succession of combined input/output ‘requests’ are added to the queue, the relative times of the inputs and outputs do not change between the first and last ‘request’. Additionally, it avoids any variations in relative times due to using ‘requests’ with different length inputs and outputs.

- That the input and output channels to be used can be specified with each ‘request’ and not all channels always have to be used. This avoids having to transfer large amounts of data which is not required between MATLAB and the utility, such as multiple channels of output samples just containing zeros.

- That the samples for all output channels are supplied to the utility in a single matrix and similarly all recorded samples for a particular ‘request’ are returned in a single matrix. This allows the number of channels used to be easily changed. Additionally in both cases each column should represent a different audio channel. This makes the utility consistent with other MATLAB functions.

- That all samples have a range of -1 to +1, supplied to the utility as either single or double precision values and returned from the utility in a suitable data type to ensure no value rounding occurs. This allows the full quantization of the soundcard to be supported.

- That the number of ‘requests’ that can be in the queue at any one time is only limited by system resources and does not need to be specified when the utility is first used. This improves ease of use of the utility by minimising the amount of initial configuration required, and also increases the flexibility of the utility, allowing it to be used for different tasks without reconfiguration.
- That all recorded input samples be accessible on demand, through reference to a particular ‘request’, and be stored by the utility until it is told to delete them. This reduces the complexity of configuring the utility by removing the need to specify the lifetime of recorded samples whilst at the same time increasing the amount of control offered to code utilising the utility.
- That the utility also provides helper functions to:
  - return a list of all available audio devices within the system;
  - determine if a particular ‘request’ has completed, including the option to wait until it has;
  - obtain a list of all queued requests and identify the point in the queue that has been reached;
  - delete any ‘request’ no matter where it is in the queue;
  - pause and resume input and output;
  - return and reset the duration, in samples, of any gaps that have occurred between ‘requests’ due to the end of the queue being reached. This is so it can determined from within MATLAB if ‘requests’ have been supplied frequently enough for gap-free audio.

In addition to these audio handling requirements, there were some more generic requirements of the utility:

- That help information should be available for the utility as a whole as well as specifically for each function the utility implements. This aids use of the utility by providing usage instructions on demand.
- That a complete list of all functions implemented by the utility should be easily accessible. This avoids time consuming searching for the name of the required function within the utility.
- That the amount of configuration required before the utility can be used is kept to a minimum. This avoids potential configuration problems and in doing so, makes the utility easier to use.
- That the number and type of parameters used with each function are validated, with warnings or errors generated as appropriate. This removes the possibility of erroneous behaviour occurring within the utility due to incorrect use.
- That the current state of the utility can be queried from within MATLAB, including whether it is initialised and if so what values were used during the initialisation. Additionally, errors should be generated when any function is used if the utility is not in the correct state. This ensures that code in MATLAB will not continue if the utility is in a different state from that expected, whilst at the same time code can be written conditionally based on the state of the utility.
- That it is possible to determine when the utility's configuration was last changed. When using the utility to run multiple tests, storing this with each test makes it possible to determine which tests were conducted without the utility being changed, such as if MATLAB has to be restarted.

- That the utility itself is designed to be easily maintainable and adaptable to specific uses, reducing the development time for future MEX-files with at least some of the same requirements.

Some of these requirements were not initially envisaged as being required by this project, such as the record only ‘request’, and some could have been achieved using alternative approaches, such as zero padding the output signal(s) so output and input for a combined ‘request’ were always the same length. However, their inclusion made the utility much more flexible both for this project and for other projects in the future.

Chapter 6

Design of MEX-file Core

Although the `playrec` utility could be designed just to meet the requirements as given in chapter 5, other approaches were likely to produce more flexible code that could also be useful in the future. For this reason the MEX-file implementation was divided into two parts: a core section, containing generic helper functions, and a section specific to the operation of the utility. Only a small amount of extra work would be required during the utility's development, yet in the future the core code could be used without any need for modification.

For a basic overview of MEX-files see appendix A.

6.1 Providing multiple functions

When considering the requirements for the utility it was obvious that multiple functions, hereafter called commands, would have to be implemented by a single MEX-file. However due to the way in which MEX-files work only the entry function can be called from within MATLAB. So instead a way to pass all command calls through this one function was required, and the chosen method was to use the first parameter to indicate the required command. This could have been implemented using a number, but a string was chosen because it would be much more user friendly, resulting in MATLAB function calls such as

```
mex_file( 'command', cmdParam1, cmdParam2, ... );
```

where `mex_file` is the name of the MEX-file, `command` is the command name and `cmdParam1, cmdParam2, ...` are all the parameters required by `command`.

If required this approach could also allow wrapper m-files to be created such as

```
function [ varargout ] = command( varargin )
%command Summary of command
%   Detailed explanation of command
    [varargout{1:nargout}] = mex_file( 'command', varargin{:});
end
```

This could then be used as

```
command( cmdParam1, cmdParam2, ... );
```

to call the function whilst

```
help command
```

would return specific help on the function. However, this would distribute command specific information between m-files and the MEX-file, making it cumbersome to make changes in the future, especially if a large number of commands are used. Additionally, this hides from the user the fact that different commands are all implemented by the same MEX-file, which could become confusing where the commands have an effect on each other. Therefore, it was accepted that all command calls would be made directly to the MEX-file and a way to include the help information within the MEX-file was investigated. The solution chosen was to include a help command within the MEX-file such that

```
mex_file( 'help', 'command' );
```

would display help information on `command`. Additionally the utility requirements specified that, to aid use of the MEX-file, a complete list of all commands should be easily accessible. Although this could have been implemented through its own command, a more intuitive approach was to provide this information whenever no command was specified.

6.2 Command Dispatching

To implement the command based operation the entry point function would have to interpret each command and run the appropriate code. Three different approaches were initially considered:

1. Place all the code for each command within the one function.

   Advantages

   - Easy to change the number and type of parameters used by each command.

   Disadvantages

   - Potential for a very large function which would be difficult to maintain.
   - Poor programming practice.

2. Use separate functions for each command, with each function's parameter list the same as that used by the command. The entry point function would then provide a mapping between the parameters it receives and those required by each command function.

   Advantages

   - Each command function would receive the relevant command's parameters as one per variable, thus making the operation of the function clearer.
   - Organises the code into a sensible structure.
   - Easy to maintain the command functions.
   - Easy for command functions to call each other within the utility.

   Disadvantages

   - The entry point function would be difficult to maintain because of the mapping between the parameters received from MATLAB and those required by the command function.

- The parameter mapping within the entry point function would need changing whenever a command's parameter list was changed.
- Implementation of optional command parameters is difficult without including the default values in the entry point function.

3. Use separate functions for each command, with all command functions having the same parameter list as the entry point function. The entry point function would then remove the command string from the parameters it receives before passing them on to the command function.

   Advantages

   - Significantly reduces the complexity of the entry point function.
   - Allows the parameter list for a command to be changed just by changing the command's function.
   - Appears to each command function as though MATLAB is directly calling the function rather than the entry point function.
   - Organises the code into a sensible structure.
   - Easy to maintain the entry point function.

   Disadvantages

   - Each command function will receive the command's parameters in a single array with no indication what each parameter represents. Therefore, care would have to be taken to make the operation of the function clear, although this would be no different from writing code directly in the entry point function.
   - Difficult for command functions to call each other within the utility.

It was considered that the last approach would be most easily maintainable, especially if modifications needed to be made by other users in the future. Additionally, because all command functions have the same parameter list a lookup table could be used between command string and function. This would avoid having to use a long `if() ... elseif() ...` construct in the entry point function, replacing it with a much more compact `while()` loop iterating through all command strings in the table. Within such a table far more information could also be included about each function, such as the help text. This way, new commands could be added just by adding a single entry to the table as well as writing the command function, a much simpler approach than having to modify the entry point function and the help command function separately.

As this approach appeared very promising, the range of additional information that could be stored for each command was considered, arriving at the following list:

name The name of the command.

> If this matches the command string supplied from within MATLAB, then this is the table entry to be used. Otherwise, the command string is compared to the next `name` entry in the table until there are no entries left. The string comparison can be selected as case sensitive/insensitive at compile time.

func Pointer to the function to be used for this command.

> This is a pointer to the function called by the entry point function if the command string matches `name`. The function is supplied all parameters in the same format as those received by the entry point function, although the command string is removed.

min_nlhs The minimum value of `nlhs` that this command requires.

> If `nlhs`, the number of return parameters expected by MATLAB, is less than this value, an error is generated within the entry point function and `func` is not called. Uses -1 to indicate there is no minimum.

max_nlhs The maximum value of `nlhs` that this command requires.

> If `nlhs` is more than this value, an error is generated within the entry point function and `func` is not called. Uses -1 to indicate there is no maximum.

min_nrhs The minimum value of `nrhs` that this command requires.

If `nrhs` *after* removing the command string from the parameter list (i.e. the number of parameters supplied to the command) is less than this value, an error is generated within the entry point function and `func` is not called. Uses -1 to indicate there is no minimum.

max_nrhs The maximum value of `nrhs` that this command requires.

If `nrhs` *after* removing the command string from the parameter list is more than this value, an error is generated within the entry point function and `func` is not called. Uses -1 to indicate there is no maximum.

desc A short, one line description of the command.

This is the text displayed when listing a summary of all the commands supported by the MEX-file. No line wrapping is implemented on this string.

help A full description of the command excluding a parameter list or specific per-parameter descriptions.

This is the start of the text returned by the `help` command and should explain the operation of the command. To speed up the writing of such text it is automatically formatted when displayed, avoiding the need to manually insert line breaks. However, if required manual line breaks can be included such as when starting new paragraphs.

paramrhs A list of all the parameters that can be supplied to the command.

For each parameter this contains the parameter name, a description and a flag to indicate if the parameter is optional. This is used to display information on each command parameter at the end of the text returned by the `help` command.

paramlhs A list of all the parameters that can be returned by the command.

For each parameter this contains the parameter name, a description and a flag to indicate if the parameter is optional. This is used to display information on each return parameter at the end of the text returned by the `help` command.

The variables min_nlhs, max_nlhs, min_nrhs and max_nrhs were included because, as stated in the utility requirements, each of the command functions would have to implement these types of checks. Therefore, by including the values in the table it would mean the entry point function could implement the checks and so avoid code duplication. Although such an implementation would mean the table would have to be updated as well as the command function if the parameter list changed, this was not seen to be a major issue because a) the help information in the table would also have to be updated, b) after a small amount of testing any omission to change the table would be observed and c) if the feature was not required it could be easily disabled using the value -1.

The full description returned by the help command was initially going to be a single string containing manual line breaks at the end of each line. However, editing text formatted in such a way can be very tedious due to the need to move line breaks. So, instead, an automatic line wrapping function, written prior to the start of the project, was utilised to remove the need for manual line breaks. This then prompted the inclusion of separate parameter lists so their layout could also be automated, something which would have taken much longer to do manually than the time required to write the automation. Consequently the three variables help, paramrhs and paramlhs were included in the table. Although this would not necessarily be suitable in very specific circumstances, such as for a single command that can take a number of completely different parameter lists, this could be easily resolved by including all the information in the help string, leaving the parameter lists empty.

6.3 Implementation

The most intuitive way to implement this command lookup table was to use an array of structures, and so two structures were defined as shown in listing 6.1.

```
typedef struct {
    char *name;      // the parameter name
    char *desc;      // description of the parameter - can be multilined
                     // and will be line wrapped if longer than one line
    bool isOptional;// true if the parameter is optional
} paramDescStruct_t;

typedef struct {
    char *name;      // Textual string used to identify function in MATLAB
    bool (*func)(int nlhs, mxArray *plhs[], int nrhs, const mxArray *prhs[]);
                     // Pointer to the function called if *name is specified
                     // The parameters are those received by the entry point
                     // function, apart from the function name is NOT supplied
                     // and so prhs starts with the second parameter and
                     // nrhs is one less. ie the parameters are as if the
                     // function was called directly from within MATLAB
    int  min_nlhs, max_nlhs, min_nrhs, max_nrhs;
                     // The minimum and maximum values of nlhs and nrhs that
                     // *func should be called with. Use -1 to not check the
                     // particular value.  This can be used to reduce the
                     // amount of input/output count checks in the function.
    char *desc;      // Short (1 line) function description - not line wrapped
    char *help;      // Complete help for the function.  Can be any length and
                     // can contain new line characters.  Will be line wrapped
                     // to fit the width of the screen as defined as
                     // SCREEN_CHAR_WIDTH with tab stops every
                     // SCREEN_TAB_STOP characters

        // descriptions of all rhs parameters in the order they are required
    paramDescStruct_t paramrhs[MAX_PARAM_COUNT];
        // descriptions of all lhs values in the order they are returned
    paramDescStruct_t paramlhs[MAX_PARAM_COUNT];
} funcLookupStruct_t;
```

Listing 6.1: Type definitions for `paramDescStruct_t`, a structure used to store the name and description of a single command parameter, and `funcLookupStruct_t`, a structure used to store all lookup information on a specific command.

Using these, a constant lookup table could then be created as shown in listing 6.2, which in this case just includes the `help` command although could include any number of commands. Additionally, by using the `sizeof` operator to determine the number of commands specified, the need to manually update a variable containing the number of commands is removed.

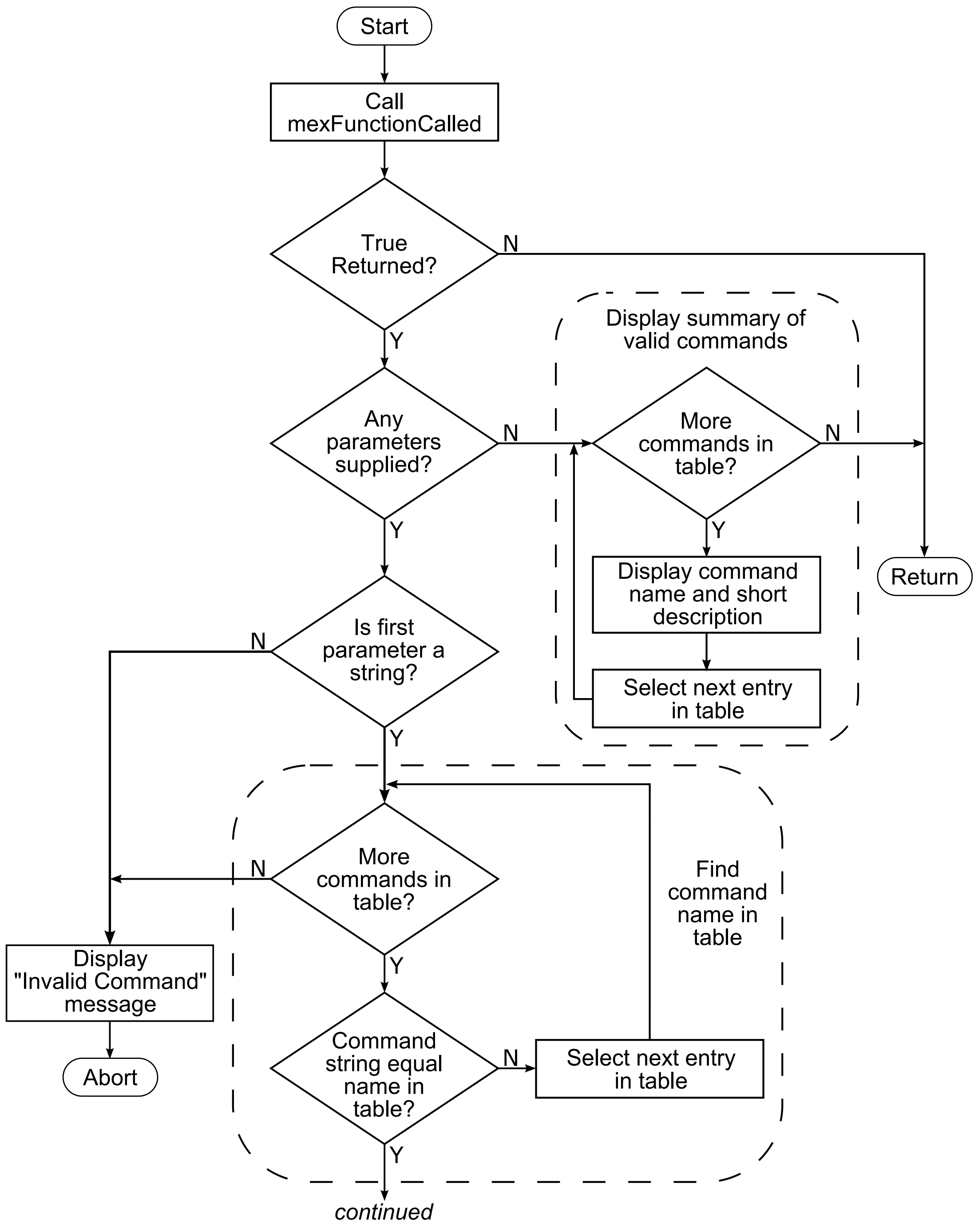
```
const funcLookupStruct_t _funcLookup[] = {
    {"help",
    showHelp,
    0, 0, 1, 1,
    "Provides usage information for each function",
    "Displays command specific usage instructions.",
    {
        {"commandName", "name of the command for which information is required"}
    },
    {
        {NULL}
    },
}

const int _funcLookupSize = sizeof(_funcLookup)/sizeof(funcLookupStruct_t);
```

Listing 6.2: Sample definition of `_funcLookup` and `_funcLookupSize`

Finally, it was realised that in some scenarios it is necessary to run the same piece of code no matter which command is specified, or even whether the specified command is valid. Such code could be placed within the entry point function, but instead a new function called `mexFunctionCalled` was added such that it is always called by the entry point function with all the same parameters as the entry point function. By doing so all the code within the entry point function could be generic and so suitable for multiple different MEX-files.

The resulting top-level flow diagram for the entry point function is shown in figure 6.1. Unlike the entry point function itself, both `mexFunctionCalled` and all the command functions were given a `bool` return type. For `mexFunctionCalled` this was to allow indication that, for whatever reason, the entry point function should not continue processing, whilst for the command functions it was to indicate if the supplied list of parameters was invalid, and thus if an error message should be generated.



Start
Call mexFunctionCalled
True Returned?
N
Y
Display summary of valid commands
Any parameters supplied?
N
More commands in table?
N
Y
Display command name and short description
Select next entry in table
Return
Y
Is first parameter a string?
N
Y
More commands in table?
N
Find command name in table
Display "Invalid Command" message
Abort
Y
Command string equal name in table?
N
Select next entry in table
Y
continued

Figure 6.1: MEX-file entry point function flow diagram.

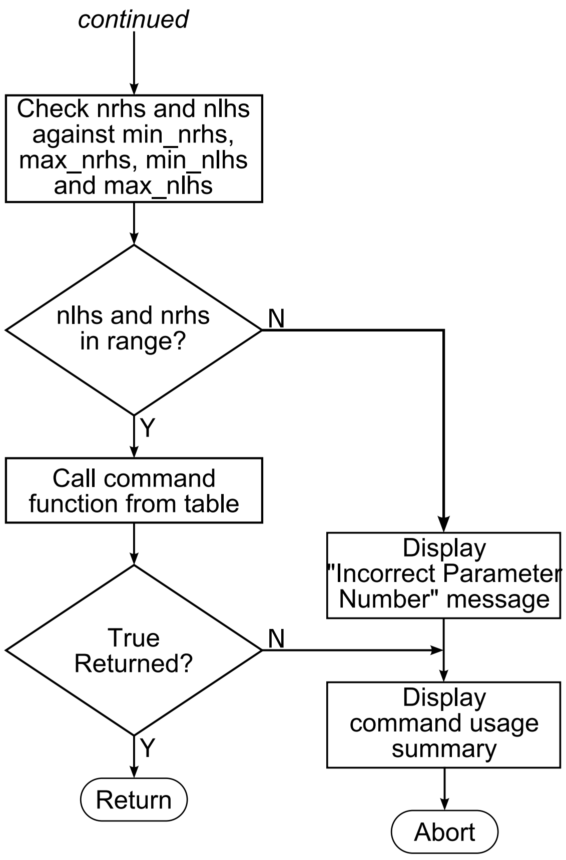

continued
Check nrhs and nlhs against min_nrhs, max_nrhs, min_nlhs and max_nlhs
nlhs and nrhs in range?
N
Y
Call command function from table
Display "Incorrect Parameter Number" message
True Returned?
N
Y
Display command usage summary
Return
Abort

Figure 6.1: MEX-file entry point function flow diagram. (continued)

The resulting source code for this function, in addition to the showHelp and linewrapString functions, can be found in the files `mex_dll_core.c` and `mex_dll_core.h` on the accompanying CD, as described in appendix G. The showHelp function is used to display help information on each command, provided it is included in the _funcLookup array, whilst the linewrapString function can be used by any function that needs to format a string and display it in the MATLAB command window.

The showHelp function's overall structure is very similar to that of the entry point function—initially the name of the function for which help is required is confirmed to be a valid string, following which the command name is found within the lookup table and, if the command is not found, an appropriate error is generated. If the command name is found, the help for the command is displayed in three stages. Initially the command definition is displayed listing the names of all input and output parameters, generated from the command name and the parameter names stored in paramrhs and paramlhs. Following this, using linewrapString the help text within the lookup table is displayed before lastly displaying all the per-parameter information stored in paramrhs and paramlhs. By displaying the help like this, it enables those who have just forgotten the order of parameters to very quickly see the required list, whilst those who require more information can read on further.

6.4 Testing

Before this core code could be used as part of the final utility it needed to be tested. Although all parts were tested, it was the command dispatching code that was most thoroughly tested because, unlike the rest of the code, any bug here would affect the whole operation of the utility rather than just the display of textual information.

Tests on the command dispatching included confirming that:

- the correct command function is called for the command name supplied, both when using case-sensitive and case-insensitive string comparison.
- all parameters supplied in MATLAB are passed to `mexFunctionCalled` and only the command parameters are passed to the required command function.
- various numbers of parameters can be returned, and all returned parameters are correctly received in MATLAB.
- the minimum and maximum parameter number limits correctly stop the command function from being called, and additionally that using a value of -1 disables each limit.
- the correct behaviour is observed for both `true` and `false` return values from both `mexFunctionCalled` and command functions.
- any incorrectly formatted entries in the _funcLookup array, such as empty or NULL strings and NULL function pointers, are handled gracefully.

Additionally, checks on the rest of the code included confirming that:

- the list of available commands correctly displays all commands.
- command help text is displayed correctly, even when including unexpected but potential scenarios such as single words longer than the length of one line. Also, correct tab alignment and line wrapping of lines including tabs was checked.
- the list and descriptions of command input and output parameters displays correctly, including scenarios where there are no parameters.

After fixing minor bugs these functions were accepted as being suitable for use in the rest of the utility.

Chapter 7

Design of playrec MEX-file

Following the completion of the core MEX-file code, the specific implementation of the utility was investigated. From previous tests, as described in chapter 4, it had been decided that PortAudio should be used to provide the utility's underlying soundcard interface. Although this offered the synchronisation level required, the utility still had to be designed carefully to ensure the channel timings did not drift internally.

7.1 Request Pages

From the requirements specified, the utility could be pictured as a buffer between MATLAB and the soundcard, or PortAudio in this case. Two common techniques used when creating routines such as this to buffer audio data are to either use a circular buffer or to double buffer the data. In both cases it is fairly easy to maintain synchronisation between channels just by ensuring the same amount of data is always added to all channels and similarly ensuring the same amount of data is always removed from all channels. However, because of the initial requirement not to have any predefined lifetime for recorded samples, such an approach would not be suitable on its own because the samples would get overwritten unless the buffers could grow indefinitely. Also, all unused channel buffers would have to be zero filled, to keep them synchronised with the channels being used, and a method to 'mark' where each record 'request' commenced would be required so that the relevant recorded samples could be returned. Although it may have been possible to use one of these techniques for output samples whilst using a different approach for recorded samples, an alternative approach that grouped all 'request' information together was considered to be more reliable as well as easier to maintain and debug should problems occur.

The resulting concept contained blocks, called pages, where each page represented a single 'request' to the utility, no matter whether it was to record only, play only, or to play and record simultaneously. Other key parts to this concept were:

- All channels used within the page start at the beginning of the page and the length of each page is the same as the length of the longest channel of data it contains.
- All pages are dynamically generated in memory when they are added. Although this requires much more dynamic memory allocation than a circular buffer or double buffer concept, it removes restrictions on the lifetime of recorded samples and the number of pages that can be in the queue at any one time.
- The minimum amount of space required by each page is used. All output channels without samples are set to zero during the PortAudio callback, avoiding the need to store and manipulate buffers just containing zeros. This is also true for any output channels shorter than the length of the page. Similarly, only the required input samples are stored.
- Pages are queued up end-to-end in the order they are added and are processed with no gap between them.
- When there are no more pages in the queue, the utility waits for the next page to be added, sending zeros to all output channels as required.

- When a page has finished, defined by the end of the page having been reached, then all output samples are automatically removed from memory leaving only the recorded samples and page-specific information. If the page contains no recorded samples then the whole page is automatically removed from memory. This is to reduce the amount of memory in use to a minimum and to remove the need to manually remove any pages just used for sample output.

- All pages are given a unique number to identify the page in all subsequent uses of the utility, such as determining if the page has finished or requesting the associated recorded data to be returned.

A diagram of three such pages, assuming the soundcard is configured to use 4 input and 4 output channels, is shown in figure 7.1. This shows when output channels are automatically zero padded as well as how continuous audio input or output can be achieved provided the page is the same length as the channel buffer and each page is added prior to the previous page finishing.

7.2 Thread safe design

Due to the PortAudio callback occurring in its own thread, the sharing of data between the callback function and all other functions within the utility had to be carefully designed. In this case, there were three main problem areas that had to be considered: working with shared variables, how to add pages reliably, and when to free memory.

From the MATLAB documentation[37] it was found that, even when using a Graphical User Interface (GUI), all code is effectively executed in the same thread. In the case when GUI events occur whilst another is being processed, the events are either dropped or queued until a time when they can be processed. This occurs when the previous event callback either finishes or calls one of a particular set of

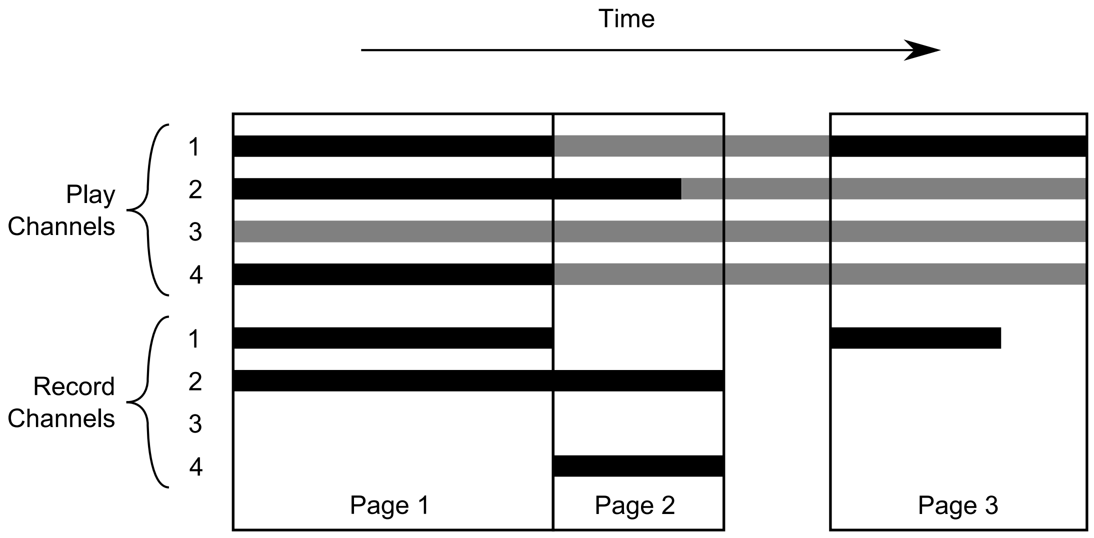


Figure 7.1: Example arrangement of channel buffers within playrec pages for a soundcard configured to use 4 input and 4 output channels. Black bars show requested audio to be either played or recorded whilst grey bars show times when output channels are automatically zero padded. Page 2 was added prior to Page 1 finishing whereas Page 3 was only added after Page 2 finished.

functions including drawnow, pause or waitfor. Thus, apart from the PortAudio callback, no two functions within the MEX-file could be executed simultaneously and therefore they would never interrupt each other. As such, the MEX-file could be considered as using two threads—one for the PortAudio callback and one for all other functions, called the *main* thread.

7.2.1 Shared Variables

When a variable is updated using a line of code such as tmpVar += 3, it cannot be guaranteed that this will occur in one machine instruction, and so is not necessarily atomic. For example it may take three instructions, read-modify-write, as shown in the following disassembly:

```
mov         eax,dword ptr [_tmpVar (0FED9504h)]
add         eax,3
mov         dword ptr [_tmpVar (0FED9504h)],eax
```

Therefore it is possible that an interrupt could occur during the execution of this line. Any changes to the value of tmpVar whilst this code is interrupted would then initially have an effect, although when the code resumes the value of tmpVar would be set to its *old* value plus 3, thus losing the other updates to the value. Furthermore, if tmpVar is a large value spread across multiple memory locations it is possible that its value could be read whilst it is half updated.

The simplest solution to this problem is to use variable types where the write instruction is atomic, such as where a write requires only one machine instruction, and ensure that the variable is only updated within one thread. With this, as many threads as required can read the variable and they will never read an incorrect value. Additionally, because only one thread writes to the variable no updates will be lost. Due to its simplicity, this was the approach used wherever possible.

Sometimes more than one thread may need to update the same variable and so, to ensure that no updates are lost, semaphores must be used so that only one thread can access the variable at once. This means that each thread must block until no other threads are accessing the variable. However, because of timing restraints on the callback function, having it block would not be ideal and instead all thread safety needed to be implemented without the callback blocking. One exception to this requirement for semaphores is when a variable is *only* ever written rather undergoing a read-modify-write cycle. In this case, provided only the latest value to be written needs to be known, no problems will occur.

A further problem with sharing variables between threads is the uncertainty as to when they might change due to the execution of code in an alternative thread. This is most noticeable if the value of a variable is relied on as being constant between the start and end of a function. In such scenarios, provided the variable never needs to be updated, making a local copy for the duration of the function avoids any problems.

When using multiple threads it is always imperative to ensure that no problems due to shared variables will occur. This is because the independent timing of threads means virtually no amount of testing can absolutely guarantee that there are no problems. Furthermore, if any problems are found, they are often very difficult to debug. Therefore, extra care must be taken to avoid this during the initial design and implementation.

7.2.2 Adding Pages Reliably

To store all the pages in a queue, a linked list was to be used because of the ease with which pages could be added and removed. To avoid potential problems due to the use of multiple threads, even a basic operation such as this had to be carefully considered. However, by ensuring that each page was completely generated before adding it to the list, the list would never be incomplete and so there would not be any possibility of a page being used by the callback before it was ready. (The addition of an element to a linked list was confirmed to alter the list using a single machine instruction, and therefore never place the list in an intermediate 'corrupt' state.)

7.2.3 Freeing Shared Memory

Removing pages also had to be carefully considered, especially because in some scenarios parts of the page would need to be freed without completely removing the page.

When completely removing pages the code implementing the removal must be certain that the other thread is not currently using the page and will not try to use it in the future. Due to timing restrictions on the callback, this would have to be implemented within the main thread. An initial approach might be to have a variable that the callback sets and clears to indicate if it is currently running. Then by using code similar to

```
while(inCallback) {
    wait();
}

removePage();
```

the page would only be removed when not in the callback. However, this is not necessarily the case. Take, for example, the case when this code starts executing whilst the callback is not running. Then just before the line `removePage()` begins executing this thread is interrupted, and the callback starts executing and gets to the point where it is using the page. However, before the callback finishes it is interrupted and this code resumes and so removes the page from memory. When the callback then resumes it will potentially try and access the page that no longer exists. Although this sequence of events might be unlikely, without thorough knowledge of how thread scheduling occurs within the operating system it cannot be ruled out.

For this reason the page removal was divided into two parts—the first would leave the page in memory but remove it from the linked list whilst the second would actually remove it from memory. Provided the callback always starts from the beginning of the linked list each time, code similar to

```
removePageFromLinkedList();

while(inCallback) {
    wait();
}

removePageFromMemory();
```

guarantees that the callback will not use the page once it has been freed from memory. This is because the page will only be removed from memory after a period when the callback is not active, and therefore the next time the callback occurs it will not have any 'memory' of the removed page.

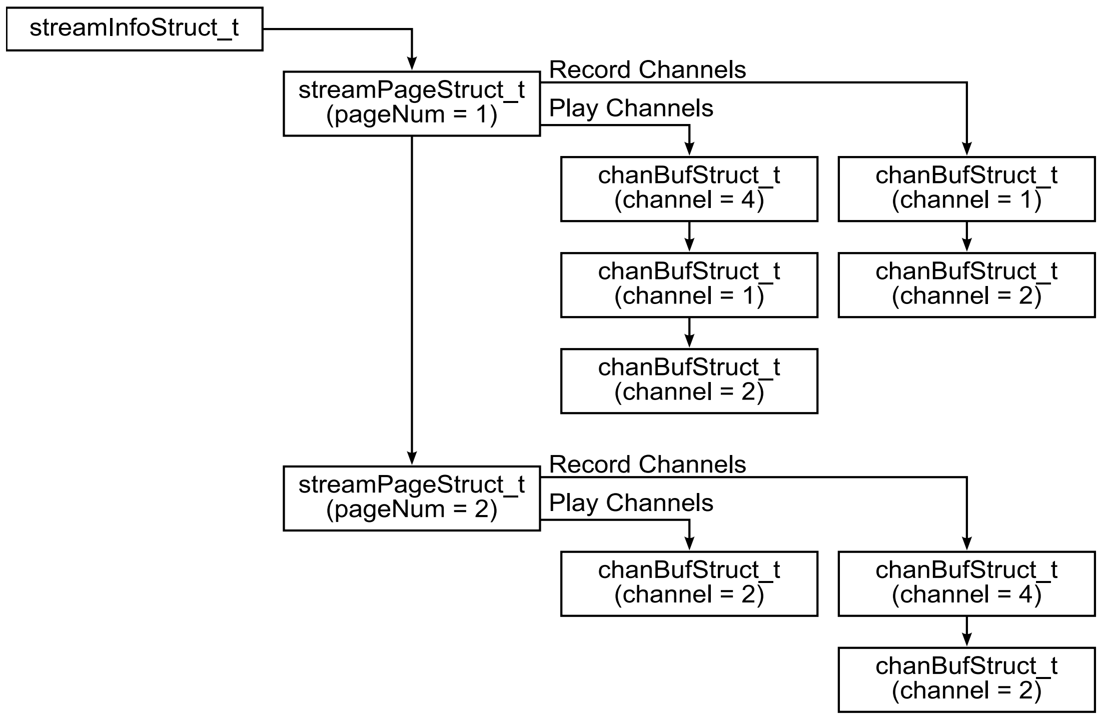
This approach introduces two timing overheads. The first is the time taken for the callback to iterate from the start of the linked list each time it is called. This could be resolved by the callback storing an additional pointer to the last active page so that each time the callback is called, it starts from this point in the linked list. Although this would reduce the execution time of the callback, it would also increase the complexity of removing pages, especially if the page being removed was the page currently being used. Therefore, because the reduction in execution time would only be noticeable if the linked list was very long, this timing overhead was deemed acceptable compared to the increased complexity, and therefore potential for problems, of the alternative approach. The second timing overhead is the delay in the page removal code waiting for the callback to finish. In code that is trying to continuously process data in the shortest time possible, having to wait for other code to complete before continuing can be very costly. This is because the code is suspended not only for the time until the other code finishes but additionally for the time until the scheduler next runs the code. One solution would be to offer page removal in two functions: one to remove it from the main linked list and add it to a 'to be deleted' list, and a second to remove it from this 'to be deleted' list. Provided these functions were called far enough apart to ensure the callback was not still using the pages to be deleted, this would avoid the need for any page removal code to wait. However, this would increase the complexity of code both within the utility and within MATLAB and therefore, because the slight timing advantage was not deemed necessary, the simpler approach was used.

Ideally as much memory as possible should be freed as soon as it is no longer required. This meant that play only pages could be deleted as soon as they had finished. However, because page removal had to occur within the main thread it would be very difficult to trigger this from within the callback and so an alternative was required. One method would be to require all pages to be explicitly deleted from within MATLAB. This would increase the time between a page finishing and it being removed and would also not be intuitively obvious—sending samples to be output from a soundcard, and then having to delete these samples *after* the output has completed. Instead of this, the chosen method was to automatically free up as much memory as possible each time the utility was used. This would slightly increase the execution time of all utility calls but at the same time memory usage would be kept to a minimum and, more importantly, the use of the utility would be simplified. For all play only pages, the page would be completely removed from memory (using the method above) whereas pages containing recorded samples would have as much memory freed as possible whilst still leaving the page and associated recorded samples in memory. To achieve this, a flag would be set by the callback indicating the page was finished. Then, once this is set, the callback must only use the page to access the next page in the linked list whilst the main thread can alter the page as required, such as freeing up parts of memory, without any concerns for thread safety. (For this to work correctly, the value of the finished flag must never be altered in the main thread and also the pointer to the next item in the linked list must be treated the same regardless of the state of the finished flag.)

In addition to automatic page removal, to meet the utility requirements MATLAB would need to explicitly remove pages. By checking that memory is not freed twice, this could be implemented using the same command, based on the method above, regardless of what play or record channels the page contained and whether or not the page was finished.

7.3 Page Structures

To implement the page queueing using a linked list, pages were represented using structures. The final design contains the three structures streamInfoStruct_t, streamPageStruct_t and chanBufStruct_t arranged as shown in figure 7.2.



streamInfoStruct_t
streamPageStruct_t (pageNum = 1)
Record Channels
Play Channels
chanBufStruct_t (channel = 4)
chanBufStruct_t (channel = 1)
chanBufStruct_t (channel = 2)
chanBufStruct_t (channel = 1)
chanBufStruct_t (channel = 2)
streamPageStruct_t (pageNum = 2)
Record Channels
Play Channels
chanBufStruct_t (channel = 2)
chanBufStruct_t (channel = 4)
chanBufStruct_t (channel = 2)

Figure 7.2: Diagrammatic representation of the arrangement of streamInfoStruct_t, streamPageStruct_t and chanBufStruct_t structures used to contain all data associated with a particular stream. This example is based on the first two pages shown in figure 7.1. Note that any order of channels within the chanBufStruct_t linked list is acceptable and only the channels used need to be included.

By grouping all data within a single top-level structure, streamInfoStruct_t, simultaneous support for multiple streams can be easily implemented should the need arise. This structure contains stream specific information, such as sample rate and the number of channels supported, as well as a pointer to the start of a linked list of the 'page' structures streamPageStruct_t. Each of these contains page specific information, including the page number, and links to the start of up to two further linked lists, one containing the buffers for the input samples whilst the other contains the output sample buffers. These linked lists are constructed using chanBufStruct_t structures with one structure for each input and output channel in the page. Using this approach it is very easy to represent a wide variety of pages, even including different length buffers for each channel within the same page.

The type definitions for these three structures, including descriptions of all the variables, are shown in listing 7.1.

```
typedef struct chanBufStruct_tag {
    SAMPLE *pbuffer;                    // The channel audio data
    unsigned int bufLen;                // Length of the audio buffer
    unsigned int channel;               // Channel number on the audio device
                                        // that this buffer is associated with
                                        // (first channel = 0)
    struct chanBufStruct_tag *pnextChanBuf; // Pointer to the next channel
                                        // buffer in the linked list. The
                                        // order of buffers in the linked list
                                        // is the order of 'channel' data
                                        // received from and returned to MATLAB
} chanBufStruct_t;

typedef struct streamPageStruct_tag {
    chanBufStruct_t *pfirstRecChan;     // First record channel within this page
    chanBufStruct_t *pfirstPlayChan;    // First play channel within the page

    unsigned int pageLength;            // The length of the page (in samples)
    volatile unsigned int pagePos;      // The current position within the page
    unsigned int pageNum;               // A unique id to identify the page

    unsigned int playChanCount; // The number of channels used to communicate
                                // with PortAudio. Must be greater than
                                // the maximum channel number used.
    bool *pplayChansInUse;      // Pointer to array type bool, size playChanCount.
                                // Each element can be:
                                //   true - the channel is in the play linked list
                                //   false - the channel is not in linked list
                                // Setting false means that the channel is set
                                // to all zeros within the callback. Any channels
                                // not included in this list (or set true) must
                                // be included in the play channel linked list.

    volatile bool pageUsed;     // Set true if the page has been used in the
                                // PortAudio callback function
    volatile bool pageFinished; // True if the page has been finished (all record
                                // buffers full and all output buffers 'empty')
                                // Once set, this and pnextStreamPage are the
                                // only variables the PortAudio callback will read
                                // (none are written)
    struct streamPageStruct_tag *pnextStreamPage;
                                // The next page in the linked list
} streamPageStruct_t;
```

```
typedef struct {
    streamPageStruct_t *pfirstStreamPage;   // First page in the linked list

    PortAudioStream *pstream;   // Pointer to the stream, or NULL for no stream

    time_t streamStartTime;     // The start time of the stream, can be used to
                                // determine if a new stream has been started.

    // Configuration settings used when opening the stream - see Pa_OpenStream
    // in portaudio.h for descriptions of these parameters:
    double sampleRate;
    unsigned long framesPerBuffer;
    unsigned long numberOfBuffers;
    PaStreamFlags streamFlags;

    volatile bool stopStream;   // Set true to trigger the callback to stop the
                                // stream.

    volatile bool inCallback;   // Set true whilst in the callback.

    volatile unsigned long skippedSampleCount;
                                // The number of samples that have been zeroed
                                // whilst there are no unfinished pages in the
                                // linked list.  Should only be modified in
                                // the callback.  Use resetSkippedSampleCount
                                // to reset the counter.
                                // If resetSkippedSampleCount is true the value
                                // of skippedSampleCount should be ignored and
                                // instead assumed to be 0.
    volatile bool resetSkippedSampleCount;
                                // Set true to reset skippedSampleCount (the reset
                                // takes place within the callback, at which point
                                // this is cleared).  Because both setting and
                                // clearing this are atomic operations there is
                                // no possibility a request to reset can be
                                // missed.

    volatile bool isPaused;     // set true to 'pause' playback and recording
                                // Never stops the PortAudio stream, just alters
                                // the data transmitted.

    PaDeviceID playDeviceID;    // Device ID for the device being used, or
                                // PaNoDevice for no device
    PaDeviceID recDeviceID;     // Device ID for the device being used, or
                                // PaNoDevice for no device
    unsigned int playChanCount; // The number of channels used to communicate
                                // with PortAudio.  Must be greater than
                                // the maximum channel number used.
    unsigned int recChanCount;  // The number of channels used to communicate
                                // with PortAudio.  Must be greater than
                                // the maximum channel number used.
} streamInfoStruct_t;
```

Listing 7.1: Type definitions for `streamInfoStruct_t`, a structure used to group all data associated with a particular PortAudio stream, `streamPageStruct_t`, a structure used to group all data associated with a particular page, and `chanBufStruct_t`, a structure to store a single channel of data.

Within this set of structures there are a few important variables not previously mentioned. The first of these are pagePos combined with pageLength within streamPageStruct_t. When a page is constructed, pageLength must be set to the length of the page which, unless a gap is required at the end of the page, must be the same as the length of the longest channel. The single variable pagePos is then used to store the current sample position within the page, making it impossible for the channels within a page to ever be temporally mis-aligned within the utility. Through a simple comparison between these two values it is also possible to determine how many samples still need to be processed in the page, avoiding the need to find the length of the longest buffer each time the page is used. Problems indexing beyond the end of each channel buffer can also be easily avoided by comparing the page position with the buffer length stored with each channel.

Two further important variables are playChanCount and pplayChansInUse within streamPageStruct_t. The first of these is used to store the size of the second to ensure indexing outside the array does not occur and, until the page has finished, should be the same as the variable with the same name in streamInfoStruct_t. pplayChansInUse is used within the PortAudio callback to indicate which output channels need to be zeroed due to not having an associated buffer of samples. Two alternative approaches to this would be to either zero all output samples at the start of the callback or to generate this list of used channels within the callback each time. The former would needlessly increase the processing time of the callback, especially if a large number of channels are in use and the page contains data for all channels, whilst the latter would require dynamic memory allocation within the callback, something which should be avoided due to the potential time it may require.

The final variable that requires further mention is stopStream within streamInfoStruct_t. This is required because, as mentioned in appendix B, although Pa_StopStream() can be used to stop the PortAudio stream, problems can be encountered freeing memory directly following this. As an alternative, when this variable is set the callback stops the stream by returning a specific value. By setting this variable and then waiting for the stream to stop before continuing, problems freeing memory are avoided.

7.4 Implementation

Based on the requirements to ensure thread safety, the functions to run in the main thread were designed. For the majority of these, one function was used for each command to be called from within MATLAB. However, to improve the structure of the code, some additional 'helper' functions were also written. These included functions to:

- create new instances of the `streamInfoStruct_t` and `streamPageStruct_t` structures, allocating the required memory and setting all the contained variables to their defaults. Although each of these is only called from one place within the utility, this neatly grouped all default values together making them easy to locate and change as necessary.
- remove structures from memory, making sure that all memory is only freed once and all necessary precautions are taken as outlined above.
- check the current state of the utility compared to that required, generating an appropriate error if required, such as when trying to add a page before the utility has been initialised.
- condense all pages that have finished, as described above.
- check the error code returned by all PortAudio functions, displaying the error number and description if an error has occurred.

Apart from these 'helper' functions and the functions specific to each command, three further functions also had to be implemented. These were:

mexFunctionCalled the function called by the core MEX-file code whenever the utility is used. This determines if the utility had been used before, and if not initialises PortAudio and registers the exit function exitFunc. This also calls the helper function used to condense all pages.

exitFunc the function to be called by MATLAB when either MATLAB is closed or the utility is cleared from memory. This ensures that all dynamically allocated memory is freed correctly, that the PortAudio stream is stopped and that PortAudio is terminated.

playrecCallback the callback function for PortAudio. This transfers all sample data between the utility's page structures and PortAudio, based on the flow diagram shown in figure 7.3.

The resulting source code for these functions, in addition to all other code specific to the operation of the utility, can be found in the files `pa_dll_playrec.c` and `pa_dll_playrec.h` on the accompanying CD, as described in appendix G. Additionally, a list of all of the commands implemented by the utility to meet the requirements can be found in appendix D along with their descriptions as returned by the 'help' command and a complete overview of the utility's operation as returned by the 'about' command.

7.5 Testing

Although the core MEX-file code had already been tested, when testing the utility as a whole many of the tests had to be repeated to ensure that the _funcLookup array had been implemented correctly, such as including the correct limits on the number of parameters. Tests like this, as well as checking for the correct return values from commands, were specifically implemented. However, to test for correct operation it was decided that this should predominantly occur through using the utility in real scenarios. This decision was made for two reasons. The first was the

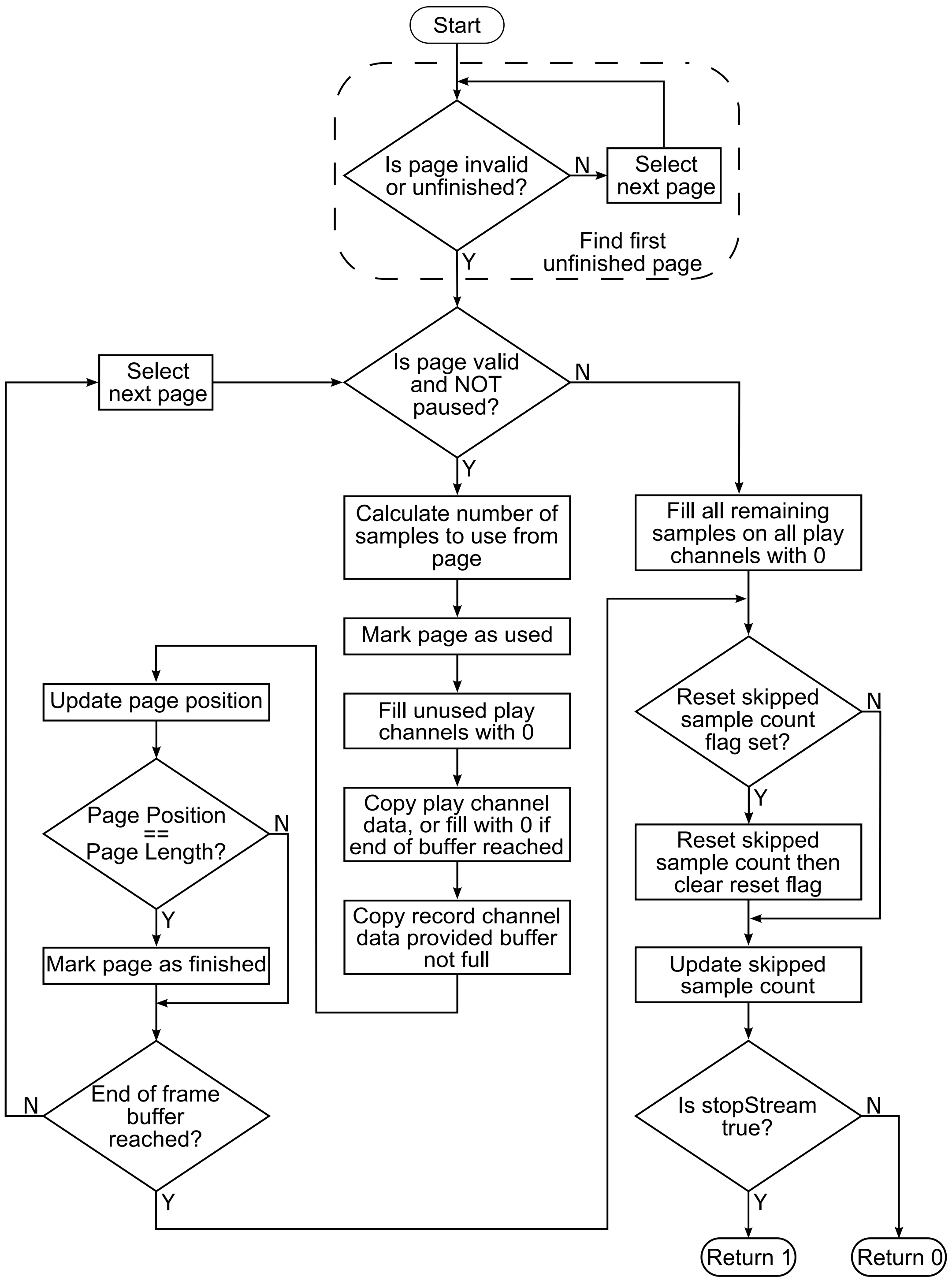

Start
Is page invalid or unfinished?
N
Select next page
Y
Find first unfinished page
Select next page
Is page valid and NOT paused?
N
Y
Calculate number of samples to use from page
Fill all remaining samples on all play channels with 0
Mark page as used
Update page position
Fill unused play channels with 0
Reset skipped sample count flag set?
N
Page Position == Page Length?
N
Copy play channel data, or fill with 0 if end of buffer reached
Y
Reset skipped sample count then clear reset flag
Y
Mark page as finished
Copy record channel data provided buffer not full
Update skipped sample count
N
End of frame buffer reached?
Is stopStream true?
N
Y
Y
Return 1
Return 0

Figure 7.3: PortAudio callback function flow diagram.

extra time already taken developing the utility, and so the reduced time available to complete the remainder of the project. The second was the likely nature of any bugs that might exist: after initial testing for underlying correct operation—such as testing continuous input and output, pausing and resuming, page list generation and correct explicit and automatic removal of pages from this list—any bugs potentially remaining would most likely be due to either incorrect dynamic memory (de)allocation or unexpected 'interaction' between threads, both of which are best found through extensive use of software.

Examples of the specific tests implemented included:

- confirming each command correctly accepted its valid range of parameters whilst rejecting both one more and one less, where applicable, than this range. Due to the testing of the MEX-file core code, further testing outside this range was not conducted.
- confirming the list of commands displayed all available commands, and the help information for each command displayed as expected based on the text entered into the `_funcLookup` array.
- confirming that audio output across multiple pages was continuous, containing no audible glitches. This was implemented by outputting a sine wave across multiple pages. To confirm the correct samples were always being used, the period of the sine wave, the page size and the callback frame size were all set to be non-integer multiples of each other. Additionally, varying page sizes and different frequencies on each channel were tested.
- confirming that audio recording across multiple pages was continuous, containing no audible glitches. To test this, sine waves, as used to test the output, were looped back to the input channels. Through listening to these recoded signals, any glitches in recording would be heard.
- confirming correct operation of automatic page removal, explicit page removal and page list generation. This was tested by rapidly adding a group

of pages—some just to record, some just to play and some to do both—to the utility in a pseudo-random order. Then whilst these pages were processed the page list returned by the utility would be monitored, ensuring that only the play only pages were automatically removed. Further to this, correct operation was tested when all three types of pages were deleted in all three possible states: prior to starting being processed, whilst they were being processed, and after they had finished being processed.

- confirming that appropriate warnings and errors were displayed when channel lists were used either containing duplicated channel numbers or channels out of the range specified during initialisation. Additionally, correct warnings were confirmed for scenarios when the number of output channels of data and the list of output channels to use were of different sizes.

One further test was to repeat the loop back tests initially conducted using the standard MATLAB audio functions and pa_wavplay, as described in chapter 4. Through the use of consecutive combined play and record pages this would test if the relative timings of the inputs and outputs remained the same over prolonged periods of time. Unexpectedly, since no glitches had been found in previous testing, the delay was found not to remain constant. On most occasions during this testing the recording always preceded the output by 584 samples, although sometimes this started at 579 samples before changing to 584 samples during the test. Whenever the delay started at 579 samples, this change was always found to occur, although the time between initialising the utility (the utility was re-initialised before each test commenced) and the change occurring did not seem to have any pattern taking from seconds through to minutes. Instead, it was found that the change could sometimes be 'triggered' through significantly increasing hard-disk access and processor usage, such as starting a new application. It was surmised that the relatively short durations of the original 'continuous output' and 'continuous input' tests were not long enough to observe this behaviour, which is why no problems were found initially.

Importantly, it was found that the change in delay would only ever occur once, after which the relative timings would remain constant—two tests each of over 12 hours demonstrated this.

As when testing the pa_wavplay utility, the delay was found to be the same on all channels. Therefore, by connecting an external signal generator to one input whilst recorded a loop back signal on another, the origin of the delay change could be narrowed down. The two resulting signals during a change in delay are shown in figure 7.4, showing that the 5 extra samples are added to the output signal, as expected by the glitch heard in the output sine wave.

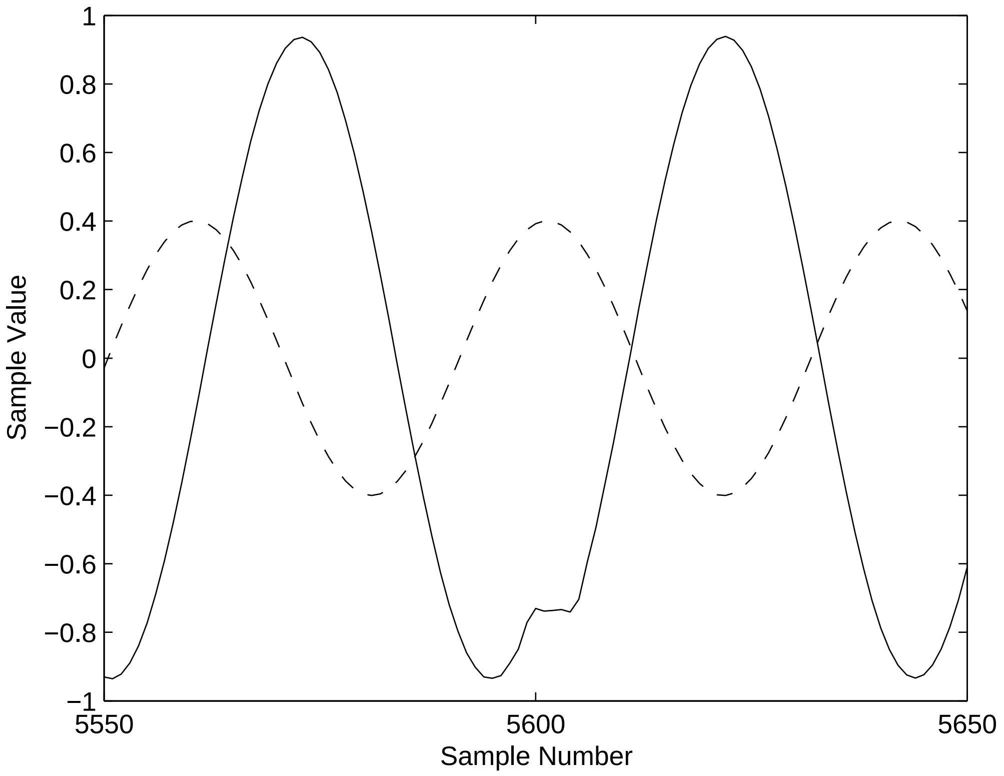


Figure 7.4: Comparison of a looped back sine wave (solid line) and an externally generated sine wave (dashed line) during a change in delay when using the playrec utility. This shows how only the looped back signal is affected during the change, and therefore the extra samples all occur within the output.

After continued testing, neither the pa_wavplay utility nor the initial functional test code could be made to reproduce this problem. However the results measured during the initial testing of the pa_wavplay utility, as described in chapter 4, were seen to indicate that this change may occur, although very occasionally compared to that with the playrec utility. To try and explain the cause of both these infrequent variations when using the pa_wavplay utility and the change of delay within the playrec utility, the effect of various different changes were investigated. This included restarting the computer multiple times to determine if there are variations in how the soundcard initialises itself as well as using different sample rates and numbers of channels to determine if the problem may be due to the soundcard configuration. None of these provided any clues to the cause of the problem and instead just revealed that there was no absolute guarantee what delay between output and input would ever occur. Also, as before, the delay measured was always the same on all channels.

Due to the use of a single variable to store the position within each page, it was realised that the change in delay could not be occurring within the utility. However, this did not rule out the utility inadvertently triggering the change. If this was the case, the most likely cause would be the execution time of the PortAudio callback although various tests, including adding delay loops into the callback, did not reveal any connection between this and the delay changes. This was not surprising because, if this was the trigger, the change in delay would be expected to occur on more than one occasion.

Although a large amount of further testing and investigation into the operation of PortAudio and Steinberg's ASIO SDK may have found the cause of this problem, this was expected to take a long time with potentially no solution at the end, especially if the problem was hardware related. Therefore, instead, the fact that the change in delay only ever appeared to occur once was utilised and a MATLAB function was written to monitor the latency between an output and input until a predetermined 'stable' delay was measured. Although not ideal, using this before all subsequent testing would allow the rest of the project to continue and, by ensuring a loop back signal was always recorded during all tests, the delay between output and input could be checked later on should anomalies appear. Furthermore, by trying to run as many tests as possible without restarting either MATLAB or the playrec utility, the possibility of this causing any problems would be reduced.

Chapter 8

Excitation Signals

To measure the distance between a sound source and microphone, the time-of-flight, t (s), of a sound between them can be used along with the speed of sound in air, c (ms^{-1}), to give the separation as

$$d = ct \text{ (m)}. \tag{8.1}$$

The speed of sound in air can be approximated by

$$c = (331.5 + 0.6T) \text{ (ms}^{-1}\text{)} \tag{8.2}$$

where T is the air temperature in degrees Celsius. In order to use this approach an accurate method of determining the time-of-flight is required. Approaches using a single stationary microphone require a known signal, the excitation signal, to be transmitted at a known time, and the processing of the recorded microphone signal to determine when the sound arrives. Many different signals can be used, although each has its own advantages and disadvantages. This chapter considers the suitability of different signals before providing implementation details for those signals to be used in all tests.

8.1 Signal Types

8.1.1 Pulsed Sine Wave

This is the simplest form of excitation signal and through the use of a matched filter can provide a high Signal-to-Noise Ratio (SNR). However, due to the signal's repetitive nature it is only possible to measure the relative phase between loudspeaker and microphone. Hence the actual separation could be one of an infinite number of values, separated by multiples of the signal wavelength. By determining the relative start and end times of the received signal the measured separation could be reduced to a much smaller subset of values, although not necessarily to just one. However, if room reflections sum together to give a signal 180° out of phase with the direct sound then it is possible that the microphone would not receive the signal at all. Although this is unlikely, a much more likely scenario is for the reflections to add such that the phase of the received signal is different from that solely of the direct signal. As such, the only reliable measurement of phase would be that made before any reflections reach the microphone—something difficult to achieve, especially in a normal environment where the loudspeakers may be placed against a wall or even in a corner. Through the use of more than one signal frequency the problems of having an infinite number of potential separations can be reduced or eliminated, although the requirement to measure the relative phase before the first reflection arrives still exists, making this approach impractical.

8.1.2 Impulse

The use of an impulsive signal enables the impulse response (IR) for a particular arrangement of loudspeaker and microphone to be measured. From this it is possible to determine the relative separation by determining the time for the direct signal to be received. Although, as with the sine wave, room reflections will interfere with the received signal this does not happen until after the direct signal has been received, and so can be ignored without any problems. However, because impulses from loudspeakers can only contain relatively small amounts of energy, such an approach can be very susceptible to even small amounts of background noise, making it difficult to identify the impulse from the noise. To try and remove this problem, multiple measurements can be taken and averaged. This requires a long enough time to be left between repeats to ensure that the previous impulse has completely decayed away, and even then may require many repeats to obtain an accurate result.

8.1.3 Maximum Length Sequence (MLS)

As with an impulse excitation signal, the IR of a room can be measured using a Maximum Length Sequence (MLS). However, this technique is much less susceptible to background noise than the impulse case. An mth-order MLS is a pseudo random signal of length $L = 2^m - 1$ samples, which can be generated using a shift register by exclusively or-ing particular registers, as shown in figure 8.1[33]. The resulting signal is periodic with period L samples which can then be scaled as required to obtain the loudspeaker signal.

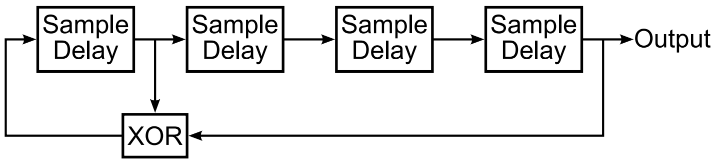


Figure 8.1: Example 4th-order binary Maximum Length Sequence (MLS) generation using a shift register.

After the response of a linear time-invariant system, such as a room, to this signal has been measured the IR can be determined in two different ways. The first uses circular correlation[34] whilst the second uses the Hadamard transform[19]. In both cases, the same result is obtained.

Importantly, the Fourier Transform of the original MLS has the same magnitude for all frequencies, apart from the dc component, whilst the phase is effectively randomised. This results in extraneous noises within the room being uniformly distributed along the resulting IR (provided the noise has a white spectrum)[34], meaning the technique is particularly suited to scenarios where people may be in the room and moving around. However, this technique does also exhibit various disadvantages, the most significant of which are distortion peaks due to nonlinearities in the system, such as within the loudspeaker[41]. Additionally, if the length of the MLS is shorter than that of the IR for the room, time-aliasing occurs with the 'tail' of the IR appearing overlaid on the start of the actual IR[34].

8.1.4 Inverse Repeated Sequence (IRS)

Based on a MLS, an mth-order Inverse Repeated Sequence (IRS) has length $2L$ samples and can be generated using the equations[34]

$$IRS[n] = \begin{cases} MLS[n], & n \text{ even} \\ -MLS[n], & n \text{ odd} \end{cases} . \tag{8.3}$$

In this case only circular correlation can be used to deconvolve the IR from the measured room response to the IRS[10, 34]. This technique has been shown to have "complete immunity to even-order nonlinearity while maintaining many of the advantages of MLS"[6]. However, it does require twice as long an excitation signal to be used compared to using an MLS because, to avoid time-aliasing, L must still be longer than the IR length.

8.1.5 Optimum Aoshima's Time-Stretched Pulse (OATSP)

The concept of using a Time-Stretched Pulse (TSP) to measure the IR of a linear system was introduced by Aoshima[1]. In this, an impulse signal with a near flat power spectrum undergoes expansion by applying a phase shift to each frequency component in the frequency domain. The resulting signal has the same total energy as the original impulse although it has a much smaller maximum amplitude in the time domain. Therefore the whole signal can be increased in amplitude and so the final excitation signal can contain much more energy than the original impulse, and so significantly increase the SNR. By applying the opposite phase shift to each frequency component within this signal, the original impulse, albeit much larger, can be obtained. Similarly, if this opposite phase shift is applied after the TSP has been passed through the linear system, the IR of the system can be obtained.

As an extension of the single specific example given by Aoshima, Suzuki *et al*[36] proposed the "Optimum" Aoshima's Time-Stretched Pulse (OATSP), specified in the frequency domain as

$$H[k] = \begin{cases} \exp\left(\frac{j4m\pi k^2}{N^2}\right), & 0 \leq k \leq \frac{N}{2} \\ H^*[N-k], & \frac{N}{2} < k < N \end{cases} \tag{8.4}$$

with the inverse, deconvolution, filter given by

$$H^{-1}[k] = \begin{cases} \exp\left(\frac{-j4m\pi k^2}{N^2}\right), & 0 \leq k \leq \frac{N}{2} \\ H^{-1^*}[N-k], & \frac{N}{2} < k < N \end{cases} \tag{8.5}$$

or more simply

$$H^{-1}[k] = \frac{1}{H[k]}$$

where m is an integer determining the 'stretch' of the OATSP and N is the signal length.

(In [36], (8.4) and (8.5) are given without the complex conjugate for $\frac{N}{2} < k < N$, although this is necessary to obtain a real time domain signal.)

The excitation signal can be obtained by taking the Inverse Discrete Fourier Transform (IDFT) of (8.4). Then, provided the IR is shorter than N samples, circular convolution of the system's response to this signal with the inverse filter will result in the system's IR. Furthermore, if the IR is longer than N samples, the time-domain excitation signal and inverse filter can be circularly rotated by

$$n_{rot} = \frac{N}{2} - m \text{ (samples)}$$

before being extended with zeros to make the total signal length longer than the IR. In this case linear convolution must then be used to obtain the IR[36].

8.1.6 Logarithmic Sweep

The logarithmic sweep excitation signal, unlike the other excitation signals discussed, contains the same energy per octave. This is achieved by ensuring “the frequency increases with a fixed fraction of an octave per unit time”, such that

$$\frac{\log\left(\frac{f_2}{f_1}\right)}{T_2 - T_1} = \mathit{constant}$$

where f_1 is the frequency at time T_1 and f_2 is the frequency at time T_2[23]. This has the advantage that a higher proportion of the excitation signal energy is at lower frequencies, and so the SNR for these frequencies will be improved.

Additionally, using this technique “each harmonic distortion packs into a separate impulse response” in the deconvolved signal, at points prior to the linear IR of the system. As such, the system’s linear IR can be separated from any distortion artefacts[8].

Unlike the OATSP, the logarithmic sweep can be generated in the time domain using the equation

$$x(t) = \sin\left[\frac{\omega_1 T}{\ln\left(\frac{\omega_2}{\omega_1}\right)}\left(\exp^{\frac{t}{T}\ln\left(\frac{\omega_2}{\omega_1}\right)} - 1\right)\right] \tag{8.6}$$

where ω_1 is the starting frequency, ω_2 is the ending frequency and T (seconds) is the duration of the sweep. The inverse filter, with which the room’s response can be convolved to obtain the IR, is then generated by “time-reversing the excitation signal, and then applying an amplitude envelope to reduce the level by 6 db/octave”[8].

Another advantage of this technique is that, as with OATSP, the sweep does not need to be longer than the IR and instead it can be extended with silence to measure longer IRs. If the resulting signal is still not long enough, time-aliasing, as would occur with both MLS and IRS, does not occur and instead the end of the IR is just ‘lost’[8].

8.1.7 Summary

A comparison of the four excitation signals MLS, IRS, OATSP and logarithmic sweep was conducted by Stan *et al* in [34]. From this it was concluded that the noise immunity of both MLS and IRS made them more suited to measurements in “an occupied room or in the exterior” and additionally these excitation signals were “more bearable and more easily masked out”. However, these signals required much more calibration to obtain optimum results and exhibited distortion peaks, something not present when using either of the other signals. OATSP was found to perform best at high output signal levels to reduce the SNR, although this was said to make it “unusable in occupied rooms”. Finally, the logarithmic sweep was found to be “the best impulse response measurement technique in an unoccupied and quiet room” even though it did not require “tedious calibration”.

Based on this assessment, none of these signals is best for all of the conditions under which the loudspeaker locations would ideally be measured—that of an occupied room where the signal level does not require any manual calibration and ideally very little automatic calibration. For this reason, in addition to determining if the direct sound ‘peak’ in the IR appears differently using the different techniques, all four techniques were investigated.

8.2 Signal Creation and Processing

Although the excitation signals to be tested required different algorithms to both create the excitation signal and process the recorded signal, the process of playing and recording would be the same in all cases. Additionally the same excitation signal would be used many times, and so to speed up the analysis as many calculations as possible associated with the post-processing stage were moved to the signal creation stage, thus ensuring they only had to occur once. To achieve this, an easy and generic method of keeping track of the excitation signal and recorded signal(s) as well as information about the type of signal was required.

The chosen method was to use three variables within MATLAB: first, a vector for the excitation signal; second, a vector or matrix for the recorded signal(s), of a length equal to or greater than the excitation signal; and third, a 'parameters' structure. This contained all the relevant information about the type of signal being used including one 'type' field indicating the type of signal, and hence the function that must be used to process it. With this approach, the function to create each test signal would return the excitation signal and the parameters structure whilst the function to process the recorded signal would just take the recorded signal and the associated parameters structure. Based on this, four `create_` and four `process_` functions were implemented, each ending in either `mls`, `irs`, `oatsp` or `sweep` to indicate the excitation signal type they created and processed respectively. Additionally a generic `process_recording` function was created to call the relevant `process_` function, based on the contents of the supplied parameters structure. The files containing these functions can be found on the accompanying CD, as described in appendix G.

8.2.1 Maximum Length Sequence (MLS)

To create the MLS excitation signal three parameters were required:

sequence order, m, which is the same as the size of the shift register required and from which the period of the sequence, $L = 2^m - 1$, can be determined. The shift register taps given by Vanderkooy[41] were used to generate the MLS due to their suggested improvement in performance compared with other possible combinations. Using these, a binary MLS between 0 and 1 was generated using code as shown in listing 8.1.

Although, provided at least one bit is set, the shift register can start in any state, a start state with all bits set was always used for consistency.

repeat count to specify how many times the length L sequence should be repeated, which can be used to improve SNR through averaging during analysis. However, due to the use of a circular deconvolution method, either the system response during the first repeat should be excluded from the analysis or the system response after the last repeat should also be included in the averaging process.

signal scaling to specify the values between which the MLS should switch. To create a symmetrical signal about zero, this was specified as a single scaling factor such that the sequence used the values ±*scaling*.

```
L = 2^morder - 1;

% Create morder-bit bitmask and set the shift register (sr) start state
bitmask = bitcmp(uint32(0), morder);
sr = bitmask;

% Create a place to store the output sequence from the shift register,
srout = int8(zeros(1,L));

for i = 1:L
    srout(i) = bitget(sr, morder);
    sr = mod(sum(bitget(sr, taps)), 2) + bitand(bitshift(sr, 1), bitmask);
end
```

Listing 8.1: MATLAB implementation of a shift register of order `morder`, where `taps` already contains the required tap numbers, as given in [41], in a row vector.

In principle circular correlation could be used to deconvolve this signal. However, due to the fact that the signal length was not a power of two and was also large, implementing this in either the time or frequency domain could be very time consuming, especially when compared to using the Fast Hadamard Transform (FHT). For this reason the FHT was used, and as a result two further vectors were created whilst creating the MLS—one to reorder the samples prior to implementing the transform and a second to reorder the output of the transform. These were generated in a similar manner to [19]: the first can be generated using the shift register value as each sample is shifted out. The second, however, is generated in two stages—first the occasions when only one bit is set in the shift register are determined (that is, contains a value which is a power of two) and then this, combined with the values in the original sequence, is used to determine the second reordering vector.

Through storing these reordering vectors within the ‘parameters’ structure, the process of deconvolving the room’s response is greatly simplified:

1. A number of repeats of the recording are averaged to obtain a single sequence L samples long. To allow comparisons between the averaging of different numbers of repeats, the range of repeats to average could be specified when calling the MLS processing function.

2. The averaged samples are reordered according to the first reordering vector, including the addition of an extra, zero, sample at the start.

3. The FHT is applied. Different methods of implementing this were evaluated including one that applied the transform on the data in-place[19], whilst another required enough space to store $m+1$ times the length of the MLS[17]. The final method chosen was based on the latter, but only requiring two buffers by bouncing the data between them. This was chosen due to the longer execution time of the in-place method within MATLAB, a result of requiring multiple nested loops.

4. The processed samples are reordered according to the second reordering vector, including the removal of the first sample, and scaling as appropriate. The resulting signal contains the room’s IR starting at the first element.

8.2.2 Inverse Repeated Sequence (IRS)

To create this signal, initially an unscaled MLS was required, generated as described above. Following this, the IRS was generated using (8.3) before it was repeated and scaled in exactly the same way as for the MLS. However, due to the need to use circular correlation to deconvolve this signal, no further variables had to be created at this stage.

The processing of this signal was no more complicated than its creation—averaging of multiple repeats, as described above, and then calculating the circular correlation between this averaged signal and the original IRS. The resulting signal contains the room's IR starting at the first element with an inverted version of the IR starting after L samples.

8.2.3 Optimum Aoshima's Time-Stretched Pulse (OATSP)

When creating an OATSP more parameters needed to be specified than for either the MLS or IRS. These were the OATSP length, N, the stretch factor, m, the length to extend the OATSP with zeros, the number of repeats of the signal, and the scaling factor (the required peak value). The most complicated of these is m, which can have significant impact on the error caused by using linear convolution instead of circular convolution[36] as well as on the amount of energy within the signal, and hence the SNR. For example, at one extreme, when $m = 0$, an impulse is obtained which contains very little energy but which would introduce no error when processed with linear convolution. In comparison, when m approaches, and is greater than, $\frac{N}{2}$, the signal can contain far more energy than the impulse but the tail of the signal 'wraps around' onto the start, introducing a significant error when using linear convolution.

One way to determine an appropriate value for m is to calculate the value which introduces an error which is just below that required, such as -98 dB, the equivalent dynamic range of 16 bit quantisation[36]. For this reason, two functions in addition to `create_oatsp` were created—one to calculate the average or maximum error due to using linear convolution, `oatsp_CalcPowErr`, and a second to find the value of m to obtain a particular error, `oatsp_FindPowErr`.

Having determined the required value for m, the unscaled OATSP was generated using the code shown in listing 8.2. To obtain the required excitation signal, this time domain signal was then scaled, extended at the *end* with zeros and repeated according to the specified values. The final inverse filter was created by adding the same number of zeros to the *start* of the time domain signal before scaling by the same factor.

```
n1 = 0:N/2;
n2 = (N/2+1):(N-1);

H(n1+1) = exp(j*4*m*pi*n1.^2/N^2);
H(n2+1) = conj(H(N - n2 + 1));

G=1./H;

rot = N/2 - m;

h = circshift(real(ifft(H, N)), [0, -rot]);
g = circshift(real(ifft(G, N)), [0, rot]);
```

Listing 8.2: Creation of an Optimum Aoshima's Time-Stretched Pulse (OATSP) of length N and stretch m using MATLAB, generating both the time (h and g) and frequency (H and G) domain representations of both the excitation signal (h and H) and inverse filter (g and G).

To process the resulting recorded signal, it was initially averaged over multiple repeats before being windowed with a Tukey window to reduce the errors when applying a DFT. This was then linearly convolved with the 'extended' inverse filter to obtain the IR, which started in the middle of the resulting signal in the sample *after* the length of the 'extended' excitation signal.

8.2.4 Logarithmic Sweep

Initially two different methods were investigated for creating the logarithmic sweep. The first worked in the time domain, as given by (8.6), whilst the second worked in the frequency domain, as described by Müller and Massarani[23]. The advantage of the latter is reduced ripple in the magnitude spectrum, although in the time domain this was found to produce more than one frequency at any one point in time and as a result would not enable the linear IR to be measured without the effect of harmonic distortion, as described by Farina[8].

To generate a logarithmic sweep within MATALB, the `chirp` function from the Signal Processing Toolbox can be used, provided an initial phase of -90° is specified such that the signal starts at zero. This signal must be windowed, to avoid errors due to transients at the ends of the signal, and scaled before being repeated as required. The inverse filter can then be generated following the steps quoted in section 8.1.6.

To obtain the IR, exactly the same process was used as for the OATSP: average, window and then linearly convolve. However, in this case the first sample of the linear IR was *at* the length of the excitation signal, not the sample after it.

8.2.5 Testing

Whilst creating all the required files to generate and process these different excitation signals, their performance was compared with that expected. This included analysing both the time and frequency domain representations of all excitation signals and inverse filters, where applicable, as well as confirming that an impulse is obtained when each excitation signal is processed directly. Further to this, each excitation signal was linearly convolved with an imaginary room IR before deconvolving to obtain the IR. The resulting IR was then compared to that convolved with the excitation signal, confirming if the functions were operating as required. Finally, to check the logarithmic sweep was implemented correctly harmonic distortion was added to the excitation signal prior to processing. The resulting IR was then visually inspected to ensure each order of harmonic distortion had packed into its own IR prior to that of the linear IR.

8.3 Test Implementation

As it would not have been possible to investigate the impact on measurement accuracy of every excitation signal parameter, a set of excitation signals to be used in all tests were specified. These all used a sample rate of 44100 samples/s, and were defined as follows:

MLS A 16^{th}-order MLS giving $L = 65535$ and so a duration of approximately

1.5 s, ensuring time-aliasing would have minimal effect. This was repeated five times to allow signal averaging.

IRS A 16^{th}-order IRS giving $L = 65535$, and therefore a signal period of approximately 3.0 s. This was repeated three times to allow signal averaging whilst maintaining an overall signal length comparable to that of the MLS.

OATSP1 An OATSP with length $N = 2^{12} = 4096$ and 90% stretch factor giving $m = 1843$, padded with $2^{16} - 2^{12} = 61440$ zeros and then repeated 5 times. The zero padding in this signal, as with OATSP2, was included so that all OATSP excitation signals had the same period, thus allowing a comparison between different values of N. A 90% stretch factor was greater than that used in [34] but still corresponded to an average power error from using linear convolution of approximately -101.3 dB, so was used to increase the total signal power.

OATSP2 An OATSP with length $N = 2^{14} = 16384$ and 90% stretch factor giving $m = 7373$, padded with $2^{16} - 2^{14} = 49152$ zeros and then repeated 5 times.

OATSP3 An OATSP with length $N = 2^{16} = 65536$ and 90% stretch factor giving $m = 29491$, repeated 5 times.

Sweep1 A logarithmic sweep with length $N = 2^{16} = 65536$, start frequency $f_1 = 10$ Hz and end frequency $f_2 = 22000$ Hz. This is the same as that used in [34], although in this case the signal was repeated five times so that all the excitation signals were of similar length.

Sweep2 A logarithmic sweep with length $N = 2^{18} = 262144$, start frequency $f_1 = 10$ Hz and end frequency $f_2 = 22000$ Hz. This excitation signal was included to allow a comparison between averaging a repeated signal and using no averaging for a signal of similar total length. Theoretically, avoiding the use of multiple averages solves problems associated with measuring slightly time-invariant systems[8].

The amplitude of each of these signals was set by measuring the level recorded when each signal was directly looped back. This ensured that any overshoot that may have occurred due to filtering within the soundcard would not introduce distortion caused by clipping[23]. The scaling factors used were 0.4 for the MLS and IRS signals and 0.9 for all other signals.

Every time an excitation signal was used in a test, the excitation signal, recorded signals and parameters structure were saved to a new file along with specific information about the test, such as the loudspeaker-microphone separation and their relative angles. Including the excitation signal and parameters structure in every file significantly increased the amount of disk space used through duplication of this data. However, it ensured that all data associated with one particular use of an excitation signal could be easily accessed in the future should the need arise. After completion of a set of tests, an index to these files was generated in which all information specific to each test was stored together with the number of the file containing the associated recorded data. This allowed fast access to specific recordings based on a certain 'search' criteria without the need to open all the original files. Although such an index could have been generated as the recordings were made, if for any reason a recording had to be stopped and repeated (such as a loud noise outside the room) then manually updating the index could have been awkward. From these files, one further set of files were created containing the IR generated by processing each recording. To enable the same test index to be used, each of these files were also given the same number as the original recording but with a different suffix. Additionally, to reduce the disk space used and the time taken to load these files, only the section of the IR around the main peak was saved along with the offset of this section within the complete IR. For multi-channel recordings, the section saved was determined using the peaks in all channels to ensure enough of the IR would always be saved.

By using the single-precision data type when saving all directly recorded data, a significant amount of disk space was saved without reducing the accuracy of the values. However, during the processing of these recordings the use of single-precision values was sometimes found to introduce significant errors, such as when calculating the mean and standard deviation with the supplied MATLAB commands. To avoid this, all values were converted to double-precision prior to any other form of manipulation.

8.4 Accuracy and Repeatability

When analysing the performance of the different excitation signals two important factors had to be determined. The first was the accuracy to which distances and angles could be measured and the second was the repeatability of measurements—that is, if the system components (microphone and loudspeaker) were repeatedly placed in exactly the same locations, how similar would the results be. For example, when measuring the separation of a particular configuration, the same result may be obtained every time but it might be completely wrong. In such case, there is poor accuracy but good repeatability.

To determine the repeatability, each time the microphone and loudspeaker were placed in a particular position the system's response to all of the different excitation signals was measured three times *without* making any physical changes. Additionally, to reduce the likelihood of any disturbances from outside the room affecting all three measurements of one signal type, the response to all of the different excitation signals was measured once before repeating them all for the second and third times. Following this, the microphone/loudspeaker would be moved to its next test location and the whole procedure would be repeated.

To help determine the accuracy of these measurements, the whole process—repeating every excitation signal three times with each physical configuration—was repeated again. Although discrepancies between this repeat test run and the original test run could be due to problems with repeatability, such as uncontrolled external factors including air temperature and pressure, they could also be due to inaccuracies in the positioning of both the microphone and loudspeaker. To be able to correctly distinguish between these sources of variation, the accuracy with which items are positioned should be greater than the expected accuracy of the measurement technique. However, due to the equipment available this was not possible and instead this error had to be minimised where possible and taken into account during the analysis.

Chapter 9

Delay Measurements

Although testing of the two target microphone configurations, a SoundField microphone or four spaced microphones, could have been conducted from the start, it was decided that initially only a single omni-directional microphone to measure separation should be used.

9.1 Influencing Factors

When designing the tests to analyse how accurately separation could be measured, there were many factors that could influence the results, some of which were specific to the target application of the technique. For example, normally when a room's IR is being measured every effort is made to keep the room quiet and, where possible, time-invariant by ensuring that no-one and nothing moves within the room. However, in a system to automatically determine where loudspeakers are within a room it is likely that there will be other people in the room who will be both making noise and moving around. Furthermore, depending on how the final system is physically implemented, the microphone itself may not stay stationary if it is being hand-held.

It was also expected that the loudspeakers and microphone used would significantly impact the results, as could the amplifier and even the Digital-to-Analogue (D/A) and Analogue-to-Digital (A/D) conversion of the signals. This was because the analysis techniques assume that the sound produced by the loudspeaker is that of the excitation signal whilst the signal recorded is that at the position of the microphone. However, this is not the case because of imperfections in the frequency response of each system component. One solution to eliminate the effect of some of these imperfections is to compare the room IR with that of a direct loop back channel, a technique used to measure the frequency response of loudspeakers. However, this does not correct for all the system components and could be very awkward and costly to implement in a distributed system where the D/A converter, amplifier and loudspeaker are in a different part of the room from the A/D converter. So, although a loop back channel would be included in all tests to determine the delay within the computer (something which could be predicted when using dedicated hardware), this channel would not be used to correct for the frequency response of any system components.

Other factors that would not initially be investigated included the effects of using different loudspeakers/microphones/amplifiers, of placing obstructions between the loudspeaker and microphone, of different air temperatures, of orientating the loudspeaker so as not to directly face the microphone, of having people moving within the room or of different levels of background noise during tests. However due to the space available, within the room where the tests were being conducted there would always be a computer running and a person, albeit staying still, making some background noise.

Although these factors would need to be considered prior to a final 'complete' system being implemented, they were all seen as extensions that should be investigated following implementation of a 'basic' system.

9.2 Test Setup

The first test setup was designed to determine the performance achievable when all room reflections could be easily ignored because both microphone and loudspeaker were in the middle of the room. It consisted of a loudspeaker placed on a loudspeaker stand such that its base was 1.10 m above the ground, its top was over 1.50 m from the ceiling and all other edges were at least 1.80 m away from the nearest wall. An omni-directional microphone was then mounted on a stand such that vertically it aligned with the middle of the loudspeaker, and this was then moved along a line perpendicular to the face of the loudspeaker, as shown in figure 9.1. The range of separations, from 0.25 m to 2.00 m, was used to cover a wide range of the distances likely to be measured when configuring Ambisonic systems, and especially those scenarios where there is no space behind the listener so the loudspeaker is placed far closer than it should be. Ideally separations greater than 2.00 m would have also been tested, but due to the size of the room these were not possible whilst keeping both the microphone and loudspeaker in the 'middle' of the room.

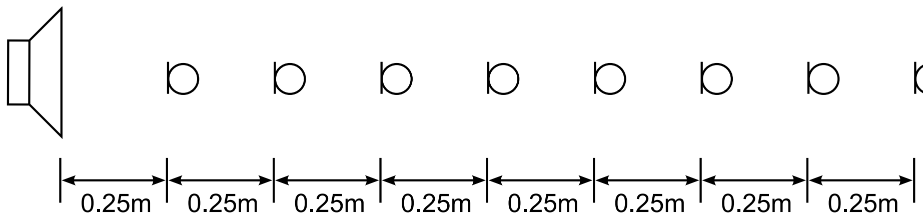


Figure 9.1: Test positions of the microphone during all single-microphone tests used to determine the performance achievable when measuring separation using propagation delays.

A more likely real-world scenario would place the loudspeaker close to a wall or even in a corner of the room, where the first room reflections arrive very shortly after the direct sound. Another potential scenario would place the loudspeaker partially facing a wall such that the amplitude of at least one reflection is greater than that of the direct sound. In a final system this would be unlikely considering the microphone should be placed at the location of the listener, but its effect should still be determined and, if possible, accounted for. Despite these both being more likely than having the loudspeaker and microphone both positioned in the middle of the room, this was used to enable the limit on accuracy to be determined, without the impact of room reflections.

Although the acoustic centre of the loudspeaker was probably not in the centre of its front face, this was used as the reference point for all measurements because it provided a rigid point from which all measurements could be accurately made. Similarly, all measurements were made relative to the very front of the microphone. An estimation of the actual location of the acoustic centre of both the loudspeaker and microphone could then be made during analysis.

Due to problems with the ruler bending when trying to measure the larger separations directly between the loudspeaker and microphone, an alternative approach of making the measurements along the floor was considered. However, this was found to cause even greater errors due to the unevenness of the floor altering the angle of the microphone stand. A second alternative was to move the microphone along a beam positioned perpendicular to the front face of the loudspeaker. This would then provide a solid surface along which to measure whilst avoiding any problems due to the uneven floor. However, this was rejected due to the likelihood of reflections from the beam and vibrations travelling down its length interfering with the signal received by the microphone.

The wiring diagram shown in figure 9.2 was used for all tests using a single microphone. Detailed configuration settings for each component are given in appendix E. Loop 1 was added to measure any variations in delay within the computer whilst Loop 2 allowed any additional delays within the amplifier and sound desk, being used in place of a microphone preamplifier, to also be measured. As such, the signal passing through Loop 2 was subject to all of the same delays as the actual microphone recording apart from those in the final stage of the amplifier, the wires to and from the microphone/loudspeaker and the propagation delay between loudspeaker and microphone. Although all three loops—Loop 1, Loop 2, and that through the loudspeaker and microphone—could have been fed from a single output on the computer, two outputs were used to make the system easily scalable should more than one loudspeaker need to be connected simultaneously. So that all the delays measured could be compared, the same signal was simultaneously output on both channels, effectively measuring the IR of the soundcard.

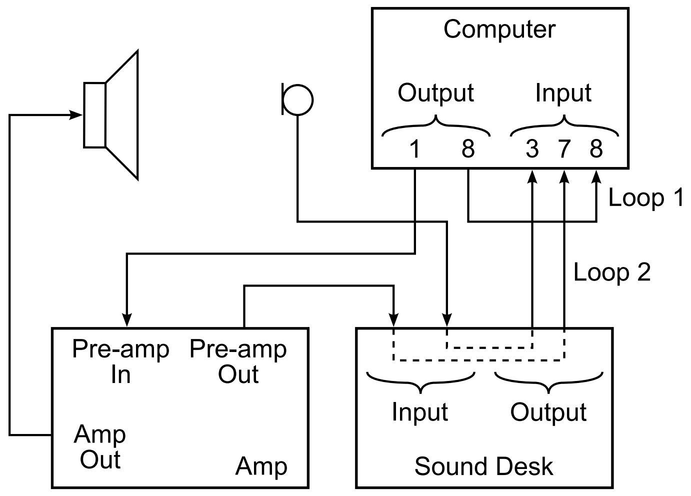


Figure 9.2: Wiring diagram for all tests conducted using a single microphone. Loop 1 allows the delay within the computer to be measured whilst Loop 2 additionally measures the delay through the amplifier and sound desk. The specification for each component can be found in appendix E.

Although the sound desk gain could have been adjusted to obtain the same peak recording level at each separation, this was seen to be manual calibration, which would not be ideal in an 'automatic' system. However, it was realised that the change in signal level between 0.25 m and 2.00 m could be significant and so a manually implemented 'automatic' adjustment procedure was used to adjust the gain: with the sound desk fader set to 0 dB each excitation signal would be tested once, if any signal clipped then the fader would be moved to -5 dB whilst if there was always at least 5 dB of headroom then the fader would be set to +5 dB. By implementing the gain change in this way the effect of not implementing it could be easily simulated using the recorded data—something not possible the other way round. Additionally, if required in the 'final' system, such automation could be practically implemented with dedicated hardware.

9.3 Loop Back Analysis

Following completion of all test runs, the loop back signals were analysed first to ensure no unexpected variations in delay had occurred within the computer, amplifier or sound desk. (All test runs were conducted over a period of 5 hours using the same instance of the playrec utility and therefore all were expected to exhibit the same loop back delay.)

When determining the location of the direct sound peak in an IR, and hence the propagation delay for the direct sound, different methods were considered including finding the location of the maximum sample, using quadratic interpolation, and using cubic interpolation.

9.3.1 Maximum Sample

Finding the maximum sample is the easiest method to determine the position of the peak within an IR, but it is also very restricted with a maximum accuracy of one sample interval. To allow for phase inversions within a system, something that should never be ruled out, the absolute maximum value can be found instead of the true maximum. This, however, was later found to cause problems due to the ripple immediately either side of the main peak having a larger amplitude than the peak itself. Because these peaks in the ripple were always of the opposite sign to the 'main' peak, during all test analysis the maximum positive value was used to locate the main peak. In a final 'complete' system an alternative solution, which would work correctly even with a phase inversion present, would need to be found. The large amplitude ringing causing these problems was attributed to the specific frequency responses of the system components, and therefore by analysing the IR after applying different filters it was expected that the presence of a phase inversion would be detectable, and thus the sign of the IR peak could be determined. However, due to the available time, this was not investigated further.

Using the maximum sample value, the delay on both Loop 1 and Loop 2 was found to always be 585 samples. Although different from the main delay value measured when testing the playrec utility, this was not seen as surprising because this value had previously been observed whilst trying to identify the source of the variation in delays.

At a sample rate of 44100 (samples/s) this level of accuracy indicated reliable timings to within 22.7 μs, although it was expected that the accuracy was much greater than this, so the use of quadratic interpolation was investigated.

9.3.2 Quadratic Interpolation

By determining the location of the peak in the quadratic equation that passes through the peak sample point as well as the sample points immediately either side, it was expected that the delay could be measured to a much greater accuracy than the nearest sample. To achieve this, initially the coefficients in the quadratic equation

$$y = ax^2 + bx + c \tag{9.1}$$

had to be determined. Assuming the peak sample, y_1, is at $x = 1$, and therefore the samples either side of this are y_0 and y_2 at $x = 0$ and $x = 2$ respectively, the coefficients a, b, and c are given by

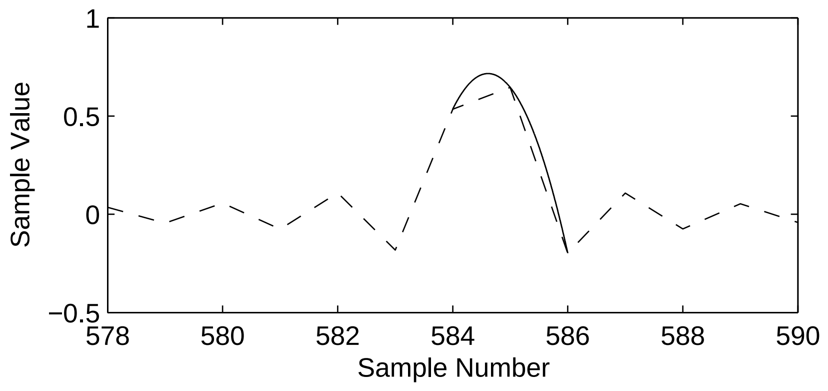
$$\begin{aligned} a &= \frac{y_0 - 2y_1 + y_2}{2}, \\ b &= \frac{-3y_0 + 4y_1 - y_2}{2}, \\ c &= y_0. \end{aligned}$$

Differentiating (9.1), the stationary point, and therefore peak position, is found to be at

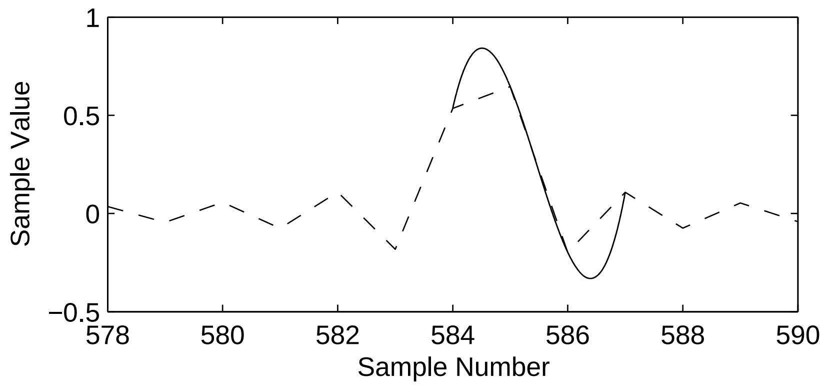
$$x = \frac{-b}{2a}$$

which can be translated back to the actual peak position within the IR by adding the sample number of the peak sample, and then subtracting one. The resulting interpolated signal, plotted alongside the original IR from a Loop 1 recording, is shown in figure 9.3(a). From this it can be seen that the peak in the interpolated signal occurs at a realistic location between the largest values of the sampled IR signal.

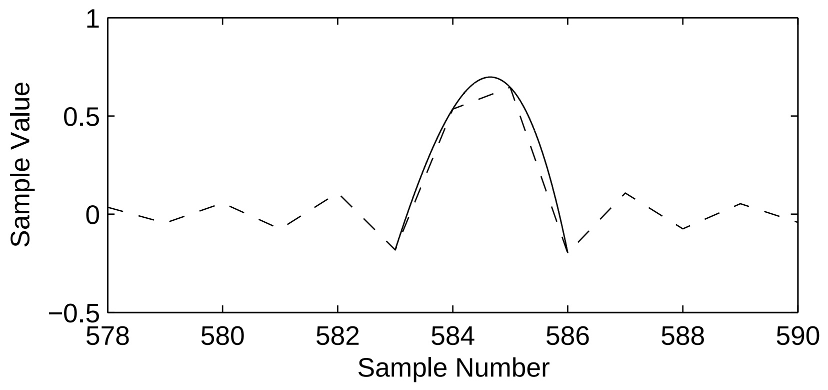
Using this technique, the delays on both Loop 1 and Loop 2 were found to vary very slightly throughout the course of the tests—Loop 1 by ≤ 0.00010 of a sample interval and Loop 2 by ≤ 0.052 of a sample interval—although the latter was found to vary with time rather than change completely randomly between tests, as shown for three excitation signals in figure 9.4. From this two significant points were observed: the delay measured is dependant on the excitation signal used and the relationship between the delays measured with different excitation signals is not linear. Due to the sequence of tests—stepping through all excitations signals for one test before moving on to the next test—the first of these could not have been caused by fluctuations in delay with time. However, this might have introduced the non-linear relationship observed between signal types because over 70 seconds would elapse between the first and last excitation signal in each test, during which time the delay could have changed. An alternative cause of this observed non-linearity is that of a time non-linearity in either the IR measurement or interpolation techniques such that the distribution of measured peak positions between samples is not even. To investigate this further, the use of higher order interpolation was considered.



(a) Quadratic interpolation



(b) Cubic interpolation using two samples *after* the peak sample.



(c) Cubic interpolation using two samples *before* the peak sample.

Figure 9.3: Comparison between an original Loop 1 impulse response (dashed line), and different interpolated signals (solid line).

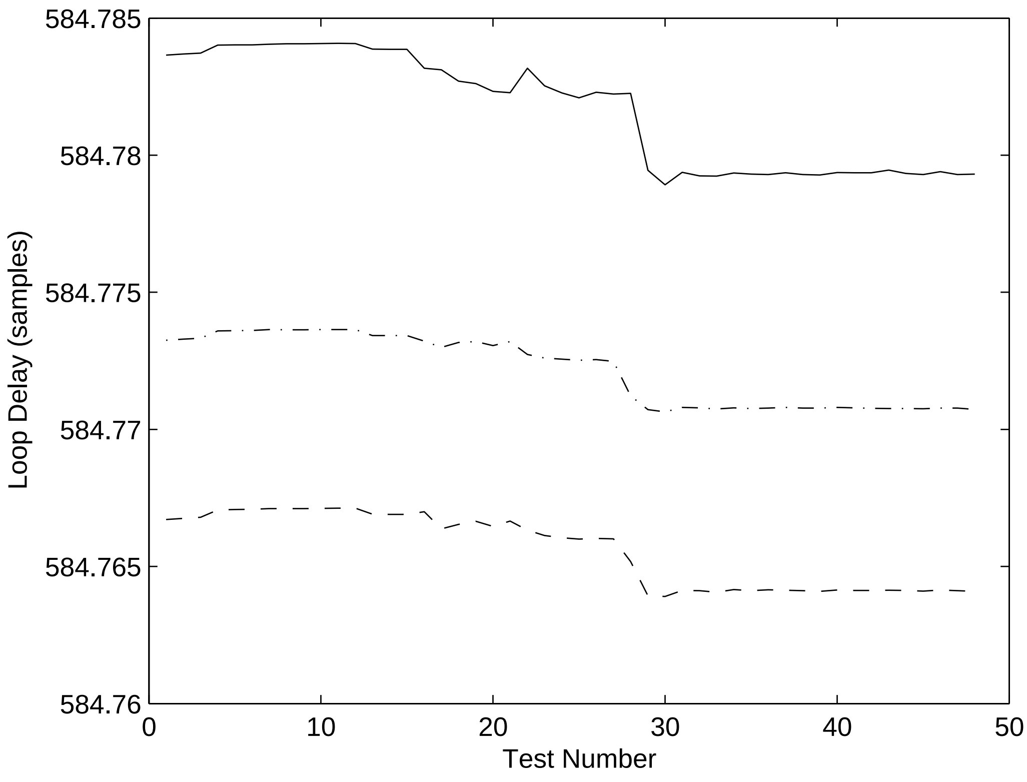


Figure 9.4: Variations in the delay measured on Loop 2 using quadratic interpolation for OATSP1 (solid line), Sweep1 (dashed line) and Sweep2 (dash-dot line). Note that the time between consecutive tests was not constant, and therefore this should only be used to observe the sequence of the delays measured.

9.3.3 Cubic Interpolation

Finding the peak position using a cubic polynomial can be achieved in a very similar manner to that using a quadratic equation. Initially the coefficients in the equation

$$y = ax^3 + bx^2 + cx + d \tag{9.2}$$

are determined for the values y_0 to y_3 at $x = 0$ to $x = 3$ (inclusive) using

$$\begin{aligned} a &= \frac{-y_0 + 3y_1 - 3y_2 + y_3}{6}, \\ b &= \frac{2y_0 - 5y_1 + 4y_2 - y_3}{2}, \\ c &= y_1 - y_0 - a - b, \\ d &= y_0. \end{aligned}$$

Differentiating (9.2), the location of the stationary point(s) are found to occur at

$$x = \frac{-b \pm \sqrt{b^2 - 3ac}}{3a}.$$

However, unlike with quadratic interpolation, there are two different ways that the values y_0 to y_3 can be aligned with the original IR. The first is to place the peak sample at y_1, as shown in figure 9.3(b), whilst the second is to place it at y_2, as shown in figure 9.3(c). From this it can be seen that the resulting interpolated signals are significantly different and as such the method chosen has a significant impact on the peak position measured. The same method cannot always be used because this introduces discontinuities in the interpolated position whenever there are two adjacent 'peak' samples with the same value. However, switching between methods based on which adjacent sample is largest was also found to experience this problem.

To avoid these discontinuities, two further possible solutions were investigated. The first was to determine the location of the peak in the average of the two cubic polynomials, and the second was to use even higher order polynomials. However, neither of these were found to significantly improve the peak position detection compared to using the simple quadratic equation, and so only quadratic interpolation was used for all further analysis.

9.4 Microphone Signal Analysis

When analysing the signal propagation delay between the loudspeaker and microphone, the extraneous delays within the system had to be taken into account and so the delay measured on Loop 2 during each test run was always subtracted from that measured using the microphone signal. As previously mentioned, in a 'final' system this loop would probably not be present and so instead an average of the loop delays would have to be used. However, this averaged delay was not used during the initial analysis so as not to introduce a further source of error—something which could easily be included in the final results.

9.4.1 Maximum Sample

As a preliminary check on the acquired test data, the propagation delays measured to the nearest sample were compared to the physical separation, as shown in figure 9.5. Apart from the first run of tests conducted at 1 m, these results indicated a very strong linear trend. A thorough investigation into the possible causes of the extraneous results at 1 m was conducted, and it was concluded that, due to the consistency of the error and the fact that it covered *all* measurements taken between positioning and moving the microphone, it must have been due to incorrect positioning of the microphone. Although no definite explanation as to how this happened could be determined, it was assumed that the microphone stand was at some point knocked after being positioned. In all further tests the position of the system components were measured both before and after the test run, thus ensuring such an error would be observed much earlier.

To avoid these extraneous measurements from incorrectly biasing further analysis,

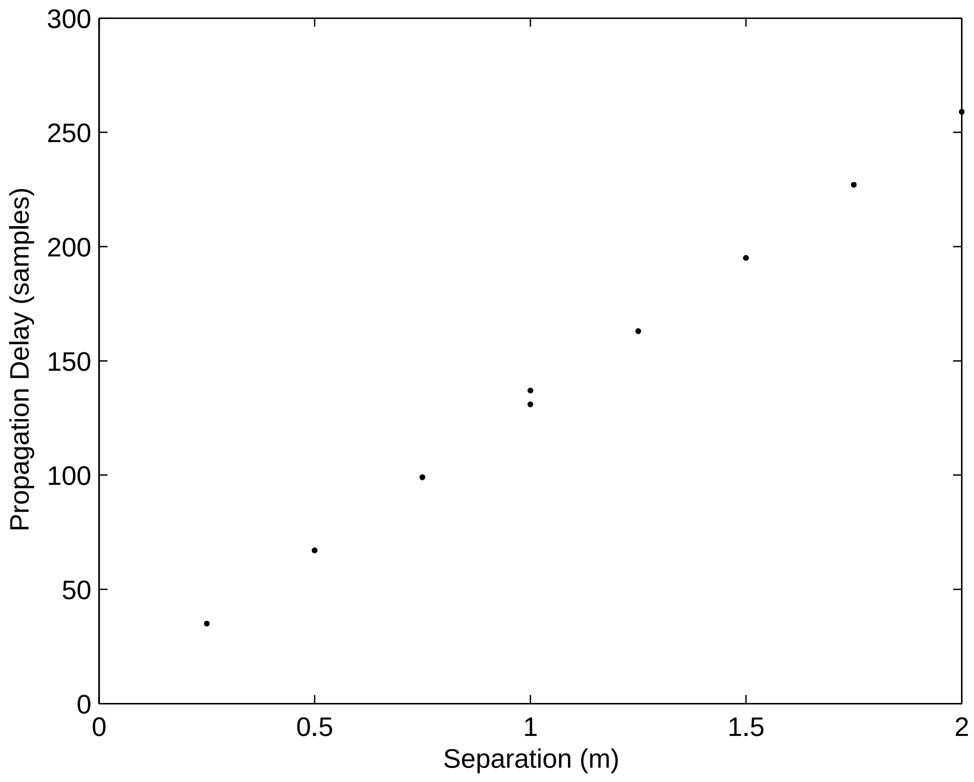


Figure 9.5: Comparison of propagation delays, based on the peak sample position, to physical separation for all excitation signals. All points, using all excitation signals, lie on the line $delay = 3 + 128 \times separation$, apart from those when the microphone was positioned at 1 m for the first time, which instead all have a delay of 137 samples.

they were excluded and replaced by measurements from a repeated test run at 1 m. Although this was conducted after some time had passed, and therefore the room conditions had changed, this was seen to be a suitable approach to allow analysis to continue provided that, should these results appear anomalous, this fact was recalled. To avoid any potential problems due to the removal of these results, ideally all tests should have been repeated. However, by initially analysing the original data the necessity of this could be determined.

Using the peak sample positions, the propagation delay at all separations (in metres) was found to be given by

$$delay \;=\; 3 + 128 \times separation \text{ (samples)}$$

where the gradient is the number of samples per metre in the propagating sound wave, and the constant offset is the discrepancy between the position of the measurement reference points and the acoustic centres of the loudspeaker and microphone.

From (8.1) and (8.2), at a sample rate Fs (samples/s), an approximation to the number of samples per metre in a sound wave is given by

$$N = \frac{Fs}{(331.5 + 0.6T)} \text{ (samples/m).} \tag{9.3}$$

Therefore, at a sample rate of 44100 samples/s, when $T = 12.9$ °C, $N \approx 130$ samples/m whilst when $T = 17.3$ °C, $N \approx 129$ samples/m. This shows that even when measuring delay to the nearest sample, a change in temperature of just 4.4 °C should alter the gradient of the trend by 1 sample/m. Additionally, this indicates that, based on the results measured, the room was approximately 21.7 °C, which was warmer than the perceived room temperature. This discrepancy was not investigated further until after a more accurate analysis of the results had been conducted using quadratic interpolation.

9.4.2 Quadratic Interpolation

For each excitation signal, the physical separations were compared with the measured propagation delays, calculated using quadratic interpolation. The parameters for the trend lines obtained are given in table 9.1. It can be seen that the gradients (proportional to the reciprocal of the speed of sound) are within 0.011 % of each other whilst the offsets are all grouped to within 0.037 of a sample interval, or approximately 0.29 mm.

	Trend line	
Excitation Signal	**Offset (samples)**	**Gradient (samples/m)**
MLS	3.116	128.058
IRS	3.115	128.059
OATSP1	3.126	128.070
OATSP2	3.141	128.068
OATSP3	3.115	128.059
Sweep1	3.149	128.068
Sweep2	3.152	128.072

Table 9.1: Parameters for the trend lines comparing separation (m) and propagation delay (samples), using quadratic interpolation, for each excitation signal where $delay = \{\text{offset}\} + \{\text{gradient}\} \times separation$.

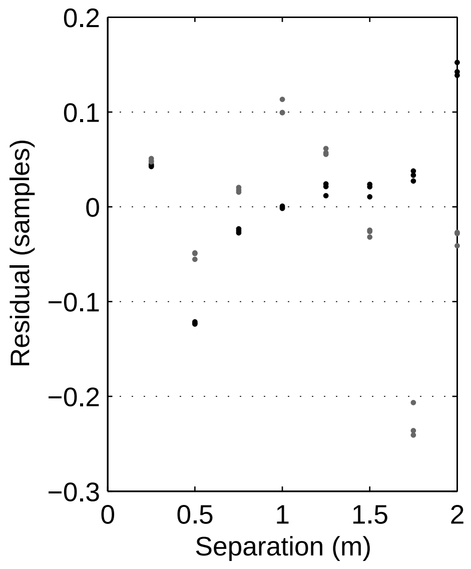
The trend's gradients indicate a room temperature of approximately 21.4 °C, which is very similar to that predicted previously. Other factors that also affect the speed of sound in air include the air pressure, humidity and CO_2 concentration[4], although temperature has by far the largest effect. Therefore, because none of these conditions were measured during the test runs, there was no way to determine the accuracy of these gradients based on a predicted speed of sound in the room. One potentially significant source of error would be experimental: if, for whatever reason, there was a consistent 0.1% error in the separation distance (i.e. 1 mm in a metre), then the room temperature approximation would change by almost 0.6 °C. A possible source of such an error is the difference between the measurement reference points and the acoustic centres of the loudspeaker and microphone. If the actual acoustic centres were positioned along the line of measurement, then the trend offsets indicate they would be approximately 24.5 mm further apart than the actual separation measured. However, at least for the loudspeaker this was not thought to be the case.

Assuming the loudspeaker's acoustic centre is positioned on its front face 0.05 m away from the measurement point, the errors in separation shown in table 9.2 occur. Although this is an overly-simplified scenario assuming the acoustic centre is the same at all frequencies, it does show the significance of the assumption made about the location of the acoustic centre. Due to other errors within the measurement process, it was not possible to predict the actual location of the loudspeaker's acoustic centre.

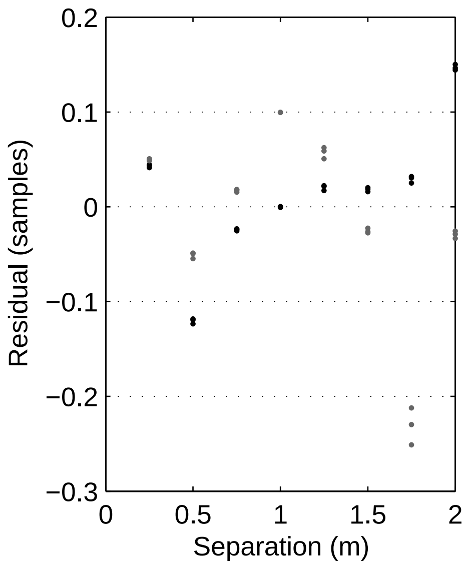
Measured Separation (m)	Actual Separation (m)	Error (%)
0.25	0.2550	1.98
0.50	0.5025	0.50
0.75	0.7517	0.22
1.00	1.0012	0.12
1.25	1.2510	0.08
1.50	1.5008	0.06
1.75	1.7507	0.04
2.00	2.0006	0.03

Table 9.2: Summary of separation measurement errors introduced by the acoustic centre of a loudspeaker being offset by 0.05 m perpendicular to the line of measurement.

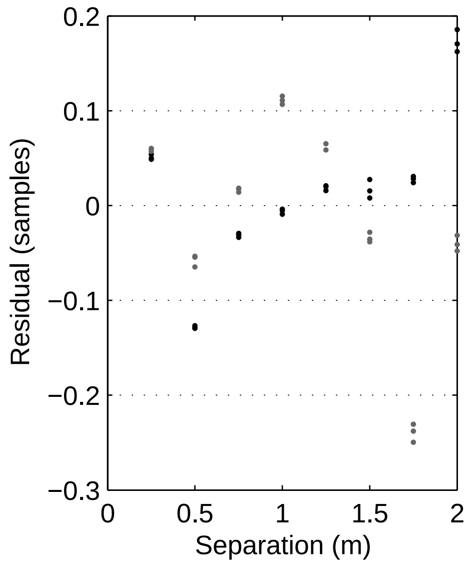
When analysing the residuals for each of the trend lines shown in table 9.1, of which four are shown in figure 9.6, it became clear that the accuracy of the propagation delay measurement was greater than that used to position the loudspeaker—at each separation there are two 'groups' of three points: one group for each time the microphone was placed in a particular position. (In some cases the points



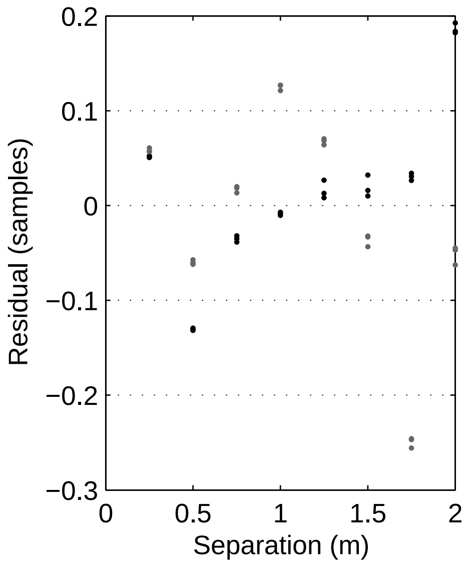
(a) MLS



(b) IRS



(c) OATSP1



(d) Sweep1

Figure 9.6: Residuals for propagation delay measurements, using quadratic interpolation, for four different excitation signals when compared to their individual trend lines as given in table 9.1. Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

are grouped so tightly that they only appear as one.) The largest separation of the three points in any of these groups was found to be 0.039 of a sample interval, occurring with 1.75 m separation using the IRS excitation signal, shown in figure 9.6(b). When comparing all excitation signals for a specific run at a particular separation, this range of values was found to increase up to 0.0581 of a sample interval, occurring at 2 m. This indicates that despite each propagation delay being calculated as the difference in delay between two signals of the same type, the signal type being used was still affecting the propagation delay measured.

To attempt to determine the source of this variation between signal types, a single trend line was fitted through the data points for all excitation signals. The resulting trend line had the equation

$$delay \;=\; 3.130 + 128.065 \times separation \text{ (samples)} \tag{9.4}$$

producing the residues shown in figure 9.7. In this case, the maximum range of any 'group' of points had increased to 0.121 of a sample interval (first group of measurements made at 2 m), indicating that the use of separate trend lines for each excitation signal had 'hidden' the true extent of the variations when comparing the different signal types.

To determine the significance of the variations in the delays measured on Loop 2 (a range of 0.0658 of a sample interval across all signal types and tests), these were excluded from the propagation delay calculations. As such, the trend between separation and delay measured directly on the microphone signal was determined, and found to have the equation

$$delay \;=\; 587.931 + 128.065 \times separation \text{ (samples)}$$

where, in this case, *delay* also includes all delays within the other system components such as the computer. Notably the gradient of this trend line remained the same as (9.4) and, more significantly, the ranges of each group of residual values either remaind approximately the same, or reduced. In some instances this reduction was significant, such as in the first group of measurements made at 2 m where the range of values changed to 0.0689 of a sample interval. The residuals in this case are shown in figure 9.8.

From this it was concluded that the inclusion of the Loop 2 delay within the propagation delay calculations appeared to *increase* the variations in values measured between signal types, as well as negatively affecting the repeatability. However, without a much more accurate method of positioning the microphone it was realised that it would not be possible to determine which of these approaches and excitation signals actually produced the most accurate results, and also whether these 'improvements' in repeatability were occurring at the expense of accuracy. Additionally if, as previously mentioned, the air temperature, humidity, pressure and, where possible, CO_2 concentration were measured then an accurate predication of the speed of sound could be compared to that determined from the results.

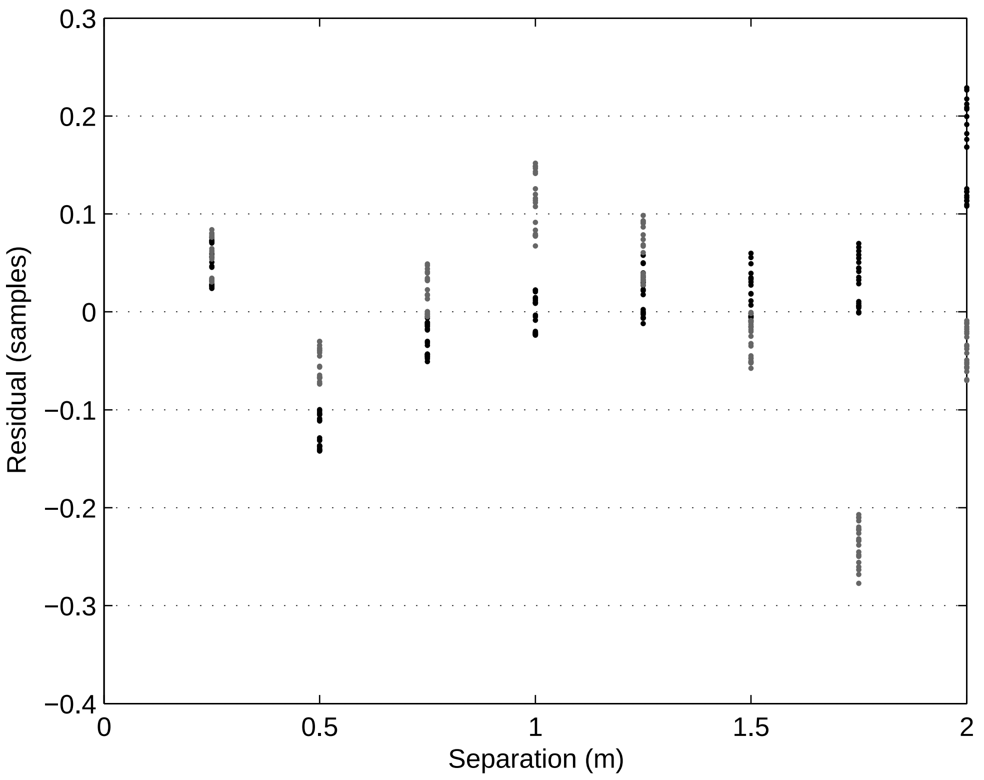


Figure 9.7: Residuals for propagation delay measurements, using quadratic interpolation, for all excitation signals when compared to the single 'common' trend line $delay = 3.130 + 128.065 \times separation$. Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

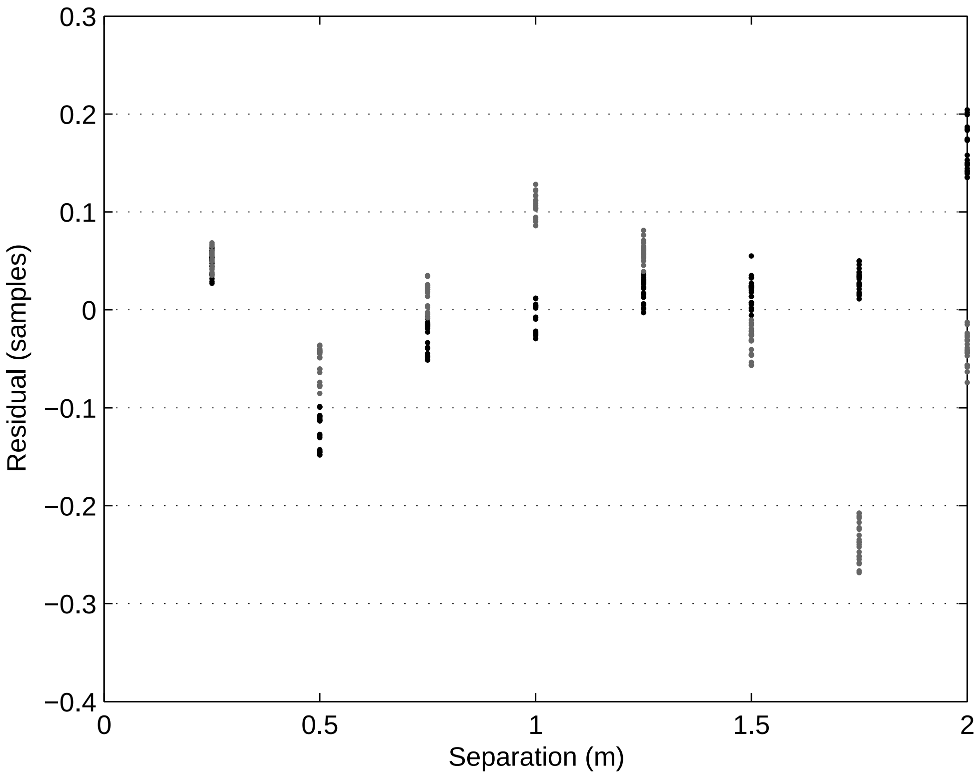


Figure 9.8: Residuals for delay measurements, using quadratic interpolation, for all excitation signals when compared to the single 'common' trend line $delay = 587.931 + 128.065 \times separation$. This includes all delays within the system components, such as the computer. Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

Chapter 10

SoundField Microphone Measurements

One method of angle measurement to be investigated was that using intensity differences in the B-Format signals produced by a SoundField microphone.

10.1 Angle Measurement Theory

Assuming a B-Format signal contains sound originating only from a point source in space, the location of the source can be determined using two different approaches which ultimately return the same results. The first of these comes from the standard B-Format encoding equations

$$\begin{aligned} W &= \frac{S}{\sqrt{2}} \\ X &= S\cos\alpha\cos\beta \\ Y &= S\sin\alpha\cos\beta \\ Z &= S\sin\beta \end{aligned}$$

where S is the mono source signal, and α and β are the azimuth and elevation of the required source location. The equations for X, Y, and Z are the same as those to convert between spherical and cartesian coordinates, where S would be the distance from the origin, and thus the x, y, and z coordinates of the sound source are proportional to the received signals X, Y, and Z. Comparing these received signals with W allows the sign of x, y and z to be determined, and hence the azimuth and elevation of the loudspeaker[11].

An alternative approach to determining the location of a sound source uses the equations to rotate and tumble a soundfield given in appendix F. From (F.1) the rotated signal X' is given by

$$X' = X\cos\theta - Y\sin\theta \tag{10.1}$$

where θ is the angle of rotation. Assuming the sound source does not lie on the z-axis, when the soundfield is rotated X' will vary with maximum and minimum occurring when the rotated sound source is at its closest and furthest points from the positive x-axis. Differentiating (10.1) with respect to θ, these stationary points are found to occur when

$$X\sin\theta = -Y\cos\theta$$

from which

$$\theta = -\arctan\left(\frac{Y}{X}\right).$$

As θ is the angle through which the sound source must be rotated to obtain a maximum/minimum, the azimuth of the original sound source, α, must be either $-\theta$, if the sign of X and W are the same, or $(\pi - \theta)$ otherwise.

To determine the elevation of the original source a similar method is used, although this time starting with an equation for X' which includes an initial rotation by $-\alpha$ before tumbling by ψ:

$$X' = X\cos\alpha\cos\psi + Y\sin\alpha\cos\psi - Z\sin\psi.$$

Differentiating with respect to ψ, the stationary points are found to be where

$$\psi = -\arctan\left(\frac{Z}{X\cos\alpha + Y\sin\alpha}\right)$$

which can be rewritten as

$$\psi \;=\; -\arctan\left(\frac{Z}{\sqrt{X^2+Y^2}}\right) \tag{10.2}$$

provided α is calculated as described above. In (10.2) ψ is the angle through which the sound source must be tumbled to position it along the x-axis, and therefore the elevation of the original sound source, β, must be either $-\psi$ if W is positive or $+\psi$ otherwise.

These equations to calculate the source azimuth and elevation are the same as those to convert from the cartesian coordinates (X, Y, Z) to spherical coordinates, thus confirming that both approaches return the same results.

To use these equations, the amplitude of the direct sound from the loudspeaker must be determined for each of the B-Format signals. This can be achieved by finding the magnitude of the direct sound peak within the room's IR for each signal, and so a set of tests were designed to investigate this using the same excitation signals as used previously.

10.2 Test Setup

To test the accuracy to which angles could be measured, either the loudspeaker would have to be moved around the microphone or the microphone would have to be rotated. With the SoundField microphone to be used (see appendix E for details) the front is indicated by a light shining through a small hole in the microphone body. Although accurate enough to position the microphone when making recordings, this was not thought to be accurate enough to allow manual rotation of the microphone during these tests. An alternative would have been to use a rotating table, such as those used to measure the polar pattern of microphones and loudspeakers. However, as this was not available the loudspeaker would have to be moved instead.

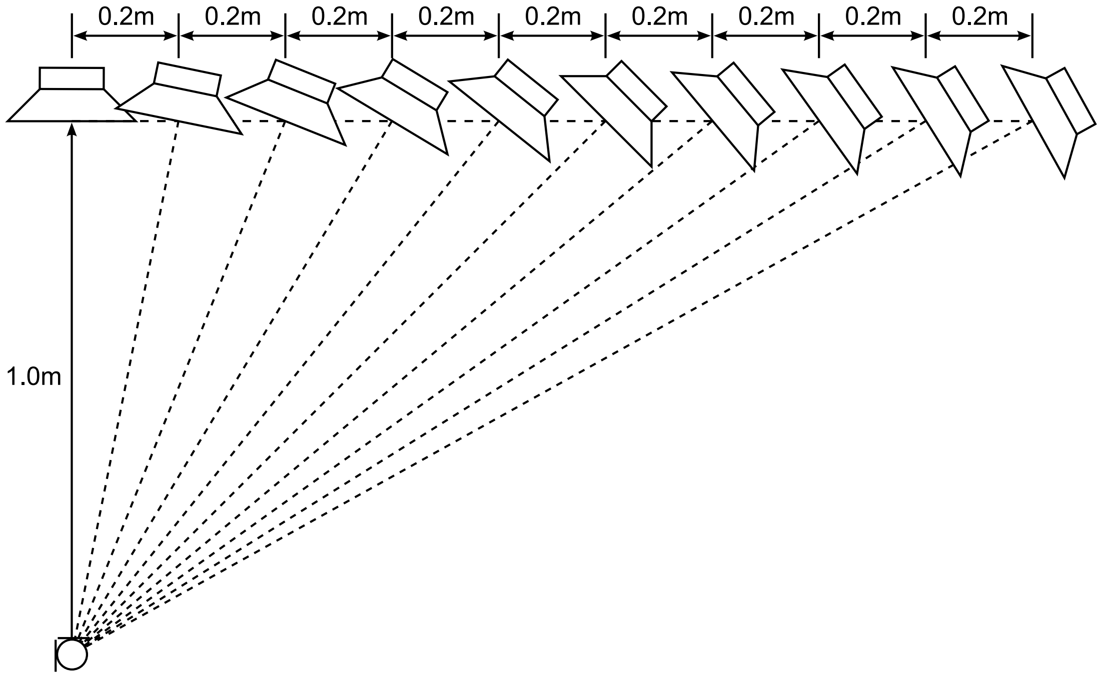


Figure 10.1: Test positions of the loudspeaker used to determine the performance achievable when measuring angles using a SoundField microphone. Dashed lines show the right-angled triangles used to calculate the angle of the loudspeaker relative to the microphone.

Two different methods of moving the loudspeaker were considered: in an arc around the microphone and in a straight line offset from the microphone. The first of these would maintain the same separation during all tests, and so the angle would be the only major factor changing. However, this required an accurate method of positioning the loudspeaker at each test angle which was difficult using the equipment available. The alternative method, however, was expected to be less prone to positioning errors, because the loudspeaker could be moved along a beam, using trigonometry to calculate the test angle. Although the use of a beam was rejected when using a single microphone, in this case it was not expected to introduce significant errors—any sound radiated from the beam would be minimal compared to that from the loudspeaker. To avoid introducing a third variable factor (in addition to separation and microphone angle) at all test positions along the beam the loudspeaker was orientated to face the microphone, as shown in figure 10.1. This was implemented by sight using alignment marks on the front and rear of the top of the loudspeaker.

This test was designed to limit the analysis to the horizontal plane so that initially the accuracy of azimuth measurements on their own could be tested. Following this, further tests including elevation could then be conducted to determine the accuracy of measurement for a loudspeaker anywhere within the room. The measurement positions used, moving along the beam from 0.0 m to 1.8 m in 0.2 m intervals, allowed a wide range of angles to be tested (from 0° to ∼-61°) as well as testing the response with changes in angle of under 3°. (The microphone was positioned at the end of the beam shown due to the available space within the room and therefore, according to the convention used with Ambisonics, all positions corresponded to negative azimuth angles.) Although specific angles could have been measured, such as even steps of 6°, this would not have introduced any significant advantages but would have increased the complexity of the test setup.

All measurements were made relative to the same reference point on the loudspeaker as for the propagation delay tests (the centre of its front face) whilst the centre of the top of the SoundField microphone was used for all horizontal measurements. Vertically the SoundField microphone was positioned so that the middle of the microphone capsules, visible through the protective grill, were aligned with the middle of the loudspeaker. As with the loudspeaker reference point, it could not be guaranteed that these points aligned with the acoustic centre of the microphone.

The microphone and beam along which the loudspeaker was to be positioned were located in the middle of the room to ensure the direct sound could easily be distinguished from any room reflections. The beam was aligned relative to the microphone by measuring the direct microphone-loudspeaker separation with the loudspeaker at three positions along the beam: 0.0 m, 1.0 m and 1.6 m. Additionally, every time the loudspeaker was moved to a new test position the separation was measured to confirm that the it had been positioned correctly. Due to the inaccurate method of determining the 'front' of the microphone, it was expected that it would not be orientated pointing directly at the first loudspeaker position. Within the results, however, any inaccuracy with the rotation of the microphone would appear as a constant angle offset although inaccuracies in the position and elevation would appear as non-linear errors.

The wiring diagram for tests using the SoundField microphone is shown in figure 10.2. Loop 1 was included to monitor the delays within the computer whilst Loop 2 would additionally measure the delay within the first stages of the amplifier, using the sound desk to amplify this signal to a suitable level. Based on the results when using a single microphone, this second loop was not expected to be used during the results analysis. However, it was still included so that should any problems arise during the tests, the source might be more easily determined and therefore the extra delay due to the signal passing through the sound desk was deemed acceptable.

As with the propagation delay tests, the effect of many factors that would have to be considered when implementing an automatic system were not evaluated. However through implementing a basic system and solving any problems that might occur, the system could then be extended to cover more of these factors as required.

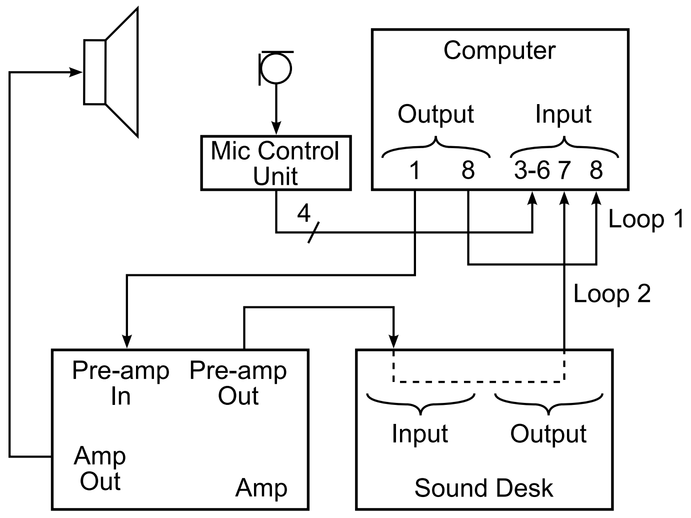


Figure 10.2: Wiring diagram for all tests conducted using a SoundField microphone. Loop 1 allows the delay within the computer to be measured whilst Loop 2 additionally measures the delay through the amplifier and sound desk. The specification for each component can be found in appendix E.

10.3 Analysis

Initially the IR for every channel of every recording was calculated. Based on the W channel being that from an omni-directional microphone, the position of the peak within this channel's IR was calculated using quadratic interpolation, as described in section 9.3.2. At this position the magnitude and sign of the X, Y and Z channel IRs were then calculated, again using quadratic interpolation. Relying on all four B-Format signals being measured at a coincident point in space, this process would produce the four values required to calculate the azimuth and elevation of the source, as described above. By using the omni-directional channel to determine the position of the peak, the accuracy of this technique would be the same regardless of the position of the sound source.

The first comparison between test angle (where the loudspeaker was positioned) and measured angle using this procedure is shown in figure 10.3. It can be seen that at some test angles there are very large discrepancies between the values measured.

However, on further inspection it can also be seen that the data contains two strong patterns: first, starting at 0° the range of angles measured diverges with increasing angle magnitude for the first four test angles; and second, this pattern appears approximately rotationally symmetrical about the points measured with a test angle of -39°. Taken together these indicate that the range of measured values increases the further the test angle is from either microphone axis. The one exception to this is when the test angle is close to mid-way between these axes, and hence X and Y are of similar magnitude, in which case the range seems to significantly reduce. This should therefore have occurred with a test angle of 45°, although a slight rotational offset in the microphones position would account for this.

To determine if these significant inaccuracies occurred for only some excitation signals, a per-excitation signal comparison was made between test and measured angles, of which four are shown in figure 10.4. From this it was found that the excitation signal used had a significant impact on the angle measured: the MLS and IRS excitation signal results had very similar performance measuring similar, albeit wrong, angles for each test; the two Sweep excitation signal results also had similar performance to each other yet were significantly different from the MLS and IRS results; finally the OATSP excitation signal results varied from being similar to the Sweep's with OATSP1 to being similar to the MLS and IRS with OATSP3. However, despite the inaccuracies in the measured angle, in the majority of cases the three angle measurements made during each test were found to produce similar results indicating that the errors were systematic rather than random.

From a comparison of the IR on all four B-Format signals, such as that shown in figure 10.5, it was found that the peak in the four channels did not occur simultaneously. Additionally, the signal amplitude observed on the Z channel was far larger than that expected considering all measurements were being made on a horizontal plane. Although microphone misplacement could have introduced a small signal on the Z channel, it should not have been this large.

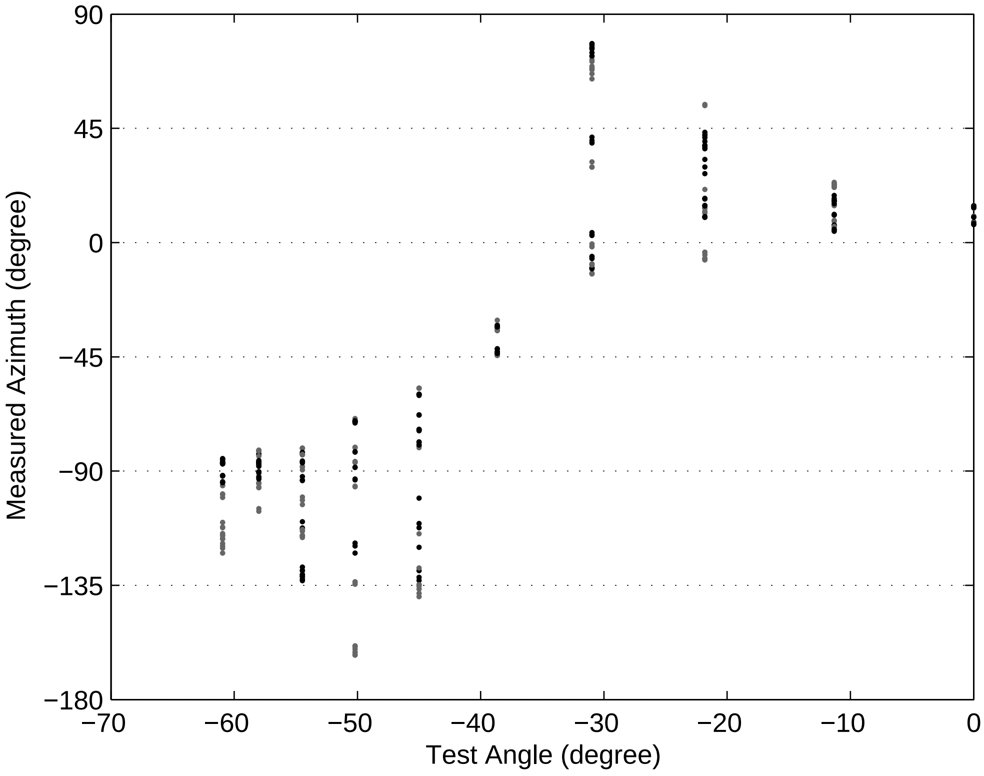
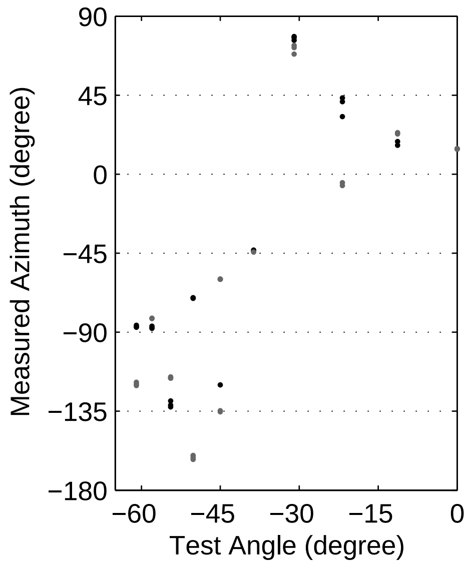
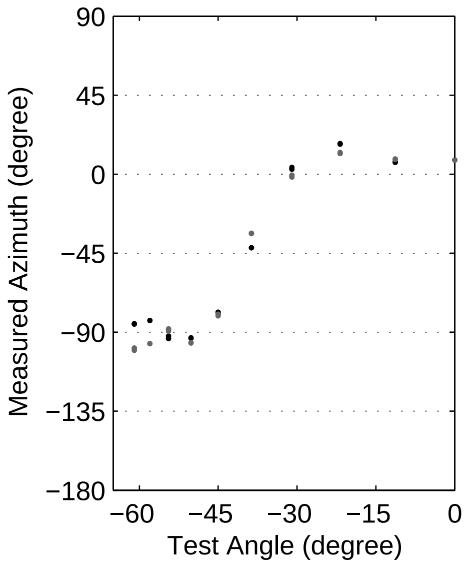


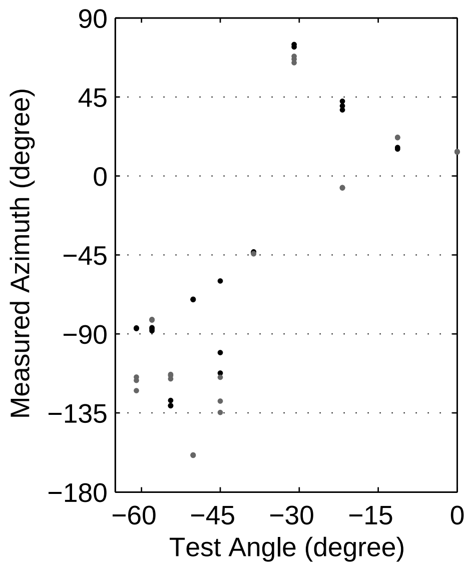
Figure 10.3: Comparison for all excitation signals between test angle and measured azimuth using quadratic interpolation to calculate the peak IR signal values. Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.



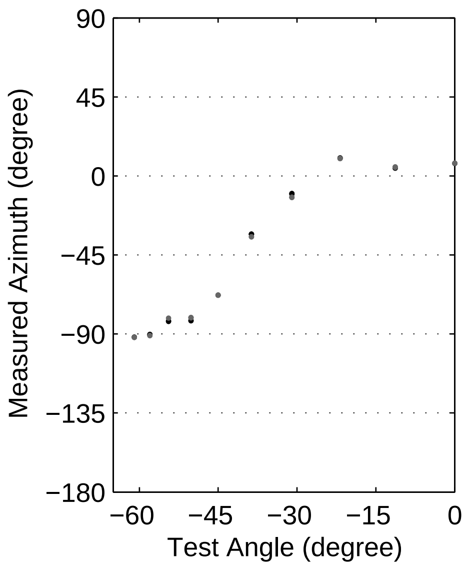
(a) MLS



(b) OATSP1



(c) OATSP3



(d) Sweep1

Figure 10.4: Comparison between test angle and measured azimuth using quadratic interpolation to calculate the peak IR signal values. Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

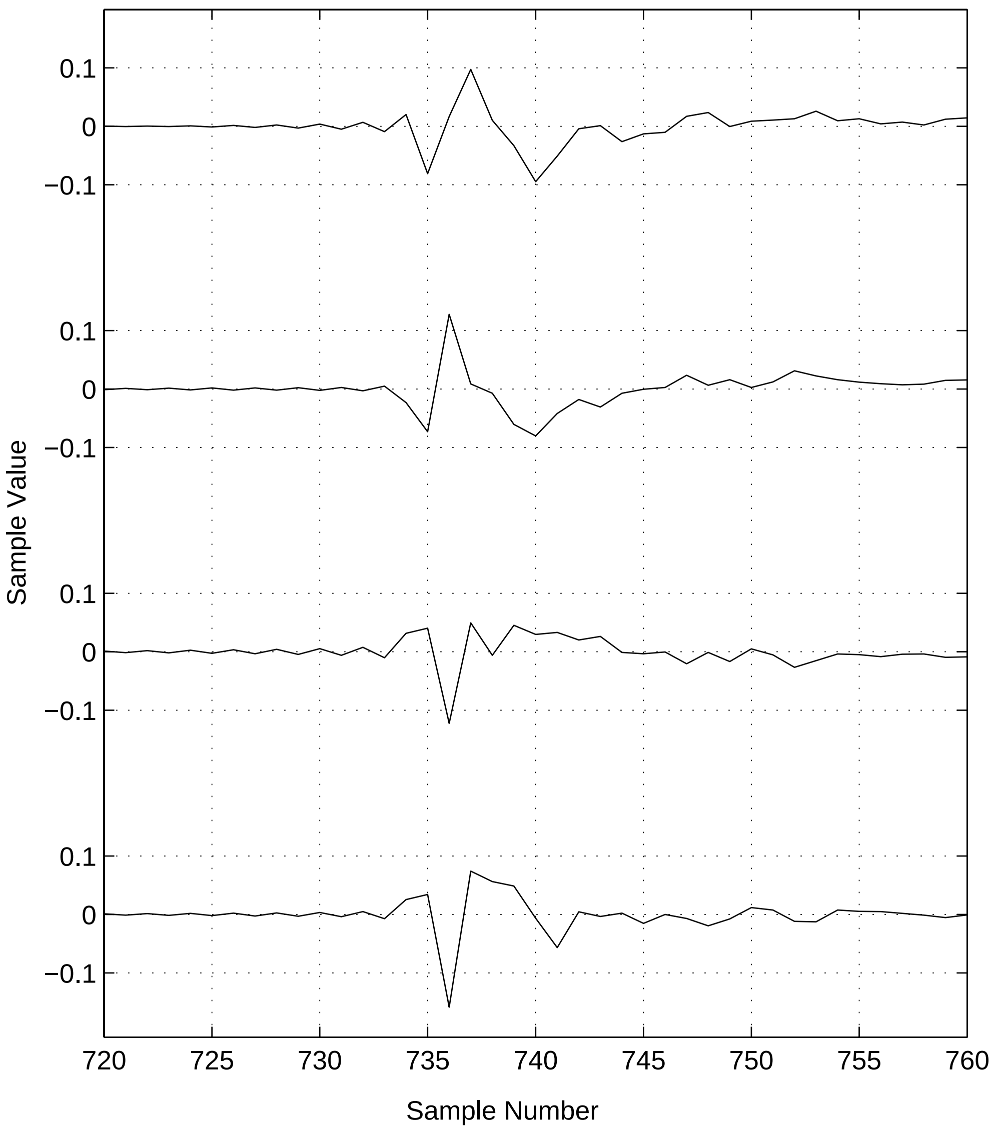
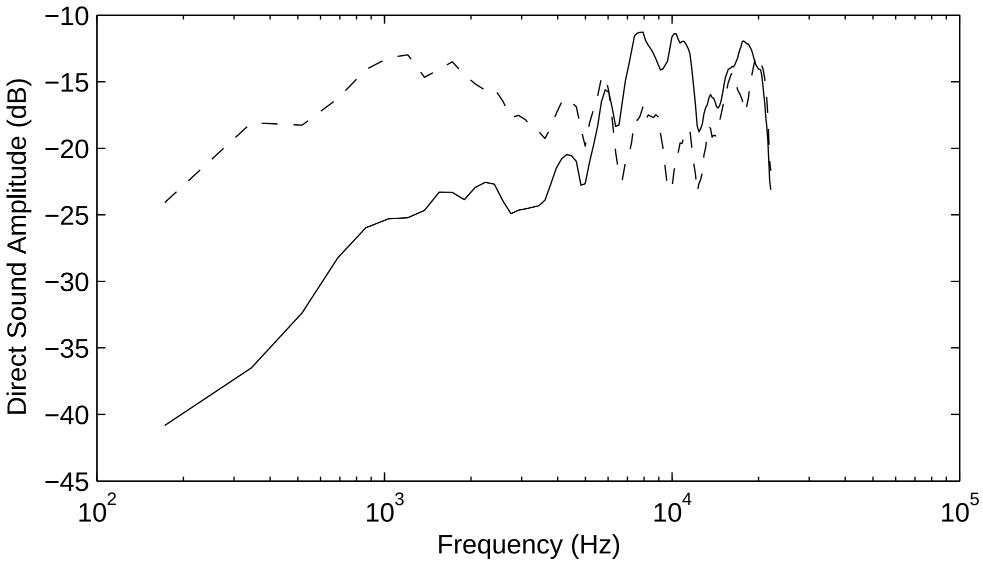


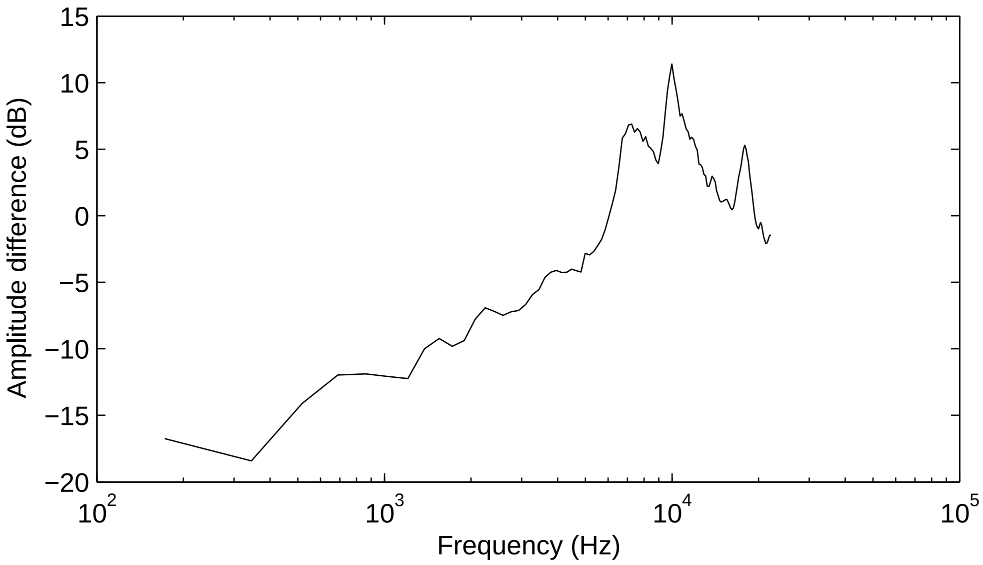
Figure 10.5: Direct sound section of an impulse response measured using an MLS excitation signal with loudspeaker positioned at 31°. Signals are, from top to bottom, W, X, Y and Z.

These two factors, the temporal misalignment and the large Z signal, indicated that the microphone was not generating the B-Format signals expected. A visual inspection of the recordings made using both the MLS and IRS excitation signals showed that the start of the Z channel recording had a different spectrum from the X and Y signals. Comparing the frequency spectra of the direct sound section of each channel's IR, such as that shown in figure 10.6(a), confirmed this. In theory the spectrum for this part of the IR should be the same for all channels, with a constant difference across all frequencies dependant on the position of the loudspeaker. However, as can be seen in figure 10.6(b), this was not the case and instead the difference between the Z channel and the X and Y channels reduced with increasing frequency until, at higher frequencies, the Z channels was largest. This explained the reason for the incorrect angle measurements, but did not explain what was causing them.

The B-Format signals generated by the SoundField microphone are created from four microphone capsules in a tetrahedral array[39]. Therefore, although it is claimed that the SoundField microphone presents a "single point source" this might not be the case, and especially not at high frequencies where the distance between the capsules is significant compared to that of the wavelength of the sound. Additionally, within the microphone control unit the conversion of the microphone capsule signals to the B-Format signals is calibrated for a particular microphone and so if this calibration has drifted then incorrect results would be expected. However, without repeating these tests with other SoundField microphones, including the same and different models to that used for these tests, it was not possible to determine if these errors were due to incorrect calibration, a limitation of the particular model of microphone used or a limitation of all SoundField microphones.



(a) X (dashed line) and Z (solid line).



(b) Difference ($Z - X$).

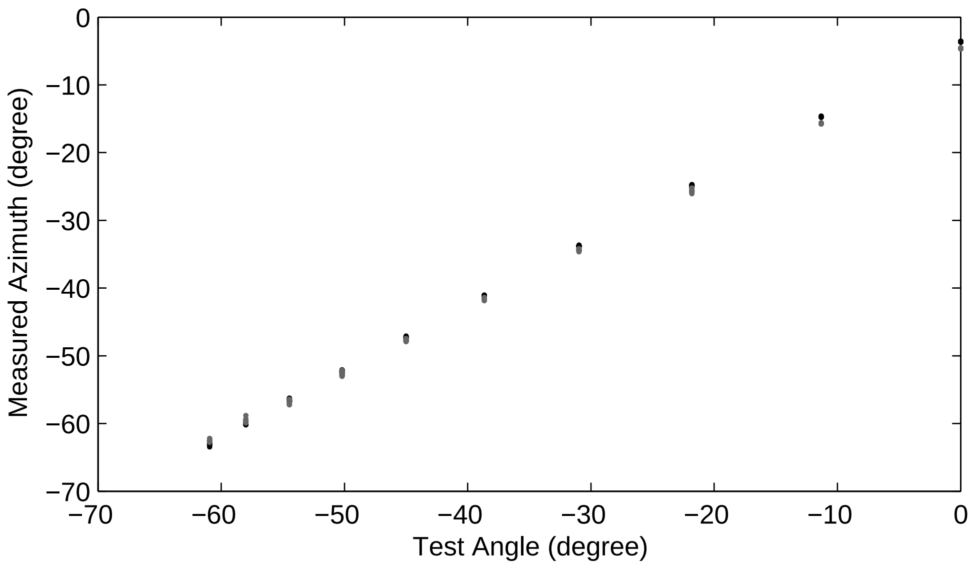
Figure 10.6: Frequency content of the direct sound section of an impulse response measured using an MLS excitation signal with loudspeaker positioned at 45°.

Although not thoroughly investigated, the possibility of low-pass filtering the B-Format signals to improve the angle measurement was considered. From figure 10.6(b) a cut-off frequency of 1500 Hz was chosen to include as much of the original signals as possible without including those frequencies where the gain of the Z channel noticeably started to rise. Filtering all signals using a 51 tap Hamming windowed Finite Impulse Response (FIR) filter prior to calculating the azimuth and elevation, the results shown in figures 10.7 and 10.8 were obtained. From these it can be seen that filtering significantly improves both the accuracy and repeatability of the results, although there are still occasional outlying points in both cases. The trend line for the measured azimuth using all excitation signals was found to be

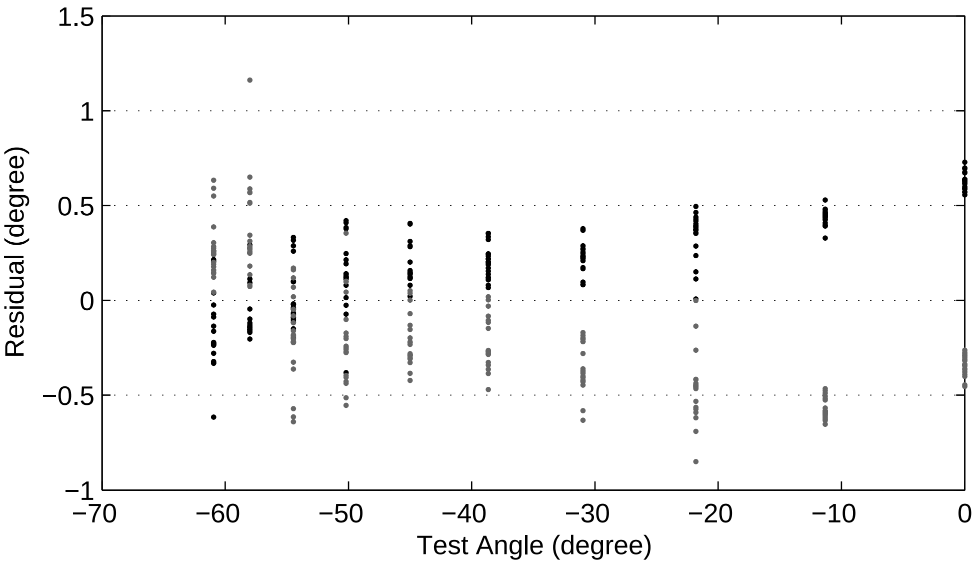
$$measuredAzimuth = 0.961 \times testAngle - 4.27$$

indicating that the front of the microphone was orientated approximately 4° away from the first loudspeaker position. From inspection of the measured elevation, shown in figure 10.8, there is evidence to suggest that the microphone was not positioned perfectly vertically but instead had its top angled slightly away from the centre of the beam. Therefore this could also explain why the gradient of the measured azimuth was not unity, as it should be in theory.

From the residuals between the trend line and the measured azimuth it can be seen that the grouping of points between first and second test runs still exists, although the density of this grouping appears to be influenced by either the microphone-loudspeaker separation or the test angle. Additionally these two groups are always centred about two different points, indicating that the accuracy of angle measurement achievable is either close to or better than that of the test setup. Finally, from these preliminary tests using low-pass filtering it can be concluded that an azimuth measurement, for a source with 0° elevation, appears to be feasible with an accuracy of greater than ±1°. However, further testing should be conducted to determine the optimum filter as well as investigating the accuracy of elevation measurement and the cause of the greater variation in repeat measurements at some angles when compared to others. Furthermore, the use of multiple points within



(a) Measured Azimuth.



(b) Residuals from common trend line $measuredAzimuth = 0.961 \times testAngle - 4.27$

Figure 10.7: Comparison for all excitation signals between test angle and measured azimuth using low-pass filtered versions of all B-Format signals (cut-off frequency: 1500 Hz). Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

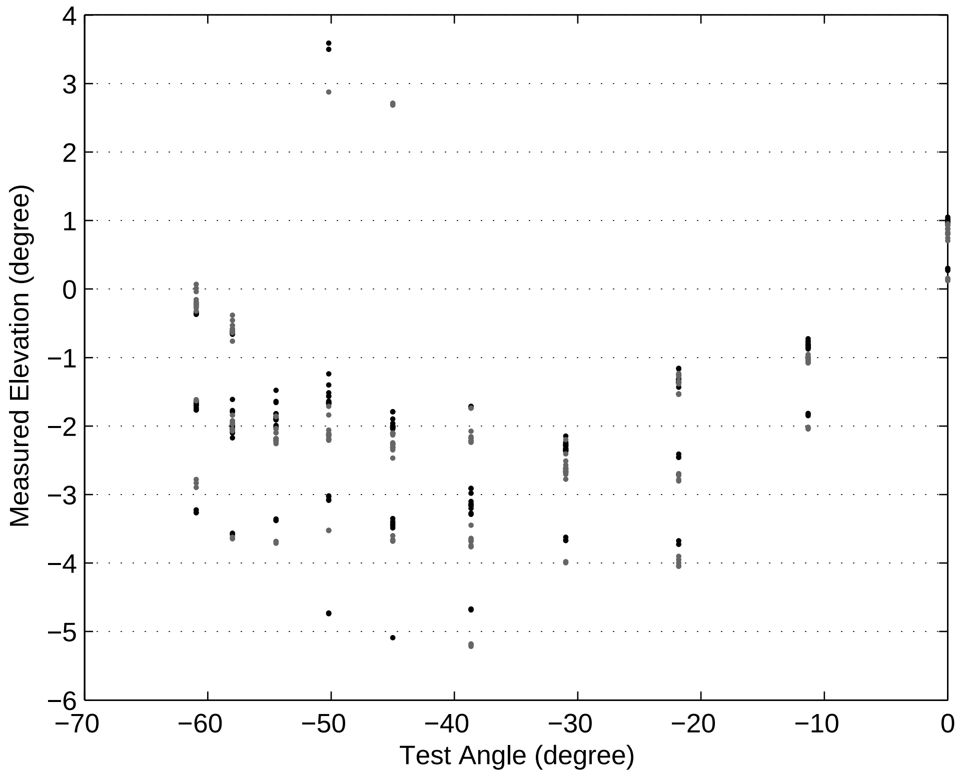


Figure 10.8: Comparison for all excitation signals between test angle and measured elevation using low-pass filtered versions of all B-Format signals (cut-off frequency: 1500 Hz). Black points mark the first group of measurements made at each separation whilst grey points mark the second group of measurements.

the direct sound section of the filtered IRs should be investigated to determine if there is any improvement over using only a single point.

To accurately measure the separation between loudspeaker and microphone, the high frequency components within the excitation signals are necessary to obtain a sharp direct sound peak. Due to time constraints, however, it was not possible to investigate the effect of the SoundField microphone's high frequency performance on such measurements, or investigate the impact of low-pass filtering.

Chapter 11

Multiple Microphone Measurements

If the arrival time of a sound is measured at enough different points in space, it is possible to determine the location of the original sound source. This chapter describes different approaches that can be used dependant on the timing information available and then gives an overview of the testing planned to investigate their relative performance.

11.1 Theory

From the propagation delay measurements using a single omni-directional microphone in chapter 9, it has been shown that the distance between a loudspeaker and microphone can be measured to a sub-sample accuracy. From just this single measurement the loudspeaker could be positioned anywhere on a sphere around the microphone. By making a simultaneous measurement with a second microphone, then a second sphere could be determined. If the distance between the two microphones is know it is possible to calculate where these two spheres intersect, thus reducing the possible set of loudspeaker positions to a circle. By using a third microphone this can be reduced to a set of two points and a fourth microphone can reduce this to a single point.

To ensure that a single point is found there are, however, certain limitations on the arrangement of the microphones. For example, if all four microphones are positioned within a single plane then there is no way to determine which of two points, one either side of the plane, is correct. Additionally, so far it has been assumed that all distance measurements are exact. Although accuracy to the sub-sample level means that the errors in distance are small, they do still exist and therefore care must be taken to avoid amplifying them. The easiest method of reducing the effect of these errors is to increase the separation between microphones—with two microphones very close together even a small change in the value measured by one would significantly change the position of the intersection between the two spheres.

One implementation of this technique positions the four microphones on rectangular axes: one at the origin and one on each axis equal distances, d, from the origin[45]. In such a setup, it can be shown that the coordinates of the sound source (x, y, z) are given by

$$\begin{aligned} x &= \frac{(d^2 + r_o^2 - r_x^2)}{2d}, \\ y &= \frac{(d^2 + r_o^2 - r_y^2)}{2d}, \\ z &= \frac{(d^2 + r_o^2 - r_z^2)}{2d}, \end{aligned}$$

where r_o, r_x, r_y and r_z are the radii of the spheres around the microphones positioned at the origin and on the x, y and z axes respectively. This has the advantage of greatly simplifying the process of determining the location of the loudspeaker compared to some approaches, although it also has the potential to produce less accurate results because it does not make full use of the available timing information. One method that does utilise this extra timing information is a least squares approach, minimising the differences between the measured and predicted loudspeaker-microphone separations.

Instead of using the direct time-of-flight to calculate the loudspeaker position, the differences in sound arrival time between the microphones can also be used. This has the advantage of not needing to know exactly when the sound was transmitted and so is far more suited to scenarios when the time delay through the complete signal path is unknown to the required level of accuracy. From these differences in time the differences in distance between the loudspeaker and each microphone can be determined. With a single pair of microphones this results in a hyperboloid on which the loudspeaker must be positioned. Through the use of more microphones, the intersection of the different hyperboloids results in a single position for the loudspeaker. The equations to determine the loudspeaker position from these differences in distance are, however, highly non-linear and so to solve them many different algorithms have been proposed[18].

An alternative arrangement of microphones that has been used is to place them at the apexes of a regular tetrahedron. This is used by the Trinnov Optimizer to determine the location of loudspeakers as well as measure the loudspeaker characteristics and loudspeaker/room interaction[40]. The quoted precision of this system is “better than 1 cm/2°” when measuring the distance, azimuth and elevation of loudspeakers, thus indicating that angle measurements using this microphone layout are likely to be of similar accuracy to those achievable using a SoundField microphone.

11.2 Planned Testing

Due to the need to revise the plan for this project, as described in chapter 14, the time initially allocated to investigating the use of multiple microphones had to be reallocated. For this reason no actual tests were implemented, although this section will briefly describe the tests that were planned to be undertaken had time allowed.

To enable a direct comparison between the use of multiple microphones and the

SoundField microphone, the initial tests were to be conducted using exactly the same methodology as for the SoundField microphone, described in section 10.2. Additionally a near identical wiring configuration to that shown in figure 10.2 would have been used, although using the sound desk instead of the dedicated SoundField microphone control unit to amplify the four microphone signals.

Two different arrangements of the microphones were planned, testing both the ‘rectangular axes’ configuration and the regular tetrahedron. Following the tests this would then allow the performance of different time of flight and time difference of arrival algorithms to be compared. However, the implementation of such algorithms was expected to take a significant amount of time and so the algorithms tested would have had to be chosen based on the time available.

Chapter 12

Ambisonics Decoding

For an Ambisonic system to be of any use, a method of generating the loudspeaker feeds from the B-Format signals, as described in section 2.2, is required. Different methods of achieving this have been proposed, although a common underlying concept is to use virtual microphone responses, as described in appendix F, for microphones pointing directly at each of the loudspeakers. This can be implemented using a 'decoder matrix', although the configuration of the virtual microphones to generate this matrix differs between approaches.

To evaluate the performance of different decoding algorithms, the 'Ambisonic equations', which are a formal representation of certain "psychoacoustic criteria" associated with sound localisation, can be used[14]. Two different theories of sound localisation to which these apply are the "Makita" theory, also known as the 'velocity vector' theory, and the 'energy vector' theory. The first is more important at low frequencies, where phase difference between the human ears is used to determine a sound's location, and provides a measure of the apparent direction of a sound based on the gains of all the loudspeakers for a known virtual source location. The second is more important at higher frequencies, where the "directional behaviour of the energy field around the listener" is important, and also provides a measure of the apparent direction of a sound source. The similarity in angle between these two vectors indicates the ‘sharpness’ of a phantom image, whilst the vector lengths indicate the stability of the sound location as the listener moves their head[12, 14].

12.1 Decoder Types

The simplest form of decoder is that in which the directivity of all virtual microphones is the same, with no filtering applied to either the original B-Format signals or the resulting loudspeaker feeds. The directivity used can then either be determined by personal preference or optimised using the theories above. Extending this, shelving filters, with different gains for high and low frequencies, can be applied to each of the B-Format signals prior to calculating the virtual microphone responses[16]. This is equivalent to using two virtual microphones to calculate each loudspeaker feed, with different directivities for high and low frequencies. Provided the loudspeakers are placed in a regular array (all loudspeakers positioned evenly on a circle around the listener), and the same directivities are used for all loudspeakers then the apparent sound direction is the same as the encoded direction. Furthermore, the perceived volume of a virtual sound source is the same regardless of the direction of the source[42].

With irregular arrays, however, this approach cannot be used without resulting in excessive decoding artefacts[44]. A range of decoding schemes that have been proposed include those for three or more “pairs of diametrically opposite loudspeakers”[13] and those for “left/right symmetric” configurations with at least two pairs of loudspeakers plus at least one further loudspeaker[14], similar to the common 5 loudspeaker layout. Instead of using shelving filters, the last of these used a “phase compensated band-splitting filter arrangement” to allow a completely different decoding matrix to be used at high and low frequencies, before summing the outputs to generate each loudspeaker feed.

An alternative method of determining the required decoding parameters for similar loudspeaker arrangements is given in [42, 44]. In these, a Tabu search (a memory based heuristic search method) is used to optimise a set of 'fitness' equations based on the velocity and energy vectors described above. Multiple potential sets of decoding parameters are then produced, with subjective listening tests required to determine the 'best' set. Although the implementation given is for the common 5 loudspeaker layout, "the methodology is applicable to any configuration"[44].

12.2 Decoder Implementation

Implementing a decoder for different loudspeaker arrangements can be divided into two stages. Initially the decoding parameters for the particular arrangement must be determined and then the B-Format signal must be decoded using these parameters. In a completely automatic system, the first of these would have to take the positional information for all of the loudspeakers, which could be anywhere in 3-dimensional space, and produce an optimal set of decoding parameters. Although, mathematically, the Tabu search algorithm in [42, 44] could be extended to cover this far more generic case, this was beyond the scope of the project and so an alternative was sought.

Investigation into other methods used to determine the decoding parameters for irregular loudspeaker arrays found that certain assumptions about the loudspeaker positions had always been made, such as left/right symmetry. Therefore, using these methods when the assumptions were not met was expected to introduce just as many decoding artefacts as using the methods for regular arrays when the arrangement is not regular. So, a decoder for regular arrays was implemented such that, provided the loudspeaker arrangement was close to being regular, a completely functioning system could be implemented.

In this case, determining the decoding parameters from each loudspeakers position requires minimal processing and so can be easily implemented alongside the B-Format signal decoding. For a loudspeaker at azimuth, θ, and elevation, ϕ, its feed can be calculated using the virtual microphone equation (F.4), where

$$\begin{aligned} r_x &= \cos\theta\cos\phi \\ r_y &= \sin\theta\cos\phi \ . \\ r_z &= \sin\phi \end{aligned}$$

To implement different decoding for high and low frequencies either shelf filtering with a single decoder or band-splitting with two decoders can be used. However, provided the directivity, D, of the virtual microphone can be specified, the same decoder block can be used in both instances. The resulting decoder block, written for MATLAB, in which all loudspeaker feeds are generated simultaneously is shown in listing 12.1.

```
if(size(BFormatSig, 2)==3)
    %Assume Z channel contains zeros
    speakerFeeds = 0.5 * (BFormatSig(:,1) * sqrt(2) * (2-directivity)...
        + BFormatSig(:,2) * (directivity .* cos(azimuth) .* cos(elevation)) ...
        + BFormatSig(:,3) * (directivity .* sin(azimuth) .* cos(elevation)) );
else
    speakerFeeds = 0.5 * (BFormatSig(:,1) * sqrt(2) * (2-directivity)...
        + BFormatSig(:,2) * (directivity .* cos(azimuth) .* cos(elevation)) ...
        + BFormatSig(:,3) * (directivity .* sin(azimuth) .* cos(elevation)) ...
        + BFormatSig(:,4) * (directivity .* sin(elevation)) );
end
```

Listing 12.1: Decoding of a B-Format signal, `BFormatSig`, for loudspeakers with the specified `azimuth` and `elevation`. `directivity` is the directivity of the virtual microphone used to calculate the loudspeaker feed. Note that multiple loudspeaker feeds can be calculated simultaneously if `directivity`, `azimuth` and `elevation` are row vectors.

Although this could have then been used to create a frequency dependant decoder, within the scope of the project this was not seen to be necessary and so instead a method of implementing continuous decoding was investigated. Using the playrec utility it had already been shown that continuous audio output was feasible from within MATLAB. Additionally, because the decoder included no 'memory' of previous samples, it could be used to decode a signal in blocks rather than all in one go. Therefore, the only remaining problem was obtaining the B-Format signal to decode, to which two different solutions were found. The first was based around a

GUI in which the mouse could be dragged to change the position of a sound source. In this case the mono sound source, S, was encoded into a B-Format signal, using the equations

$$\begin{aligned} W &= \frac{S}{\sqrt{2}} \\ X &= S\cos\alpha \\ Y &= S\sin\alpha \\ Z &= 0 \end{aligned}$$

where α is the azimuth of the virtual sound source. By encoding and then decoding the signal in small blocks of samples (typically 512 samples long), the sound output from the loudspeakers would move with only a small delay relative to the mouse movement. However, it was found that if this latency was reduced too far, the audio output would be stable the majority of the time although sporadically glitches would occur due to other background tasks running on the computer.

An alternative source of B-Format signals were those available for download from a range of websites[1]. Using these, combined with a version of the MATLAB file `wavread.m` modified by Sylvain Choisel[2] to support the WAVE-FORMAT-EXTENSIBLE (WAVE_EX) file format used by these files, it was possible to implement a decoder that would, in theory, decode a file of any length. This was achieved by reading a small block of samples from the file, decoding to the appropriate loudspeaker feeds and then sending these samples to the playrec utility before repeating the cycle. The two files used to implement this, `playAmbiWav.m` and `BFormatDec.m`, as well as a file to encode a mono signal to a B-Format signal, and one to rotate, tilt and tumble a B-Format signal can all be found on the accompanying CD, as described in appendix G.

[1]Sources of files include SoundField Ltd. (`http://www.soundfield.com/downloads/b-format.php`), Angelo Farina's public files (`http://pcfarina.eng.unipr.it/Public/B-format/`), and the Ambisonic Bootlegs website (`http://www.ambisonicbootlegs.net/`). The 'Composers' Desktop Project' Multi-Channel Toolkit provides useful command line tools to reformat these files where necessary (`http://www.bath.ac.uk/~masrwd/mctools.html`).

[2]`wavexread.m` available online at `http://acoustics.aau.dk/~sc/matlab/wavex.html`.

Chapter 13

Overall System Design

For an Ambisonics system to be easy to install and upgrade, one possibility is to distribute the signal processing between a central unit and the loudspeakers using wireless communications. However, for this to work effectively care must be taken during the initial stages of the design. In this section the overall design of such a system is considered, assuming the microphone(s) used for loudspeaker calibration are attached to the central unit.

13.1 Signal Decoding

As a wireless surround sound system Ambisonics has the distinct advantage that the loudspeaker feeds are all generated from the same B-Format signals. Therefore, instead of transmitting each loudspeaker signal separately from the central unit it should be possible to transmit the B-Format signals simultaneously to all loudspeakers. By including loudspeaker specific decoding information within the signal, each loudspeaker can then generate its own signal. This reduces the amount of processing required in the central unit and also allows very simple system expansion—the only real limit is the amount of time required to configure all the loudspeakers with their own decoding parameters. Additionally, when only using a first-order Ambisonic system, this reduces the total bandwidth required if there are more than four loudspeakers.

Two further considerations are the latency through the system and the synchronisation between the different loudspeakers. To achieve minimal latency, important when the system is part of a larger multimedia setup, the buffering of data must be kept to a minimum. If each loudspeaker could request retransmission of corrupted data from the central unit, this could improve the quality of audio when used in 'noisy' RF environments. However, this requires enough data to be buffered to allow the whole retransmission cycle to occur. Furthermore, if some data was corrupted on arriving at one loudspeaker it may also have been corrupted at others, thus meaning multiple loudspeakers may try and send re-transmission requests simultaneously.

An alternative approach that would simplify both the loudspeakers and the central unit would be to use a uni-directional link. This would allow for much smaller buffers to be used but would increase the chance of the loudspeaker having either no data or incorrect data to output. Using some form of error correction scheme each loudspeaker would be able to correct for a limited number of received errors and so reduce, although not eliminate, the effect of these. The 'best' compromise between latency, resilience to errors and bandwidth required is one that can only be made with knowledge of the equipment and intended environment.

Accurate synchronisation between the loudspeakers and the central unit is important for two reasons: it ensures that the 'same' sound is not reproduced by one loudspeaker before another, and it helps avoid either a buffer underrun or overrun within the loudspeaker. The former does not impose a very strict requirement provided any offset between loudspeakers in small and remains constant—this is the same as an inaccuracy in the positioning of the loudspeaker. Inclusion of timing information within the transmitted radio signal meets these requirements: an internal clock within each loudspeaker would control the output of individual samples whilst the timing information from the central unit would be used to fine tune this clock on a regular basis.

To decode a B-Format signal, the decoding parameters must be based on the location of the loudspeakers. Using the basic decoder implemented during the project, the central unit would only need to tell each loudspeaker where it is positioned. However, to determine the optimum decoding parameters for an arbitrary arrangement it is necessary to know where all loudspeakers are positioned. Implementing this within the loudspeakers would therefore not be sensible because it would rely on all the other loudspeakers calculating the same set of results. Thus these calculations should be made by the central unit, which introduces a potential problem: how to make this part of the system scalable. If bi-directional links were used with each loudspeaker then a method of distributing this processing could be implemented, although leaving the central unit processing for longer would be another viable alternative, depending on how much longer it would actually take. Using this approach a very basic decoding could be available almost immediately with 'better' decodings becoming available over time. The system should not just switch between these decodings, but they could be added to a list from which the user can select.

13.2 Loudspeaker Location Detection

Assuming the use of uni-directional links, each loudspeaker would have to be given a unique identifier. In the simplest system design the user would set these using sequential numbers, telling the central unit how many loudspeakers exist. The order of these numbers would not be important provided none were duplicated. An alternative would be for each loudspeaker to be given its unique identifier during manufacture. The central unit could then determine which loudspeakers were present by using sound signals as a return link from the loudspeaker. Obviously if bi-directional links are in use then this problem is greatly reduced.

As has been shown, the location of a loudspeaker can be determined when it produces a known excitation signal. By putting all loudspeakers into a 'configuration' mode it would be possible to send this signal using one of the audio channels within the wireless link. Alternatively the excitation signal could be generated within the loudspeaker and just triggered by the central unit. The former of these has two advantages: every loudspeaker does not have to be capable of generating the required excitation signal, and the loudspeakers would be capable of working with different central units, even if they use different algorithms to locate the loudspeakers.

No matter where the excitation signal is actually generated, the latency and synchronisation between central unit and loudspeaker would have a significant impact, under some techniques, on the possible accuracy of loudspeaker location. Using a SoundField microphone, or equivalent intensity difference approach, then the latency and synchronisation would impact the separation measurement but they would not affect the angle measurement. In comparison, both angle and separation measurements would be affected when using time-of-flight with four microphones whilst neither would be affected when using time difference of arrival with four microphones. However, without testing all three of these methods it is not possible to determine if this independence from latency and synchronisation is outweighed by the technique being prone to larger errors.

13.3 Summary

It can be concluded that, theoretically, a system using a common uni-directional link between a central unit and all loudspeakers could be implemented. Each loudspeaker would need to be given a unique identifier and the central unit would require a means of determining how many loudspeakers are present. Assuming a first-order Ambisonics system, this link would need to carry four channels of audio data at the required sample rate in addition to any data required for error correction. Further to this a control data stream would be necessary including information such as the decoding parameters and volume as well as the current mode for each loudspeaker: calibration, mute or decoding. Should a loudspeaker either temporally lose power or receive corrupted data, continuously repeating this information in the control stream would then allow it to quickly resume as required.

To calibrate the whole system the central unit would iterate through all loudspeakers placing them into a calibration mode and then measuring their location using one of the techniques previously discussed. With this information a very crude decoding could be determined immediately, with better decodes produced over a certain period of time, dependant on the number of loudspeakers, the algorithm used and the processing power within the central unit. The resulting decoding parameters would be sent to the loudspeakers and the Ambisonics system would be functional.

Provided the necessary bandwidth to carry the extra data is available, this could be expanded in different ways. For example a higher order Ambisonic system could be implemented, requiring a larger number of audio data channels and more decoding parameters to be determined, or loudspeaker frequency compensation could be provided, requiring the central unit to tell each loudspeaker its required compensation.

Chapter 14

Project Management

The large number of aims for this project meant that successful time management was always going to be critical for the project to succeed. This chapter outlines some of the decisions that were made during the project that affected what would be achieved and by when.

An initial timetable was created which was believed to allow the project to be completed on time. However, following the testing of different applications to find one suitable for the project, as described in chapter 4, it was decided that the playrec utility should be written to provide the required audio handling. Initially the development of audio handling routines was planned to occur throughout the project as necessary, but implementing this utility was seen to be beneficial to both this and future projects and so the project timetable was revised. It was expected that including the software development would mean not all of the aims of the project could be achieved, so the different parts of the project were prioritised.

To generate a complete system implementation the utility was seen to be necessary and so, combined with the advantages of using the same audio handling for the duration of the project, this was given highest priority. Investigation into, and implementation of, the different excitation signals was placed next because without this no tests could be implemented. Following this the different microphone tests and data analysis were seen to have similar priorities to the implementation of a basic B-Format decoder. Although such Ambisonic decoders had been implemented before, implementing it within MATLAB would help confirm correct operation of the playrec utility, this being one of its target applications, as well as show what could be achieved with the utility, opening up the opportunity for many future projects. Because the different microphone tests would also have to occur sequentially they were prioritised relative to each other. The first of these was the separation measurements using a single microphone. Although this could have been conducted directly as part of either of the other microphone configurations, by initially implementing this on its own a test setup specifically for measuring separation could be used and any problems with the excitation signals could be eliminated before further tests. Next on the list was testing angle measurement using the SoundField microphone. Its cost would probably prohibit its use in a final commercial system, but it was decided that advantage must be taken of the availability of this relatively rare and expensive piece of equipment. This resulted in the tests using multiple microphones being given lowest priority.

The project timetable was re-created so that tasks would be completed in priority order. However, to avoid unexpected problems preventing any further work some items were still timetabled to occur simultaneously. At this stage all of the sets of testing were still included, at the expense of removing the two spare weeks previously allocated to allow parts of the project to overrun. Therefore, should any problems arise it was possible that the project timetable would require further updating.

Utility development and testing took longer than planned due to various problems, such as the unexpected random addition of 5 output samples. Almost simultaneously, the implementation and testing of the creation and processing files for the different excitation signals also took longer than expected. Therefore it was necessary to revise the project timetable one last time. In this revision all tests using multiple microphones were removed for two reasons: first, a long setup time was expected for this configuration, accurately positioning all the microphones and running preliminary tests; and second, allocating longer times for two sets of tests rather than rushing three sets was deemed to be more beneficial for the project as a whole. Even then it was found that far more time could have been spent on each of the other sets of tests, such as analysing the effect of the room conditions or loudspeaker orientation. This additional work had to be left until future projects although that planned was completed according to the timetable.

Chapter 15

Further Work

The work undertaken during this project can be considered under four categories: audio handling within MATLAB, loudspeaker location detection, Ambisonics decoding, and complete Ambisonic system implementation. There is more work that could be done in each area, some of which is described in this chapter.

15.1 Audio Handling within MATLAB

The playrec utility, implemented to provide continuous audio input and output within MATLAB, was designed to be versatile enough to support a wide range of target situations, computers, soundcards and even operating systems. Therefore further work based on the this utility could include:

- testing for correct functionality when using different computers and soundcards from those used during the initial development of the utility. By repeating the loop back tests described in section 7.5 the extent of the observed unusual behaviour (the extra output samples) could then be determined.
- recompiling the utility to utilise an alternative soundcard interface or even to

operate on a different operating system, utilising the cross-platform support offered by PortAudio.

- modifying the utility to utilise the additional functionality offered by PortAudio V19.
- adding simultaneous support for multiple streams to allow more than one soundcard to be used.
- writing MATLAB functions to determine the true extent of what can be achieved using the utility such as the minimum latency that can be used reliably, limits on the number of channels or sample rates (apart from those inherent within PortAudio), or possible implementation of a real-time filter or a spectrum analyser.

15.2 Loudspeaker Location Detection

The tests conducted to measure the accuracy with which the location of loudspeakers can be determined included many assumptions. In a completely automatic system, depending on the level of accuracy required, no assumptions can be made and so further work should investigate the significance of these assumptions, which included:

- the room being quiet and time-invariant;
- the speed of sound in the room remaining constant and hence the room's temperature, pressure, and even humidity and CO_2 concentration not changing;
- the microphones, loudspeakers, amplifies, and D/A and A/D converters all being identical to those used in the tests;
- the loudspeaker always being orientated to point directly at the microphone;

- the loudspeaker and microphone being positioned in the middle of the room, ensuring the direct sound section of each IR could be easily distinguished from room reflections.
- only one loudspeaker producing the excitation signal at any one time, something that should always be possible unless the user has wired two loudspeakers to be fed with the same signal.

Analysing the results obtained when using a single microphone highlighted the need for further work in the following areas:

- Peak sample detection. It was found that in some IRs the ripple either side of the 'peak' sample had a larger magnitude than the peak itself. In a final system it should not be assumed that the peak will always be positive and therefore a method of correctly identifying the peak or reducing the magnitude of the ripple should be investigated.
- Signal interpolation. After an investigation into different interpolation methods quadratic interpolation was used. However it was not determined if this approach, when combined with the different IR measurement techniques, provides a linear distribution of interpolated points between samples. Therefore if very high levels of accuracy are required this should be practically investigated using a test setup capable of making very accurate and small movements of either a loudspeaker or microphone. Additionally a comparison between quadratic interpolation and other techniques, such as oversampling, could be made.
- Measurement accuracy. The grouping of the results at each test separation indicated that the technique used to position the microphone was less accurate than the accuracy of the measurements made with sound. Therefore the same tests should be repeated in a much more strictly controlled environment so that any variations can be attributed to the measurement technique and not the test methodology. As a part of this, a comparison between the

speed of sound determined from the room conditions and that predicted by the results can be made. Furthermore this would enable a more accurate comparison between the different excitation signals to be made—are some more accurate than others when measuring distances, or were the discrepancies observed due to other factors, such as slightly different acoustic centres for different excitation signals.

- Multiple Acoustic Centres. Within a loudspeaker it is expected that its acoustic centre is not in the same place at all frequencies. Therefore by determining the location of the sound source using different ranges of frequencies, it might be possible to determine not only the position but also the orientation of the loudspeaker. However, this also introduces the extra unknown of the maximum accuracy possible with only a limited range of frequencies.

Using a SoundField microphone it was found that at high frequencies the B-Format signals produced were significantly different from those expected. Therefore, as described in section 10.3, by repeating these tests with another microphone of the same model as well as different models of SoundField microphone the source of this problem—calibration, the model of microphone, or SoundField microphones in general—could be determined. Additionally the use of filtering, as preliminarily tested, could be investigated further to find the optimum filter for measuring the angle of the loudspeaker as well as that for measuring the separation. This could also further the work presented by including variations in elevation as well as azimuth.

To be able to compare the SoundField microphone approach to that using multiple microphones, tests such as those described in section 11.2 could be conducted. Further to these, the effect of inaccuracies in the relative positions of the microphones could be investigated, relating this to the accuracies required during microphone manufacture and therefore indirectly the overall cost of the system.

In addition to determining the actual location of the loudspeakers within a room, further work could investigate how to measure other features of a particular Ambisonics setup, such as the apparent source location of early reflections and the frequency response of the loudspeakers.

15.3 Ambisonics Decoding

When determining the decoding parameters for an Ambisonics decoder, such as that implemented for the project, various assumptions about the loudspeaker locations are always made, such as left/right symmetry or equal distances from the listener to all loudspeakers. In a system where the loudspeaker locations do not fit these assumptions, using them to determine the decoding parameters could introduce significant errors in the reproduced soundfield. Therefore a method of determining the optimum decoding parameters for a completely arbitrary set of loudspeaker locations should be investigated. For some sets of locations, such as positioning all loudspeakers on one side of the listener, the results would obviously be atrocious and so positioning the loudspeakers ‘sensibly’, which would need to be defined, would always be required. However, even when including this there would still be the need to determine the optimum value for many decoding parameters. For a system such as that proposed the investigation into this should consider not only the performance of the resulting decoder but also the processing power and time required to obtain it. The possibility of including frequency and distance compensation within the decoder could also be considered.

15.4 Complete System Implementation

Although a top-level design for a complete system using wireless communication has been proposed, until some of the further work described above has been completed such a system would be limited to using specific loudspeaker arrangements. Therefore a practical implementation could be investigated further and constructed, although if this were to happen the design should be very flexible to allow changes to be made based on the results of other work. For example, until the comparison between the different excitation signals and microphone arrangements is completed the required processing within the central unit is unknown, as is the impact of varying latency through the system. Additionally, until a suitable method of calculating the decoding parameters has been determined, the required resources to implement the algorithm efficiently is unknown.

Chapter 16

Conclusion

A suitable test system, based around a Personal Computer (PC), was designed. A utility to be used within MATLAB was implemented and refined, allowing continuous multi-channel audio input and output. This has been shown to be versatile: it was used for all testing in determining the location of loudspeakers and also allowed a continuous Ambisonics decoder to be implemented, running simultaneously with a MATLAB GUI.

The accuracy to which microphone-loudspeaker separations can be measured using sound was investigated with seven excitation signals based on Maximum Length Sequences (MLS), Inverse Repeated Sequences (IRS), Optimum Aoshima's Time-Stretched Pulses (OATSP) and logarithmic sweeps. These were all found to measure propagation delays to a sub-sample level accuracy, with a strong indication that the accuracy achievable is greater than that of the experimental method used. The repeatability of the results using any one excitation signal was found to be very good, with a maximum range of 0.039 of a sample interval across three readings. Minor differences were observed between the different test signals, although these, at under 0.10 of a sample interval, were also less significant than the inaccuracies within the test procedure. The MLS and IRS excitation signals were found to be more bearable than the other signals, although with this there was also a reduced feeling of the ‘need’ to remain quiet, something which could favour the alternative signals despite their poorer noise immunity. The accuracy of a room temperature reading was found theoretically to have a significant impact on the accuracy with which separation can be measured, although this was not tested.

Using a SoundField microphone and the same seven excitation signals it was found that azimuth angle measurements could be accurate to within $\pm 1^\circ$. The B-Format signals produced by the microphone deviated significantly from their theoretical values at high frequencies, requiring low-pass filtering to obtain this level of accuracy. Additionally, the results indicated that the accuracy achievable was equal to, or better than that of the test procedure. Due to time limitations, no tests could be conducted to determine the accuracy achievable using four omni-directional microphones.

An Ambisonics decoder was implemented within MATLAB based on virtual microphones orientated towards each loudspeaker. This was shown to allow continuous decoding of B-Format signals stored within a file as well as decoding B-Format signals generated dynamically through controls on a MATLAB GUI.

A potential top-level design for a complete Ambisonic system using wireless communications has been proposed. This system consisted of as many loudspeakers as required and a central unit, into which the B-Format signals are fed and to which the calibration microphone is attached. Sequentially the location of each loudspeaker would be determined before converting this information into appropriate decoding parameters and starting the decoding. Through implementing the decoding within each loudspeaker, expansion of the system would be possible. However, system expansion was seen to be a potential source of problems when calculating the decoding parameters—something which would most likely be implemented solely in the central unit. Before the system can be fully implemented, further work into the detection of loudspeaker positions and the calculation of decoding parameters is required.

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Appendices

Appendix A

MEX-file Overview

MEX-files are shared libraries which can be executed from within MATLAB. They can be compiled from C, C++, or Fortran source code using the `mex` function within MATLAB [38], or alternatively they can be compiled using a separate compiler creating a ‘dll’ file within Windows. For instructions on how to use Visual Studio to achieve this see Appendix C.

Throughout this project only C was used to create MEX-files and so this overview is based on using C, although many of the concepts are similar when using Fortran.

A.1 Entry Point Function

Within a C MEX-file there must be an entry point function called `mexFunction`. This is executed by MATLAB whenever the library is used, achieved by typing the name of the MEX-file in the same way as m-files can be used. No matter how many variables are supplied to or expected from the MEX-function within MATLAB, the entry point function always receives four parameters:

nlhs The number of variables expected to be returned.

plhs A pointer to an array of pointers of length nlhs, used to return the expected number of variables.

nrhs The number of variables supplied to the MEX-function.

prhs A pointer to an array of length nrhs of pointers to the supplied variables.

All variables are transferred between the MEX-file and MATLAB using C structures of type `mxArray`. A set of functions are then supplied to operate on these `mxArrays`—creating, modifying and destroying them. Examples of such functions include `mxCreateStructMatrix()`, `mxSetField()`, `mxGetScalar()` and `mxDestroyArray()`. Additionally there are a set of functions used to determine the type of variable stored within a `mxArray`, such as `mxIsChar()`, `mxIsComplex()` and `mxIsStruct()`. Utilising these functions, the type of variables supplied as parameters from within MATLAB can be determined, the required data can be extracted from them and new variables can be created for use as return parameters.

A.2 Memory management

With MEX-files, as with any applications written in C, correct memory management is very important. However, to achieve this the lifetime of dynamically allocated memory must be taken into account:

- All `mxArrays` passed to the entry point function are managed by MATLAB and must not be freed or modified within the MEX-file.
- All `mxArrays` created to be used as return parameters and added to `plhs` are managed by MATLAB and are automatically freed once they are no longer required.

- All other `mxArrays` are also freed by MATLAB when the entry point function returns *unless* the `mxArray` has been marked as 'persistent' using `mexMakeArrayPersistent()`. In this case MATLAB will *never* free the memory and instead it must be freed by the MEX-file once it is no longer required.

- Memory allocation using the standard C functions, such as `calloc()` and `malloc()`, is not recommended and instead the 'mx' equivalents, such as `mxCalloc()` and `mxMalloc()`, should be used where possible. Such memory has the same lifetime as `mxArrays` and can, if required, be made persistent using `mexMakeMemoryPersistent()`. In some scenarios, such as when including source code that has already been written, the standard C functions might need to be used. Any memory allocated in this way will never be freed by MATLAB and so must always be freed once it is no longer required.

Once a MEX-file has been used, it remains loaded within MATLAB and so all global variables and all persistent dynamically allocated memory remain unchanged between uses of the MEX-file. To ensure all memory is freed after the MEX-file has been used for the last time an exit function can be registered using `mexAtExit()`. Once registered, the function will be called by MATLAB before the library is unloaded, such as when MATLAB is closed or the MATLAB `clear` command is used. The function must free all memory that has been dynamically allocated and not freed either explicitly or automatically by MATLAB.

A.3 Textual Display

From within MEX-files it is possible to display text within the MATLAB command window using similar commands to those used in m-files:

mexPrintf() This is very similar to the C function `printf` and the MATLAB command `fprintf()` when `fid` is omitted.

mexWarnMsgTxt() mexWarnMsgIdAndTxt() These display a warning although do not terminate the MEX-file. They are similar to different versions of the MATLAB command `warning()`.

mexErrMsgTxt() mexErrMsgIdAndTxt() These display an error, terminate the MEX-file and return control to the MATLAB prompt. They are similar to different versions of the MATLAB command `error()`.

A.4 Providing Help

Although text can be displayed within the command window when a MEX-file is used, it cannot be used to display help information when the MATLAB `help` command is used. Instead, a m-file with the same name as the MEX-file must be created to contain the help information. This will not affect the operation of the MEX-file and will only be used when help is requested.

More information on MEX-files can be found within the "External Interfaces" and "External Interfaces Reference" sections of the MATLAB help[1].

[1]Supplied with MATLAB and available online at `http://www.mathworks.com/access/helpdesk/help/techdoc/matlab.html`

Appendix B

PortAudio Overview

PortAudio is a “free, cross platform, open-source, audio I/O library” written in C, providing a “very simple API for recording and/or playing sound using a simple callback function”[25]. For the duration of this project V18.1 was used because it was the latest non-development versions available and it provided all the functionality required.

The library operates by calling a registered function with two buffers—one containing recorded samples and the second to be populated with samples to play. This callback function is automatically called by the library and so it cannot be predicted when it will run, just that it will run regularly enough to avoid glitches in audio. This assumes, of course, that the buffers have been configured correctly and the function does not take too long to execute. For applications that only play samples from a larger play buffer and record samples to a larger record buffer, such an interface makes this a simple case of copying memory. Additionally, through this type of interface ‘pausing’ the audio stream can be achieved by ignoring the supplied record buffer and zeroing all values in the play buffer. Due to the buffering of recorded samples this approach effectively pauses recording slightly before playing but then it also resumes recording before playing by the same amount. However, in most applications this would not be problematic.

To configure the library to start using the callback, the following procedure must be followed[26]:

1. Initialise PortAudio using `Pa_Initialize().`
2. Open a PortAudio Stream using either `Pa_OpenDefaultStream()` or `Pa_OpenStream().` This configures all the stream settings such as which device, channels, sample rate and bit depth to use. Additionally it registers the function to be used as the callback function described above.
3. Call `Pa_StartStream()` to start the library calling the callback and hence start audio input and output. Once the stream has started this function returns for the thread to continue.

When the library is no longer required, the following procedure must be followed[26]:

1. Stop the stream by either returning `1` from the callback function or calling `Pa_StopStream().` The former always stops the callback from being called although in some circumstances when using the latter the callback might be called one more time, despite `Pa_StreamActive()` reporting the stream as inactive. This was found to cause problems when stopping the stream and then immediately freeing memory that the callback used. One solution is to always use the callback to stop the stream, and then wait for the stream to stop according to `Pa_StreamActive()` before freeing memory.

   The stream can be started again using `Pa_StartStream().`
2. Close the stream using `Pa_CloseStream().` New streams can still be opened as described above.
3. Terminate PortAudio using `Pa_Terminate().` The library should not be used once this has been called unless it has been initialised again.

If required multiple streams can run simultaneously using either the same or different callbacks. To identify which stream is calling the callback a pointer can be supplied when opening the stream. This pointer can be of any type and is passed to the callback.

Although the C code required to use PortAudio does not have to be changed to use different operating systems or audio interfaces, the files used from the PortAudio SDK do have to change. Additionally, when using the ASIO implementation a SDK from the developers section of Steinberg's website[1] must be obtained.

For more information on the PortAudio functions refer to `portaudio.h` within the PortAudio SDK[2] and for configuration information see the PortAudio tutorial[26].

[1]`http://www.steinberg.net/`

[2]Available online at `http://www.portaudio.com/download.html`

Appendix C

Configuring Visual Studio

This section provides configuration information to allow Microsoft Visual Studio to create MEX-files that can use PortAudio. For a basic overview of MEX-files and PortAudio see Appendices A and B respectively.

MEX-files can be compiled either within MATLAB or using a separate compiler. The latter is initially more complicated to configure although once configured it allows for easier development through the use of an Integrated Development Environment (IDE), something especially useful when multiple files are being used.

Due to the need to include source code from the PortAudio SDK and the Steinberg ASIO SDK, the latter approach was considered much more suitable. The available IDE was Microsoft Visual Studio 2005 Professional Edition, which was configured as follows[1] to produce suitable MEX-files for use within MATLAB:

1. Install MATLAB and Visual Studio.

[1]Configuration is based on a combination of the pa_wavplay utility source code (`http://sourceforge.net/projects/pa-wavplay/`), the PortAudio tutorial (`http://www.portaudio.com/docs/`), MATLAB documentation "Custom Building on Windows" (`http://www.mathworks.com/access/helpdesk/help/techdoc/matlab_external/f24571.html`) and modifications found to be necessary throughout the course of the project.

2. Download and extract the PortAudio SDK[2] and the Steinberg ASIO SDK[3] to suitable locations. This configuration is based on PortAudio SDK V18.1 and Steinberg ASIO SDK 2.1.

3. Create a new Visual C++ Win32 Console Application, specifying an 'Empty project' and 'DLL' application type.

4. Add to the project the files `pa_common\pa_lib.c`, `pa_common\portaudio.h`, `pa_common\pa_host.h` and `pa_asio\pa_asio.cpp` from the PortAudio SDK as well as `host\asiodrivers.cpp`, `host\pc\asiolist.cpp` and `common\asio.cpp` from the ASIO SDK. None of these files should need modifying so they can be used from their original location within the extracted SDKs.

5. Within the project properties:

   - add additional compiler include directories: `pa_common` from the PortAudio SDK; `host`, `host\pc` and `common` from the ASIO SDK; and `extern\include` from the MATLAB installation directory.
   - Change the character set to 'Use Multi-Byte Character Set' from the default 'Use Unicode Character Set'. This is to avoid compile errors within `asiolist.cpp`.
   - Add the C/C++ Preprocessor definitions MATLAB_MEX_FILE and WIN32 if not already included.
   - Add linker dependencies: `winmm.lib`, `libmex.lib`, and `libmx.lib`. Depending on which MATLAB functions are used some of these may not be required.
   - Add an additional linker library directory for the MATLAB `.lib` files. Choose the most appropriate directory within `extern\lib\` of the MATLAB installation directory. Within MATLAB R14SP3

[2]Available online at `http://www.portaudio.com/download.html`

[3]Available from the 3rd Party Developers section of Steinberg's website at `http://www.steinberg.net/`

there is no directory specifically for Visual Studio 2005 so `extern\lib\win32\microsoft\msvc71` should be used instead.

- Add the additional linker command line option `/export:"mexFunction"`

6. Add a new `.c` or `.cpp` file to the project to contain the entry point function from MATLAB. Add the include statements

```
#include "mex.h"
#include "portaudio.h"
```

and the entry point function

```
void mexFunction(int nlhs, mxArray *plhs[], int nrhs, const mxArray *prhs[])
{
}
```

on which more information can be found in the MATLAB documentation[4].

Following these steps the compiler should produce a file with the `.dll` extension. To use the file within MATLAB either select the directory containing the file as the 'Current Directory' or add it to the MATLAB search path. On typing the name of the file no warnings should be produced—if warnings or errors are produced they should be rectified before continuing. To check the correct configuration of the PortAudio SDK and the ASIO SDK, the code shown in listing C.1 can be used instead of the file created in step 6 above.

A further advantage of using an IDE is the ease with which code can be debugged. To start debugging, the 'Local Windows Debugger' option should be selected in the Debugging section of the project properties and the path to the MATLAB executable[5] added as the 'Command'. From within the IDE selecting 'Start Debugging' will then start MATLAB and the MEX-files can be debugged using all the tools provided by Visual Studio. When starting debugging, a warning that no debugging information could be found for MATLAB.exe may be displayed. Despite this, debugging of the MEX-file can continue without problems.

[4]See "mexFunction (C)" within the help documentation supplied with MATLAB and online at `http://www.mathworks.com/access/helpdesk/help/techdoc/apiref/mexfunction_c.html`.

[5]`\bin\win32\MATLAB.exe` within the MATLAB installation folder

```
/* pa_dll_test.c
 *
 * A single-function file to be used in conjunction with the PortAudio SDK
 * to create a MATLAB MEX-file.  Returns a structure matrix containing
 * information of all available devices.  Although this does not test audio
 * throughput, it can be used to test for correct compiler and linker
 * configuration.
 *
 */

#include "mex.h"
#include "portaudio.h"

void mexFunction(int nlhs, mxArray *plhs[], int nrhs, const mxArray *prhs[])
{
    const char *field_names[] = {"deviceID", "name", "inputChans", "outputChans"};
    const PaDeviceInfo *pdi;
    int i;
    PaError err;

    if(nlhs > 1){
        mexErrMsgTxt("Too many output arguments expected");
    }

    err = Pa_Initialize();
    if( err != paNoError )
    {
        Pa_Terminate();
        mexPrintf("Error number: %d\nError message: %s\n",
                err, Pa_GetErrorText( err ) );
        mexErrMsgTxt( "The above error occured when initializing PortAudio" );
    }

    plhs[0] = mxCreateStructMatrix(1, Pa_CountDevices(),
            sizeof(field_names)/sizeof(char*), field_names);

    for( i = 0; i < Pa_CountDevices(); i++)
    {
        pdi = Pa_GetDeviceInfo(i);

        if (pdi != NULL)
        {
            mxSetField(plhs[0],i,"deviceID", mxCreateDoubleScalar(i));
            mxSetField(plhs[0],i,"name",mxCreateString(pdi->name));
            mxSetField(plhs[0],i,"inputChans",
                    mxCreateDoubleScalar(pdi->maxInputChannels));
            mxSetField(plhs[0],i,"outputChans",
                    mxCreateDoubleScalar(pdi->maxOutputChannels));
        }
    }
    Pa_Terminate();
}
```

Listing C.1: Sample test file which can be used to confirm correct configuration of Visual Studio to produce MEX-files including ASIO Driver support.

Appendix D

playrec Utility Help Information

Within the playrec utility an overview of the utility's operation is provided by the 'about' command, and help information for each of the commands is provided through the 'help' command. This section reproduces all of this information directly as it is displayed by the utility within the MATLAB Command Window.

D.1 Utility overview

This playrec utility has been written to provide versatile access to soundcards via Steinberg's ASIO API. It is based on PortAudio, a free, open-source audio I/O library, so should easily be portable to either WMME or DirectSound under Windows, or even to different platforms just by recompiling with the relevant files.

A basic outline of how to use this utility is provided below. For more information on any command type 'help' as the first parameter followed by the command of interest as the second parameter.

All commands are accessed through the one function in MATLAB. This is achieved by specifying the name of the command to run as the first parameter in the function call followed by any additional parameters as required by the command. A list of all available commands can be displayed by supplying no parameters when calling the function.

Before any audio can be played or recorded, the utility must be initialised to use the required sample rate and device(s). Initialisation can be achieved using the ‘init’ command, supplying the ID of the required audio device(s) as returned by ‘getDevices’. Once successfully initialised, the sample rate or device(s) to be used cannot be changed without first resetting the utility using the ‘reset’ command. This clears all previously recorded data so use it with care. To check if the utility is currently initialised, use the ‘isInitialised’ command.

The utility divides time up into “pages”, with no restrictions on the duration of any one page, although with very short pages skipping in the audio may occur. Additionally there can be as many pages as required, provided the utility can allocate memory to store all the pages. Pages are joined together sequentially in the order they are added, with each page starting the sample after the previous page finishes. The duration of a page is determined by the longest channel contained within the page. Therefore if, for example, the record channels are 1000 samples long whilst output channels are only 900 samples long, the page will be 1000 samples long and the final 100 output samples of the page will automatically be set to 0.

When each page is added, the channels that are to be used for recording and/or output are specified (depending on the command used to add the page). The channels used must be within the range specified during initialisation and no channel can be duplicated within a channel list. Within these limits, the channel list for each page can be different and each list can contain as many or as few channels as required in any order. All output channels not provided with any data within a page will output 0 for the duration of the page. Similarly, during any times when there are no further pages to process 0 will be output on all channels.

Each page has a unique number which is returned by any of the commands used to add pages (‘playrec’, ‘play’ or ‘rec’). When a page is added, the utility does not wait until the page has completed before returning. Instead, the page is queued up and the page number can then be used to check if the page has finished, using 'isFinished'. Alternatively a blocking command, 'block', can be used to wait until the page has finished. To reduce the amount of memory used, finished pages are automatically condensed whenever any command is called in the utility. If a page contains any recorded data, this is left untouched although any output data within the page is removed. If the page does not contain any recorded data, the whole page is deleted during this page condensing. For this reason if either 'isFinished', 'block' or 'delPage' indicate the page number is invalid this means the page either never existed or has already finished and then been deleted during page condensing.

For pages containing recorded data, the data can be accessed using the 'getRec' command. This does not delete the data so it can be accessed as many times as required. To delete the recorded data, the whole page must be deleted using the 'delPage' command. This command will delete pages no matter what their current state: waiting to start, currently active or finished. If no page number is supplied, all pages will be deleted, again regardless of their state.

To ascertain which pages are still left in memory, the 'getPageList' command can be used, returning a list of the pages in chronological order. NOTE: there may have been gaps of silence or other pages between consecutive pages in this list due to pages either being automatically or explicitly deleted as detailed above. To determine if there were gaps between pages due to all pages finishing processing before new ones are added, the commands 'getSkippedSampleCount' and 'resetSkippedSampleCount' can be used.

The page that is currently being output is returned by 'getCurrentPosition', along with an approximate sample position within the page. Additionally, the page number of the last completed page still resident in memory is returned by 'getLastFinishedPage'. NOTE: this might not be the most recent page to finish if that page has been deleted either during page condensing (ie contained no recorded data) or through the use of 'delPage'.

Finally, the utility can be paused and resumed using the 'pause' command. This will manipulate all output and recording channels simultaneously to ensure synchronisation is always maintained. This command can also be used to ascertain if the utility is currently running or paused.

D.2 Command specific help

D.2.1 help

```
[ ] = help(commandName)
```

Displays command specific usage instructions.

Input Parameters

commandName name of the command for which information is required

D.2.2 about

```
[ ] = about()
```

Displays information about this utility

D.2.3 getDevices

```
[deviceList] = getDevices()
```

Returns information on the available devices within the system, including ID, name, and number of channels supported.

Output Parameters

deviceList Structure array containing the following fields for each device: ‘deviceID’ - ID used to refer to the device; ‘name’ - textual name of the device; ‘inputChans’ - number of input channels supported; ‘outputChans’ - number of output channels supported

D.2.4 init

```
[ ] = init(sampleRate, playDevice, recDevice, {playMaxChannel},
{recMaxChannel}, {framesPerBuffer})
```

Configures the utility for audio output and/or input based on the specified configuration. If successful the chosen device(s) will be running in the background waiting for the first pages to be received. If unsuccessful an error will be generated containing an error number and description.

All channel numbers are assumed to start at 1. If the maximum channel number to be used is not specified, the maximum number of channels that the device supports is used. Specifying a maximum number of channels verifies the device supports the required number of channels as well as potentially slightly reducing the utility’s processor usage.

If an optional value is specified, all previous optional values must also be specified.

Input Parameters

sampleRate the sample rate at which both devices will operate

playDevice the ID of the device to be used for output (as returned by ‘getDevices’), or -1 for no device (ie output not required)

recDevice the ID of the device to be used for recording (as returned by 'getDevices'), or -1 for no device (ie recording not required)

playMaxChannel *(optional)* a number greater than or equal to the maximum channel that will be used for output. This must be less than or equal to the maximum number of output channels that the device supports. Value ignored if playDevice is -1.

recMaxChannel *(optional)* a number greater than or equal to the maximum channel that will be used for recording. This must be less than or equal to the maximum number of input channels that the device supports. Value ignored if recDevice is -1.

framesPerBuffer *(optional)* the number of samples to be processed in each callback within the utility (ie the length of each block of samples sent by the utility to the soundcard). The lower the value specified the shorter the latency but also the greater the likelihood of glitches within the audio.

D.2.5 reset

```
[ ] = reset()
```

Resets the system to its state prior to initialisation through the 'init' command. This includes deleting all pages and stopping the connection to the selected audio device(s). Generates an error if the utility is not already initialised - use 'isInitialised' to determine if the utility is initialised.

Use with care as there is no way to recover previously recorded data once this has been called.

D.2.6 isInitialised

```
[currentState] = isInitialised()
```

Indicates if the system is currently initialised, and hence if ‘reset’ or ‘init’ can be called without generating an error.

Output Parameters

currentState 1 if the utility is currently initialised, otherwise 0.

D.2.7 playrec

```
[pageNumber] = playrec(playBuffer, playChanList, recDuration,
recChanList)
```

Adds a new page containing both sample input (recording) and output (playing). Generates an error if the required memory cannot be allocated or if any other problems are encountered.

The length of the page is equal to whichever is longer: the number of samples to play or the number of samples to record.

Input Parameters

playBuffer a MxN matrix containing the samples to be played. M is the number of samples and N is the number of channels of data.

playChanList a 1xN vector containing the channels on which the playBuffer samples should be output. N is the number of channels of data, and should be the same as playBuffer (a warning is generated if they are different but the utility will still try and create the page). Can only contain each channel number once, but the channel order is not important and does not need to include all the channels the device supports (all unspecified channels will automatically output zeros). The maximum channel number cannot be greater than that specified during initialisation.

recDuration the number of samples that should be recorded in this page, or -1 to record the same number of samples as in playBuffer.

recChanList a row vector containing the channel numbers of all channels to be recorded. Can only contain each channel number once, but the channel order is not important and does not need to include all the channels the device supports. This is the same as the order of channels returned by 'getRec'. The maximum channel number cannot be greater than that specified during initialisation.

Output Parameters

pageNumber a unique number identifying the page that has been added - use this with all other functions that query specific pages, such as 'isFinished'.

D.2.8 play

```
[pageNumber] = play(playBuffer, playChanList)
```

Adds a new page containing only sample output (playing). Generates an error if the required memory cannot be allocated or if any other problems are encountered.

The page is the same length as that of playBuffer.

Input Parameters

playBuffer a MxN matrix containing the samples to be played. M is the number of samples and N is the number of channels of data.

playChanList a 1xN vector containing the channels on which the playBuffer samples should be output. N is the number of channels of data, and should be the same as playBuffer (a warning is generated if they are different but the utility will still try and create the page). Can only contain each channel number once, but the channel order is not important and does not need to include all the channels the device supports (all unspecified channels will automatically output zeros). The maximum channel number cannot be greater than that specified during initialisation.

Output Parameters

pageNumber a unique number identifying the page that has been added - use this with all other functions that query specific pages, such as 'isFinished'.

D.2.9 rec

```
[pageNumber] = rec(recDuration, recChanList)
```

Adds a new page containing only sample input (recording). Generates an error if the required memory cannot be allocated or if any other problems are encountered.

The page is the same length as that specified by recDuration.

Input Parameters

recDuration the number of samples that should be recorded on each channel specified in recChanList.

recChanList a row vector containing the channel numbers of all channels to be recorded. Can only contain each channel number once, but the channel order is not important and does not need to include all the channels the device supports. This is the same as the order of channels returned by 'getRec'. The maximum channel number cannot be greater than that specified during initialisation.

Output Parameters

pageNumber a unique number identifying the page that has been added - use this with all other functions that query specific pages, such as 'isFinished'.

D.2.10 pause

```
[currentState] = pause({newPause})
```

Queries or updates the current pause state of the utility. If no parameter is supplied then just returns the current pause status, otherwise returns the status after applying the change to newPause.

Input Parameters

newPause *(optional)* the new state of the utility: 1 to pause or 0 to unpause the stream. This can be either a scalar or logical value. If newState is the same as the current state of the utility, no change occurs.

Output Parameters

currentState the state of the utility (including the update to newPause if newPause is specified): 1 if the utility is paused or otherwise 0.

D.2.11 block

```
[completionState] = block({pageNumber})
```

Waits for the specified page to finish or, if no pageNumber is supplied, waits until all pages have finish. Note that the command returns immediately if the utility is paused!

This uses very little processing power whilst waiting for the page to finish, although as a result will not necessarily return as soon as the page specified finishes. For a faster response to pages finishing use the ‘isFinished’ command in a tight while loop within MATLAB, such as

while(playrec(‘isFinished’, pageNumber) == 0) end

This will run the processor at full power and will be very wasteful, but it does reduce the delay between a page finishing and the MATLAB code continuing, which is essential when trying to achieve very low latency.

Input Parameters

pageNumber *(optional)* the number of the page to wait until finished

Output Parameters

completionState 1 if either pageNumber is a valid page and has finished being processed or pageNumber was not specified and all pages have finished being processed. Note that page validity refers to when the function was called and so now the page has finished it may no longer be a valid page.

0 if the stream is currently paused and neither return values of 1 or -1 apply.

-1 if the specified page is invalid or no longer exists. This includes pages that have automatically been condensed, and hence have finished.

D.2.12 isFinished

```
[completionState] = isFinished({pageNumber})
```

Indicates if the specified page is finished or, if no pageNumber is supplied, indicates if all pages are finished.

Input Parameters

pageNumber *(optional)* the number of the page being tested

Output Parameters

completionState 1 if either pageNumber is a valid page that has finished being processed or pageNumber was not specified and all pages have finished being processed.

0 if either pageNumber is a valid page that has not finished being processed or pageNumber was not specified and not all pages have finished being processed.

-1 if the specified page is invalid or no longer exists. This includes pages that have automatically been condensed, and hence have finished.

D.2.13 getRec

```
[recBuffer, recChanList] = getRec(pageNumber)
```

Returns all the recorded data available for the page identified by pageNumber. If the page specified does not exist, was not specified to record any data, or has not yet started to record any data then empty array(s) are returned. If the page is currently being processed, only the recorded data currently available is returned.

Input Parameters

pageNumber used to identifying the page containing the required recorded data

Output Parameters

recBuffer a MxN matrix where M is the number of samples that have been recorded and N is the number of channels of data

recChanList a 1xN vector containing the channel numbers associated with each channel in recBuffer. These channels are in the same order as that specified when the page was added.

D.2.14 delPage

```
[completionState] = delPage({pageNumber})
```

Deletes either the specified page or, if no pageNumber is supplied, deletes all pages. Pages can be in any state when they are deleted - the do not have to be finished and they can even be deleted part way through being processed without any problems (in this case the utility will automatically continue with the next page in the page list). If a problem is encountered whilst deleting a page

Input Parameters

pageNumber *(optional)* the number of the page to be deleted.

Output Parameters

completionState 0 if nothing is deleted (either there are no pages in the page list or, if pageNumber was specified, no page with the specified number exists), otherwise 1 is returned.

D.2.15 getCurrentPosition

```
[currentPage, currentSample] = getCurrentPosition()
```

Returns the instantaneous current page number and sample number within this page.

Output Parameters

currentPage the current page number, or -1 if either the utility is not initialised or no page is currently being processed (there are no pages in the list or all pages are finished).

currentSample the current sample number within currentPage, or -1 if currentPage is also -1. This is only accurate to framesPerBuffer samples, as returned by 'getFramesPerBuffer'

D.2.16 getLastFinishedPage

```
[lastPage] = getLastFinishedPage()
```

Returns the page number of the last finished page still resident in memory. Due to automatic condensing/removal of pages that are no longer required, such as those with just play data after they have finished, this may not be the most recent page to have finished. Put another way, this returns the page number of the last finished page in the pageList returned by 'getPageList'.

Output Parameters

lastPage pageNumber of the most recently finished page still resident in memory.

D.2.17 getPageList

```
[pageList] = getPageList()
```

Returns a list of all the pages that are resident in memory. The list is ordered chronologically from the earliest to latest addition.

Due to automatic condensing/removal of pages that are no longer required, such as those with just play data after they have finished, this will not be a complete list of all pages that have ever been used with the utility.

Output Parameters

pageList a 1xN vector containing the chronological list of pages, where N is the number of pages resident in memory.

D.2.18 getFramesPerBuffer

```
[framesPerBuffer] = getFramesPerBuffer()
```

Returns the number of frames (samples) that are processed by the callback internally within the utility (ie the length of each block of samples sent by the utility to the soundcard). This is either the value specified when using ‘init’, or the default value if the optional parameter was not specified in ‘init’.

Output Parameters

framesPerBuffer the number of frames returned by the utility internally during each callback, or -1 if the utility is not initialised.

D.2.19 getSampleRate

```
[sampleRate] = getSampleRate()
```

Returns the sample rate that was specified when using ‘init’.

Output Parameters

sampleRate the current sample rate or -1 if the utility is not initialised.

D.2.20 getStreamStartTime

```
[streamStartTime] = getStreamStartTime()
```

Returns the unix time when the stream was started (number of seconds since the standard epoch of 01/01/1970).

This is included so that when using the utility to run experiments it is possible to determine which tests are conducted as part of the same stream, and so identify if restarting the stream (and hence the soundcard in some scenarios) may have caused variations in results.

Output Parameters

streamStartTime time at which the stream was started (in seconds since the Epoch), or -1 if the utility is not initialised.

D.2.21 getPlayDevice

```
[playDevice] = getPlayDevice()
```

Returns the deviceID (as returned by ‘getDevices’) for the currently selected output device.

Output Parameters

playDevice the deviceID for the play device or -1 if no device was specified during initialisation or the utility is not initialised.

D.2.22 getPlayMaxChannel

```
[playMaxChannel] = getPlayMaxChannel()
```

Returns the number of the maximum play channel that can currently be used. This might be less than the number of channels that the device can support if a lower limit was specified during initialisation.

Output Parameters

playMaxChannel the maximum play channel number that can currently be used, or -1 if either no play device was specified during initialisation or the utility is not initialised.

D.2.23 getRecDevice

```
[recDevice] = getRecDevice()
```

Returns the deviceID (as returned by 'getDevices') for the currently selected input device.

Output Parameters

recDevice the deviceID for the record device or -1 if no device was specified during initialisation or the utility is not initialised.

D.2.24 getRecMaxChannel

```
[recMaxChannel] = getRecMaxChannel()
```

Returns the number of the maximum record channel that can currently be used. This might be less than the number of channels that the device can support if a lower limit was specified during initialisation.

Output Parameters

recMaxChannel the maximum record channel number that can currently be used, or -1 if either no record device was specified during initialisation or the utility is not initialised.

D.2.25 resetSkippedSampleCount

```
[ ] = resetSkippedSampleCount()
```

Resets the counter containing the number of samples that have been 'missed' due to no new pages existing in the page list. See the help on 'getSkippedSampleCount' for more information.

D.2.26 getSkippedSampleCount

```
[skippedSampleCount] = getSkippedSampleCount()
```

Returns the counter containing the number of samples that have been 'missed' due to no new pages existing in the page list. The term 'missed' is specifically referring to the case where multiple consecutive pages are used to record a continuous audio stream (and so input samples are missed), but is the same also for output samples because the input and output samples within a page are always processed simultaneously.

This value is incremented by one for every frame (ie one sample on every input/output channel) of data communicated between the utility and soundcard that occurred whilst there were no new pages in the page list. Using this it is possible to determine, from within MATLAB, if any glitches in the audio have occurred through not adding a new page to the page list before all other pages have finished, such as in the case where the code within MATLAB is trying to play/record a continuous stream.

Output Parameters

skippedSampleCount the number of samples, since last being reset, that have occurred when there are no more pages in the pageList or -1 if the utility is not initialised

Appendix E

Equipment List

Throughout this project the following equipment was used:

Personal Computer A Dual Pentium 4 2.8 GHz with 512 MB RAM running Windows XP Professional SP2. A M-Audio Delta 1010LT soundcard (driver version 5.10.00.0051) was used for all audio input and output with both inputs and outputs set to "consumer" signal levels (-4dBu) with all bass management turned off. Software used included MATLAB (version 7.1.0.246 R14 Service Pack 3) and Microsoft Visual Studio 2005 Professional Edition (version 8.0.50727.42).

Amplifier A Marantz AV Surround Receiver SR4300, using the front and surround channels through the direct signal 6.1 channel input connections. Volume set to -30dB during all tests.

Loudspeakers Audio Pro Focus SA-2 with grill removed and all distance measurements made relative to the horizontal centre of the front of the cabinet.

Microphones

TCM110 Tiepin Microphone manufactured by AV-JEFE with the following quoted specification[2]:

Element Back electret-condenser

Polar Pattern Omni-directional

Frequency 50 Hz $\sim$ 18 kHz

Sensitivity -65 dB $\pm$3 dB

CM4050 SoundField Microphone and Control Unit (serial number 615) manufacture by Calrec. Gain set to 40 dB, and all soundfield manipulation disabled.

Microphone pre-amplifier For the TCM110 and one loop back signal an Allen & Heath GL2000 was used. In both cases, both the channel equaliser and high-pass filter were disabled, and the respective group faders were set to +5 dB. The loop back channel used the line input and had +20 dB of gain whilst the TCM110 channel used the microphone input with a gain of +60 dB and a fader value between -5 dB and +5 dB dependant on the loudspeaker-microphone separation.

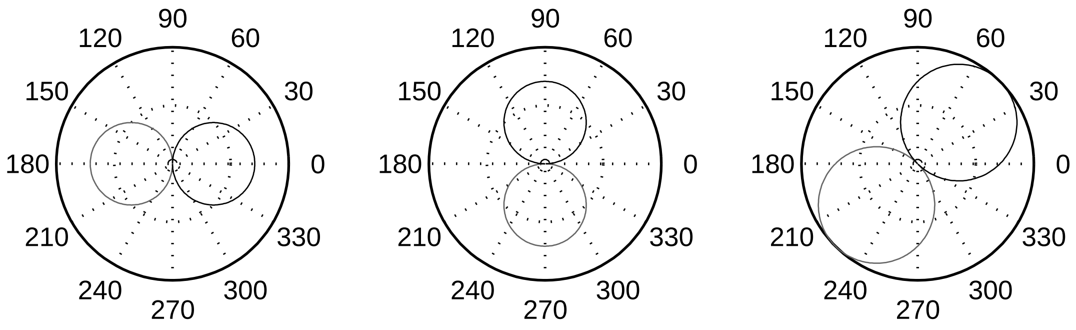
Appendix F

B-Format Signal Manipulation

When the B-Format signals used to represent a soundfield at a single point in space, as described in section 2.2, are summed together in different proportions there are two important effects. The first of these allows the soundfield to be rotated into any orientation required whilst the second allows virtual microphone signals to be generated.

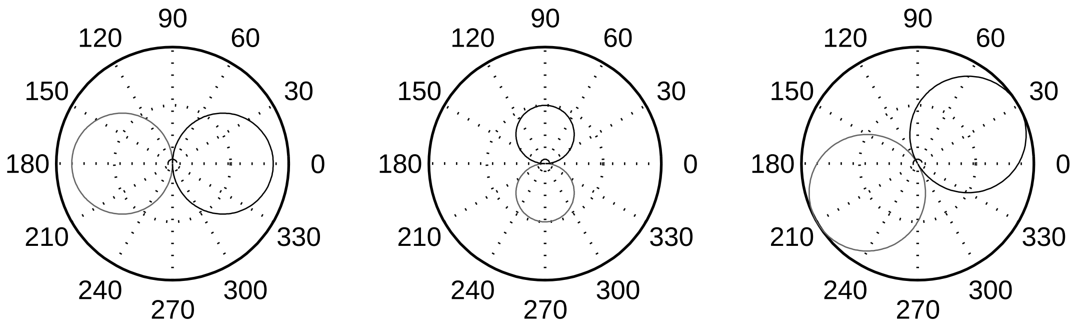
F.1 Soundfield Rotation

By summing together the first-order spherical harmonic components (X, Y and Z) in different proportions it is possible to rotate, twist and tumble a recorded soundfield such that it can be given any orientation within space[21]. This occurs because the summing of weighted figure-of-8 polar patterns results in a single figure-of-8 pattern with a different orientation, as shown in figures F.1 and F.2. These both show rotation within a plane, although through the use of all three first-order components 3-dimensional positioning of the figure-of-8 polar pattern can be achieved. Thus, by creating three new orthogonal signals (X', Y' and Z') a new B-Format signal can be created with the front pointing in any direction compared to that of the original.



(a) X, scaled by cos(45°). **(b)** Y, scaled by sin(45°). **(c)** Resulting sum of signals.

Figure F.1: Creation of a figure-of-8 polar pattern at 45° by summing two weighted polar patterns. Black lines represent in-phase signals whilst grey lines represent out-of-phase signals.



(a) X, scaled by cos(30°) **(b)** Y, scaled by sin(30°) **(c)** Resulting sum of signals.

Figure F.2: Creation of a figure-of-8 polar pattern at 30° by summing two weighted polar patterns. Black lines represent in-phase signals whilst grey lines represent out-of-phase signals.

Rotating the soundfield about the Z-axis by θ is achieved using

$$\begin{bmatrix} X' \\ Y' \\ Z' \end{bmatrix} = \begin{bmatrix} \cos\theta & -\sin\theta & 0 \\ \sin\theta & \cos\theta & 0 \\ 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} X \\ Y \\ Z \end{bmatrix}. \tag{F.1}$$

Similarly, tilting by ϕ about the x-axis is given by

$$\begin{bmatrix} X' \\ Y' \\ Z' \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 \\ 0 & \cos\phi & -\sin\phi \\ 0 & \sin\phi & \cos\phi \end{bmatrix} \begin{bmatrix} X \\ Y \\ Z \end{bmatrix}, \tag{F.2}$$

and tumbling by ψ about the y-axis by

$$\begin{bmatrix} X' \\ Y' \\ Z' \end{bmatrix} = \begin{bmatrix} \cos\psi & 0 & -\sin\psi \\ 0 & 1 & 0 \\ \sin\psi & 0 & \cos\psi \end{bmatrix} \begin{bmatrix} X \\ Y \\ Z \end{bmatrix}. \tag{F.3}$$

Therefore, to implement rotation about the z-axis followed by tilting about the x-axis, these operations can simply be implemented sequentially, giving

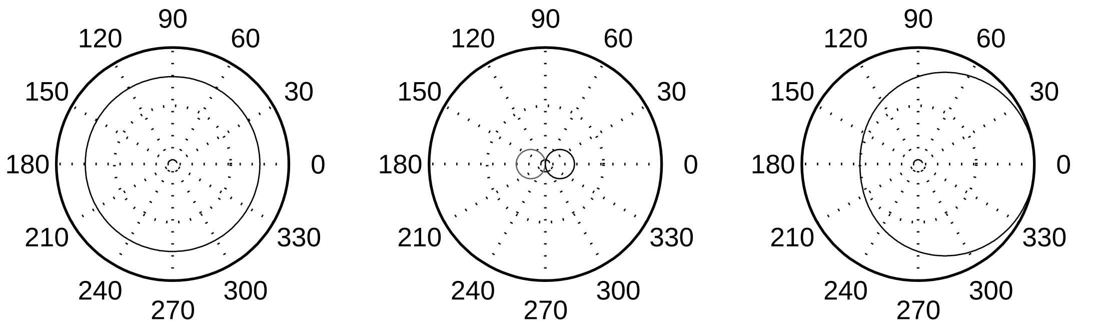
$$\begin{bmatrix} X' \\ Y' \\ Z' \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 \\ 0 & \cos\phi & -\sin\phi \\ 0 & \sin\phi & \cos\phi \end{bmatrix} \begin{bmatrix} \cos\theta & -\sin\theta & 0 \\ \sin\theta & \cos\theta & 0 \\ 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} X \\ Y \\ Z \end{bmatrix}.$$

F.2 Virtual Microphones

An alternative to summing only the first-order spherical harmonics is to include the zeroth-order spherical harmonic (W). This not only allows the orientation of the resulting signal to be changed, but also the polar pattern—ranging from that of an omni-directional microphone (only includes the W signal) to a figure-of-8 microphone (not including the W signal). This is controlled by the directivity factor D in

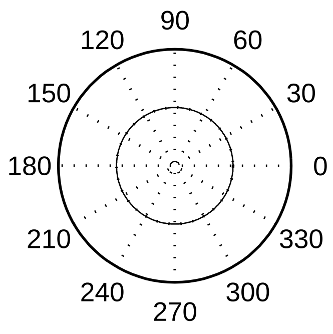
$$V(\vec{r}) = \frac{1}{2}\left[\sqrt{2}(2-D)W + D\left(r_x X + r_y Y + r_z Z\right)\right] \tag{F.4}$$

where $\vec{r}$ is a unitary vector pointing in the direction of the required virtual microphone pattern, $V(\vec{r})$ [9]. (Note that the $\sqrt{2}$ factor is included as the B-Format standard includes a 3 dB reduction in W compared to X, Y, and Z.) D can take any value in the range 0 to 2, with 0.5 producing a sub-cardioid polar pattern (see figure F.3), 1.0 producing a cardioid polar pattern (see figure F.4) and 1.5 producing a hyper-cardioid polar pattern (see figure F.5).

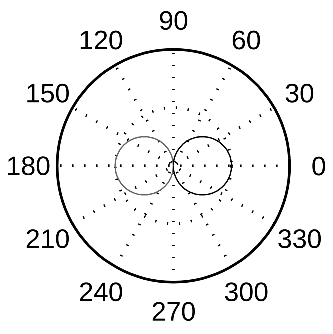


(a) W, scaled by 0.75. **(b)** X, scaled by 0.25. **(c)** Resulting sum of signals.

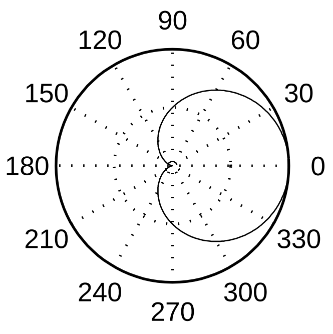
Figure F.3: Creation of a virtual sub-cardioid microphone polar pattern using a directivity, D, of 0.5. Black lines represent in-phase signals whilst grey lines represent out-of-phase signals.



(a) W, scaled by 0.5.

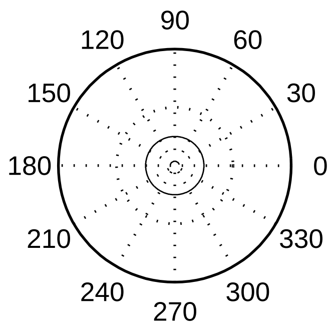


(b) X, scaled by 0.5.

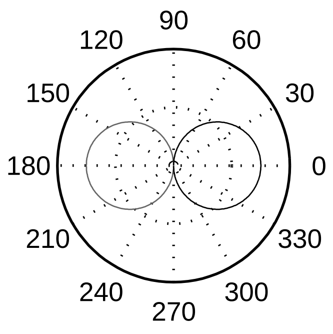


(c) Resulting sum of signals.

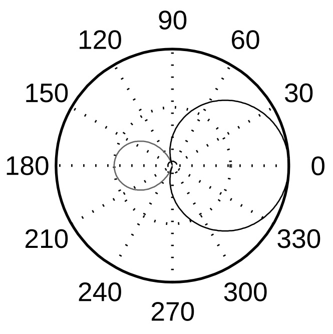
Figure F.4: Creation of a virtual cardioid microphone polar pattern using a directivity, D, of 1.0. Black lines represent in-phase signals whilst grey lines represent out-of-phase signals.



(a) W, scaled by 0.25.



(b) X, scaled by 0.75.



(c) Resulting sum of signals.

Figure F.5: Creation of a virtual hyper-cardioid microphone polar pattern using a directivity, D, of 1.5. Black lines represent in-phase signals whilst grey lines represent out-of-phase signals.

Appendix G

CD Contents

The CD accompanying this project contains three folders: `Documentation`, `MATLAB_code` and `playrec_utility`.

G.1 Documentation

This folder contains an electronic copy of this report and a copy of the initial report submitted in February 2006.

G.2 MATLAB_code

This folder contains the final MATLAB code used during the project. It is divided into the following subfolders:

BFormatProcessing contains files to manipulate B-Format signals including creating them from a mono source, rotating, tilting and tumbling them, and decoding them to generate loudspeaker feeds. This also contains a GUI to demonstrate decoding using an encoded mono source, and a file to allow WAVE-FORMAT-EXTENSIBLE files containing B-Format signals to be decoded in a block-wise manner.

CoreFunctions contains core files that were used throughout the project under different circumstances. Many of these are 'helper' files that were used to aid figure plotting. Other files include one to window a signal and another to return the index of specific values (needles) within an array (haystack). Files to implement linear and circular correlation and convolution are also included, allowing other files to operate without the need for the Signal Processing Toolbox.

IR_ExcitationSignals contains all files required to create and process the four different excitation signals used: MLS, IRS, OATSP and logarithmic sweep.

IR_PeakDetection contains files used to detect the position and value of peaks within an Impulse Response, using either peak sample, quadratic interpolation or cubic interpolation. A file to trim an IR around its peak, including support for multi-channel IRs, is also included.

PresentationGUIs contains GUIs used during a presentation given at the Department of Speech, Music and Hearing (TMH), at the Royal Institute of Technology (KTH), Stockholm on 23 May 2006.

Test_and_Analysis contains all files used to manage and manipulate the data associated with all tests conducted.

UsefulFiles contains the file `wavexread.m` created by Sylvain Choisel and based on the MATLAB file `wavread.m`. This file is used when playing WAVE-FORMAT-EXTENSIBLE files containing B-Format signals and is also available online at `http://acoustics.aau.dk/~sc/matlab/wavex.html`.

Some of the files included are dependant on files contained in other subfolders.

G.3 playrec_utility

This folder contains all of the source code used to compile the playrec utility. A sample Microsoft Visual Studio Solution containing two projects is also included. The first project is for the playrec utility whilst the second is for the test file shown in listing C.1. This folder is divided into the following subfolders:

asiosdk2 For the sample Visual Studio Projects to compile the Steinberg ASIO SDK must be downloaded and extracted to this folder. The SDK can be downloaded from the 3rd Party Developers section of the Steinberg website (`http://www.steinberg.net/`).

pa_ASIO_dll contains the sample Visual Studio Solution and three subfolders:

> **playrec** containing the utility specific source code in the four files `mex_dll_core.c`, `mex_dll_core.h`, `pa_dll_playrec.c` and `pa_dll_playrec.h`;
>
> **release** containing the files generated from compiling both the utility and the test project, including the `.dll` files used by MATLAB;
>
> **testProject** containing the test project source code in `pa_dll_test.c`.
>
> To use the `.dll` files in the `release` folder within MATLAB either add the folder's path to the path list or set it as the current directory. The playrec utility can then be used by typing `playrec` at the MATLAB command prompt whilst the test project can be used by typing `testProject`.

portaudio_v18_1 contains the PortAudio SDK V18_1 as downloaded from their website (`http://www.portaudio.com/download.html`). This is required to compile either of the included projects.