Assessing the Spatial Diffusivity of Sound Fields in rooms using Ambisonic techniques

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A thesis submitted in fulfilment of the requirements for the degree of Doctor of Philosophy

Faculty of Architecture, Design and Planning
AUTHOR’S DECLARATION

This is to certify that:

I. this thesis comprises only my original work towards the Doctor of Philosophy Degree
II. due acknowledgement has been made in the text to all other material used
III. the thesis does not exceed the word length for this degree.
IV. no part of this work has been used for the award of another degree
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Abstract

This thesis explores the means of objectively measuring diffuse sound fields contained within music performance auditoria. Although the diffuse field is considered to be an important component of the reverberant sound field there is currently no widely accepted method for its measurement.

A review of methods shows that attempts to characterize the field may be divided into those methods that seek to directly measure the state of the field and those that indirectly indicate the existence of the state.

The primary focus of this thesis is the application of Ambisonic techniques to capture the spatial aspects of the sound field.

Initial work explores the rotation of a directional microphone in three measurement spaces. The results and modeling in idealised simulated sound fields indicate that the method may have some efficacy.

The method is extended through the application of signal processing to the output of an Ambisonic microphone array. The method is tested firstly in a reverberation room that is modified progressively to produce a series of room states with incrementally increasing reverberation time.

The extents of the measurement system were tested by measuring the degrees of diffusivity reached in a reverberation room. Diffusing panels were progressively added in the expectation the increases in diffusivity would be detected. The measurement was carried out in conjunction with standard absorption coefficient measurements outlined in Appendix A of ISO 354. Comparison was made between the measured field and the standard method for achieving a diffuse field in a reverberation room test facility.

The final stage attempts to find correlation between physical measures of diffuse fields and listener’s subjective assessment of those fields. To that end a paired comparison test was conducted where listeners were presented music samples rendered through simulated halls where the scattering coefficients and consequently the sound field diffusivity was varied. Subjects were asked to choose which pair they preferred.
Acknowledgements

I wish to convey my gratitude to Andrew and Jean Wong for their provision of the Andrew and Jean Wong fellowship. The award allowed me to conduct a listening tour of many of the great concert halls of the world. Extending my listening experience beyond the relatively small pool of concert halls in Australia has assisted me enormously over the years.

Of course I’d like to thank my supervisors Densil Cabrera and Fergus Fricke. In particular I am grateful to Fergus for staying on as my associate supervisor long after he retired. I decided to embark on a PhD when I heard Fergus was to retire in a few years, I’m glad he stayed around over the lengthy journey.

Craig Jin and Andre van Schaik at the CARLab were most helpful in the early stages in assisting me with Matlab coding. More recently, Andrew Wabnitz, Nicolas Epain and Roman Kosobrodov have been exceptionally helpful in providing advice during the setup of the subjective tests within CARLab.

Luis Miranda was most helpful in providing the Max/MSP system for subjective tests.

David Gilfillan and Densil Cabrera were most helpful in conducting extra measurements during their modification of the reverb room.

Thanks also to Peter Alway for providing access to the National Acoustics Laboratory for measurement of the Soundfield mic characteristics.

The second reverb room measurement was conducted in conjunction with David Spargo. It was a pleasure to work with a colleague with a good ear who’s attention to detail far surpasses mine. The coffee breaks were good to.

Of course I have to thank my wife Paddy and my son George who’s patience has seemed endless. Having a child in the last year of the thesis should be a recipe for disaster but George has proven to be entirely charming throughout.

No doubt that will end…….. but finally so has this thesis.
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Chapter 1

Introduction

Reverberation time has been the primary measure of a room’s acoustic character for over a century. Sabine’s theory for the calculation of Reverberation Time (Sabine 1900/1964) established the way forward for designers to create rooms for which the acoustics of the space could be predicted with some accuracy prior to construction. Usually this was achieved through specification of room volume and the amount of absorptive material within in the room. It was found, however, that variance between the predicted and measured values could, in part, be attributed to the basic assumption in Sabine’s work that the sound field is diffuse (Eyring, 1930). The field created in the designed room rarely approaches this ideal condition and as a result the assumption that all absorbing surfaces may be averaged was shown to produce errors. Eyring suggested that Sabine’s calculation was workable for reasonably ‘live’ rooms but was inadequate for calculation of the Reverberation Time in highly absorbent or ‘dead’ rooms. The inaccuracy of the results produced by this assumption can be significant even for highly reflective ‘live’ rooms. This problem has been reflected in the use of reverberation chambers to measure the absorption coefficient of materials. To this day the method of assessing whether a field is diffuse in a reverberation chamber is through assessment of the variation in the measured absorption coefficient for different placements of the sample in the space. The room is assumed to contain a diffuse field when the absorption coefficient remains constant, independent of placement of the sample in the room (International Organisation for Standardisation, 2008). There is no clear criterion for the direct measurement of the diffusivity of the field in the test facilities that are used to measure the absorption coefficients.
Although reverberation time is still considered to be an important room criterion there is a growing number of other measures considered important in assessing a room’s quality. Various authors have shown a link between subjective and objective room criteria (Beranek, 2004; Barron, 1993), yet despite some recognition that diffusivity of a room sound-field is an important measure for assessing listener envelopment there is currently no widely accepted objective measure for soundfield diffusivity.

This thesis will explore a means of objectively measuring the degree to which a sound field is diffuse. The primary interest is in its application to room acoustics and as such the measure will be tested against people’s subjective response to simulated auditorium sound fields with various degrees of diffusion.

1.1 What is diffusion?

In the text *Music Acoustics and Architecture* (Beranek, 1962), numerous subjective terms are introduced which are deemed to be important factors in assessing what he refers to as the Musical-Acoustical Quality for music performance spaces. There are eighteen subjective factors listed. Beranek then sought to find some correlation between these subjective terms and objective measures applicable to performance venues. The process involved the gathering of technical, structural and acoustical data for 54 halls in various parts of the world. In addition to these objective measures he sought the opinion of a range of musicians, including conductors, and music critics. The interviewees were asked to rank the halls that they knew. Based on the interview results and his own experience Beranek created a ranking scheme for the 54 halls ranging from A+ (Excellent) to C+ (Fair). A number of positive attributes were derived with the intention of finding objective values that would produce good correlation to the halls considered to be A+. These attributes were; Intimacy, Liveness, Warmth, Loudness of Direct Sound, Loudness of Reverberant Sound, Diffusion, Balance and Blend and Ensemble. Of these, many had objective measures that related to the attributes, others did not.
In the four decades between this original study and the production of the text *Concert Halls and Opera Houses* (Beranek, 1996) objective measures had been developed that closely correlated with the attributes originally proposed by Beranek.

The one exception is Diffusion. In the original text, the means of assessing the diffuseness of the performance space, the Surface Diffusivity Index, is based on a subjective visual assessment of the hall surfaces with the assumption that irregular surfaces produce a more diffuse sound in the space.

Prior to and in the intervening years there have been many attempts to develop a measure of the diffuseness of a sound field. These measures are discussed in Chapter 2. Arguably, there is still no widely accepted means of doing so. This author suggests that only when we have an objective measure are we able to assess to what degree a diffuse sound field is a desirable aspect of sound in performance spaces. Obviously in order to measure the state we must first understand what constitutes a diffuse field. To do so the following sections will explore methods of defining the sound within an enclosure with the aim of building an understanding of how we may approach the definition of a diffuse sound field.

### 1.2 Sabine’s definition

The obvious point of departure is Sabine’s work in deriving the reverberation time criterion (Sabine, 1900/1964). In his initial investigations of the Jefferson Physical Laboratory he found that the decay time of sound in the room was almost entirely independent of position in the room. Following extensive testing of his ‘hyperbolic law’ in a range of rooms he concluded that “[...] the efficacy of an absorbent is independent of its position when the problem under consideration is reverberation, and that the sound, dispersed by regular and irregular reflection and diffraction, is of nearly the same intensity at all parts of the room soon after the source has ceased [...]” We may view this as an early definition of a diffuse field.
The flaw in Sabine’s reverberation time formula started to be realised as practitioners began designing spaces based on Sabine’s calculations. As Eyring (1930) pointed out, the assumption in Sabine’s calculation of reverberation time, where the average absorption coefficient is calculated from the sum of absorbent areas, implies that each area of material with the same absorption coefficient will absorb sound equally. This implies that the sound impinging on the material is coming from all directions equally.

When absorption coefficients were measured in newly constructed reverberation chambers it was found that there was often a discrepancy between expected and measured results. The most obvious example of this was where the measured absorption coefficients are greater than one. In a paper exploring a range of methods for calculating the reverberation time (Eyring, 1933) it was shown that, where the absorption coefficient is high and the sample is large, it is not possible to maintain a diffuse state in a reverberant chamber. Consequently the sound in the room does not impinge equally on all elements in the room resulting in an error in the averaging of the total room absorption. This caused Eyring and Andree (1932) to separately warn that the diffuse state may not be assumed without ‘good evidence’.

1.3 The wave equation

In the work of Strutt and Schuster and Waetzmann (Knudsen, 1932), reports suggested that the geometrical approach used by Sabine, Eyring and others was “unsatisfactory” proposing instead that the wave equation should be applied to the enclosure, with the reverberation formula derived by the introduction of boundary conditions. Here the proposal is that the enclosure is excited by a great many resonances, the number of which is a function of the room’s geometry and the frequency of the sound radiated into it. Morse and Bolt (1944) propose that a room may range from simple modal behaviour through to an ergodic state.
1.4 The ergodic state

This term ergodic is of interest in defining the diffuse state. Based on the work of Sinai (1970), Joyce (1975) used particle theory to propose that a phonon $x$, reflecting in an enclosure, will eventually have spent equal time in each position, travelling in every direction within that enclosure. This is referred to as an ergodic state. Further to this, Joyce suggests that, at equilibrium, an equal number of phonons on random paths will cross any point within a second. This is defined as an isotropic field. (It should be noted that Joyce attributes the above definition to Maxwell in 1879.) From this we can interpret Joyce as saying that an isotropic field is one in which the field is ergodic within a particular time frame.

1.5 The isotropic state

This leads us to another definition – the isotropic state, which may be simply defined as; moving equally in all directions. This definition was adopted by Beranek (1954) in defining the diffuse state as one where, at any moment, sound is travelling in a room in all directions equally.

This of course raises the question of the duration of a ‘moment’ but follows from the clearer definition made by Joyce.

From the above we can see that a definition of a diffuse state is possibly made unclear by the two conflicting ways of approaching room acoustics the modal response versus the statistical estimation of the field. This confusion may be cleared by reference to Morse and Bolt (1944) or Kuttruff.
1.6 Modal and diffuse room states

In his text *Room Acoustics*, Kuttruff (1973) explores sound in rooms by starting with the simplest sound environment, the free field, and then progressively adding complexity by first exploring the effect of sound arrival at a barrier and following through to subjective responses to the combination of multiple sound fields. The ‘correct’ solution is presented as the wave acoustics approach which begins with the free propagation of sound in a gas and then bounds a volume of gas with surfaces forming an ideal rectangular enclosure. The normal modes of resonance or eigenfrequencies created by the reflection between boundaries between 0 and an upper frequency ($f$) limit may be calculated from;

$$N_f = \frac{4\pi}{3} V \left( \frac{f}{c} \right)^3 + \frac{\pi}{4} S \left( \frac{f}{c} \right)^2 + \frac{L}{8} \frac{f}{c}$$

where; $c =$ speed of sound (m/s)

$V =$ volume of enclosure (m$^3$)

$S =$ surface area of enclosure (m$^2$)

$L =$ length of enclosure edges (m)

The average spacing of adjacent eigenfrequencies may be found from;

$$\delta f = \frac{c^3}{4\pi V f^2}$$

where; $\delta f$ is the average spacing of eigenfrequencies in Hz.

Kuttruff (2007) examines two limiting cases, one where the spacing between eigenfrequencies is large compared with the average resonance half-power bandwidth, and the opposite where many eigenfrequencies occur within a resonance half-power bandwidth.

In the first case the transfer function of the room will consist of clearly distinguishable resonances whereas the second case will produce a response curve with overlap between adjacent frequencies. Kuttruff assumes, for this case, that the contribution from each reflection has random phase so it follows that the sound pressure is randomly distributed. The first case will occur in the low frequency range where the individual room resonances are clearly defined and detectable. At higher frequencies the second case occurs, where strong overlap of the resonant modes results in an inability to excite individual modes.
The frequency at which ‘significant’ modal overlap occurs is deemed to be one where there are at least three resonant modes within the half-power bandwidth of one of those resonances. That frequency is known as the ‘Schroeder frequency’ (Schroeder, 1954/1987); 

\[ f \approx 2000 \sqrt{\frac{T}{V}} \]

where; \( T \) = reverberation time (sec)

As Morse and Bolt indicated, any room may be seen to ‘act’ in different ways depending on the frequency range of sound introduced into it. Above the Schroeder frequency the room may be viewed as approaching a diffuse state whereas below that frequency the room is dominated by strong resonant room modes.

Waterhouse and Cook (1976) take another view of this in their comparison of diffuse sound fields where they show that the Eigenmode model and the free-wave model will produce the same space correlation results. They propose that the different models may be more appropriate for different room states but that they converge as the energy density becomes uniform.

This was confirmed through the application of Asymptotic Modal Analysis (Kubota and Dowell, 1992), which demonstrated that the results for the wave theory approach those of statistical acoustics where the number of modes in the room become large.

It can be seen by applying the calculation for resonant frequencies, above, that for a room of reasonably large size the number of resonant frequencies becomes very large, making it almost impossible to apply the wave equation due to the number of calculations required. Treatment of a room in such circumstances may be simplified by applying the geometric acoustics proposed by Sabine. In doing so the phase interactions produced by locally reactive elements in the room are ignored.

This is an important point to note; that the phase interaction between the incident waves is ‘averaged out’ in that the number of in-phase additions is equal to the number of out of phase cancellations and all partial additions and cancellations are of equal magnitude. In this way any effects of phase interaction in a complex sound field are ignored.
This is deemed acceptable where the sound in the room is itself sufficiently wide bandwidth, that is, non-tonal. In such circumstances it is acceptable to simply sum the energy of the arrivals and consequently to treat the various sound reflections as rays. In doing so the assumption is that the interaction of the sound rays occurs in circumstances where they are “mutually incoherent”. This idea of an incoherent field will be considered later.

1.7 Kuttruff’s definition

Kuttruff (1973) considers this case where the sound incident on a room surface is arriving from all directions as the diffuse state containing the following characteristics:

[...] the amplitudes of the incident waves .. distributed uniformly over all possible directions of incidence in such a way that from each element of solid angle the same amount of energy arrives on the wall per second and per unit of area element perpendicular to the respective direction. Furthermore we can assume that the phases of the elementary waves are distributed at random so that the interference effects can be neglected and we can simply add the energies of the waves impinging simultaneously.
1.8 Defining the diffuse state

In summary, the terms that are used to define the diffuse state are:

*Sabine* (from Eyring) - If "the duration of audibility of the residual sound is nearly the same in all parts of an auditorium," if it is "nearly independent of the position of the source," and if "the efficiency of an absorbent in reducing the residual sound is, under ordinary circumstances, nearly independent of its position," the sound will be at least approximately diffuse.

*Beranek* - at any moment, sound is travelling in a room in all directions equally.

*Kuttruff* - the amplitudes of the incident waves are distributed uniformly over all possible directions of incidence in such a way that from each element of solid angle the same amount of energy arrives per second and per unit of area element thought perpendicular to the respective direction. Furthermore we can assume that the phases of the elementary waves are distributed at random so that the interference effects can be neglected and we can simply add the energies of the waves impinging simultaneously.

*Schultz* (1971) sought to encapsulate the various definitions;

(i) In a diffuse sound field there is uniform total (potential plus kinetic) energy density at all points in the room and each volume element radiates equally in all directions.

(ii) In a diffuse sound field there is equal probability of energy flow in all directions and random angle of incidence of energy upon the boundaries of the room.

(iii) A diffuse sound field comprises a superposition of an infinite number of plane progressive waves, such that all directions of propagation are equally probable and the phase relations of the waves are random at any given point in the space.
The terms above are drawn together to establish the criteria against which a sound field may be assessed for diffuseness;

1) The amplitude of incident waves is distributed evenly over all angles of incidence.
   - contained within the definitions provided by Beranek, Kuttruff and Schultz.

2) Equal energy is arriving from each direction over a designated period of time.
   - Kuttruff defines a period of time, it is implied by Beranek and Schultz.

3) The phase relationship of the incident waves is randomly distributed so that phase effects may be ignored.
   - contained within the definitions provided by Kuttruff and Schultz.

These definitions provide a clear indication of the properties of a diffuse field. As such they should be directly measurable. The definition provided by Sabine is an implicative measure that does not directly measure the field but implies that a diffuse field is present when other, measurable conditions, are met.

In this thesis a measurement system is developed that attempts to directly measure the sound field encapsulating the three criteria above.

The Sabine definition is used in the standard for measuring the absorption coefficient of a material in a reverberation room. In Chapter 5 this method is tested against the developed measurement system.

1.9 What is a desirably diffuse listening condition?

We began by indicating that a diffuse field is considered to be an important listening condition for music performance auditoria but we find that most of the literature on the diffuse field focuses on achieving the state in measurement facilities. This is unsurprising in that the Sabine calculation, and subsequent variations, for measurement of reverberation time form the basis of any approach to designing the aforementioned facilities. It should also be unsurprising that little work has been conducted into what constitutes a desirable diffuse listening condition when there is little agreement on how to measure the diffuse state. This is best illustrated by, the following, reverberation room measurement method.
Reference to ISO354 (International Organisation for Standardisation, 2008), the standard for measurement of absorption coefficient of materials in reverberation rooms shows the method for establishing the required diffuse sound field consists of an iterative method changing room conditions until the desired criterion is met.

It is clear that in order to understand what is a desirable listening condition we must first find a useful method for measuring sound field diffusivity. The potential means of doing so will be considered in the following chapter. The aim of achieving steady state diffusion in a reverberation chamber, however, may not be completely desirable in a performance space.

Obviously a steady state field is not the condition that we experience in most music performance rooms where the field may be characterised as primarily transient. (Gregorian chant performed in large cathedrals may be an exception.) This further illustrates the lack of detailed criteria for establishing a desirably diffuse state in auditoria.

Barron (1993), for example, states;

For the reverberant sound, it is important to perceive some sound both from the side and behind. A diffuse sound field satisfies these requirements and in design terms requires space above the audience and possible paths for reflections to arrive at listeners from most directions.

There is an implication of some separation between the direct and diffuse sound. So already it is clear that some temporal aspects of the diffuse soundfield must form part of this examination. Kuttruff, in his definition of the diffuse field, proposes the state be measured over the duration of one second. Beranek (2004) separates the diffuse field into early and late, with diffusion of the early sound reducing the affect of “acoustic glare” and late diffusion producing a “homogenised” sound. These temporal aspects provide a starting point for further exploration.

This distinction between the early and late reverberant sound has been illustrated by Moorer (1979) in his experiments with the simulation of reverberant fields. In attempting to develop realistic sounding reverberation simulators the impulse of the models was compared to the impulse responses of actual halls. Moorer observed that the reverberant tails illustrated in the impulse responses were similar in character to white noise.
A Gaussian pseudo-random sequence was generated with an appropriate exponential decay envelope. A single impulse was added at the beginning, to represent the direct sound, and the simulated impulse response was convolved with ‘dead’ music samples. Moorer reported the results to be “astonishing” in their realism, compared to those of actual concert halls.

The implication of this is that in addition to the direct physical measurement of the diffuse field we may pursue other aspects of the field that may be important. Based on Moorer’s work other areas that may bear consideration are the level of incoherence of the field and the time it takes to reach a state of diffusion.
1.10 Aims and Objectives

This thesis aims to explore the following issues;

The primary focus of the work is the establishment of an objective measurement of a diffuse sound field. In exploring the means of making such a measurement the various aspects that constitute a diffuse sound field outlined above will be examined in Chapter 2. The isotropy of the field at a specific point in the room will be examined in Chapter 3. Following that the degree to which the field meets that isotropic criterion throughout the measurement space, referred to as homogeneity, will be examined in Chapters 4 and 5.

It is expected that the spectral character of the field may vary, where the field may be diffuse within some frequency range while exhibiting a more modal character at lower frequencies. Consequently a frequency dependent criterion for assessment of the field is developed in Chapter 3 and explored in Chapter 5.

As has been discussed above, there may be an argument for also assessing the development of a diffuse field over time. Therefore the temporal character of the diffuse field will be considered in Chapter 3 and expanded on in Chapter 5.

There are currently some measures that indirectly measure diffusion. The developed measurement method will be compared with existing measurements in Chapter 6.

Finally, any objective measure may not be considered effective if it does not adequately correlate with people’s subjective responses to stimuli. To this end a subjective test will be carried out, in Chapter 6, to test the efficacy of the measurement in assessing desirable listening conditions for music performance auditoria.
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Chapter 2

Methods for measuring the diffuse state – a review

In considering the methods of measuring a sound field for diffuseness, an obvious starting point is to explore methods that aim to establish an objective measure of how well a sound field meets the criterion of having the amplitude of incident waves distributed evenly over all angles of incidence, with equal energy arriving from each direction over a designated period of time. Additionally the phase relationship of the incident waves is randomly distributed so that phase effects may be ignored. That is, a measure of the isotropy of the field.

It is important to recognise that the focus of this work is on the diffuse state in auditoria. In considering the measurement of diffuseness we find that the majority of the work has focussed on the establishment of the said field in reverberation chambers for the reliable measurement of the absorption coefficient of materials. In each case the degree of variability in the measured diffuse state for different positions in the enclosure, the homogeneity of the field, provides a more comprehensive measure of the state of diffusivity of the room.

One area that has focussed on auditorium measurement has been the variability in measured decay curves. An example is Schroeder’s (1964) documentation of the variability in results in measuring the decay in Philharmonic Hall, New York. The problem was caused by the random signal used to excite the room and possibly by a non-diffuse state resulting in different modes, producing different rates of decay. The random noise problem has been resolved by the use of either pseudo-random noise or swept-sine signals but the non-linear decay curve remains an indication of a non-diffuse room state. This is an example of where a defined objective measure is used as an implicative measure for a related sound field characteristic. In exploring methods of measuring the sound field such indicative measures may provide an acceptable means of distinguishing varying states of diffuseness in a space.
In the following exploration of methods this chapter will initially consider the historical method proposed by Wente. Although this method may be seen as an indicative measure it informs two of the streams of thought that follow, the direct measurement of the sound field and measurements that imply that the field is diffuse. After that the methods that seek to directly measure isotropy of the sound field at a point in space will be explored. That will be followed by methods that sample the overall space and seek to derive a measure of homogeneity. Finally the more indicative measurements and methods that surmise that a diffuse field is present are considered.

### 2.1 Frequency Irregularity

Wente (1935) explored the characteristics of a room through application of communication engineering by treating the room as a transmission medium. He measured the transmission characteristic of a room by continuously varying the frequency of a tone radiated by a loudspeaker into the room. A microphone detected the resulting pressure level, with the output plotted by a level recorder. Where the tone was varied quickly in frequency and the recorder operated at a low speed the frequency averaging produced a ‘frequency response’ of the room that provided some indication of the support provided by the room to a range of frequencies. Of more interest were the resultant plots when the frequency was varied slowly and the recorder speed was set to high. This produces a curve that more closely represents the steady-state condition of the room. Wente plotted the results over a narrow frequency range, 900-1000Hz, showing the variability of level received by the microphone for a constant output level from the loudspeaker. His focus was on measuring the room characteristic in a way that provided more information about the room than the reverberation time measure. He proposed that the subtraction of the sum of all minima on the curve from the sum of all maxima on the curve would provide a measure of the quality of the reverberant properties of the room. He did, however, make the point that although there was some correlation between his transmission irregularity measure and the reverberation time of a range of room states the placement of absorption, loudspeaker and microphone in a room are significant influences on the result.
Hunt (1939) made the connection between the measurement of a room’s steady state response and its modes of vibration but it was Bolt and Roop (1950) who extended Wente’s original idea to create the Frequency Irregularity measure in dB/Hz.

They proposed that the $F_v$ (Frequency Irregularity) criterion is affected by room geometry and absorption and consequently may be a measure of the diffuseness of the soundfield in the room. $F_v$ was measured by plotting the transmission characteristic of a room in 25Hz bands. They were able to show, experimentally, that an irregularly shaped studio produced a smooth, relatively flat curve whereas two hard walled, rectangular spaces produced curves with one or two strong peaks. They suggest that below the peak in the curve there are progressively less room modes and consequently less peaks and dips in the response curve per measurement band. Above the peak the modal density increases resulting in a reduction in difference between the peaks and dips. Bolt and Roop recognised that it was difficult to correlate the criterion due to the large amount of data required to be collected and smoothed.

Randall and Ward (1960) pursued this approach, in part, testing the placement of diffusors and the arrangement of absorbent materials. Their tests explored the spatial and frequency variation of the reverb time, irregularity of the sound decay and curvature of the decay as well as directional characteristic of the reverberant field. Of particular interest was their demonstration of the variation in the decay curve when observed over short (50Hz) segments. The decay response was more even, with less large variations, when diffusors were present in the measurement space.
2.2 Directional measurements in the sound field

2.2.1 Meyer and Thiele – Directional Diffusivity

Meyer and Thiele (1956) conducted a series of measurements on a range of spaces to explore methods that might provide useful objective criteria for the assessment of auditoria. One of the methods used was the frequency irregularity method outlined above. Their measurements supported a theory proposed by Schroeder (1954/1987), that the $F_v$ measure bore a simple linear relationship to reverberation time such that; $F_v/R_t = 1.45$. This was based on an understanding that the number of maxima in a specified frequency range is proportional to the reverberation time and, as Bolt and Roop had shown, the level difference between maxima and minima is approximately 10dB. This was supported experimentally in the measurement of three different room conditions where it was found that $F_v/R_t \approx 1.5$. Meyer concluded that frequency irregularity, at least for large rooms, provided no greater insight to a room’s character than the reverberation time.

Another element of Meyer and Thiele’s work was a measurement method that could capture the sound coming from all directions in the room. This technique appears to have been developed earlier by Thiele (1953).

A capture system capable of an acceptance angle of $10^\circ$ was constructed by mounting a microphone at the focal length of a parabolic reflector. A room was then ‘scanned’ by rotating the dish in $10^\circ$ increments through a full sphere. A rather novel approach was taken to representing the data by mounting rods in a hemisphere with the length of the rod indicating the strength of the signal measured in that direction.

These so-called hedgehog plots still have some currency, being offered as a plotting option in the room simulation software Catt-Acoustic.
Quantitatively they proposed a directional diffusivity measure by firstly calculating the mean of the levels measured over all directions (M) along with the average deviation from the mean (ΔM). The ratio $m = \frac{\Delta M}{M}$ was then used to find the directional diffusivity $d = 1 - \frac{m}{m_0}$ where $m_0$ is the value measured in a room with no reflections. In a room with no reflection $m = m_0$ and $d = 0$, where the room is ideally diffuse $\Delta M = 0$ (equal level from all directions) and consequently $d = 1$.

Damaske (2008) subjectively tested Meyer’s measure $d$ by replicating the relative sound levels from one of Meyer’s hedgehog plots to a 65 loudspeaker hemispherical array arranged around the listener. The measured value of $d$ was low, 0.06, but subjects reported that the sound was evenly distributed above their head. Damaske’s conclusion was that $d$ was not a good measure of subject’s perception of diffuseness. Whether this conclusion is based on a failure of Meyer’s method to effectively measure a diffuse field, Damaske’s possible oversimplification of Meyer’s results or, as Damaske proposes, it is unclear that the subjective response to a diffuse sound field is significantly different to the objectively measureable state. He does introduce the possibility that other means of measuring the diffuse field may be relevant. This point will be explored later in the chapter.
2.2.2 Furduev and T’ung – Directional Microphone Measurement

Furduev (1955) conducted a review of methods for the measurement of the diffuse field in 1955 that included the previous method proposed by Thiele. Following that Furduev and T’ung published a proposal for measuring the diffuse field in Akusticheskii Zhurnal (Vol 6, Jan-Mar 1960). The paper was translated to English by the American Institute of Physics (Furduev and T’ung, 1960) in the same year. They proposed a simple measure of diffuseness that is very close to that published by Thiele.¹

The method is based on the expectation that a diffuse field will present as equal sound pressure levels in all directions of azimuth and elevation. Their proposal is to use a directional microphone with a directivity of D(θ). The microphone is initially aligned with the source and the output voltage \(U_0\) measured. The microphone is then rotated through various angles of \(θ\) from 0 - 2π with the output voltage recorded for each angle of rotation. The output relative to angle of rotation, normalised to the \(0°\) level, \(Rθ = U/U_0\) is plotted as a polar diagram. If the sound field is perfectly diffuse the resulting plot will be a perfect circle, if the sound field is characterised as a free field plane wave the plot will be that of the direction characteristic of the microphone. Of course a perfectly diffuse field is probably impossible to create so the resultant plot, from a measurement, will be somewhere between these two idealised conditions. The degree to which a sound field is diffuse can then be measured as the difference \((S_1)\) between the area of a unit circle (fully diffuse – \(S_0\)) and the area of the directional plot of the microphone (free field – D(θ)).

¹ There is no reference to the German work in the 1960 paper.
Their measure of diffuseness can then be found from the simple calculation:

\[ d = \frac{S_t - S}{S_t} \]

where; \[ S = \pi - R(\theta) \]

**Figure 2.2:** Plot used by Furduev demonstrating the two limiting states, the directional plot and the unit circle, with a semi-diffuse state plot (dashed line).

This difference between the measured state and the two idealised states produces a 0–1 scale for the measured diffuse state, when the field is fully diffuse the difference between the measured area plot and the ideal area is 0 and \( d = 1 \). When the measured area is equal to that of a free field \( S_t = S_1 \) and \( d = 0 \).

This method found some acceptance within the Soviet acoustics community (Zakharin, 1968; Penkov, 1970) but appears not to have been pursued further due to the time required to perform the measurement and the cumbersome process of measuring the area of the complex plots.

### 2.2.3 Abdou and Guy – 3D Intensity Probe

Another extension of Meyer and Thiele’s work was developed by Abdou and Guy (1994) in their work with the 3D intensity probe. The premise of Abdou’s work was that the most accurate method of measuring directional energy flows was to use an intensity probe, either with a 3D probe or by taking measurements of the three Cartesian axes with a standard intensity probe. In chapter 5 of his thesis a series of possible quantifiers are considered.

The first was a visual examination of the measured directional characteristics. Here the direction and magnitude of the sound arrivals, within a particular time window, are plotted on a circle representing either the horizontal or vertical plane.
The plots produced are reminiscent of Meyer’s hedgehog plots. A visual inspection of intensity plot in time-slices of the decaying field provides a reasonably clear indication of the growth in complexity of the sound field. A numeric value was proposed by calculating the standard deviation of the magnitude of all reflections.

**Figure 2.3:** Abdou’s directional plots of energy arrival in the horizontal and vertical planes at 1kHz and 4kHz.

The second approach was based on the dubious assumption that after the direct sound, all angles of arrival would be at approximately the same level. A calculation is performed that effectively sums all angles of direction of the measured field and divides it by the total number of angles measured. This is similar to the approach taken by Furduev and T’ung but the lack of magnitude variation makes this method scarcely serviceable.
The third approach is based on the definition of a diffuse field having equal energy in all directions of arrival for all arrival times. This equates to zero net acoustic energy flow. A Directional Diffusion (DD) criterion is proposed which takes the average energy flow in a designated direction and divides it by the total energy flow. The result is normalised to the maximum to produce a value between +/- 1. To find the Spatial Diffusion (SD) value the DD values of the three Cartesian values are squared and summed. The Spatial Diffusion is then obtained by taking the square root of the sum of the squares. If, in each direction, there is close to an equal number of +ve and –ve peaks in the impulse response the DD will produce a low overall level. If this is so in all three directions then SD will have a low value, indicating a field that is close to diffuse. If the field has a significant directional component then one of the DD values will be high, due to the bi-directional response of the intensity probe. This will result in a high SD value, indicating a low sound field diffusivity. Abdou proposes that the first 5ms of the impulse be trimmed to remove the influence of the source on the measurement.

Alternatively, by using a sliding time window Abdou suggests that the sound energy flow in the room may be characterised by plotting the SD against time. The expectation is that the SD value will start at 1, indicating the direct sound and move towards 0 as the sound field becomes more diffuse, as the sound decays in the room.

The fourth method considers the degree to which the sound field is isotropic by examining the directional decay curves in a room. The assumption is that if the sound field is even in the room then the decay rate will be the same for each of the Cartesian directions. Variation in decay curves indicates different levels of reflection, assumed to be due to different amounts of absorption in one direction relative to the others.

This approach is useful in providing a visual indication of a lack of isotropy but does not produce an objective criterion that may be applied across a range of spaces.

The fifth and final approach proposed extends the one above where the total energy, excluding the direct sound, from a direction is divided by the total energy arriving at all directions. The ratio of energy arriving in six directions (front, back, left, right, up and down) is compared. Where there is little variation in the directional results the sound field is considered to be diffuse.
2.2.4 Hanyu – Directional microphone arrays

Measurement of directional information presents a difficult problem in that the capture of directional information is either highly generalised, such as the use of a cardioid response microphone, with an acceptance angle in the vicinity of 120° or very tedious, such as the space scanning approach proposed by Meyer. The rotation of a directional microphone, proposed by Furduev addresses this to some degree but still suffers from a low directivity. Abdou’s intensity probe method resolves the directivity problem but fails to establish a means of characterising the field, beyond simple visual representations.

Sekiguchi, Kimura and Hanyu (1992) utilised a method pioneered by Yamazaki and Ito that used four microphones mounted at the points of a regular tetrahedron. Sekiguchi, et al, placed the omnidirectional microphones 17cm apart. By conducting a cross-correlation between channels the direction of energy arrival may be found. To obtain sufficient data between arrivals at the spaced microphones a sample rate of 50kHz was used. With this measurement system two methods for presenting the data were applied; the first plotted the virtual sources in the space by plotting circles with radii indicating level magnitude on three Cartesian planes. A reflective room exhibits a dense array of virtual sources scattered over the measurement plane whereas an absorptive room will exhibit the source and a few discrete reflection plots.

![Figure 2.4: Hanyu’s circle plots, on the left, indicating signal strength and direction by the size and location of the circle. (Front is at the top of the plot)](image)
Overall impulse response is plotted on the right hand side of Figure 2.4. The direct sound, plotted as the high level, single impulse is represented on the left figure by the large, central circle.

Alternatively radar plots were used with the direction and magnitude plotted as straight lines on the concentric level circles. This is reminiscent of the ‘hedgehog’ plots. Hanyu and Kimura (2001) applied this measurement technique in further work that sought to establish a measure of listener envelopment.

A number of subjective tests were conducted and along with Bradley’s findings on Listener Envelopment (LEV), explored later in this chapter, a series of conclusions were made:

i. LEV increases as lateral reflections increase.
ii. There is an influence from reflections arriving from in front of the listener on LEV.
iii. The contribution of individual reflections to LEV is dependent on the arrival direction of other reflections.
iv. LEV increases where there is a broad spatial distribution of arrivals and those arrivals are spatially balanced.
v. LEV increases where the reflection energy is high, relative to the direct energy level and where the reflection delay increases relative to the direct signal.
vi. LEV increases as reverberation time increases.
vii. As listening level increases so does LEV.

Hanyu and Kimura propose that a means of capturing all the above criteria is to measure Spatially Balanced $T_s$ (centre time) a means of incorporating both the spatial and temporal aspects of an enveloping sound field based on measurements using an omni directional and uni-directional microphones. This measure can be viewed as a more detailed Lateral Fraction measurement that measures the arrival energy from a range of directions which is then compared to the overall arrival level.
2.3 Cook/Balachandran – Correlation coefficient

A method that seeks to measure the distribution of sound within a space is the measurement of the correlation coefficient between sound pressures measured by two spaced microphones, in the reverberant field. Balachandran (1959) reported such a measure was proposed by Furduiev (1955). In the same year Cook et al (1955) introduced the correlation coefficient \( R \) as a measure of the diffusivity of the sound field. They propose that, for a random sound field,

\[
R = \frac{\sin kr}{kr} \text{ where; } k \text{ is the wave number } 2\pi/\lambda.
\]

\[
\lambda = \text{wavelength (Hz)}
\]

\[
r = \text{distance between measurement microphones (m)}
\]

**Figure 2.5:** Cook et al’s plot of correlation coefficient against theoretical curve \( \sin(kr)/kr \).

The circles with error bars indicate measured values in a reverberation room. The field is deemed diffuse where the measured results show a close correlation to the curve, that is, the measured value averages follow the theoretical curve and the error bars are small.

Balachandran found that plotting measurement results against the theoretical curve, a first order Bessel function, provided some indication of the degree to which the field was diffuse but when the results varied significantly from the theoretical ideal it became difficult to quantify the variance from the ideal state where he used standard deviation as the best measure of the variance.
2.3.1 Lubman

Lubman (1974) proposed a variation on the two microphone spatial correlation method above by using a single microphone rotated in the reverberant field. A single tone is fed into the measurement space with the output of the traversing microphone directed to a narrow band power spectral density analyser. This produces a Fourier transform of the variation in the sound pressure as the microphone traverses the room. This means the frequency spectrum will relate to the spatial correlation of the microphone traverse. The autocorrelation signal, like the Cook/Balachandran method, will be a Bessel function but in Lubman’s version the transformation to the frequency domain allows a simpler representation of the results.

The theoretical result will be a rectangular plot, centred on the driving frequency \( f_0 \), with a width of: \[ \frac{2Vf_0}{c} \]

where; \( \frac{V}{c} \) = the Mach number

\( (velocity \ of \ traverse/ \ speed \ of \ sound) \)

Like the two microphone method, this approach allows a visual indication of when the field is diffuse but provides little means of quantifying the degree of diffuseness.
2.4 Methods that imply a diffuse field is present

Measures that imply a diffuse field have been developed based on an exploration of listener envelopment a subjective state that indicates that the field is reasonably diffuse. There has been some confusion in the past between listener envelopment and spatial impression. This is understandable because a wide apparent source width will potentially give the listener the impression of being immersed in the sound. Morimoto and Maekawa (1989) made the distinction between the listener being immersed in sound and having a sense of width of the source. In reviewing these implied measures we shall see some fluidity in the use of these two terms.

2.4.1 Barron – Lateral Fraction

This possible confusion between the two terms is made clear in the examination of Barron’s work in this area. Based on some earlier work by Marshall that recognised the importance of lateral sound in auditoria and Haas’ work on speech with the effect of delay, Barron conducted some subjective tests that indicated that lateral energy contributed to what he referred to as Spatial Impression (Barron, 1971). The results are hardly conclusive because they are based on the subjective response of two or three people. Barron, with Marshall, revisited this work a decade later using a larger number of subjects (10) and testing their sensitivity to reflection delay, direction and level. In that paper the effect of azimuth on reflection threshold is explored through reference to work by Schubert, Damaske and Reichardt & Schmidt (all published in German). Barron’s translation of Reichardt and Schmidt showed that their approach took the logarithm of direct sound over the lateral reflections. This resulted in a curve response. Barron proposed that the lateral energy be divided by the total energy, producing a linear relationship between ‘lateral energy fraction’ and spatial impression. This Lateral Fraction was defined as;

$$LF = \frac{\int_{0.005}^{0.80} p^2(t) \cos \phi dt}{\int_{0}^{0.80} p^2(t) dt}$$
This translates as the energy received in the period 5ms to 80ms, at a bi-directional receiver with its null aligned with the source direction relative to the measured sound pressure level over the 0 – 80ms interval of an omni-directional receiver. Barron recognises that \( LF \) is not capable of measuring the diffuseness of a field but rather is a measure of the subjective degree of spatial impression.

Bradley and Soulodre (1995) showed that there was some confusion in what spatial impression meant with Barron using the SI descriptor but describing a broadening of the source. Through a series of subjective tests, influenced by the work of Morimoto and Maekawa (1989) Bradley and Soulodre demonstrated that spatial impression can be divided into two dimensions; Apparent Source Width (ASW) – a measure of the perceived ‘size’ of the source over the azimuth and Listener Envelopment (LEV) – the degree to which the listener feels immersed in the sound. They showed that ASW is affected by the level of early lateral arrival to the listener while LEV is derived from late lateral arrival. The term used was \( LG_{80}^x \) which uses the same calculation as LF but shifts the timeframe of the lateral capture to all energy after 80ms while the omni-directional capture is over the complete period of the decay.

Furthermore a high LEV will reduce a listeners sensitivity to variations in ASW. This division between the two spatial ‘effects’ appears to have been widely accepted with LEV being defined not as a measure of sound field diffusivity but as the listeners sense of immersion in the sound field. A possible flaw in the measure is that it significantly reduces the effect of rear arriving sound. There is some evidence that listeners have a very poor ability to detect such sound, as demonstrated by Damaske. There is contrary evidence, however, presented by Morimoto et al (2001) that LEV is influenced by the Front to Back Ratio, defined as \( 10\log(E_f/E_b) \).
2.4.2 Keet/Ando – IACC

In the late 60’s Keet (1968) suggested that the broadening of sound was an important factor in the quality of sound in a concert hall proposing the term Apparent Source Width as a measure of binaural listening quality. He used a pair of cardioid microphones spaced 21cm apart at an angle of 90° to record some ‘dry’ recorded orchestral music in four different positions in three concert halls. The recordings were played back to subjects in an anechoic room with the loudspeakers placed at 100° relative to the listener (assumed to be +/- 50°). The loudspeakers were placed behind a screen that was numbered. Subjects were asked to indicate the boundary of the sound by indicating the number on the screen that coincided with their auditory impression.

Following the works of Sayers & Cherry; Burgtorf, Wagener & Damaske; Keet plotted the Cross Correlation Factor (K), of the two channels, in 50ms time windows. He found that the order of the measurement positions inversely corresponded with the subjects’ ASW responses, in the 0-50ms window and the 150-200ms window. This indicates that a highly coherent signal corresponds to a ‘narrow’ source image. Keet proposed that the measure $1 - K_0^{50}$ be used to measure ASW.

Damaske and Ando (1972) extended this idea, introducing the Inter-Aural Cross-correlation Coefficient as a measure of sound field diffuseness. The IACC is measured with a dummy head with the microphones placed at the ends of simulated ear canals. This form of measurement seeks to replicate the physical hearing system of a human. The premise is that a diffuse sound field will have signals arriving at each of the ears in a random distribution, resulting in zero correlation between the signals at each ear. Therefore the measure of field diffusivity is $1 - \text{IACC}$. A degree of subjective preference was shown by subjects to bi-directional stimuli where there was a low IACC value.

Hidaka, Beranek and Okano (1995) refined the measure by removing the first 80ms from the measured impulse responses. The reasons for doing so are manifold, ranging from detection of delays becoming perceptible to the possible fusion time of music in the auditory system. Essentially the assumption is that the first 80ms consist of early, and to some degree, discrete reflections whereas the sound field after 80ms is the reverberant field.
2.4.3 Polack/Defrance – Mixing Time

An area of research that is related to the measurement of diffuse sound fields, proposed by Polack (1993), is the ‘mixing time’ of a space as an indicator of the onset of the diffuse field. In a room with perfectly reflecting boundaries, the assumption is that within a particular period of time, the energy from a source will be equally distributed throughout the room. As the equal distribution of energy within the space is deemed to signify a diffuse state then the time taken to reach such a state is of interest. If we invert the logic, the estimation of the mixing time implies the onset of a diffuse state. An approximation of the mixing time may be calculated from the room volume:

\[ t_{\text{mixing}} = \sqrt{V} \]

where \( V \) = Room volume (m\(^3\))

\( t \) = time (ms)

This is based on Wente’s Frequency Irregularity and Schroeder’s extension of that work (below) in making a connection between the modal density of a sound field and the volume of the room in which the field exists.

A series of measures have been proposed for estimating the mixing time (Defrance and Polack, 2008);

The extensive Fourier Transform (XFT) – provides a cumulative integration of the signal with the same window size taken for each time step. The result is, for the first step, the phase bears a linear relationship to frequency but as the windows step along the impulse response the phase to frequency plot becomes progressively more ragged. The mean regression error (\( D(t) \)) for each window is calculated, relative to the initial window. The value for \( D(t) \) will increase in value over time but usually exhibits a sharp inflection at the mixing time.

Kurtosis – is a measure of a random process’ Probability Density Function. Essentially a measure of how much a result set varies from a normal distribution. It is calculated from the mean and standard deviation of a data set, such that;

\[ k = \frac{E(x - \mu)^4}{\sigma^4} - 3 \]

where; \( E \) = expected value of \( x \)

\( \mu \) = the mean of \( x \)

\( \sigma \) = the standard deviation of \( x \)
For a normal distribution \( k = 0 \). Defrance and Polack recommend using a window of 24ms, which they based on the integration time of the hearing system. Calculating \( k \) over this time window, the mixing time is found to be the point where \( k=0 \) and consequently the field is deemed to be diffuse.

Of the two methods presented, Defrance and Polack considered the XFT method the more reliable due to the sensitivity to window width exhibited by the Gaussian measure.

Although the mixing time implies the diffusive state, it is unable to provide any information on the degree to which the field is diffuse. The measures introduced here may have some efficacy when used in conjunction with other criteria that more directly measure the sound field, such as establishing the time taken to reach an acceptable level of diffuseness where that level is enumerated.

### 2.4.4 Schroeder – the Schroeder frequency

A method of establishing whether a diffuse state exists in a room that is, one could argue, based on a set of assumptions, has been presented by Schroeder (Schroeder and Kuttruff, 1962, Schroeder, 1954/1996). Starting with the assumption that we are dealing with large rooms (what constitutes a large room is unclear), Schroeder originally proposed that above a certain frequency the number of independent room modes that are simultaneously excited is sufficiently large to produce a complex sound pressure that has a Gaussian distribution. Originally the value for what was to become known as the Schroeder frequency was:

\[
f = 4000 \sqrt{\frac{T}{V}}
\]

where; \( T = \) reverberation time (T60 secs)

\( V = \) room volume (metres\(^3\))

This calculation was based on the assumption that 10 normal modes overlapping within the half-power bandwidth of a single mode would produce a low frequency irregularity value. The constant was later revised to 2000 (Schroeder and Kuttruff, 1962). This ensures a minimum of three overlapping resonant modes within the half-power bandwidth of a mode.
Fazenda and Wankling (2008) have since shown, through subjective testing, that listeners detect an unevenness in the sound field where there are less than four modes at frequencies below 250Hz, with an increasing number required as the frequency is increased.

This measure of modal density or acceptable spacing is expressed as:

$$\Delta f = \frac{c^3}{4\pi Vf^2}$$

where; $c =$ speed of sound (m/sec)

$$f = \text{arbitrary frequency (Hz)}$$

By applying Sabine’s reverberation time formula and converting frequency to wavelength Schroeder derives:

$$\lambda_c = \sqrt{\frac{A}{6}}$$

where; $A =$ Absorption Area ($m^2$)

This is a simple conversion between frequency and wavelength that allows the transition region from modal to stochastic dominance of the sound field to be expressed by one variable. $\lambda_c$ expresses a wavelength for which greater values will produce less than three overlapping modes within the half-power bandwidth.

A second assumption applied to this model is that the distance between the source and the receiver is such that there is “negligible direct power transmission”. This implies the measurement microphone is placed beyond the reverberation radius. Schroeder refers to this distance as the “diffuse-field distance”, $r_c$ such that:

$$r_c = \sqrt{\frac{\ln 10^6 V}{4\pi T}}$$

which equates to; $$\sqrt{\frac{A}{16\pi}}$$ when Sabine’s formula is applied.

Through a process of substitution the following relationship between $r_c$ and $\lambda_c$ is arrived at:

$$r_c = \sqrt{\frac{3}{8\pi}}\lambda_c$$

which suggests there is a correlation between the Schroeder frequency and the reverberation radius.

The third assumption made by Schroeder is that the enclosure consists of a simple geometry, allowing the application of the image-source method. In applying an impulse response model Schroeder approximates the number of arrivals at a point in the enclosure within time $t$, after the initial impulse, to be;
so that the average spacing between reflections is equal to;

\[ \Delta t = \frac{V}{4\pi c^3 t^2} \]

Schroeder then proposes that where \( \Delta t \) is smaller than the echo decay time, \( T/\ln 10^6 \), and where the reflections arrive from “many different directions” the field at the measurement point may be deemed to be diffuse. Following these assumptions allows Schroeder to present the diffuse field time interval;

\[ t_c = \sqrt{\frac{\ln 10^6 V}{4\pi c^3^r T}} \]

Jeong (2010) suggested that \( t_c \) does not ensure a diffuse field where the room has a complex mix of absorbing and diffusing elements.

Nonetheless Schroeder proposed that the onset of a diffuse field may be viewed from the following aspects:

i. \( f_c \) – the frequency above which the sound field is deemed to be diffuse

ii. \( t_c \) – the time interval of the enclosure’s impulse response, beyond which the field is considered diffuse and

iii. \( r_c \) – the distance from the source beyond which the field is assumed to be dominated by the reverberant (diffuse) field.

The criteria, stated above are based on a series of simplifications; acceptable degree of modal overlap, measurement position is placed beyond the room radius and the enclosure in question has a simple geometry. These simplifications allow the development of the criteria and the consequent correlation between them. Usually the spaces that we consider are of greater complexity than is presented here. Schroeder’s criteria are therefore useful as indicators of transition between simple and complex field states but are unable to specifically indicate where the diffuse state has been reached.

As such the criteria are useful in providing a broad indication of whether a diffuse state in a room may exist.
2.4.5 Haan and Fricke – Visual Assessment

A component of Haan and Fricke’s evaluation of concert hall surface diffusivity (Haan and Fricke, 1997) was to apply visual assessment of the halls surfaces in assessing how diffuse the sound field in the auditorium may be. The first stage of this assessment was to establish an Acoustic Quality Index based on a survey of 35 musicians and conductors who had experienced a wide range of performance halls. The scores from all participants were averaged to produce the AQI. The Musicvereinsaal in Vienna rated highly while the Avery Fisher Hall in New York rated poorly.

An assessment of the halls was then made by examining photographs of the rooms sorting them into one of three categories. A series of descriptors were used to inform the viewer as to what surface treatment was deemed to be diffusing. For example; “random diffusing elements over the full area of the ceiling” for high values to “large flat and smooth surface” for low values. A regression analysis was applied to the results and it was found that there was a reasonably strong correlation ($r^2 = 0.631$) between the subjective AQI and the equally subjective visual assessment of the halls surfaces.

Although the method is fairly subjective it is reasonable to expect that a number of people assessing a hall visually would reach similar conclusions about the potential diffuseness of the field, where their assessment is made according to a broad set of visual criteria. Distinguishing between complex surfaces and flat surfaces is reasonably simple. The method does not incorporate the complexity of the room volume or provide a means of distinguishing small increments in surface scattering. As such the method may only be useful where only large increments in sound field diffusivity are distinguishable by the average listener.
2.5 Diffuseness vs Diffusivity

Conducting listening tests in an anechoic room with 65 loudspeakers Damaske (2008) found that listeners judged a sound field to be immersive when they were presented with direct and delayed sound from seven loudspeakers mounted on the azimuth plane. The realisation that listeners may perceive the sound field in an entirely different way to how the field is actually arriving at the listener caused him to propose that diffuseness, a subjective term, be applied. The argument is that many acoustic criteria start out as pure physical measures of a characteristic sound feature. Over time, Damaske argues, we come to realise that the physical measure does not map directly to how we experience the feature being examined. We then move towards some measure that correlates with the average psychological response to the stimulus.

In making the point about diffuseness Damaske appears to ignore the fact that there is currently no widely accepted measure of the diffuse field as a physical entity. He may be correct that we need to explore listener’s perception of a diffuse field but the argument here is that some physical measure must be found as the starting point, against which listener preference is then assessed.
2.6 Conclusion

A review of methods used by researchers to assess the diffuseness of a sound field indicates a wide range of methods may be applied. This researcher’s attempt to find a method for the measurement of a diffuse field will explore some of the following measures or methods. Other methods will not be explored due to a lack of resources. The following list indicates the methods that will be explored in this thesis;

i. Furduiev and T’ung – Directional Microphone Measurement is the primary method explored in this work. The development of the ambisonic microphone lends itself to exploration of the sound field through the B-Format signals. The measure will be assessed against some of the other measures outlined in this chapter.

ii. Directional Diffusivity – Lateral Fraction and IACC will all be compared against the ambisonic method for efficacy.

iii. Frequency Irregularity – although the measurement method was cumbersome at the time of its proposal the means of assessing Frequency Irregularity is now fairly simple. Schroeder’s (Schroeder, 1954/1987) calculation of required number of modes per frequency band is basically the same approach in that if there is an acceptable modal overlap within each frequency band the Frequency Irregularity measure will, most likely, produce a result indicating the field is diffuse. A colleague at the University of Sydney is considering these aspects in his work and consequently, will not be explored further in this work.

iv. 3D Intensity Probe – this device is not available to this researcher.

v. Directional microphone arrays are also unavailable.

vi. Mixing Time – may provide a useful check of the onset of the diffuse state in exploring the temporal aspects of diffusivity.

vii. Visual Assessment – is easily dismissed as entirely subjective and arguably unrepeatable. However, Fricke and Haan only chose three states for assessment. Doing so created fairly clear demarcation between the states being assessed. The result is a numerically low range of variability, Not Diffuse, Reasonably Diffuse or Very Diffuse.
Part of the work in this thesis intends to explore the subjective assessment of diffuse sound fields. The author suspects that the average listener’s capacity to detect different degrees of diffusivity may support Fricke and Haan’s small range of variables.
References


Chapter 3

Initial work

As we saw in the previous chapter there are several methods for directly and indirectly measuring a diffuse sound field. The major interests of this work are the methods that seek to directly measure the state of the sound field. The 3-D Intensity Probe method establishes accurate directional information in a way similar to that pioneered by Thiele and Meyer. As previously indicated, the method becomes difficult to interpret due to the large amount of data generated. A clear method for interpreting that data is currently unavailable. The intention here is to explore the degree to which a sound field approaches that of a perfectly diffuse field, as defined in the introduction to Chapter 2. Where the field is close to that of a fully diffuse state the degree of variation between sampled angles is expected to approach zero. It is only in cases where the field is non-diffuse, dominated by strong modes for example, that the field will change significantly over the sample angles. It may be argued that a capture system that is simpler and less accurate may be adequate in producing an acceptable measurement of the field. In this chapter the rotating microphone method will be explored. Initial works tests the efficacy and limitations of a microphone directly rotating in the sound field. This is followed by an exploration of virtual rotation techniques through modeling and measurement.

3.1 Exploring Furduev and T’ung’s approach

The intention of this work is to focus primarily on one method that seeks to measure the sound field in a way that addresses the definition that:

The amplitudes of the incident waves are distributed uniformly over all possible directions of incidence in such a way that from each element of solid angle the same amount of energy arrives per second and per unit of area element thought perpendicular to the respective direction. (after Kuttruff)
Furduev and T‘ung’s method of assessment is based on the rotation of a directional microphone in the room under test. This method tests the uniformity of the amplitude arriving at a point in the room by measuring the level of sound present through a $360^\circ$ rotation of the measurement microphone.

For an anechoic room the output of the rotating microphone would be that of the directional characteristic of the microphone, whereas the rotating microphone’s output would be constant in a diffuse field. In most rooms the measured characteristic will be somewhere between these two ideal states. This is illustrated in Figure 3.1b below.

This thesis will focus primarily on this method, comparing it with other methods to confirm its validity.

**Figure 3.1:** (a) Area difference ($S_1$) between diffuse and free-field measurements (b) Area difference ($S$) between measured room and ideal diffuse field

(After Makrinenko (Makrinenko 1994))

Their measure $\partial$ provides a $0 – 1$ measure of the variation between these two ideal states. This is achieved through the simple calculation:

$$\partial = \frac{S_1 - S}{S_1}$$

where:

- $S_1$ is the area difference between the measured polar response of the microphone and the ideal state ($\pi$ for a unit length radius)
- $S$ is the area between the measured plot and the circle.
(Note: Furduev and T’ung used the character $d$ to symbolise their diffusivity measure. To avoid confusion between that and other uses of the italic $d$ the Cyrillic lowercase $\partial$ will be used.)

For an anechoic room $S = S_i$ so $\partial = 0$ and for a perfectly diffuse field $S = 0$ and $\partial = 1$.

![Diagram](image)

**Figure 3.2:** Comparison of polar responses for anechoic [$\phi_1 (\theta)$] and semi diffuse [$\phi_2 (\theta)$] rooms. (Makrinenko 1994)

The simplicity of the measurement technique, and the fact that it does not demand expensive or elaborate equipment, suggests that further investigation of the method is warranted. The obvious problem with such a technique is the measurement of a complex area bounded by the response plot.

### 3.2 Updating the method

The availability of computer based measurement and analysis systems provides an opportunity to revisit the work. To do so a series of readings were taken in three rooms at the Acoustics Laboratory at the University of Sydney using a Bruel & Kjaer 4011 cardioid microphone mounted on a B&K 3921 turntable. The turntable completed a full rotation in 80 seconds. An omni directional sound source radiating pink noise was used. A sound file was recorded for each $360^0$ rotation in the spaces under test. These sound files were analysed using the Psysound program (Cabrera 2001). Sound pressure level in octave band intervals were recorded for each $2.25^0$ of rotation. These levels were normalised to the level at $0^0$. 


This produces a value $r$, which is equivalent to the height of an isosceles triangle where the base of the triangle is approximately that of the circumference of a circle divided by the number of segments of that circle, in this case 160. The area of the sector is calculated from this value $r$.

The area calculation is simply the area of a triangle where the length = $r$ and the width = sector width/2. If we treat the circle as a unit circle then the sector width is approximately the circumference of the circle divided by the number of sectors measured. As the circumference of a circle = $2\pi r$ then we can estimate the sector width based on the ‘radius’/height of that sector, resulting in sector width = $2\pi r$/ (number of measurement sectors). The area of the sector is then calculated from; $r * (0.5*2\pi r$/ number of measurement sectors) = $\pi r^2$/ no. measurement sectors. (Alternatively, viewed as the area of a circle of radius $r$ divided by the number of sectors where each sector may be of a different radius.) The sum of the segment areas produces a $S_{\Delta}$ value for the reading that is input into the calculation - $\vartheta = \frac{S_1 - S}{S_1}$.

Measurements were taken in an anechoic room, with the microphone placed 2 metres from the source, to provide the plot of the directional characteristic of the microphone. Readings were then made in a reverberation room and a lab space which was assumed to have a diffusion coefficient somewhere between that of the anechoic and reverberation rooms. Three readings were performed at the same point in each space to assess the repeatability of the test.

In the reverberation room the loudspeaker was placed in an axial corner of the room with the microphone placed in the corner diagonally opposite the loudspeaker and set 1.2 metres above the floor, an overall distance of 8.1 metres.

The Reverberation room under test is a rectangular, painted concrete and rendered masonry room measuring 6.36 m (l) x 5.12 m(w) x 3.98 m (h), producing a volume of 130 m$^2$. The $RT_{20}$ measured in the room is shown below:
Figure 3.3: Octave band RT$_{20}$ results for bare Reverberation Room.

The lab is a T shaped space that is basically 8 metres long and 3.4m wide with a ceiling height of 3.9 metres. The ‘top’ of the space interconnects to other areas in the lab by a corridor that extends approximately 6 metres in each direction. Within the lab space a bench runs the length of the space on each side. The loudspeaker was placed on the floor at the apex of the “T” in the lab space. The microphone was placed at a distance of 5 metres from the source, at a height of 1.2 metres from the floor.

Figure 3.4: Lab area used for medium diffuse space. The microphone was placed in the centre of the area in the photograph on the left, the loudspeaker was placed where the black pentagon is in the photograph on the right.
The results of the Psysound readings are exported to spreadsheets where the following calculations were performed:

\[ S_D \] is calculated from the sum of the sector areas for the anechoic room readings
\[ S_I \] is found by subtracting \[ S_D \] from \[ S_0 \] (\( \pi \))
\[ S \] is calculated from the sum of the sector areas for the Lab and Reverb room readings respectively.

The value for \( \vartheta \) was then calculated in octave bands for the two rooms.

**Figure 3.5**: \( \vartheta \) values in octave bands measured in Reverb Room and Lab space.

(Three measurements in each position)

Between 500 Hz and 4 kHz the average \( \vartheta \) value for the Lab space is 0.37. Within the same frequency band the \( \vartheta \) value for the Reverb room is 0.99. The \( \vartheta \) value \(< 0\) for the lab occurs where the area plot of the readings in the lab produce a smaller area than the plot measured in the anechoic room. This may be possible due to the greater distance from the source in the lab measurement if the absorption at 8 kHz, in the lab, was equivalent to that in the anechoic room. Where the area plot of the test room exceeds that of the perfect circle, at the lower frequencies in the reverb room, a value of \( \vartheta > 1 \) is returned.
This is at least partly attributable to fluctuations in \textit{rms} amplitude, due to the random nature of the noise source, which are more pronounced in the lower octave bands (such fluctuations are increasingly averaged out as the measurement duration increases). This may also be exacerbated by strong modal activity in the low frequency range. This effect was illustrated in the Furduev and T'ung paper, reproduced here in Figure 3.6, where a strong side reflection produces a plot that extends beyond the unit circle. They resolved this by separately calculating the area outside the circle and deducting it from the area inside, producing a lower value for \( \vartheta \). An alternative approach is to normalise levels to the maximum level rather than to 0\(^\circ\) level producing a plot that is entirely within the unit circle.

\begin{figure}[h]
\centering
\includegraphics[width=0.5\textwidth]{figure3_6.png}
\caption{Illustration of a strong directional reflection causing a strong variation in the polar response, from (Furduev and T'ung 1960)}
\end{figure}

### 3.3 Extending the work

This initial series of tests provide a discernible difference in the value \( \vartheta \) for the different spaces tested. Furthermore the results for the rooms reflect the value, based on subjective assessment, expected for rooms of their type. The test does not provide a detailed indication of how isotropic the sound field is as it only assesses level with respect to the 0\(^\circ\) measurement direction - any directional information in the method is of a rather general nature based on the wide directional pattern of the cardioid microphone.

Clearly methods developed by other researchers, detailed in section 2.1, may be more accurate in providing the means of assessing the degree to which a sound field meets the criteria for a diffuse field.

Nevertheless, it should be borne in mind that such methods are quite complex and consequently could be expected to be rather time consuming.
By comparison Fricke and Haan’s method used a 3-state criterion based on visual inspection. Their results indicated that the diffusion of the sound was a significant criterion in assessment of a hall. This raises the question as to what level of accuracy is required in diffuse field measurement that equates to listener preference.

Furduev and T’ung’s method provides a method of assessment that is more direct than the visual assessment of a space but is simpler and quicker than the variations of the Thiele method, for example. For this reason the author considers the method bears further investigation.

The technique itself requires more refinement, particularly with respect to the assessment software. In particular, automation of the recording process requires improvement to streamline the cumbersome measurement process.

Results greater than unity point to two problems with applying Furduev and T’ung’s method. Firstly, fluctuations in the sound field demand a much longer duration measurement, implemented with a slower rotation speed (or averaging many rotations). Secondly, comparing the anechoic and test polar patterns by matching the $0^\circ$ value can yield a result greater than 1 when a ‘notch’ is found in the test polar pattern at this angle.

### 3.4 Considering the Soundfield™ ambisonic microphone

The technique used in the preliminary work is limited primarily to rotation on the azimuth. Bradley and Soulodre (1995), through their work in listener envelopment, present an argument for assessment of the diffuse soundfield on the horizontal plane only. So there may be an argument for making measurements only on the azimuth. Furuya, Fujimoto et al (2001) however, suggest that energy arrival from above and behind the listener contribute to their sense of envelopment.

Furduev’s original work proposed examination of the field through all angles of rotation. Rotation of a directional microphone through axes other than the horizontal becomes an exercise in constructing complex electro-mechanical systems. It is therefore desirable to use a microphone array that will allow rotation of a virtual directional microphone. The periphonic/ambisonic microphone is one device that produces such an output. The object of this work is to explore the potential to update the Furduev method with the use of contemporary technology.
3.5 The virtual rotation of a directional microphone

An ambisonic microphone is such a device that allows virtual rotation of a directional microphone by mixing respective B-Format outputs. In this way a directional response transducer may be rotated through the full unit sphere.

The commercially produced Soundfield™ microphone provides four outputs in B-format, W, X, Y and Z derived from an array of four sub-cardioid microphones mounted on a tetrahedron (Rumsey 1994). The response of the capsules is: \(2 + \cos \theta\); where \(\theta\) is the polar angle of incidence relative to the direct axis (0°) of the capsule.

A sub-cardioid response is usually achieved by mixing the outputs of a pressure microphone with a pressure gradient microphone, where the level from the pressure microphone is boosted 6db relative to the pressure-gradient output.

The four sub-cardioid capsules are aligned left/front (LF), right/front (RF), left/back (LB) and right/back (RB). It should be noted that the axis of each of the capsules is at 45° to the horizontal plane, as illustrated in figure 3.7.

The outputs of the sub-cardioid capsules are electronically processed to produce an output equivalent to a point receiver up to approximately 10kHz, as reported by the manufacturer. Kan (2009) has shown that the directional character of the B-Format outputs does not match the expected directivity pattern above 8 kHz.

The B-Format outputs, the equivalent to pressure and pressure-gradient transducer outputs are derived by firstly adding the opposite capsules outputs, with the rear capsule electrically phase inverted, ie: (LF-RB) and (RF-LB). This produces two bi-directional responses aligned at 45° to the horizontal and rotated 45°.

Finally, by combining these two bi-directional responses the B-Format outputs are achieved, as follows;

\[
\begin{align*}
W &= \text{input signal} \times 0.707 \\
X &= \text{input signal} \times \cos A \times \cos B \\
Y &= \text{input signal} \times \sin A \times \cos B \\
Z &= \text{input signal} \times \sin B
\end{align*}
\]

where \(A\) is azimuth angle and \(B\) is elevation angle. (Malham elevation angle. (Malham 1998)

Figure 3.7: A-Format configuration of Soundfield™ microphone.
The B-format outputs are equivalent to an omnidirectional microphone (W output) and three bi-directional microphones (X,Y,Z outputs) aligned Front/Back (X), Left/Right (Y) and Up/Down (Z).

By combining the B-format outputs a range of pick-up patterns and orientations may be derived. For example; combining the W output, the equivalent of a pressure transducer output, with the X output, the equivalent of a pressure-gradient transducer, will produce a range of pick-up patterns orientated on the $0^\circ$ axis as shown in figure 3.8:

\[
\begin{align*}
2W+X & \quad W+X & \quad W+2X & \quad X \\
\text{SubCardioid} & \quad \text{Cardioid} & \quad \text{Hypercardioid} & \quad \text{Bi-directional}
\end{align*}
\]

Additionally, combining X and Y outputs with the W output allows ‘steerage’ of the pick-up pattern:

\[
\begin{align*}
(X+W)+(Y+W) & \quad (-X+W)+(Y+W)
\end{align*}
\]

**Figure 3.8:** Commonly used combinations of pressure and pressure-gradient microphones

By these means a virtual directional microphone may be aligned on any angle of azimuth and elevation.
### 3.6 Modeling the microphone response

To explore the directional character of the microphone a series of two-dimensional response plots are examined. Although the microphone array is capturing sound in the three-dimensional sound environment, 2-D representation is easier to present in the format of this thesis. The microphone may be modeled in idealised sound fields quite simply by creating a vector array in Matlab into which values for the fields are inserted. The simplest of these is where a 1 is input at the first element with the rest set to zero. This may be characterised as a signal coming from the 0 degrees direction for the duration of a ‘rotation’. The microphone array is then rotated within the ‘field’ producing the characteristic cardioid plot in Figure 3.9a. Similarly, the vector array may be set to all ones, simulating a field which, over the duration of the ‘rotation’, presents equal level of signal from all directions. The resultant plot is illustrated in Figure 3.9b. The high number for the ‘output’ level is the sum of the 360 plots of the directional character of the microphone. This will be explored later in the chapter.

![Figure 3.9](image)

**Figure 3.9:** (a) Idealised free field rotation, Area = 1.18  
(b) Idealised diffuse field rotation, Area = $\pi$ (normalised)

These two simple cases illustrate the extents of the measurement system, the area enclosed by the rotation of a directional (cardioid) microphone and the area enclosed by the rotation of the same microphone in a field where the energy level arriving from all directions is equal, providing us with an ideal $S_1$ value of 1.96.
Some further idealised cases may then be explored, firstly a plane wave simplified as a set of ones in the first half of the rotation. If the microphone is treated as a point source then the assumption is that a plane wave, at the point of arrival, will register input to the microphone in the ‘front’ hemisphere while the ‘rear’ hemisphere will be silent. The resultant plot is displayed in Figure 3.10.

![Figure 3.10](image_url)

**Figure 3.10:** (a) Idealised rotation of directional microphone, front arrival field, A = 1.41

(b) Idealised measurement of a frontward arriving wave, A = 1.57

Here we find the wide capture range of the cardioid microphone produces a 10% error relative to the idealized case, when the results are normalized. This may be improved by the use of higher directivity capture systems such as those utilised by Meyer or Abdou. Recent work in higher order Ambisonics (Moreau, Daniel et al. 2006; Parthy, Jin et al. 2008) have produced narrower acceptance angles. A fourth order circular plot (Kolundzija, Faller et al. 2010) is shown in Figure 3.11(a). Rotating such a pick-up pattern through the same idealised plane wave produces the plot in figure 3.11(b). The area inside the plot is 1.49, closer to the idealized state above, so we can say that higher directivity pattern microphone/arrays may produce more accurate measures of $\partial$. 
Figure 3.11: (a) 4\textsuperscript{th} Order circular harmonic pick-up pattern, $A = 0.57$, $S_{1(4OA)} = 2.56$

(b) Idealised measurement of a front arriving wave, using a 4\textsuperscript{th} order circular harmonic pick-up, $A = 1.49$

As Furduev indicated strong lateral reflections may produce output that is outside the unit circle. This may be resolved by normalizing to the maximum level of the rotation measurement but the lack of distinction in the cardioid pattern may result in overly smoothed results that don’t fully expose strong reflections in the measurement space.

To explore this possibility ones are placed in a narrow band at right angles to the azimuth, producing an idealised resonant mode while all other angles are allocated a 0.1 value.

Figure 3.12: (a) Cardioid response to sharp peak, $A = 2.17$, $S = 0.96$, $\vartheta = 0.51$

(b) 4\textsuperscript{th} Order circular harmonic response to peak, $A = 1.46$, $S_{4OA} = 1.68$, $\vartheta_{4OA} = 0.35$
Clearly the method of rotating a cardioid microphone does not distinctly indicate the presence of strong reflections in an otherwise ‘diffuse’ field. From this we may assume that the Furduev method appears to be an inadequate measure of these significant variations in the sound field. This however is not the point of the measurement technique. We will now explore the method’s efficacy in ‘measuring’ a uniform field against that of the 4th order circular harmonic method. To do so the method of setting a reflection level is again utilised, this time by setting ones between 225 and 235 degrees. The level of the remaining ‘field’ is then adjusted by setting values that approximate to levels 3, 6, 12, 18, 24 and 40dB below that of the reflection level. A comparison is made between the 1st order ambisonic cardioid pattern and the 4th order circular harmonic pattern.
Figure 3.13: Comparison of Cardioid (1a-c, d-f) and 4th order circular harmonic pickup (4a-c, d-f) of a range of fields approaching a model of a diffuse state, from left to right.
There is convergence between the two methods at the extremes of the simulated fields. Where the level difference between the ‘strong reflection’ and the remainder of the field is large (-40dB and -24dB) the $\varphi$ value is low for both pick-up methods. Likewise where there is a small level difference (-3dB) between the ‘reflection’ and the remainder of the field the $\varphi$ value is high for both methods. Overall the higher order measurement produces a lower $\varphi$ value than the first order.

This supports, to a reasonable degree, the suggestion that the cardioid method is applicable to medium to high diffusivity states. The results are similar to those presented by Furduev and T’ung who used cardioid and figure-8 patterns in measuring a range of spaces.

**Figure 3.14:** $\varphi$ value variation of cardioid and 4$^{th}$ order ambisonic pickup.
3.7 Analysis system for Ambisonic microphone arrays

Based on the work above, an analysis system was developed using Matlab to analyse the output from the B-format Soundfield\textsuperscript{TM} microphone system. The specific scripts used are documented in Appendix A. A general overview of the process is outlined below.

3.7.1 ‘2-D’ measurements

In Furduev and T’ung’s original paper measurements were made separately in the horizontal and vertical ‘planes’. Obviously, due to the directional character of the microphone the measurements weren’t capturing the sound only in the horizontal direction but the output was weighted on that measurement plane. This is the primary method that will be applied in the following measurements.

Recordings of the four B-format outputs of the system are imported into Matlab using the wavread command and are concatenated to a single data matrix. Directivity factor of the resultant output may be set using a df\textsuperscript{1} value ranging from 0 for omnidirectional to 2 for a bi-directional output. For this work the value has been set to 1 to produce a cardioid response. The directional weighting of the four outputs is set using the sph2cart command, converting the spherical coordinates to the equivalent cartesian coordinates on a unit sphere where the position is set by theta (horizontal) and phi (vertical).

For a horizontal rotation the theta vector is set for five degree intervals with phi set to zero. For a vertical rotation the values for theta and phi are reversed. The output level is then calculated on the weighting of the X, Y and Z outputs added to the W output. The results are plotted through 360\textdegree to produce the response of the 'cardioid' microphone rotated within the measurement space. The area enclosed by the plot is then calculated for each segment and summed, the \( \delta \) value is then found by calculating the area differences, as detailed for the Furduev method.

\footnote{1 The use of df here is used to designate the directional factor of the combined microphone characteristics. This should not be confused with df (or DRF) used for distance factor of microphones, which is a measure of the direct to reverberant ratio of the microphone pick-up pattern. (Eargle, 2005)}
3.7.2 ‘3-D’ measurements

To explore the directivity offered by the ambisonic microphone we need to expand the analysis to a ‘3-D’ measurement. To follow Furduev’s method the area of a three dimensional object needs to be calculated. To achieve an equal weighting of the measurements the sphere must be divided into equal sections. To do so an Icosahedron, the platonic solid with the largest number of individual surfaces is required. This solid consists of 20 triangular faces with 12 vertices. A property of the Icosahedron is that its inscribed sphere closely matches the surface of the sphere.

The centre lengths of the triangles coincide with the vertices of the dodecahedron. To estimate the total of the complex volume produced by a directional microphone the level measured for each spherical angle of rotation is normalized to the maximum level. The spherical angles used are the vertex angles for the dodecahedron. Those vertices coincide with the centre points of the triangles that construct the icosahedron. So the circumradius of the dodecahedron is equal to the inradius of the icosahedron.

The vertex angles of a dodecahedron may then be used to establish the centre points of the triangle faces of the icosahedron. The length \( h \) of the inradius may then be used to calculate the area of the associated triangle element.

The area of the face is calculated from;

\[
a = \frac{\sqrt{3}}{4} s^2 \quad \text{where;} \quad s = \text{length of triangle side}
\]

and

\[
h = \frac{\Phi^2}{2 * \sqrt{3}} * s \quad \quad h = \text{inradius length}
\]

\[
\Phi = \frac{1 + \sqrt{5}}{2}
\]

by substitution we find;

\[
s = h \times \frac{2 * \sqrt{3}}{\Phi^2}
\]

So

\[
a = \frac{\sqrt{3}}{4} \left( \frac{h * 2 * \sqrt{3}}{\Phi^2} \right)^2
\]
This calculation produces a surface area approximately 20% larger than the surface of the equivalent sphere but this is consistent so it is acceptable for these calculations which are based on ratios.

For the derivation of the Furduev measure the same method is used as for the two dimensional method in that the level is found for twenty directions on the sphere that are then normalised to the maximum level. That value is then input to the area calculation above and the area of the triangles summed to produce an approximation of the area of the measured surface. The same calculation of $\delta$ is then applied as that used in the two dimensional measurements.

### 3.8 Conclusion

The application of this method shows some promise in providing a reasonably simple method of quantifying the degree to which a sound field meets the criterion of having equal energy level in all directions of arrival. In Section 3.6 it was demonstrated that Higher Order Ambisonic methods may provide more accurate measurement of the field but this greater accuracy requires measurement systems of greater complexity. It is therefore interesting to test the B-format measurement system to ascertain whether the level of accuracy provided by the system is adequate for practical measurement of sound field diffusivity. This will be explored in the following chapter.
References


Chapter 4

Measurement of Diffusivity in a Reverberation-Controlled Room

In order to test Furdjev’s method an obvious point of departure is to assess whether the method will provide an adequate measure of the variation in a sound field as the reverberation time of a room is varied. To test this, the reverberation room at the University of Sydney was altered through the addition of absorbent materials, producing reverberation times ranging from 0.15 to 4.12 seconds. It is assumed that in varying the reverberation of the space the diffusion of the sound in the room will also be altered. It is this assumed variation in diffusivity that is explored. The reverberation time of a room was progressively altered with an ambisonic microphone measuring the state of the field in each room state. Diffusivity indexes were produced in the horizontal and vertical planes for each state. Further to this some additional measurements were taken to explore the temporal aspects of the diffuse fields in the ‘rooms’.

4.1 Reverberation Room States

As indicated previously, the room under test is a rectangular, painted concrete and rendered masonry room measuring 6.36 m (l) x 5.12 m(w) x 3.98 m (h), producing a volume of 130 m².

Six room states were established through careful placement of absorbent material, Tontine Acoustisorb3, so that a relatively linear reverberation time spectrum was achieved. A dodecahedral loudspeaker was placed in the axial corner of the reverb room with the Soundfield™ microphone, set at a height of 1.2 metres, placed at distances of 0.9, 1.8 and 3.6 metres from the source. An additional measurement was taken at the furthest position with the microphone set at a height of 1.8 metres. The placements were chosen to firstly establish a 6dB difference in direct sound level between each position. This then places position 3 close to the centre of the room, on the horizontal plane and position 4 close to the centre on both horizontal and vertical planes. Steady-state measurements were made using broadband pink noise as the source material.
**Figure 4.1:** Measurement positions in test room.

**Table 4.1:** Transducer positions in test room (metres).

<table>
<thead>
<tr>
<th></th>
<th>Sp1</th>
<th>S/F1</th>
<th>S/F2</th>
<th>S/F3</th>
<th>S/F3</th>
</tr>
</thead>
<tbody>
<tr>
<td>(x)</td>
<td>0</td>
<td>0.56</td>
<td>1.13</td>
<td>2.25</td>
<td>2.25</td>
</tr>
<tr>
<td>(y)</td>
<td>0</td>
<td>0.7</td>
<td>1.40</td>
<td>2.8</td>
<td>2.8</td>
</tr>
<tr>
<td>(z)</td>
<td>1.2</td>
<td>1.2</td>
<td>1.2</td>
<td>1.2</td>
<td>1.8</td>
</tr>
</tbody>
</table>
4 Measurement of Diffusivity in a Reverberation-Controlled Room

Figure 4.2: Reverberation Time Plots of the six room states.

A simple calculation of room radius, averaged over the 500Hz and 1kHz octave bands, may be applied for each room state;

\[ r_H = 0.057 \frac{\sqrt{V}}{T} \]

where; \( r_H \) = the distance from the source where the energy density of the direct sound is equal to the energy density of the reverberant field.

\( V \) = room volume (\( \text{m}^3 \))

\( T \) = reverberation time of the room (seconds)

(From: (Cremer and Muller, 1978))

It may be argued, however, that the source has an effective directivity coefficient of 4 in the low and mid frequencies due to its placement in the corner of the room. This modifies the calculation;

\[ r_H = 0.057 \sqrt{\gamma} \frac{\sqrt{V}}{T} \quad \text{or} \quad 0.057 \sqrt{\frac{QV}{T}} \]

where; \( \gamma \) = directivity coefficient (Q)
Table 4.2: Room radius for an omnidirectional source and for the source placed in a corner of the room.

<table>
<thead>
<tr>
<th></th>
<th>Omni directional</th>
<th>Directivity = 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Room 1</td>
<td>1.68</td>
<td>3.36</td>
</tr>
<tr>
<td>Room 2</td>
<td>1.17</td>
<td>2.33</td>
</tr>
<tr>
<td>Room 3</td>
<td>0.99</td>
<td>1.98</td>
</tr>
<tr>
<td>Room 4</td>
<td>0.68</td>
<td>1.37</td>
</tr>
<tr>
<td>Room 5</td>
<td>0.45</td>
<td>0.91</td>
</tr>
<tr>
<td>Room 6</td>
<td>0.32</td>
<td>0.64</td>
</tr>
</tbody>
</table>

This places measurement position 1 inside the room radius for room states 1 and 2. All other measurement position-room states are deemed to be in the reverberant field, where the source is assumed to be omnidirectional.

If the corner placement is taken into consideration only position 3 is in the reverberant field for rooms 1 to 3, positions 2 and 3 are in the reverberant field for rooms 4 and 5 while all positions are for room 6.
4.2 Anechoic Room Measurements

Measurements where also taken in the anechoic room at the Australian National Acoustic Laboratory, providing a free field response of the microphone and consequently a value for $S_1$, the free field directional characteristic of the microphone.

This spring suspended anechoic room has a low frequency cut-off frequency of 50Hz. Its dimensions are 11.1m long, 9.1m wide and 7.1m high (wedge tip clearance). A Bruel and Kjaer dodecehederal loudspeaker was set at a height of 1.5m with the microphone placed, at the same height and 1.5m from the source. Broadband pink noise was used as the source material.

As outlined in the previous chapter, the area $S_x$ of each $360^\circ$ rotation is found by calculating the area of the triangle where the height of the triangle is the rms value for each $5^\circ$ rotation position and the base is the distance $2.5^\circ$ either side of the measurement position. The areas are then summed to produce a close approximation of the area contained by the directional plot. The horizontal and vertical plots for the six room states and four microphone positions are shown in Appendix B.

Figure 4.3 shows the resultant plots of a horizontal rotation measured in the anechoic room, the most absorbent room state (Room 1), with a reverberation time of 0.15 seconds, and the most reflective state (Room 6) with a reverberation time of 4.12 seconds. The solid line plot indicates a close to ideal cardioid plot measured in the anechoic room. The heavy broken line is a response that is close to that of the anechoic state, indicating that the room with the low reverberation time is close to the anechoic state. The light broken line plots a response that indicates that energy is present in all directions, approaching equal level, indicating that the most reverberant room state is close to that of a diffuse state.
Figure 4.3: Plots of horizontal rotation for anechoic room and measurement room with greatest variation in reflective states.

In Furduev's paper areas outside the unit circle were simply deducted from the area enclosed by the measured plot. In this way strong reflections off the direct source to microphone axis were accounted for in the consequent reduction in $\partial$. In this work the angular measured levels are normalised to the maximum level in order to standardise the presentation of the results. $\partial$ was calculated from the measurements using the normalised to maximum method.
4.3 $\phi$ Measurements

The $\phi$ values calculated for each measurement state are examined in the horizontal and vertical.

![Horizontal Graph]

**Figure 4.4:** Measured horizontal $\phi$ at positions 1-4 for the 6 reverberant room states.

![Vertical Graph]

**Figure 4.5:** Measured vertical $\phi$ at positions 1-4 for the 6 reverberant room states.
There is a clear trend of increasing $\delta$ values, indicated in Figures 4.4 and 4.5, as the reverberation time increases with each room state. This may be made clearer by plotting linear regressions for each of the measurement position $\delta$ values against the reverberation time averaged over the 125Hz to 4kHz octave bands.

Figure 4.6: Average reverberation time plotted against vertical $\delta$ for each room. (Note Rt for the 6 room states is indicated on the right axis of the chart)

As stated previously, there is an expectation that the field in the room will be more diffuse the higher the reverberation time is in that room. The $\delta$ measure shows some correlation to an increasing reverberation time. Also, as mentioned above, there is an expectation that some of the measurement positions in some of the room states are outside the reverberant field. In the case of measurement position 1 the measured $\delta$ values are consistently lower than the other positions implying a greater effect from the direct field. Positions 2 and 3 are quite close, at least in the regression lines, while position 4 shows a consistently higher result but converging on the values for positions 2 and 3 as the reverberation time increases.
There appears to be a point at which the room reaches a diffusivity maximum. In the horizontal measurements this occurs where Room 6 shows little increase over the Room 5 value. For the vertical measurements this occurs after the Room 4 value. Reference to the plots in Appendix B reveals that there is a skew in the vertical plots with a higher level arriving from above the measurement position than below it.

This is attributable to the difficulty in suspending the absorbent material from the roof compared with the ease of laying it on the floor. In these two room cases the distribution of absorbent material in the room was not even. Figure 4.7 illustrates the top bias in the results for the vertical plots with the consequent reduction in vertical $\delta$.

**Figure 4.7:** Polar plots for Room states 5 and 6 – posn. 3, vertical plot is dotted.
There is an expectation that the measured field within the room, for the less reflective room states, will vary depending on the distance between the source and receiver. As the measurement position is moved away from the source the ‘mix’ between the direct and reverberant energy will increase. This is illustrated in Figure 4.8 where, for Room state 2 there is a low diffusivity index measured but the value increases for each increase in distance from the source. The assumption here is that the direct sound dominates in level at position 1 but progressively the reverberant (diffuse) energy increases in relative level. By comparison all the positions in the highly reverberant Room 6 are similar in level. In this room state we can say that the field is tending toward homogeneity.

The degree to which a room is considered diffuse is dependent firstly on how isotropic the field is at a measurement position and then how homogeneous the field is throughout the room. A variance test is applied to the horizontal and vertical $\tilde{\varnothing}$ values for each of the six room states. (Figure 4.9)
There is a general trend in rooms 2 to 6 of a reduction in variance of the \( \hat{\partial} \) value. This is reversed for the room state with the highest absorption. This may be attributable to the measurement position, regardless of position in the room, being in the near field of the loudspeaker with the direct level dominating.

\[ \text{Figure 4.9: Variance in } \hat{\partial} \text{ values over 4 measurement positions} \]

4.4 Closer analysis of the \( \hat{\partial} \) plots

Reference to Figures 4.4 and 4.5 show that there is an almost linear increase in \( \hat{\partial} \) values found for position 1 as the room states become more reverberant. As indicated above, this is attributed to the changing balance between direct and reverberant sound at this microphone position/room state. This characteristic breaks down as the microphone position moves away from the source. Position 2 exhibits a similar characteristic up to Room state 4 but there is no significant increase in measured diffusivity for the more reverberant room states. Reference to the plots shows a strong reflection in the \( 150^0-180^0 \) direction again skewing the plot and consequently producing a lower \( \hat{\partial} \) value. This and the lower \( \hat{\partial} \) values in the vertical measurements, outlined above, are attributable to strong off-axis reflections producing asymmetrical plots. This is most obvious in Room 3 where a strong reflection is evident at the \( 330^0 \) direction.
4.4.1 Frequency analysis

A general expectation of a room, that is to some degree diffuse, is that the field’s diffusivity increases with frequency. In the assessment of the diffusivity index results above we have seen where a strong reflection can reduce the value of $\delta$. The question that is unable to be answered from the analysis of broadband noise is whether strong modal effects are causing the overall result to be shifted downward.

A further means of checking the diffusivity of the sound field is to assess the frequency variation in the results. Octave band filters were applied to the broadband measurements and the results plotted in Figure 4.10 for Position 3 in Room States 1, 4 and 6.

![Figure 4.10: Frequency Dependent $\delta$ for Room position 3 in three Room states.](image)

The lack of diffusion in the high frequencies for Room 1 indicates a lack of reflections due to the absorptive material in the room. The value for $\delta$ rises as the frequency is lowered due to reduced effectiveness of the absorbent material at the lower frequencies.

Room 6 exhibits a reasonably even spread of $\delta$ values, lowering in the 125Hz and 250Hz octave bands. Room 4 however exhibits an uneven diffusivity characteristic in the three lowest octave bands. This room may be characterised as a mid level diffuse space that has been shown in Figures 4.4, 4.5 and 4.10 to have an uneven diffusion character both in the frequency and spatial domains.
4.4.2 Temporal Diffusivity Measurements

Although a diffuse state is considered desirable in music performance auditoria, it is also evident that such a state would make it difficult to enjoy the music performance. A soundfield that instantly presents the source signal from all directions would make it impossible to locate the source potentially confusing the listener.

It has long been recognised (Randall and Ward 1960) that an uneven distribution of absorbent material may produce unevenness in a room’s decay slope. This is attributed to some resonant room modes being absorbed faster than others, effectively leading to the collapse of the diffuse field and the dominance of some room resonances as the sound in the room decays. Beyond recognition of a rise-time in a soundfield, characterised by the image source method (Eyring 1930), there appears to have been little investigation of the on-set of a diffuse soundfield.

Bradley and Soulodre (1995) propose that lateral energy arriving after 80ms provides a strong sense of envelopment. Therefore a more useful measure of the diffuse state may be the time taken for the sound field to establish the degree of diffusivity characteristic of that space.

Another series of measurements were taken at the same measurement positions and room states with sine sweeps generated by the Lake Huron™ as the source material. In this case a more directional Tannoy DMT15 loudspeaker was used as the source. The four channel measurements were analysed in the Huron DSP and Impulse Response wav files were generated. These files were then divided into 30mS sectors and analysed using the same Matlab code. As the ‘rise time’ of the diffuse state is the primary interest here only the first 270ms is examined. Position 3 for three room states was selected. Room 1 with a Rt of 0.15 seconds, Room 4 with a Rt of 1 second and Room 6 with a Rt of 4 seconds.
In Figure 4.11 we see that the average to high reverberant room states achieve a relatively high diffuse state within the first 90ms. The most absorbent room state (Room 1) reaches the highest level of diffusivity within a similar time-frame but the state collapses equally quickly. In the case of Room 1 any results beyond 0.15 seconds are indicative of the dominance of noise in the system rather than sound in the room. This is best illustrated by examining the room impulse response in Figure 4.12 where we find that the sound in the room has diminished to such a degree that it becomes difficult to separate it from noise.

Figure 4.11: $\partial$ measurements in 30ms duration time windows.
It is clear from the plot that after 90ms the field in the room has decayed significantly and that beyond 250ms, where the $\partial$ is showing a rise in value, there is no significant signal.

Further to this, examination of the polar plots in Appendix B illustrates the unevenness of the field relative to the initial 30ms window compared with the more even plots for the two more reverberant states. The results call into question the assumption that there may be a varying rise time for different diffuse room states. In this case it appears that the room has a particular rise time with the degree of diffusivity becoming established within the first 90ms. This may be a function of room volume rather than room reflectivity.
4.5 Conclusions

An updated version of a technique pioneered by Furduev was tested in a room in which the level of reverberance was varied by careful addition of absorptive material. The $\hat{\vartheta}$ measure was found to provide an adequate differentiation between room states but appeared to produce variable results for different positions in the room. It was unclear whether this is due to irregularities in the sound field in the room.

Frequency band $\delta$ measurement provides a more accurate assessment of the field in the room and helps to explain variations in the broadband results.

Temporal measurements present interesting if inconclusive results suggesting further research may be fruitful. There appears to be a good argument for establishing a form of diffuse field rise time criteria similar to the rise time of a reverberant field.

As indicated at the end of Section 2.2.2 this technique doesn’t appear to have been pursued beyond the Russian acoustic community with the last paper published in 1970. Of the two papers produced by Zakharin the 1968 paper is the most relevant. In that work he measured field diffusivity $d$ of a hall at different positions, finding that the measured results varied between 0.47 and 0.7 depending on the measurement position. It should be noted that the $d$ value was an average both of horizontal and vertical measurements and of select frequency bands between 100Hz and 1.4kHz. Although the results are from a real auditorium the span of results is similar to those produced in the experiment detailed in this Chapter.

Overall the method, due to its relative ease in performance and analysis of the measurements, appears to have some value in the assessment of sound fields in rooms. Further research into this method, in a larger number of rooms, such as auditoria, is required.
References


Chapter 5

Testing the limits of a Reverberant Field

The measurement of the diffusivity index based on the Furduev method exhibits some promise in establishing to what degree a sound field is isotropic. The question arises as to whether the method provides an adequate degree of accuracy in measuring the limits of a diffuse field.

It would therefore be interesting to compare the method using an ambisonic microphone with the standard method for establishing a sound field that is considered to be diffuse. To this end a comparison is made between the method of characterising a diffuse field, outlined in International Organisation for Standardisation ISO 354 – “Acoustics - Measurement of sound absorption in a reverberation room” and the previously established diffusivity index measurement method.

When measuring the absorption coefficient of a material or item in a reverberation room a fundamental assumption of the measurement is that the sound field in the reverberation chamber is diffuse (International Organisation Standardisation, 2008). The method of confirming that the field is diffuse is to measure the absorption coefficient of a sample in accordance with Annex A of the Standard. The Annex outlines a method of progressively adding diffusing elements to the reverberation room with the absorbing sample in place. Diffusing panels, equivalent to five square metres, are added with the average reverberation time over the frequency range 500Hz to 4kHz being noted. The expectation is that, as the panels are added to the room, the average measured absorption coefficient will increase. This increase in the absorption coefficient will reach a maximum beyond which the addition of further panels will not cause an increase in the measured average absorption coefficient. The number of diffusing panels at which the absorption coefficient reaches the maximum value is deemed to be the optimum number and the sound field in the reverberation room is considered to be diffuse.
This method is compared with the diffusivity index measured by the Furduev method. Correlation is sought between the Standard method and the Furduev method to establish the level of accuracy with which the $\delta$ measure can effectively measure the isotropy of the sound field.

### 5.1 Reverberation Room Standard measurement

The measurements were carried out in the reverberation room in the Architecture faculty at the University of Sydney. This is the same room that was utilised for the previous absorbent room measurements. In this case, rather than progressively adding absorptive materials to the room, reflective panels were added.

The room under test is a rectangular, painted concrete and rendered masonry room measuring 6.36 m (l) x 5.12 m(w) x 3.98 m (h), producing a volume of 130 m$^2$ and a total surface area of 156.5 m$^2$.

The basic assumption in measuring the absorption coefficient of a sample in a reverberation room is that the sound field is diffuse. This is based on the expectation that such sound absorbing materials will be subject to unpredictable sound fields in the environments they are placed in. Consequently how the material reacts when exposed to a random incidence or diffuse sound field is considered to be a close approximation of the acoustic spaces in which the material is placed (International Organisation Standardisation, 2008). As we will see this is not actually possible but we nonetheless work towards this ideal state.

International Organisation for Standardisation ISO 354 specifies the following conditions are to be met in the space used to measure sound absorption. An indication will be made of where the test room does not meet the specification.
5.1.1 Volume and shape of the room

The room should have a volume of at least \(150\text{m}^3\). (The test room has a volume of \(130\text{m}^3\).) \(I_{\text{max}}\) (the largest dimension of the room) \(<1.9 \text{ V}^{1/3}\). For a room volume of \(150\text{m}^3\) this limit would equate to 10.09m. (The room under test, with dimensions; \(x = 6.366\text{m}\) (length), \(y = 5.121\text{m}\) (width), \(z = 3.985\text{m}\) (height), produces an \(I_{\text{max}}\) of 9.09m.)

As Shultz has indicated the volume of the room is fundamental to the establishment of a diffuse field (Schultz 1971). The number of resonant modes and directions of arrival in the room increases with frequency. The low frequency limit of diffusivity in a space is therefore dependent on the room size. Where the lower frequency modes are scarce there are a reduced number of angles of incidence to the material. Different angles of incidence to the sample result in the sound traversing different thicknesses of the material. Angles of incidence close to parallel to the sample surface will travel through more of the material before reflection compared with sound impinging close to perpendicular to the material. Where the angles of incidence to the material approach a large number the varying amounts of absorption due to the sound traversing different thicknesses of the material average out to the random incidence absorption coefficient. Where there are a reduced number of incident angles, due to a lack of modal density, the measured absorption coefficient becomes unreliable.

Further to this Kosten (1960) has found that rooms smaller that \(180\text{m}^3\) have difficulty in maintaining a diffuse field when a highly absorptive sample, with an area of 10 to 12 \(\text{m}^2\) is used. There is a trade off between using a sample large enough to minimise the influence of the ‘edge effect’ whilst maintaining a room volume large enough to sustain a diffuse field in the presence of a relatively large area of absorptive material.

In the case of the room under test the smaller volume is offset by the smaller sample area. The degree to which a diffuse field may be maintained will be tested while the smaller sample size may result in a measured absorption coefficient that is higher due to diffraction effects from the edge of the sample.
According to ASTM 423-66 (after Schultz) if the volume of the room is four times the cube of the wavelength there will be at least 20 modes in the room within a third octave band centred on that frequency. Calculating the cube of the wavelength based on a room volume of 130m$^3$ produces a frequency of 106Hz (101Hz for a volume of 150m$^3$), implying that the test room will be adequately diffuse in the 125Hz third octave band.

Kuttruff (Kuttruff 1973) on the other hand, proposes the cut-off frequency to be;

$$f_g = \frac{100}{\sqrt[3]{V}}$$

where; $V =$ room volume (m$^3$)

For the test room this equates to 197Hz (188Hz for a volume of 150m$^3$). For reasons indicated below frequencies below 125Hz will not be considered in this work.

Schroeder has proposed a calculation that has become known as the ‘Schroeder frequency’;

$$f_c = 2000 \frac{T}{\sqrt{V}}$$

where; $T =$ Rt$_{60}$, Reverberation time (secs)

$V =$ room volume (m$^3$)

For the empty room with Rt averaged between 500Hz and 4kHz of 6 seconds, $f_c = 429$Hz. (For a room with the same reverberation time that met the standard requirement of 150m$^3$ $f_c$ would be 400Hz). Clearly the rooms would be deemed to have adequate modal density in the frequency range above 500Hz, according to Schroeder.

It has been argued (Kuttruff 1998) that the Schroeder frequency is intended to be used in large spaces such as concert halls and is less appropriate for ‘small’ rooms. A ‘small room’ approach is found in the application of the Bonello criteria. (Bonello 1981) proposes, for rectangular rooms, that the number of resonant modes in a third octave band must be equal to or greater than the third octave below it. The expectation is that with each increase in frequency band the number of resonant modes will increase. The resonant frequencies for the test room were calculated producing a mode count in the low frequency range below 300Hz.
Figure 5.1: (a) Calculated reverberation room modes. (b) Bonello criteria for the room.
In a room suffering from poor modal distribution a dimensional mode in one part of the room coincides in frequency with another resonant mode. For example an axial mode may ‘support’ a tangential mode of a similar frequency. Where such support occurs the relative mode support, plotted on the Y-axis of 5.1(a) will be 2 at that frequency.

From figures 5.1 above we can see that there is no modal coupling due to dimensional support of mode harmonics. It is clear that the room is modally dense above 130Hz. It also fits the Schultz criterion of having more than 20 modes within a third octave band above 125Hz. The room also meets the Bonello criteria in having more modes in each one-third octave band than the proposed ‘Ideal’ criteria.

From this we can conclude that the room under test is modally dense above 400Hz, according to Schroeder, and may be adequately diffuse above 100-130Hz, according to Schultz and Bonello. However, as we will see below, the poor signal to noise ratio is in fact a greater limiting factor in the measurement than the room’s capacity to support the lower frequency sound.

### 5.1.2 Equivalent Sound Absorption Area

The method for finding the equivalent sound absorption area of the sample is by calculating the difference between the equivalent sound absorption area of the room with and without the sample. The room must meet the criteria that its equivalent absorption area is below specified values and the variation between third octave bands be less than 15%.

To check that the measurement set-up meets these criteria the measured equivalent absorption area of the empty room is plotted against the maximum permissible values.

The standard requires an adjustment where the room volume is less than 200m²:

\[
\left(\frac{V}{200}\right)^{\frac{2}{3}}
\]

= 0.75

That adjustment has been made for both the measured values of equivalent absorption area (A₁) and the maximum values. It should be noted that the standard specifies a value at the frequency 100Hz. That measurement was not made due to the poor signal to noise ratio. The dip from 125Hz to 160Hz exceeds the acceptable 15% variation but beyond that the higher frequency values are within that range specified by the standard.
Figure 5.2: Measured Equivalent Absorption area (A1) plotted against the standard required by ISO354

5.1.3 Test specimen

The test specimen used was a plane absorber, which according to the standard, needs to have an area of between 10 and 12m$^2$. The sample is to be rectangular with a length to width ratio of between 0.7 and 1. It should be placed so that no part of the sample is parallel to a wall and must be more than a metre from any boundary edge.

Waterhouse (1957) has shown that the sample must be placed at least half a wavelength from a reflecting wall and other reflecting surfaces. In this case, where the sample is one metre from the sidewall, the lowest frequency that would not suffer from interference is 171.5Hz. The sample must be mounted in accordance with one of the methods outlined in Appendix B of the standard in order to avoid the “edge effect”.

The specimen used was three pieces of Tontine Acoustisorb3, a 50 mm thick polyester batt. The samples were butted together and mounted in accordance with Type A outlined in appendix B of the standard. The area covered by the sample was 3.12 m x 2.16 m, a total area of 6.74 m$^2$, producing a length to width ratio of 1: 0.69. The absorption coefficients of the sample, published by the manufacturer, is listed in Table 5.1;
Table 5.1: Reported random incidence absorption coefficient of sample

<table>
<thead>
<tr>
<th>Oct Band Centre Frequency (Hz)</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1k</th>
<th>2k</th>
</tr>
</thead>
<tbody>
<tr>
<td>Absorption Coefficient</td>
<td>0.40</td>
<td>0.70</td>
<td>0.91</td>
<td>0.95</td>
<td>0.94</td>
</tr>
</tbody>
</table>

The specimen is smaller than outlined by the standard but when fitted into the smaller room met the criteria that it not be mounted parallel to the walls and that it be at least one metre from the walls.

The Standard specifies minimum room and sample size to ensure that the relationship between the absorbent surface in the room and the overall room allows a diffuse field to be created within the room without the sample overly reducing the energy in the room or producing significant diffraction effects. Although the room under test does not meet the criteria set out by the it can be expected to act in accordance with the Standard above some frequency. It has been shown that the placement of the sample in the room makes the measurement space reliable above 170Hz but the modal density and the signal to noise ratio indicate taken into consideration in the work that follows.

The Standard applies a test of the reliability of the measurement system as a whole. This test, outlined in Section 5.1.2 above, shows that the measurements are reliable within the frequency range under test.

5.1.4 Temperature and Relative Humidity

The standard specifies the temperature and relative humidity should not change significantly over the period when measurements are made in the empty room and the room with sample, to avoid significant differences in air absorption in the different measurement states. Overall recommendations are; relative humidity should be between 30% and 90% and room temperature must be at least 15°C.
In accordance with the standard the temperature and relative humidity were measured over the several days that it took to perform all the measurements. The temperature ranged from 23.6º to 24.1ºC with the relative humidity varying between 51.9 and 62.6%. Barometric pressure data for the measurement days were drawn from the Australian Bureau of Meteorology.

5.1.5 Noise and source level

The standard states that the source signal must be 10 dB above the noise level at the end of the evaluation range and that the decay be measured from -5 to -25 dB ($T_{20}$). However, in this work the measurements were made from -5 to -35 ($T_{30}$) to provide a more accurate measurement of the room decay. Using the $T_{30}$ measure consequently requires the stimulus signal to be 45dB above the noise floor of the room. The noise level was established by recording the output of one of the microphone channels to the Bruel and Kjaer data recorder. A five minute recording was made at the same time of day as the measurements were made. The recording was then analysed using the CPB analyser of the B&K Pulse system.

Figure 5.3 below, plots the measured pink noise output of the dodecahedral loudspeaker, measured in an anechoic room at a distance of 1 metre, with the same amplifier settings as those used in the measurements. Against this is plotted the noise measurement of the reverberation room. The capacity of the loudspeaker to output low frequency signal diminishes significantly below 250 Hz. To maintain a difference greater than 45 dB between signal level and room noise results in all measurements below 125 Hz and above 16 kHz were ignored. (The actual difference at 125 Hz is 43 dB.)
5.1.6 Measurement Method

The standard specifies that the loudspeaker and measurement microphones be omnidirectional. The microphones should be 1.5 to 2 metres from the source and at least 1 metre from any room surface or test specimen. The source positions are recommended to be at least 3 metres apart. There should be at least twelve spatially independent measurements. Either the interrupted noise or the integrated impulse response method may be used to generate the decay curves. In the first section of tests the interrupted noise method was used. The integrated impulse response method was used in the second stage.

There are two stages to the measurements in the room. Firstly the reverberation time was measured using three loudspeaker positions and four microphone positions, producing a total of 12 spatially independent decay curves, in accordance with the Standard. The test geometry is detailed below;

**Figure 5.3:** Signal and background noise levels in the reverberant room.
Figure 5.4: Measurement plan of the test room.

Table 5.2: Transducer positions in test room (m)

<table>
<thead>
<tr>
<th></th>
<th>Sp1</th>
<th>Sp2</th>
<th>Sp3</th>
<th>Mic1</th>
<th>Mic2</th>
<th>Mic3</th>
<th>Mic4</th>
<th>S/F1</th>
<th>S/F2</th>
<th>S/F3</th>
</tr>
</thead>
<tbody>
<tr>
<td>x</td>
<td>4.12</td>
<td>1.12</td>
<td>2.02</td>
<td>3.02</td>
<td>2.12</td>
<td>4.02</td>
<td>1.32</td>
<td>4.27</td>
<td>2.56</td>
<td>1.7</td>
</tr>
<tr>
<td>y</td>
<td>2.8</td>
<td>1.2</td>
<td>5.2</td>
<td>1.0</td>
<td>3.0</td>
<td>4.8</td>
<td>3.3</td>
<td>2.1</td>
<td>3.18</td>
<td>5.36</td>
</tr>
<tr>
<td>z</td>
<td>1.55</td>
<td>1.9</td>
<td>2.4</td>
<td>1.1</td>
<td>2.8</td>
<td>3.0</td>
<td>1.4</td>
<td>2.5</td>
<td>2.0</td>
<td>0.65</td>
</tr>
</tbody>
</table>

Table 5.2 provides the length (y), width (x) and height (z) coordinates for each of the transducers in the room. All measurements are in cm. The position and orientation of the sample on the floor of the room is illustrated by the large skewed rectangle within the enclosure.
5.1.7 Measurement Procedure

The Bruel and Kjaer Pulse Labshop 12.6 system was used with the reverberation time template. The source, an Outline dodecahedral loudspeaker, can be divided into four separate output sections. For this set of measurements the signal from the Pulse system was sent to all four zones via a Brüel and Kjær 2716C amplifier. The microphones were B&K 4189 transducers attached to the Pulse 3109 interface. The source material was interrupted noise generated through the Pulse front-end. Three loudspeaker positions were selected in places that were calculated to be away from dominant nodes or antinodes of the lowest room resonances. Likewise four B&K 4189 microphones were placed to avoid obvious nodes or anti-nodes in the space. The interrupted noise source was run three times for each position with the Pulse system automatically averaging the results. Measurements were made for each of the loudspeaker–microphone configurations, a total of 36 measurements for each room state, output to twelve reverberation plots for each room condition. This set-up was used as the control for the room measurements by providing standard reverberation time measurements.

The second stage of the measurement procedure was to run three test sources from a ProTools digital audio workstation running four separate channels of audio through a Digidesign 003 audio interface. The four outputs were amplified with a Crown CP6600 amplifier connected to the four separate zones of the Outline loudspeaker. Correlated and de-correlated noise and swept sinusoids were output to the dodecahedral loudspeaker. The noise sources were intended for other work carried out by researchers in the Faculty and will not be considered here. The 20s swept-sine signal was replicated across the four channels, effectively producing the same signal from all components of the loudspeaker. A Soundfield™ SPS422B microphone was placed in three positions in the room. Like the B&K microphone positions, two of the ambisonic microphone placements were established in positions that aimed to avoid dominant resonance nodes or anti-nodes. The second position for the microphone (S/F2) however was in the centre of the room. The microphone was aligned with the x-axis of the room. The outputs of the Soundfield B-Format processor were recorded, along with the B&K microphone outputs, with the Brüel and Kjær Pulse Time Data Recorder for later processing and analysis.
The four recorded sinusoidal sweeps from the B-Format processor were imported into Auditec Audition and deconvolved from the original sweep signal using the Aurora (Farina 2000) package. The resulting four impulse responses were loaded into Matlab for analysis.

5.1.8 Room states

The reflectors used were Perspex panels 1220 mm x 915 mm x 5 mm, 1.1163 m² per panel (on each side). The standard recommends adding 5m² of panels per room state variation. For this test 5 panels were added for each room increment.

There is a question of how the panels effect the field in the room. Three distinct effect stages may be identified although it is understood that there will be transition zones between the following three states;

1) relative to the wavelength of the sound the panels dimensions are insignificant so there is negligible effect on the sound due to the panels.
2) around a range of frequencies the panel will act as a diffracting element resulting in some localised cancellation of energy due to phase effects.
3) where the panel is large relative to the frequency of the sound it will act as a reflector.

Primakoff, Klein et al. (1947) showed that a ‘shadow region’ occurs behind a circular disk where;

\[ x_s = 1.5 \left( \frac{x^2}{\lambda} \right) \]

\[ x_s = \text{distance from disk 'in shadow', } x = \text{radius of the disk} \]

Calculating the two dimensions of a panel: 0.915metres and 1.22metres against frequency provides an indication of the frequency range where the panel face becomes a significant obstacle to sound energy in the room. The frequencies at which a shadow region of 1 metre is created are 1.1kHz and 609Hz respectively for 0.915m and 1.22m. We can therefore assume that the panels are significant reflectors for frequencies above 1kHz with them having some effect in the lower range of the measurement.
5.1.9 Reverberation time measurements

The reverberation time was measured in the empty room, the room with the sample and with reflecting panels added five at a time. The panels were ‘randomly’ suspended in the room. The means of achieving this ‘random’ distribution was to purposely not think about where each panel was placed. The intention is to avoid any tendency to place the panels in an ordered fashion, consciously or otherwise.

The maximum number of panels in the room was 25. At this state the distance between measurement microphones and reflective panels was no longer greater than 1 metre. The Standard does not clearly state this as a requirement but it specifies that the measurement microphones must be at least 1 metre from the walls. The panels are significant reflectors and as such the proximity to the microphones, particularly in the high frequencies would be expected to produce unreliable results due to modal effects in the frequency range related to the distance between the reflector and the measurement microphone. A final reference measurement was made with 25 panels in the room, without the sample to establish the room plus panels absorption.

The process for establishing whether the sound field in the reverberant room is diffuse, outlined in AS ISO 354, is to progressively add reflective panels to the room with a sample present. The area of panels added, in each instance, should be in the vicinity of 5 m². The field is deemed diffuse when the measured absorption coefficient reaches a maximum and remains unchanged if even more reflectors are added to the room.

To confirm this is the case the equivalent absorption coefficient of the sample in the room and the empty room must be calculated in each frequency band. This is a three-stage process, firstly calculating the sound absorption area of the empty room, based on the following formula;

\[ A_1 = \frac{55.3V}{cT_1} - 4Vm_i \]

where; \( V \) = volume of empty reverberation room (m³)

\( c \) = the speed of sound (ms⁻¹)

\( T_1 \) = reverberation time of empty room (sec)

\( m_i \) = power attenuation coefficient calculated from;

\[ m_i = \frac{\alpha}{10\log(e)} \]

where; \( \alpha \) = sound attenuation coefficient (dB/m)
The source to receiver distance does not exceed 6km and hence the pure tone method of calculating $\alpha$ is an acceptable approximation; ref ISO9613-1.

The calculation for the speed of sound is now required. The Standard recommends;

$$C = 331 + 0.6t$$

where; $t =$ temperature ($^0C$ ) of the room during the measurement.

The room temperature varied over the three days the measurements were taken: from 23.6$^0C$ to 24.1$^0C$. This produced a variation in the speed of sound between 345.16 and 345.46m/s. The sound absorption area of the room follows this with the sample in it. The new values of $T$ and $m$ are substituted.

The equivalent sound absorption area of the specimen is found from;

$$A_t = A_2-A_1 = 55.3V\left(\frac{1}{c_2T_2} - \frac{1}{c_1T_1}\right) - 4V(m_2-m_1)$$

In the calculation above it is assumed that the value of $m$ will not change significantly making the second half of the calculation, $-4V(m_2-m_1)$ $\approx$ 0. The assumption is made on the basis that this component of the calculation accounts for the effect of sound absorption in the air. To test that the small variations in temperature and humidity over the course of the measurements did not have a significant effect on the overall room absorption the following calculations were made according to ISO 9613.

The total absorption of sound in air consists of a number of effects influenced by temperature, humidity, ambient air pressure and the frequency of the sound. The first stage of the calculations is to convert the relative humidity, the molar concentration of water vapour in the air ($h$), outlined in Annex B of ISO 9613.
5 Testing the limits of a Reverberant Field

\[ h = h_r \left( \frac{p_{\text{sat}}}{p_r} \right) / \left( \frac{p_a}{p_r} \right) \]

where; \( h_r \) = relative humidity

\[ p_{\text{sat}} = \text{water vapour saturation} \]

\[ p_r = \text{reference ambient pressure, 101.325kPa} \]

\[ p_a = \text{atmospheric pressure on the day of the measurement} \]

\[ \frac{p_{\text{sat}}}{p_r} = 10^C \]

where; \( C = -6.8346 \left( \frac{T_{01}}{T} \right)^{1.261} + 4.6151 \), \( T = \text{temp (}^\circ \text{K)} \)

\( T_{01} = \text{triple point isotherm temperature, 273.16}^\circ \text{K} \)

From this we find \( C_{\text{day1}} = -6.8346 \left( \frac{273.16}{(273.15+23.9)} \right)^{1.261} + 4.6151 \)

\[ = -1.53351 \]

\( C_{\text{day2}} = -1.5285 \]

\( C_{\text{day3}} = -1.5416 \]

\[ \frac{p_{\text{sat}}}{p_r} (\text{Day1}) = 10^C = 10^{-1.53351} = 0.0295 \]

\[ \frac{p_{\text{sat}}}{p_r} (\text{Day2}) = 0.0296 \]

\[ \frac{p_{\text{sat}}}{p_r} (\text{Day3}) = 0.0287 \]

\[ h_{\text{Day1}} = h_r \left( \frac{p_{\text{sat}}}{p_r} \right) / \left( \frac{p_a}{p_r} \right) = 53.7(0.0295)/(101.76/101.325) = 1.564 \]

\[ h_{\text{Day2}} = 1.521 \]

\[ h_{\text{Day2}} = 1.781 \]

The attenuation coefficient is then calculated based on the ‘classical’ attenuation of sound in air plus the relaxation effects of the two primary elements in air, Oxygen and Nitrogen. The ‘classical’ attenuation of sound refers to frictional losses in the medium. The relaxation effects are caused by the rotation of Oxygen and Nitrogen particles when impacted by particles in a sound wave. This non-linear rotational effect causes additional losses to the energy flow. This is illustrated in the following formula where the \( fo \) and \( fn \) components are calculated from the ambient air pressure over the measurement period.
\[
\alpha = 8.686 \times 10^{-2} \left[ 1.84 \times 10^{-11} \left( \frac{p_u}{p_r} \right)^{-1} \left( \frac{T}{T_0} \right)^{1/2} \right. \\
\left. + \left( \frac{T}{T_0} \right)^{1/2} \times 0.01275 \left[ \exp \left( -2239.1 \frac{1}{T} \right) \right] \left[ f_o + \left( \frac{f^2}{f_o} \right) \right] + 0.1068 \left[ \exp \left( -3352.0 \frac{1}{T} \right) \right] \left[ f_N + \left( \frac{f^2}{f_N} \right)^{-1} \right] \right]
\]

where; \( T \) = room temperature in \(^0\text{K}\)

\( T_0 = \) reference air temperature, 293.15\(^0\text{K}\)

The results of the calculations above were plotted against the data presented in ISO9613 for expected results at 25\(^0\text{C}\) and 60% humidity. There is a close correlation in the results.

**Figure 5.5:** Calculated air attenuation coefficient and ISO 9613 quoted results for 25\(^0\text{C}\), 65% humidity.

The absorption coefficient of the sample is then found from;

\[ \alpha = At/S \quad \text{where; } S \text{ = surface area of the sample} = 6.74 \text{m}^2. \]
5.1.10 Reverberation Time results

Three measurements were taken for each measurement state. The results were averaged within the Pulse system to produce a single reverb time – frequency plot. The four microphone position averages are plotted for each of the microphone positions producing twelve plots for the total thirty six measurements in each room state. The plot for the empty room is shown below, plots for all room states are displayed in Appendix C;

![Reverb Times - Empty Room](image)

**Figure 5.6:** Measured Reverberation Times for all Source – Receiver positions in the empty Reverberation Room.

The average reverberation time measured at each of the room positions is then averaged to produce the plots for each room state below. The numbers in the legend (+5, +10, etc) indicate the number of diffusing panels in the room.
5 Testing the limits of a Reverberant Field

![Figure 5.7: Spatially averaged reverberation time for each room state.](image)

![Figure 5.8: Standard deviation of reverberation times across measurement positions.](image)
There was significant variation in the measured results attributable to spatial variance in each of the room states. Of particular interest is the difference in average reverberation time for each of the room conditions. There is approximately a 1 second difference between the empty room reverberation time and the T for the room with no sample but 25 reflective panels. This may be explained by the reduction in mean free path resulting in the sound being absorbed by the greater number of reflections introduced by the diffusing elements. There may, however, be other effects occurring in the empty room with diffusing panels. Reference is made to the effect by Cox and D’Antonio, based on the work of Hargreaves. Bennedetto et al. (Bennedetto, Brosio et al. 2002) has also reported similar results with measured reverberation times varying by up to 60% in some frequency bands by adding diffusors to an empty room.

At a macro level it is interesting to note that the Rt is less for the empty room with panels than without. This is similar to the results reported by Bennedetto. This would be expected within a limited frequency range, where the panels absorbed energy due to panel resonance but these results indicate a wideband reduction in Rt. Further to that we can see that the Rt doesn’t vary a great deal above 2000Hz once some panels are placed in the room. It is this later case that this report will focus on later in the chapter.
5.2 The diffuse state – ISO 354

The standard specifies, in Annex A, the conditions under which a diffuse state is achieved. It proposes that a sample be placed in the room in accordance with the standard and its absorption coefficient be measured. Panels are added to the room in increments of 5 m² for each room state. The absorption coefficient for all measurement positions is averaged. The mean absorption coefficient should increase to a point where it reaches a maximum. This is the point where random incidence is achieved and the sample presents a range of ‘depths’ to the impinging sound, depending on the angle of incidence. Beyond that point the addition of further panels causes no further increase in mean absorption coefficient. The room state where the maximum is reached is deemed to be diffuse.

![Average absorption coefficient, all room states](image)

**Figure 5.9:** Mean absorption coefficients of sample with varying numbers of diffusing panels in the reverberation room.

The figure above indicates that the room is sufficiently diffuse when 15 panels are placed in the room. The measured absorption coefficient is above 1 for frequencies above 500Hz when more than 5 panels are placed in the room. This result is usually attributed to the edge effect of the panel placed within the room (Kosten 1960).
The process of averaging multiple measurements masks the degree of variance in the results. An alternate means of interrogating the data is to plot the variation in the results in the box-plots below;

![Box-plots showing variation in absorption coefficient](image)

**Figure 5.10:** Plot of absorption coefficient variation over all measurement positions in octave bands for each room state.
The box plots in Figure 5.10 illustrate the distribution of measured results over the 12 measurement configurations plotted against the number of reflective panels placed in the room. The line inside the box is the median with the upper and lower result quartiles enclosed within the box. The upper and lower extents of the results are marked by the T marks. Possible outliers are indicated by the + sign.

In the octave bands above 1kHz there is a clear trend indicated with the absorption coefficient reaching a maximum value from 15 panels upward. The amount of variation in results at these frequency bands is low, as indicated by the standard deviation in Figure 5.11 below.

**Figure 5.11:** Standard deviation of the absorption coefficients for each room state.

It is clear that when the standard states that the panels are added until there is no further change in absorption coefficient this needs to be read within the context of acceptable variation in results in the experimental condition.
In Davern and Duboit’s (1980) round-robin tests of reverberation rooms, in Australia and New Zealand, they found that the ratio of diffusor area to floor area was a contributing factor in the repeatability of measurements.

They found for the value \( \Delta = \frac{S_d}{S_f} \) where \( S_d = \) area of diffusors (both sides)

\[ S_f = \text{floor area of test room} \]

This was reflected, at the time and now, in ISO 354 where it provides a general guideline that the diffusor area (both sides) should be 15 - 25% of the total surface area of the room. They found that a value greater than 1.4 was desirable but reported that both the Australian and International standards recommended a value of 2.

**Table 5.3:** Comparison of diffusor area vs floor area and surface area of the test room.

<table>
<thead>
<tr>
<th></th>
<th>5 panels</th>
<th>10 panels</th>
<th>15 panels</th>
<th>20 panels</th>
<th>25 panels</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sd/Sf</td>
<td>0.068</td>
<td>0.685</td>
<td>1.028</td>
<td>1.371</td>
<td>1.714</td>
</tr>
<tr>
<td>Sd/St</td>
<td>0.071</td>
<td>0.143</td>
<td>0.214</td>
<td>0.285</td>
<td>0.357</td>
</tr>
</tbody>
</table>

According to the two criteria above, 20-25 panels are required to produce a diffuse field using the Davern and Dubout calculation while 10-15 panels are required according to the rough calculation proposed in the standard.

The experimental results show, for the purpose of producing a reliable absorption coefficient measurement, the placement of 15 panels within the room is adequate. The variation in the results beyond the point where, according to the standard, the room is deemed to be diffuse indicates that the sound field has not reached a homogenous state.
5.2.1 Conclusions on ISO 354

It must be remembered that the intention of the Standard is to advise on the means of establishing a reliable measurement condition within which the absorption coefficient of a material may be measured. Such measured coefficients are rarely quoted with an experimental error, so the user isn’t provided with a clear indication of the potential variance in the final result.

The test has shown that the recommendations in Appendix 1 provide adequate guidance to allow a reasonably reliable measurement of the absorption coefficient of a sample. The variation in the results across the different measurement positions in the room, however, indicate that the sound field in the room has not reached a truly diffuse state with even 25 panels in the room. This is clear from Figure 5.12 below where we can see the variation in measured absorption coefficient dependent on source-receiver position. If the sound field in the room was truly diffuse the assumption is that the sound energy in the room would be absorbed equally regardless of source-receiver position. Of course, once an absorptive element is placed within even a truly diffuse field that state of diffuseness will be reduced through the action of the sample’s removal of energy from the field. This reduction in the state of diffusivity is borne out, to a degree, in the Ambisonic microphone measurements.

![Sample + 25 panels- Abs Coeff](image)

**Figure 5.12:** Measured absorption coefficient for reverb room with 25 panels and all measurement configurations.
5.3 The $\bar{\delta}$ measurements

To assess how well the $\bar{\delta}$ measurement indicates the presence of a diffuse field, the room impulse responses measured by the B-format Soundfield microphone were processed in Matlab to produce the equivalent of a cardioid microphone pointing in 72 directions on the horizontal plane, the equivalent to $5^0$ rotations on the circle. The output was filtered in third octave bands and the amplitude at each orientation was normalised to the maximum amplitude. The normalised amplitudes in the 72 directions, for each frequency band, were then used to calculate the area inside the plot of the virtual rotating microphone. This value was used to produce $S$ for calculation of the diffusivity index originally proposed by Furduev and T’ung. The results in Figure 5.13 indicate that there is a fair degree of variation across the frequency bands for each of the room states but the overall trend shows that, according to this method, the sound field in the reverberation room is not diffuse but is approaching the state, indicated by a value of 1, in each of the room conditions. The results, like the absorption coefficients, are quite variable making it difficult to detect a trend.

![Diffusivity Index for all room states](image)

**Figure 5.13:** Diffusivity Index for loudspeaker 1 – Soundfield 1 configuration in all room conditions.
The box plot method of interrogating the data may be useful in this case but it should be recognised that box plots are usually used to represent large data sets. In this case there are only nine data points within any column.

Examination of the horizontal and vertical diffusivity index values plotted in Appendix D reveals that, with the exception of the 125Hz octave band, the room is most diffuse when it is empty. There is a tendency for the $\delta$ to drop when the sample is introduced and then rise as the panels are added. This is best demonstrated in the plots for the 1kHz octave band below in the vertical plane but in the horizontal plane the mean $\delta$ remains fairly constant as the panels are added. This may be explained by the sample acting to reduce the diffuseness of the field in the room. In horizontal plane of measurement the sample has less direct influence than in the vertical.

Figure 5.14: Horizontal and vertical box plots of df in the 1kHz octave band for all room states.
(Note; On the horizontal axis ‘1’ represents the room+sample, 0-25 represents the sample plus 0-25 panels)

The Soundfield directivity plots provide us with a visual indication of this. Producing a polar plot of the normalised levels from the 72 directions sampled in the horizontal and vertical planes respectively generates these plots.
In Figures 5.15 and 16 below, the amplitude plot in both the horizontal and vertical planes show difference between the room with the sample and the empty room with 25 diffusing panels for the Speaker 1 – Soundfield 1 measurement configuration. This measurement configuration places the loudspeaker at about 30° in the horizontal plane 30° on the vertical plane.

**Figure 5.15**: Horizontal and vertical plots of amplitude from a virtual rotation of a cardioid microphone in the empty reverberant room with sample, for Speaker 1-Soundfield1 orientation. (Horizontal plot on left, vertical on right)

**Figure 5.16**: Horizontal and vertical plots of amplitude from a virtual rotation of a cardioid microphone in the reverberant room containing 25 diffusor panels, without sample, for Speaker 1-Soundfield1 orientation.
The difference is most marked in the vertical plots where there is greater amplitude in the lower hemisphere when there are diffusing panels in the room but no sample. The field, according to this measurement, is close to diffuse with less energy arriving at the microphone in the 230° to 270° direction. This may be attributed to two factors; firstly the loudspeaker direct signal dominates the level measured in each case and the sample, in this case is placed in the hemisphere below the ambisonic microphone.

5.3.1 Ambisonic measurements conclusion
The intent of this chapter was to test whether the Ambisonic microphone method could provide an accurate measure of a field that was approaching diffusivity. The measurements, in Figure 5.14 shows that the highest diffuse state is measured in the empty room. The results here show that the reverse may be true, that the measurement provides a clear indication that the field does not reach that of a truly diffuse state but the sample acts to prevents the room from reaching that state. This is unsurprising as the absorbent sample will effectively prevent the field from reaching complete diffusivity as it removes energy from one surface in the room. It is therefore interesting to test the space without an absorptive sample.

5.4 Reverberation Room measurements without sample
The \( \delta \) measure produces values around 0.75 to 0.9 for the various sample plus reflective panel states when a sample is present. The question remains as to whether the measurement method will be accurate enough to adequately resolve small enough changes as the room approaches a highly diffuse state. It is also interesting to explore the apparent absorptive effects of the panels in the room, highlighted earlier. To test this, another series of measurements were conducted in the same reverberation chamber. For these tests a sine-sweep signal was played from the same playback system outlined above but through a Bruel and Kjaer 4292 dodecahedral loudspeaker. The sine-sweeps were recorded using the same method, measurement microphones and positions as the previous tests. These recordings were exported as Matlab files for later analysis.

The impulse responses were derived using the Matlab script in Appendix A and then processed using the same Matlab script as was used in the previous analysis.
5.4.1 Standard measurement

To assess the apparent absorptive effect of the panels the absorption coefficient calculation was applied in accordance with ISO 354. The difference, in this case, is that the surface area of two sides of the panels is substituted for the sample area in the calculation. As outlined previously, the Standard proposes that the surface area of both sides of the panels should be calculated, this calculation produces the results as shown in Figure 5.17. Overall, the apparent absorption of the panels reduces toward the empty room coefficient as the number of panels in the room is increased. This trend reverses for the 25 panel case.

The negative absorption coefficients are produced when the total absorption in the room with panels installed is less than the empty room. This occurs in the case of 5 panels in the room at frequencies below 160Hz. This may be attributable to the panels enhancing modal support in that frequency range. In the same frequency range where more panels are placed in the room there is an increase in absorption. This would be expected where the panels act as absorbers. Kuttruff (2007) suggests that even a smooth surface will have an absorption coefficient of about 0.01 due to surface effects effectively absorbing sound impinging on the surface. What is unexpected is the strong peak in absorption centred around 1.6kHz for the 5 panel case and to a lesser degree in the other cases.
5.4.2 \( \varphi \) measurements – room without sample

Bradley and Soulodre (1995) have suggested that the early section of the impulse response should be removed from the analysis of the diffuse or enveloping field. This was seen in Chapter 4 where the development of the diffuse field occurred over a specific time interval. In order to check this the \( \varphi \) values for each room state were measured, for each speaker-Soundfield\textsuperscript{TM} position, with both full impulse response data and the response trimmed by the removal of the first 80ms of data. The results are plotted in Appendix D with the two cases, no panels and 15 panels, displayed in Figure 5.18 (a + b) below;
Figure 5.18: Horizontal $\vartheta$ values for three measurement positions in; (a) empty room, (b) empty room with 15 diffusing panels.

The significant dip in the loudspeaker 1 – Soundfield$^{TM}$ 2 value may be attributed to the placement of the microphone close to the centre of the room resulting in a source-receiver geometry that produces strong modal support at the measurement position. Generally the trimmed (80ms – end) results produce a slightly higher value for $\vartheta$ relative to the full impulse response result but there is little difference in the two cases, in that the average value is 0.85.

This raises the question of the validity of Bradley and Soulodre’s assumption. To explore the time based measure of $\vartheta$, initially explored in the previous chapter, the B-Format outputs were again divided into 30ms time slices. The outputs were then analysed using the same Matlab script for the vertical plane as the previous analyses.
The Speaker 3 – Soundfield™ 3 measurement geometry was chosen for the proximity of the loudspeaker to the microphone and equally the proximity of the microphone to the reflecting floor surface. For the following reasons the vertical case is examined.

The expectation is that the measurement will be influenced by strong specular reflections in the early stages of the decay. This is borne out by the results, to some degree, as can be seen in Figure 5.18. In each room state the value of $\hat{\varepsilon}$ in the first 30ms is low, rising to a value near to the overall broadband $\hat{\varepsilon}$ (80ms trimmed value indicated in chart legend) within 60ms.

![Figure 5.19: 30ms time windowed $\hat{\varepsilon}$ results for Speaker 3 to Soundfield™ 3 measurement condition in each room state.](image)

The growth of the diffuse field may be best illustrated in Figure 5.20 where the signal from the loudspeaker, above the microphone, dominates (blue plot) in the first 30ms. The results show the field approaching the diffuse state through the 30-60ms (green plot) and 60-90ms (red plot), maximising in the 90-120ms period (black plot).
This bears out Bradley and Soulodre’s argument for the trimming of the initial direct sound from the impulse response but suggests that trimming earlier, at the 30ms point may be valid.

### 5.4.3 Octave Band measurements

The Standard measurements explored in Figure 5.17 indicated variation in apparent absorption of the panels over the octave bands. The B-format outputs were filtered in octave bands to examine the frequency dependent diffusivity of the room states. The average $\partial$ values over the three measurement configurations is plotted against frequency in Figure 5.21.
Two notable factors are indicated in the results. Firstly, the peak in measured absorption for the 5 and 10 panel cases, at 2kHz, are reflected in the diffusivity index results. Secondly, the diffusivity of the room reduces significantly in the 8kHz band when more than 15 panels are placed in the room.

Figure 5.21: Average $\vartheta$ values for each room state in octave bands.

Figure 5.22 Standard deviation of measurement positions in each octave band.
**Figure 5.23** Standard deviation of octave band $\vartheta$ values for each room state.
5.5 Conclusion

Comparison has been made between the standard procedure for establishing a diffuse state in a reverberation room, outlined in AS ISO 354, and the measurement proposed by Furduev and T’ung. Each method has been assessed against the two criteria for a diffuse field proposed by Makrinenko. That is, that energy arrives at the measurement point from all directions, over time at the same level and that this isotropic state can be measured in all parts of the enclosure.

The measurement regime outlined by the Standard is a pragmatic approach intended to achieve reliable and consistent results in the measurement of the absorption coefficient of materials and objects. The isotropic nature of the field is implied by the normalisation of the measured absorption coefficient. There is an assumption that the sound field is diffuse because the measured absorption coefficient is consistent regardless of source position. If the source produced a sound field that was different for different placements in the space then the sound energy absorbed by the sample would differ dependent on the source position. We would expect the variation in results to diminish as the diffuse state is approached. Further to that the degree to which the field is homogeneous may be assessed through examination of the results for one source position and multiple receiver positions. If the field is homogeneous, the measured reverberation time and consequent absorption coefficient would approach the same value regardless of receiver position. It is through assessment of these criteria that the deduction is made that the soundfield in the room is deemed to be diffuse.

In each of the cases above we have shown that the room reached a state where the field was adequately diffuse in that the variation in the measured absorption coefficient varied by a small quantity (StdDev 0.04 – 0.06 at 2kHz) when there were 15 or more panels in the room. The overall variation in the measured values, for absorption coefficient, indicate that the sound field in the room is not completely diffuse.

Measurements conducted in the same room without the sample present show that the maximum level of diffusivity is achieved by the addition of 5 panels. This illustrates the effect of adding an absorptive panel into the room in that an increased number of reflective panels must be placed in the room to achieve similar levels of diffusivity.
The above conclusion is supported by the directional microphone method that indicates that the field in the room approaches an adequately diffuse state with $\vartheta$ values of 0.8 to 0.9 in both the room with and without the sample. In the case of the empty room the high diffusivity state is realised with the addition of 5 panels and remains at that level for 10 panels but drops when 15 – 25 panels are placed in the room. This may support Kuttruff’s theory that an ‘excess’ of diffusing elements in a room ‘prevents’ the energy from arriving at the receiver. One might expect that the energy would in fact reach the receiver at a later time due to the greater reflection paths introduced by more reflectors. This is not clearly supported in the temporal diffusivity measurements illustrated in Figure 5.19. There is an early peak in the 10 panel case suggesting that the states with more panels do have some shift in energy to later arrival at the measurement position but this is not supported by a higher $\vartheta$ value in the plots later in time.

The directional microphone method is useful in providing a visual analysis of the field which may assist in the placement of reflecting panels to more effectively achieve an isotropic and homogeneous sound field.

However, the variation in results does not allow direct correlation between the directional microphone method and the standard method of establishing the diffuse field. The method does indicate that the field is not completely diffuse when it reaches a state acceptable for measurement of absorption coefficients, according to Standard ISO 354. It does not indicate higher levels of $\vartheta$ when the room has no absorptive sample but diffusing panels. The highest $\vartheta$ value measured was 0.95 for the room plus 10 panels in the 60 – 90ms time window. Whether this is due an error in the measurement system or extreme difficulty in achieving a truly diffuse state is unclear.

The results again raise the question of what degree of accuracy is required in measuring the diffuse field. The test in Chapter 4 showed that the distinction between differing states of the diffuse field could be measured. This chapter has shown that the directional microphone method is capable of indicating a reasonably high diffuse state but is incapable of providing fine detail measurements at the upper end of the diffusivity index scale. This was suggested by the modelling in Chapter 3.
The final question is whether people are sensitive to small variations in sound field diffusivity, or whether the generalist directional microphone method is adequate for measuring diffuse field states in the music performance auditoria. This will be pursued in the following chapter.
References


Chapter 6

Subjective Preference for a Diffuse Sound Field

At the commencement of this thesis it was recognised that there is currently no widely accepted measure of how diffuse a sound field is. Ensuing chapters have explored the means of making such an objective measurement. This leaves the question of how well the average person perceives a diffuse state and consequently how well the objective measure correlates with subjective preference.

The purpose of the following work is to explore the degree to which a diffuse sound field is desirable as a listening state in auditorium conditions. This requires subjective testing of a number of room ‘states’ where the degree of diffusivity has been measured in order that the listener preference may be represented to a specific numeric value. Such methods have been in use for several decades. Schroeder, Gottlob et al. (1974) and Conant (1978) explored methods of presentation of music material to explore the relationship between subjective response to music and physical characteristics of a hall. Schroeder et al recognised the extreme difficulty in comparing halls due to the reliance on a listener’s memory of a hall and the difficulty in making comparison when several days if not weeks may span the period between listening to performances in different venues. More importantly the variation in performance of an ensemble and how they may respond differently to different halls calls into question the capacity to make meaningful comparison between halls, even when the same piece of music, performed by the same ensemble, is heard. It is desirable then to ‘capture’ the sound of a number of halls so that they may be presented to a listener consecutively. To achieve this firstly, the music presented to the listener must be free of reflections, it must be as close to anechoic as possible, to avoid the sound of the room that the music was recorded in, affecting the listeners judgement of the subsequent hall sound presented to the listener. The series of ‘rooms’ that the material is played through should provide a realistic representation of the hall. Finally, the method by which the material, is played through the hall, to the listener should be natural, without distractions to the subject.
This approach has been used by a great many researchers over the decades. The work of Damaske and Ando (1972) and Barron and Marshall (1981) have been explored in earlier chapters. Kleiner, Dalenback et al. (1993) have provided an extensive overview of the methods used to simulate music performed in concert halls and other spaces.

In order to present material untainted by room reflections, recordings of music made in anechoic conditions are required. Most of the work in this area has been carried out by three groups. In each case the recordings were made to assist in listening experiments. The Archimedes Project (Hansen and Munch 1991) needed material to explore timbral differences in loudspeaker to listening room interactions. Most of the material recorded consisted of solo instruments and solo voices. Another set of recordings were made as part of the Charisma Project to compare room simulations with actual spaces (Rindel and Christensen 2003), in part to confirm the efficacy of the model. In a later paper Rindel, Otondo et al. (2004) suggested that the omni-directional radiation of the sources into the models was resulting in an overstated representation of the high frequencies in the auralisation. It was proposed that a directional model needed to be applied to the sources to avoid the high frequency component of a voice, say, being overemphasised in the rear stage reflections. This has been addressed by the third group. An extensive set of recordings were undertaken that sought to capture the anechoic material for a series of orchestral pieces by recording the instruments one at a time in an anechoic chamber (Lokki, Patynen et al. 2008). To establish the directivity of the instrument a number of recordings were made simultaneously in a measurement ‘mesh’ around the performer to establish the directivity of the instrument (Patynen and Lokki 2010).

The material must then be presented to the listener rendered through the listening space. There is now a wide range of room impulse responses available and the methods for their capture (Stan and Embrechts 2002). Alternatively, the ‘room’ may be simulated either by a simple construction of reflections or through a range of software packages (Bork 2002) that seek to simulate acoustic spaces with varying degrees of sophistication.
The material must then be presented to the listener’s ears either by binaural rendering directly to the ear canals or through loudspeakers of varying complexity. A limiting factor in binaural presentation is the requirement that each subject’s Head Related Transfer Function should be used to successfully recreate the correct spatial imaging (Wenzel, Arrunda et al. 1993). The alternative, loudspeaker presentation in anechoic spaces will be the method explored here.

To this end the 32 channel loudspeaker array in the Faculty of Electrical Engineering at the University of Sydney (CARLab) (Epain, Guillon et al. 2010) is utilised. A series of room states are simulated using anechoically recorded instruments which are assembled within the room simulation to provide as realistic a representation of an orchestra as possible. The reverberation time and program level within each room was held within the just noticeable difference while the scattering coefficients of the room surfaces were modified. Two pieces of music are used, one from the Classical era the other from the Romantic era.

The intention of this work is to test the response of a range of subjects to the different pieces of music rendered through concert hall simulations with varying degrees of surface scattering. Correlation will then be sought between their subjective response and a range of measures used to characterise the diffuse field.

6.1 Room Simulation

The rooms for the simulation were classic ‘shoe-box’ halls constructed within the Matlab software platform using MCRoomSim written by Andrew Wabnitz (Wabnitz, Epain et al. 2010). The MCRoomSim script produces Impulse Responses for source and receiver positions within the designed ‘room’. In Matlab the RunMCRoomSim script may be used in the form; $\text{RIR} = \text{RunMCRoomSim} (\text{Sources}, \text{Receivers}, \text{Room}, \text{Options})$

The inputs to this script are generated by separate scripts; AddReceiver, AddSource, SetupRoom, SetupOptions. Each of these scripts specifies the conditions of each component that makes up the simulation.

Add Source and AddReceiver scripts set characteristics of the virtual transducers in the model such as location, orientation and type. An extensive range of directional characteristics are available.
SetupOptions establishes the general options for the room simulation, such as sample frequency, speed of sound and air absorption including distance attenuation. SetupRoom establishes the characteristics of the room simulation including the dimensions of the space and the absorption and scattering coefficients of the room surfaces. Following that the generated IRs are rendered to the 32 channel array through Nicolas Epain’s scripts; Create MozartHOAsignals and RenderHOASignalsto CarlabArray.

Three room states and two orchestral samples were chosen for the test. The room states are approximations of Angel Place (volume 6,048m³), the Herkulessaal (volume 13,590m³) and Boston Symphony Hall (Volume 18,750m³), basically small, medium and large ‘shoe-box’ concert halls.

The simulation package is a simple model, only capable of producing such rectangular halls with the capacity to alter the absorption and scattering coefficients on the six surfaces of the ‘box’. The particular halls were chosen for their volumetric relationship and the fact that this researcher had attended concerts in each of the auditoria. This ‘aural memory’ was considered an advantage when making initial assessments of the auditory models of the spaces.

Angel Place is an auditorium in Sydney, the acoustic characteristics of which have been measured by this researcher, with room dimensions provided by the venue. The data for the other two halls, including volumes and dimensions, are drawn from Beranek’s text (Beranek 2004). His method provides the volume of the main hall and the orchestra enclosure but the length dimensions of the halls are quoted from the stage front to the rear of the hall. To estimate the total length of the hall the stage area (S_o) is divided by the hall width to find an approximate stage depth, which is then added to the hall length. For the Herkulessaal the overall dimensions arrived at are; length 39.5m, width 22m, height 15.5m producing a volume of 13,469.5m³, a close approximation of the quoted hall volume.

In the case of the Boston Symphony Hall, the volume quoted is 8,750m³. The quoted room dimensions produce a significantly higher value. Reference to Beranek’s first text on Auditoria (Beranek 1962) quotes a volume of 662,000ft³ (18,745m³) indicating a typographical error in the later text. The stage house of the Boston hall is lower than the main hall.
Where we are using a simple ‘shoe-box’ model we can expect the volume to be larger, when the linear room dimensions are used. In this case the stage depth is calculated to be approximately 6.6m but was reduced to 6m, providing overall hall dimensions; length 45m, width 23m, height 18.6m producing an overall volume of 19,250m$^3$. This is considered a reasonable approximation from the complex hall to the simple model.

The absorption coefficients of the hall surfaces in the room simulations are adjusted to as closely as possible emulate the published occupied Rt of the halls. The Rt data was gathered from Beranek for the Herkulessaal and Boston Symphony Hall and measured for Angel Place.

**Table 6.1:** Quoted reverberation time in octave bands for the hall simulations.

<table>
<thead>
<tr>
<th>Octave band Rt (sec)</th>
<th>125 Hz</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1 kHz</th>
<th>2 kHz</th>
<th>4 kHz</th>
</tr>
</thead>
<tbody>
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<td><em>Angel Place</em></td>
<td>1.8</td>
<td>1.8</td>
<td>1.7</td>
<td>1.9</td>
<td>1.9</td>
<td>1.8</td>
</tr>
<tr>
<td><em>Herkulesaal</em> (occupied)</td>
<td>2.14</td>
<td>2.15</td>
<td>2.01</td>
<td>1.94</td>
<td>1.71</td>
<td>1.55</td>
</tr>
<tr>
<td><em>Boston</em> (occupied)</td>
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<td>1.9</td>
<td>1.9</td>
<td>1.95</td>
<td>1.59</td>
<td>1.43</td>
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Absorption coefficient data was adjusted in SetupRoom.m to create a close comparison of the characteristics of the rooms. An absorption coefficient reported in Kuttruff (1973) for occupied audience area was used as the floor absorption coefficient in all the halls.

In order to measure the room characteristics of the hall simulations a sweep signal was used with an omni-directional source emulation, placed at the conductor position, in the room model. A fourth order ambisonic receiver simulation was placed at the listener position to produce the multichannel signal that is then rendered through the CARLab fourth order ambisonic loudspeaker array. This produces a 25 channel audio output (representing 25 spherical harmonics) that may be then played through the 32 channel loudspeaker system. An Earthworks microphone placed at the centre of the array recorded the room simulated sweep through a Digidesign Pre 8 microphone preamplifier to Adobe Audition recording software.
The sweep file is deconvolved using the Aurora plug-in (Farina 2000) within Audition and then exported to Matlab where the reverberation time is calculated using a script written by Nicolas Epain (Appendix A). The absorption coefficients are then adjusted in the model, the sweep rendered through the model and the measurement repeated until the reverberation time measured in the CARLab room simulation is within the difference limen of the desired room by the listener.

### 6.1.1 Room characteristics

Three diffusivity states have been established for the subjective tests; high (scattering coefficient of 0.8), mid (0.5), low (0.2) for each room. The absorption coefficients required adjustment to meet the desired reverberation criteria for each of the diffuse states. The intention of this test is to modify the diffuse field in the models whilst holding the reverberation time as close as possible to the pre-determined values for the existing halls. The aim is to produce natural sounding simulations, allowing the subject to be absorbed in the test, while altering the diffuseness of the sound in the model to test the subjects response to that characteristic.

For example; in SetupRoom;
- X0 is the Stage end wall, X1 the opposite wall
- Y0 and Y1 are the side walls
- Z0 is the floor and Z1 is the ceiling
### Table 6.2: Absorption coefficients applied to Boston Symphony Hall room surfaces.

(a) Boston Symphony Hall - correct Rt for high $\bar{\rho}$

<table>
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<tr>
<th></th>
<th>125 Hz</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1000 Hz</th>
<th>2000 Hz</th>
<th>4000 Hz</th>
<th>Surface</th>
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<td>Abs 11</td>
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<td>0.17</td>
<td>0.09</td>
<td>0.03</td>
<td>0.15</td>
<td>0.09</td>
<td>X0</td>
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<td>0.17</td>
<td>0.09</td>
<td>0.03</td>
<td>0.16</td>
<td>0.09</td>
<td>X1</td>
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<td></td>
<td>0.22</td>
<td>0.18</td>
<td>0.08</td>
<td>0.03</td>
<td>0.16</td>
<td>0.08</td>
<td>Y0</td>
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<td>0.08</td>
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(b) Boston Symphony Hall - correct Rt for mid $\bar{\rho}$

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<td>0.12</td>
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<td>0.09</td>
<td>Y0</td>
</tr>
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<td>0.12</td>
<td>0.06</td>
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<td>0.09</td>
<td>Y1</td>
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<td>0.85</td>
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(c) Boston Symphony Hall - correct Rt for low $\bar{\rho}$

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</table>

The room absorption in each case was adjusted to match, as closely as possible the measured occupied reverberation time of the existing hall. This Rt is plotted in black with $\pm$ 5% values in Figures 6.1 (a-c). The ‘measured’ results derived from an omni-directional source and receiver in the MCRoomSim model is plotted in red.
Figure 6.1(a): Reverberation Time (s) plotted against octave band centre frequency (Hz) for the high diffusion Boston model and as measured in the CARLab simulator.

Figure 6.1(b): Reverberation Time (secs) plotted against frequency for the mid diffusion Boston model and as measured in the CARLab simulator.

Figure 6.1(c): Reverberation Time (secs) plotted against frequency for the low diffusion Boston model and as measured in the CARLab simulator.
Table 6.3: Absorption coefficients applied to Herkulessaal room surfaces.

(a) Herkulessaal correct Rt for high $\tilde{\alpha}$

<table>
<thead>
<tr>
<th>Surface</th>
<th>125 Hz</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1000 Hz</th>
<th>2000 Hz</th>
<th>4000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>X0</td>
<td>0.15</td>
<td>0.10</td>
<td>0.04</td>
<td>0.08</td>
<td>0.02</td>
<td>0.08</td>
</tr>
<tr>
<td>X1</td>
<td>0.15</td>
<td>0.10</td>
<td>0.05</td>
<td>0.08</td>
<td>0.02</td>
<td>0.09</td>
</tr>
<tr>
<td>Y0</td>
<td>0.15</td>
<td>0.10</td>
<td>0.04</td>
<td>0.08</td>
<td>0.02</td>
<td>0.08</td>
</tr>
<tr>
<td>Y1</td>
<td>0.15</td>
<td>0.10</td>
<td>0.04</td>
<td>0.08</td>
<td>0.02</td>
<td>0.08</td>
</tr>
<tr>
<td>Z0</td>
<td>0.60</td>
<td>0.74</td>
<td>0.88</td>
<td>0.96</td>
<td>0.89</td>
<td>0.85</td>
</tr>
<tr>
<td>Z1</td>
<td>0.15</td>
<td>0.09</td>
<td>0.04</td>
<td>0.07</td>
<td>0.01</td>
<td>0.08</td>
</tr>
</tbody>
</table>

Note: the reduction of audience absorption at 2k for the floor absorption coefficient in order to achieve the desired Rt.

(b) Herkulessaal - correct Rt for mid $\tilde{\alpha}$

<table>
<thead>
<tr>
<th>Surface</th>
<th>125 Hz</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1000 Hz</th>
<th>2000 Hz</th>
<th>4000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
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<tr>
<td>Y0</td>
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<td>0.10</td>
<td>0.005</td>
<td>0.08</td>
</tr>
<tr>
<td>Y1</td>
<td>0.14</td>
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<td>0.04</td>
<td>0.10</td>
<td>0.005</td>
<td>0.08</td>
</tr>
<tr>
<td>Z0</td>
<td>0.60</td>
<td>0.74</td>
<td>0.88</td>
<td>0.96</td>
<td>0.93</td>
<td>0.85</td>
</tr>
<tr>
<td>Z1</td>
<td>0.14</td>
<td>0.08</td>
<td>0.04</td>
<td>0.10</td>
<td>0.001</td>
<td>0.08</td>
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(c) Herkulessaal Abs - correct Rt for low $\tilde{\alpha}$

<table>
<thead>
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<th>500 Hz</th>
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<th>2000 Hz</th>
<th>4000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
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<td>0.13</td>
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</tr>
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<td>0.16</td>
<td>0.13</td>
<td>0.26</td>
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<td>0.19</td>
</tr>
<tr>
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<td>0.16</td>
<td>0.13</td>
<td>0.26</td>
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<td>0.19</td>
</tr>
<tr>
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<td>0.88</td>
<td>0.96</td>
<td>0.93</td>
<td>0.85</td>
</tr>
<tr>
<td>Z1</td>
<td>0.15</td>
<td>0.15</td>
<td>0.13</td>
<td>0.25</td>
<td>0.15</td>
<td>0.18</td>
</tr>
</tbody>
</table>
6 Subjective Preference for a Diffuse Sound field

Figure 6.2(a): Reverberation Time (secs) plotted against frequency for the high diffusion Herkulessaal model and as measured in the CARLab simulator.

Figure 6.2(b): Reverberation Time (secs) plotted against frequency for the mid diffusion Herkulessaal model and as measured in the CARLab simulator.

Figure 6.2(c): Reverberation Time (secs) plotted against frequency for the low diffusion Herkulessaal model and as measured in the CARLab simulator.
Table 6.4: Absorption coefficients applied to Angel Place room surfaces.

(a) Angel Place - correct Rt for high $\delta$

<table>
<thead>
<tr>
<th>Abs 31</th>
<th>125 Hz</th>
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<th>1000 Hz</th>
<th>2000 Hz</th>
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<tr>
<td></td>
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<td>0.01</td>
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<td>0.05</td>
<td>0.005</td>
<td>Y1</td>
</tr>
<tr>
<td></td>
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<td>0.74</td>
<td>0.88</td>
<td>0.96</td>
<td>0.93</td>
<td>0.85</td>
<td>Z0</td>
</tr>
<tr>
<td></td>
<td>0.17</td>
<td>0.15</td>
<td>0.12</td>
<td>0.07</td>
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<td>0.005</td>
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(b) Angel Place - correct Rt for mid $\delta$

<table>
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<th>2000 Hz</th>
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<th>Surface</th>
</tr>
</thead>
<tbody>
<tr>
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<td>0.06</td>
<td>0.05</td>
<td>0.01</td>
<td>X0</td>
</tr>
<tr>
<td></td>
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<td>0.14</td>
<td>0.15</td>
<td>0.06</td>
<td>0.05</td>
<td>0.01</td>
<td>X1</td>
</tr>
<tr>
<td></td>
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<td>0.06</td>
<td>0.02</td>
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</tr>
<tr>
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<td>0.14</td>
<td>0.15</td>
<td>0.05</td>
<td>0.06</td>
<td>0.02</td>
<td>Y1</td>
</tr>
<tr>
<td></td>
<td>0.60</td>
<td>0.74</td>
<td>0.88</td>
<td>0.96</td>
<td>0.93</td>
<td>0.85</td>
<td>Z0</td>
</tr>
<tr>
<td></td>
<td>0.18</td>
<td>0.14</td>
<td>0.15</td>
<td>0.07</td>
<td>0.07</td>
<td>0.02</td>
<td>Z1</td>
</tr>
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</table>

(c) Angel Place - correct Rt for low $\delta$

<table>
<thead>
<tr>
<th>Abs 33</th>
<th>125 Hz</th>
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</thead>
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<tr>
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<td>0.15</td>
<td>0.12</td>
<td>0.12</td>
<td>0.10</td>
<td>0.08</td>
<td>0.001</td>
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</tr>
<tr>
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<td>0.13</td>
<td>0.12</td>
<td>0.10</td>
<td>0.08</td>
<td>0.01</td>
<td>X1</td>
</tr>
<tr>
<td></td>
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<td>0.12</td>
<td>0.13</td>
<td>0.11</td>
<td>0.09</td>
<td>0.01</td>
<td>Y0</td>
</tr>
<tr>
<td></td>
<td>0.15</td>
<td>0.12</td>
<td>0.13</td>
<td>0.11</td>
<td>0.09</td>
<td>0.01</td>
<td>Y1</td>
</tr>
<tr>
<td></td>
<td>0.60</td>
<td>0.74</td>
<td>0.88</td>
<td>0.96</td>
<td>0.93</td>
<td>0.85</td>
<td>Z0</td>
</tr>
<tr>
<td></td>
<td>0.15</td>
<td>0.12</td>
<td>0.12</td>
<td>0.11</td>
<td>0.09</td>
<td>0.001</td>
<td>Z1</td>
</tr>
</tbody>
</table>
Figure 6.3(a): Reverberation Time (secs) plotted against frequency for the high diffusion Angel Place model and as measured in the CARLab simulator.

Figure 6.3(b): Reverberation Time (secs) plotted against frequency for the mid diffusion Angel Place model and as measured in the CARLab simulator.

Figure 6.3(c): Reverberation Time (secs) plotted against frequency for the low diffusion Angel Place model and as measured in the CARLab simulator.
6 Subjective Preference for a Diffuse Sound field

6.1.2 Directional measures

In addition to the reverberation time characteristics measured with the omnidirectional microphone the periphonic microphone was used to derive a series of direction related measures. The obvious measurement is the $\delta$ measure but additionally the W and Y, B-Format outputs may be utilised to perform a Lateral Fraction measurement. Further to those measurements a Brüel & Kjaer 4128C Head and Torso simulator was used to perform a series of measurements. In each case the measurement device was placed at the listener position. This allows a comparison between Furduev’s $\tilde{\delta}$, Meyer’s $m$, Bassett/Cabrera’s StdDev, Barron’s LF and Ando’s IACC measures of the diffuse field.

Table 6.5: Horizontal and vertical directional measures of directional energy arrival measured in the 32 channel loudspeaker room simulator.

<table>
<thead>
<tr>
<th></th>
<th>$\tilde{\delta}$H</th>
<th>$\tilde{\delta}$V</th>
<th>H Meyer</th>
<th>V Meyer</th>
<th>H BC</th>
<th>V BC</th>
</tr>
</thead>
<tbody>
<tr>
<td>Angel HiD</td>
<td>0.76</td>
<td>0.76</td>
<td>0.92</td>
<td>0.92</td>
<td>0.92</td>
<td>0.92</td>
</tr>
<tr>
<td>Angel MiD</td>
<td>0.58</td>
<td>0.58</td>
<td>0.86</td>
<td>0.86</td>
<td>0.86</td>
<td>0.86</td>
</tr>
<tr>
<td>Angel LoD</td>
<td>0.57</td>
<td>0.57</td>
<td>0.86</td>
<td>0.86</td>
<td>0.85</td>
<td>0.85</td>
</tr>
<tr>
<td>Hurk HiD</td>
<td>0.70</td>
<td>0.70</td>
<td>0.87</td>
<td>0.87</td>
<td>0.87</td>
<td>0.87</td>
</tr>
<tr>
<td>Hurk MiD</td>
<td>0.43</td>
<td>0.43</td>
<td>0.78</td>
<td>0.78</td>
<td>0.76</td>
<td>0.76</td>
</tr>
<tr>
<td>Hurk LoD</td>
<td>0.41</td>
<td>0.41</td>
<td>0.78</td>
<td>0.78</td>
<td>0.76</td>
<td>0.76</td>
</tr>
<tr>
<td>Boston HiD</td>
<td>0.61</td>
<td>0.61</td>
<td>0.84</td>
<td>0.84</td>
<td>0.84</td>
<td>0.84</td>
</tr>
<tr>
<td>Boston MiD</td>
<td>0.39</td>
<td>0.39</td>
<td>0.76</td>
<td>0.76</td>
<td>0.75</td>
<td>0.75</td>
</tr>
<tr>
<td>Boston LoD</td>
<td>0.39</td>
<td>0.39</td>
<td>0.77</td>
<td>0.77</td>
<td>0.75</td>
<td>0.75</td>
</tr>
</tbody>
</table>

Based on the previous work the above measurements were made on impulse responses with the initial 80ms trimmed from the measurements. The following measures are presented from the full signal.
Figure 6.4: Lateral Fraction measured within the Boston Symphony Hall simulation plotted against frequency.

Table 6.6: Comparison of measured LF values for the actual room and the simulation of Boston Symphony Hall (data from actual hall drawn from (Beranek 1996)).

<table>
<thead>
<tr>
<th>Source</th>
<th>125 Hz</th>
<th>250 Hz</th>
<th>500 Hz</th>
<th>1 kHz</th>
<th>2 kHz</th>
<th>4 kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation</td>
<td>0.043</td>
<td>0.052</td>
<td>0.074</td>
<td>0.13</td>
<td>0.073</td>
<td>0.356</td>
</tr>
<tr>
<td>High diff.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Bradley</td>
<td>0.23</td>
<td>0.19</td>
<td>0.17</td>
<td>0.22</td>
<td>0.25</td>
<td>0.18</td>
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<tr>
<td>Gade</td>
<td>0.15</td>
<td>0.25</td>
<td>0.28</td>
<td>0.25</td>
<td>0.24</td>
<td>0.25</td>
</tr>
</tbody>
</table>

The results from the simulator are lower than those measured in the hall itself. The hall measurements were in the unoccupied hall whereas the model simulates the absorption of an audience. Nonetheless, it is unlikely the presence of an audience would cause such a significant variation in LF.
Figure 6.5: Lateral Fraction measured within the Herkullessaal simulation plotted against frequency.

Figure 6.6: Lateral Fraction measured within the Angel Place simulation plotted against frequency.
Figure 6.7: IACC measured within the Boston Symphony Hall simulation plotted against frequency.

Table 6.7: Comparison of measured IACC values for the actual room and the simulation of Boston Symphony Hall (data from actual hall drawn from (Beranek 1996))

<table>
<thead>
<tr>
<th>Source</th>
<th>125Hz</th>
<th>250Hz</th>
<th>500Hz</th>
<th>1kHz</th>
<th>2kHz</th>
<th>4kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation</td>
<td>0.98</td>
<td>0.83</td>
<td>0.34</td>
<td>0.29</td>
<td>0.52</td>
<td>0.32</td>
</tr>
<tr>
<td>High diff.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Bradley</td>
<td>0.94</td>
<td>0.76</td>
<td>0.27</td>
<td>0.24</td>
<td>0.21</td>
<td>0.22</td>
</tr>
<tr>
<td>Takenaka</td>
<td>0.91</td>
<td>0.73</td>
<td>0.29</td>
<td>0.14</td>
<td>0.11</td>
<td>0.11</td>
</tr>
</tbody>
</table>

Unlike the Lateral Fraction measurements the IACC values in the simulation are similar to those measured in the actual hall. The degree of diffusivity in the real Boston hall is unknown but the measurements from the hall fall within the range and trend of the IACC values measured over the three hall states for that particular hall.
Mason, Kim et al. (2009) have shown that the IACC value changes with head orientation. This was checked in the room simulation by taking additional measurements at plus and minus 10-degree azimuth positions.
Figure 6.10: Measured IACC for different azimuth orientations of the head for the Boston Symphony Hall simulation plotted against frequency.

The IACC measure produces a close correlation with the reported values for at least the Boston Hall. The variation in the results over what is assumed to be a reasonable degree of head movement within the listening context brings into question the efficacy of the measure for assessment of the diffuse field.

Finally, the possible diffusivity index measures discussed in this work are plotted against each other. Due to the similarity between horizontal and vertical results only the horizontal results for the directional measures will be presented here.
Figure 6.11: Various measures plotted indicating variation (vertical spacing) between high and low surface diffusivity in the room models.

The group on the right hand side (7-15) are all derived from the same measurement method so we would expect them to exhibit similar trends. It is clear that for the same scattering coefficient there is an increase in measured diffusivity in the models that is inversely related to hall volume. The main interest here is in the values output by each method. Although there is little difference in measured results between the mid and low diffuse states the $\partial$ measure produces results that span the expected range of the room states. Meyer’s $d$ and Bassett/Cabrera’s statistical measure may be normalised to produce similar results. The Lateral Fraction measure (1-3) also tracks the increase in diffusivity for the smaller volumes implied by the $\partial$ group of measurements but the broadband results exhibit little variation between room states for each of the simulations. The IACC results (4-6), like the LF measurement, exhibit little variation between the room states but it is interesting to note that, for the Angel Place simulation, shown to be the more diffuse of the three room states by the other measures, the results indicate a narrowing of the result span. This result may imply that a reduction of the variation in IACC results is an indication of the presence of a diffuse state.

This assumption is borne out when the rotated head results are considered in Figure 6.12. Here the IACC is measured after the first 80ms of the signal is removed.
The low diffuse states for each simulation consistently produce a high IACC(late) value. Furthermore while the low diffuse states result in significant variation as the head simulator is rotated there is a reduced variation in the results for the higher diffuseness simulations. This indicates that an IACC(late) measurement that incorporates some rotation of the head simulator may provide a useful measure of the soundfield isotropy.

Figure 6.12: IACC(late) results for each room state at each rotation of the dummy head.

In establishing the room states for the subjective test it is clear that the low and middle diffusivity states are very similar in character. The test could have proceeded using only two states for each room simulation but the variation in results produced by the IACC and to a lesser degree by the LF measures indicate there may be a difference in the quality of these simulations. This was supported by pilot listening tests where the subjects reported distinguishable differences in the quality of the sound in the simulations. The Low and Mid diffusivity states were therefore used in the subjective test to explore the listeners capacity to distinguish between small variations in the sound field.
6.2 Music samples

The music samples used are the Donna Elvira aria from Mozart’s Don Giovanni and Mahler’s First Symphony, Fourth movement (Lokki and Päätynen 2009). The intention of using these two samples is to attempt to reveal any tendency in the listener to show preference for a particular music form and a possible association with a room sound. For example; we may expect a classical period piece to sound best in a drier more direct sounding hall that has a less diffuse state while a romantic era piece may be preferred in a more reverberant and diffuse sounding hall.

The orchestra has been simulated within the space by placement of the individual sound files in a traditional arc layout relative to the conductor. Instruments are placed at a height that approximates that of a performance position. The original recordings were made based on the score.

“Where there was, for example, one first violin line on the score one violin was recorded. The number (usually 6) in the track name means the number of microphone with which the sound is picked up. Most instruments are recorded in frontal direction (mic 6, azi = 0, ele = 11), but for some instruments the main radiation direction is also included (mic 8, azi = 144, ele = 11, for French horn) and (mic 5, azi = 288, ele = 53, for bassoon and tuba).


Some of the files from TKK are quite noisey. To reduce the noise level each file was normalised to -0.3dB and then processed using the iZotope(2010) noise cancelling software. Due to the proprietary nature of the software, it is unclear what process is used to achieve the noise reduction. Thiemann (2001) provides a good overview of methods for noise reduction.

In each case a sample was taken of a section of ‘silence’ with the noise present. The software then produced an inverse filter of the noise that was applied to the overall file. The software provides a function that allows the user to listen to the material removed from the file. In each case the audio file was checked by this method to ensure that no harmonic material was removed along with the noise. This entailed adjusting the threshold at which the filter operated, in some cases resulting in the retention of some noise but the retention of all the musical components of the sound file.

This process alters the overall level of the signal, destroying the level relationship between instruments established by the score and replicated in the recording process.
To re-establish the correct level the RMS level of the processed files were found. These were compared with the original files and a gain correction made.

When that file was ‘placed’ in several positions within the simulator, to replicate an orchestral section, the listener was most likely to hear the instrument sound from one location. Similar results have been reported by (Vigeant, Wang et al. 2007).

Of course ten violins, for example, playing the same thing don’t all sound exactly the same. A method of altering the sonic character of each instrument was proposed by Lokki (2007) where each file is phase shifted slightly. The method chosen for this work was to convolve each sound file with different samples of 20ms duration white noise. The noise was faded to zero over the last 10ms. This randomisation of the phase between each of the source files may produce a similar effect to that of a diffuse field. In each case the files were played through the simulator so any diffusing effect caused by the randomisation of the files would be presented through each of the further diffusing fields. As such the difference between the diffuse room states is maintained due to the inputs remaining the same.

The first file of each group is original ie not convolved with noise. The slight alteration of each file by this method produced a more realistic ensemble sound in the model.

6.2.1 Ensemble configurations

Mozart

The score, accessed from the International Music Score Library Project (2008), indicates there is a line each for first, and second violins and violas. There are two lines for Clarinet, two lines for Bassoon and one line for two French Horns. The number of each instrument is not specified.

The anechoic files provided by TKK(Pätyinen, Pulkki et al. 2008) consist of; first violins, second violins, violas, cellos, basses, flutes, clarinets, bassoons and two French Horns and a soprano.

To construct a realistic representation of an ensemble performing the piece a traditional chamber ensemble, consisting of ten first violins, ten second violins, six viola, six celli and four basses were created by decorrelating the single audio track recorded for the individual instrument at TKK. In addition an extra flute, clarinet and bassoon were added by the same process. (The TKK group note that the Mozart piece only calls for
high strings so it is unclear why low strings, celli and basses, were recorded for the piece. It was decided to include them in the simulation).

The virtual instruments were placed within the modelled rooms by first establishing a conductor position 1 metre from the edge of the stage. For each of the rooms this equates to a distance from the stage wall; Angel Place – 7m, Herkulessaal – 17m, Boston – 12.5m. The ensemble was then arranged in a traditional ‘American orchestra’ configuration; first violins, second violins, violas and celli in an arc from left to right. In most cases the recording from the microphone placed in front of the instrument in the anechoic room recording was used. The exceptions to this were the French Horns, Bassoons and Tuba where the microphone aligned to the dominant radiation axis was used. For the French Horns the microphone 8 channel, recorded at the azimuth angle of 144 degrees, was used. For the Bassoon and Tuba microphone channel 5 with an azimuth angle of 288 degrees and an elevation angle of 53 degrees was used.

Figure 6.13: Mozart ensemble arranged in the large hall. Note the Soprano placed in front of the conductor, the French horns (sources 11 and 15) facing sideways and the listener position, designated by the blue arrow.

The ensemble was placed at heights equivalent to their position on the raised stage in each of the virtual auditoria. Again the stage heights of the actual venues are used in the model; Angel Place and Herkulessaal – 1m, Boston – 1.37m.
Figure 6.14: Listener placed at 1.2m above the floor in the Boston simulator. Instruments are placed on the 1.37m high ‘stage’ at varying heights, ranging from standing soprano to sitting cellists.

Mahler

The score, again accessed from the International Music Score Library Project calls for 2 Piccolo, 2 Flute, 3 Oboe plus another Oboe playing a different line, 3 Clarinets in C, 1 Clarinet in E♭, 3 Bassoon, 4 Horns plus another 3 playing a different line, 2 Trumpets plus another 2 playing a different line, 2 trombones plus another playing a different line, a Tuba, 2 timpani each playing a different line, 1 cymbal set, 1 bass drum plus the standard string grouping. The size of the string sections can be variable. For this work the size has been chosen to fit between the early and later Romantic periods; 16 1st, 14 2nd, 10 viola, 8 celli and 6 bass. As noted previously, one track was recorded for each of the lines of the score. This meant that the 1st violin, 2nd violin, viola, cello and contrabass channels had to be convolved with white noise a number of times dependent on the number within the ensemble. TKK provide two tracks for the first violins. The first track, after convolution with various noise files, is mapped to the first two rows of violins (a,b,e,f,i,j,m,n). The second file maps to (c,d,g,h,k,l,o,p). A similar arrangement is used for the second violins.
It is unclear from the Lokki and Päätynen (2009) paper as to why multiple tracks were recorded for a single score line, in the case of the Mahler recordings. There are three trombone tracks, 4 trumpet tracks, 4 Oboe tracks, 4 Clarinet tracks, 3 bassoon tracks a Tuba and two Timpani tracks which is how many there is on the score. None of these files need to be decorrelated, it is assumed that there is an adequate difference in each performance of the piece to produce the required variation in each file. There are two percussion tracks in the fileset, Perc1 is the cymbal and perc2 is the bass drum, as called for on the score. There are 2 Piccolo and 2 flutes called for on the score. The TKK track set has Flute 3 which is the higher pitch track and Flute 1 the lower pitch. Each of the tracks was convolved once with noise to produce the second instrument track for each group. There are 7 ‘corno’ tracks which is how many French Horns are called for in the score.

Figure 6.15: Ensemble layout for the Mahler orchestra.

Due to the size of this ensemble the instruments have been arranged on risers on the stage, emulating the kind of stage set-up expected for such an ensemble. The first row of woodwinds are an additional 0.2m high, the second row of woodwinds are 0.4m and the percussion are 0.6m higher. The size of the ensemble could be accommodated on the Herkulessaal and Boston stages but in the case of Angel Place the stage extension was used, shifting the conductor to 9.5metres from the rear wall.
6.3 CARLab level measurements

In order to reduce the chance of listeners showing preference to the music samples based on level differences rather than diffusivity states the samples were level matched to within just noticeable difference of the SPL. It is also desirable to present the material at a level that would be experienced in an actual performance. Obviously the level of the performance will vary depending on the conductor’s direction but the score provides an indication of expected level from the ensemble.

Reference to the score shows, for the Mahler at bar150 the ensemble should play from $p$ to $fff$. This clearly sets the upper level to around 100dB, (Campbell and Greated 1987). At bar 166 the notation indicates a dynamic level of $ppp$ with much of the ensemble silent, establishing a sound level in the vicinity of 30dB. For the Mozart the aria has a dynamic range from $p$ to $f$ approximating to a decibel range of 50dB – 80dB.

Firstly the background noise level was measured in the CARLab room by making a 15 minute statistical measurement, using a calibrated B&K 2250 sound level meter. During the measurement short duration indistinguishable noises were the only audible sounds. The LAeq, the measured effective continuous level, was 19.5 dB (although, considering that the self-noise of the sound level meter is 16.6 dBA, the acoustic component of the
noise may have been about the same, calculated as 16.4 dBA). This means that, if the
correct levels are established in the room, the quietest passages will be 10dB above the
noise floor, an acceptable level difference.

The samples were then played with the sound level meter measurement running for the
duration of the piece. Three levels were examined;

I. \( LAf1.0 \), a fast response, A weighted statistical measurement that indicates the
level exceeded 1% of the time

II. \( LAf99 \), the level exceeded 99% of the time

III. \( LAeq \), the equivalent continuous level.

As the sample pieces were of different duration the measurement procedure involved the
stopping of the measurement at the end of the piece. This meant that the period of silence
measured at the end of the piece varied between measurements. As such the L99 results
should be treated as indicative only. The sample levels are checked to confirm that they
are within just noticeable differences.
Table 6.8: Measured levels for the Mozart sample in nine room states.

<table>
<thead>
<tr>
<th>Mozart Angel Pl.</th>
<th>High scattering</th>
<th>Mid scattering</th>
<th>Low scattering</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>76.1</td>
<td>76.4</td>
<td>76.3</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>83.6</td>
<td>84</td>
<td>83.9</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>26.8</td>
<td>30.6</td>
<td>24.9</td>
</tr>
</tbody>
</table>

To achieve the levels for the Low replay a gain of 1dB was required.

<table>
<thead>
<tr>
<th>Mozart Herculessaal</th>
<th>High scattering</th>
<th>Mid scattering</th>
<th>Low scattering</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>76.6</td>
<td>76.0</td>
<td>75.7</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>83.6</td>
<td>82.6</td>
<td>82.9</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>27.3</td>
<td>27.6</td>
<td>29.4</td>
</tr>
</tbody>
</table>

No adjustment of level was required.

<table>
<thead>
<tr>
<th>Mozart Boston</th>
<th>High scattering</th>
<th>Mid scattering</th>
<th>Low scattering</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>76.7</td>
<td>76.0</td>
<td>76.7</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>84.9</td>
<td>83.8</td>
<td>84.4</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>23.1</td>
<td>27.0</td>
<td>27.8</td>
</tr>
</tbody>
</table>

To achieve the levels for the Mid and Low replay a gain of 2dB was required.

It is interesting to note that additional gain is required for some room models where the surface diffusivity is altered. It may be speculated that the lower scattering coefficient results in a reduction of energy in the room model. The reduced scattering is likely to result in the sound in the room being absorbed earlier by the significant floor absorption. There is some evidence of this, for example, in the ‘Herculessaal’ model where an additional 3dB and 6dB respectively was added to the Mid and Low scattering states to match that of the High state.

An overall gain increase of 24dB was made to the Mahler samples to bring the levels in line with expected values. Further level adjustments for individual room states are noted below.
Table 6.9: Measured levels for the Mahler sample in nine room states.

<table>
<thead>
<tr>
<th>Mahler Angel Pl.</th>
<th>High $\hat{\delta}$</th>
<th>Mid $\hat{\delta}$</th>
<th>Low $\hat{\delta}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>88.0</td>
<td>88.2</td>
<td>88.1</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>94.4</td>
<td>94.5</td>
<td>94.6</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>30.0</td>
<td>26.6</td>
<td>26.8</td>
</tr>
</tbody>
</table>

To achieve the levels for the Mid and Low $\hat{\delta}$ replay an additional gain of 1dB and 2.5dB respectively was required.

<table>
<thead>
<tr>
<th>Mahler Herkulessaal</th>
<th>High $\hat{\delta}$</th>
<th>Mid $\hat{\delta}$</th>
<th>Low $\hat{\delta}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>88.3</td>
<td>87.9</td>
<td>88.6</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>94.2</td>
<td>94.0</td>
<td>94.4</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>26.7</td>
<td>26.3</td>
<td>30.2</td>
</tr>
</tbody>
</table>

To achieve the levels for the Mid and Low $\hat{\delta}$ replay an additional gain of 3dB and 6dB respectively was required.

<table>
<thead>
<tr>
<th>Mahler Boston</th>
<th>High $\hat{\delta}$</th>
<th>Mid $\hat{\delta}$</th>
<th>Low $\hat{\delta}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>LA eq (dB)</td>
<td>88.5</td>
<td>88.2</td>
<td>88.0</td>
</tr>
<tr>
<td>LA 1.0 (dB)</td>
<td>94.7</td>
<td>95.0</td>
<td>94.3</td>
</tr>
<tr>
<td>LA 99 (dB)</td>
<td>29.2</td>
<td>30.8</td>
<td>31.3</td>
</tr>
</tbody>
</table>

To achieve the levels for the Mid and Low $\hat{\delta}$ replay an additional gain of 1dB and 4.5dB respectively was required.
6.4 Subjective Preference

The subjective test involved 26 subjects who were invited to participate in the test through contact via email or personal invitation. The subjects were played the two music samples and given a choice of whether to commence the test with the Mahler or Mozart set of samples. The listener was then positioned in a central sitting position within the loudspeaker array and given a screen based interface on which to carry out the test. The test consisted of the subject stepping through 30 paired-comparison tests. In each trial the subject could switch between two simulations of the same room with different diffusivity states. The three pairs were; High-Low, High-Mid and Mid-Low. For each set the order of paired presentation was randomised as was the order within the pair. The first three pairs were not included in the analysis to ensure that unfamiliarity with the interface did not affect their decisions. The resultant 27 paired comparisons comprised all possible states being presented to the listener 3 times. This allowed a consistency check to be carried out on the subjects response to each pair.

On completion of the tests the subject is given a short questionnaire that seeks to ascertain their experience in critical listening and how realistic they considered the simulations. (Refer to Appendix E)

Based on the responses to the enquiry into the subject’s experience with acoustic music and their involvement with music listening it was found that 7 identified as performers, 2 as associated with performance, 15 as audio professionals, 7 as audio students and 6 as general music lovers. There were many cases where the subjects identified in more than one category. A clearer distinction could be made between the listeners by examining their experience with the musical forms presented in the test where they could be grouped in thirds under the general categories of Frequent, Occasional and Infrequent for performance attendance.
Table 6.10: Experience of listeners—the number of listeners identifying with each category.

<table>
<thead>
<tr>
<th>Performance attendance</th>
<th>Weekly</th>
<th>Monthly Few times a year</th>
<th>Less than once a year</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>8</td>
<td>2</td>
<td>8</td>
</tr>
<tr>
<td>Involvement</td>
<td>Performer Associated with music performance</td>
<td>Audio Professional</td>
<td>Audio Student</td>
</tr>
<tr>
<td></td>
<td>7</td>
<td>2</td>
<td>15</td>
</tr>
</tbody>
</table>

No one identified as a music lover only, the category intended for ‘the average listener’. On that basis the category was removed from the final findings resulting in approximately 50% audio professionals and 25% each, musicians and audio students.

Table 6.11: Listener’s impression of the simulated listening environment.

<table>
<thead>
<tr>
<th>Percentage choice</th>
<th>Very realistic/ Very Easy</th>
<th>Realistic/ Easy</th>
<th>Unrealistic/ Difficult</th>
<th>Very unrealistic Very difficult</th>
</tr>
</thead>
<tbody>
<tr>
<td>Room simulations</td>
<td>27</td>
<td>57</td>
<td>17</td>
<td>0</td>
</tr>
<tr>
<td>Orchestra simulations</td>
<td>37</td>
<td>43</td>
<td>20</td>
<td>0</td>
</tr>
<tr>
<td>Overall impression</td>
<td>21</td>
<td>58</td>
<td>21</td>
<td>0</td>
</tr>
</tbody>
</table>

A common response from the subjects was to rate the room simulations as both realistic and unrealistic as a means of indicating that some rooms within the test sounded realistic while others did not. The final question asked how difficult it was for the subject to imagine they were in the space presented. A table was constructed with a 1 assigned to a subjects response to each question. The percentage of responses to each option are shown in Table 6.11.
There is a clear indication (82%) that the subjects found the simulations realistic to very realistic. Where ‘Unrealistic’ was chosen it was coupled with ‘Realistic’ indicating that only some of the states were considered realistic sounding. The case was similar for the listeners ability to imagine they were in the space (81% easy or very easy), although four of the five who chose ‘Difficult’ chose it exclusively.

The subjects were also asked whether there was anything in the test that distracted them. Eight reported no distractions, six reported clicks some of which was intentional clicking as the tracks switched and some appeared to be caused by processor overload by quickly switching between tracks. Three reported distortion mostly in the Mahler sample and three found the visual environment distracting. Other distractions included concerns about ensemble balance, timbral difference between samples and difficulty using the interface. In each case the respondent reported that the distractions were minor and did not significantly affect the listeners capacity to perform the test.

### 6.4.1 Results

The test interface output a text file for each sample set. These results were transferred to a spreadsheet where each diffusivity pair was arranged so that a 1 was input where a subject preferred that state and a 0 where they did not. The results for all subjects were averaged and these averages were input to a P-matrix indicating the proportion of times each state was chosen.
Table 6.12: Probability matrix from subject responses to all sample pairs.

<table>
<thead>
<tr>
<th>P-Matrix</th>
<th>Overall</th>
<th>Hall 1</th>
<th>Hall 2</th>
<th>Hall 3</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>High</td>
<td>High</td>
<td>High</td>
</tr>
<tr>
<td>Overall</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>High</td>
<td>0.41</td>
<td>0.52</td>
<td>0.46</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.39</td>
<td>0.50</td>
<td>0.49</td>
<td>0.54</td>
</tr>
<tr>
<td>Low</td>
<td>0.53</td>
<td>0.52</td>
<td></td>
<td>0.45</td>
</tr>
<tr>
<td></td>
<td>0.47</td>
<td>0.48</td>
<td>0.55</td>
<td></td>
</tr>
</tbody>
</table>

In addition to the overall results the cases for the Mozart and Mahler samples were separated.
Table 6.13: Probability matrix from subject responses to Mozart sample only.

<table>
<thead>
<tr>
<th>Hall 1</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.32</td>
<td>0.38</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.68</td>
<td>0.55</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.62</td>
<td>0.45</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 2</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.58</td>
<td>0.56</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.42</td>
<td>0.54</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.44</td>
<td>0.46</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 3</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.49</td>
<td>0.58</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.51</td>
<td>0.45</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.42</td>
<td>0.55</td>
<td></td>
</tr>
</tbody>
</table>

Table 6.14: Probability matrix from subject responses to Mahler sample only.

<table>
<thead>
<tr>
<th>Hall 1</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.50</td>
<td>0.40</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.50</td>
<td>0.50</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.60</td>
<td>0.50</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 2</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.46</td>
<td>0.44</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.54</td>
<td>0.50</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.56</td>
<td>0.50</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 3</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.49</td>
<td>0.58</td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.51</td>
<td>0.45</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.42</td>
<td>0.55</td>
<td></td>
</tr>
</tbody>
</table>
Where a value of 0.5 appears, such as in the Mid – Low comparison for the Mahler in Hall 2, the results indicate that there may have been preference shown by individual subjects but overall the group of subjects showed an even distribution of preference between the two states. To clarify the data a X-matrix is produced by taking the inverse of the normal distribution function.

**Table 6.15: X-matrixes for Overall, Mozart and Mahler results.**

<table>
<thead>
<tr>
<th>X-Matrix</th>
<th>Overall</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Hall 1</td>
<td>Hall 2</td>
</tr>
<tr>
<td>High</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Av</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Overall</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
<th>Check</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hall 1</td>
<td>-0.23</td>
<td>-0.28</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hall 2</td>
<td>0.23</td>
<td>0.06</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hall 3</td>
<td>0.28</td>
<td>-0.06</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Av</td>
<td>0.25</td>
<td>-0.15</td>
<td>-0.11</td>
<td>0.00</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 1</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>0.05</td>
<td>0.00</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>-0.05</td>
<td>0.05</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.00</td>
<td>-0.05</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Av</td>
<td>-0.02</td>
<td>0.00</td>
<td>0.02</td>
<td>0.00</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hall 2</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>-0.10</td>
<td>-0.03</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.10</td>
<td>-0.13</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.03</td>
<td>0.13</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Av</td>
<td>0.06</td>
<td>0.02</td>
<td>-0.08</td>
<td>0.00</td>
</tr>
</tbody>
</table>
6 Subjective Preference for a Diffuse Sound field

The chance value 0.5 is placed as the mean in the distribution with values greater showing positive, values lower are negative. The resultant average in each column is the variation of a standard distribution for that diffusivity state. The higher the positive number the greater the number of room states chosen by the subjects. The higher the negative number the greater negative preference shown by the subjects. A value above 0.25 indicates a reasonable preference is shown by the subjects, for a particular room state, as it indicates that more than 20% of the subjects indicated preference for that state.

### 6.4.2 Result analysis

Reference to the overall results shows that the only results that show even faint significance is the subjects preference for the High diffusivity state in Hall 1. As indicated previously, there is a question of whether the different musical forms may influence a listener’s preference. Separating the two pieces produces a slightly stronger preference for the High diffusion state in Hall 1 for the Mozart and a small indication of preference, arguably insignificant, for the High state in Halls 1 and 2 for the Mahler. This result is surprising because subjects were reporting significant differences in preference.
for the room sound in at least some of the sample pairs. The difference in the Lo and Mid states is borne out, to a degree, in the results in that there is difference between the two states in most room simulations. The results may be showing negative preference but they are not showing the same preference for each state. To explore the results further the responses will be sorted into categories.

6.4.3 ‘Concert-going’ Listeners

At the commencement of the test a decision was made to draw subjects from as wide a range of the populace as possible. The reasoning was that a range of people attend concerts whereas subjective testing often focuses on a subset of the community. This approach is reflected in ITU-T Recommendation P.800 (ITU 1997) where the recommendation is that subjects be drawn randomly from the telephone using public for assessment of listening quality tests. Although this test did not draw randomly from the concert going public a division may be made in the listeners based on the regularity of their attendance at acoustic music concerts. This subset was drawn from the subjects who identified their attendance of concerts at least monthly. This group contained 10 subjects. The same data analysis was performed on this group as for the larger group.

Table 6.16: X-matrixes for Overall, Mozart and Mahler results from concert goers.

<table>
<thead>
<tr>
<th>Overall</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Hall 1</td>
<td>High</td>
<td>Mid</td>
</tr>
<tr>
<td>High</td>
<td></td>
<td>-0.34</td>
</tr>
<tr>
<td>Mid</td>
<td>0.34</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.30</td>
<td>-0.13</td>
</tr>
<tr>
<td></td>
<td>0.32</td>
<td>-0.23</td>
</tr>
<tr>
<td>Hall 2</td>
<td>High</td>
<td>Mid</td>
</tr>
<tr>
<td>High</td>
<td></td>
<td>-0.13</td>
</tr>
<tr>
<td>Mid</td>
<td>0.13</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.30</td>
<td>-0.04</td>
</tr>
<tr>
<td></td>
<td>0.21</td>
<td>-0.08</td>
</tr>
<tr>
<td>Hall 3</td>
<td>High</td>
<td>Mid</td>
</tr>
<tr>
<td>High</td>
<td></td>
<td>-0.17</td>
</tr>
<tr>
<td>Mid</td>
<td>0.17</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>0.04</td>
<td>0.08</td>
</tr>
<tr>
<td></td>
<td>0.10</td>
<td>-0.04</td>
</tr>
</tbody>
</table>
These results slightly reinforce the previous indication that the experienced listeners are showing a preference for the High diffusivity state for Hall 1 over the Medium and Low states, particularly for the Mahler sample. There is also some overall preference being shown for the High state in Hall 2, again mainly for the Mahler sample. There is a slight, bordering on insignificant preference shown for the Mid level state for Hall 3. Overall the results do not clearly indicate a preference for different room states.

### Table 6.4.4: ‘Consistent’ Listeners

<table>
<thead>
<tr>
<th></th>
<th>Hall 1</th>
<th>Hall 1</th>
<th>Hall 2</th>
<th>Hall 2</th>
<th>Hall 3</th>
<th>Hall 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td></td>
<td></td>
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<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Examination of the results show that even the listeners experienced in attending music performances similar to that simulated are not making consistent preference decisions. As Shlien and Soulodre (1996) indicated expert listeners may focus on different aspects of the material presented, potentially producing inconsistent results in a small sample size such as the one presented here. Another approach to the data is the examination of subjects who make consistent decisions for the material presented. Each pair was presented to the subjects three times. Where a subject chose a state two of three times it is not clear whether the subject is indicating that they generally prefer that state or are randomly choosing a sample. It is only where the subject makes consistent choices that
we may consider them to be reliable subjects. To this end the results were examined and all subjects who chose the same sample in the pair for all three presentations of that pair for at least six of the nine pair sets, were selected. Of the 26 subjects 12 met this criterion.
Table 6.17: X-matrixes for Overall, Mozart and Mahler results from consistent listeners.

<table>
<thead>
<tr>
<th>X-Matrix</th>
<th>Overall</th>
<th>Hall 1</th>
<th>Hall 2</th>
<th>Hall 3</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>High</td>
<td>Mid</td>
<td>Low</td>
<td>High</td>
</tr>
<tr>
<td>Hall 1</td>
<td>-0.56</td>
<td>-0.35</td>
<td>0.23</td>
<td>0.35</td>
</tr>
<tr>
<td>High</td>
<td>0.56</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mid</td>
<td>0.35</td>
<td>0.19</td>
<td>0.04</td>
<td>0.04</td>
</tr>
<tr>
<td>Low</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Av</td>
<td>0.45</td>
<td>-0.39</td>
<td>-0.06</td>
<td>0.17</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Mozart</th>
<th>Hall 1</th>
<th>Hall 2</th>
<th>Hall 3</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>High</td>
<td>High</td>
<td>High</td>
</tr>
<tr>
<td></td>
<td>Mid</td>
<td>Mid</td>
<td>Mid</td>
</tr>
<tr>
<td></td>
<td>Low</td>
<td>Low</td>
<td>Low</td>
</tr>
<tr>
<td></td>
<td>-0.80</td>
<td>-0.52</td>
<td>-0.31</td>
</tr>
<tr>
<td></td>
<td>0.35</td>
<td>-0.35</td>
<td>0.31</td>
</tr>
<tr>
<td></td>
<td>0.52</td>
<td>-0.35</td>
<td>0.04</td>
</tr>
<tr>
<td></td>
<td></td>
<td>-0.57</td>
<td>0.16</td>
</tr>
<tr>
<td></td>
<td></td>
<td>-0.08</td>
<td>0.19</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
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<tr>
<td></td>
<td>-0.80</td>
<td>-0.04</td>
<td>-0.44</td>
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<td>0.35</td>
<td>-0.04</td>
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<td>0.04</td>
<td>-0.04</td>
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<td>-0.04</td>
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<tr>
<td>Mahler</td>
<td>Hall 1</td>
<td>Hall 2</td>
<td>Hall 3</td>
</tr>
<tr>
<td></td>
<td>High</td>
<td>High</td>
<td>High</td>
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<tr>
<td></td>
<td>Mid</td>
<td>Mid</td>
<td>Mid</td>
</tr>
<tr>
<td></td>
<td>Low</td>
<td>Low</td>
<td>Low</td>
</tr>
<tr>
<td></td>
<td>-0.35</td>
<td>-0.04</td>
<td>-0.31</td>
</tr>
<tr>
<td></td>
<td>0.35</td>
<td>-0.35</td>
<td>0.31</td>
</tr>
<tr>
<td></td>
<td>0.52</td>
<td>-0.35</td>
<td>0.04</td>
</tr>
<tr>
<td></td>
<td></td>
<td>-0.57</td>
<td>0.16</td>
</tr>
<tr>
<td></td>
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<td>0.19</td>
</tr>
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</tbody>
</table>
The results are similar to the previous two explorations, the preference for High diffusivity is stronger overall for Hall 1 with some small, probably insignificant, preference shown for Mid level diffusivity in Hall 2 for the Mozart.

### 6.4.5 ‘Consistent’ Results

It may be concluded from the results above that even when the listeners are making fairly consistent choices the overall results are producing output that is close to evenly distributed. This raises the question of whether the subjects are producing results that are internally contradictory. To check this only the results where the subject chose the same sample in all three instances was selected for analysis. Of the 468 data sets from the cohort 137 were consistently selected. A detailed statistical analysis is not required to illustrate the point. Reference to the Overall Sum for Hall 3 shows that the subjects were almost evenly showing preference for the High and Low states. Although there are other cases where there is a strong preference shown, such as the strong preference shown between High and Mid for Hall 1, the Hall 3 example shows us that the subjects are making a consistent choice for one state but within the cohort the preference from individual to individual is almost equally divided.
Table 6.18: X-matrixes for Overall, Mozart and Mahler results where samples were consistently chosen. Sample pairs read from top to bottom.

<table>
<thead>
<tr>
<th>Subjects</th>
<th>Overall</th>
<th>Mozart</th>
<th>Mahler</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Sums</td>
<td>Sums</td>
<td>Sums</td>
</tr>
<tr>
<td>Roomstates</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hall 1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hi</td>
<td>4</td>
<td>1</td>
<td>3</td>
</tr>
<tr>
<td>Mid</td>
<td>11</td>
<td>7</td>
<td>4</td>
</tr>
<tr>
<td>Hi</td>
<td>3</td>
<td>3</td>
<td>0</td>
</tr>
<tr>
<td>Lo</td>
<td>16</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>Mid</td>
<td>10</td>
<td>6</td>
<td>4</td>
</tr>
<tr>
<td>Lo</td>
<td>7</td>
<td>3</td>
<td>4</td>
</tr>
<tr>
<td>Hall 2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hi</td>
<td>7</td>
<td>5</td>
<td>2</td>
</tr>
<tr>
<td>Mid</td>
<td>5</td>
<td>1</td>
<td>4</td>
</tr>
<tr>
<td>Hi</td>
<td>8</td>
<td>6</td>
<td>2</td>
</tr>
<tr>
<td>Lo</td>
<td>10</td>
<td>4</td>
<td>6</td>
</tr>
<tr>
<td>Mid</td>
<td>9</td>
<td>5</td>
<td>4</td>
</tr>
<tr>
<td>Lo</td>
<td>7</td>
<td>4</td>
<td>3</td>
</tr>
<tr>
<td>Hall 3</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hi</td>
<td>4</td>
<td>1</td>
<td>3</td>
</tr>
<tr>
<td>Mid</td>
<td>5</td>
<td>3</td>
<td>2</td>
</tr>
<tr>
<td>Hi</td>
<td>10</td>
<td>3</td>
<td>7</td>
</tr>
<tr>
<td>Lo</td>
<td>11</td>
<td>6</td>
<td>5</td>
</tr>
<tr>
<td>Mid</td>
<td>2</td>
<td>2</td>
<td>0</td>
</tr>
<tr>
<td>Lo</td>
<td>8</td>
<td>5</td>
<td>3</td>
</tr>
</tbody>
</table>
6.4.6 Subjective tests review

It is clear from the results that the listeners do not show a strong overall preference for higher states of diffusivity for the auditorium conditions tested. There is some indication that different states of diffusivity are preferred depending on the hall simulation. The Hall 1, the ‘Angel Place’ simulation does indicate some preference shown by listeners for High diffusivity. This result may be viewed in the context of the measured results that indicate that this hall simulation may have the highest diffusivity state of the three ‘halls’. In the case of Hall 2, the ‘Herkulesaal’ model there is some small preference shown by the ‘expert’ listeners for the Mid level diffuse state for the Mozart.

The results certainly call into question any assumption that a high state of diffusivity is desired in music performance auditoria. The results suggest that different halls may sound best with different degrees of diffuseness. This raises the question of whether the form of the music affects the listener’s expectation of desirable levels of diffusivity. Examination of the results indicate that the musical form may have an influence on listener preference. For example, there is a stronger preference shown for the Mahler sample in the high diffusion Hall 1 than for the Mozart in the same ‘hall’. The Mahler sample was chosen for its ensemble size and sound level with the hypothesis that romantic era music would be more likely to be preferred by listeners in an enveloping auditory experience the results may support the hypothesis although the results are hardly convincing.

What is reasonably clear from the results is that people are making individual preference judgements when presented with the samples in the hall simulations.
6.5 Conclusion

A set of three simulated halls were created in the MCRoomSim package in order to test subjective response of varying diffuse states. The reverberation time and program level were held constant for each of the rooms, allowing the subjects to respond via a simple preference test to the material presented. The decision to hold the reverberation time and sound level as constant as possible was based on the understanding that most subjects would detect difference in reverberation time or overall level. As preference judgements were sought in the subjective test it was deemed undesirable for change to occur in the primary cues to a room’s character. In addition to that it was also desirable to maintain a character to the reverberation that was similar to that experienced in real halls, in order to maintain ‘believability’ in the presented soundfield. To achieve that the absorption coefficient of the ‘floor’ surface was maintained constant for all rooms. This resulted, in some cases, in an unrealistic absorption coefficient for some surfaces, to maintain the desired reverberation time. This was considered a reasonable compromise in order to achieve the overall listening conditions in the model where the primary auditory cue the subjects were responding to was the diffuseness of the soundfield.

A range of measurements were taken in each of the room simulations with the aim of finding any correlation between the subjects response and the measured criteria.

The subjective responses were less than consistent but this in itself provides an indication that the current view, that a highly diffuse sound field is desirable, is overly simplistic. As the results have shown there were many cases where subjects were unable to show a clear preference for different room states. This may have been expected where the difference between mid and low diffusivity was presented but the results show the lack of clarity to be more widespread. Within the results, however are indications that some listeners are showing quite clear preferences for some room states and, in the example given, those strong preferences may be divided fairly evenly between two choices. The separation of the results is made more difficult by the difference in measured diffusion. It may be argued that listeners do show preference for a high diffuse state because they preferred in Hall 1, the simulation that measured the higher \( \delta \) values. Reference to Table 5.2 shows that the lowest ‘Angel Place’ measurement was 0.57 whereas the highest ‘Boston’ measurement was 0.61.
This of course raises the question of whether the $\partial$ measure is an effective measure of the diffuse field. It may be argued that the trend in results between the measurement of the fields created high and low surface scattering in the model is supported by a similar trend in the Lateral Fraction results. There is an inverse relationship exhibited also in the mid–band IACC results. Reference to Tables 5.6 – 5.8 show a variation beyond the difference limen for the three diffusivity states. (The IACC Just Noticeable Difference is held to be 0.075 (International Organisation of Standardisation 2009). The significant variation between IACC values for the different states further confirms the variation in the different room models is measureable.

The initial question was whether the $\partial$ measure is a viable measure of a diffuse field. Where it does provide an arguably suitable range of results between high and low diffusivity states, produced in the simulation, the lack of distinction between the mid and low diffuse states causes its efficacy to be questioned.

From the results we find that IACC provides a better method of distinguishing between these states. The subjective results do not provide any significant indication of whether that level of distinction is required in measuring the diffuse state. Overall the broadband $\partial$ measure is useful in measuring the higher levels of a diffuse field but where a room presents a lower degree of diffuseness the $\partial$ measure should be used in conjunction with other measurement criteria.
References


Chapter 7

Summary and Future Work

This thesis set out to explore methods of assessing diffuse sound fields. The aim was to find a method of objectively measuring the sound in a room and finding some correlation between a physical measure of the sound field and the general public’s desire for such listening states. The measurement originally proposed by Furduev and T’ung showed some promise due to the relative simplicity of the method. An analysis system was developed that allowed measurement of the field in both planar and spherical formats, although the bulk of the work in the thesis concentrated on the two dimensional measurements, due in part to the presentation format.

7.1 Appraisal of the measurement system

The measurement system allows assessment of the field in both frequency and temporal components. Based on the measurements conducted in the reverberation rooms it is clear that both aspects bear consideration in assessing the diffuse state. Further development of the analysis software to incorporate these factors into the measurement system, whilst maintaining the relative simplicity of the system appears to be justified.

The diffusivity index has proven to be a fairly general measurement. Fine detail variations in the sound field are not detectable by this method. The work in the two reverberation room configurations have shown that the measure is capable of providing a numerical value for macro variation in the state of the field. This formed part of the hypothesis, that only large scale variation, such as low mid and high, were needed to be quantised. The method was not able to make a distinction between the variations in the sound field, as it approached a completely diffuse state.

The thesis opened with the view, put by Beranek and others, that a high state of diffusion is desirable in concert halls. The subjective tests have introduced a contrary view. The listeners showed that in many cases they were unable to distinguish or decide which state they preferred. There was an indication, however, that where they showed strong preference the results were quite subjective. This should not be surprising.
7.2 Further Work

What the results are telling us is that an individual may show a preference for music that is often quite different to the next person. The suggestion that everyone should like a highly diffuse state is like suggesting that everyone wants to listen to orchestras, say, playing in halls with a reverberation time of 2.2 seconds. Clearly this is an oversimplification. Further work should be carried out in this area by simply changing the question put to the subjects. The test in this thesis sought to question the assumption about the diffuse state. It would be interesting to ask which of the states they feel more immersed in, in order to distinguish between the state experienced and the state preferred.

The measurements and tests conducted in this work were carried out in highly idealised rooms, either a reverberant room or a simple auditorium model. The obvious next step would be to conduct measurements in a range of actual auditoria. Analysis of this greater range of rooms would be useful in providing a larger data set required for a detailed analysis of the measure’s efficacy, over a wide range of room states.

The accuracy of the Ambisonic microphone was explored in 3.6. There was an indication that higher order ambisonic arrays would produce more accurate results. In Chapter 2 other microphone arrays, including the 3-D intensity probe showed promise as measurement devices. The work in this thesis may be extended by the use of such measurement arrays.

Overall the work raises more questions than provides answers – as it should.
List of References


1 Introduction


Appendix A;  Matlab code used in sections of this thesis.

soundfield_DIplot forms the core of the Soundfield™ microphone analysis. The code has explanation sections throughout.

```matlab
%soundfield_DIplot  (last update 10/3/06)
% This file is written in order to analyse data supplied
% by the B_Format Soundfield microphone leading to the
% production of a plot of directional information
% according to the Furduiev method.
% The code in this file is based on rmsplotloop and
% soundfield_direction
% written by Craig Jin and Andre van Schaik.
% This file has been compiled by John Bassett
% Start by placing the output wav files into a dedicated
directory. The files should be of the format W_'roompos'.
% This maintains some meaning in the filenames as the
% process creates further files. Replace the XXXX in
% roompos below with the correct label.
% roompos='XXXX';

theta=zeros(1,72);
%theta sets angle of rotation for each measurement in the
%horizontal plane.
phi=[0:5:355];
%phi sets angle of rotation for each measurement in the
-vertical plane.
% Use X=zeros(1,72); to remove the horizontal or
-vertical plane from the calculation
% Use X=[0:5:355]; to set the angle of rotation in the
-plane under test.
% In this case, every 5 degrees.

df=1;

% Directivity factor, df, can be set as:
% omnidirectional: df = 0;
% subcardiod: df = 0.5;
% cardiod: df = 1.0;
% hypercardiod: df = 1.5;
% figure-of-eight: df = 2.0

data_w=wavread(['W_','roompos','.wav']);
data_x=wavread(['X_','roompos','.wav']);
data_y=wavread(['Y_','roompos','.wav']);
data_z=wavread(['Z_','roompos','.wav']);
%This reads in the data from the B-Format audio files.
%The four vectors are then concatenated into the data
%matrix
```
data=[data_w data_x data_y data_z];
clear data_w data_x data_y data_z;

%soundfield_direction is a separate file that sets the
%directional weighting for the data

rms_vals=soundfield_directionDI(data,theta,phi,df,roompos);
    clear data;
    norm_rms_vals=rms_vals/rms_vals(1);

%The output from soundfield_direction is normalized.
%In the line above it is normalised to the first, 0deg
%value. Alternately, use max(rms_vals)

polar([((pi*phi/180) 2*pi],[norm_rms_vals
norm_rms_vals(1)]),
    title(roompos),xlabel({'DI = ',A});

    %NOTE for a horizontal plot use theta in the above
    %line, for a vertical plot use phi

norm_rms_vals_shift = [norm_rms_vals(2:end)
norm_rms_vals(1)];

A=0.5*sin(5/180*pi)*sum(norm_rms_vals.*norm_rms_vals_shift);
    eval(['A_',roompos,'=A'])

% This section calculates the Directivity Index based on
%the surface area plot of the cardioid rotation measured
%in the National Acoustics Laboratory in Sydney.
%NOTE: be sure to use the correct value for the NAL
%reading horizontal or vertical?

%sd=1.1203   %area Horizontal A_NAL reading
sd=1.1297   %area Vertical A_NAL reading
S1=pi-sd

v = evalin ('base',['A_','roompos','=A'])

s=pi-v
d=(S1-s)/S1
disp (d)

%This outputs the calculated ∂ value of the measurement

This script imports the variable indicated below and a weighting is applied to the signal
% function signal_out =
% soundfield_direction(data,theta,phi,df,filename)
%
% Directivity factor, df, can be set as:
% omnidirectional: df = 0;
% subcardiod: df = 0.5;
% cardiod: df = 1.0;
% hypercardiod: df = 1.5;
% figure-of-eight: df = 2.0

function rms_val =
soundfield_direction(data,theta,phi,df,roompos)

num_directions = length(theta);

for ii=1:num_directions
% The weighting for the Cartesian coordinates is set for
% each direction of rotation by the following line
% [weight_x,weight_y,weight_z] = sph2cart(-
% theta(ii)*pi/180,phi(ii)*pi/180,1);

% The following sets the values to zero where they are
% small to reduce unnecessary processing
    if (abs(weight_x)) < 1e-10
        weight_x = 0.0;
    end
    if (abs(weight_y)) < 1e-10
        weight_y = 0.0;
    end
    if (abs(weight_z)) < 1e-10
        weight_z = 0.0;
    end

    x(ii) = weight_x;
    y(ii) = weight_y;
    z(ii) = weight_z;

    signal_out = 0.5*((2-df)*data(:,1) +
    df*(weight_x*data(:,2) + weight_y*data(:,3) +
    weight_z*data(:,4)));

% signal_out sums the data with the direction weighting

    if false
        savefile = [roompos,'_direction_'.num2str(ii)];
        cmd = ['save ',savefile,' signal_out'];
        eval(cmd)
    end

    rms_val(ii) = sqrt(mean(signal_out.^2));
end
Appendix A Matlab Codes used in sections of this thesis

%soundfield_DI3D  (last update 10/8/11)
%This file is written in order to analyse data supplied by the
B_Format Soundfield microphone leading to the production of a
plot of directional information according to the Furduev
method.
%This code is designed to analyse the surface area of a
icosahedron as a measure of the 3-D capture allowed by the
Soundfield mic.
%The code in this file is based on rmsplotloop and
soundfield_direction
%written by Craig Jin and Andre van Schaik.
%This file has been compiled and modified by John Bassett
%Start by placing the output wav files into a dedicated
directory. The
%files should be of the format W_'roompos'. This maintains
some meaning in
%the filenames as the process creates further files. Replace
the XXXX in
%roompos below with the correct label.
%roompos='XXXX';
%theta=[[18;36;54;72;90;108;0;36;-
18;0;13.6138224408033;36;58.3861775591967;81.732301447702;108
;144;144;36;9.732301447702;];
%theta sets angle of rotation for each measurement in the
horizontal plane.

phi=[[92.7065186640873;97.4774023159273;92.7065186640871;78.86
08750411093;57.2957795130823;30.1221738831454;0;30.12217388314
55;57.2957795130822;78.8608750411090;121.710871392264;127.5995
76199073;121.710871392264;104.73754887812;78.8608750411093;48
.7387011579639;30.1221738831455;48.7387011579639;78.8608750411
094;104.73754887812;];
%phi sets angle of rotation for each measurement in the
vertical plane.

df=1;

% Directivity factor, df, can be set as:
% omnidirectional: df = 0;
% subcardiod: df = 0.5;
% cardioid: df = 1.0;
% hypercardioid: df = 1.5;
% figure-of-eight: df = 2.0

%data_w=wavread(['W_',roompos,'.wav']);
data_x=wavread(['X_',roompos,'.wav']);
data_y=wavread(['Y_',roompos,'.wav']);
data_z=wavread(['Z_',roompos,'.wav']);
data=[data_w data_x data_y data_z];
clear data_w data_x data_y data_z;
rms_vals = soundfield_direction_sph(data, theta, phi, df, roompos);
% clear data;
norm_rms_vals = rms_vals / max(rms_vals);

fi = (1 + power(5, .5)) / 2;
s = (2 * (power(3, .5))) / (power(fi, 2)) * norm_rms_vals
AIcosTri = ((power(3, .5) / 4) * (power(s, 2)))
Ameas = sum(AIcosTri)

% This section calculates the Directivity Index based on the
% surface area plot of the cardioid rotation measured in the
% Auditory Neuroscience Laboratory’s anechoic room in Sydney
% University.

Sanec = 7.172
S1 = 15.1622 - Sanec

s = 15.1622 - Ameas
d = (S1 - s) / S1
disp (d)
Reverberation Time calculation code, used in CARLab measurement of room simulation Rts.

```matlab
function [RvbTme,frq] = ReverberationTime(ImpRsp,smpFrq,bndOpt,pltRvbTme,pltSchCur)
%Written by Nicolas Epain

% [RvbTme,frq] = ReverberationTime(ImpRsp,bndOpt,pltOpt) ;
% Calculate the RT60 reverberation time.
% Input: - ImpRsp is an impulse respo [ne ([Nx1]*vector) or array of impulse
% - smpFrq is the sampling frequency.
% - This parameter can be omitted, the default value is 48kHz.
% Output: - if ImpRsp is a vector, RvbTme is the vector of the RT60s for different frequency bands. If ImpRsp is a matrix, RvbTme is the matrix of the RT60s for each impulse response.
% - frq is the vector of the frequency band center frequencies.
% Options: - bndOpt can be set to 'oct' (default) to obtain the RT60 for 9 octave bands with center frequency from 62.5 to 16kHz, or '3rdoct' in which case the RT60s are calculated for 25 third-octave bands.
% - if pltRvbTme is set to true (default), the RT60s are plotted.
% - if pltSchCur is set to true, the Schroeder curves are plotted.
% Note: Requires MATLAB's Signal Processing Toolbox
% N.Epain, 2011

% Don't plot the Schroeder curve by default
if nargin < 5
    pltSchCur = false ;
end

% Plot the reverb time by default
if nargin < 4
    pltRvbTme = false ;
end
```
% Default "frequency band option": 'oct'
% (octave bands with center frequencies from 62.5 to 16 kHz)
if nargin < 3
    bndOpt = 'oct';
end
bndOpt = lower(bndOpt);

% Default sampling frequency: 48kHz
if nargin <= 2 || isempty(smpFrq)
    smpFrq = 48e3;
end

% Min and max frequencies at which the order 6 and 8 Butterworth filters
% can be used
switch bndOpt
    case 'oct'
        minFrq = smpFrq/200;
        maxFrq = smpFrq/8;
    case '3rdoct'
        minFrq = smpFrq/80;
        maxFrq = smpFrq/8;
end

% Vector of the center frequencies
switch bndOpt
    case 'oct'
        frq = 1000 * 2.^(4:4);
    case '3rdoct'
        frq = 1000 * 2.^(4:1/3:4);
end
nmbFrq = length(frq);

% Initialise the output
nmbImp = size(ImpRsp,2);
RvbTme = zeros(nmbFrq,nmbImp);

% Loop on the center frequencies
for J = 1 : nmbImp

    % Create a new figure for the Schroeder curves
    if pltSchCur == true
        figure
    end

    % Loop on the impulse responses
    for I = 1 : nmbFrq
Appendix A Matlab Codes used in sections of this thesis

% Band-pass filter the impulse response.
% Check if the current center frequency is in the
fs/200 -> fs/5
% interval. If not, resample the impulse first.
if frq(I) < minFrq
    % The frequency is too low, resample the signal
    rat = 2^nextpow2(minFrq/frq(I));
    % Resampled impulse response
    imp = resample(ImpRsp(:,J),1,rat);
    % Band-pass filter
    switch bndOpt
        case 'oct'
            [num,den] = ... 
            butter(3,[2^(-1/2)]
    end
2^(1/2)*frq(I)/smpFrq*2*rat);
    case '3rdoct'
    [num,den] = ...
        butter(4,2^(-1/6)]
    end
2^(1/6)*frq(I)/smpFrq*2*rat);
    else
        % Band-pass filter
        switch bndOpt
            case 'oct'
                [num,den] = ...
                butter(3,[2^(-1/2)]
        end
2^(1/2)*frq(I)/smpFrq*2*rat);
            case '3rdoct'
            [num,den] = ...
                butter(4,2^(-1/6)]
        end
2^(1/6)*frq(I)/smpFrq*2*rat);
    end
% Band-pass filter the impulse response
imp = filter(num,den,ImpRsp(:,J)) ;

end
% Schroeder curve
schCur = 10*log10(flipud(cumsum(flipud(imp.^2))));
schCur = schCur - max(schCur);

% Estimate the slope of the Schroeder curve between -5 and -35dB
fst = find(schCur<=-05,1,'first');
lst = find(schCur<=-35,1,'first');
if frq(I) < minFrq
    coe = polyfit((fst:lst)'/smpFrq/srat,schCur(fst:lst),1);
elseif frq(I) > maxFrq
    coe = polyfit((fst:lst)'/smpFrq*rat,schCur(fst:lst),1);
else
    coe = polyfit((fst:lst)'/smpFrq,schCur(fst:lst),1);
end

% Plot the Schroeder curve and the estimated slope
if pltSchCur == true
    switch bndOpt
    case 'oct'
        subplot(3,3,I)
    case '3rdoct'
        subplot(5,5,I)
    end
    if frq(I) < minFrq
        plot((1:length(imp))/(smpFrq/rat),schCur,'k','linewidth',2)
        hold on
        plot((1:length(imp))/(smpFrq/rat),..., 
             coe(2)+coe(1)*(1:length(imp))/(smpFrq/rat),... 
             ':r','linewidth',2)
    elseif frq(I) > maxFrq
        plot((1:length(imp))/(smpFrq*rat),schCur,'k','linewidth',2)
        hold on
        plot((1:length(imp))/(smpFrq*rat),..., 
             coe(2)+coe(1)*(1:length(imp))/(smpFrq*rat),... 
             ':r','linewidth',2)
    else
        plot((1:length(imp))/(smpFrq),schCur,'k','linewidth',2)
        hold on
        plot((1:length(imp))/smpFrq,... 
             coe(2)+coe(1)*(1:length(imp))/smpFrq,... 
             ':r','linewidth',2)
    end
    title(['f = ' num2str(round(frq(I))) ' Hz'])
xlabel('Time [s]')
ylabel('Energy [dB]')
axis([0 1.05*size(ImpRsp,1)/smpFrq -95 5])
end
% RT60
RvbTme(I,J) = -60/coe(1); 

end

% Plot the RT60
if pltRvbTme == true
    figure
    semilogx(frq,RvbTme(:,J),'-ok','linewidth',2)
    xlim([sqrt(2)*frq(end)])
    title('Reverberation time');
    xlabel('Frequency [Hz]')
    ylabel('RT60 [s]')
end
Appendix B: Reverb Room Plots

Code: R# denotes room state, P# denotes position in room, Solid line denotes Horizontal plot, Dashed line denotes Vertical plot

Room Condition 1:
Room Condition 2:
Room Condition 3:
Room Condition 4:
Room Condition 5:
Room Condition 6:
Room Condition 1, 30mS intervals
Room Condition 3, 30mS intervals

Directional Plot R4P3

---

Room Condition 3, 30mS intervals

Directional Plot R4P3

---

Room Condition 3, 30mS intervals

Directional Plot R4P3

---
Room Condition 6, 30mS intervals
Appendix C – Room state Rt plots

Reverberation Times - Empty Room

Reverberation Time Empty Room + 25 panels
Appendix C Room State Rt Plots

Reverberation Time - Room + Sample

Reverberation Time Sample + 5 panels
Appendix D – Room state box plots and charts

0 = Empty room, 1 = Sample in room, 5-25 = Sample +x panels in room.

Horizontal δ 125Hz

Vertical δ 125Hz

Horizontal δ 250Hz

Vertical δ 250Hz

Horizontal δ 500Hz

Vertical δ 500Hz
Appendix D Room state $\vartheta$ box plots and charts

- Horizontal $\vartheta$ 1kHz
- Vertical $\vartheta$ 1kHz
- Horizontal $\vartheta$ 2kHz
- Vertical $\vartheta$ 2kHz
- Horizontal $\vartheta$ 4kHz
- Vertical $\vartheta$ 4kHz
Appendix D Room state box plots and charts

Horizontal $\delta$ 8kHz

Vertical $\delta$ 8kHz
ETHICS AND PRIVACY APPLICATION
FORM FOR RESEARCH INVOLVING
HUMANS

INSTRUCTIONS FOR ALL SUBMISSIONS

Original Application signed [all signatures required before submitting] X Y

10 copies of the signed Original Application plus a soft PDF copy X Y

Please (X) to indicate either “Y” or “N” that the following documents are attached to the Original and Copies:

Participant Information Statement (s) X Y N

Consent Form (s) X Y N

Copy of questionnaire(s), survey questions, interview topics to be covered etc. X Y N

Research references Y X N

Recruitment advertisement / circular Y X N

Evidence of permission to conduct research in other locations Y X N

One copy of the grant application with appropriate clearance forms as requested by the Research Office Y X N
Please Note: Each question on this form has instructions and links to relevant documents and guidelines on how to answer that particular question as hidden text. To show the text with the hidden text effect, click symbol “¶” (Show/Hide) (situated next to the “Zoom” button) on the “Standard” toolbar. When hidden text is shown it is marked with a dotted underline. This text will not be seen on the printed version.

Please note the following:

1. This application must be completed electronically or typewritten
2. Complete all sections except those specifically not applicable
3. Use lay terms wherever possible
4. Do not alter the order of questions or layout of the application form
5. “Y” signifies Yes, “N” signifies No, and “N/A” signifies Not applicable
6. Some “Y”/“N” boxes have been reversed so take care in answering the questions
7. HREC refers to Human Research Ethics Committee

SECTIONS:

Section 1: Administration 3
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Section 3: Participants and Recruitment 7
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Section 6: Risks and Benefits 15
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Section 8: Conflict of Interest and Other Ethical Issues 17
Section 9: Description of Project 18
Section 10: Field Based Research or Research Conducted Outside Australia 20
Section 11: Declaration of Researchers 22
   Checklist 23

This form has been prepared in collaboration between Ms G Briody, Associate Professor M Grimm, Professor A Lloyd, Associate Professor J Watson and Ms M Wright of the Human Research Ethics Committees (HRECs) of the Universities of New South Wales and Sydney.
SECTION 1: ADMINISTRATION

This section is obligatory

1.1 (a) Full project title

Assessment of listener preference for a diffuse sound field for music listening spaces.

(b) Short name by which the project will be known

Listener's preference for an immersive sound when listening to music

(c) Name of Chief Investigator

Craig Jin

(d) Provide a brief summary of the project in lay language (approximately 100 words)

The study aims to test listeners for their subjective preference to being immersed in sound when listening to music. The subjects will be presented with recordings of orchestral music in a listening environment that simulates that of a series of performance venues. One criterion, the diffuseness of the sound, will be altered in each of the listening samples. The subjects will be asked to indicate their preference for one of two presented samples.

(e) Outline the academic/scientific merits of this study (including potential contributions to the body of knowledge and methodological rigor) (approximately 100 words)

Currently there is little correlation between listeners’ subjective preference for their sense of immersion in sound and an objective means of measuring the immersive sound field. This study will attempt to link the participants subjective response to an objective measure taken within the sound field.

1.2 Indicate the institutional ethics committee that you consider to be the primary one for this project. (In general, if the Chief Investigator is a University employee, then the University should be considered to be the primary site. If the Chief Investigator or participants are from a health care service, then the Area Health Service ethics committee should be considered as the primary site.)

University of Sydney

1.3 (a) Has this project already been submitted to any other HREC(s)?

N Y

(b) Will this project be submitted to any other HREC(s)?

N Y

If you answered YES to (a) or (b), give the name of the HREC(s), and indicate the status of the application at each (i.e., submitted, approved, deferred or rejected). Attach copies of the correspondence with each of the other HREC(s). Please do not submit to more than one HREC concurrently.
1.4 List the following details of the Chief Investigator/Supervisor, any Co-Researcher(s), Associate Researcher(s) and Student(s).

### Chief Investigator/Supervisor

<table>
<thead>
<tr>
<th>Name</th>
<th>Craig Jin</th>
</tr>
</thead>
<tbody>
<tr>
<td>Title</td>
<td>Senior Lecturer ARC QEII Fellow</td>
</tr>
<tr>
<td>Qualifications</td>
<td>J03 – Electrical Engineering, The University of Sydney NSW, 2006, Australia</td>
</tr>
<tr>
<td>Full mailing address (including building number)</td>
<td></td>
</tr>
<tr>
<td>Telephone</td>
<td>9351 7208</td>
</tr>
<tr>
<td>Fax</td>
<td>9036 5449</td>
</tr>
<tr>
<td>E-mail</td>
<td><a href="mailto:craig.jin@sydney.edu.au">craig.jin@sydney.edu.au</a></td>
</tr>
</tbody>
</table>

### Co-Researcher(s), Associate Researcher(s), Student(s) or other Personnel involved in the study

(If appropriate indicate for each named person whether they are University staff, student or neither). If the named person is a student, nominate (in the Qualifications section) the degree for which he/she is enrolled.

<table>
<thead>
<tr>
<th>Name</th>
<th>John Bassett</th>
</tr>
</thead>
<tbody>
<tr>
<td>Title</td>
<td>PhD Student</td>
</tr>
<tr>
<td>Qualifications</td>
<td>M Des Sci</td>
</tr>
<tr>
<td>Full mailing address (including building number)</td>
<td></td>
</tr>
<tr>
<td>Telephone</td>
<td>9351 2769</td>
</tr>
<tr>
<td>Fax</td>
<td>9036 5449</td>
</tr>
<tr>
<td>E-mail</td>
<td><a href="mailto:jbassett@usyd.edu.au">jbassett@usyd.edu.au</a></td>
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<table>
<thead>
<tr>
<th>Name</th>
<th>Roman Kosobrodov</th>
</tr>
</thead>
<tbody>
<tr>
<td>Title</td>
<td>Research Assistant</td>
</tr>
<tr>
<td>Qualifications</td>
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<table>
<thead>
<tr>
<th>Name</th>
<th>Densil Cabrera</th>
</tr>
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<tbody>
<tr>
<td>Title</td>
<td>Senior Lecturer</td>
</tr>
<tr>
<td>Qualifications</td>
<td>G04 – Wilkinson, The University of Sydney NSW, 2006, Australia</td>
</tr>
<tr>
<td>Full mailing address (including building number)</td>
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</tr>
<tr>
<td>Telephone</td>
<td>9351 5267</td>
</tr>
<tr>
<td>Fax</td>
<td>9351 3031</td>
</tr>
<tr>
<td>E-mail</td>
<td><a href="mailto:densil.cabrera@sydney.edu.au">densil.cabrera@sydney.edu.au</a></td>
</tr>
</tbody>
</table>

Insert additional boxes if necessary.
1.5 Who is the nominated Contact Person (from those listed in 1.4 above) for this protocol?

<table>
<thead>
<tr>
<th>Name</th>
<th>Telephone Number</th>
<th>Email</th>
</tr>
</thead>
<tbody>
<tr>
<td>John Bassett</td>
<td>9351 2769</td>
<td><a href="mailto:jbassett@usyd.edu.au">jbassett@usyd.edu.au</a></td>
</tr>
</tbody>
</table>

1.6 Who is the person preparing this document?

<table>
<thead>
<tr>
<th>Name</th>
<th>Telephone Number</th>
<th>Email</th>
</tr>
</thead>
<tbody>
<tr>
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<td>9351 2769</td>
<td><a href="mailto:jbassett@usyd.edu.au">jbassett@usyd.edu.au</a></td>
</tr>
</tbody>
</table>

1.7 Are there students involved in this project?

If you answered YES, indicate the number of students covered by this study and the degrees which this study will contribute towards (i.e., Honours, Masters, PhD, etc.) If the names are already known please include them.

John Bassett, study will contribute to PhD

1.8 (a) Indicate the proposed date of commencement of the project.

Projects may not commence without the prior written approval of the HREC.

Date: 1/12/2010

(b) Indicate the proposed completion date of the project.

Date: 1/3/2011

1.9 Indicate all location(s) at which the research will be undertaken.

The CARLAB, Level 8, Electrical Engineering building

1.10 (a) Has this protocol received research funding/contracting or is this submission being made as part of an application for research funding/contracting?

If you answered YES, list the funding/contracting bodies to which you have submitted, or intend to submit, this project. Attach a copy of the grant application(s), contract(s) or similar agreement(s).

Funding/Contracting body 1:
Funding/Contracting body 2:
Funding/Contracting body 3:

(b) What is the outcome of these funding/contracting application(s) (please tick the appropriate box)

<table>
<thead>
<tr>
<th>RIMS_ID</th>
<th>Funding/Contracting body 1</th>
<th>Funding/Contracting body 2</th>
<th>Funding/Contracting body 3</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Approved Pending Refused</td>
<td>Approved Pending Refused</td>
<td>Approved Pending Refused</td>
</tr>
</tbody>
</table>

(c) Will this study still be undertaken if funding is not successful?

N Y

(d) If the title of the project submitted for funding is different from that listed
under Q1.1(a), state it below.

Proceed to Section 2.
SECTION 2: NATURE OF RESEARCH
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 23-45)

This section is obligatory

2.1 The nature of this project is most appropriately described as research involving:-
(more than one may apply):

- behavioural observation
- self-report questionnaire(s)
- interview(s)
- qualitative methodologies (e.g. focus groups)
- psychological experiments
- epidemiological studies
- data linkage studies
- psychiatric or clinical psychology studies
- human physiological investigation(s)
- biomechanical device(s)
- human tissue (see Section 11 – Medical Form)
- human genetic analysis (see Section 11)
- a clinical trial of drug(s) or device(s) (see Section 12)
- Other (please specify in the box below)

Proceed to Section 3.
SECTION 3: PARTICIPANTS AND RECRUITMENT
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 25-34)

This section is obligatory

3.1 (a) What is the age range of all participants involved in this study?

20 - 60

(b) If the participants include children (defined by statute for this purpose as anyone under 18) has a Prohibited Employment Declaration Form for the researchers (“criminal record check”) been lodged with the University or hospital? (see http://www.kids.nsw.gov.au/check/)

Y N

If you answered NO, give reasons why not.

There will be no children involved in this research.

3.2 Are the participants:-
(more than one may apply)

– in a teacher–student relationship with the researchers or their associates? X N

– in an employer–employee relationship with the researchers or their associates? N X Y

– in any other dependent relationship with the researchers or their associates? X N Y

– wards of the state? X N Y

– prisoners? X N Y

– refugees? X N Y

– members of the armed services? X N Y

– mentally ill? X N Y

– intellectually impaired? X N Y

– unconscious or critically ill patients? X N Y

– under the Guardianship Act 1987 (as amended)? X N Y

– in a doctor–patient relationship or a health giver–receiver relationship with the researchers or their associates? X N Y

– Aboriginal or Torres Strait Islanders? N X Y

If you answered YES to any of the above, provide details.

Some of the potential subjects are students of the study’s supervisor.
At least one of the potential subjects is aboriginal.
3.3 (a) What is the sample size for the study? Comment on how this sample size will allow the aims of the study to be achieved.

The sample size will be 24. The Australian Bureau of Statistics quotes a figure of 1.5 million people in Australia attending classical concerts in 2005. By the current population level this would approximate to 1.65 million in 2010. A sample of 24 will give the results a confidence level of 95% with a confidence interval of 20.

(b) How will the participants be recruited?

Investigators should note that the initial contact with participants should be at “arm's length” to avoid real or perceived coercion.

Potential participants from within a network of people recognised as potential concert goers will be approached by email. It is the intention of the study to survey a wide range of subjects, from professional musicians through to occasional concert goers. They will be asked to participate in a listening test of between one and two hours duration.

3.4 (a) Does recruitment involve a direct personal approach from the researchers to the potential participants? [ ] N [ ] Y

If you answered YES, explain how the real, or perceived, coercion from researchers for potential participants to enrol has been addressed.

The approach will be made by email to avoid the perception of personal pressure from the researchers. The email will clearly state that participation is entirely voluntary and the subject may withdraw from the study at any time.

(b) Does recruitment involve the circulation/publication of an advertisement, circular, letter, email letter etc? [ ] N [X] Y

If you answered YES, provide a copy. If recruitment involves an advertisement, please indicate where and how often it will be published.

3.5 Will participants receive any reimbursement of out-of-pocket expenses, or financial or other “rewards” as a result of participation? [X] N [ ] Y

If you answered YES, what is the amount or nature of the reward and the justification for this?
3.6 Is the research targeting any particular ethnic or community group?  

If you answered YES, which group is being targeted?

If you answered YES, is there an investigator who is a member of the Particular ethnic or community group?

If you answered YES to 3.6, has this project been planned in consultation with a representative of this group?

If you answered YES, who have you consulted and how do they represent this group?

If you answered NO, give reasons why you have not consulted.

Proceed to Section 4.
SECTION 4: PRIVACY


This section is obligatory

4.1 Is there a requirement for the researchers to identify, collect, use, or disclose information of a personal nature (either identifiable or potentially identifiable) about individuals without their consent?

(a) from Commonwealth departments or agencies?  \[ \square \] \[ \times \] \[ N \] \[ Y \]

(b) from State departments or agencies?  \[ \square \] \[ \times \] \[ N \] \[ Y \]

(c) from other third parties, such as non-government organisations?  \[ \square \] \[ \times \] \[ N \] \[ Y \]

If you answered YES to (a), (b) or (c), state what information will be sought and how many records will be accessed.

4.2 (a) Is there a requirement for the researchers to identify, collect, use, or disclose personal health information about individuals without their consent, which is identifiable or potentially identifiable?  \[ \square \] \[ \times \] \[ N \] \[ Y \]

IF YOU ANSWERED NO, YOU DO NOT NEED TO COMPLETE ANY MORE OF SECTION 4. GO TO SECTION 5

If you answered YES, indicate the reason(s)

– The project involves linkage of data  \[ \square \] \[ Y \]

– Scientific deficiencies would result if de-identified information was used  \[ \square \] \[ Y \]

– Other  \[ \square \] \[ Y \]

Please provide details
4.3 Will the health information that is identifiable or potentially identifiable with respect to individuals be collected, used or disclosed without the consent of the individual(s) concerned?  

If you answered YES, indicate the reason(s)

- The size of the population involved in the research.
- The proportion of subjects who are likely to have moved or died since the health information was originally collected.
- The risk of introducing bias into the research, affecting the generalisability and validity of the results.
- The risk of creating additional threats to privacy by having to link information in order to locate and contact subjects to seek their consent of the results.
- The risk of inflicting psychological, social or other harm by contacting subjects with particular conditions in certain circumstances.
- The difficulty of contacting individuals directly when there is no existing or continuing relationship between the organisation and the individuals.
- The difficulty of contacting individuals indirectly through public means, such as advertisement and notices.
- Other

Please provide details

4.4 Was this research the primary purpose of collecting the health information?  

If you answered YES, you do not need to complete any further questions in Section 4. Go to Section 5  
If you answered NO, please provide details

4.5 Would the subjects have expected the researchers to use or disclose their health information for the purposes of this project?  

Please provide details
4.6 Explain why the collection, use or disclosure of this information is in the public interest, and why the public interest in the project substantially outweighs the public interest in the protection of privacy.

Proceed to Section 5.
SECTION 5: COLLECTION OF DATA AND DISSEMINATION OF RESULTS
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 52-53)

This section is obligatory

5.1 Will any part of the study involve recordings using audio tape, film/video, or other electronic medium? 

If you answered YES, what is the medium and how it will be used?

5.2 Does your research involve the secretive use of photographs, tape-recordings, or any other form of record-taking? 

If you answered YES, provide details and a justification for the secrecy.

5.3 (a) How will the results of the study be disseminated (e.g. via publication in journals and presentations in scientific meetings)?

The primary output of the study will be a chapter in a PhD thesis. A conference or journal paper may be produced at a later stage.

(b) How will feedback be made available to participants (e.g. via a lay summary or newsletter)?

Please cross (X) the appropriate box:

X One (1) Page Lay Summary
- Written Transcripts
- Newsletter
- Report
- Web-based Feedback
- If NO feedback will be given, provide details below

5.4 How will the confidentiality of the data, including the identity of participants, be ensured during collection and dissemination?

Participants will be allotted a letter to identify the individual in the study. The data will be retained by the researcher. It will not be made available to anyone else. Due to the nature of the study no individual will be mentioned in any report. All data will be averaged across the group.
5.5 Is there any possibility that information of a personal nature could be revealed to persons not directly connected with this research?

If you answered YES, provide details.

X \ N \ Y

5.6 (a) What is the proposed storage location of, and access to, materials collected during the study (including files, audiotapes, questionnaires, videotapes, photographs)?

Please cross (X) the appropriate box:

- Chief Investigator/Supervisor’s Office
- Faculty / Departmental Office
- Other (Please provide details below)

Material will be stored in a padlocked locker in the researcher’s office.

(b) On completion of the study, where will the materials that were collected during the study (including files, audiotapes, questionnaires, videotapes, photographs) be stored?

Please cross (X) the appropriate box:

- Chief Investigator/Supervisor’s Office
- Faculty / Departmental Office
- Other (Please provide details below)

(c) Specify how long materials collected during the study (including files, audiotapes, questionnaires, videotapes, photographs) will be retained after the study, and how they will ultimately be disposed of.

Please ensure that the period of data retention stated here is appropriate to the nature of the proposed study. If the project involves clinical trial(s), the data should be kept for a minimum of 15 years (please refer to http://www.fda.gov/oc/ohrt/irbs/websites.html). If the projects do not involve clinical trial(s), the data should be kept for a minimum of 7 years after which time the data may be disposed of. (Please also refer to National Statement on Ethical Conduct in Research Involving Humans, 12.11 for further requirements).

Please cross (X) the appropriate box:

- 15 years for clinical trials
- 7 years
- Other (Please provide details below)

Please cross (X) the appropriate box/es:

- Paper / CD / DVD Shredding
- Audio / Video Tapes Erased
- Other (Please provide details below)
Proceed to Section 6.
**SECTION 6: RISKS AND BENEFITS**  
*(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 51)*

This section is obligatory

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<tr>
<td>6.1 (a)</td>
<td>Could participation in the research adversely affect the participants?</td>
<td>X</td>
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If you answered YES, complete 6.1 (b) and 6.1 (c). If you answered NO go to 6.2

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<td>6.1 (b)</td>
<td>Could the research induce any psychological distress in the participants?</td>
<td>X</td>
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<tr>
<td>6.1 (c)</td>
<td>Could the research cause any physical harm to the participants?</td>
<td>X</td>
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(e.g. from physically invasive procedures or from drug administration, etc)

If you answered YES to (b) or (c) describe the aspect(s) of the research and all the risks involved. Indicate the rate at which these risks are expected to occur. Indicate what facilities and trained personnel are available to deal with such psychological or physical problems.

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<tr>
<td>6.2</td>
<td>Will the true purpose of the research be concealed from the participants?</td>
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If you answered YES, outline the rationale and provide details for the concealment. Provide details of the debriefing. (If you do not intend to debrief, give reasons why not).

To avoid the possibility of leading the subjects to conclusions based on their expectations in the listening space the listeners will not be advised of the true nature of the test prior to their involvement. On completion of the test the researcher will explain the intention of the test and discuss those aspects with the subject. Their responses will be recorded in a debriefing sheet.

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<td>6.3</td>
<td>Are you doing research on patients (i.e. subjects receiving health care)?</td>
<td>X</td>
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If you answered YES, list the procedures/techniques which would not form part of routine clinical management.

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<tr>
<td>6.4</td>
<td>Is this research expected to benefit the participants directly or indirectly?</td>
<td>X</td>
</tr>
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</table>

If you answered YES, provide details.

Listener’s preference for an immersive sound when listening to music  
Page 1 of 2

Version 1, 29th October 2010
Proceed to Section 7.
SECTION 7: PARTICIPANT INFORMATION AND CONSENT
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p.12-13, p.28-29, p. 40-42, p.44-45, p.47-50, p.54)

This section is obligatory

7.1 Will a Participant Information Statement be provided?  
[ ] X  [ ] Y  [ ] N

7.2 Will written consent be obtained?  
[ ] X  [ ] Y  [ ] N

If you answered NO to either 7.1 or 7.2, give reasons why not.

7.3 In the case of participants who may not be fluent in English or who have difficulty understanding English, will arrangements be made to ensure comprehension of the Participant Information Statement and Consent Form?  
[ ] X  [ ] Y  [ ] N

If you answered NO, give reasons. If you answered YES, what arrangements have been made?

There is an expectation that all participants will have reasonably fluency in English.

7.4 (a) Do the Participant Information Statement and Consent Form have:-  
[ ] X  [ ] Y  [ ] N

- the first page of the Participant Information Statement and Consent Form printed on appropriate institutional letterhead?  
[ ] X  [ ] Y  [ ] N

- the title of the project on every page, including the Revocation of Consent? (if one is required) (Use a short title as appropriate)  
[ ] X  [ ] Y  [ ] N

- the page numbers expressed as page 1 of .., 2 of .., 3 of .. etc?  
[ ] X  [ ] Y  [ ] N

- an assurance that participation is voluntary and participants are permitted to withdraw from the project at any time without penalty?  
[ ] X  [ ] Y  [ ] N

- the name and telephone number of an appropriate researcher?  
[ ] X  [ ] Y  [ ] N

- a telephone number, fax number and E-mail address for the HREC, should a participant wish to make a complaint about the conduct of the research project?  
[ ] X  [ ] Y  [ ] N

(b) How has the possibility of withdrawal from the study been addressed in the Participant Information Statement and Consent Form?  

Participants are clearly advised that they may withdraw from the study at any time.

Proceed to Section 8.
SECTION 8: CONFLICT OF INTEREST AND OTHER ETHICAL ISSUES
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 51–54, Appendix 2)

This section is obligatory

8.1 Are any “conflict of interest” issues likely to arise in relation to this research?  
\[X\] \[N\] \[Y\]  
If you answered YES, provide details.

8.2 Do the researchers have any affiliation with, or financial involvement in, any organisation or entity with direct or indirect interests in the subject matter or materials of this research?  
\[X\] \[N\] \[Y\]  
(Note that such benefits must be declared in the Participant Information Statement.)  
If you answered YES, provide details.

8.3 Do the researchers expect to obtain any direct or indirect financial or other benefits from conducting this research?  
\[X\] \[N\] \[Y\]  
(Note that such benefits must be declared in the Participant Information Statement.)  
If you answered YES, provide details.

8.4 (a) Have conditions already been imposed upon the use (eg. publication), or ownership of the results (eg. scientific presentations) or materials (eg. audio-recordings), by any party other than the listed researchers?  
\[X\] \[N\] \[Y\]  
(b) Are such conditions likely to be imposed in the future?  
\[X\] \[N\] \[Y\]  
If you answered YES to (a) or (b), provide details.

Proceed to Section 9.
SECTION 9: DESCRIPTION OF PROJECT
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p. 13)

This section is obligatory

9.1 Describe the project using lay terms wherever possible, including the aims, hypotheses, research plan and potential significance. Where relevant, provide the projected number, sex, and age range of participants (including inclusion/exclusion criteria). You must satisfy the HREC that the study is scientifically valid (include at least four (4) research references) and conducted in accordance with the accepted ethical principles governing research involving humans.

The description must be no longer than 2 pages and must be in a font size of at least 10 points.

Aim

Architectural acousticians, such as Beranek[1], have claimed that a high diffusivity in sound within music performance venues is desirable. Soundfield diffusivity is best described as the degree to which a listener is immersed in the sound, where the sound comes from all directions rather than from directly in front of the listener.

This view has been supported, to some degree, by Bradley and Soulodre[2] who showed that listeners are able to detect variations in lateral or immersive sound. Their subjective tests show that listener envelopment is fairly independent of reverberation time, the primary measure of acoustic quality. There have been numerous attempts to establish an objective measure of listener envelopment, implied by high sound field diffusivity. The two dominant measures are Lateral Fraction (LF), proposed by Barron[3], and InterAural CrossCorrelation Coefficient (IACC) proposed by Ando[4]. In the case of LF the sound field measurement is highly simplified while the IACC uses a dummy head to measure the soundfield at the two ears.

Bassett (the PhD candidate) has developed a measure of the sound field based on the work of Furduev and T’ung[5] which seeks to measure the field in greater complexity than the LF method and at a point in space rather than the more listener-centric IACC. In order to test the efficacy of the measure the proposal is to test for correlation between listeners’ preference for immersive sound fields and this physical measure of the field.

Research plan

The venue for the tests is the Electrical Engineering multi-channel listening room. This space is a 5 x 5 x 2.5metre room treated to be almost anechoic. The anechoic room is an important component of the test because anechoic rooms are completely ‘dead’, they have no room reflections. In that facility test material may be played to the listener through 32 loudspeakers arranged into a spherical array around the subject. The lack of room reflections means the sound presented through the loudspeakers is not modified by the sound of the room. The implementation of the test is to establish a series of performance space simulations within which a virtual music ensemble is played. The reason for this is two-fold. The use of a room simulator allows the manipulation of some basic room characteristics so that the listener is presented with a ‘room’ where only the sound diffusion is altered by the researcher. Secondly, previous work has presented listeners with recordings made in anechoic rooms where the material is played through multiple loudspeakers. A failing of those anechoic ensemble recordings is that the ensemble effectively is being played from a single point in the simulator. It is desirable to ‘place’ individual instrument anechoic recordings within the virtual room. In this way the individual sources interact with the ‘room’ to produce a more complex sound which is a better emulation of a real listening environment.

The test

The subjective test will consist of listeners being presented with a paired comparison test where they are requested to select which of two music pieces presented is preferable. There is a possibility that different music may be deemed more or less appropriate in particular rooms. To minimise listener preference based on musical form two pieces will be presented, a segment of a Mozart opera, a classical era piece and a segment of a Mahler symphony. The assumption is that the listeners will prefer a more direct, less diffuse sound for the Mozart while the opposite will be preferred for the Mahler.
21 subjects will be chosen as much as possible to maintain gender balance and a range of both experience and age. The subjects will be provided with an interface that will allow them to progressively select the samples. Within each sample pair they will be able to switch between the music piece rendered through two different room simulations.

The CARLAB Audio Facility hosts two-way audio contact with the subject at all times. As well, there is video monitoring of the test room. Proper protocols and procedures have been established to ensure safe acoustic levels are present at all times. This has been verified with OH&S.

Proceed to section 10.
SECTION 10: FIELD-BASED RESEARCH (i.e., CONDUCTED OFF CAMPUS OR OUTSIDE A HEALTH SERVICE) INCLUDING RESEARCH CONDUCTED OUTSIDE AUSTRALIA
(refer to the National Statement on Ethical Conduct in Research Involving Humans, p.14, p.31-32)

This section must be completed for all applications involving EITHER field-based research OR research to be carried out in countries outside Australia (eg. in a school, a corporation, a government department an Aboriginal and Torres Strait Islander community or research in a another country).

10.1 Is your research conducted

(i) Outside Australia

(ii) Off Campus

(iii) In an Aboriginal and Torres Strait Islander Community

(iv) In a School

(v) In a Corporation

(vi) In a Government Department

(vii) In a Hospital

If you answered NO to all of the above, go to Section 11

10.2 Have you obtained formal permission from relevant authorities for entry to the area to carry out research (e.g., national or local government bodies, organisations of local communities)?

If you answered YES, name the relevant authorities and attach the relevant correspondence.

If you answered NO, give reasons.

10.3 If research is proposed among members of specific organisations, have you sought approval from those organisations (e.g., church groups, national associations, etc)?

If you answered YES, name the relevant authorities and attach the relevant correspondence or letter of support.

If you answered NO, give reasons.
10.4 Does the research involve individuals or groups of people who are not formally organised (e.g., people living in a village or town, etc)?

If you answered YES, indicate the context of the research. How will you obtain access to participants? Indicate any ethical issues that you can foresee in this approach.
10.5 Will your research necessarily involve the acquisition of objects of valuable cultural property (e.g., carvings, paintings, etc)?

If you answered YES, give details of arrangements with owners of the property with regard to access to/acquisition of these items, where appropriate.

N  Y

10.6 Will your research necessarily involve any activities that are likely to be seen by research participants and/or members of their local communities as in conflict with local practices and customs (e.g. regarding religious or ritual participation)?

If you answered YES, provide details.

N  Y

Proceed to Section 11.
SECTION 11. DECLARATION OF RESEARCHERS

I/we apply for approval to conduct the research. If approval is granted, it will be undertaken in accordance with this application and other relevant laws, regulations and guidelines.

Signature of Chief Investigator or Supervisor

Name: Craig Jin.................................................

Signature: ...................................................... Date: 19/11/2010

(print)

Signature of Associate Researcher(s) or Student(s)

Name: John Bassett...........................................

Signature: ...................................................... Date: 19/11/2010

(print)

Name: Roman Kosobrodov..........................

Signature: ...................................................... Date: 19/11/2010

(print)

Name: Densil Cabrera.................................

Signature: ...................................................... Date: 19/11/2010

(print)

Name: ..........................................................

Signature: ...................................................... Date: ..................

(print)

Signature of appropriate senior officer NOT ASSOCIATED with the research (e.g. Head of School OR appropriate).

After careful consideration and appropriate consultation, I have reviewed the attached HREC application, including the Participant Information Statement and Consent Form. I am satisfied that the scientific merit of this work justifies its being performed and that the information which will be obtained justifies the inconvenience and risks to participants.

Name: ..........................................................

Title: ..........................................................

Position: .......................................................

(print)
PARTICIPANT INFORMATION STATEMENT

(1) What is the study about?

(2) Who is carrying out the study?

The study is being conducted by student John Bassett and will form the basis for the degree of PhD in Architecture at The University of Sydney under the supervision of Senior Lecturer Densil Cabrera.

(3) What does the study involve?

The test involves a listener making a comparison between two identical pieces of music played within a room simulator. The subject will be asked to indicate which of the two sounds they prefer.

(4) How much time will the study take?

Depending on the speed with which the subject steps through the tests the study will take between one and two hours.

(5) Can I withdraw from the study?

Being in this study is completely voluntary - you are not under any obligation to consent and - if you do consent - you can withdraw at any time without affecting your relationship with The University of Sydney.

(6) Will anyone else know the results?

All aspects of the study, including results, will be strictly confidential and only the researchers will have access to information on participants. A report of the study may be submitted for publication, but individual participants will not be identifiable in such a report.

Listener’s preference for an immersive sound when listening to music
(7) Will the study benefit me?

There will be no direct benefit to you from participating in the study.

(8) Can I tell other people about the study?

You can tell other people about your experience in the study but please don’t reveal to anyone specific details about the test. This means other participants will enter the test with open minds.

(9) What if I require further information?

When you have read this information, John Bassett will discuss it with you further and answer any questions you may have. If you would like to know more at any stage, please feel free to contact PhD student, John Bassett, on 9351 2769.

(10) What if I have a complaint or concerns?

Any person with concerns or complaints about the conduct of a research study can contact The Manager, Human Ethics Administration, University of Sydney on +61 2 8627 8176 (Telephone); +61 2 8627 8177 (Facsimile) or ro.humanethics@sydney.edu.au (Email).

This information sheet is for you to keep
PARTICIPANT CONSENT FORM

I, ..........................................................[PRINT NAME], give consent to my participation in the research project

TITLE:       Listener’s preference for an immersive sound when listening to music

In giving my consent I acknowledge that:

1. The procedures required for the project and the time involved have been explained to me, and any questions I have about the project have been answered to my satisfaction.

2. I have read the Participant Information Statement and have been given the opportunity to discuss the information and my involvement in the project with the researcher/s.

4. I understand that my involvement is strictly confidential and no information about me will be used in any way that reveals my identity.

5. I understand that being in this study is completely voluntary – I am not under any obligation to consent.

7. I consent to: Receiving Feedback YES ☐ NO ☐

If you answered YES to the “Receiving Feedback Question”, please provide your details i.e. mailing address, email address.
Post test questions

1) How often do you attend acoustic music performances, such as orchestral concerts?
Weekly  Monthly  A few times a year  Less than once a year

2) Are you a:
   Performer
   Associated with music performance
   Audio professional
   Audio student
   General music lover

3) Did you find that the room simulations sounded realistic?
   Very realistic
   Realistic
   Unrealistic
   Very unrealistic

4) Was there anything distracting in the room simulation that prevented you from focusing on the test? If so, what?

5) Overall did the orchestra sound realistic in the simulation?
   Very realistic
   Realistic
   Unrealistic
   Very unrealistic

6) Overall was it easy or difficult to imagine that you were in a concert hall when the simulation was played?
   Very easy
   Easy
   Difficult
   Very difficult
Bassett ethics application - Proposed email to potential subjects;

During the first three months of 2011 I will be conducting subjective tests as part of my thesis at the University of Sydney. I am writing to you to ask whether you would be interested in participating in the study.

The test involves a listener making a comparison between two identical pieces of music played within a room simulator. You will be asked to indicate which of the two sounds you prefer. The average time required to perform the test will be one to one and a half hours. Being in this study is completely voluntary, if you are willing to be a subject in the study you can withdraw at any time without affecting any relationship you may have with The University of Sydney.

All aspects of the study, including results, will be strictly confidential and only the researchers will have access to information on participants. A report of the study may be submitted for publication, but individual participants will not be identifiable in such a report. If you would like to know more about the study, either prior to or following your participation in the study, please don’t hesitate to contact me.

Kind regards,
John Bassett
PhD candidate,
Faculty of Architecture, Design and Planning
The University of Sydney
9351 2769

Email addresses will be obtained from three mailing lists that Mr Bassett belongs to;
Sydney Opera House Technical Department
Conservatorium staff via a request to post on the staff bulletin
Audio students within the Architecture faculty via the Postgrad mailing list.
Appendix F: Publications associated with this thesis

Presented at the 18th International Congress on Acoustics, 2004, Kyoto, Japan.

Measuring the directional variation of spectral energy across time in the Concert Hall of the Sydney Opera House

John Bassett\(^{(1)}\), Craig Jin\(^{(2)}\), Densil Cabrera\(^{(1)}\), Riduan Osman\(^{(1)}\)

\(^{(1)}\)School of Architecture, Design Science and Planning
\(^{(2)}\)School of Electrical and Information Engineering
University of Sydney, Australia

jbassett@mail.usyd.edu.au
Abstract
This paper assesses a measurement method that seeks to provide information on the directional flow of energy in a space. Impulse responses on orthogonal Cartesian axes are derived through convolution of the received signals of a Soundfield microphone and the original test signal. The impulse responses are processed in Matlab to produce a spherical plot of energy level over 5 ms timeframes. Further analysis is possible through third octave filtering of the energy time curves. The results are plotted providing a visual analysis tool for assessment of the directional information associated with the reflection pattern, contained within the impulse response of a room.

1. Introduction
Measurement criteria for auditoria emphasise temporal characteristics of the sound-field. Spatial sound-field characteristics can be represented using binaural or bipolar ratio measurements such as the interaural cross correlation and lateral fraction. Such measures have been useful in predicting aspects of subjective spatial impression, including apparent source width and listener envelopment [1, 2]. Three-dimensional representations of the sound-field, possible with multiple microphone channels, can be seen as extensions of lateral fraction measurements. Such an approach has been taken, for instance, using a three-dimensional intensity probe [3], arrays of four pressure microphones [4, 5], and potentially using many other multiple channel techniques. Such techniques may be used to represent acoustical energy flow over time, thereby giving an indication of the direction of discrete reflections, and the changing spatial distribution of the sound-field.

The present study illustrates the analysis potential of a four-channel coincident microphone through example measurements in the Sydney Opera House Concert Hall. It seeks to extend the methods outlined above by suggesting a visual analysis approach for energy flow representation over time for a position in an auditorium.

2. Method
Measurements were carried out using a Soundsphere loudspeaker and a purpose built sub-frequency loudspeaker at six positions on the stage. A range of test signals, including seven sine sweeps from 20 Hz to 22 kHz with a duration of approximately 5 seconds were recorded on an Alesis HD24 Hard Disk Recorder via a B-Format Soundfield microphone. The Soundfield microphone was placed at a height of 1.2 metres in a range of seat positions throughout the hall. Using Matlab impulse responses corresponding to a hyper-cardioid directional microphone were derived for 188 directions around the sphere (approximately every 20 degrees spherical angle) using the impulse responses recorded by the Soundfield microphone. This provides the opportunity to examine the energy flow at the receiver position, over time, by plotting the level measured at each position on the sphere. From this it is possible to examine the effect of reflecting surfaces relative to a particular seat in the hall.

2.1. Analysis system
The RMS energy of the impulse signal as a function of time was calculated using a 0.4 ms time window. The average energy corresponding to larger time windows was derived from the data averaged over the 0.4 ms time window. The degree of variation in level over each window is illustrated below for the measurement position J21(Figure 1). This figure shows the difference between the maximum and minimum levels plotted against time.
Figure 1: Variation in level between maximum and minimum levels in each window across the sphere.

Analysis of the directional variation of the RMS energy shows that the range of variation is about 15 dB across the sphere, for the 5 ms window, and is relatively constant across time, even during the late reverberation tail.

The directional energy pattern around the sphere was derived using a spherical thin-plate spline interpolation that was fit to the data corresponding to the 188 directions. The data were plotted using the ‘Eckert6’ map projection in MATLAB. Furthermore the derived impulse responses were filtered using a bank of 1/3-octave band-pass filters. The filters consisted of a cascade of three second-order Chebyshev band-pass filters. Through examination of the Energy Time Curve for a particular seat position individual reflections may be selected and their directional character plotted either for a full range signal or for individual one third octave bands.

2.2. Energy plots

A resultant plot of the initial impulse arrival for seat J21 in the stalls of the Concert Hall is shown in Figure 2. This figure illustrates the arriving energy of the spherical plot represented in the two dimensional figure by establishing the zero degree point in the vertical and horizontal axes as the centre point of the figure. The equivalent point on the rear of the sphere can be found at the horizontal extents of the figure at the 180 degree rotation points. The pole points of the sphere, at the top and bottom of the sphere have been laterally stretched to allow a more visible representation of the z-axis.
The first reflection plot (55ms) indicates the direction of the reflection to be from 30° above and to the left of the direct signal. Armed with the directional information and the reflection path length, derived from the arrival time, the reflecting surface most likely to be contributing to the reflection may be identified in the hall. Similarly, the second reflection, arriving 45° above and to the left of the direct signal may be attributed to the side walls of the roof space in the hall. The third arrival plot in figure 3, part of a cluster of reflections arriving between 125 and 160mS show a broadening of the energy arriving at seat J21, within this time window. This implies a complex series of reflections from multiple surfaces to the left and above the seat position.

Deeper analysis is possible by plotting the energy arrival of the filtered impulse responses as shown in Figures 4 and the right hand side of Figure 3. Here the energy arrival in the third octave bands 250Hz and 1kHz are co-plotted showing significant differences in energy level at particular time windows.

Figure 2: Plot of direct sound arrival at seat J21

This seat is in the stalls area of the hall slightly off the centreline. The stage in the Concert Hall is 1.3 metres above the auditorium floor placing the source approximately 1 metre above the receiver in this position. This is clear from the plot for the initial impulse where the light area, indicating highest signal level is marginally above the centre point of the plot.

Figure 3 below, illustrates the ability to read the energy flow in the room at the measurement position, within a 5ms time window. The full range plots on the left hand side of the figure show the plots developed for the strong reflections arriving 55, 90 and 149ms after the initial impulse.

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Figure 3: Full range plot of direct sound arrival at seat J21 and octave band plots at Box W26
Figure 4: Energy Time Curves for 250Hz and 1kHz

Plotting the direction of the energy flows in the time windows 160mS and 375mS show that there is significant variation in the spectral content of the field relative to time. At 160mS there is a strong reflection approximately 22dB below that of the initial impulse at 1kHz. The level of the 250Hz component of the energy, in this timeframe, is approximately 15dB below the 1kHz component. Examination of the plot for this time window show that whereas the 1kHz signal is arriving directly in front of the receiver the 250Hz signal is arriving 60° to the right in the upper hemisphere. Conversely, at 375mS the energy at 250Hz is arriving from the front and overhead at a level 23dB below the direct signal while the 1kHz component has a more complex arrival pattern some 10dB lower in level.

3. Conclusions

The method outlined may provide a useful tool for analysis of energy flows within spaces such as auditoria. The ability to locate the direction of the energy flow related to a particular reflection, detected in the impulse response of a hall, gives the practitioner or researcher the means of visualising the reflections arriving at a receiver position. This may allow improved analysis of faults in halls particularly where they are of a spectral nature. The approach indicates potential areas for investigation and refinement particularly in development of a means of measuring the degree to which a field is diffuse.
4. Acknowledgements

The authors gratefully acknowledge the contributions of H K Cho, Y J Choi, K Koralage, G Pope, K Stewart, and J Xu in planning, making and analysing the measurements.
They thank the Sydney Opera House for allowing the measurements to be made.
They thank Lake Technology for the loan of equipment.

5. References