Application of Auralisation and Soundscape Methodologies to Environmental Noise

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Abstract

The thesis investigates how auralisation and qualitative research methodologies developed in the field of soundscape research, may be applied to the issue of environmental noise in urban areas. The project brings together research and assessment methodologies from traditionally distinct areas in acoustics, and attempts to develop a multidisciplinary approach to solving noise related urban design problems. Within this process, the author also explores the theory that part of a sustainable solution to noise annoyance lies in challenging negative attitudes to noise, particularly in urban environments, where attempts to reduce sound pressure levels are either not feasible or inappropriate.

To demonstrate the application of the approach, a case-study is presented involving the auralisation and subjective evaluation of a real-world soundscape before and after the insertion of a sonic crystal noise barrier. The results of the case-study suggest that, with minimal adaptation, certain soundscape assessment methodologies are compatible with virtual acoustic environments. This leads to the conclusion that, using the virtual assessment methodology, one is able to predict with reasonable accuracy the impact of various physical types of noise intervention on the perceived sound quality in urban environments prior to their construction. The author also argues that a greater awareness and appreciation for the multiplicity of sound in urban environments might, in conjunction with sustainable noise control strategies, offer more long-term benefits to society than over-speculation concerning the adverse effects of noise.

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And last, but by no means least, I would like to thank Maxwell, for the doggedness that has ultimately brought this thesis to its conclusion.

Declaration

I hereby declare that I am the sole author of this thesis and that it was produced without making use of aids other than those specified; notions taken over directly or indirectly from other sources have been identified as such. I also declare that some parts of this program of research have been presented previously at conferences. These publications are listed as follows:

- **3-D FDTD Simulations of Sonic Crystals Based on Triply Periodic Minimal Surfaces**, Sorrel Hoare and Damian Murphy, paper and oral presentation at the *Baltic-Nordic Acoustics Meeting (BNAM)* , Odense, 2012
- **Prediction of Scattering Effects by Sonic Crystal Noise Barriers in 2-D and 3-D FDTD Simulations**, Sorrel Hoare and Damian Murphy, paper and oral presentation at *Acoustics 2012*, Nantes, 2012
- **Auralisation of Sonic Crystals through Simulation of Acoustic Band Gaps in 2-Dimensional Periodic Scattering Arrays**, Sorrel Hoare and Damian Murphy, poster presented at *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, NY, 2011

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1

Introduction

Sound is all around us. Outside, the everyday sounds of nature and human endeavour merge to form the rich sonic tapestries that give an environment its own distinctive character. While we go about our day-to-day lives, preoccupied with whatever thoughts and plans are evolving in our conscious minds, unconsciously, *environmental sounds* are informing us and influencing our behaviours, attitudes and moods. As the thinking brain emerges in infancy, so absorbed do we become by our own internal monologue, that, as adults, we are almost permanently disengaged from the symphony that is occurring all around us - of which we are principal players. So much so, that we rarely notice environmental sounds at all except on the occasions when they encroach on our inner space, our 'head space'. At which point, we throw up our hands and ask indignantly 'what is that *noise*? I cannot *hear myself think*!' Having previously been barely acquainted with the sound environment, suddenly we are at odds with it; attaching to it feelings of irritation, anxiety and loss.

Yet, in making this abrupt transition, it seems we devalue environmental sound, by neglecting its many positive aspects. For along with the capacity to annoy, environmental sound defines and enriches the places we inhabit - environments which we have invariably had a hand in shaping, whether they be rural, urban, industrial, or a lone cabin on a remote hilltop. So, by extension, it can be said that environmental sound defines *us*; as a species, as societies, as communities - even as individuals. Just as the visual landscapes can be seen as illustrations in the ongoing history of nature and humankind, so the *soundscapes* have their stories to tell. And because they are so fundamental to who we are, we come to associate the sounds of our environments with experiences in our lives; attaching them to places, events and emotions like page markers in a book. Yet, in spite of this meaningful and complex relationship between humans and the sound environment, there is growing concern that the volume of environmental sound affecting some areas - mainly urban - is not only harmful, but may become unmanageable if measures are not taken to suppress it. In fact, so great is the concern over *environmental noise*, that some have dubbed it the *modern plague* of society [1].

Environmental noise is generally considered to be *unwanted sound* in the environment, and is regarded in psychological terms as an environmental *stressor*. As such, the impact of noise on the behaviour, mood, cognitive function, physical health and/or psychological well-being is one that can be proven - particularly in the case of noise-sensitive groups such as children, the elderly, and those with existing mental health problems. But how does one define unwanted sound? How do we decide what is dispensable, and what has value? The author would argue that it is not satisfactory to consider noise in terms of pleasant or unpleasant sound, as even unpleasant sounds have value. Moreover, the notion of pleasant or unpleasant sound is highly subjective, and while there may be a strong inclination towards certain sounds based on physiological or social factors, reactions to sound are also inextricably linked to an individual's personal and cultural experiences, present mood and situation. In which case, is it even possible to distinguish between *good and bad environmental sound?* Or, if it is a question of quantity, then *how much is too much?* It is considerations such as these which have been given a great deal of attention in recent years, particularly in the interdisciplinary field of *soundscape research*. For example, the notion of a *positive soundscape* conceived by soundscape researchers seeking to re-evaluate environmental sound [2] presents a question in itself which many have sought to answer.

The term *soundscape* can be defined in a number of different ways, although here the author uses it to mean *all the sounds that are heard in a particular location, considered as a whole*. Soundscape research not only recognises but pays close attention to the reciprocal relationship humans share with the environment, and seeks sustainable ways to manage environmental sound, while considering the social and economic needs of a population. In this respect, there is an element of overlap with environmental psychology, where attention is given to building and testing theories in order to explain and predict human-environment interaction [3]. While efforts to improve our understanding of positive soundscapes have so far proved fruitful, to the author's mind, a great deal remains to be done, particularly in terms of testing and implementing these theories in practice. For example, *what tools are there that can be used to inform the design and regeneration of urban soundscapes?* And, if such tools exist, *who is going to use them?*

1.1 Motivation

The role of the acoustician is well established in the field of architecture. He or she will invariably be consulted at some stage in the process of planning a new building and in some cases, such as buildings primarily intended for musical performance, may even influence the design to some extent. However, this is in stark contrast to what typically goes on in the case of outdoor developments. In the development or regeneration of outdoor spaces, the acoustic environment is frequently overlooked until problems come to light requiring remedial treatment. That said, it is often argued that more attention needs to be given to the acoustics of urban spaces, or that creative use of sound could form part of the 'cure' for other unfavourable characteristics of the space. For example, it has often been posited that music may act as a deterrent in places where antisocial behaviour and vandalism are prevalent [4]. While there is no proven scientific rationale behind this argument, there is plenty of evidence to suggest that aesthetically pleasant environments can help promote happiness and well-being in a population (e.g. [3, 5]), which would undoubtedly lead to other benefits.

There is, therefore, the hope that the modern *urban designer* will consider the soundscape as an integral part of the the landscape they are seeking to transform, and afford the same care that is already devoted to other aspects. In many cases, the soundscape represents a valuable opportunity to impart to a space its own distinctive character, and it is the premise behind this thesis that *auralisation* could go some way towards achieving this. It is thought that the ability to create diverse and structurally unique soundscapes offers other practical benefits to society too. For example, the incorporation of distinctive acoustic features, or *soundmarks*, into the design of urban spaces might serve as a navigational tool for the visually impaired.

Another factor motivating this research concerns current attitudes to environmental sound, which can appear to be somewhat polarised between separate disciplines - i.e. those of environmental acoustics and sections of the arts and social sciences. The author asks how soundscape methodologies might form part of a more unified approach to dealing with the negative effects of noise - one that is scientifically robust, yet sensitive to human-environment relationships. Such an approach requires a well-developed understanding of how humans interact with the sound environment - something that auralisation may be able to assist with.

A final aspect of this research has been to consider the technical challenges associated with the acoustic modelling and auralisation of outdoor spaces. Whereas most advanced auralisation techniques are typically aimed at reproducing the acoustics of rooms and other resonant spaces, their application to outdoor environments is less widely practised. Part of the reason for this may be the incompatibility of an exclusively physics-based approach . For example, in architectural acoustics, the whole acoustic system may be modelled to a high level of precision, which would be computationally unfeasible in the case of complex outdoor environments - at least with today's technology. Furthermore, these systems are less well suited to the assumptions of linearity and time-invariance which tend to be applied to indoor spaces. However, the ability to observe the influence of isolated structures or specific noise sources on the perceived sound quality in outdoor environments is considered to be entirely achievable using existing auralisation techniques, provided a perceptually integrated approach can be taken.

1.2 Statement of Hypothesis

The hypothesis guiding this thesis can be summarised as follows:

> *The application of auralisation and qualitative test methodologies to environmental sound can be an effective aid in the design, evaluation and promotion of positive urban soundscapes.*

In much the same way as auralisation is already employed by acousticians in relation to indoor spaces, here it is proposed that auralisation may be used to inform the design of outdoor spaces. For this to happen, the auralisation must go beyond the realms of communicating an idea, but should be combined with appropriate qualitative test methodologies in order to test that specific idea. In this sense, it may also be seen as a medium though which we may develop our understanding and appreciation of environmental sound.

1.3 Thesis Outline

The thesis starts with an introduction to the fundamental principles of acoustics. It then gives a comprehensive review of current research and assessment methodologies in three traditionally distinct areas of acoustics: soundscape studies, environmental acoustics and auralisation. Throughout the literature review, the author considers the potential for auralisation to be used as an 'evaluation tool' in urban design and environmental noise control. Urban design, at the time of writing, is an emerging discipline which strives to unite the roles of architect and acoustician at an early stage in the design of urban spaces. In the context of this document, the term *urban space* shall refer to an open or partially open public area within a large town or city, such as a park, city square, courtyard or concourse.

Regarding the auralisation of urban spaces, there are two main areas for consideration. Firstly, there is the suitability of current acoustic modelling techniques to be considered. These are typically used in relation to indoor problems (i.e. small, reverberant spaces), and may require adaptation to make them more applicable to outdoor problems (i.e. large-scale, unbounded spaces). These problems are addressed in Chapter 6. Secondly, there is the question of how to evaluate the auralisation in subjective experiments. While there exists a well defined set of acoustic parameters for the evaluation of indoor spaces (e.g. [6, 7]), as yet there is no established convention by which to assess the validity of soundscape auralisations. It is this second requirement of the auralisation that has led the writer to a much broader question regarding the possibility of using auralisation as a practical 'evaluation tool' in relation to real-world environmental noise problems and urban soundscape design.

Chapter 7 focuses on the design and auralisation of a specific form of noise treatment in a real urban space, such as a playground, park or city square. The treatment chosen for the case study is a variety of noise barrier belonging to a class of acoustic meta-materials known as *sonic crystals*. A sonic crystal is a physical structure consisting of a periodic array of solid inclusions embedded in air. Depending on the geometry of the inclusions and the wavelengths of incident sound, it is theoretically possible to observe a number of interesting filtering effects in the transmitted signal. These include the selective attenuation or reinforcement of sound over a narrow range of frequencies, as well as the formation of sound beams. The prospect of selective attenuation has led some researchers to suggest that the devices might be used to help combat unwanted sound in the environment. While various types of sonic crystal noise barrier have been examined both empirically and analytically in the literature, the audibility of the filtering generally remains unclear until the device has been constructed and exhibited in the space for which it is intended. It is therefore posited that auralisation and an appropriate form of qualitative test methodology might offer a convenient solution to this problem. The structural design of the barriers will be founded on well established scientific theory, with reference to the relevant European guidelines [8] for control of environmental noise. The results of these auralisations are subsequently evaluated in carefully conceived subjective experiments based on methodologies developed in the field of soundscape research. The results of the experiments should support or refute the idea that such an approach offers significant benefits to the health and enjoyment of users of the space.

2

Fundamentals of Acoustics

2.1 Introduction

Acoustics is a broad subject area involving the study of all types of mechanical wave that are supported in solids, liquids and gases. The term is derived from the Greek word $\alpha\kappa o\upsilon\sigma\tau\iota\kappa\acute{o}\varsigma$ (akoustikos), meaning 'of or for hearing, ready to hear' and 'heard, audible' [9], hence its natural association with audible sound, despite its usage now extending to applications involving the entire range of frequencies. The Latin synonym for acoustics is *sonic*, leading to the familiar terms *ultrasonic* and *infrasonic* in reference to frequencies of sound above and below the audible range respectively.

The field of audible acoustics or sonics can be further broken down into several sub-fields. Three which are of particular interest in relation to this thesis are the fields of psychoacoustics, concerning the sensation and perception of sound; environmental acoustics, concerning the propagation of sound outdoors; audio signal processing, concerning the representation, alteration, transmission and reproduction of auditory signals in either the digital or analogue formats; and architectural acoustics, concerning the behaviour and perception of sound in buildings. It is thus the aim of this chapter to deliver a general introduction to acoustics and to outline relevant principles in each of these areas.

2.2 Basic Properties of Sound

2.2.1 Mechanical Description

Sound occurs when some applied physical force causes particles in a medium to vibrate at a rate that is within the range of frequencies to which the human ear is sensitive. The displacement of a particle from its stable equilibrium causes a local fluctuation in the ambient pressure, which, due to certain physical properties associated with the medium, causes the particle to exert either a push or a pull force on its nearest neighbours. This effect is then reciprocated by the neighbouring particles, and so on in a ripple effect, until the resulting pressure wave is ultimately dissipated by the medium. Areas of high pressure where particles are more densely packed are referred to as compressions, while areas of low pressure where the particles are more sparsely distributed are referred to as rarefactions (see Figure 2.1).

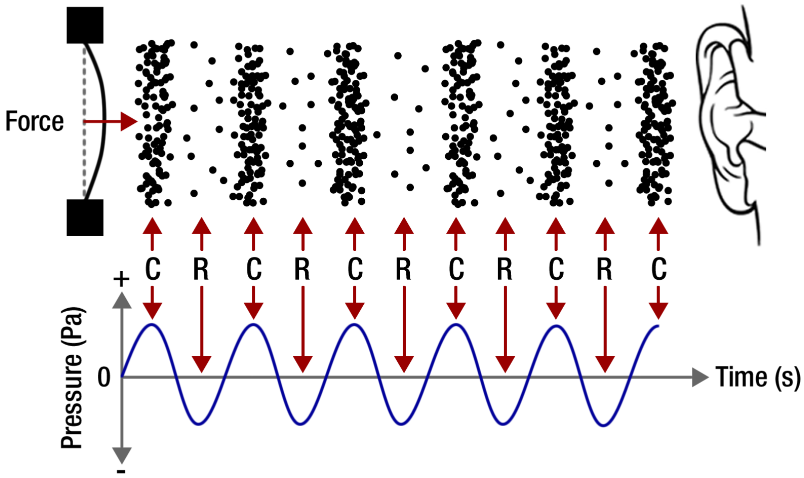


Figure 2.1: Sound is a pressure wave consisting of regions of compression (C) and rarefaction (R). Reproduced from [10].

The human auditory system is responsive to these small fluctuations in local ambient pressure, provided the rate at which they occur is within a certain range of frequencies. Pressure waves within this range or spectrum are thus referred to as sound waves.

Longitudinal Sound Waves

In an ideal fluid the movement of the particles is assumed to be a smooth and repetitive parallel or anti-parallel oscillatory motion. This type of sound wave is referred to as a longitudinal wave due to the one-dimensional nature of the oscillation. In a longitudinal wave, the pressure at a given point in time may be expressed by the sine function:

$$p(t) = A \cdot \sin(2\pi f t + \varphi) = A \cdot \sin(\omega t + \varphi) \tag{2.1}$$

in which the following definitions apply:

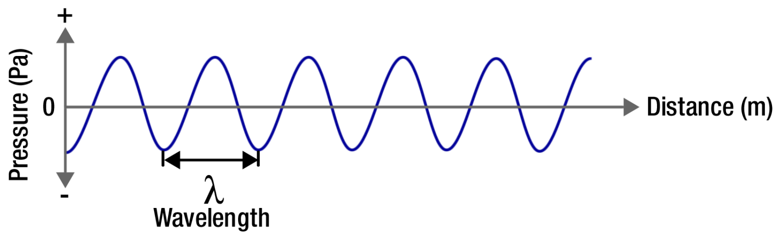
- p is the instantaneous sound pressure or force per unit area at a specific moment in time, t.
- A is a value associated with the peak amplitude or maximum displacement of the function from zero.
- f is the temporal frequency, or number of oscillations per unit of time, which is generally taken to be cycles per second or Hertz (Hz).
- ω is the angular frequency or rate of change. ω is defined in radians per second and is related to temporal frequency by the expression $\omega = 2\pi f$.

- φ is the phase, also expressed in radians, and indicates where in its cycle the oscillation is at $t = 0$.

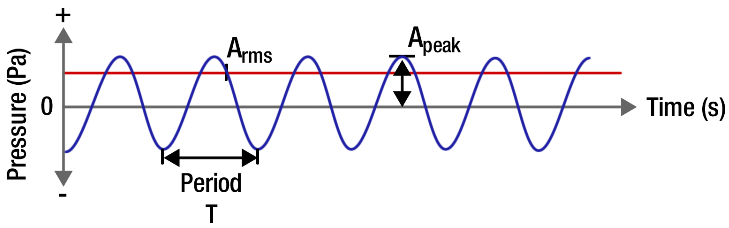
The period of time in seconds over which the cycle repeats is denoted by T. The spatial analogue of T is the wavelength or distance over which the cycle repeats denoted by λ (see Figure 2.2a). The speed of propagation is thus derived from the wavelength and the temporal frequency of the oscillations by the relationship,

$$c = \frac{distance}{time} = \frac{\lambda}{T} = \lambda f \tag{2.2}$$

Since particle displacement can result in both positive and negative changes in atmospheric pressure, quantifying the degree of change in terms of the maximum or peak displacement is not generally an indication of how loud the sound will be perceived to be by a human, particularly in the case of complex sounds. A more meaningful measure of the degree of change over a given period of observation would be to take the mean average of the displacement at all points along the waveform. However, in the case of a sine wave this would obviously result in zero displacement, hence it is necessary to square the amplitude at each point before averaging, and then take the square root of the result. This is known as the *root mean square* (RMS) amplitude of the waveform, which in the case of a pure sine wave is simply the peak amplitude multiplied by $\sqrt{2}$.



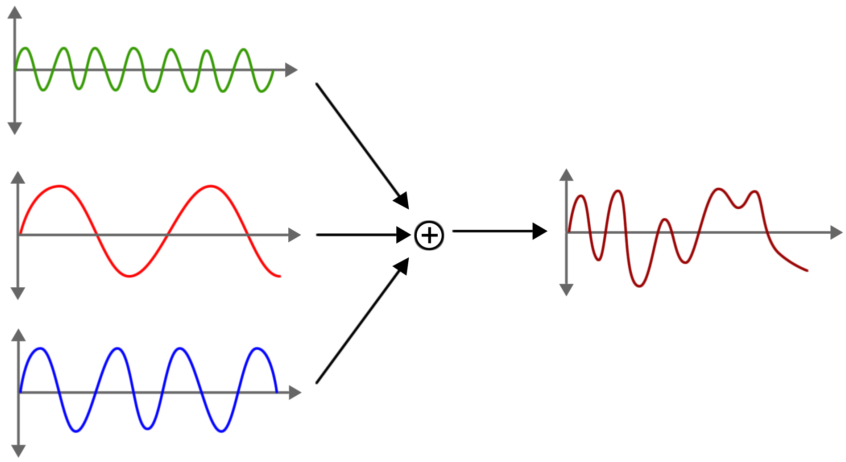
(a) Amplitude-distance graph of a longitudinal sine wave.



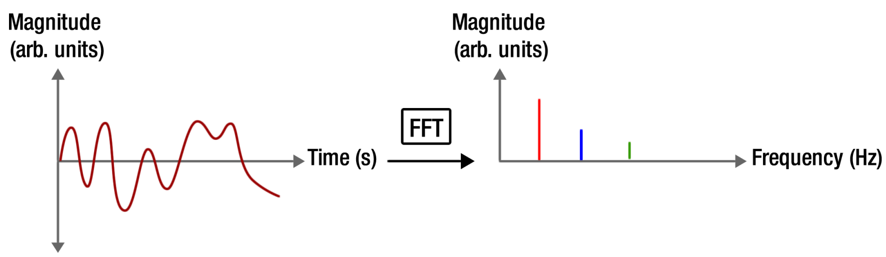
(b) Amplitude-time graph of a longitudinal sine wave showing the peak (A_{peak}) and RMS (A_{rms}) amplitudes.

Figure 2.2: Temporal and spatial representations of a longitudinal sine wave.

Longitudinal sound waves share the same mathematical principles as sine waves, including that of wave superposition, a principle which describes the interaction of multiple sine waves propagating through a medium at the same time. According to the theory of wave superposition, the total displacement of a medium at a particular location and time is equal to the net sum of the individual wave displacements. This means that sound waves of the same frequency travelling in opposite directions will effectively pass through one another without disturbing the waveform. It also leads to the supposition that any sound - other than a pure tone - may be expressed as the sum of multiple sinusoidal sound waves of different amplitude, phase and frequency. Thus it is said to be the superposition of many individual wave components that give a sound texture of character - attributes one tends to associate with 'timbre'. The theory of wave superposition also leads to Fourier analysis - the mathematical process by which a complex waveform may be broken down into its constituent sinusoids.



(a) According to the theory of wave superposition, any complex waveform is considered the sum of multiple sinusoidal components.



(b) The sinusoidal components that make up a complex waveform may be recovered through the process of Fourier analysis.

Figure 2.3: Wave superposition theory is a fundamental mathematical concept in acoustics.

2.2.2 Speed of Sound

The speed of sound may be affected by a number of factors associated with the medium. In a fluid, the primary factors affecting the speed of propagation are its density ρ and bulk modulus K - both of which are volumetric measures, and affected by temperature. Density is defined as the mass (m) per unit volume (V) according to (2.3). The bulk modulus is a measure of stiffness, or the resistance of the medium to a uniform compression, and is defined according to (2.4) in which $\frac{\delta p}{\delta V}$ denotes the derivative of pressure with respect to volume.

$$\rho = \frac{m}{V} \tag{2.3}$$

$$K = V\frac{dp}{dV} \tag{2.4}$$

The bulk modulus may also be derived from the density according to the following relationship, wherein $\frac{\delta p}{\delta \rho}$ denotes the derivative of pressure with respect to density.

$$K = \rho\frac{dP}{d\rho} \tag{2.5}$$

The bulk modulus is in fact one of several elastic moduli which describe the *elasticity* of a substance i.e. its ability to oppose deformation, and to restore itself to equilibrium in response to a deformation. For small deformations, most elastic materials exhibit linear elasticity, in which case it is generally sufficient to characterise the elasticity of the medium in terms of a single elastic constant. Longitudinal waves are thus commonly modelled by the mass-spring systems of classical mechanics wherein the masses assume the role of the particles, while the spring represents the coupling between adjacent particles. Hooke's law states that the force (F) needed to restore a mass (m) to its equilibrium (x_0) following its displacement by an amount (dx) is dependent on the characteristic stiffness of the spring, as denoted by a constant factor (k) known as the elastic or spring constant.

$$F = -kx \tag{2.6}$$

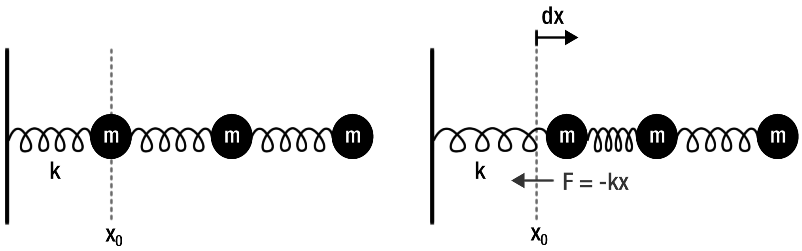
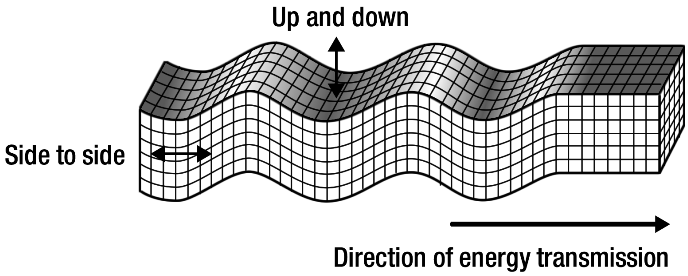


Figure 2.4: A coupled mass-spring system.

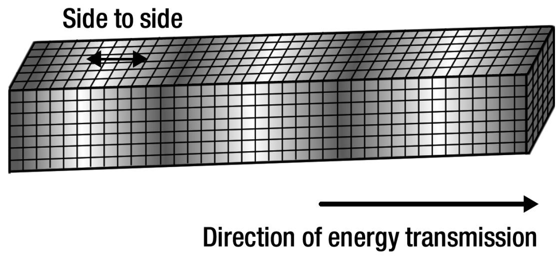
In acoustic systems, the elastic constant may be represented by one or more elastic moduli depending on the properties of the medium and its ability to sustain wave motion of different types. In a compressible medium, only longitudinal waves are supported, hence the speed of sound may be approximated by the following relationship in which c denotes the speed of sound, and K and ρ are the bulk modulus and density of the medium respectively.

$$c = \sqrt{\frac{K}{\rho}} \tag{2.7}$$

In dry air at 20° this equates to a speed of 343 metres per second and is constant for all frequencies. In non-ideal fluids, the speed and dissipation of sound waves may also be affected by other physical factors such as friction and turbulence, as well as thermal and molecular discontinuities in the medium. However, in most practical situations it is sufficient to consider the medium - usually air - in terms of an ideal fluid of constant temperature and humidity. In incompressible media such as solids and liquids, particles may also vibrate in directions other than that of the direction of travel, meaning additional wave modes such as transverse (shear) waves (see Figure 2.5) are also supported.



(a) In a transverse wave the motion of the wave is perpendicular to the direction of propagation, so for a wave propagating from left to right, its oscillations would be up and down.



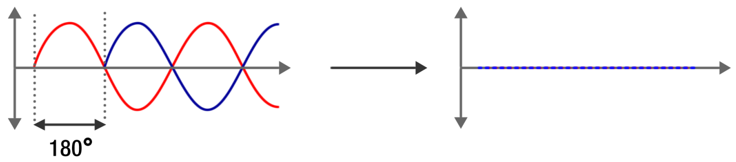
(b) In a longitudinal wave, particles oscillate in the direction of wave motion.

Figure 2.5: Comparison of longitudinal and transverse pressure waves.

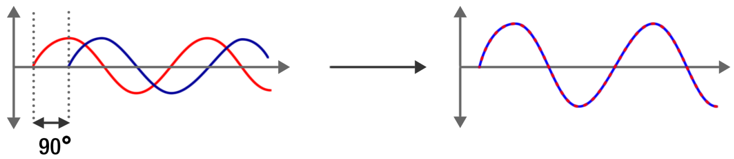
The mathematical description of sound propagation through solids is potentially very complex compared with that of fluids and requires additional elastic moduli. This essentially marks a divergence in acoustics, between those areas which deal mainly with 'sonics' - i.e. audible sound transmitted through air - and other areas such as ultrasonics and structural acoustics which deal with transmission through solid structures.

2.2.3 Wave Interference Effects

The principle of wave superposition means that the phase relationship between sound waves propagating through a medium at the same time can lead to both constructive and destructive interference, resulting in a sound wave of higher or lower amplitude within a certain region of space (see Figure 2.6). Destructive interference occurs when an area of low pressure overlaps with an area of high pressure. In this case the waves are said to be 'out of phase'. Conversely, constructive interference occurs when the two waves are 'in phase' - i.e. their peaks and null points coincide resulting in addition. These peaks are referred to as anti-nodes, whereas the null regions are referred to as nodes.



(a) Complete phase cancellation results when two waveforms are 180° (π radians) out of phase.



(b) Interaction between sound waves of the same frequency may result in constructive or destructive interference depending on the phase relationship between them.

Figure 2.6: Constructive interference results from interaction between two waveforms in phase.

A surface over which a wave has constant phase is referred to as a *wavefront* (see Figure 2.7). The simplest wave interference patterns are thus formed when longitudinal plane waves - i.e. waves of constant frequency with characteristically flat or 'planar' shaped wavefronts - intersect at an angle θ (see Figure 2.8a). Point sources radiate sound over a spherical surface, hence the interference from point sources is dependent on the distance between the sources as well as the wavelength and phase of the waves (see Figure 2.8b). In both of these examples the sources of sound are said to be coherent because they are phase-linked i.e. they have a constant phase difference. Wave interference due to coherent sound sources is an important consideration in most areas of acoustics and is associated with a large number of acoustical problems and remedies. For example, coherent source positioning can result in elevated sound levels at a given receiver position, or it can be exploited in the design of directional transmitters and receivers by the application of Huygen's Principle.

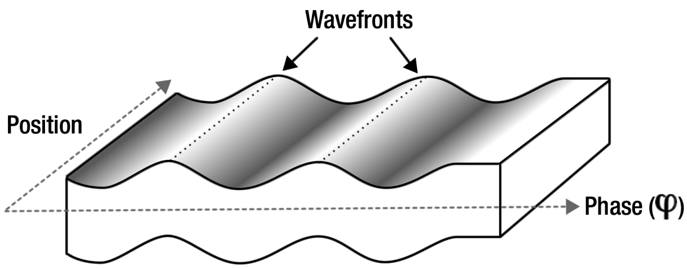
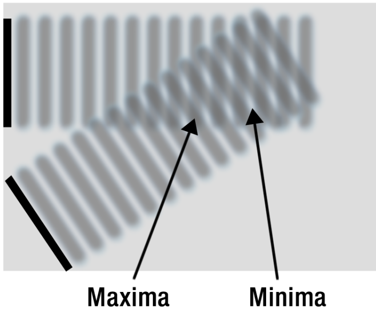
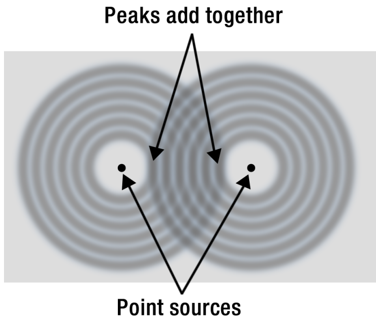


Figure 2.7: A wavefront is a surface of constant phase perpendicular to the wave direction.



(a) plane wave interference.



(b) Spherical wave interference.

Figure 2.8: Wave interference patterns.

Huygen's Principle

In the 17th Century, Christaan Huygen published a theory describing how a wavefront moves in space [11]. According to this principle, each point on a wavefront acts as a point source radiating spherical wavelets of a constant velocity. At any given point in time, the total wavefront may be formed by drawing a tangential line connecting the front of all the individual wavelets (see Figure 5.17). It therefore follows that an idealised wavefront of any shape can be constructed from a number of simple spherical sources - a concept that is highly significant in acoustics and is the basis behind many source modelling and spatial reproduction techniques.

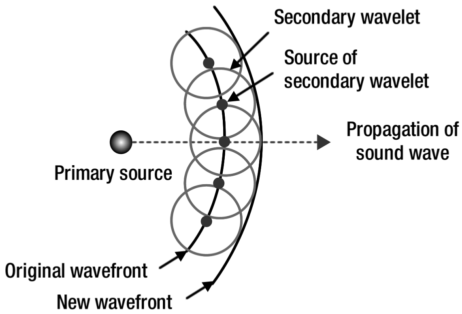
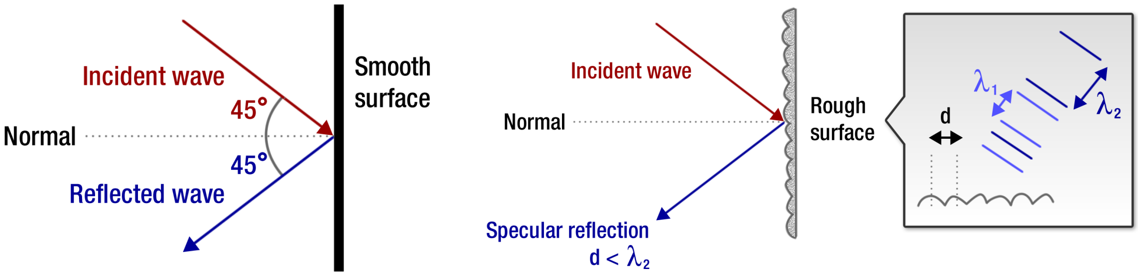


Figure 2.9: According to Huygen's Principle, a wavefront is considered the superposition of an infinite number of secondary wavelets.

While interaction between multiple sound sources is one possible cause of wave interference, in real world situations, wave interference is more often the result of interaction with obstacles and boundaries. The following wave phenomena are all linked to wave interference, being either a potential cause or an effect.

Reflection

When a sound wave encounters a boundary between two media with different material properties, only a proportion of the total incident energy is transmitted, while the rest is reflected back. The basic laws of reflection attributed to sound waves are the same as those attributed to light waves. If the surface is very smooth and any irregularities are minute relative to the wavelengths of incident sound, the reflections are said to be specular, whereas if it is very rough with geometrical irregularities of a comparable scale to that of the incident wavelengths, the reflections may be classed as diffuse [12] (see Figure 2.10). Specular reflection is modelled by drawing an invisible line at a perpendicular angle to the surface. This line is referred to as the surface normal. If a specular reflection occurs, the angle at which the incident wave meets the surface normal is equal to the angle between the reflected wave and the surface normal. When diffuse reflection occurs, the sound wave is reflected at many different angles. Diffuse reflection tends to occur more at higher frequencies where surface contours may be large relative to the wavelength - hence there is a lower chance of audible phase interference close to a boundary at high frequencies.



(a) Specular reflection

(b) Diffuse reflection

Figure 2.10: Interaction with a boundary leads to specular or diffuse reflection depending on the surface conditions. Here d denotes the distance between surface contours and λ denotes the wavelength of the incident sound.

Reflection Coefficient

The reflection coefficient R is defined as the ratio of the complex amplitude of the reflected pressure wave to that of the incident pressure wave.

$$R = |\bar{R}| \exp^{i\varphi} = \frac{A \exp_r^{i\varphi}}{A \exp_i^{i\varphi}} \tag{2.8}$$

where φ represents the phase change on reflection. Strictly speaking this makes the reflection coefficient a complex number, meaning both the amplitude and phase of the reflected wave may therefore differ from that of the incident wave. However, it is quite common in acoustics to express R as a real number between -1 and +1 denoting the ratio of reflected intensity over the incident intensity.

$$R = \frac{I_{reflected}}{I_{incident}} \tag{2.9}$$

In this representation the phase change is not explicitly defined, yet an approximate value is implied by considering certain ‘idealised’ scenarios. For example, within the range of values -1 to

+1, one can identify four general types of reflection:

1. $Z_2 >> Z_1$, $R => 1 \rightarrow$ Rigid boundary, i.e. most of the acoustic energy will be reflected without a change in phase.

2. $Z_2 << Z_1$, $R => -1 \rightarrow$ Soft or pressure release boundary, i.e. most of the acoustic energy is reflected with a 180 degree phase change.

3. $Z_2 = Z_1$, $R = 0 \rightarrow$ No Reflection.

4. $Z_2 \sim Z_1$, $-1 << R << 1 \rightarrow$ Some phase change.

Since the total incident energy is a sum of the reflected and transmitted sound energy, a related coefficient, the transmission coefficient (T), is obtained by subtracting the reflection coefficient from 1. For convenience, both R and T are often expressed in decibels (e.g. $R_{dB} = 10\log_{10}(R)$), or sometimes as percentages (multiplied by 100).

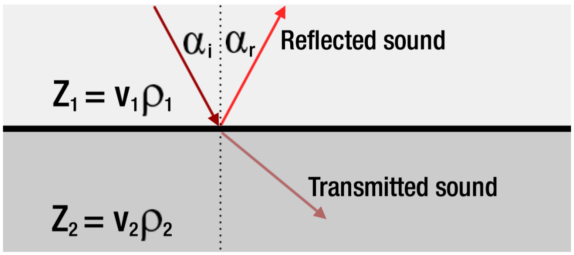


Figure 2.11: Reflection at a boundary between two semi-infinite lossless media.

From the above, it is clear that both R and T share a close relationship with another important property of the media known as the *characteristic impedance*, Z_0. The characteristic impedance, usually denoted Z_0, is an inherent property of a medium, defined as the ratio between pressure ($\bar{p}$) and complex amplitude of the fluid velocity ($\bar{v}$) in the direction of propagation when a plane wave is propagating through a substance [13].

$$Z_0 = \frac{\bar{p}}{\bar{v}} \tag{2.10}$$

In an ideal, lossless fluid, the characteristic impedance may be defined more simply as the product of the density and the speed of sound of the medium regardless of frequency or position.

$$Z_0 = \rho_0 c_0 \tag{2.11}$$

Since velocity and local pressure must be continuous across a boundary, the proportion of reflected sound energy relative to the transmitted sound energy may be derived if the characteristic impedances of the media either side of the boundary are known. Reflection is therefore said to occur as a result of a mismatch in the impedances between two media separated by a boundary (see

Figure 2.11). The full expression for the sound reflection coefficient based on the characteristic impedance is given by

$$R = \frac{(Z_2/Z_1) - \sqrt{1 - (n-1)\tan^2 \alpha_i}}{(Z_2/Z_1) + \sqrt{1 - (n-1)\tan^2 \alpha_i}} \tag{2.12}$$

where Z_1 and Z_2 are the characteristic impedances of the media, n is the ratio between their respective sound velocities squared ($n = \left(\frac{v_2}{v_1}\right)^2$), and α_i is the vertical angle of incidence. For normal incidence (i.e. $\alpha_i = 0$) this can be simplified to

$$R = \frac{(Z_2 - Z_1)}{(Z_2 + Z_1)} \tag{2.13}$$

It is important to distinguish between the characteristic impedance of a medium, and two similar parameters, the acoustic impedance Z and the specific acoustic impedance z. Acoustic impedance is defined as the ratio between sound pressure (p) and velocity (v) over a given surface area (S), or, volumetric flow, and indicates the amount of opposition to the flow of sound across a given surface at a given frequency (2.14). Z therefore varies strongly with frequency.

$$Z = \frac{p}{vS} \tag{2.14}$$

For infinite or semi-infinite media, it is common to see the acoustic impedance referred to as the specific acoustic impedance which can lead to confusion as these two quantities have different meanings. The specific acoustic impedance, $z(\omega)$, refers to the ratio between the sound pressure and particle velocity at a specific frequency at the connection point between two acoustic components and is defined by

$$z(\omega) = \frac{\bar{p}}{\bar{v}_{in}} \tag{2.15}$$

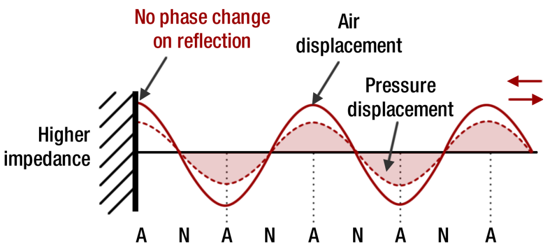
where $\bar{p}$ is the complex amplitude of a single frequency component of p at any point on a surface, and $\bar{v}_{in}$ is the corresponding complex amplitude of the fluid velocity component directed into the interface at the same point [14]. Ignoring the phase component, (2.15) may be simplified to

$$z = \frac{p}{v} \tag{2.16}$$

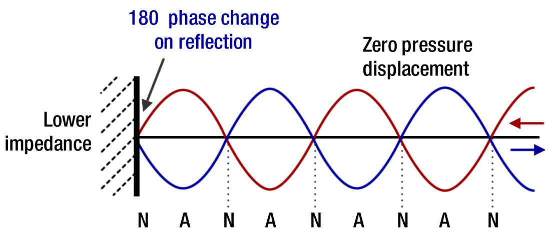
Specific acoustic impedance is measured in rayls (i.e. $Pa \cdot m$ or $N \cdot m^3$) unlike the acoustic impedance which is expressed in rayls per square metre (or pressure per velocity per unit area). On the assumption of normal incidence, z may be expressed as the ratio between the characteristic impedance of a substance and that of air [15]. While this general rule is valid for surfaces of infinite or semi-infinite area, when considering the propagation of sound waves through confined regions of space such as ducts or pipes, a transmission-line model which considers the acoustic impedances at the connection points between individual acoustic components would normally be needed.

The Boundary Effect

From (2.12) and (2.13) it is apparent that for two media with very different impedances, almost all of the sound energy is reflected back from the boundary. Furthermore, when the second medium (Z_2) has a much higher impedance than the first (Z_1), there is no phase reversal upon reflection. This can lead to an effect known as the *boundary effect*, which is the result of constructive interference between incoming and reflected sound waves. The boundary effect is more noticeable at lower frequencies where the wavelengths are large relative to any surface defects. Conversely, when a sound wave is incident upon a medium of a much lower impedance, as would be the case for a sound wave propagating through an open window for example, the consequence is a phase reversal (see Figure 2.12b). In reality, the difference in impedance between two media is rarely as extreme as the examples given in Figure 2.12b, meaning part of the wave is reflected from the boundary while the rest is transmitted across the boundary. The relationship between the characteristic impedance and the propagation speed given in (2.11) means the transmitted part of the wave will either speed up or slow down, depending whether the transition is from high density to low density or vice versa.



(a) A sound wave striking a surface with a much higher characteristic impedance will be reflected back with no phase change.



(b) A sound wave striking a surface with a much lower characteristic impedance will undergo a 180° phase change.

Absorption

When sound interacts with a boundary, the portion that is not reflected back from the interface is said to have been absorbed. Of that which has been absorbed, some will be transmitted and a small amount will be converted into thermal energy due to friction. Being a reciprocal of reflection, absorption is also dependent on the structural properties of the medium relative to the wavelength of the incoming sound. Porous materials absorb more sound than dense materials provided the cavities are of a comparable dimension to the wavelength (see Figure 2.13). While these materials are good at dampening sound - i.e. reducing the amount of reverberant sound energy in a space, they do not substantially reduce the transmission of sound.

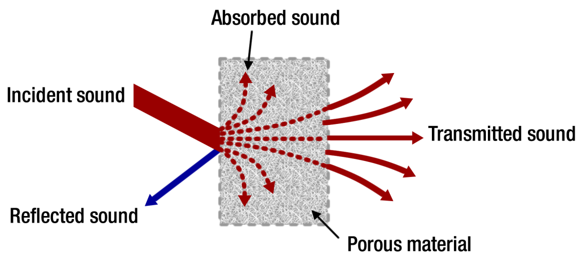


Figure 2.13: When sound enters a porous material, part of the sound is reflected, part is absorbed, and part is transmitted.

The absorption of a material at a given frequency is expressed in terms of an absorption coefficient, α, which is a number between 0 and 1. Values of α are normally given in octave or third octave bands and are relevant to a variety of acoustical assessment procedures, including estimation of reverberation time (room acoustics), and ground attenuation (atmospheric acoustics). Absorption coefficients are generally measured using the two-microphone method first described in ASTM-E1050 [16] and later in ISO 10534-2 [17]. This method involves placing a small sample of the test material at one end of a closed rigid tube and connecting a sound source to the other end. A multi-channel spectrum analyser is then used to obtain the transfer function between two microphones mounted inside the tube walls close to the sample. An alternative to this method that is more suitable for measuring the absorption of large objects and building elements is given in [18]. In this method the sample is placed in a reverberant room and α is derived from comparison of the reverberation time before and after insertion.

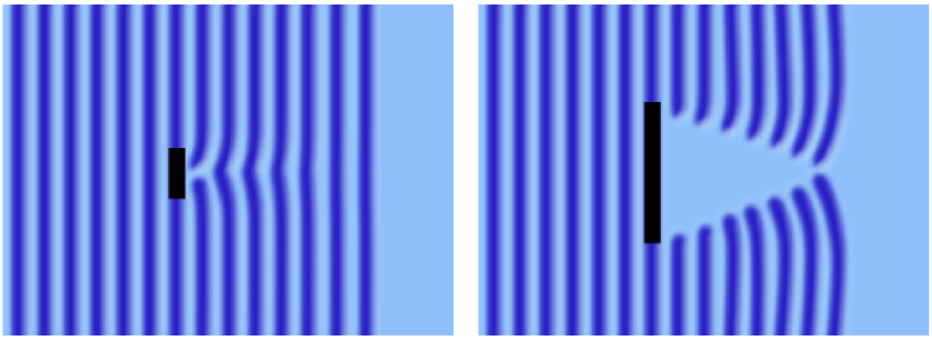
Diffraction

Diffraction is the term used to describe the 'bending' or 'spreading' of sound waves around the edge of an obstacle. The amount of spreading depends on the size of the object relative to the wavelength. For large wavelengths encountering small objects, diffraction is minimal, whereas a small wavelength encountering a large object will be diffracted much more (see Figure 2.14). Diffraction essentially describes a form of wave interference and, as such, there are no important physical differences between these two phenomena, other than usage [19].

Refraction and Dispersion

Refraction is the change in the direction of a wave due to a change in its speed. This can occur when sound encounters a boundary between two different media at an angle. The ratio between the original angle of incidence and the refracted wave is a constant which is equivalent to the ratio of velocities in the two media, or the inverse ratio of the refractive indices (the refraction index being a measure of how much the speed of the sound wave is reduced in the medium). The law that describes this relationship is known as *Snell's Law*.

$$\frac{\sin\theta_1}{\sin\theta_2} = \frac{v_1}{v_2} = \frac{n_1}{n_2} \tag{2.17}$$



(a) When a sound wave encounters an obstacle that is small in relation to its wavelength, it will pass around unperturbed.

(b) When a sound wave encounters an obstacle that is large in relation to its wavelength, it will bend around it.

Figure 2.14: Diffraction describes the tendency for waves to bend around obstacles in their path.

where θ_i is the grazing angle, α_i is the vertical angle (see Figure 2.15), v_i the velocity and n_i the refractive index of the medium.

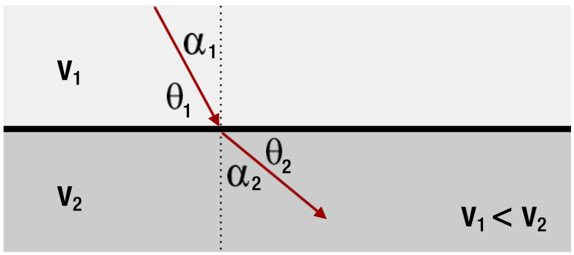
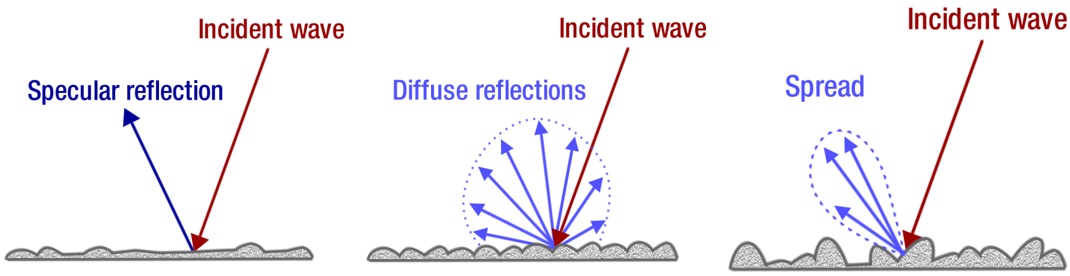


Figure 2.15: Refraction is the term used to describe the change in direction of a sound wave due to a change in its velocity.

Refraction may also be described as the observable effect of wave dispersion. Dispersion is a term used to describe a phenomenon whereby the speed of wave propagation through a medium varies with frequency. Therefore waves of different frequencies will be refracted by different amounts as they pass through mediums of different refractive indices. In electromagnetics, dispersion may be observed when light passes through a prism and is split up into its constituent frequencies. Media are said to be non-dispersive if sound can be transmitted without deformation and thus at a speed which is consistent for all frequencies. Air is effectively a non-dispersive medium over the audible frequency range, although strictly speaking it becomes dispersive at ultrasonic frequencies due to the presence of small amounts of carbon dioxide and other atmospheric effects [20, 21].

Scattering

Contrary to absorption, in the case of scattering there is no transformation of sound energy. Acoustic scattering simply describes the redistribution of sound energy in space by sending it in many different directions. In this sense it can be considered a property of diffuse reflections. The causes of scattering can be wave interaction with uneven surfaces (see Figure 2.16), local variations of homogeneity, or changes in acoustic impedance at boundaries. For example, as a wave passes from one medium to another of a different refractive index, the wave interacts with particles of matter in the medium and a portion of the energy is scattered in all directions. A scattering coefficient is an objective parameter used in acoustics to describe the tendency of a surface to scatter incident sound. It is defined as the ratio between the acoustic energy reflected in non-specular directions and the total reflected acoustic energy [22]. However, the scattering coefficient does not consider the distribution of the non-specular reflections, hence it is sometimes necessary to refer to a second parameter known as the *diffusion coefficient* to give a better estimation of the diffuseness of a sound field. The diffusion coefficient is defined as the ratio of uniformly distributed reflections to total reflected sound energy. The standard measurement procedures for estimation of the scattering and diffusion coefficients are outlined in ISO 17497-1 [23] and ISO 17497-2 [24] respectively.



(a) Specular reflection *(b)* Diffuse scattering *(c)* Non-diffuse scattering

Figure 2.16: Diffuse and non-diffuse scattering due to surface interaction.

2.3 Sensation and Perception of Sound

2.3.1 Physiology of Hearing

The ear consists of three main parts: the outer ear, middle ear and inner ear (see Figure 2.17). The outer ear consists of the fleshy external pinna and the outer ear canal. Their main function is to funnel sound toward the middle ear, as well as assisting in the localisation of sound.

The middle ear consists of the tympanic membrane (the eardum) and three very small bones - the incus, malleus and stapes - which are known collectively as the ossicles. The role of the middle ear is to transform sound into mechanical vibrations and conduct them to the cochlea. The cochlea is a bony shell-like structure comprised of three hollow chambers. Its purpose is to separate the sound into its constituent frequencies ready for transmission to the central nervous system. Sound enters the cochlea in the form of mechanical vibrations which are then propagated up its length from the base nearest the oval window to the apex.

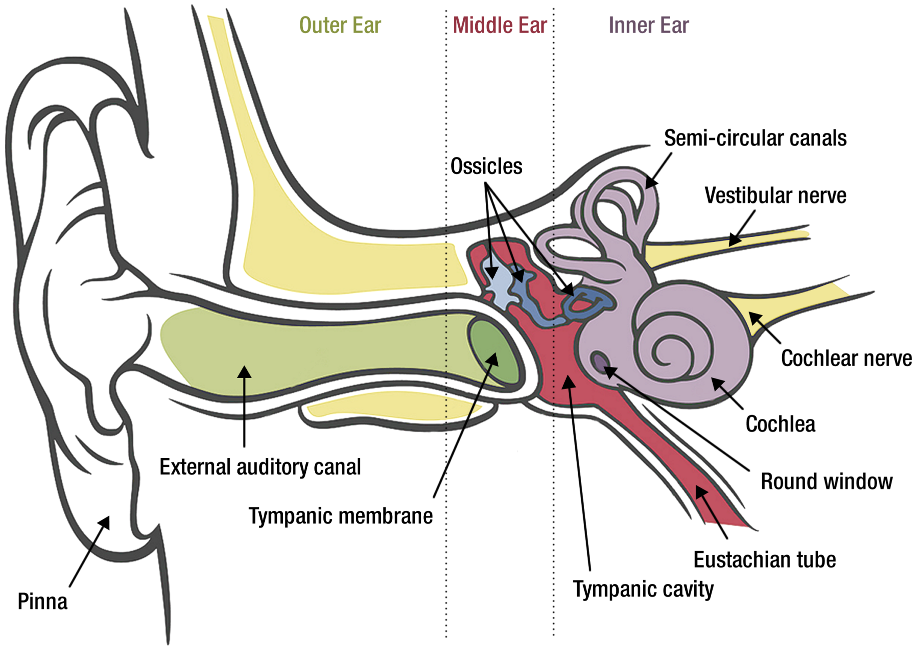


Figure 2.17: Section through the right ear.

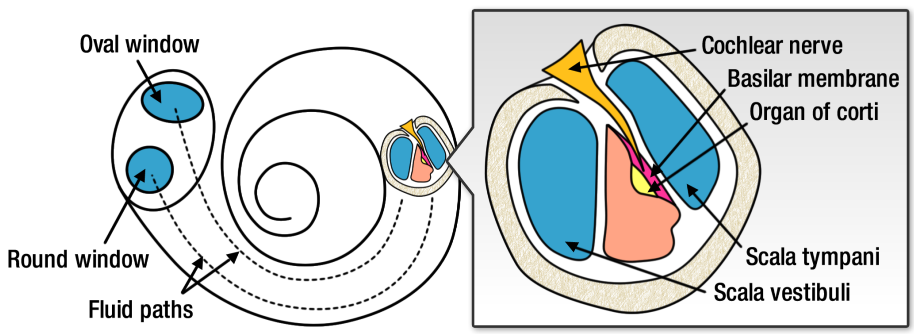


Figure 2.18: Section through the cochlea.

The separation of sound into frequency bands is accomplished by a flexible membrane known as the *basilar membrane* which separates two of the three cochlean chambers. The basilar membrane is tapered, so that it has a different resonant frequency at different points along its length (see Figure 2.19). High frequency sounds set the base into motion where it is narrowest, while lower frequency sounds stimulate the apex of the basilar membrane where it is widest and most flexible[25]. Upon the basilar membrane lies a cellular layer known as the *organ of corti* which is lined with motile sensory hair-like cells known as *stereovilli*. The function of the stereovilli is to transduce the vibrations of the basilar membrane into a code of neural impulses. The neural code is then transmitted to the cortex down the vestibular and cochlear nerves, known collectively as the *vestibulocochlear nerve*. The stereovilli are said to be motile due to their ability to move spontaneously and to change their length in response to sound - a factor which greatly increases the sensitivity of the ear and explains why damage to the outer cells leads to hearing loss.

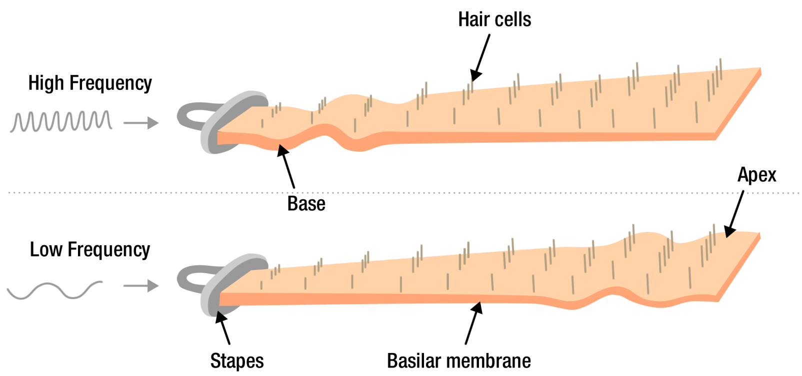


Figure 2.19: The basilar membrane. Due to its varying mass and stiffness, high frequencies stimulate the lower end nearest the oval window, while low frequencies stimulate the top end nearest the cochlea apex.

Hearing Loss

Hearing loss covers a broad spectrum of physiological problems, most of which are directly related to the functioning of one or more parts of the ear. Hearing loss can occur gradually - due to the ageing process, for example - or it may be the result of a sudden trauma, such as a loud explosion. Depending on the circumstances, it may be temporary or permanent, partial or complete. There are two main types of hearing loss, each of which is associated with a different part of the ear. Deafness or hearing loss due to defects in the middle ear is known as *conductive deafness*, and until recently this was the only type of hearing loss that could be treated. Conductive deafness can be caused by a ruptured eardrum (following an explosion, for example), or it can be due to damage to the ossicles. Conductive deafness is most often the result of a bacterial infection of the middle ear which often originates in the throat and spreads to the ear via the eustacian tube. If caught early enough, a bacterial infection can be treated with antibiotics and will not result in any permanent hearing loss. However, if left untreated, it can cause the ossicles to fuse due to a build up of scar tissue, and in children it can hinder speech development. Conductive hearing loss is readily treated with the insertion of a hearing aid inside or behind the pinna. The second type of hearing loss is called *nerve deafness* or *sensorineural hearing loss*, and is a result of physical damage to the nerve cells, vestibulocochlear nerve or the auditory cortex. Damage to the nerve cells in the cochlea may occur as a result of sudden or excessive exposure to very loud sounds. Often the effect is temporary, though the ability of the cells to recover declines with age. In recent years success in treating patients with severe cases of nerve deafness due to nerve cell disfunction has advanced considerably with the development of cochlear implants. Cochlear implants are electrical transmitter and receiver devices which are surgically implanted into the ear in order to stimulate the auditory nerve. While these devices are not yet capable of replicating the sensation of sound experienced by a normal hearing individual, they are able to provide useful auditory feedback of sufficient detail to facilitate speech recognition for those who are profoundly deaf. Nerve deafness due to a damaged auditory nerve or auditory cortex is not generally treatable. This damage may be present from birth, or in rare cases it can occur following a stroke or other illness, or as a side-effect of some types of medication.

2.3.2 Psychoacoustics

Limits of Perception

Dynamic Range

A person with good hearing is sensitive to changes in pressure ranging from approximately 20 μPa ($20 \times 10^{-}5Pa$) to $20Pa$. In decibels this equates to sounds in the range of 10 dB (the threshold of audibility), to around 130 dB (the threshold of pain). Very loud sounds can cause hearing loss and, in extreme cases, total deafness, although the onset of hearing damage varies between individuals and is related to both the sound pressure level and the length of exposure. For example, sounds in excess of 130 dB can cause permanent hearing loss after just a few seconds or milliseconds, while exposure to sounds of around 85 dB can cause hearing loss if sustained over a much longer period of time (i.e. for several hours on a recurring basis).

Table 2.1: Common sounds and their typical sound pressure levels. Approximations based on values given in [26] and [27].

L_p (dB)	Sound
160	12 gauge shotgun blast
	Jet plane taking off at engine at 30m
140	Loud club/rock concert
	Threshold of pain
120	Jet plane taking off at 500m
	Power tool, symphony orchestra
100	Lawn mower, tractor
	Car horn at 5m
80	Road traffic noise in a city
	Car at 20m, typical ambient noise level in an urban street
60	Laughter
	Conversation at 1m, average office noise
40	Typical ambient noise level in a sub-urban/residential area
	Typical ambient noise level in a rural area
20	Quiet room at night, whispering, watch ticking
	Threshold of good hearing
0	Threshold of very good (youthful) hearing

Frequency

The audible range of frequencies is broadly dictated by the physiology of the ear, but is also influenced to a variable degree by factors such as the age and physical health of the individual. Generally, the range of frequencies extends from around 20Hz to 18kHz [28] in a person with good hearing which equates to wavelengths of less than 2cm to over 17m. However, the ear is not equally sensitive to all frequencies. It is generally most sensitive to mid-frequency sounds of between 500-2500Hz which also happens to coincide with the range of human speech. The variation in hearing sensitivity with frequency can be measured in laboratory tests by asking subjects to adjust the loudness of one pure tone against a reference tone of a different frequency until they are perceived to be of equal loudness. The process is repeated for different frequency tones until a pressure-frequency curve has been derived. Fletcher and Munson [29] were the first to measure a set of equal-loudness contours (see Figure 2.20), although these have since been updated. The current definitive set of equal-loudness contours are defined in ISO 226:2003 [30].

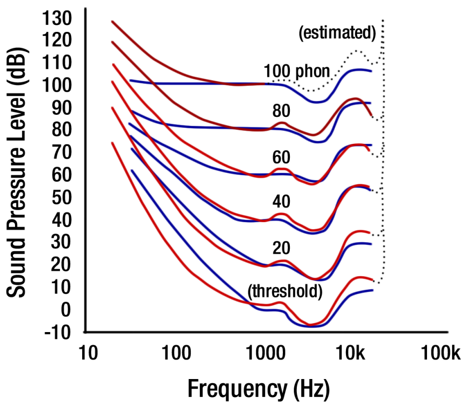


Figure 2.20: Equal loudness contours from ISO 226:2003 [30] (red) shown against the original Fletcher-Munson curves [29] (blue) for comparison.

When speaking generally about the behaviour and perception of sound, it is often convenient to group frequencies into regions based on the changing sensitivity of the ear. From here on, where the author refers to low frequency sound, this alludes to sounds within the range of 20 Hz - 1 kHz. Mid-frequency sounds are considered to be those frequencies at which the ear is most sensitive. These start at 1kHz and extend to 5kHz, with a maximum sensitivity around 2-5kHz. Where the author makes reference to high frequencies, this refers to sounds above 5kHz.

Temporal Resolution

In psychoacoustics, temporal resolution generally refers to the auditory system's ability to follow rapid changes in the envelope of sound [31]. Deficits in temporal resolution are associated with deficits in speech understanding, and may also be indicative of other neuralogical disorders. There is no definitive way to measure temporal resolution since the temporal resolving abilities of the auditory system are manifold, and thus results will vary depending on the nature of the test and stimuli that are used. A popular procedure that is particularly useful in assessing patients with cortical and brain stem lesions is the Gaps-In-Noise (GIN) test whereby temporal resolution is measured in terms of a *gap detection threshold* - i.e. the duration of a just-detectable interruption in a sound [32]. For adults and infants with normal hearing this is thought to be within the range of 2-10ms depending on the frequency distribution and bandwidth of the stimuli (for example, in [32]the stimuli was low-pass filtered white noise with a filter cut-off around 7kHz).

A slightly different neurological interpretation of temporal resolution is one that ignores the cognitive aspect of temporal perception, but considers the auditory system's ability to follow rapid changes in a sound over time in terms of the highest frequencies that it is capable of responding to - whether consciously or otherwise. The wavelength of an 18kHz tone is just under 2cm, which implies that the temporal resolution of hearing is in the region of 9 microseconds ($\frac{1}{2\pi 18000} = 9\mu s$). However, it has long been suspected that sounds outside the audible range of frequencies have the capacity to alter certain aspects of auditory perception, from which it may be inferred that the temporal threshold is potentially higher than originally suspected. Investigation of this hypothesis has lead to the conclusion that neural processing beyond the cochlea can permit extraction of temporal information at time scales shorter than $\frac{1}{2\pi f_{max}}$ [33].

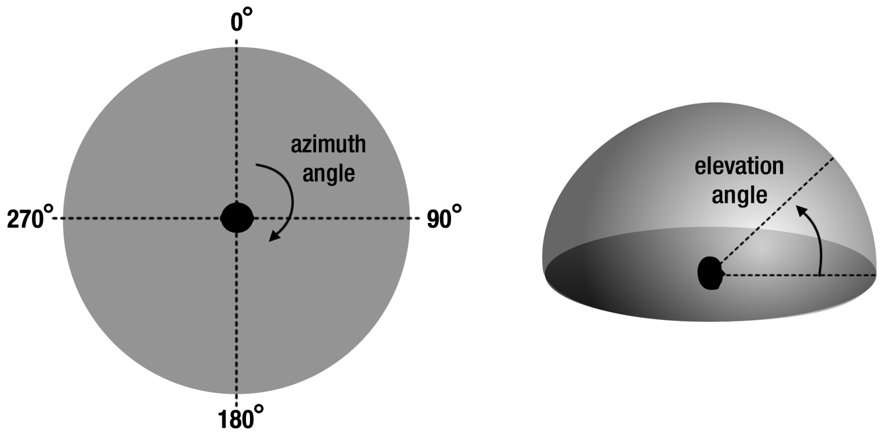
2.3.3 Binaural and Spatial Hearing

The perception of sound with two ears is referred to as binaural hearing, and it offers a number of advantages over monaural hearing, many of which are related to the spatial perception of sound. Spatial hearing is an extremely important aspect of auditory perception covering various different mechanisms, including the ability to localise sounds in space, to perceive certain spatial characteristics of both the sources and their surrounding environment and to distinguish between different sources of sound. It also encompasses the ability of the brain to integrate two separate signals to produce a single, coherent auditory 'scene'. Without spatial hearing it becomes much more difficult to extrapolate meaningful information from the complex soundscapes encountered in everyday life - information which, at a much earlier point in human history, would have been all the more critical.

Spatial hearing is largely attributed to the subtle timing and level differences between signals reaching either ear, and spectral variations caused by the pinna, head and torso [34], as well as the processes by which the auditory cortex interprets these 'inter-aural cues' to derive information about the sound sources and surrounding space. Of course, there are other advantages to binaural hearing that are not directly related to spatial hearing. For example, it has been proven that the binaural summation of low level sounds enhances the threshold of audibility by some 3 dB [35] over that of monaural hearing. For sounds above threshold level, the difference in perceived loudness is even more pronounced. Fletcher and Munsun reported that a sound of a given SPL will seem twice as loud binaurally as monaurally [29], although it has since been shown that the difference in sensitivity between monaural and binaural hearing increases as the amplitude of the stimulus increases (e.g. [35]). It has also been shown that binaural listening leads to an an improvement in differential sensitivity for both intensity and frequency compared with monaural hearing [36]. Differential sensitivity refers to the ability to detect a difference between two stimuli, usually referred to as the difference threshold or difference limen (DL).

Source Localisation

Source localisation is a critical aspect of spatial hearing, both in terms of identifying the location or origin of a detected sound in direction and distance, and as a contributing factor in Auditory Scene Analysis (ASA). The ability to locate a sound in 3-D space may be attributed to subtle timing, intensity and spectral differences in signals arriving at either ear, broadly referred to as inter-aural cues. To enable accurate localisation of a sound in 3-D space, these inter-aural cues must convey a substantial amount of information pertaining to the specific source, including its horizontal or azimuth angle, the vertical angle or elevation, the distance (static sources) and velocity (moving sources) relative to a listener. A number of factors contribute to this information being ‘encoded’ into the signals. These include the difference in arrival times between the ears, relative amplitude differences due to the shadowing of the head, and by the asymmetrical spectral reflections from various parts of the body, including torso, shoulders, and pinnae [37].

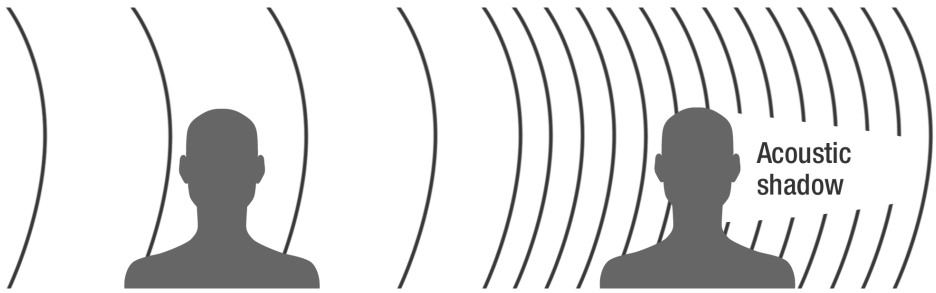


(a) Top down view *(b)* Lateral view

Figure 2.21: Localisation in 3-D space can be described in terms of the azimuth or horizontal angle, the elevation or vertical angle, and the distance or velocity of the source [37].

Azimuth

With respect to the localisation sound sources on the horizontal plane, reliance on inter-aural timing differences (ITD) and inter-aural level differences (ILD) varies according to the frequency of the sound. For example, localisation of high frequency sounds (above 1600 Hz) is more dependent on ILDs due to the shadowing effect of the head, whereas localisation of lower frequencies is more reliant on ITDs in the form of phase delays. This is due to the fact that as the wavelength approaches the diameter of the head, the phase difference between ears becomes ambiguous tending to lead to localisation errors, whereas longer wavelengths will tend to bend around the head resulting in an unambiguous phase difference and no acoustic shadow (see Figure 2.22). However, for very low frequencies (typically < 80 Hz) where sounds exhibit a proportionally very small phase delay between ears relative to the total wavelength, it may be very difficult or impossible to discern the direction of the sound [38].



(a) Large wavelengths are diffracted around the head, hence localisation is more dependent on inter-aural phase differences.

(b) Small wavelengths are shadowed by the head and torso, hence localisation is more reliant on inter-aural level differences.

Figure 2.22: Source localisation due to inter-aural differences is dependent on the wavelength of sound relative to the head diameter.

In addition to inter-aural time and level differences, localisation on the horizontal plane is also aided by the spectral transformation of signals due to the diffraction, absorption and reflection properties of the outer ears (pinnae), head and torso. These particular filtering effects are characterised by complex frequency response functions known as *Head-Related Transfer Functions* (HRTFs). The corresponding impulse response function (IRF) or time-domain representation of an HFTF is referred to as a Head-Related Impulse Response (HRIR) [39]. HRTFs are particularly important for ensuring accurate localisation of near-field sources where ILDs and ITDs may be minimal, and generally contribute to the extracranialisation of sound reproduced through headphones - i.e. the sensation of sound being 'all around' the listener, as opposed to inside the head.

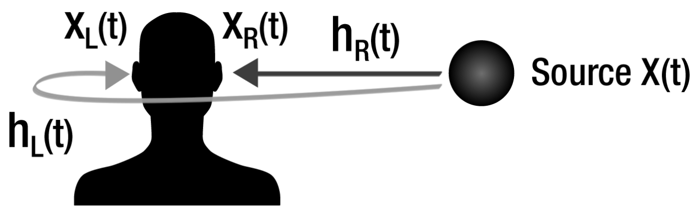
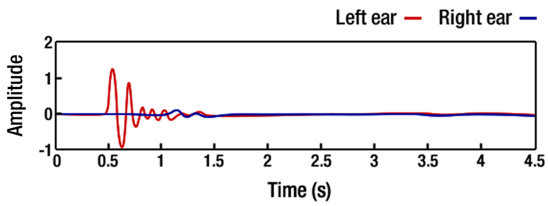
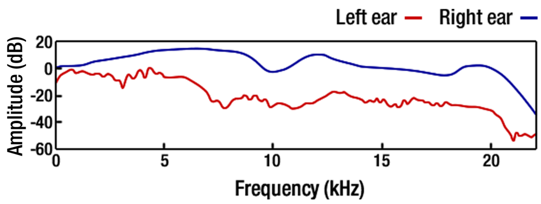


Figure 2.23: A pair of Head-Related Impulse Responses (HRIRs), $h_L(t)$ and $h_R(t)$, can be measured experimentally. The HRTFs may then be derived from the HRIRs via the process of Fourier transformation.



(a) HRIR



(b) HRTF

Figure 2.24: HRIR and HRTF for the left and right ear when source is directly in front of the right ear and at a distance of 1m from the centre of the head. Reproduced from [40].

Elevation

Localisation on the vertical plane is substantially weaker, by comparison, than on the horizontal plane. This is due to the horizontal symmetry of the ears which leaves an unresolved region known as the *cone of confusion*, i.e., a cone extending outward from each ear and centered on the lateral axis connecting the two ears of the listener [41]. Monaural pinna cues are therefore primarily responsible for vertical localisation, as well as being important contributors to front-back differentiation [42], though they are only effective at frequencies above 3 kHz where the wavelength becomes comparable to the pinna size [43]. At lower frequencies, HRTF features are primarily due to the torso reflection and head diffraction, both of which are thought to be involved in vertical localisation, albeit to a lesser extent due to the comparative weakness of these reflections [40]. Other mechanisms for vertical localisation include the analysis of ground reflections, head tilting, and visual cues.

Distance

The perception of distance is primarily affected by the sound level, spectral and reverberant characteristics of the sound, all of which are dependent to some extent on prior knowledge of both the source and its surrounding environment. Estimation of distance is complicated further by various subjective factors, such as the spatial acuity, expectations and past experience of the listener. Consequently, the margin for error is very high - in the region of 20% according to one source [44].

Sound level cues may be a useful indicator of absolute distance if the source is moving, or there are multiple representations of the same source, yet in the case of a single static source, sound level cues are unlikely to prove useful without prior knowledge about the source and the environment.The ratio of direct-to-reverberant energy can also serve as a useful indicator of distance, as sounds that are close to a listener will generally contain more of the direct sound than a sound that is further away. Again, the usefulness of this information is affected to a certain degree by other subjective factors. Lastly, the change of spectral content due to the selected attenuation of frequencies by absorption is thought to influence the perception of distance to some extent [45], assuming there is some knowledge of the environment.

Environment

In most real-world situations, the sound arriving at a listener's ears is a combination of both direct and reverberant (reflected) energy due to the presence of objects and reflective surfaces which act to alter the distribution of sound energy in space. The reverberant sound can impede localisation in both the horizontal and vertical planes, particularly if the reflected energy is very strong and long-lasting relative to the direct sound [41].

Visual cues are another aspect of the environment that can be important contributors to the localisation of sound, particularly with respect to front-back confusion. One instinctively looks for the source of sound in the place where one would expect it to be, hence there may be some hope of locating sources that could otherwise be extremely difficult to place - a bird tweeting or woodpecker boring, for example. They are also responsible for conveying a large amount of information about the reflective/absorptive properties of objects and surfaces, guiding expectation and thus assisting in both distance and directional localisation. In the absence of visual cues, a group of individuals might each build very different internal spatial represention s of sound objects in a relatively non-reverberant sound field, such as a park or other open outdoor space.

Auditory Scene Analysis

Another important aspect of spatial hearing relates to ASA. ASA examines the processes by which the brain is able to separate a complex auditory environment into its constituent sounds to create an image that is perceptually meaningful. It is therefore predominantly focussed on the neurological aspects of spatial hearing, although the physical aspects are of course also highly significant. A number of acoustical parameters are thought to contribute to ASA, including spectral profile, spectral separation, harmonicity, spatial separation, temporal onsets and offsets, temporal modulation and temporal separation [46]. A model proposed by Bregman suggests ASA involves the grouping of sounds into streams based on certain criteria or grouping principles [47]. Within this model, there are considered to be two different groups of cognitive processes involved in auditory streaming. The first of these are primitive or innate processes that occur unconsciously, while the second are referred to as schema-based processes and are said to involve cognitive effort and learned information [47]. Of the primitive processes, sounds which are similar in one or more attributes such as frequency, intensity, harmonicity etc are thought to be grouped together, which could help to explain why it it is often difficult to discern a single voice amongst many - an effect often referred to as the 'cocktail party effect'. Bregman's theory would seem to be supported by the psychoacoustic phenomenon known as *auditory masking*, which might then be viewed as a side-effect of auditory streaming processes.

Auditory Masking

Sounds encountered in real world listening situations are rarely heard in isolation, yet it can sometimes be very difficult or impossible to distinguish between separate sounds due to auditory masking. Masking is the tendency for one sound (the masker) to conceal another (the maskee), particularly when the sounds in question have some similar characteristics. For example, one might struggle to communicate with someone in a next-door room whilst the television was on, yet have no difficulty hearing the telephone ring. Essentially the listener experiences a threshold shift in frequency region of the maskee, due to the presence of the masker. The amount of masking can be expressed as the difference between the unmasked threshold and the masked threshold. For example, if, with the TV off, the *unmasked threshold* for hearing the person next door is 30 dB, but after turning it on the person next door has to speak twice as loudly to be detected (i.e. a *masked threshold* of 40 dB), then the amount of masking is 40-30 dB, or 10 dB [34]. How much a sound is masked is largely dependent on its frequency and intensity in relation to the masker.

Frequency masking is a well-known psychoacoustic phenomenon thought to have first been observed by Mayer in 1894 [48]. Mayer noted that low frequency sounds are better at masking higher frequency sounds, whereas higher frequencies are much poorer maskers of low frequencies. The first masking patterns were obtained by Ehmer [49] (see Figure 2.25), from which one observes the tendency for masking to be greatest at frequencies in the vicinity of the masker, while beyond this central spike, the masking tapers with frequency. It is also possible to observe from these curves the tendency for masking to increase with intensity.

Temporal masking is the characteristic of the auditory system where sounds are hidden due to maskers which have just disappeared or maskers which are about to appear [50]. Post-masking is when a quiet sound which immediately follows a loud sound is not heard, and can occur up to 200 ms after the offset of the loud sound. Pre-masking is when a quiet sound immediately preceding a loud sound is not heard. The time span over which this can occur is relatively short by comparison with post-masking - up to around 50 ms before the onset of the loud sound (see Figure 2.26).

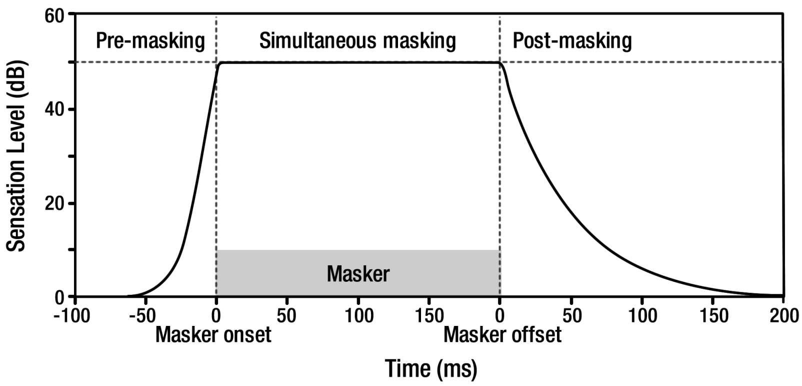
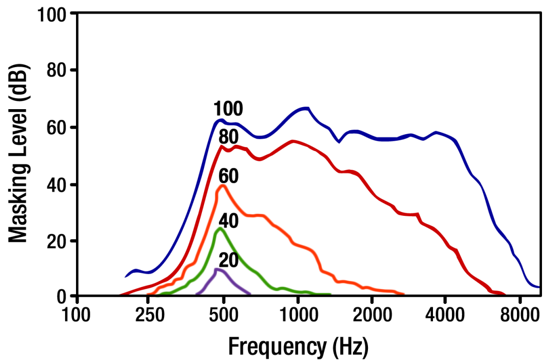
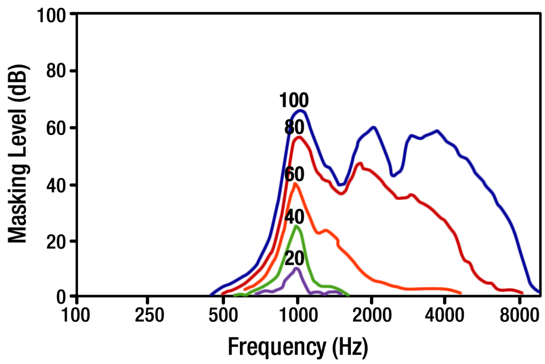


Figure 2.26: Temporal masking describes the effect whereby quiet sounds are not perceived after or preceding a loud sound.



(a) Average masking patterns for 500 Hz



(b) Average masking patterns for 1000 Hz

Figure 2.25: Masking patterns at 500 Hz and 1000 Hz based on three listeners. The sensation level of the masking tone is attached to each curve. Reproduced from [49].

Binaural Integration

Of course, as well as being able to separate sources, the auditory system must also have measures in place to prevent sound that is received binaurally from being perceived as two distinct channels of sound. The ability of the brain to fuse together sounds incident on two ears into one coherent image is often referred to as *binaural fusion*. In audiology, binaural fusion is one of several objective measures that are used in the diagnosis of various hearing and related disorders. In the binaural fusion test mono-syllables are separated into high and low frequency pass-bands, one of which is fed to the left ear and the other to the right. In normal hearing individuals, word recognition when presented with either signal in isolation is generally poor but much higher when presented with both signals together [51].

A well-known psychoacoustic phenomenon that can be attributed to binaural integration is that of binaural beats, an effect that is observed when two tones of a similar frequency are presented separately to each ear via stereo headphones. Rather than hearing two distinct tones, the sounds are perceived as a single pulsating tone. Binaural beats will only occur when the two sounds are below 1000 Hz and differ in frequency by 30 Hz or less, otherwise they are perceived as two separate tones. While they are one of the oldest observed psychoacoustic phenomena, binaural beats are still of interest to neurologists and audiologists today seeking to further their understanding of auditory processes, and to exploit these capabilities in emerging technologies. For example, a recent study proposes that binaural beat noise may be able to induce a sensation of auditory motion [52], which may have implications in terms of virtual audio reality. Various similar studies have been conducted which aim to elicit a sense of auditory motion through signal processing techniques, hence the more that is understood about auditory processing and psychoacoustics, the more successful one might be at 'fooling' the brain into perceiving 'sound objects in space' as opposed to flat, dimensionless sound.

2.4 Atmospheric Acoustics

Atmospheric acoustics deals with the transmission of sound in open air, where the inherent structural discontinuities of the medium and other atmospheric phenomena affect how sound is propagated. This distinguishes it from indoor acoustics, where one is typically dealing with the propagation of sound over much shorter distances with very little atmospheric variation. Furthermore, due to the protection afforded by the building or enclosure, sound in indoor spaces is largely unaffected by external meteorological phenomena if one ignores the obvious yet indirect link with background noise levels. Propagation of sound in open air is primarily affected by the following factors:

- Geometrical spreading
- Obstructions
- Terrain type and contour
- Meteorological conditions
- Atmospheric absorption

2.4.1 Geometrical Spreading

Divergence Theorem

As sound is radiated away from a source, the area over which the original sound energy is distributed becomes larger, so that the sound intensity decreases proportionally with distance. The conservation of acoustic energy is thus described by the divergence theorem, also known as *Gauss's theorem*, which states that the outward flux of a vector field through a closed surface is equal to the volume integral of the divergence over the region inside the surface [53]. In acoustics this translates as the energy density or energy per unit volume resulting from a wave disturbance. The divergence theorem can be applied in any number of dimensions, with geometrical spreading over 1-D, 2-D and 3-D surfaces corresponding to plane wave, cylindrical wave, and spherical wave propagations respectively. For a plane wave, the energy density may be expressed by the following:

$$\omega = \frac{I}{\rho c^2} p^2 \tag{2.18}$$

where ω is the energy density, I is the intensity or energy flux vector, ρ, p and c are the density of the medium, sound pressure and speed of sound respectively.

The Inverse Square Law

The inverse square law is a general form of Gauss's law, stating that the vector intensity, or power-per-unit-area, at any given point on the surface is inversely proportional to the square of the distance from the source. In general form, it is written as

$$I \propto \frac{1}{d^2} \tag{2.19}$$

where I is the vector intensity and d is the distance from the source.

Source Geometry

The attenuation of sound due to geometrical spreading is primarily dictated by the source geometry and type, and the presence of any nearby reflective surfaces. Source characterisation is approximated according to the distance and relative dimensions of the source and an observation point. Due to this dependency, the overall sound field is generally separated into distinct regions within each of which the sound from a given source may be treated differently. The region of the sound field nearest to a source tends to be the most difficult to predict, being highly dependent on specific characteristics of the source. Beyond this *near-field* region, there is a roughly linear operational range wherein prediction is generally much easier, assuming interference from surface and/or atmospheric discontinuities is minimal. This relatively stable region is also referred to as the *free-field*, or *semi free-field* if it is assumed that sound is being radiated across an infinite acoustically hard surface. In either case, the attenuation shares a linear relationship with the distance travelled. The free-field may, in theory, extend infinitely. In practice, there is generally a point at which the linear relationship between distance and sound intensity fails, due to the presence of reverberant sound energy or residual noise in the system - residual noise referring to whatever noise may be present in a system in the absence of any applied input. This region is often referred to as the reverberant sound field, although the nomenclature is more relevant to bounded systems where the levels of reverberant energy may rapidly exceed that of the direct sound.

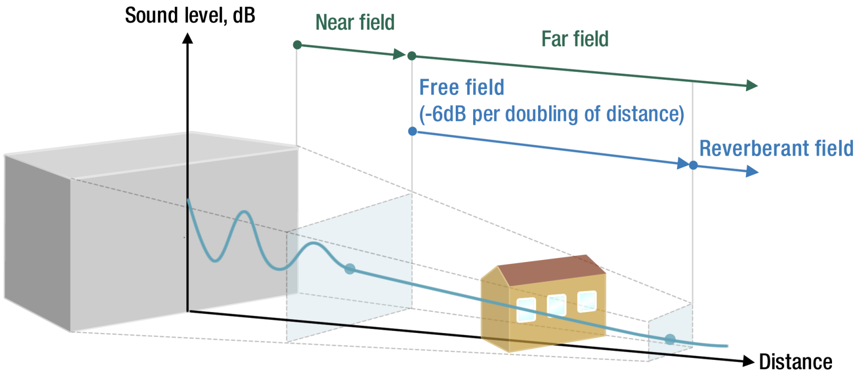


Figure 2.27: Regions of a sound field.

Point Sources

A sound which is assumed to originate from an infinitely small point in three-dimensional space is radiated outward over a spherical closed surface in accordance with the inverse-square law. The form of the inverse square law that is applied to point sources states that the vector intensity, or power per unit area, at any given point on the surface is inversely proportional to the square of the radial distance from the point of origin (see Figure 2.28).

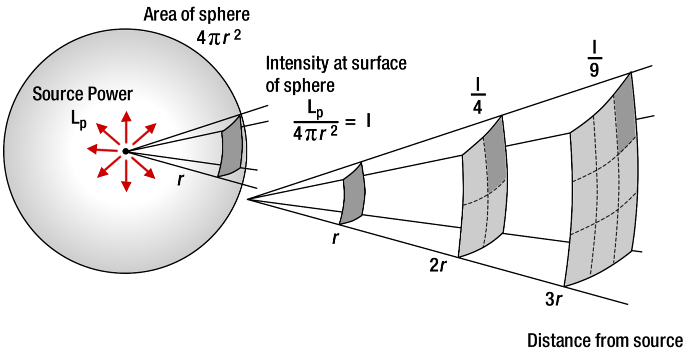


Figure 2.28: In an ideal fluid, sound is radiated from a point source equally in all directions. Sound levels are therefore spread evenly over spherical surfaces and the resulting loss of intensity with distance follows the inverse square law.

According to the inverse square law, the intensity of a sound drops off rapidly as distance increases. In logarithmic form, the reduction in sound pressure level L_p with radial distance r from a point source with the rated sound power level L_w is expressed by the following relationship:

$$L_p = L_w - 20\log_{10} r - 11\ dB \tag{2.20}$$

This equates to a reduction of 6 dB per doubling of distance, assuming the source is omnidirectional and free to radiate in all directions without interference from nearby reflecting surfaces. In the event that a point source does happen to be close to one or more flat and acoustically hard surfaces, the surface area over which the sound energy may spread is effectively restricted according to the number of surfaces present. The situation thus described is essentially equivalent to one in which the source has inherent directivity. In either case, corrections should be made which take into account the reduced surface area of the propagation sphere. In environmental noise prediction, this takes the form of a directivity index, DI, which is incorporated into (2.20) as follows:,

$$L_p = L_w + DI - 20\log_{10} r - 11\ dB \tag{2.21}$$

The directivity index is calculated according to (2.22) in which Q is the directivity factor and defined as the ratio of the intensity at a certain distance and angle from the source (I_θ) with that which would be measured at the same distance under perfect free-field conditions (I). The values of Q for some ideal geometry and assuming perfect reflection are illustrated in Figure 2.29.

$$DI = 10\log_{10}\left(\frac{I_\theta}{I}\right) = 10\log_{10}(Q)\ dB \tag{2.22}$$

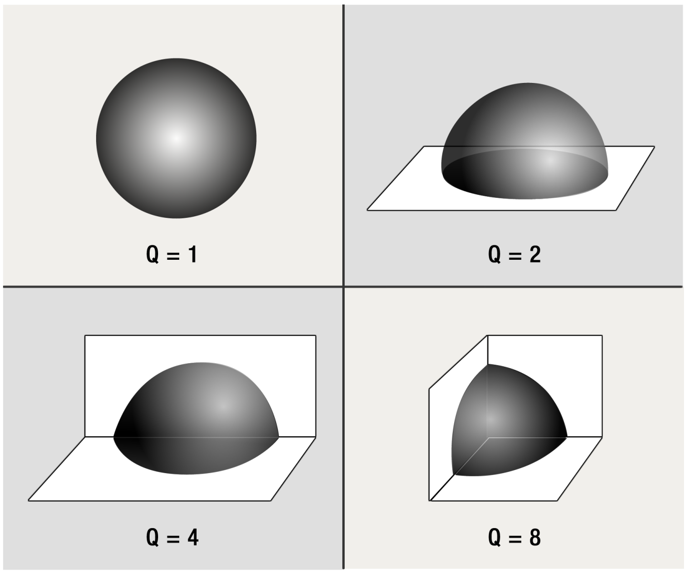


Figure 2.29: In environmental acoustics, the associated directivity of the source is approximated by a directivity factor Q.

Line Sources

An ideal line source is considered to be an infinitely long array of point sources, to each of which the inverse square law applies. The net contribution from all sources results in a cylindrical radiation pattern (see Figure 2.30). Thus the sound energy in any perpendicular direction is inversely proportional to the increasing circumference of the cylinder, resulting in a 3dB reduction in relative sound intensity per doubling of distance (2.23).

$$L_p = L_{w(line)} - 10\log_{10}\left(\frac{I_1}{I_0}\right)\ dB \tag{2.23}$$

where I_1 and I_0 are the relative sound intensity levels at the radial distances r_1 and r_0 respectively and are related according to (2.24).

$$I_1 = I_0\left(\frac{r_0}{r_1}\right) \tag{2.24}$$

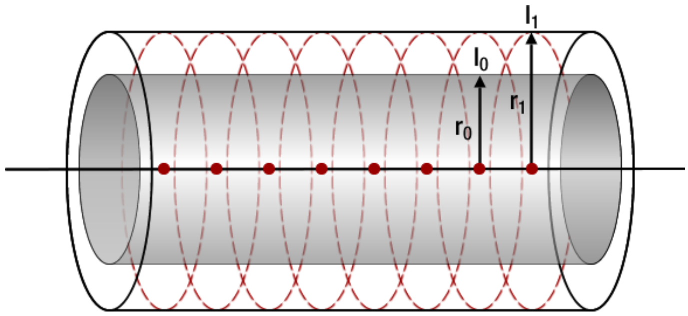


Figure 2.30: Sound radiated from an infinite line of point sources results in cylindrical spreading and equates to a 3dB reduction in sound intensity per doubling of distance.

Road traffic noise can be modelled by a finite line of incoherent point sources on an acoustically hard surface [54]. The sound pressure level at an observation point halfway along a finite line source of length l, and at a certain distance, d, from the source can be approximated according to the following [13]:

$$L_p = L_w - 10\log_{10} d - 8 + 10\log_{10}\left[2\tan^{-1}\left(\frac{l}{2d}\right)\right]\ dB \tag{2.25}$$

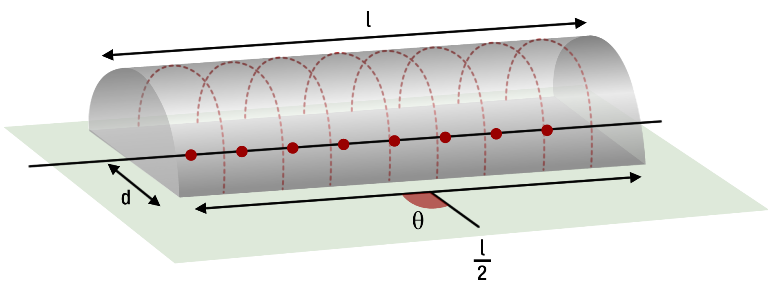


Figure 2.31: Multiple uncorrelated sources along an acoustically hard surface may be approximated as a finite line source.

From Figure 2.31 it is apparent that the angle θ subtended by the observation point and the source edge should be within a certain range ($0° << \theta << 90°$) for the above to yield a reliable approximation. For very small values of θ, a point source model should be used. For very large values of θ, an infinite line or plane source model may be more appropriate, although, in the case of the latter, this also means taking into account the effective height of the source in relation to the observation point.

Plane Sources

If an infinite source array is extended into two dimensions, the radiation pattern ressembles that of a plane wave. Assuming there are no other losses in the model, the effective sound pressure level produced by an ideal plane source is therefore independent of distance from the source. Although plane sources are rarely encountered in real world scenarios, it may be feasible to assume plane wave propagation from a flat surface which is vibrating at a single frequency and which is very large in relation to a nearby observer. It may also be reasonable to combine source models in some situations. For example, in environmental noise prediction, when predicting the increased SPL at a noise sensitive receptor (such as a residential dwelling) due to noise transmitted by a nearby factory facade (see Figure 2.32), it may be appropriate to make independent predictions for losses in the near and far field regions of the sound field.

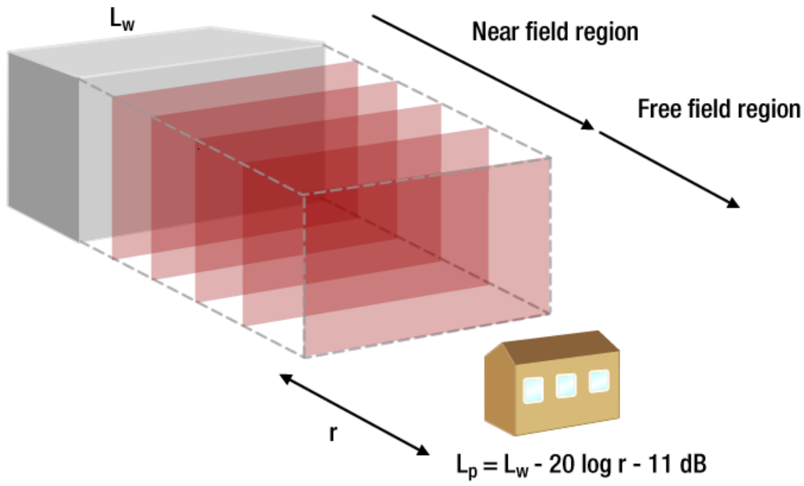


Figure 2.32: Plane waves radiated in a lossless medium from an ideal plane source may theoretically propagate indefinitely without any loss of intensity. Some sources may approximate ideal plane wave sources within the near field region, while in the far field they approximate point sources.

2.4.2 Obstructions

Barriers

Barriers are objects that prevent the transmission of sound, either by reflecting the incident sound, or by absorbing it internally. Generally speaking, an obstacle or barrier is only acoustically relevant in outdoor sound propagation if the sightline between source and receiver is interrupted. The simplest barriers are typically rigid walls or panels with high mass and stiffness that reflect sound. Ignoring the effects of diffraction, the frequency dependent transmission loss of such structures can typically be defined in terms of 4 distinct regions (see Figure 2.33). The first is a stiffness controlled region, whereby the transmission loss of the barrier is primarily affected by the stiffness or inertia of the material. This increases by approximately 6 dB per doubling of stiffness. The second region is a resonant region, whereby the attenuation is difficult to predict due to natural resonant frequencies of the barrier. The lowest resonant frequency can be estimated from:

$$f_r = 0.454 h v_b [\frac{1}{x^2} + \frac{1}{y^2}] \tag{2.26}$$

in which f_r is the resonance frequency in Hz, h is the thickness of the barrier in metres, v_b is the velocity of longitudinal waves in the barrier material in m/s, and x and y are the dimensions of the barrier in metres [55]. Beyond the resonant frequency region, the attenuation is easier to predict and follows a linear progression, increasing at a rate of approximately 6 dB per octave. There is then a coincidence region, the onset of which may be marked by a pronounced narrow dip in attenuation which coincides with the point at which the frequency of the sound matches the natural frequency of the barrier. The coincidence effect is caused by bending waves which can coincide with the incident sound waves resulting in a marked increase in sound transmission. The lowest frequency at which coincidence occurs is known as the *critical frequency*, f_c, and can be calculated from:

$$f_c = \frac{c^2}{1.8 h c_b} \tag{2.27}$$

where f_c is the critical frequency in Hz, c is the velocity of sound in air in m/s, c_b is the velocity of longitudinal waves in the barrier material in m/s and h is the thickness of the panel in metres [56]. While the sound insulation performance of building elements can be improved with damping, this is generally less applicable in the case of noise barriers which tend to have a much higher mass and thickness and are limited primarily by diffraction and other edge effects.

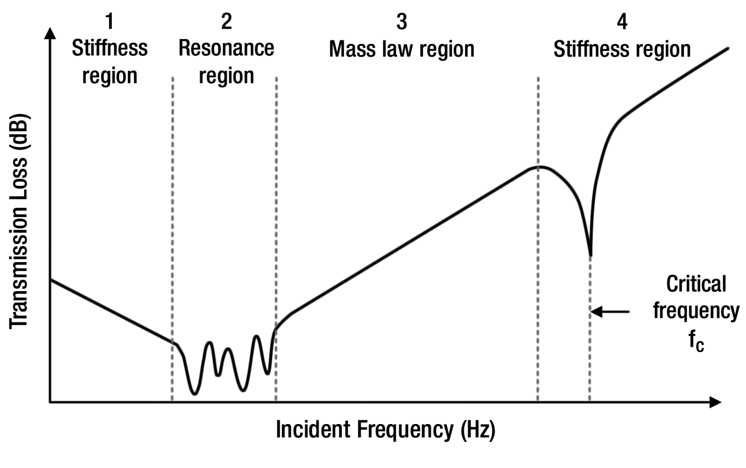


Figure 2.33: Four distinct regions showing the way a rigid barrier will react to different frequencies of incident sound. Reproduced from [57]

The attenuation afforded by noise barriers is primarily affected by their size and the relative distance from both source and receiver due to frequency dependent diffraction (see Figure 2.34). Within a certain region of proximity, the barrier casts an acoustic shadow, the extent of which is dependent on the wavelength of incident sound.

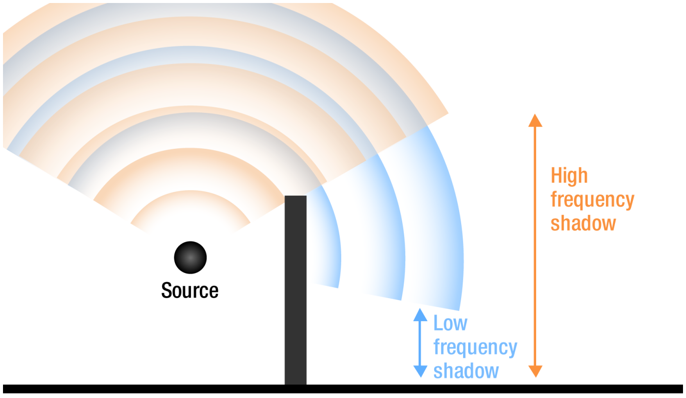


Figure 2.34: Frequency dependent diffraction over a rigid noise barrier.

The simplest and most widely used method for predicting the loss in sound pressure due to diffraction around the edge of a rigid barrier uses something known as a *Fresnel number* [13]. The Fresnal number method, based on the Kirchoff-Fresnal diffraction theory developed by Kirchoff in relation to electromagnetic rays, simply calculates the minimum increase in path length a wave must travel from source to receiver in the presence of the barrier, divided by half the wavelength of the frequency of interest - diffraction being a frequency dependent phenomenon. The Fresnel number is defined as:

$$N = \frac{2d_p}{\lambda} \tag{2.28}$$

where N is the Fresnal number, λ is the wavelength, and d_p is the path difference between the transmitted and diffracted sound which can be calculated using simple trigonometry provided the heights and relative positioning of source, receiver and barrier are known. A second Fresnel number, N_2, is defined similarly to the first Fresnel number, representing the relative positions of an image source and the receiver [58].

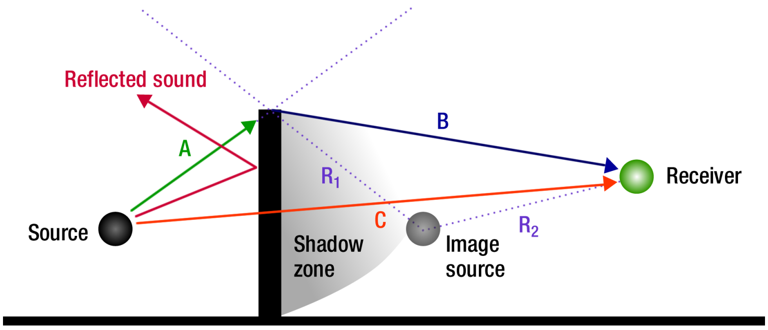


Figure 2.35: From (2.28), the path difference d_p is equal to $A + B - C$.

Maekawa [59] attempted to use Fresnal number theory to express the attenuation provided by a thin rigid barrier which he based on a curve derived from empirical measurements. Many subsequent attempts have been made to express this curve in a mathematical formula, the simplest of which is Maekawa's original formula (2.29) which uses just one Fresnel number.

$$Att = 5 + 20 \log_{10} \left(\frac{\sqrt{2\pi N_1}}{\tanh \sqrt{2\pi N_1}} \right) \tag{2.29}$$

An improved version of the formula that includes both Fresnel numbers is given in (2.30). This version has become the standard empirical method for barrier calculations, including the one outlined in ISO 9613-2:1996 [60].

$$Att = Att_s + Attb + Att_{sb} + Att_{sp} \tag{2.30}$$

Where

$$\begin{aligned}
Att_s &= 20 \log_{10} \left(\frac{\sqrt{2\pi N_1}}{\tanh \sqrt{2\pi N_1}} \right) - 1 \\
Att_b &= 20 \log_{10} \left[1 + \tanh \left(0.6 \log_{10} \frac{N_2}{N_1} \right) \right] \\
Att_{sb} &= \left(6 \tanh \sqrt{N_2} - 2 - Att_b \right) \times \left(1 - \tanh \sqrt{10 N_1} \right) \\
Att_{sp} &= -10 \log_{10} \frac{1}{(R'/R_1)^2 + (R'/R_1)}
\end{aligned} \tag{2.31}$$

In the above, Att is the total attenuation afforded by the barrier. Att_s is a function of N_1, which is a measure of the relative position of the receiver to the source. Att_b depends on the ratio of N_2/N_1, Att_{sb} depends on the proximity of the receiver to the shadow region, and Att_{sp} is a function of R_1 and R' which denote the image source to barrier edge distance and shortest source-edge-receiver path respectively (see Figure 2.35). Att_{sp} accounts for the diffraction effect due to spherical incident waves [15].

Transmission may also be prevented if the sound that enters a structure is absorbed internally. Structural discontinuities within a medium can cause destructive phase interference and increased classical losses due to internal reflections. The amount of attenuation provided by a structure may thus be improved substantially at certain frequencies without increasing its mass, simply by layering materials of different characteristic impedances. For example, by leaving air gaps between layers of rigid panels in a barrier or building partition. If holes need to be present in a barrier for ventilation or cables, the transmission loss may be less compromised by offsetting the holes between layers, which serves to lengthen the pathway of transmitted sound.

Vegetation

When sound interacts with foliage it may be scattered, reflected, diffracted or refracted, depending on the wavelength and size and surface of the leaves and branches. Generally speaking, small objects like leaves do not produce these effects unless they are very numerous and dense, in which case the acoustic properties of the medium may be changed [61]. Dense forests are essentially a stratified medium, whereby the different layers (sky, canopy, below canopy, ground etc.) can affect the propagation of sound and cause excess attenuation at some frequencies resulting in *shadow zones*. Belts of trees may thus act as sound barriers, the effectiveness of which at any given wavelength is largely dependent on ground type, belt depth, distribution of the trees, and the presence and nature of any foliage. Wind and temperature gradients can either contribute to or reduce the barrier effect by refracting the sound path up or down [62].

Generally attenuation due to vegetation is only relevant when making acoustic predictions for vegetation of depths of more than 10m, in which case the scattering by leaves and shrubs contributes to the attenuation of sound in the mid-frequencies, while absorption is more prevalent in the high-frequencies. At lower frequencies, attenuation is mainly the result of wave interference between different layers in the tree belt. Scattering and absorption by tree bark contributes mainly to mid-frequency attenuation, depending on the type of tree and the diameter of the trunk. Trunk scattering is also thought to reduce the ground effect by reducing the coherence between waves [13].

The absorption coefficient α of various different types and configurations of trees and shrubs have been derived empirically using the reverberation chamber method (for example, [63, 64]), currently defined in ISO 354:2003 [18]. These experiments have resulted in the development of some general formulae that allow one to approximate the attenuation produced by absorption from a belt of trees.

$$Att_{absorption} = -10\log_{10}\left(1 - \frac{1}{8}GS\varnothing f^{\frac{1}{2}}\right)\, dB \tag{2.32}$$

where S is the total surface area of leaves per unit volume, $\varnothing$ is the diameter of the trees, f is the frequency of interest and G is the attenuation factor and is equal to $\frac{\alpha}{f^2}$ [61].

Buildings

In general, a building is effectively a barrier in the pathway of sound, and can thus be treated as such - i.e. by considering its size and position relative to the source-receiver geometry and wavelengths of sound. However, it is difficult to refer to the impact of buildings on the propagation of sound in the atmosphere without stepping into an almost entirely separate field in acoustics - i.e. that of urban acoustics. The propagation of sound in urban areas is characteristically quite different from that of rural areas and other open air spaces where buildings and roads are much sparser. This is due to the multipath propagation of sound between source and receiver, due to the presence of buildings and hard reflective surfaces. The additional paths serve to enhance the sound above that which would be received in an open area, albeit to a lesser extent in wide streets where the buildings are further apart. Furthermore, the surfaces of buildings are not perfectly smooth, hence scattering by irregularities reduces the amplitude of the specular reflections and contributes to the diffuseness of the sound field, the extent of which may be related to the building geometry and other architectural features. In urban environments, the propagation of sound is thus primarily dictated by geometrical spreading, atmospheric absorption, reflection, surface absorption and scattering, and to a lesser degree by meteorological effects [65].

2.4.3 Terrain Type and Contour

A combination of ground absorption, reflection and refraction all serve to modify the geometrical spreading of sound according to the physical laws outlined in Section 2.4.1. Losses due to absorption will depend on the surface type and its characteristic impedance. For example, smooth, hard surfaces will absorb very little sound, whereas thick grass may result in a significant reduction in sound levels at mid-high frequencies, despite being less effective at low frequencies. Reflections from the ground may also result in another more complicated mechanism by which sound levels are reduced - an effect known as the *ground effect*. Estimation of the ground effect as an approximate indicator of the combined contributions of these effects is generally the approach taken in atmospheric acoustic assessments.

The Ground Effect

When broadband sound propagates over a flat absorbing ground, an effect known as the ground effect can occur, whereby the sound is attenuated excessively within a narrow range of frequencies, and negatively attenuated at others. Ground effect is the result of interactions between direct sound radiated from a source and sound that is reflected from the ground resulting in constructive or destructive wave interference. In the case of a plane wave with a reflection coefficient R, the total pressure at the receiver may be given by

$$P = P_d + R_p P_r \tag{2.33}$$

where P_d is the direct contribution and and P_r is the specularly reflected contribution [66]. The phase difference between the direct sound and the reflection which leads to constructive or destructive wave interference is caused by both the path length difference (see Figure 2.36) and the phase change on reflection with the ground. Since the degree of phase change is connected to the characteristic impedance of the ground, the type of ground - i.e. its density and porosity - thus affect the frequencies at which interference occurs. For porous surfaces with low impedance, phase change tends toward π, while for acoustically hard surfaces with very high impedance, phase change tends toward 0. The frequencies at which destructive interference occurs for a given source-receiver geometry can be derived from the following:

$$f_n = c\left[\frac{(2n+1)\pi - \varphi}{2(r_2 - r_1)}\right] \tag{2.34}$$

where r_1 and r_2 are the path lengths of the direct and reflected sound as per Figure 2.36, φ is the phase change on reflection from the ground, c is the speed of sound, and $n = 0, 1, 2, 3\dots$ [66].

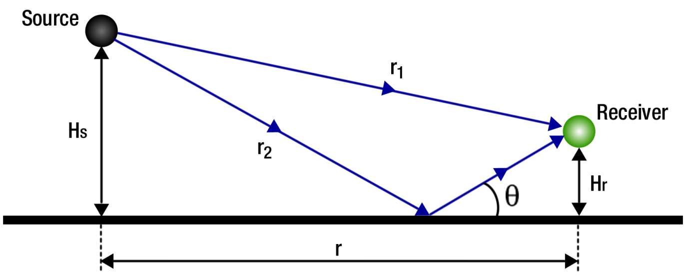


Figure 2.36: Source-receiver geometry above a flat absorbing ground. Reproduced from [66]

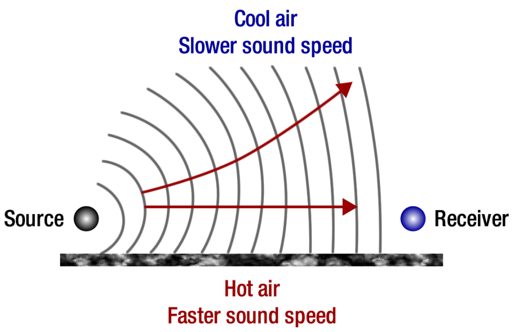
For the purpose of predicting the sound pressure level at a receiver relative to a specific source, the attenuation due to the ground effect may be approximated according to the following empirically derived formula, as outlined in ISO 9613-2 [60].

$$A_{ground} = 4.8 - \frac{2h}{d}\left(17 + \frac{300}{d}\right) \geq 0\,dB \qquad (2.35)$$

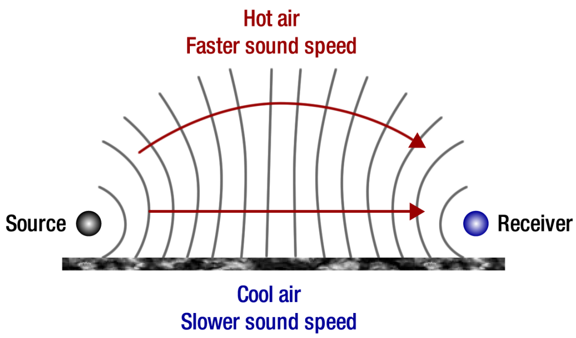
where h is the average height above ground of the direct sound propagation path in metres, and d is the distance from source to receiver [60].

2.4.4 Meteorological Conditions

The most common meteorological effects encountered in atmospheric acoustics are temperature and wind gradients, both of which affect the speed of sound propagation and lead to refraction. If the gradient is positive, the sound is refracted downward, whereas a negative gradient will cause the sound to bend upwards. Usually the air near the surface of the earth is warmer than the air above it, which produces a negative vertical temperature gradient. Under certain conditions, this gradient can become inverted, for example, when the thermal energy radiated by the earth's surface exceeds that which is received from the sun, as tends to happen at night. This means that, under normal daytime conditions, more sound energy is lost to atmospheric absorption that would be at night, leading to higher noise levels at night for the same source-receiver configuration (see Figure 2.37) [13].



(a) Positive temperature gradient.



(b) Negative temperature gradient.

Figure 2.37: Temperature gradients can cause refraction and affect the noise heard on the ground.

Similarly, sound propagating upwind will travel more slowly than sound propagating downwind, hence a receiver that is downwind of a source is likely to receive a higher dose of sound than one that is the same distance away in the opposite direction - depending on the surface conditions. The relationship between the speed of sound (c), temperature (T) and wind speed (u) in the direction of propagation with respect to the height above ground level (z) may be expressed generally by the following:

$$c(z) = c(0)\sqrt{\frac{T(z) + 273.15}{273.15}} + u(z) \tag{2.36}$$

where T is given in degrees centigrade, z is given in metres, and u is given in metres per second [13].

Other meteorological conditions that can have a marginal effect on the speed of sound propagation include precipitation such as snow, fog and rain due to the associated rise in humidity. However, the effect is only small, typically between 0.1-0.6% [67], and therefore unlikely to be of any significance in most real world situations. However, the accumulation of precipitation on any surface over which sound is propagated is much more significant due to its influence on the reflection and absorption properties of the surface.

2.4.5 Atmospheric Absorption

When sound waves travel through air they are primarily attenuated due to geometric spreading, although a small amount of attenuation results from atmospheric absorption. Because it is only a small amount, atmospheric (or air) absorption is generally only taken into account when considering propagation over longer distances, or where high frequencies or turbulent airflow are concerned. Atmospheric absorption is caused by a combination of classical losses (i.e. heat loss and shearing forces), and molecular relaxation losses [15]. Classical losses are mainly affected by changes in temperature and atmospheric pressure, while relaxation losses are primarily affected by the composition of the medium, particularly its humidity. On the basis of experimental data and theoretical principles originally developed by Knudsen and Kneser [68], a standard working model has been developed for estimation of atmospheric absorption in outdoor noise assessments. The amount of atmospheric absorption in decibels over a path of length d is expressed in terms of an attenuation coefficient, A_a, according to the basic expression:

$$A_a = -20\log_{10}\left[\frac{p_d}{p_0}\right] = -20\log_{10}\left[\exp(-\alpha d)\right] = ad \tag{2.37}$$

where p_d is the sound pressure after travelling the distance d, p_0 is the initial sound pressure at $d = 0$, α is the attenuation coefficient in Nepers per metre, and a is the attenuation coefficient in decibels per metre ($a = 0.868\alpha$) [15]. The procedure for calculation of the frequency dependent attenuation coefficient α in decibels per metre is outlined in ISO 9613-1:1993 [69] and involves the four variables temperature, atmospheric pressure, relative humidity and frequency. The formulae also use two relaxation frequencies f_{rO} and f_{rN} which are the relaxation frequencies of the two main constituents of air, oxygen and nitrogen. Relaxation frequency is a term used to describe the rate at which a perturbed medium returns to its equilibrium.

2.5 Room Acoustics

In room acoustics the principles governing sound propagation are effectively the same as those attributed to atmospheric acoustics, although there is generally a more dominant psychoacoustic element involved in their assessment due to the activities associated with them. The distances involved tend to be much shorter and are not affected by factors such as ground attenuation or atmospheric absorption, at least not to a noticeable extent. Room acoustics is therefore predominantly concerned with the reverberant and absorptive characteristics of a space and what effect this has on the perception of sound in the space, relative to the nature of the sound and the purpose the room serves. This tends to lead to more detailed estimations of both the physical behaviour and perception of sound in the space, potentially involving a large number of objective and subjective parameters. Whilst this thesis is mainly concerned with outdoor sound propagation, the behaviour of sound indoors is also significant in relation to the reproduction aspect.

2.5.1 Room Impulse Responses

In rooms and other enclosures, the sound that reaches a listener's ears is made up of a combination of direct and indirect sound. The indirect sound is a combination of early and late reflections that reach the listener via multiple different routes, having bounced around through interaction with boundaries and other obstacles. The frequency dependent absorption by obstacles and surfaces, as well as wave interference effects act to alter the spectral characteristics of the sound, and the geometrical design of the room and the roughness of the surfaces alter the distribution of the reflections. The combination of these effects give a room its own unique character or 'acoustic fingerprint'. This acoustic fingerprint is generally defined as the response of a room to an impulse input or sudden high energy broadband noise and is therefore something which can be measured relatively easily. Its representation in the time domain is known as an *Impulse Response* (IR) (see Figure 2.38) which is typically characterised by three distinct regions. These are the direct sound; the early reflections - i.e. sound that has been reflected once or twice from parts of the room; and the late reverberant 'tail' consisting of many overlapping resonances. While the impulse response can give a general approximation of the acoustics of a room, strictly speaking a single impulse response is valid only for a specific source-receiver configuration.

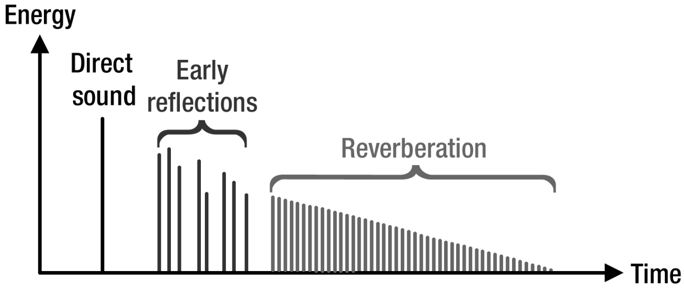


Figure 2.38: The acoustic fingerprint of a room or other enclosure is represented in the time domain by an Impulse Response (IR) which is typically characterised by three distinct regions: direct sound, early reflections, and a reverberant tail.

2.5.2 Reverberation Time and Other Objective Parameters

A single monaural RIR can be used in the estimation of various objective acoustical parameters, including the reverberation time (RT) and early decay time (EDT). For estimation of other more advanced acoustical parameters such as the Interaural Cross-correlation Coefficient (IACC), Interaural Cross-correlation Function (IACF) and Lateral Fraction (LF), a binaural impulse response or stereo pair of impulse responses are needed. Reverberation time is defined as the amount of time it takes for the sound level at a receiver to reduce by a certain amount after the main source of sound has stopped. The most commonly used objective measure of reverberation time is the T_{60} time (also written RT60), which is the number of seconds it takes for the sound level at a receiver to reduce by 60 dB after cessation of the source. There are various methods for measuring the reverberation time in rooms of different sizes. The most common of these are the integrated impulse response method [70], interrupted noise method and impulsive impulse response method. The latter two methods are very similar procedures, the only real difference being the form of excitation. Both are easy to implement, and for this reason are commonly used to estimate the reverberation of rooms with relatively simple geometries. While all of these methods are thought to yield similar results in large, regular shaped rooms, in smaller and/or irregular shaped rooms, more care should be taken over the choice of measurement method and excitation source [71]. Furthermore, when the extrapolation of other acoustical parameters is also a requirement of the measurement, an advanced method would tend to be favoured. Various types of excitation signals are suitable for use with the integrated impulse response method, including Maximum Length Sequences (MLS), sine sweeps and pink noise, each of which have their associated advantages and disadvantages. The standard procedure for estimation of the reverberation time and other acoustical parameters from the room impulse response is outlined in BS ISO 3382 [72].

Before modern technology facilitated the widespread use of impulse response methods, empirical formulae were developed in order to allow estimation of the reverberation time in rooms. These are based on the volume and a number of absorption units which are derived from the surface area and absorption coefficients of the surfaces. The most well-known of the empirical methods are the Sabine (2.38) and Norris-Eyring (2.39) equations.

$$T_{60} = 0.161\frac{V}{A} \tag{2.38}$$

where V is the volume of the room in cubic metres and A is the total number of absorption units in the room, measured in Sabins, and calculated according to $S_1\alpha_1 + S_2\alpha_2 + S_3\alpha_3 \cdots S_n\alpha_n$ where $S_1 \cdots S_n$ are the areas of the surfaces in square metres, and $\alpha_1 \cdots \alpha_n$ are their respective absorption coefficients.

$$T_{60} = \frac{0.049V}{-S_T \ln(1-\bar{\alpha})} \tag{2.39}$$

where S_T is the total area of all room surfaces and $\bar{\alpha}$ is the arithmetic average of the surface elements S_i multiplied by their absorption coefficients α_i according to (2.40).

$$\bar{\alpha} = \frac{1}{S}\sum_i S_i\alpha_i \tag{2.40}$$

Both of the above can give a reasonable estimation of the reverberation time in small-medium sized rooms when more advanced measurement technology or acoustic design software is unavailable. In acoustically 'dead' rooms, i.e. rooms with a lot of absorption, the Norris-Eyring is usually the more accurate of the two, as the Sabine equation will always give a non-zero result [73]. Neither are these methods generally reliable for rooms with irregular aspect ratios or non-uniform absorption distribution which are characteristically non-diffuse. This is due to the assumptions that sound waves travel between surfaces in straight lines and that surfaces have an average absorption coefficient - both of which can lead to inaccurate estimations of the reverberation time [74].

2.5.3 Standing Waves and Room Resonance Modes

Modal resonances or *room modes* are a collection of resonant frequencies related to the wall-to-wall and ceiling-to-floor dimensions of the room and can cause patterns of standing waves, particularly in rectangular 'box-like' rooms.

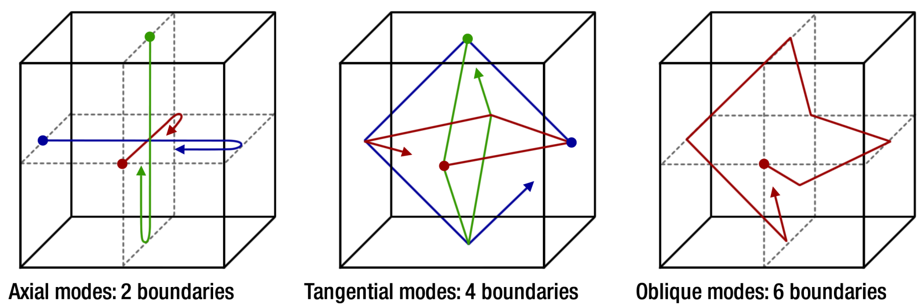


Figure 2.39: In regular shaped rooms there are three types of room mode, each of which involves a different number of room boundaries.

A standing wave is a sound wave that does not move due to its reflection between two parallel walls separated by a distance that is either equal to or some multiple of the wavelength (see Figure 2.40). It is thus characterised by areas of no displacement (nodes) and areas where the particles in the medium oscillate about their equilibrium with maximum amplitude (anti-nodes). Standing waves may be experienced as audible peaks and dips in the sound level at particular frequencies and can be problematic in room acoustics, particularly if the distribution of standing waves is such that the peaks and dips are well-defined without much overlap. Generally speaking this tends to occur at low frequencies, while at higher frequencies the distribution of the room resonances is progressively denser with statistical (Gaussian) properties.

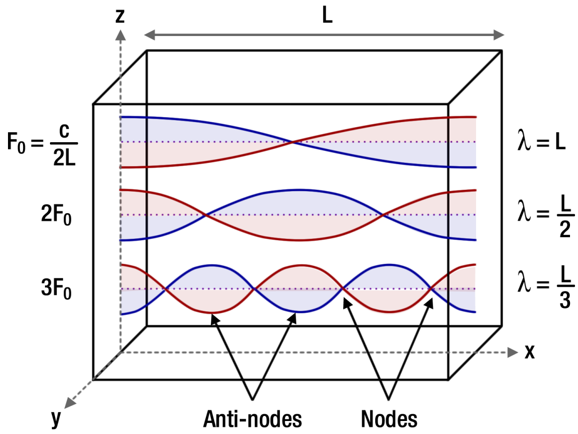


Figure 2.40: The fundamental and first two harmonics of axial mode standing waves.

The Schroeder frequency is an approximate crossing-point below which the behaviour of the system is characterised by individual, well-separated resonances, as opposed to to many overlapping normal modes [75]. For airborne sound in reverberant enclosures, the Schroeder frequency is given by:

$$f_c \cong 2000\sqrt{\frac{T_{60}}{V}} \tag{2.41}$$

where T_{60} is the $60 - dB$ reverberation time in seconds and V is the volume of the enclosure in cubic metres [76].

2.5.4 Acoustic Treatments

Consideration for the reverberant behaviour of rooms is important for ensuring the sound that reaches a listener's ears does not suffer from undesirable spectral colouration or other adverse acoustical effects such as flutter echoes. Whereas the signal colouration due to reverberation is very often a desirable feature of some rooms - provided it is properly managed and appropriate for the intended application - this is not generally the case when replicating complex sound fields over loudspeakers. Where the aim is to preserve the spatial cues and spectral information inherent in one or more channels of audio, one would normally seek to suppress room reflections to the greatest extent possible, thus minimising interference effects between direct and indirect sound and reducing the likelihood of echoes. This can be achieved by increasing the absorption outside the reproduction area, and/or through the installation of acoustic diffusers.

Absorption

Sound absorption in a room depends mainly upon the surface area and material properties of boundaries and other objects in a room. Increasing the absorption in a room intended for sound reproduction can thus be achieved though the installation of acoustical panels or porous materials such as curtains and acoustical tiles outside the reproduction area. Porous materials are effective at absorbing high frequencies due to heat losses through interaction with fibres within the material, whereas thin panels of wood and various other solid materials can provide absorption at low frequencies due to their ability to flex in response to sound waves of large wavelengths [77].

Another type of absorber based on the principle of the Helmholtz resonator can be designed to give selective absorption over a narrow range of frequencies. These generally take the form of perforated panels with air cavities behind, such that each perforation behaves as an individual Helmholtz resonator. A Helmholtz resonator is a device originally used by Helmholtz in the 1800s to identify musical tones [78] and is characterised by a narrow tube or ‘neck’, connected to a resonant chamber. It can be modelled on a classical mass-spring system, whereby the air in the neck behaves like a mass on a spring, which, following an initial displacement, may be caused to vibrate by the compression and rarefication of the air inside the chamber. The resonance frequency f_r of a Helmholtz resonator is given approximately by:

$$f_r = \frac{c}{2\pi}\sqrt{\frac{A}{VL}} \tag{2.42}$$

where c is the speed of sound, A is the cross-sectional area of the neck, V is the volume of the cavity and L is the length of the neck (see Figure 2.41). From (2.42) it is possible to approximate the resonance frequency of a perforated panel absorber by:

$$f_r = \frac{200 l_h p_p^{0.5}}{l_c} \tag{2.43}$$

where l_h is the effective hole length with the correction factor ($panel\ thickness\ +\ 0.8\ \times\ hole\ diameter$), p_p is the perforation percentage ($hole\ area/panel\ area \times 100$), and l_c is the length of the air cavity [77].

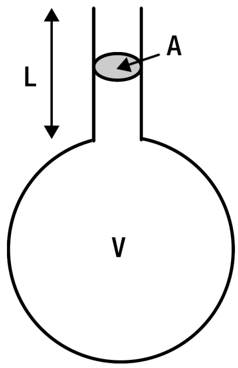


Figure 2.41: A Helmholtz resonator is a container of air with a narrow neck ending in an open aperture. The air in and near the aperture vibrates due to the 'springiness' of the air inside.

Diffusers

Where it is impossible to eliminate problematic room reflections through absorptive treatments, another solution is to redistribute the reflected sound energy so that it is spread evenly throughout the space - something that can be achieved with diffusers. Diffusers are irregular surfaces that scatter incident sound in a diffuse manner - i.e. in all directions. Historically diffusers were incorporated into a structure in a fairly haphazard way through the inclusion of structural ornamentations such as statues and cornices. It was not until that the late 1970s that diffusers were designed to reflect sound in a predictable way, largely due to the pioneering work of Schroeder whose phase-grating diffuser based on number theory sparked a series of developments in diffuser technology [79, 80]. Most modern diffusers are thus characterised by probability distribution functions that define the range and type of randomness associated with the scattered sound energy, and can be optimised for a particular room and/or delivery system.

2.5.5 Subjective Sound Quality

Depending on the intended use(s) of a given room, a group of subjective parameters may be selected to assess the sound quality of the room through controlled listening tests. Generally speaking, subjective parameters are adjectives that have been found to correlate well with individuals' perceptions in a specific environment. Much effort has been made to find correlation between subjective parameters and measurable acoustic quantities. The resulting group of objective acoustical parameters are thus important for enabling the prediction of subjective sound quality in a room prior to its construction. Some of the more common subjective parameters and their associated objective parameters are given in Table 2.2. All are applicable to a given receiver position and are typically used in relation to musical performance venues - hence the musical connotations with several of the adjectives.

For obvious reasons more care and attention has historically been granted to the subjective sound quality in spaces intended for music, whereas other types of rooms tend to have much simpler and more easily quantifiable requirements. For example, in a teaching room, the aim is for speech to be audible and intelligible anywhere in the room, without the teacher having to strain their voice. Relevant objective parameters are therefore the Reverberation time (T_{60}), Background Noise Level (BNL), Signal to Noise Ratio (SNR) and Early Decay Time (EDT), all of which contribute toward speech intelligibility [81]. Acoustical measurements of speech and noise are thus involved in the calculation of several physical measures that are used to assess the intelligibility of speech, the most common of which is the Speech Intelligibility Index (SII) [82].

Subjective Parameter	Objective Parameter	Definition
Liveness	Reverberation Time (T_{60})	The time it takes for the steady state sound level to decrease by 60dB after cessation of the source.
	Early Decay Time (EDT)	The reverberation time measured over the first 10 dB of the decay.
Warmth	Bass Ratio (BR)	The ratio of the average reverberation times at 125 and 250 Hz to those at 500 and 1000 Hz $\left(\frac{RT_{125}}{RT_{500}}\frac{RT_{250}}{RT_{1000}}\right)$
Brilliance	Treble Ratio (TR)	The ratio of the average reverberation times at 2000 and 4000 Hz to those at 400 and 1000 Hz $\left(\frac{RT_{2000}}{RT_{400}}\frac{RT_{4000}}{RT_{1000}}\right)$
Intimacy	Initial Time Delay Gap (ITDG)	The difference between the arrival time of the direct sound, t_d, and that of the first significant reflection t_r ($t_d - t_r$).
Ratio D/R	Direct to Reverberant Ratio (RDR)	The logarithmic ratio of the energy of the direct sound to the energy of the rest of the impulse response $\left(\log_{10}\left(\frac{E_d}{E_r}\right)\right)$
Clarity	C50 C80	The logarithmic ratio of early to late sound energy, or more specifically, the ratio of all the sound energy that arrives within the first 50ms (speech) or 80ms (music) to all that arrives after $\left(C80 = \log_{10}\left(\frac{E_{0-80ms}}{E_{80-\infty ms}}\right)\right)$
Definition	D50 D80	The ratio all the sound energy that arrives within the first 50ms (speech) or 80ms (music) to the total incident sound energy $\left(D80 = \frac{E_{0-80ms}}{E_{0-\infty ms}}\right)$
Sound strength	G	The logarithmic ratio of the sound pressure of the impulse response to the response of the same source placed 10 meters away in a free-field $\left(\log_{10}\left(\frac{L}{L_{free}}\right)\right)$
Spatial Impression	IACC	The maximum value for the correlation between signals arriving at the left and right ears, and is expressed in values ranging from -1 (left and right signals 180° out-of-phase) to +1 (identical signals). The more dissimilar the signals, the more spaciousness is perceived.
	Lateral Energy Fraction (LF)	The ratio of sound arriving from a lateral plane over sound arriving from all directions $\left(\frac{E_{5-80mslat}}{E_{0-80msomni}}\right)$.

Table 2.2: Some well-known subjective and associated objective acoustical parameters [83, 84].

2.6 Summary

In this chapter the author introduced some of the fundamental physical and psychoacoustic principles that are applicable to the areas covered in this thesis. It is clear from this relatively short foray into spatial hearing that the mechanisms by which sources of sound are localised in 3-D space are numerous and varied according to the nature and frequency of the sound and its surrounding environment. It would also appear to be the case that while there is a well-developed awareness and base knowledge of different psychoacoustic phenomena, a complete understanding of the cognitive processes leading to auditory perception is still a long way off. Nevertheless, what has been learned through observation can and does serve a valuable purpose in acoustics and audio engineering by informing the development of a wide range of audio technologies, from audio compression algorithms, to cochlear implants. In terms of spatial audio reproduction, knowledge of spatial hearing has an important part to play in ensuring that the audio material is represented in the best possible light.

The author also presented a brief introduction to atmospheric and room acoustics in order to gain an appreciation of their respective differences. It became apparent that, whereas assessing the subjective sound quality in rooms is a well studied and relatively constrained problem, outdoors the problem is a much less well understood and more convoluted one. For example, the study of room acoustics has led to the widespread adoption of meaningful objective measures of sound quality, while work on establishing an equivalent set of parameters for outdoor environments is at a much less advanced stage. In fact it is only in recent years that research has begun to shed light on the many uncertainties involved in the perception of outdoor sound environments - often referred to as *soundscapes*. The area of soundscape research is thus a relatively new discipline and one which will be the focus of Chapter 4. Before that, the next chapter will look at how the acoustics of outdoor environments are currently assessed and regulated using established empirical methods.

3

Environmental Noise

3.1 Introduction

The purpose of this chapter is to examine the topic of environmental noise from a practical and regulatory perspective. This means looking at how it is monitored and mitigated on a day-to-day basis at the local, national and international government levels. The intention is to examine the available evidence regarding adverse effects of environmental noise, and to look at how this has been implemented in practice, both in terms of the legislation and relevant assessment methodologies.

In acoustic terms, *noise* is generally taken to mean any unwanted sound, which often happens to be sound that is considered by a listener to be excessively loud, unexpected or unpleasant. *Environmental noise* is thus the term used by governments and local authorities to refer to unwanted or harmful outdoor sound created by human activities, including noise emitted by road traffic, rail traffic, air traffic, and from sites of industrial activity [8]. The World Health Organisation (WHO) defines environmental noise, also referred to as community, residential or domestic noise, as noise emitted from all sources, except noise at the industrial workplace [85]. As such, it is a blanket term which does not give much consideration to the nature and perception of individual sounds in their wider sociological and/or ecological context. By this definition, attitudes to environmental noise tend - unsurprisingly - to be negative, hence the approaches taken in environmental noise control make it an area of acoustics that is conceptually and methodologically distinct from that of soundscape research which considers environmental sounds as a resource rather than a waste [86].

Environmental noise is now recognised by the European Union as being ‘one of the main local environmental health problems in Europe’ [8]. Whilst the effects of environmental noise are not fully understood, it is known to cause annoyance which may lead to increased stress and its associated physical symptoms in some individuals. Sudden loud noises and low level continuous noise are both known to cause sleep disturbance, which may have further health-related consequences. There is reportedly some evidence to suggest there could be a connection between environmental noise and cardiovascular and mental health problems, and reduced performance at work or school [87]. However, the relationship is not a straightforward one, as noise is just one amongst many stressors encountered in day-to-day living. Furthermore, attitudes to noise are highly influenced by the personal circumstances and cultural background of an individual, hence the inherent difficulty in predicting the overriding response of a population. Nevertheless, the overriding attitude amongst governments and health organisations worldwide is somewhat bleak, ranking noise alongside other forms of air pollution as a serious threat to health and well-being. One source warns,

> [Environmental noise] is more severe and widespread than ever before, and it will continue to increase in magnitude and severity because of population growth, urbanisation, and the associated growth in the use of increasingly powerful, varied, and highly mobile sources of noise [1].

Consequently, there are a number of governmental policies effective in the UK which make provisions for the prevention and mitigation of environmental noise. Over the years these have gradually been extended to cover different types of noise, and to impose certain responsibilities on local authorities. The most important of these are outlined in this chapter.

3.2 Noise Effects

The negative effects of environmental noise on human health and well-being can be separated into two categories: auditory and non-auditory.

3.2.1 Auditory

While the association between environmental noise and hearing impairment is weaker, by comparison, than other sources of noise such as occupational and leisure-time noise, its prevalence is thought to be increasing, albeit predominantly in developing countries [85]. Of the various types of noise-induced hearing impairment, the one most likely to be attributed to environmental noise is that of permanent threshold shift (PTS), also referred to as permanent hearing loss, caused by continuous exposure to sound levels of 80 dB(A) or more. PTS is a cumulative process that is not normally noticeable until it is severe enough to interfere with routine activities. It can be quantified in terms of a subject's ability to understand conventional speech in common levels of background noise. While it affects different people differently, noise-induced hearing impairment occurs predominantly in the high-frequency range of 3000-6000 Hz, with the effect initially being greatest at 4000 Hz [85, 88] before gradually spreading over other frequencies as noise level and/or exposure time increases.

Often linked to hearing loss is the condition referred to as tinnitus, whereby the sufferer experiences sounds that have no external physical cause, within their own heads - usually described as ringing, hissing or buzzing. In some cases tinnitus is thought to occur when the stereovilli of the inner ear are damaged, possibly due to noise exposure, giving rise to random noises [89, 90]. Interestingly, environmental sound can actually be a benefit to sufferers of tinnitus. Among the treatments for tinnitus is Tinnitus Masking (TM), which involves playing masking sounds such as white noise or environmental sounds to reduce the perception of tinnitus. Masking can also be achieved by amplifying environmental sound through the use of a hearing aid [91, 92]. In a related study, it was even suggested that exposure to high and low frequency environmental sound can lead to diminished hearing losses due to the neuro-plasticity of the brain [93].

Other types of hearing impairment that could in rare cases be attributed to environmental noise include acoustic trauma - i.e. sudden hearing damage following a short burst of extremely loud noise (>140 dB) - and temporary threshold shift (TTS) immediately proceeding exposure to a high level of noise. TTS is experienced as a temporary loss of sensitivity at certain frequencies which gradually recovers once the source has stopped. Recovery time varies depending on the length of the exposure, the nature of the noise and the age of the sufferer.

3.2.2 Non-auditory

While the auditory effect of hearing impairment is well-known in connection with prolonged exposure to loud noise, non-auditory effects are more numerous and much less well understood. These are sometimes sub-divided into performance and physiological related effects, although, in relation to environmental noise, it is difficult and not particularly expedient to attempt to group them in this way. The main classes of non-auditory effects are listed below. The list is based on the *WHO Guidelines for Community Noise* [85].

Interference with spoken communication Various problems can arise as a result of speech interference. Problems with concentration, fatigue, uncertainty and lack of self-confidence, irritation, misunderstandings, decreased working capacity, problems in human relations, and a number of stress reactions have all been identified [94]. The problem occurs when other, simultaneous noises mask the speech in the critical frequency range (300-3000 Hz), to the extent that it is rendered unintelligible. For complete intelligibility in listeners with normal hearing, the signal-to-noise ratio (SNR) should be between 5 and 10 dB [95, 96]. Of course, it is not only our comprehension of speech that suffers as a result of excessive noise. Other sounds that are considered an important part of our everyday lives can be affected too, such as fire alarms, door bells, music and telephone signals.

Sleep disturbances There are various kinds of sleep disturbance that may be attributed to noise. These include difficulty in falling asleep; awakenings; and alterations of sleep stages or depth, especially a reduction in the proportion of REM-sleep (REM = rapid eye movement) [97]. Other physiological effects that can be induced by noise during sleep include increased blood pressure; increased heart rate; increased finger pulse amplitude; vasoconstriction; changes in respiration; cardiac arrhythmia; and an increase in body movements [85]. Needless to say, these are all effects which can contribute to the development of more serious conditions is some individuals.

Cardiovascular disturbances Environmental noise is considered by the WHO to be a potential contributor to instances of coronary heart disease (CHD) in the US and Europe, based primarily on evidence published by the WHO Noise Environmental Burden on Disease working group in 2005 [98]. Acute noise exposures activate the autonomic and hormonal systems, leading to temporary changes such as increased blood pressure, increased heart rate and vasoconstriction [85]. Provided exposure is temporary, the physiological system usually returns to normal after a period roughly equivalent to the duration of the exposure. However, after prolonged exposure, susceptible individuals may develop permanent effects, such as hypertension and ischaemic heart disease. Some studies have suggested there could be other psychophysiological effects directly linked to noise pollution[85]. These include changes in stress hormones, magnesium levels, immunological indicators, and gastrointestinal disturbances, although the evidence is inconclusive.

Mental health effects While environmental noise is not thought to be a cause of mental illness, it may accelerate development of latent mental disorders in some succeptible individuals. Studies on the adverse effects of environmental noise on mental health cover most of the common symptoms associated with mental health disorders, including anxiety; emotional stress; nervous complaints; nausea; headaches; instability; argumentativeness; sexual impotency; changes in mood; increase in social conflicts, as well as general psychiatric disorders such as neurosis, psychosis and hysteria [85].

Impaired task performance Several studies have indicated that noise can have adverse effects on cognitive and motivational performance in children [85]. It is also assumed that noise interference can have similar effects on adults, although there is currently no data to support this. It has also been suggested in [85] that noise might increase the incidence of accidents by causing a distraction or provoking a startle response. Again, no data is available to support this assumption.

Negative social behaviour and annoyance reactions Noise has been used as a noxious stimulus in a variety of studies because it produces the same kinds of effects as other stressors [99]. Although noise itself does not cause aggressive behaviour, as a stressor it can provoke aggression and other forms of antisocial behaviour, especially among those with a pre-existing tendency toward violent or hostile behaviour.

3.2.3 Measurement and Prediction of Non-Auditory Effects

Due to the incompleteness of our understanding, predicting the occurrence of non-auditory reactions to environmental noise is still problematic. There are two main approaches considered here: those which are based on socio-acoustic surveys, typically of the community noise survey sort, and those that are based on epidemiological and laboratory-based studies. However, research is still ongoing in this area and at present the data obtained from socio-acoustic surveys seems to be the most common basis for prediction, although both types of study are instrumental in arriving at recommended noise limits.

Socio-acoustic

Predicting the onset and severity of annoyance reactions to environmental noise can be problematic due to the inherent complexity of the problem. It is well known that the degree of annoyance produced by noise is highly variable and may be affected by a number of different factors, including the time-of-day, the unpleasant characteristics of the noise, the duration and intensity of the noise, the meaning associated with it, and the nature of the activity that the noise interrupted [99]. To complicate matters further, reactions to noise are inherently subjective and thus dependent on a whole host of individual and cultural sensitivities. Nevertheless, noise researchers are continuously working to establish and clarify relationships between measurable characteristics of noise and their negative effects on human health and well-being. Moreover, because of the association between annoyance and stress (stress being a known contributor to a large number of health problems), there is the tendency to assume a link between noise annoyance and adverse health effects. Of the numerous scientific studies conducted over the years, many have attempted to define these relationships in terms of *dose-response* or *dose-effect* curves - i.e. simple X-Y graphs relating the magnitude of a stressor to the response of a receptor (e.g. [100, 101]). These studies have typically been in the form of socio-acoustic surveys in which residents' reactions to noise are analysed in relationship to objectively determined acoustical parameters associated with their noise environment. The earliest attempts at defining a purely descriptive dosage-effect relationship in order to assess the effect of transportation noise on communities are attributed to Schultz who synthesised a set of curves in which average day and night sound levels were plotted against the age of residents who were 'highly annoyed' by the noise [100, 102] (see Figure 3.1). As in any social science research that relies purely on semantic descriptors, the wording of the questions can have strong implications on the results of community noise surveys, hence a lot of effort was made over subsequent years to construct high-quality survey questions that are able to yield internationally comparable measures of overall reactions to noise sources[103]. However, the usefulness of these curves, particularly from a policy making perspective is somewhat contentious, given that the correlation between the predicted annoyance prevalence rates and the rates actually observed in communities is often unsatisfactory, partially due to their failure to take into account or explain the great variability of community reaction [104]. There is also the danger of misinterpreting the results if one is operating on the assumption that greater annoyance shares a direct relationship with negative health effects, whereas, in reality, the relationship is not a straightforward one. In fact, the issue of whether or not noise policies should be based around the likelihood of complaints at all is historically a matter of some controversy, despite being a widely adopted approach (for example, BS 4142 [105] - perhaps the most widely used standard in the UK for community noise assessments - is founded on this approach). Yet, as Fidell points out, the lack of a strong or simple relationship between noise exposure and its effects is neither a consistent nor a persuasive rationale for ignoring noise complaints in policy analyses, especially given the current lack of alternatives and limited understanding of reactions to noise.

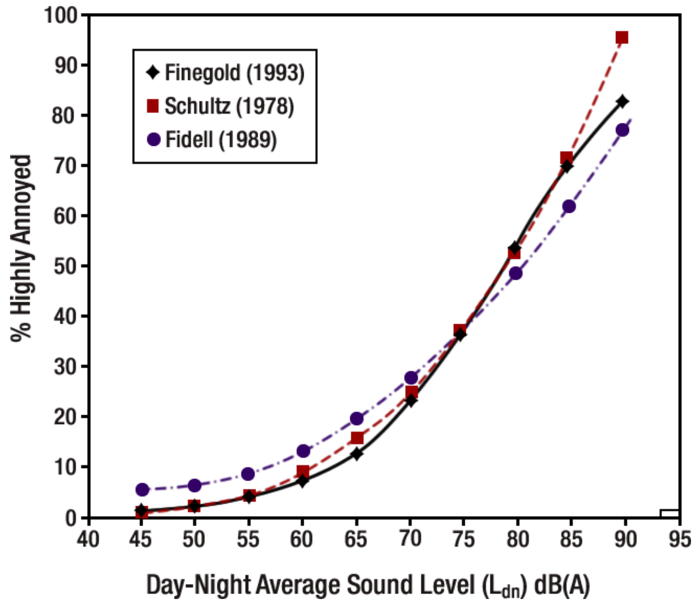


Figure 3.1: Dose-effect curves produced by Schultz[100], Fidell[102] and Finegold[106] to predict community response to noise based on socio-acoustic surveys.

Despite the difficulty in forming generalisations on the basis of auditory data, it is possible to draw some useful conclusions. For example, a greater level of annoyance is consistently observed in response to continuous low frequency noises, or noise which contains impulses such as the noise of gunshots. It has also been found that annoyance is greater when noise progressively increases rather than remaining constant. Thus, in environmental noise assessments, sources which exhibit one or more of these characteristics may incur noise penalties or ‘correction values’ (i.e. additional decibels are applied to the predicted sound level at the assessment location) [105].

Epidemiological

Over the past two to three decades, technological advances have made measuring non-auditory reactions to noise more readily practicable, including the determination of dosage-effect relationships based on epidemiological studies. These may take the form of electrocardiography (ECG) scans, clinical questionnaires, and monitoring blood pressure, hormonal and enzymatic changes amongst other things (e.g. [99, 107]). However, it should be noted that a large proportion of the epidemiological studies which show strong relationships between exposure to noise and health effects are based on occupational exposure to noise. Results of such studies should not be taken as evidence that environmental noise such as traffic and street noises induce the same effects, as even physiological reactions to noise are both source-dependent and tightly linked to activity and other health and lifestyle factors that may be prevalent amongst any given group of workers. In contrast to the acute noise effects observed in noisy work environments, physiological effects of relatively low environmental noise levels primarily occur when the sound level disturbs cognitive functions, causes emotional reactions, or interferes with activities of the individual such as mental tasks, relaxation or sleep [108]. According to Babisch there is very little evidence of increased occurrence of hypertension or high blood pressure in adults in response to noise, although, with regard to ischaemic heart disease, there is some evidence in the literature of an increased risk in subjects who live in noisy areas with outdoor noise levels of greater than 65-70 dB(A) [85]. Another recent study by Sørensen et al. [109] reports an increased risk of stroke due to exposure to noise greater than 60 dB(A), particularly amongst the elderly. Their report is based on statistical analysis of health questionnaire data obtained from a population-based cohort of 57,053 participants between 1993 and 1997, amongst which 1881 were recorded as having had a first-ever stroke in a national hospital register between the time they joined the survey and 2006. The data recorded for stroke and non-stroke participants was analysed in association with estimated noise exposure data based on the address history of the participants used in the study. As surveys of this type are still rare, it is difficult to draw any firm conclusions from it, although one would expect the frequency and scale of these sorts of studies to increase over the coming years as more and more electronically stored data becomes available.

3.3 Noise Regulation

In much of the developed world, particularly amongst EU constituents, development of national-level regulations has undergone a rapid evolution over the past decade, particularly in certain areas such as night-time aircraft noise emissions [110]. While there are some exceptions (Mexico, Brazil etc.) in the majority of developing countries, regulations are either much less stringent or non-existent, raising questions as to whether the approaches adopted by the Western world are appropriate, affordable, or technologically feasible in countries that are economically and culturally very different. According to Finegold, differences in noise sources, available finances and noise control technologies, cultural norms, climate, views concerning the role of the government, etc. seem to be suggesting that different approaches might be needed in developing countries for their noise policies to be effective [111].

In addition to the development of national-level regulations, formal standards development activities related to noise emission have also progressed at the international level by organisations such as the International Standards Organisation (ISO) and the European Commission. While the influence of such organisations and the general tendency toward scientific collaboration ensures policies and practices are relatively common - at least amongst developed countries - this section deals specifically with legislation that is largely applicable throughout the UK (although there is some variation in Scotland).

3.3.1 Environmental Protection Act 1990 (EPA)

Since the introduction of the Control of Pollution Act 1974[112] and which was later superseded by the Environmental Protection Act of 1990[113], noise may be treated as a statutory nuisance.

As such, persistent perpetrators of harmful or anti-social noise may be found guilty of a criminal offence should the matter be brought to court, although such occurrences are exceedingly rare with the vast majority of cases being settled out of court. Section 79 of the Environmental Protection Act 1990 places a duty on local authorities to inspect their areas periodically for statutory nuisances, and to take such steps as are reasonably practicable to investigate complaints of statutory nuisance[114]. If, following a reasonable form of investigation, the local authority feel satisfied that a statutory nuisance does exist, they may serve the perpetrator an abatement notice. An abatement notice may impose all or any of the following requirements (from [113]):

1. Requiring the abatement of the nuisance or prohibiting or restricting its occurrence or recurrence;

2. Requiring the execution of such works, and the taking of such other steps, as may be necessary for any of those purposes.

Anyone accused of causing a noise nuisance under the EPA has a right to appeal within 21 days of receiving the notice, otherwise failure to comply with the terms therein may result in criminal prosecution and a fine of up to £20,000. However, it should be noted that the term 'statutory nuisance' has no precise formal definition, hence it is largely at the discretion of the investigating officer to decide whether the noise constitutes a statutory nuisance - a decision which they must base on their professional experience and whatever evidence is available to them.

3.3.2 Noise and Statutory Nuisance Act 1993

The Noise and Statutory Nuisance Act was an act passed in 1993 to make provision for certain street noises which were not previously covered under the EPA. Included amongst the noises which may be considered under this act to be a statutory nuisance are loudspeakers in a street and audible intruder alarms. The act also makes provision for expenses incurred by local authorities in abating, or preventing the recurrence of a statutory nuisance to be charged to the premises to which they relate [115].

3.3.3 Noise Act 1996

A further extension to the Noise and Statutory Nuisance Act of 1993 which makes specific provisions for noise emitted from dwellings at night was introduced in 1996 under the Noise Act 1996. This also details actions to be taken regarding the forfeiture and confiscation of equipment used to make noise unlawfully and for connected purposes [116].

In cases where regulatory noise levels have not been exceeded, the accusation of statutory nuisance may still be upheld if the amount of annoyance and/or sleep disturbance affecting the complainant is deemed to be significant. Such cases rely on the experience and expertise of acoustics and environmental health professionals to determine whether or not further action needs to be taken against the perpetrator. In cases such as these where the noise is intermittent and experienced over a long period of time, or at hours outside the active local authority, it may be appropriate to rely on noise nuisance data obtained by the complainant themselves. Traditionally this was done on paper, by asking the complainant to keep a noise diary over a period of time. It is now generally more convenient to log details of any noise-related disturbances electronically. In 2012 a private environmental health consultancy launched a noise nuisance mobile phone application, which has since been published on some local authority websites (e.g. [117]). This has advantages over the paper-based method, namely that it enables the complainant to make audio recordings of the noise and provides convenient access to their local authority and other relevant information. It should be noted that such recorded evidence may only be given as evidence of the occurrence of a noise, but is not considered reliable proof of the level or spectral content of the noise. Section 7 of the Noise Act 1996 clearly states a measurement of noise made by a device is not admissible as evidence of the level of noise unless it is a device that has been approved in writing by the Secretary of State, and that any conditions subject to which approval was given have also been satisfied [116].

Figure 3.2: The Noise Nuisance 1.0 application interface. ©2012 ENcentre and Three Spires acoustics [118].

3.3.4 EU Environmental Noise Directive

The European Noise Directive (END) was adopted by the European Parliament and council in June 2002 at the recommendation of the European commission. Its overall aim is to define a common approach to avoid, prevent or reduce on a prioritised basis the harmful effects, including annoyance, due to the exposure to environmental noise [8]. The END is thus combative in its approach, being primarily focussed on reducing and restricting sound levels in populated areas, particularly sound that is generated by road, rail and air traffic, and industrial sources. Amongst its strategies are the requirements for local authorities to draw up ‘strategic noise maps’ for major roads, railways, airports and agglomerations, using noise indicators L_{den} and L_{night} (see section 3.4.2) and to devise appropriate action plans for reducing noise and maintaining environmental noise quality in places where it is considered good [119]. Another of its strategies is to provide information to the general public about noise and its effects, and where applicable, to make strategic noise maps and comprehensible action plans readily accessible.

The END also focusses on establishing common assessment methodologies for determination of L_{den} and L_{night}, although it does not set any binding limits or prescribe the measures to be taken by local authorities when devising their noise action plans. Nor do its recommendations apply to noise that is caused by the exposed person themself, noise from domestic activities, noise created by neighbours, noise at work places or noise inside means of transport or due to military activities in military areas [119].

3.3.5 Clean Neighbourhoods and Environment Act 2005

Section 7 of the Clean Neighbourhoods and Environment Act (CNEA) of 2005 [120] makes futher provisions for night time noise to be dealt with as a statutory nuisance. Prior to these amendments, the Noise Act of 1996 only applied to night noise from dwellings, whereas under the CNEA a complaint may be made by someone within a dwelling concerning noise emitted from 'any premises in respect of which a premises licence or a temporary event notice has effect' [114]. It also makes further provisions in relation to noise intruder alarms, including allowing a local authority to issue fixed penalty fines [120].

Other relevant legislation includes part 7 of the Crime and Disorder Act 1998 [121], which grants police, in collaboration with local authorities, the power to obtain an Anti-Social Behaviour Order against a resident who causes harassment or distress to others, which can be noise related. The Anti Social Behaviour Act of 2003 [122] also grants greater powers to local authorities for investigating and dealing with noise at night.

3.4 Assessment Methodologies

There are a number of standard measurement parameters used in environmental noise assessments. These have been grouped here into expressions of sound levels which have more general applicability, and noise descriptors which are specific to environmental acoustics.

3.4.1 Sound Levels

Sound Pressure

Sound pressure is defined as a local deviation from the atmospheric pressure as a result of an acoustic disturbance. It is usually expressed algebraically by lowercase p and is measured in Pascals (Pa). Sounds that are audible to humans cover a very broad range of values, from around $20\mu Pa$ ($20 \times 10^{-5} Pa$) to $20Pa$. Furthermore, on account of the logarithmic sensitivity of the human auditory system, it is generally more meaningful to refer to sound pressure in terms of an effective Sound Pressure Level (SPL). SPL is expressed in decibels which is a logarithmic unit widely used in science and engineering to express the ratio of two relative quantities - in this case, the effective sound pressure relative to the smallest change in pressure that can be detected by the human ear. A ratio in decibels is ten times the logarithm to base 10 of the ratio of two power quantities. In actual terms, the sensitivity of the human ear is something which varies between individuals as well as being dependent on both frequency and atmospheric conditions. However, based on empirical data measured using normal hearing subjects, a reference value of 20 μPa is typically used which is taken to be the threshold of human hearing for a 1 kHz tone, as observed under normal atmospheric conditions[34]. The decibel scale is therefore a convenient way to represent sound pressure levels within the human range of hearing which, due to its logarithmic basis, enables one to make predictions and make meaningful comparisons between sets of sound pressure measurements. Relative sound pressure levels are usually denoted by the letter L_p and expressed in decibels according to (3.1), in which p_1 is the measured RMS or instantaneous sound pressure level, and p_0 is the reference sound pressure level (generally taken to be $20 \times 10^{-5} Pa$).

$$L_p = 10 \log_{10} \left(\frac{p_1^2}{p_0^2} \right) = 20 \log_{10} \left(\frac{p_1}{p_0} \right) dBSPL \tag{3.1}$$

In environmental acoustics it is also common to see decibels written with a suffix indicating that a frequency weighting has been applied. The most common of these weightings is the A-weighting, defined in the International Standard IEC 61672[123] which is intended to correct for the frequency dependent sensitivity of the human ear based on equal loudness contours. A-weighting is currently the standard frequency weighting used in environmental noise measurement and is also a standard design feature of most Sound Level Meters. Other weightings include the B-, C-, D- and Z-weightings, although the B- and D- weightings are not mandated by IEC 61672. To indicate that a frequency weighting equation has been applied to a measurement, the relevant letter is usually included in brackets after the dB suffix (eg. dB(A)).

Sound Intensity

Sound intensity, I, is a vector quantity which describes the acoustic sound power (P_{ac}) per unit area (S). It is related to pressure by the velocity - also a vector quantity - and therefore has both magnitude and direction. Measuring intensity typically involves the use of two pressure transducers at a known distance apart and orientated parallel to the direction of propagation. The intensity is then obtained by averaging the pressure and derived velocity components from the two signals over the period of measurement T. For sound propagating outwards from a spherical source, the intensity at the radial distance r from the centre of the source is given as,

$$I = \frac{P_{ac}}{S} = \frac{P_{ac}}{4\pi r^2} \tag{3.2}$$

where S is the surface area of a sphere of radius r. Sound intensity is measured in Watts per square metre (W/m^2), although, for practical purposes, it is more usual to refer to the *Sound Intensity Level* (IL) - a logarithmic measure of the sound intensity and expressed in decibels. The reference value used for expressing the IL is the same as that used for measuring sound power - $10^{-12}W$.

$$L_I = 10\log_{10}\left(\frac{I_1}{I_0}\right)dB(IL) \tag{3.3}$$

Another important property of sound intensity is its proportional relationship to the square of the sound pressure:

$$I \propto p^2 \tag{3.4}$$

Note that it is possible to perform basic mathematical operations of addition and multiplication on sound intensity levels, but not on sound pressure levels.

When expressing sound levels in decibels, it is important to clarify whether the sound level is derived with respect to intensity or pressure due to the different reference values that are used. Unless otherwise specified, the SPL decibel values given in this thesis shall assume the conventional reference values of $ref = 20\mu Pa$, while IL values given in decibels shall assume a reference value of $ref = 10^{-12}W.m^2$.

Sound Power

Sound power or acoustic power P_{ac} expresses the rate of energy per unit of time and is measured in watts (W). Sound power can be considered the cause of sound, whereas sound pressure is the effect. As a dimensionless quantity, it is used to rate the sound emissions of individual sources, irrespective of propagation effects. The Sound Power Level (SWL, L_w or L_{Pac}) of a given source is a logarithmic measure of the difference between two sound powers, P_1 and P_0, where P_0 is a reference power corresponding to the threshold of human hearing. In air this is generally taken to be 10^{-12} W.m^2. In environmental acoustics the equation for sound power is written as:

$$L_w = 10\log_{10}\left(\frac{P_1}{P_0}\right)dB(SWL) \tag{3.5}$$

There are two main approaches to measuring the SWL. The first is based on sound pressure measurements taken at points surrounding a source, while the second derives the SWL from sound intensity measurements taken over an imaginary closed surface surrounding the source. Whereas the former should ideally be undertaken in an anechoic chamber, the latter has the benefit that it may be performed in situ - i.e. in the presence of background noise from other acoustic sources, provided it is relatively constant. These methods are defined in ISO 3745[124] and ISO 9614 [125] respectively.

3.4.2 Noise Descriptors

The most common of the environmental noise descriptors defined in ISO standard 1996 [126] are listed below.

$L_{Aeq,T}$ refers to the A-weighted, equivalent continuous sound pressure level averaged over the measurement period, T. Through Fourier analysis it is possible to obtain separate $L_{Aeq,T,f}$ readings for octave or 1/3 octave bands centred on f (Hz). Most sound level meters have this functionality built in.

L_{Amax} L_{Amax} is the maximum value that the A-weighted sound pressure level reaches in the measurement period. L_{AmaxF} or Fast indicates the measurement has been averaged over a 0.125 second interval, whereas L_{AmaxS} or Slow means the measurement was averaged over 1 second. The averaging interval may also be referred to as the response time.

L_{90} is the noise level exceeded for 90% of the measurement period. It is generally used to quantify the background noise level in a particular environment.

L_{10} is the noise level exceeded for 10% of the measurement period. In the UK it is commonly used in relation to road traffic noise.

L_{50} is the noise level exceeded for 50% of the measurement period and represents the median of the fluctuating noise levels.

L_{den} is a noise indicator used to predict the severity of the noise effects that are likely to be experienced by someone situated in the assessment position over a 24 hour period ('den' being an abbreviation of day-evening-night). The term 'noise indicator' is defined in the END [8] as a physical scale for the description of environmental noise, which has a relationship with a harmful effect - these being assessed by means of the dose-effect relations referred to in Annex III of the END. L_{den} is thus a measure of the daily exposure to noise, and may be calculated from the following formula:

$$L_{den} = 10\log\frac{1}{24}\left(12\times 10^{\frac{L_{day}}{10}} + 4\times 10^{\frac{L_{evening}+5}{10}} + 8\times 10^{\frac{L_{night}+10}{10}}\right) \tag{3.6}$$

wherein L_{day}, $L_{evening}$ and L_{night} are the A-weighted long term average day, evening and night sound levels respectively, derived over the daytime, evening and night time periods of a year in accordance with ISO 1996-2 [126]. The daytime period is 12 hours long (by default it extends from 0700-1900 local time), evening is 4 hours (1900-2300), and night time 8 hours (2300-0700). The addition of 5 and 10 dB at evening and night are made to allow for the increased likelihood of annoyance and sleep interference at these times due to the relatively low background noise levels. L_{night} is also used in isolation to assess the likelihood of noise-related sleep disturbance during night-time periods.

3.4.3 Noise Limits

In the UK there are no intrinsic legal limits on the cumulative or overall ambient sound levels in an environment, partly because the difficulty in enforcing such limits would mean they were of questionable value. Limits are instead applied to noise generated by specific sources, and there are various guideline limits in place which are used in relation to land-use, planning and noise insulation programs.

In relation to planning, most governments base their assessment criteria on safe limits recommended by the WHO in *Guidelines for Community Noise* [85] and sometimes those of other professional bodies such as the Institute of Acoustics (IOA) - both of which base their recommendations on internationally peer reviewed research. Current UK assessment criteria for planning is stipulated in Planning Policy Guidance 24 (PPG 24), which considers both the acceptability of a noise climate for new noise sensitive developments to be situated in, and the acceptability of a new noise source to be situated in an existing noise climate [127]. It also makes recommendations regarding the standard assessment methodology to be used, although, as is the case with the assessment criteria, these are for guidance purposes only.

While the European Noise Directive 2002/49/EC [8] does not stipulate absolute limits for outdoor noise levels, it does make a number of stipulations regarding the monitoring and reporting of areas exposed to levels above those recommended by the WHO. The guideline $L_{Aeq,24h}$ values for annoyance stipulated by the WHO have been set at 50-55 dB(A), with maximum permissible values of 65-70 dB. This is on the basis that long term exposure to $L_{Aeq,24h}$ values of 70 dB or more is considered to lead to increased risk of hearing impairment and cardiovascular and psychophysiological effects [85]. Indoors the permissible levels are dependent on the purpose of the building and the nature of the activities performed within it - for example, for moderate-high levels of speech intelligibility, indoor ambient noise limits should not exceed 35-45 dB(A), which obviously has implications in terms of outdoor permissible noise levels in the immediate vicinity of noise sensitive buildings such as hospitals and schools. At night time, indoor noise levels are a greater concern given the association with sleep disturbance. The WHO specifies that, where noise is continuous, the equivalent sound pressure level should not exceed 30 dB(A) indoors, if negative effects on sleep are to be avoided. Furthermore, to avoid the probability of noise induced awakenings, they recommend limiting the number of noise events with an L_{Amax} exceeding 45 dB. A more detailed and up to date table of WHO threshold levels for observed effects in relation to sleep are to be found in [128], although it is noted in the document that, as the evidence for some of the effects listed is limited, their values have limited weight.

Industrial Noise

With regard to industrial sources affecting mixed residential and industrial areas, absolute and/or relative limits may be adopted at the discretion of the relevant local authority depending on the type and circumstances of the assessment. Assessments are generally aimed at establishing the noise levels outside noise sensitive buildings - i.e. dwellings, offices and schools - due to noise generated by factories, industrial premises, fixed installations or sources of an industrial nature in commercial premises [105]. Where relative limits are used, these are more often than not related to the predicted increase in the background noise level (L_{90}) after the introduction of a specific source, or, in the case of bass heavy noise, relative differences in the L_{Aeq} and L_{90} in 1/3 octave bands. Absolute limits tend not to be used except where background noise levels are very low, or in relation to night-time and low frequency noise emissions, in which case they are generally determined on the basis of noise rating (NR) curves developed by the ISO and implemented in ISO 1996 [126] (see Figure 3.3). In relation to some specific problems, night-time noise thresholds may also be adopted from the German Standard DIN: 45680 [129] which is based on investigations in the region of industrial installations conducted by Piorr et al. [130]. The main way in which these criteria differ from those of the conventional assessment methods is their adjustment for the standardised A-weighting filter that is normally applied to environmental noise measurements. A-weighted measurements de-emphasise the high and low frequency components due to the varying sensitivity of the human ear with frequency which many researchers agree makes them poor indicators of annoyance in cases where there is a high amount of energy in the low frequency range [131]. The night-time (L_{night}) noise limit values given in DIN: 45680 are thus determined by extrapolating the threshold of hearing from the equal loudness contours given in ISO 226 from 20 Hz down to 8 Hz.

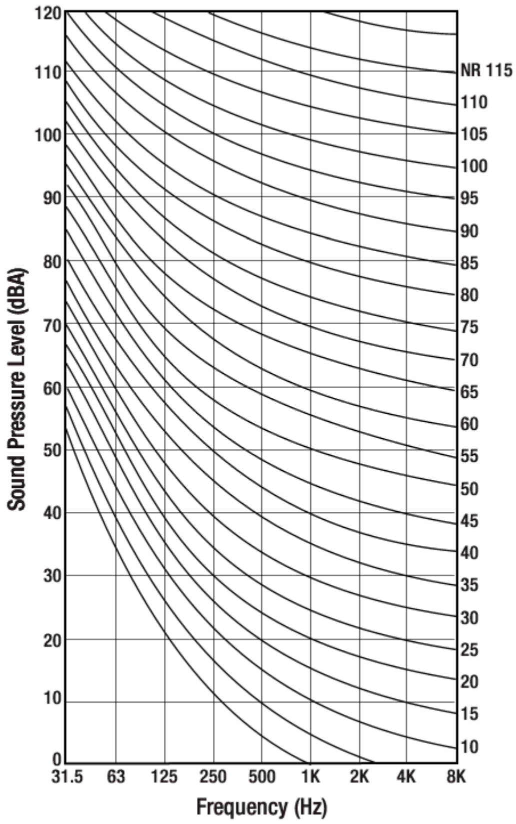


Figure 3.3: ISO 1996 [126] Noise Rating (NR) curves are used to determine the acceptable indoor environment for hearing preservation, speech communication and annoyance in different octave bands. For example, the NR 30 curve will often be adopted for private dwellings and hospitals, whereas the NR 40 curve would be more appropriate for venues such as restaurants and shops [132].

In the UK, relative noise limits tend to be used in the majority of noise impact assessments - i.e. surveys made in relation to new developments or construction work aimed at assessing whether the noise is likely to give rise to complaints from people residing in nearby buildings.

The British Standard referred to in the vast majority of assessments of this type states that an increase of around 10 dB(A) or more ‘indicates that complaints are likely’; a difference of around 5 dB(A) is of marginal significance; and a difference of -10 dB(A) or more is ‘a positive indication that complaints are unlikely’ BS 4142 [105]. The ambiguity of the objective criteria specified in BS 4142 leaves them open to interpretation and, ultimately, the decision whether or not to refuse a planning application or impose certain conditions on the applicant on this basis is made at the discretion of the local authority. For bass heavy noise such as music from pubs and clubs and road traffic noise, an additional objective parameter, L_{10}, is often used as it is believed to correlate well with perceived loudness [133]. For example, in relation to noise from pubs and clubs, the IOA recommend that L_{10} should not exceed L_{90} by more than 5dB in any 1/3 octave band between 40 and 160 Hz [133]. Craik [134] has suggested that subjective criteria is more appropriate for certain types of environmental noise such as noise generated by pubs and clubs. On this basis some councils will adopt the *inaudibility* criterion, although again this is somewhat ambiguous, leading others to adopt alternative approaches. Dibble proposes a measurement method using a 1/3 octave analyser, with a criterion that L_{Aeq} should not exceed the L_{90} in each band.

For transportation noise, including road, rail and aircraft traffic noise, $L_{Aeq,24hr}$ is the preferred objective parameter, although the percentile measures L_{10} and L_5 and peak sound pressure level L_{Amax} may also be used to assess the probability of noise-induced awakenings. With respect to a proposed housing development, the site is rated with reference to four Noise Exposure Categories (NECs). Normally this is achieved using noise contour or *noise zone* maps generated using an appropriate methodology. The noise exposure categories outlined in PPG 24 and their associated limits are given in Table 3.1. Category A indicates that ‘noise need not be considered’; B indicates planning authorities should take noise into account and require noise control measures; C indicates there should be a strong presumption against permitting the development, unless there are other important factors involved, such as a lack of alternative site; and D indicates planning permission should normally be refused. For example, in relation to aircraft noise, noise zones with an estimated $L_{Aeq,24hr}$ of $\leq 55dB$, have no restrictions on land use, 55-64 insulation measures must be on new dwellings, and for >70 dB there are to be no new dwellings [127].

It is perhaps worth noting that while guideline noise limits such as those given in BS 4142 are aimed at protecting the inhabitants of buildings from the harmful effects of transportation and industrial noise, they do not serve to protect members of the general population who elect to be within audible proximity of such sources while away from their homes or places of work. To protect members of a population from excessively loud or annoying noise and to limit overall levels of ambient noise present in an environment, various other measures are in place which relate to a variety of non-industrial sources. Where these measures fail to prevent annoyance or other health effects, the affected party may register a complaint with their local Environmental Health Service, following which the Noise Act may be brought into force [114].

Table 3.1: Noise Levels Corresponding to the Noise Exposure Categories for New Dwellings ($L_{Aeq,T}$ dB(A)) [127].

Noise	Period	Noise Exposure Category			
		A	B	C	D
Road Traffic	Day	<55	55-63	63-72	>72
	Night	<45	45-57	57-66	>66
Air Traffic	Day	<57	57-66	66-72	>72
	Night	<48	48-57	57-66	>66
Rail Traffic	Day	<55	55-66	66-74	>74
	Night	<45	45-57	57-66	>66
Mixed Sources	Day	<55	55-63	63-72	>72
	Night	<45	45-57	57-66	>66

Noise from Vehicles

Since 1970, all motor vehicles have been required under EU law to meet certain noise emission limits, and these have become progressively more stringent over the years as automotive technology has advanced. Current EU Vehicle Noise Directive 70/157/EEC stipulates a maximum sound pressure level of 74 dB(A) for passenger vehicles and 80 dB(A) for vehicles of engine power not lower than 150 kW or 81 dB(A) if they are equipped with direct injection diesel engine [135]. Measurements are currently made in accordance with ISO 10844 [136]. Similar stipulations are made in relation to other classes of vehicle. (N.B. the European Commission is currently in the process of revising the existing vehicle noise limits. Although the decision has not yet been made as to what the new limits will be, it is likely there will be a 4 dB(A) reduction for light vehicles and 3 dB(A) reduction for heavy vehicles [137].)

In the UK, the Vehicle Certification Agency (VCA) enforces noise emission based on information received from outside sources and investigations carried out by their own enforcement officers. Non-compliance with VCA regulations carries substantial penalties, including a fine of up to £5,000 and/or imprisonment for up to three months. The impetus is mainly on manufacturers, importers, suppliers and fitters of tyres and exhausts to ensure compliance [138]. Stationary noise tests may also be included in the annual Ministry of Transport (MOT) inspection in accordance with the Vehicle and Operator Services Agency (VOSA) recommendations if the examiner is not satisfied that the noise emissions meet with those specified in the manufacturer's documentation, or if the vehicle has been modified [139], although it does not seem likely that many examiners would go to any great length to fail a vehicle on these grounds due to the implications on future business.

Noise Break-out from Buildings

In the UK there are no limits for the sound insulation performance of exterior façades and windows. Only the airborne and impact sound insulation of walls and floors between habitable rooms may be required to meet certain criteria. These are dependent on the nature and situation of the building in question. In the case of residential dwellings, these are dealt with under Approved Document E of the Building Regulations [140]. In some countries a minimum value may also apply to the exterior façade, though this would be for the purpose of preventing outside-inside transmission, rather than the other way round. In the case of commercial or industrial premises, target sound reduction indices may form part of an agreed strategy, aimed either at satisfying the initial planning conditions, or conditions imposed at a later date in response to noise complaints.

Outdoor Equipment

Following recommendations outlined in the EU Directive 2000/14/EC [141], the UK government have passed the Noise Emission in the Environment by Equipment for use Outdoors Regulations Act 2001 which requires manufacturers, sellers and users of certain types of outdoor equipment such as lawnmowers, building machinery and power generators to bear the CE mark indicating compliance with EU legislation, as well as an indication of the guaranteed sound power level. Whenever a piece of equipment is placed on the market, it should also be accompanied by an EC declaration of conformity. EU Directive 2000/14/EC makes recommendations for the maximum permissible sound power levels of 22 classes of outdoor equipment which are to be be rated in accordance with the assessment procedures outlined in Annex III of the directive. The maximum permissible limits range from around 94 dB(A) to 106 dB(A) depending on the type of equipment and various other factors such as the net installed power, electrical power, cutting length (in the case of lawnmowers) and mass of the machine. There are, however, some exclusions from the directive; for example, certain devices operated by military, emergency services and police personnel may be exempt from the regulations. Furthermore, as sound power is a dimensionless quantity, such limits are poor indicators of how a sound will be perceived in real-world conditions, and are primarily in place to prevent dangerously loud devices from entering the market.

Noise from Outdoor Events

As is the case with environmental noise generated by pubs and clubs, organisers of outdoor music events have no responsibility to members of the general public with regards to noise-induced hearing loss or annoyance, yet they do have certain responsibilities toward their employees and residents of nearby buildings. In short, there is no limit to the level of environmental noise a person may willingly subject themselves to, only to that which they inflict on others. With regard to noise affecting nearby residents, the limits are typically set with reference to the Noise Council's *Code of practice on environmental noise control at concerts* [142] which stipulates relative limits that are adjusted according to the type of venue and the number of events per year. In general, for all venues with 4-12 concert days per year, the music noise level should not exceed the background noise level at a noise sensitive building by more than 15 dB(A) over any 15 minute period. It is also advised to place absolute limits on the low frequency bands 63 Hz and 125 Hz at the noise sensitive buildings as levels of 80 dB or more in either of those bands is thought to cause significant disturbance. As with other industrial sources, specific noise control measures are generally agreed between the event organisers, noise professionals and the local authority during the planning stages of an event and form part of the licensing conditions.

3.4.4 Assessment Methodologies

There are a number of standard procedures for measuring and predicting the value of a noise indicator and its harmful effects, each of which is applicable to a given scenario. Generally speaking, the methodologies used are dependent on the nature of the noise and the type of activity it interrupts. Noise may be classified as steady, non-steady or impulsive, depending upon the temporal variations in sound pressure level, all of which impinge on the appropriateness of the noise indicators and limit values adopted in the assessment procedure. Likewise the frequency and tonal content of the noise also affect the choice of assessment methodology. The following standard assessment methodologies are currently the most prevalent in UK environmental noise assessment, although the list is by no means comprehensive given there are no definitive stipulations regarding the assessment methodologies that may be used.

ISO 1999

ISO 1999 [88] outlines a method for calculating noise-induced hearing impairment for populations exposed to continuous, intermittent, or impulsive noises during working hours. Although the standard was initially intended to be used in relation to occupational noise exposure which is characterised by L_{Aeq} over 8 hours ($L_{Aeq,8h}$), studies have shown that the procedure can also be accepted for environmental and leisure-time noise exposures of adults and older children, provided the exposures are not too extreme and are expressed in $L_{Aeq,24h}$ [143]. In the Standard, the relationships between $L_{Aeq,8h}$ and noise-induced hearing impairment are given for frequencies of 500-6000 Hz, and for exposure times of up to 40 years. However, it is noted that, whereas an employee's exposure to occupational noise over a normal typical working day may be estimated with relative accuracy, estimating environmental noise exposure is more problematic. This is mainly due to high levels of variability in most people's day-to-day lives, and the logistical challenges associated with measurement over a longer period.

ISO 1996

ISO 1996 parts 1 and 2 [126] are intended as a basis for assessing environmental noise. The standard outlines procedures for determining octave or 1/3 octave band sound pressure levels by direct measurement, by extrapolation of measurement results by means of calculation, or exclusively by calculation. It also specifies methods to assess environmental noise and gives guidance on predicting the potential annoyance response of a community to long-term exposure from various types of environmental noises, although it does not specify limits for environmental noise.

As noted in a previous section, the A-weighting applied to measurements made in accordance with ISO 1996 is thought to underestimate the potential for annoyance due to low frequency noises with significant tonal components, low frequency noise of a fluctuating nature, or in areas with very low background noise levels (i.e. an unbalanced frequency spectrum) [131]. Variations on this procedure may sometimes be used in relation to specific noise problems.

ISO 9613

To predict the levels of environmental noise at a distance from a variety of industrial type sources, the standard procedure recommended by the European Parliament under the Environmental Noise Directive is that outlined in ISO 9613 [60], where the noise emission data has been derived in accordance with one of the following: ISO 8297 [144], ISO 3744 [145] or ISO 3746 [146].

ISO 9613 is based on using octave band source sound power level data to predict the equivalent continuous A-weighted sound pressure level (as described in ISO 1996) at a receiver point by the subtraction of an attenuation term, A, taking into account different meteorological conditions and various atmospheric phenomena. In the absence of barriers or other miscellaneous obstacles, the overall attenuation term A is approximated as the sum of three independent terms:

$$A_{total} = A_{div} + A_{air} + A_{env} \tag{3.7}$$

where A_{div} is the attenuation due to geometrical divergence, A_{air} is the attenuation due to air absorption, and A_{env} is the attenuation due to all other effects including ground effects, refraction due to atmospheric inhomogeneities, and scattering due to wind turbulence [69].

BS 4142

The UK has for a long time relied heavily on the British Standard BS 4142 *Method for Rating Industrial Noise Affecting Mixed Residential and Industrial Areas* [105] as a way to assess the likelihood of noise complaints. The standard first involves making L_{Aeq} measurements of all the noise in the assessment location, *including* the problem source. This is termed as the *ambient* sound level. It then involves making L_{Aeq} and L_{90} measurements of all the noise in the location *excluding* the problem source. These values are referred to as the *residual* and *background* noise levels respectively. The residual noise level is then subtracted logarithmically from the ambient sound level to give an estimation of the contribution from the problem source alone. This is termed the *specific* noise level. If the problem noise contains audible tones, impulsive noises, or if it is intermittent, it will incur a further 5 dB penalty. This yields the *rating* noise level. Finally, the rating level is compared with the background level to predict the likelihood of annoyance due to the problem source. If the rating level exceeds the background by 10 dB(A) or more this 'indicates that complaints are likely'. A difference of around 5 dB(A) is 'of marginal significance'; and a difference of -10 dB(A) or more is 'a positive indication that complaints are unlikely'.

While it is still widely used, BS 4142 has received an increasing amount of criticism from noise practitioners and noise campaigners in recent years, and is currently undergoing revisions. There are two main issues with the standard. The first is what to do when the difference between the rating and the background level is >-10 dB(A) and <5 dB(A), which is neither 'a positive indication that complaints are unlikely', or 'of marginal significance'. The second is that the standard often underestimates the annoyance caused by low frequency noises in the 20 - 160 Hz region due to the A-weighting that is applied to the measurements [147]. Moreover, the standard cannot be used when both the background noise and the problem noise levels are low, which is often the case in rural areas.

Calculation of Road Traffic Noise 1988

In the UK, the environmental assessment of road traffic noise is based on the procedures described in the publication *Calculation of Road Traffic Noise* (CRTN) [148] issued by the UK Department of Transport in 1988. The procedure applies to both measurements made manually, and those obtained through calculation using noise prediction software such as CadnaA [149]. Noise levels are expressed either as L_{10} hourly, or $L_{A10,18hr}$ dB(A).

The procedure can be broken down into five main steps which are as follows:

1. Divide the road into small enough segments so that the variation in noise level between adjacent segments is small (i.e. < 2 dB(A)).

2. Calculate the basic noise level at a reference distance of 10m from the nearside carriageway.

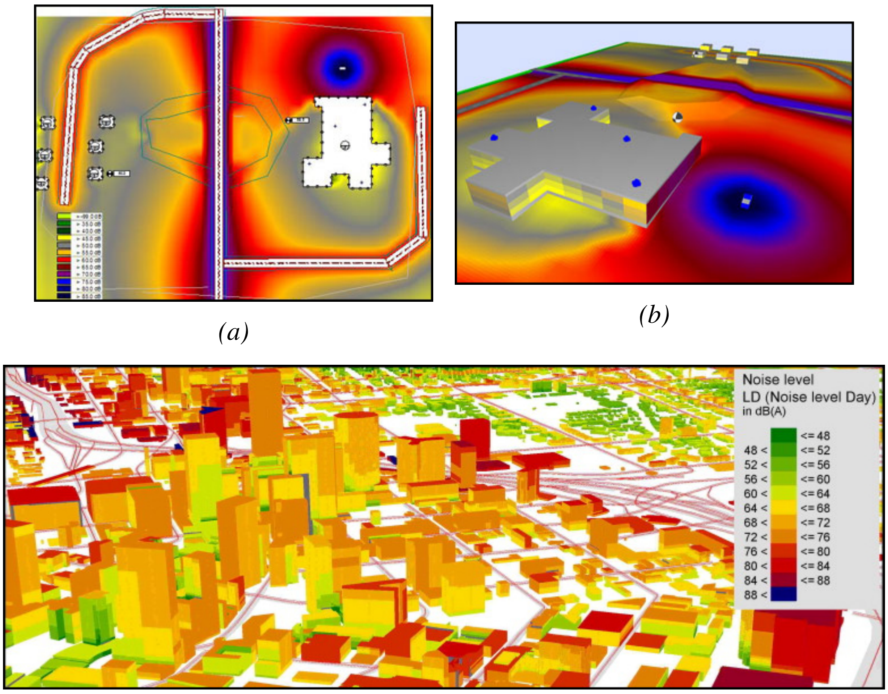
3. Assess the noise level at the receptor for each segment, taking into account the effects of distance attenuation and screening due to any barriers present. The size of each segment is dependent on the angle of view, A_s, which is defined as the angle subtended by the segment boundaries at the receptor,
$$A_s = 10\log_{10}\left(\frac{\theta}{180}\right) \tag{3.8}$$

4. Apply corrections to the noise levels at the receptor for site layout features and the size of the segment.

5. Combine contributions from all segments using simple decibel addition to give a single value of L_{10},
$$L_{10total} = 10\log\left(\sum_{i=1}^{N} 10^{\frac{L_1}{10}}\right) \tag{3.9}$$

Noise Mapping

The European Commission have defined noise mapping as ‘the presentation of data on an existing or predicted noise situation in terms of a noise indicator, indicating breaches of any relevant limit value in force, the number of people affected in a certain area, or the number of dwellings exposed to certain values of a noise indicator in a certain area [8]. The term *strategic noise map* can also refer to a map designed for the global assessment of noise exposure in a given area due to different noise sources, or for overall predictions for such an area [150]. In more basic terms, a noise map is a graphical representation of the sound pressure level (SPL) spatial distribution in a region, and is usually represented using noise contour lines - i.e. curves of a constant specified value, much like those on a topographic map. The general noise mapping procedure involves dividing the space under observation into a grid of receivers and carrying out a number of $L_{Aeq,T}$ measurements at each coordinate on the grid, ideally throughout the year and under different conditions from which the noise indicators L_{den} and L_{night} may be derived. While this data can, in theory, be determined either by computation or by direct measurement at the assessment positions, technological advancements have made it unusual to perform noise mapping by direct measurement. It is generally much more convenient to use computational tools based on noise models to generate the maps due to their higher accuracy, lower cost and the option of controlling certain parameters (i.e. times of day, atmospheric conditions etc.) to produce a large amount of yearly averaged readings.



(a) *(b)*

(c)

Figure 3.4: Examples of noise maps. *(a)* and *(b)* show aerial and façade noise maps of an industrial area produced in CadnaA [149] by Eckel et al. [151]. *(c)* is a façade noise map of the Atlanta downtown area and was produced in SoundPLAN [152] by Seong et al. [153].

While there are no strict guidelines for noise mapping, recommendations are given in ISO 1996 [154] and the *EU Good Practice Guide for Strategic Noise Mapping* [150]. The latter document stipulates that the input data used to generate the contour lines should be in the form of the noise indicators L_{den} and L_{night}. In the strategic noise mapping of UK roads, these values can also be obtained by conversion of $L_{A10,18h}$ values made in accordance with CRTN using the formulae given in [155].

In relation to aircraft noise in the vicinity of airports, various procedures have been identified. The EU recommended method, as stipulated in the Environmental Noise Directive (2002/49/EC) (END), is ECAC.CACA Doc 29 [156]. Other possibilities include an Integrated Noise Model (INM) based on the algorithm and framework from SAE AIR 1845 [157].

Regarding railway noise, the UK currently bases its assessments on the *Calculation of Railway Noise 1995* (CRN)[158]. The method was originally developed in relation to the Noise Insulation (Railways and other Guided Transport Systems) Regulations 1996 and, as such, applies only to new or improved railways. A report produced by Hardy et al[159] for Defra proposes revisions to this standard to make it more appropriate for noise mapping in line with the END.

Advanced Modelling Techniques

Whereas conventional standards such as ISO 9613 attempt to account for physical phenomena empirically, in recent years the application of advanced modelling techniques has led to significant advances in the prediction of outdoor sound propagation. In the industrial arena, such methods are more typical of other branches of acoustics such as ultrasonics, automotive and aerospace engineering owing to the scale and multiplicity of outdoor environments - i.e. the need to account for a variety of complicating factors such as ground roughness and topology, atmospheric inhomogeneity and meteorological effects. Moreover, the issue of boundary termination in open-domain problems is an ongoing challenge in relation to the wave-based numerical techniques, contributing to the higher computational costs associated with these methods. However, significant advances have been made in recent years, to the point where even real-time interactivity is a feasible goal in modern simulation engines (e.g. [160, 161]).

To give some examples, time-domain numerical solutions of the linearized Euler equations are becoming increasingly popular for studying broadband noise propagation outdoors, since they can accurately take into account the interactions of the acoustic waves with local wind and temperature fluctuations in the atmospheric boundary layer [162]. In recent years there have also been attempts to model complex outdoor environments using geometrical methods (e.g. [163, 164, 165]), taking into account diffraction effects. The parabolic equation (PE) is another popular numerical method for modelling outdoor sound propagation, particularly in relation to road and rail traffic. However, in its simplest form, the parabolic wave equation may be considered unsuitable for modelling complex outdoor environments due to its inability to take backscattering (backward reflections due to ground interference) and complex topologies into account [166]. Corrections to the equation and various hybrid modelling techniques have been proposed in recent years which attempt to deal with these limitations (e.g. [166, 167, 168, 169, 170]).

Nevertheless, despite these advances and the apparent feasibility of using numerical tools in some types of environmental noise assessment, empirical methods are still largely favoured amongst most industry professionals. One explanation for this is that they are procedurally very straightforward, and can be performed by consultants with little to no knowledge of the underlying physics. For a commercial outdoor acoustic modelling software to be a worthwhile resource in a small-medium sized acoustics consultancy, it would have to at least match the empirical methods, not only in terms of the accuracy of its predictions and ability to cope with a wide range of different scenarios, but also in terms of its ‘user-friendliness’. Furthermore, while there are no strict regulations over what assessment methods a consultant may use in their assessments, the reality is that in order to justify their actions before clients, local authorities, and, ultimately, a court of law, they rely on approval from governmental or international organisations. Finally - and often more importantly - there is a strong subjective element involved in many of the day-to-day problems encountered by those working in the field of environmental acoustics, which often requires the consultant to draw on experience gained while on site in a variety of different scenarios. This makes field work - i.e. getting first-hand experience of the problem - among the more essential aspects of the work, and implies that a highly desirable feature of any outdoor acoustic modelling application would be high quality auralisation capabilities. As yet there is no such package that meets all of these demands, although, given the level of research and development activity in this area, it seems increasingly likely that modelling and auralisation software suitable for performing a wide range of environmental noise assessments will emerge in the relatively near future. Furthermore, once endorsed by government departments, such tools could potentially negate the need for a significant proportion of on site measurement work which can be costly, time-consuming and subject to various uncontrollable factors that can impede or prevent measurements being carried out, including adverse weather conditions, site inaccessibility and hostility from members of the general public.

3.5 Environmental Noise Control

There are considered to be four principal approaches to the control of environmental noise [171]:

1. **Reduction at source** - attenuation or containment of the source.
2. **Separation** - ensure adequate distance between source and receiver.
3. **Administrative** - limiting operating time of source; restricting activities allowed on the site; specifying an acceptable noise limit.
4. **Screening** - either by natural barriers, other buildings, or non-critical rooms in a building.

Control of noise by separation is generally the least feasible of the four approaches in relation to environmental noise, as one is generally not able to control the distance between source and receiver, either because they cannot be moved, or because one or both are mobile. Reduction at source is often the most straightforward and inexpensive approach to noise control. This may be remedial - as is the case when attempting to isolate or insulate noise from industrial machinery - or it may be preventative - i.e. by designing quieter vehicles and making sure noise emitting devices such as air condenser units are well maintained. When none of the first three options are possible - as tends to be the case where traffic noise is concerned - then screening may be a feasible alternative. In recent years, there has been a significant growth in use of noise barriers, reflecting the mounting concern with respect to noise pollution and the need to find methods of noise control that will not inhibit industrial or economic development.

3.5.1 Noise Barriers

The contribution to overall attenuation during the propagation of sound outdoors due to the insertion of noise barriers is dealt with in ISO 9613 by the inclusion of a screening algorithm as a portion of the standard's overall attenuation term, A [69]. This method is based on the work of Maekawa [59] and uses straight ray paths to predict the insertion loss (IL) of the barrier due to acoustic shadowing. For a basic description of this method the reader is referred to section 2.4.2. A more precise way to assess the IL of barriers in-situ is given in ISO 10847 [172]. However, this method is not suitable for predicting the performance of barriers prior to installation. A more up to date standard was recently published by the European Commission and has been implemented in the UK under BS EN 1793 [173]. This standard specifies two methods by which a single number rating for airborne sound insulation is derived from A-weighted measurements made before and after insertion. The first is the direct method which can be only used if the barrier has not yet been installed or can be removed for the 'before' measurements. The second method is the indirect method using measured 'before' levels at an equivalent site.

Needless to say the empirical methods require an actual barrier or prototype from which to obtain the 'after' levels, which is not always ideal in the research and development of new types of noise barrier given the costs involved in their construction and, where applicable, in accessing adequate testing facilities. The ability to make accurate predictions of barrier performance prior to construction is therefore highly desirable - something that increased computational power has helped to progress considerably in recent years. This, along with mounting concern over noise pollution, has lead to the development of a wide range of different noise barrier systems, some of which offer various advantages over the traditional straight-edge or wedge shaped barriers. Furthermore, advanced analytical and numerical modelling methods can facilitate the prediction of various diffraction and atmospheric effects that are neglected by simple ray-based models like that used in ISO 9613. For example, atmospheric turbulence is an important factor affecting the performance of noise barriers due to its tendency to scatter sound into the shadow zone [58].

Strategic Noise Barriers

Theories related to a straight-edge vertical barrier have been presented in Section 2.4.2. This can be considered the simplest form of noise barrier, the only major design requirements being that it is tall enough and long enough to maximise the shadow zone, and sufficiently dense and rigid to suppress structural resonances. In practice the design has a number of disadvantages. For example, it is often highly reflective which can have the effect of increasing the sound levels on the near side. What is more, to provide effective screening the barrier often needs to be of a height and thickness that is impractical to build and visually intrusive. To compensate for these shortcomings, efforts have been made to optimise the design, leading to various different types of strategic noise barrier.

Due to their multiplicity, strategic noise barriers are classified in the literature in a number of different ways. For example, Garg et al. [174] have proposed the design morphology for noise barriers shown in Figure 3.5 which serves to illustrate the level of variation in their design and construction. Four of the most common classes of strategic barrier design shall be considered here. These are the multiple-edge, random-edge, dispersive and absorptive types of noise barrier.

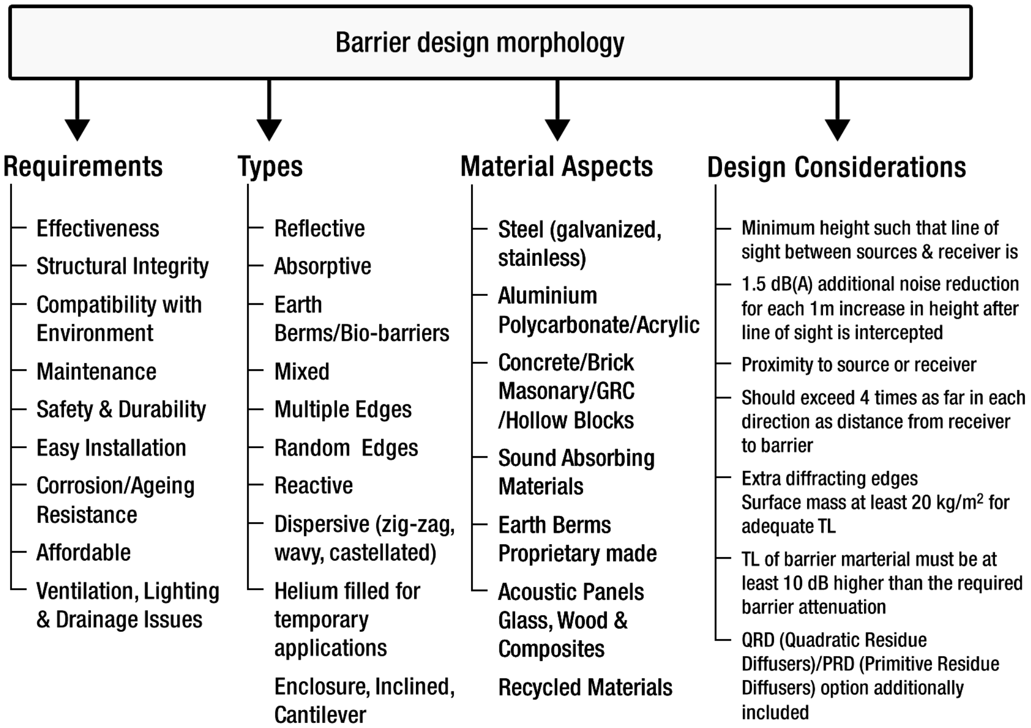


Figure 3.5: A design morphology for sound barriers [174].

Multiple-Edge Noise Barriers

When sound is incident on a noise barrier or acoustic screen, some of that sound reaches the acoustic shadow zone by diffraction or scattering around the barrier edge (see Figure 2.35). Efforts to improve the performance of noise barriers are therefore often aimed at refining the top profile to maximise the attenuation through diffraction. The benefits of adding additional diffracting edges has been demonstrated by the Y-profile [175], T-profile [176], various multiple-edge [177] (Figure 3.6) and fan-shaped head barriers [178] among others.

Random-Edge Noise Barriers

By redirecting incident sound into the shadow zone, the edge of a thin rigid barrier effectively acts as a line source [179]. In the case of a straight-edge barrier, the source is coherent - i.e. can lead to the formation of a continuous pattern of sound waves - meaning there is the possibility that in some areas behind the screen, sound levels are increased rather than decreased. One way to reduce the coherence of the source and thus improve the barrier performance is by cutting irregularly shaped notches from the edge of the barrier. This form of barrier is known as a *random-*

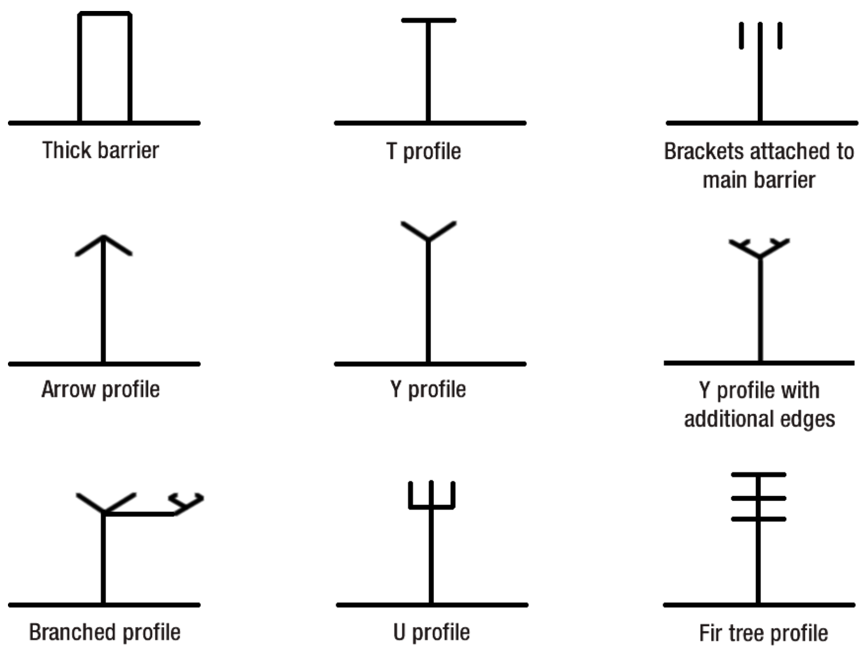


Figure 3.6: Vertical cross-sections of multiple-edge noise barriers with different top profiles. Adopted from [22].

edge or *jagged-edge* noise barrier and has been found to offer improvements of up to 6 dB over straight-edge barriers [179, 180, 181].

Dispersive Noise Barriers

Reflections from flat screen barriers can be problematic, particularly when the barriers are erected in parallel to each another or to nearby buildings. A way to reduce this problem is to make the surface of the barrier irregular in order to disperse the reflections in different directions (e.g. [182]). Some examples of dispersive barriers are pictured in Figure 3.7.

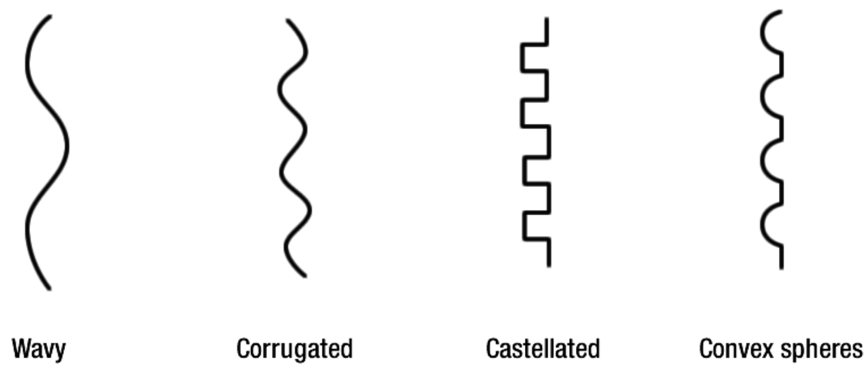


Figure 3.7: Horizontal cross-sections of dispersive noise barriers with different surface irregularities.

Absorptive Noise Barriers

Absorptive noise barriers are designed for two purposes. Firstly to reduce reflections from the barrier surface; and secondly, to maximise attenuation by reducing the energy of incident sound waves. Normally this involves treating the surface of the barrier with some form of absorbing material, although other forms of passive absorptive barriers have also been investigated. For example, Mizuno et al. introduced the concept of a phase-interference type barrier. This device consists of an open-topped box section fitted to the top of a vertical barrier in which a number of passages of different lengths are cut at regularly spaced intervals forming a structural delay circuit [183, 184]. The basic operating principle relies on the assumption that sound waves entering the device will be refracted internally resulting in a phase shift. Upon exiting the passages, the phase-shifted waves will then interfere destructively with the sound propagated over the barrier, resulting in additional sound reduction.

The physical principles behind phase-interference barriers are not wholly dissimilar from another form of noise barrier to have received interest in recent years known as a *sonic crystal* noise barrier. This form of noise barrier is quite unique as, while most noise barriers have a reputation for being rather unsightly, the discovery of the sonic crystal noise barrier came about when the scientist Martínez-Sala observed some interesting acoustic effects in the vicinity of a sculpture by the Spanish artist Eusebio Sempere [185]. Upon further investigation, Martínez-Sala was able to reveal that, with his somewhat bizarre construction of polished stainless-steel tubes mounted on a rotating base, Sempere had inadvertently produced the world's very first sonic crystal noise barrier.

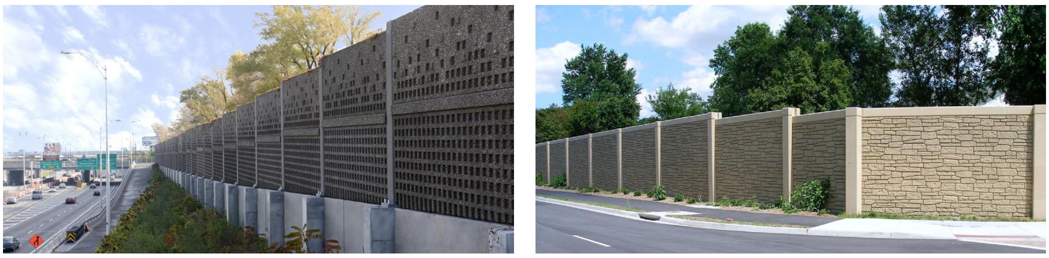


Figure 3.8: Examples of sound absorbing noise barriers. Images used with permission from Whisper Wall [186].

Sonic Crystals

Belonging to the class of acoustic meta-materials, sonic crystals are artificial composite materials consisting of periodically spaced arrays of solid inclusions embedded in air [187]. Much of the theory and terminology used in relation to these structures originated within the fields of electromagnetics (e.g. [188]) and crystallography (e.g. [189, 190]) yet is largely applicable in the acoustic case. The concept of a 'sonic crystal' stems from the basic mechanical principle that waves propagating within a crystalline structure may be scattered through interaction with atoms or elements in the material, leading to positive and negative phase interference. This type of wave phenomenon was studied intensively by father and son William Henry and William Lawrence Bragg who observed that, depending on the lattice configuration and direction of the incident waves, either or both of the following scenarios may ensue: intense peaks of reflected radiation at certain frequencies; a pattern of standing waves signifying a band gap in the transmitted signal [189]. This observation led to the formulation of Bragg's Law - a formula expressing the relationship between the wavelength of the incident wave, the spacing between the lattice planes, and the angle between the incident wave and the scattering planes (3.10).

$$2d \sin\theta = n\lambda \tag{3.10}$$

where n is an integer determining the reflection order, d is the spacing between the lattice planes (see Figure 3.9), λ is the wavelength of the incident wave and θ is the scattering angle. Bragg's law adheres to Rayleigh scattering theory, which assumes that scattering elements are small in relation to the wavelength.

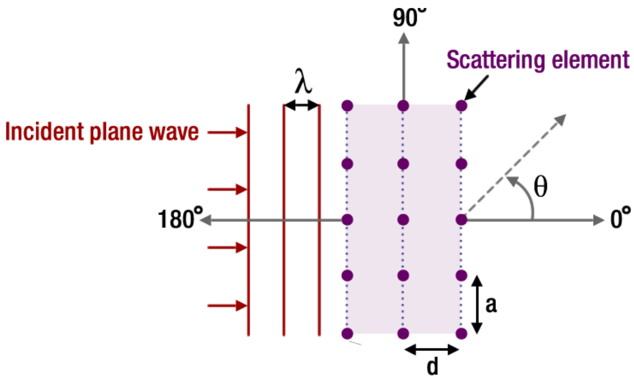


Figure 3.9: Parallel planes in a square 2-D lattice. Peaks and dips in transmitted signal occur when the angle of incidence is equal to the scattering angle and the path length difference between successive rays is equal to an integer number of wavelengths.

While the analogy can be extended into an arbitrary number of dimensions in Euclidean space, the dimensionality of a sonic crystal generally refers to the number of dimensions in which the structure has periodicity (see Figure 3.10).

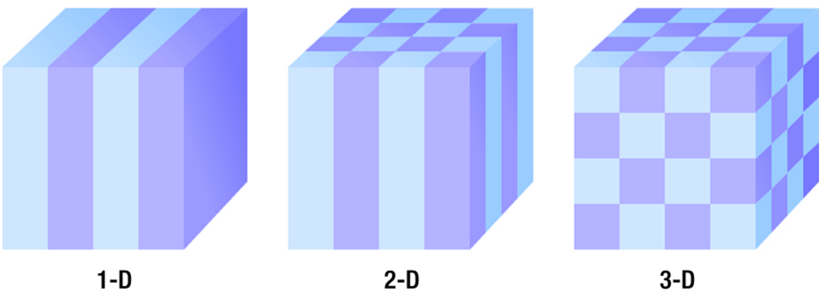


Figure 3.10: 1- 2- and 3- dimensional periodicity in a cubic crystal lattice.

The Miller indices are a system of notation used in crystallography to denote the crystal directions and family of lattice planes for three-dimensional crystals [191]. Each set of planes is determined by the three integers h, k, and ℓ, the Miller indices, wherein each index relates to a fraction of the unit cell edge that is intersected by the set of planes [192]. So, in the case of a plane which lies parallel to a cell edge and thus never intersects it, the corresponding Miller index is 0. One can draw a number of planes in a simple cubic structure but the most common are [100], [110] and [111] [193].

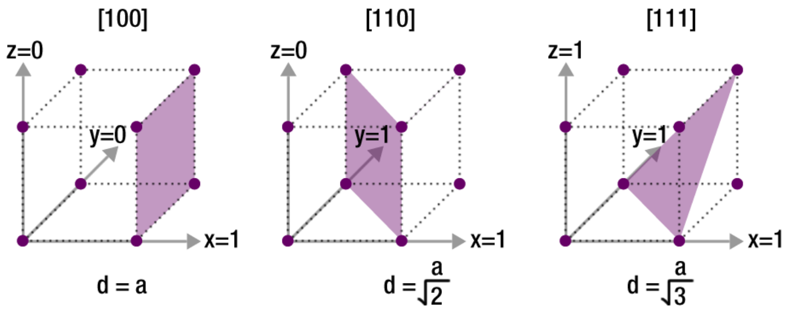


Figure 3.11: Common planes in a simple cubic structure. Adapted from [193]

For cubic crystals with lattice constant a, the perpendicular distance d between adjacent Miller planes may be derived from the Miller indices by the following:

$$d_{hk\ell} = \frac{a}{\sqrt{h^2 + k^2 + \ell^2}} \tag{3.11}$$

Since Martínez-Sala et al.'s fortuitous discovery, sonic crystals have been investigated by scientists and artists with a mutual interest in the soundscape, both separately - as in the work of Perez et al. [194] - and collaboratively. For example, Crow and Prior's widely acclaimed *Organ of Corti* was produced in collaboration with Keith Attenborough, Professor of Acoustics at the Open University (UK) and a key figure in sonic crystal research (e.g. [195, 196]). Crow and Prior's work consists of 4 metre high sonic crystal composed of acrylic tubes and is described by the artists as an 'experimental instrument that recycles noise from the environment' [197].

(a) Órgano - a sculpture by Eusebio Sempere widely regarded as the world's first sonic crystal. (Image ©2009 Nature[198])

(b) **The Organ of Corti** - sculpture by Dr David Prior and Frances Crow (Image ©2012 Liminal)

Figure 3.12: Sonic crystals in art.

3.6 Summary

This chapter delivered a broad overview of current environmental noise control regulations and recommended practices. The final section of the chapter introduced the four main approaches to environmental noise intervention, with particular attention given to noise barriers. Among the types of noise barrier discussed was a novel form known as a sonic crystal noise barrier which is to become a significant element later on in this thesis.

Whilst the information presented in this chapter is by no means an exhaustive review of the large amount of scientific and legal literature connected to environmental noise, it does serve to highlight the breadth and complexity of the subject, and the potential for caveats in terms of its regulation. It is also noted that the aim of the legislation is to keep sound pressure levels within the limits recommended by medical and environmental health experts, though preferably not at the expense of industrial, social or economic development that is a benefit to society. Regarding community and industrial sources, the environmental noise legislation is generally quite flexible, and it is rare that a court will order complete cessation of a noisy activity. Furthermore, these sources are less of a wide-spread threat to public health and can be treated relatively easily, either by limiting operating hours, or with remedial acoustic treatment. What has emerged from the literature as being a far greater threat is traffic noise. The scale of the problem makes it one that is far more challenging in terms of its mitigation.

It has also been found that research into forms of environmental noise treatment such as highway noise barriers is predominately motivated by strong negative attitudes toward man-made sound. The popular model of man-made sound as a pollutant has ensured that governments, urban planners and acoustic practitioners are primarily concerned with reducing sound pressure levels to the greatest extent possible. However, there is evidence that attitudes are slowly shifting in favour of a more holistic and sustainable approach to environmental noise control, a good example of this being the considerable amount of effort that is invested in developing new forms of noise reducing devices such as sonic crystal noise barriers. Such advances would seem to suggest a possible fifth approach to environmental noise control - that of *noise augmentation*. By this the author is referring to measures that would aim to enhance or improve the subjective sound quality in noisy environments, without necessarily seeking to alter the intensity of the sound in places where it is considered to be within safe limits.

4

Soundscape

4.1 Introduction

For several decades there has been growing concern over the effects of human-made sound on the health of the planet and its various inhabitants. Along with light pollution and global warming, environmental sound is another inevitable side-effect of population growth, technological evolution and urbanisation, and while it lacks the urgency of some of the more notorious environmental health concerns, it has nevertheless provoked its own surge of scientific activity. Yet the scope and complexity of the environmental noise problem make it one that is difficult to solve, particularly as existing approaches aimed at sound pressure level reduction are rarely compatible with the ever-increasing demands of modern life. It is thus apparent that new and complementary approaches are needed - ones that seek to understand and to work with, and not necessarily against, the sounds present in an environment.

Consequently, a multidisciplinary field of research to have emerged in the past quarter of a century, and still considered by most to be in its infancy, is that of soundscape research. Present-day soundscape research is the product of a convergence of fields, principally those of acoustic ecology, musicology, sociology, psychology, architecture and acoustics, and this convergence would appear to be the result of a shared philosophy or intellectual interest amongst individuals. This shared philosophy is one that takes an holistic view of the acoustic environment, considering all sound in a space - desirable or otherwise - to be part of what characterises it, alongside the visual landscape and other distinguishing features (i.e. meteorological, ecological, anthropological etc.). A soundscape researcher's approach to environmental noise control is thus one which addresses more than sound pressure levels, but also looks at how a given acoustic environment is perceived by a population, as well as considering its wider impact on society and the ecosystem.

It is often argued that the soundscape - considered here as all sounds characteristic of a community or outside space - has an even greater impact on human health and well-being that the visual landscape. This may relate to the fact that the visual, being relatively static and easy to disengage from, occupies a more peripheral field, and thus lacks the physical potency of sound.

Furthermore, while it is neither possible nor realistic to disassociate the emotional from the physiological, by attempting to ignore the aesthetics of environments and viewing sensory input from a purely physical, cause and effect standpoint, it is clear that sound has more immediate and measurable physiological effects. Yet, in spite of this, humans are often concerned first and foremost with the appearance of places, hence it is hardly surprising that they are generally much less active in terms of their aesthetic involvement with the soundscape, whether with the intention of creating a more positive aural impression, or of conserving an already pleasant one for future generations. Whilst a complete understanding of the soundscape is very likely beyond human reach, given it is as much a philosophical edifice as it is a science, the collective knowledge that has so far been amassed in different fields has already reached the stage where its implementation in urban design is feasible.

The focus of this chapter is twofold: firstly the author examines the history, meaning and significance of soundscape research in a broad social and environmental context. In doing so, they hope to demonstrate how the concept has succeeded in penetrating the physical sciences, despite being founded on ideas and techniques more common to art, philosophy and the social sciences, and why this is justified. Secondly the author presents an overview of existing soundscape analysis and design practices, considering their compatibility with environmental noise methodologies.

4.2 History

Soundscape was originally a concept introduced by Canadian composer, environmentalist and music educator Murray Schafer in the 1970s, referring then, as it does now, to all constituent sounds of an environment, whether man-made, animal or natural [199]. A contemporary of Schafer, the acoustic ecologist and composer Barry Truax, defines the soundscape as an environment of sound with an emphasis on the way it is perceived and understood by individuals or a society [200]. According to Truax, the term can be applied to artificial environments and abstract constructions, as well as actual environments. Usage of the term may also be extended to fields such as biological acoustics, whereby the presence and perceptions of humans are not integral to its analysis. Here, the soundscape is considered in its broadest sense as the aural equivalent of landscape, thus encompassing all kinds of acoustic phenomena present in an environment.

In an attempt to draw attention to the sonic environment and the social effects of noise pollution, Schafer established the World Soundscape Project (WSP), an educational and research group based at the Simon Fraser University. The group attracted several young composers, who sought to disseminate the soundscape through contemplative sound art. Since then the term has found its way into a wide variety of disciplines, from the quantitative (e.g. engineering and physical sciences), to the qualitative (social and ecological). The Vancouver Soundscape project was a series of soundscape compositions produced by Schafer on behalf of the WSP at the Sonic Research Studio of Simon Fraser University [201]. In the accompanying commentary Schafer refers to ‘good and bad acoustic design in Vancouver’ [202], which is perhaps the first documented evidence of the idea that environmental sound could be ‘designed’ rather than being impressed upon a population of passive recipients. The pieces are composed entirely of sounds he recorded in-situ between May 6th and June 5th 1996. Throughout the seventies and eighties, soundscape research was predominantly the remit of the musicologists, although it gradually accumulated broad outside interest following the publication of several seminal texts by Schafer and Truax, most famously *The Handbook for Acoustic Ecology* [203], and *The Soundscape* [204]. It was around this time that the World Forum for Acoustic Ecology (WFAE) was established in 1993. The organisation describes itself as 'an international association of nine affiliated organisations that share a common concern with the state of the world's soundscapes' [205]. Their annual publication *Soundscape: Journal of Acoustic Ecology* [206] pulls together current soundscape research from around the world and from across disciplines.

Schafer posited that there are three main classes of sound constituting any given soundscape: *key notes*, *sound marks*, and *sound signals* [207]. According to Schafer, keynotes are those sounds 'which are heard by a particular society continuously or frequently enough to form a background against which other sounds are perceived. Examples might be the sound of the sea for a maritime community or the internal sound of the combustion engine in the modern city.' [207] Sound signals, by contrast, are sounds to which the attention is particularly directed. In soundscape studies sound signals are contrasted by keynote sounds in much the same way as figure and ground are contrasted in visual perception [207]. In essence, the key notes give us the basic perspective, the 'map' of the situation, and guarantee its continuity, while the sound signals are isolated fragments that float freely in the universal medium of the sound aquarium [208]. These are in fact the words of Slovene philosopher and cultural critic Žižek, who described the role of soundtrack in cinema as the medium of 'the real' in which the symbolic - the imagery - is supported. Yet the same also applies to the soundscape when one considers sound signals as the signifiers, and key notes as the continuous backdrop creating the sense of time, place and context. Sound marks are somewhere between these two extremes. They are unique sounds 'imbued with strong associations within a shared acoustic community' [209] - the Minster bells in York for example.

More recent contributions to soundscape research have taken a more objective approach to improving our understanding of sonic environments. In response to the The publication of the EU Directive Relating to the Assessment and Management of Environmental Noise (END) in 2002 [8] (see Chapter 3) the European Co-operation in Science and Technology (COST) launched Action TD0804: *Soundscape of European Cities and Landscapes* [210]. The main aims of TD0804 were 'to provide the underpinning science for soundscape research and make the field go significantly beyond the current state-of-the-art, through coordinated international and interdisciplinary efforts', and 'to promote soundscape into current legislations, policies and practice, aiming at improving/preserving the sonic environment [86]. Within the documentation detailing the proposed action, COST outlines the importance of a multidiciplinary approach to soundscape:

> Reducing sound level, the focus of EU environmental noise policy, does not necessarily lead to improved quality of life in urban/rural areas, and a new multidisciplinary approach is essential. Soundscape research represents this paradigm shift as it involves not only physical measurements but also the cooperation of human/social

sciences (e.g. psychology, sociology, architecture, anthropology, medicine), to account for the diversity of soundscapes across countries and cultures; and it considers environmental sounds as a 'resource' rather than a 'waste'[86].

2009 saw the culmination of the 'Positive Soundscape Project' (PSP), a three year collaborative research effort funded by the Engineering and Physical Sciences Research Council (EPSRC) which sought to re-evaluate environmental sound. What is significant about this project, is that it gave formal recognition to soundscape research as a multi-disciplinary science, as opposed to the small and isolated area in the arts where it began. In fact, much of the PSP was dedicated to developing new empirical techniques for soundscape design and evaluation. The group's final results and findings were communicated at the 'Applied Soundscapes Symposium' held in September 2009, and some of the papers presented at the symposium were later published in a special edition of the Applied Acoustics journal in 2011 [211]. Since the PSP came to an end, soundscape research has continued to flourish as a science, becoming what is now a well established branch in acoustics, as well as maintaining a strong presence in other disciplines. In the Acoustics 2012 conference held in Nantes, an entire section of the conference was dedicated to soundscape, which, as keynote speaker Truax remarked upon at the time, was quite extraordinary given the idea originated with a small group of artists in Vancouver.

4.3 Philosophy

4.3.1 A Sense of Place

> The thin grasses, more or less coating the hill, were touched by the wind in breezes of differing powers, and almost of differing natures - one rubbing the blades heavily, another raking them piercingly, another brushing them like a soft broom. The instinctive act of humankind was to stand and listen, and learn how the trees on the right and the trees on the left wailed or chanted to each other in the regular anti-phonies of a cathedral choir; how hedges and other shapes to leeward then caught the note, lowering it to the tenderest sob; and how the hurrying gust then plunged into the south, to be heard no more.
> - Thomas Hardy, *Far From the Madding Crowd*[212].

It is often felt that vision eclipses sound as a primary sensory mechanism, leaving sound to play a peripheral role in the perception of multi-stimuli environments. Yet, in the above passage, it is sound and motion breathing life into Hardy's landscape while the visual merely serves as a backdrop; a stage that remains set long after the actors have left. Though one is not always aware of it, sound is at least as well disposed as vision to facilitate interaction between humans and the physical world. After all, imagery is very often static, merely hinting at what has already happened or could conceivably happen in the future, whereas sound is a by-product of a physical event, and an indicator that *something is happening now*. Thinking in particular about urban outdoor spaces, where events happen with a frequency and urgency that demand attention - whether or not it is given willingly - sound is thus a constant and critical source of feedback from the environment; an unbroken dialogue between the space and those who inhabit it. It is this *dynamism* that distinguishes auditory perception from vision, in its ability to convey the progression of time and to stimulate changes in one's emotional and physiological response patterns, often to a more palpable extent than is achieved by vision. Sound is thus considered an emotionally-rich stimulus, as opposed to vision which tends to be more information-rich [22], and, like the thoughts and feelings that accompany us unremittingly throughout our lives, so innate is it to our existence, that an absence of sound can be profoundly disconcerting.

This leads one to consider the importance of auditory feedback in creating a 'sense of place' - a phrase commonly used by environmental psychologists and cultural geographers amongst others to describe a phenomenon whereby spaces are imbued with meaning to an individual or to a particular group of society. While the sense of place can mean different things to different people, it tends to relate to the perception of an identity - i.e. unique characteristics which make an environment different from anywhere else. This may be grounded in the physical geography of the environment, its cultural or natural history, or the feelings of an individual. Likewise, it can occur unconsciously - for example, children develop their own sense of place through play and the influence of cultural and family traditions [213]; or it can be attained intellectually. For example, Seddon speaks of his search for a sense of place in a harsh and featureless environment which he comes to realise through scientific observation and analysis [214]. Sense of place is also considered here to be an essential step in a process that may ultimately lead to a relational connection with a space - spaces having the potential to bring a person 'outside of the self', away from the noise of 'inner space'. For this to happen, a space needs to have its own identity, as one cannot connect to something that is not perceived to be separate from the self.

Sense of place is considered pertinent to modern soundscape research, as this is a field which also seeks to build positive associations in places where they are lacking, through a combination of scientific and philosophical processes, and yet it is not something that is altogether new. The sense of having an intrinsic connection or affiliation with an environment, whether past or present, is something which has both fascinated and perplexed philosophers for many centuries and is common across all cultures to a variable extent. Whether a person is aware of it or not, maintaining a positive connection with their environment can be enormously beneficial at both the individual and societal levels, promoting a sense of stability and 'oneness', and other feelings associated with general well-being. This is an idea that is supported by numerous psychological studies that have attempted to unravel the complex relationship between environment and mental health (e.g. [215, 216, 217, 218]). One particular empirical study performed by Weinstein et al. [219] provides evidence that environmental perception not only affects how people feel, but may also affect how people act. The experiments performed in the study involved exposing subjects to images of natural and man-made environments and getting them to perform certain cognitive tasks. It was observed that the different environments not only affected how the participants felt, but also affected significant differences in their attitudes and behaviour. While these particular studies were aimed at observing the psychology of natural environments in contrast to man-made ones without any direct connection to soundscape, they do serve to highlight the extent to which humans are influenced by their environment - of which sound is undoubtedly a significant part. What is also interesting about this particular experiment, is the fact that the participants were not presented with auditory or other non-visual stimuli, but encouraged to imagine the sorts of sounds and smells they might encounter in each of the spaces. It is supposed that this approach might lead to an exaggerated estimation of the differences between the environments, due to a likely discrepancy between subjects' expectations and what they might actually experience under real conditions. Whereas relying on memory in order to stimulate a somewhat artificial and exaggerated impression of an environment could be seen as a positive when the goal is to understand the psychology behind sensory experience, it is important to distinguish between expectation and real experience in relation to soundscape perception.

Nevertheless, the apparent preference for natural environments is something that has been corroborated in numerous soundscape related studies which have repeatedly shown that natural sounds such as waterfalls and birds tend to be preferred over man-made sounds (e.g. [220, 221, 222]). This is further evidence that the creation and conservation of green spaces in urban environments is a mutual concern across disciplines, despite the underlying psychology being less well understood in some fields than in others. One suggestion is that the preference may be related to what in philosophical terms can be referred to as a greater sense of personal autonomy, or the tendency for natural environments to bring people closer to their 'authentic selves' - i.e. stripped of the pressures and frustrations associated with structured environments which are thought to alienate people from themselves and each other[219]. However, there may be a danger in drawing too simplistic a conclusion from the results of some studies, namely that 'natural' is 'good' and man-made is 'bad'. Whereas in the short-term nature may well be the 'safe bet', there is a hidden layer of complexity that science has barely begun to expose. This is related to the fact that reactions to a particular environment are tightly bound to a whole host of other variables, including, for instance, current mood and situation, past experience and cultural attitudes. In sociology, the often subtle and intimate relations between environment and the human condition are frequently linked to the concept of *habitus* - the set of dispositions that one may hold individually or in groups or in common with members of various groups or strata, and which are mutually sustained by the built environment, though neither has complete control over the other [223]. This could be related to the fact that perceptions are tied, not only to the present, but to a culmination of past events and experiences which together define both the individual and collective *habitus*. As Bourdieu writes,

> 'The habitus - embodied history, internalised as second nature and so forgotten as history - is the active presence of the whole past of which it is the product.' - [224]

The concept of a collective *habitus* as a condition which evolves over time could have different implications for those involved with soundscape. If the absence of structure and formalism in natural environments is to be construed as an *escape from habitus*, then getting as far 'back to nature' as possible in the design of urban spaces seems entirely logical. Alternatively, if the tendency of a population to rank 'natural' artefacts over man-made ones is construed as an *embodiment of habitus*, then, whilst durable, it is an attitude that could conceivably change - albeit over a period of time. The author finds this an interesting proposition, and one that has a certain relevance in relation to soundscape research. It also seems that the way to test it, is to adopt a stance which opposes the notion of a fixed disposition toward natural environments, and to try to stimulate a collective shift in attitudes - or at least the very beginnings of one. From looking at how architectural features, art, and fashions have evolved over time, it seems clear that humans are indeed capable of adapting their aesthetic preferences and tolerances in tune with their moral and intellectual *habitus* - why should this not also apply to the sonic environment? In fact there is already plenty of evidence in the media that changing attitudes toward environmental sound reflect the changing values of communities - a factor that can add to the fervency of noise-related disputes. For example, sounds that have historically been a familiar and reassuring presence in the soundscape, such as the crowing of a cockerel, church bells and clock chimes, are often perceived as annoying by visitors and newcomers to an area (e.g. [225, 226, 227]). This has resulted in mitigation of sounds that would at an earlier point in history have played an important role in our lives but which have now taken on different associations. This also raises the issue of environmental sounds being part of our natural and cultural heritage. There are many who feel that, as these changes occur, efforts should be made to document and preserve elements of the soundscape for posterity (e.g. [228, 229]).

4.3.2 Compatibility of Different Perceptual Models

Auditory perception is broadly acknowledged as something which extends beyond the physiological processes of sensation, to being something which is inextricably tied to the 'self' - i.e involving the thoughts, feelings and the personal and collective habitus of the individual. In other words, perception can be considered a philosophical construct, in the sense that it exists in the mind of a subject, as opposed to 'real' objects which exist outside the mind [230]. A percept is thus defined as a concept or impression that is developed through recognition by the senses of some external object or phenomenon [231]. This suggests it is something transitory, on account of the continuous flow of sensory information connecting humans to the physical world. It also implies there is more to perception than the passive conversion of physical stimuli into sensory-neural information; it implies something of an intellectual gnosis of the events which have taken place. How one listens to sound affects how one perceives it - an idea that is common across disciplines, albeit expressed in different terms. Among the first to realise this was Pythagoras, who famously delivered lectures from behind a curtain so that his pupils might better concentrate on his teachings [232]. The idea of abstracting sounds from their causes in order to facilitate a heightened appreciation of the aural dimension became a popular theme in electroacoustic music in the 1940s, leading to the development of Musique Concrète [233]. This style combines elements of music with 'acousmatic' sounds - i.e. sounds one hears without seeing - which can be synthesised, derived from instruments, objects or voices, or recorded in the environment. In electroacoustic music, the loudspeaker is often thought of as the Pythagorean 'veil', isolating perception of sound from other senses [234].

Schaeffer [233] extends the acousmatic theory to define four modes of listening - 'les quatre écoutes' - which he uses to differentiate between the various ways in which humans perceive sound, between the conscious and the subconscious, understanding and reasoning [235]. The idea which has become ubiquitous in musicology is not inconsistent with a popular perceptual model proposed several years earlier by the psychologist Ernest Schachtel [236]. Schachtel posited that there are two modes of perception: the autocentric and allocentric. Autocentric perception is said to be subject-centred, in that it concerns the thoughts and feelings of the individual. In this sense it is more readily associated with sensory quality and pleasure. Reflecting on Schachtel's model in relation to popular soundscape terminology, one might consider the autocentric mode is applicable to the perception of keynotes [204] or background ambiences in a sound environment - i.e. sounds which, though not always heard, contribute to the impressions one unconsciously associates with a place. The second mode of perception, allocentric, is said to be object-centred and thus concerned with objectification and transference of knowledge. Allocentric perception might then apply to the sound signals in a soundscape, which involve attention and directionality.

While it is difficult to substantiate such theories in scientific terms, there are a number of studies which support the view that voluntary and involuntary shifts in auditory attention do indeed alter the perception of sound. For example, recent empirical studies in the fields of psychoacoustics and neurology have observed changes in cochlear response in relation to focused auditory attention [237]. Sussman et al. have also attempted to explain why our experiences of the environment are changed by attentional processes, by measuring the on-going electrical activity of the brain in a series of Event Related Potential (ERP) and Event Related Field (ERF) experiments [238, 238]. The results have indicated that auditory attention not only modulates the grouping and storage of sound representations in the auditory cortex, but also affects the accuracy for discriminating stimulus information.

4.3.3 Connection With Environmental Aesthetics

It is evident that many of the ideas motivating soundscape research are reminiscent of a sub-field of philosophical aesthetics known as environmental aesthetics. Among their similarities is the notion that human-environment relations are gradually deteriorating in the prevailing climate of constant consumption and waste - to which noise is a significant contributor - along with the raft of social problems still faced by more economically developed countries (MEDCs). In his comprehensive essay on the subject, Porteous writes,

> [The environment] is clearly in danger unless we change our lives of endless having, the development of a better appreciation of environmental aesthetics is one step toward the restoration of world-order and earth-order [5].

The study of soundscapes could be seen as a movement toward this, although it is important to underline its contribution need not be limited to the creation and conservation of positive soundscapes, but might also serve to revive the appreciation of environmental aesthetics through art and other forms of expressive media. Furthermore, while the association between soundscape research and conservation is self-evident, it is noted that conservation need not always be interpreted as the literal preservation or restoration of the physical environment into a present or former condition; it can also mean conserving the connections between people and environments. As Carlson remarks, 'what one wishes to conserve is not nature, but the human-nature relationship' [239].

A similar recurring theme in environmental aesthetics, which is particularly pertinent in relation to urban design and soundscape research, is the belief that objects which are modest and familiar have equal worth with those which are grand and rare. Leopold's remark that, 'the weeds in a city lot convey the same lesson as the redwoods' [240] is a reminder that what is lacking, and which humans therefore need to invest in above all else, is an ability to learn these lessons and to recognise the latent value in the environments around them. It would be a mistake to think that this naïve and effortless appreciation of the environment requires one to journey to the far reaches of the world, to places that are widely regarded as exceptional and 'unspoilt' by human interaction. Moreover, there is the danger that such encounters will further diminish the value one finds in the day-to-day, perpetuating those all too common feelings of boredom and dissatisfaction with the 'here' and the 'now'.

One also finds an inherent association between environmental aesthetics and environmental ethics through the philosophical study of value. In basic terms, environmental ethics seeks to articulate reasons why the natural environment has an intrinsic value or worth which cannot be reduced down to economic value [241], from which one might infer that humans therefore have a certain moral obligation or responsibility to protect it. This is something that, if not expressed explicitly, is often alluded to in acoustic ecology and environmental noise literature. From an aesthetic view point, one might also argue that a sense of duty toward the environment ought not to be motivated by feelings of guilt or fear - factors which fluctuate according to social, political and economic changes - but because they have aesthetic value, and what's more, that this value is entirely sustainable because it originates with humans. Here, one is reminded of the quote by Hume, that 'beauty in things exists merely in the mind which contemplates them' [242] - an idea that is echoed in environmental aesthetics and one which the author believes can be extended to soundscape research. To quote Berleant,

> Environment does not stand separate and apart to be studied and known impartially and objectively. A landscape is like a suit of clothes, empty and meaningless apart from its wearer. Without a human observer, it possesses only possibilities. [243]

Furthermore, there is the shared assumption that what is aesthetically pleasing is 'better for you', physically and mentally, than that which arouses distaste.

> The assumption that preferred scenes are somehow good for us, with the corollary that disliked ones have the opposite effect, has rarely been tested. Yet it would seem especially important to discover if scenes that are 'good to see' are simply good aesthetically (i.e. good to think or good to feel) or whether such aesthetic satisfactions have deeper implications, such as the promotion of physical well-being and mental health. [5]

Whether scenes that are 'good to hear' are simply good aesthetically, or whether they too have deeper implications, is a question that has fuelled and continues to fuel soundscape research. While there is already mounting evidence which favours this supposition (e.g.

4.4 Modes of Analysis

Modes of soundscape analysis typically fall into one or other of three main classes. These are considered to be the class of quantitative techniques, which it shares with environmental noise control, and involves the measurement or calculation of objective parameters; a class of qualitative techniques, generally in the form of socio-acoustic surveys, interviews, psychometric tests etc. which deal with data of a subjective nature; and a class of contemplative techniques, as one finds embodied in musical composition, creative writing, visual arts and other expressive media. Within each of these classes it is possible to examine the soundscape at different scales. For example, analysis can take place at the macro or composite level, wherein one extrapolates information from observing the whole system or structure over a length of time; or it may take place at a constituent or 'object-based' level, wherein one considers the characteristics and significance of individual sound objects, both in isolation and in context.

4.4.1 Quantitative Techniques

Quantitative techniques seek to classify soundscapes and interpret behavioural response based on mathematical and statistical modelling and the measurement of objective parameters. These range from relatively simple procedures such as noise mapping based on long or short-term averaged sound levels and the measurement of physical indicators such as L_{10} and L_{90} (e.g. [244, 245]), to the development of complex mathematical models (e.g. [246]). Also included are empirical and computational tools used in bio-acoustics for species classification and identification, which also have applications in environmental monitoring and conservation (e.g. [247, 248, 249]).

Source Identification and Classification

Approaches to soundscape analysis that are primarily aimed at separating, localising and identifying sounds within complex outside environments have broad applicability, in areas such as environmental studies, planning guidance and biodiversity. These methods combine techniques developed in different areas of research, such as spatial sound and bio-acoustics. To give an example, a recent publication by Chesmore et al. [250] presents a method of source separation and direction of arrival estimation in a 3-D soundscape environment from B-format recordings. The recordings used in the source separation part of the study were made using three speakers under anechoic conditions, while those used for direction-of-arrival (DOA) estimation were made using six chirps - again under anechoic conditions. This would seem to suggest the method has limited applicability in relation to urban planning and environmental noise impact assessments due to the prevalence of low frequency sources and relative complexity of real urban soundscapes. Nevertheless, such developments are key agents informing the design of tools suitable for these applications and more. For example, the study has contributed to the development of an advanced type of sound level meter known as an 'instrument for soundscape recognition, identification and evaluation' (ISRIE) which is said to be capable of characterising a sound field in terms of the relative contributions of the different noise sources [251]. The instrument uses a linear array of three acoustic sensors (omni-directional or 3-D Soundfield microphones) to capture the sound simultaneously before implementing a source separation algorithm, and finally a combination of a time-domain analysis method [252] and the artificial neural network approach known as Learning Vector Quantization (LVQ) [253] to perform the classification. The usefulness of tools like ISRIE in environmental noise impact assessments, such as PPG 2417, BS 414218 and noise nuisance applications is discussed by the authors in [251, 254].

Objectification of Subjective Impressions

As well as being aimed at the classification and characterisation of soundscapes, most quantitative studies are also concerned with describing how physical data correlates with subjective impressions and/or physiological responses, and might therefore be used to explain and predict human reactions to a given soundscape.

Due to the time-consuming nature of subjective assessment procedures, efforts have been made to devise measurement systems based on objective sound parameters, along the lines of those used in the automotive industry for the purpose of improving vehicle interior noise quality. In one recent study, the psychoacoustic parameters sound pressure level, loudness, sharpness, roughness and fluctuation strength, as well the binaural parameter, urban interaural level difference (uILD), were used to identify common features of a soundscape and differences in their quality [255]. uILD is a new parameter proposed by the authors which compares level differences at the left and right ears to show which was exposed to higher sound levels and other acoustical parameters for more of the time. The study analysed 250 binaural audio samples according to these parameters, and a clustering methodology was used for subsequent categorisation of the soundscapes.

Other approaches have looked for a possible correlation between physiological and subjective responses from listeners exposed to different sound environments. These have involved monitoring various physiological functions before and during exposure, including the heart rate, respiritory rate, neural and muscular activity, the latter having been performed using the techniques of functional Magnetic Resonance Imaging (fMRI) [256] and electromyography (EMG) [257] respectively.

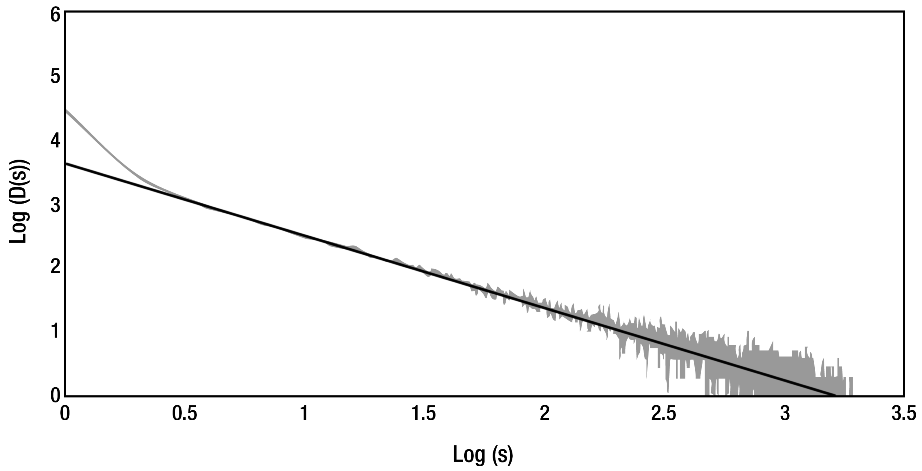


Figure 4.1: A linear relation between the number of avalanches (N) and size of the avalanches (s) is revealed in a logarithmic plot from [260]. SOC is used to predict the behaviour of all kinds of complex phenomena, from earthquakes and land formations, to brain signals and stock market dynamics.

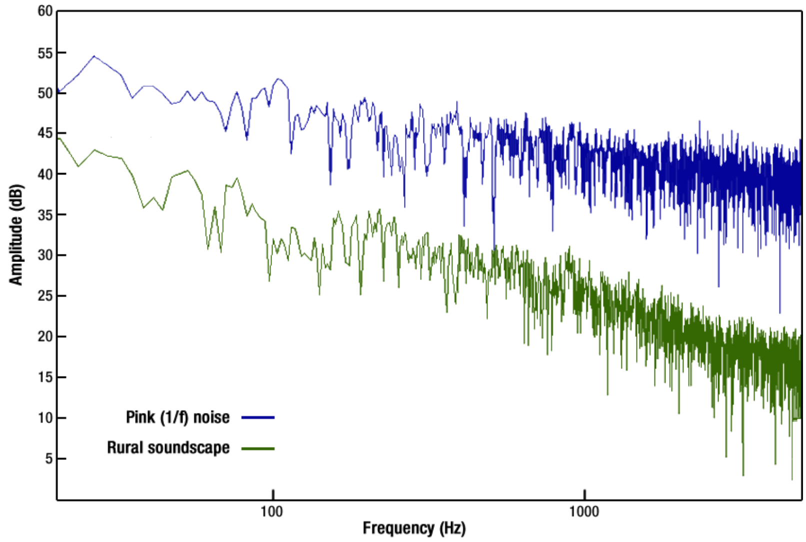
Efforts have also been made to explain subjective response through spectral and temporal comparison with natural systems and music. Spectral analysis at the composite level reveals a varying degree of similarity between soundscapes and pink noise. Pink noise, also known as $1/f$ noise, can refer to any stochastic process with a power spectral density of the form:

$$S(f) \propto \frac{1}{f_n} \tag{4.1}$$

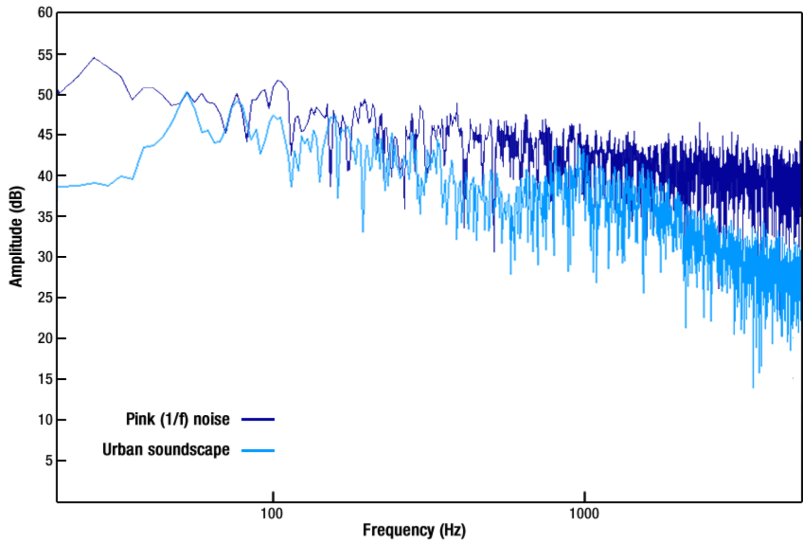
where f is frequency, on an interval bounded away from both zero and infinity. $1/f$ noise is widely studied on account of its ubiquitous occurrence in many physical and biological systems. It is also observed in music, which some believe may be an unconscious imitation of natural systems [258]. The notion of Self-Organised Criticality (SOC), initially presented by Bak et al. [259], is now a widely accepted explanation for the straight lines that result from plotting the distribution $D(s)$ of data points in many complex systems on a logarithmic scale. For example, Figure 4.1 shows the log-log distribution of the number of sites involved in an avalanche when grains of sand are dropped onto the top of a pile of sand at a very slow rate [260]. The straight line indicates that the distribution follows a power law of the form $D(s) = s^t$ which is, by definition, implicit in $1/f$ noise [261]. SOC recognises that a dynamic system which at a micro-level exhibits randomness will, above a critical point, adhere to a scale-invariant spatial distribution curve [262] - subject to certain limitations within the system. It has also been suggested that both sounds and music which exhibit a frequency spectrum close to $1/f$ noise are preferred by most listeners [261]. Moreover, the calming effects of $1/f$ noise are also well-documented and thought to be related to a reduction in brain wave complexity due to the noise stimulus [263].

Botteldooren et al. have also investigated the temporal structures of the urban soundscape, which they view as a manifestation of a rhythmic system and thus capable of exhibiting different characteristic behaviour at different temporal scales. For example, at the micro level the soundscape is composed of individual *sound particles* at the threshold of perception, progressing to single events or *sound objects*, and finally to the overall structure or architecture spanning minutes, hours or days [261].

Figure 4.2: Log-log frequency plots of typical 10 second segments of (a) rural and (b) urban ambient sound compared with $1/f$ noise. In this example the power spectrum of the rural soundscape follows a straight line closely resembling that of $1/f$ noise. This is slightly less evident in the urban soundscape due to the (relatively) high proportion of low frequency energy. Nevertheless, both soundscapes are spectrally quite similar.



(a) Rural versus $1/f$ noise



(b) Urban versus $1/f$ noise

While it may be possible to derive very generalised theories on the basis of composite soundscape analyses, as Botteldooren et al. point out, characterisation of soundscapes in terms of their temporal and dynamic structure alone does not give a complete picture, as two time varying sequences which produce statistically similar results may be perceived very differently by a subject

[264]. This is clearly evident in Figure 4.2 in which there is a distinct similarity between two frequency spectra, one of which is recorded in an urban environment, and the other in a rural setting. It is thus apparent that, in order to draw any realistic conclusions concerning the significance of culture and individual experience on soundscape perception, the subject should not be precluded from the analysis completely. Nevertheless, recent computational techniques based on human-like auditory processing models (e.g. [246]) may mark the beginning of an effort to automate these processes.

4.4.2 Qualitative Techniques

Qualitative techniques are concerned with furthering our understanding of soundscapes through subjective observation of emotional and behavioural responses, generally with a view towards developing systems of classification based on perceived sound quality. Qualitative methods not only ask which sounds are preferred over others, but also ask *how*, *when* and *why* it is that they are preferred, typically through questionnaires, in depth interviews [265] and psychometric tests. The hope is that a deeper understanding of subjective response to environmental sound supported by standardised terminologies and assessment procedures would be beneficial to urban planning processes, noise impact assessments, and encourage the creation and conservation of acoustically healthy environments. However, with the field still in its infancy, there is as yet no universal consensus concerning soundscape methodologies or metrics, although efforts have been made in recent years to remedy this (e.g. [255, 266]).

Soundscape Metrics

An output from the Positive Soundscapes Project was a set of 12 new soundscape metrics intended for soundscape evaluation. Each of these metrics relates to one of the perceived components of an urban sound environment [266]. There is still a degree of contention surrounding the use of psycho-semantic descriptors in soundscape research as some feel that the difficulty in achieving a consistent interpretation of the terms amongst subjects from different linguistic and cultural spheres means they have limited scientific value. It is thought that a reasonable understanding of psychological research methods is needed to produce results from soundscape metrics which are statistically meaningful, which could be cause for concern given that many of those involved with soundscape research are from traditionally unrelated fields such as acoustics and architecture.

A novel approach to the analysis of sound quality of soundscapes from an urban planning perspective was recently proposed by Jennings et al. [267, 268]. The approach is basically an adaptation of the Kano customer satisfaction model, a theory of product development and customer satisfaction developed in the 80s [269], applied to the soundscape under study [270]. Jennings believes that the users of the space have different requirements to those planning the space, and that these tend to change as they become more engaged with the soundscape. For example, the authors propose that the meaning of ‘positive’ for a public space is quite different for three types of people, each with a different level of direct engagement with the soundscape: planners; serious listeners; users of the space. Another interesting outcome of this research is the discovery that soundscapes rated as 'calm' or 'vibrant' by a user do not necessarily differ in terms of sound pressure levels [271] and provide further evidence in the case for new assessment methods. Another study that supports this argument is Hall et al.'s recent exploration of perceptual, psychoacoustic and acoustical properties of urban soundscapes [272] which involved subjective and statistical analysis of soundscape recordings. Amongst its findings were the statistical validity of two new soundscape metrics, emotional-valence and arousal. Others attempt to validate subjective parameters by comparison of physiological responses to soundscape elements with subjective estimates. For example, in a study performed by Hume et al. [257] the effects of individual soundscape elements on the subjective assessment of pleasantness and arousal were compared with associated physiological responses: Heart Rate (HR), Respiratory Rate (RR) and forehead electromyography (EMG) levels.

Crowd-sourcing

Other noteworthy developments in relation to soundscape perception have included a community-based environmental monitoring project known as the 'Sounds Around You' [273] project. Those behind the project developed a smart phone application that would enable a user to record and subjectively evaluate the 'sounds around them'. The data that was generated would then be uploaded to a shared network. The view was to build up a comprehensive geo-mapped soundscape database that could be tapped by applications such as google maps.

Influence of Visual Stimuli

It is generally accepted that visual and other sensory input has a significant, and at times, confounding, influence on soundscape perception, perhaps more so at the auto-centric perceptual level. The influence of specific physical features and visual ambience on perception of sound have both been the object of a number of qualitative surveys. Of particular interest in light of the fact that it seems to have been ahead of its time is a 1984 study conducted by Anderson et al. [274] which sought to uncover the influence of vegetation on perception of sound. Among its conclusions was the idea that vegetation can affect both the perceived visual quality and acoustic expectations in some urban spaces which, in turn, affects tolerances to noise. A more recent study performed by Marry [265] would seem to corroborate these findings. She also investigated the influence of housing type on sound perception, which is thought to have a similar explanation, possibly with an even greater deal of experiential interference. Marry attempted to look for correlation between acoustic indicators and verbal feedback obtained from in depth interviews. She found that those spaces considered pleasant tended to be characterised by long sequences for which the sound level was below 60 dB (taken to be an indicator of tranquillity), and relatively low levels of L_{Aeq}, L_{10}, L_5 and L_1. In contrast, results indicated that high levels of L_{50}, L_{10} and L_5 and long periods for which the sound level was above 50 dB were characteristic of soundscapes that were considered unpleasant by the interviewees. Soundscapes rated as neither pleasant nor unpleasant were thought to be characterised by fluctuating sound levels.

Restorativeness

The psychological concept of restorative experiences [275] is something which has been introduced to soundscape research in recent years, the most basic aspect of which is related to recovery from mental fatigue. Hartig et al. have paid considerable attention to the preference for natural over man-made features in their investigations of the restorative value of different environments [276]. Therein restoration is defined as 'the process of renewing or recovering physical, psychological and social capacities that have become depleted in meeting ordinary adaptation demands', and in that sense it could be seen as a re-expression of an idea mentioned previously, namely that natural environments have the ability to 'bring us closer to our authentic selves' [219]. Restorativeness is thus considered an important component in the quest for a deeper understanding of subjective preferences which may help to guide land managers, planners, politicians and others in effecting changes to the environment. From a visitor's perspective, the ability to locate restorative environments, and negate those that are not, might also be considered a desirable goal. Some effort has thus been made to devise a system that will enable the classification of soundscapes in terms of their restorative value. One study proposes the use of 'quality labels' based on a combination of physical and perception-based indicators intended to help visitors to a region identify restorative environments [277], while a later study proposes the introduction of a 'perceived restorativeness scale' based on sound level and socio-acoustic data [278].

4.4.3 Modelling and Re-Synthesis

Modelling and/or re-synthesis of the soundscape is typically approached in one of three ways: an empirical approach may be taken, as with noise mapping software; an impressionistic approach may be taken, by which a soundscape 'composition' is rendered, through the manual or mathematical sequencing of recorded and/or synthesised audio material; finally the approach may be aimed at rendering a perceptually convincing auditory environment using advanced numerical modelling methods typically associated with auralisation applications (see Chapter 5). Auralisation is considered to be an important tool in acoustic design which, in the context of soundscape, refers to the realistic, artificial simulation and reproduction of the various sound sources that can be heard in an environment [279].

For example, various systems have been devised which aim to generate compelling artificial soundscapes through the automated sequencing, processing and playback of large quantities of audio files in real-time (e.g. [280, 281]). Some have also tried to make this process interactive, either by designing systems which are responsive to user generated input in a virtual reality context (e.g. [282]), or by allowing the user to take conscious control of the composition in the real world. For example, members of the Positive Soundscapes group developed a 'sound sequencer toy' [283]

- a device with which a listener can create their own soundscape by triggering pre-recorded sound files via a graphical user interface.

Another approach to soundscape composition has been to computationally emulate self-organized biological or natural acoustic environments. Fornari et al. have proposed the use of an Evolutionary Sound Synthesis method, the ESSynth, for the dynamic creation and control of sound dispersion in soundscape [284]. The ESSynth is a mathematical model for interactive sound synthesis based on the application of Genetic Algorithms (GA). In evolutionary sound synthesis, sequences of waveform variants are generated by applying GA to an initial population of waveforms [284].

Arup Acoustics consultancy reportedly uses geographical information systems (GIS) for the purpose of auralising the soundcape. The technology is used 'to guide the location and sound levels for new soundscape features' [285]. Another technology mentioned on the company website is iXDLab, a collaboration between Arup and the Glasgow School of Arts Digital Design Studio. The lab combines acoustic modelling with 3-D imaging and is used in the context of soundscape design. Several more papers have been published in relation to modelling and auralisation of outdoor urban environments (e.g. [286, 287]). While such publications have typically tended toward applications in computer gaming, virtual reality systems and military education, soundscape modelling in relation to urban design has started to receive greater attention in recent years. For example, a relative newcomer to the market, IPL SDK [288], reportedly offers sufficient flexibility to cope with modelling real-world urban acoustic problems. This particular acoustic modelling application is based on research conducted by GAMMA Research Labs (e.g. [286, 289]).

4.5 Implementation

In this last section, the author drew on some examples of how the ideas and methodologies of the previous two sections either have been, or can conceivably be, implemented in real urban environments. All are thought to be connected to soundscape research in some sense - whether this is through a recognition of its scientific findings, or out of a mutual appreciation for the underlying philosophy.

Audition is often said to be a fairly passive sense compared with vision [5, 22], which might explain why some implementations attempt to merge with the existing environment and create new sonic ambience, without necessarily drawing attention. It therefore seems necessary to distinguish between the use of sound in urban design - i.e. treatments or interventions aimed at enhancing subjective experiences - and sound art which has its own agenda and, as a consequence, may or may not be sympathetic to the existing soundscape.

4.5.1 Urban Design

Urban design is an inter-disciplinary subject which seeks to unite many different fields, from the physical and social sciences, to economics and engineering. In essence it can be considered the architecture of urban spaces. Urban design might also be linked to environmental design, or rather, certain environmental design processes might also be used in urban design. Soundscape design would be seen as an environmental design process in that it balances aesthetics with deep consideration for the environment and the desire to create harmony between people, nature and man-made artefacts.

Urban public spaces can be designed to encourage activities which generate unique sounds or *soundmarks*, that attract attention and reflect traditional or cultural elements [279]. This might take the form of sound installation, or the introduction of masking sounds such as water fountains. Altering flow methods and fountain design has proven to provide great potential in shaping the spectrum of water features [290], making them more effective at masking undesirable sounds and thus a positive contribution to the aesthetics of the space. The careful arrangement of vegetation in urban spaces can also enhance the visual ambience of the environment and attract songbirds, as well as offering some amount of absorption and scattering of sound energy. It has also been posited that, as a more drastic measure, concealed loudspeakers could be introduced into the space, which could play back fitting environmental sounds, such as singing birds in an urban park [291], or even complimentary man-made sounds. In this respect, urban spaces can provide the stages for a plethora of aesthetically and interactive sound installations.

An urban designer might also ask how behaviour can be influenced by the design of a space - or indeed, how behaviour might influence the design - including the acoustic elements. An example of an urban design installation which utilises sonic elements was the musical stairs in Stockholm. This was a temporary 'thought experiment' produced by Volkswagon for advertising purposes, which sought to encourage travellers to choose the stairs over the escalator [292]. Whilst this particular example is unlikely to have exhibited any long-term social effects, were it to become a permanent fixture, it may serve to inspire urban designers to find other ways of using sound to influence the way humans interact with the spaces around them.

4.5.2 Sound Installation

The term sound installation is used here to refer to works of sound art incorporated into public spaces - mostly urban. These are founded on the philosophies of the artists responsible for producing them, their primary motivations being linked to personal expression. This contrasts with the previous examples which could be seen in a more purposeful and altruistic light, although there is inevitably some crossover.

The German experimental musician Max Neuhaus began to explore the concept of bringing sound art into public spaces in the 1960s, disguising them within their environments in such a way that people discovered them for themselves and took possession of them, led by their curiosity into listening [293]. Following Neuhaus, sound installation is founded upon the idea of making a sound work more public, or rather, making the experimental strand of musical practice more susceptible to a different set of conditions and questions [294]. One of his later works was a sound installation - which he refers to as a 'sound sculpture' - located in Times Square [295]. The sculpture was installed beneath the grating over a subway ventilation shaft and was deliberately unmarked so as to surprise the listener and enable them to take ownership of their own, unexpected sensory discovery.

The sound, which has been described a continuous moan, a reverberating bell or an organ-like drone [296], was not conceived with the intention of improving the perceived sound quality of the space, though it is interesting to consider what might have been achieved had this been on the agenda. For example, studies investigating the influence of live music on the perceived sound quality in urban environments have suggested that, while the low frequency content is not generally loud enough to mask traffic sound, the high frequency components can make the music stand out from the background and make the soundscape more pleasant [279, 290]. Another of the artist's earlier pieces, dated 1967 and entitled *Drive In Music*, consisted of seven radio transmitters located intermittently along a stretch of road, each one broadcasting at a particular frequency. Drivers could assemble the piece for themselves by tuning their car radios as they traversed the different zones. The resulting music would therefore be different depending on the speed and position of the vehicle, together with the weather conditions on the day. Here the listener himself becomes an active participant in the assembly of the soundscape. Again, this leads one to wonder what might be achieved if urban designers were to adopt the philosophies of sound installation in relation to soundscape, especially given the benefits of modern technology and current understanding of the psychological and physiological processes behind auditory perception.

An example of an artificially engineered soundscape that could be seen as a step in this general direction is a sound and light installation in the dark arches beneath Leeds City Rail Station. The purpose of this installation was to improve the look and feel of an area that connects the once dilapidated but now up and coming district of Holbeck with the town centre, via a rather insalubrious tunnel dominated by road traffic noise and the intermittent rumble of overhead trains (Figure 4.3). Yet, while the concept seems highly progressive, its execution lacks the drama that would perhaps have created a more invigorating sensory experience, and which could be advocated on the basis that the space is transitional in its nature.

Figure 4.3: *Dark Arches* sound and light installation by Hans Peter Kuhn in Neville Street, Leeds. Photograph: Kippa Matthews

Other examples can be found in the work of the sound artist Bill Fontana. In his 2011 work entitled White Sound [297] the artist rigged up a live audio feed from Chesil beach in Dorset (Southern England) to Euston road in central London. The sound of the sea has spectral qualities similar to white noise, and is thus assumed to have some auditory masking properties, although there does not appear to have been any attempt at scrutinising or quantifying its positive effects from a scientific angle.

4.6 Discussion

The area of soundscape research has undergone a rapid evolution over the past decade or so, and is one which continues to prosper, ever-increasing the likelihood it will become a major influence in future urban design processes. What is perhaps most exciting about this prospect, is the potential for acousticians and urban designers to impart on a space its own unique character - a factor that sets it apart from environmental acoustics which seems driven toward conformity. That said, it is not uncommon to see brevity and originality in visual architecture and landscaping, making bold statements in soundscape design is more hazardous. After all, it is possible to turn the head or close the eyes to avoid an eyesore, whereas unwelcome sound is much harder to avoid.

Another way in which soundscape research is thought to differ from environmental acoustics, is that it recognises that efforts aimed at SPL noise reduction are not always viable in urban areas, and that alternative approaches may ultimately prove more successful. Yet this difference is not thought to have arisen due to any disagreement or evidential discrepancy; it simply reflects the fact that there is still relatively little overlap between these two fields at the occupational level. The job of environmental acoustic consultant is primarily to ensure that noise guidelines are met; beyond this they are not generally concerned with the subjective sound quality in the environment - although there are of course exceptions to this rule. By contrast, those involved in soundscape research are not so much concerned with limiting noise as they are with the creation of healthy and aesthetically pleasant living, working and recreational spaces, and the restoration of human-environment relationships. Furthermore, in soundscape research, assessment does not follow a prescribed format. In fact, it has been demonstrated here that there are a multitude of different ways to approach the analysis of the soundscape, and that these are justified on the basis of what one is seeking to learn or to demonstrate. However, if there is to be any hope of soundscape methodologies becoming a mainstay in environmental noise assessment procedures, it seems that some degree of uniformity is needed in order to streamline these processes. For example, it would be helpful if there was some consolidation and simplification of the large quantity of terms and procedural variations, as these can at times be a source of confusion.

It is thus clear that approaches to soundscape design and evaluation need to be adaptive, giving appropriate consideration to the intended use of a space and thus ensuring it is 'fit for purpose'. This is based on the assumption that negative responses to soundscapes are generally indicative of a discord between what people want to do or to feel in a space, and what the sound and/or other stimuli in that space is conducive to. For example, with regards to living, working and restorative spaces, greater consideration may be given to the composite soundscape - as observed over an appropriate length of time - than would be given in transitional spaces. The term transitional is intended to refer to spaces whose primary function is to convey subjects from A to B, and relatively short-term passing places. Such spaces would be required to meet a different set of design criteria than those in which the subject is seeking to relax, recuperate, or focus their attention elsewhere - a proposition thought to be linked to how people attend to their environment perceptually under different conditions. Spaces required to serve multiple functions, such as parks and city squares, might thus benefit from the creation of distinct acoustic ‘micro zones’ and by organising pedestrian routes and structural elements in such a way as to facilitate this. On a slightly larger scale, acoustic zones are a long established approach, the earliest evidence of which is thought to date back to farming communities of ancient Mesopatamia who are believed to have constructed their dwellings in concentric circles for acoustic reasons [298]. Attempting to create zones within a relatively small outdoor space may be more challenging on the physical level, but given the manifold nature of auditory perception, it does not seem an unrealistic proposition.

What has also emerged over the past two chapters, is that there are two quite distinct conceptual models that can be applied to environmental sound. On the one hand, sound can be seen as a part of an evolving landscape, one which is continuously adapted and reinvented to reflect the current technologies and social and political trends of the era. In this sense, the urban landscape can be considered an extension of the human organism, and thus something of ethical value. From another perspective, much of the sound in urban environments can be seen as a contaminant and, as such, is both undesirable and threatening. Since both views are legitimate and neither precludes the other, today’s urban designers face the difficult task of giving each the respect and attention it deserves, without compromising the other. Effectively what this means is that there can be two approaches to dealing with negatively perceived sound in the environment. The first is to address the sound itself, asking how the space or sources of sound may be modified in some way to create a more positive impression; the second is to confront negative reactions to it, as if they originated from somewhere other than the sound itself. While this might seem counter-intuitive at first, by considering sound as an instrument of stress (the real epidemic), it seems plausible that breaking this relationship might lead to significant improvements. The scope of the problem is not then limited to the treatment and prevention of noise; it extends into other areas, such as art, education, and public engagement activities - all aimed at stimulating the imagination and intellect of the general public. Such an approach would attempt to dispel the general tendency to group sonic events into two categories - i.e. sound that is desirable, and sound that is not - and would try instead to elicit an appreciation for sounds that would otherwise be ignored or perceived negatively. Of course, the effectiveness of such an approach would vary from person to person and according to the nature and severity of the problem.

It was proposed in the previous chapter that new forms of environmental noise treatment which aim to alter the perception of sound in noisy environments might be termed *noise augmentation*. Examples of noise augmentation might include the introduction of masking sounds (e.g. [299, 300], physical filtering devices such as the sonic crystals mentioned in Chapter 3, and outdoor electro-acoustic sound installations such as the ones mentioned in this chapter. While the widespread implementation of noise augmentation technology might still seem futuristic, it may not be beyond the realms of technological feasibility in some urban spaces. However, what was not clear in the literature was whether these particular examples actually had a measurable effect on the perceived sound quality in the environment, as one might reveal through qualitative analysis before and after treatment. Due to the inherent difficulties associated with performing subjective analyses in situ, and the with the move toward automating the process still in its early stages, it seems a solution to this problem may be to perform the analysis via the medium of auralisation.

4.7 Summary

In this chapter, the history, science and philosophies behind soundscape research were outlined. The author then discussed how soundscape research ideas and methodologies differed from those of environmental acoustics, which were prevously covered in Chapter 3. Within the discussion, the suggestion was made that part of an integrated and sustainable approach to environmental noise control might be noise augmentation, and that this could be investigated through auralisation. The remainder of this thesis is therefore dedicated to testing this theory, and in doing so, attempts to prove the underlying hypothesis that auralisation can be an effective aid to the design, evaluation and promotion of positive urban soundscapes.

5

Auralisation

5.1 Introduction

This chapter presents an introduction to the process of auralisation and a broad overview of associated acoustic modelling and spatial reproduction techniques. In its broadest sense, auralisation can be considered the aural analogue of visualisation, although the term has taken on quite a specific meaning in relation to virtual acoustics. It is thought to have first made its appearance in the field of computer science when Francioni and Jackson used it to refer to the use of sound to represent different aspects of the run-time of parallel program behaviour[301]. Around that time it was often used synonymously with the term *sonification*, which is said to refer to 'the use of data to control a sound generator for the purpose of monitoring and analysing data' [302]. The meanings of the two terms have diverged somewhat since auralisation was later adopted into the field of virtual acoustics - a move which seems to have coincided with the publication of an article by Kleiner et al[303] in which the authors define auralisation as:

> 'The process of rendering audible, by physical or mathematical modelling, the sound field of a source in a space, in such a way as to simulate the binaural listening experience at a given position in the modelled space.'

Within this definition, the authors have essentially imposed three conditions that must be met in order for an audio signal (or group of signals) to be classified as an auralisation. The first of these conditions pertains to the process of generating the audio, namely, that it must be achieved through computational means; secondly, the source must be represented in the context of the wider acoustic system, thus conveying to the receiver some acoustical clue as to the nature of the space it occupies; and thirdly, the receiver is presumed to be a binaurally equipped organism who also happens to be present, at a specific location and time, in the acoustic scene. One is therefore reminded of the fact that audible sound is the product of a complex relationship between one or more acoustic sources, the space in which the sound is propagated, and a receiver - all of which have their own unique characteristics.

It should come as no surprise then that auralisation is most commonly encountered in relation to room acoustics, where one is dealing with relatively easy to model, bounded acoustic systems that are also large enough to accommodate one or more human receptors. This has led many to consider the term almost exclusively in relation to building acoustics, with particular emphasis on musical performance venues - after all, it seldom happens that the finer acoustic details of a space are of such interest. Yet, despite its historical association with building acoustics, auralisation has developed much broader applicability in recent years. For example, the emergence of real-time auralisation tools (e.g. [304, 305]) has distinct implications in terms of creating interactive and immersive virtual environments - a development that can largely be attributed to improvements in computer processing architecture. This has lead to its association with Virtual Reality (VR) systems, the objectives of which are highly compatible with the possibilities afforded by advanced auralisation techniques.

5.2 Auralisation for Virtual Reality Applications

Virtual Reality (VR) is commonly defined as,

> 'The computer-generated simulation of a three-dimensional image or environment that can be interacted with in a seemingly real or physical way by a person.' [306]

By this definition, the imagery generated by a VR system ought to be perceived by a subject in three dimensions. Whereas this has historically only applied in the case of the visual imagery, *Acoustic Virtual Reality* (AVR), also referred to as *Virtual Audio Reality* (VAR), extends this requirement to the aural imagery.

It can be said that the aim of a VR system is to deliver a faithful representation of an environment - real or imagined - that is capable of deceiving the senses into perceiving one is physically present in that environment. Thus *presence* is a term one often encounters in relation to VR systems, referring to the subjective experience of being in one place or environment, even when one is physically situated in another [307]. It also implies that the virtual environment is one that is both *dynamic* and *immersive* - thus satisfying conditions for an unbroken and unconscious dialogue between the subject and their physical surroundings. A system that possesses these two qualities therefore has wide applicability and can readily facilitate user interaction.

Immersion in the virtual environment is said to be related to the quantity and quality of sensory data from that environment, and can be measured by the extent to which the computer system shuts out sensations from the real world and accommodates different sensory modalities [308]. Next to be considered are the qualities of audio data that contribute to listener immersion. Since one is primarily interested in the aural modality here, other modalities are largely overlooked, despite there being a degree of interdependence between the senses.

Perhaps most critically, an immersive auditory environment relies on the presence of spatial cues in the acoustic signals - these being indicative of both the source(s) of sound, and the space occupied by the listener. In normal hearing individuals, the vast majority of spatial information is derived from a combination of subtle time, level and spectral differences between the acoustic signals that arrive at each ear, and the quality of this information needs to be particularly high for the listener to resolve the acoustic image in three dimensions.

In virtual acoustics, there are two principle ways in which spatial information can be associated with a 'dry' source signal - dry signals being generally defined as source signals that are free of reverberation and other clues introduced by sound transmission [309]. The first of these deals with the influence of the listener and their physical presence in the environment on the sound that reaches the inner ear - this being a consequence of the diffraction, absorption and reflection properties of the outer ears (pinnae), head and torso. This particular filtering process can be characterised by the Head-Related Transfer Function (HRTF) (see Section 2.3.3). A pair of HRTFs can be used to synthesise a binaural sound at a particular point in space, and for a specific orientation of the listener (see Figure 5.1). Whilst HRTFs are unique to an individual, generic HRTFs are often found to perform well for the majority of listeners.

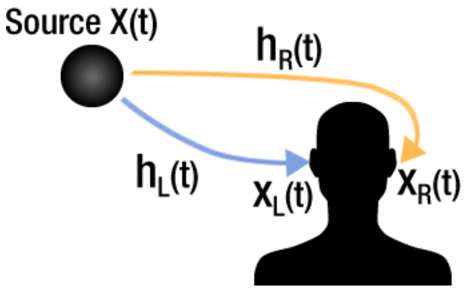


Figure 5.1: A pair of Head-Related Transfer Functions (HRTFs) can be used to describe the filtering of a source signal ($x(t)$) before it is received at the left and right eardrums as $x_L(t)$ and $x_R(t)$, respectively.

The second way in which spatial information can be associated with a given signal (or group of signals), involves looking at the wider acoustic system - ie. the influence of obstacles and surfaces on sound making its way from source to receiver. In Chapter 2 it was mentioned that the filtering properties of a closed acoustic system - be it a room or outdoor space - can be considered in terms of the Impulse Response (IR), which is essentially an energy histogram representing the response of the system to a sudden, high energy input for a given source-receiver configuration. However, listener immersion is not easily achieved with a single impulse response, given the binaural nature of the human auditory system. A minimum of two distinct receiver positions are required to replicate the inter-aural differences, corresponding to the relative positions of the left and right eardrums. While the binaural approach is applicable to headphone listening, where multichannel loudspeaker systems are concerned, the binaural signal can no longer be considered an independent entity, but now consists of a network of contributions from each element in the loudspeaker array. In other words, the requirement is now to replicate the sound field across the presentation domain so that the signals incident on a listeners eardrums are indistinguishable from those they would receive if the virtual environment were in fact real. While in theory this would require a loudspeaker and associated impulse response for every point in the space, in practice the limits and idiosyncrasies of human perceptual fidelity can be exploited to find workable compromises (see Section 5.5.

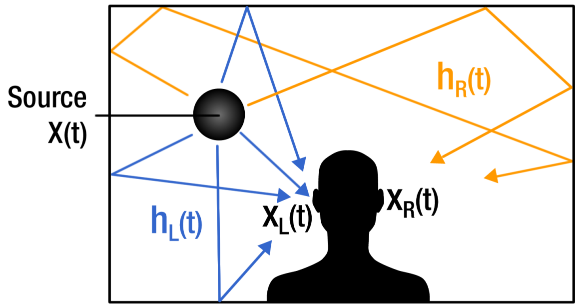


Figure 5.2: The modelling of spatial effects such as scattering and absorption with multiple impulse responses is widely practised in virtual acoustics to create a sense of space.

Another contributor to listener immersion is the exclusion of outside interferences - or, as Slater calls them, 'sensations from the real world' [310]. It is often the case that the sense of immersion is compromised by the presence of unwanted stimuli, such as the reverberation characteristics of the listening room, or noise emitted by the equipment. Distraction may also be caused by non-acoustic stimuli - the sight of the loudspeakers, for example, or the haptic sensation of having headphones attached to one's ears. Hence it can be beneficial to suppress other stimuli where possible. For example, by rendering an auralisation in darkness where visual cues are not a necessity, and by selecting the most appropriate form of delivery based on the desired outcome.

To understand what is meant by a *dynamic* auditory environment, it can help to consider its opposite; namely, an environment that is static and unresponsive. In other words, it will always deliver the same output, regardless of whatever else is occurring in the auditory scene. A dynamic auditory environment, on the other hand, is one that changes relative to some other time-varying input, such as the actions or motion of an object or receiver, or some other physical phenomena. As this implies, the signal that is delivered to the output can no longer be prepared in advance, but must undergo some degree of processing in near real-time. As Lokki writes,

> 'For the acoustic environment to be called dynamic, temporal changes in both the input (stimuli) and output (rendering) systems must be resolved to within a perceptually acceptable resolution. This implies knowledge of the sensory mechanisms is needed in order to make computationally cost-effective trade-offs where applicable [311].'

Here one is introduced for the first time to the notion that something can be 'perceptually acceptable' whilst being physically or conceptually imperfect, yet the latter is an aspect of acoustics that is virtually unavoidable. In fact, one of the greatest difficulties an acoustician or audio engineer may be faced with, is establishing where to draw that all-important line between the acceptable and the unacceptable when it comes to auditory perception. Knowledge of human resolving power therefore has a certain influence over the design of AVR systems, governing what information to include at both the input and output stages, and what can reasonably be thrown away. Fortunately, in the auditory case, understanding of resolving power is relatively advanced, whereas for some other modalities - the haptic system for example, which deals with the sensation of touch - knowledge is much less developed [312]. Further, a complicating factor is the illusory nature of perception. As Durlach remarks, perception often represents a stimulus configuration that is different from the actual one [312], due to the natural inclination of the sensory mechanisms to create 'illusions' from sensory input. A better understanding of the neurological processes involved in auditory perception might then play an important part in the development of future generations of AVR systems.

According to Lentz, when considering the audio capabilities of VR systems, the ultimate aim for the resulting sound is not to be physically absolutely correct, but perceptually plausible [313]. While this may well be the case in the majority of VR applications, there may also be situations where the demand for physical accuracy of one or more aspects of the resultant sound may equal the demand for plausibility. Consider the possibility of a VR system capable of delivering high fidelity muti-sensory information: one might then assume that, by rendering a physically faithful model of an environment, the recipient will respond naturally and little more will be left open to subjective interpretation than would be in a real-world scenario. This is what Lokki refers to as the 'physically-based' [311] approach to rendering virtual environments, the alternative being the 'perceptually-based' approach, whereby one exploits psychosensory mechanisms to improve the efficiency of the system. However, there comes a point in any physically-based auralisation when knowledge and technological limitations force one to defer to the latter of these approaches; the difficulty is often in deciding when that point has been reached. When dealing with acoustic environments of a generic or fictitious nature, or in situations that call for real-time interactivity, this point tends to be arrived at relatively early. In such cases, plausibility with minimum computational cost is generally the main aim, meaning it is often sufficient to include only basic spatial information, such as source position and generic reverberant information - things that can be achieved through relatively simple signal processing techniques such as amplitude panning and algorithmic reverb plug-ins. Algorithmic reverbs essentially try to emulate the convolution approach by filtering sound with an artificially generated impulse response which is controlled by parameters on the Digital Audio Workstation (DAW).

On the other hand, where specific details concerning the acoustics of the environment are of significant interest, it may be then that a physically-based approach should be attempted more rigorously. For example, an architect or acoustician may be less concerned with perception of a specific source than they are with the audible characteristics of the environment [311], in which case one might forego real-time interactivity in favour of a more computationally demanding auralisation methodology. Thus it is apparent that a significant aspect of auralisation consists of striking the right balance between three main criteria: objective accuracy, subjective accuracy, and the available resources - a balance that happens to be highly dependent on the application.

5.3 Fundamentals of Auralisation

The theoretical basis for auralisation can be traced back to a visionary paper presented by Schroeder et al. in 1962 [314] in which the author outlines a methodology that will enable one to generate and subjectively evaluate the frequency and space responses of rooms prior to their construction [315]. Whilst the computational techniques have advanced considerably since the paper's publication, the fundamental mathematical principles behind Schroeder's methodology remain central to the process now known as auralisation.

The goal of auralisation is to alter the perceived spatial characteristics of one or more acoustic signals, either by removing unwanted features, or by imposing new ones on the signal(s). This can be achieved using purpose-built digital filters applied to a signal in either the time or frequency domain. In the majority of cases, the attributes of these filters are derived from impulse responses. In signal processing, an impulse response - also known as a *unit-sample response* - describes the response of a discrete-time linear system to a unit impulse function. A unit impulse function, referred to in mathematics as the Dirac delta function, is a function that is zero everywhere except at zero [316]. It is defined abstractly as:

$$\delta(x) = 0\text{for} \quad x \neq 0 \tag{5.1}$$

$$\int_{-\infty}^{\infty} \delta(x) = 1 \tag{5.2}$$

This can be thought of as a very tall and thinÂŤ spike with unit area located at the origin, as shown in Figure 5.3 [317]. An important property of an ideal unit impulse function is that it has a flat frequency response. Of course, an ideal unit impulse is a physical impossibility, as this would imply the signal was of an infinitely short length in time. Likewise, in digital systems, the width of the spike, δx, is limited by the sampling interval, which also dictates the upper and lower frequency bounds of the system. Nevertheless, the response of a system to an approximation of the ideal unit impulse function can reveal useful information about the behaviour of the system, such as its frequency response. To understand exactly how this is possible and why it is relevant to auralisation, one must first consider the fundamental characteristics of acoustic systems.

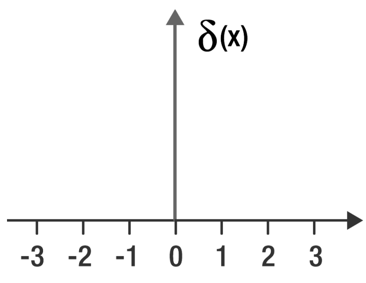


Figure 5.3: The unit impulse or delta function.

5.3.1 Acoustic Systems

In the majority of auralisation applications, the acoustic system, be it a room, musical instrument or loudspeaker enclosure, can be modelled as a Linear Time-Invariant (LTI) system, and thus characterised by three basic mathematical properties: *linearity*, *time-invariance* and *superposition*.

Linearity refers to the fact that the input and output of the system are mapped to one another linearly in time, such that input $x_1(t)$ yields the response $y_1(t)$, and input $x_2(t)$ yields the response $y_2(t)$ and so on (see Figure 5.4). For this to hold true, the system must demonstrate the mathematical properties of *homogeneity* and *additivity*, corresponding to the linear operations of scaling and summing respectively. For example, the system is said to be homogeneous if an input of $kx(t)$ results in an output of $ky(t)$, for any input signal and constant scalar, k. Likewise it is said to be additive, if an input of $x_1(t)+x_2(t)$ results in an output of $y_1(t)+y_2(t)$, for all possible input signals [318]. This can be extended to an arbitrary number of terms according to the following general rule:

$$\sum_n k_n x_n(t) \rightarrow \sum_n k_n y_n(t) \tag{5.3}$$

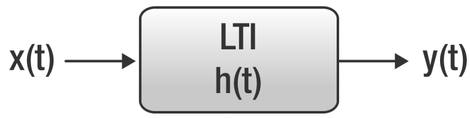


Figure 5.4: In linear system theory, the system can be characterised by its unit sample response, $h(t)$, which is its response to a unit input.

The time-invariance of an LTI system means that its response to an input is always the same, regardless of when the input is applied. So, for any continuous input signal $x(t)$, a delay of T seconds to the input sequence will produce a delay of T seconds in the output:

$$x(t-T) \rightarrow y(t-T) \tag{5.4}$$

Finally, the mathematical principle of superposition means that if the input sequence is made up of the sum of individual components, then the overall response of the system can be considered the sum of the unit-sample responses produced by each individual component [319]. Thus for a discrete-time system where $x[n]$ is an arbitrary input, $h[n]$ is the unit sample response of the system, then its output, $y[n]$, can be expressed by the superposition sum:

$$[H]y[n] = \sum_{m=-\infty}^{\infty} x[m]h[n-m] \tag{5.5}$$

For continuous-time signals, the output of the system, $y(t)$, to an arbitrary input, $x(t)$, and where $h(t)$ is the unit sample response can be expressed by the superposition integral:

$$y(t) = \int_{-\infty}^{\infty} x(\tau)\cdot h(t-\tau)d\tau \tag{5.6}$$

The properties of linearity and time-invariance mean that an LTI system can be characterised entirely by its impulse response and, even more importantly, means it is possible to derive the output of the system in response to any arbitrary input - provided the unit impulse response of the system is known. Furthermore, the principle of superposition enables one to perform this operation algorithmically on signals of indeterminate length, as each signal component may be updated independently without prior knowledge of past or future values. In this sense, the modelled system is said to be memoryless, in that the current value of $y(t)$ does not depend on any past value of $x(t)$.

Convolution

The superposition integral is synonymous with the mathematical operation known as *convolution*. For example, it can be said that $y(t)$ is a result of the convolution of $x(t)$ with $h(t)$. The operation of convolution is denoted by the symbol ⊗. An equivalent expression of (5.6) is thus:

$$y(t) = x(t) \otimes h(t) \tag{5.7}$$

Non-linear Artefacts of Convolution

In the majority of cases there will also be some uncorrelated noise inside the system. For example, in computer modelled systems this can be due to numerical artefacts in the simulation, or, in the case of real systems, noise may 'break in' from external sources. When a noise signal, $n(t)$, is combined with the deterministic part of the output signal, the signal chain is expressed by the following:

$$y(t) = n(t) + x(t) \otimes h(t) \tag{5.8}$$

Of course, real acoustic systems do not adhere to the simple rules attributed to LTI systems. In addition to noise, real-world systems also consist of non-linear components that produce undesirable artefacts in the output (see Figure 5.5). These can have many causes, including non-linear mechanisms occurring in loudspeaker systems, background noise, or atmospheric non-linearities. The measurement process therefore aims to suppress the non-linearities to the greatest extent possible, as the removal of these artefacts at a later stage is difficult to achieve without compromising the integrity of the output.

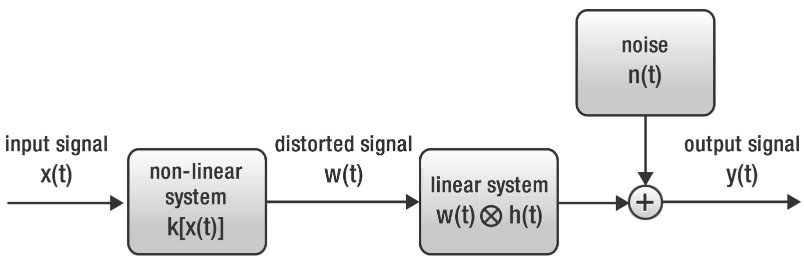


Figure 5.5: In this non-linear system, a distorted version of the input signal, $w(t)$, is now convolved with the unit-sample response.

The problems associated with noise and non-linearity are particularly significant when considering the propagation of sound outdoors where there are a large number of variables to consider. This is obviously an issue that warrants further attention and shall therefore be addressed more fully in Chapter 3. Nevertheless, LTI system theory is applicable to the majority of auralisation applications, whether or not there is a need to compensate for its shortcomings at some stage in the process.

Frequency Domain Representation

Whereas an impulse response can be used to characterise an LTI system in the time domain, its equivalent representation in the frequency domain is something known as the *transfer function*. The steady-state transfer function $\underline{H}(f)$ and the impulse response $h(t)$ are related through the process of Fourier transformation according to the following rule:

$$\mathcal{F}\{h(t)\} = \underline{H}(f) \tag{5.9}$$

where $\mathcal{F}$ denotes the Fourier integral operator [320].

Fourier analysis theory states that any complex periodic waveform can be expressed as the sum of an infinite number of individual frequency components. In relation to the LTI model, whereas the system has so far been depicted in terms of a single input (x) and a single output (y), Fourier analysis breaks the signal down into its constituent sinusoids, allowing the response for each frequency component to be calculated individually. A magnitude frequency plot of these individual responses thus yields the frequency spectrum of the signal.

The Fourier transform is a reversible transform, hence, if the output and steady-state transfer function of the system are known, the inverse Fourier transform can be used to reconstruct the original signal. Hence conversion from the impulse response to the steady-state transfer function and vice versa is achieved by the following rules:

$$\underline{H}(f) \int_{-\infty}^{\infty} h(t) \cdot \exp^{-j2\pi f t}\, dt \tag{5.10}$$

$$h(t) \int_{-\infty}^{\infty} \underline{H}(f) \cdot \exp^{j2\pi f t}\, df \tag{5.11}$$

The relationship of the impulse response to the steady state transfer function via Fourier transformation means that, in practice, the process of convolution is more rapidly computed in the frequency domain where the equivalent operation is a simple multiplication (see Figure 5.6).

It should also be noted that Fourier transformation is a linear process, in which the signal is treated as the linear combination of its constituent sinusoidal functions. As such, its integrity is reliant on the principle of superposition. However, there are invariably situations in the real world where the superposition principle does not hold. These are considered to be situations when the system is ‘overstretched’ so that waves no longer oscillate in a sinusoidal manner - following an explosion for example. In the digital realm, the failure of the system to transmit a signal pattern

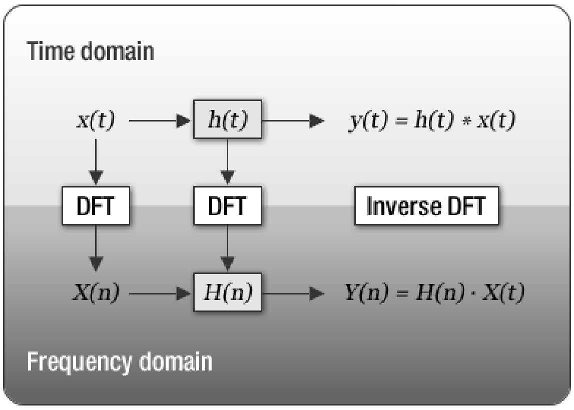


Figure 5.6: The time and frequency domain representations of the system response are related via the Discrete Fourier Transform (DFT)

without altering its shape in some way - and, by association, its frequency response - is referred to as a distortion. Distortion can be particularly problematic when performing computer simulations over discretised regions of space, where numerous factors can corrupt the waveform.

5.4 Acquisition of the Impulse Response

Impulse responses can be obtained using a wide variety of techniques. These can be grouped under three main headings: direct measurement, simulation and synthesis.

5.4.1 Direct Measurement

In the past decade or so, research that was initially aimed at studying the acoustic properties of concert halls and opera houses has led to the development of advanced techniques for the measurement of the room impulse response (room impulse response) in all kinds of buildings. Whereas earlier methods did not yield good enough results for auralisation applications, the newer techniques are much less susceptible to background noise and variance effects like wind or temperature drift. Furthermore, spatial recording techniques have been developed which will enable the approximation of room impulse responses from many different receiver positions in a single measurement (e.g. [321, 322]). Although the measurement methods have emerged in relation to room acoustics, much of the methodology is readily applicable to other problems in acoustics - for calculating the Sound Transmission Class (STC) of a building element [323], for example, or predicting the acoustic shadowing due to finite sized noise barriers [324, 325].

Essentially the procedure for measuring the room impulse response involves exciting the system with a known input and measuring its output at one or more receiver locations. There are various ways to approach this, and these may differ in one or more of the following respects: the type of input, the deconvolution technique that is used to obtain the room impulse response from the measured output, and the recording configuration. Whereas the first and second of these are contributors to the objective quality of the recorded data, the latter impacts primarily on the perceived spatial quality (PSQ) of the auralisation and is thus inextricably linked to the reproduction system. They are therefore considered as two separate issues, and, as such, warrant individual discussion.

The choice of methodology should aim to satisfy the following conditions to the greatest extent possible [326]:

- The excitation signal should be perfectly reproducible.
- The excitation signal and the deconvolution technique should achieve a very good signal-to-noise ratio (more than 80dB).
- The excitation signal and the deconvolution technique should enable the elimination of non linear artefacts in the deconvolved impulse response.
- The excitation signal should extend over the entire range of audible frequencies (20 Hz to 20000 Hz).

Input Signals

Until relatively recently, it was common in room acoustics to use simple devices such as balloons or woodblocks to generate broadband impulses [327]. These sources attempt to emulate the ideal unit impulse function with a very short, high energy burst of broadband sound. One of the main advantages of impulsive input signals is that there is no need to perform inverse filtering - also referred to as *deconvolution* - post-measurement. In such cases, deconvolution is necessary to reverse the unwanted convolution of the input signal with the system's impulse response, thus allowing the unit sample impulse response and the original source waveform to be separately estimated. Deconvolution may be implemented quite easily in the frequency domain by constructing a deconvolution filter to undo the amplitude and phase changes that resulted from the convolution [318]. However, deriving the appropriate filter coefficients for the deconvolution filter using mathematical operations is not trivial, hence the accuracy of the solution can vary quite significantly depending on the deconvolution algorithm [328].

Whilst impulsive sources are often adequate for prediction of reverberation time in small-medium size rooms, the poor signal-to-noise ratio of the measured impulse response makes them unsuitable for auralisation purposes, or for estimation of more advanced acoustical parameters for that matter. This is mainly due to their uncontrolled spectral response and poor repeatability. Balloon bursts in particular are known to have very poor spectral content at low frequencies [329]. An alternative to these devices that was found to offer a significant improvement is the Maximum Length Sequence (MLS) signal introduced by Schroeder [330].

MLS signals are pseudo random bursts of noise that can be generated using linear feedback shift registers [331]. The resulting signal has a frequency power distribution resembling that of white noise, whereas it is in fact constructed of recognisable sequences that can be removed from the recorded signal through decorrelation. Whilst the signals are more difficult to implement, they offer a much better tolerance to background noise than the earlier methods. To reduce the effects of background noise in the measurement chain, a number of periodic bursts are used which are then averaged prior to deconvolution. This averaging process, known as *Time Synchronous Averaging* (TSA), is a relatively simple technique that can be very effective at extracting periodic waveforms from noisy data - provided the noise is uncorrelated with the source [332]. It also means that the distribution of non-linear distortions in the resulting impulse response is relatively even across the measurement period.

A drawback to using MLS signals is the possibility that a phenomenon known as *time-aliasing* may occur if the impulse response is longer than the delay between pulses, thus causing the tail of one decay to overlap with the proceeding one [333]. For this reason, the MLS technique is not very well suited to measuring the impulse response in spaces that have very long reverberation times. A variation on the MLS excitation signal, known as the Inverse Repeated Sequence (IRS), attempts to alleviate the inevitable effects of non-linearities in the measurement chain by using two identical MLS sequences, the second one being an inverted copy of the first. In principle, IRS exhibit complete immunity to even-order non-linearity while maintaining many of the advantages of MLS [334], although the process of deconvolution is less straightforward with IRS. Furthermore, the length of the sequence needs to be twice as long as the impulse response, thus doubling the measuring time.

Another type of input signal to have been adopted by acousticians in recent years, and which tends to be preferred over MLS/IRS for impulse response measurement, is the *Exponential Sine Sweep* (ESS) or *Log Sine Sweep*. The ESS is a sine signal which has a frequency that rises exponentially with time. Mathematically, it is defined by the following expression, wherein ω_1 and ω_2 are angular frequencies equivalent to $2\pi f_1$ and $2\pi f_2$ respectively, T is the sampling interval in seconds, and $x(t)$ denotes the sine signal which progresses in frequency from f_1 to f_2.

$$x(t) = \sin\left[\frac{\omega_1 \cdot T}{\ln\left(\frac{\omega_2}{\omega_1}\right)} \cdot \left(e^{\frac{t}{T}\cdot\ln\left(\frac{\omega_2}{\omega_1}\right)} - 1\right)\right] \tag{5.12}$$

The ESS supersedes an earlier method introduced in [335] that utilises a Linear Sine Sweep (LSS). A problem with LSS signals, as indeed with any signal captured or reproduced using electromechanical elements, is that of the distortion artefacts introduced by the loudspeaker and, to a lesser extent, the microphone. The worst of these transients occur in the earlier part of the signal at onset, thus, for a sweep which ramps in frequency from low to high, achieving a high SNR in the low frequency region is particularly important. By this logic, if more energy is applied in the earlier part of the signal, one can afford to apply progressively less power as the frequency increases. Thus it was proposed in [333] that an improvement to the LSS would be a sweep with a logarithmic frequency-power distribution - this, of course, being the ESS or Log Sine Sweep.

Impulse responses measured using sine sweep signals can be obtained in a single measurement, thus eliminating the need for synchronous averaging and dramatically increasing the signal's robustness against time-variance. Furthermore, using the sine sweep method it is possible to substitute the circular deconvolution process of the MLS method with a ’normal’ or linear deconvolution process, thus enabling the complete removal of distortion-induced artefacts that would otherwise appear as spurious peaks in an MLS signal[333]. The tolerance sine sweeps have to harmonic distortion permits the use of much higher signal levels, which in turn increases the signal-to-noise ratio (SNR) of the recorded impulse.

The deconvolution technique used to obtain the impulse response from a recorded ESS involves a technique referred to in [336] as the ‘time reversal mirror’ technique. The process is covered in more detail in Chapter 6, but it essentially involves convolving the measured signal with a time-reversed version of the input signal. The measured signal should also be filtered with an appropriate bandpass filter prior to deconvolution, to minimise the effects of time-smearing.

Source Specifications

It is standard practice to use an omnidirectional sound source for measurement of reverberation time and other acoustical parameters to ensure the equal spatial excitation of the subject room. The source should also be able to produce an appropriate level of sound power relative to the size of the measurement domain. This is to ensure the sound pressure levels produced are sufficiently high to provide decay curves with the required minimum dynamic range without contamination by background noise [72]. The same principles generally apply in impulse response measurement for auralisation applications, although it is now widely recognised that, in some cases, the use of directive sources can enhance the perceived spatial quality of the auralisation [337, 338].

While it is normally the case that a single source will be used to excite the system, the number of source positions is largely dependent on the size of the room and the intended application. For example, when the intention is to render a dynamic auralisation with moving sources, multiple source positions are generally needed to create the impression of a smooth spatial transition.

Receiver Specifications

The microphone should in theory demonstrate a relatively broad and flat frequency response and an appropriate directivity pattern relative to the intended configuration. For some spatial recording techniques involving multiple microphones, it is also appropriate to ensure the microphones closely resemble one another in these respects. As a general rule, the recommendations given in BS ISO 3382 [72] serve as a good guide:

> ‘The microphone should be as small as possible and preferably have a maximum diaphragm diameter of 14 mm. Microphones with diameters up to 27 mm are allowed, if they are of the pressure response type or of the free field response type but supplied with a random incidence corrector. The octave or one-third-octave filters shall conform to IEC 61260.’

Regarding the recording set-up, several approaches are considered here which vary in their complexity. The simplest of these involves the placement of a single omnidirectional microphone at the assumed listening position or another suitable location within the region of interest. A suitable location is one that is considered representative of the space, i.e. avoiding any null or points, and which is a sufficient distance from any reflective surfaces. The reason for maintaining a sufficient distance from any surfaces is in order to avoid a reflection effect known as the *boundary effect* which is caused by constructive interference between the direct and reflected wave front. In ISO 3382 [72] this is considered to be a height of 1.5m above ground and at least 1m from any reflective surface. It is also a stipulation in ISO 3382 that a minimum of twelve receiver positions should be used from which an average reverberation time is then taken. However, for auralisation applications, this particular requirement would not normally be relevant, thus the number and positioning of the receivers is determined on a case by case basis.

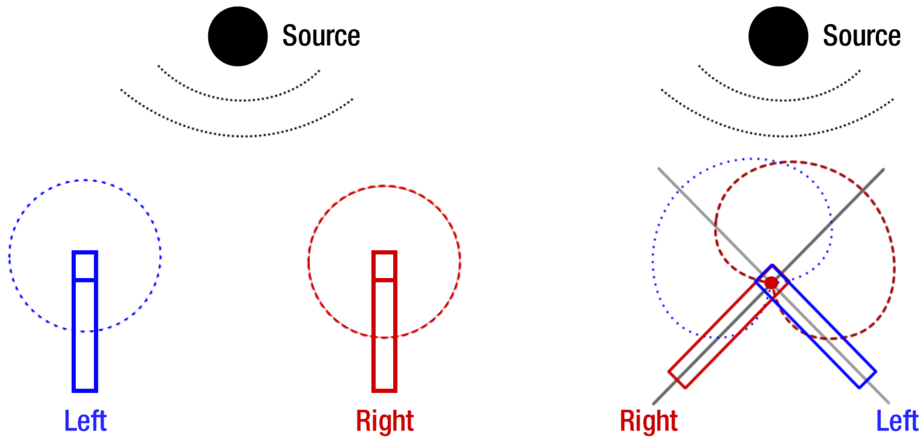
The mono-receiver approach can work reasonably well for some applications, particularly if the sound field is very diffuse at the receiver location, in which case there is no real benefit to using a more sophisticated set-up. From a practical standpoint, it may also be preferred in situations that call for a large number of receiver positions distributed throughout the space - as is often the case when rendering dynamic and interactive environments. However, creating a sense of spaciousness depends on information that cannot be delivered by a single impulse response such as inter-aural amplitude, phase and spectral differences.

A step-up from the mono-receiver set-up is therefore to record two impulse responses at different locations simultaneously, thus rendering a 'stereo-image' of the room response. Stereo recording techniques typically involve using a matched pair of condenser microphones that can be arranged in a number of different configurations. Two of the more popular classes of stereo recording technique which shall be mentioned here are the A-B or 'spaced pair' configuration, and the X-Y or 'coincident pair' configuration (see Figure 5.7).

A-B recording techniques involve the use of two identical (often omnidirectional) microphones positioned in parallel. The timing and amplitude differences captured at either microphone act as spatial cues and, depending on the relative distances between the microphones and the source, can facilitate reasonably accurate source localisation on the azimuth plane. A main consideration when using this technique is the separation distance between the microphones which should be chosen with reference to the dimensions and proximity of the source, the acoustic properties of the recording space, and the desired width of the stereo image.

A disadvantage of the A-B technique is that the timing differences between signals arriving at either microphone can result in undesirable phase ambiguities. The X-Y configuration differs in this respect as it relies only on differences in sound pressure level and not time-of-arrival. It involves placing two identical microphones, typically with cardioid polar responses, as close as possible to one another and facing each other at an angle of $90 - 135°$, depending on the size of the source and the desired effect [339]. The stereo image produced by this technique is generally less 'spacious' than that of the A-B technique, although the elimination of phase anomalies make it a more suitable technique for recording relatively narrow sources.

Both of these stereo recording techniques permit source localisation on a single plane, yet they are unable to capture both vertical and azimuth information simultaneously due to the two-



(a) The A-B or 'spaced pair' *(b)* The X-Y or 'coincident pair'

Figure 5.7: Popular stereo recording techniques.

dimensional arrangement of the microphone capsules. For headphone reproduction, it may still be possible to achieve a complete three-dimensional reconstruction of the acoustic image using two microphones - provided the filtering effects of the head and torso can be incorporated into the signal. This can be achieved most readily using a matched pair of omnidirectional condenser microphones placed either just outside or, better still, within the ear cavities of a real or artificial head. However, in the case of multichannel loudspeaker systems where one is potentially dealing with a large number of loudspeakers in some as yet unknown configuration, creating a realistic three-dimensional impression requires a more sophisticated recording technique by which one can extrapolate the requisite room impulse response for each loudspeaker feed. For obvious reasons it is not generally feasible to try to mirror the loudspeaker positions in the recording set-up, as tended to be the case for the previous techniques. Furthermore, if the objective behind the measurements is to render a dynamic auditory environment, then in order to match the resolving power of a human subject and create smooth motion transitions, a potentially large set of impulse responses may be required. Whilst measuring all these room impulse responses is obviously not practicable, there are a variety of spatial recording techniques in existence that were conceived with the intention of allowing reasonable approximations of the room impulse response to be made for any given point in the problem domain. These are typically based on a set of mathematical functions known as spherical harmonics which are solutions to the spherical harmonic differential equation [340]. Through the correct application of these functions, it is possible to interpolate the impulse response data at positions between the actual measurement points [341].

In order to apply the principles of spherical harmonics, both the sound pressure and particle velocity components of the incoming signal are needed, hence the receiver must essentially function as a kind of three-dimensional intensity probe - intensity being the product of sound pressure (p) and particle velocity (u) according to the following relation: $I = pu$. There are various microphones on the market which have been specially designed for this purpose, the most popular of which to date are the range of Soundfield microphones. These include models based on Gerzon and Craven's original patented design [342], as well as several more recent variations.

The early Soundfield microphones modelled on Gerzon and Craven's design are multichannel microphones comprising of four near-coincident capsules arranged in a tetrahedron. The first of the capsules is a pressure microphone with an omnidirectional polar response while the remaining three are pressure gradient microphones with a bi-directional or 'figure of eight' response (see Figure 5.8). These measure the sound pressure and particle velocity components along the three Cartesian axes respectively. These four signals are mapped to four audio channels, WXYZ, which together form the *B-format* signal. While the B-format has been defined specifically in relation to ambisonic surround sound systems (see Section 5.5), it can also be decoded to a variety of other formats [343]. The B-format impulse response therefore offers the potential for 3-D reconstruction of the sound field, and, theoretically, over any kind, number and position of loudspeakers.

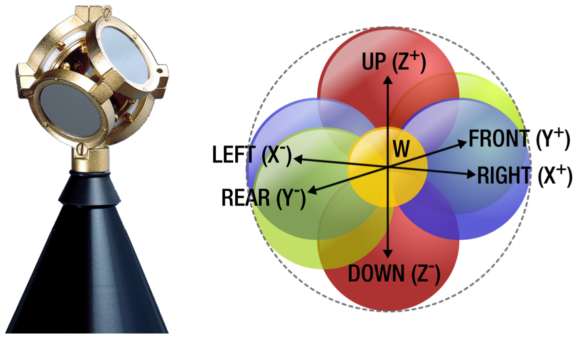


Figure 5.8: The early Soundfield microphones consist of 4 coincident capsules arranged in a tetrahedron. Photograph copyright ©Soundfield 2012.

An alternative way to measure B-format impulse responses that does not require the use of such specialised microphone technology has been presented by Farina and Tronchin [322]. Their method makes use of a single omnidirectional microphone, and involves making seven successive room impulse response measurements from closely spaced positions located on a 3-D crux. These correspond to a central reference position and a pair of positions along each of the 3 Cartesian axes. The pressure gradient along each axis may then be calculated using a similar algorithm to that employed in an ordinary pressure-difference intensity probe [322].

However, while B-format room impulse responses can create a good impression of the sound field at a central listening position, the stability of the image over a larger area may not be as good as if one were to involve a greater number of measurement channels - potentially one for each loundspeaker channel. This may relate to a lack of signal decorrelation which can have various effects on the perception of spatial imagery [344]. From a practical point of view, the acquisition of impulse responses for different loudspeaker layouts can present a problem, particularly if the intention is to render a dynamic auralisation, such as a walkthrough, which may require a very large number of impulse responses. There have been several different approaches suggested to the interpolation of room impulse responses from a minimal number of measurement positions, although none are considered wholly satisfactory. One such method presented by Adjler et al.

[345] involves rotating a microphone at constant speed along a circular trajectory. Their technique is based on the supposition that the maximal angular speed of the microphone can be written as a function of the length of the impulse response, its maximal temporal frequency, the speed of sound propagation, and the radius of the circle. Thus, with the right selection of set-up parameters, the room impulse response can be reconstructed for any point along the microphone trajectory.

The last of the recording techniques to be considered here is one which attempts to deal with a problem that is often encountered in commercial applications of auralisaion - namely, that the reproduction system is often an unknown entity. In these situations it may be necessary to make multiple measurements to better accommodate the end-user, which can prove very time-consuming, particularly when measuring in large venues or auditoria. Farina et al. have therefore proposed a method whereby three different recording systems are mounted on a rotating boom, thus allowing a complete set of impulse responses to be measured simultaneously for a number of different angular positions [346]. The systems used in the set-up are a Soundfield microphone, a binaural dummy head, and a stereo pair in the ORTF configuration (see Figure 5.9). This particular method is not only compatible with the standard reproduction formats, but also offers the potential for use with non-standard loudspeaker arrays, due to the flexibility afforded by the Soundfield microphone.

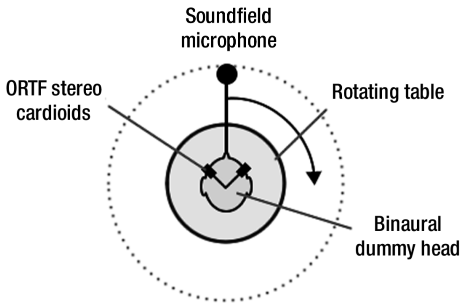


Figure 5.9: An approach involving simultaneous impulse response measurements with a Soundfield microphone, binaural dummy head and crossed-pair at different angles of rotation offers greater flexibility in terms of the reproduction system.

5.4.2 Simulation

In situations where the direct measurement of an impulse response is either not possible or not economically viable, it may be more appropriate to develop a numerical model of the acoustic system and derive the impulse response through a process of simulation. The numerical modelling techniques typically used in virtual acoustics for the generation of synthetic impulse responses tend to fall into one of two main groups. The first of these groups consists of the geometrical models, while the second comprises the wave-based algorithms (see Figure 5.10). Geometrical and wave-based models each have their associated limitations, and, as a result, it can sometimes be feasible to use both in complement with each other.

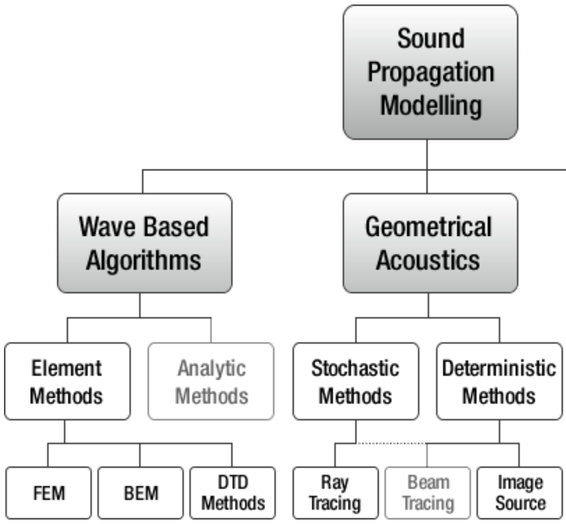


Figure 5.10: Simulation models for sound propagation. Reproduced from [309].

Geometrical Models

The first group of simulation models to be considered here are geometrical models. These models are based on the assumption that, under certain conditions, sound may be characterised by the movement of particles of energy in straight lines which decrease in intensity according to the inverse square law. This makes them approximately analogous to the ray-based methods used in optics whereby narrow beams of light are used to model the propagation of light through an optical system. So-called *sound rays* adhere to the same laws of reflection as light rays, and consequently suffer from the same limitations. Namely, that the ray analogy is only valid for situations where the wavelengths of sound are small compared to the area of any reflecting surfaces and large compared to their roughness. For this reason, frequencies are generally limited to those above the Schroeder frequency - an approximate crossing-point below which the behaviour of the system is characterised by individual, well-separated resonances, as opposed to many overlapping normal modes[75]. For airborne sound in reverberant enclosures, the Schroeder frequency is given by:

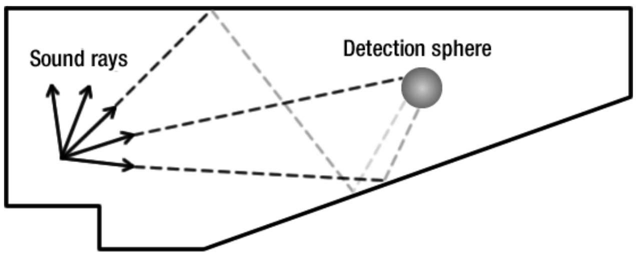
$$f_c \cong 2000\sqrt{T_{60}/V} \tag{5.13}$$

where T_{60} is the -60dB reverberation time in seconds and V is the volume of the enclosure in cubic metres [347].

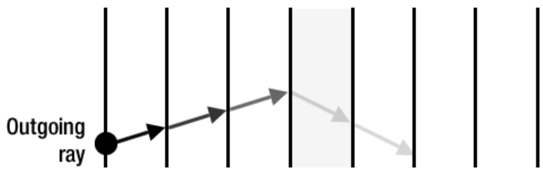
Furthermore, ray theory negates the wave theory of sound, and, by implication, phenomena such as diffraction and phase interference are precluded. Nevertheless, geometrical models can predict the earlier portions of the impulse response with reasonable accuracy for relatively simple room geometries, and are generally very computationally efficient compared with other techniques[348]. Two of the most popular implementations of the geometrical model will be considered here: the stochastic ray tracing method and the image source method. These are generally distinguished as stochastic and deterministic methods respectively.

Stochastic Ray Tracing The Stochastic Ray Tracing (SRT) method attempts to determine the spreading and depreciation of sound energy by considering the sound radiated from a source, as bunches of particles travelling in straight lines and in various directions[309]. Each particle has initial energy and direction of propagation values associated with it which are updated at discrete intervals according to the time elapsed since radiation and the absorption coefficient of any boundaries encountered. An air attenuation coefficient may also be included in the model.

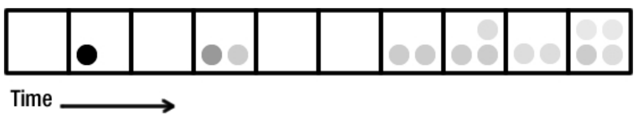
The receiver is represented by a detection sphere, which counts the crossing particles and stores their values in an energy histogram. The histogram can thus be considered an approximation of the system's impulse response at the detector location. Generally speaking, sounds are assumed to originate from point sources which are characterised by sound power and directivity. The dimensions of the source are usually ignored.



(a) In SRT, sound is radiated in regular impulses from a specified point in the simulation domain. These sound 'particles' have initial energy and direction of propagation values associated with them.



(b) At discrete intervals, a particle is advanced over a short distance and its values are updated according to the time elapsed since radiation and the absorption coefficient of any boundaries encountered.



(c) A detector counts the crossing particles and stores their values in an energy histogram.

Figure 5.11: The Stochastic Ray Tracing (SRT) simulation method.

In SRT sound is radiated according to Lambert's law of intensity, also referred to as the cosine law, which states that the intensity observed from an ideal diffusely reflecting surface is directly proportional to the cosine of the angle θ between the observer's line of sight and the surface normal[349] (see figure 5.12). For a ray-tracing case, the cosine law may be stated as:

$$I_r = \frac{I_0 dS\, cos(\psi) cos(\theta)}{\pi r^2} \tag{5.14}$$

where I_r is the reflected intensity at the surface, I_0 the incident intensity at the surface, θ is the angle of the receiver to the surface normal, ψ the angle of the source to the surface normal, dS the area of the surface being considered, and r the receiver radius[350]. Advantages of the technique are its speed and efficiency, coupled with its consideration of surface effects such as absorption and scattering. Furthermore, computation time does not necessarily increase with the number of sources.

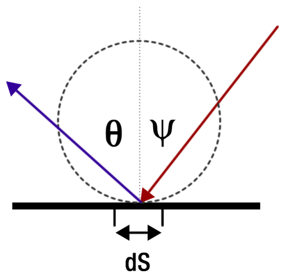


Figure 5.12: Cross-section view of Lambert's law of intensity. Reproduced from [350].

The main drawbacks associated with SRT are its neglect for wave effects, and the need to execute the SRT algorithm repeatedly for individual frequency bands due to the frequency dependence of sound absorption. Furthermore, the assumption of a diffuse sound field - i.e. one in which the energy density is the same everywhere, and all directions of sound propagation occur with the same probability[351] can cause the accuracy of the model to suffer for source and receiver positions close to boundaries or discontinuities[352]. This is a consequence of a well known effect sometimes referred to as the Waterhouse effect, whereby an increase in the sound pressure level is observed at distances less than $\frac{\lambda}{4}$ from a boundary surface.

Image Source Method A second popular geometrical acoustic modelling technique is the Image Source Method (ISM). The ISM attempts to synthesise the impulse response at a given point by treating reflections as independent direct sounds originating from phantom 'image sources'. The sound paths can be represented as unidirectional rays if the room boundaries are removed and the path of the ray extended outside the propagation area. In this way the spreading of energy is determined by distance as a function of the divergence law, and the absorption coefficient of the boundary. The divergence law, also known as Gauss's law, is a conservation law originating in electromagnetics which states that the net flow of a fluid flowing over some area is the sum of all sources minus the sum of all sinks. In acoustics this translates as the energy density or energy per unit volume resulting from a wave disturbance. For a plane wave, the energy density may be expressed by the following:

$$\omega = \frac{I}{\rho c^2} p^2 \tag{5.15}$$

where ω is the energy density, I is the intensity or energy flux vector, ρ, p and c are the density of the medium, sound pressure and speed of sound respectively.

Mirror images are created of the room and the source from each room boundary. Tracing the path of the direct sound from the image source to the receiver results in a ray of energy density and angle of incidence equivalent to a real specular reflection (see figure 5.13). However, at corners there are regions where the image source is not visible. Orders of image sources are grouped according to the number of walls hit (ie. image sources of n^{th} order correspond to rays hitting n walls). This method produces an impulse response for the room in which the direct and reflected sounds are easily distinguishable.

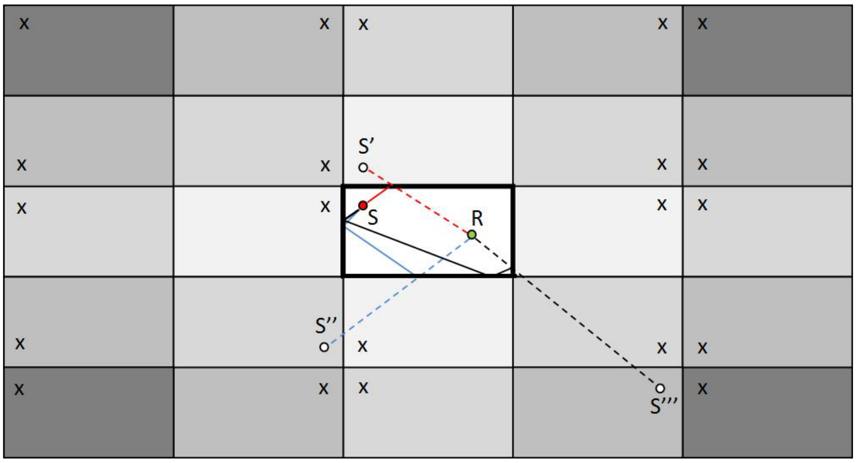


Figure 5.13: 2-D graphical interpretation of the Image Source Method. Source and receiver positions are shown within the boundaries of a simple rectangular space. Examples of 1st - 3rd order reflection are shown with their respective image sources positioned in fictitious spaces. The reflection order related to each space is denoted by shade where white denotes 0th order and dark grey denotes 4th order). Further examples of valid image source positions are denoted by black crosses [353].

Some limitations associated with the image source model include the assumptions that sound travels in a straight line, and that the room reflections are purely specular. Nor does it genuinely take account of phase changes on reflection. Therefore it is most accurate for rooms with hard flat surfaces where phase changes are minimal. Furthermore, the practicalities of implementation mean that it is generally not feasible to compute the entire impulse response using this method. There are two main reasons for this: firstly, the number of audible image sources increases by a third power with time; and secondly, the overall number of image sources increases exponentially with n order of reflections. Therefore there must be a truncation of the image source at a certain order so as to limit computation time. Computation time can be estimated beforehand from the time taken to complete the point-in-polygon test [354] - an algorithmic method to find out whether a point is inside or outside a polygon judging by the number of times a path traced from the point crosses one of the shape's boundaries. An odd number signifies it is inside the polygon, while an even number signifies the point is outside. Hence the more complex the room geometry, the longer this test will take to perform.

In a typical room the scattered energy will dominate the specularly reflected energy from $n \geq 3$ [309]. The reflection coefficient is given to be a constant (i.e.. the reflection factor is angle independent). This implies the angle of incidence is also a constant as is consistent for a quasi-plane wave. This assumption only holds when the distance between the source and the boundary and the receiver is large. For smaller distances the resulting diffraction cannot be determined by the grazing angle with any certainty. The audible consequences of this uncertainty cannot be predicted since the errors would depend not only on the room properties, but also on the nature of the signal.

Most image source implementations neglect diffraction, for the simple reason that diffraction is a wave phenomenon and, as such, is precluded in basic ray-based algorithms. There have been various adaptations to the model which attempt to include diffraction effects, often for the purpose of predicting noise barrier performance where diffraction is obviously a main concern. For example, Svensson et al. [355] have proposed an analytical method to obtain the impulse response behind a barrier. Their method applies the concept of secondary edge sources, wherein every point at the edge of the barrier is considered a sound emitter. They show that the analytical directivity functions for such edge sources can be derived. The other cause of diffraction in room acoustics is sound encountering edges at surroundings of the room boundaries. In such cases diffraction may be due to changes in characteristic surface impedance, or the physical geometry of the room boundary. Despite these limitations, the method is preferred by a lot of commercially orientated CAD software due to the relative accuracy and speed of computation in most rooms.

Since both types of geometrical model have their associated limitations, it therefore makes sense to combine them, typically relying on a deterministic model for calculation of the first 100ms of the impulse response, and a stochastic method for later reflections [309]. These so-called hybrid models can offer a distinct improvement over the traditional techniques, and are the approach taken by some of the more popular room acoustic modelling software such as CATT-acoustic [356], ODEON [357] and EASE [358].

Wave-based Models

In contrast with geometrical models, wave-based models are founded on the wave theory of sound, much of which is derived from classical mechanics and thus common to all forms of wave phenomena (i.e. light, water etc.). In wave theory, the propagation of sound is described by the acoustic wave equation, also referred to as the *Helmholtz equation*, which is a second order partial differential equation describing the change in acoustic pressure p or particle velocity u as a function of position r and time t [359]. There are numerous formulations of the wave equation describing systems of different complexity. A simple form of the equation describing the motion of sound waves in one spatial dimension (x) in an ideal fluid is given by the following:

$$\frac{\delta^2 p}{\delta x^2} - \frac{1}{c^2}\frac{\delta^2 p}{\delta t^2} = 0 \tag{5.16}$$

where c denotes the speed of sound according to the relation $c = \sqrt{\frac{\kappa}{\rho}}$ in which ρ and κ are the density and compressibility of the medium respectively.

Wave-based methods essentially attempt to approximate solutions to the wave equation for all points across a region of space. While they are the most theoretically correct group of methods, they can also be the most costly to compute depending on the size and density of the coordinate system used in the model - a factor which, by association, also dictates the upper frequency bounds of the system. For this reason they are generally better suited to modelling low frequencies and/or spaces lacking in fine geometric detail. This is due to the need to resolve very small wavelengths of the scale of the surface discontinuities in order to describe the redistribution of high frequency sound due to scattering.

Within the group of wave-based methods are those that are implemented in the frequency domain, such as FEM and BEM, and those that are performed in the time domain, such as FDTD and DWM. The latter are broadly referred to as Difference Time Domain (DTD) methods and tend to be preferred in acoustics due to their efficiency and ease of implementation. For the sake of completeness, all four are briefly considered here.

Finite Element Method The Finite Element Method (FEM) provides a formalism for generating discrete (finite) algorithms for partial differential equations [360]. With this technique, the entire area over which the sound propagates is essentially broken down into a network of discrete, finite size elements. Each element is modelled as a damped mass-spring system and is connected to adjacent elements at junctions called nodes. Together the elements form a mesh which may be programmed to respond under certain loading conditions (i.e. acoustic pressure). Nodes in the mesh should be of a sufficient density to represent all relevant geometrical details in the simulation domain - i.e. every corner or surface detail that is not small compared to the shortest wavelength. The FEM can also be optimised by the introduction of additional variables (e.g. temperature, magnitude). In this way the whole structure can be represented by a large number of simultaneous equations which, in matrix form, allow for equations of motion to be applied to the system as a whole. A detailed description of the formulation for acoustic scattering problems is given in [361].

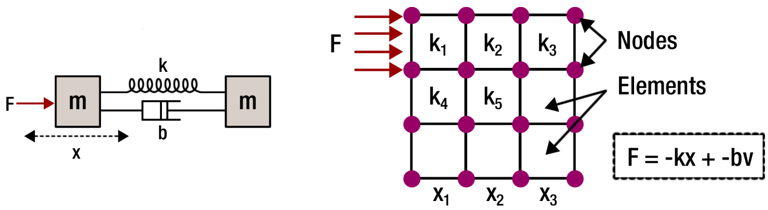


Figure 5.14: In Finite Element Method (FEM) each element is modelled as a damped mass-spring system (a). Elements are connected at nodes and can be programmed to respond under different loading conditions (b). In the above, x denotes the displacement from equilibrium, F is the force applied to system, m is the mass of an element, b is the damping constant, v is velocity and k is the spring constant.

The mass loading of the spring and the spring constant k of each element determine the resonant frequency of the system. Hence the system of equations must be solved for discrete frequency steps. The frequency response of the system is limited according to the proximity of the nodes, meaning that high frequency vibrations are more computationally expensive as a much larger number of elements are required to model the wave accurately. A consequence of the direct relationship between frequency resolution and computation time is that a trade-off must be made between mesh density in the frequency domain and computation time. A higher nodal density may also be a consideration in areas where a greater degree of mechanical stress is anticipated, such as those areas immediately surrounding a source.

Boundary Element Method A method that attempts to solve the high computational expenditure of the FEM whilst maintaining the geometrical integrity of the model is the Boundary Element

Method (BEM). The BEM is a technique for solving partial differential equations that have been formulated in boundary integral form. In acoustics, the traditional approach is to numerically approximate the Kirchoff-Helmholtz (K-H) integral equation which can be derived from the original Helmholtz wave equation using Green's theorem [14].

$$c(x)p(x) = -\int_s i\rho\omega v_n(x_s)g(x_s|x) + p(x_s)\frac{\delta g(x_s|x)}{\delta n}ds \tag{5.17}$$

where $c(x)$ is a position dependent constant, $p(x)$ is the complex pressure amplitude (with $\exp^{i\omega t}$ time dependence) at location x, $i = \sqrt{1}$, ρ is the fluid density, ω is the angular frequency, $v_n(x_s)$ is the normal surface velocity at location x_s and $g(x_s|x)$ is the free space Green's function relating locations x and x_s [362].

In BEM, it is the boundary or surface of the problem domain which is discretised, as opposed to the cavity or space inside. The key idea is that the acoustic field is represented as a superposition of fields due to elementary (monopole, dipole) sources located on the surface [363]. An integral equation taking the values of variables specified for each of these boundary elements represents a solution to the governing Helmholtz wave equation, thus allowing the values at any point in the volume to be derived. While this method is extremely useful for modelling relatively small systems where surface detail is a non-trivial concern, it is less well suited for room acoustic modelling where there is typically a much higher volume to surface ratio.

Disadvantages of BEM are storage requirements as a complex function must be stored with each element - a problem which is exacerbated in the event of diffraction effects and specularities being included in the model. Furthermore, the upper frequency limit is dependent on the size of the elements. As a general rule, a minimum of six elements per wavelength are required in 3-D BEM [362].

Finite Difference Time Domain Method In traditional FEM an approximate solution to the wave equation is solved for a specific frequency. By contrast, FDTD methods may be performed across a wide range of frequencies in a single simulation, thus saving on computation time. With these methods the vector components in a volume of space are solved at a given instant in time. The process is then repeated for the next moment in time and so on. For this to happen the wave equation is discretised and numerically approximated using second order finite differences. The acoustic pressure at each grid point is thus determinable at each step in time.

Discretisation of the 1-D wave equation given in (5.16) yields the following FDTD update equation, in which p is sound pressure, ρ is the fluid density, c is the speed of sound, v is the particle velocity, and n is a discrete-time index.

$$p^n[x] = p^{n-1}[x] - \rho c^2\frac{\Delta t}{\delta}\left(v^{n-\frac{1}{2}}[x] - v^{n-\frac{1}{2}}[x-1]\right) \tag{5.18}$$

The above can be readily extended for higher spatial dimensions. The method is covered in depth in Chapter 6.

Digital Waveguide Mesh A variant of the FDTD method is the Digital Waveguide Mesh (DWM). In this technique, the acoustic pressure at each grid point is calculated using a travelling wave solution to the wave equation as opposed to finite differences, meaning each grid point has associated incoming and outgoing pressures. The difference between the incoming and outgoing pressure values corresponds to the particle velocity at that point. The method is based on a multi-dimensional extension to the 1-D digital waveguide. A digital waveguide is a computational model of a physical structure through which sound waves propagate. It is based on the discrete form of D'Alambert's solution to the one-dimensional wave equation which realises that the vibration of an ideal string can be described as the superposition of two travelling waves of constant velocity moving in opposite directions along the x-axis. In a lossless media the 1-D digital waveguide can be expressed as:

$$u(x,t) = F(x+ct) + G(x-ct) \tag{5.19}$$

where $u(x,t)$ is the transverse displacement of the string as a function of time and position, ct is a constant and F and G are arbitrary functions representing right and left-travelling waves respectively. A Finite Difference Time-Domain (FDTD) approximation of (5.19) leads to (5.20) wherein the sound pressure (p) at any point can therefore be approximated by the sum of neighbouring values at the previous time-instant, minus its own value two time-instants before. n is a temporal index according to $t = nT$ and m is the spatial index according to $x = mX$ in which X and T are the spatial and temporal sampling intervals respectively.

$$p(m,n) = p(m+1,n-1) + p(m-1,n-1) - p(m,n-2) \tag{5.20}$$

Thus a bidirectional delay line (a digital waveguide), may be used to model a one-dimensional linear acoustic system. The approach is an intuitive one, particularly when one considers the system in terms of its characteristic sound paths. In the simple case of a plucked string, this equates to two travelling waves moving in opposite directions and being reflected back at each of the string terminals. In Figure 5.15, two waveforms travel through the delay lines and are reflected at reflection filters $R_f(z)$ and $R_b(z)$ which produce phase inversion and slight frequency-dependent damping. The net flow of energy through the system is evaluated at an arbitrary pick up point at the sampling rate f_s (samples per second), yielding the output signal $y(n)$ represented in the DWG by the delay line. Applied to a linear system, this form of implementation retains the mathematical property of commutability, such that the losses can be lumped to one place in the system, thus minimising computational and hardware expenditure. However, the benefits of commutability are lost when moving to higher dimensions, as when modelling 2-D and 3-D meshes. The so-called 'lumped model' is also inappropriate when dealing with complex systems for which different interacting components must be modelled separately and combined.

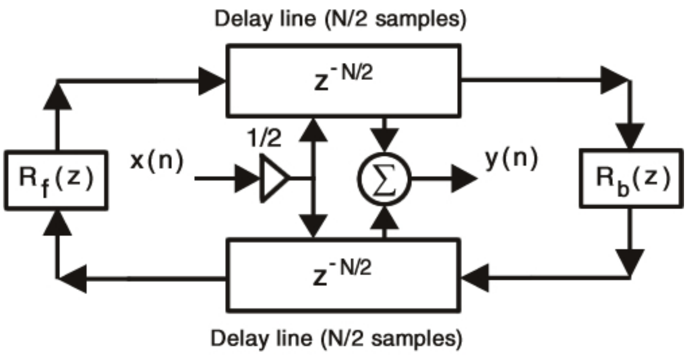


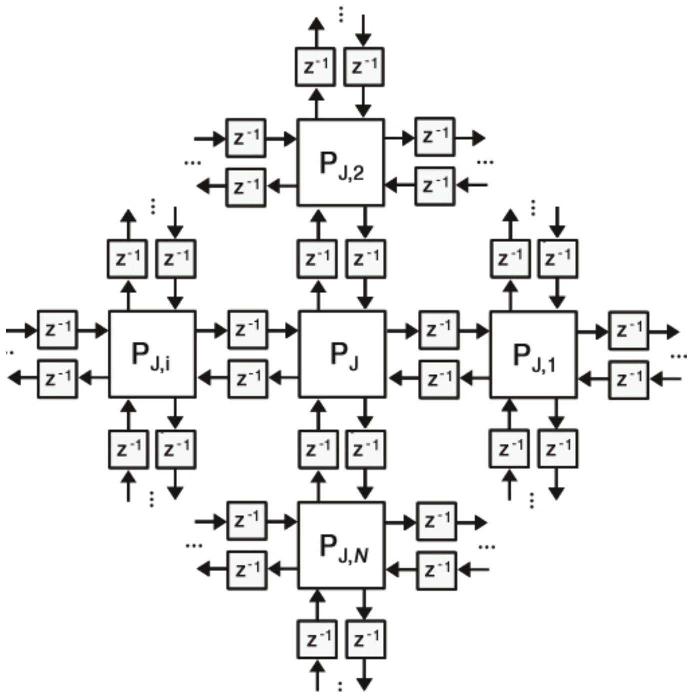
Figure 5.15: A bidirectional delay line.

Following the same method as in (5.20), a FDTD approximation of the 2-D acoustic wave equation can be constructed [364]. Spatial coordinates x and y are indexed as m_x and m_y respectively and n is the discrete time step, such that the pressure p at a point is:

$$\begin{aligned} p(m_x, m_y, n) = p(m_x, m_y + 1, n-1) \ + \ p(m_x + 1, m_y, n-1) \ + \ p(m_x, m_y - 1, n-1) \\ + \ p(m_x - 1, m_y, n-1) \ - \ p(m_x, m_y, n-2) \end{aligned} \tag{5.21}$$

In a 2-D rectilinear mesh, each incoming wave component is the same as the outgoing component from the adjacent node at the previous time-step. The future value of any given node can be calculated from the four outgoing waves at the surrounding nodes. The nodes are connected at multi-port junctions known as *scattering junctions* via bidirectional delay elements, as in the 1-D case. Hence at any given instant in time, a given junction J with N neighbouring nodes will have incoming wave variables $P_i^+(x_0, n)$ and outgoing wave variables $P_i^-(x_0, n)$ [365]. The sound pressure P_J at junction J for N connected waveguides of admittance Y_i can be expressed as:

$$P_J = \frac{2\sum_{i=1}^{N} Y_i \cdot P_{J,i}^+}{\sum_{i=1}^{N} Y_i} \tag{5.22}$$



Whilst the 2-D DWM lends itself well to modelling the modal resonances of stretched membranes, a 3-D model can be used to simulate sound propagation inside resonant enclosures. Both have been found to have useful applications across various fields in acoustics, including room acoustic modelling [366, 367]. However, the DWM is perhaps best suited to modelling small systems such as vocal tracts and musical instrument cavities, as the very dense meshes needed to measure wavelenths that are very small in comparison to the size of the system, become more computationally feasible [12]. Further efforts to improve the efficiency of the DWM have yielded the Kirchhoff variable DWM (K-DWM), a FDTD formulation of the traditional W-model (W-DWM). In the K-DWM, junction pressure values are expressed purely as physical quantities rather than sampled travelling wave components [368].

5.5 Spatial Sound Representation

The final stage of auralisation concerns the reproduction of the audio over a suitable delivery system. Sound is a spatial event, and so to experience it in its authenticity, the reproduction system should aim to preserve the distance and directional cues that are unique to a given source - space - receiver configuration. Advanced spatial sound reproduction systems tend to belong to one of three main categories:

- Binaural synthesis
- Sound field synthesis, which includes techniques such as Wave Field Synthesis and Higher Order Ambisonics

- Perceptually motivated approaches, which include techniques such as Vector Based Amplitude Panning (VBAP) and Directional Audio Encoding (DIRAC) [369].

5.5.1 Binaural Synthesis

Binaural technology is based on the theory that, if the sequence of sound pressure variations at the eardrums can be captured and reproduced exactly, then the listener will perceive the sound as if they had been present at the time when the event took place. Binaural signals may be produced artificially by convolution of the anechoic signal with a Binaural Room Impulse Response (BRIR), in which case the technique is referred to as binaural synthesis. To achieve the best possible sense of realism the BRIR must consider the influence of the environment or room, as well as the influence of the human body on the perception of sound. The former is dealt with by the room impulse response, while the latter is dealt with by a pair of Head Related Transfer Functions (HRTFs). These are generally obtained empirically using either a real or artificial head and torso, but may also be obtained through simulation (eg. [370, 371]). Synthesis of dynamic, interactive auditory environments is also possible using binaural synthesis if a head-tracker or other form of motion sensor is worn. In dynamic binaural synthesis, the BRIR is updated from a bank of BRIRs according to the location and/or orientation of the listener.

Binaural signals are generally presented over headphones, but may also be presented over loudspeakers, in which case the influence of the presentation room must also be considered. Given the complications associated with loudspeaker delivery, binaural synthesis is normally more successful when presented over headphones, although there are drawbacks to this method too. These are predominantly related to the physical and psychological restrictions associated with headphone wearing. For example, some listeners find the physical sensation of wearing headphones uncomfortable and/or unnatural, which can interfere with their sense of immersion in the virtual environment. It may also be restrictive in terms of their movement and vocal interaction with other listeners. Furthermore, headphone listening is potentially a very isolating experience, which can be undesirable in some situations. This isolation may be reinforced by the sensation of sound originating ‘inside the head’ when the HRTFs are not particularly well suited to those of the individual. In fact, the success of the binaural synthesis method is dependent to a large extent on how well the HRTFs match those of the listener, and it is for these reasons in particular that multispeaker systems are often preferred for many auralisation applications.

5.5.2 Sound Field Synthesis

Sound field synthesis techniques attempt to reproduce the sound field over a region of space using an array of loudspeakers. A successful three-dimensional reconstruction of the sound field within a given listening area ought to create an impression of the virtual source being fixed in space and sharing no spatial relationship with the listener, such that the listener may move freely within the space whilst the source does not appear to shift. Implementation is of course complicated by the need to suppress the non-linear characteristics of the listening space.

The more traditional approaches to spatial sound rendering require a measurement position in the corresponding location for each loudspeaker in the array. This approach has largely been superseded by the coincident recording techniques from which one derives the loudspeaker feed and/or related impulse response using mathematical techniques. This is typically achieved using either spherical harmonic decomposition (as in the case of ambisonics) or Kirchhoff-Helmholtz integral (used in Wave-field Synthesis). These each have their associated limitations, hence in recent years new methods of spatial encoding have been developed which are considered to sound more natural in listening tests.

Ambisonics

Ambisonics is a series of spatial sound recording and reproduction techniques developed by Gerzon and Fellgett in the 1970s [372, 373], and was intended to overcome the main problems associated with the quadrophonic surround sound systems around at that time. In its simplest form, known as first-order ambisonics, the sound information is encoded into four channels: W, X, Y and Z, corresponding to the sound pressure and pressure gradient components of the sound field at a point in space. As mentioned in Section 5.4, these four channels form what is known as the B-format signal - a format based on a spherical harmonic decomposition of the sound field.

Although initially designed for a square speaker array (see Figure 5.16), the B-format signal can be decoded to any number of loudspeakers, after which it is referred to as the D-format. D-format signals are derived from the B-format signal using simple matrix multiplication [374], which for regular loudspeaker layouts is given as:

$$D = \frac{\sqrt{2}}{N} \begin{vmatrix} \frac{1}{\sqrt{2}} & \dots & \frac{1}{\sqrt{2}} \\ \cos(\phi_1) & \dots & \cos(\phi_L) \\ \sin(\phi_1) & \dots & \sin(\phi_L) \\ \cos(2\phi_1) & \dots & \cos(2\phi_L) \\ \sin(2\phi_1) & \dots & \sin(2\phi_L) \\ \dots & \dots & \dots \\ \cos(L\phi_1) & \dots & \cos(L\phi_L) \\ \sin(L\phi_1) & \dots & \sin(L\phi_L) \end{vmatrix}^T \tag{5.23}$$

where the L is the number of loudspeakers, N is the decoding order, and ϕ is the angle of the i^{th} loudspeaker [375].

Higher order ambisonics use higher orders of spherical harmonics and require more channels than are consistent with the B-format signal. As the measurement of higher order signals is inherently limited by the microphone technology, it is often necessary to approximate the higher order signals from a B-format signal. Different decoders achieve this with varying degrees of success.

The main advantages that ambisonic multichannel surround sound systems offer over some of the more common systems such as 5.1, are the inclusion of height information, and the ability to place the loudspeakers in convenient positions - albeit to a limited extent. It can also provide more natural localisation of sources due to the fact that all speakers are generally used to localise a sound, unlike the studio panning techniques used by mixing engineers to position sounds for

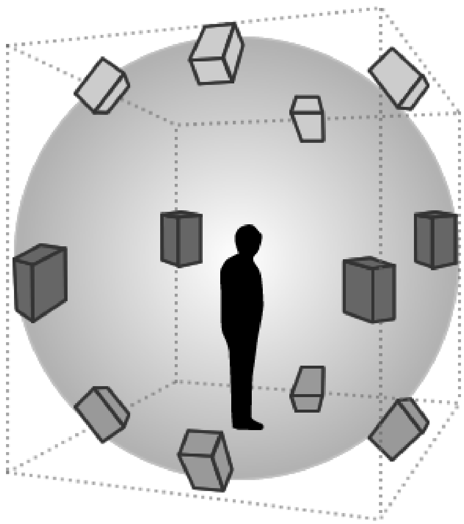


Figure 5.16: A first order ambisonics surround sound system.

playback over conventional systems. It was initially claimed by early proponents of the system that the surround-sound image produced by an ambisonic system is largely unaffected by listener position, although, nowadays, this is not generally considered to be the case.

In recent years ambisonics has been criticised on account of its inherent instabilities which are partially attributed to the disruption caused by the presence of the listener, and partly due to the fact that the reconstruction is theoretically correct in a central 'sweet spot', outside which the listener may experience strange phase effects. Although these effects are reduced for higher order systems, the format has failed to win widespread approval, particularly since more recent technologies have been found to produce a more natural sounding reconstruction in subjective evaluations. Yet despite its limitations, one advantage the ambisonic format has over some of the more recent technologies is its ease of implementation and the availability of freely downloadable decoders - something which is largely due to the expiration of most of the patents associated with the format. This still makes it an obvious contender for a lot of spatial sound rendering applications, particularly those concerned with the reconstruction of relatively noisy and diffuse sound fields, where the ability to localise individual sources with fine precision may be less of a priority.

Wave Field Synthesis

Wave Field Synthesis (WFS) is an approach to spatial sound rendering, the theoretical basis of which was originally developed by Berkhout [376] in the late 1980's. The technique is based on Huygen's principle which states that any wave front can be regarded as the superposition of an infinite number of point sources (see Figure 5.17). Mathematically it is founded on the Kirchhoff-Helmholtz integral theorem (KHT), which expresses the wave field in a volume inside or outside a given surface in terms of the fields value on the surface [377].

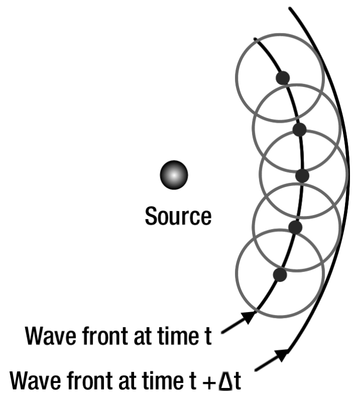


Figure 5.17: Wave Field Synthesis is founded on Huygen's Principle, which states that each point on the wave front gives rise to another spherical wave front and due to interference between these elementary wave fronts, a new composite wave front ensues.

Using the KHT, it is theoretically possible to reconstruct any sound field, provided that sound pressure and acoustic velocity are restored on all points of the surface of its volume, though this would require an infinitely large array of individual loudspeakers. In practice, a moderately large array can be used to good effect. In principle, the loudspeakers used in the reconstruction should be of the same spatial configuration as that of the receiver array used to measure the original wave front. In practice the positions of the receiver and loudspeaker arrays are different, thus a process of numerical extrapolation of the positions needs to be applied.

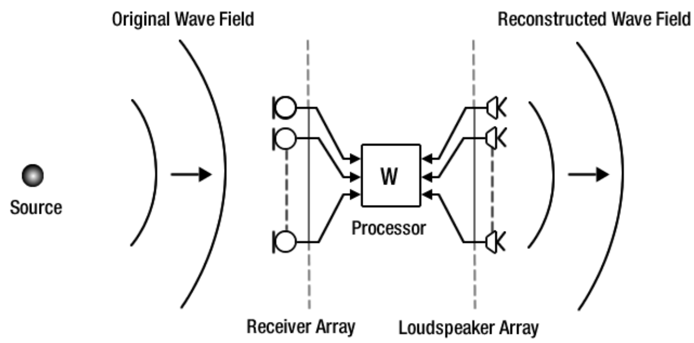


Figure 5.18: Wave Field Synthesis. Reproduced from [376].

WFS has the obvious disadvantage of being costly in terms of the number of transducers needed, and the need to make corrections for the resonant characteristics of the presentation room which can badly affect the quality of the reconstruction.

5.5.3 Perceptually-Based Spatial Encoding Techniques

Due to the limitations associated with the previous two techniques, other spatial sound rendering techniques have emerged in recent years which, rather than attempting to produce an exact replica of the original sound field over a region of space, attempt to stimulate a spatial impression through perceptually based processes. These typically involve analysing real or synthetic room impulse responses in order to derive perceptually relevant information about the sound field. The information can be used to generate the impulse responses that are associated with each loudspeaker feed. Spatial Impulse Response Rendering (SIRR) is one such method, in which time-frequency analysis is performed on room impulse responses in frequency bands to obtain directional and diffuseness information about the sound field. The non-diffuse sound is then reproduced as point-like sources using a vector base amplitude panning method [378], while the diffuse sound is reproduced using a combination of two different stochastic processes [379]. Another spatial encoding method developed more recently involves representing every sample in a set of room impulse responses recorded using a spaced array of microphones in order to extrapolate the required spatial information [380].

Both of these systems offer a distinct improvement over ambisonics in terms of their ability to create a more natural perception of space. What is more, they can be used with any loudspeaker system. However, these techniques are intended to reproduce the acoustics of reverberant spaces when applied to dry signals. Their applicability to the spatial rendering of outdoor sound fields which are inherently unstable as well as being dominated by direct sound is uncertain. It is thus considered that the sound field synthesis techniques may be more appropriate for rendering soundscape auralisations.

5.6 Summary

In this chapter, auralisation is described as the end product of a sequence of processes, placing particular emphasis on its association with virtual reality applications. The writer has then proceeded to outline the basic theoretical concepts underlying most auralisation processes, namely the characterisation of an acoustic system by its impulse response, from which one may predict the response of the system to an arbitrary input via the mathematical process of convolution. It has also been established that the requisite impulse responses are typically obtained through direct measurement or numerical simulation - both of which were subsequently addressed in Section 5.4.

All things being equal, impulse responses obtained from direct measurement remain unrivalled in terms of their authenticity, physical accuracy and perceived ‘naturalness’. Nevertheless, the time and effort required to obtain them means they are not feasible for many auralisation applications, particularly those concerned with rendering large-scale interactive environments and walk-throughs, which invariably require a large number of impulse responses. There may of course be other factors prohibiting direct measurement. For example, if the space of interest no longer exists, or indeed, has never existed; or if the conditions are unsuitable for measurement, as is often the case with public spaces. In situations such as these, the ability to generate synthetic impulse responses is highly advantageous.

The reader was thus introduced to the two main classes of acoustic modelling technique commonly used in auralisation for the simulation of impulse responses, each of which offers distinct advantages over the other. In brief, the wave-based approaches offer higher accuracy over the lower range of frequencies and include wave propagation effects such as diffraction and occlusion, while the ray-based techniques are more readily adapted for real-time implementation making them more appropriate for applications that demand real-time interactivity [381].

Assuming one has obtained the requisite impulse responses by whatever means best suit the application, the next stage in the auralisation process concerns the delivery format. Thus, in Section 5.5, a number of different reproduction formats and spatial sound rendering techniques were presented, and their respective shortcomings discussed. In light of those observations it is concluded here that the most natural approach to soundscape evaluation would involve the reconstruction of the sound field over a multichannel loudspeaker system. In this instance, producing an accurate spatial impression necessitates making suitable adjustments to compensate for the order and positioning of the loudspeaker array. Ideally the response of the listening room should also be taken into account, or alternatively, appropriate absorptive treatment installed to suppress the reverberation in the room.

There are of course numerous variations on the relatively small selection of auralisation techniques touched upon in this chapter, and while it would be ineffective to try to cover every variant, a general understanding of the main classes of technique and their respective differences is expected to justify the experimental approaches taken in this thesis. It is therefore concluded here that, in devising a strategy for auralisation, one should always consider one's primary objectives in relation to several important factors. These include - but are by no means limited to - the nature of the material being evaluated, the requirements of the system in terms of real-time interactivity, the operator expertise, and the available technology. All of these factors are considered over the remainder of this thesis in relation to the overriding research aims.

6

Experimental Methods

6.1 Introduction

Based on the review of soundscape and environmental noise literature presented in Chapters 3 and 4, the author considers there to be a strong case for the development and deployment of advanced auralisation techniques in relation to urban soundscapes. Moreover, they would argue that the auralisation of urban soundscapes holds possibilities beyond the domain of urban planning and environmental noise assessment procedures, to a plethora of potential public engagement and artistic initiatives aimed at strengthening human-environment relationships. The remainder of this thesis is therefore devoted to exploring how such techniques might be implemented in practice, and whether or not such approaches are really effective, either as reliable indicators of subjective impressions, or as a means of communicating a message or concept.

An experimental process has therefore been devised to assess the impact of less conventional noise treatments in a virtual acoustic environment, through a combination of acoustic modelling and spatial sound reproduction. The experiments performed here draw on one particular example of an unorthadox form of noise intervention: that of the sonic crystal noise barrier. The aim of this chapter is therefore to document the process of auralising the structure in an urban soundscape. The purpose of the auralisation is to enable a comparison of the perceived sound quality in the environment before and after insertion of the barrier. Whereas Chapter 5 presented an overview of auralisation and its associated techniques, this chapter goes on to give a more detailed explanation and justification for those that have been adopted here.

The chapter is organised as follows: firstly it documents the simulation methodology used to derive the transfer functions of the barriers; secondly it deals with estimating the ratio of incident to diffracted sound energy; and finally it deals with the auralisation and spatial sound reproduction methods used. While the process described here is considered flexible enough to deal with a variety of different scenarios, testing it involves tailoring it for one particular scenario. It shall therefore be implemented in the final chapter of the thesis involving a real-world case-study.

6.2 Experimental Design

A sonic crystal is a physical object composed of an array of periodically spaced scattering elements embedded in a fluid medium. The structure can be designed to give selective attenuation at certain wavelengths and angles of incident sound. In fact, depending on the nature of the material and the geometry of the array, it is theoretically possible to achieve a complete gap in the transmission over a select range of frequencies and multiples thereof [382].

The purpose of the experiment will be to model the propagation of sound through a simple two-dimensional sonic crystal array using wave-based numerical simulations. During simulation the impulse response function of the structure shall be measured, from which it will be possible to observe the acoustic band gaps in the frequency domain and verify the efficacy of the model with respect to simple theoretical considerations presented in Chapter 3. Acquisition of the impulse response is ultimately intended for the purpose of auralising the structure by convolution of the impulse response with appropriate source material.

The simulation set-up is loosely modelled on the experiments performed in [383] in which empirical results were obtained for an array of rigid cylindrical scatterers arranged in a square lattice configuration (see Figure 6.1). Essentially it involves modelling the propagation of sound through the structure by solving the discretised wave equation in algorithmic form across a three-dimensional numerical mesh. An array of rod-like scattering elements are included in the model by manipulating the governing update equations at the corresponding points in the mesh. The surfaces of the scatterers are assumed to be acoustically hard with a reflection coefficient tending toward 1. Results are obtained by exciting the mesh with an appropriate input function at a point in front of the array and recording the transmitted signal at the rear. The simulation is also performed without the sonic crystal insertion to enable a comparison of the transmitted signals before and after insertion.

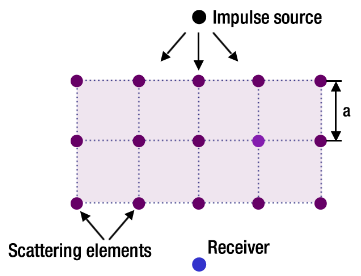


Figure 6.1: Measuring the transfer function of scattering elements arranged in a square lattice using the impulse response method.

Based on Bragg's law, the centre frequency of the band gap can be approximated using

$$f_c = \frac{c}{2 \cdot a} \tag{6.1}$$

where c is the speed of sound in air and a is the centre-centre distance between adjacent cells known as the lattice constant. Using a lattice constant of $a = 0.11cm$ and the assumption that

$c = 343m/s$, the centre frequency of the Bragg reflection event for the {1 0 0} set of planes in the square lattice is expected to occur at approximately 1559 Hz. The spacing between the {1 1 0} set of planes is $a/\sqrt{2}$, corresponding to a band gap centred at 2205 Hz. The bandwidth of any band gap is determined by the filling fraction - i.e. the fraction of the structure occupied by the cylinders [383]. For a square lattice arrangement, the filling fraction, F, is given by

$$F = \frac{\pi d^2}{4a^2} \tag{6.2}$$

where d is the diameter of the cylinders and a is the lattice constant. Experiments performed in [383] and [187] suggest that a filling fraction of around 0.3-0.4 is optimum for the appearance of a complete band gap in a square lattice.

6.3 Numerical Modelling with the FDTD Method

Of the various numerical modelling techniques presented in Chapter 5, it was decided that a Finite Difference Time Domain (FDTD) method would be used to model the propagation of acoustic waves through the sonic crystal barrier. This method has proved highly effective in a variety of problems intended for auralisation, largely on account of its speed, efficiency and the relative simplicity of the computer algorithms by comparison with other wave-based methods such as the Finite Element Method (FEM). Furthermore, it is more suitable for problems concerned with wave interference effects than geometrical methods more commonly used in auralisation. However, with this being an open domain problem, as opposed to the closed domain problems typically encountered in architectural acoustics, the basic formulation used for a static linear system requires some adaptation which can potentially compromise its efficacy. A significant proportion of the methodology that follows is thus aimed at demonstrating how this particular challenge has been dealt with here.

6.3.1 Mathematical Basis

The normal FDTD formulation for describing the propagation of longitudinal waves in an ideal fluid (i.e. zero viscosity and maximum homogeneity) uses the scalar Helmholtz acoustic wave equation or reduced wave equation, a linear second order partial differential equation based on Newton's second law of conservation of momentum [384]. Here, instead of acceleration there is the derivative of the velocity, instead of mass there is the mass density, and instead of force there is the derivative of pressure.

$$mass \times acceleration = force = -pressure\ gradient \tag{6.3}$$

The dynamics of a linear acoustic system may thus be described as local perturbations in the density $\rho(r, t)$, pressure $p(r, t)$, and mass velocity $v(r, t)$ of the medium where t is time and r

represents the position in one, two or three dimensional vector space. It is also assumed that the perturbations are small, so that

$$p = p_0 + p'; \; \rho = \rho_0 + \rho'; \; p' \ll p_0; \; \rho' \ll \rho_0; \; \left|v'\right| \ll c \sim \sqrt{\frac{p_0}{\rho_0}} \tag{6.4}$$

where p_0 is the atmospheric pressure of the medium at rest, ρ_0 is its characteristic density, p' and ρ' are the first derivatives of p and ρ with respect to position, $|v'|$ is the first derivative of the vector velocity with respect to position, and c is the speed of sound in the medium, assumed here to be much greater than the velocity of the medium [385].

Given the assumption that, in an ideal fluid, pressure is a function of density only (i.e. $p = p(\rho)$, it can be said that,

$$p' = c^2\rho'; \; c^2 = \frac{\delta p}{\delta \rho} \tag{6.5}$$

The governing acoustic equations in three dimensions are thus,

$$\frac{\delta p'}{\delta t} = -\rho c^2 \nabla \cdot v' \tag{6.6}$$

$$\frac{\delta v'}{\delta t} = -\frac{1}{\rho}\nabla p' \tag{6.7}$$

in which the invariant nabla operator is represented in Cartesian coordinates where i_x, i_y, and i_z are the Cartesian basis vectors [385].

$$\nabla = i_x\frac{\delta}{\delta x} + i_y\frac{\delta}{\delta y} + i_z\frac{\delta}{\delta z} \tag{6.8}$$

Taking the divergence of (6.7) and interchanging the order of temporal and spatial differentiation yields

$$\frac{\delta}{\delta t}\nabla \cdot v' = -\frac{1}{\rho}\nabla^2 p' \tag{6.9}$$

Differentiating (6.6) with respect to t and using (6.9) yields

$$\frac{\delta^2 p'}{\delta t^2} = -\rho c^2 \frac{\delta}{\delta t}\nabla \cdot v' = c^2\nabla^2 p' \tag{6.10}$$

Finally, after rearranging 6.10, one arrives at the scalar wave equation

$$\frac{1}{c^2}\frac{\delta^2 p'}{\delta t^2} = \nabla^2 p' \tag{6.11}$$

The velocity is a vector and satisfies the vector wave equation [385]

$$\frac{1}{c^2}\frac{\delta^2 v'}{\delta t^2} = \nabla^2 v' \tag{6.12}$$

6.3.2 Computer Implementation

To obtain an FDTD algorithm for acoustic propagation, the scalar pressure field $P(x, y, z, t)$ and vector velocity field $v(x, y, z, t)$ must be discretised in both time and space. The region of interest is thus represented in the digital domain by a matrix of points describing the spatial coordinates of connected elements in a grid (2-D) or mesh (3-D). The arrangement of the elements within the grid or mesh can be described in terms of its unit cell - i.e. the smallest spatial unit that is formed between adjacent elements. The material properties of the medium are the speed of sound c_0 and the density ρ, both of which can vary as a function of position by controlling the pressure and velocity coefficients associated with each element or node.

2-D Implementation

In the 2-D implementation, a suitable arrangement of nodes is given in Figure 6.2 wherein unit cells are arranged in a square grid. This is the simplest node arrangement to implement algorithmically as it assumes the spatial sampling interval Δ is uniform in both the x and y Cartesian axes (e.g. $\Delta x = \Delta y$).

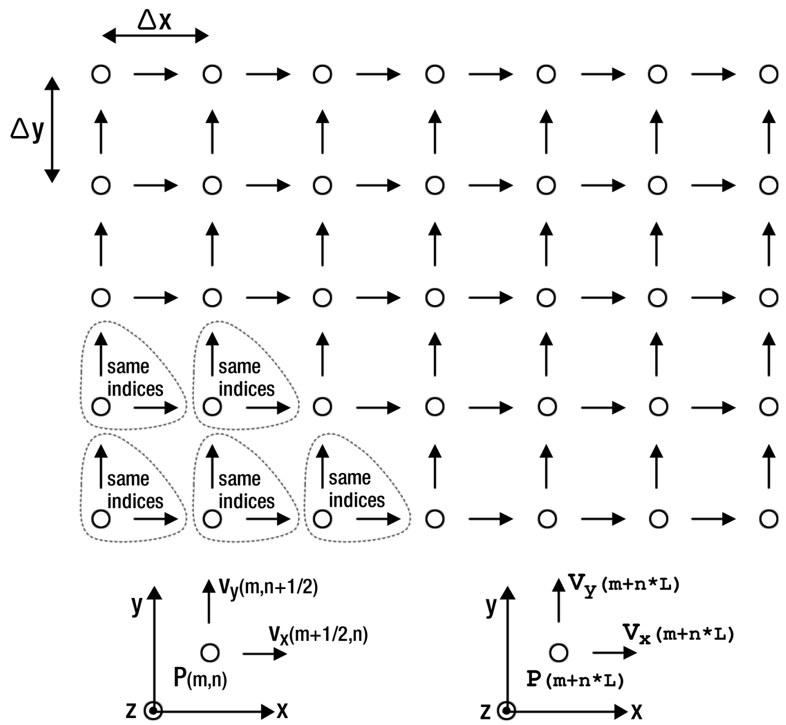


Figure 6.2: The arrangement of pressure nodes (circles) and components of velocity (arrows) in a 2-D FDTD grid. Below the grid are two depictions of a unit cell. The one on the left shows the nodes with the spatial offsets as per (6.16) while the one on the right shows the corresponding spatial indices used in the computer implementation. Reproduced from [386].

In each unit cell a pressure node is surrounded by velocity components such that the components are orientated along the line joining the component and the pressure node. It can be seen in Figure 6.2 that the pressure nodes are offset by a half spatial step from the velocity nodes. To obtain the necessary update equations, the pressure and velocity fields are also assumed to be staggered in time by a half temporal step, although the velocity components are not staggered relative to one another. However, when implementing the update equations in computer code, it is not necessary to make explicit reference to the spatial offset as it is is implicitly understood [386]. Hence the update equations that are used in the 2-D FDTD simulations consist of the following quantities, defined with an implicit understanding of spatial offsets:

$$p(x,y,t) = p\left(m\Delta_x, n\Delta_y, q\Delta_t\right) = p^q(m,n) \tag{6.13}$$

$$v_x(x,y,t) = p\left([m+1/2]\Delta_x, n\Delta_y, [q+1/2]\Delta_t\right) = v_x^{q+1/2}(m,n) \tag{6.14}$$

$$v_y(x,y,t) = p\left(m\Delta_x, [n+1/2]\Delta_y, [q+1/2]\Delta_t\right) = v_y^{q+1/2}(m,n) \tag{6.15}$$

Assuming a uniform grid in which the spatial step sizes are all the same, i.e., $\Delta x = \Delta y = \delta$, replacing the derivatives in (6.6) with finite differences and using the discretisation of (6.11) yields the following update equation [386]:

$$p^q(m,n) = p^{q-1}(m,n) - \rho c^2 \frac{\Delta t}{\delta}\Big[v_x^{q-1/2}(m,n) - v_x^{q-1/2}(m-1,n) + v_y^{q-1/2}(m,n) - v_y^{q-1/2}(m,n-1)\Big] \tag{6.16}$$

Similarly the x and y components of velocity are updated according to the discretisation of (6.12). The update equation for the x component is given below.

$$v_x^{q+1/2}(m,n) = v_x^{q1/2}(m,n)\frac{1}{\rho}\frac{\Delta t}{\delta}\left[p^q(m+1,n) - p^q(m,n)\right] \tag{6.17}$$

The next step is to translate the update equation given above into a computer algorithm that can be iterated over a discrete representation of the problem domain. Firstly, the supersript q may be dropped as time shall be represented by a single variable updated in an iteration statement or loop. Assuming a uniform grid, the pressure and xy velocity components are held in the arrays `P`, `Vx` and `Vy` respectively. Within the main iteration statement, the update algorithms for pressure and velocity are executed in nested statements over elements in the pressure and velocity arrays. In algorithmic form, the update equations can be written as follows:

```
P[m,n] = P[m,n] - ((dt/dx)*K)*(Vx[m,n] - Vx[m,n-1] + Vy[m,n] - Vy[m-1,n]);
Vx[m,n] = Vx[m,n] - ((dt/dx)/Rho)*(P[m+1,n] - p[m,n]);
Vy[m,n] = Vy[m,n] - ((dt/dx)/Rho)*(P[m,n+1] - p[m,n]);
```

in which `dx` and `dt` are constant variables denoting the spatial and temporal intervals respectively. Note that due to the relation $c = \sqrt{\frac{K}{\rho}}$, the coefficient of pressure ρc^2 may be represented by the single variable `K`, the bulk modulus of the medium.

3-D Implementation

Having implemented the FDTD in a 2-D grid, the additional steps needed to implement it in a 3-D mesh are fairly trivial [386]. A 3-D mesh is essentially a stack of 2-D meshes, offset by a half spatial step in the z direction, hence the methodology is almost identical to that of the 2-D FDTD. The only difference is the addition of a third velocity array housing the V_z component of the 3-D update equation given in (6.18).

$$\begin{aligned} p^q(l,m,n) = p^{q-1}(l,m,n) - \rho c^2 \frac{\Delta t}{\delta} \Big[& v_x^{q-1/2}(l,m,n) - v_x^{q-1/2}(l-1,m,n) \\ & + v_y^{q-1/2}(l,m,n) - v_y^{q-1/2}(l,m-1,n) + v_z^{q-1/2}(l,m,n) - v_z^{q-1/2}(l,m,n-1) \Big] \end{aligned} \tag{6.18}$$

In algorithmic form, the 3-D update equations thus appear as follows:

```
P[l,m,n] = P[l,m,n] - Rho * pow(c,2) * dt/dx * (Vx[l,m,n] - Vx[l-1,m,n]
+ Vy[l,m,n] - Vy[l,m-1,n] + Vz[l,m,n] - Vz[l,m,n-1]);
Vx[l,m,n] = Vx[l,m,n] - 1/Rho * dt/dx * (P[l+1,m,n] - P[l,m,n]);
Vy[l,m,n] = Vy[l,m,n] - 1/Rho * dt/dx * (P[l,m+1,n] - P[l,m,n]);
Vz[l,m,n] = Vz[l,m,n] - 1/Rho * dt/dx * (P[l,m,n+1] - P[l,m,n]);
```

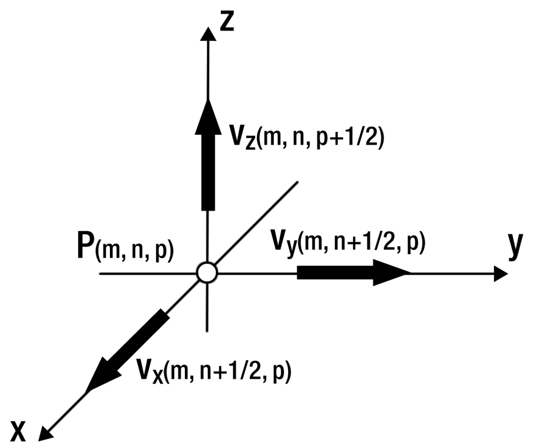


Figure 6.3: An acoustic unit cell in three dimensions showing the arrangement of velocity nodes relative to the pressure node with the same spatial indices. Reproduced from [386].

A complete C++ implementation of the acoustic 3-D FDTD method is included in the DVD accompanying this thesis (Directory 9.7.1).

Performance Limitations

A primary limitation of the FDTD method which is shared by other wave-based models, is a gradual loss of fidelity at high frequencies [387]. This is related to the fact that the grid must be sufficiently fine to resolve both the smallest wavelength and the smallest geometrical feature in the model. Intuitively, this points to a minimum spatial sampling interval of half the wavelength of the highest frequency of interest, in accordance with the Nyquist-Shannon sampling theorem.

The Nyquist-Shannon theorem is a fundamental theorem in communications theory which states that a continuous band-limited signal may be represented by a discrete sequence of samples without any loss of information [388]. It also specifies the lowest rate at which the samples must be recorded to allow an accurate reconstruction of the signal - a rate referred to as the *Nyquist rate* or *Nyquist frequency*. In mathematical terms, the Nyquist frequency may be expressed as:

$$f_s \geq 2f_c \tag{6.19}$$

where f_s is the sampling frequency and f_c is the highest frequency contained in the signal. Considering a continuous signal as the summation of a large number of sinusoids according to Fourier's theorem, this allows for a minimum of two samples per cycle constituent sinusoid. When a digital signal is sampled below the Nyquist rate, an effect known as *aliasing* can occur, whereby *image* or *alias* frequencies are introduced upon reconstruction of the original waveform from the samples (see Figure 6.4).

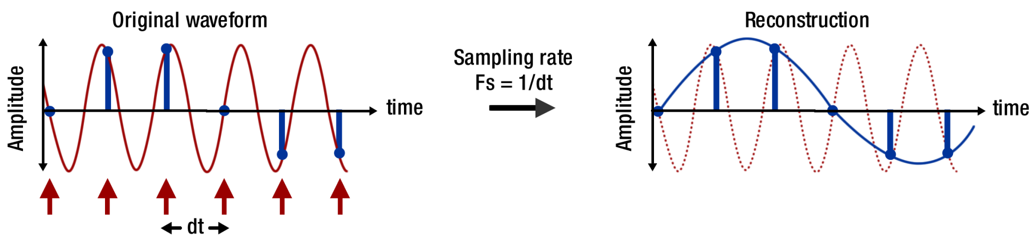


Figure 6.4: When a waveform is sampled at an insufficient rate, its reconstruction can result in *alias* signals.

In FDTD simulations, the upper frequency limit of the system is further compromised by a numerical effect known as *numerical dispersion* [389, 390]. Numerical dispersion is an undesirable artefact of the discretisation process which essentially equates to a direction-dependent variation in wave propagation speed relative to the mesh topology. The effect is readily observed in a square mesh due to the fact that waves propagating in the orthogonal directions require only a single time-step to travel the end-to-end distance d of a unit cell, while a wave propagating in the diagonal direction requires twice the length of time to cover a distance of $d\sqrt{2}$. At low frequencies, this has a negligible effect on the resultant waveform, but at higher frequencies, the velocity discrepancy causes visible distortion of the waveform. For broadband signals, numerical dispersion causes the high frequency content of the wave to lag in some directions more than others, causing spherical waveforms to become slightly cubical. This deformation of the waveform can result in cumulative phase error. In other words, a small error in the waveform propagation velocity will, over a period of time, result in an unacceptable degree of phase error manifested as a misallocation of resonances in the frequency domain [391].

The simplest way to reduce dispersion error is to increase the sampling rate of the system. However, this approach is not necessarily the most practical as it can make computation very costly in terms of both time and memory requirements, particularly in the 3-D case. According to Kowalczyk et al., a sampling rate twice that of the Nyquist frequency is needed to reduce the dispersion error in the rectilinear mesh to an acceptable level [392]. Another approach would be to use a higher order mesh topology such as a triangular or hexagonal mesh. In this context, the order of the mesh topology relates to the rotational symmetry of the geometric shape upon which it is built. While more difficult to implement, higher order topologies offer greater resistance to numerical dispersion than the traditional rectilinear mesh [366].

The essential stability of the FDTD method is also limited by a condition known as the Courant-Friedrichs-Lewy (CFL) condition [393]. The Courant number is a ratio between the maximum distance an acoustic wave can travel in one time-step to the spatial step.

$$S_c = \frac{c\Delta t}{\Delta x} \tag{6.20}$$

where S_c, Δt, Δx and c are the Courant number, time-step, length interval and speed of sound respectively. In standard FDTD schemes the Courant number should generally satisfy the following to ensure the essential stability of the simulation:

$$S_c \leq \frac{1}{\sqrt{D}} \tag{6.21}$$

where D is the number of dimensions [392]. Ignoring the effects of dispersion error, the upper frequency bound of the system is in principle half the sampling frequency, f_s, satisfying the following:

$$f_s = \frac{1}{\Delta t} < \frac{c_0}{S_c \delta} \tag{6.22}$$

The imposition of the CFL condition also means that, wherever a fine feature needs to be described in terms of either geometry or frequency, very fine discretisation is required in the problem domain, leading to even greater computational expenditure. Thus a considerable amount of effort has been made to reduce the computational costs associated with this group of methods. Some of the approaches that have been taken focus on speeding up the run-time of the FDTD program using advanced computational techniques (e.g. [394, 395]), whereas others look at adapting the FDTD scheme to make it less computationally demanding. One such example involves the implementation of a non-uniform orthogonal FDTD scheme [391]. In other words, by using a dense mesh in areas where there are fine features, and a coarse mesh in areas where fine discretisation is not necessary.

Implementing FDTD algorithms can be computationally expensive in terms of speed and memory, hence there is inevitably some trade-off between achieving the optimum physical accuracy and reducing the demands of the program. In these experiments, it has been assumed that the material properties c and ρ are constant, thus eliminating the need for additional arrays housing the spatial dependent coefficients of pressure and velocity that were proposed by Schneider et al. [386]. Furthermore, a square mesh has been chosen for its low computational demands and its ability to satisfy conditions for a single spatial differential $\delta x = \delta y = \delta z = \delta$. Though several values of δ were considered, a spatial sampling interval of $\delta = 0.005$ (5mm) was found to be optimal for achieving sufficient geometrical detail while maintaining acceptable levels of dispersion error and simulation times. In the 3-D case this equates to a sampling frequency of approximately 119 kHz - nearly 6 times the desired frequency range.

6.3.3 Boundary Modelling

One reason why the FDTD method is often favoured in virtual acoustics relates to the fact that, in the majority of cases, acoustics problems are closed domain problems - i.e. there are inherent restrictions on the size of the domain imposed by the finite dimensions of the room or other enclosure. Open domain problems on the other hand - i.e. sound propagating in a free-field - are more problematic due to the introduction of unwanted reflections at the terminating elements in the mesh. There are two general approaches one can take to mesh truncation. The first approach is to extend the mesh until it reaches sufficient proportions to eliminate boundary reflections. However, the computational cost associated with this approach renders it unfeasible for almost all realistic simulations. The second and more popular approach is to implement an Absorbing Boundary Condition (ABC). ABCs are typically subdivided into three main groups: global ABCs, local ABCs, and absorbing media [396].

Global ABCs provide a precise solution to the boundary problem through the application of Green's Theorem. They are global in the sense that the pressure field at a given boundary node is computed through integration of all points on the boundary. Whilst this approach is nominally exact, it is computationally very costly, as it relies on having a complete time-history of pressure and velocity fields at the boundary. Local ABCs differ from global ABCs in that they rely on local integral-differential operators to approximate a solution to the wave equation, meaning boundary nodes are linked only to their immediate neighbours. Although far less costly to implement, local ABCs are inherently imperfect and will always reflect some energy back into the simulation domain [397].

Absorbing media, often referred to as 'sponge layers', are effectively artificial absorbing layers surrounding the region of interest. They are broadly analogous to real-world absorbing materials, behaving in much the same way as the foam wedges lining an anechoic chamber. Like local ABCs, sponge layers are unable to eradicate unwanted reflections completely. The tolerance of the solution to contamination by low-level boundary reflections is therefore an aspect for consideration when implementing a suitable ABC. There are several factors affecting their performance, the main ones being the proximity of the source to the boundary layer, the incident angle of the outgoing wave (ABCs generally perform less well at grazing incidence), and the depth of the absorbing layer. Determining the depth of the absorbing layer is generally a trade-off between performance and increased memory cost.

Amongst the most successful and widely used of the sponge layer ABCs is an approach proposed by Berenger referred to as a Perfectly Matched Layer (PML). Although originally developed for use with Maxwell's equations in relation to electromagnetic waves, there have been several adaptations of the technique for acoustic waves (e.g. [398]). In an electromagnetic model, the magnetic field is created around the moving electric charge (denoted H) which, in Berenger's original formulation, is split into two non-physical fields, H_x and $-H_z$, in the boundary layer [399]. The electric field (E_z) depicts the force of an electrically charged particle at a given position. These can be considered approximately analogous to the pressure and velocity components in an acoustic model. Thus Berenger's formulation can be adapted for the acoustic case using the following symbol substitutions [398]:

$$\{E_z, -H_z, H_x, \epsilon, \mu, \sigma, \sigma*\} \leftrightarrow \left\{P, v_x, v_y, \kappa, \rho, \alpha, \alpha*\right\} \tag{6.23}$$

where P is the pressure field, v is the velocity vector field, ρ is the mass density of the medium, and κ is the compressibility of the medium ($c = \frac{1}{\sqrt{\kappa\rho}}$) - compressibility being the inverse of the bulk modulus, K. The attenuation coefficient, α, is the usual attenuation due to compressibility in acoustic media. There is a second attenuation coefficient associated with density. This is generally assumed to be zero in acoustic media, although it may be used to approximate mass-proportional damping when modelling sound waves in an elastic solid. It is used here since a PML relies on the gradual introduction of non-physical density attenuation in the absorbing layers. This coefficient increases from 0 to its maximum value, α_{max}, from the innermost to the outermost layer of the PML respectively. For example, in a PML m samples thick, the attenuation coefficient is given by:

$$\alpha_i = \alpha_{max}\frac{i}{m}^2 \tag{6.24}$$

for $i = 1 \rightarrow m$ where i is the index of the current layer in the PML.

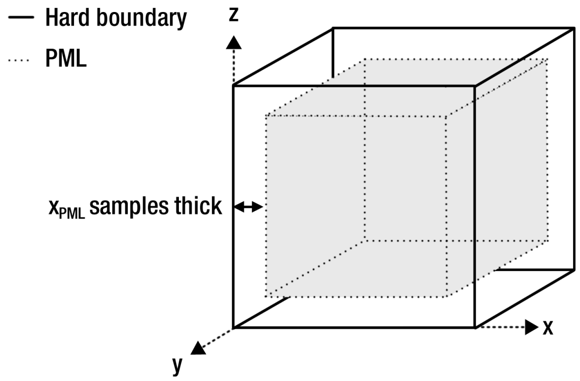


Figure 6.5: Geometry of the 3-D FDTD array showing the PML boundary regions. Adapted from [398]

Computer Implementation of PML

In a 1-D FDTD acoustic model, the update equations in the PML region are:

$$p^{q+1}(n) = \exp(-\alpha_i K\delta t)p^q - \left(\frac{1-\exp-\alpha_i K\delta t}{\alpha_i K}\right)K\frac{1}{\delta x}\left[v_x^{q-1/2}(n) - v_x^{q-1/2}(n-1)\right] \quad (6.25)$$

$$v_x^{q+1/2}(n) = \exp(-\alpha_i K\delta t)p^{q-1/2} - \left(\frac{1-\exp-\alpha_i K\delta t}{\alpha_i K}\right)\frac{1}{\rho\delta x}\left[p^q(n+1) - p^q(n)\right] \quad (6.26)$$

where α_i is the attenuation coefficient associated with the corresponding layer of the PML. The value of α_i is calculated at each update according to (6.27) in order to achieve a gradual introduction of the absorption across the boundary layer.

$$\alpha_i = \alpha_0 \frac{x_i}{x_{PML}} \quad (6.27)$$

In the above, x_i is the depth moved into the PML region, x_{PML} is the depth of the PML, and α_0 denotes a maximum attenuation coefficient which is a user-defined value between 0 and 1 [400]. Assuming the depth of the PML region is 10 nodes deep, the maximum amplitude reduction that will be achieved over a propagation distance of dx is 1/10. The maximum attenuation will therefore be

$$\alpha_0 = \frac{1}{K\delta t}\ln(10) \quad (6.28)$$

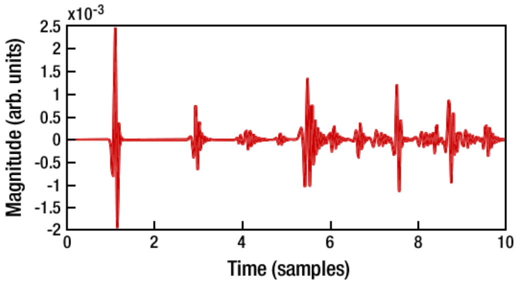
The update equations (6.25) and (6.26) can of course be readily extended into 2 and 3 dimensions.

Performance

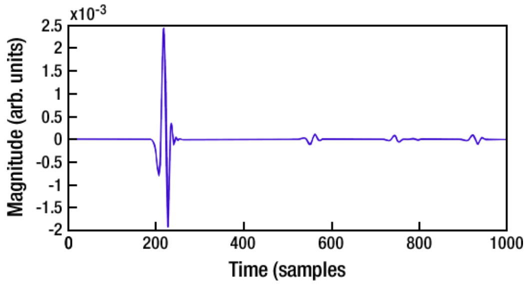
The performance of the PML is assessed by comparing the energy in the first wave front to that of the reflections, before and after its implementation. In acoustics this quantity is referred to as the direct-to-reverberant ratio (DRR) and it can be calculated in a number of different ways. A simple technique presented by Larsen et al. [401] calculates DRR directly from the impulse response.

$$DRR = 10log\frac{\int_0^{T_d} h^2(t)dt}{\int_{T_d}^{\infty} h^2(t)dt}\, dB \quad (6.29)$$

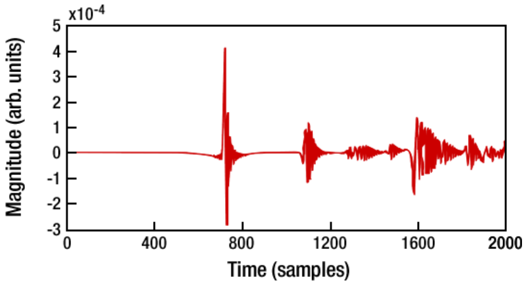
where T_d is the duration of the direct sound and h is the impulse response. Since the distances between source and receiver, spatial sampling interval and signal length are all known quantities, it is possible to determine the precise value of T_d. In a 300 x 450 x 300 rectilinear mesh, the values of DRR before and after implementation of a 10 element deep PML were 1.3 dB and 22.7 dB respectively, indicating a reduction of 21.4 dB due to the PML.



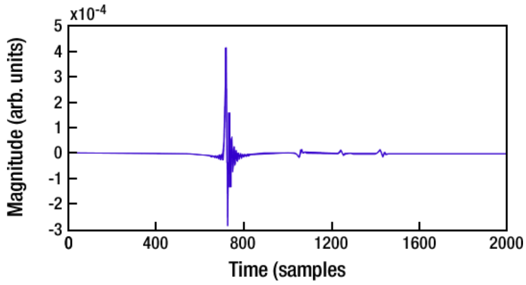
(a) Output before implementation of PML



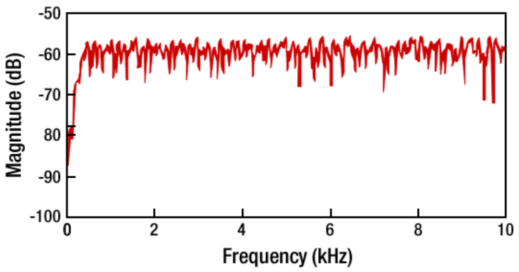
(b) Output after implementation of PML



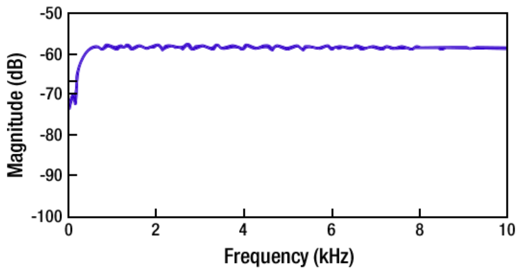
(c) Impulse response before PML



(d) Impulse response after PML



(e) Frequency spectrum of impulse response before PML



(f) Frequency spectrum of impulse response after PML

Figure 6.6: Time and frequency representations of Ricker impulses recorded before (red) and after (blue) implementation of the PML in 3-D FDTD simulations. Size of mesh is 300-by-450-by-300 elements, PML depth is 10 elements, and the spatial sampling interval in each axial dimension is 5mm ($\delta = 0.005$). Source and receiver locations are given in Figure 6.7.

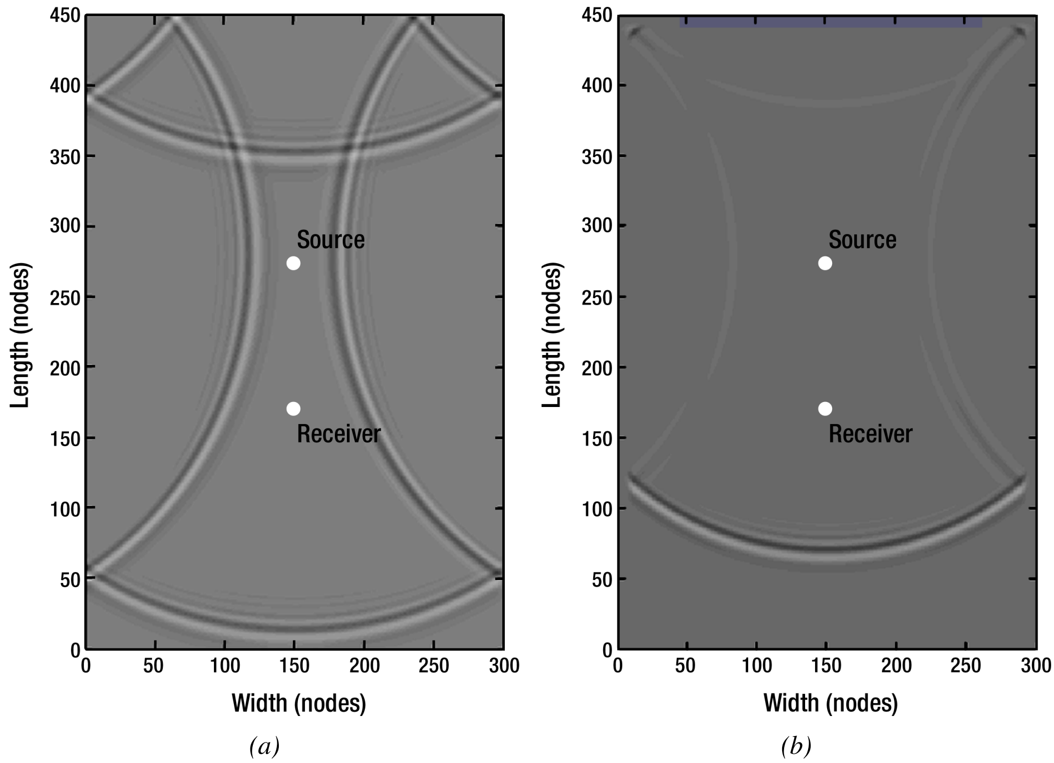


Figure 6.7: Horizontal cross-section through the simulation domain at the 400^{th} time step before *(a)* and after *(b)* implementation of the PML.

6.3.4 Discontinuities

As well as the external mesh boundaries, the surfaces of the inclusions also represent discontinuities in the material properties of the medium. The difference in propagation speed between two materials of very different densities presents a significant problem in the FDTD simulations if the Courant condition is still to be satisfied. This is due to the fact that the temporal resolution of the mesh is forced to become extremely fine in order to maintain stability at the interface between the media. In terms of memory requirements and computation time, this is out of reach of a single processor. The solution here has been to assume a perfectly reflective boundary at the boundaries of the inclusions, which can be achieved computationally by forcing the values in the pressure array to zero at the inclusion nodes. There are thus three regions of the simulation domain: the host medium, in which the ordinary inverse square law applies; the boundary layer, in which there is direction dependent absorption; and the inclusions, which are assumed to be perfectly reflecting. Within each of these separate regions, a different set of governing update equations apply. This can be achieved very simply by incorporating a switch statement into the computer code which serves a different update equation depending on the node type.

6.3.5 Source Characterisation

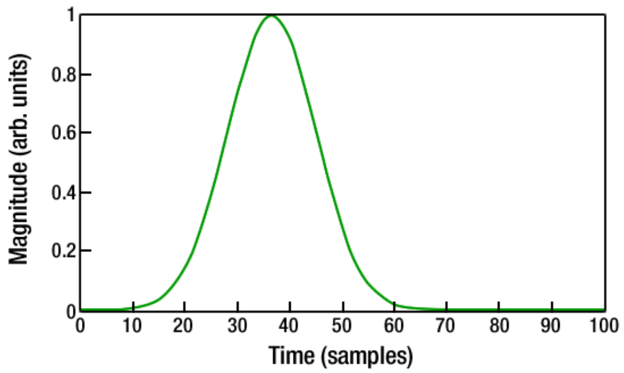
There are various types of input signals that are appropriate for acoustic profiling, each with its own advantages and disadvantages. The most traditional of these is the Dirac delta function or unit impulse. In the ideal case, this signal has infinite amplitude over an infinitely short space of time, although, in practice, the signal bandwidth is limited by the sampling interval. Whereas in a real-world situation this signal is impossible to implement, in digital simulations it is the most straightforward. Furthermore, since the Fourier transform of an ideal unit impulse is theoretically flat across all frequencies, the signal requires no deconvolution. However, as is the case when performing real-world impulse response measurements, results are compromised by the presence of noise - i.e. undesirable signal components that limit the dynamic range and cause spurious artefacts in the frequency domain. In acoustic simulations, noise can accumulate during run-time due to a build up numerical error in the solutions to the equations, making it much more difficult to remove. To limit the effects of noise there should be significantly more power in the measured signal than that of the noise - i.e. to maximise the signal-to-noise ratio (SNR). Since the dimension of power is energy divided by time, the unit impulse typically achieves a poor SNR in FDTD simulations, particularly in a rectilinear mesh which is prone to numerical dispersion error. However, since the system is linear, it is possible for the frequency content of an impulse to be spread out over time using a longer input signal, thus allowing more energy to be put into the system and a higher SNR to be achieved [402]. Several alternatives to the unit impulse were thus explored in simulations performed in an empty 3-D mesh. The first of these to be considered was the Gaussian pulse function.

Gaussian Pulse

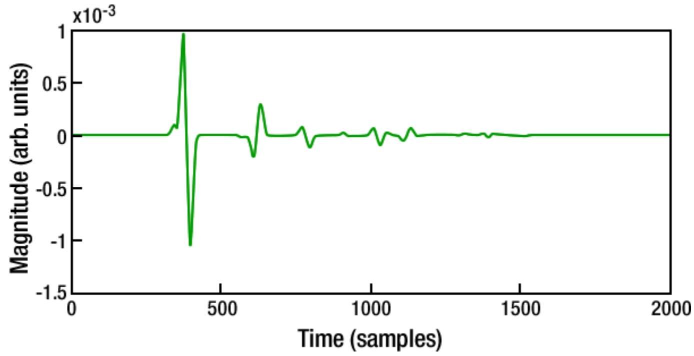
The 1-D Gaussian pulse may be defined by

$$g(t) = -\exp\frac{(t - t_0)^2}{2\sigma^2} \tag{6.30}$$

where the pulse width, σ, determines the frequency range of the filter [400]. A Gaussian pulse has the unusual property in that the shape of its response curve is the same in both the time and frequency domains. However, a disadvantage of using the function as an input signal in FDTD simulations is that it contains a DC component which can have the undesirable effect of introducing artefacts into the simulation [386].



(a)



(b)

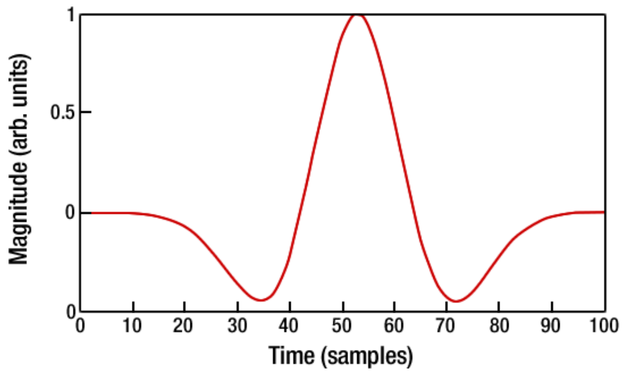
Figure 6.8: Time domain representations of the input *(a)* and output at the receiver node *(b)* using a Gaussian source function with a 10 kHz cut-off frequency.

Ricker Wavelet

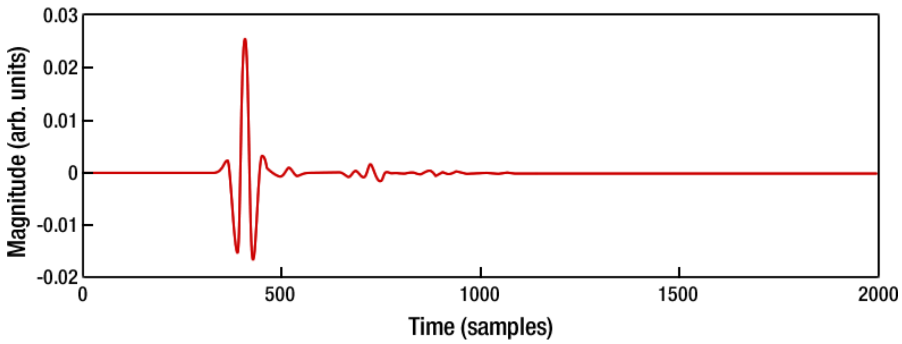
Another possibility would be to use the 2^{nd} derivative of a Gaussian pulse, also known as a Ricker wavelet function. The Ricker wavelet is defined by

$$r(n) = \left(1 - 2\left\{\pi f_p\left[t - d_r\right]\right\}^2\right)\exp\left(-\left\{\pi f_p\left[t - d_r\right]\right\}^2\right) \tag{6.31}$$

where f_p is the peak frequency and d_r is a temporal delay consisting of some multiple of $\frac{1}{f_p}$ [386]. Unlike the Gaussian impulse, this type of input does not contain a DC component. Moreover, comparison of their outputs in the simulations suggest it is also more tolerant to noise.



(a)



(b)

Figure 6.9: Time domain representations of the input *(a)* and output at the receiver node *(b)* using a Ricker wavelet source function with a centre frequency of 2.5kHz and a roll-off at 10 kHz.

Both the Gaussian and Ricker wavelet source functions require deconvolution in order to obtain a useful representation of the system impulse response. This is accomplished most simply and efficiently in the frequency domain when the Fourier transform of the recorded signal is divided by the Fourier transform of its inverse filter. However, the process of deconvolution can lead to a significant degradation of the signal-to-noise ratio (SNR) due to the introduction of parasitic high frequency components [403]. A third source signal was therefore attempted which, while unproven in FDTD simulations, is highly regarded in empirical studies on account of its excellent tolerance to noise. The signal referred to is the exponential sine sweep signal (see Chapter 5).

An exponential sine sweep is a sine wave in which the instantaneous frequency increases exponentially from ω_1 to ω_2 over a period of time T[404]. It is defined in the time domain as

$$s(t) = \sin\left[\theta(t)\right] = \sin\left[k \cdot (\exp^{-t/L} - 1)\right] \tag{6.32}$$

where

$$k = \frac{\omega_1 T}{\log\left(\frac{\omega_1}{\omega_2}\right)} \tag{6.33}$$

and

$$L = \frac{T}{\log\left(\frac{\omega_1}{\omega_2}\right)} \tag{6.34}$$

The main advantage of this type of test signal in real-world measurements is that it introduces more energy into the system than a standard Dirac impulse in a relatively short space of time and therefore achieves a much higher SNR across a continuous spectrum of frequencies. An important consideration when using this technique is that of the relationship between energy and the rate of change ($\theta(t)$), associated with a given instantaneous frequency ($\omega(t)$). Since the signal energy at a specific frequency is dependent on the length of time the signal oscillates at that frequency, it therefore follows that more energy will be concentrated in the lower frequencies due to the fact that the instantaneous frequency changes slowly at low frequencies and much faster at higher frequencies. The resulting spectrum is therefore one which decreases by 3 dB per octave, a factor that can be compensated for when generating the inverse filter by applying amplitude modulation to the reversed sweep signal. This step must be taken if one is to obtain an impulse response in which energy is distributed evenly over the full range of frequencies.

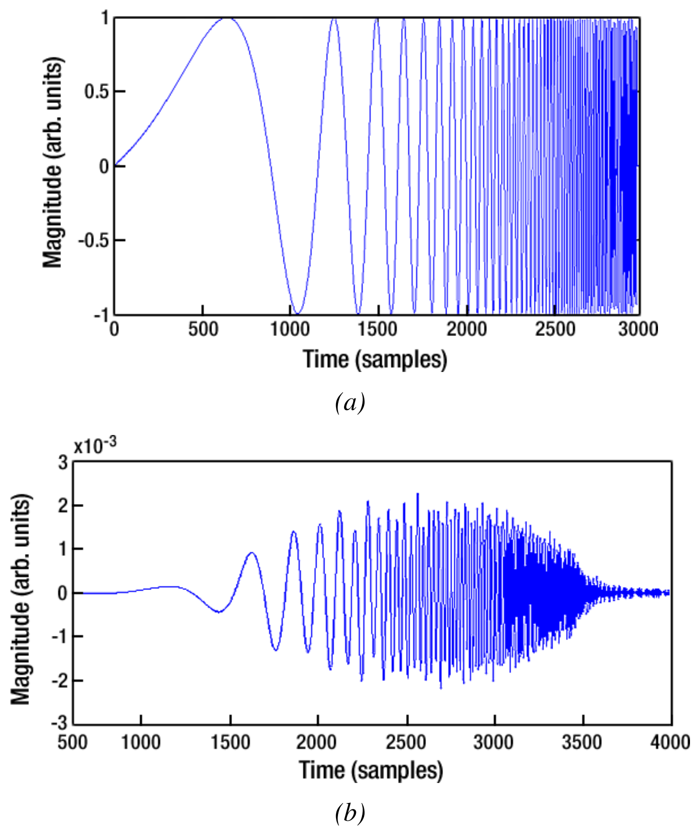


Figure 6.10: Time domain representations of the input *(a)* and output at the receiver node *(b)* recorded in simulations using a swept sine source signal.

While the sine sweep is a mainstay in room acoustics, its application in acoustic FDTD simulations is uncommon. It was attempted here to identify whether or not it is a feasible candidate for a source signal in FDTD based acoustic profiling.

Source Transparency

In FDTD simulations the source is embedded into the mesh by forcing the value of the source node at a given time step - denoted by the superscript n - to that of the source function, f. The value recorded in the pressure array is thus,

$$p^n(r_{src}) = f^n \tag{6.35}$$

where p is a pressure field of arbitrary dimensions and r_{src} is an index corresponding to the location of the source node in a scheme of arbitrary dimensions. This means the update equation is not performed at the node where the injection of energy occurs, which effectively makes the node a scatterer. This type of source is referred to as a 'hard' source, and it is not necessarily a problem unless boundaries and discontinuities are introduced such that reflections to and from the source node are able to reach the receiver within the time it takes for the simulation to run.

A solution to this problem - albeit at a higher computational cost - is to implement a transparent source. A transparent source radiates the same field as a hard source, yet, once executed, is invisible to the returning wave front. In other words, sound that is reflected back to the source by the array or mesh boundaries will pass through it unperturbed. The simplest way to implement such a source is using a technique presented by Schneider et al. [405], whereby a preliminary simulation is performed to capture the impulse response of the empty mesh. This is achieved by performing the update equation at the source node and recording the returned value as part of the impulse response. Since the value of the source node that is stored in the pressure array remains fixed to the source function, those stored in the impulse response array, h, are ignored by all neighbouring nodes. Once the impulse response has been captured, assuming the dimensions and other properties of the mesh remain unchanged, source transparency is achieved by subtracting the convolution of the impulse response and the source function from the source - a process which can be performed at run-time in subsequent simulations. Furthermore, the impulse response need only be captured once for a given mesh configuration and is then valid for all subsequent simulations [405].

The impulse response can be found using the standard update equation at the source node. In a 1-D scheme this is given as,

$$h^{n+1}(i_{src}) = p^n(i_{src}) - \delta t\left[v_x^{n-1/2}(i_{src}) - v_x^{n-1/2}(i_{src}-1)\right] \tag{6.36}$$

where h is the impulse response function, and i_{src} is the spatial index associated with the source node. Note the first value of the impulse response is zero.

During subsequent simulations, rather than being fixed according to (6.35), the update equation is performed at the source node and the source function is applied by addition. Finally, to remove the field that will be reflected back to the source node and ensure the radiated field is the same as that of the hard source, the convolution of the impulse response and the source function are subtracted. Specifically, a transparent field source for an arbitrary Courant number, S_c, is obtained using Schneider's formula [405],

$$p^{n+1}(i_{src}) = p^n - \rho c S_c \left[v_x^{n+1/2}(i_{src}) - v_x^{n+1/2}(i_{src} - 1) \right] + f^{n+1} - \sum_{m=0}^{n} h^{n-m+1} f^m \tag{6.37}$$

A more general form of (6.37) that is applicable to an arbitrary number of dimensions, D, is given by,

$$p^{n+1}(r_{src}) = [N - D\; update\; equation] + f^{n+1} - \sum_{m=0}^{n} h_N^{n-m+1} f^m \tag{6.38}$$

In the following computer code, `H` is the impulse response (IR) recorded at the source node in a previous simulation, and `src_corr` is a correction value that is added to the source node with each iteration of the update equation. Note the '+1' from (6.38) is removed from the time index as the future time step is referenced implicitly.

```
for (int t = 0; t <= T; t++) // Main update loop spanning T iterations
{
  if (t < sizeof(H)) // Checks time step is less than length of compensating IR
  {
    for (int k = 0; k <= t; k++)
    {
      src_corr += H[t-k]*excitation[k]; // Compensation value, summed over k
    }
  }
  else
  {
    src_corr = 0.0;
  }
  ... update pressure and velocity arrays
}
```

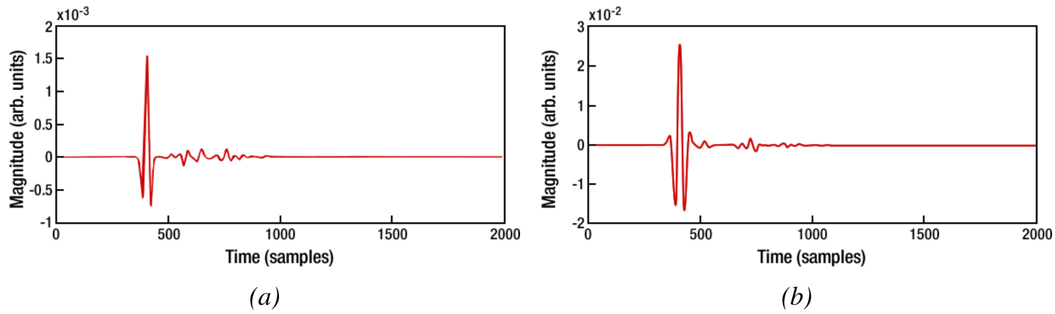


Figure 6.11: The output signal recorded with a Ricker wavelet source function in a 314-by-314-by-300 element mesh, before *(a)* and after *(b)* implementation of the transparent source.

Deconvolution

To derive the impulse responses from the signals recorded at the receiver locations, each signal must be convolved with the inverse filter of the relevant source function. Deconvolution, like all signal processing techniques utilised in this thesis, is implemented in Matlab version r2010a [406]. The inverse filter associated with the Ricker wavelet function is calculated in the frequency domain using a script by Matthes [407] (see Figure 6.12).

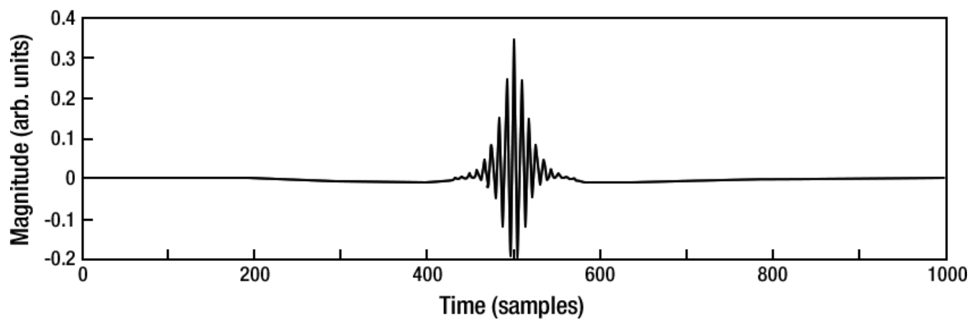


Figure 6.12: The inverse Ricker wavelet filter

The inverse filter associated with the sine sweep is a time-reversed version of the original logarithmic sweep to which an amplitude envelope is applied. The amplitude envelope attenuates by 6 dB per octave to compensate for the logarithmic power-frequency distribution in the original sweep. Both the sweep and inverse sweep were generated in Matlab using a script by Wells [408]. The only slight modification is the removal of the 100 ms ramp in and out that are needed in real-world measurements for the prevention of undesirable ‘start and stop’ transients produced by the loudspeakers. Due to the high sampling rate of the simulation and the implication on the run time of the program, it would not be feasible for the sweep to have lasted the full 10 seconds recommended for real measurements. However, since one does not have to consider loudspeaker capabilities, it is possible to perform the measurement with a much shorter sweep, albeit with some loss of smoothness in the frequency response curve. Furthermore, as the sweep duration approaches 1/20 of a second, the low frequency response is severely compromised.

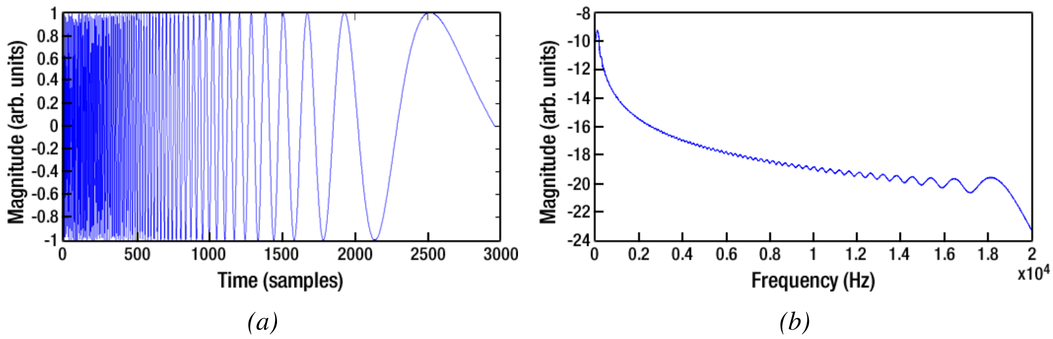


Figure 6.13: Time *(a)* and frequency *(b)* domain representations of the inverse sweep function.

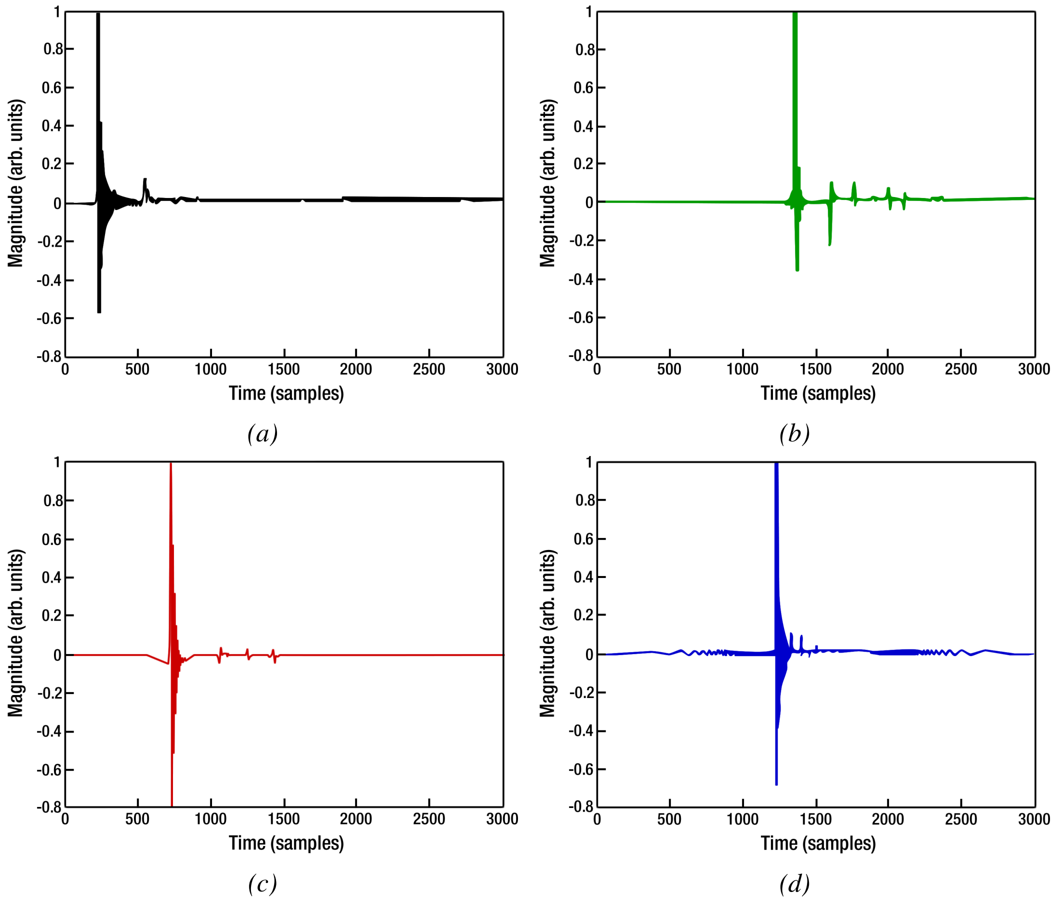


Figure 6.14: Four impulse responses obtained from 3-D FDTD simulations using the Dirac delta *(a)*, Gaussian pulse *(b)*, Ricker wavelet *(c)* and sine sweep *(d)* source functions.

Figure 6.14 shows four impulse responses recorded in an empty 3-D rectilinear mesh using four different source functions, all with a maximum amplitude of 1. All the impulse responses have been normalised here to enable a visual comparison. The impulse response recorded with the Dirac delta source function required no deconvolution, although it does require low-pass filtering to remove high frequency artefacts to which it is prone. The low-pass filter was a Butterworth filter with a cut-off frequency of 10 kHz. The audio files are available on the DVD accompanying this thesis (Directory 9.7.3.2).

Performance Comparison

The performances of the three different sources are assessed on the basis of their signal-to-noise (SNR) ratio, which can be considered a measure of how much corruption there is in the signal relative to the meaningful part. How one quantifies the SNR in practice is somewhat dependent on the nature of the problem. With respect to the synthetic impulse response, the SNR shall be defined as the ratio expressed in dB between the average power of the signal recorded by the receiver and the average power of the noise and distortions present in the tail of the deconvolved impulse response [326]. By this definition the SNR takes into account all sources of noise, including that which is introduced by the PML, dispersion error and other sources of numerical error inherent to the model. The basic formula for calculating the SNR is thus:

$$SNR = 10 \log \left(\frac{\left\langle y^2(t) \right\rangle}{\left\langle h_{tail}^2(t) \right\rangle} \right) dB \tag{6.39}$$

where $\left\langle y^2(t) \right\rangle$ is the average power of the output signal, $y(t)$ and $\left\langle h_{tail}^2(t) \right\rangle$ is the average power in the tail of the deconvolved signal. Both are defined as functions of time according to the following general equation:

$$\left\langle f^2(t) \right\rangle = \lim_{x \to \infty} \frac{1}{2T} \int_{-T}^{T} |f(t)|^2 dt \tag{6.40}$$

For a signal in discrete form, power can be calculated from the following:

$$P_w = \frac{1}{N} \sum_{n-0}^{N-1} f^2[n] \tag{6.41}$$

where N is the number of samples in the signal from $n = 0 \rightarrow N - 1$. It was found that of the four signals tested, the Ricker wavelet yielded the best SNR at 38.5 dB, followed closely by the Gaussian impulse at 36.6 dB. After low pass filtering the Dirac delta function yielded a SNR of just 8.5 dB, while the sine sweep achieved only an 6.8 dB SNR. The frequency response curves for each of the deconvolved impulse responses are presented in Figure 6.15, where it is noticeable that the signals which yield the better signal-to-noise ratios do not necessarily yield the flattest response curves. For example, on the basis of the frequency response curves, it appears the Dirac delta function may be the most satisfactory choice of input, yet its SNR suggests otherwise - hence the choice of input signal should not be based on the frequency response, but on the SNR. Most of the variation in these results can be attributed to the frequency content of the input signals, and the imperfection of the inverse filtering process (in the case of the Ricker, Gaussian and sine sweep signals).

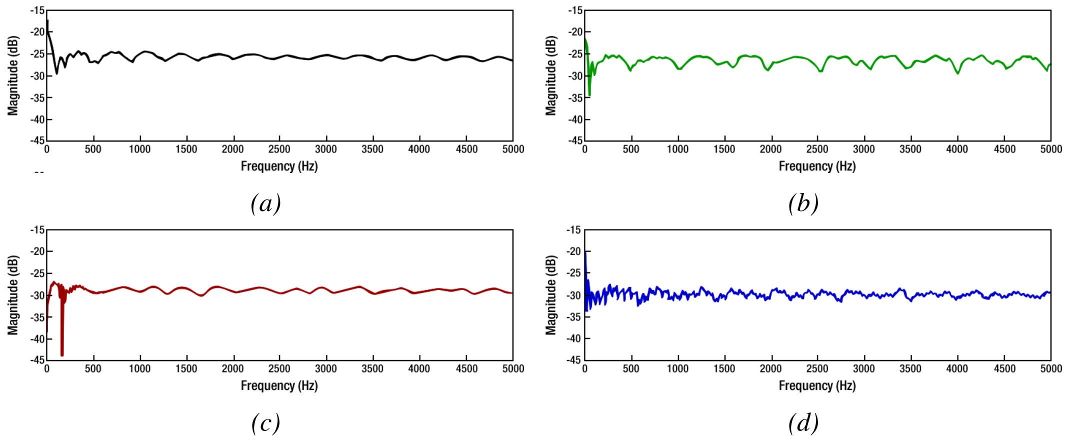


Figure 6.15: Frequency response curves of the deconvolved Dirac delta *(a)*, Gaussian *(b)*, Ricker wavelet *(c)* and swept sine *(d)* impulse responses.

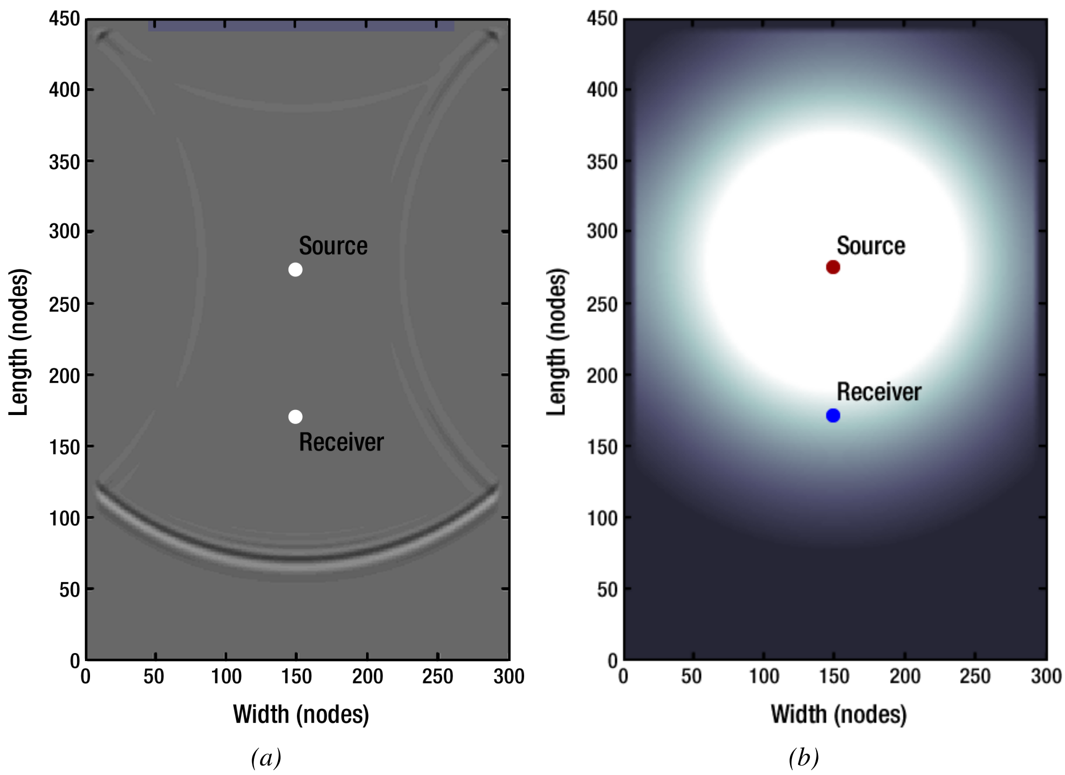
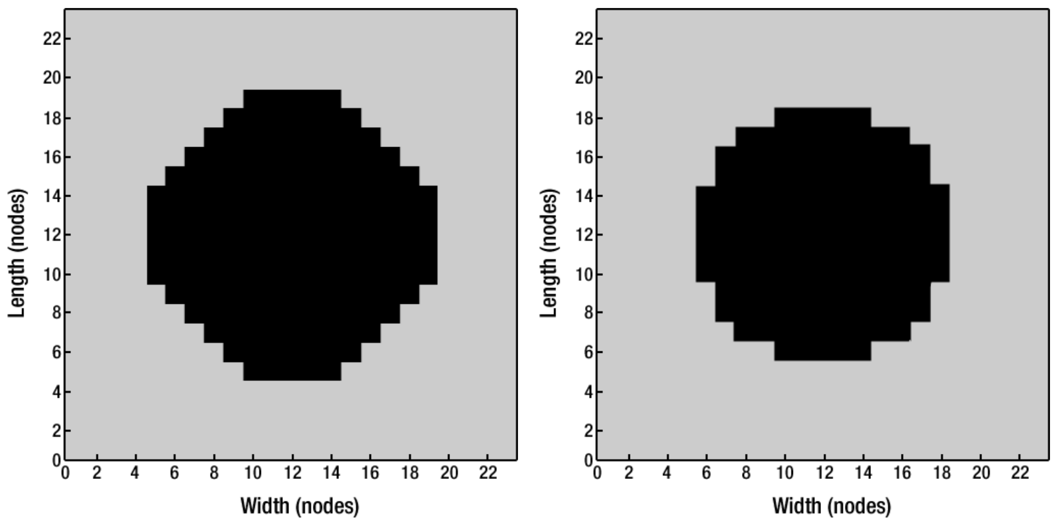


Figure 6.16: Horizontal slices through the pressure field in simulations of an empty mesh using Ricker wavelet *(a)* and sine sweep *(b)* sources. The time step, T, in each of the above is 400.

From the results of the simulations performed on an empty mesh, it is concluded that the Ricker wavelet is the most reliable input function of those tested and is the one that shall be used in the following simulations with the periodic array.

6.3.6 Results

The following impulse measurements have been obtained from 3-D FDTD simulations of 2-D periodic scattering arrays in a rectilinear mesh with a PML absorbing boundary condition applied. The results from three different *m*-by-*n* array configurations are presented, where *m* is the number of rows and *n* is the number of columns. All are arranged in the square lattice configuration. Results are also given for for two different cylinder radii: 35mm and 30mm. From (6.2) the radii correspond to filling fractions of 0.32 and 0.23 respectively (see Figure 6.17).



(a) *(b)*

Figure 6.17: Cross section through unit cells of the mesh for two different radii, *r*, of the scattering elements. In *(a)* $r = 35mm$, and in *(b)* $r = 30mm$.

The arrays were positioned in the centre of the simulation domain and extended the entire height of the mesh (excluding the PML). The space around the arrays in the front-back and left-right directions was consistently 0.5m. This meant that the dimensions of the mesh and the computation times varied depending on the size of the array being simulated. All simulations were implemented in C++ in Visual Studio Express 2010 [409] and executed on a single desktop computer. The hardware specification is given in Table 6.1, and a complete list of simulation parameters is given in Table 6.2. The simulation times are approximate as the actual run-time depended on what other processes were running at the time.

Table 6.1: Hardware specifications for 3-D FDTD simulations

Processor	Intel Core 2 Duo E8500 3.16 GHz
RAM	4GB
System type	Windows 7 64-Bit Operating System

Table 6.2: Simulation parameters. δx is the spatial sampling interval, F_s is the equivalent sampling rate, N is the number of iterations, and L is the length of the output signal.

Array dimensions	Mesh dimensions [L; W; H]	Mesh size (nodes)	δx (m)	F_s (Hz)	N (samples)	L (seconds)	Run-time (hours)
2-by-4	[266; 310; 300]	24,738,000	0.005	118,670	4000	0.034	3-4
4-by-4	[310; 310; 300]	28,830,000	0.005	118,670	4000	0.034	4-5
6-by-4	[354; 310; 300]	32,922,000	0.005	118,670	4000	0.034	5-6

In all of the the simulations, a transparent Ricker wavelet input function with a peak amplitude of 2.5 kHz and a cut-off frequency of 10 kHz was radiated from a point source at a distance of approximately 30cm from the array. There are several reasons why limiting the source bandwidth was considered to be justified. Firstly, high frequencies are attenuated more rapidly in air than lower frequencies, hence the overall contribution of high frequency sound arriving at the receiver from sources which are obscured by the barrier is unlikely to be perceptually significant, particularly given the decreasing sensitivity of the human ear from 5kHz. This loss of high frequency sound is reinforced further by the likelihood of auditory masking (see Section 2.3.2) due to low frequency noise such as wind and traffic. There is also the consideration that the Bragg reflection events are prohibited at wavelengths smaller than the gaps between the inclusions (8.5 kHz). It therefore makes sense to aim for the low-mid range frequencies at the expense of the high frequencies.

Three receiver array positions were used to capture impulse responses on and off axis with the centre of the array (see Figures 6.18 and 6.19). The frequency spectra of the deconvolved impulse responses (see Figure 6.20) are depicted up to maximum frequency of 5 kHz as the majority of the reflection events were found to be within the 0.5-5 kHz region. Furthermore, above 8.5 kHz the space between inclusions is of the scale of the wavelength, hence no significant Bragg reflection events are expected above this frequency.

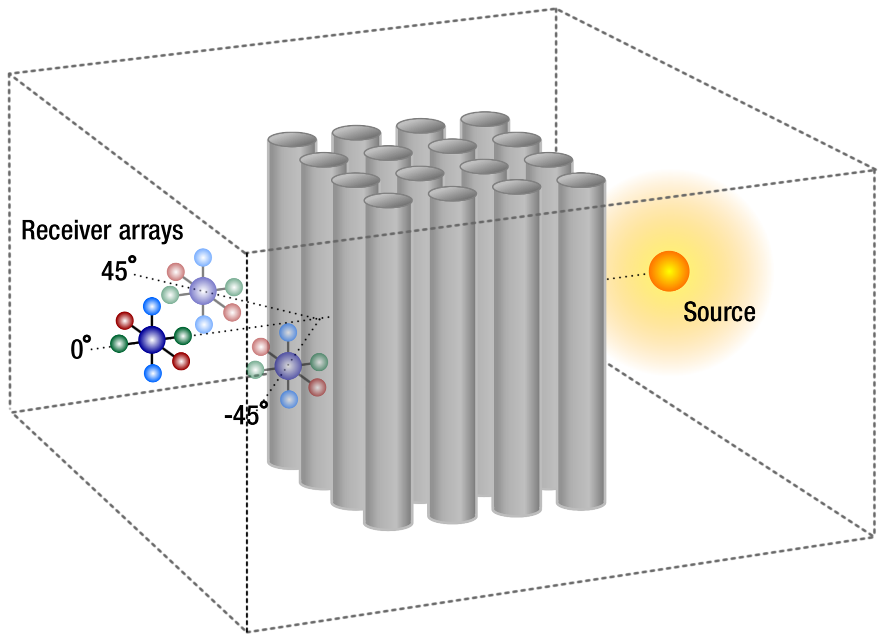


Figure 6.18: The simulation set-up for a 4-by-4 array. Each receiver array consists of 7 adjacent nodes and are positioned so as to capture impulse responses on and off -axis. An aerial projection is given in Figure 6.19.

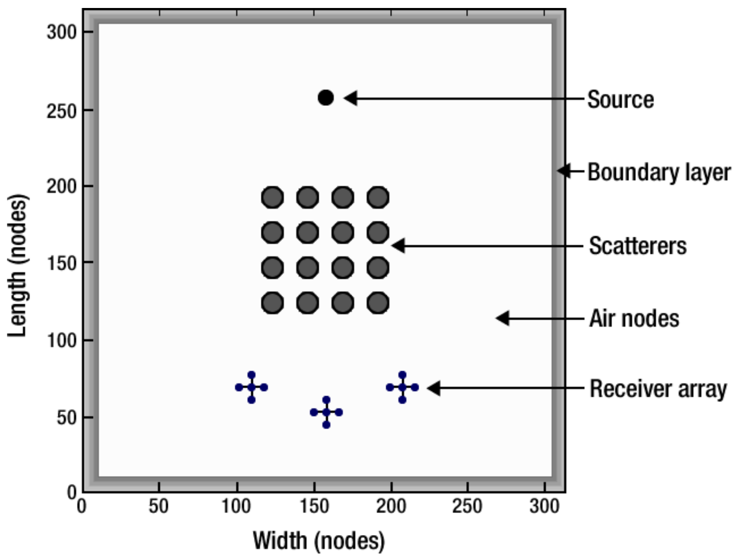
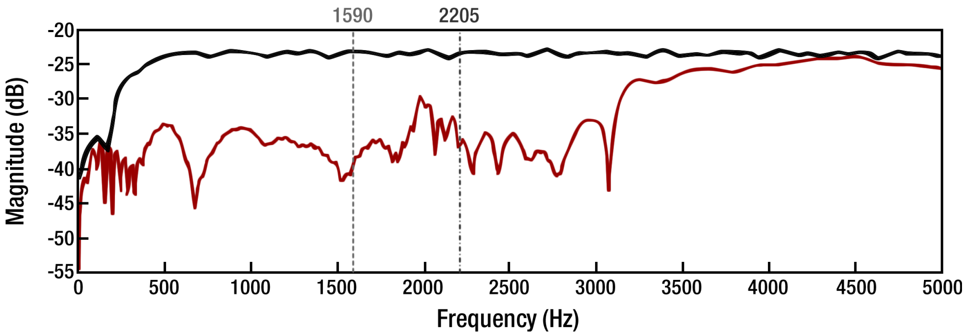
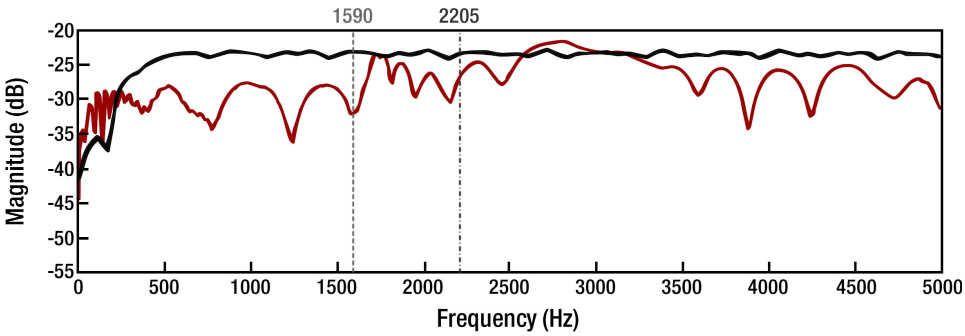


Figure 6.19: Horizontal cross-section through the simulation domain with a 4-by-4 array included.

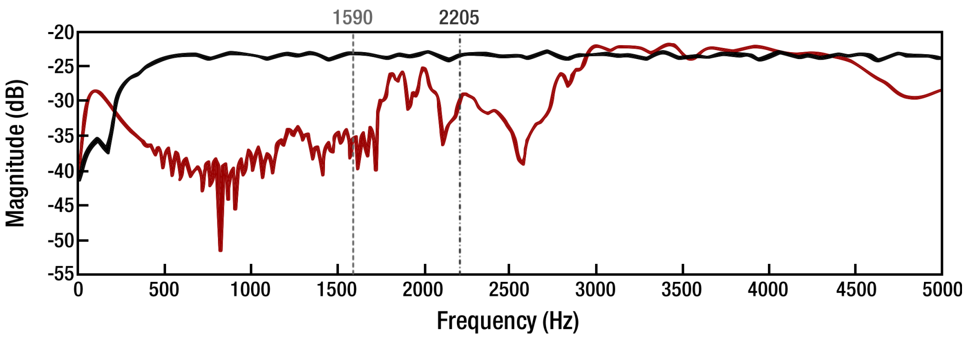
Figure 6.20: Frequency spectra of impulse responses recorded in different arrays. The vertical lines indicate the centre frequencies of the first theoretical band gaps associated with the first and second set of Bragg planes. For the 4-by-6 arrays shown in *(e)* and *(f)* the first theoretical harmonic is also depicted. While the theoretical band gap is quite pronounced in *(e)*, there is a substantial amount of variation between the spectra whereas one would expect a more consistent arrangement of peaks and gaps.



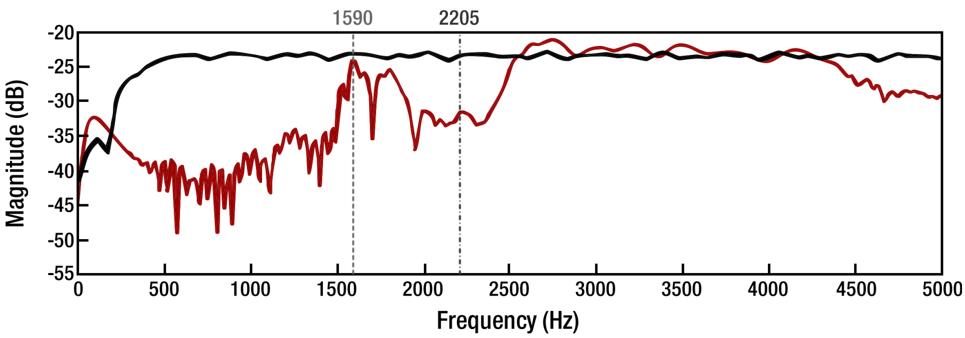
(a) 4-by-2 array, r = 35mm



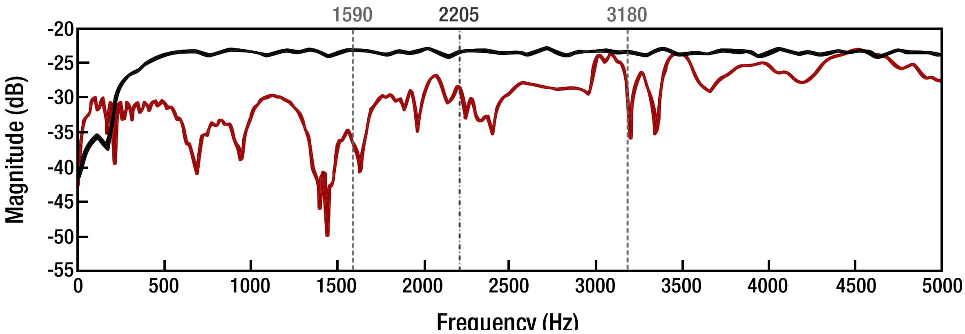
(b) 4-by-2 array, r = 30mm



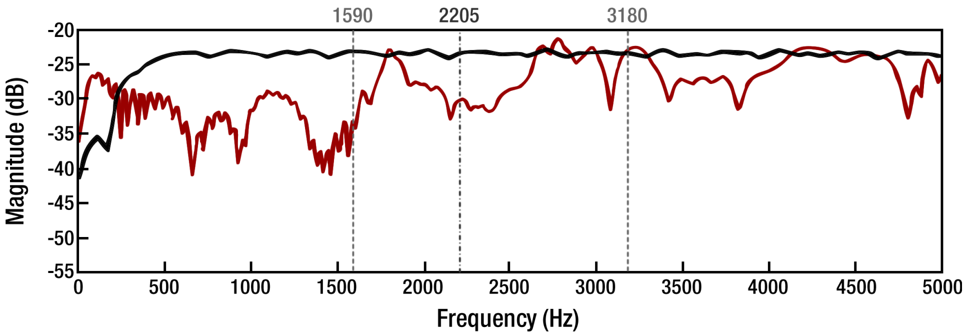
(c) 4-by-4 array, r = 35mm



(d) 4-by-4 array, r = 30mm

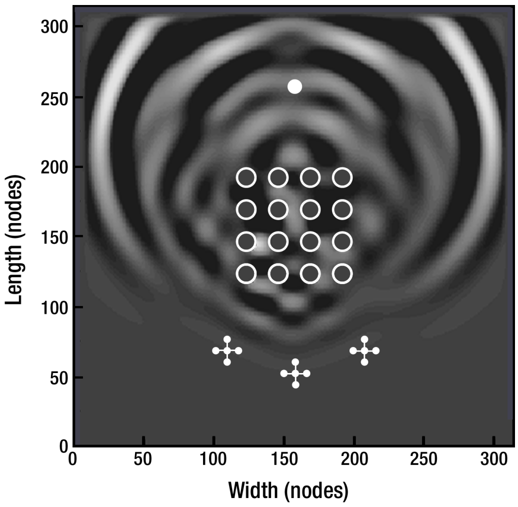


(e) 4-by-6 array, r = 35mm

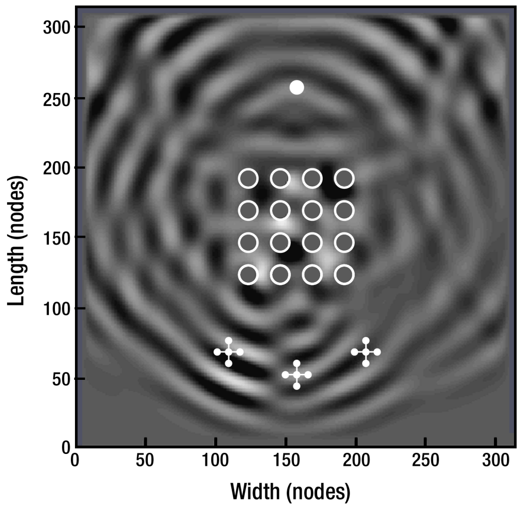


(f) 4-by-6 array, r = 30mm

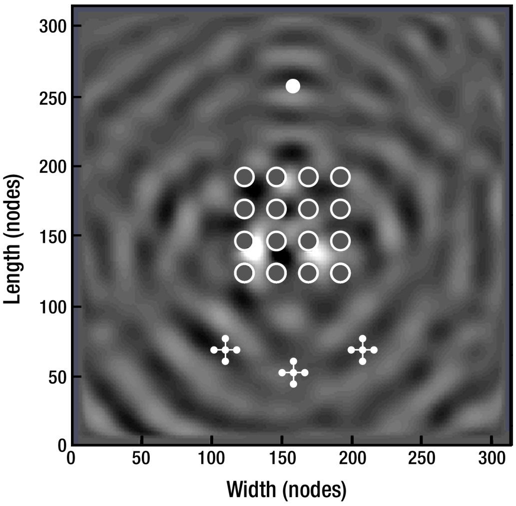
Figure 6.21: Horizontal slice through the pressure field with a 4-by-4 array. T denotes the current time step.



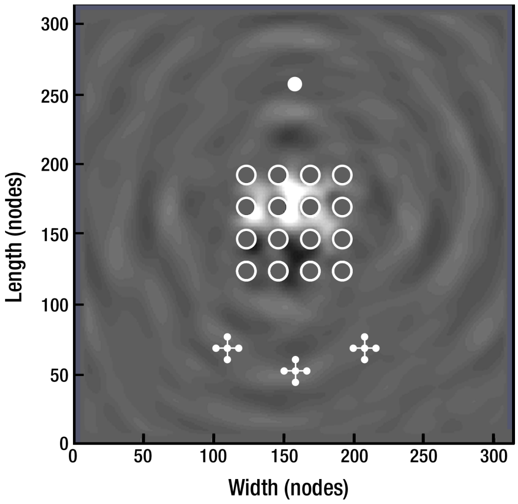
(a) T = 400



(b) T = 500



(c) $T = 1000$



(d) $T = 2000$

6.3.7 Comparison with Real World Measurements

To gain a deeper insight into the filtering properties of the structure, real world measurements are performed on a 6-by-10 element sonic crystal composed of wooden rods. For a direct comparison with the simulation results, the radius of the real rods were also 30 mm and the lattice constant 11 cm. The intention behind the measurements was not only to enable the comparison of the simulation results with real world measurements, but also to observe how closely a real structure corresponds with the theoretical predictions. Should there be evidence of inconsistencies or vulnerabilities under close to ideal theoretical conditions, it would suggest that such a structure in real world environment might also exhibit erratic behaviour, as conditions in the real world are generally not ideal. For example, the presence of wind and temperature gradients, atmospheric turbulence, variable source characteristics and various other factors which would tend to compromise the reliability of theoretical predictions.

Method

The measurements are made using the swept sine impulse response technique presented by Farina [333], albeit without synchronous averaging at the deconvolution stage. The length of the sweeps was 10 seconds and covered a frequency range of 20 - 20,000 Hz. The procedure was performed in a 5.5 m cube anechoic chamber where 5.5 m is the wall-to-wall distance and 4 m^3 is the usable space taking into account the 75 cm foam wedges. The relatively small dimensions of the chamber make it approximately analogous to the simulation domain surrounded by the artificial absorbing boundary layer. The experimental set up is pictured in Figures 6.23 and 6.24. The sine sweeps were recorded using an Haun MBC550 measurement microphone at a distance of 30 cm from the sonic crystal - as per the simulation. The Haun MBC550 is a small-diaphragm condenser with an omnidirectional pick-up pattern and a linear response between 20 Hz and 20 kHz. A Genelec

8040A loudspeaker was used to approximate a point source. This speaker exhibits a near flat frequency response between 48 Hz and 21 kHz on axis (see Figure 6.22).

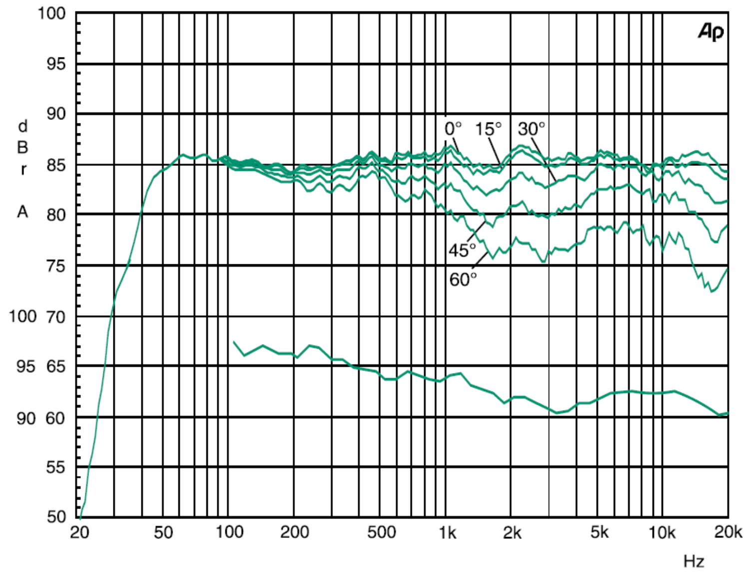


Figure 6.22: Genelec OY 8040A horizontal off-axis response level (dBr) vs frequency (Hz) from [410]. The upper curves show the horizontal directivity characteristics of the 8040A measured at 1 m. The lower curve shows the system's power response.

The sine sweeps were both triggered and recorded within the Reaper Digital Audio Workstation (DAW) running on a laptop computer. The laptop was connected to an external sound card and audio interface, the RME Fireface 800, in and out of which the microphone and loudspeaker signals were fed. The project sampling rate selected in the DAW was 96 kHz and the bit-depth was 32. Three sets of measurements were made with the sonic crystal present for three source distances of 0.3 m, 0.5 m and 1 m. Measurements were also made without the structure present.

The array of rods was held in place by a supporting top and base made of 2.5 cm thick MDF into which the rods were mounted. The vertical distance between them was approximately 88 cm - the length of the rods minus 1 cm each end where they had been embedded. These parallel surfaces could be a confounding factor in the results, hence efforts were made to reduce their effect by angling a pair of foam wedges to try to absorb sound headed in the direction of these surfaces (see Figure 6.24). The structure was also placed on top of a 20 cm thick foam block to give it greater stability. Initially it was intended that all but 4 of the rods could be unscrewed and removed to enable a measurement to be made with the top and base alone, but in practice the difficulty in getting the rods into position made disassembly unfeasible.

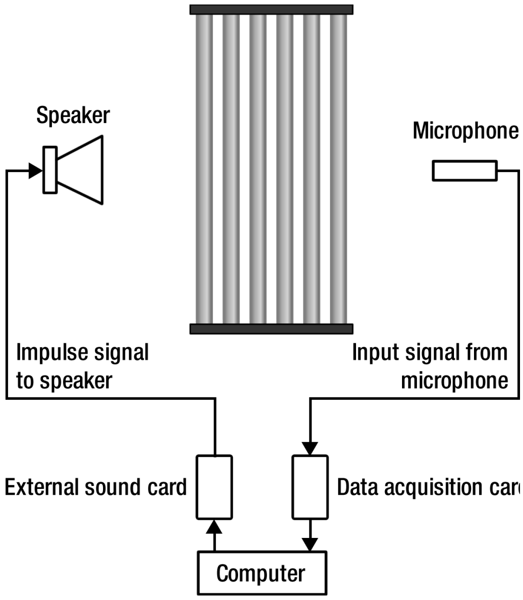


Figure 6.23: Schematic diagram of the experimental configuration. Reproduced from [383].

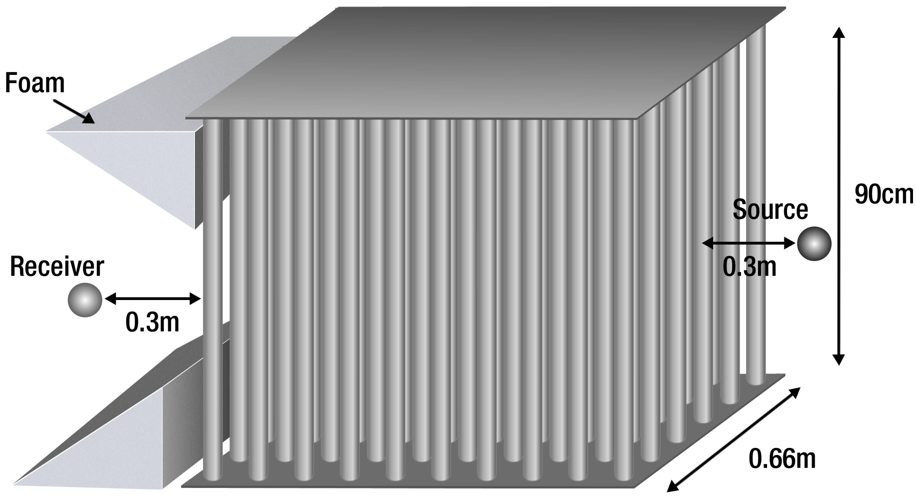


Figure 6.24: The source-receiver configuration in the real-world measurements of the sonic crystal sample. Impulse response measurements were made before and after the insertion of the foam wedges pictured, which were intended to reduce the interference effects due to the parallel top and base of the structure.

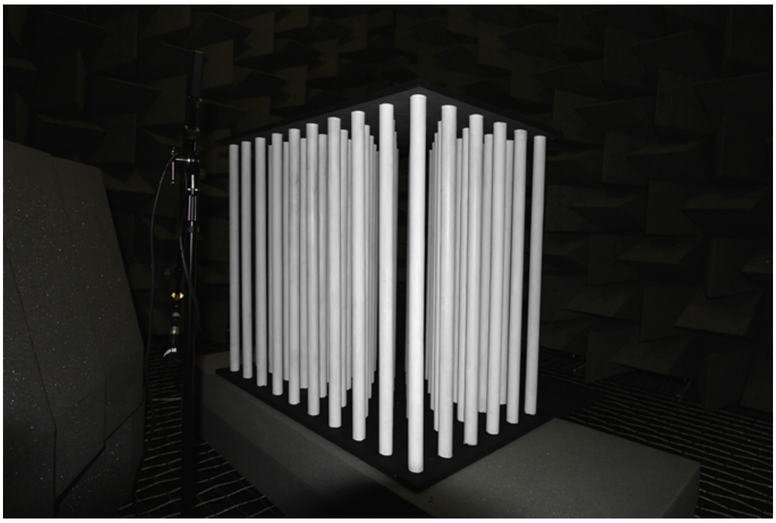
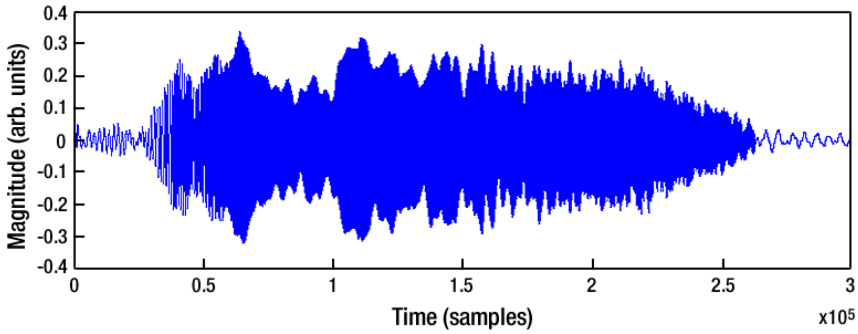
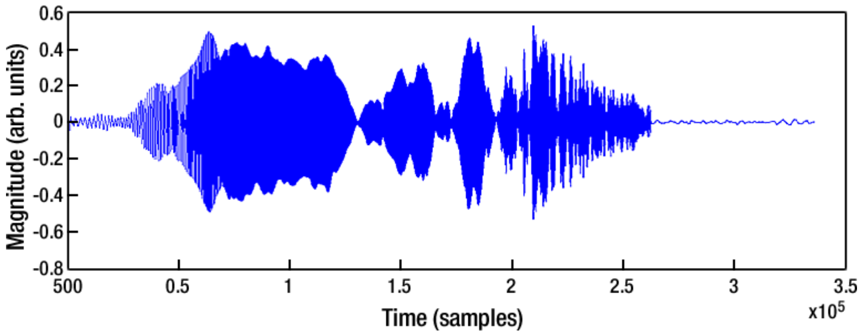


Figure 6.25: The sonic crystal sample pictured in the anechoic chamber in which measurement took place.

Results and Discussion



(a)



(b)

Figure 6.26: Output signals recorded by receiver before *(a)* and after *(b)* both the sonic crystal and foam wedges were inserted.

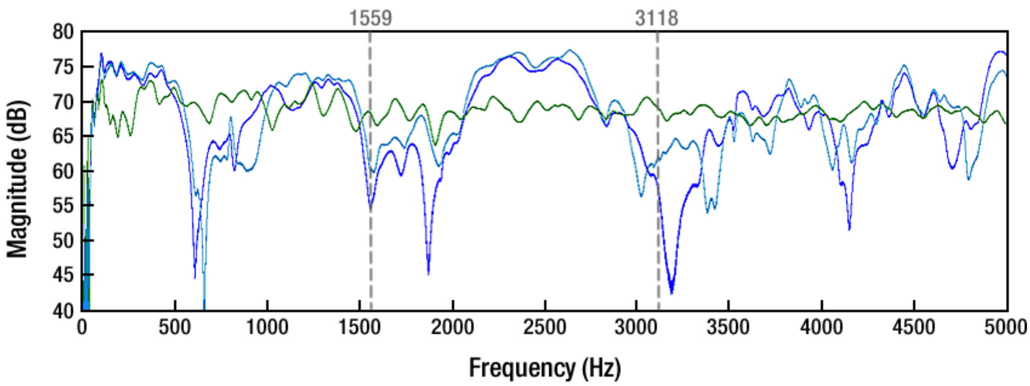


Figure 6.27: Frequency response curves of the impulse responses recorded without the sample (green), with the sample (light blue), and with both the sample and the foam wedges (dark blue).

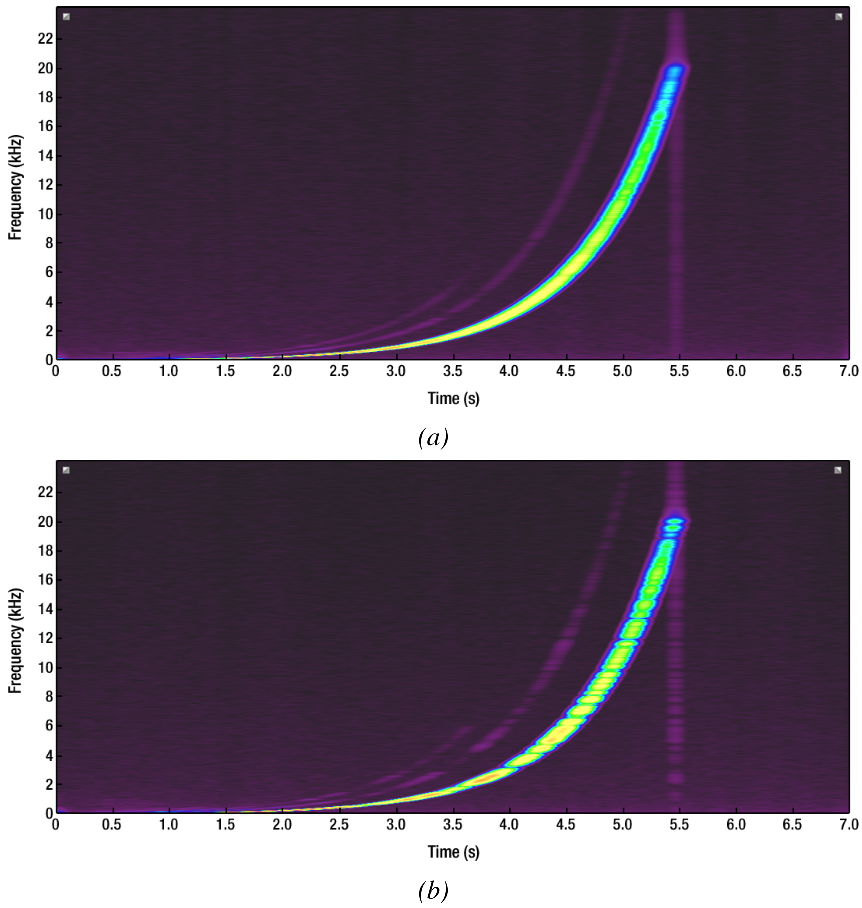


Figure 6.28: Frequency-time spectra of signals recorded before *(a)* and after *(b)* the sonic crystal sample and foam wedges were inserted. The horizontal lines in *(b)* indicate gaps in the transmission.

The results of the simulations with the Ricker wavelet show some correspondence with the theoretical predictions and the real-world measurements. For example, the fundamental band gap appears approximately at 1590 Hz as predicted. Furthermore, both the simulations and real world measurements suggest the presence of an earlier band gap at a half wavelength of the fundamental frequency (i.e. around 790 Hz). However, it also appears that the simulations are very sensitive to the discretised shape of the inclusions. For example, varying the radii of the inclusions leads to significant differences in the layout of the discretised unit cells (see Figure 6.17), which are manifested as noticeable differences in their respective band structures. A higher spatial resolution may therefore have yielded results which were even closer to the predictions. It is also apparent from the visualisations that the scattered field radiated by the structure consists of a pattern of peaks and null areas, hence there was found to be a significant variation in the band structure of the impulse response depending upon the measurement position. This effect may have been exacerbated by the shape of the wave front incident on the array. The theoretical predictions assume a one-dimensional plane wave source, which suggests the need to adjust the source position (or rotate the array) in order to observe Bragg reflection events associated with the second set of Bragg planes. One theory is that the spherical wave front produced by a point source may produce confounding effects by entering the array at different angles.

Also of interest while performing the simulations was the performance of the sine sweep as an input function as it is not thought to have been documented in the literature. While the signal-to-noise ratio was not as good as the Ricker wavelet on this occasion, it is possible that refining the technique may make it a feasible contender for some kinds of problems. For example, in free-field problems such as barrier estimation where the simulation does not need to be performed over a long time window. On the other hand, it is probably not suitable for room acoustic simulations or for modelling complex environments which, as well as having much larger problem domains, must also be performed over a longer period of time in order to capture the reverberant tail. This would not only prove computationally expensive with a sweep function, but may also result in more cumulated error.

Another limiting factor of the simulations is the long execution time of the update equation - a factor which increases significantly with each additional layer added to the array. Furthermore, increasing the depth of the array increases the path length between source and receiver, meaning a greater number of samples are needed to obtain the same level of detail in the impulse response. Performance of the programme could be improved significantly by utilising a GPU and by optimising the code with object oriented programming principles. While such changes would make the computation of much larger arrays feasible, the memory requirements of the program would still be prohibitively high.

6.4 Auralisation

Having obtained an approximation of the transfer function of the sonic crystal in the FDTD simulations, it is now possible to impose its filtering characteristics on any given audio signal through the mathematical process of convolution (see Section 5.3.1). The question is now, how to do this in a manner that stimulates a realistic subjective impression of a finite sized sonic crystal barrier in a complex environment. In other words, an environment in which the filtering effects of the sonic crystal may be obfuscated due to the fact it is only active on a small proportion of the total sound reaching a listener. In the simulations, the approximation of a free-field, coupled with the proximity of the source and receiver to the sample, ensured that most of the sound captured at the receiver had been transmitted through the crystal. In an auralisation intended to mimic real-world listening conditions, it would not be reasonable to apply the sonic crystal filter to a complete recording with no further consideration of sound that arrives at the receiver by alternative routes.

6.4.1 Barrier Approximation

Thinking first about sound which originates from sources that are occluded by the barrier, one recalls from Chapters 2 and 3 that there are in fact four possible outcomes due to its interaction with the barrier: transmission, diffraction, reflection and absorption (i.e. transmitted sound that is not re-radiated). The latter two of these can be neglected at this point since they are dealt with implicitly by the sonic crystal impulse response. Thus what remains to be established is the ratio of transmitted to diffracted energy at different frequency bands and under normal, free-field conditions. For this, one refers to the standardised procedure for estimation of the frequency dependent attenuation afforded by a thin rigid barrier, ISO 9613-2:1996 [60]. In the standard a formula is given for calculating the attenuation of sound due to a thin rigid barrier (see Section 2.4.2). The formula is based on a curve which has been derived from empirical measurements [59] and is used here to generate a frequency weighted amplitude envelope. When applied to a recording of the source sound, the envelope yields an approximation of the sound which reaches the receiver, despite the presence of the barrier. Since the formula assumes the barrier has a smooth, reflective surface, the majority of the energy that reaches the receiver must therefore be a result of diffraction around the barrier edges. The signal that results from this filtering process is therefore referred to here as the *diffracted signal*.

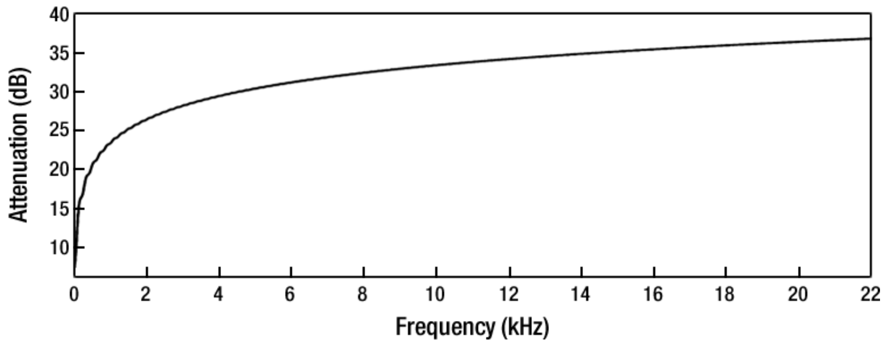


Figure 6.29: Frequency response of the amplitude envelope corresponding to the thin rigid barrier pictured in Figure 6.30.

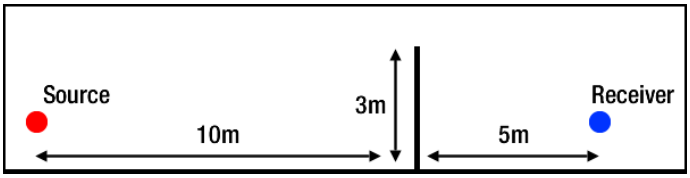


Figure 6.30: The source receiver scheme of the virtual barrier.

The *incident signal* is thus considered to be that which was removed by the filtering process - i.e. all the energy that was absorbed or reflected away from the barrier surface. It was first considered that the incident signal could be obtained simply by subtracting the diffracted signal from the original recording. However, this approach would only be valid if the two signals were perfectly aligned in time and phase - to the sample. Unfortunately these conditions are not met as convolution yields a signal of length $L+1$ where L is the length of the original signal. Furthermore, the filter derived from (2.30) is effectively a form of low pass filter, and a characteristic of low pass filters is that they cause the phase of the output to lag[411]. Thus an alternative approach to obtaining the residual signal involves inverse filtering - i.e. filtering the original signal with the inverse of the filter that was used to obtain the diffracted signal.

Once both the incident signal and the diffracted signal have been derived, the next stage in the process of auralisation is the convolution of the incident signal with the sonic crystal filter. Before this, the impulse responses must be re-sampled to match the sampling rate of the recorded material that forms the incident signal. Next the two signals are recombined by addition, although, this time, any phase differences are ignored since they are now considered to be a genuine effect of sound interacting with the barrier, as opposed to an artefact of the audio processing. Finally, the signal is combined with unprocessed ambient sound wherein the source sound is either absent or heavily suppressed. For clarification, this entire procedure is illustrated in Figure 6.31.

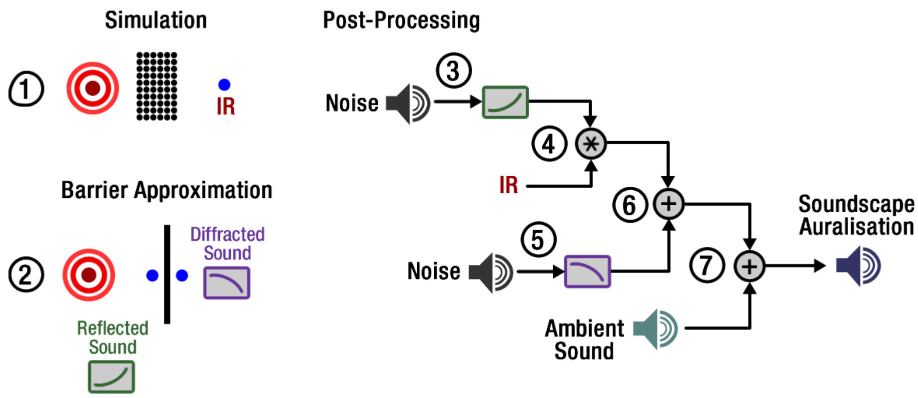


Figure 6.31: The step-by-step process of auralising a sonic crystal barrier. In Step 2. Maekawa's formula is used to derive the filter coefficients needed to obtain the diffracted and reflected portions of the signal.

6.4.2 Presentation

To achieve the most convincing spatial impression, the process given in Figure 6.31 is preferably performed using 3-D sound field recordings reproduced over a suitable multichannel system. Furthermore, as the sound field radiated by the crystal varies with position on the horizontal plane, one would prefer to capture the spatial cues in a B-Format impulse response which may then be convolved directly with a B-Format recording. Reproducing the resulting B-Format audio signal is possible using an appropriate decoder and loudspeaker array.

Recording the B-Format Impulse Response

In [322] Farina describes a technique for obtaining B-Format impulse responses using a single omnidirectional microphone. The technique involves making seven separate monophonic impulse response measurements, starting from a central reference position (R) and moving to six other closely spaced positions along the three Cartesian axes in both directions (X+, X-, Y+, Y-, Z+, Z-), as shown in Figure 6.32. Using this technique the B-Format impulse responses can be calculated using a numerical technique similar to that implemented in pressure-difference sound intensity probes. In a conventional pressure-difference sound intensity probe a signal proportional to the component of particle velocity is obtained by applying a finite difference approximation to the local spatial gradient of sound pressure [412]. The component of particle velocity may be approximated from

$$X(t) \approx \left(\frac{1}{\rho_0 \cdot d}\right) \int_{-\infty}^{t} [p_2(\tau) - p_3(\tau)]\delta\tau \tag{6.42}$$

where d is the distance separating the acoustic centre of the microphones. The resulting signals are processed using formulae based on the finite difference approximation of sound velocity between two closely spaced pressure sensors, as per (6.44). Hence the four signals corresponding to the WXYZ channels are as follows:

$$W(t) = P(t) \tag{6.43}$$

$$X(t) \approx \left(\frac{1}{\rho_0 \cdot d}\right) \int_{-\infty}^{t} [p_2(\tau) - p_3(\tau)]\delta\tau \tag{6.44}$$

$$Y(t) \approx \left(\frac{1}{\rho_0 \cdot d}\right) \int_{-\infty}^{t} [p_4(\tau) - p_5(\tau)]\delta\tau \tag{6.45}$$

$$X(t) \approx \left(\frac{1}{\rho_0 \cdot d}\right) \int_{-\infty}^{t} [p_6(\tau) - p_7(\tau)]\delta\tau \tag{6.46}$$

where the four components of the B-Format signal, $W(t)$, $X(t)$, $Y(t)$ and $Z(t)$, are formed using receivers $p_1(t)$ to $p_7(t)$, ρ_0 is the normal density of air, and distance d is the space between two associated receivers. The technique may also be performed using seven omnidirectional microphones arranged in a 3-D cross configuration and making the measurements simultaneously. While this approach is not generally the most practical to implement in real-world measurements, its implementation in FDTD simulations is straightforward - as demonstrated by Southern et al. [381] - and is achieved simply by defining additional receiver positions in the computer code. Having processed this array of virtual receivers using (6.46) in order to arrive at a complete B-Format signal, the sound field may be reproduced over a multichannel loudspeaker array using a suitable decoder to derive the loudspeaker feeds.

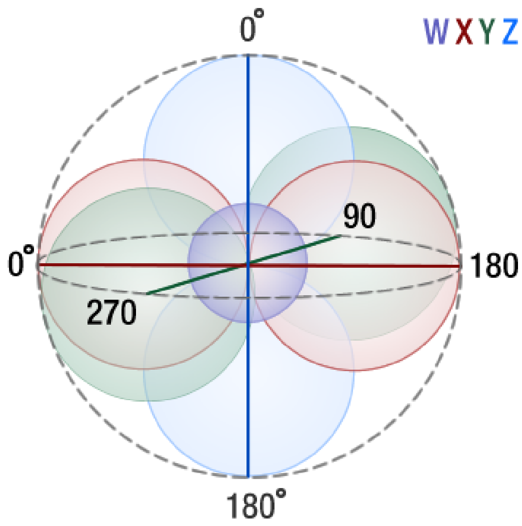


Figure 6.32: The 3-D microphone array required to encode the WXYZ channels of a B-Format signal.

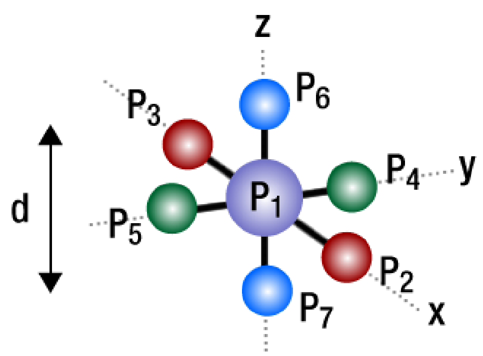


Figure 6.33: The 3-D receiver array required to encode the WXYZ channels in an FDTD scheme.

6.5 Summary

In this chapter a procedure was described whereby the transfer function of a 2-D periodic array of cylindrical scatterers was derived through 3-D FDTD simulations. While the 3-D FDTD scheme was arguably not necessary in the case of a 2-D periodic array, the scheme permits the investigation of higher order structures, as well as enabling the direct measurement of B-Format impulse responses from the sound field.

The FDTD scheme incorporates a Perfectly Matched Layer (PML) to emulate free-field conditions which unavoidably introduces artefacts into the simulation domain. For optimum results, four different source types were evaluated according to their Signal-to-Noise (SNR) ratios. On this basis it was found that a Ricker wavelet source produces the most reliable synthetic impulse response measurements for this particular FDTD scheme.

The chapter then went on to describe how the synthesised impulse responses would be used to render auralisations of sonic crystal noise barriers in complex urban sound environments through convolution with an appropriate B-Format source signal. It was suggested that the formula given in ISO 9613 [60] for estimating the frequency dependent attenuation curve of a finite sized rigid barrier could be used to weight the source signal prior to its convolution with the impulse response. The auralisation methodology has been conceived to facilitate the subjective evaluation of sonic crystals in the context of a real-world soundscape - an objective which shall be the focus of the next chapter.

7

A Case Study

7.1 Introduction

In this chapter a case study is presented to illustrate the application of the auralisation scheme outlined in Chapter 6. The scheme has been used in conjunction with selected soundscape and environmental noise assessment methodologies from Chapters 3 and 4 to facilitate the subjective evaluation of soundscapes in a virtual acoustic environment before and after the insertion of a sonic crystal noise barrier. The case study has been conceived in order to test the hypothesis that soundscape auralisation can be a reliable indicator of the perceived sound quality in urban environments, thus enabling the design and evaluation of noise treatments prior to their construction.

Here auralisation is seen as just one part of a multi-stage process, one that spans the design, simulation and evaluation of a new form of noise treatment in the context of a real world soundscape. The scheme can be broken down into three main components: first, the initial site survey work; second, the design and auralisation of the soundscape before and after insertion of the noise treatment; and third, the subjective analysis of the auralisation. As details concerning the technical aspects of the auralisation have already been covered in Chapter 6, they are not covered in depth here. More attention has instead been given to the design and compositional aspects involved in producing the auralisation material. A substantial portion of this chapter is also dedicated to the test methodologies used in the subjective analysis. These include a combination of laboratory style listening tests, and a phenomenologically based assessment consisting of a series of 'virtual sound walks' intended to emulate the experience of being in a real soundscape. The success of the virtual sound walk methodology is verified by comparison of the results obtained without the insertion with real sound walks carried out in the environment under observation. It has also been suggested that auralisation may permit a deeper insight into the perception of environmental sound when it is disassociated from visual stimuli and, to some extent, the learned emotional and intellectual connections that subjects attach to real places. It is expected that a comparison between real and virtual sound walks may offer some indication as to whether or not this is the case.

7.2 Site survey

In the following case-study, virtual augmentation of the soundscape in an urban green space is attempted through the selective placement of a bespoke form of sonic crystal sculpture. Before this can be realised, a certain amount of preparatory work must take place so that the design of the structure may be tailored to the space for which is it intended. First and foremost, a suitable location was needed for the case study. The main requirements were that it was a public, recreational space located within a large urban area. Ideally the space should have a vibrant soundscape, and one that is known to be adversely affected by environmental noise. The first to be considered for the study were spaces within the city of Leeds. Leeds is situated within the West Yorkshire urban area (Figure 7.1) and is estimated to be the third most populous city in the UK according to the 2011 census [413]. Following recommendations made by Leeds City Council's environmental protection team, it was decided that the area of Woodhouse Moor in Leeds would be an ideal location for the case study. Woodhouse Moor, also known as Hyde Park, is an area of green space approximately one mile from Leeds city centre. It comprises three parts: a formal park of around 26 hectares on the west of Woodhouse Lane, and two other open areas to the east of it. These are known as the Monument (or Upper) and Cinder (or Gravel, or Lower) Moors which are used for events such as circuses and sporting matches, and sometimes car parking [53] (Figure 7.2).

As of 2005 the park had just under 3 million visits a year and is the second most popular urban park in Leeds according to a report published by Leeds City Council in 2006 [414]. Being situated close to the Universities makes it a popular social venue for the city's large student population. The park is also host to a variety of public events, including Unity Day - an annual live music and cultural activities festival. The park has five main paths which meet in the centre, each of which is tree-lined, and together they divide the park into different areas of usage. For example, the north-eastern corner of the park houses the children's play area, skate park, picnic area, basketball courts and public conveniences; to the west are the allotments and tennis courts; and to the east are gardens and dedicated nature areas. The Woodhouse Moor area is known to be adversely affected by environmental noise, its two primary sources being road traffic (the park is bordered by roads, one of which is dual carriage-way and a main artery to Leeds city centre); and aircraft flying to and from Leeds-Bradford Airport.

Having selected a suitable location for the case study, the next step was to carry out an acoustic survey of the site in order to identify any specific noise concerns - particularly those in breach of the European guidelines [8]; and to gain a general impression of the soundscape, highlighting both its strengths and its weaknesses. After the site survey, it will then be possible to outline the purpose of the noise intervention more precisely and to consider its desired effect on the soundscape. For example, which are the areas that would benefit most from treatment? And will the aim of the treatment be to make an acoustically active area more conducive to relaxation, or to make a common and monotonous soundscape more stimulating? In order to answer these questions, it was considered that two types of acoustic survey would be needed. The first of these is a quantitative survey, involving detailed noise mapping of the affected area; and the second a qualitative survey, in the form of a sound walk and questionnaire.

Figure 7.1: The West Yorkshire urban area showing Leeds and Woodhouse Moor. ©Google 2013.

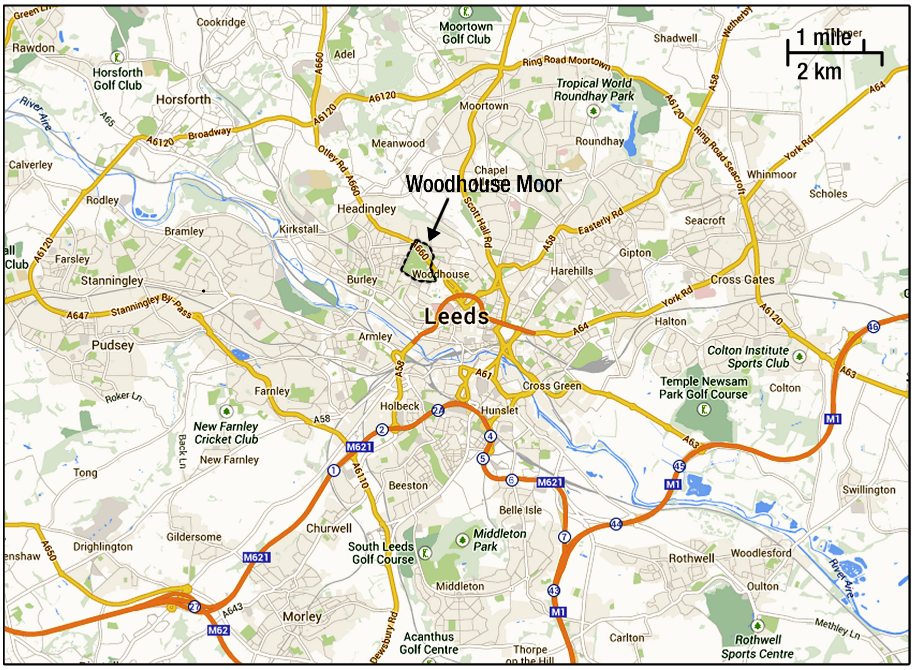


Figure 7.2: Satellite view of Woodhouse Moor divided into 3 main areas. ©Google 2013.

(a) Entrance from Clarendon Road

(b) The lower moor

(c) View north from centre of park

(d) Tennis courts

(e) Children's play area

(f) Hyde Park Corner

Figure 7.3: Images of Woodhouse Moor.

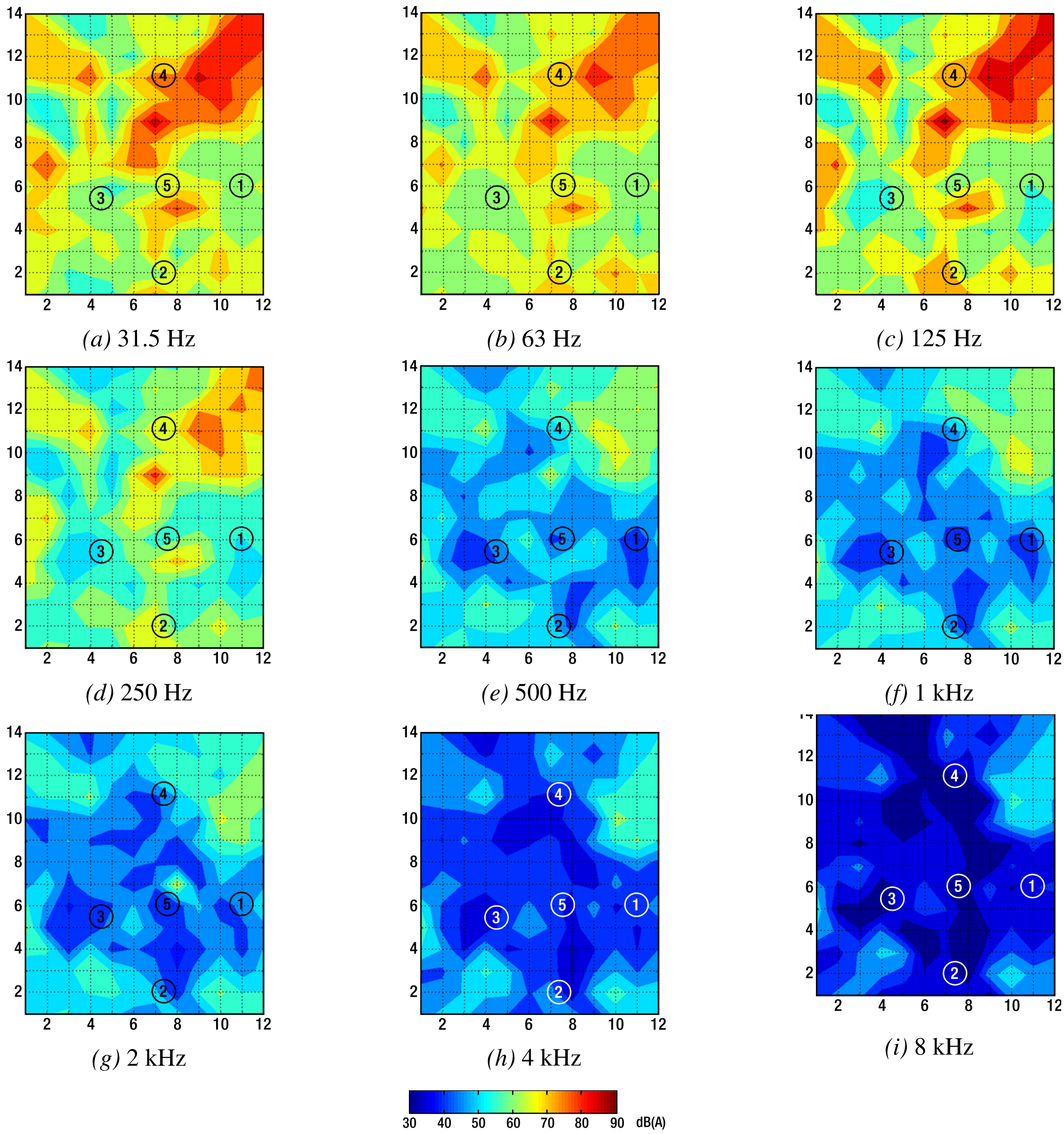
7.2.1 Quantitative Survey: Noise Mapping

A noise map is a graphic representation of the sound pressure level (SPL) spatial distribution in a region, and is usually represented using contour lines (see Section 3.4.4). While there are no strict guidelines for noise mapping, the general procedure involves dividing the space under observation into a grid of receivers and carrying out a number of $L_{Aeq,T}$ measurements at each coordinate on the grid. It is then possible, using Fourier analysis, to obtain $L_{A,f}$ readings for octave or 1/3 octave bands. Noise mapping is almost always performed using specialist software, here the decision was made to carry out the procedure by direct measurement. There are two main reasons for this. Firstly, the area is relatively small, thus it was presumed that greater accuracy could be achieved through field measurements. Secondly, the time spent performing the measurements enabled the researcher to become very familiar with the soundscape in different areas of the park, which would prove valuable when devising the route for the sound walk amongst other things.

A minimum of two $L_{Aeq,2min}$ measurements were recorded in each measurement position using a Norsonic NOR110 sound analyser. The meter was configured to record $L_{A,f}$ at octave bands ranging from 31.5 Hz - 8 kHz. In most circumstances it would be considered good practice to carry out multiple measurements throughout the year and at different periods in the day. However, within the context of the case study, the author was mainly interested in observing the park at its busiest. Hence it was considered appropriate to make all of the measurements during daylight hours and in fair weather.

The coarseness of the noise mapping grid generally depends on the size of the area being mapped, and can range from tens to hundreds of metres. In an area as small as Hyde Park, a decent noise mapping tool might set the distance between receivers at 10 m, but this would result in an impracticably large number of measurement positions. A rectilinear grid with a resolution of 50 m was therefore chosen, and a least squares method of interpolation was subsequently used to refine it by a factor of 5. Measurements were made approximately 1.5 m above tread level, as recommended in ISO 1996 [126]. After interpolation, the contour lines of the noise map were plotted in Matlab in 5 dB(A) ranges (see Figure 7.4).

The results of the noise mapping survey indicate that noise was distinctly more dominant in the lower frequencies, particularly around the 31.5 and 63 Hz regions where it was often above 70 dB(A). At these levels - and for the relatively short periods of time one would typically spend in such a location - low frequency noise (LFN) is unlikely to cause any lasting physiological side effects. Furthermore, the overall levels in the park are mostly below the recommended threshold given by the WHO of 55-65 dB(A). Yet even over a relatively short exposure time, it is possible that some people might experience one or more of the symptoms of LFN annoyance, which include feelings of irritation and unease, headache and fatigue. One of the reasons why LFN can be particularly annoying is due to its effect on speech intelligibility. At sufficient volumes, the sort of low frequency hums and drones one typically encounters in urban areas can have an undesirable masking effect in the fundamental frequency range of speech. This tends to start occurring when the signal-to-noise ratio is less than 0 dB - i.e. when the SPL of the masking noise is higher than that of speech. Speech intelligibility is defined as the percentage of the words that are understood correctly in a speech intelligibility test and is typically expressed as a function of the signal-to-noise ratio (SNR) [415]. Poulsen et al. suggest a minimum SNR of -3dB is needed to achieve 80% word intelligibility, while the speech recognition threshold (defined as the SNR corresponding to a 50% score in a speech intelligibility test), is said to be around -8dB. This means that, in the presence of background noise exceeding 80 dB(A), the level of speech needs to be in the region of 72 dB(A) - and that is assuming the listener has normal hearing and is being addressed in their native language, which of course may not be the case. Given that the most comfortable listening level (MCL) for speech in terms of both loudness and intelligibility lies somewhere in the region of 64 dB SPL [416], the effort of raising one's voice some 10 dB above that in order to compete with high levels of background noise can be damaging to the larynx (a condition medics refer to as *mechanical laryngitis*), particularly to one who is unused to projecting their voice. Added to that is the general irritation which may be felt, both by participants in the conversation and by those in their immediate vicinity.



(a) 31.5 Hz *(b)* 63 Hz *(c)* 125 Hz *(d)* 250 Hz *(e)* 500 Hz *(f)* 1 kHz *(g)* 2 kHz *(h)* 4 kHz *(i)* 8 kHz

Figure 7.4: Noise maps generated using $L_{Aeq,T}$ measurements recorded in Woodhouse Moor. The overlaid numbers show the approximate locations that are later used in the sound walk (Figure 7.5).

From looking at Figures 7.4, it may be noted that, barring one or two exceptions in the low frequency regions, in most areas of the park the SPLs are within the safe limits recommended by the WHO. Locations 3 and 5 benefit from the lowest SPLs being situated further away from the surrounding roads. The noisiest area of the park is around the skate park and children's play area, and while this is not necessarily a problem in itself, its effects on other areas in the park could perhaps be ameliorated to a certain extent by better separation between acoustic zones. It is therefore concluded that, while the noise levels in the park are insufficient to warrant immediate concern, the quality of the sound environment and its conduciveness to social and relaxation activities could be improved.

7.2.2 Qualitative Survey: Soundwalks

Subjective approaches to soundscape analysis vary significantly depending on the purpose of the research and the discretion of the researcher. The sound walk methodology adopted here is largely based on Berglund and Nilsson's 'structured listening walk' which they proposed as a tool for measuring the perceived soundscape quality in urban residential areas [417]. A sound walk essentially involves leading a small group of participants along a predefined route, during which the aim is to concentrate on what can be heard and the physical environment. It was felt in this case that the 90 minute duration of Berglund et al.'s structured listening walks would be inappropriate given the relatively small area being studied. It was therefore decided that an approximate duration of 60 minutes would be adequate for the sound walk, allowing 30 minutes walking time, 15 minutes for a short introductory talk and pre-sound walk questionnaire, and 15 minutes for shared reflection and concluding remarks. The purpose behind the pre-sound walk questionnaire was to obtain some general information about the participant, such as their age, gender and profession, as well as their precepts about Woodhouse Moor and expectations regarding the sound walk. While on the sound walk, at each of five pre-determined locations (see Figure 7.5) participants were instructed to stop and listen in silence for a short period of time while answering some location specific questions. In the first of these questions they were asked to rate the soundscape with respect to nine soundscape semantic differential scales. The scales employed were the same as those originally proposed by Raimbault for carrying out qualitative judgements of an urban soundscape [418] (see Figure 7.6). Following that, participants were encouraged to record their comments and observations about the soundscape in their own words. At the end of the sound walk, they were then asked to comment on the sound walk more generally. A copy of the blank questionnaire is given in Appendix 9.1.

A total of 16 participants were involved in this survey, 5 female and 11 male, mostly within the 21-35 age range and from mixed professions. It would not have been practicable for all participants to have attended a single sound walk, hence they were split into three groups of 4-5 and conducted the walks over 3 consecutive days during late June 2011.

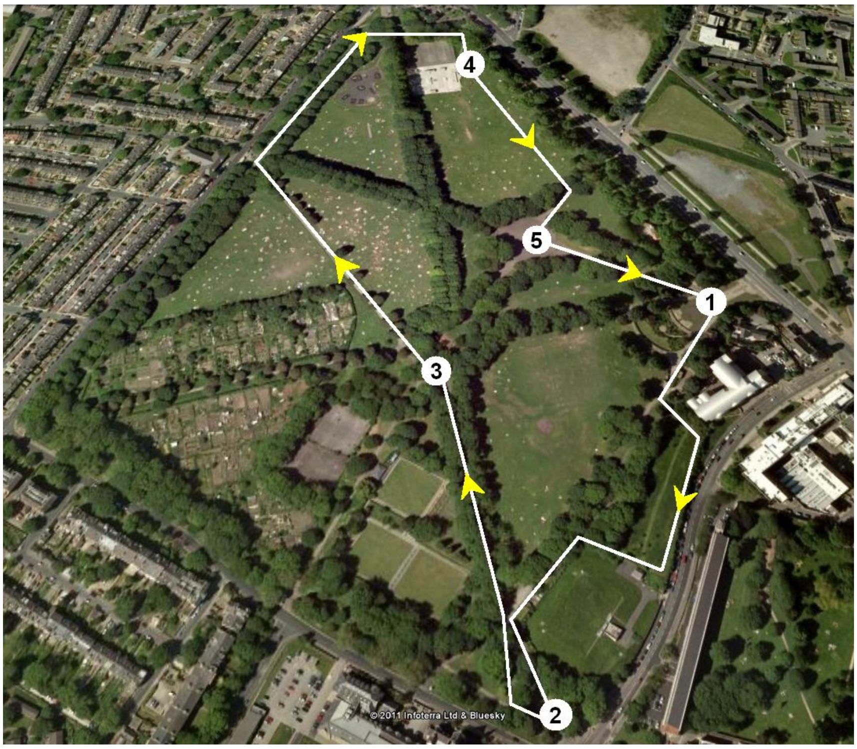


Figure 7.5: Aerial view of Woodhouse Moor overlaid with sound walk route and 5 stopping points. Image © Google 2013.

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

Figure 7.6: Semantic differential scales used in sound walk survey.

Each of the five stopping locations on the sound walk were identified by the author as having quite distinctive characteristics - visually and aurally. With reference to Figure 7.5, these are summarised in the following.

Location 1

Geography	Located near the south west entrance to the park, close to the Woodhouse Lane and Clarendon road junction.
Visual Feaures	Colourful flower beds surrounded by tarmac with park benches and a large statue of Queen Victoria facing the road. A well tended area in a prominent position of the park.
Soundscape	Regular stream of traffic down Woodhouse Lane is a dominant feature of the soundscape, particularly the frequent buses. Birdsong, pedestrians, and the distant activities of people in the park are also noticeable. Occasionally aircraft and church bells may be heard.
SPLs	**Hz** 31.5 / 63 / 125 / 250 / 500 / 1k / 2k / 4k / 8k **dB(A)** 58-63 / 57-63 / 55-60 / 48-53 / 38-43 / 40-45 / 38-43 / 38-43 / 30-35

Figure 7.7: Location 1 facing east onto Victoria monument and toward Woodhouse Lane

Location 2

Geography	Located at the south east corner where Moorland Road meets Clarendon road.
Visual Features	Statue of the Duke of Wellington situated on a small island surrounded by tarmac. Lines of trees, hedges and grass verges separate interior of park from this outer corner and connected paths. A pedestrian crossing is situated here.
Soundscape	The sounds of intermittent traffic turning into and exiting Moorland road and the regular beeping of a pedestrian crossing are distinctive features of this soundscape. Due to traffic lights, traffic sounds are irregular, with vehicles often idling, accelerating or braking.
SPLs	**Hz** 31.5 / 63 / 125 / 250 / 500 / 1k / 2k / 4k / 8k **dB(A)** 60-65 / 65-70 / 65-70 / 59-64 / 47-52 / 45-50 / 42-47 / 40-45 / 30-35

Figure 7.8: Location 2, facing south west with Clarendon road to the left

Location 3

Geography	Central circular area through which two of the park's main footpaths intersect.
Visual Features	Area surrounded by trees, raised beds and foliage with benches arranged in an arc around the eastern perimeter of the circle.
Soundscape	Birdsong and distant traffic noise provide a constant ambience, which is regularly interspersed with pedestrians and cyclists. It is close to tennis courts and a flat grassy area, hence the sounds of people playing tennis and field sports are frequently heard. During summer the grounds keepers' machinery is also a regular feature of the soundscape.
SPLs	**Hz** 31.5 63 125 250 500 1k 2k 4k 8k **dB(A)** 57-62 58-63 54-59 47-52 42-47 40-45 36-41 35-40 30-35

Figure 7.9: Location 3, facing south west

Location 4

Geography	Area located to the north west of the park, near the busy junction at Hyde Park Corner.
Visual Features	Part of a large, open grassy area and next to the skate park, basketball courts and children's play area. Public toilets, trees and shops also visible nearby. There is usually a lot of litter and areas of burnt ground following sunny spells.
Soundscape	Strong and varied, also the loudest area in the park. The sounds of skateboards, basketballs, children playing and students picnicking on the main grassy area, along with the occasional ice-cream van and game of kabaddi are all features of this soundscape.
SPLs	**Hz** 31.5, 63, 125, 250, 500, 1k, 2k, 4k, 8k **dB(A)** 71-76, 69-74, 70-75, 62-67, 48-53, 47-53, 40-45, 30-35, 30-35

Figure 7.10: Location 4, facing Hyde Park Corner (north east)

Location 5

Geography	Large circular area of tarmac situated slightly west of centre.
Visual Features	Around the perimeter are a few benches, trees, and a pink bottle bank that has been covered in graffiti and fly posters. Facing it is a boarded-up Indian restaurant. The tarmac is badly damaged with grass growing through in some areas.
Soundscape	Regular flow of road traffic and birdsong with occasional pedestrians and cyclists passing. Sounds from other locations in the park can be heard more faintly.
SPLs	**Hz** 31.5, 63, 125, 250, 500, 1k, 2k, 4k, 8k **dB(A)** 65-70, 67-72, 61-66, 55-60, 40-45, 35-40, 36-41, 37-42, 30-35

Figure 7.11: Location 5, facing west

Results and Discussion

The range and frequency of sounds that participants singled out in their written feedback is presented in Figure 7.12. This is not only an indicator of the diversity of the soundscape, but also reflects the overall balance of the soundscape in Woodhouse Moor in the sense that those sounds considered most dominant and/or evocative are those most often remarked upon. It may also serve as a useful source for comparison between the real and virtual sound walks.

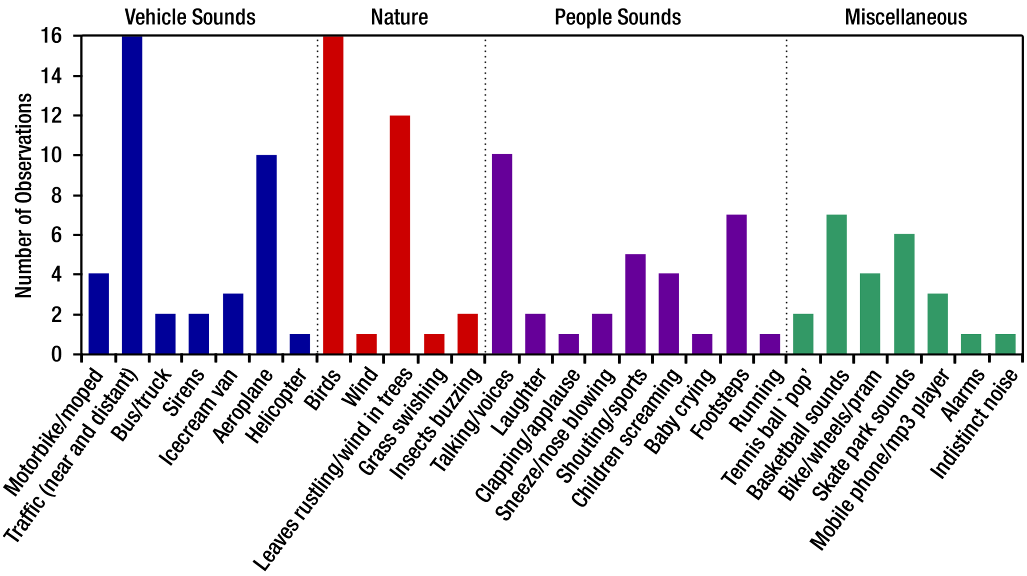
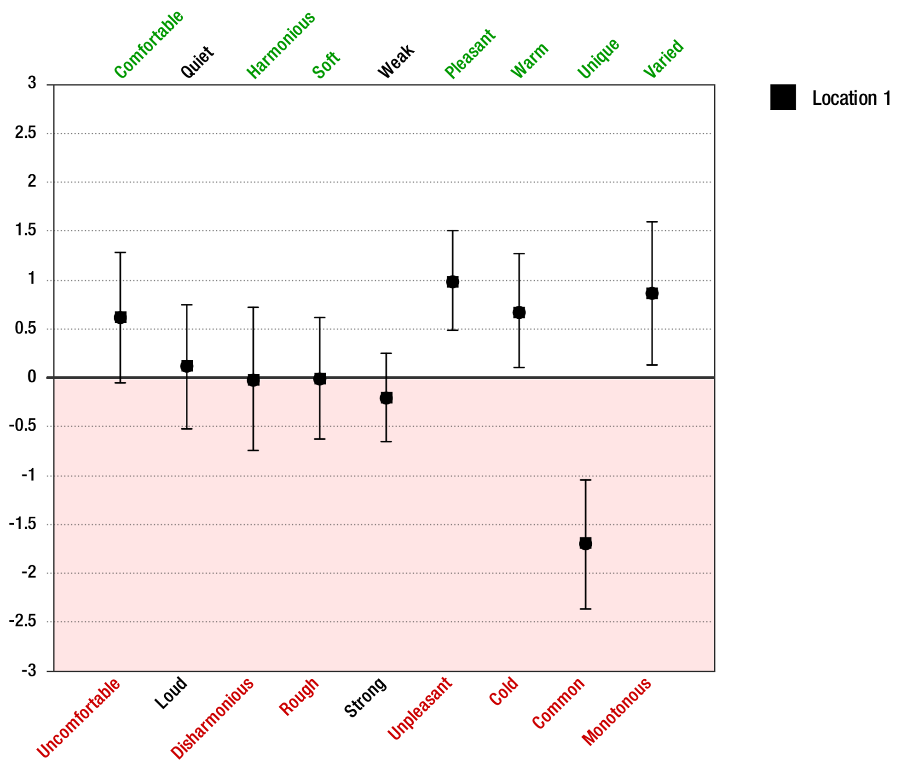


Figure 7.12: Chart showing range and frequency of sounds identified by subjects in written feedback.

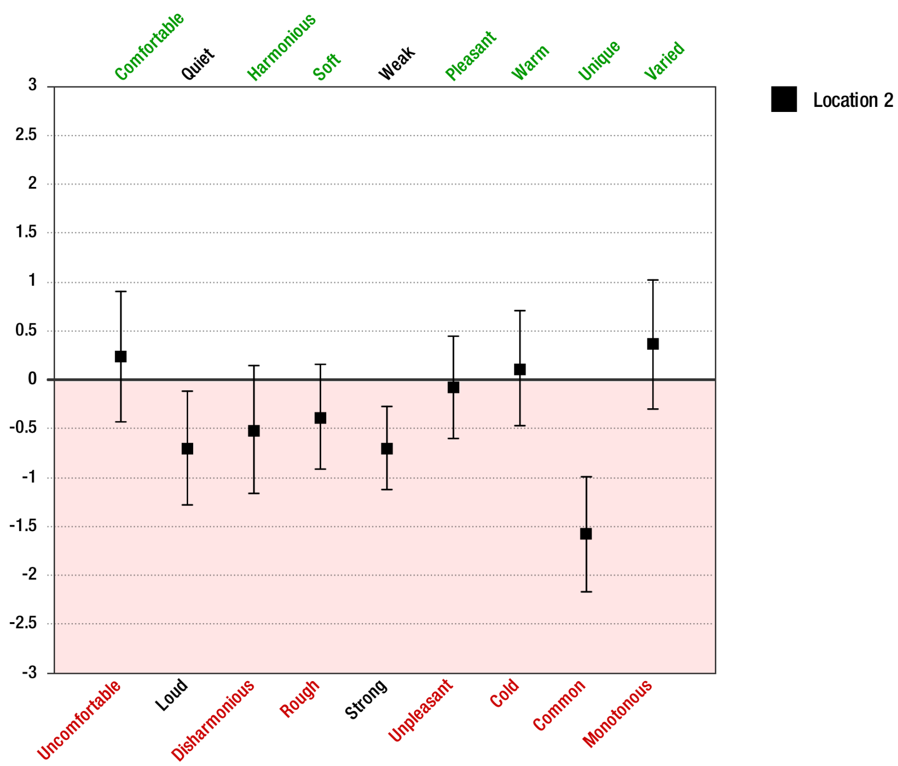
Next, the responses to the semantic differentials were collated and analysed for statistical significance. To qualify the use of these data sets in follow up experiments, it should be demonstrated that these differentials are able to produce statistically meaningful results. Interpretation of the results is less critical at this stage; one simply wants to confirm that they provoke responses amongst subjects that are significantly different from those that would occur from random selection. A good indicator of this would be if the responses recorded in different locations were significantly different from one another, particularly if this was then corroborated by the written responses given in the questionnaires, as well as the verbal feedback recorded by the interviewer. A summary of the questionnaire data is presented in Appendix 9.2.1. The data is available in its entirety on the DVD accompanying this thesis (Directory 9.7.4.1).

From observing the mean values and respective error bars in a series of box and whisker plots (see Figure 7.13), it appears there may be significant differences between locations 3 and 4 compared with locations 1, 2 and 5 which appear quite similar. In order to confirm or disqualify this theory, it is necessary to perform suitable statistical hypothesis tests. Before doing so, the data must be subjected to normality tests to check for normal distribution in the data, as this will impact on the choice of hypothesis test. Initially the data sets were tested for normal distribution using the Chi Square method [419]. In this method a high Chi Square value is an indicator that a data set is probably not normally distributed about the mean average. The results of these tests indicated that in almost all cases, the hypothesis of normal distribution cannot be rejected. However, in the majority of cases, the histograms of the data sets do not appear to be normally distributed, which leads one to suspect that the Chi Square test may not have been the best choice of normality test for this data, most probably due to the small sample size. For small data sets, the Shapiro-Wilks [420] test is considered a more reliable test for normality. Using this method, the majority of data sets were found not to be normally distributed when tested with a 5% significance level. There are several explanations for this. Looking at the histograms (see Figure 7.14), it is apparent that, in some cases, there is an obvious skewness in the data which is thought to have resulted from responses tending toward the extreme ends of the differential scale. It is also apparent that in some other cases the distribution may be bimodal, indicating that the population under test is split between two different extremes. This may be a genuine division due to the relative nature of the descriptors - for example, one would expect that participants who spend more time in the country might have different aural perceptions from those who spend more of their time in big cities - or it may be due to how the individuals have interpreted some of the more ambiguous descriptors. The pair of adjectives that stood out as being the most difficult to interpret were harmonious-disharmonious, suggesting musical attributes may not be an appropriate choice in soundscape studies. This was closely followed by warm-cold, which in room acoustics tends to be associated with the reverberant characteristics of the space. Again, in relation to outdoor environments, the terms are much less intuitive - these being characteristically un-reverberant environments. Furthermore, both of these descriptors seem prone to occupational bias, in the sense that applicants for whom music or acoustics is a main occupation might apply an advanced understanding of these adjectives in a way that a layperson would not. It may also be due to variations in the test conditions between successive sound walks. While effort was made to ensure the meteorological conditions of tests were consistent, this is of course no guarantee that the sound field was identical in all respects. Detailed statistics and a complete set of frequency histograms are given in Appendix 9.2.2.

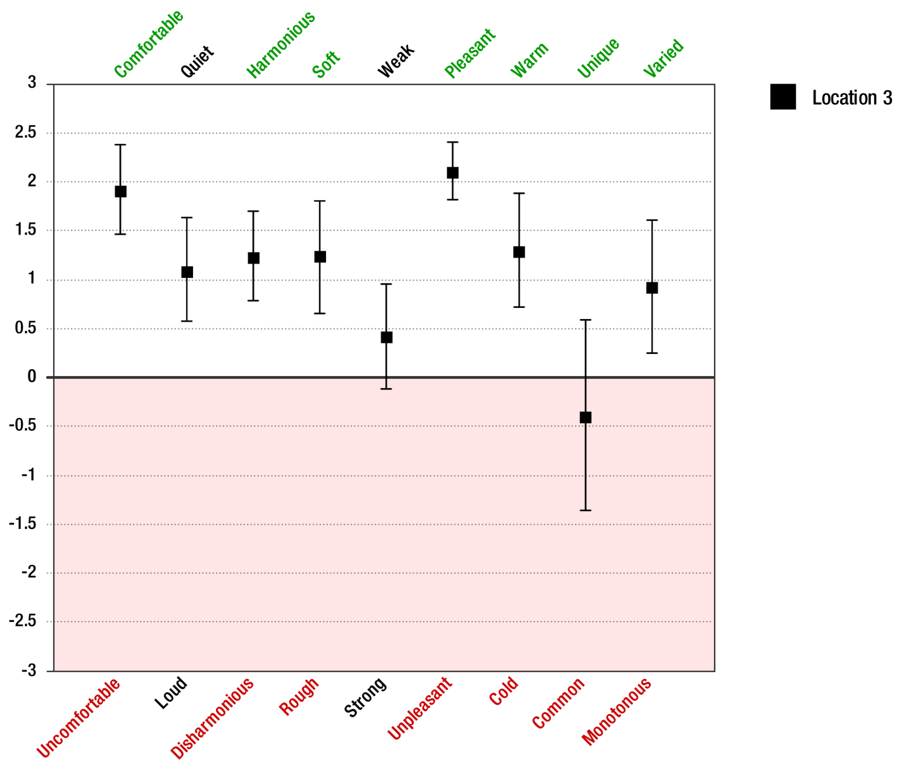
Figure 7.13: Rated sound quality in 5 areas of Woodhouse Moor using soundscape quality metrics. Sample size: 16.



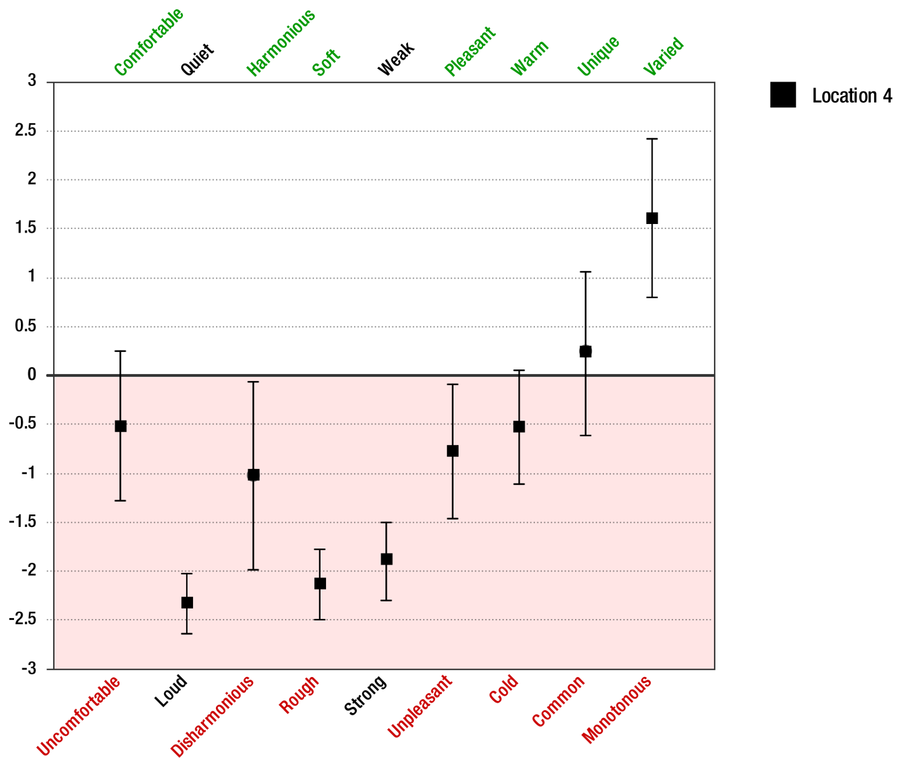
(a)



(b)


Comfortable
Quiet
Harmonious
Soft
Weak
Pleasant
Warm
Unique
Varied
3
2.5
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
-2.5
-3
Uncomfortable
Loud
Disharmonious
Rough
Strong
Unpleasant
Cold
Common
Monotonous
Location 3

(c)


Comfortable
Quiet
Harmonious
Soft
Weak
Pleasant
Warm
Unique
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3
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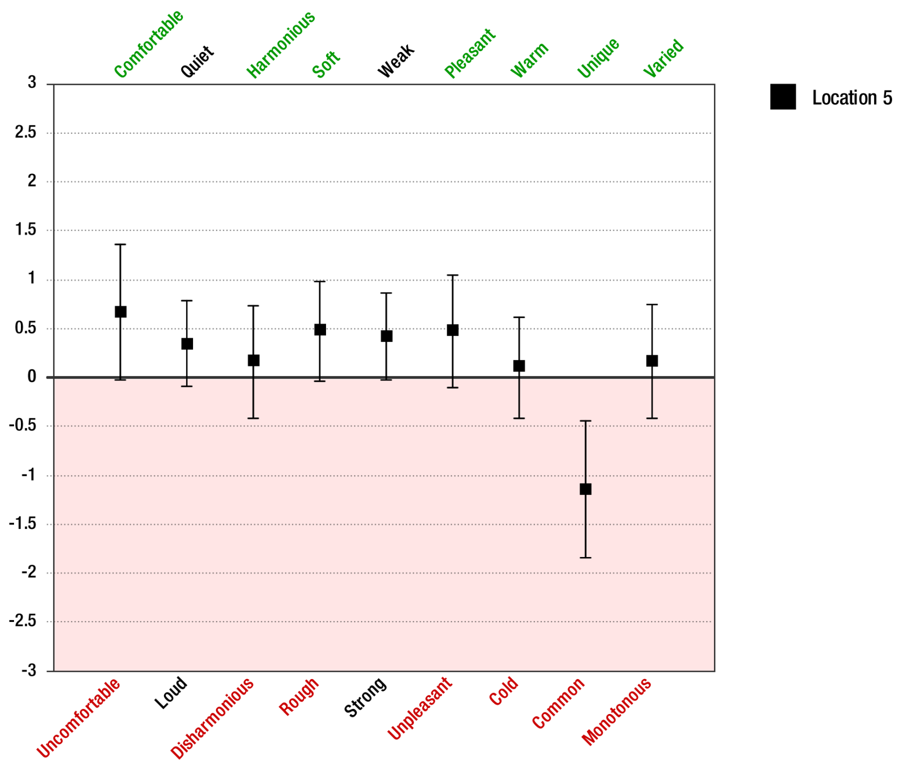
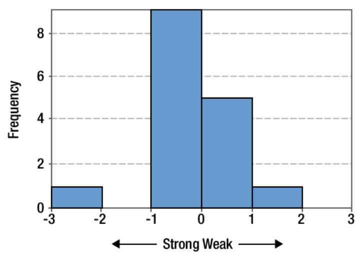
0.5
0
-0.5
-1
-1.5
-2
-2.5
-3
Uncomfortable
Loud
Disharmonious
Rough
Strong
Unpleasant
Cold
Common
Monotonous
Location 4

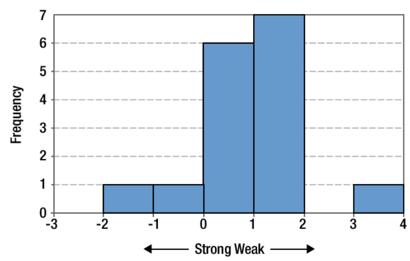
(d)



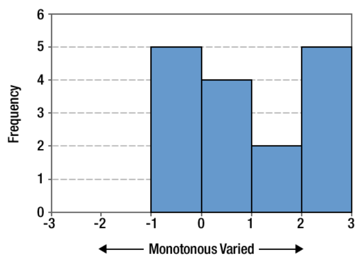
(e)



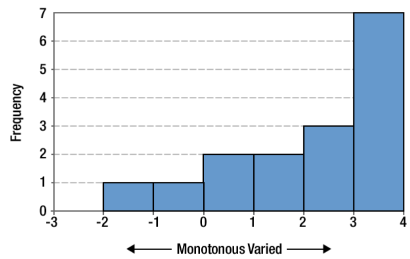
(a) Location 2 - Weak/Strong



(b) Location 3 - Weak/Strong



(c) Location 2 - Monotonous/Varied



(d) Location 4 - Monotonous/Varied

Figure 7.14: Some examples of histograms showing distribution of responses to semantic differential scales. A complete set is given in Appendix 9.2.3.

Due to the distribution of the data, it was necessary to use a non-parametric form of hypothesis test to determine whether the apparent differences between locations are more likely to have occurred by chance, or are in fact statistically significant. Rather than comparing every differential for every possible combination of locations, only the sets which emerged from verbal and written feedback as being most different were tested. These also happened to be the sets which appeared most different in the box plots. Since the soundscapes in locations 1, 2 and 5 were deemed to be quite similar on this basis, it was decided that location 2 would be contrasted with both location 3 and location 4 independently. The type of test chosen was the Mann-Whitney U-test [421], a well known non-parametric statistical hypothesis test suitable for small and arbitrary sample sizes. However, the U-test does assume that the distributions of the data sets under comparison are similar, which may make some of the results less reliable than others. The results of the U-tests are interpreted as follows: if the null hypothesis is rejected, one data set tends to have significantly higher or lower values. If the null hypothesis cannot be rejected, it is assumed that the data sets are not significantly different. The results have been calculated with a probability of error value of $\alpha = 0.01$ and are given in Figure 7.15. H_0 indicates the null hypothesis has not been rejected, while H_1 is an indicator that the null hypothesis can be rejected with a 1% probability of error, which is equivalent to a 99% confidence level. There were also 3 cases where the corresponding data sets in both locations 2 and 4 were believed to contain normally distributed data. A second kind of statistical hypothesis test, the Welch t-test [422], was thus performed on those data sets to see if the results corresponded with those of the U-test. In all 3 cases the result of the Welch t-test corresponded with that of the U-test.

Location	Comfort/discomfort	Quiet/loud	Harmonious/disharmonious	Soft/rough	Weak/strong	Pleasant/unpleasant	Warm/cold	Unique/common	Varied/monotonous
2 vs 3	H1	H1	H1	H1	H1	H1	H0	H0	H0
2 vs 4	H0	H1	H0	H1	H1	H0	H0	H1	H1

Figure 7.15: Results of significance test between data sets in different locations. H0 indicates null hypothesis not rejected (ie. no significant difference), H1 indicates null hypothesis rejected with 99% confidence (ie. difference is significant).

Despite the relatively high levels of variation in the data sets, the results of the U-test have indicated that there are likely to be significant differences in the perceived sound quality between locations 2 and 3, as well as between locations 2 and 4. In terms of the verbal feedback, there was a very positive response to location 3, which was widely considered the more pleasant, comfortable and harmonious of all the locations. Interestingly, despite scoring highly in the semantic differential test on loudness, roughness and strength, location 4 - the area recorded as having the highest sound levels in the noise mapping survey (see Figure 7.4) - was not considered by many of the participants to be significantly unpleasant or uncomfortable. Verbal remarks on the soundscape in this location were often quite positive and included adjectives such as ‘active’ and ‘vibrant’, although it was not somewhere they would choose to spend time relaxing. Locations 1 and 2 were considered to be quite similar, although location 2 was perceived to be slightly more peaceful than location 1. Location 5 was not too dissimilar from 1 and 2, though generally perceived to be more relaxing - which may also have been related to the fact it immediately proceeded location 4, the least relaxing of the locations. These responses seem to bear a relationship with the sound levels recorded in the noise mapping survey in which locations 1, 2 and 4 had the highest sound pressure levels in the most sensitive regions of human hearing (2-4 kHz)), while locations 3 and 5 were somewhat quieter. Athough the sound levels were similar between locations 3 and 5, location 5 in fact bore a closer resemblance to locations 1 and 2 in the semantic differential test. This might be related to the visual ambience of the locations - location 3 being considered quieter and more 'remote' on account of the visual separation created by the surrounding trees.

On the basis that significant differences have been observed between data sets, it can be concluded that, in the majority of cases, the data is unlikely to have occurred through random selection. This suggests that these particular semantic differentials probably have some statistical value - albeit some more than others. However, whether or not one agrees with the choice of adjectives is less relevant, since the analysis was primarily aimed at establishing whether or not the data produced by these differentials has statistical value in order to qualify them for use in the next phase of the case study. Later on in the case study, the results from the semantic differential scales are used as a basis for determining whether or not similar experiences can be expected in virtual soundscape listening experiments. One should therefore be satisfied that the differentials have some merit or any subsequent comparison between data sets would also be meaningless.

It would thus seem that variation and contrast are important components of a soundscape and the key to ensuring noisy areas are perceived positively. It also seems that visual cues are highly influential on auditory response, which is a particularly important consideration when designing noise interventions for recreational spaces. While the simpler and more 'traditional' styles of noise barrier may well be acceptable along a motorway, they are unlikely to be acceptable in a residential area or green space. The reasons for this are partly the aesthetic implications, as well as the possibility that imposing a visual barrier between the park's inhabitants and the outside world might lead to an increase in crime. The third major issue that was raised by participants was the effect that their own involvement had on their perception of the soundscape. In the group discussion which followed the sound walk, almost all participants agreed that the experience had increased their appreciation of the soundscape as a result of taking the time to actively listen to the sounds around them. Many also noted their surprise at the variety of sounds. This particular observation reinforces the argument that better management of environmental noise should include aiding and encouraging engagement with the soundscape. This could come from public art installations, informative displays, soundscape events, and creative forms of noise intervention such as the sonic crystals demonstrated here.

A possible weakness in the results could be attributed to the small sample size used in the sound walks. There is a degree of uncertainty regarding the smallest acceptable sample size in qualitative research, although it generally depends on the methodology and form of analysis one intends to carry out on the data. Bertaux [423] argues fifteen is sufficient in qualitative research, whereas Morse suggests as few as six are sufficient for a phenomenological study [424]. One might argue that sections of the questionnaire where differential scaling has been used make this a psychometric test, and as such the sample size is perhaps too small to derive any statistically significant results. However, one could also argue that the written comments and post-walk discussion are more reminiscent of a phenomenological study, where the goal is to identify phenomena through the study of experience from the perspective of an individual [425]. Furthermore, pure phenomenological research seeks to describe rather than explain, and to start from a perspective free from hypotheses or preconceptions [426]. As far as the phenomenological aspect of the survey is concerned, the sample size was likely to be adequate, and so it is this aspect one is inclined to emphasise in the analysis of the results - particularly given the inherent ambiguity of semantic differentials. Another factor to be aware of when considering the sound walk methodology, is the fact that, in contrast to most other qualitative research methods, a lot of modern phenomenological studies emphasise the role of the researcher in the 'frame' of the research, not only as the interpreter, but as an interested and active participant - as opposed to a detached and impartial observer [425]. In other words, acknowledging the researcher has a certain influence in the outcome of the study does not necessarily devalue it, but it does need to be made as transparent as possible.

7.3 Design

In this section, the process of designing the sonic crystal noise barrier is detailed. This includes the investigation of a higher order sonic crystal structure in FDTD simulations. Based on the results of these experiments and the site survey, it is hoped that the final design will complement the space for which it is intended both acoustically and aesthetically.

7.3.1 Location

Through careful consideration of results from both the noise mapping and sound walk elements of the site survey, it was decided that the area of Woodhouse Moor which stands to benefit most from the sonic crystal installation is location 5. A lack of either strong positive or negative reactions by respondents in this location mean it is considered the most neutral, and in this respect it seems to be the most wasted. The fact that it has little in the way of visually distinguishing features may also have been a contributing factor. Not only is it a fairly large area, but it serves no obvious purpose besides being located at an intersection between several pedestrian routes. While location 4 was perhaps a more obvious candidate in terms of the noise levels (the results of the noise mapping survey indicated levels in excess of 70 dB in the 31.5 Hz region, which is above those recommended by the WHO [98]), the area has a distinctive character and its soundscape was not considered by participants to be out of keeping with the activities that take place there. On the other hand, it is felt that location 5 could be enhanced both aurally and visually to create a different yet equally distinct environment.

7.3.2 Layout

The hypothetical sonic crystal noise barrier is based on a honeycomb structure and consists of a series of interconnected hexagonal cells (see Figure 7.16). The barrier is designed to be placed in parallel with a primary noise source (ie. a road), thus achieving both visual and aural abstraction of the source without obscuring the view in and out of the structure completely. Each cell will be large enough for groups of visitors to sit in, the intention being to encourage interaction with the sculpture by imparting different characteristics on the soundscape in each of the cells. For example, the outside facing cell walls could each be 'tuned' differently, and the interior landscaping might also vary between cells. It is considered that coating the barrier with a luminous paint might improve the safety of the area at night, as well as being a welcome addition at the various festivals and parties that take place in the park during the summer months.

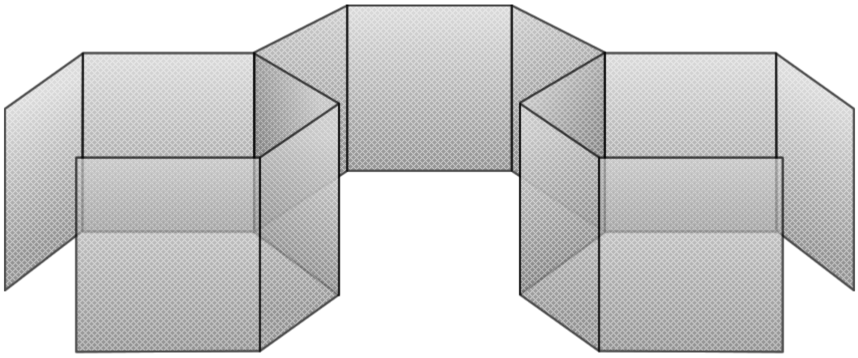
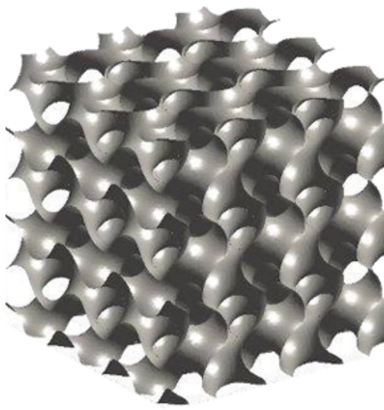


Figure 7.16: 'The Comb' - a hypothetical sonic sculpture consisting of interconnected hexagonal cells separated by gyroid structures.

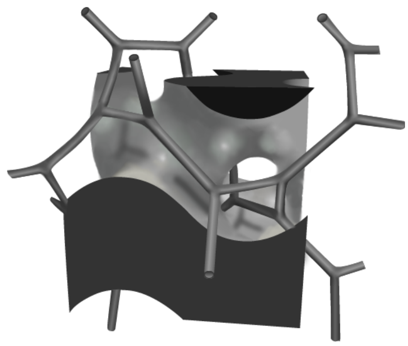
7.3.3 Composition

The intention is to evaluate two types of sonic crystal structure in the following auralisations in order to determine which is more suitable, or indeed, whether there are any discernible differences between them. The first is of the conventional cylindrical rod variety that was enountered in the previous chapter, while the second is a more experimental form of periodic structure known as a 'gyroid'.

Discovered by Schoen in 1970 [427], the gyroid is an infinitely connected triply periodic minimal surface containing no straight lines. Essentially, it consists of a single surface folded in such a way that the mean curvature at the interface between solid and pore space is zero. The pore space is composed of distinct channels, which interlock to form a labyrinthine structure that is very homogeneous in terms of channel diameter variations. The name 'gyroid' alludes to the spiralling or 'gyrating' movement that results when traversing a given channel (Figure 7.17).



(a)



(b) Close

Figure 7.17: The gyroid - a triply periodic minimal surface (a). Folds in the surface form a network of routes, creating a labyrinthine structure (b).

The gyroid is thought to occur in nature, having recently been discovered on the wings of several species of insect [428]. In such cases it is believed that the influence of the structural geometry on the physical interactions of incident light produces some extreme filtering effects at visible wavelengths - to which the creature's vivid colouration may be attributed. Gyroids have also been investigated on the nano scale in the field of metamaterials [429], that is, artificial materials with unusual magnetic responses such as negative permeability [430]. It is thus speculated here that, on the sonic scale, these structures may exhibit similarly interesting filtering effects when applied to acoustic sources.

The transfer functions of both varieties of sonic crystal were modelled in acoustic simulations using an FDTD scheme identical to that which was described in Chapter 6. While meshing the 2-D cylindrical array is relatively trivial, mapping a triply periodic minimal surface is slightly more complex. In computational geometry it is standard practice to use the level set method [431] for mapping continuous surfaces, wherein three-dimensional shapes are built up from layers of level surfaces represented by functions $F : R^3 \rightarrow R$ of points $(x, y, z) \in R^3$, which satisfy the equation $F_{(x,y,z)} = t$, where t is a constant [432]. A value of $t = 0$ defines the boundary of the shape (referred to as the zero level set of F), while the interior of the shape is the set of points on each level surface for which F is positive. Thus, the surface of a gyroid may be defined by Schoen's gyroid level set equation:

$$F_{(x,y,z)} = \sin(x)\cos(y) + \sin(y)\cos(z) + \cos(x)\sin(z) = A \tag{7.1}$$

where the parameter A determines the volume fraction of the gyroid labyrinth by the relation $\frac{A}{\varphi}$ in which φ is the golden ratio (approximately 1.61). A theoretically perfect gyroid in which $A = 0$ implies the surface has a mean curvature of zero. In the discretised gyroid of our simulations, exact solutions to (7.1) do not occur, hence one looks for transitions between positive and negative curvature when mapping the surfaces.

Within a fixed volume, the simulation is reiterated for gyroids of different compositions. Using an additional parameter referred to here as the step-size, it is possible to vary the density of the channels (essentially by shifting the frequency of the output in (7.1). Hence the function used to generate the gyroid dataset becomes

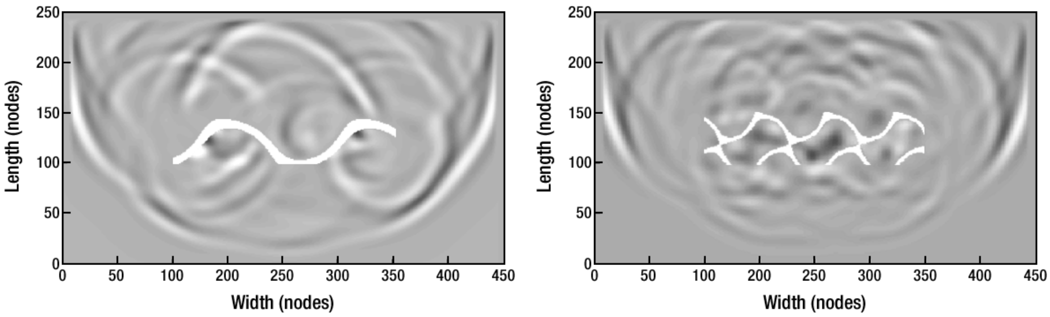
$$F_{(x,y,z)} = \sin(g \cdot x)\cos(g \cdot y) + \sin(g \cdot y)\cos(g \cdot z) + \cos(g \cdot x)\sin(g \cdot z) \tag{7.2}$$

where g is the step-size. If one considers x, y and z are linear input functions, then step-size can also be thought of as the gradient of these input functions (ie. their rate of change). A small step-size results in a low frequency output (ie. fewer channels), whereas a large step-size has the opposite effect. By association, assuming the sampling rate is fixed, then a large step-size implies a lower surface resolution; hence it is necessary to increase the resolution of the mesh for higher step sizes to maintain surface fidelity. Here it was found a step-size > 0.2 necessitated a mesh resolution that was beyond the capabilities of the single CPU.

In order to place the gyroid in the simulation domain, we need to determine which elements in the dataset are surface nodes and which are air nodes. We do this by assigning each element in the mesh an absorption coefficient α, the value of which is determined by the gyroid function (7.2) such that

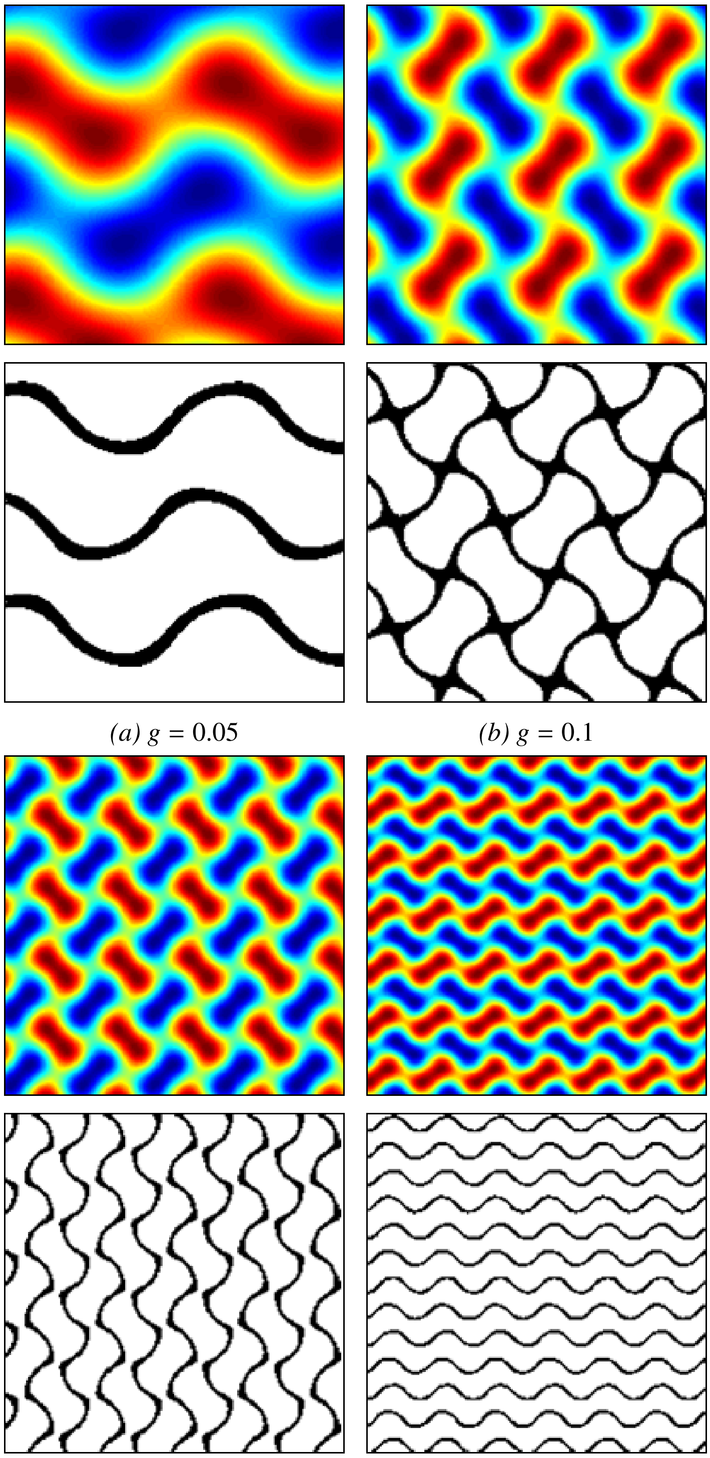
$$\alpha_{(x,y,z)} = 1 \text{ if } |F|_{(x,y,z)} > A, \text{ and } 0 \text{ if } |F|_{(x,y,z)} \leq A \tag{7.3}$$

The aim is to keep the volume fraction consistent by fixing the value of A at 0.25. In each case, this yields a volume fraction of approximately 0.16, which can be verified by dividing the number of surface elements by the total number of elements making up the gyroid.



(a) $g = 0.05$ *(b)* $g = 0.1$

Figure 7.18: Horizontal slices through simulation domain at $T = 1000$.



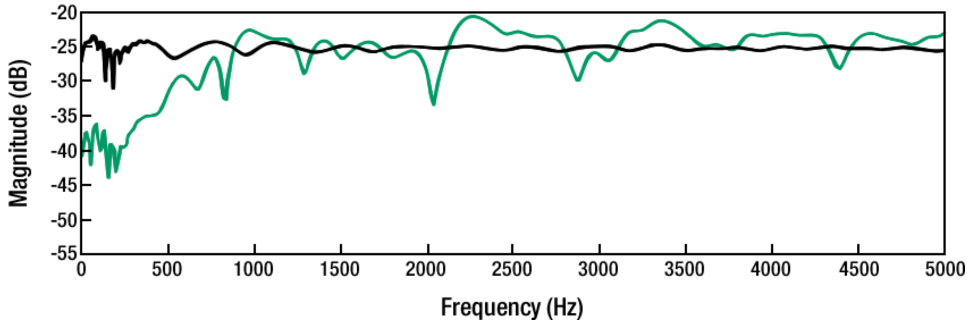
(a) g = 0.05

(b) g = 0.1

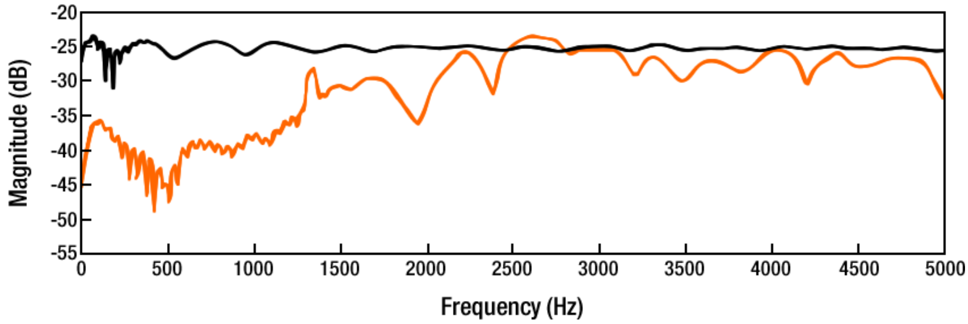
(c) g = 0.15

(d) g = 0.2

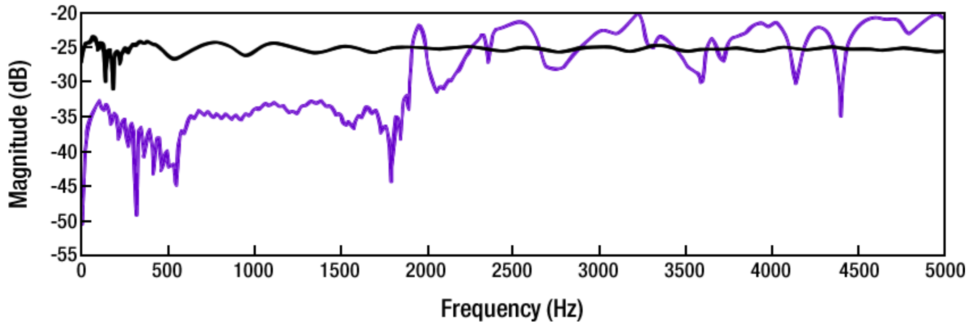
Figure 7.19: Vertical cross-sections through gyroids of different 'step-size' (g) generated by the continuous function (7.2), and the corresponding slices through the simulation domain after meshing the surface. Each slice consists of 200 x 200 nodes. In the top set of figures, values in (7.2) range from approximately 1.4 to -1.4 with zero crossings mapped to green, positive values (indicating positive curvature) mapped to red, and negative values (indicating negative curvature) mapped to blue.



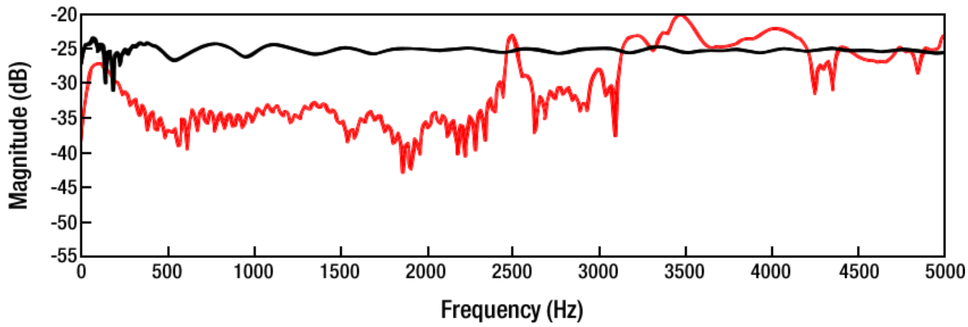
(a) $g = 0.05$



(b) $g = 0.1$



(c) $g = 0.15$



(d) $g = 0.2$

Figure 7.20: Frequency response curves obtained in FDTD simulations of gyroid samples of the same size but with different width channels. The channel width is dictated by the step size, g. A higher value of g indicates the structure has more convolutions per unit volume (see Figure 7.19). The depth of each sample is 25cm.

Discussion of Results

The frequency response curves generated by the gyroid structure indicate that there is a strong attenuation of frequencies where the wavelength is large relative to the channel opening. This produces an effect similar to that of a high-pass filter, whereby the smaller the channel opening, the higher the cut-off frequency of the filter. However, the strong attenuation of low frequencies is thought to be exaggerated in the simulation due to the lack of consideration for structurally borne sound. Above the cut-off frequency, there are some irregularly spaced peaks and gaps in the transmitted signal, though these do not appear to follow any obvious pattern. While it is thought that further work is necessary to clarify the acoustic filtering effects of the gyroid sonic crystal, a thorough comprehension is not considered critical to the objective behind the following experiments. This would therefore be the focus of an independent study.

7.4 Auralisation

This section deals with the auralisation of the sonic crystal in a soundscape composed of multiple sound field recordings made in parts of Woodhouse Moor. Having already obtained the filters that will be applied to elements of the soundscape composition, the following details the procedure by which the soundscape composition is generated and processed in order to render the auralisation.

There have been various approaches to soundscape composition, from the stochastic methods developed by Xenakis [433], to more recent experiments with genetic algorithms [434]. Such methods typically require a large database of sound files and a level of computation that exceeds the requirements of this case study. Hence the approach taken here was to take genuine B-format field recordings in the actual soundscape and to produce an arrangement from the recorded material that is considered to be representative of the real soundscape.

B-format sound recordings were made using a Soundfield microphone and a portable 4-track recorder. Three recording locations were used: one close to the primary source of noise (i.e. the road), one at the site of the barrier (location 5), and a third in location 3 where the primary source was less dominant. This is due to the assumption that sound incident on the barrier consists mainly of the primary source (both being at ground level), whereas the sound captured by the microphone consists of sound from all angles and elevations - thus implying a degree of separation is needed at the filtering stage. Since it is not possible to record the source in isolation, it was therefore desirable to obtain material with varying amounts of the primary source in order to produce a well balanced mix of filtered noise and ambient sound.

The B-format recordings were subsequently auditioned and annotated according to their content. This involved making records of all the unique and intermittent sound events that were captured, as well as any noticeable variation in the continuous background ambience, which were considered to be the flow of traffic on Woodhouse Lane, birdsong and wind in the trees. Individually distinct vehicle pass-bys were classed as unique sounds. An arrangement was then built up of these elements which would give a realistic reflection of the soundscape in the proposed site at both quiet and busy periods. The original sound files used in the arrangement are given on the DVD accompanying the thesis (Directory 9.7.3.3).

One of the issues faced when producing the arrangement was how to convey the diversity of the soundscape within a relatively short time frame. Whereas participants in the real sound walks spent over an hour in the environment and therefore had much more opportunity to observe its multiplicity, participants in the virtual sound walk will have only a few minutes in which to form their impressions. Restricting the length of the virtual sound walk to a few minutes was deemed necessary, both for logistical reasons related to the test methodology, and to prevent the onset of fatigue amongst participants. In a pilot demonstration of the composition which took place in an auralisation workshop held in the University of Derby [435], it was noted that, in an artificial environment lacking visual stimuli, listeners become bored and develop listening fatigue much more quickly than in a real soundscape. Essentially one needs to compress the soundscape in time, without compromising its authenticity. The arrangement therefore tries to create a balanced impression by including busier periods when a lot of sound events are taking place, either side of a less active period in the middle. It is hoped that structuring the composition in this way will not only provide balance, but will also convey a sense of movement, reminiscent of the transitions that occur in a real soundscape during its daily life cycle. The score presented in Figure 7.21 illustrates the overall structure of the composition.

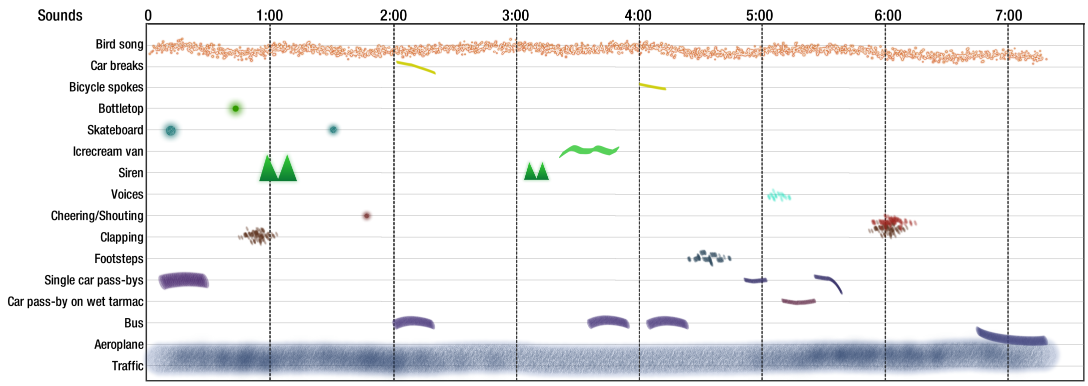


Figure 7.21: A graphical score illustrating the overall structure of the soundscape composition.

The arrangement is assembled in the Steinberg Nuendo DAW, and three separate B-format mixes are exported separately. The first consists of all tracks together (this will form the control mix); the second consists of the tracks recorded close to the primary source (these will undergo further processing in Matlab); and the third consists of the tracks recorded furthest away from the primary source (this ‘ambient mix’ will later be recombined with the source mix). In accordance with the procedure presented in Chapter 6, Figure 6.31, the source mix is copied so that one copy is processed with the filter derived from Maekawa’s barrier attenuation equation, while the second is filtered with the inverse of this filter. The former is subsequently convolved with the sonic crystal filter, before being recombined with both the residual of the source mix, and the ambient mix. It is performed three times: once with the impulse response recorded in the gyroid simulations;

once with the response from the cylindrical array; and finally with the response from an empty mesh. This last version will be compared with the untreated soundscape composition to assess the audibility of any artefacts which would lower the credibility of the other auralisations.

Satisfied that the auralisations of the soundscape with the sonic crystal insertion are as authentic as possible, the next stage in the case-study involves the subjective evaluation of the virtual soundscape before and after the insertion.

7.5 Subjective Analysis

The subjective evaluation consisted of a series of listening tests, the objectives of which were two-fold: first, to answer certain questions relating specifically to the use of sonic crystals as environmental noise interventions; and second, to comment on the methodology and the suitability of the approach in relation to the design and perceptual evaluation of urban soundscapes. The outcome of the latter objective is highly dependent on the extent to which the former is achieved. If the analysis yields results which are persuasive, whether in favour or against the application of sonic crystal noise barriers in urban environments, it will be considered a positive indication that the methodology has worth. If, on the other hand, the results are inconclusive, it may suggest that parts of the process need refining, or that they are inappropriate to the application.

The primary objectives can be summarised as follows:

1. To ascertain whether or not there is a perceivable difference between soundscapes rendered with and without the presence of a sonic crystal.

2. If objective 1 is true, is the difference independent of artefacts in the simulation?

3. If objectives 1 and 2 are true, in what way are these differences manifested and is the overall effect considered favourable by untrained listeners? Is it possible to tell?

Responses to these questions shall then form the basis for comment on the suitability of sonic crystals as a form of environmental noise intervention.

Secondary objectives related to the test methodology are summarised as follows:

1. To comment on the success of the test methodology by comparison of the results with those attained in the real-world soundscape survey.

2. To comment on the absence of visual stimuli on perception of the soundscape amongst adults with normal hearing and vision.

3. To comment on the potential this method has both in relation to environmental noise and urban design problems, and in valuing soundscape perceptions.

Two styles of subjective listening test were performed. The first was a laboratory style perceivable difference test, while the second was a less tightly controlled phenomenological type survey, closely modelled on the sound walking methodology, but this time performed in a virtual acoustic environment. The aim of the first group of tests is simply to verify that filtering portions of the audio material with the sonic crystal impulse response produces an audible effect, while the second group aims to establish whether or not the effect is actually significant in terms of the perceived sound quality of the environment when encountered in more natural listening conditions.

7.5.1 Perceivable Difference Test

The first group of listening tests was devised to give a statistically meaningful indication of whether or not there are perceivable differences between the auralised soundscape material in which the presence of the sonic crystals has been emulated, and the untreated soundscape recordings. It was also intended to confirm that any apparent differences are not in fact artefacts of the FDTD simulation. It was not, however, for the purpose of making any inference as to which of the versions presented is preferred.

The test took the form of a category judgement test in which the participants were asked to compare pairs of sounds and rate their respective difference on a category scale according to a specified criterion. In this experiment it was considered appropriate to use experienced listeners in a controlled, laboratory-type assessment. The test was therefore taken by 13 normal hearing participants with general expertise in audio processing and critical listening.

The test was conducted in two parts, each associated with a different difference criterion. The criteria that were used were ‘loudness’ and ‘timbre’, and the category scale was a 5 point Rohrmann scale with the categories ‘not at all’, ‘slightly’, ‘moderately’, ‘very’ and ‘extremely’ [436]. Questions were grouped according to the difference criterion, so that all questions rated according to one criterion were presented sequentially, followed by all those rated according to the next. It was felt that switching back and forth between difference criterion might confuse the participant’s internal ‘differential scale’ and therefore bias the results.

It is readily conceded that ‘timbre’ is not an obvious choice of difference criterion and therefore warrants some explanation. Timbre has been referred to as ‘the psychoacoustician’s multidimensional waste-basket category for everything that cannot be labelled pitch or loudness’ [437]. However, this so-called ‘waste-basket’ was exactly what was needed here - a rather ambiguous criterion that encompasses everything related to the texture or spectral character of a sound yet is not related to loudness. The reason why this ambiguity was desirable is because the sonic crystal filter effectively involves applying a spectral envelope to the signal, and the psychoacoustic effect of that is, as yet, an unknown quantity. In other words, it is not yet possible to tell how this change in spectral content might be perceived - whether as a change in tonal content, colouration, attack or so on - and one therefore wants to cover all bases. Furthermore, as the material in question is not of a musical nature, it could be difficult for those used to using such subjective parameters in a musical context to ascribe them to a non-musical sound. However, a drawback with using timbre as a descriptive term, particularly in the presence of participants with a high level of expertise in psychoacoustics, is that it does require some disambiguation to ensure it is interpreted correctly. A clear, contextual definition was therefore included in the test instructions (see Appendix 9.4).

Quality control

Using a category scale can incur bias if participants are inconsistent with their answers - an effect which can be exacerbated when successive pairs are opposite extremes. In order to identify participants who did not give consistent answers, the test was structured so that each phase in the test would be preceded by a training phase - an approach which has the added benefit of permitting a period of listener 'calibration'. If it was found that answers given in a training phase did not correlate well with the answers given in the relative test phase, the answers to questions in that phase would then be excluded from the results. The threshold for elimination was determined based on the average variance in listener response after all the data had been collated, so that if any given individual was found to have an above average level of variance for any given phase, their answers were deemed unreliable. Furthermore, if a participant failed a test phase pertaining to one criterion but passed the other, only answers from the phase in which they had failed would need to be excluded from the results. Although the training phases effectively meant that each pair was presented twice, they were intended for screening purposes only and not included in the final results.

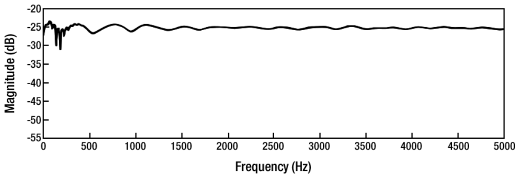
A second form of 'quality control' involved the inclusion of anchors in the guise of identical pairs. Participants who consistently failed to identify the anchors as 'not at all' different were also be excluded from the results.

Training

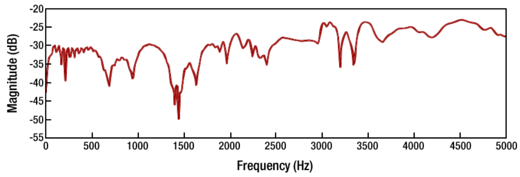
As all the participants were deemed to be experienced listeners, and given the relative simplicity of the test, a minimal amount of training was given. Clearly worded and identical written instructions were presented to each participant before taking the test (see Appendix 9.4). Since the participants were not made aware of the training phases, and were therefore under the impression that the test consisted of 64 questions in 2 parts, it was considered appropriate to include a number of 'example' questions at the start of each half of the test. It was hoped these example questions - which also served as a preview of the test interface - would relax the participant and fully prepare them for what was to follow. Answers to these questions were not submitted to the database for analysis.

Audio material

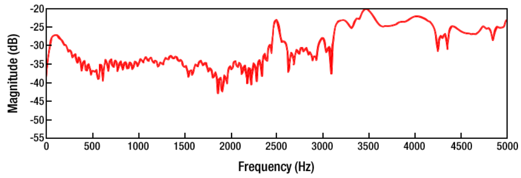
The audio material consisted of 3-4 second extracts of the soundscape composition that would later be used in the virtual soundscape experiment presented in 7.5.2. To make the test slightly more interesting for participants, two different extracts were used for each difference criterion - one consisting of the background noise, and another with some tonal content. From each extract a total of 4 unique sound files were produced, each of which was assigned a letter (A-D) denoting the filter that was used. Filter A was the impulse response recorded in an empty mesh; B was the



(a) Filter A: Empty mesh



(b) Filter B: 2-D Cylindrical Array

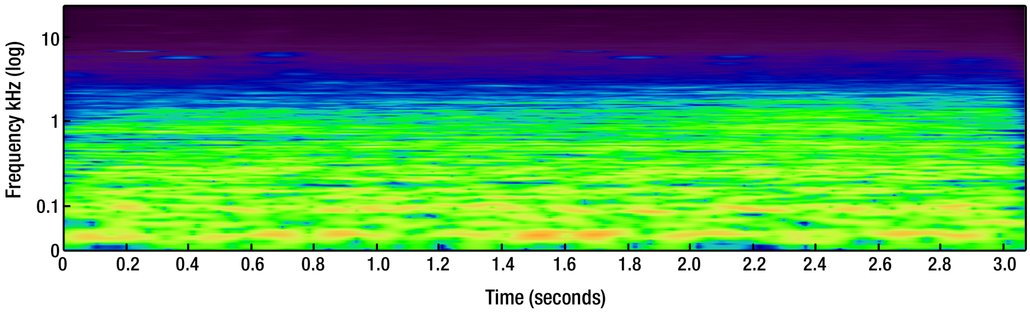


(c) Filter C: Gyroid

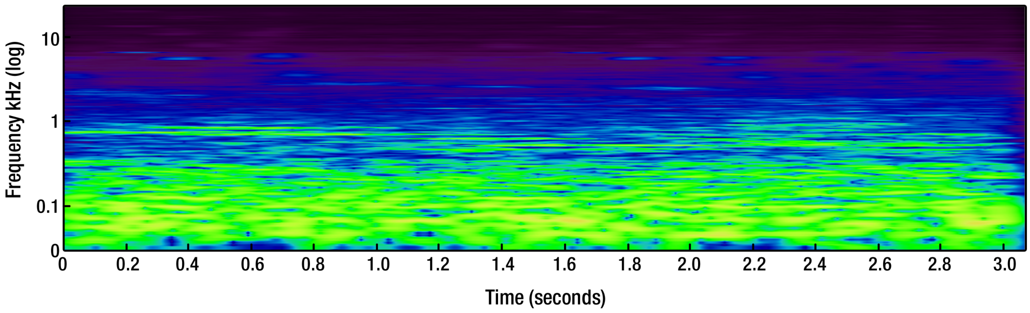
Figure 7.22: Frequency response curves of the impulse responses that were used to filter the soundscape extracts.

2-D cylindrical array type sonic crystal with a row depth of 6; C was a gyroid; and D had no filter applied at all. The frequency responses of the filters are pictured in Figure 7.22.

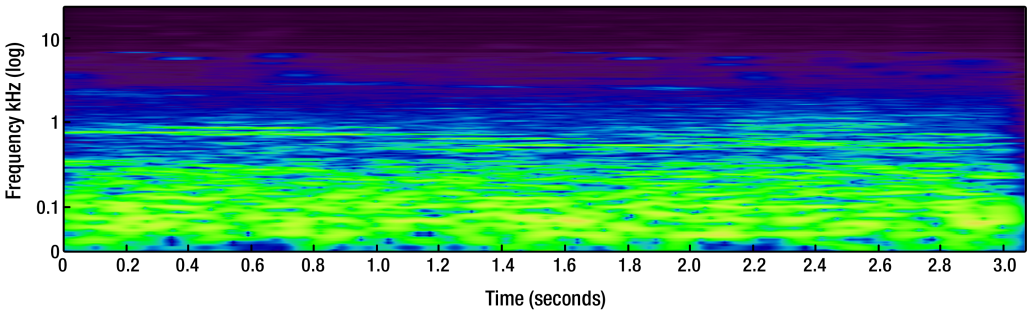
Figure 7.23 shows the spectrograms of the first extract after application of the different filters. A complete set of spectrograms are given in Appendix 9.3. Provided the artefacts introduced by the simulation and filtering processes are minimal, the differences between the control and the extract filtered with the impulse response recorded in the empty mesh should barely be noticeable. Note that the differences between the versions with the sonic crystal insertions and those without are more marked for the first extract than for the second. Moreover, there is little visible difference between the extracts filtered with the 2-D sonic crystal impulse response and those filtered with the gyroid impulse response. It is thus the intention to determine whether these observations are also reflected in the results of the listening test.



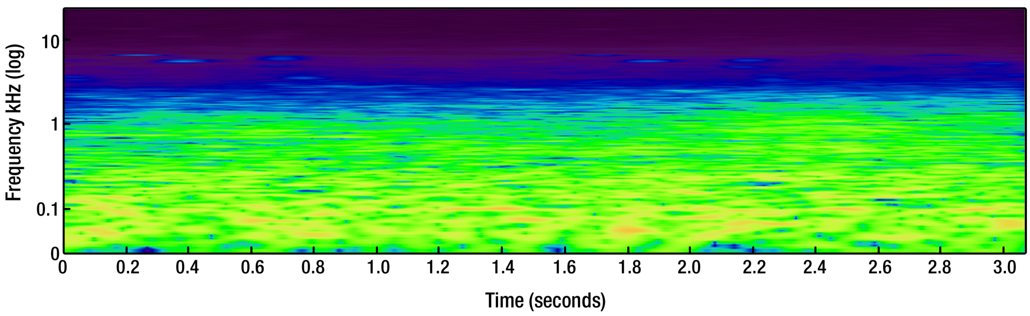
(a) A (Empty)



(b) B (2-D Cylindrical Array)



(c) C (Gyroid)



(d) D (Control)

Figure 7.23: Spectrograms of the first audio extract used in the perceivable difference tests after the application of different impulse response filters. The complete set of spectrograms is given in Appendix 9.3 and the corresponding audio files are included on the DVD accompanying this thesis (Directory 9.7.3.4).

A disadvantage with this test method is that the number of pairs for comparison increases exponentially with each additional difference criterion or audio sample, rapidly leading to very lengthy tests. It has been recommended in [438] that any listening test duration should not exceed 20 minutes to prevent the onset of listener fatigue. It was therefore decided that, rather than spanning the test over multiple sessions which could be problematic in terms of logistics and the rise in uncertainty, a limit should be imposed on the number of pairs presented. Given that each audio extract lasts between 3 and 4 seconds, it was felt that 20 seconds would be an ample amount of time to allow for each pair. This allowed for up to 60 pairs to be compared within the 20 minute test period. Using a total of 2 audio extracts to produce the test material and 2 difference criteria, plus the 5 practice questions presented in each section of the test, the total number of pairs is 50. Allowing a further 5-10 minutes for reading, the total length of the test was 25-30 minutes. A breakdown of the test structure is presented in Table 7.1.

Table 7.1: Perceivable difference test structure

Phase	Extract	Criterion	No. Pairs	Duration (minutes)
Instruction	N/A	N/A	N/A	5-10
Training 1	1 & 2	Loudness	5	2-3
Phase 1	1 & 2	Loudness	20	6-8
Training 2	3 & 4	Timbre	5	2-3
Phase 2	3 & 4	Timbre	20	6-8

Test environment

The tests were held in an empty teaching room. A photograph of the test set-up is presented in Figure 7.24. The test interface was presented in a web page displayed on a desktop computer. It allowed participants control over the order of playback and number of repetitions of each relative pair of sounds but they were not able to control the order in which the pairs were presented. The sequence in which the pairs were presented was randomised for each participant to distribute any bias in the data that might be linked to presentation order. Participants were allowed to complete the test in their own time, although they were advised beforehand of a reasonable length of time to spend on each question. Answers were submitted to a SQL database so that they could be accessed remotely by the researcher from another workstation. Screenshots of the test interface are given in Appendix 9.4.

Results and Discussion

From Figure 7.25, it can be seen that the combined results of the perceivable difference test for all extracts and criteria tested show good correspondence with the predictions. As anticipated, there is a slight perceivable difference between the control and the auralisation with the empty mesh impulse response which is attributed to artefacts in the simulation. However, the fact that this difference is smaller than the differences perceived between the 2-D array and the gyroid is taken as a positive indication that the unique transfer functions of the two structures lead to unique and audible colouration of the signals, and that this is primarily responsible for the differences

Figure 7.24: The perceivable difference listening test set-up.

perceived. It was also noted that the difference perceived between identical samples was higher than anticipated, which suggests all of the ratings may be slightly exaggerated.

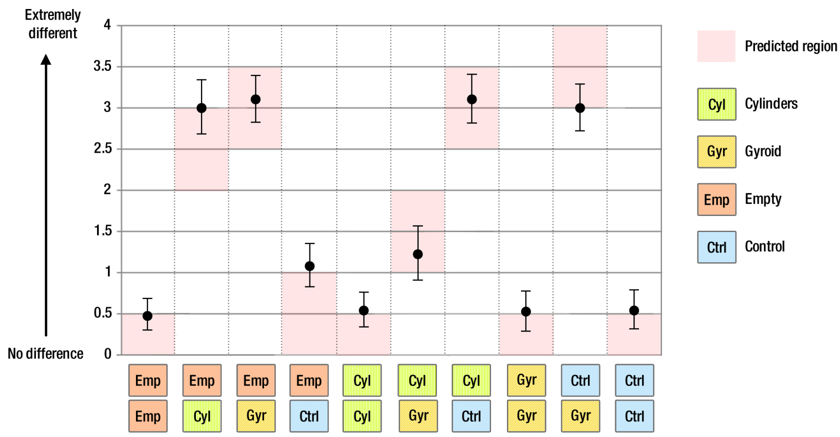


Figure 7.25: Results of paired perceivable difference test. *Ctrl* indicates the unprocessed soundscape material, *Emp* indicates the material was filtered with impulse response recorded in an empty mesh, and *Cyl* and *Gyr* indicate the material filtered with the cylindrical array and gyroid impulse responses respectively.

Figure 7.26 compares the averaged perceived differences between the different audio extracts when assessed according to different criteria. The differences due to the filtering were more noticeable in extracts 1 and 4 which may have been related to their predominantly broadband spectral content (extracts 2 and 3 each had discernible high frequency sound events). There were two possible anomalies in the results: for extract 3 larger than anticipated differences were perceived between the empty and the control when compared against timbre; and for extract 4, the differences perceived between the gyroid and the cylinder when compared against timbre were unusually high. These apparent inconsistencies may be related to the ambiguity surrounding the timbre criterion, or there may be another explanation which would require further investigation to expose.

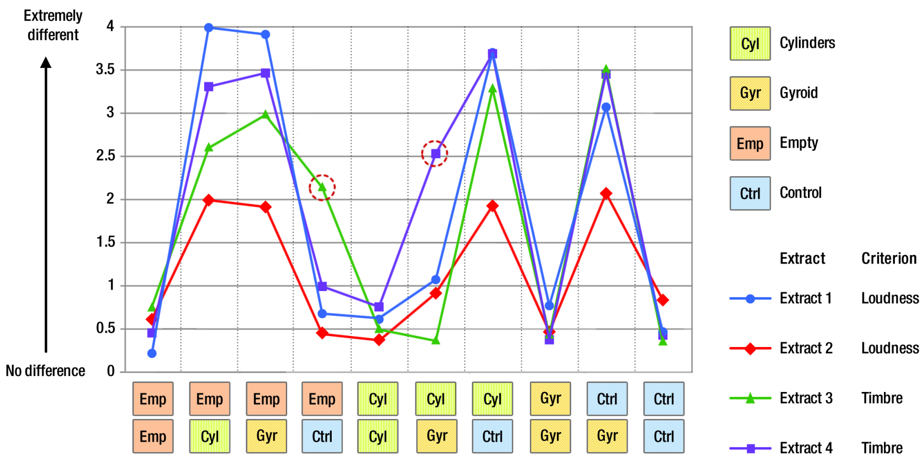


Figure 7.26: Comparison of average perceived differences using different audio extracts and difference criterion. The data points circled in red indicate possible outliers.

While the results of the perceivable difference test have indicated there are quite significant perceivable differences between the auralisations with the sonic crystals and the unprocessed audio, the statement only applies unequivocally to audio that is evaluated under laboratory conditions and where the duration of a sample does not exceed that of the average auditory short-term memory. The results of these tests cannot give any real indication as to whether these differences are significant under more natural listening conditions where the listener is not consciously seeking to compare one environment against another in rapid succession. To try to ascertain whether or not the differences translate to perceived sound quality of the soundscape, a different experimental approach is subsequently taken.

7.5.2 Phenomenological Test

In this second group of tests, a condensed version of the original sound walk questionnaire was used, in which the psychometric tests were retained, along with the space for free comments. As in the original sound walk questionnaire, participants were explicitly asked to list all sources of sound they could pick out, including individual words or phrases. Since there is no actual 'walking' involved, the questionnaire need only concern the location of interest - hence the duration of the test is much shorter, comprising of around 7 minutes of listening material and a further 10 minutes or so for answering questions.

The listening material was presented over a 3$^{\text{rd}}$-order ambisonic array, whereby the higher-order signals were extrapolated from the 1$^{\text{st}}$-order signals by the Harpex [439] ambisonic decoder plug-in in Reaper [440]. The array consisted of a total of 16 loudspeakers distributed in a 4-by-8-by-4 circular configuration (see Figure 7.27). Using a higher-order ambisonic set up extends the sweet spot, thus enabling several listeners to be tested at once. The questionnaire data consists of two sets: A group of respondents (A) who were played the material processed with the empty mesh impulse response (i.e. no barrier present), and second group (B) who were played the material assimilating the 2-D sonic crystal barrier. The spectrograms of the audio material are presented in Figure 7.28 and stereo versions of the sound files are available on the DVD accompanying this thesis (Directory 9.7.3.5). The decision was made to exclude a third group of auralisations in which the gyroid barrier was assimilated on the basis that the material was only perceived to be slightly different from that of the 2-D sonic crystal barrier in the perceivable difference tests. If the results of the first two groups pointed to a significant change in the perceived sound quality, then a third set of tests to incorporate the gyroid would be performed at another time.

Figure 7.27: The virtual sound walk listening test set-up showing a subject seated in the centre of a 4-by-8-by-4 loudspeaker array. The height of the seat is adjusted so that their ears are approximately level with the centre of the array. Acoustic curtains surround the array and absorptive materials are installed on the ceiling and floor of the studio to reduce room reflections.

Some of the questions presented to participants in the virtual soundscape questionnaire were different from those that were presented in the original sound walk questionnaire. This is partially related to the fact that information pertaining to the nature and whereabouts of the soundscape location had been purposefully occluded from the participants. The reason for making the soundscape anonymous in these tests was to try to gain some insight into how expectation and learned association affects the way people value the sound around them. As before, the questions were divided over two sections: a section of pre-sound walk questions, and a section to be answered during and/or after presentation of the material.

In the pre-sound walk questions, participants were asked to provide some general information about themselves, such as their age group, nationality, sex and profession. They were also asked whether they had any involvement with acoustics or urban planning in their profession, and if so to give details. They were then asked about their general impressions of urban soundscapes in the UK, and their current attitudes towards environmental sound. If further clarification was needed on a question, the researcher would elaborate, or make suggestions on how to approach the question.

The second half of the questionnaire included the differential scales used in the real sound walks, as well as several fairly broadly worded questions designed to encourage spontaneous observations, without being too suggestive. To ease them into the process, the first of these questions asked the participant to list all the sounds they were able to hear, and where they were unable to identify the source of the sound, to attempt to describe it in their own terms. The intention behind this question was to allow some speculation as to whether there was any change in clarity or definition due to the sonic crystal. For example, if participants tended to observe more different sounds with the sonic crystal present, it might suggest the masking effect of the traffic noise had been reduced.

The remaining questions focused on the participants' impressions of the soundscape, how they imagined it might look, what they might do there, and finally their feelings about the virtual soundscape experience, including any technical observations. A copy of the questionnaire is given in Appendix 9.5.

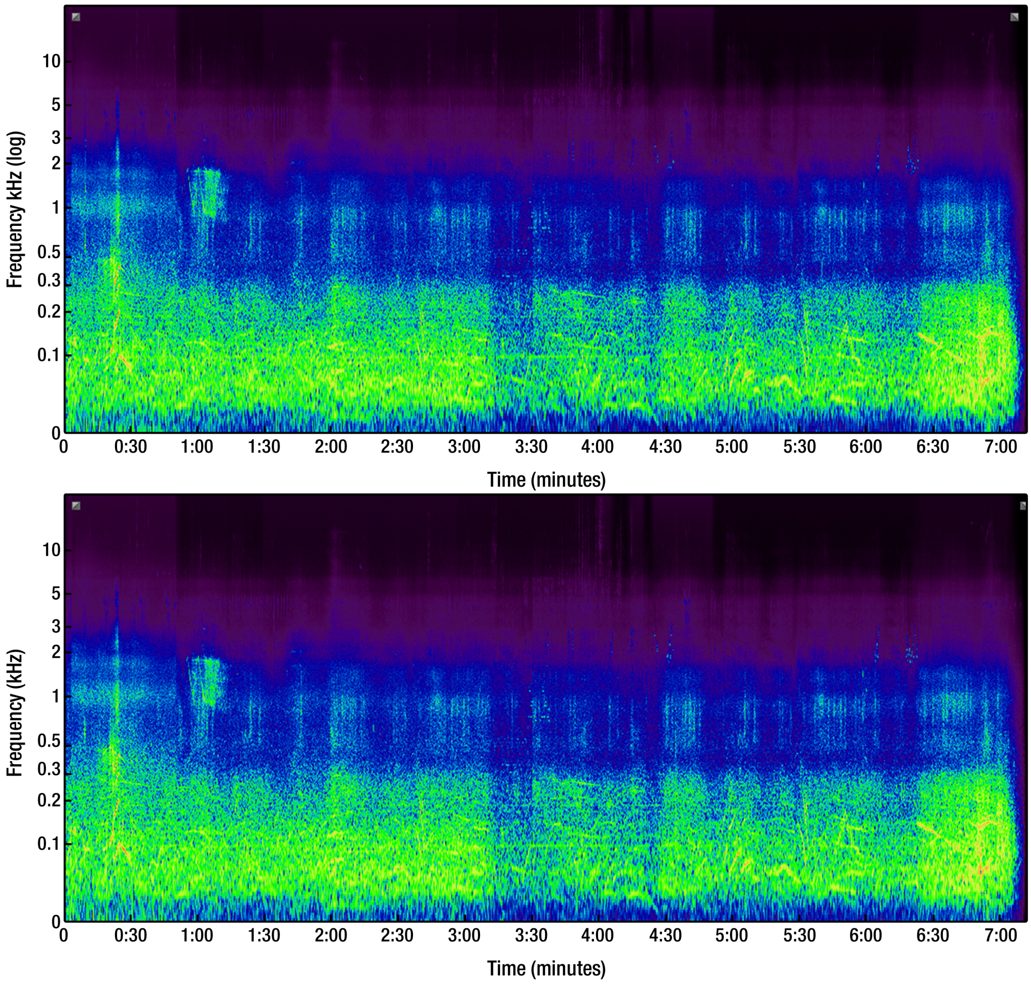
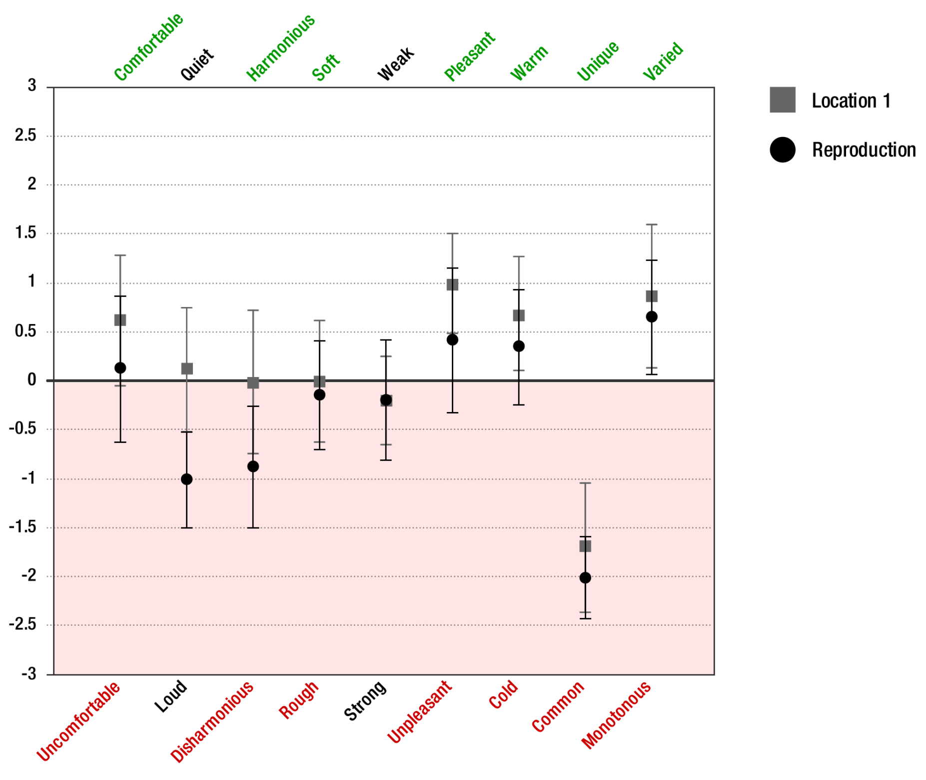


Figure 7.28: Time-frequency spectrum of the soundscape composition before *(a)* and after *(b)* the auralisation of the sonic crystal barrier. On this temporal scale, it is difficult to identify any visible differences in the spectrograms. The differences are much more easily observed in the very short extracts used in the perceivable difference tests (see Figure 7.23).

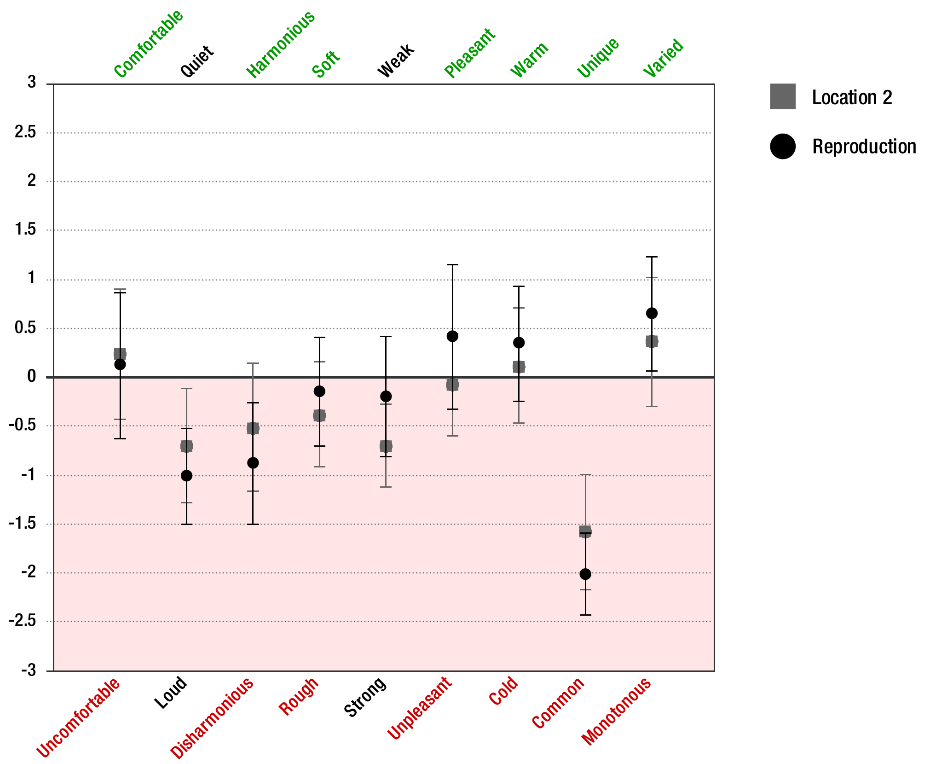
7.5.3 Results and Analysis

The results from the virtual soundscape listening experiments are analysed in several phases. In the first phase of the analysis, the results of the semantic differential test from the real sound walk are compared with those of the virtual sound walk without the presence of the sonic crystal and analysed for significant difference. This set of significance tests also took the form of the Mann-Whitney U-test [421]. Locations 1, 2 and 5 are expected to be most similar to the virtual tests on the basis that they contained very similar elements. Location 3 is expected to be quite different because the visual aesthetics and slightly lower noise levels in this location seemed to contribute to it being perceived more positively in the real sound walks, despite the fact that it was structurally similar to locations 1, 2 and 5. Location 4 is also expected to be different due to the uniqueness of the area and the fact that none of the material used in the compositions was recorded there.

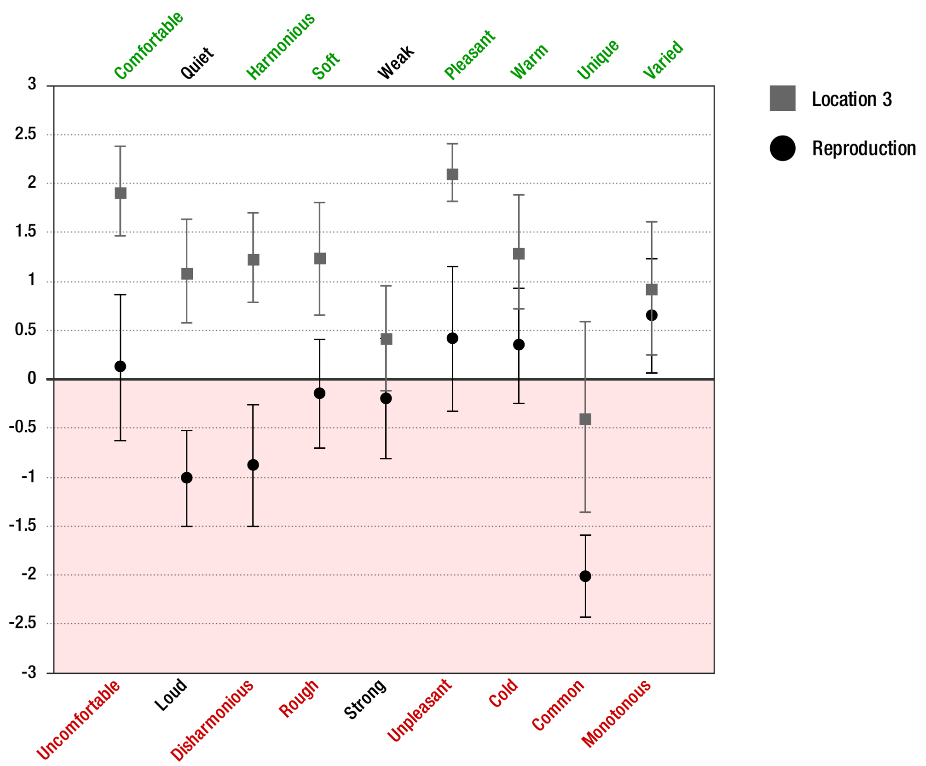
Figure 7.29: Rated sound quality in 5 areas of Woodhouse Moor versus results from virtual sound walk (VSW) with no sonic crystal insertion.



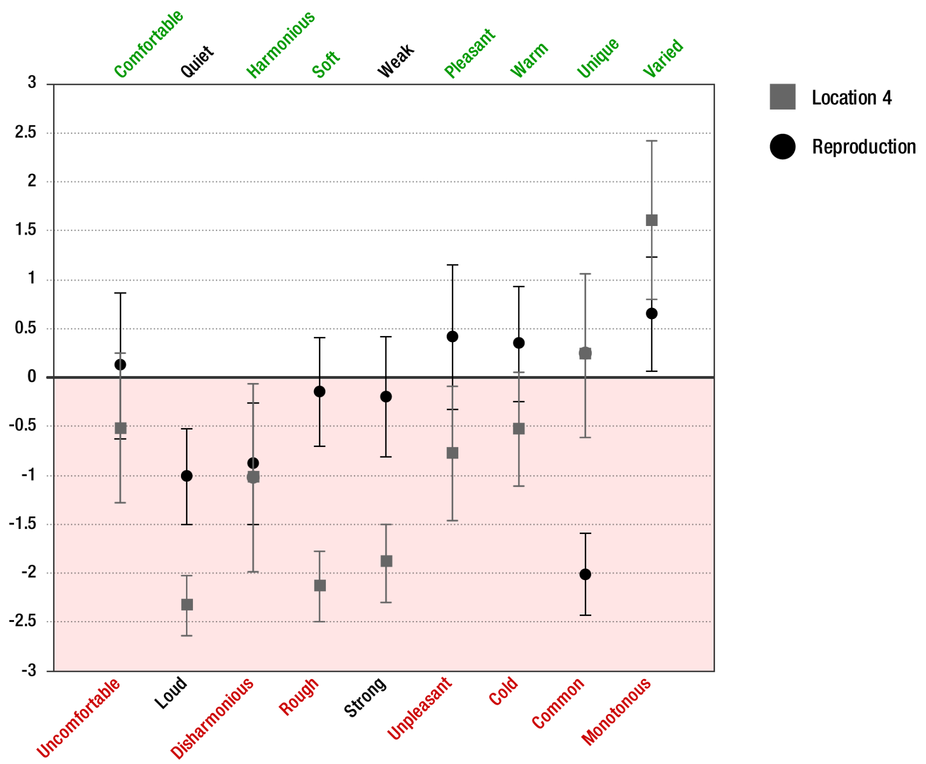
(a) VSW compared with location 1



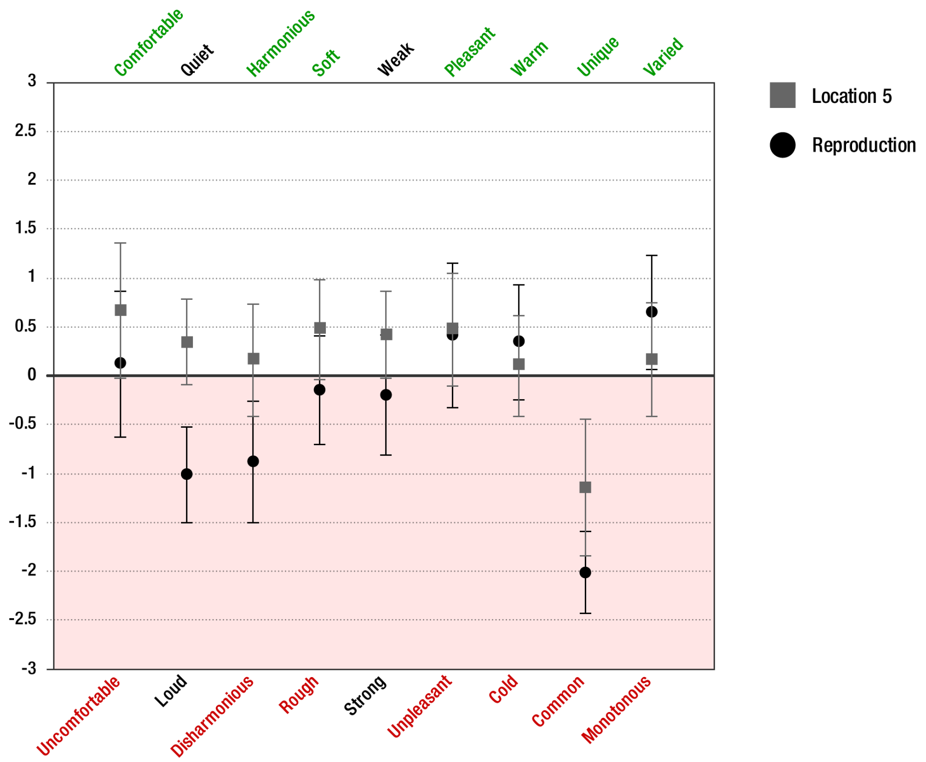
(b) VSW compared with location 2



(c) VSW compared with location 3



(d) VSW compared with location 4



(e) VSW compared with location 5

In Figure 7.30, the results of U-tests between different real locations (real versus real) and real with virtual (real versus virtual) are presented for comparison. The relationship between the virtual soundscape and location 2 is of particular interest as this location appears most similar to the virtual soundscape based on the box plots. Whether or not the null hypothesis for significant difference is rejected should be the same for both forms of test if one is to conclude with reasonable confidence that the auditory experience of a listener in a spatially reproduced soundscape is close to that which they would have in the real soundscape.

Location	Comfort/discomfort	Quiet/loud	Harmonious/disharmonious	Soft/rough	Weak/strong	Pleasant/unpleasant	Warm/cold	Unique/common	Varied/monotonous
VSW vs 2	H0	H0	H1	H0	H0	H0	H0	H0	H0
2 vs 3	H1	H1	H1	H1	H1	H1	H0	H0	H0
VSW vs 3	H1	H1	H1	H1	H0	H1	H0	H0	H0
2 vs 4	H0	H1	H0	H1	H1	H0	H0	H1	H1
VSW vs 4	H0	H1	H0	H1	H1	H0	H0	H1	H0

Figure 7.30: Comparison of the results from the significance tests between locations on the real sound walk, with the real versus the virtual sound walk (VSW). H0 indicates the null hypothesis - i.e. there is no significant differences between the data sets, while H1 indicates the data sets are significantly different. All are expressed with 90% confidence interval. These results suggest that subjects perceived the soundscape on the VSW and location 2 of the real sound walk similarly.

Results of the U-test between the virtual soundscape and location 2 of the real sound walk indicate that with one exception the null hypothesis that neither set was significantly greater than or less than the other could not be rejected. The single exception was for the harmonious-disharmonious descriptor. These results suggest that, in this case, listeners did have a qualitatively similar auditory experience in the virtual soundscape as they would in the real one, despite the absence of visual cues or preconceptions about the space. Whether or not the same can be expected of other soundscapes remains to be investigated. When asked to describe how they imagined the space to look, most of the participants had a fairly accurate impression, having experienced similar environments in the past. In fact, it could be that the generic nature of this soundscape contributed to the success of its reproduction.

Also interesting was that a number of participants mentioned hearing water, or initially mistaking the traffic for water. This also resonated with the real sound walk, as a few participants had then commented that from a distance they could imagine the road traffic to be water. It supports the notion that disassociation - whether due to acousmatic listening or a conscious effort - might lead to an enhanced qualitative experience. If this is the case, then while the sonic crystal barrier may not have had a significant effect on the sound that was heard, its visual impact may well have done. This theory is supported by a number of past studies into the effect of visual stimuli on auditory experience (e.g. [215, 441, 442]).

In the next phase of the analysis, U-tests were performed between the virtual soundscape data obtained before and after auralisation of the sonic crystal (see Figure 7.31. The results of these tests appear to indicate that there are no significant differences in the perceived sound quality of either environment under realistic listening conditions. This is an interesting result, as it is known from the paired comparisons that there is in fact an audible difference between the two soundtracks when a subject compares two short extracts in close succession. Yet it would appear that these differences do not constitute a significant improvement or degradation of the overall quality of the sound environment. However, given the small sample size that was used in this comparison (15 in the untreated group and 12 in the treated group), it is possible that different results may be obtained for a larger sample size. Verbal and written feedback seemed to suggest the treated soundscape was perceived slightly more positively, although there did not seem to be any difference in the quantity of sounds that were detected. There were fewer comments about the loudness of the treated soundscape, and this particular soundscape metric was just short of significance in the U-test. It is possible that with a larger sample size, the difference may be confirmed. Nevertheless, the results are a good indicator that subtle changes in the spectral content of a complex urban soundscape have a negligible impact on the perceived sound quality.

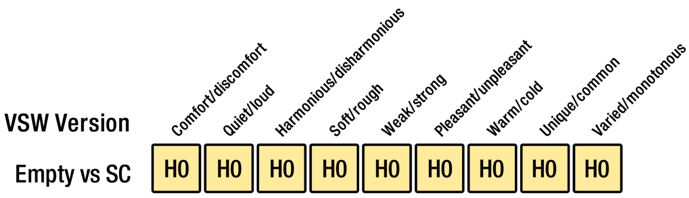


Figure 7.31: Significant difference tests between data sets recorded in the virtual soundscape experiments before and after inclusion of the sonic crystal barrier.

The variety and frequency of different sounds perceived by the participants in the virtual sound walk are presented in Figure 7.32. In both groups the number of unique sounds identified by each participant averaged around 8, suggesting the sonic crystal neither heightened nor reduced the masking effect of the traffic sound. However, the figures given in Table 7.2 indicate there was a difference in the range of sounds identified by each group collectively. For example the control group jointly identified a total of 18 different sounds, whereas the sonic crystal group jointly identified a total of 26 different sounds - an increase of approximately 80% after correcting for the difference in the group sizes. This would appear to suggest that, while the listeners in the sonic crystal group could hear the same variety of sounds as the control group, they were less certain about the causes of the sounds - in other words, there may have been a rise in perceptual ambiguity as a result of the sonic crystal. It can also be noted that the figures are very similar to those of the real sound walking experiment, despite the virtual sound walks being significantly shorter in duration. This might suggest an increased aural awareness due to the absence of visual stimuli, or it may simply be an indicator that the composition successfully reflected the diversity of the soundscape in Woodhouse Moor.

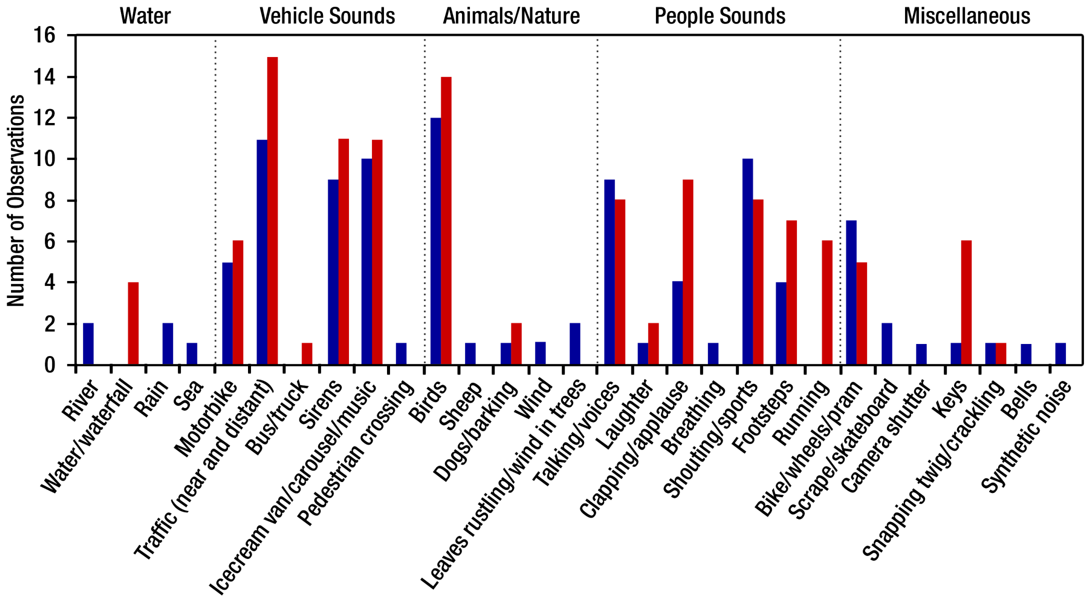


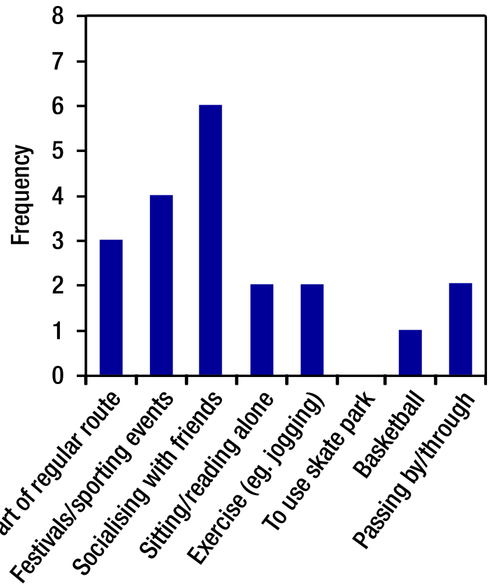
Figure 7.32: Sounds identified in virtual soundscape experiments before and after inclusion of the sonic crystal barrier. The red bars correspond to the group played the soundtrack with no sonic crystal, while the blue bars correspond to the group played the soundtrack with the sonic crystal included.

Table 7.2: Comparison of unique sound events identified in virtual sound walks before and after insertion of a sonic crystal noise barrier. Results of real sound walk are also given for comparison.

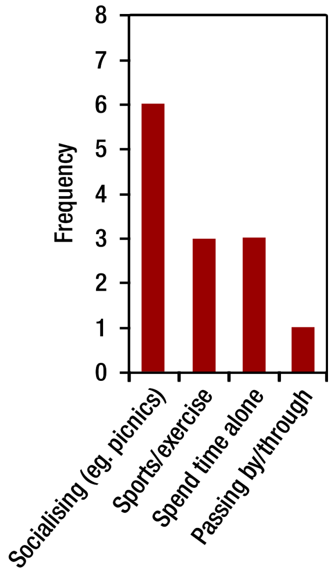
Group	Group Size	Total Number of Sounds Identified	Number of Unique Sounds	Average Number of Sounds Per Participant
Control	15	117	18	8
Sonic crystal	12	101	26	8
Real sound walk	16	128	28	8

Attitudes regarding the activities subjects imagined they might partake in while visiting Woodhouse Moor were somewhat varied, yet showed some correspondence with those recorded by participants on the real sound walk. Figure 7.33 shows the activities participants on the real soundwalk associated with visits to Woodhouse Moor, compared with those that participants on the virtual sound walk pictured themselves taking part in. In both instances it appears the area was most favoured as a place for socialising, and was less popular for relaxation. Bearing in mind that, in the majority of cases, participants in the virtual sound walk experiments were not aware of the location, the similarity between these sets of data is taken as further evidence that the virtual soundscape successfully captured the ambience of the space.

Figure 7.33: Activities participants on the virtual sound walk imagined they might do in Woodhouse Moor compared with those participants on the real sound walk reported to engage in.



(a) Activities associated with Woodhouse Moor (real sound walk)



(b) Activities subjects in the virtual sound walk associated with the location

Verbal and Written Feedback

Summaries of both the verbal feedback recorded by the researcher in the interviews, and the written feedback recorded by participants in the questionnaires are presented in Appendix 9.6. Copies of the questionnaire papers are also included on the DVD accompanying this thesis (Directory 9.7.4.2). Generally speaking, subjects' impressions of urban soundscapes in the UK ranged from mostly quite positive, to ambivalent, where the ambivalence tended to be surrounding traffic and other man-made sources of sound. While strong negative responses were infrequent, they were almost always connected to road traffic sound. It also transpired that negative feelings were often aroused when the subject's expectations regarding the soundscape of a particular area were not met. For example, the sound of road traffic and people talking were generally acceptable in an urban setting, but not in a rural location. There did not seem to be any obvious correlation between a participant's country of origin and their impression of UK soundscapes, other than a couple of comments by Chinese participants who considered UK cities to be quite quiet by comparison with Chinese cities. Although the questionnaire did not ask the participants to comment on the soundscapes they had grown up in, the issue sometimes arose in discussion. It seemed that subjects who had at some point experienced a dramatic change in their acoustic landscapes, whether from rural to urban or vice versa, were able to adapt after a period of time. The ability people have to adapt to their surroundings, along with the lack of any clear relationship between address history and present attitudes, seemed to suggest that reactions to environmental sound may be more closely linked to personality and personal experiences than they are to culture. However, a larger scale investigation might have revealed cultural trends that were not picked up on here due to the small sample size. Furthermore, all of the participants who took part in the real and virtual sound walks were aged between 18 and 56, which may have resulted in a certain amount of bias given the most sensitive age groups are considered to be young children and the elderly [99]. However, what was more important with respect to the hypothesis, was that the subjects who took part in the virtual sound walk were representative of those who took part in the real sound walk. As both groups consisted of a similar range of ages, cultural and socio-economic groups, it was considered that they were.

Some, though not all, participants were also asked about their musical preferences and it was interesting to note the variety in their responses. One participant who stood out as having a particularly strong aversion to traffic and industrial noise stated their preference for classical and folk music, with a strong dislike for jazz and rap music, whereas other participants who were not disturbed by noise stated a preference for electronic music and hip-hop. While their responses were not considered evidence of any link between musical taste and reactions to noise, they do seem to support the idea that, like musical tastes, reactions to noise and environmental sound are very personal to an individual.

What was also quite remarkable about the virtual sound walks was the level of detail and consistency with which some participants were able to describe the visual aspects of the landscape. While this was not true of all the participants - for example, some found visualisation very difficult - most participants described a large green area close to a main road and a short distance from a city centre. Although several perceived the sound of running water or the sea, there was otherwise quite close agreement between their descriptions. As well as aiding visualisation, the apparent familiarity participants felt with the soundscape may have led some to perceive sounds that were not in fact present. However, this is just a theory, as it is not possible to distinguish here between 'phantom sounds' and instances where a subject has guessed incorrectly the source of a sound.

While comments relating to the technical aspects of the composition and virtual soundscape methodology were generally favourable, there were a few negative issues that arose. Two of the participants mentioned the feeling of being in a tunnel at some stage and that there were cars overhead. There were also some issues over image stability, as some participants experienced strange phase effects when moving their head around. While broadband noise is particularly susceptible to phasing issues when reproduced over multi-speaker systems, it is possible that both of these problems may also be linked to the Harpex decoder's attempts at source separation, which were never intended for dealing with complex soundscapes. However, the Harpex was used on all of the virtual soundscapes that were presented, thus its limitations are not considered to have impacted on the outcome of the experiment. Furthermore, as all ambisonics decoders suffer from similar issues, consistency was the primary concern here.

A number of subjects commented that they felt they were static and that the soundscape moved around them rather than the other way around. While it was not actually the intention of the author to imply movement from place to place, their reference to a 'sound walk' in the questionnaire was perhaps unfortunate as it implied movement to the participants. Thus what were intended to be temporal transitions in the soundscape were invariably interpreted as spatial transitions - i.e. the listener felt they were transported to another place. Two of the subjects commented that these transitions were quite abrupt, which suggests there may have been room for further improvement on the structuring of the piece and wording of the questionnaire. The length of the composition and test were generally considered appropriate.

7.6 Summary

In this chapter a case-study was presented which attempted to demonstrate the application of auralisation and the virtual sound walk test methodology in relation to a real-world environmental noise problem. In the example, the perceived sound quality of an urban soundscape was evaluated over a third order ambisonic array, before and after the insertion of a hypothetical sonic crystal noise barrier. The faithfulness of the auralisation and suitability of the test methodology is supported by comparison of the results before insertion of the sonic crystal with results obtained on a real sound walk.

To ensure the validity of the virtual sound walk experiments, several versions of the audio material were also tested for audible difference beforehand. The results have indicated that, while there is an audible difference between the sound files when evaluated in a paired comparison test, the differences are not manifested as a significant change in the perceived sound quality of the soundscape when evaluated under more natural listening conditions. There was also some evidence to suggest that the presence of the sonic crystal may have had an obfuscating influence on aural perception, there being less agreement amongst subjects in the sonic crystal group as to the causes of some sounds. While confirmation of this theory requires further investigation, any supposed obfuscation does not appear to have influenced the overall perceived sound quality.

This apparent tolerance of perceived sound quality to audible changes in the soundscape may relate to the fact that subjective responses to environmental sounds are, on the whole, very varied. This variation makes the detection of statistically significant patterns in the data difficult. It may also relate to a subject's ability to adapt to different acoustical conditions - both in the short-term and also in the long-term. For example, while some participants found the initial sound levels very loud, they generally found they 'got used to it', and sometimes attributed their heightened sensitivities to listening to an outdoor soundscape in an indoor space. The fact that humans find it so easy to adapt and reconcile such incongruities, whether on a psychoacoustic or an intellectual level, is thought to support the idea that expectation and personal preference are among the most significant contributors to the perception of sound quality in urban environments. In fact, under some circumstances - and within certain boundaries - they may have as much if not more relevance on perceived sound quality than quantifiable characteristics such as sound pressure level and spectral content.

While there were some sources of weakness identified in the test methodology, the results of the experiments are nonetheless a positive indication that auralisation can be used successfully to evaluate the impact of new strategies in environmental noise control and urban design, as well as offering a degree of insight into the multiplicity of attitudes and reactions to sound in urban spaces. For example, a key strength of the technique is that it allows a subject to evaluate the sound environment in relative isolation, with less interference from other stimuli and personal precepts. It also facilitates a more natural auditory experience than is possible using laboratory-style listening tests.

8

Conclusions

8.1 Thesis Summary

In this thesis the author has explored the application of auralisation and soundscape research methodologies in relation to a real-world environmental noise problem. The premise behind the investigation was that auralisation can be used to generate reliable qualitative feedback regarding the acoustics of outdoor spaces, which could be useful to designers, urban planners, and anyone else with an interest in understanding how humans interact with their sound environments.

The author began by outlining those fundamental acoustic principles that are pertinent to the modelling of environmental sound. Chapter 3 described how these principles are implemented in practice for the purpose of monitoring and controlling undesirable sound, also known as *environmental noise*. Particular attention was given to current legislation regarding environmental noise and knowledge with respect to its adverse effects.

In Chapter 4 the subject of environmental sound was considered from an holistic and exploratory perspective, as opposed to the more restrained and quantitative approaches encountered in the previous chapter. The intention was to draw together evidence from the fields of soundscape research, environmental psychology and philosophy in order to justify the need for integrated and sustainable approaches to environmental noise control, of which auralisation and subjective analysis might form an integral part.

In anticipation of the experimental work in which such an approach would be demonstrated, Chapter 5 focused on the key theoretical principles and techniques associated with auralisation. This chapter also highlighted the challenges that lie ahead in terms of modelling outdoor spaces and with rendering three-dimensional sound fields over multi-speaker arrays.

In Chapter 6 the process of developing and refining an auralisation process that will permit the subjective evaluation of a specific form of noise treatment known as a *sonic crystal noise barrier* was considered. While the simulations performed in this chapter focused on the relatively simple example of a 2-D sonic crystal array, the intention was to devise an auralisation scheme that could later be used in relation to different and novel forms of sonic crystal noise barrier.

Finally, in Chapter 7, the ideas and methods developed in the preceding chapters were brought together and evaluated in a case study-based assessment. In the case study a combination of quantitative and qualitative techniques have been used to evaluate the impact of a sonic crystal noise barrier on the perceived sound quality in a real urban green space. The example of the sonic crystal was considered appropriate to the objectives behind the thesis as it not only facilitated the application of auralisation to an acoustic design problem, but it is also a rare case of a noise treatment which, as an object of some curiosity, has the potential to stimulate deeper consideration for the sound environment by those who encounter it.

In the case study, the sonic crystal barrier was modelled in Finite Difference Time Domain (FDTD) simulations and the system's impulse response was subsequently used to filter a soundscape composition. The composition consisted of authentic B-format field recordings made in the case study location. Measures were taken to ensure the effects of barrier edge diffraction were fully accounted for, and that an authentic balance of filtered and ambient sound was evident in the final composition. The resulting auralisations were subsequently evaluated in subjective listening tests conducted in a 3rd-order ambisonic array. The tests took the form of a *virtual sound walk* - a novel approach to qualitative assessment closely modelled on the popular sound walking methodology used in soundscape research.

To validate the virtual sound walking methodology, the results of the virtual sound walk *without* the barrier insertion were compared with results obtained from real sound walks carried out in the location under observation. Satisfied that both sets of results were in close agreement, the results from the virtual sound walk *with* the barrier insertion were then contrasted with results obtained using the unprocessed version of the composition.

8.2 Restatement of Hypothesis

The case study carried out in the thesis was conceived with the intention of proving or disproving the following hypothesis:

> *The application of auralisation to environmental sound can be an effective aid in the design, evaluation and promotion of positive urban soundscapes*

The experimental work presented in the thesis has succeeded in demonstrating a useful application of auralisation relating to the design and implementation of a sonic crystal noise barrier. In the thesis, it was shown using auralisation that, under normal listening conditions, the device has no significant impact on the perceived sound quality in an urban soundscape. This result was dependent on a second finding, namely, that both the real and the virtual sound walks conducted in Woodhouse Moor produced results which were were statistically similar. These two findings make auralisation a realistic proposition in socio-acoustic research and urban soundscape design, and are viewed as strong evidence in favour of the above overlying hypothesis.

8.3 Contributions to the Field

In the thesis, the research that has been described with a view to proving the hypothesis has resulted in the following contributions to the field:

1. **The application of established auralisation and qualitative soundscape research methods to a real-world environmental sound design problem.**
   Environmental noise control and soundscape studies are two areas of acoustics that, until recently, have seen relatively little cross-over. While the multi-disciplinary approach to understanding reactions to sound in the environment has started to attract more attention in recent years, there are still few examples where this research has been practically implemented in noise control and urban design strategies.

2. **The development of the *virtual sound walk* test methodology.**
   Analysis of the results of the virtual sound walks have indicated that, not only is the approach suitable for generating realistic subjective feedback about a proposed noise treatment, but it also allows for deeper speculation concerning human-environment interactions. This goes back to the elementary principle that, in devising an effective solution to a problem, the problem should first be understood as far as is possible in the context in which it occurs. In other words, the more that can be learnt about human relationships with the sound environment, the better we will become at managing both the noise itself and our reactions to it.

   The success of the *virtual sound walk* listening test framework was considered to be very high, with subjects reporting similar experiences to those who were involved in the real sound walks (see Appendix 9.2.1 and 9.6.1). In both of these scenarios, it transpired that attentive listening led to an overall positive perception of the soundscape, in spite of certain negative elements, such as the road traffic. The author uses the term *attentive* here as it implies two things: firstly, that the listener is giving the sound their full attention; and secondly, that they are doing so in a way that is careful and considerate. It is careful in the sense that the object of their attention is something outside of him or her -self. However, the converse of this particular observation is that it may be difficult to judge the likelihood of annoyance from tests involving attentive listening. This makes the sound walking methodology unsuitable for assessing the likelihood of noise complaints. However, the success of the virtual sound walk experiment suggests that, with appropriate adjustments to the test format, auralisation might yet be used for this purpose - for example, by enabling researchers to measure the level of disruption to a particular task or activity due to environmental sound under controlled test conditions.

3. **The application of the virtual sound walk methodology to the 'in-situ' assessment of a specific form of noise treatment.**
   Normally the performance of a noise barrier is based exclusively on relative sound pressure

levels before and after its insertion. For example, one would first predict or measure the frequency dependent attenuation afforded by a particular type of barrier. Then, based on the source-barrier-receiver geometry and source characteristics, a further prediction would be made with regard to the rise or fall in each octave or third octave band due to the barrier. The drawback to this approach is that it does not afford a reliable indication of what effect the treatment will have on the perception of sound in a particular environment. The sound environment is a complex system, and it is difficult to tell how changing one aspect will impact on another aspect, without actually involving humans in the process. While it would be not be feasible to attempt such an extensive investigation with respect to all outdoor noise treatments, in high-cost projects of a large scale, such as the regeneration of an urban space, investing more time in subjective testing in the early stages may ultimately save money and lead to better acoustic design.

4. **The modelling and auralisation of a sonic crystal noise barrier.**
Prior to the thesis, the auralisation of a sonic crystal had not yet been attempted. To this end, the Finite Difference Time Domain (FDTD) numerical modelling method was used for its simplicity and flexibility. In the process of devising a suitable FDTD scheme, several different source types were evaluated according to their respective signal-to-noise ratios (SNRs). This was a novel approach, and one that is thought to have proved successful, having helped the author to identify the optimum driving function in the simulations. It was felt that the results of the simulations showed sufficient correlation with the theoretical predictions to satisfy the requirements of subsequent experiments, despite there being scope for further development in this area.

5. **The proffering of novel insights into the perception of urban soundscapes.**
A number of observations have been made with regards to the perception of urban soundscapes and attitudes to environmental sound. For example, while it has been shown that auralisation can assist in the design of innovative forms of noise treatment in a very practical sense, it is also suggested that designing or redesigning the soundscape does not necessarily have to be realised externally - it can occur internally. The act of designing can be defined as the ability to conceive or to fashion in the mind. It is thus posited here that, in some cases, it may be possible to promote positive soundscapes without actually changing anything about the physical environment, other than the way humans listen to and interact with it. That said, it is also felt that some spaces may be made more conducive to creative listening through well-considered urban design strategies, and, where necessary, effective forms of noise control. For example, when conducting the real sound walks in Woodhouse Moor, differences in the perceived sound quality were observed in locations 1, 2 and 5, despite there being little to distinguish them from one another acoustically. The differences are thus thought to be attributable to other aspects of the space, such as its aesthetic qualities, and the way that it is used and treated by those who visit it. On the other hand, when the soundscape was abstracted from other sensory stimuli and from its social context through auralisation, its

modification by the inclusion of the sonic crystal barrier appears to have had little effect on people's perception of it.

It is also suggested that the over-emphasis of the negative aspects of environmental sound, particularly in relation to man-made sources of sound, may impact negatively on soundscape perceptions. This relates to the proposition that creating positive soundscapes is not necessarily about making physical changes to the sound environment, but is also about improving human-environment relationships. In this sense, drawing too much attention to undesirable aspects of the sound environment may serve to reinforce negative reactions, rather than ameliorate them. Furthermore, how one relates to the environment has a lot to do with how one perceives oneself and others within a much broader context. This was something that came to be realised through the listening surveys that were performed in the thesis, and is something that the author feels could be explored further, both in real and in virtual listening environments.

8.4 Personal Reflections

Here the author reflects on several observations that were made over the course of this thesis which, while not directly connected to the experimental work that was performed, are nevertheless salient with respect to the wider issue of environmental noise. For example, an important point was made regarding the way the problem of environmental noise is represented in the literature, namely, the over-dramatisation of its negative effects. For instance, The World Health Organisation has attempted to quantify the 'burden of disease' attributable to noise in terms of Disability Adjusted Life Years (DALY). They have estimated that 'at least one million healthy life years are lost every year from traffic related noise' [443]. Such statistics are of course readily seized upon by journalists and noise campaigners who present them to a lay audience as if they are unequivocal, despite the statistical methods used to derive them being imperfect at best and, at worst, fundamentally flawed (e.g. [444, 445]). Furthermore, the appropriateness of DALY in relation to noise is perhaps questionable given the almost impossible task of isolating noise from a myriad of other environmental and sociological factors.

The author is not attempting to argue that there are insufficient grounds for concern over noise, only that the way this information is presented to the public and used to influence government policy should be treated with due care and clarity. It seems that much of the current legislation and recommendations are based on a relatively small amount of empirical evidence, and that there may be a danger in exaggerating the harmful effects of environmental noise if the limits of our understanding are not made transparent. The apparent pessimism may be partly related to the fact that noise is an emotive issue, and consequently there can be some difficulty in maintaining a composed and unbiased perspective regarding its adverse effects. This is compounded by the fact that noise is generally regarded as a pollutant, which automatically places it in a similar mental category as more serious chemical pollutants. For example, the authors of a popular article entitled *Noise: A Modern Plague* [1] which was published by the Southern Medical Journal do not shy away from the use of evocative language in their writing, even going so far as to describe noise as an air pollutant 'analogous to second-hand smoke'. They place particular emphasis on the idea that 'people have the right to choose the nature of their acoustical environment', and that 'it should not be imposed by others'. What is also worrying is that it tends to be these rather biased and emotively charged articles which have penetrated the popular media. While the intention here is not to deny the prevalence of negative reactions to noise, or that links to sleep disturbance are a serious matter and should be given due attention, it is being suggested that some of the approaches taken by a lot of 'noise activists' should be questioned. Noise is an unavoidable by-product of society, and belonging to a society means having to tolerate one another, and accept that one may not find everything that comes with it quite to one's satisfaction, all of the time. By drawing too much attention to noise, there is the possibility that we may be inadvertently feeding the problem, by increasing the likelihood of annoyance and encouraging hostility toward benign sources of sound and its perpetrators. What the author would like to see instead is a more balanced and honest assessment of the problem, and one which also highlights the positive aspects of environmental sound.

After all, noise - if one is forced to call it that - actually has the potential to be a very positive aspect of our lives, and this is a claim which the evidence presented in this thesis supports. For example, not one of the participants in either the real or virtual sound walks reported feeling persistently bothered by noise in their day-to-day life, and the vast majority found the experience of attentive listening pleasurable and relaxing. Furthermore, listening attentively to environmental sound as a means of relaxation and spiritual contemplation is by no means a new phenomenon, and has been an exercise in spiritual and therapeutic practices for many centuries. In this context, a distinction need not be made between natural and man-made sounds, as all sound has equal status and represents an opportunity to connect to the environment while taking attention away from whatever negative patterns of thought may be occurring inside the head. While there are undeniably grounds for concern over excessive amounts of environmental noise, it is also important to keep a sense of perspective. Stress is the real epidemic, and this of course has countless other causes, with the overall contribution from noise likely to be small in the majority of cases. By over-emphasising its contribution, we risk damaging relationships further and losing many of the positive aspects sound holds for both individuals and society, provided one is receptive to them.

It was also observed that there is a strong tendency in the soundscape literature for researchers to separate natural and man-made sounds, placing a negative emphasis on the latter. In this model, humanity becomes the pollutant and the despoiler of all that is wholesome and pure. While various studies undoubtedly reveal a preference for non-human sounds, it should also be acknowledged that this may be caused by learned attitudes and social context. This research suggests that, while there are times when an individual may seek an exclusively 'natural' soundscape in order to temporarily disengage with the pressures and constraints of cultural life, there are also times when the hustle and bustle of urban environments can be a comfort. It is the author's opinion that the self-imposed disparity between natural and man-made sound may have a negative effect on human-environment relations. Furthermore, it is not always relevant in terms of subjects' reactions. While it is generally accepted that traffic noise and loud machinery such as drills are disturbing, there seemed to be little consistency in subjects' responses to many other sounds. For example, some subjects regarded intermittent man-made sounds to be disturbing, while others regarded them as interesting; some found people sounds pleasurable, while others found them intrusive. Thus, what is often perceived by some to be a battle between two soundtracks out of tune with one another, the author would perceive to be one soundtrack with many voices and textures, all of which are subject to the learned values and current whims of the individual.

Also evident in the qualitative research that was conducted in this thesis is the importance of expectation on listener experience. For example, subjects would often remark that traffic noise was 'to be expected' in urban areas. This implies there is a sense that certain sounds belong in certain places, and provided they do not break the mental model, they can be tolerated. However, there did also seem to be a slight discrepancy between expectation and reality. For example, participants often reported feeling surprised at the amount of variation in the soundscape - something which may be related to the selective auditory attention of the brain. Normally, sound is unconsciously filtered by the brain, so that anything that is not considered relevant is ignored. This explains why our memories of soundscapes may be quite different from what is perceived in real life, at least on those occasions when we actually pay attention to them. Following on from this, it has occurred to the author that the impact of selective auditory attention on listener expectation may be reinforced to some extent by the entertainment and electronic arts industries. For example, it is common in these industries for sound designers to adopt a *hyper-realistic* approach in their work (e.g. [446]). This basically means they 'intensify the normal' for dramatic effect. It seems conceivable to the author that excessive exposure to hyper-realistic sound might serve to widen the gap between expectation and reality. In the extreme, humans may come to expect and strive for what is in fact artificial, as they already do in relation to so many other aspects of their lives - from home interiors and holiday destinations to their own bodies and relationships. It is quite conceivable that the human obsession with the unreal has also penetrated the acoustic domain, or if it has not happened yet, perhaps it is something for the next generation to be aware of when considering the soundscape.

8.5 Future Work

In this section, suggestions are made for improving and expanding on the methodologies used in the thesis in future work. For example, an area for concern regarding the sound walking methodology relates to the semantic differentials that were used in the soundscape questionnaires, despite the fact they were based on prior work. While the differentials appear to have satisfied the objectives of this thesis, there was a degree of ambiguity in relation to some of the adjectives, within the context in which they were used. The greatest ambiguity was associated with the terms harmonious-disharmonious, rough-smooth, warm-cold and strong-weak. It is thought this could be due to their musical connotations, which is probably no coincidence since it is common in soundscape studies for researchers to draw parallels with music. However, in relation to environmental sound, the author considers them to be less appropriate, or at least to warrant further explanation. It is therefore suggested that replacing these terms with ones that are more intuitively attributable to outdoor sound environments might reduce this ambiguity. For example, warm-cold might be replaced with vibrant-dull, harmonious-disharmonious with balanced-unbalanced, and rough-smooth with disjointed-flowing. This would very probably require a separate investigation to determine the most suitable choice of terms, and is therefore something to be considered in future work. For example, this might be achieved by asking a group of listeners to evaluate recordings of several unique soundscapes according to a list of semantic differentials. Analysing the amount of deviation in their responses ought to be an indicator of how consistently the terms were interpreted. It could also be investigated whether explaining the intended meaning of the terms prior to performing the test reduces the amount of deviation in the results.

Also an issue was the instability of the 3-D image in the ambisonic presentation of the virtual sound walk, which was noticeable when listeners moved their heads. Some of those who were positioned outside the sweet spot also reported a slightly uneven mix, although generally the sense of realism and immersion was thought to be high. Both of these issues are well known characteristics of ambisonic multichannel systems. The phase and instability issues may also have been exacerbated by the Harpex decoder which attempts to perform source separation based on a psychoacoustical model to deliver a higher order rendering. Given the complexity and heavy broadband noise content of the recordings, it may have resulted in worse phase issues than would have occurred using a more basic decoder. However, no ambisonic decoder is without its limitations, and, since the Harpex decoder was used consistently throughout the series of virtual sound walk experiments, it is not considered to have biased the results. Nevertheless, the choice of reproduction system is something that might be re-evaluated in future work.

While there is strong evidence in the literature to support the case for monitoring and mitigating against excessive amounts of environmental noise, there is another side to the story that, given sufficient attention, might hold equally positive benefits for society. When one considers that negative reactions to environmental sound often originate in the mind - i.e. they have a psychological explanation rather than a physiological one - then it seems possible they might be managed more effectively by the individual. The author would like to investigate this proposition further, giving particular attention to the benefits of attentive listening on the perception of environmental sound. It seems possible that re-learning how to tune into one's senses out of curiosity rather than defence, would not only improve the quality of human-environment interactions, but may also hold personal benefits for the individual. A first step in this process might involve the disassociation of stressful sounds from their causes - something which might be investigated through auralisation and acousmatic listening.

With respect to the simulations, while there was some agreement between the band structure of the impulse responses derived through FDTD simulation and the theoretical predictions, it was felt that the limitations of the computer model prohibited a closer match. For example, it was found that the band structures produced by the simulations were quite sensitive to small changes in the mesh geometry which may be related to the coarse discretisation of the inclusions. Similarly, the scope for comparison between the simulated results and the real-world measurements were also limited by the inherent differences in the models. For example, in the computer model, the surface of the inclusions was assumed to be perfectly reflective - hence no consideration was given to structure-borne sound and local resonances. While these limitations are unlikely to have had a major impact on the outcome of the virtual sound walks performed here, in the interests of extending the method to accommodate problems of a different nature, it is considered that some further work would be needed. For example, it may be advantageous to make provision for elastic wave propagation through the structures. This would require different computational strategies, and is therefore something that is being considered for future work.

Finally, the work presented was a single illustrated example of the creative application of auralisation to an environmental noise problem. However, there are endless other possibilities for the imaginative use of audio technology in environmental acoustics and urban design. The hope is therefore to conceive and investigate other examples in future work using similar techniques to those adopted in this thesis.

9

Appendices

9.1 Sound Walk Questionnaire

Pre-soundwalk Questionnaire

Please answer the following questions as honestly as possible. You may leave blank any questions that do not apply.

Age group (circle one):	15-24	25-44	45-64	65+	**Sex (circle one):**	M	F
Profession:					**Postcode:**		

How often do you visit Woodhouse Moor? (tick one)

Daily	
Very frequently (i.e. once or twice a week)	
Quite frequently (i.e. a few times a month)	
Not very frequently (i.e. once a month)	
Infrequently (i.e. less than once a month)	
Rarely	
This is my first visit	

For what purpose do you visit? (tick all that apply)

It is part of my regular walking/cycle route	
Festivals and sporting events	
For socialising with friends (e.g. picnics, BBQs)	
By myself for reading/sitting quietly	
Exercise (e.g. jogging, circuits)	
To use the skate park	
Other (please state)	

In your profession, are you involved with any aspect of urban planning and/or acoustics?

Still thinking about your professional life, in what way(s) do you engage with sound?

What are your general impressions of the Woodhouse Moor area?

A map of the route has been provided on page 2. What do you expect you will hear on the soundwalk? Are there any particular feelings or attitudes you would generally associate with these sounds?

About the Soundwalk

What to expect

- The soundwalk route is approximately **1 mile** long and will be taken together, with the guide leading.
- At the start, there will be a **5 minute warm-up** to prepare you for soundwalking.
- The walk is to be conducted in **silence**. If possible, it is best to **spread out** whilst walking.
- At each of the **points 1-5**, we will **pause 4-5 minutes** before continuing.
- At the end of the walk, there will be a **10-15 minute group discussion** after which you will be asked to hand over your questionnaire.
- The whole procedure is expected to last approximately **1 hour.**

Route map

On the Soundwalk

In each of the locations marked 1-5, please answer the following questions about the sound environment. The empty boxes are for you to record your thoughts as and when they occur. You may wish to list/describe individual sounds and note your reactions to them, and/or to the soundscape as a whole. You needn't use structured sentences, odd words are fine too. You may leave a question blank if you are unsure what to write. After the walk there will be a brief discussion during which you may wish to add further comments. Remember, there are no right or wrong answers!

Location 1

What were your impressions of the sound environment in this location?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

In your own words...

Location 2

What were your impressions of the sound environment in this location?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

In your own words...

Location 3

What were your impressions of the sound environment in this location?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

In your own words...

Location 4

What were your impressions of the sound environment in this location?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

In your own words...

Here you may wish to list/describe individual sounds, and/or note your reactions to them or the soundscape as a whole.

Location 5

What were your impressions of the sound environment in this location?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

In your own words...

General comments

This space is for any general comments about the sound environment on Woodhouse Moor.

General comments about the soundwalk

You may use this space to further comment on your experiences today. For example, has it done anything to change your feelings and/or behaviour towards the sound environment, both on Woodhouse Moor and in general? Did you enjoy the experience? Could anything have been done better? Etc.

Thank you for taking part.

9.2 Sound Walk Data

9.2.1 Written Feedback

Presented in Tables 9.1-9.8 is a summary of the written comments that were made during the real sound walk experiments. All of the completed questionnaires are available on the DVD accompanying this thesis (Directory 9.7.4.1).

While every attempt has been made to ensure the following text is true to the original questionnaire text, in some cases some minor amendments have been made for ease of reading. These are either grammatical or syntactical are are not considered to have affected the meaning of the text.

Table 9.1: Question: 'What are your general impressions of the Woodhouse Moor area?'

Comments	Age Range	Gender
'It looks like a very old park...the trees I see bring back memories from the past.'	15-24	M
'It is green, surrounded by roads and frequented by students.'	25-44	M
'A nice place to spend time alone or with friends...a social place.'	25-44	F
'Urban - a little bit of nature in the city.'	25-44	M
'Nice and natural at first, but then as more of a community place.'	25-44	M
'Very young and active. Lots of space and a good location.'	15-24	M
'Green urban space, a student hang out in the summer term.'	25-44	F
'Nice place to have a rest.'	15-24	M
'Quite a pretty area.'	25-44	M
'Vibrant and lively.'	25-44	F
'It is nice and green, though not looked after well enough.'	25-44	M
'A very pleasant park. Peaceful, with a good atmosphere.'	25-44	M
'Very pleasant.'	45-64	F
'Beautiful flowers, interesting people, and too much litter.'	25-44	M
'Nice surprise. A variety of beautiful trees along path - too bad people take it for granted and litter.'	25-44	M
'Spacious, natural, garden, peaceful, fun.'	25-44	M

Table 9.2: Question: 'What are your impressions of the sound environment in location 1?'

Comments	Age Range	Gender
'A very strong sound of swinging leaves which nearly overtakes the inbound sounds from the streets...The sound of people talking makes me feel a bit nervous.'	15-24	M
'Road noise dominates this location.'	25-44	M
'It is near the street, so there are many annoying sounds when you listen carefully, such as cars and people talking loudly.'	25-44	F
'Wind in the trees versus tyres on tarmac. There's lots of movement, which I like.'	25-44	M
'Various kinds of noise, but not too noisy.'	25-44	M
'Too much traffic, although wind sound is soothing.'	15-24	M
'There was too much car noise.'	15-24	M
'I hear talking and the wind through the trees.'	25-44	M
'It is quiet. I hear leaves rustle, cars, wood pigeons...it feels calm and hidden.'	25-44	F
'The aeroplanes are intrusive, but the wind rustling the trees is nice. Although it is close to the road it feels peaceful.'	25-44	M
'Traffic noise is consistent and distant. It did not disturb me mostly, except the occasional loud motorbike or bus.'	25-44	M
'Interesting quite gentle environment, with children's voices, traffic and birds.'	45-64	F
'The sound seems softer as we are surrounded by grasses and flowers. There is birdsong and the buzzing of insects.'	25-44	M
'Surprised how little we needed to walk for the cars to become silent.'	25-44	M
'Sounds were distant, ambient. I noticed natural sounds - the birds and wind which changes the loudness of certain sounds. Some sounds were constant, such as the traffic.'	25-44	M

Table 9.3: Question: 'What are your impressions of the sound environment in location 2?'

Comments	Age Range	Gender
'Road noise dominates the other sounds. Cars are slower, so there is less tyre noise, which makes it more peaceful than location 1.'	25-44	M
'Even though it is quite loud because it is near the street, it is nicer than location 1. There are not so many people talking, so the birds sound louder. It is a noisy place, but the sounds are very different. It is interesting to listen to them all.'	25-44	F
'The traffic is on more than one side - it surrounds us. The bird song has lessened.'	25-44	M
'Too noisy with the car noise, even though I am in a park. It feels a bit unpleasant.'	25-44	M
'Pretty much the same as location 1.'	15-24	M
'Road closer but less intrusive noise and less human sources.'	25-44	F
'There were more natural sounds which made the place more pleasant, but the street noise made me annoyed.'	15-24	M
'There are cars passing by very close, and the wind in the trees.'	25-44	M
'A varied spot for sounds when cars are not vrooming past. The birds are singing and people are strolling peacefully through alone or with friends chatting.'	25-44	F
'There are people laughing, shouting, birdsong, footsteps, cars...It felt good to hear people laughing - pleasant, energising. The cars are annoying but to be expected. They almost become the background noise. The sound of wind in the trees is calming. There are lots of languages being spoken.'	25-44	M
'It is much closer to the road. The noise of cars driving past seems to be in the forefront. I find it more disturbing than location 1. The fact it is on a path and a bike lane means the sounds of people walking are too close, making me uncomfortable.'	25-44	M
'I heard my footfall, leaves and twigs cracking as I walked. There were harsher, abrasive sounds of traffic noise. I was aware of the buildings on my left acting as a barrier or hard edge and I did not like it. Fluctuations were more marked as the traffic was less constant. I heard running footsteps, and an aeroplane which was soft and slow.'	45-64	F
'I heard jogging, a bicycle clicking, traffic and a musical interlude drifting by...the distant screams of aeroplanes, blackbirds singing and in distress...I heard voices but not their words, a wood pigeon, and the buzz of a fly near my ear.'	25-44	M
'Nice to hear people walking and talking as they enter the park at this point, but the proximity to traffic lights meant engines were louder.'	25-44	M
'The natural elements are disrupted and taken up by traffic sounds. I am aware I am not in a natural environment. I am aware of the city and the need to pay more attention. It is not relaxing.'	25-44	M

Table 9.4: Question: 'What are your impressions of the sound environment in location 3?'

Comments	Age Range	Gender
'As it is located a bit more centrally, the surrounding trees cover this area from the distinct sounds of the cars. Singing birds give it a nice tone and helps to relax, although the sound of a plane overhead overtakes it.'	15-24	M
'Road noise is masked by the wind in the trees. The bird calls are more easily heard. The noises of people on the park are also less prominent as this area is cut off from them by trees and the geography of the location. There is more bass engine noise than the higher pitched tyre noise from the cars.'	25-44	M
'This place makes me feel relaxed. It is not very loud, and the sounds are very calm. There are less different sounds, but they are not monotonous. Even the street sounds and plane sounds are nice to hear here.'	25-44	F
'Everything sounded distant and removed, other than the wind in the trees and the birds...and the plane passing over! I found this to be intrusive.'	25-44	M
'Only wind and singing birds touched me. There is less car noise, which makes me feel more relaxed.'	25-44	M
'There is less traffic and I can hear birds. It feels much more comfortable.'	15-24	M
'Wind through trees and birdsong is quite restful. The occasional passing jet and urban sounds are more disharmonious but are interesting.'	25-44	F
'It was the most pleasant place. You could hear the harmony of nature. There were mainly soft, peaceful sounds with a few interrupting annoying sounds.'	15-24	M
'There are distant cars passing by, birds, a plane passing by which is quite loud, and the constant wind in the trees.'	25-44	M
'Two men sit drinking wine and chatting in an unusual language. There are birds nearby and people passing through.'	25-44	F
'Aeroplanes are intrusive. There are the sounds of people talking, walking and training for sports, and lots of bird song. The space feels nicer...it is a comforting environment. Sounds of sirens disturb the silence and bring me back to reality.'	25-44	M
'There are lots of paths around. People meeting, walking past and chatting or cycling past made me feel slightly uncomfortable. There is much less traffic noise than previously, and there seems to be more birdsong.'	25-44	M
'I hear the slap of a hand on a ball, laughter, more human voice sounds, the 'pop' of a tennis ball on a racket, a mobile phone tone. The bird song is stronger with more variety of tones. There is the harsh aeroplane noise, the click of hard shoes on a hard surface, loud and harsh sirens, different languages being spoken, the rustle of wind in the trees and the swish of feet on grass.'	45-64	F
'I imagine African voices as percussion, a blackbird as soprano, an aeroplane as the bass. Heels on brick are insistent and quick...the occasional pop of tennis balls, the usual sporadic emergency alarms, seemingly from all directions.'	25-44	M
'There was all natural noise - people playing and talking, the birds doing the same. No cars audible. Beautiful. Happy again!'	25-44	M
'It has a natural feel. There is little traffic noise. All sounds can be heard and identified...conversations can be heard. The distant traffic feels slightly relaxing...slows things down. Peaceful, although sirens and aircraft disturb the peace.'	25-44	M

Table 9.5: Question: 'What are your impressions of the sound environment in location 4?'

Comments	Age Range	Gender
'A mixture of urban sounds erases harmony from the soundscape and makes me feel a bit distracted.'	15-24	M
'The sounds of human activity dominate here. There are men playing basketball, teenagers shouting, and skaters and bikers jumping ramps.'	25-44	M
'Very active sounds. It's hard to concentrate on each of them. Urban sounds of cars and people are easier to hear. The wind is not blowing, the birds are not singing...but it is nice and social. An active place, not for relaxing.'	25-44	F
'I am surprised by how much sound skateboards make. It is interesting to try to detach the sound of speech from language and to hear it as any other sound - it becomes more invasive when I hear the words.'	25-44	M
'There is not only car noise, but also people shouting, children screaming, and noise from doing sport.'	25-44	M
'It is very crowded. It gives you good emotions if you are in a good mood yourself. I find the sound of basketball bouncing very inviting.'	15-24	M
'The noise would have been more intrusive had we stood closer to a noisy group.'	25-44	F
'I felt I was in a big town in peak hours. I wanted to go to another place.'	15-24	M
'There is a phone ringing, cars, screams, conversations, basketball, the wind, skateboards, people walking by and a plane passing by.'	25-44	M
'There are loud, quite hard, man-made sounds, but it is not unpleasant. It is good to hear movement and lots of clanging. It is not somewhere I would stop for lunch, but somewhere to people watch and be entertained.'	25-44	F
'It is a mish-mash of sounds - all man-made. There is a drollness to it - a lack of uniqueness. The sound of people talking and laughing is nice, but is drowned out by man-made sounds The sound of balls bouncing is quite monotonous but the rhythm varies. I feel agitated here.'	25-44	M
'A lot of activity. The traffic and basketball are most prevalent. It would be quite disturbing if I wanted to relax and shut my eyes, but it is not so disturbing that I could not sit and chat or watch the sports perfectly happily.'	25-44	M
'There is the burr of a helicopter, the chatting of male and female voices, the tinny sound of music from mobile devices, magpies squawking, the clack of the skate park, children shouting, a baseball pump, the musical 'plang' of metal railings, the deep rumble of traffic, a heavy lorry and a bus purring...the cars close by have a higher tone.'	45-64	F
'There is a rhythmic stutter of basketballs, the drone of a passing moped above the lower hum of larger vehicles, the crunching and scraping of skateboard wheels, the squawking of a disturbed magpie, the excited chatter and squeals of children in the playground, the rackety whir of a bicycle being pushed, and the higher whir of a ridden one.'	25-44	M
'Nice to hear people enjoying themselves as we walk through, but I could not rest here as it is too noisy and too close to the road. Even the nicer noises of people enjoying themselves are too abrasive to enjoy for any longer than passing through.'	25-44	M
'Human sounds - traffic, people. Frequently changing sounds, but a mix of repeated sounds. Feels active, interesting and busy, but predictable and human-like. I still feel relaxed though. The traffic is ambient, while the people and sports are prominent.'	25-44	M

Table 9.6: Question: 'What are your impressions of the sound environment in location 5?'

Comments	Age Range	Gender
'I would say this location is a bit unique as it is really warm and harmonious even though I can see many cars and people close by.'	15-24	M
'The wind drowns out and blends in with the road noise. Children's shouts compete with the occasional alcoholic and conversing Arabs.'	15-24	M
'Typical urban sounds. No people are talking, which makes the place seem much more quiet. The wind is blowing again...it is a nice place - harmonious.'	25-44	F
'I find the novel ice-cream van tune less annoying than the ones I hear everyday at home...though I am sure the novelty would wear off. The traffic is softer, and the sound of trees seems to be in front.'	25-44	M
'I hear echoes from different sources, such as children shouting and vehicle noise, but it feels more pleasant than at location 4.'	25-44	M
'Most of the sounds are further away. It makes me feel more isolated in a pleasant way.'	15-24	M
'It was good to get some silence again.'	15-24	M
'I hear music from ice-cream vans, the wind in the trees, a bird flapping, a few distant shouts, kids, and a person passing by.'	25-44	M
'Quiet, not many people about. There is a couple on a bench laughing...there is the breeze in the trees which creates a calming rustle, and when it is constant the vroom of the traffic could be a river.'	25-44	F
'The car noise provides an ambient background. There are people playing sports, trees rustling, birds chirping, and motorbikes intruding on the peace. I can hear the distant echoes of skateboards hitting the concrete. I feel neutral towards this spot.'	25-44	M
'The traffic is similar to location 1. At these levels it is not too disturbing - at least not for a short time. There are relatively few people walking past or cycling. Most sounds are quite distant, making it feel very peaceful.'	25-44	M
'I heard a nose being blown, the rumble of a plane, the constant sound of traffic, the burble of voices, the swish of grass, the squeak of wheels, the crunch of shoes on gravel and tarmac, loud voices, clapping, the rumble of a motorbike, and sometimes a lull.'	45-64	F
'There was enthusiastic shouting and laughing of rugby players, growling motorcycles, reedy blackbirds and soft coo of pigeons. There was the soft crunching of feet and the distant hum of indistinct sounds.'	25-44	M
'Boring. It has a bit of everything - nice and bad in equal measures, but no charm (though this is probably due to the concrete ground that makes one immediately uncomfortable).'	25-44	M
'It is quiet...there is less sound altogether, but the usual mix of sounds. The traffic is constant, with some louder traffic and some distant people sounds. Feels natural - bird sounds can be identified.'	25-44	M

Table 9.7: General comments about the sound environment of Woodhouse Moor.

Comments	Age Range	Gender
'More varied than I expected.'	25-44	M
'It is a big place, so everybody can choose a place they like to spend time in. However, there are not many silent places, so I'd say it's not a place to relax, but for socialising and spending time with people.'	25-44	F
'It is possible to escape the sounds of traffic, which explains the circle of seats in location 3. The sound of wind and trees featured highly for me, which will change with the weather and time of year.'	25-44	M
'It is not very quiet. I would feel more comfortable in a forest, but it is a good place for socialising if you are feeling like it. If you felt like meditating or relaxing then it would not be the best place to go.'	15-24	M
'There is nowhere without sounds from the street and people, although this is not necessarily a bad thing. There was more variety than I expected.'	25-44	F
'It was different in all spots. Some were pleasant, others not so.'	15-24	M
'Quite an interesting variety of sounds, especially when concentrating. I liked the way the content is varying so much.'	25-44	M
'A very lively place! Lots of life...the sounds of cars were intrusive, but not overly so.'	25-44	M
'The traffic was occasionally quite loud and annoying but not consistently so.'	25-44	M
'I found it really engaging around the play area (location 4). The road to the east is less distracting than the one to the south. I found the bird sounds and some of the people sounds very pleasant. I appreciate the value of the park more now - it is restful, but also stimulating. A good 'transition' space.'	45-64	F
'I had anticipated variety but was still surprised by it. Almost all noises were pleasant, except for the engine noise of road traffic and aeroplanes.'	25-44	M
'A beautiful park is easily let down by the people who enjoy it and kill it at the same time. The litter saddened me greatly. The road noises were bearable on the whole, and the park is quite odd as it felt like several parks at the same time, depending on the location - something for everyone.'	25-44	M
'There seem to be three main sound types: people, nature and traffic. Each seems to vary in general level with spikes in each, occasionally. Generally the Moor is a relaxing place, cut away from the city.'	25-44	M

Table 9.8: General comments about the sound walk.

Comments	Age Range	Gender
'It is nice to take the time to absorb environmental sounds.'	25-44	M
'I enjoyed it a lot. It's nice to be silent while being amongst so many sounds.'	25-44	F
'Well done and enjoyable. I could happily spend longer sound walking.'	25-44	M
'Very interesting. I would have expected more information about what is expected for the result, and a more simplified vision of the project.'	15-24	M
'I wanted to stop in the middle to talk and drink a juice, as it felt odd to have silence for so long.'	25-44	F
'I had time to understand how many different sounds are surrounding us.'	15-24	M
'It has changed my view of the Moor positively. I am becoming more sensitive in terms of making sense of my audio environment.'	25-44	M
'It made me want to fall asleep! I felt very relaxed. It is nice to stop and listen to sounds as I never do it.'	25-44	M
'Very enjoyable. I actually found it quite relaxing!'	25-44	M
'Enjoyed it very much indeed - found it more interesting/engaging/enjoyable than I expected. Will do it again! Felt the timing was good - just right.'	45-64	F
'It made me want to fall asleep! I felt very relaxed. It is nice to stop and listen to sounds as I never do it.'	25-44	M
'I was very pleased to focus on listening to such a degree that I felt able to link some sounds together as if parts of a song or members of an orchestra...I thoroughly enjoyed the experience and hope that it will change how I view the park.'	25-44	M
'Very interesting. Made even more interesting by not talking and being alone with one's thoughts. A friendly gathering and a nice walk.'	25-44	M
'The sound walk has highlighted the individual sounds of the Moor for me. Just the simple change of place changes the 'mix'. It made me take more notice and take peace from the sounds - something I would not have thought of. Good! The times between places were fine.'	25-44	M

9.2.2 Normality Testing

Tables 9.9-9.13 show the results of the Shapiro-Wilks statistical tests which were performed on the real sound walk data in Microsoft Excel using the Analysis Toolpack. In statistics, the Shapiro-Wilk test tests the null hypothesis that a sample $[x_1, ..., x_n]$ came from a normally distributed population. If the p-value is less than the chosen alpha value (α), then the null hypothesis is rejected - i.e. it can be assumed that the data are not from a normally distributed population. If the p-value is greater than alpha, the null hypothesis that the data came from a normally distributed population is not rejected. The p-value is a statistical value indicating the probability of obtaining data at least as extreme as that which was measured [447].

Table 9.9: Statistical data pertaining to location 1 of the real sound walks in which $\alpha = 0.05$.

	Comfort-discomfort	Quiet-loud	Harmonious-disharmonious	Soft-rough	Weak-strong	Pleasant-unpleasant	Warm-cold	Unique-common	Monotonous-varied
Sample size	16	16	16	16	16	16	16	16	16
W statistic	0.94	0.90	0.88	0.93	0.88	0.83	0.88	0.85	0.89
p-value	0.3080	0.0671	0.0426	0.2350	0.0415	0.0067	0.0434	0.0151	0.0540
Mean	0.6	0.1	0.0	0.0	-0.2	1.0	0.7	-1.7	0.9
Std. error	0.34	0.33	0.38	0.32	0.23	0.26	0.30	0.34	0.38
Std. deviation	1.4	1.3	1.5	1.3	0.9	1.0	1.2	1.4	1.5
Variance	1.9	1.7	2.3	1.6	0.8	1.1	1.4	1.8	2.3
Skewness	-0.3	-0.3	0.1	0.0	-0.2	-0.8	-0.6	0.6	0.1
Kurtosis	-0.47	-1.16	-1.42	-0.91	-0.68	-0.21	0.01	-0.89	-1.40
Distribution	Normal	Normal	Not Normal	Normal	Not Normal	Not Normal	Not Normal	Not Normal	Normal

Table 9.10: Statistical data pertaining to location 2 of the real sound walks in which $\alpha = 0.05$.

	Comfort-discomfort	Quiet-loud	Harmonious-disharmonious	Soft-rough	Weak-strong	Pleasant-unpleasant	Warm-cold	Unique-common	Monotonous-varied
Sample size	16	16	16	16	16	16	16	16	16
W statistic	0.94	0.86	0.88	0.88	0.81	0.93	0.91	0.91	0.88
p-value	0.3652	0.0202	0.0408	0.0388	0.0036	0.2176	0.1056	0.0996	0.0386
Mean	0.3	-0.7	-0.5	-0.4	-0.7	-0.1	0.1	-1.6	-0.4
Std. error	0.34	0.30	0.33	0.27	0.22	0.27	0.30	0.30	0.34
Std. deviation	1.3	1.2	1.3	1.1	0.9	1.1	1.2	1.2	1.4
Variance	1.8	1.4	1.7	1.2	0.8	1.1	1.5	1.5	1.9
Skewness	0.0	0.9	0.7	-0.2	-0.7	0.1	0.3	0.6	0.1
Kurtosis	0.11	0.23	-0.26	-1.15	2.79	-0.49	-0.63	-0.32	-1.34
Distribution	Normal	Not Normal	Not Normal	Not Normal	Not Normal	Normal	Normal	Normal	Not Normal

Table 9.11: Statistical data pertaining to location 3 of the real sound walks in which $\alpha = 0.05$.

	Comfort-discomfort	Quiet-loud	Harmonious-disharmonious	Soft-rough	Weak-strong	Pleasant-unpleasant	Warm-cold	Unique-common	Monotonous-varied
Sample size	16	16	16	16	16	16	16	16	16
W statistic	0.66	0.88	0.87	0.91	0.87	0.78	0.93	0.80	0.82
p-value	<0.0001	0.0393	0.0321	0.1151	0.0300	0.0014	0.2180	0.0031	0.0047
Mean	1.9	1.1	1.3	1.3	0.4	2.1	1.3	-0.4	-0.9
Std. error	0.23	0.27	0.23	0.30	0.27	0.15	0.30	0.50	0.35
Std. deviation	0.9	1.1	0.9	1.2	1.1	0.6	1.2	2.0	1.4
Variance	0.9	1.2	0.9	1.4	1.2	0.4	1.4	4.0	1.9
Skewness	-2.1	-0.6	-0.6	-0.3	0.0	-0.1	-0.2	0.0	1.6
Kurtosis	6.88	0.40	1.47	-0.86	2.12	0.05	-0.65	-1.96	3.75
Distribution	Not Normal	Not Normal	Not Normal	Normal	Not Normal	Not Normal	Normal	Not Normal	Not Normal

Table 9.12: Statistical data pertaining to location 4 of the real sound walks in which $\alpha = 0.05$.

	Comfort-discomfort	Quiet-loud	Harmonious-disharmonious	Soft-rough	Weak-strong	Pleasant-unpleasant	Warm-cold	Unique-common	Monotonous-varied
Sample size	16	16	16	16	16	16	16	16	16
W statistic	0.93	0.76	0.85	0.81	0.85	0.88	0.91	0.95	0.83
p-value	0.2170	0.0008	0.0140	0.0042	0.0120	0.0416	0.1122	0.5725	0.0063
Mean	-0.5	-2.3	-1.0	-2.1	-1.9	-0.8	-0.5	0.3	1.6
Std. error	0.39	0.15	0.48	0.18	0.20	0.35	0.29	0.42	0.41
Std. deviation	1.5	0.6	1.9	0.7	0.8	1.4	1.2	1.7	1.6
Variance	2.4	0.4	3.7	0.5	0.7	1.9	1.3	2.9	2.7
Skewness	-0.2	0.2	1.1	0.2	0.6	1.2	-0.6	-0.4	-1.0
Kurtosis	-1.06	-0.38	0.55	-0.82	0.75	2.72	-0.07	-0.63	-0.03
Distribution	Normal	Not Normal	Not Normal	Not Normal	Not Normal	Not Normal	Normal	Normal	Not Normal

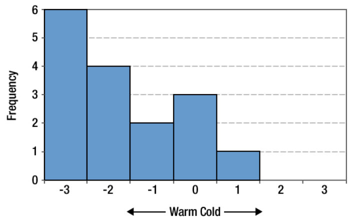
Table 9.13: Statistical data pertaining to location 5 of the real sound walks in which $\alpha = 0.05$.

	Comfort-discomfort	Quiet-loud	Harmonious-disharmonious	Soft-rough	Weak-strong	Pleasant-unpleasant	Warm-cold	Unique-common	Monotonous-varied
Sample size	16	16	16	16	16	16	16	16	16
W statistic	0.93	0.87	0.90	0.89	0.69	0.91	0.87	0.90	0.87
p-value	0.2785	0.0271	0.0729	0.0600	0.0001	0.1122	0.0299	0.0902	0.0271
Mean	0.7	0.4	0.2	0.5	0.4	0.5	0.1	-1.1	0.3
Std. error	0.35	0.22	0.29	0.26	0.22	0.29	0.26	0.35	0.27
Std. deviation	1.4	0.9	1.2	1.0	0.9	1.2	1.0	1.4	1.1
Variance	2.0	0.8	1.4	1.1	0.8	1.3	1.1	2.0	1.1
Skewness	-0.4	-0.2	-0.4	0.0	-1.7	-0.6	0.1	0.3	0.2
Kurtosis	-0.67	-0.65	0.11	-0.99	2.70	-0.07	0.76	-1.08	-1.18
Distribution	Normal	Not Normal	Normal	Normal	Not Normal	Normal	Not Normal	Normal	Not Normal

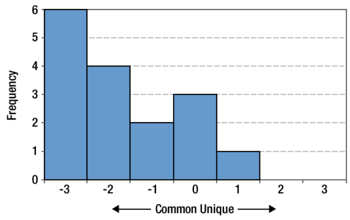
9.2.3 Frequency Histograms for Semantic Differential Scales

The following frequency histograms resulted from the semantic differential scales rated by participants in the real sound walk.

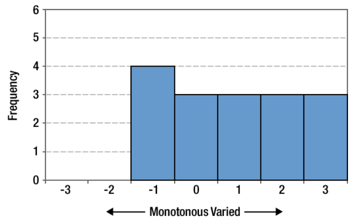
Location 1



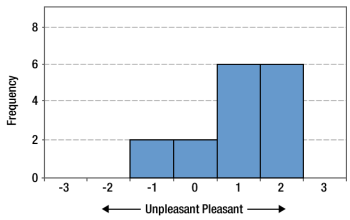
(a) Warm-Cold



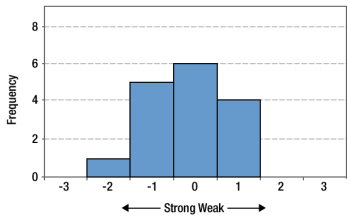
(b) Unique-Common



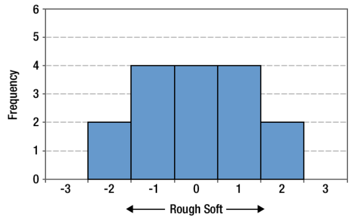
(c) Monotonous-Varied



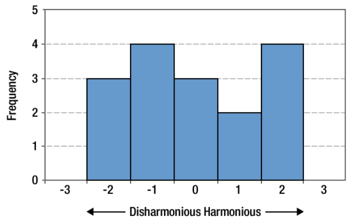
(d) Pleasant-Unpleasant



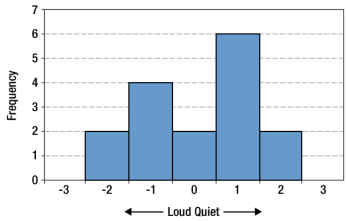
(e) Weak-Strong



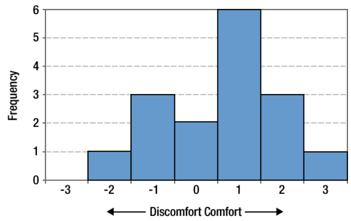
(f) Soft-Rough



(g) Harmonious-Disharmonious



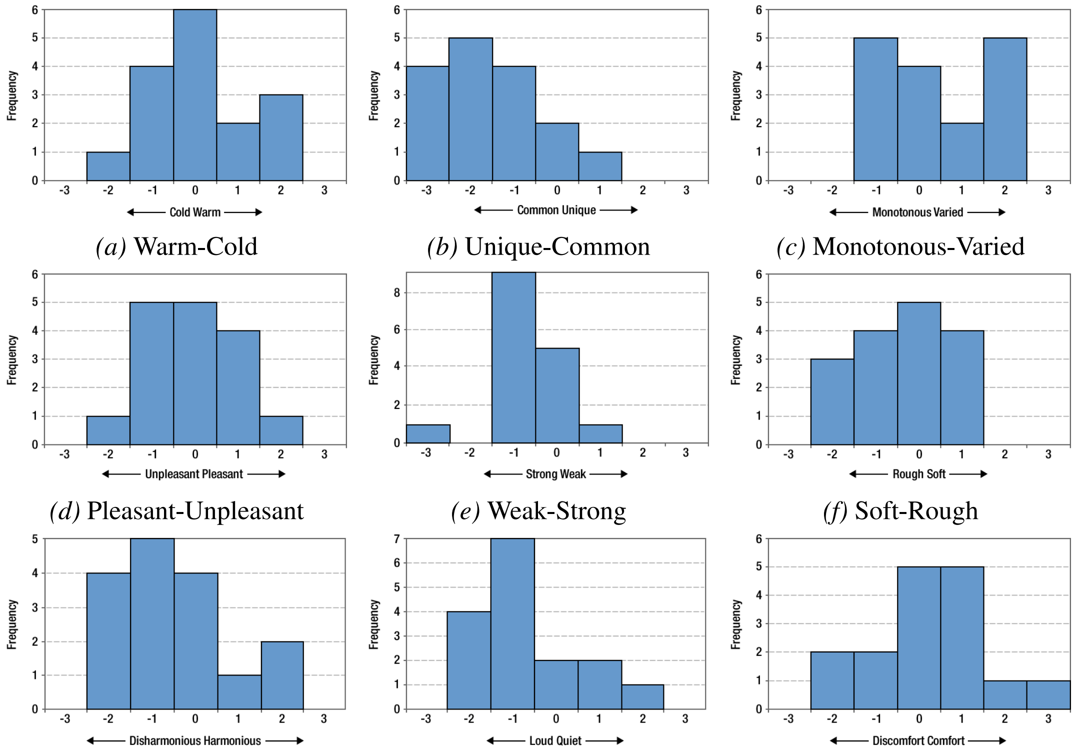
(h) Quiet-Loud



(i) Comfort-Discomfort

Figure 9.1: Histograms corresponding to location 1 of the sound walk.

Location 2



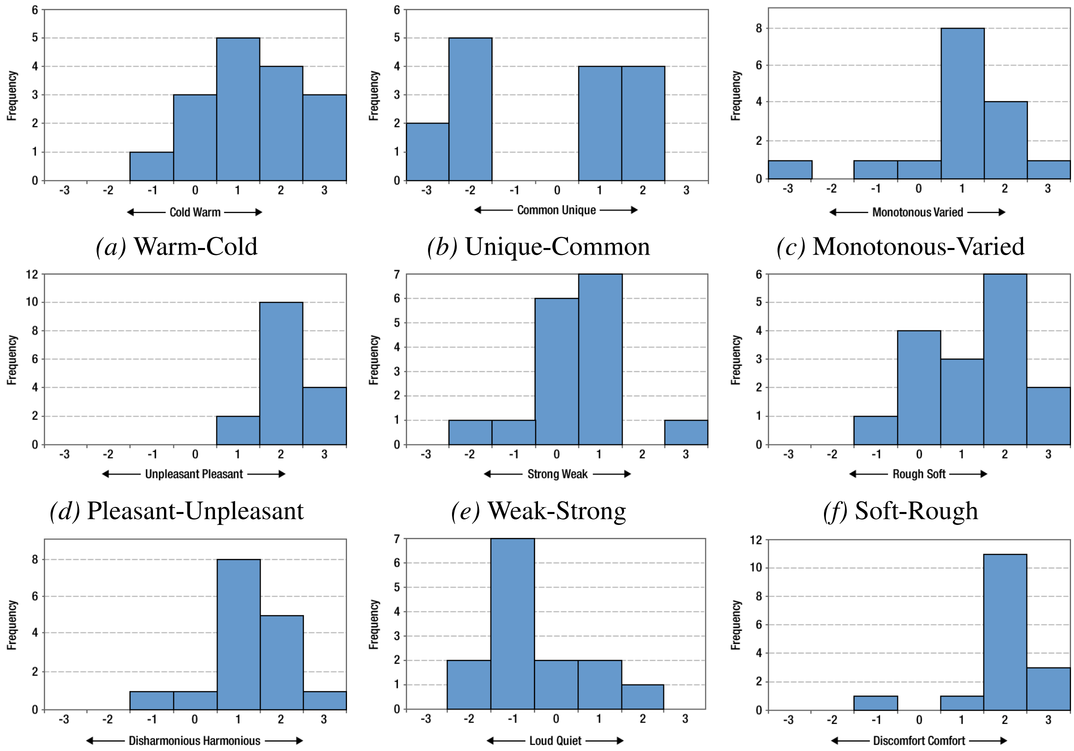
(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied

(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough

(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

Figure 9.2: Histograms corresponding to location 2 of the sound walk.

Location 3



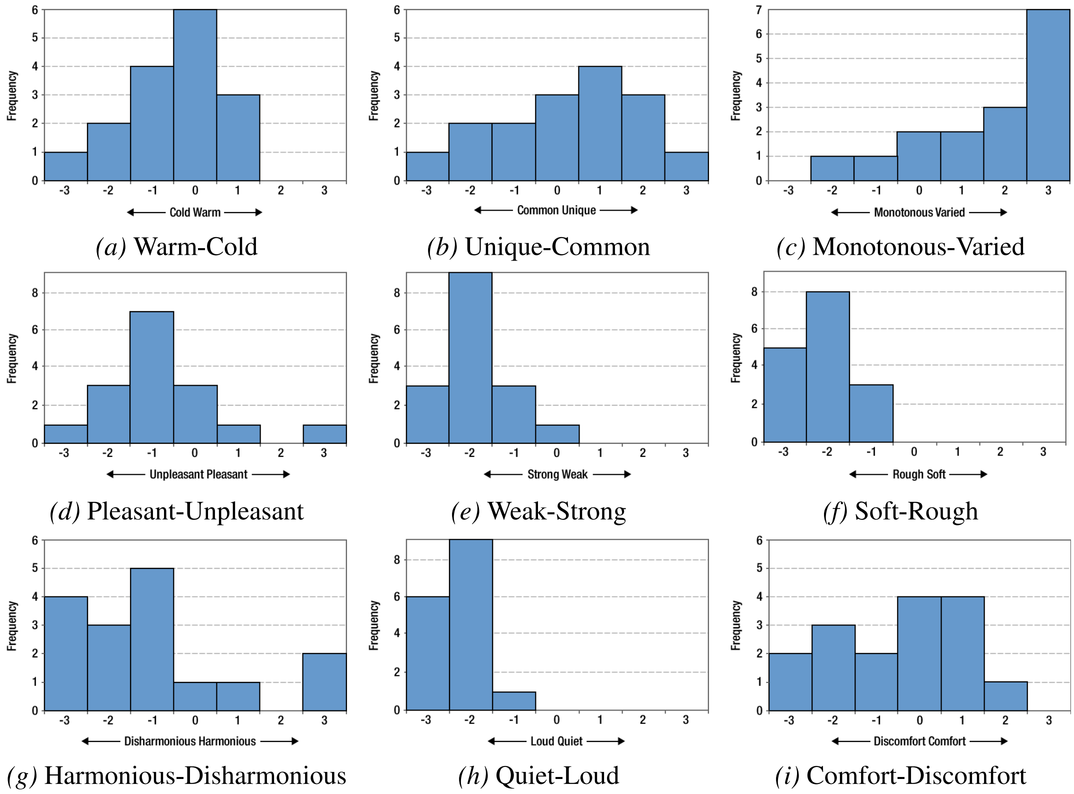
(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied

(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough

(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

Figure 9.3: Histograms corresponding to location 3 of the sound walk.

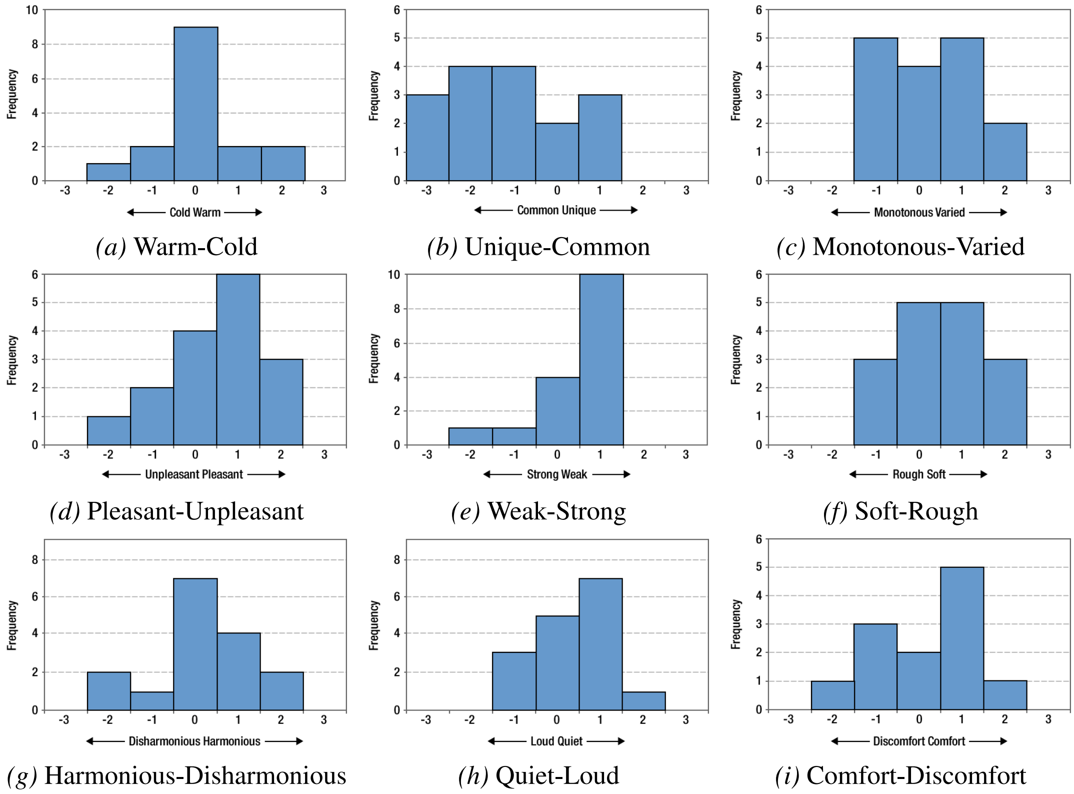
Location 4



(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied
(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough
(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

Figure 9.4: Histograms corresponding to location 4 of the sound walk.

Location 5



(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied

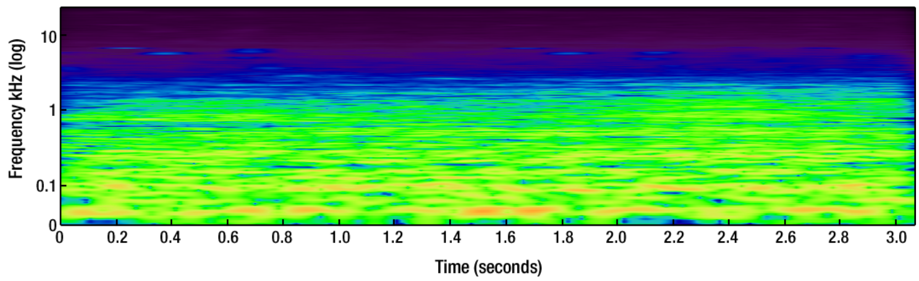
(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough

(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

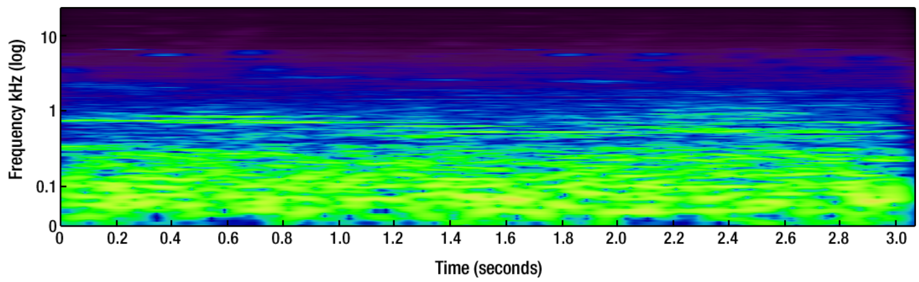
Figure 9.5: Histograms corresponding to location 5 of the sound walk.

9.3 Perceivable Difference Tests

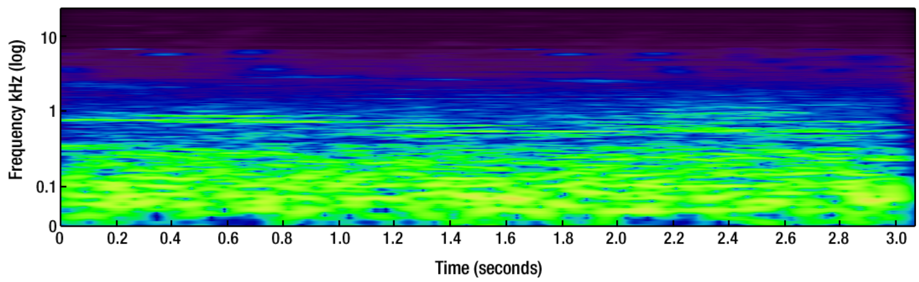
The following spectrograms relate to the audio extracts used in the perceivable difference test. There are four versions of each of the four extracts used. These consist of the extract filtered with the impulse response recorded in the empty FDTD simulation (A), the extract filtered with the 2-D sonic crystal array response (B), the extract filtered with the gyroid response (C), and the unfiltered control extract (D).



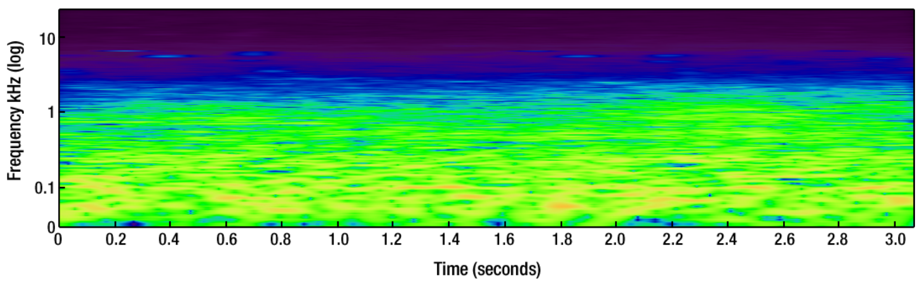
(a) A (Empty)



(b) B (2-D Sonic Crystal)

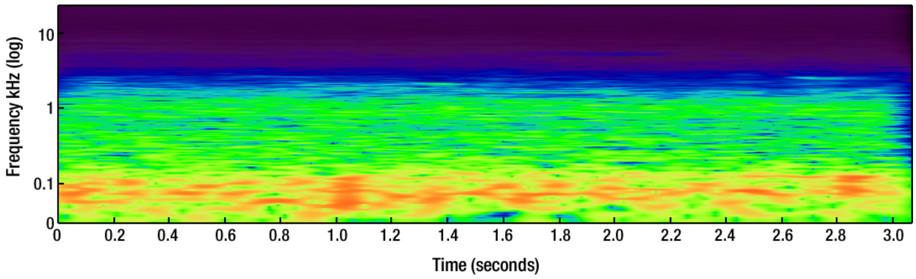


(c) C (Gyroid)

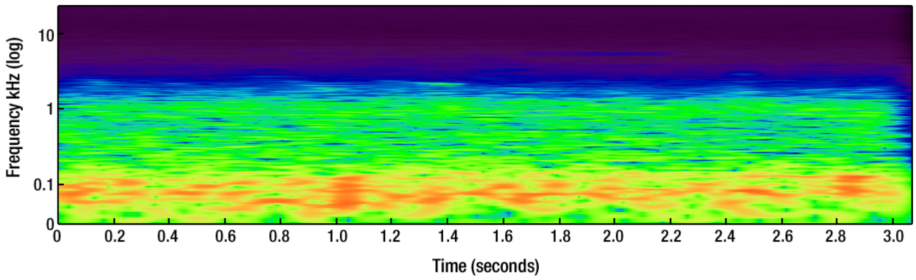


(d) D (Control)

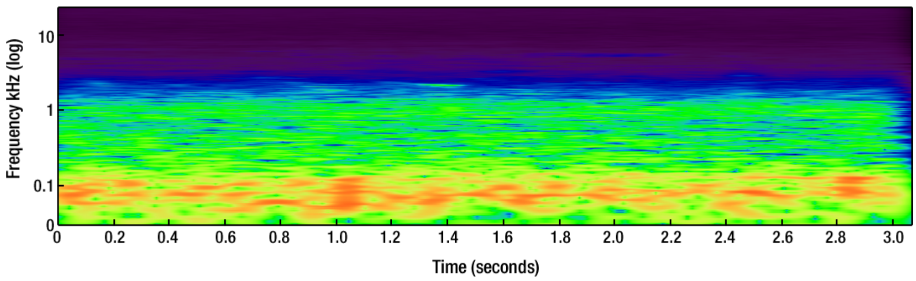
Figure 9.6: Spectrograms of the first audio extract used in the perceivable difference tests after the application of different impulse response filters.



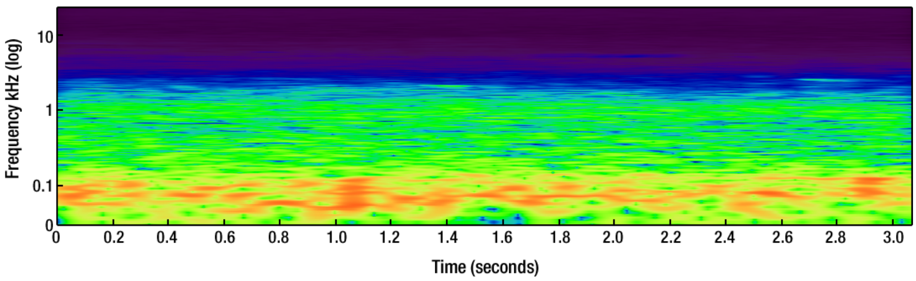
(a) A (Empty)



(b) B (2-D Sonic Crystal)

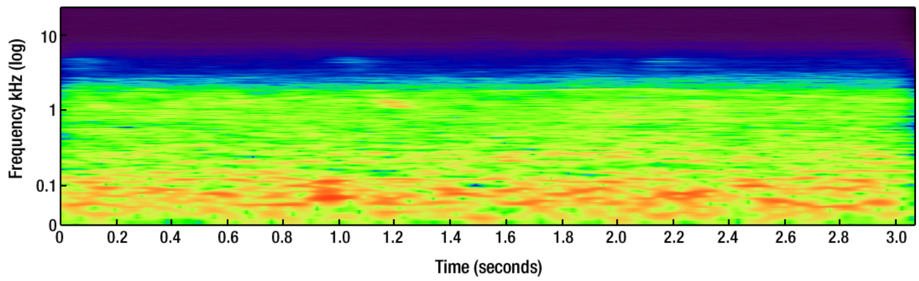


(c) C (Gyroid)

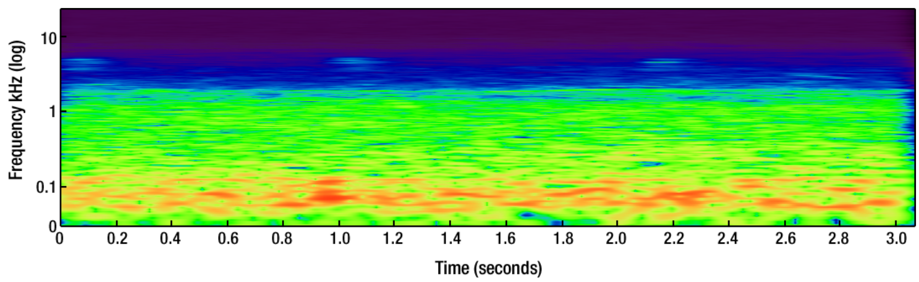


(d) D (Control)

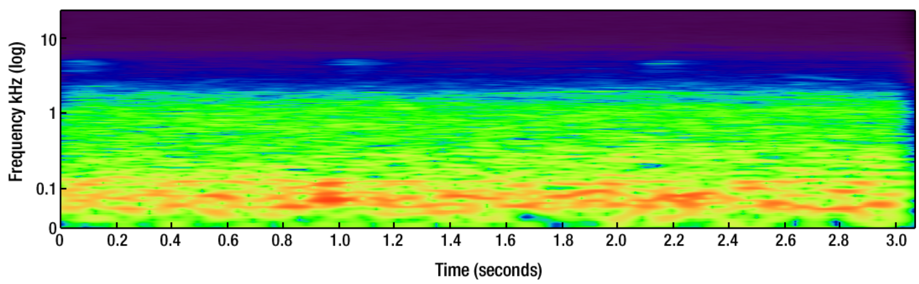
Figure 9.7: Spectrograms of the second audio extract used in the perceivable difference tests after the application of different impulse response filters.



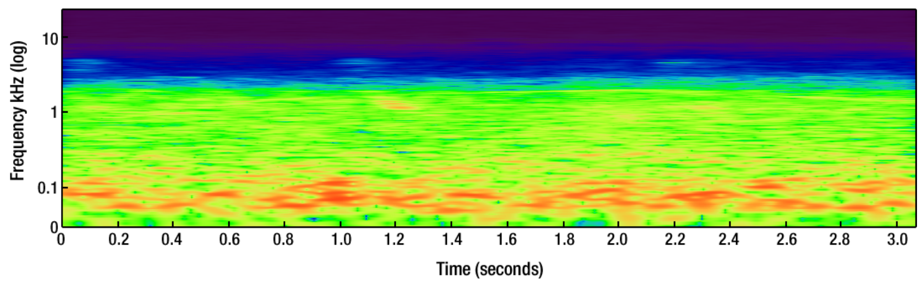
(a) A (Empty)



(b) B (2-D Sonic Crystal)

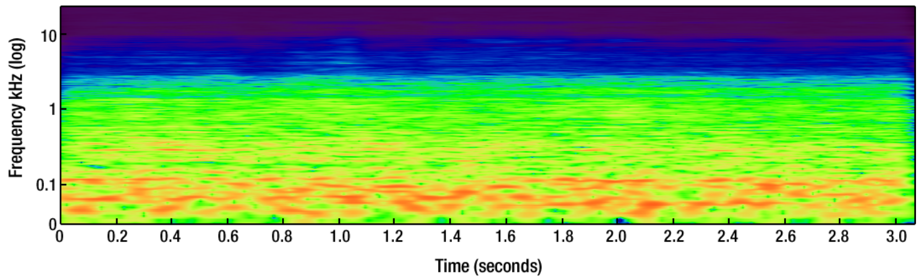


(c) C (Gyroid)

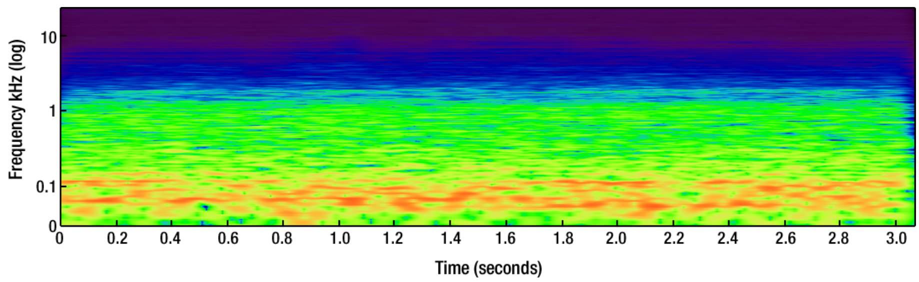


(d) D (Control)

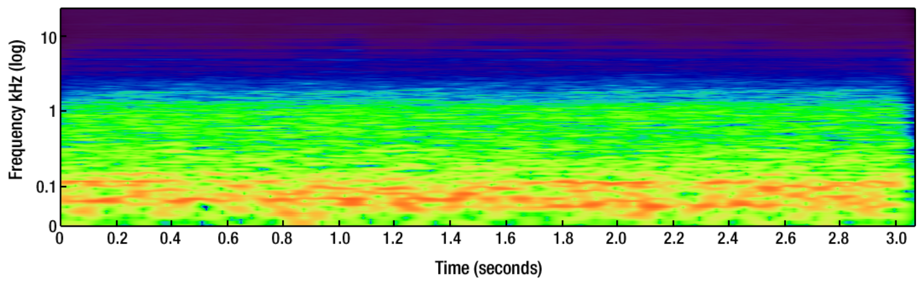
Figure 9.8: Spectrograms of the third audio extract used in the perceivable difference tests after the application of different impulse response filters.



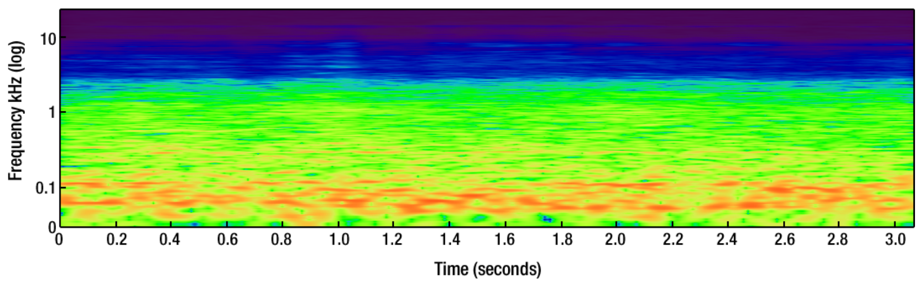
(a) A (Empty)



(b) B (2-D Sonic Crystal)



(c) C (Gyroid)

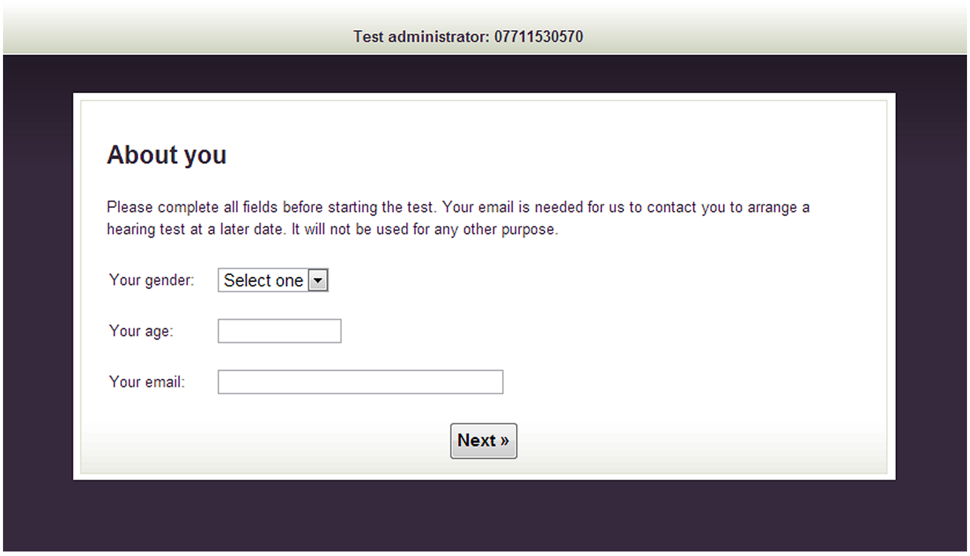


(d) D (Control)

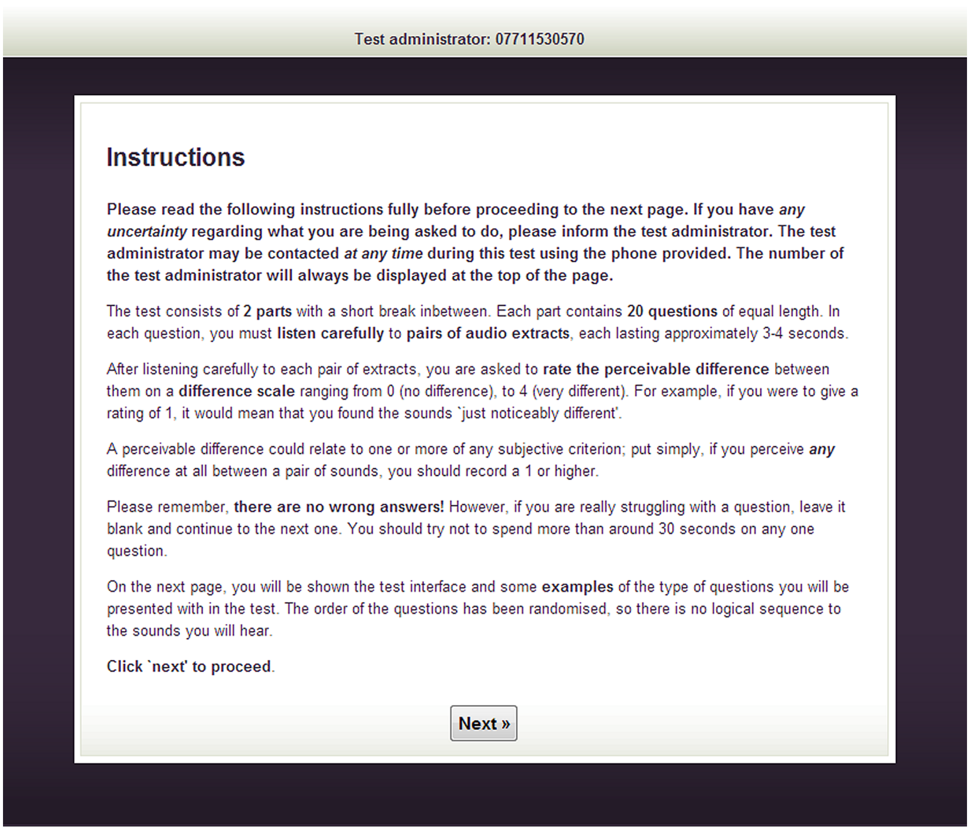
Figure 9.9: Spectrograms of the fourth audio extract used in the perceivable difference tests after the application of different impulse response filters.

9.4 Perceivable Difference Test Interface

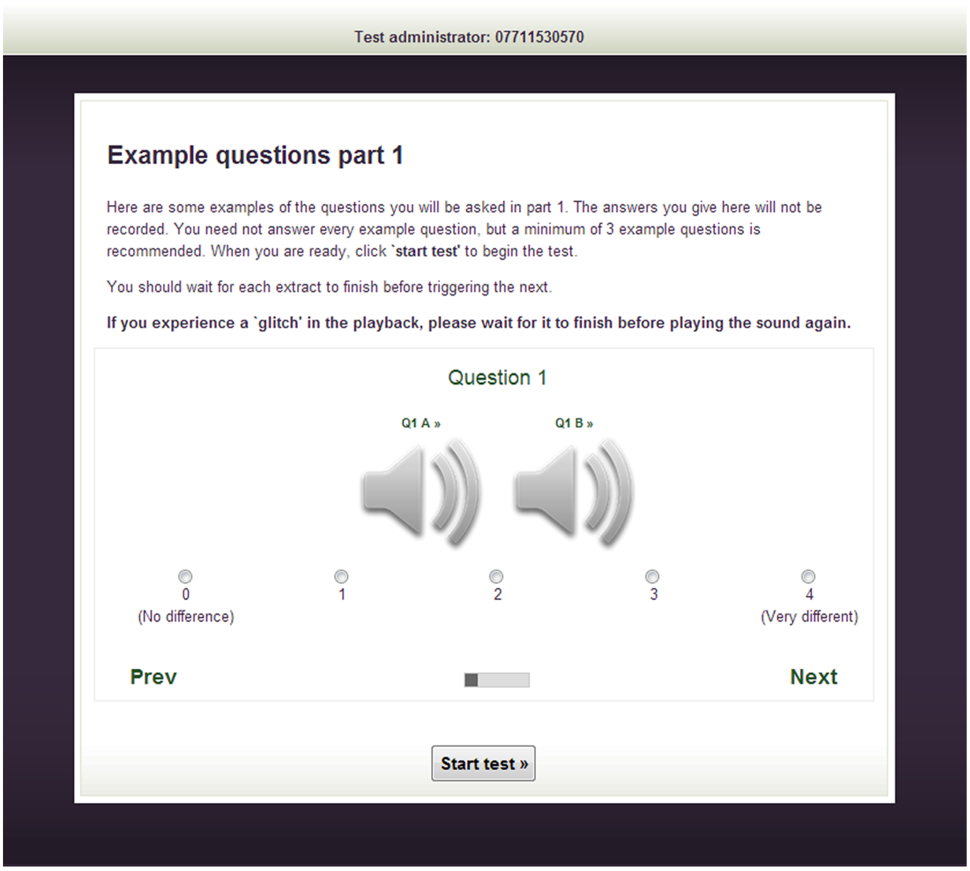
Figure 9.10: Screen shots of the perceivable difference test interface shown in order of appearance. The instructions are visible in *(b)* and include a definition of *timbre* in the context of the test. An electronic version of the test interface is included on the CD accompanying this thesis.



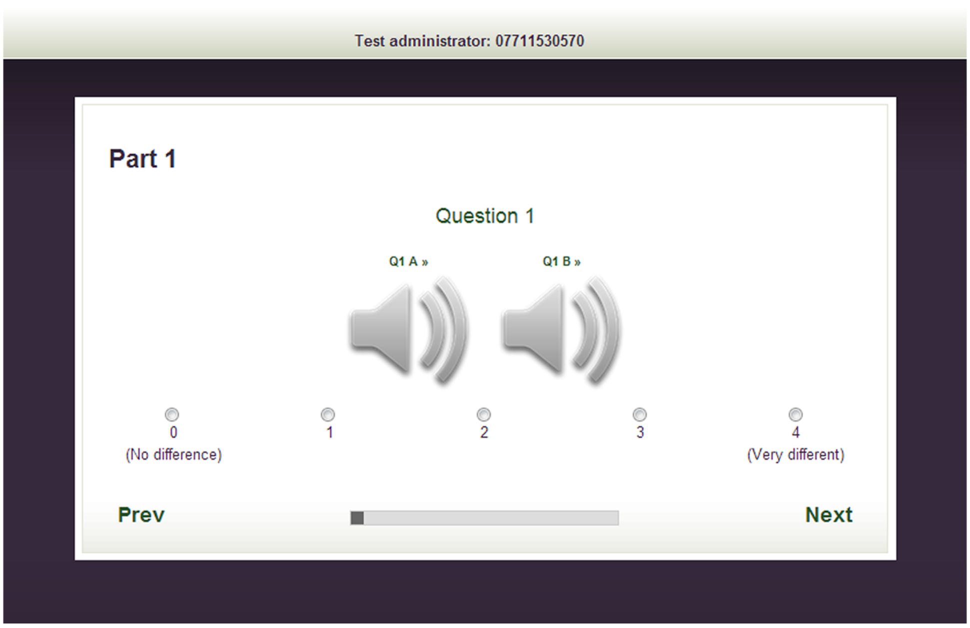
(a)



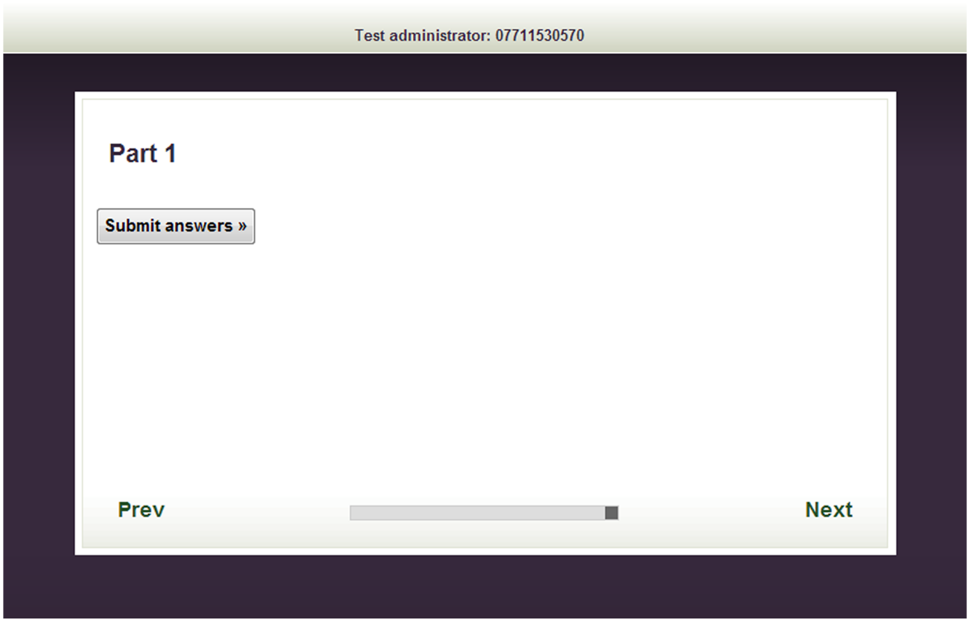
(b)


Test administrator: 07711530570
Example questions part 1
Here are some examples of the questions you will be asked in part 1. The answers you give here will not be recorded. You need not answer every example question, but a minimum of 3 example questions is recommended. When you are ready, click `start test' to begin the test.
You should wait for each extract to finish before triggering the next.
If you experience a `glitch' in the playback, please wait for it to finish before playing the sound again.
Question 1
Q1 A »
Q1 B »
0
(No difference)
1
2
3
4
(Very different)
Prev
Next
Start test »

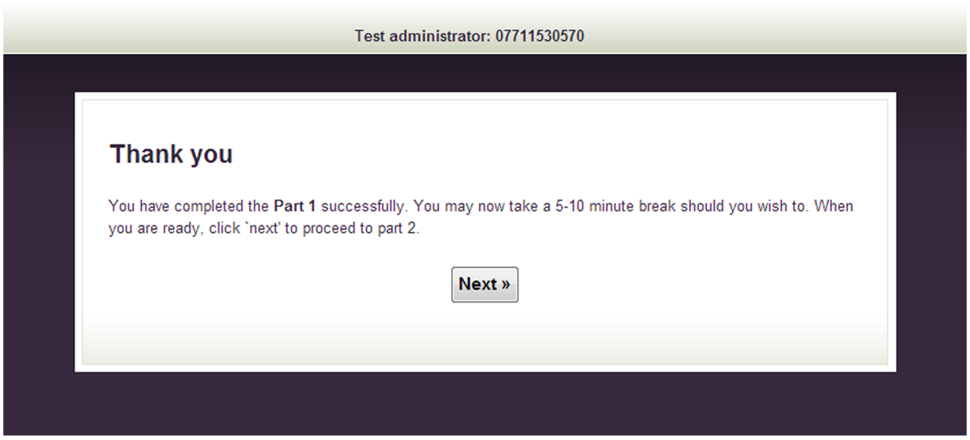
(c)


Test administrator: 07711530570
Part 1
Question 1
Q1 A »
Q1 B »
0
(No difference)
1
2
3
4
(Very different)
Prev
Next

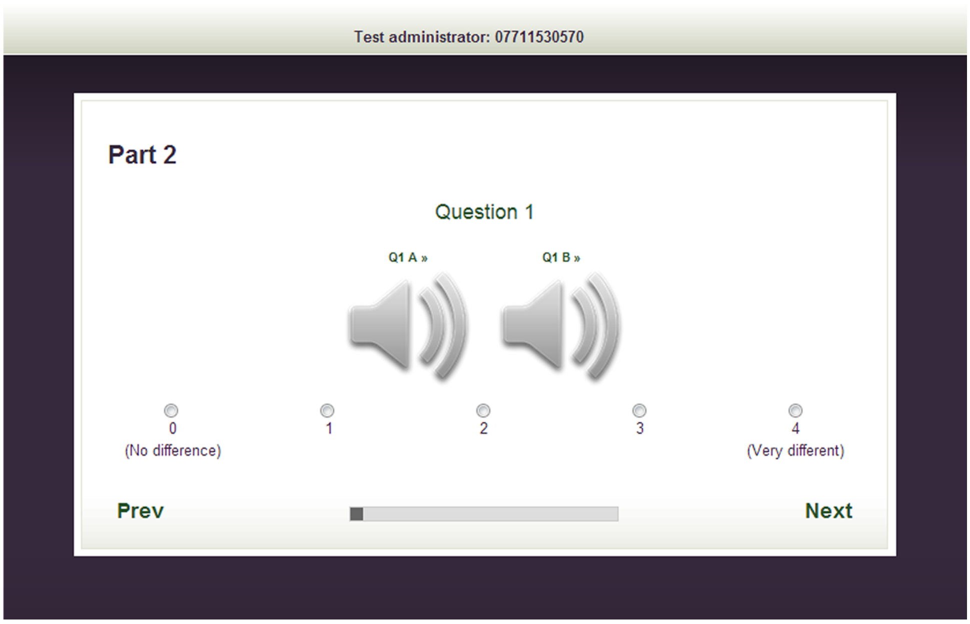
(d)


Test administrator: 07711530570
Part 1
Submit answers »
Prev
Next

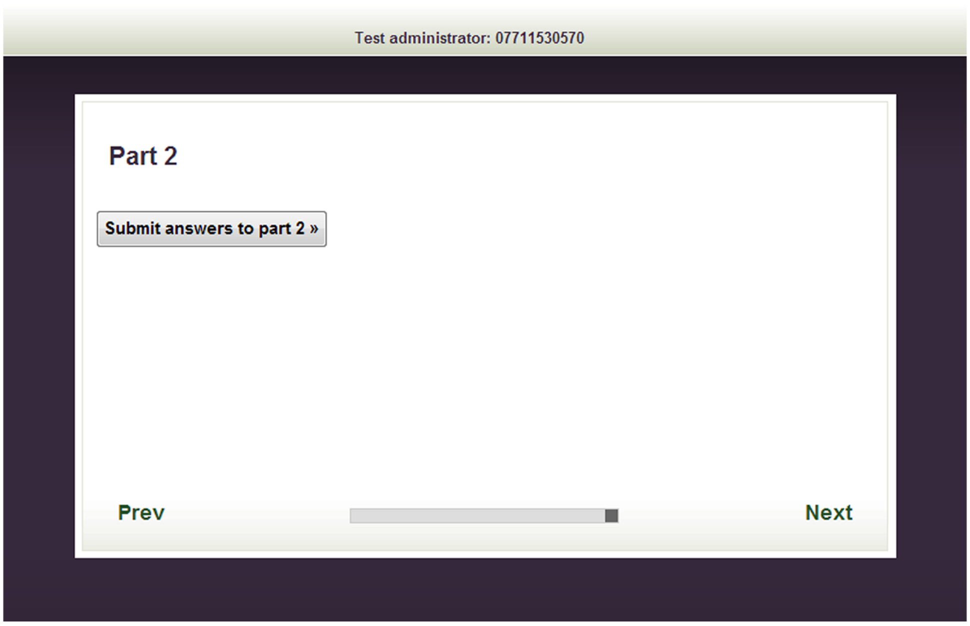
(e)


Test administrator: 07711530570
Thank you
You have completed the Part 1 successfully. You may now take a 5-10 minute break should you wish to. When you are ready, click `next' to proceed to part 2.
Next »

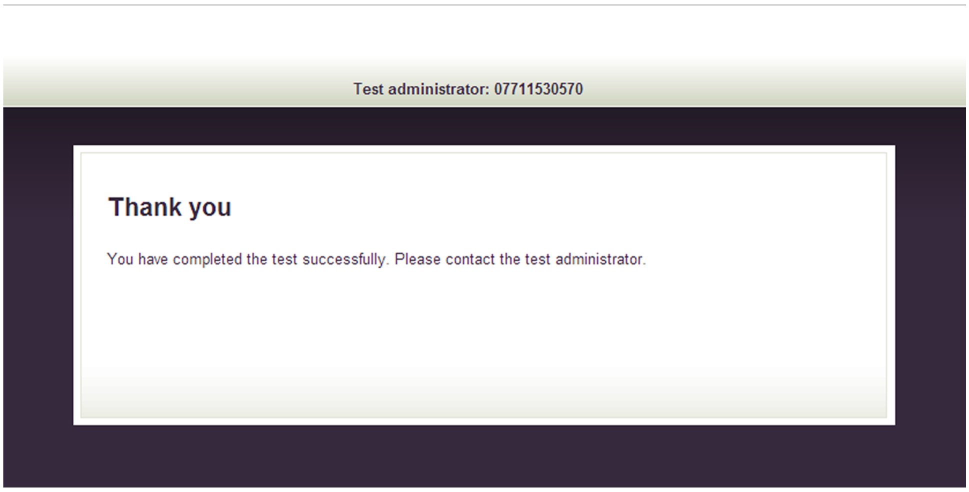
(f)


Test administrator: 07711530570
Part 2
Question 1
Q1 A »
Q1 B »
0
(No difference)
1
2
3
4
(Very different)
Prev
Next

(g)


Test administrator: 07711530570
Part 2
Submit answers to part 2 »
Prev
Next

(h)



(i)

9.5 Virtual Sound Walk Questionnaire

Virtual Soundwalk Experiment [Date] [Time]

Please answer the following questions as honestly as possible. You may leave blank any questions that do not apply. Feel free to seek assistance if you are unsure of anything. Should you require more space, please use the back of the questionnaire.

Pre-Soundwalk Questionnaire

Age group: (circle one)	15-24	25-44	45-64	65+	**Gender : (circle one)**	M	F
Profession:					**Nationality:**		

In your profession, are you involved with any aspect of urban planning and/or acoustics?

In what way(s) do you engage with sound in your working life?

What are your general impressions of the sound in UK urban areas?

Do you tend to view environmental sound as part of the landscape and akin to visual landscape, or do you consider it noise and something that should therefore be mitigated? Please comment...

On the Soundwalk

What were your impressions of the sound environment?

	3	2	1	0	1	2	3	
Comfort								Discomfort
Quiet								Loud
Harmonious								Disharmonious
Soft								Rough
Weak								Strong
Pleasant								Unpleasant
Warm								Cold
Unique								Common
Monotonous								Varied

What sounds can you hear? (You may list sources, or try to describe the sound if the source is not known.)

Your impressions of the soundscape

This space is for any general comments about the virtual sound environment.

Your impressions of the space

Based on what you have heard, what are your impressions of this area? For example, how do you imagine it to look? Is it somewhere you would feel comfortable to spend time? What might you do there?

Your feelings about the virtual soundwalk

This space is for any general comments about the virtual sound walk.

Technical comments

When you have completed the `soundwalk', you may record any technical observations or suggestions here.

Thank you for your participation!

9.6 Virtual Sound Walk Data

9.6.1 Written Feedback

Presented in Tables 9.14-9.19 are typical examples of the written comments that were made during the virtual sound walk experiments. All of the completed questionnaires are available on the DVD accompanying this thesis (Directory 9.7.4.2). The 'Group' column points to which version of the soundscape the participant was played. Group A were the group without the barrier insertion, while group B were played the version with the barrier simulation.

While every attempt has been made to ensure the following text is true to the original questionnaire text, in some cases some minor amendments have been made for ease of reading. These are either grammatical or syntactical are are not considered to have affected the meaning of the text.

Table 9.14: Question 3: 'What are your general impressions of the sound in UK urban areas?'

Group	Comments	Gender	Nationality	Age Range
A	'Generally very quiet. Many bird sounds everywhere. Car sounds are very common. Compared with where I'm from, UK urban areas enjoy a much quieter acoustic condition. '	M	Chinese	25-44
A	'Loud though not always in a bad way. Layered and various.'	F	British	25-44
A	'Generally annoying, but no strong opinion.'	M	British	25-44
A	'I generally find large cities very noisy. A friend lives next to a dual-carriageway - I find her house noisy, but she seems used to it. '	F	British	25-44
A	'I've never thought about it, but it does not disturb me.'	M	Greek	25-44
A	'I find areas with human voices worse than areas with traffic, which you get used to. My grandmother lived in a city centre and used to complain about the birds when she stayed with us. '	M	British	45-64
A	'Generally it is not too noisy or annoying.'	F	Greek	25-44
A	'I find they are mostly made up of traffic noise masking many other sounds. In parks they are generally different, and you can hear birds, people, and wind in the trees. Industrial sound can add to soundscape in some areas, though I generally prefer areas dominated by natural sounds. '	M	British	25-44
A	'They are OK, pretty standard.'	M	Dutch	15-24
A	'I get the impression sound has not been considered as a priority in many urban areas, but I cannot say I often really notice or am disturbed by urban soundscapes. '	M	British	25-44
A	'They can be stressful and take some getting used to...I am rarely in urban areas. '	M	German	15-24
B	'They are a melange of sounds that can make for a confused soundscape, but not always a loud one. For those used to it, the absence of noise can be an issue as much as its presence. '	M	British	25-44
B	'Usually manageable, but in cities with massive roads running through, like in the US, the noise can be draining. '	M	British	25-44
B	'Loud, complex, and multi-directional. Generally very broadband - from traffic rumble up to pelican crossings. '	M	British	15-24
B	'It is generally quite messy and noisy - sometimes too noisy. '	F	Thai	25-44
B	'In a busy city centre, the sounds are not pleasing to my ears. I enjoy being outdoors in a more 'natural' environment. '	M	Greek	15-24
B	'Depending on the city, it varies. Usually I find heavily populated areas with less open space louder due to transport. '	M	British	15-24
B	'Informative. '	M	British	15-24
B	'Generally quite noisy. '	M	American	15-24

Table 9.15: Question 4: 'Do you tend to view environmental sound as part of the landscape and akin to the visual landscape, or do you consider it something that should be mitigated?'

Group	Comments	Gender	Nationality	Age Range
A	'I enjoy environmental sound very much, apart from the traffic noise. High street sounds are very interesting to me as well.'	M	Chinese	25-44
A	'If in the countryside, I'd rather not hear man-made noise.'	F	Italian	25-44
A	'I view sound as part of the landscape - it adds to the experience.'	F	British	25-44
A	'I don't think general noise like traffic should be mitigated, but certain noises can be very annoying like drilling and mopeds.'	M	British	25-44
A	'I see it as part of the landscape. Most of the time it augments the experience of being in a place, although there are times when I would like it to be mitigated. Usually it's mechanical sounds I find annoying.'	M	Greek	25-44
A	'I often associate traffic noise with danger and pollution. On the other hand, occasional planes overhead are interesting rather than annoying. I find constant noise with no visual source irritating.'	F	British	25-44
A	'I believe environmental sounds are part of the landscape, especially if they occur naturally.'	M	Greek	25-44
A	'I love natural sound and tend not to like urban sound, but I think it's what you get used to.'	M	British	45-64
A	'I think indoor sound should be mitigated, but outdoors sound is part of the landscape. I think its importance is not always obvious until something spoils it.'	M	Portuguese	25-44
A	'Generally urban noise is familiar to me and I am used to it. I enjoy the countryside but I cannot stay there for long.'	F	Greek	25-44
A	'I generally do not have a problem and consider them part of the landscape. Disturbing noises are generally loud, temporary sounds with a strong high frequency content such as the machinery used in road works. Busy roads can also be disturbing.'	M	British	25-44
A	'I definitely consider it part of the landscape, although some of it may be unpleasant. Depending how prominent it is, it may or may not be a large part of the overall experience in an environment.'	M	Dutch	15-24
A	'I generally consider it to be part of the landscape, not as noise. I am not necessarily negatively affected by sound that is part of day-to-day urban living, but I do notice its absence when spending time in quiet countryside.'	F	German	25-44
A	'When it is particularly disruptive and prevents people working and living peacefully then I think it should be mitigated, but generally I view it as part of the landscape and it does not bother me.'	M	British	25-44
A	'I definitely feel it is part of the environment, but there are some places that should remain quiet, such as homes and private ground. '	M	German	15-24
B	'Very much part of the landscape in the majority of circumstances.'	M	British	25-44
B	'In general it is all part of the landscape, but traffic noise is offensive and should be reduced where possible.'	M	British	25-44
B	'Though annoying and noisy, I believe it is part of the landscape. Mitigating it is a good idea, provided no essential parts are removed.'	M	British	15-24
B	'I think sound, including noise is part of the landscape. If I stand in an open space and close my eyes, all noise definitely represents the environment surrounding me.'	F	Thai	25-44
B	'I definitely do not consider the sounds of the environment as noise.'	M	Greek	15-24
B	'I would argue that it is part of the landscape. I usually find places of visual beauty are accompanied by a supportive level of sound. I would expect high levels of noise in built up areas.'	M	British	15-24
B	'It depends on the sound...traffic, machinery and intrusive sounds should be countered if possible. Noise in an environment is as important, if not more important, than the visuals. '	F	British	25-44
B	'Without it, there would be little information about the landscape, and it would be hard to visualise. I am usually quite aware of my environment, so anything that does not take up my attention is a bonus.'	M	British	15-24
B	'Depends on the context, but generally I view sound as part of the landscape.'	M	American	15-24
B	'It is part of the landscape. You expect it to be noisy in a city, and in the countryside it is meant to be quiet, but it is not really something I have thought about.'	F	British	15-24

Table 9.16: Question 5: 'What were your impressions of the soundscape?'

Group	Comments	Gender	Nationality	Age Range
A	'I imagine there are many trees because of the bird sound. Traffic sound is acceptable but not pleasant...I think it sounds very common, with not much good or bad about it. '	M	Chinese	25-44
A	'It sounded like being at the edge of a park or playing field, with cars passing near. I felt that I was moving, and that I went into a tunnel. '	F	British	25-44
A	'Relatively varied. The traffic was present throughout but not unpleasant. '	M	Greek	25-44
A	'There was a whitish noise. To start with I was unsure what it was, but later I thought it was traffic. There were many familiar sounds. I was surprised by how much birdsong was present, I don't often consciously notice it in daily life. '	M	British	25-44
A	'At first I thought I was in the countryside, in a valley with rushing water, but later I realised the background 'noise' was traffic. '	M	British	25-44
A	'The continuous 'blur' of traffic sounds a bit like waterfalls, but the individual sounds are invasive and a bit threatening. '	M	British	45-64
A	'Largely dominated by traffic noise, though it was not immediately obvious what it was...at first I thought it was the sea. There's running water of some sort...a fountain or waterfalls. The traffic and water dominated the soundscape, while other sounds such as birds singing were calming and pleasant and did not sound as they would be expected to. '	M	Portuguese	25-44
A	' Sounds like a park. A relatively quiet area, quite peaceful and pleasant. There were no overly disturbing sounds or consistently annoying sounds. '	M	British	25-44
A	'Common and everyday...sounds that are expected, nothing unusual or out of the ordinary. Was hard to ignore background sound. '	F	German	25-44
A	'Sounded like a normal soundscape I would expect in a city park. '	M	German	15-24
B	'Generally soothing, though dissonant at first.'	M	British	25-44
B	'It was quite generic.'	M	British	15-24
B	'To begin with the background noise was kind of annoying, but after a while I got used to it, and felt it was flowing smoothly with the other sounds of the environment. '	M	Greek	15-24
B	'There was constant 'noise' throughout, although it did fade and return. Other sounds were heard through the haze of noise. Their clarity increased when this background noise was reduced. '	M	British	15-24
B	'Quite satisfying - would be peaceful without traffic and sirens.'	F	British	25-44
B	'There were conflicting sounds, like morning bird sounds and overhead cars, as if I were inside a tunnel.'	M	British	15-24
B	'Very pleasant. '	M	American	15-24
B	'My impressions would depend how it looks. It is not an unpleasant soundscape, and if I could be visually separated from traffic noise I would not mind it.'	F	British	25-44
B	'Sounds typical, but very bassy. The bird sounds are quite repetitive...I would not typically associate road noise and industrial sound with bird sounds. The background sound could be mistaken for a river or waterfall as it is quite repetitive. Might prefer a bit more variation. '	F	British	15-24

Table 9.17: Question 6: 'What were your impressions of the space?'

Group	Comments	Gender	Nationality	Age Range
A	'I imagine a street with terraced houses on one side, and alongside the houses there are many trees. There may also be an empty space such as a park nearby. There are not many people. '	M	Chinese	25-44
A	'An open space with trees around...maybe a big garden or park. A busy road nearby, and small streets around. I would not feel comfortable to spend much time here.'	F	Italian	25-44
A	'It felt pleasant and like a good place for walking...There was more traffic than I would like but that is inevitable in a city.'	F	British	25-44
A	'I imagine a large park in a city next to a busy road with lots of trees, maybe a fountain, and a playing ground. I'd feel comfortable to work, read, eat or sleep here.'	M	Greek	25-44
A	'I could not easily visualise the environment. It seemed quite a busy area.'	M	British	25-44
A	'There are trees, a football pitch, a road nearby, hedges, grassy areas and footpaths...I think it is a park.'	M	British	25-44
A	'It sounds like I am in a park, close to a highway or many roads. The sound was pleasing and not disturbing...I would definitely feel comfortable to spend time here, reading a book or relaxing.'	M	Greek	25-44
A	'I felt I was at the edge of an urban environment, at an interface between nature and the man-made...maybe a parkland. I think of a bleak landscape with roads, tarmac and flyovers, but there are also green areas. I think it is a place where people are on the go, rather than relaxing and 'being' here.'	M	British	45-64
A	'I imagine an urban park with trees, birds and small groups of people though not many. There's a busy road and a fountain behind the place where I am sitting. I imagine it to be a sunny day. I would not be uncomfortable to be here, reading a book or waiting for someone.'	M	British	25-44
A	'I had the impression I was in a park in the middle of a city...the 'life' of the park was around me, although I was not a part of it, and the city noise was further away.'	F	Greek	25-44
A	'Parkland surrounded by roads in an urban environment. There were a few people but not many, and mostly distant. The weather was dry and not too windy.'	M	British	25-44
A	'Due to the birds, it sounded like a small park, but because of the very high background noise, it also seemed to be in a city or next to a main road. For that reason, it did not seem like a very nice place to spend much time.'	M	Dutch	15-24
A	'A common urban area, close to a main road. Definitely a 'concrete' environment, but there must be some green as well...perhaps a few trees or close to a park - a green space within an urban area.'	F	German	25-44
A	'Sounds like a park in a city. It might be relaxing or stressful, depending on the exact location. I might go there to do sports, walk the dog, have a picnic or hang out with friends. '	M	German	15-24
B	'A suburban area - heavily honed with a lot of green space near a major motorway.'	M	British	25-44
B	'An urban park, wooded, and relatively near a main road. There may have been buildings surrounding it. If I were stuck in the city, I might spend time there, but otherwise I would go somewhere quieter.'	M	British	25-44
B	'A big inner city park with a small road 70-100 metres away. Not a great place to live due to the constant background noise.'	M	British	15-24
B	'A big park in the middle of an urban city. The park could be surrounded by trees.'	F	Thai	25-44
B	'I believe I could be listening through an open window, and that outside the house there is a place where people are able to stop and chat. Or I could be on a balcony near a not too busy street. There may be a kind of athletics pitch nearby.'	M	Greek	15-24
B	'I imagined I was sitting on a park bench on the periphery of an open space, next to a busy road. It was a blend of urban and green environments. After a short time, the traffic would become irritating, so I would pause here but not stay long as I would find it uncomfortable.'	M	British	15-24
B	'A seaside resort - quite busy - with sea and a beach on one side but a busy road very close. I felt it was somewhere to go and have fun rather than work, but busy rather than relaxing. I imagine a bustling environment down the road with shops and pubs.'	F	British	25-44
B	'I could be in a park. I felt like I could read a book, potentially relax.'	M	British	15-24
B	'By or in a park with trees, in a city area. I would play Frisbee or have a picnic here.'	M	American	15-24
B	'I imagine I am in peaceful countryside next to a busy road, but at times it seems to be a busy town.'	F	British	15-24

Table 9.18: Question 7: 'What were your feelings about the virtual sound walk?'

Group	Comments	Gender	Nationality	Age Range
A	'The bird sounds were pleasant...I could not tell whether the location was changing.'	M	Chinese	25-44
A	'I felt I was there. It felt very realistic.'	F	British	25-44
A	'I felt I was sitting still and that different things were happening in the same area.'	F	British	25-44
A	'I felt I was sitting and things were happening around me...It felt quite pleasing.'	M	Greek	25-44
A	'I enjoyed the experience - not sure I'd want to stay in the place, but am intrigued by how much nature is in an urban environment.'	M	British	45-64
A	'I had the feeling of staying in the same position while everything was moving around me. I think another participant's suggestion of a map of the route would have helped create the impression of walking.'	F	Greek	25-44
A	'Pleasant experience, not too short or too long. I had a sense of movement as the soundscape changed.'	M	British	25-44
A	'There was a very high level of background noise, and I am curious to know if this is actually a very noisy environment and we normally just ignore these sounds.'	M	Dutch	15-24
A	'Could picture the place and could disconnect from laboratory setting.'	F	German	25-44
A	'It was a very interesting experience, but I could not forget the fact that I was in an artificial listening space.'	M	German	15-24
B	'Ultimately relaxing - although sound is complex to ears at first, it becomes soothing.'	M	British	25-44
B	'It gave a good sense of place, and of being surrounded by things happening - envelopment.'	M	British	25-44
B	'It made me aware of how much noise there is in major cities, and made me think about how we deal with this.'	M	British	15-24
B	'I did not feel like I was walking, but that different environments were changing in time.'	F	Thai	25-44
B	'It was a nice experience overall.'	M	Greek	15-24
B	'There was a point where I felt I had moved from one situation to another of a less urban nature.'	M	British	15-24
B	'Feels realistic, as if I am somewhere else. I felt the space was changing, but I did not get the sense I was walking.'	F	British	25-44
B	'It was good, though some parts were quite repetitive. Some parts felt like it leapt into different environments.'	F	British	15-24

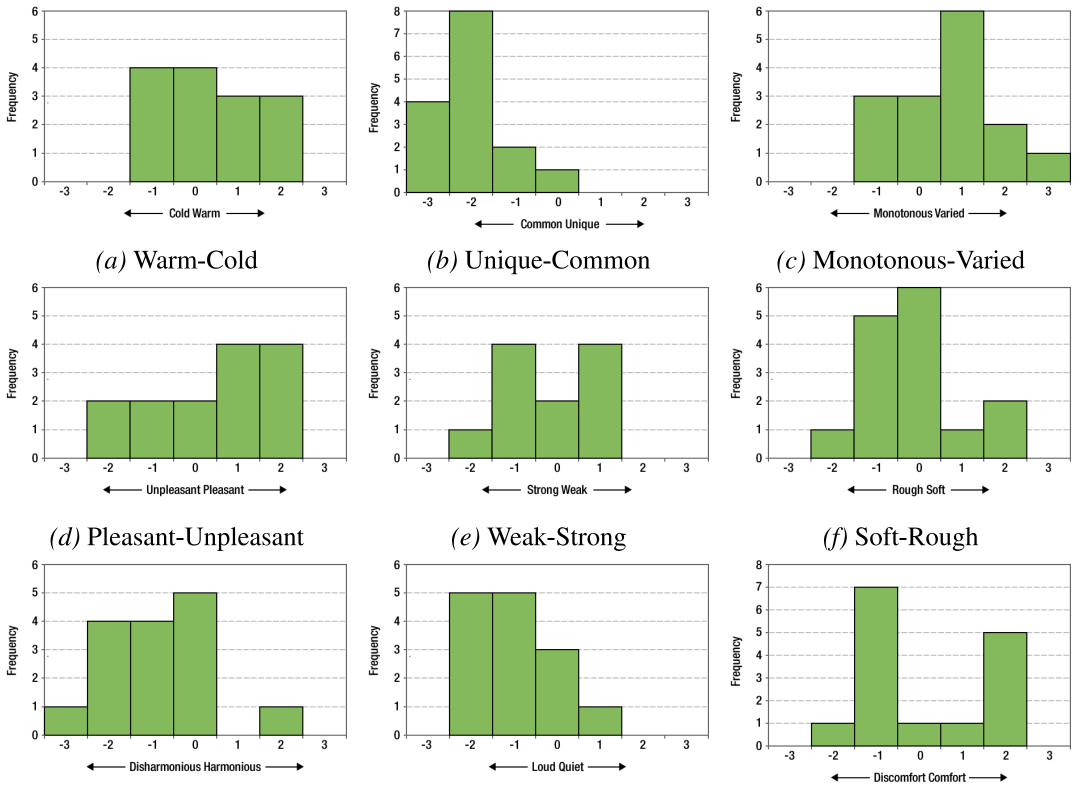
Table 9.19: Technical comments about the virtual sound walk.

Group	Comments	Gender	Nationality	Age Range
A	'I think that in a real environment the background noise won't be that loud. At a quieter volume it might sound more real. '	M	Chinese	25-44
A	'The level initially seemed high, but I think it was a normal amount of noise for a city. Being indoors makes it appear louder, as one would not normally hear those noises indoors.'	M	British	25-44
A	'I had a real sense of being enveloped in the space.'	F	British	25-44
A	'I noticed some phasing when I moved my head around.'	M	Greek	25-44
A	'The darkness was a good idea. I would not have the same impressions if the lights were on. The traffic noise stood out more than I expected.'	F	Greek	25-44
A	'Environment changed about a third of the way through...There seemed to be a change in amplitude but not quality of sound, in that the sounds were the same but not as loud.'	F	German	25-44
A	'Some transitions seemed to be too abrupt. I was not in the sweet spot though, so probably did not experience the technique's full potential.'	M	German	15-24
B	'The test environment was very good - it focussed the listener on the noise.'	M	British	25-44
B	'You need to stay very still for a stable image.'	M	British	25-44

9.6.2 Frequency Histograms for Semantic Differential Scales

The following frequency histograms resulted from the semantic differential scales, as rated by participants in the virtual sound walk.

Group A - No Sonic Crystal



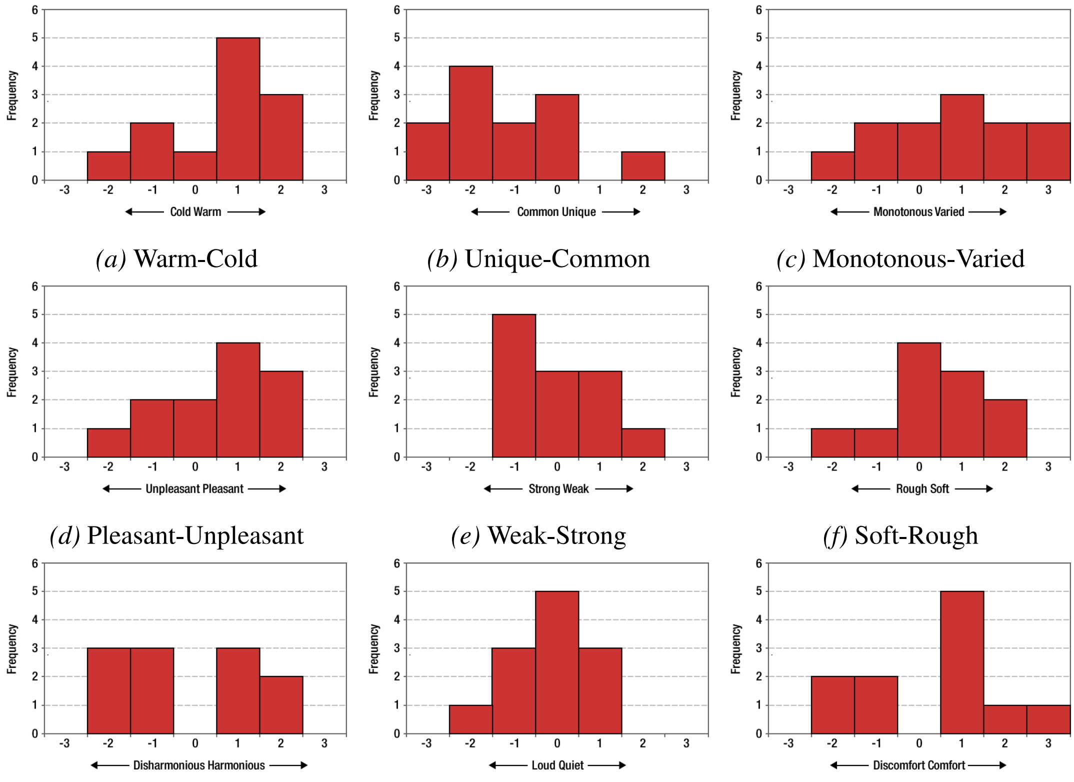
(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied

(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough

(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

Figure 9.11: Histograms corresponding to group A (no sonic crystal) of the virtual sound walk.

Group B - 2-D Sonic Crystal



(a) Warm-Cold *(b)* Unique-Common *(c)* Monotonous-Varied

(d) Pleasant-Unpleasant *(e)* Weak-Strong *(f)* Soft-Rough

(g) Harmonious-Disharmonious *(h)* Quiet-Loud *(i)* Comfort-Discomfort

Figure 9.12: Histograms corresponding to group A (no sonic crystal) of the virtual sound walk.

9.7 Digital Assets

The following items are included on the DVD accompanying this thesis. The indexes match those of the file directories on the DVD.

9.7.1 Computer Code

`3d-fdtd.txt` The C++ script needed to run the 3-D FDTD simulations included as a text file. To run, create a new project in Visual Studio and copy and paste the code into a source file. There are various simulation parameters that must be specified in the source file, including whether to use a transparent or a hard source. If the source is transparent, the user must record the correction signal first. A copy of should be made of this file as subsequent simulations will overwrite it. The first half of the script is concerned with defining the mesh, the second half actually performs the simulation. If simulating a 2-D SC, a pre-prepared csv file of the 2-D grid is required. This is then replicated vertically to create the 3-D mesh. The grid can be generated using this 2-D SC Matlab Script which was originally written for running 2-D simulations.

`2d-sc-geom.txt` This script was originally written for running 2-D FDTD simulations in Matlab. Because Matlab could not run the simulations in 3-D, the code was transported into C++ (see `3d-fdtd.txt`). The parts of the original code that generated the 2D-SC geometry were retained, since it need only be replicated vertically to create the 3-D mesh. To run it, copy and paste into a matlab script.

9.7.2 Video

The following videos are included in directory 9.7.2 on the DVD. The parameters used in the 3-D FDTD simulations all correspond to those used in the experiments presented in the thesis. The spatial resolution of the mesh is consistently 5mm, corresponding to a sampling frequency of 118,670 Hz. All depict a Ricker wavelet source. The space surrounding the inclusion is consistently 0.5m, meaning the size of the simulation domain varies depending on the inclusion geometry. Vertical and horizontal cross-sections through the simulation domain have been included.

`empty.avi` 3-D FDTD simulation in an empty mesh

`vert-empty.avi` Vertical view of 3-D FDTD simulation in an empty mesh

`4x2-0-03.avi` 3-D FDTD simulation of 4-by-2 sonic crystal array with 3cm radii

`4x6-0-035.avi` 3-D FDTD simulation of 4-by-6 sonic crystal array with 3.5cm Radii

`vert-4x6-0-035.avi` Vertical view of 3-D FDTD simulation of 4-by-6 sonic crystal array

`gyroid-0-1.avi` Gyroid simulation with ‘step size’ parameter of 0.1

`gyroid-0-15.avi` Gyroid simulation with ‘step size’ parameter of 0.15

`gyroid-0-2.avi` Gyroid simulation with ‘step size’ parameter of 0.2

`gyroid-0-25.avi` Gyroid simulation with ‘step size’ parameter of 0.25

`vert-gyroid-0-15.avi` Vertical view of gyroid simulation with ‘step size’ parameter of 0.15

`SonicCrystal.avi` Short time-lapse video showing the building of the sonic crystal used in the empirical experiments

9.7.3 Audio

9.7.3.1 B-Format Impulse Responses

In each of the following directories are the W, X, Y and Z components of the B-Format impulse responses.

empty Recorded in an empty mesh.

2D-SC Recorded with the 4-by-6 sonic crystal sample.

gyroid Recorded with the gyroid sample.

9.7.3.2 Sine Sweeps from Real-World Measurements

The following impulse responses were recorded during the real world measurements described in Section 6.3.7.

`SC_NO_FOAM_30CM.wav` Impulse response recorded 30cm from the edge of the sonic crystal.

`SC_NO_FOAM_50CM.wav` Impulse response recorded 50cm from the edge of the sonic crystal.

`SC_NO_FOAM_1M.wav` Impulse response recorded 1m from the edge of the sonic crystal.

`NO_SC_30CM.wav` No sonic crystal. Recorded 30cm from where the edge of the sample was before it was removed.

`NO_SC_50CM.wav` No sonic crystal. Recorded 50cm from where the edge of the sample was before it was removed.

`NO_SC_1M.wav` No sonic crystal. Recorded 1m from where the edge of the sample was before it was removed.

`SC_FOAM_30CM.wav` Impulse response recorded 30cm from the edge of the sample after foam wedges were inserted.

9.7.3.3 Original Sound Field Recordings

The original, unedited sound field recordings from which the extracts used in the composition were taken. The recordings were made on a 4 track Tascam recorder which outputs the 4 channels as two stereo sound files. These have been split into 4 mono files to give the W, X, Y and Z channels which make up the B-Format. For ease of listening, the stereo master mixes from the Tascam have also been presented here along with the B-Format signals.

loc-2 Recordings made in location 2.

loc-3 Recordings made in location 3.

traffic Recordings made between location 3 and the A660.

9.7.3.4 Perceivable Difference Test Material

This directory contains four sub-directories relating to the four audio extracts used in the perceivable difference tests. Within each sub-directory there are four versions of the relevant audio extract: A, B, C and D. A is the extract convolved with the empty mesh impulse responses, B with the 2-D sonic crystal, C with the gyroid, and D is untreated.

9.7.3.5 Virtual Sound Walk Material

`vs-control.wav` Stereo mix-down of the soundscape composition with no convolution or barrier approximations applied.

`vs-2D-SC.wav` Stereo mix-down of the soundscape composition simulating a 2-D sonic crystal insertion. Delivered to group B in the virtual sound walk.

`vs-empty.wav` Stereo mix-down of the soundscape composition where convolution was performed with impulse responses recorded in an empty mesh. Delivered to group A in the virtual sound walk.

9.7.4 Questionnaires

9.7.4.1 Real Sound Walk Data

9.7.4.2 Virtual Sound Walk Data

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