Body Space Interaction

Sound as a Spatial Mnemonic

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Declaration of Authenticity

I declare this thesis is my work and that all sources herein have been acknowledged.

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Abstract

"...in the white wake of my steps" is a sound art installation that seeks to heighten a participants awareness of the interaction between sound, body and space. The installation uses information from electronic sensors in conjunction with the amplified reverberations and resonances of the chosen space to create a layering of tones that are directly linked with ones physical presence within the space and the nature of the space itself.

This sonic mirror of the body as an object in space, seeks to promote contemplation upon the ideas we form regarding of our place in the world. This thesis explores the various concepts of space that influence everyday life, such as art, science and philosophy. By defining what we recognize as the spaces that make up our world, it is my hope that through interaction with the installation, participants will develop a personal understanding of the concepts reflected within it.

Acknowledgements

I wish to thank my friends and family for their love, support and encouragement throughout the years.

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Introduction

Throughout history, our perception of space has been shaped by the influences of art, science and philosophy. Consequently, mankind has sought to categorise the various spaces that form our ever changing world picture. Within this image lies the space of our physical environment, as described by mathematics and physics, the spaces that we construct around us in the form of architecture and the sociological systems developed within them, and the spaces as seen through the eyes of the artist and philosopher, whose representation seeks to find meaning of our role within it.

Is our environment merely a construct of our senses, or is it something fundamentally concrete?

In addressing these questions, I begin with a brief discussion concerning the evolution of various spatio-temporal concepts. What follows serves only as an introduction to the topics of later chapters and by no means constitutes the sum total of mankind's concept of space.

1.1 The Mobile and Purposeful Self

> *The whole history of life had been characterized by an incessant diversification and intensification of the interaction between inside and outside.*[1]

Historical and anthropological study has formed a vivid yet incomplete picture of how mankind has perceived the world. The physiological development of binocular vision, and the associated increase in the perception of depth, signalled a radical shift in mankind's spatial notions. The concurrent development of binaural hearing and bimanual touch helped to further enrich our sensory perception of the world[2]. Accompanying maturation of the sensory system were significant changes in the brain, namely of the cortex (that which is responsible for sensory processing and the generation of consciousness)[3]. A world experienced through our senses and enriched by emotive thought, began to high-light the significance of the body as a distinct object within space.

By observing objects in our environment, a sense of movement or stasis in relation to the body can be perceived. Moving from one place to another, one acquires a sense of direction and bodily orientation. Through movement mankind became distinctly aware of the notions of left and right, front and back, upper and lower. Space assumed *"a rough coordinate frame that was centred on the mobile and purposeful self."*[4]

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[1] Lefebrve, H. *The Production of Space.* p176.

[2] Akundov, M. D. *Conceptions of Space and Time.* p15.

[3] Carter, R. *Mapping the Mind.* p301.

[4] Tuan, Y. *Space and Place: The Perspective of Experience.* p12.

> *We have a sense of space because we can move and of time because, as biological beings, we undergo recurrent phases of tension and ease. The movement that gives us a sense of space is itself the resolution of tension. When we stretch our limbs we experience space and time simultaneously – space as the sphere of freedom from physical constraints and time as the duration in which tension is followed by ease.*[5]

On a primitive level the world can be experienced as the relative location of objects within the environment, which can in turn be thought of as an area defined by a network of places.[6] The world appears as a number of interconnected, locally continuous spaces, each associated with its own local time yet functioning as an interrelated whole.[7]

Through active influence upon the environment mankind began to gain an understanding of the essence and structure of objective reality.[8] What began as undifferentiated space slowly gave rise to a sense of place as it was endowed with value through meaningful activity.[9] The coordination of ones own spatial and emotive experiences, with those of others, also helped to develop the notion of time. The flow of time is evident in many facets of life, in the rise and setting of the sun, the passing of the seasons, and the cyclic nature of birth and death. This abstract notion of a homogeneous and continuous time brought with it, the ability to make educated predictions about the future.

Armed with the physiological, cognitive and societal developments gained over time, man evolved into an entity not only capable of directly affecting his environment, but one that is also distinctly aware of the temporal nature of the world. Man became conscious of the continuum of life that combines past, present and future.[10]

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[5] Tuan. p118.

[6] Ibid. p12.

[7] Akundov. p23.

[8] Ibid. p16.

[9] Tuan. p6.

[10] Akundov. p15.

1.2 The Birth of Measurement

The notions of space fostered through physical movement within the environment, contemplation, and memory, laid much of the ground work for the sophisticated spatial concepts that were to follow in the development of art, science and philosophy.

Although preceded by the notions formed by civilizations such as the ancient Babylonians, Egyptians, Hindu and Chinese, in regards to the western world, no single civilisation provided greater influence upon our modern concept of space than that of the ancient Greeks. Beginning with Thales (600 BC) and continued by Pythagoras (540 BC)[11] and his followers, the concept of space was strongly tied to the idea of matter (objects) and the void (air). The void was seen as *"constituting a kind of separation and division between things next to each other"*.[12] Space and matter were considered independent of one another. Space was the container of all matter and thus incapable of being within something else.

> *...space is therefore not some pure extension, lacking all qualities and force, but is rather a kind of primordial atmosphere, endowed with pressure and tension and bounded by the infinite void.*[13]

The Pythagoreans also believed in a mystical significance held by numbers, in particular, the role that mathematics and geometry played in the articulation of space. Upon these notions were built the beginnings of what would form the basis of future efforts in the measurement and quantification of space.

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[11] Whiteman, M. *Philosophy of Space and Time and the Constitution of Nature.* p19.

[12] Jammer, M. *Concepts of Space*. p7.

[13] Jammer. p8.

The concepts explored by the Pythagoreans opened the way for the development of atomism by scholars including Leucippus (450 BC), Democritus (420 BC), Aristotle (340 BC) and later by Epicurus (290 BC).[14] Although of differing approaches and opinion, each based their beliefs upon the presence of an indivisible unit of matter called the atom. Much debate ensued over the implications the atom posed upon the structure of the universe and the presence of the void. Some believed the universe to be a compact, continuous and unchanging plenum, while others maintained the existence of the void as simply unoccupied space.

> *Space, in this sense, is complementary to matter and is bounded by matter: matter and space are mutually exclusive.*[15]

Although imperceptible to our senses, the atom was seen to have a certain magnitude and extension. Hence, space was composed of an infinite multitude of atoms in combination. However, the problem remained of how one could apply a system of measurement to the unbounded container of matter that was space. Around 300 BC, a system of measurement was proposed by the Alexandrian geometer, Euclid[16], that would come to influence all measurement and study of natural law for the next 2000 years: Euclidean geometry.

Incorporating the theories and experience of his predecessors, Euclid presented his definitive treatment of geometry, spanning thirteen volumes, in his work titled *Elements*. Within it, Euclid presented a number of definitions, common notions and propositions that could be used to represent and analyse the abstract properties of space. Although too great to discuss at length in the scope of this introduction, in essence, the volumes detailed principles of abstract spatial concepts such as the point, line, and surface.[17] Armed with these fundamental notions, through a process of logic, one could irrefutably prove the relationship between geometric objects in two dimensions and ultimately apply them to the analysis of three-dimensional space.

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[14] Whiteman. p21-22.

[15] Jammer. p9.

[16] Greenberg, M. J. *Euclidean and Non-Euclidean Geometries: Development and History 2nd ed.* p7.

[17] For a detailed summary of Euclid's postulates refer to Faber, R. L. 1983. *Foundations of Euclidean and Non-Euclidean Geometry.* Marcel Dekker Inc. Appendix A. p291.

Although of significant scientific and philosophical importance, Euclidean geometry only began to touch on the science of solid geometry. It was not until the seventeenth century and the efforts of French philosopher and mathematician, René Descartes, that the use of a three-dimensional coordinate system for the analysis of space would become commonly accepted.

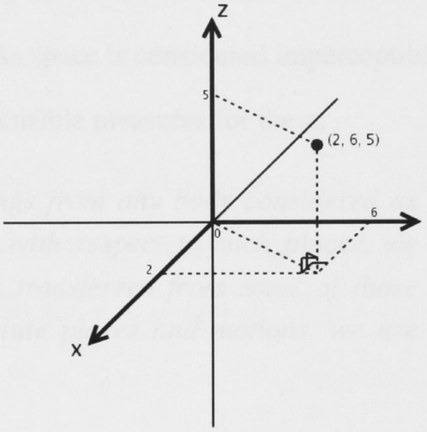

Figure 1: Cartesian coordinate system.

Descartes contributed greatly to the geometrisation of space with the introduction of a system based upon rectangular coordinates. The Cartesian system (Figure 1) allows the definition of a point in space by calculation of the perpendicular distances of the point from three intersecting axes.

Descartes regarded empty space as a non-material extension.

> *From the sole fact that a body is extended in length, breadth and depth we rightly conclude that it is a substance: because it is entirely contradictory for that which is nothing to possess extension. And the same must also be concluded about space which is said to be empty: that, since it certainly has extension there must necessarily also be substance in it.*[18]

This new approach to the measurement and analysis of physical space was further enhanced by the implications it posed for the notions of time. Throughout antiquity and the Middle Ages, time had been separated into two categories: absolute and relative. Descartes elaborated on these notions describing time as consisting of three components: eternity, which was associated with the nature of God; duration, which was intrinsic to the physical material world; and time, which was considered as a mode of thought imposed by man.[19]

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[18] Akundov. p108.

[19] Akundov. p109.

This concept of time was further developed by Sir Isaac Newton with the publication of Philosophiae Naturalis Principia Mathematica in 1687.[20] In it, Newton combines space, time, force and mass to form the fundamental concepts of mechanics. In contrast to Descartes' concept of spatial extension, Newton gives priority to mass as the fundamental property of matter, *"since every real body has both* (extension and mass) *and is inconceivable apart from either."*[21] As space is considered imperceptible to our senses, Newton suggested that we substitute sensible measures for them.

> *For from the positions and distances of things from any body considered as immovable, we define all places; and then with respect to such places, we estimate all motions, considering bodies as transferred from some of those places into others. And so, instead of absolute places and motions, we use relative ones...*[22]

However, these relative motions and spaces were still considered part of absolute space and time. In 1885, Ludwig Lange began to take steps in disproving the presence of absolute space with the introduction of an inertial coordinate system.

> *a coordinate system with respect to which three free particles, projected from a single point and moving in non-coplanar directions, move in straight lines and travel mutually-proportional distances. The law of inertia then states that relative to any inertial system, any fourth free particle will move uniformly.*[23]

The world of physics was ready to abandon the concept of absolute space and focus only upon relative spaces. While no longer an absolute entity, space remained Euclidean in nature. It would not be until the twentieth century with the development of Einstein's general theory of relativity and its application to the newly formed notions of non-Euclidean geometries that our present day concept of space would begin to take shape.

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[20] Hawking, S. W. *A Brief History of Time.* p5.

[21] Jammer. p97.

[22] Ibid. p98.

[23] DiSalle, R. *Space and Time: Inertial Frames.* The Stanford Encyclopedia of Philosophy (Summer 2002 Edition). [Internet ref.].

Since its inception, the application of Euclidean geometry to the measurement of space remained unquestioned. With the development of non-Euclidean geometry, the distinction between the concepts of physical and mathematical space became evident, i.e. there is no means of determining through mathematics and logic alone, which type of geometry does indeed represent the spatial relations among physical bodies. In order to validate the application of non-Euclidean geometries, measurements can be performed in the real physical world.

Engineering, architecture and art are all evidence that Euclidean geometry is useful in the articulation of space when the distances involved are small. It is when we deal with very large distances, such as those concerning the greater universe, that the accuracy of traditional Euclidean geometry comes into question. The results of experiments performed in the proof of Euclidean geometry both terrestrially and astronomically have always been inconclusive due to the inherent error within our measuring devices which are unfortunately, never 100% accurate.

Returning to the work of Einstein, in simple terms, relativity allows the combination of three spatial dimensions with a fourth of time, to form the geometry of space-time. Space-time is invariably affected by matter so that light becomes curved by the gravitational attraction of masses.[24] This suggests that physical space would be more accurately represented by spherical or elliptical geometry.[25]

Although the large scale nature of space is perhaps best described using a non-Euclidean 4-dimensional space-time geometry, our immediate surroundings are most conveniently described using the simpler Euclidean geometry.

> *Thus we put Euclidean geometry into the world approximatively for the purpose of measuring it.*[26]

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[24] Greenberg. p238.

[25] Faber, R. L. *Foundations of Euclidean and Non-Euclidean Geometry.* p284.

[26] Whiteman. p279.

1.3 From the Exterior to the Interior

With so much of the physical world quantifiable in the terms of mathematics and physics, how do we begin to measure the world of our individual experience? The answer lies perhaps in our various sensory and cognitive processes.

Our senses are the means by which we explore our environment. The sensations of light, sound, texture, flavour and scent combine to stimulate and enrich our perception of the world. Almost magically, our senses pull information from the ocean of stimuli that surrounds us, and the functions of our mind then endow it with meaning.

> *The senses don't just make sense of life in bold or subtle acts of clarity, they tear reality apart into vibrant morsels and reassemble them into a meaningful pattern.*[27]

Most often this meaning is useful, but at other times we can be deceived. Our senses serve to inform us, but simultaneously limit our understanding of the true nature of space. Our sense organs selectively extract and filter what is needed but also leave us with a tantalising glimpse of what remains unsensed. Undoubtedly, we rely heavily upon our senses. Without them we would be lost.

> *If the sensations of time and space can give man a biologically purposive orientation, this can only be so on the condition that these sensations reflect an objective reality outside man: man could never have adapted himself biologically to the environment if his sensations had not given him an objectively correct presentation of that environment.*[28]

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[27] Ackerman, D. *A natural history of the senses.* p xvii.

[28] Akundov. p14.

Sensory cognition is concretely organised in space and time. The formation of distinct and useful sense images demands intimate contact between the subject and objects of perception, through visual, auditory, tactile and other stimuli.[29] All forms of stimuli are absorbed over time, each operating upon different time-scales but with some degree of overlap present between the senses.

Our sense organs (excluding those associated with taste and smell which are determined by chemical processes) divide our environment into specific energy ranges (electromagnetic and mechanical waves), each being sensitive only to a particular range.

Our visual sense relies on the presence of light. The visible (light) spectrum extends from the extreme violet (400 nm) to the extreme red (750 nm) wavelengths, hence composing only a small fraction of the entire spectrum of electromagnetic radiations.[30] Radiations of wavelengths outside these limits cannot be seen by the human eye.

Our auditory sense relies on the presence of mechanical vibrations. The human ear is typically sensitive to waveforms spanning the range of 20Hz – 20 kHz, although the actual boundaries vary according to the individual.[31] Sounds above this range fall into the category of the ultrasonic. While imperceptible to the human ear, ultrasonic waves have found use within science and industry as a means of measurement and acoustic imaging. Conversely, below 20 Hz the continuity of sound breaks down and waveforms become perceived as distinct pulses, otherwise known as infrasonic impulses. Sounds within this range, such as percussive rhythms, become not only perceivable with the ear but also the body.

Our sense of touch is determined by changes in pressure and temperature. Our skin senses these changes in pressure via the action of mechanical vibrations, and temperature via electromagnetic radiation in the form of heat.

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[29] Akundov. p27-28.

[30] Alpern, M., M. Lawrence, and D. Wolsk. *Sensory Processes.* p14.

[31] Roads, C. *Microsound.* p7.

In addition to our senses, information regarding the position of our limbs, and equilibrium is received in the form of proprioception. The processes of such, involve the integration of information received through sensations from skin and muscles; visual and motor information from the brain; and regarding our balance from the inner ear.[32]

The time scale of our sensory processes is relatively short. As a matter of survival, once we have seen, heard, smelled, or tasted something, our cognitive processes quickly move on. Hence, our specialisation in detecting rapid changes in our environment somewhat diminishes our ability to perceive slower changes. E.g. at dusk when focused upon events immediately before us we are often surprised to realize how dark it has become once the distraction has passed.[33]

To gain meaning from events that occur over extended time periods we rely on cognitive processes that direct sensory information into paths of attention, abstract thought and memory.

During waking hours, our sensory system is bombarded with information from our surroundings. To find meaning, our brain must be held in a state of alertness. The cognitive processes of attention most likely began as a survival mechanism, warning us of danger present within the environment. Nowadays they are also regarded as necessary for abstract thought and the development of consciousness.[34]

As the environment is scanned for stimuli, the brain undergoes processes of arousal, orientation and focus.[35] It is these processes that signal the coupling of environmental stimuli with a sense of meaning and purpose. Upon reflection of our experiences, these processes begin to leave a lasting impression. The cyclic nature of our learning processes serve to strengthen these bonds, thus cementing our sensory and emotional experiences in the form of memory.

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[32] Carter. p187.

[33] Alpern, Lawrence and Wolsk. p126.

[34] Carter. p 304

[35] Ibid. p305.

Memories generally exist in two forms: semantic and episodic. Semantic memories are comprised of the things that we know independent of our personal relationship to them.[36] Although first learnt in conjunction with a strong personal association, unless these attachments hold particular significance, their retention begins to fade over time. Once the original personal association has dissolved we are left only with the bare fact.

Perhaps of greater interest here, are the role of episodic memories. These types of memories retain their personal attachment and are firmly entrenched within our experiences of space and time. Upon recollection, these memories serve to recreate the state of mind we were in when their impression was first made. The vivid nature of these memories is highlighted within a passage of Proust's *A la recherche du temps perdu.* In it, the protagonist recalls a moment of epiphany felt in the sensation of eating and drinking.

> *An exquisite pleasure had invaded my senses, something isolated, detached, with no suggestion of its origin. And at once the vicissitudes of life had become indifferent to me, its disasters innocuous, its brevity illusory – this new sensation having had on me the effect which love has of filling me with a precious essence; or rather this essence was not in me, it was me. I had ceased now to feel mediocre, contingent, mortal. Whence could it have come to me, this all-powerful joy? I sensed that it was connected with the taste of the tea and the cake, but that it infinitely transcended those savours, could not, indeed, be of the same nature. Whence did it come? What did it mean? How could I seize and apprehend it?*[37]

The feelings evoked by these kinds of memories show the powerful ability of our experiences in space and time to leave lasting impressions upon us. We gain a sense of how we felt in the first instance, even many years later, due to the deep emotional association of an event with a particular time and place.

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[36] Carter. p265.

[37] Proust, M. *A la recherche du temp perdu (Rememberance of Things Past: Volume I).* p48.

Thus we can see that our sensory system and cognitive processes effectively filter pertinent information from the vast resource that is present within our environment. By combining sensory impressions with the processes of abstract thought, emotional association and memory we begin to foster an awareness of an internal space. This could be said to constitute the realm associated with the philosophical notion of the self and what is regarded by some, as the spiritual space of the soul.

Analysis and discussion of the space of the interior has been a subject of continuous debate and development by the numerous philosophers and religious leaders that have populated our history. It is perhaps not in my interests to begin discussion upon these topics in the context of this introduction but instead, have it serve as the conclusion of a journey that began with the concepts that define our measurable, external world, and moved progressively inwards to a world enriched by the experiences that mark us as distinct entities within space.

2

Conceptual development and related works

In this chapter I discuss my developing interest in the various spatio-temporal concepts previously mentioned. I continue with a discussion of several artistic influences and the role that their works played in my own creative investigation of the interior and exterior.

2.1 In Contemplation of the Transient

It is the sound that surrounds us in life that intrigues me. The ebb and flow of waveforms that envelope us as we move through space. In this work, I have used sound as a medium to draw focus upon the role that our physical presence has upon the development of sound within space. It was my desire was to draw attention to the way our body acts as an object in space; essentially becoming a part of the space. By listening to the sound of their spatial presence and influence, it is my hope that participants will observe a distinct relationship between their body and space and that this will help them to reflect upon the interior space of their thoughts and memories.

The notion of using sound to demonstrate my ideas was fostered by numerous influences, both natural and man made. In particular, my appreciation of the rhythmic patterns formed by disturbances in bodies of water. Stones thrown into a mirror-smooth pond, rain wet trees dripping into puddles and the complexity of form evoked by wind playing upon its surface. Profound moments spent in observation of vibrations rendered visible. Sounds that are otherwise unnoticed until imposed upon another medium. In each case I was intrigued by the state of symmetry and structure formed by waves in resonance.

Concurrent to these moments of contemplation were the experiences gained when viewing the works of several artists. In particular, the works: *Aeriology - wiretap* by Joyce Hinterding and *A position between two curves* by David Cunningham. When first viewing and later reading at depth about the concepts explored by these artists I felt a strong desire to further investigate my own ideas and notions of space using the medium of sound.

2.1.1 Hinterding

In my contemplation of the movement of water, I began investigating the nature of waves, in particular the presence of different waveforms within seemingly empty space.

In the series of works, *Aeriology*, Hinterding explores the idea that our environment is filled with unseen sensory information. Hinterding investigated the notion of revealing the unseen through the use of large wire coils.

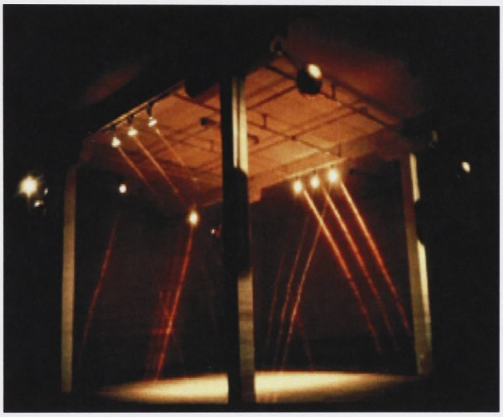

Figure 2: *Aeriology – wire tap,* Hinterding (1995)

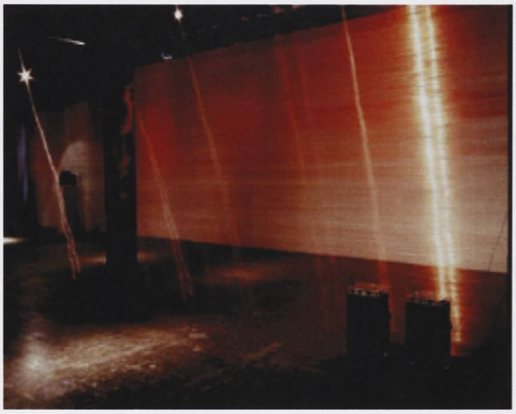

Figure 3: *Aeriology – i_tone,* Hinterding (1997)

> *....a project for an unfolding of the ethereal. 'A machine for a techne of the invisible'. These harmonising coils, reveal through sympathetic amplification activity of the unseen.*[38]

In these works, 20 km of wire is used to wrap the space. The large coils formed, act as antennae capable of resonating with passing electromagnetic waves. An electrical current is produced within the conductor, which in turn, induces a small voltage of the same waveform as the electromagnetic wave.[39]

> *....the coil resonates and reconfigures static as productive electrical potential.*[40]

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[38] Joyce Hinterding. [Internet ref.].

[39] Jacobowitz, H. *Electronics Made Simple.* p199.

[40] Aeriology : catelogue essay. [Internet ref.].

Works within the *Aeriology* series utilise different methods for uncovering these hidden waveforms. In the piece *wire tap*, the waveforms are rendered audible by tapping into the coil with a pair of headphones.

> *The electric nothing was made to speak, however inaudibly, in a crackling on the line.*[41]

Within *i_tone*, the waveforms are made visible by displaying them upon a cathode ray oscilloscope. The indistinct, but real, waveforms present within the space are rendered in a frequency range that our senses are capable of interpreting, providing us with a window into a hidden facet of our surroundings.

The notion of space being filled with waves, unsensed, but continuously surrounding us throughout our lives, held a particular fascination to me. With this concept in mind, I decided to further investigate the nature of waveforms and the mechanisms involved in our sensory interpretation of sound.

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[41] I_TONE aeriology : catelogue essay. [Internet ref.].

The matter that makes up our surroundings is complex in nature. Even seemingly empty spaces hide an intricate tapestry of interacting atoms and molecules. Consequently, even when regarding the simplest of sound-emitting objects (the hypothetical point source), the sound-field produced throughout a space rapidly increases in both dimensional and spectral complexity. As the waveforms encounter objects within their path, they undergo processes of absorption, reflection and diffraction due to their interaction with the physical properties of the object. Air alone, is capable of absorbing sound. The component gases and presence of absorbed water vapour, influence the spectral content of sounds travelling through it. Sound waves of high frequency are absorbed to a greater extent than those of low frequency. A common example of this behaviour is the sound of thunder. At close range it is heard as a sharp cracking sound composed of a wide spectrum of frequencies. As the sound expands outwards, the air, filters out the higher frequencies leaving only the low frequency rumble which is heard by distant listeners. This effect is further exaggerated when considering the transmission of complex sound waves through irregular spaces and materials.

> *Even in free space, where there is nothing else to interact with, real sound sources which have extended sound-emitting surfaces have a more complicated behaviour, since the radiation of sound will vary in a non-simple manner with both position and frequency.*[42]

As illustrated in Figure 4, interference occurs at locations where the original and reflected wave-fronts coincide. When two peaks coincide, the wave becomes amplified.

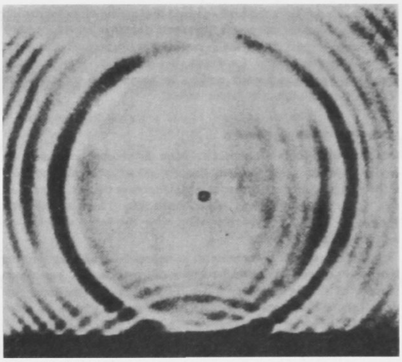

Figure 4: Interference pattern formed by wave-front colliding with plane surface. Analogous to a sound source close to the ground.

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[42] Malham, D. G. *Approaches to Spatialisation*. Organised Sound, Issue2, 1998. p167.

The same effect is observed with the coincidence of two troughs. Also, when a peak of one wave coincides with a trough of another, cancellation occurs (Figure 5).

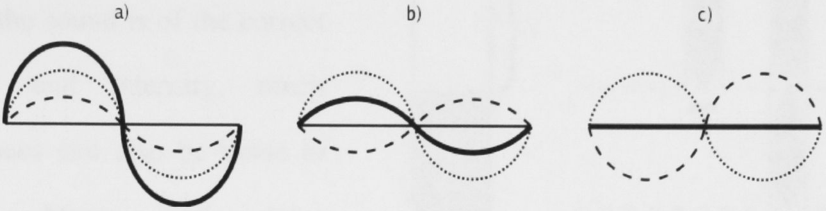

Figure 5: Waveform interference a) Reinforced b) Diminished c) Cancelled.

Rarely do sounds of the same frequency and intensity coincide exactly, resulting in complete silence or reinforcement. It is during the intermediary cases that waveforms begin to exhibit the characteristics that give a sound its particular tonal qualities.

Figure 6 illustrates what happens when waveforms of slightly different frequencies are produced together. As the two waveforms progress they shift in and out of phase with each other, producing rhythmic changes in the intensity of the combined sound. The effect can be heard as a slow pulsation, otherwise known as 'beating'.

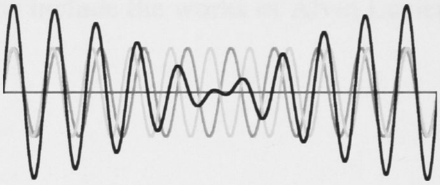

Figure 6: Reinforcement and cancellation between two waveforms of 10 and 11 Hz.

Just as matter is comprised of atoms, sounds can also be broken down into basic building blocks. These pure tones are known as sine waves. Although rarely heard in the natural world, the effect of many sine waves in combination can be heard in the form of resonance. Resonance occurs when a sound present within a space becomes reinforced by its own reflection. The numerous waves vibrate in phase with each other resulting in an intensification of the original sound. The resonant qualities of a space are largely determined by its structure and physical properties. The observation of resonance within pipes serves as a simple example (Figure 7).

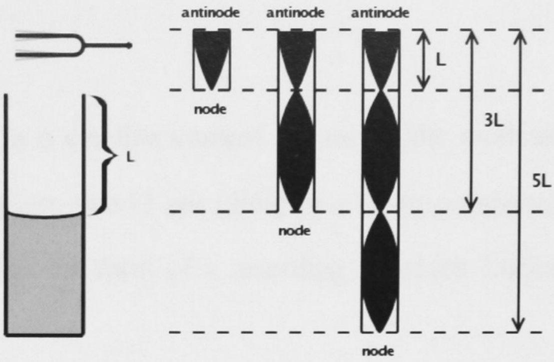

Figure 7: Resonance occurs within a partially filled pipe when minimum length of air column L = λ / 4.

Resonance is not only displayed within confined volumes. Provided the sound is of the correct frequency and intensity, much larger spaces can also be made to resonate. Most will have experienced the impressive sound heard within a cathedral or natural cavern. The shape and presence of hard reflective surfaces within these spaces serve to amplify and sustain sounds present within them.

Resonance within space has been explored by various artists throughout the last century. Those providing particular inspiration to me include the works of Alvin Lucier, David Cunningham and Bernhard Leitner.

2.1.2 Lucier

My interest in resonance and its use in a creative context led me to the work of Alvin Lucier; perhaps best known for his piece titled *I am sitting in a room*, composed for voice and tape in 1970. The piece takes the form of a recording in which Lucier narrates the following text.

> *I am sitting in a room, different from the one you are in now. I am recording the sound of my speaking voice and I am going to play it back into the room, again and again until the resonant frequencies of the room reinforce themselves. So that any semblance of my speech, with perhaps the exception of rhythm, is destroyed. What you will hear then are the natural resonant frequencies of the room articulated by speech. I regard this activity not so much as a demonstration of a physical fact, but more as a way to smooth out any irregularities my speech might have.*[43]

Lucier proceeds to re-record the piece as it is played back into the room. This new recording is then replayed and recorded again. As the process is repeated, the words slowly become unintelligible. The sounds once recognisable as speech become replaced by the *"pure resonant harmonies and tones of the room itself."*[44]

The piece unfolds in this fashion due to the characteristic resonance present within all architectural spaces. Certain frequencies become emphasised, and others negated, as sound waves undergo processes of reflection and absorption. The physical structure of the room, the materials from which it is made and the objects present within it all play a part in determining which frequencies become reinforced and those that are attenuated. In an effort to demonstrate this principle, Lucier specified that this piece need not follow its original from. The performance could take place in any room using any form of sound input. Thus, a completely difference piece would evolve depending on the nature of the space and the type of sound placed within in.

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[43] Lucier, A. 1970. *I am sitting in a room* for voice and tape, (45m 21s).

[44] Alvin Lucier – Wikipedia, the free encyclopedia, [Internet ref.].

Other intriguing facets of this work were the notion of time and role of memory within these resonating spaces. The acoustic influence of the space occurs on a very short time-scale. The lifetime of reverberations and resonances are short, yet in musical terms the piece evolves slowly, taking several minutes before an effect is recognisable. These short lived sonic events become exaggerated over time through the successive addition of layers. The processes of reinforcement and attenuation within each layer remain hidden when observed in the immediate sense, but slowly reveal their influence over time. The gradual evolution of the piece shifts our perception of the room's acoustic colouring into the realm of memory. Hence we perceive a change only by recognising that the piece sounds different to as it did several minutes ago.

As sounds pass from one time-scale to another their perceptual qualities also change. These changes can be categorising into nine different time-scales: Infinite, Supra, Macro, Meso, Sound object, Micro, Sample, Sub-sample and Infinitesimal.[45] Of particular interest, are those occurring on time-scales from macro through to micro.

Type of Space	Reverberation time (s)
Outdoors	0.0
Average bedroom	0.4
Theatre for speech	0.9
New Glyndebourne Opera house	1.3
Symphony Hall Birmingham (concert hall)	2.4
St. Paul's Cathedral	13

Figure 8: Reverberation times within various spaces.

The sound of reverberation occurs within the micro time-scale, extending from several hundred microseconds up to the duration of short sound objects (~100ms).[46] Although these initial reflections border upon the threshold of auditory perception, they become more recognisable as successive reflections combine over time. Figure 8 shows some typical reverberation times exhibited within different types of architectural spaces.[47] As sound waves bounce around the room, the combined reflections begin to be perceived within another time-scale: that of the sound object.

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[45] Roads, C. *Microsound.* p4.

[46] Ibid. p4.

[47] Reverberation in concert halls. [Internet ref.].

A sound object can be regarded as a form of elementary unit, much like that of the note, when speaking in the terms of traditional music structure. A sound object typically spans a range from a fraction of a second up to several seconds. It is within this time-scale that the sensation of tone begins. Although different for each person, a sense of continuity is generally established at ~30 Hz, with an impression of definite pitch occurring at ~40 Hz.[48]

Beyond that of the sound object, lie events which occur within a mesostructural time-scale. Within it, sound objects are grouped locally to form phrases, with a duration typically measured in seconds. It is on the meso level that important compositional elements occur. The basis of sequence, combination, harmony, melody and rhythmic development are all formed within this time-scale.[49]

Moving further, we reach a time-scale where locally arranged sounds are combined in the form of a macrostructure. These sounds are global in nature, corresponding to the notion of form and architecture with a composition. Hence, the macro time-scale is generally measured in minutes. Since macrostructural elements develop slowly over time, they are generally perceived in retrospect, through the action and recollection of our memories. The human brain tends to remember past events in pieces rather than in a continuous, linear fashion.[50] Consequently, the nature of memory is fraught with discontinuities and distortions. As in many facets of life, significance is placed upon sounds within musical structures by linking them with those memories that hold emotional significance.[51]

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[48] Roads. p17.

[49] Ibid. p15.

[50] Carter. P274.

[51] Roads. p11-12.

To bridge the discontinuities of our memories, a listener can be reoriented within the spatial framework of our environment through the use of sequence and repetition. When faced with unfamiliar territory we can quickly become confused and disoriented. Through repetition and the application of spatial markers we are able to gain an understanding of our surroundings. We learn to locate ourselves in space through repeated travel of the same familiar paths. Just as we are able to orient ourselves within physical space through our memories, so too can we locate ourselves in the temporal spaces created through sound.

The culmination of these concepts led me to examine the layering process exhibited within Lucier's work. Could it be possible to render the process in real-time? What would happen if the time-scale of acoustic reverberations were shifted to the point of continuous tones? It was these questions that led me to explore the works of David Cunningham.

2.1.3 Cunningham

When visiting London I had the good fortune of experiencing Cunningham's *A position between two curves* (Figure 9). The piece was a site-specific installation presented within the alcoves of the Manton Entrance at the Tate Britain Gallery as part of the exhibition *Days Like These, The Tate Triennial of Contemporary British Art*, 27 Feb - 26 May 2003.

Figure 9: *A position between two curves,* Cunningham. (2003)

The installation consists of two identical systems incorporating a microphone, loudspeaker and noise gate. The microphones capture the ambient sound of the space, which becomes acoustically focused by the two curved alcoves at either side of the building entrance. The sounds are amplified to a point where feedback is developed between the microphones and speakers. The two noise gates serve to cut off the sound once a particular volume is reached. As the loudspeakers are silenced, the space continues to resonate. Once the sound falls below the threshold of the noise gate, the system is reactivated and the process repeats. Thus, the sound produced is determined by the physical presence of the listener and the effect they have upon the acoustic qualities of the space.

> *He* (Cunningham) *provides the conditions and the equipment but it is the presence of people moving through the space which moves the air and alters the characteristics of the sound.*[52]

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[52] A position between two curves. [Internet ref.].

The curved surfaces of the two alcoves cause sound waves to be focused into an elliptical pattern similar to that shown in Figure 10. Sounds generated at one focal point are reflected and converge upon the other. The mirrored microphone/speaker pairs simultaneously amplify and re-project the sounds produced by each other. This process of recirculation aids the development of a feedback tone that is directly related to the patterns formed by interfering wave-fronts present within the space.

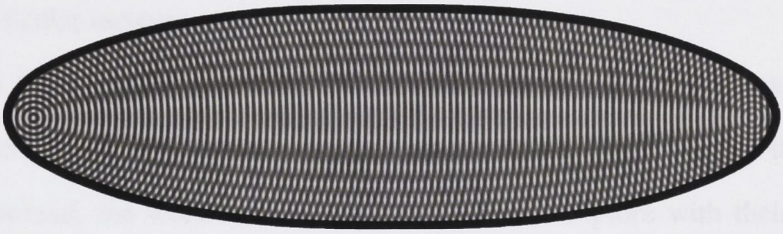

Figure 10: Elliptical waveform reflection.

Cunningham applied this system to several other spaces in the works titled *The Listening Room,* (Figures 11 and 12). The principle remained the same but instead of the highly focused effect produced by the curved alcoves, the system is used to emphasise the resonant frequencies present within everyday architectural spaces.

Figure 12: *The Listening Room,* University of Newcastle - Cunningham (2002).

Figure 11: *The Listening Room,* Ikon gallery, Birmingham - Cunningham (2003).

> *This process is modulated by very slight acoustic changes as people move around the room, by ambient sound, by humidity, by anything that causes air to move.*[53]

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[53] The Listening Room. [Internet ref.].

As the wave-fronts travel throughout the space they become altered by the physical properties of the air itself. Temperature, humidity and atmospheric pressure all play a part in changing the speed at which sound can travel through air. Even slight changes in these properties imparted by the presence of people, can alter the sound produced within it. This, coupled with the shape of the space itself and the types of sounds produced within it, help to develop a unique array of frequencies which are characteristic of that space at a particular moment in time.

Although the physical presence of participants within the work has a significant effect upon the sound of the piece, Cunningham refrains from terming the work as interactive. Instead, the works encourage participants to explore with their ears. It is only when the participant begins to actively listen, that the cause and effect of their presence becomes quantifiable.

2.1.4 Leitner

The notion that our physical presence actively influences the acoustic signature of a space, led me to the works of architect and artist, Bernhard Leitner.

Leitner's works deal essentially with the experience of choreographed spaces that are created and shaped using sound. Sound becomes a sculptural material used in the production of *"acoustic-haptic"*[54] spaces, such as those of *Sound Cube* and *Sound Space* (Figures 13 and 14). Emphasis is placed upon the movement and fundamental perceptual qualities of sound in space.

Throughout his works, Leitner uses sounds that that are perceived as *"Leading, pulling, swinging, opening, sinking, stretching, turning, shoving, ascending, (and) bouncing"*.[55] The sense of movement fostered in these sounds serve to emphasise the temporal nature of sound. Movement, sequence and repetition orient the listener within spaces that possess a distinct beginning, middle and end.

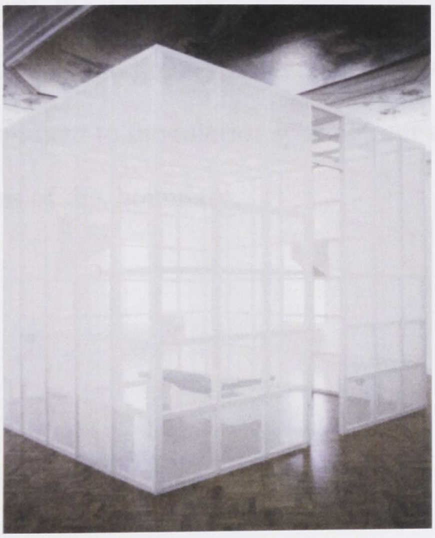

Figure 13: *Sound Cube* - Leitner (1981)

Figure 14: *Sound Space* - Leitner (1984)

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[54] Leitner B. *Sound:Space.* p23.

[55] Ibid. p23.

My interest in Leitner's work lay not so much in the methods employed, but the ideas and perceptions formed through the bodily experience of sound. Unlike many traditional artworks, Leitner's installations place an emphasis upon the role of the listener. This process of active listening fosters an awareness of the link between the physical presence of our body in space and our role as a perceiving entity within it.

> *The power of acoustics has its roots in the way a person is tied into the sound of a room, into the particular time of a room. Yet it is man himself who must make the room resound-with his steps, his speech, with any activity that generates sounds, even with his breathing. This interconnection between man and space, which is achieved with sounds and affects even our innermost being, is like a kind of dialogue which is determined by the acoustic premises. This dialogue enables us to experience ourselves in the sound of a room.*[56]

The act of focused listening, fosters within us, a heightened state of awareness. The creation of a sonic mirror allows us to picture ourselves listening to our own existence. Through contemplation and the development of a state of reverie we are able to gain a sense of the immensity of space. In the reflection of ourselves we internalise the grandeur of the greater universe. It is this shift of focus from the exterior space of our environment to the interior space of our thoughts that inspired me to investigate my own notions of this boundary.

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[56] Leitner. *Sound:Space.* p293.

2.2 Realisation through Technology

Prior to approaching the task of developing the installation, research was performed into how other artists had accomplished similar works through the use of technology. While by no means limited to, my focus was primarily drawn to Australian artists and instrument builders. By examining works produced locally, I was provided with valuable insight into technologies that were readily available and how I may be able use similar techniques in the realisation of my own work. The following sections discuss the works of three artists that provided particular inspiration during the development of *...in the white wake of my steps.*

2.2.1 Garth Paine

Composer, academic and installation artist, Garth Paine has been a prolific and passionate exhibitor of sound art works in Australia and abroad. In the course of my research, I was particularly drawn to the installation: *Map 1.* (Figure 15)

Paine proposes the notion that "*human expectations, frustrations, desires and experiences are usually expressed to the outside world as a physical or aural response.*"[57] Thus, through the use of sound, the work explores "*the ways in which humans develop and re-evaluate cognitive mappings of personal relationships with their environment.*" [58]

Figure 15: Paine demonstrating the interaction between movement and sound.

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[57] Ibid.

[58] Map1 Sound Installation [Internet ref.]

The installation makes use of the immersive and emotive qualities of sound to develop a rich, and continually evolving environment. This effect is enhanced via the use of a system capable of moving the apparent sound source through the space. In doing so, a relationship is formed between the position of the viewer and the paths taken by the sounds. The ability to move sound throughout the space imbues it with a physicality, i.e. the sound becomes as if it were another physical presence within the space. Building upon this principle, a range of aural qualities can be mapped to regions within the space, in relation to the quality of movement exhibited within that region.

The system (SoftVNS)[59] used to track the physical movements of viewers within the installation space provides information concerning the mass, dynamic and direction of movement within predefined regions. This data is passed through artificial intelligence software that is programmed to respond using diverse range of possible outcomes. The resulting control data is used to drive a realtime granular synthesis engine (using SuperCollider3 (SC3))[60] that processes the sounds made by those within the installation. Additional sounds, such as elements of text and prepared piano, are introduced according to the patterns of movement detected within the space.

The underlying source of inspiration gained from this work was the the idea of tracking and influencing the movement of participants through the use of sound. The concept of mapping of sounds to a position in space that can be directly linked with the memory of a action associated with it generation, appealed to me greatly. In doing so, viewers of the work could begin to forge links between their presence and the influence of their body upon the production of sound within that space. The work also presented solutions for the generation of control data (via video tracking) and the reconfiguration of captured sounds using granulation techniques. The concept of using sound that is continually captured, reconfigured and reflected upon the viewer was again reinforced.

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[59] David Rokeby : softVNS.html [Internet ref.]

[60] SuperCollider >> About [Internet ref.]

2.2.3 Rene Christen and Jasper Streit

Figure 16: Viewers interacting with the Listening Glass.

Whilst residing in Sydney I had the good fortune of viewing the interactive audiovisual installation by co-creators Rene Christen and Jasper Streit titled: *Listening Glass*.[59] I was also fortunate enough to hear Jasper Streit discuss the work at the Electrofringe festival.[60] What appealed to me most about the work was the reappropriation of the boundary between an enclosed architectural space and the outside environment.

Built into the front window of the First Draft Gallery in Surry Hills, the installation was designed to capture sounds from the outside environment, process the input, and relay the interaction through sound and video. The window glass (and security grill) acts as an interface through which viewers can create sounds for the instrument to process.

> *The work pulsates with the collective pulse of the public. Like a DNA donation to the evolution of the entity, affecting the work in such a way that it becomes unique through the interminable interaction specific to its location.*[61]

From a technical standpoint the system consists of:

- Two microphones – one a piezo transducer secured to the glass with silicon, and the other a vocal microphone placed through a crack in the wall.
- A computer that buffers the audio input, performs granular processing, randomisation, pitch shifting, and manipulation video content.
- Two speakers, attached to the window and used for sending vibrations through the glass to the street outside, and
- A projector that displays the video image.

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[59] :: Clatterbox :: [Internet ref.]

[60] Electrofringe [Internet ref.]

[61] :: Clatterbox :: [Internet ref.]

The visual component of the work consisted of three video loops controlled by Max/MSP and Jitter.[62,63] The manipulation of video images was dependent on the dynamic level of the captured sounds. The first loop which appears during moments of quiet, was based upon an extract from an implementation of John Conway's *Game of Life* (1970). The second consisted of visual white-noise and appeared during periods of medium dynamic level. The third loop produced intermittent pulses of white light during periods of high dynamics. All three loops were combined with a black mask to produce an aesthetic akin to looking through a microscope.

I was immediately intrigued by the use of granular synthesis and loops to form a sound/videoscape that was representative of what was occurring outside the gallery over the course of time.

> *The fragmentation of the incoming sound is important to the aesthetics of the soundscape - where the physical world sound is 'frozen', re-arranged, stretched and pitched; the original altered, yet still discernible.*[64]

The concept of taking a very small slice of time, repeating it and stretching it out to form a representation of past events helped to reinforce my own ideas regarding the use of sound to convey the effect of our actions upon the environment.

The other facet of this work that provided inspiration, was the idea of using sound as an interface between the body and environment. We are all capable of producing sound, whether it be through our voice or the movement of our bodies e.g. clapping, stamping etc. As such, we are all capable of influencing our environment through the production of sound. With this in mind, I began to investigate ways of representing the link between the environment and the influence our bodily presence has upon the production of sound with it.

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[62] Cycling '74: Max/MSP [Internet ref.]

[63] Cycling '74: Jitter [Internet ref.]

[64] :: Clatterbox :: [Internet ref.]

2.2.4 Alex Davies

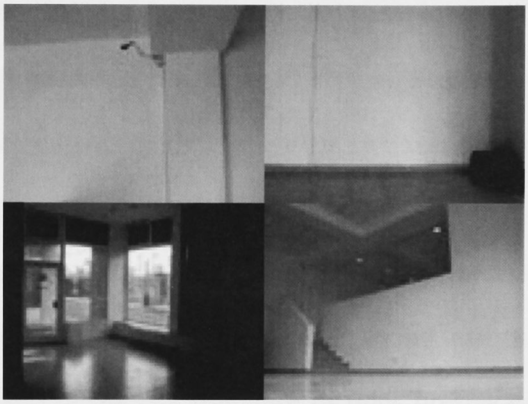

Figure 17: Sonic Displacement, Kunstraum (2002)

Sonic Displacement is an installation that explores the manipulation of sound in space and the affect it has upon our perception of the environment.[65] It operates by manipulating the live acoustics present within space using extended delay lines and the fragmentation of its sonic content. Sounds within the space are sampled and recirculated in an endless cycle.

The principle bares similarities to the Luciers *I am sitting in a room*, however instead of using analogue tape, manipulation of the sound is performed using the Max/MSP development environment.

In the case of the Kunstraum installation (Figure 17), the system captured (via a stereo microphone) a wide range of material over a period of approximately 4 hours, "*ranging from a subtle cough to the dense chatter of gallery patrons.*"[66]

These sound were fragmented, combined with the original audio content and then replayed back into the space via four speakers after a period of 3 minutes. Due to the proximity of the speakers to the microphone, the sounds were subsequently re-recorded causing "*subtle cyclic feedback loops.*"[67] Over time, the sound evolves as the sonic space is continually re-recorded, fragmented and transformed.

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[65] Alex Davies – installation [Internet ref.]

[66] Ibid.

[67] Ibid.

Although unable to witness the installation first-hand, Sonic Displacement did provide some insights into how I may approach my own work.

Like those used in *Listening Glass,* Davies had employed techniques of fragmentation and recirculation as the primary modes of sound manipulation. From this observation it became apparent that the software I was to choose for the creation of my own work, must provide the means for creation of similar (re)synthesis building blocks.

The use of delay lines to recirculate an audio input stream and the idea of cyclic feedback being capable of emphasising particular frequencies and/or sound events prompted me to think of ways in which I could develop a signature of ones presence, and yet at the same time show a relationship between the sounds we make and the type of feedback that is heard.

Another point of interest in Davies' work was the use of a stereo microphone. This prompted me to investigate the use of recording devices that mimic the action of the human ear.

2.3 Drawing Focus

Reflection, resonance, repetition and recollection. It is these processes that provided the conceptual ground work for the development of my installation work. Through the application of sound, I investigate a means of articulating the unseen forces acting within the spaces of our environment. Sound becomes the medium for exploring space.

In this work I place emphasis not so much upon the significance of architecture, but instead explore the use of its acoustic properties as a type of instrument. I attempt this by using the continuous recirculation of sounds formed through movement and the application of a non-contact object tracking system. In this fashion, the installation produces a sonic mirror of ones physical presence within the space. The use of resonance serves to condition our listening, and the ability to retrace ones steps sonically promotes contemplation upon what it means for us to exist within a certain point in space and time. Memory, action and expectation combine to orient the listener within the past, present and future.

With the elimination of a tangible interface, it is my hope that participants are able to interact with the work in an intuitive manner. As users gain familiarity, a sense of instrumental control can be established, thus extending the capacity for interaction with the environment and each other. The culminating aim of the system is the development of a heightened awareness of the seemingly empty spaces that surround us.

The remainder of this thesis details the various processes involved in the creation of the installation piece, *...in the white wake of my steps*, developed as a culmination of the ideas and concepts discussed throughout the preceding chapters. It is at this point that I recommend the reader of this text to view the contents of the attached DVD. Within it is video evidence of the installation in operation is shown in addition to a brief outline of the underlying technology used in its construction.

3

Hardware

This chapter details the selection processes, component prototyping and construction of the sensor system employed in this project. Each aspect of the development process was carefully considered against a strict criteria. In this fashion, I hoped to objectively reason the use of one particular method over another. The following discussion is based upon experiences gained through my own personal investigation and by no means constitutes the only method by which the concepts discussed throughout this thesis could be explored.

3.1 Selection criteria

To capture the interaction between body and space, an interface was required. To accomplish this, the decision was made to employ some form of object tracking system. Following a thorough literature search, a basic outline of the different types of tracking systems that could be applied to this project was formed.

The methods deemed most appropriate were as follows:

- Video tracking
- Ultrasonic
- Capacitive
- Infra-red
- Vibration and pressure switches
- Sound activated switches

Other methods were also considered, including electromagnetic tracking (such as FASTTRAK by Polhemus[68] and Motionstar by Ascension[69]) and radio frequency identification (RFID) tag based systems. Although effective, these systems were deemed unsuitable as I wanted participants to be free to move, unhindered by wires or the distraction of something they would be required to carry. A system was required that would provide accurate, responsive, and wireless tracking of people within three dimensional space.

Each of the above mentioned object-tracking methods possesses advantages and disadvantages. Accordingly, in selecting the system that would be employed in this project I compared each against the following criteria:

- Ease of use and installation
- Range and precision
- Reliability, and
- Cost

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[68] Fastrak. [Internet ref.].

[69] Ascension Products – MotionStar Wireless 2. [Internet ref.].

The system was required to be simple enough for me to apply without the need of highly technical knowledge or the assistance of a particular individual. The system also had to be flexible enough so that it could be easily adapted to the environment in which it would be tested, and ultimately installed. As I would be using spaces utilised by others, the system was required to be easy to assemble and remove. Also, installation of the system could not cause destructive alteration of the environment and could not rely on long-term fixture within the space. Excessive wiring or crowding of the space with equipment was undesirable due to its effect upon the movement of participants and the overall aesthetic of the piece. When using electronic equipment, powering the system is always an issue. Consequently, the system was required to be powered using existing outlets or to be easily adapted to battery operation provided the cost was not prohibitive.

The system was required to track objects over a reasonably large area, therefore it needed to possess an effective range and level of precision. Taking an average room to be 5 x 5m, the system was required to reliably track the movement of objects over a significant proportion, if not the entirety of the space. When designing interactive spaces, reference is made to three distinct ranges: intimate, body-sphere and architectural (spatial).[70] (Figure 18)

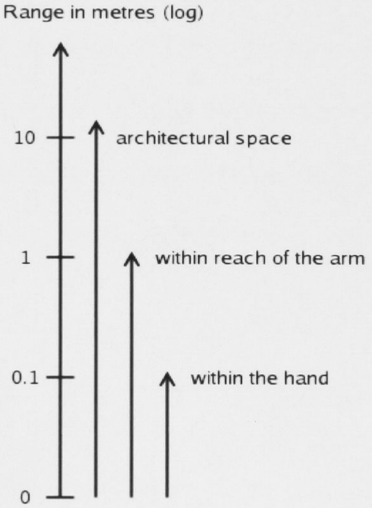

Figure 18: Designation of design space with increasing range

The needs of the project dictated a usable range spanning from body-sphere to architectural space, i.e. 0.5m and above. In an effort to minimise the number of discrete sensing modules, each was required to have a usable range and depth of precision over the entirety of the scale.

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[70] Harris, Y., and B. Bongers. *Approaches to creating interactivated spaces, from intimate to inhabited interfaces*. Organised Sound, Issue 3, 2002. p241-242.

Although it had not yet been decided whether I would be using the Musical Instrument Digital Interface (MIDI) or Open Sound Control (OSC) protocol, an estimate was made of the range and precision required according to the MIDI scale. The MIDI protocol provides MIDI control change (CC) messages ranging from 0 to 127. If the required range and precision could be attained using a protocol with low resolution, it could always be adapted to one of higher resolution if the needs of the project changed. For a detailed discussion of the MIDI and OSC protocols, refer to section 3.4.

The final consideration made was that regarding the cost of the system. Working with limited funds meant that I had to adhere to a strict budget. I calculated that the object tracking system could cost no more than $500AU. I felt that this was not an unreasonable estimate, and keeping within this figure would not effect the successful completion of the work.

3.2 Selection of Object-tracking Method

As previously stated, each object-tracking method that appeared applicable to this project possessed its own advantages and disadvantages. Due to the scope of each method, not all of the pros and cons could be explored to their fullest extent within the allocated time-frame. Therefore what follows is a detailed description of the methods and results obtained in my experience. In some instances, results obtained by others when implementing a particular method could not be reproduced by myself due to environmental differences, inexperience, lack of time, and/or financial restrictions.

3.2.1 Video tracking

An object-tracking method that has exhibited increasing popularity over the last decade is one based upon the use of video cameras. With the ever increasing processing power of personal computers, video tracking has become an effective method for tracking objects over moderately sized areas. Figure 19 shows a simple set-up for the tracking of objects within the camera's field of view.

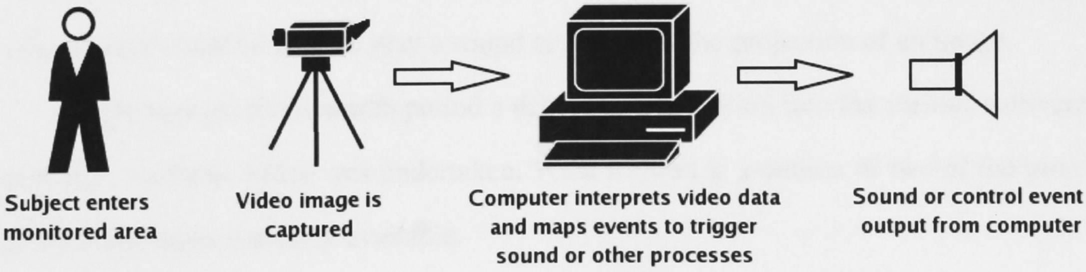

Figure 19: Basic video tracking system.

The system consists of a video camera, computer and processing software. Objects within the cameras field of view are captured and sent to the computer. The image data is processed by the computer, and a result returned depending on factors such as the objects size, shape, colour, position and speed of movement.

Various software packages are available that help to simplify this process, including:

- EyesWeb[71]
- BigEye[72],
- Isadora[73],
- Max/MSP, Jitter and its numerous extension libraries[74], and the
- PiDiP[75] extension for PD/GEM.

Each application employs its own method of interpreting and manipulating video data but all rely upon similar techniques. The most common approaches include:

- the use of a reference grid to measure position and displacement of objects,
- measurement of pixel differences (optical flow),
- measurement of colour and/or luminosity, and
- pattern recognition.

Regardless of what method is employed the end result is the production of data that can be used to trigger controllable elements of a software or hardware device. For example, the presence of a particular colour within a designated region of the captured video image could be used to play a sound or to trigger the projection of an image.

Throughout the research period a detailed investigation into the various software packages available today was undertaken. What follows is a outline of two of the more popular packages currently available.

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[71] InfoMus Lab – Laboratorio di Informatica Musicale [Internet ref.]

[72] BigEye. [Internet ref.].

[73] TroikaTronix. [Internet ref.].

[74] Cycling '74: Jitter. [Internet ref.].

[75] Index of /pd/pdp. [Internet ref.].

3.2.1.1 EyesWeb

EyesWeb was originally developed as a visual programming language tool for the Windows operating system, to aid research into expressive interfaces and interactive multimedia systems.[76] It was designed with a focus on the analysis and processing of expressive gesture in movement, midi data, audio signals, and music. It can be obtained free of charge from *http://www.infomus.dist.unige.it/EyesWeb/Eyw4New.html*

EyesWeb consists of a development environment and set of libraries that can be assembled in a fashion similar to other graphical programming environments such as Max/MSP and AudioMulch.[77]

The EyesWeb libraries include modules capable of performing the following functions:

- Input: via video devices, hardware sensors, audio, keyboard and mouse.
- Math and filters.
- Communication: via, MIDI, OSC, TCP/IP, serial, DCOM and scripting languages such as Visual Basic, Python, and JavaScript.
- Image processing.
- Motion analysis: (e.g., feature/blob tracking, movement and spatial usage).
- Output: visual, audio, and data.

The Expressive Gesture Processing Library is separated into three categories:

- Motion Analysis: modules for real-time motion tracking and extraction of movement cues based upon background subtraction, segmentation, centroid and optical flow techniques.
- Space Analysis: modules for analysis of 2-D spaces via a grid tracking method.
- Trajectory Analysis: modules for extraction of features from trajectories in 2-D spaces, i.e. trajectory direction, length, velocity, acceleration and curvature.

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[76] InfoMus Lab – Laboratorio di Informatica Musicale [Internet ref.]

[77] AudioMulch Interative Music Studio [Internet ref.]

3.2.1.2 Extensions libraries for Max/MSP/Jitter

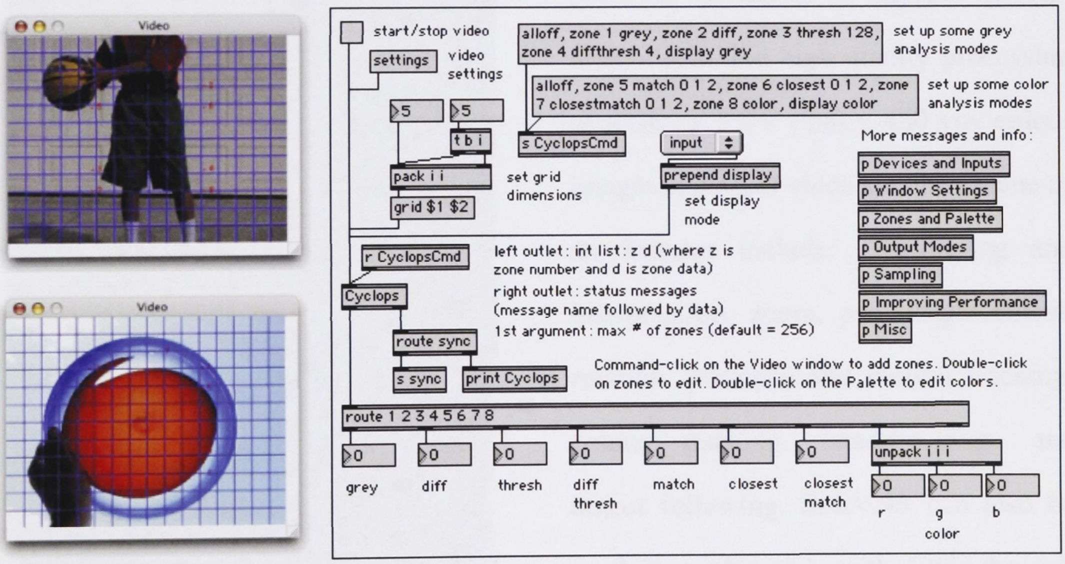
In the course of my research into computer vision based tracking methods, it soon became clear that the Max/MSP and Jitter patching software was a popular choice throughout the arts community. On its own Max/MSP and Jitter cannot be used as a complete video tracking solution. However, several extension libraries have been made available that provide this functionality. When used in conjunction with the standard Max/MSP and Jitter objects these libraries become capable of performing quite sophisticated video tracking functions. As with most patching software the end result is primarily left to the users creativity, ingenuity, and desired usage.

The following subsections provide a outline of the most widely used extension libraries and their key functions.

3.2.1.3 Cyclops

The Cyclops extension utilises a simple interface (Figure 20) through which the user can track objects with a video camera.[78] By analyzing the live image, the resulting data can be used to control MIDI, audio, video or any other controllable parameter accessible within Max/MSP.

Video is captured via a QuickTime input source, i.e. capture cards, USB and FireWire devices. The image is divided into a user defined grid of rectangular zones which can be analyzed for changes in greyscale (4 modes), colour (4 modes), threshold and/or pixel difference (used for tracking motion). Zones within the grid can be targeted individually to facilitate multiple types of analysis upon the same stream. The Cyclops objects are also fully compatible with Jitter, via the included jit.cyclops object.



alloff, zone 1 grey, zone 2 diff, zone 3 thresh 128, zone 4 diffthresh 4, display grey

set up some grey analysis modes

alloff, zone 5 match 0 1 2, zone 6 closest 0 1 2, zone 7 closestmatch 0 1 2, zone 8 color, display color

set up some color analysis modes

left outlet: int list: z d (where z is zone number and d is zone data)

right outlet: status messages (message name followed by data)

1st argument: max # of zones (default = 256)

Command-click on the Video window to add zones. Double-click on zones to edit. Double-click on the Palette to edit colors.

Figure 20: An example patch illustrating commands that can be used for video tracking.

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[78] Cycling '74|| Product cyclops [Internet ref.]

3.2.1.4 SoftVNS

SoftVNS 2 is a set of external objects for Max/MSP developed by artist David Rokeby, as part of his interactive sound installation series titled, The Very Nervous System (VNS) (Figure 21).[79,80] Within it, the installation employed the use of video cameras, image processors, computers, synthesizers and speakers to form a space in which the movements of one's body create sound and/or music.

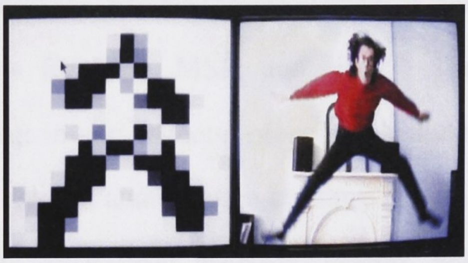
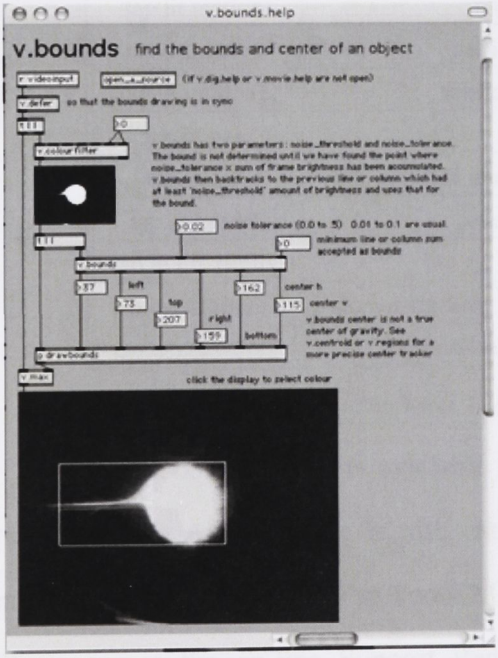


Figure 21: An image from The Very Nervous System and the movement tracked within it.



(if v.dig.help or v.movie.help are not open)

so that the bounds drawing is in sync

v.bounds has two parameters: noise_threshold and noise_tolerance. The bound is not determined until we have found the point where noise_tolerance x sum of frame brightness has been accumulated. v.bounds then backtracks to the previous line or column which had at least 'noise_threshold' amount of brightness and uses that for the bound.

noise tolerance (0.0 to .5) 0.01 to 0.1 are usual.

minimum line or column sum accepted as bounds

v.bounds center is not a true center of gravity. See v.centroid or v.regions for a more precise center tracker

click the display to select colour

Figure 22: An example patch utilising some of the SoftVNS objects.

Forming the core of the system, SoftVNS is said to be capable of real-time, stable and high quality processing of RGB or YUV (640 x 480 yuv colour images at 30 fps) video streams. Some of its features include: anti-aliasing and interpolation, zoom, pan, displacement, rotation, presence and motion tracking, colour tracking, head-tracking, and object following. SoftVNS can also be used in conjunction with Jitter through the provided compatible plug-ins (Figure 22).

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79 David Rokeby : softVNS.html [Internet ref.]

80 David Rokeby : Very Nervous System [Internet ref.]

3.2.1.5 Tap.Tools 2

In addition to the Teabox sensor interface hardware, ElectroTap also produces one of the largest collections of third-party objects for Max, MSP, and Jitter.[81] The collection is divided into categories including: audio effects processing, audio filter/spectral processing, audio analysis and conversion, audio plugin-building objects, XML file utilities, miscellaneous control objects, JSUI (user interface) objects, and perhaps of most interest in this case, the graphics, video and jitter objects (Table 1).

Object	***Function***
tap.colorspace	a colorspace conversion utility.
tap.jit.delay	a matrix stream (video) delay.
tap.jit.getattributes	an object to assist with building Jitter abstractions.
tap.jit.grayscale	reduction from 4-plane colour to 1-plane greyscale matrices.
tap.jit.sum	sums the contents of a matrix.
tap.jit.motion	motion detection and tracking.
tap.jit.motion	motion tracking with visualization.
tap.jit.motion2	motion detection, output as a matrix.
tap.jit.pan	a video panner
tap.jit.ali	matrix-based interpolation between multiple parameter sets
tap.jit.kernel	generation of 2D Gaussian kernels
tap.jit.proximity	a UI-building utility
tap.jit.colortrack	a color-tracking for up to 4 different colors simultaneously
tap.windowdrag	for creation of custom title-bars and drag regions

Table 1: Tap.Tools 2 objects for video tracking.

The Tap.Tools package is available in two versions: as a Artist license (a student version of the Artist license is also available) and a Pro license. The Artist license entitles the use of all of the Tap.Tools 2.0 components in Max, MSP, and Jitter and also allows access the on-line Support Resources. The Pro license provides all of the privileges of the Artist license as well as the ability to build collectives, standalone applications and Pluggo-based plug-ins. It also provides access to some of the Tap.Tools source code.

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[81] Electrotap [Internet ref.]

3.2.1.6 cv.jit

The cv.jit library is a group of objects that perform functions including: Statistics, Motion (optical flow), Image transformation, Image segmentation, and Image analysis.

The statistics objects allow the user to perform math based analysis (sum, mean, standard deviation, variance/covariance matrices) and transformation of video data.

Motion based analysis and their associated objects are based upon the notion of optical flow, defined as *"the measure of how much a pixel moves from one frame to another."*[82] This is based upon the assumption that each pixel within a frame was obtained by translating a pixel in the frame previous to it. Optical flow is therefore determined by measuring the displacement of all of pixels relative to the X and Y axes. Since there is no absolute method for determining whether a pixel in one frame corresponds to another, estimation is performed by two of the most common optical flow algorithms: the Lucas-Kanade technique, and the Horn-Schunk technique.

The image transformation objects are used to perform functions such as removing noise, or the smoothing of data. Functions similar other image editing applications are provided including: edge detection, dilation, erosion, open and close.

Image segmentation is designated as the task of identifying and isolating connected components, otherwise known as blobs. Using this premise, a blob can be thought of as a region of continuously ON pixels located within an area of OFF pixels. The detection of such regions is handled by the cv.jt.floodfill and cv.jit.label objects respectively.

Measurement, orientation and pattern recognition functions are handled by the image analysis objects. They are useful for quantifying the relationship between what is captured within a video image and the real physical world. Such information can be used as control data for other aspects of a Max/MSP patch.

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[82] cv.jit Documentation.pdf [Internet ref.]

3.2.1.7 Summary

As discussed, several options are available the purpose of video tracking. However, the principle issues concerning its use in this case were latency, and cost. Issues concerning accuracy and responsiveness due to system latency and environmental influences also become apparent. The vast amount of data that requires processing, places large demands on available CPU cycles. Although a modestly powered computer is capable of processing real-time video data, resources become burdened when running concurrent audio and MIDI I/O. To alleviate this, a second computer can be employed. Additional computers were available through the resources of the CNMA, yet my primary concern lay in the additional cost of video cameras.

In addition to system latency, accuracy and responsiveness due to environmental influences must also be considered. When tracking objects, strict control of the light within the space is required, as false triggering may occur when light levels are inconsistent, e.g. the presence of shadows or reflections. Many of these problems can be minimised through software but typically increase the complexity of the system and load placed upon the processor.

Despite its apparent problems, video based object-tracking appeared to be a suitable solution. At this early stage of the project I chose first to investigate other solutions and return to video tracking later if required. My attention now turned to an area in which I had some prior experience: electronics.

3.2.2 Capacitive

My first attempt at construction of a sensor device was one based upon the capacitive effect that is displayed by the human body when brought into close proximity to a metal plate or wire. This effect was first brought to light in the musical world with the development of the theremin in 1919[83]. The instrument was comprised of electronic oscillators that could be controlled by moving the hands over sensor plates; one affecting pitch and the other, amplitude.

Having recently purchased a theremin in kit form, I took the opportunity to apply the same principle to the development of an object tracking device. I began by taking the amplitude controller section of the kit theremin[84] (Figure 23), and proceeded with the addition of further circuitry. Figure 24 shows the completed circuit board incorporating two identical controller sections and power supply. Two long wires (~3m each) were soldered to the board to act as sensing elements. Also pictured in the lower half of Figure 24 is an early prototype of the MIDIsensor controller board, (refer to section 3.5 Construction of MIDIsensor controller for details).

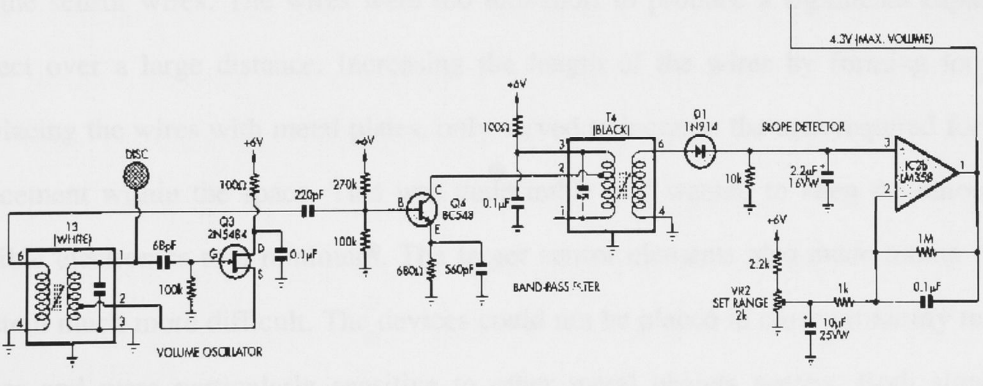

Figure 23: Excerpt of theremin kit schematic depicting volume controller components.

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83 Leon Theremin – Wikipedia, the free encyclopedia. [Internet ref.].

84 Clarke, J. *Silicon Chip Magazine - August 2000*. p16-24.

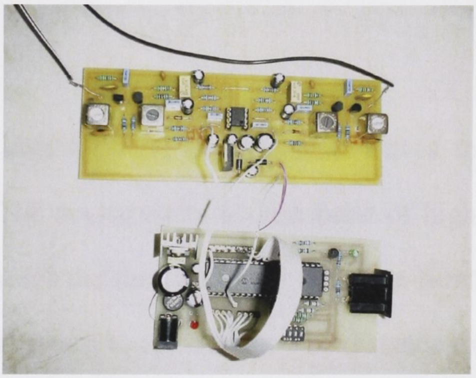

Figure 24: Capacitive sensor prototype with MIDIsensor controller

Initial tests showed moderate success. The device was able to output a steadily changing voltage in relation to proximity to the sensor wires. Unfortunately, the range was to limited to ~75 cm. Several changes were made in an effort to improve the range and linearity of the device. The power supply was altered to increase the gain from the op-amp, allowing the output to swing between 0-5V compared to 0-4.3V. This allowed the full MIDI scale (0-127) to be utilised without the need to alter the code or voltage references within the microcontroller. A further improvement was gained by substituting the silicon diodes within the peak detector circuit for germanium. The voltage drop across a germanium diode (~0.4) is slightly less than that of a silicon diode (~0.7) thus allowing more of the signal to pass from the output of the band-pass filter into the peak detector. This reduced the gain required of the op-amp, resulting in a slight improvement in output linearity.

The major restriction upon the range of the device was due to the width and length of the sensor wires. The wires were too thin/short to produce a significant capacitive effect over a large distance. Increasing the length of the wires by forming loops or replacing the wires with metal plates, only served to increase the area required for their placement within the space. This was undesirable as I wanted to keep the amount of visible electronics to a minimum. The larger sensor elements also made tuning of the system much more difficult. The devices could not be placed in close proximity to each other and were particularly sensitive to other metal objects nearby. Both situations produced erratic output from the device.

Although performance was ultimately unsatisfactory, the development process provided technical knowledge that would prove useful in the design of other devices.

3.2.3 Ultrasonic

In its simplest form, an ultrasonic sensor uses high frequency sound waves to measure distances and perform object tracking. The device emits a short burst of high frequency sound (typically 40-100kHz) and measures the time taken for the wave-front to propagate outwards, reflect from the target and return to the device. The distance to the target object can be calculated using the following equation:

Distance to target = (time of flight x speed of sound in air) / 2

Accurate measurements can be made using this method, but the process does possess some significant disadvantages. The main source of error is due to reception of multiple reflections. If a sound wave reflected from an object close to, or in the path of the target, is received before that of the intended target, a false reading will occur. Problems also occur when using multiple ultrasonic devices within the same area, as pulses transmitted from one sensor can be received by another. This effect can be minimised by ensuring that only one sensor is capable of transmitting and receiving at a time. Switching between sensors can be accomplished electronically but this method only serves to decrease the rate at which measurements can be performed.

Many commercially available systems utilise multiple receiver devices to triangulate the location of the target using sophisticated digital signal processing (DSP) algorithms. Although much more accurate and reliable, the implementation of such a method at this point in time was beyond the scope of my programming experience.

Despite the apparent disadvantages, work began on the construction of an ultrasonic based sensor device to test these issues first-hand. Internet and literature research yielded a number of possible solutions based upon discrete transmitter/receiver pairs. Another possible solution was the use of a pre-built Polaroid® 6500 ultrasonic ranging module. The module has been a popular choice within the amateur robot building community and is available for purchase or can be salvaged from discarded Polaroid® cameras (Figures 25 and 26).[85]

Figure 25: Various Polaroid® cameras that utilise an ultrasonic ranging device.

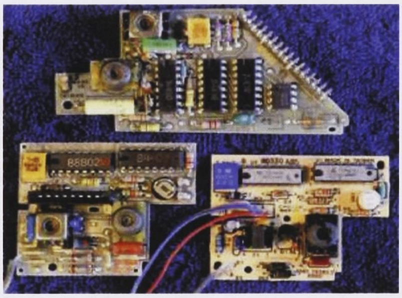

Figure 26: Polaroid® ultrasonic ranger modules.

Although the Polaroid® module seemed an attractive solution, when activated, a clearly audible click can be heard from its transducer. In some cases the reflected wave can also be heard. The high frequency sound is made audible to the human ear as it undergoes absorption and diffraction by the material structure of the room and objects within it.

After much consideration a decision was made to built a sensor device from scratch. Work began on the construction of a simple proximity detector based upon a continuously, pulsed transmitter (Figure 27). Figure 28 shows the ultrasonic receiver amplifier based upon a design for an ultrasonic ruler.[86] Figure 29 shows the completed prototype device. Note the re-use of the peak detector circuit from the capacitive sensor.

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[85] Polaroid Sonar Ranger Modifications. [Internet ref.].

[86] Dawson, C. *Ultrasonic rule with digital readout*. Electronics Australia. p77.

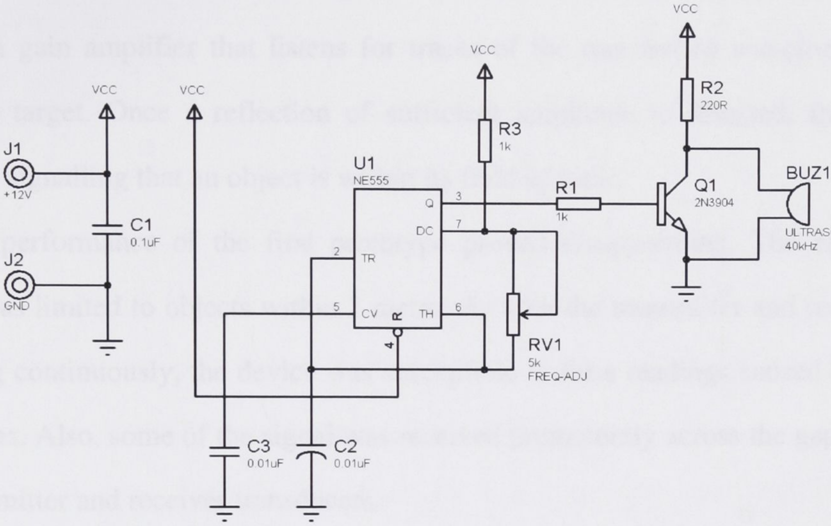

Figure 27: 40 kHz ultrasonic transmitter.

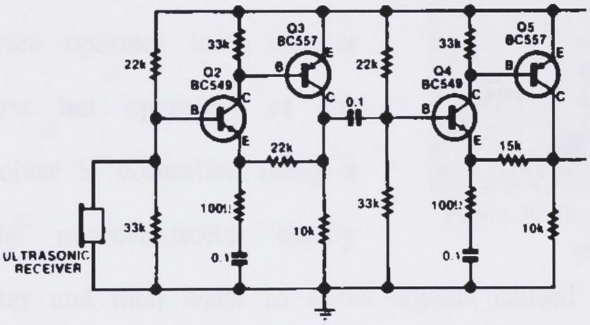

Figure 28: Ultrasonic receiver.

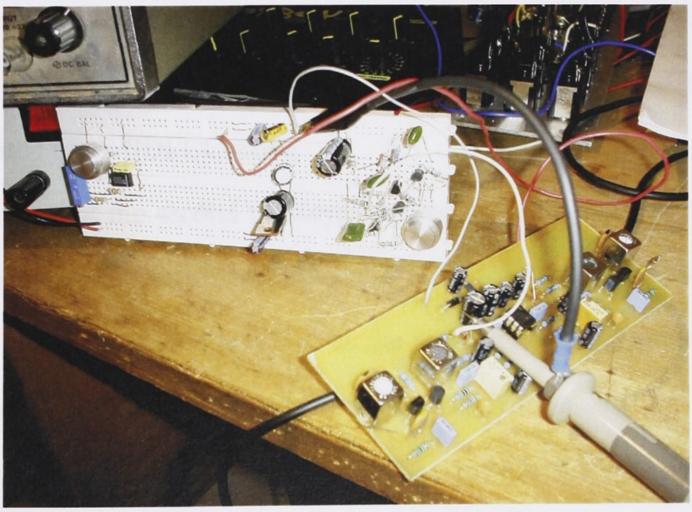

Figure 29: First ultrasonic sensor prototype.

The device emits a continuous square wave at a rate of 40kHz. The receiver consists of a high gain amplifier that listens for traces of the transmitted waveform reflected from the target. Once a reflection of sufficient amplitude is detected, the sensor is triggered signalling that an object is within its field of view.

The performance of the first prototype proved disappointing. The range of the device was limited to objects within 1 metre. As both the transmitter and receiver were operating continuously, the device was susceptible to false readings caused by multiple reflections. Also, some of the signal was received prematurely across the gap separating the transmitter and receiver transducers.

A second prototype (Figure 30) was constructed in an attempt to address these problems. The device operates in a similar fashion to the first but operation of the transmitter and receiver is controlled using a microcontroller. The microcontroller briefly pulses the transmitter and then waits to allow signals caused by ringing of the transducer and propagation across the gap separating the transmitter and receiver to pass. The microcontroller continuously polls the output of the receiver until an echo signal is detected. The time taken between transmission and reception is calculated and the result converted to a MIDI CC value.

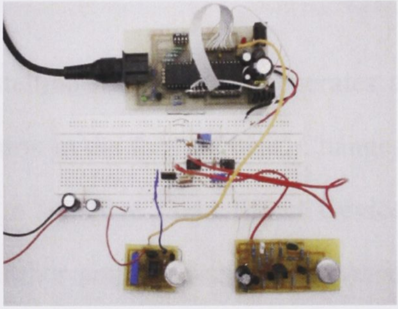

Figure 30: Second ultrasonic sensor prototype.

The performance of the second prototype device was an improvement over the first but still produced unsatisfactory results. False readings was greatly reduced but the detection range was still limited to ~1 m as the strength of the transmitted signal was insufficient to cause adequate reflection from objects other than a plane surface. Several attempts were made to increase the strength of the transmitted signal using high voltage pulses but adequate results could not be obtained within the allocated time-frame.

3.2.4 Infra-red

The properties of infra-red (IR) light (750-1000 nm)[87] can be exploited to create sensor devices. As infra-red light cannot be detected by the human eye, it can be used to cast beams of light that can be detected by electronic devices but otherwise remain invisible. The human body also emits IR radiation in the form of heat (~940 nm). These properties can be used to build sensors that react to the presence of a person passing within the field of view of the device.

IR sensors can generally be separated into two categories: pyroelectric, and transmitter/receiver pairs.

Pyroelectric sensors (Figure 31) consist of a crystalline material that generates a surface electric charge when exposed to infra-red radiation in the form of heat. Changes in the amount of charge produced is measured using a sensitive Field Effect Device (FET) built into the sensor.[88] The resulting signal is further amplified and conditioned with additional electronics. Pyroelectric sensors generally incorporate a filter window that restricts incoming radiation to wavelengths between 800-1400 nm. With the addition of shaped lenses (Fresnel lenses, Figure 32), pyroelectric sensors can be very effective at detecting the presence of human body radiation. When an individual moves within the sensors field of view, a pulse is output from the device that can be used to trigger other electronic devices.

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[87] Infrared – Wikipedia, the free encyclopedia. [Internet ref.].

[88] Pyroelectric Infrared Sensor. [Internet ref.].

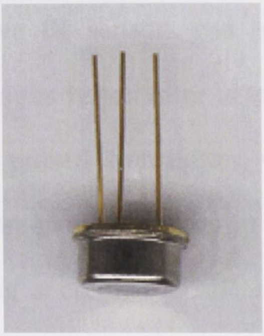

Figure 31: Pyroelectric sensor.

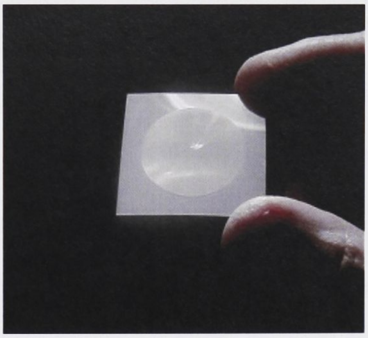

Figure 32: Polymer Fresnel lens.

Upon investigation of readily available PIR security sensors, it was found that all utilised a trigger pulse type output. While effective at triggering an on/off type event, these devices were unable to give an indication of proximity. Consequently, my attention turned to another form of IR sensor: transmitter/receiver pairs.

Transmitter/receiver type sensors function in a similar fashion to pyroelectric but instead of relying on body heat, a pulsed IR light beam is emitted by the device. A portion of this transmitted light is reflected back towards the device by objects entering the beam path. The reflected light is measured using a photo-diode or photo-transistor.

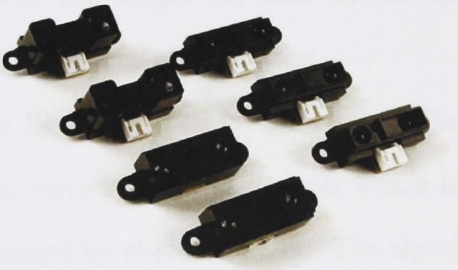

Figure 33: Various Sharp IR sensor modules.

Research provided information on the use of commercially available modules, and devices constructed using discrete components. The Sharp General Purpose Type Distance Measuring module[89] (Figure 33) has found widespread use throughout the robotics community. The module features a digital output that can be used to trigger other devices. The major disadvantage of these modules is their limited range (up to 80cm). The range can be increased with the use of multiple transmitting LED's and collimating lenses but better results could be obtained through the construction of a purpose built device.

---

[89] Demystifing the Sharp IR Rangers. [Internet ref.].

Effective IR sensors can be constructed using discrete components. Testing of various designs began after obtaining several IR LED's and frequency matched photo-diodes. A pulsed transmitter was constructed utilising a modified version of the ultrasonic transmitter shown in Figure 27. Pulsed IR light is used for several reasons. When using a continuous IR source, the effectiveness of the device is substantially reduced due to its increased susceptibility to interference caused by visible light. On the other hand if the IR source is rapidly switched on and off, the receiver can be tuned to detect only the IR light emitted at the frequency of the transmitter. By restricting the bandwidth of the receiver, greater levels of amplification can be obtained thus increasing the range of the device. Another benefit of pulsed IR light is that a beam of greater intensity can be produced by carefully controlling the amount of time that the LED emitter stays on and off. A greater current can be passed through the LED by ensuring that the amount of time spent in the on state, is considerably shorter than that spent off. The short on time allows the LED to be driven at a current higher than that for which they are rated for continuous operation. The longer off time allows for sufficient cooling of the LED material thus preventing damage of the device.

Figure 34 displays schematic diagrams for the IR sensor used in this project. The device functions as follows. U2 produces a square wave at a rate of 10 kHz which is used to drive 7 IR LEDs. The reflected light is detected by photo-diode D5. The signal undergoes various stages of amplification and filtering until it is passed through a peak detector circuit consisting of D6, D7 and U1. The resulting DC voltage is buffered and finally clamped to a maximum of 5.1V using zener diode D8. This serves to prevent damage to the input of the microcontroller used to convert the voltage into a signal that can be understood by a computer.

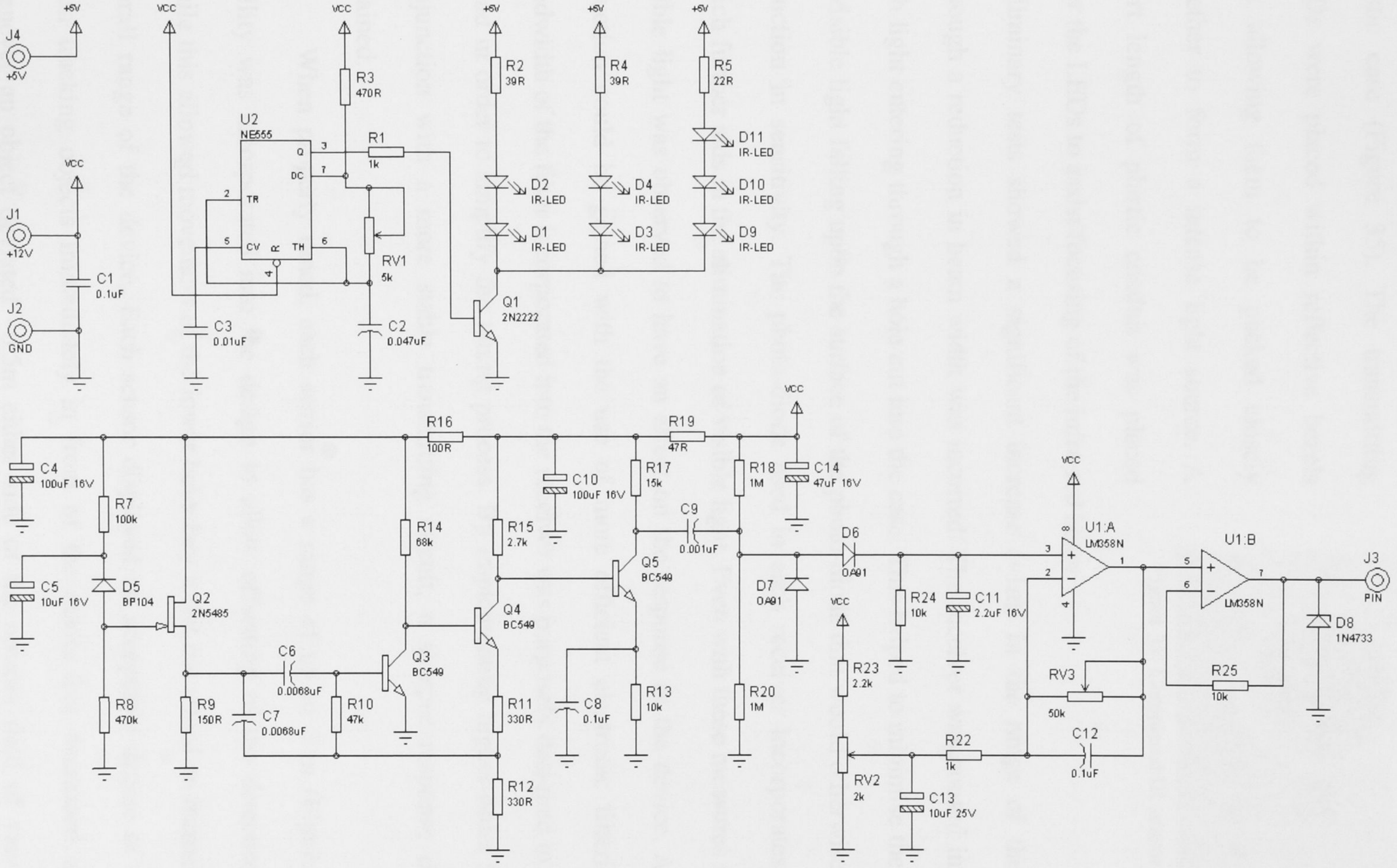

Figure 34: Schematic diagram of IR transmitter and receiver used in final sensor device.

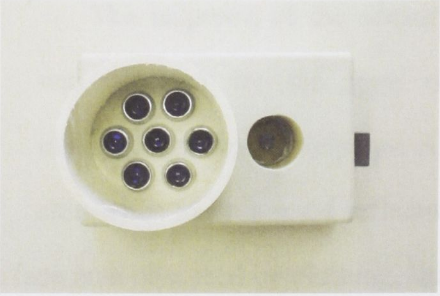

Figure 35: Completed IR sensor device.

Each sensor is housed within a small plastic case (Figure 35). The transmitting LEDs were placed within reflective bezels that allowing them to be packed closely together to form a intense light source. A short length of plastic conduit was placed over the LEDs to assist focusing of the infra-red beam. preliminary tests showed a significant increase (~1m) in the range of the device although a reduction in beam width was incurred. The receiver was housed internally, with light entering through a hole cut into the case. This helped to minimise the amount of visible light falling upon the surface of the photo-diode that would otherwise cause a reduction in sensitivity. The photo-diode used in each receiver incorporates a filter which further aids in the elimination of visible light. Even with these measures in place, visible light was observed to have an effect on the response of the device. A further reduction could be gained with the use of more efficient electronic filtering. The bandwidth of the filter incorporated into the receiver was purposely designed to be quite broad in order to simplify the tuning process. By implementing higher-order filters in conjunction with a more stable transmitting circuit, a sharper response could be obtained.

When properly tuned, each sensor has a range of up to 3 m (Figure 36). A facility was incorporated into the design to allow offsetting of the detection range. While this allowed movement of the lower boundary it did not afford an increase in the overall range of the device. Each sensor displayed an acceptable degree of linearity when tracking objects immediately in front of the device but decreased markedly whenever an object deviated ~0.5m either side of the sensors field of view. While imperfect, the response shape was deemed satisfactory for use in this project.

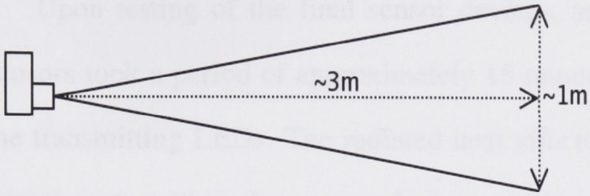

Figure 36: Approximate IR sensor field of view.

The response of the sensors was affected by the type of clothing worn by participants. As skin tends to be a good reflector of IR radiation, it was found that the greater the surface area obscured by clothing, the greater the decrease in sensitivity of the device. The type of material used in the clothing also had an effect. Items such as woollen jumpers were considerably less reflective than a cotton t-shirt due to its thicker and uneven texture.

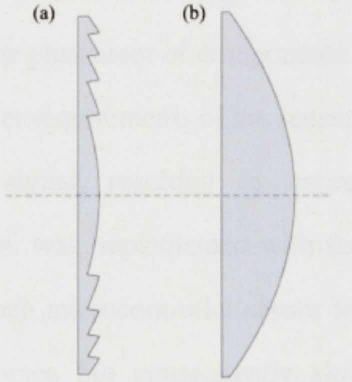

Figure 37: a) Fresnel lens b) Spherical lens.

Several attempts were made to increase the range of the sensors using various lenses and reflectors. Results of experiments performed with glass and plastic spherical lenses were inconclusive. Upon further research, my attention was turned to the use of Fresnel lenses[90] (Figure 37). Fresnel lenses can be used to focus IR light provided they are constructed of a material that allows transmission of IR radiation. Commercially available lenses are generally made from low density polyethylene due to its optical and mechanical properties, and low cost.

Tests performed with a lens obtained from a PIR security sensor yielded unsatisfactory results. To obtain correct focussing, the lens was required to be bent into a curve. Unless properly mounted, the lens only served to attenuate the IR light falling upon the photo-diode. Although literature on the subject stated that Fresnel lenses could be applied effectively in this situation, their use was eventually abandoned due to the time constraints of the project.

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[90] Fresnel lens – Wikipedia, the free encyclopedia. [Internet ref.].

Upon testing of the final sensor devices, several problems became apparent. The sensors took a period of approximately 15 minutes to stabilise due to heat produced by the transmitting LEDs. The radiated heat affected the electrical properties of the other components within the sensor device, causing the transmitter pulse clock and filter circuits to drift. This was easily corrected by retuning each sensor after a period of approximately 30 mins. While initially not of great concern this would pose problems if the installation was to be left unmonitored for extended periods.

Interference was observed in the form of energy radiated by the transmitter. This could be minimised by isolating the transmitter and receiver sections with the use of metal shielding and applying further attention to the placement of components. Further reductions could also be made by lowering the power requirements of the sensor.

Noise was observed within each sensor signal, resulting in unpredictable fluctuations in sensor output. A temporary solution was implemented with the use of RC low-pass filters placed in close proximity to each microcontroller sensor input pin. This helped to minimise the effects of interference but consequently slowed the response time of the sensor. While not longer instantaneous, the response speed was still sufficient for the purposes of this project.

3.2.5 Vibration/Force sensors

Sensors that react to an applied force, be it a small vibration, sharp impact or slow steady pressure can be used to track movement within a space. These sensors are commonly placed upon the floor where they pick up the vibrations of foot-fall. Many different vibration/force sensors are available but most operate using the same principle.[91] Figures 38 depicts a typical commercially available piezo film sensor. Figure 39 shows the internal component of an electronic buzzer that is commonly reappropriated for use as a sensing device.

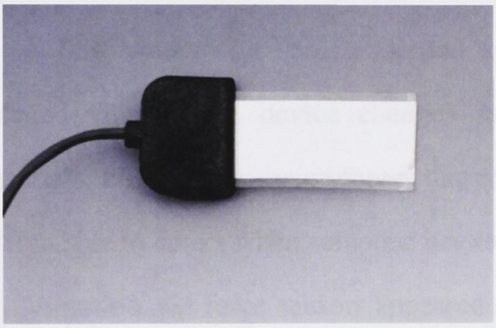

Figure 38: Measurement Specialities Inc. SDT1-028K piezo film sensor.

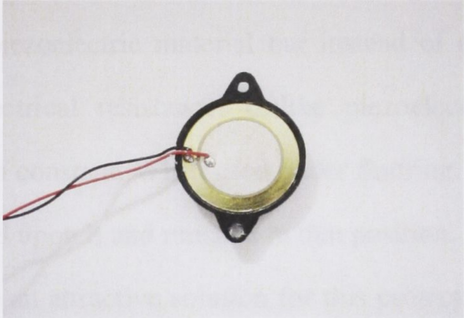

Figure 39: Piezo buzzer element.

A potential difference can be induced within the sensor material when it is subjected to mechanical deformation. The greater the deformation, the greater the induced potential difference. This is known as the piezoelectric effect. Piezoelectric materials have some disadvantages in regards to their use as sensor devices. Piezoelectric materials tend to be hard and brittle making them unsuitable for measuring direct impacts or bending forces. Also, the induced voltage does not remain constant when a force is applied continuously. For example, an induced voltage increases rapidly when a footfall is detected but quickly returns to zero as the subject remains in contact with the device.

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[91] Piezo Film Pulse Sensor. [Internet ref.].
FlexiForce®: Single Button Force Sensing Resistor. [Internet ref.].
fsrdatasheet.pdf. [Internet ref.].

Another type of sensor device used to measure applied forces is the Force Sensing Resistor (FSR). Typical examples are shown in Figures 40 and 41.

Figure 40: Interlink® FSR 400 and 402, Force Sensing Resistors.

Figure 41: FlexiForce® A201 FSR, Force Sensing Resistor.

A FSR works in a similar fashion to a piezoelectric material but instead of an induced voltage, the device changes its electrical resistance. Unlike piezoelectric materials, FSR's can measure forces that remain constant. When used under flooring, an FSR is able to detect when someone has stepped upon it and remains in that position.

Vibration and force sensors appeared to be an attractive solution for this project. A large area could be monitored by using multiple sensing devices. By arranging the sensors in a grid within the flooring of a room, a persons position could be determined. An installation that utilises this type of system to great effect is, *Gravity and Resistance* by Seiko Mikami and Sota Ichikawa (Figure 42).

> *On the floor are placed 225 units of 40cm x 40cm cell-like grids, in which specially developed sensors (not on/off switches) are fixed to detect instantly and continuously the changing position, weight, and speed.*[92]

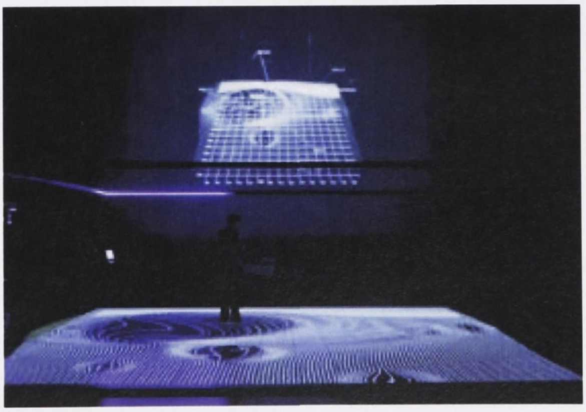

Figure 42: *Gravity and Resistance* - Mikami and Ichikawa, (2004)

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[92] Gravity and Resistance project. [Internet ref.].

An important consideration when using a great number of floor mounted sensors is the extensive wiring required to interface the devices to a computer or microcontroller. Assuming an average distance between footsteps to be ~50cm, 100 sensing elements would be required to cover a space of 5 x 5m. To ensure detection of each footstep, additional sensing elements would be required on the corners and edges of each tile. One can see that this very quickly adds to the complexity of the system. With such extensive wiring, the likelihood of faults developing over time would increase. Also the amount of time required to set up the installation would pose a significant problem in the case of this project as it is required to be easily transported.

Due to the overall complexity and reliability issues regarding piezo and FSR based sensors, neither option was deemed suitable for this project.

3.2.6 Sound activated switches

Sound activated switches fall into the category of hardware or software, but both function in a similar way. A microphone or piezoelectric transducer converts sound energy into a small fluctuating electrical potential. Using electronics or computer software, a switch is triggered when the amplitude of this waveform reaches a desired level.

The flexible nature of modern audio processing software applications makes it relatively simple to create a software based sound activated switch. Applications typically include some form of envelope follower function that can be used to trigger events upon different sound levels. A software based approach has been used to great effect in works such as *Listening Glass* by Rene Christen and Jasper Streit[93], and *Wire Field* by Shawn Decker[94].

Research revealed various circuit designs that could be used in the construction of a hardware based sound switch. The majority operate as follows. The sound input received by a microphone is amplified, rectified and filtered to obtain a DC voltage that increases with the increasing amplitude of the sound source. As the voltage reaches a desired level an electronic switch is triggered. The switching device often takes the form of a transistor, relay or solenoid.

While relatively simple to implement, both hardware and software switches are subject to the same disadvantages in regards to tuning. False triggering can occur when attempting to use more than one device within the same space as a particularly loud sound can cause several devices to be activated outside of the desired area.

After assessment of both hardware and software systems, it was decided that a software based approach would provide greater flexibility and ease of implementation.

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[93] Clatterbox. [Internet ref.].

[94] Shawn Decker – Wire Field. [Internet ref.].

3.2.7 Conclusions

Following a thorough assessment of each object tracking method it was found that infra-red based sensors were the most suitable option for this project. Despite the problems faced in the construction and implementation of such sensors, the range, precision and speed of the devices were considerably better than that provided by other object tracking methods. Use of software controlled sound activated switches was also considered due to their ease of implementation.

3.3 Sensor Interfaces (Past and Present)

Throughout the research period, an investigation was made into the various hardware devices that have been employed in the past for the capture and translation of sensor data. Many different systems have been implemented throughout the years and now, more than ever, a wide variety of new and accessible tools are available to the arts community.

Essentially most systems consist of the following components:

- Sensor devices that transform a physical action into an analogue signal, e.g. switches, variable resistors, IR transmitter/receiver pairs, etc.
- A digitizer (typically a microcontroller or microprocessor) that captures the output of the sensor device and translates it into representative binary number.
- An interface that transmits the resulting data to another device (usually a computer) via a standard communications protocol such as MIDI, RS-232 and OSC.

A selection of such systems include: Greg Schiemer's MIDI Tool Box, the STEIM Sensorlab, IRCAMs EtherSense, La Kitchens Toaster and Kroonde, Electrotap Teabox, I-CubeX, Eobody, Angelo Friaettas Smart and Dumb Controller, DIEM Digital Dance System, and Gluion. An outline of some of these past and present systems is discussed in the following section.

3.3.1 Greg Schiemer's MIDI Tool Box (1989-1994)

The MIDI Tool Box (MTB) is a Motorola 68HC11 microcontroller based hardware platform developed by Australian artist and composer, Greg Schiemer (Figure 43).[95] The MTB was conceived for the purpose of live-performance algorithmic composition, and to assist the creation of new and innovative electronic instruments. Its features made it a unique and useful device capable of delivering new uses of the MIDI protocol into the hands of artists and away from its previous commercial exclusivity.[96] At a time when algorithmic composition was predominantly confined to computer software systems, the MTB helped to remove some of their inherent restrictions. Applications could boot automatically on power-up, thus eliminating the need of a disk-operating system or console terminal for input/output (I/O) control. The enhanced I/O peripherals and memory features of the microcontroller, made it suitable for translating between various signal formats and MIDI. User-written applications, assembled on a host machine (IBM-PC, Atari, Amiga or Macintosh), were loaded, run and debugged via a terminal emulator into the MTB's software-monitor ROM. The monitor ROM also contained some commonly used MIDI utilities that could be called from user programs. The MTB found use in a number of pioneering artworks such as Jon Drummond's *Spiral and Sheet*, and Densil Cabrera's *Orbits*.[97,98]

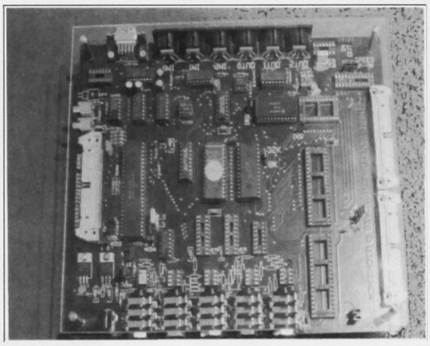

Figure 43: The MIDI Tool Box (MTB) hardware platform.

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[95] Schiemer, G. *Interactive radio.* Leonardo Music Journal, Vol. 4. (1994), pp. 21-22.

[96] Ibid.

[97] Jon Drummond – Past Projects [Internet ref.]

[98] Densil Cabrera [Internet ref.]

3.3.2 I-CubeX (1995)

The I-CubeX by Infusion Systems, is a sensor interface system capable of translating analog and digital signals into MIDI messages.[99] The I-CubeX Digitizer (Figure 44) can be purchased separately or as a package comprising a selection of different sensors, a software editor (for configuration) and a suite of Max/MSP and jMax plugins. Configuration is also possible with the use of MIDI system exclusive messages.

The Digitizer can function in two different modes: as a stand alone unit, or as a host when used in conjunction with a computer capable of processing MIDI messages.

In addition to its ability to translate input signals, the Digitizer is also able to output voltages that can be used to control devices such as relays, motors and solenoids.

With the increasing popularity of radio frequency and wearable devices spurring the market over the last decade, wireless and miniature hand-held I-CubeX systems are now also available.

Figure 44: The I-CubeX sensor interface hardware and wireless components.

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[99] I-CubeX Online Store [Internet ref.]

3.3.4 Dumb and Smart Controller (2002)

The Dumb and Smart Controller are the creation of performer and instrument builder, Angelo Fraietta.[100,101]

The Dumb Controller is a MIDI I/O board that also forms part of the Smart Controller. Based on a PIC16F877 microcontroller, the device can be used to convert control voltages to MIDI and vice versa. The device features 8, 10-bit analogue inputs and 8, digital I/O pins. The device can be battery operated and is also available with a class 1 Bluetooth wireless option.


Figure 45: The Smart Controller sensor interface.

The Smart Controller is a portable hardware device based upon a Real-Time Operating System (RTOS) called RTEMS.[102] The device operates in a similar fashion to the Dumb Controller yet with extended functionality. The Smart Controller is able to process input and output control voltages, MIDI, and Open Sound Control (OSC) messages via the optional Ethernet connection. The board features 16 analogue inputs, 16 digital inputs, 16 analogue outputs, and 16 digital outputs. Configuration is accomplished using the supplied Patch Editor, which bares similarities to the application Max/MSP.

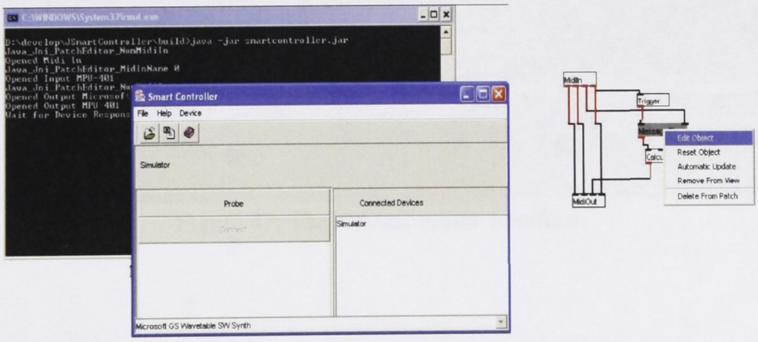
```
D:\develop\JSmartController\build>java -jar smartcontroller.jar
Java_Jni_PatchEditor_NumMidiIn
Opened Midi In
Java_Jni_PatchEditor_MidInName 0
Opened Input MPU-401
Java_Jni_PatchEditor_Nu
Opened Output Microsof
Opened Output MPU-401
Wait for Device Respons
```

Figure 46: The Smart Controller Patch Editor.

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100 Dumb Midi Controller [Internet ref.]

101 Smart Controller [Internet ref.]

102 RTEMS [Internet ref.]

3.3.5 La Kitchen Toaster and Kroonde (2003-04)

La Kitchen, is known for producing two high speed, high precision, sensor systems: Toaster and the Kroonde.[103,104]

The Toaster is capable of translating analog voltages into UDP Ethernet packets and/or MIDI messages. Up to 16 sensors can be connected to the device using 6.5mm RTS jacks. When transmitting over Ethernet, the sensors are updated approximately every 5 ms, with 16-bit precision (of which the first 14 bits are stable). When using MIDI, the speed of transmission is determined by the type of sensor employed, with updates typically occurring between 3 and 12 ms with a precision of 14-bits via MIDI Control Change messages.

The Toaster allows communication with a computer via four different protocols: OSC, binary, MIDI A and MIDI B. The OSC protocol allows the transmission of sensor data to any other compatible software such as Max/MSP, PD, and SC3. The binary transmission mode can be used in conjunction with the PD netreceive object. Messages are received as an ASCII string containing the keyword, Toaster, followed by the value of each sensor. In this fashion, data can be separated out from other netsend/netreceive messages using the PD route object.

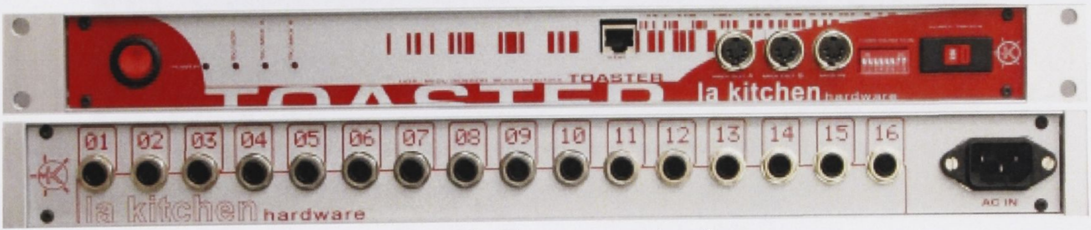

Figure 47: The Toaster sensor interface.

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103 La kitchen [Internet ref.]

104 La kitchen [Internet ref.]

In order to overcome the limitations of wired sensors, La Kitchen produced a wireless implementation of the Toaster, called the Kroonde Gamma.

The system is comprised of small (4.5 x 2.7 x 1 in) battery operated unit that can integrate up to 16 sensors simultaneously. The device transmits the captured sensor data using radio frequency waves (433 or 914 MHz) over a range of up to 300 ft. Its small size and wireless capability makes it useful tool when capturing and analysing data from sensors mounted on the body for applications such as gesture tracking. The base station (1U rack) receives information from the wireless unit and transmits the data via Ethernet to a computer, with 10-bit precision. Like the Toaster, the Kroonde can also send sensor data via MIDI.

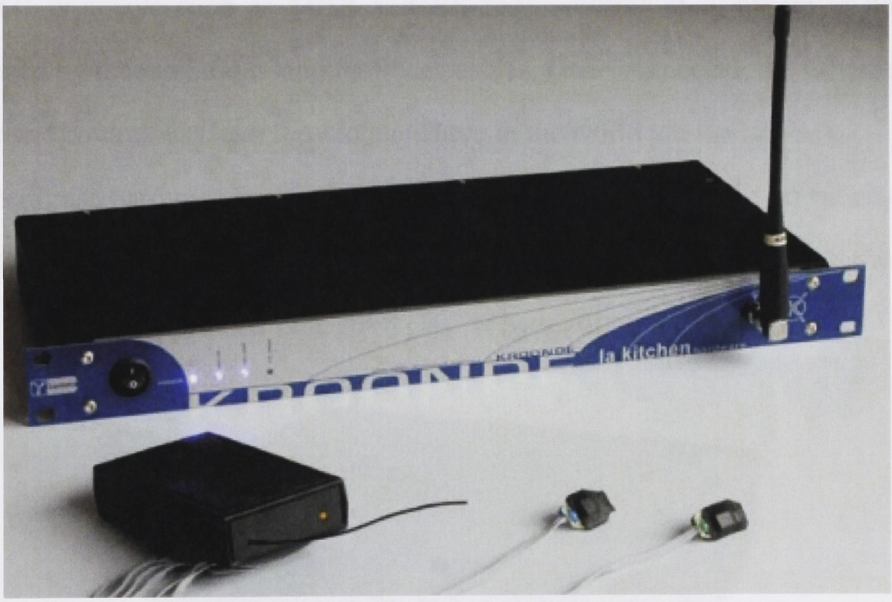

Figure 48: The Kroonde Gamma wireless sensor interface system.

3.3.6 EtherSense (2005)

The EtherSense system (developed at IRCAM by the Performing Arts Technology Research Team) is a high-resolution, high-speed, sensor digitization and acquisition tool that was created in an attempt to simplify the processes involved in gesture analysis. The interface gives the user the choice of both MIDI and the Open Sound Control (OSC) protocol.[105]

As the name suggests, the unit features a built-in Ethernet controller. Sensor signals can be digitized and transmitted over a network using standard UTP cables, via OSC. (For a more detailed explanation of the OSC protocol please refer to Section 3.4.4.) Unlike traditional protocols such as MIDI and RS-232, control of a system is no longer constrained by the maximum length of ones cables. Once connected, the EtherSense can monitor and control a system located anywhere in the world via the Internet.

The EtherSense hardware is able to simultaneously digitize up to 32, 16-bit channels at a rate of 500 Hz (or 16 channels at 1 kHz), features an LCD and joystick interface and is housed within a standard half rack sized case. (Figure 49)

Figure 49: The EtherSense OSC/MIDI sensor interface.

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[105] Ethersense, [Internet ref.]

3.3.7 Electrotap Teabox (2007)

Figure 50: The Teabox sensor interface.

The Teabox is a high-resolution, low latency sensor interface that connects to a computer using the digital audio S/PDIF standard.[106] The use of a S/PDIF for the transmission of data makes the Teabox quite unique. This method was employed in an attempt to remove the latency and resolution problems commonly associated with the MIDI protocol. The Teabox provides 8, 12-bit analog inputs and 16 digital inputs. These inputs are accessible via a variety of connectors including XLR, ¼ inch TRS, RJ11, and pin headers. The use of 12-bit ADCs presents a significant improvement in resolution when compared to MIDI, i.e. 0 to 4095, compared to 0 to 127 for MIDI data. In addition, all 24 sensors are capable of being updated at a rate of 4000 Hz. Once processed, data for each sensor is transported over S/PDIF as a 16-bit word. Connection can be made between the Teabox and computer using a 75-Ohm RCA-style or TOSLINK (optical) cable. The use of an optical cable helps to minimise the effect of electrical interference.

Configuration of the device is handled via the choice of a Max/MSP external, and/or the Teabox Bridge. In the case of the Max/MSP external, once the correct adc~ channel is patched into the teabox~ object, the sensor data is output as individual signals. Values range between 0.0 and 1.0, which can be further manipulated using the vast array of objects provided within Max/MSP. As the name suggests, the Teabox Bridge application acts as a bridge between the sensor hardware and any other software package that conforms to the OSC and/or MIDI protocols.

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[106] Electrotap [Internet ref.]

3.3.8 Conclusions

As illustrated in the previous section, a number of ready made sensor interface hardware solutions are available to the present day artist and instrument builder. While based around similar core functions, each system primarily seeks to simplify the process of interfacing the physical world to the world of 1's and 0's within the computer. While some are no longer available, there are undeniably, more tools available today than ever before. As technology continues to advance new tools will no doubt become available that offer even greater flexibility and performance.

Although a ready-made system appeared an attractive option in the case of this project, most were still prohibitively expensive or difficult to obtain. Perhaps the most likely candidate was the Dumb and/or Smart Controller. At a price of $240 AU, the Dumb Controller was certainly within the scope of my budget and appeared to offer the features that I required. In addition, having observed the device in operation at the Electrofringe festival, I felt convinced that support for the device would be easily attainable.

Conversely, having some prior experience in programming microcontrollers, I was also convinced that undertaking the task of building my own purpose built device would be within my abilities. With this in mind I set out to research how this might be achieved and set myself a time frame of two months in which to develop a working prototype. If it was deemed that a device could not be successfully constructed within the designated time frame, I still had the option of purchasing a ready-made unit.

3.4 Selection of Microcontroller

Following selection of a suitable object-tracking method, the next step in the development process involved an investigation into a means of controlling the vast amount of data produced by the sensors. The options deemed most suitable were:

- An analogue to digital converter board interfaced to a computer, or
- A microcontroller.

The first option could be realised with the use of a peripheral controller added to my notebook computer or the use of a second dedicated computer. Peripheral cards are capable of high speed, high accuracy analogue to digital conversion but generally require the user to implement their own code for processing the data. The notebook computer had no provision for adding an ADC PCI or ISA card and the extra financial outlay of a PCMCIA adapter made this an unsuitable option. On the other hand, the use of a second dedicated computer would not pose a significant financial problem as an obsolete system such as an early Pentium or 486 PC would be sufficient. However, the addition of a second computer would add to the space required for the system.

The second option, involving the use of a microcontroller appeared to be a more suitable solution. A microcontroller is essentially a simplified computer built into a single chip. This integrated design enables the development of highly sophisticated applications whilst minimising the complexity of external circuits. While not as powerful as the typical microprocessor found in personal computers, microcontrollers can be applied in situation where the use of a full-sized computer is undesirable.

For the successful implementation of a microcontroller based system, several important design issues required consideration. The microcontroller must provide easily implemented and effective functionality for a minimal financial outlay.

The average price of a single microcontroller is quite low (starting from several dollars), therefore, cost was not considered of great importance during the the selection process. The microcontroller would preferably be available within Australia, to ensure minimal delay in obtaining the device. Many manufacturers distribute development software specific to their device and a vast selection of prototyping boards are available direct from the manufacturer or via third-party distributors. Also, the Internet provides a valuable resource for freely available programmer designs and software.

To reduce the complexity of the circuit, selection of a microcontroller was limited to those with the required functionality built into the device. The microcontroller was required to have an analogue-to-digital converter (ADC) with suitable resolution and precision, a Universal Asynchronous Receiver Transmitter (USART), and sufficient input and output (at least 8 analogue and 8 digital) pins.

I wanted to be sure that the device had a simple, effective instruction set that was well documented and easy to learn. This also applied to the tools (compiler, assembler, debugger and hardware programmer) required for development, and upload of code to the microcontroller. As a final consideration, it was also required that the device had a well established on-line user-base and knowledge resource that could be accessed in the event that I ran into problems during the development process.

Literature and Internet research presented a number of suitable solutions including:

- the Motorola 68HC11
- Atmel AVR®
- Parallax Basic Stamp® II, and
- Microchip PICMicro®.

3.4.1 Motorola 68HC11

The 68HC11 is an 8-bit data, 16-bit address microcontroller manufactured by Motorola that has seen extensive use throughout the engineering community. The series offers built-in EEPROM/OTPROM, RAM, digital I/O, timers, A/D converter, PWM generator, synchronous and asynchronous communications channels.[107]

Since their introduction, the 68HC11 has undergone a number of revisions but is starting to show its age when compared to products offered by other manufacturers. Although still capable of performing sophisticated tasks the major hurdle presented in my case was the lack of user-friendly development tools. Unlike other devices, the 68HC11 series does not have an integrated development environment and hardware programming tools were hard to locate and/or prohibitively expensive. Literature research revealed many books on the topic of programming but I failed to locate any code examples that could aid in the development of my specific application.

3.4.2 Atmel AVR® AT90S8535

The Atmel AVR® 8-bit microcontrollers have been a popular choice amongst professional and hobbyist electronics enthusiasts. The AT90S8535 appeared to be a viable option as it provided a multi-channel 10 bit-ADC, numerous digital I/O pins and an on-board USART.[108] The freely available development environment, *AVR Studio*, appeared to be an attractive solution for code development and several designs for hardware programming devices were located via the Internet. However, a lack of available code examples did prove to be a problem, although it was conceded that with more time, relevant examples could be found amongst of the on-line AVR community.

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[107] About MC68HC11. [Internet ref.].

[108] at09s8535.pdf. [Internet ref.]

3.4.3 Parallax Basic Stamp® II

Parallax offer a number of single board computer solutions in their popular Basic Stamp® range.[109] The series, so called due to their similarity in size to a postage stamp, aims to simplify both electrical and software development through the use of a prefabricated circuit board (consisting of the microcontroller, memory, clock, and power supply) and the PBASIC language. Like other microcontroller solutions, the Basic Stamp II (BS2-IC) is able to control electronic devices using its 16 digital I/O pins. The BS2-IC has proven popular amongst hobbyists and has also found wide-spread use in industrial applications and educational institutions. The PBASIC language was perhaps the most attractive point when first considering the Basic Stamp. The PBASIC language seeks to speed development through its quick learning curve compared to the more powerful but less user friendly, assembler language. Although initially attractive, as I looked further into the capabilities of the BS2, I became concerned with the amount of available code space, and lack of on board ADC. After much thought, I decided to investigate the PIC16 series of microcontrollers that the BS2-IC was based upon.

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[109] Basicstampfaq.pdf. [Internet ref.]

3.4.4 Microchip PICMicro® PIC16F877

Microchip PICMicro® microcontrollers have been in widespread use since the 1980's. Microchip offer a diverse range microcontrollers, from the simplified low-range PIC12 series, through the PIC16 mid-range devices up to the feature-rich PIC18 high-end devices.[110] After performing an extensive literature and Internet search I became aware of the popularity of two particular devices within the mid-range PIC16 series: the PIC16F84 and PIC16F877. The PIC16F877 was of particular interest as it provided a multichannel 10-bit ADC, USART, and numerous I/O pins. During my research, it became apparent that others had used the PIC16F877 specifically for MIDI related projects. As such, a number of well documented code examples were located. Microchip offer a code development environment, MPLAB, free of charge and numerous examples of chip programming devices were located via the Internet. This provided great encouragement and ultimately confirmed my decision to use this device for my own project.

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110 Microchip [Internet ref.]

3.5 Selection of a Communication Protocol

This section details the processes involved in the selection of a suitable protocol used for communication between the microcontroller and computer. A discussion is presented on the implementation of protocols including: serial (RS-232), Parallel, MIDI and Open Sound Control.

3.5.1 Serial

Serial communication involves the transmission of data one bit at a time in a sequential manner. The timing is coordinated between the transmitter and receiver with use of a clock signal.[111] Serial communication can be easily implemented using the PIC16F877. Once the required data transmission (baud) rate is ensured, the required bit patterns can be sent from the built-in USART. To ensure successful communication between the microcontroller and computer, the data signal must first be converted to the correct voltage level. This is generally performed with the use of a RS-232 converter. After conversion, the serial data must then be translated into a format that can be understood by the signal processing software used within the computer. Many applications provide a facility for input of serial information as ASCII data but information can also be conveyed using a serial MIDI driver.

Unfortunately my computer did not possess a serial port. This meant that the process would be further complicated by the necessity of a USB to serial converter. While not cause for great concern, it did present the possibility of time wasted in locating a compatible converter and its successful installation under the Linux OS. Consequently, my attention turned to the implementation of a parallel communication protocol using the computers available printer port.

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[111] Predko, M. *Programming and Customizing PICMicro® Microcontrollers 2^{nd} Ed.* p293.

3.5.2 Parallel

Parallel communication appeared to be a viable option since data can be easily sent from PORTD of a PIC16F877. Although my computer provided a built in parallel port, the lack of a parallel communication object within the audio processing software considered for this project (PD, Max/MSP, SC3) or the presence of a parallel MIDI driver within Linux, meant that implementation would rely upon the development of my own driver. The time constraints of the project made this an unsatisfactory solution.

3.5.3 MIDI

The Musical Instrument Digital Interface (MIDI) became the standard method of communication between instruments and computers in the 80's. It was developed as an industry wide protocol to allow transmission of information from a keyboard, sequencer, or personal computer to a group of synthesizers[112]. The protocol allows the transmission of parameter and configuration settings in the form of control change (CC), program and channel change, and system exclusive messages. Of particular interest in this case were the possibilities offered through control change messages. CC messages provided 128 (0-127) message streams, each with a resolution of 128 steps (0-127). This would provide a suitable level of precision for this project.

The PIC16F877's 10-bit ADC provides sufficient resolution to translate voltages output by the sensors into the full MIDI scale. Since the MIDI scale ranges from 0 to 127, each increment can be expressed with use of an 7-bit number (or 8-bit with a leading 0). The PIC16F877's instruction set does not natively support multiplication and division but is capable of performing these types of calculations using bit rotation instructions[113].

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[112] Chamberlin, H. *Musical Applications of Microprocessors* 2^{nd} *ed.* p313.

[113] Predko. p123.

An 8-bit number can be formed by dropping the last two digits from the 10-bit ADC result (0-1023). This number (0-255) is then rotated one bit to the right to obtain a value ranging from 0 to 127. This process is illustrated in Figure 51.

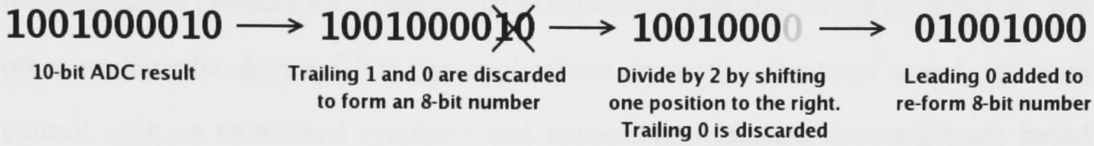

Figure 51: Conversion of a 10-bit ADC value into an 8-bit MIDI value.

This method greatly simplifies the microcontroller code but presents a significant decrease in the resolution of the A/D conversion. This does not pose a problem when using the somewhat limited MIDI scale but would be unsatisfactory if using the OSC protocol (Refer to Section 3.5.4).

Each sensor outputs a voltage of 0-5V. Thus, for every ~19.6 mV increase in the sensor output voltage, the MIDI CC value increases by 1. For complete coverage of an area ~5 m^2 each sensor is required to have a range of at least 2.5 metres. Utilising the full 0-5V sensor range, each sensor would possess a resolution of ~2cm for every increment in MIDI CC value. This was deemed more than adequate for the purposes of this project.

The effective and flexible communication between devices presented by the MIDI protocol has seen it in widespread use until the present day. However, the appearance of new protocols such as Open Sound Control has signalled a considerable step towards solving some of the issues faced when using MIDI.

3.5.4 Open Sound Control

Open Sound Control (OSC) was developed as an open, transport-independent, message-based protocol for communication between computers, sound synthesizers, and other multimedia devices.[114] It has applications in many aspects of sound and data control such as networked synthesis and musical performance, sensor/gesture based instrument design, web-based control interfaces and virtual reality.

The protocol functions on a client/server basis with data sent and received between the two in the form of OSC packets. A packet consists of a contiguous block of binary data, and the number of 8-bit bytes that make up that block. A packet may consist of one or more data units called messages.

A OSC message consists of the following elements:

- An address pattern - a string that specifies the type of message that is transmitted and the destination within the OSC server to which the message is directed.
- A type tag string - a string that designates the type of data present in each argument, and
- Arguments - the data contained within the message.

In order to understand how a message is interpreted by an OSC server, an explanation of its methods and address space is required.

Every OSC server has a number of potential destinations for OSC messages that correspond to the points of control available within the application. An OSC method is invoked when a server supplies arguments that cause the method's effect to take place. Unlike other protocols (e.g. MIDI), all controllable elements of an OSC server are

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[114] OpenSound Control Specification. [Internet ref.].

organised in a tree-like hierarchy (similar to a URL or file system path), with each node, or container, having its own symbolic name within the servers address space. The address space can also be dynamic, i.e. its contents and structure can change over time.

Whereas the MIDI protocol dictates that only particular messages (i.e. Note on/off, Control Change, Program Change, etc) can be sent and/or received by a device, an OSC server defines its address space according to the features that are available within the device, and the implementers idea of how these features should be arranged.

For example, the address "/lpfilter/2/cutoff" refers to a node named "cutoff" (in this case denoting the cutoff frequency of a low-pass filter) that is a child of a node named "2" that is a child of a top-level node named "lpfilter."

Upon definition of the address space and its nodes, the server is then ready to accept arguments. However, before a messages arguments can be transmitted, it must first be grouped with an address pattern string and a type tag string. An OSC address pattern is essentially an OSC address, except that it may contain special characters (such as ?, *, !, [012abc], {abc,def}) for regular expression pattern matching. When an message's address pattern matches more than one of the addresses in the server's address space, it behaves as if individual messages (all with the same arguments) were sent to each of the matched addresses.

Table 2 displays the various OSC data types and associated tag that make up an OSC message.[115]

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[115] Wright, M., A. Freed, and A. Momeni. *Open Sound Control: State of the Art.* Proceedings of the 2003 Conference on New Interfaces for Musical Expression (NIME-03), Montreal, Canada. p153.

OSC Type Tag	**Associated Argument**
i	32-bit integer
f	32-bit floating point
s	OSC string - A sequence of non-null ASCII characters followed by a null, followed by 0-3 additional null characters to make the total number of bits a multiple of 32.
b	OSC blob - An int32 size count, followed by that many 8-bit bytes of arbitrary binary data, followed by 0-3 additional zero bytes to make the total number of bits a multiple of 32.
h	64-bit big-endian two's complement integer
t	OSC-timetag - 64-bit big-endian fixed-point time tag
d	64-bit (double) IEEE 754 floating point number
S	Alternate type represented as an OSC-string (for example, for systems that differentiate symbols from strings”
c	ASCII character, sent as 32 bits
r	32-bit RGBA colour information
m	4 byte MIDI message. Bytes from MSB to LSB are: port id, status byte, data1
T	True. No bytes are allocated in the argument data.
F	False. No bytes are allocated in the argument data.
N	Nil. No bytes are allocated in the argument data.
L	Infinitum. No bytes are allocated in the argument data.
[	Indicates the beginning of an array.
]	Indicates the end of an array.

Table 2: The various OSC type tags and their associated arguments.

Using the previous example, a message can be formed using the address pattern, /lpfilter/2/cutoff, its type string, i, indicating that there is a single integer argument, and that the argument has a value of 440 (Hz).

Once a message is assembled, a sequence of messages can be strung together in the form of a bundle. A bundle is prepended with the OSC string “#bundle” to denote that the data following it is in the form of a bundle.

Since the OSC protocol does not provide any mechanism for clock synchronization, a server must have access to a representation of the correct current absolute time. Thus, each bundle is also accompanied by a 64-bit Time Tag (based on the Internet Network Time Protocol) that specifies the absolute time at which the messages in the bundle should take effect. When a received packet contains a single message, the server invokes the corresponding methods immediately upon receipt of the packet. In the case of a bundle the Time Tag determines when the corresponding methods should be invoked. If the time is set to a period before or equal to the current time, the server should invoke the methods immediately. In all other cases the Time Tag represents a time in the future, and the server must store the bundle until the specified time.

Bundles may also contain other bundles nested within themselves. These are known as sub-bundles. In this fashion, all messages in the same bundle are interpreted as though the individual messages within it were processed in a single instance.

Another type of message worthy of mention is the OSC query. A query is a message that requests information be returned back to the client from the server. An example could be message that requests information about what types of arguments are accepted by a given node or a request for documentation about a function located at a particular address.

3.5.5 Conclusions

Of the protocols investigated during the research period, the two that provided the most appeal were MIDI and OSC.

Although the OSC protocol offers much greater flexibility and functionality when compared to MIDI, a lack of readily available code examples specific to the PICMicro® family made it a less attractive option. The vastly simpler nature of MIDI convinced me of a short projected time for completion of a working prototype and ultimately confirmed my decision to use the protocol for this project. In the event that the resolution requirements of the project changed and the MIDI protocol could no longer provide the necessary performance, the option of purchasing a ready made OSC compliant device still remained.

3.6 Construction of the MIDI sensor interface

3.6.1 Theory of Operation

Upon selection of a suitable microcontroller and communication protocol, work commenced on the construction of a device that would perform the translation of sensor voltages into MIDI messages. To do this, the microcontroller was required to perform a number of tasks, including:

- Initialisation of various PICMicro® features, e.g. Clearing variables, ADC and baud rate settings,
- MIDI channel selection,
- Analogue to digital conversion of sensor voltages,
- Calculation of MIDI CC value,
- Formation and output of a MIDI byte pack.

Following initialisation, the bulk of the work performed by the microcontroller lies in the analogue to digital conversion of sensor voltages, and the formation of MIDI byte packs. These tasks are performed as follows.

Each pin of the ADC (PORT A) is sampled and the result stored in a table within the microcontroller's memory. The ADC result is then manipulated (Figure 51) to produce a MIDI CC value representative of the sampled sensor voltage. This value is combined with the MIDI channel number formed using a 4-way DIL switch (Refer to Section 3.6.3). Upon each loop, the microcontroller tests the voltages applied to the respective pins of PORTC and stores the 4-bit number. This allows the microcontroller to continuously update the selected MIDI channel without the need to reset the device.

To understand the steps taken in the translation of sensor voltages into MIDI messages, a brief discussion of the MIDI byte pack is required.

A complete MIDI byte pack consists of three bytes (Status, Data1 and Data2) containing the MIDI message information, plus a number of individual (Start, Stop) bits that are used to signal the beginning and end of each byte[116].

The first byte begins with its MSB set to 1. This signals the MIDI device that this is a Status byte. The next three digits represent the type of MIDI event. Of interest in this case are codes '000', '001', and '011' representing note off, note on, and control change messages respectively. The remaining four bits represent the MIDI channel.

The second byte begins with its MSB set to 0, signalling the presence of a Data byte. The remaining seven bits represent a MIDI number (e.g. Note or controller number) between 0 and 127.

The third byte also begins with its MSB set to 0, denoting a Data byte. The remaining seven bits represent a MIDI value (e.g. velocity or controller value) between 0 and 127.

Prior to transmission of the MIDI pack, the order of the bits within each byte are reversed. Finally, start (0) and stop (1) bits are placed at the beginning and ending of each byte respectively.

An example of this process can be found in Appendix A.

The completed MIDI byte pack is then output using the microcontroller's addressable USART at a baud rate of 31.250 kHz. Upon each loop of the code, the microcontroller tests whether the voltage output of each sensor has changed. If so, a new CC value is substituted for the old and the newly constructed byte pack is transmitted.

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[116] JDP Homepage - Practical MIDI packs generating/transmitting. [Internet ref.].

3.6.2 Code Upload

The completed code was loaded into the microcontroller using a JDM[117] programming device (Figure 52) and the application ICProg[118] (Figure 53).

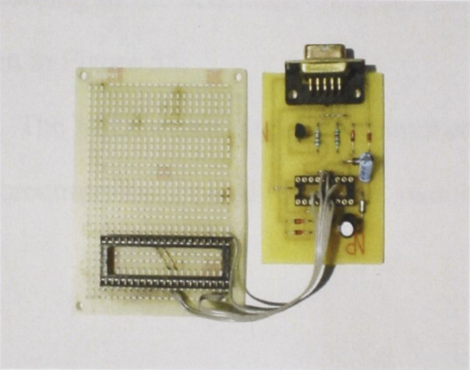
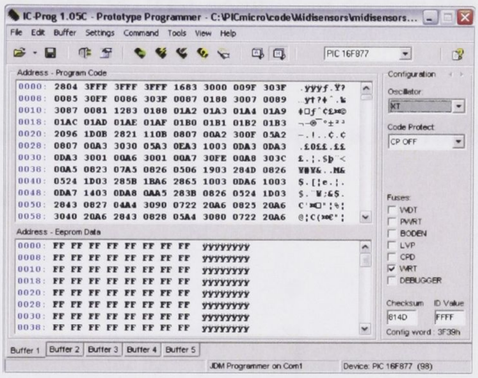


Figure 52: JDM programmer and adapter.



```
0000: 2804 3FFF 3FFF 3FFF 1683 3000 009F 303F  .ÿÿÿf.Ÿ?
0008: 0085 30FF 0086 303F 0087 0188 3007 0089  .ÿ†?‡ˆ.‰
0010: 3087 0081 1283 0188 01A2 01A3 01A4 01A9  ‡□ƒˆ¢£¤©
0018: 01AC 01AD 01AE 01AF 01B0 01B1 01B2 01B3  ¬-®¯°±²³
0020: 2096 1D0B 2821 110B 0807 00A2 300F 05A2  –.!..¢.¢
0028: 0807 00A3 3030 05A3 0EA3 1003 0DA3 0DA3  .£0££.££
0030: 0DA3 3001 00A6 3001 00A7 30FE 00A8 303C  £.¦.§þ¨<
0038: 00A5 0823 07A5 0826 0506 1903 284D 0826  ¥#¥&..M&
0040: 0524 1D03 285B 1BA6 2865 1003 0DA6 1003  $.[¦e.¦.
0048: 0DA7 1403 0DA8 0AA5 283B 0826 0524 1D03  §.¨¥;&$.
0050: 2843 0827 04A4 3090 0722 20A6 0825 20A6  C'¤□"¦%¦
0058: 3040 20A6 2843 0828 05A4 3080 0722 20A6  @¦C(¤€"¦
```

```
0000: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0008: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0010: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0018: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0020: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0028: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0030: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
0038: FF FF FF FF FF FF FF FF  ÿÿÿÿÿÿÿÿ
```

Oscillator: XT

Code Protect: CP OFF

Fuses:
- ☐ WDT
- ☐ PWRT
- ☐ BODEN
- ☐ LVP
- ☐ CPD
- ☑ WRT
- ☐ DEBUGGER

Checksum: 814D

ID Value: FFFF

Config word : 3F39h

JDM Programmer on Com1

Device: PIC 16F877 (98)

Figure 53: IC-Prog programming software.

The JDM programmer is a free-ware based design capable of programming PICMicro® and serial EEPROM devices. This device was selected due to its low parts count and compatibility with the PIC16F877.

The programmer was connected to a computer via RS-232 and the code uploaded to the microcontroller using IC-Prog. IC-Prog was used in preference to the supplied software due to its user friendly interface.

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117 PIC Programmer 2, 16C84, 12C508. [Internet ref.]

118 IC-Prog Prototype Programmer, programs : 12C508, 16C84, 16F84,..., 90S1200, 59C11, 89C2051, 89S53, 250x0, PIC, AVR, 80C51 etc. [Internet ref.].

3.6.3 Design features

Prototypes were constructed to determine the various functions required by the controller. After testing and debugging, the final MIDIsensor controller was constructed according to the schematic diagram shown in Figure 54. The completed device can be seen in Figure 55.

The MIDIsensor circuit is comprised of a regulated 5V power supply, a PIC16F877 microcontroller, crystal oscillator, switches and several connection jacks.

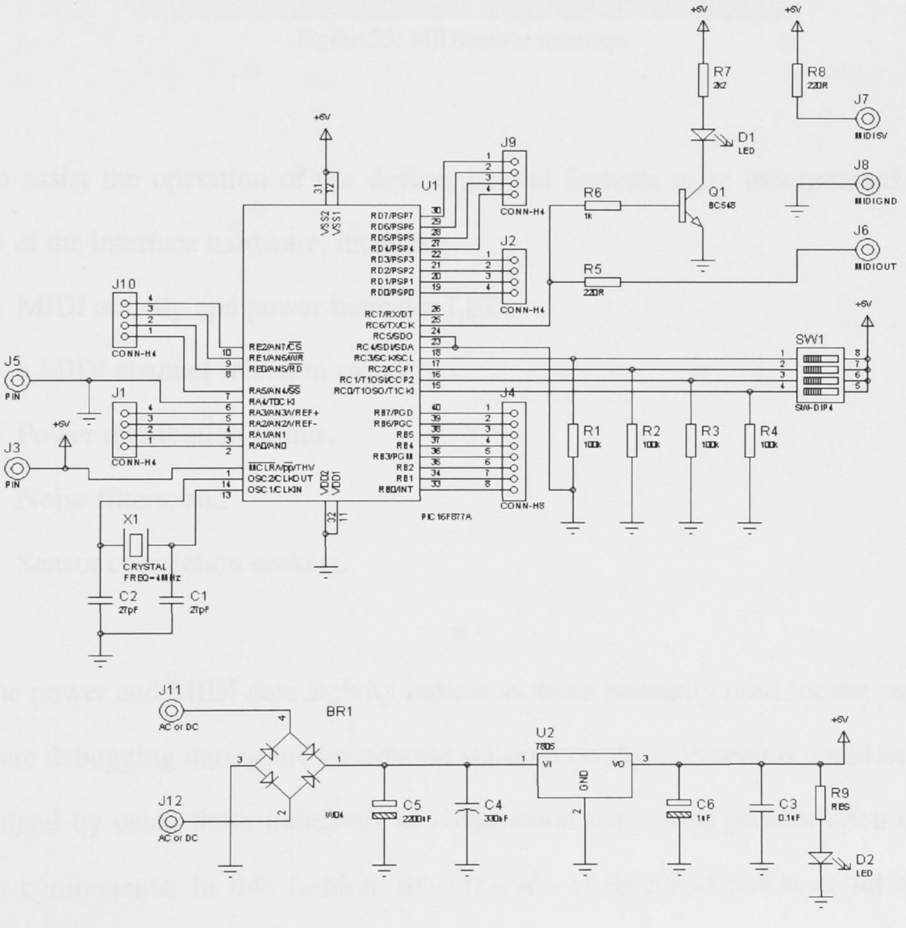

Figure 54: Schematic diagram of MIDIsensor controller.

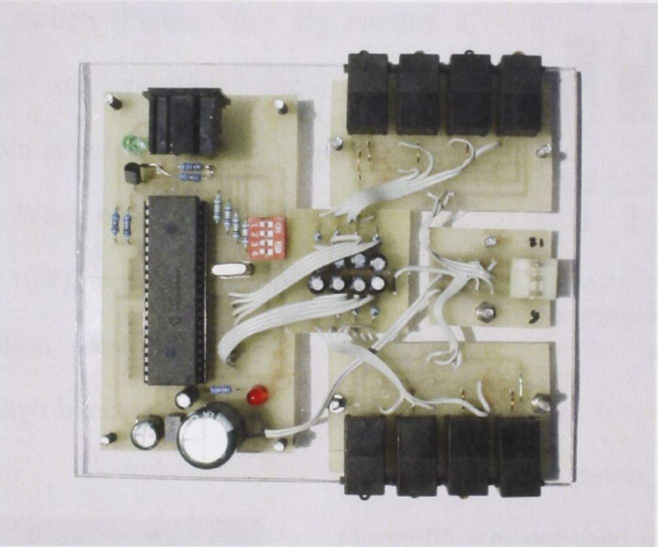

Figure 55: MIDIsensor interface.

To assist the operation of the device, several features were incorporated into the design of the interface hardware, including:

- MIDI activity and power indicator LEDs,
- a MIDI channel selection switch,
- Power distribution points,
- Noise filters, and
- Sensor connection sockets.

The power and MIDI data activity indicators were primarily used for the purpose of hardware debugging during the installation set-up procedure. Problems could be quickly ascertained by using these indicators in conjunction with those present upon the other system components. In this fashion, the presence of data and power could be easily traced throughout the system.

MIDI channel selection was performed via the use of a 4-way DIL switch (Figure 56). By moving a switch to the on position the respective microcontroller pin is pulled high (5V) representing a binary state of 1. When placed in the off position, the pin is pulled low (0V), representing a binary 0. When used in combination, these four switches are able to represent a four digit binary number.

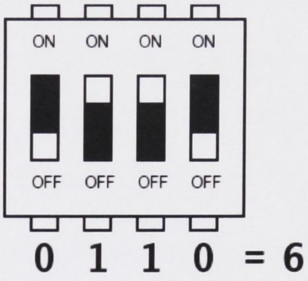

Figure 56: MIDI channel selection using a 4-bit binary number.

The on-board power supply shown in Figure 55 was not used in the final version of the MIDIsensor interface. Instead, a more robust power supply was constructed, capable of supplying +12V and +5V at a maximum of 1.5A (Figure 57). This was used to power both the MIDIsensor board and eight IR sensors.

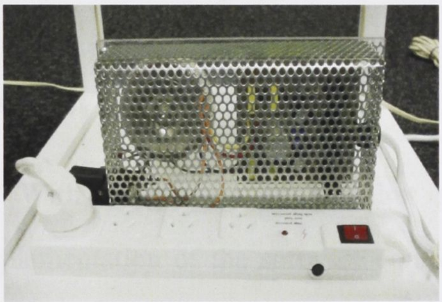

Figure 57: +5V and +12V power supply and distribution board.

As discussed in section 3.2.4, an RC low-pass filter was placed upon each of the eight sensor data lines to assist in the removal of noise.

Connectors were provided for the eight IR sensors on two separate boards, each with four RJ11 jacks. These jacks allowed connection of +5V, +12V, Ground and Data lines between the sensors and the MIDIsensor board.

4

Software

With completion of the first sensor system prototype, work began on the implementation of the software controlled functions of the piece. This chapter details the selection processes and evaluation of several software solutions that appeared to meet the needs of the project.

4.1 Selection and Evaluation

Since I possessed little computer programming experience, and limited time in which to complete the work, research was focused upon readily available software options. The viability of a particular application was determined according to: compatibility, ease of implementation, functionality, performance, and cost.

It was deemed necessary that I should be able to work on the project in my own time outside of the university environment and the computer equipment provided there. Hence, the application was required to be compatible with the computer currently in my possession. Prior to commencing the project I purchased a Toshiba Satellite M30 notebook computer. Perhaps the greatest influence upon my selection of this system was my increasing interest in the Linux operating system (OS). Research performed on the Internet, and through first hand experience with a previous computer, convinced me that it would provide excellent compatibility with the Linux OS. The computer was also capable of running the Windows operating system, therefore provided compatibility with software that I had used previously. Most importantly it was capable of running the MPLAB IDE used for development of PICMicro® applications.

The application was required to be simple to learn due to the time constraints of the research period. I did not want to waste time struggling to learn an application well enough to be able to apply it effectively. Hence, this eliminated many of the traditional methods of digital signal processing that require extensive technical training (such as C) and those that were not well documented or had a limited user base. For this reason I chose to use either a graphical patching application such as Max/MSP and PD or a text based language like SC3 and Csound. These applications were selected due to their popularity and proven success when used by other artists and musicians. By no means were they the only options available for the production of realtime audio processing. Other options explored included: jMax, Reaktor, RTMix, AudioMulch, and the Kyma/Capybara system.[119,120,121,122,123]

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[119] jMax [Internet ref.]

[120] Native Instruments [Internet ref.]

[121] RTMix [Internet ref.]

[122] AudioMulch Interactive Music Studio [Internet ref.]

[123] Capybara [Internet ref.]

Of the two graphical patching applications, Max/MSP appeared to be more user friendly. The Max/MSP environment was initially easier to understand due to its well designed interface and extensive documentation. Max/MSP also benefits from a large on-line community, although PD has been steadily increasing in popularity since its conception. Both offer a wide array of processing objects. However, on first impressions, Max/MSP appeared to provide a greater number of general purpose objects. This did not pose a significant advantage over PD as the majority of these could be easily implemented using several PD objects in combination. Both Max/MSP and PD benefit from a vast array of libraries that provide extended functionality. In addition, specific libraries can be used to provide patch exchange between the two applications.

When comparing the two text based applications, I found the SC3 language to be more user friendly compared to that used of Csound. As I had little experience in high-level programming languages it was important that I should be able to easily comprehend the syntax used in each application. Csound has been used to great effect in many applications and benefits from an extensive user base community. However, it is perhaps beginning to show its age when compared to more modern sound processing languages. SC3 is based upon the Music N and Csound languages[124] and utilises a more user friendly syntax that aims to overcome the disadvantages of its predecessors. Although only a relatively new language, SC3 is quickly becoming a popular choice amongst the wider community. Consequently, documentation for the language is continuously improving.

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[124] Cottle, D. M. *Digital Synthesis using SC3.* p9. [Internet ref.].

Regardless of the type of application to be chosen, it was required to be flexible and provide adequate functionality. It must be capable of satisfying key technical requirements such as low latency audio and MIDI I/O, advanced signal processing tasks such as FFT, filtering, and complex switching and routing. The options provided in both graphical patching and text based applications appeared to meet these requirements. The only differences apparent between the two occurred in their level of user friendliness and performance. As a general observation the text based applications provided greater flexibility and performance at the expense of the user friendliness of graphical patching applications.

The performance requirements of the application were also taken into consideration. I was aiming to run all of the necessary software upon a single computer to help minimise the amount of equipment required for the project. Therefore, the application was required to perform reliably for extended periods of unmonitored use, within the constraints of available CPU cycles.

The final consideration was that of cost. The application would preferably fall with the scope of the allocated budget, although provision could be made to designate more funding to the application if it provided a clear advantage over a less expensive option. Of the four applications currently in consideration, the Max/MSP environment was the only one that would incur a significant fee.

As I was considering the use of the Linux OS my attention naturally turned to the choices available to me in the Open Source community. The term source refers to "*a set of principles and practices that promote access to the design and production of goods and knowledge.*"[125] It is most commonly applied to the source code of software made available to the general public. The limited or complete lack of intellectual property restrictions allows others to create software through individual or collaborative effort.

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[125] Open source – Wikipedia, the free encyclopedia [Internet ref.]

Many of the options presented by open source software were capable of providing the necessary functionality whilst also having the benefit of being available free of charge.

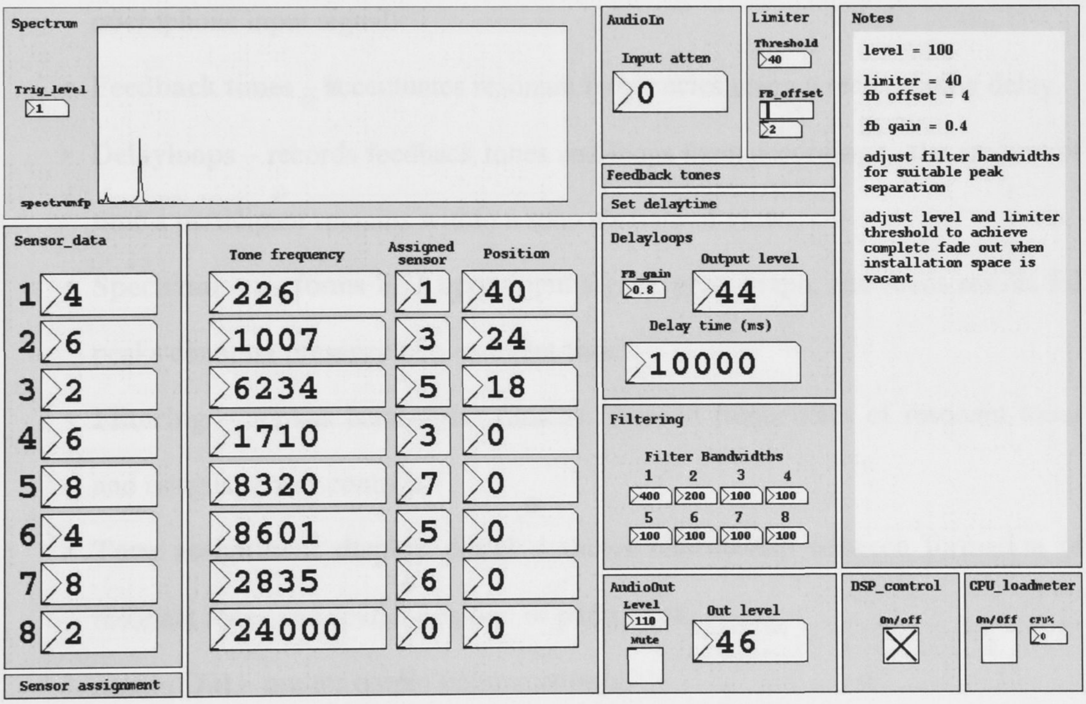
PD, SC3 and Csound are all available for the Linux OS as free open source software. PD and Csound are both well established for the Linux OS whereas SC3 has only recently been ported to the platform. Consequently, not all of the features provided in SC3's OSX version are available in the Linux port.

After careful consideration of the selection criteria I decided to implement the software based functions of the installation using PD. The features offered by text based applications were initially attractive, and ultimately appeared to be the more suitable option when compared to the graphical environments. Unfortunately, preliminary tests performed using SC3 reaffirmed the importance of adhering to a strict time frame. Although my initial progress was encouraging, I was not completely confident that I would be able to learn the language sufficiently within the designated period. Consequently, the use of Csound was also deemed unviable.

When comparing the differences displayed between the two graphical environments it was clear that one application did not present a significant advantage over the other. As I had used PD successfully in past instances, and considered myself to have a good understanding of its implementation, my final choice was inevitably made as a matter of personal preference.

4.2 PD patch

Using the ideas discussed throughout the previous chapters, work began on the implementation of a program patch within PD. The patch was to perform the bulk of the signal processing work required for the piece. Figure 58 shows the final version of the patch used in the installation. The patch can be divided into two main functions; the interpretation of MIDI sensor data and, the input, manipulation and output of audio. Within these main functions are numerous signal processing objects, each tailored to perform a specific task. Each component shown on the front panel incorporates several sub-patches. The operation of each component is summarised as follows.



level = 100

limiter = 40
fb offset = 4

fb gain = 0.4

adjust filter bandwidths for suitable peak separation

adjust level and limiter threshold to achieve complete fade out when installation space is vacant

Figure 58: Pure Data patch front panel.

4.2.1 Components

The PD patch consists of a following components:

- **Sensor data** - controls input and display of messages for each sensor.
- **Sensor assignment** – detects triggering of each sensor and assigns them to most dominant tones.
- **Set delaytime** – detects triggering of each sensor, measures speed of movement and time spent within field of view.
- **Audio In** – input of microphone signal. Also shows the current level of attenuation imposed by limiter.
- **Limiter** – maintains system at point of feedback by controlling the amplitude of microphone input signal.
- **Feedback tones** – accentuates resonant frequencies using a recirculating delay.
- **Delayloops** – records feedback tones and loops them according to the amount of time a participant remains within a sensor's field of view.
- **Spectrum** – performs FFT upon input signal, plots graph, and scans results for peaks denoting presence of a resonant tone.
- **Filtering** – applies band-reject (notch) filters at frequencies of resonant tones and assigns sensor control.
- **Tone assignment display panel** – shows relationship between formation of resonant tones and spatial position of participant.
- **Audio Out** – master output volume control.
- **DSP control** – master on/off switch.
- **CPU loadmeter** – displays current CPU load.
- **Notes** – area for leaving comments.

4.2.2 Theory of Operation

The patch utilises several signal processing functions. Some are run in a continuous and parallel fashion, while others operate only upon the output of another process. Figure 60 outlines the basic operation and signal paths used in the PD patch. The patch performs three main functions concurrently: the manipulation of sensor data, processing of audio, and frequency spectrum analysis.

The MIDI CC messages sent by the MIDIsensor board are received by the software patch and are then distributed to the front panel, sensor assignment, and delaytime setting functions. The sensor data is used to track the position and movement of participants located within the speaker-field. An approximate position can be determined in relation to the subjects proximity to a particular sensor (Figure 59). The higher the peak value, the closer the subject is to the sensor. The speed and direction of movement can also be established by monitoring the rate at which sensor values rise and fall. The output of the sensor increases as the subject moves closer towards the centre of the beam. Hence, the value increases as the subject moves into the beam path, reaches a maximum value at the centre, and then begins to decrease as the subject moves out of the beam path. Since the beam is shaped like a cone, an approximate position can be established by observing the rate of increase in these values.

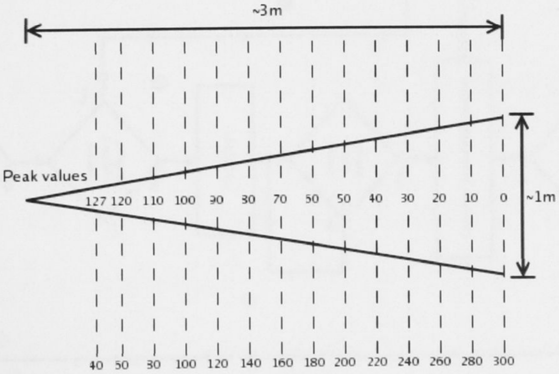


Figure 59: Approximate sensor response.

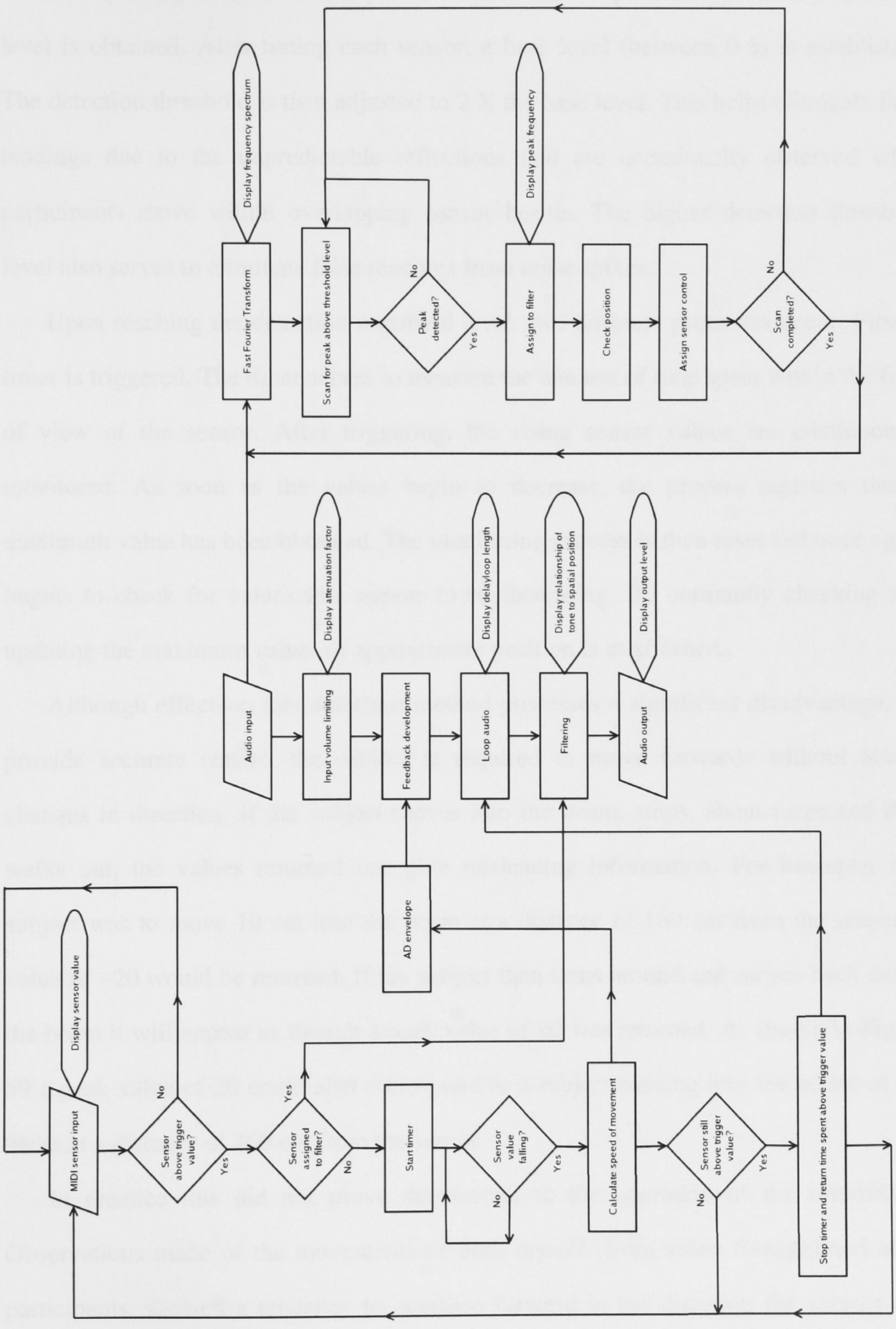


Figure 60: Flow diagram of Pure Data patch operation.

The first step in the detection process occurs when a particular proximity threshold level is obtained. After tuning each sensor, a base level (between 0-5) is established. The detection threshold is then adjusted to 2 X the base level. This helps eliminate false readings due to the unpredictable reflections that are occasionally observed when participants move within overlapping sensor beams. The higher detection threshold level also serves to eliminate false readings from noise spikes.

Upon reaching the detection threshold level, two different processes occur. First, a timer is triggered. The timer serves to measure the amount of time spent within the field of view of the sensor. After triggering, the rising sensor values are continuously monitored. As soon as the values begin to decrease, the process registers that a maximum value has been obtained. The monitoring process is then reset and once again begins to check for values that appear to be increasing. By constantly checking and updating the maximum value, an approximate position is established.

Although effective, this detection method possesses a significant disadvantage. To provide accurate results, the subject is required to move forwards without abrupt changes in direction. If the subject moves into the beam, stops, about faces and then walks out, the values returned can give misleading information. For example, if a subject was to move 10 cm into the beam at a distance of 160 cm from the sensor, a value of ~20 would be returned. If the subject then turns around and moves back out of the beam it will appear as though a peak value of 20 was returned. As shown in Figure 59 a peak value of 20 could also correspond to a subject moving into the centre of the beam at a distance of 260 cm from the sensor.

In practice this did not prove detrimental to the operation of the installation. Observations made of the movements of both myself (from video footage) and other participants, showed a tendency to continue forward in the direction the subject was facing. Abrupt direction changes generally occurred only after a subject had come to a complete stop. The subject would pause shortly to listen to the sounds produced through their proceeding movements, and then move off in a different direction. The effects of the false proximity readings produced by these kinds of movements are minimised with the use of the timing function previously mentioned .

As the subject moves out of the beam path and the sensor value falls below the detection threshold level, the timer is stopped and the total time returned. The time spent within the field of view of a sensor is used to control two important functions: the setting of audio loop length and the extent of filter attenuation.

During the processing of sensor data, audio is continuously piped into the software patch via the computers ADC. The digitised audio branches off to a spectral analysis function before being passed to a limiter. The frequency spectrum analysis is accomplished by performing a Fast Fourier Transform (FFT) upon the input audio stream. The spectrum is recorded and displayed on the front panel. Each block is scanned to determine the most dominant frequencies present within the audio stream. The total number of peaks detected above the set threshold level is calculated, and the corresponding frequencies distributed to the filter function and front panel display.

Meanwhile, the volume of the input audio stream is controlled with the use of a limiter function. The limiter serves to control the level of feedback obtained between the microphone and speaker array. If left unchecked, the system quickly falls into an uncontrollable feedback state. The volume increases dramatically resulting in an unpleasant speaker howl. Hence, the limiter monitors the input amplitude and constantly checks whether a set threshold level has been reached. Once exceeded, the limiter begins to decrease the volume of the input audio. The limiter also monitors the rate of increase in amplitude and adjusts its function accordingly, i.e. The faster the amplitude rises, the faster the limiter attenuates the signal. In this fashion the limiter is able to keep the system at a point of equilibrium. When the system is at rest (i.e. background noise) the input signal is maintained at full volume. As a participant enters the speaker field and introduces further sound input, a feedback tone begins to evolve (discussed below). As the intensity of the tone increases, the limiter decreases the volume accordingly. As the feedback tone begins to decay, the limiter is able to sense how much attenuation is required to re-establish the resting state of the system.

Following the input limiting process, the audio stream is split into two. One path is passed through a short recirculating delay loop, and the other to a longer delay loop.

The short delay loop process is used to accentuate the most dominant frequencies present within the audio stream much like the technique employed in Lucier's *I am sitting in a room.* The input audio stream is continuously loaded into a buffer, and looped over and over. To avoid oversaturation and distortion of the sound, the amplitude is decreased with each successive loop, much like the decay of an echo. Each replaying of the buffer over the incoming audio stream serves to add together the most dominant frequencies present within the sound. The reinforcement of these frequencies continues until they become the only ones perceived within the audio stream.

Where Lucier applied the layering process to a sound source several minutes in length, I have chosen to greatly accelerate the process by using only a small portion. By delaying and looping a very short piece of sound, very quickly, it becomes possible to render a similar effect in real-time.

An attack-decay (AD) envelope is applied to the output of the Feedback tones function. The total time span and extent of amplification is determined by the amount of time spent within a sensor's field of view. The length of the attack section is set using the time period measured between triggering of a sensor and the first peak sensor value obtained, i.e. the rate at which a participant enters a sensors field of view. Provided that there is sufficient sound input, the amount of amplification is altered to ensure that a tone is able to evolve. Thus, with a short attack the amplification level is increased.

Conversely, the amplification level is decreased during a long attack period. This process is employed because during short attack periods less recirculations are performed, whereas during long attack periods many more recirculations can be made before the output is reduced to zero.

The envelope decay period operates in a different manner to that of the attack. The decay period is initially set to the same value as that of the attack. Subsequent movements re-trigger the decay period with an updated measurement of the time spent within the sensor's field of view. Hence, the longer a participant remains, the longer the decay period.

The output of the feedback delay loop is recombined with the original audio stream and passed to the second set of delay loops. This delay loop functions as a global recording device. All of the sounds captured within the speaker field pass through this buffer. The audio is recirculated in the same fashion as in the feedback development process, with the amplitude of each loop decaying over time. The length of each loop is set using the value calculated for the total time spent within a sensor's field of view.

Looped selections of audio are used to build up layers of sounds over time. The resonant tones formed during the feedback process are played repeatedly with each layer creating additional rhythmic complexity. Although the audio within the buffer is looped continuously, only those frequencies which are allowed to pass through the filtering process can be heard by participants.

The output of the combined loops are passed through eight sequential band-reject (notch) filters. The filters are initially set to pass all frequencies below 24000 Hz with a bandwidth of 100 Hz. In some situations an excessive amount of low frequency sound is heard, resulting in a monotonous drone. To correct this, the bandwidths of the first two or three filters are adjusted to allow more mid and high frequencies to be heard. The principle cause of excessive low frequency sound was due to the proximity of the microphone to the subwoofer. The subwoofer volume was adjusted in an attempt to limit the amount of low frequency sound reaching the microphone but it was found that a greater degree of precision could be obtained by also adjusting the filter bandwidths.

Each band-reject filter frequency is set according to the peaks detected upon scanning of the FFT frequency spectrum. The data points of each spectrum plot are scanned sequentially from 0-1023. As the first peak is detected it is assigned to the first filter, the second peak to the second filter and so on until the scan is completed.

No more than eight peaks can be assigned to the filters. In practice, it was rare that more than six peaks were ever detected above the threshold level. Therefore, a total of eight assignable filters was deemed sufficient for successful operation of the system.

Upon assignment of a peak to a filter, the corresponding frequencies heard within the recirculating audio become attenuated. Control of the extent of attenuation is simultaneously switched to the sensor that was active at the time. The recorded position of the subject is retrieved and the attenuation factor is adjusted accordingly (Figure 61).

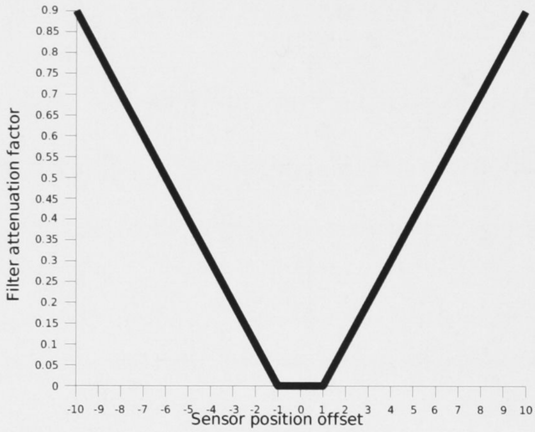


Figure 61: Relationship between filter attenuation and last recorded position of subject.

The output of the filters is passed to the final process in the signal chain: the Audio out function. This function provides master volume control before routing the signal to the computers DAC. It was my intention that the projection of sound into the installation space should be as uniform as possible. Hence, although audio was captured and manipulated as stereo waveforms it was mixed down to one channel prior to output.

In addition to the signal processing functions, three non-critical functions are implemented within the patch. A front panel toggle switch is used to turn operation of the patch on and off. Also, a CPU load meter is used to measure the amount of processing cycles utilised by the patch. This is used to periodically check that the current load does not exceed the capabilities of the audio server. During normal operation the CPU usage of the patch did not exceed 15 %.

Finally, a region was created upon the front panel for the purpose of writing notes. These were used to provide information regarding the set-up procedure and tips acquired from previous sessions.

5

Configuration and Aesthetic Development

The highly technical nature of the system ultimately dictated that a substantial amount of electronic equipment required deployment within the installation space. This chapter discusses issues concerning the configuration of sensors and speakers, and the aesthetic development of the final piece.

5.1 Sensor and Speaker Configuration

Following development of the technical aspects of the system, a means of conveying its function to the viewer was required. To function as an artwork, the system components required deployment within a defined space that would also leave participants both physically and visually unhindered. Therefore, the arrangement of the technology would ultimately determine the visual aesthetic of the piece.

Testing first began on the configuration of IR sensors and speakers. The arrangement of these critical components would determined aspects such as, the dimensions, appearance, and creative focus of the piece.

The first configurations tested, were those used during the PD patch development process. At this preliminary stage, work on the patch progressed by placing several sensors throughout an area defined by four speakers arranged in a square. Sensors were generally placed upon any available flat surface approximating waist height. Tests were also performed with the sensors placed upon the floor.

Observations were made of each sensors response as I moved about the space. It was during this period that I first formulated a method for measuring the rate of movement of a subject. As discussed in section 4.2.2, the rising and falling edges of the sensor's response curve can be used to measure the time spent within a sensor's field of view. Also, by observing the transition between rising and falling values, an indication of speed can be determined. Particular attention was paid to the size of the area that could be monitored by the sensors whilst maintaining freedom of movement within the space. The presence of dead areas (unmonitored spaces between sensors) was also undesirable. The most significant influence upon the configuration of system components was the range of the IR sensors. Each sensor possesses a range of ~2.5 m, therefore the effectiveness of a particular configuration was largely determined by the space between sensors.

After performing these preliminary tests it became clear that a more ordered system of sensor placement was required. Various arrangements of eight sensors were devised (Figure 62 a – f). Cross-talk between sensors is minimised by ensuring that no sensor faces directly towards another.

The most significant issue concerning the success of a particular configuration was the amount of cross-talk exhibited between sensors. As each sensor uses the same method for transmitting and receiving infra-red light, problems would occur when a sensor received the light transmitted by another. This predominantly occurred in those configurations where sensors were facing directly toward each other. Cross-talk was also observed in situations where the projected beams of two or more sensors overlapped. In these cases, the light transmitted from one sensor is reflected towards the receiver of another when a subject enters the overlapping area.

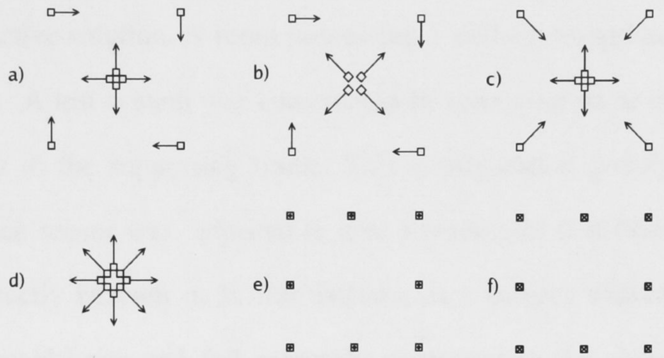


Figure 62: a) - f) Various sensor configurations. Figures e) and f) indicate sensors pointing upward and downward respectively.

Complete elimination of these effects would require a comprehensive redesign of the sensor receiver circuits. Unfortunately, due to the time constraints of the project this could not be undertaken. However, the effects could be minimised through careful consideration of each sensors placement. A configuration was required where none of the sensors would be facing each other. Figure 62 e) and f) show two configurations with sensors pointing upward from the floor and downward from the ceiling respectively. Locating the sensors upon the floor proved an unsuitable option. Only a small fraction of the sensors usable range could be utilised by a participant walking around or past it, as most of the transmitted light was focused directly in front of the device. The response of the sensor was also unsatisfactory if a participant stepped directly over the device, resulting in abrupt changes in sensor output. Ultimately, interpretation and utilisation of these fast moving values proved exceedingly difficult.

Conversely, a configuration where the sensors point downward from the ceiling appeared to be a much more attractive solution. A room possessing a ceiling comprised of removable panels was located. A test system was constructed by removing some of the panels and mounting sensors to the supporting frame. This configuration greatly simplified the tuning process. Each sensor was adjusted to give a reading of 0 at floor level and 127 when standing directly beneath it. In this fashion, as a subject walked beneath the sensor, the output would rise and fall smoothly compared to the sharp spikes observed when stepping over it.

Locating the sensors and other system components within the ceiling provided a tidy solution, effectively hiding all of the technology from view. Participants could move about the space unrestricted and unaware of the working components of the system. However, in practice this approach presented some significant problems. I did not have unrestricted, long-term access to a site in which to build, test and install the system. Consequently, all of the equipment was required to be transported, installed and then disassembled at the end of each day. Therefore, installing all of the equipment into the ceiling each time I wanted to use the system proved to be a time consuming and ultimately an impractical solution.

As I would be installing the system within spaces used by others, I was required to minimise the level of disruption and inconvenience. I was also unable to make destructive changes to the room. Therefore, a different method of housing the system components was required. Throughout the research period I had observed examples of how system components could be suspended from the ceiling. One such example is illustrated in the work, *Interdependent System* by Akabane, Tamai and Ogura[126] (Figures 63 and 64).

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[126] Interdependent System. [Internet ref.].

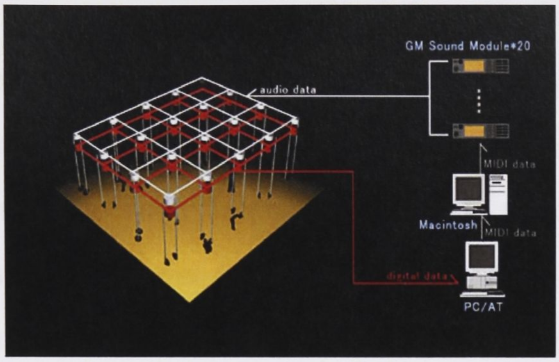

Figure 63: Suspended frame utilised in *Interdependent System.*

Figure 64: *Interdependent System* - Akabane, Tamai and Ogura, (1996).

My thoughts turned to the development of a modular frame that could be suspended from an existing ceiling (Figure 65).

The frame consists of three layers, each formed by joining four smaller sections together. This modular design would allow the system to be easily transported compared to a design of fixed dimensions and also eliminates the need to make destructive changes to the original ceiling.

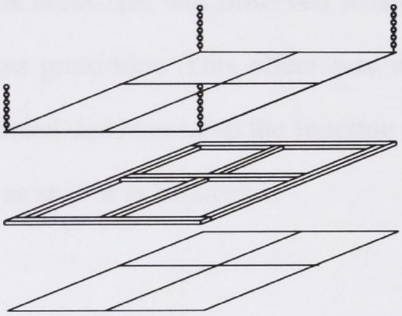

Figure 65: Suspended housing frame.

The upper layer is used to support the various system components and the middle provides strength and rigidity. The bottom layer takes the form of a thin panel, film or fabric covering that serves to hide the sensors and speakers from view. Sensors are attached to the frame as depicted in Figure 62 f), with wiring routed across the surface of the upper layer as required. Holes would possibly require cutting in the bottom layer to allow for the transmission and reception of IR light. Finally, chain links are fixed to each of the four corners to assist in securing the structure to the ceiling.

Securing such a frame proved more difficult than anticipated. As individuals would be walking beneath the structure, safety became of primary concern. Therefore, a location was required that possessed a ceiling of adequate height that also provided the means to safely support the weight of the frame. All of the locations immediately available to me did not possess any pre-existing means for the hanging of heavy structures and/or did not permit the installation of such fixtures. After careful consideration, the application of a hanging frame structure was abandoned.

Investigation returned to the arrangement of sensors at ground level. Testing began in determining the effectiveness of configuration shown in Figure 62 d). The effects of cross-talk between sensors in this arrangement were almost entirely removed since no sensor faces directly towards another. However, some cross-talk was observed when a subject passed between two adjacent sensors at close proximity. This effect was not observed in all instances and therefore was not considered detrimental to the function of the system. Testing began with a rudimentary set-up as shown in Figure 66.

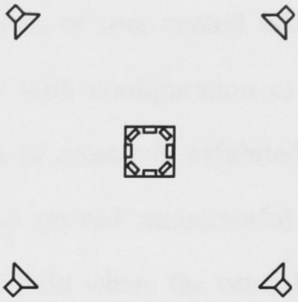

Figure 66: Test set-up based on configuration (d).

Eight outward facing sensors were placed on top of a box approximately 0.5 m in height. Four inward facing speakers were arranged in a square surrounding the sensors. The remaining components (computer, midi hardware, subwoofer, etc.) were placed to one side. This configuration allowed freedom of movement throughout the space with the exclusion of the very centre and speaker locations. The majority of usable space within the speaker field was covered by the sensors. Those areas not within the sensors' field of view were typically less than the size of an average person.

Although effective, this configuration presented several technical and aesthetic compromises. Of primary concern was the affect of the central structure upon the behaviour of participants. The presence of an object within an otherwise empty space immediately draws the attention of the viewer. Participants were generally attracted directly to the structure and would try to provoke an immediate response from the sensors. While undesired, this did not prove detrimental to the operation of the system. The offset of the sensors meant that very little response was observed at such close proximities since the output would remain fixed at maximum value. Hence, quick hand movements directly in front of the sensor were not registered by the system. Once participants realised that an immediate response could not be provoked they would lose interest in the central structure and begin to explore the types of movements that would produce an effect. Movement about the space generally occurred in a circular fashion as participants were unable to move from one side to the other without passing around the central structure. While this did not have an adverse affect upon the operation of the system, I was somewhat unsatisfied that movement was not completely unrestricted. Consequently, I decided to experiment further with the configuration of sensors in an attempt to encourage movement towards the perimeter of the speaker field.

Configurations a), b) and c) of Figure 62 show combinations of four central and four perimeter sensors. Problems were immediately apparent with configuration c). Although not facing directly towards one another, the amount of cross-talk exhibited meant that it was an unsuitable option. Configuration a) also proved unsuccessful. Problems occurred at the midpoint of each side of the speaker field where the central and perimeter sensor beams overlapped. The excessive overlap caused the system to rapidly switch between delay time values and envelope triggering of each sensor resulting in an unpleasant stuttering effect.

The final configuration tested was that depicted in Figure 62 b). Initially I was sceptical that this configuration would function correctly due to the central sensors pointing almost directly at those on the perimeter. Much to my surprise it was this configuration that produced the most satisfactory results. The conduit used to collimate the IR beam prevented direct cross-talk between sensors although a small amount still remained due to reflections. In practice this had little effect upon the operation of the system. Also, the problems produced by excessive sensor overlap did not occur with this configuration and the area covered suffered from minimal unmonitored areas.

5.2 Construction and Aesthetic Considerations

With the fulfilment of sensor and speaker arrangement, my attention turned towards the construction of a frame (Figure 67) that would house the four central sensors and remaining system components.

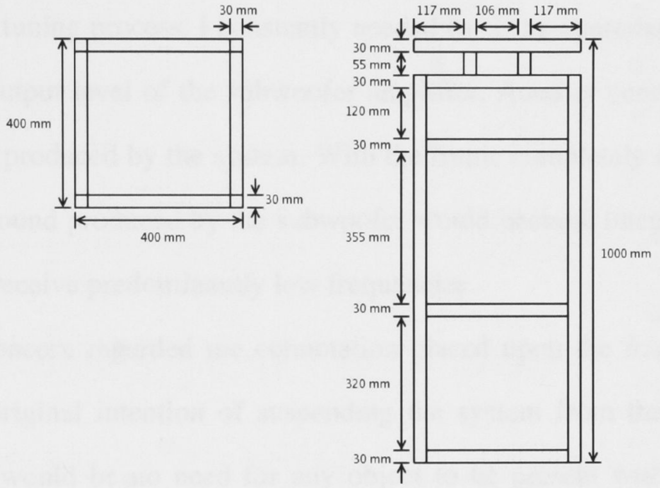

Figure 67: Housing frame dimensions.

The dimensions of the frame were chosen upon the following considerations. During the testing process, the sensors were mounted at a height of ~0.5 m. While adequate for testing purposes, the range of each sensor suffered. To correct this, each sensor was raised to a height of 0.9 m. This enabled the sensors to receive a more intense reflection due to the IR beam hitting a greater surface area of a subject's body.

A width of 0.4 m allowed the computer and protruding cables to be adequately housed, while the height of each tier was selected to allow easy placement of the various components. The top panel was made removable to allow access to the sensors for tuning purposes. The windowed shape of the upper tier allows the four central sensors to be placed either upon the sides or the corners of the frame. An added benefit of the frame is that each of the four perimeter sensors can be located upon the same stand used to support the speaker. This allows the wiring of both sensors and speakers to be run along the same paths therefore minimising the amount of visible cabling.

Following population of the frame, a question was raised concerning the visibility of the system components. I had originally intended to hide as much of the technology as possible from view to minimise the amount of distraction caused by the presence of the frame. I considered enclosing the box with panels but soon abandoned the idea after I began using the system. I quickly discovered that access to the system components was vital to the tuning process. I constantly needed to change parameters within the PD patch and the output level of the subwoofer amplifier. Another concern was the effect upon the sound produced by the system. With the frame completely enclosed there was a risk that the sound produced by the subwoofer would become intensified, causing the microphone to receive predominantly low frequencies.

My final concern regarded the connotation placed upon the frame as a sculptural object. If my original intention of suspending the system from the ceiling had been possible, there would be no need for any object to be present within the space. By enclosing the frame and removing the technology from view, the frame immediately becomes the focal point of the space. This was not my intention. I wanted participants to ignore the presence of the technology and focus more upon the empty space surrounding it. Hence, it became preferable to show that the frame was simply for housing the technology and did not carry any symbolic meaning. By leaving the technology exposed, participants can immediately identify its purpose and are then free to question the role of its function rather than the meaning of its presence as a sculptural object.

The final aesthetic consideration was the colour of the frame, as I was unsatisfied with the natural appearance of the materials used in its construction. The colour was not regarded of critical importance, but was to conform with those of the system components. Thus, the frame was painted white to avoid placing further unwanted symbolic meaning upon the frame.

6

Integration of System Components

With completion of the hardware and software aspects of the project, work began on the final construction and application of the installation piece. This chapter details the processes involved in the selection of various components required for its successful operation.

6.1 Interconnection

Components required for operation of the system, included:

- Roland UM-4 USB MIDI interface,
- Sony ECM-DS70P binaural microphone,
- Digitor speaker system,
- Speaker stands,
- Various audio and power cables, and
- Housing frame.

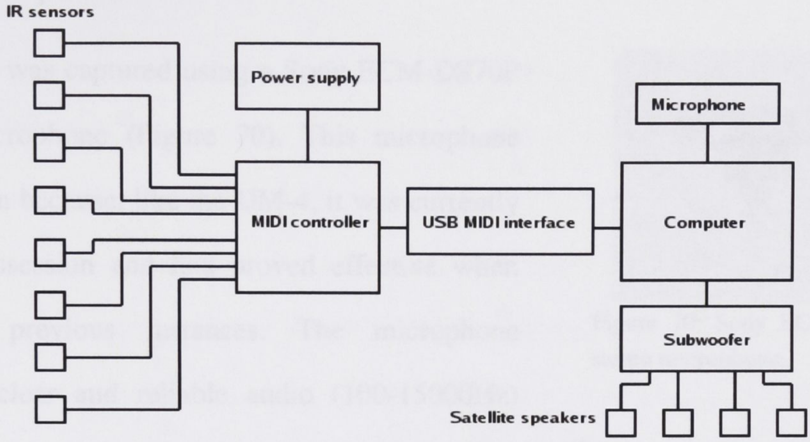

Figure 68: Integration of installation system components.

6.2 Components

6.2.1 MIDI Interface

A Roland UM-4 USB MIDI interface (Figure 68) was used to convey MIDI data generated by the MIDIsensor controller to the computer. The UM-4 provides four MIDI inputs and outputs although only one input port was required for the correct operation of the system. The UM-4 was used in this project as it was currently in my possession and had proved to be effective and reliable in past instances when used with Linux and PD. Therefore, further financial outlay upon a single port interface was deemed unnecessary.

Figure 69: Roland UM-4 USB MIDI interface.

6.2.2 Microphone

Figure 70: Sony ECM-DS70P stereo microphone.

Audio was captured using a Sony ECM-DS70P stereo microphone (Figure 70). This microphone was chosen because, like the UM-4, it was currently in my possession and had proved effective when used in previous instances. The microphone provided clear and reliable audio (100-15000Hz) with a reasonably low background noise level (-38.0 db)[127].

The use of a stereo microphone was not critical for the correct function of the system. Tests performed with both mono and stereo microphones showed no discernible difference between the sound captured by each microphone.

The electret capsules within the microphone contain an integrated preamplifier that require a small power source to function. A power supply (Figure 71) was constructed according to the schematic diagram shown in Figure 72.

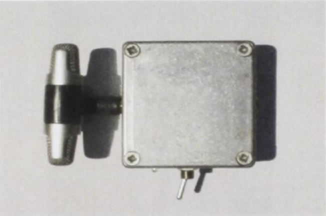

Figure 71: Microphone power supply.

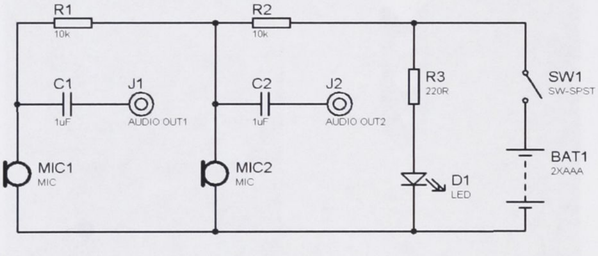

Figure 72: Schematic of binaural microphone power supply.

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[127] Sony : ECM-DS70P (ECMDS70P) Technical Specifications : United Kingdom. [Internet ref.].

6.2.3 Speaker System

Audio was projected into the installation space using a Digitor home theatre speaker system (Figure 74) consisting of a subwoofer and four satellite speakers. The system was inexpensive, light, and provided adequate sound quality.

The inexpensive nature of the system provided piece of mind when dealing with loud feedback processes as I was able to freely experiment without fear of damaging the system. The low weight of the components allowed easy transport of the system and reduced the amount of time spent setting up the system. Also, the small size of the satellite speakers allowed them to be placed upon the stands without the need for heavy securing bolts. The output level and frequency response was deemed sufficient for the needs of this project. Although originally purchased solely for testing purposes, the speaker system maintained satisfactory performance after extended periods of use. Therefore, purchase of a more powerful and sophisticated system was deemed unnecessary at this point in time.

Figure 73:
Speaker/sensor stand.

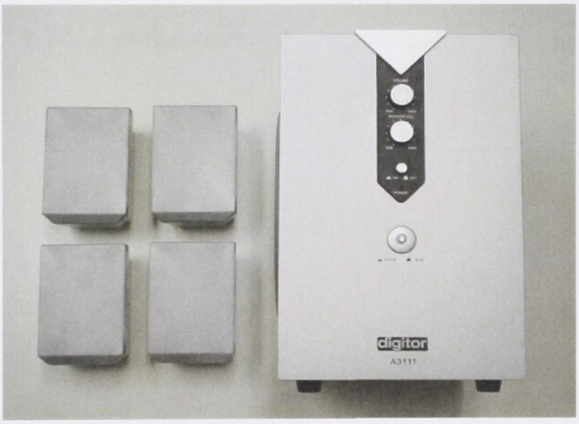

Figure 74: Speaker system.

Four adjustable stands (Figure 73) were used for mounting of the satellite speakers. The heavy steel base allowed each stand to be left unsecured with minimal risk of it falling if touched by viewers of the work. The height of each stand was adjusted to 900mm.

6.2.4 Power and Cabling

Three 240V AC outlets were required for operation of the computer, speakers and sensor system. These outlets were provided by a 4-way power distribution board. A 10 m extension cord was run from the distribution board to a nearby wall outlet. This method ensured that only one power cable would be present within the installation space therefore minimising possible trip hazards and its effect upon the aesthetic of the piece.

Approximately 22m of 4-core telephone cable was used to provide power and data connections to the eight IR sensors.

Other electrical cables required for operation of the system, included:

- 10 m of 2-core speaker cable,
- 30cm MIDI cable,
- 1m USB cable,
- 1m, 3.5mm stereo male to male audio cable, and
- 0.5m, 3.5mm stereo male to 2 x RCA audio cable.

6.2.5 Housing Frame

A wooden frame (Figure 75) was constructed for the purpose of housing the various system components and to facilitate their interconnection. The frame consists of four tiers (Figure 76). From highest to lowest the frame was populated as follows:

- Four IR sensors and microphone,
- MIDIsensor controller board and USB MIDI interface,
- Computer,
- Subwoofer, power supply and distribution board.

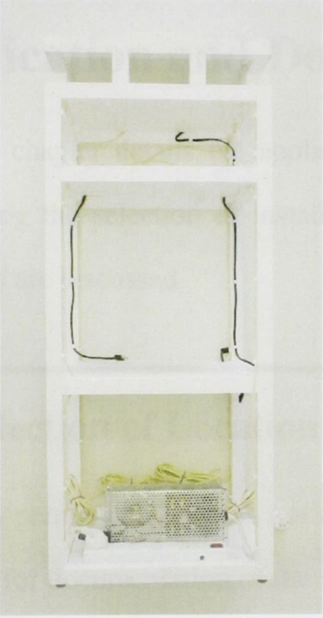
Figure 75: Housing frame.

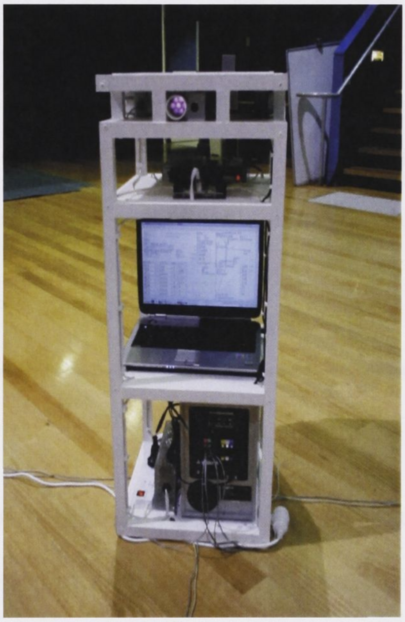
Figure 76: Frame with components in-situ.

All of the required cables were secured to the frame using plastic U-clips. The power supply and distribution board were secured to the base of the frame using cable ties. This served not only to maintain a neat and tidy appearance but helped to minimise the time taken to set-up and dismantle the system. The speaker and perimeter sensor cables were bundled and secured with rubber bands, allowing them to be unravelled when required.

7

Application and Documentation

This chapter details the application and documentation of the installation. Issues concerning the selection of installation space, set-up procedure and operation of key functions are discussed.

7.1 Selection of Location

As a permanent site was not available for exhibition of the piece, a location was required for documentation of the work. Selection of a suitable location was determined upon the following considerations:

- Physical dimensions,
- Availability,
- Access to power facilities,
- Security, and
- Acoustic qualities.

Foremost, the location was required to be of adequate physical dimensions. To ensure successful operation and documentation of the system, the location was required to accommodate the housing frame and speaker field whilst also providing sufficient surrounding space to allow unrestricted movement within and around the piece. As shown in Figure 77 the speaker field was to be arranged in a square with with a diagonal length of 5 m.

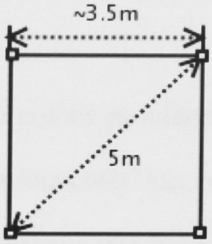

Figure 77: Speaker field dimensions.

This arrangement was chosen due to the length and shape of each sensor's field of view. Each sensor has a range of approximately 2.5 m. As participants typically never pass immediately beside a sensor, there remains approximately 0.5 m of unused space surrounding each sensor. Therefore by applying an offset of 0.5-1 m, the 3.5 m range between sensors on the outer perimeter is adequately covered. The sensors placed within the central frame are able to utilise their full range without the need for offsetting due to the shorter distance between the centre and corners of the speaker field.

Although not all of the usable space within the speaker field is covered by the chosen configuration, the amount of unmonitored space is kept to a minimum. Participants seldom stand perfectly still, tending to shift their weight between feet whilst standing. These small movements are sometimes sufficient to trigger a sensor as a subject moves partially into its field of view whilst remaining within an unmonitored area. Also, as the human body is wider than it is in depth, instances may occur where a subject may be present within an unmonitored area when facing in one direction, yet by turning, enters a triggerable area.

After much deliberation, a room was located that provided the physical dimensions, power, privacy and security required for the documentation process.

One concern in choosing this location was the lack of acoustic colour provided by the room. The space is normally used for band rehearsal and consequently has been constructed with the acoustic qualities associated with traditional instruments in mind. The walls (Figures 78 and 79) and ceiling (Figure 80) were constructed from sound absorbing materials and the floor was carpeted to minimise reflections from the wooden flooring. Although acoustically engineered, by clapping ones hands or speaking loudly, a level of reverberation, albeit small, could be discerned.


Figure 78: Sound adsorbing panel.

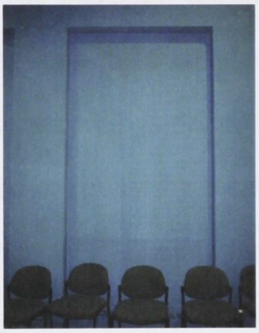
Figure 79: Sound adsorbing panel.

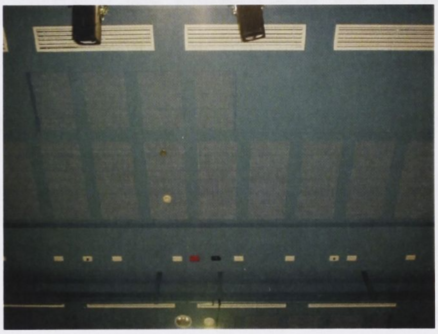
Figure 80: Sound adsorbing materials within ceiling.

Despite the apparent acoustic disadvantages, the room satisfied the primary requirements of adequate dimensions and availability. Access was acquired and the documentation process commenced.

7.2 Video and Audio Recording

The documentation process was centred around video and audio recordings of the system in operation. Two video cameras were used. The first (Sony DCR-TRV16E) was positioned upon the balcony/gallery, providing a clear view of the installation from above. The second, (Canon MXV3I) was located to one side of the system at ground level. This camera was equipped with an external microphone (Rode video mic).

Neither camera possessed a sufficiently wide viewing angle to capture the entirety of the installation. This meant that each camera had to be situated a distance of at least 8-10m from the system. The increased distance between the camera and the speaker field had an adverse affect upon the quality of the captured audio. As a consequence, noise was introduced to the recordings due to the high levels of amplification required. This situation could not be remedied by locating the microphones within the speaker field. Doing so added clutter, thus affecting the aesthetic of the piece and restricting the movement of participants. Consequently, the captured audio lacked a sense of what it truly sounded like when standing within the speaker field. To correct this and facilitate the capture of higher quality audio recordings, another method was required.

The notion of capturing audio as I moved within the speaker field appealed to me greatly. I first considered the use of wireless microphones placed in a fixed position within the speaker field but soon abandoned the idea in preference to the use of a binaural microphone carried upon my person.

> *This is perhaps the most obviously correct way of approaching the problem of full three-dimensional (3D) spatialisation of sound. Exact duplication of what the ear would hear in a natural situation should produce the best reproduction. In fact, under a certain limited set of circumstances there can theoretically be no better or closer approach to real-world performance.*[128]

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[128] Malham. *Approaches to Spatialisation.* p171.

The binaural effect created by the ears is a result of three different cues; inter-aural arrival time differences, inter-aural intensity differences, and the effects of the pinnae (outer ear)[129]. These cues, combined with other physical and cognitive processes, determine how the brain locates a particular sound in space. The most common approach taken when performing binaural recordings, is with the use of dummy head microphones. These systems have been in widespread use since the 1920's and generally take the form of a simplified model of the human head with microphones inserted into the ears[130]. By arranging the microphones in this fashion, sound is captured as it would be by the ears of a listener. When the recorded sound is played back, the listener is able to gain a sense of how the sounds were perceived by the person present during the original recording. Although these simplified systems are capable of reproducing very realistic sound fields, they are not without their limitations.

Every individual is different, therefore what sounds correct for one person may not be so for another. In practice, the unique physiology of each person ultimately determines how sounds are perceived by the individual. Also, dummy head recordings generally must be listened to with the use of headphones. When replayed using loudspeakers, sounds originally intended only for the left ear are combined with that of the right (and vice versa). This is otherwise known as inter-aural crosstalk cancellation[131]. These effects can be corrected using digital signal processing but pose a significant computational burden. Despite the disadvantages, it was deemed that binaural recordings would provide satisfactory results for the purposes of the documentation process.

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[129] Runstein and Huber. *Modern Recording Techniques 2nd Ed.* p37.

[130] Malham. *Approaches to Spatialisation.* p171.

[131] Ibid. p172.

A pair of binaural microphones were obtained, and clipped to the ear-pieces of a pair of headphones. In this fashion I was able to listen to the sounds formed as I moved about the speaker field and could simultaneously monitor the recording levels. Audio was captured using a Sony MZ-R910 MiniDisc recording device. The small size of the device allowed me to carry it upon my person without feeling encumbered or distracted during the recording process. By clapping loudly at the beginning of each recording, the audio could be synchronized with the captured video during the editing process.

Video was captured as one continuous take, with each camera running over the course of approximately 1 hour. The captured video was then transferred to a computer and cut into individual takes.

7.4 Digital still images

Still images were recorded using a Sony DSC-F55E digital camera. Images were taken of each of the installations components, including; the housing frame, speaker field, camera locations, and typical features of the room. These images were used to supplement the video and audio recordings and can be seen as figures mentioned throughout this thesis.

7.5 Discussion

During the documentation process a number of observations were made regarding the operation of the system, including:

- installation of the various system components,
- tuning processes,
- general operation, and
- assessment of key system functions.

7.5.1 Installation

Installation of the system proceeded by first positioning the housing frame within the centre of the space. The various system components were then placed in their designated positions within the frame (Figure 81). Next, the speaker field was assembled by placing the four stands in a square surrounding the frame. The perimeter sensors and speakers were then placed upon each stand (Figure 82). Wiring to each speaker/sensor pair was passed through the shaft of the stand and routed to the frame. Power was applied to the system via an extension cord routed to an outlet located in the floor (Figure 83). Figure 84 shows the assembled installation.

Figure 81: Installed housing frame.

Figure 82: Stand with sensor and speaker.

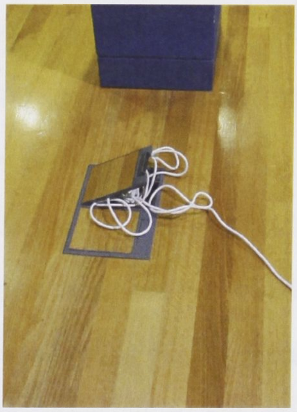

Figure 83: Power outlet with extension cable.

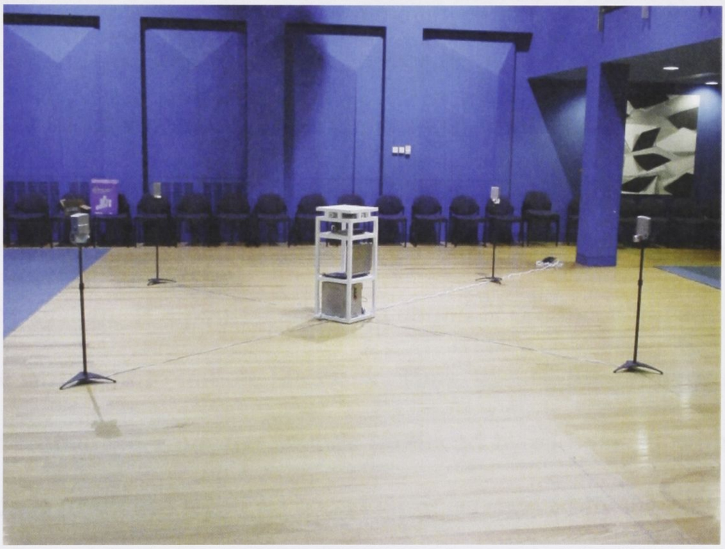

Figure 84: Assembled installation.

7.5.2 Tuning

Upon installation, tuning of the system was commenced. Each IR sensor was adjusted to give its maximum range and offsets were then applied where appropriate. The system was checked for the presence of cross-talk and an assessment made of the location and size of any unmonitored areas. The position and direction of each sensor was then adjusted accordingly.

Retuning of each sensor was undertaken approximately 30 mins after initial assembly to correct for any drift in output caused by heating. After the system was deemed sufficiently stable, tuning of the feedback process was begun.

First, the output volume of the computer and speaker amplifier were adjusted to achieve a comfortable listening level. The sensors were then triggered by walking within the speaker field. An equilibrium point is established by adjusting the trigger and volume level of the limiter. The triggering level is adjusted by observing its behaviour during instances of abrupt volume change (such as clapping) and the rate at which it returns to its idle state. The limiter must respond with sufficient speed to prevent fast, loud sounds from over-saturating the microphone input, resulting in distortion.

Conversely, a sufficient amount of amplification must be maintained for the system to initiate feedback. The influence of ambient noise can be minimised by adjusting the limiter trigger level, but excessive noise levels will have an adverse effect upon the operation of the system. If a particularly loud sound is produced outside of the speaker field, it may be sufficient to initiate feedback. Thus, successful tuning is achieved when feedback can be obtained from the sounds produced by movement within the speaker field, but also fade to silence when no sound input is received by the computer.

Tuning of the frequency content of the sound produced by the system is also required. Observations were made regarding the types of tones produced by different sound sources. The response of the system to various frequency ranges was tested by whistling, speaking, hand clapping, and stamping the feet. In this fashion, an assessment of high, mid and low frequency ranges is made. If the system appeared to be favouring a particular range, independent of the types of sound input, the bandwidths of each band-reject filter were adjusted accordingly. Following these steps, if an excessive amount of low frequency sound is still observed, the output volume of the subwoofer amplifier is reduced until a satisfactory level can be obtained.

7.5.3 Operation

Upon completion of the tuning process, video and audio documentation of the installation was undertaken. During this process, an assessment of the key functions of the system was made.

Figure 85: Video documentation - Scene 1.

Development of the piece begins as a participant enters the speaker field (Figure 85).

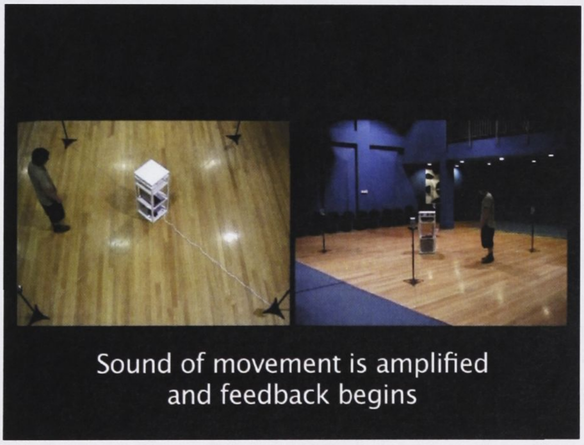

Figure 86: Video documentation - Scene 2.

Sounds produced by the movement of the subject (foot fall, movement of clothing, speech, etc.) are amplified by the system and fed into the feedback delayloop process of the software patch (Figure 86). It is here that the relationship between the presence of the participant and the evolution of sound is formed.

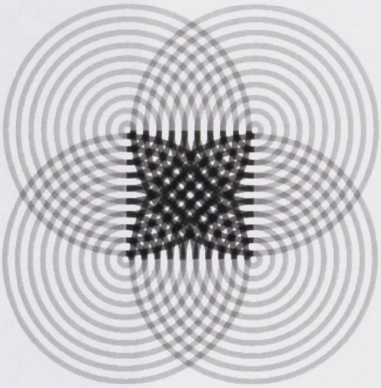

Figure 87: Simplified representation of waveform interference patterns within speaker field.

When the speaker field is unoccupied, the waveforms produced by the speakers should theoretically collapse into a central point (Figure 87). The microphone situated at this point therefore receives sound of the same intensity and frequency content from each of the four corners of the space.

Upon entering, the sound waves become disrupted. The body acts as a object which absorbs, reflects and diffracts the waveforms travelling around it. Also, any sound produced by the subject will spread out from their position, adding further complexity to the interference patterns of the sound field. The effect of this disruption upon the natural resonant state of the system serves to produce a unique signature of ones presence at a particular point in space and time. This process is illustrated in Figure 88.

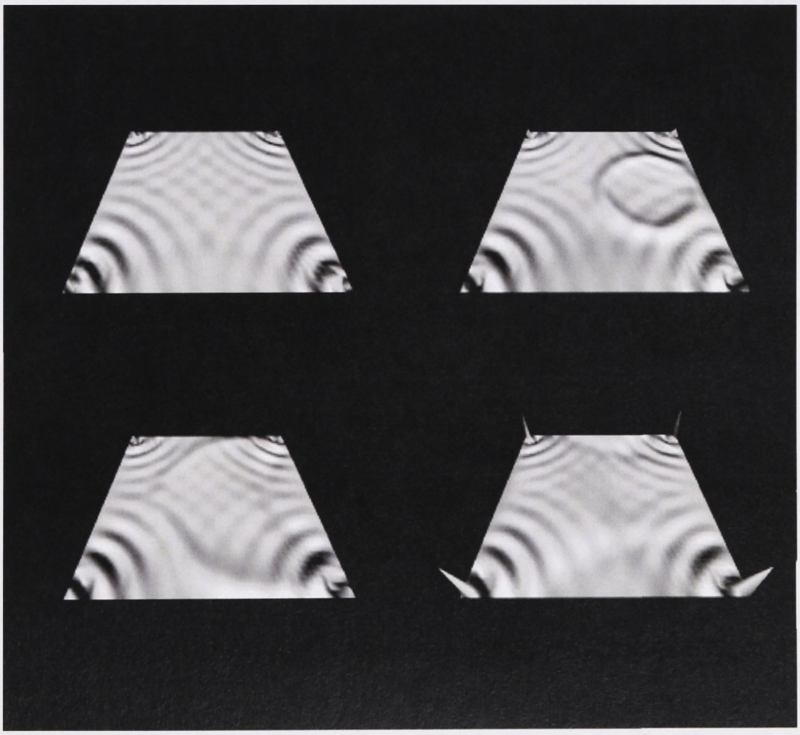

Figure 88: Disruption of interference patterns caused by presence of subject.

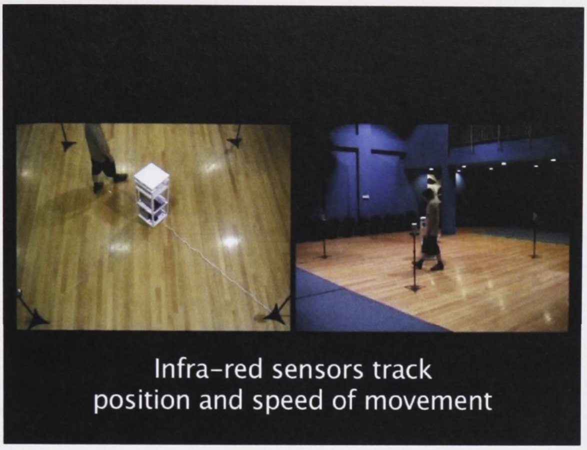
Infra-red sensors track
position and speed of movement

Figure 89: Video documentation - Scene 3.

The sensors located within the space register the presence of the subject and begin reporting the position, speed of movement and total time spent within each sensor's field of view (Figure 89). These values are used to set variables within the different functions of the software patch. Switching and routing of sensor and filter control is dictated by the output of these functions.

Figure 90 a-d) shows typical response curves as a participant moves within a sensor's field of view.

a) Subject begins outside of the sensor's field of view. The sensor response and filter attenuation remains unchanged.
b) As the subject enters, sensor values begin to increase. As the subject stops, the peak value is recorded. During this time, given sufficient sound input, a tone begins to evolve. Once detected, the sensor is switched to control the level of filter attenuation.
c) The sensor value decreases as the subject changes position. As the subject moves away from where the tone was created, the level of filter attenuation is increased. The further the subject moves from this point the greater the attenuation. As attenuation increases the tone begins to be removed from the recirculating audio.
d) The sensor value increases as the subject returns to the original position. The level of filter attenuation decreases and the tone can once again be heard.

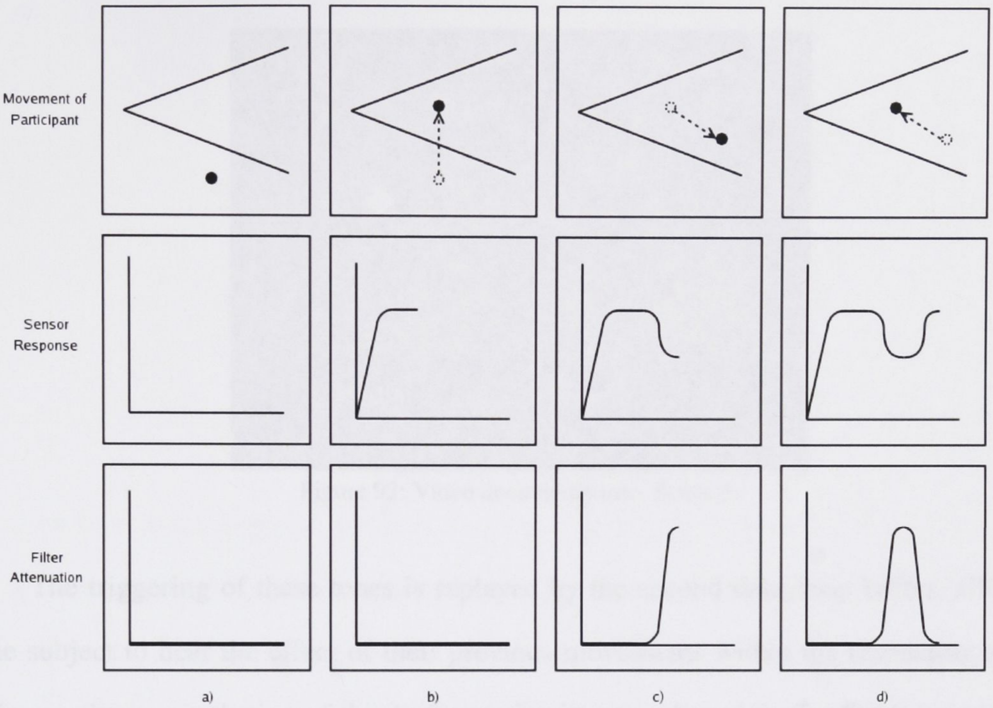

Figure 90: Typical sensor and filter response as subject moves within sensor's field of view.

Figure 91: Video documentation - Scene 4.

Meanwhile the continuous recirculation of audio within the feedback loop begins to accentuate the most dominant frequencies contained within the input stream. As the subject continues to move within the speaker field, the amplitude envelope of the feedback loop is triggered, causing the tones to become audible (Figure 91).

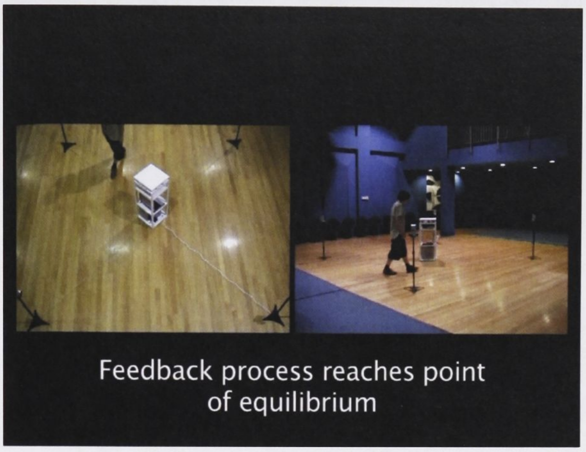

Figure 92: Video documentation - Scene 5.

The triggering of these tones is replayed by the second delayloop buffer, allowing the subject to hear the effect of their previous movements within the resonating space. The continuous replaying of this buffer maintains stimulus of the feedback process and allows the system to reach a point of equilibrium Figure 92). The persistence of the most dominant frequencies within the space means that very little subsequent sound input is required on behalf of the participant in order to return the system to a state of feedback.

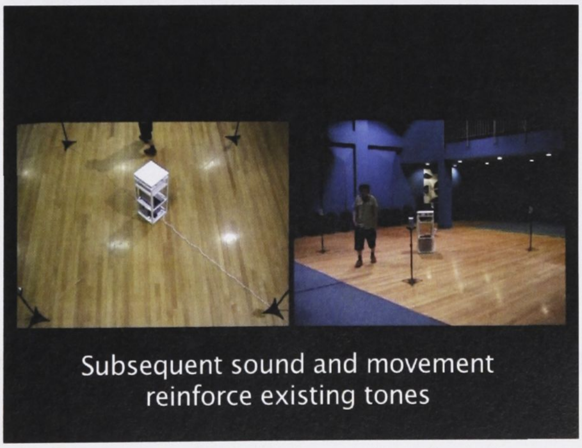

Figure 93: Video documentation - Scene 6.

As an equilibrium is formed, the participant is free to explore the effect of their presence within the space (Figure 93).

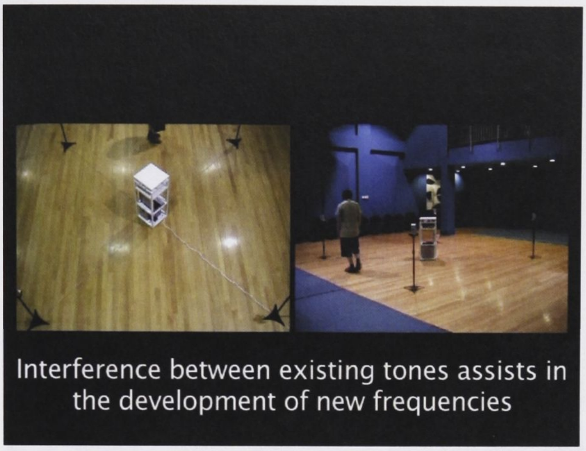
Interference between existing tones assists in the development of new frequencies

Figure 94: Video documentation - Scene 7.

As layer upon layer of sound is developed, interference between the various tones begins to alter the frequency content of the input stream (Figure 94). This allows the development of new tones. The effect may manifest itself as beating between tones of similar frequency, or may appear very different and seemingly unrelated to the tones already present. This process encourages participants to explore the space further, by experimenting with different locations, movements and sound input.

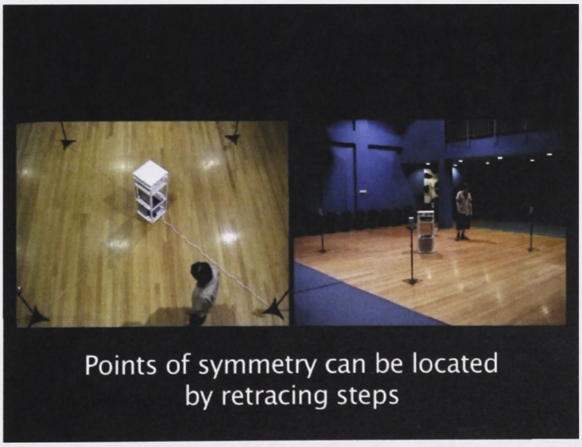
Points of symmetry can be located by retracing steps

Figure 95: Video documentation - Scene 8.

Conversely, points of symmetry can be located by developing a tone, letting it decay, and then proceeding to a new location. Those positions where similar tones are created, give an indication of the level of symmetry within the space. This can be explored further by retracing the paths between these positions and observing the extent of similarity exhibited between tones produced at these locations (Figure 95).

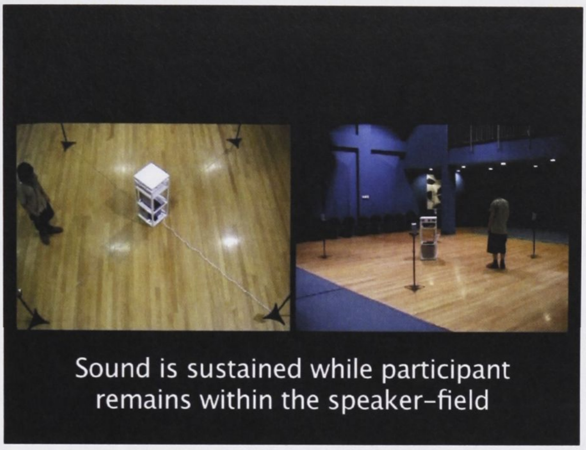

Figure 96: Video documentation - Scene 9.

The evolution of sound is sustained for as long as the participant remains within the speaker field (Figure 96). The sound may be reduced to a very simple state if the participant remains in one location for an extended period of time. However, it will continue to recirculate if the participant remains within the field of view of a sensor, as even very small movements will cause re-triggering of feedback tones.

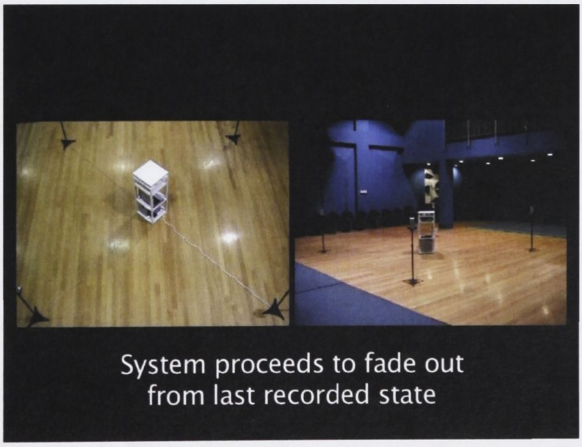

Figure 97: Video documentation - Scene 10.

As the participant exits the speaker field, the system continues to operate using the last recorded sensor data (Figure 97). In the absence of further triggers, the recirculating audio begins to fade out and the system eventually returns to its original state.

7.5.4 Assessment of Key Functions

Although success of the piece as an artwork is ultimately left to the individual, an assessment of the various technical and aesthetic functions of the piece can be made.

The principle concerns lay in the function of the infra-red sensors. In particular, the stability and range exhibited by each device. The system required retuning several times following the initial set-up, due to drift observed in output values. As previously stated, this was attributed to the heat produced by the IR LEDs. Ultimately, this meant that the system could not be left unmonitored for extended periods of time. If the system was to be installed without my supervision it would be required to function flawlessly. Therefore, a redesign of the electronics to improve stability would be highly beneficial.

The shape and range of the beam transmitted from each sensor also played a significant role in the performance of the installation. The shape of the beam allowed several useful measurements to be obtained simultaneously but also imposed limitations upon the physical arrangement of the sensors. While the chosen configuration assisted in minimising cross-talk and unmonitored areas, movement within the speaker field was somewhat restricted. To increase the sense of freedom felt when moving about the space, the dimensions of the speaker field should be increased.

If the sensors had been located above the heads of participants, the size of the monitored area could be simply increased with the addition of more sensors. This was not a viable option when locating the sensors at ground level due to the increased amount of wiring involved. Therefore, the size of the speaker field in its current configuration could only be increased by improving the range of the existing sensors. However, it may instead be worth reconsidering the use of a suspended frame system, or an alternative object tracking method such as video tracking.

Ultimately, problems are evident in the application of any electronic sensor system. For the purposes of this project, the infra-red based system provided satisfactory performance and would easily lend itself for use in future applications.

The quality of the microphone and speaker system was another point of concern. Although the sound produced by the system was of an acceptable quality, questions could be raised as to the true spectral content of the sound within the space. To gain an accurate representation, the microphone and speakers should ideally possess a flat response spanning a large frequency range. Also, the ADC and DAC of the computers sound card should be of the highest quality to ensure accurate digitisation of the input stream and subsequent analogue conversion upon output. Sound was captured and replayed as 16-bit samples at a rate of 48 kHz (the maximum attainable for this system). It is suggested that in future efforts, to further improve the quality of reproduction, a system capable of providing 24-bit, 96 kHz sampling or higher should be used.

The amplifier in both the computer and speaker system should also provide a background noise level as low as possible to avoid the amplification of noise inherent within the electronics. Excessive background noise only served to complicate the tuning process, as feedback would be initiated without the need for additional sound input. In practice, this was only noticeable when the volume of the computer and amplifier were set to maximum levels. Although not critical, by reducing the background noise level a higher level of amplification could be obtained without compromising the sound quality or general operation of the system.

The performance of the MIDIsensor board was excellent. The PIC16F877 provided fast and reliable operation with seemingly instantaneous conversion of sensor voltages into MIDI messages. Performance of the Linux operating system and PD software was also of a high standard. The system provided error free audio I/O whilst maintaining CPU usage below 15%.

The presence of the central frame, while unavoidable, proved to be a considerable distraction. Although measures were taken to minimise its impact, my initial concerns regarding its presence within the speaker field ultimately proved correct. The presence of a central structure places emphasis upon the interaction between the viewer and the object, rather than the creation of a seamless interface between their body and the space surrounding it. To fully appreciate this interaction, movement within the speaker field should be unrestricted. Also, the visibility of any underlying technology should, if not completely eliminated, at least work in a harmonious fashion with the existing aesthetic of the space. If the presence of objects within the space is unavoidable, then they should immediately inform the viewer that they are an integral part of the experience. The visual aesthetic should therefore be closely tied to the desired aural experience.

Although the installation involves considerable technical complexity, it is not in my intention that the viewer should require a detailed understanding of the underlying processes of the system to fully appreciate the work. That said, a certain degree of patience is required of the viewer. It is only by spending time with the system that a participant can develop a sense of instrumental control. Although flaws were evident in its execution, the elimination of a traditional tangible interface encourages interaction in an intuitive manner. Interaction with the system is therefore formed through a state of activated listening and the stimulation of decision making processes, rather than any prerequisite need of manual dexterity or overt physical skill.

With the cues formed in exploration of these key functions, participants are presented with a means for investigation of not only physical space, but of the interior space formed in contemplation of their role within it.

Yet listen well. Not to my words,
but to the tumult that rages in
your body when you listen to yourself.[132]

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[132] Bachelard, G. *The poetics of space.* p182.

8

Development and Application of Version 2

This chapter details the development and execution of a second version of the installation. Key issues discussed within it include: how problems faced during execution of version 1 (V1) were addressed, the development process of version 2 (V2), the documentation of the installation process and assessment of the revised system.

8.1 Revision of System

Following completion and assessment of the installation, a number of concerns were raised. While the system performed well from a technical standpoint it was concluded that certain aspects regarding its execution required further exploration. In particular, how the configuration of the system components affected the participants interaction with the installation, and the degree to which an enhanced sense of spatial awareness was formed. In an attempt to address these issues, a second version of the installation was developed.

The presence of the central frame structure clearly imposed significant restrictions upon the freedom of movement within the space. While it could be argued that its presence encouraged participants to move about the space with a certain degree of caution, it was my original intention that the interaction of participants with the space should be as seamless as possible. I wanted to encourage participants to engage with the space in the manner that they deemed most appropriate and that the system itself should impart its influence purely through the use of sound. Thus, the development of a revised system began with the removal of the central frame structure. Since no major problems were encountered with the technical aspects of the system, it was deemed that the electronic components of the system could be re-used and that emphasis would be placed upon the re-configuration of its various components.

In chapter 7 various sensor/speaker configurations were discussed. Of particular note, the proposal of a hanging frame system (Figure 65) was made. While the initial concerns I had regarding how such a frame could be mounted were still valid, it was this configuration that I was most keen to revisit. In order to overcome these problems, an investigation was undertaken of available locations in which such a configuration could be realised.

8.2 Selection of Location

Throughout the development of V1 it became clear that the location in which it was to be installed played a significant role in determining how the sensors and other equipment could be placed.

For the V2 system to function as intended, a number of considerations regarding the installation space were made. Firstly, the space was to be of a size, large enough to encourage freedom of movement, yet small enough to develop a sense of intimacy and enclosure. The ceiling would have the provision for the mounting of system components above the heads of participants and would allow access with a minimum of difficulty. The space would preferably have hard reflective surfaces to encourage reverberation, yet be located in an area with minimal background noise. With this criteria in mind, an investigation into suitable locations was begun.

A location was eventually found that provided the stated requirements. The space was a single room approximately 5m x 8m (Figure 98). The room possessed high ceilings (~7m) with a frame structure that supported light and power fixtures (Figure 99). The pre-existing frame provided a convenient means of housing the system components as the pattern of the cross beams allowed the sensors to be attached between the light fittings. The room also possessed concrete walls and floor that would be ideal for the encouragement of reverberation and the formation of resonance. The location was secured for a period of 7 days during which the installation was installed, exhibited and documented.

Figure 98: Overview of installation space as seen from rear corner.

Figure 99: Overview of the frame structure present within the ceiling space.

8.3 Configuration and installation

Configuration of the system components was primarily dictated by the location of the ceiling frame beams and light fittings. In order to maximise the area covered by the infra-red beams, the sensors required placement approximately 2.5m apart. The spacing between the beams and the location of the light fittings allowed the sensors to be placed in the configuration shown in Figure 100. In this fashion each beam covered an adjacent area effectively minimising the amount of space left unmonitored.

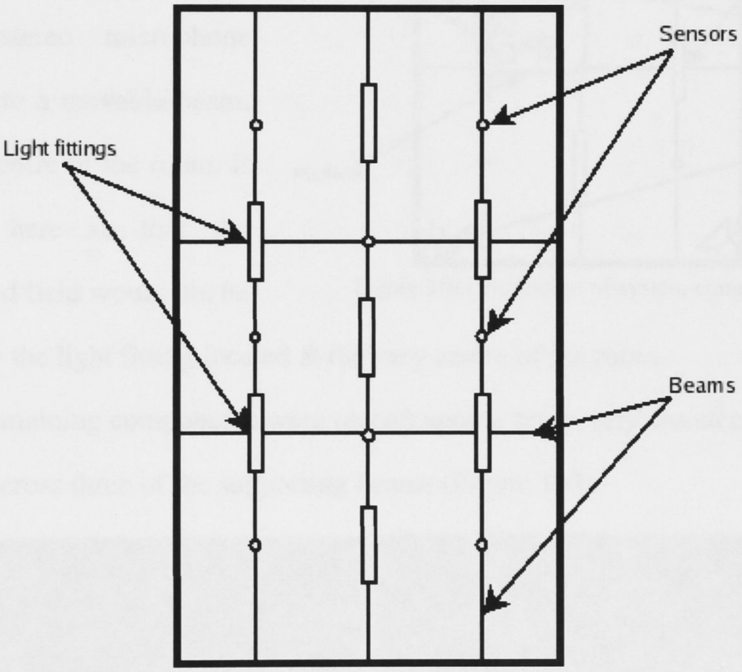

Figure 100: Sensor configuration utilising frame beams.

Securing the sensors and cabling to the supporting frame initially proved problematic. Since no destructive form of securing method could be employed (i.e. nails, screws, etc) an alternative method was required. After experimenting with cable ties and Velcro®, it was concluded that adhesive tape provided the best option for installation and did not pose any danger of damaging the frame when removed.

The frame beams also proved useful for the securing the cables for the sensors, speakers and power. Since the beams ran the length and width of the room, cables could be easily routed by following an appropriate beam in one direction and then continuing at right angles when an intersection was encountered.

The satellite speakers were placed at the four corners of the room upon ledges that conveniently ran the length of each wall as shown in Figure 102.

The stereo microphone was attached to a movable beam, close to the centre of the room. It was placed here so that the captured sound field would not be interrupted by the light fitting located at the very centre of the room.

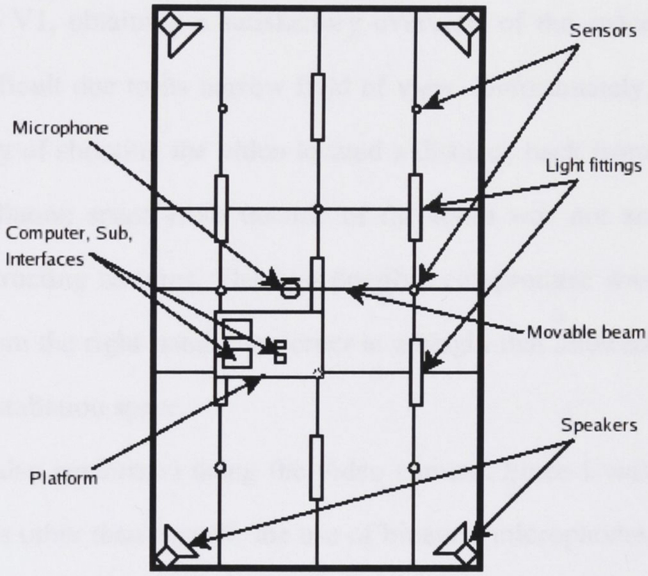

Figure 101: Placement of system components.

The remaining components were placed upon a temporary wooden platform that was secured across three of the supporting beams (Figure 103).

Figure 103: Ledges used for positioning of speakers.

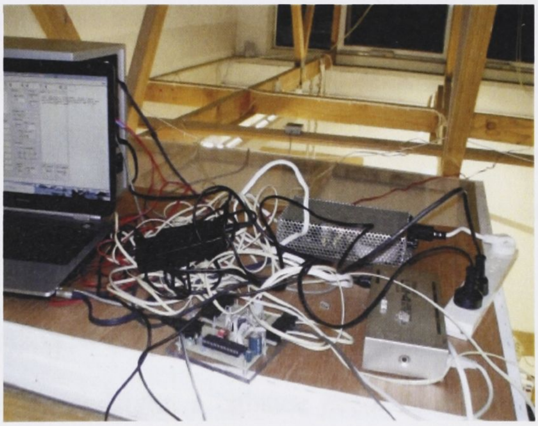

Figure 102: System components concealed upon platform.

8.4 Documentation

Documentation of the installation was undertaken in similar fashion to that performed for V1. Video, audio and still images were recorded using a Sony HDR-HC3 video camera. As was the case in V1, obtaining a satisfactory overview of the space using the video camera proved difficult due to its narrow field of view. Unfortunately, unlike V1 I did not have the luxury of shooting the video located a distance back from the installation. Filming the installation space from outside of the room was not an option due to the presence of obstructing features. The best possible compromise was obtained by recording the video from the right-hand rear corner at a height that allowed the greatest possible view of the installation space.

The capture of audio was also performed using the video camera. Since I was recording the interaction of viewers other than myself, the use of binaural microphones and a portable recorder could not be employed. It was my intention that viewers of the work were to interact with the space unencumbered. Also, I did not want to influence the behaviour of participants by reinforcing the knowledge that their actions were to be recorded.

During the installation period, the work was made available for public viewing. Although people were intrigued, it became apparent that the majority of people only spent a very short time within the space. Without direct instruction, most entered, walked around briefly and then continued on their way. People also felt reluctant to interact with the work when others were present.

The conditions under which the installation is required to be installed does not lend itself to a high throughput of viewers. In hindsight, a balance needs to obtained between having a location that is quiet for lengthy periods and yet still affords easy access.

In order to obtain a large pool of information regarding viewer experiences, the work perhaps requires installation within the same space for a considerable length of time. The session discussed here remained in-situ for a period of 7 days. It is my opinion that for viewers to form an understanding of the works subtleties, repeated viewings over an extended period are required. I would suggest a minimum period of 1 month to be a good starting point.

Much deliberation was had, regarding how I was to record the experiences of the viewers. I had considered asking viewers of the work to answer prepared questionnaires. On the advice of my supervisor and other members of staff and the CNMA, I was discouraged from doing so, due to the ethical considerations that must be undertaken when participant responses are recorded. I was advised that getting clearance from the university ethics council would be very difficult unless I was in direct consultation with trained psychologists that could prepare suitable questionnaires for me. Although this was disappointing, I was confident in obtaining useful information from participants through casual conversation. The participants shown in the attached DVD had given their express permission to be filmed and proved more than willing to share their opinions of the work with me.

Arrangements were made for the two independent viewers of the work to be present during the documentation process. The individuals selected, had very little exposure to the work prior to the session other than a brief period of interaction during the days spent installing the work. Although I had spoken to both of them about the work, no set instructions were given as to how they should interact with it. Beyond informing them that the work utilised a number of sensors to track their position and that the work made use of the sounds they produced while within the space no further instruction was given.

8.4 Application and Observations

Once the system had been physically installed and a plan formulated for the documentation process, a brief period of testing was performed before it was made available for viewing.

It is at this point that I suggest the reader view the contents of the DVD labelled *...in the white wake of my steps. V2.* The disc is comprised of three video recordings depicting the actions of myself and two participants.

8.4.1 Testing the System

The first track, titled *Testing the System,* shows myself interacting with the installation for a brief period to ensure that the system was functioning as intended. During the test I perform a number of simple actions that allow me assess the current state of the system. The footage begins with myself entering the room (Figure 104).

Figure 104: Test subject enters.

Note that the system is already in a resonating state. This was indicative of normal function. The system was encouraged into this state by the sound of the door closing as I exited the room prior to the beginning of the recording. This state was developed further as I re-entered the space. I begin the testing process by walking through the path of several sensors in succession (Figure 105). This serves to reset the extent of audio recirculation, resulting in the cancellation of the majority of sound heard.

Figure 105: Test subject changes position.

As I pause, it can be noted that the recirculating delays lengthen once again. I then proceed to encourage the production of new tones by introducing sound into the space by clapping my hands and stamping my feet (Figure 106).

Figure 106: Test subject introduces sound by clapping.

The sounds produced can be heard recirculating, but without further encouragement via additional sound input and movement the sound levels begin to decay. With the system left in a neutral state, I exit the room (Figure 107).

Figure 107: Test subject exits.

Adjustments were made to amplification levels before the first participant (Subject A), was instructed to enter the room.

8.4.2 Subject A

Upon entering, Subject A produces sounds for the system by opening and closing the door (Figure 108).

Figure 108: Subject A enters, producing sound in doing so.

Subject A then proceeds to move beneath one of the sensors, scuffing her feet in the process (Figure 109). This serves to produce a dominant tone associated with the coinciding movement and sound input.

Figure 109: Subject A evolves first sounds.

The subject begins to explore the space, changing position several times (Figure 110).

Figure 110: Subject A changes position within the space.

Upon changing position the sounds begin to decay resulting from a lack of sound input produced by the subject movements (Figure 111).

Figure 111: Subject A cancels sounds by moving quietly.

Noticing that she is not producing enough sound to promote the evolution of sounds the subject proceeds to cough, causing the system to react (Figure 112).

Figure 112: Subject A coughs provoking a reaction from the system.

The subject continues to change position, then exits the room (Figure 113)

Figure 113: Subject A exits the room.

8.4.3 Subject B

Figure 114: a) Subject B enters. b) Listens attentively and considers next move.

Subject B enters the room cautiously, after appearing to expect an immediate response from the sound produced by the closing the door. The subject moves quietly at first but then continues less cautiously as sounds begin to appear (Figure 114 a) and b)).

Figure 115: Subject B performing exaggerated movements.

In an attempt to provoke a response from the system, the subject commences highly exaggerated movements including a handstand, the stamping of feet and small consecutive leaps (Figure 115).

Figure 116: a) Subject observes rhythmic patterns in the sound. b) Begins finger snapping.

Following his previous movements, Subject B begins to notice rhythmic patterns within the sound (Figure 116 a)). In an effort to produce different sounds the subject begins snapping his fingers, clapping and stamping (Figure 116 b) and Figure 117).

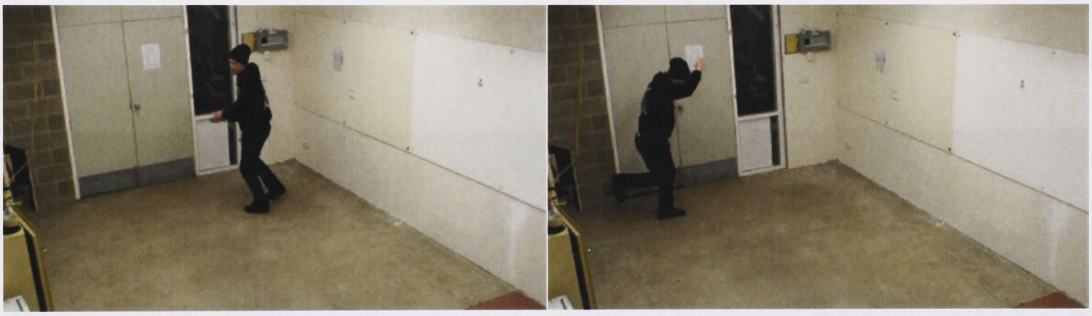

Figure 117: Subject B producing sound through clapping and stamping.

Figure 118: Subject B investigating the effects of repetitive movements by moving back and forth.

After observing the effects of exaggerated movement and sounds, the subject begins to investigate the effect of repetitive movements (Figure 118).

Figure 119: Subject experiments with a) localised but vigorous movement. b) slow expansive movements.

The subject also experimented with movements that were vigorous but confined to a small area, by jumping up and down on the spot (Figure 119 a)). Conversely, the subject also tried stretching out to cover more space (Figure 119 b)).

Figure 120: Subject performs a) Large jump b) use of shelf to create sound.

The subjects final investigations before exiting the room included: a large jump from one side of the room to the other, and the effect produced by banging loudly on a nearby shelf (Figure 120 a) and b) respectively).

Figure 121: Subject B exits the room.

8.5 Discussion

As shown in the video recordings, the two participants exhibited very different modes of interaction with the system and the space itself.

Subject A primarily made use of reserved, understated movements and subtle incidental noises. The paths taken, being those of a casual stroll rather than a focussed journey. The subject appeared content to wander slowly about the room and listen for possible changes in the sound.

In direct contrast, Subject B chose to impose exaggerated movements and sounds in an effort to provoke an immediate and clearly recognisable change in the state of the system.

When questioned on what drove them to move in particular patterns or create certain sounds, both participants replied that their interaction was based upon what they wanted to hear. Subject A expressed that she was more interested in hearing the interplay between the different frequencies present in the sound. She chose not to enforce her own ideas of what sounds she would like to hear but instead let the room dictate what frequencies would dominate. Subject B on the other hand was more intent on forming an association between an action and the types of sounds that could be produced by it. Hence the type of interaction displayed by the viewer is a product of their personality. The viewer chooses to interact with the system in a way that comes naturally to them.

A distinct pattern of behaviour was displayed by both subjects, where the viewer would gravitate towards the area directly below the sensors. This was not completely unexpected since these areas appeared to have the greatest influence upon the development of sounds within the space. The pattern took the form of a period of action, followed by inaction coupled with observation, followed by another period of action. It was not immediately discernible whether this was due to the fact that the locations of each sensor could be ascertained if the viewer looked upwards, or if it coincided with points that were deemed a comfortable length of travel. Certainly in the case of Subject B, who is seen to look upwards on numerous occasions, it was clear that the sensors could be easily observed. However, although Subject A displayed similar behaviour, the effect was less obvious due to the more relaxed, wandering approach. This raised questions as to how the size and shape of the room may have influenced viewer behaviour. Since the room was relatively small, the distance from one side of the room to the other was traversed in the space of several steps. Thus over the course of a perceived journey, undertaken so as to cause a change in the state of the system, the viewer would naturally gravitate to the edges of the room. Once the viewer had remained still, spent some time considering their next move and then recommenced their journey, it was only a matter of taking several steps before the viewer was once again situated at the edge of the room. This behaviour provided an intriguing insight into the thought processes formed in the exploration of our environment. A note was made to explore this activity again at a later date following further research into the cognitive processes responsible for the formation of behavioural patterns.

In terms of the effect that sound played in the development of a sense of spatial awareness, it was unclear whether the viewer understood how much influence the qualities of the room imposed upon the types of sound present with it. Both subjects seemed content to assume that the frequencies that were exhibited were associated with the type of input rather than the nature of the resonant qualities of the space. That is not to say that the types of sounds produced by viewers did not influence how the work evolved, but that it is chiefly the physical structure of the room that dictates which frequencies will remain in circulation. This can be observed when both subjects purposefully introduce sounds into the space by coughing, clapping or scuffing their feet. The sound can be heard repeated briefly in its original form but quickly decays into the dominant tones present within it. Upon questioning, both subjects were unsure whether they regarded the use of resonant tones as the primary method for sound generation and development of spatial markers as an effective means for doing so. Both subjects also stated that at some points it was, difficult to discern whether their movements were playing a role in determining the sound or not. They also expressed difficulties in remembering what types of movements produced a particular response due to the similarity exhibited between sounds. When provided with insight into the underlying concepts that were examined during the course of the research period, my choice to use resonant tones became clearer. However, I had hoped that the link between the formation of resonance and the bodily presence of the viewer within the space was clear and unmistakable. Perhaps, through multiple viewings of the work, the relationship would have become more obvious.

From a technical standpoint, the system operated in the way that it was intended. What can be regarded as a hindrance to some viewers may be considered beneficial to others. Thus, from this viewing of the work it is unclear whether the concepts discussed in this thesis were conveyed effectively. That said, it is the nature of art that what one individual deems as an appropriate method for the conveyance of an idea is regarded as incorrect to another. Thus, the success of the installation can only truly be determined on an individual basis.

9

Conclusions

This chapter provides some concluding remarks and presents suggestions for further development and alternative uses of the system employed within the installation.

Throughout this thesis I have discussed the processes of inspiration, development and application involved in the creation of a semi-interactive installation. *...in the white wake of my steps* was initiated as a means for drawing focus upon the interaction between sound, body and space. To achieve this I chose to implement the use of an infra-red sensor tracking system. Information drawn from the movement of participants, in conjunction with the recirculation of sounds produced in those movements, served as an interface between the viewer and the space.

Ultimately, the success of the installation is left to the individual. The effect of our physical presence, the influence of sensory data and the action of our memories, all combine to form, what is I hope, a unique experience of space and the ideas we form regarding our place within it.

The experience gained through interaction with the installation poses questions regarding the nature of our engagement with our spatial environment. The use of an interface places emphasis upon the role of the viewer, whose presence plays an integral part of the work and of space itself. The removal of traditional, tangible elements from the interface encourages participants to explore the work in an intuitive manner through natural, everyday movements. In doing so, our physical and cognitive processes become rendered in musical sequence and structure, assisting in orientation of the listener within the environment. Exploration of the space thus exposes the various decision-making processes involved with investigation of the unknown and the reaffirmation of memories.

The knowledge gained throughout the research period and its application in a creative context constitutes a solid foundation for the development of future works. The construction of an interface that encourages exploration and experimentation, positions the work well for application within a performance environment, or to an investigation of sociological and behavioural systems. Essentially, it may be applied in any situation where emphasis is to be placed upon the production of spatial information without the need for development of skills specific to control of the interface. In the case of a dance or theatre environment, participants would require little training and could therefore rely more upon the skills associated with movement and gesture. In the analysis of social behaviour or spatial utilisation, participants need not know that any conscious interaction is required on their behalf. By simply tracking the natural movements of participants, a detailed map could be constructed showing how we utilise our surroundings as individuals, and when interacting with other people.

As discussed throughout this volume, many factors influence the effectiveness of any particular object tracking method. Therefore, the application of technology used in this project is ultimately left to the discretion of the user. Efforts have begun on refinement of both technical and aesthetic aspects of the system. Although too great to discuss at length here, new sensor designs currently in development have shown improvements in both range and stability. Consequently, this allows the implementation of sensor configurations previously unsuitable due to cross-talk effects.

Although the system remains in its infancy, it is my hope that through experience of the installation and exploration of the concepts discussed in its development, the reader will have gained sufficient understanding to apply similar methods in the creation of new works.

Although only briefly touched upon in this work, the idea of using sound as a tool for the examination of social interaction and spatial utilisation has inspired the creation of a new, government funded installation titled, *Remnant Encounters.* Currently scheduled for exhibition in October 2007, the work utilises aspects of the work discussed in this thesis, in addition to new techniques such as video tracking and projected images to examine how we use space in social environments.

In defining the various spaces that constitute our modern world picture, I chose to explore the use of technology as a means for the articulation of abstract concepts; some expressible in the quantifiable terms of mathematics and physics, and others that retain an ethereal and seemingly immeasurable quality. As technology continues to mature and tools become increasingly accessible, the role of the artist as a medium for the expression of abstract notions will also continue to evolve. It is my hope that the concepts discussed in this volume will not exhaust themselves purely in demonstration but serve as a platform for the continuing development of abstract spatial notions.

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Appendices

Appendix A

Generation of MIDI byte pack

E.g. A sensor voltage of 3.5V is translated into a MIDI pack by the following process:

1. With voltage references of 0V and 5V, 3.5V represents a ratio of 3.5V / 5V = 0.7
2. As a 10-bit sample this ratio becomes, 0.7 x 1023 = 716.1 = 716 rounded to whole number.
3. Expressed as binary 716 = 1011001100
4. Dropping the last two digits to form an 8-bit number, 1011001100 = 10110011
5. Divide by 2 by rotating each digit once to the right, 10110011 >> 01011001
6. Expressed as decimal, 01011001 = 89. This becomes the MIDI cc value.
7. A Status byte is formed from a Status bit '1', control change event '011' and MIDI channel, in this case channel 1, '0001'. Status byte = '10110001'.
8. The first Data byte is formed from a Data bit '0', and control change number, in this case sensor 4 is assigned controller 4, '0000100'. Data1 byte = '00000100'.
9. The second Data byte is formed from a Data bit '0', and control change value, in this case 89, '1011001'. Data2 byte = '01011001'.
10. Bit order is reversed in each byte, '10001101', '0010000' and '1001101'.
11. Start and stop bits are inserted, '0100011011', '000100001' and '010011011'.
12. Completed MIDI byte pack = '0100011011000100001010011011'

Appendix B

Midisensor controller PICMicro® assembly code

```
;Midisensor controller
#define _version 1.11"
; Date: 9/6/04
; Update History:
;       14/6/04  - changed demomidi.asm from sending (midi note 60, velocity 64), to (cc 1, 0, 64, 127
;                  and repeat)
;       15/6/04  - changed midi channel selector DIL switch to port D to free up analogue ins
;                - changed port A to analogue in and turned on ADC
;       16/6/04  - connected pot to AO, takes voltage and converts to midi controller value, loops
;       24/6/04  - extensive rewrite
;                - sequential reading of ADC channels
;                - changed code to make it easier to scale number of inputs
;                - general housecleaning, made things a bit neater, more modular
;                - now uses indirect addressing to save ADC results into a table
;       30/6/04  - combined ADC input and digital input sections
;       2/7/04   - added cc and note transpose function using PORTC
;                - changed PORTD to outputs
;                - midi channel is now checked on every loop rather than just at startup
; Author: David Hirschausen
; elements of this program are based on demomidi.asm by Derek Johnson
; http://homepage.ntlworld.com/derek.johnson
;****************************************************************
;* This program converts analogue sensor voltages (0-5V) and digital       *
;* inputs to midi controller and note on/off messages                      *
;****************************************************************

; Hardware Notes: clock speed 4 or 20 MHz


        __CONFIG _CP_OFF & _WDT_OFF & _BODEN_OFF & _PWRTE_OFF & _XT_OSC &
_WRT_ENABLE_ON & _LVP_OFF & _DEBUG_OFF & _CPD_OFF

        list p=16F877,r=dec                     ; list directive to define processor
        #include <p16f877.inc>                  ; processor specific variable definitions
                                                ; r=dec tells the compiler to treat all


#DEFINE PAGE0       bcf     STATUS, RP0         ;these two lines make it easier to
#DEFINE PAGE1       bsf     STATUS, RP0         ;change between pages

                                                ;first availiable user memory location (PIC 16F877)
```

```
TEMP                    equu                    33          ;temporary storage file
CHAN                    equ                     34          ;file used to store MIDI channel
TRANSPOSE       equ             35
NOTEFLAG        equ             36                          ;stores note on flags
NOTEINCR        equ             37                          ;increments the note output on each sequential read
                                                            ;of PORT B
TESTFLAG        equ             38                          ;sets the PORT B input channel to be read
FLAGTOSET       equ             39                          ;designates the flag that needs to be set depending
                                                            ;on which channel is being read
FLAGTOCLR       equ             40                          ;designates the flag that needs to be cleared
                                                            ;depending on which channel is being read
ADCRESULT       equ             41                          ;file used to store ADC input
CCNUM           equ             42                          ;continuous controller number incremented on each
                                                            ;loop
FSRINCR         equ             43                          ;used to increment pointer to PREVADCRESTAB
                                                            ;on each loop
PREVADCRESTAB           equ                                 ;table used to store previous ADC results
;LASTFILE       equ             127                         ;not used, this is the last memory location on page 0

;****************************************************************************************
;****************************************************************************************

                org             0                           ;START OF PROGRAM MEMORY
                goto    START

                org             4                           ;INTERRUPT VECTOR ADDRESS
                                                            ;interrupts are not used in this program

;****************************************************************************************
;****************************************************************************************
;Initialize PIC
START

        PAGE1
        movlw   B'00000000'                 ;Left justify, 8 analog channels (A0-7)
        movwf   ADCON1 ^ 0x080              ;sets port A as ADC inputs
        movlw   B'00111111'
        movwf   TRISA ^ 0x080               ;all pins inputs
        movlw   B'11111111'
        movwf   TRISB ^ 0x080               ;all pins inputs, switches
        movlw   B'00111111'
        movwf   TRISC ^ 0x080               ;RC0-5 inputs, RC6-7 outputs
        clrf    TRISD ^ 0x080               ;all pins outputs
        movlw   B'00000111'
        movwf   TRISE ^ 0x080               ;all pins inputs
```

```
        movlw   B'10000111'             ;TMR0 prescaler, 1:256
        movwf   OPTION_REG ^ 0x080
        PAGE0
        clrf    PORTD                   ;all pins zero
        clrf    CHAN                    ;all bits zero
        clrf    TRANSPOSE               ;all bits zero
        clrf    NOTEFLAG                ;all bits zero
        clrf    ADCRESULT               ;all bits zero
        clrf    PREVADCRESTAB           ;clear table
        clrf    PREVADCRESTAB + 1
        clrf    PREVADCRESTAB + 2
        clrf    PREVADCRESTAB + 3
        clrf    PREVADCRESTAB + 4
        clrf    PREVADCRESTAB + 5
        clrf    PREVADCRESTAB + 6
        clrf    PREVADCRESTAB + 7
        call    SETBAUD                 ;set up usart

;*************************************************************************

MAIN    btfss   INTCON,T0IF             ;Wait for Timer0 to timeout
        goto    MAIN                    ;
        bcf     INTCON,T0IF             ;clear flag


;set midi channel
        movfw   PORTC                   ;read the bit pattern on port c
        movwf   CHAN                    ;save it in CHAN
        movlw   B'00001111'
        andwf   CHAN, f                 ;make sure bits 4-7 are clear and save in file labelled CHAN

;set cc and note value range
        movfw   PORTC
        movwf   TRANSPOSE
        movlw   B'00110000'
        andwf   TRANSPOSE, f            ;make sure bits RC0, 1, 2, 3, 6 and 7 are clear and save RC4
                                        ;and 5 to TRANSPOSE
        swapf   TRANSPOSE, f
        bcf     STATUS, C
        rlf     TRANSPOSE, f
        rlf     TRANSPOSE, f
        rlf     TRANSPOSE, f

;*************************************************************************
;PORTB digital input section
```

```
BEGIN
          movlw   1
          movwf   TESTFLAG              ;load TESTFLAG with 1 for RB0
          movlw   1
          movwf   FLAGTOSET             ;load FLAGTOSET with 1 for RB0
          movlw   254
          movwf   FLAGTOCLR             ;load FLAGTOCLR with 254 (11111110), AND with
                                        ;NOTEFLAG to clear bit
          movlw   60
          movwf   NOTEINCR              ;load NOTEINCR with first note 60 (C4)
          movfw   TRANSPOSE
          addwf   NOTEINCR, f

TEST
          movfw   TESTFLAG
          andwf   PORTB, w              ;is switch on?, i.e. pulled low, if switch is off zero flag is set
          btfsc   STATUS, Z             ;yes then process, otherwise skip
          goto    NOTEON

          movfw   TESTFLAG
          andwf   NOTEFLAG, w           ;is flag set?
          btfss   STATUS, Z             ;yes then process, otherwise skip
          goto    NOTEOFF

NEXTNOTE
          btfsc   TESTFLAG, 7           ;was RB7 the last bit tested?, yes then goto ADC section
          goto    ADCSTART
          bcf     STATUS, C
          rlf     TESTFLAG, f           ;rotate set bit in TESTFLAG for next PORT B channel
          bcf     STATUS, C
          rlf     FLAGTOSET, f          ;rotate set bit in FLAGTOSET for next PORT B channel
          bsf     STATUS, C
          rlf     FLAGTOCLR, f          ;rotate clear bit in FLAGTOCLR for next PORT B channel
          incf    NOTEINCR, f           ;increment midi output note for next PORT B channel
          goto    TEST

NOTEON
          movfw   TESTFLAG
          andwf   NOTEFLAG, w           ;test flag, has note on been sent?
          btfss   STATUS, Z             ;yes then do nothing
          goto    NEXTNOTE

          movfw   FLAGTOSET
          iorwf   NOTEFLAG, f           ;set flag

          movlw   B'10010000'           ;note on event
```

```
        addwf   CHAN, w                 ;add channel number
        call    OUTBYTE                 ;send first byte
        movfw   NOTEINCR                ;midi note number
        call    OUTBYTE                 ;send second byte
        movlw   64                      ;velocity value
        call    OUTBYTE                 ;send third byte
        goto    NEXTNOTE                ;test next switch

NOTEOFF
        movfw   FLAGTOCLR
        andwf   NOTEFLAG, f             ;clear flag
        movlw   B'10000000'             ;note off event
        addwf   CHAN, w                 ;add channel number
        call    OUTBYTE                 ;send first byte
        movfw   NOTEINCR                ;midi note number
        call    OUTBYTE                 ;send second byte
        movlw   64                      ;velocity value
        call    OUTBYTE                 ;send third byte
        goto    NEXTNOTE                ;test next switch

;*****************************************************************************
;ADC input section
ADCSTART
        bcf     PIR1, ADIF
        movlw   B'01000001'             ;Fosc/8, ADC enabled (A0)
        movwf   ADCON0                  ;turn on ADC
        clrf    TEMP
        clrf    FSRINCR
        movlw   1
        movwf   CCNUM                   ;load CCNUM so A0 = controller number 1
        movfw   TRANSPOSE
        addwf   CCNUM, f
        movlw   PREVADCRESTAB           ;point fsr to top of table
        movwf   FSR                     ;

NEW_AD
        call    SAMPLE_DEL              ;provide necessary sampling time
        bsf     ADCON0,GO               ;Start A/D conversion

WAIT
        btfss   PIR1,ADIF               ;Wait for conversion to complete
        goto    WAIT                    ;no? then loop
        movf    ADRESH, W               ;Write A/D result to TEMP
        movwf   TEMP                    ;
        bcf     STATUS, C               ;clear carry flag to ensure MSB=zero
        rrf     TEMP, W                 ;divide by 2 put result in w
```

```
        movwf  ADCRESULT              ;save in ADCRESULT file
        movf   INDF, w        ;get value pointed to by current FSR in PREVADCRESTAB
        subwf  ADCRESULT, w           ;checks to see if input has changed
        btfsc  STATUS, Z              ;if value is the same get next channel value
        goto   TESTCHAN
        movfw  ADCRESULT              ;recall ADC result
        movwf  0                      ;save in table

MIDIOUT
        movfw  CHAN                   ;move the value stored in CHAN into the working register
        addlw  B'10110000'            ;status bit 1, control value event 011, channel number 0000
        call   OUTBYTE                ;output the first MIDI byte
        movfw  CCNUM                  ;controller number 1
        call   OUTBYTE                ;output the byte
        movfw  ADCRESULT              ;control value (0 - 127 range)
        call   OUTBYTE                ;output the byte

TESTCHAN
        btfss  ADCON0, 3              ;tests channel select bits
        goto   NEXTCHAN
        btfss  ADCON0, 4
        goto   NEXTCHAN
        btfss  ADCON0, 5              ;if 111 then go back to 000
        goto   NEXTCHAN               ;otherwise select next channel
        goto   MAIN

NEXTCHAN
        movlw  8                      ;select next ADC channel
        addwf  ADCON0, F              ;
        bcf    PIR1, ADIF             ;reset interrupt flag bit
        incf   CCNUM, f
        incf   FSRINCR, f             ;later added to table address to increment pointer to next


ADC                                   ;result register
        movlw  PREVADCRESTAB          ;get table address
        addwf  FSRINCR, w             ;add to FSRINCR
        movwf  FSR                    ;move into indirect
        goto   NEW_AD

;*********************************************************************************
;Set midi baud rate (31.250 KHz)

SETBAUD
        PAGE1
        movlw  1                      ;1 = spbrg value for 31.250 KHz @ 4 MHz clock
        movwf  SPBRG ^ 0x080          ;change to 9 if using 20MHz clock
```

```
        movlw   160                     ;val for txsta 10100000 brgh = 0 (64 bit division)
        movwf   TXSTA ^ 0x080           ;asynchronous mode transmit enabled brgh low speed

        bcf     TRISC ^ 0x080, 6        ;set RC6 as output
        bcf     PIE1 ^ 0x080, 4         ;clear transmit interrupt

        PAGE0

        movlw   B'10010000'             ;val for rcsta 10010000 serial port enabled
        movwf   RCSTA

        return

;*******************************************************************************
;Software delay of 10uS for the a/d setup.
;At 4Mhz clock, the loop takes 3uS, so initialize TEMP with a value of 3 to give 9uS, plus the move etc
;should result in ;a total time of > 10uS.

SAMPLE_DEL
        movlw   3
        movwf   TEMP

SD
        decfsz  TEMP, F
        goto    SD
        return
;*******************************************************************************
;Output midi byte from usart

OUTBYTE
        btfss   PIR1,4                          ;wait for bit 4 = 1 showing txreg empty
        goto    OUTBYTE
        movwf   TXREG                           ;put byte into txreg for auto transmission
        return
;*******************************************************************************
        end
```