Acoustic Phenomena as a Model for Music Composition

Representing acoustic phenomena in musical composition through the extrapolative and interpolative modelling of the phenomena’s characteristics and behaviour

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STATEMENT OF ORIGINALITY

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Signed_________________________

David Tracy

July 2013
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1.0 Introduction

1.1 Abstract

The purpose of this research is to explore the relationship between acoustic phenomena and their incorporation into musical compositions. It seeks to identify composers that have represented acoustic phenomena in musical compositions through the examination of their work and compositional practices. This research attempts to explore new and existing methods of modelling acoustic phenomena for music through the creation of new musical works.

Referencing acoustic phenomena is evident in the works of recent music composers such as Steve Reich, Alvin Lucier, Karlheinz Stockhausen, Gerard Grisey and La Monte Young. These composers either utilise the acoustic phenomena as a musical focal point or represent them in their compositional approach; these two approaches are achieved through the use of instrumental techniques, electronic mediums or by presenting phenomena as a natural acoustic occurrence.

Compositional practices involve the modelling of the behaviour and characteristics of acoustic phenomena and the types of modelling approaches observed will be categorised as either extrapolative or interpolative.

In addition to the observation of techniques utilised by these composers, this research addresses the potential use of acoustic phenomena such as phase shifting, resonance and delay as models for musical works. Models for particular acoustic phenomena are created in order to demonstrate their use in both interpolative and extrapolative ways and the resultant musical works will be used as case studies in order to analyse the effectiveness of the phenomena as models.
1.2 Outline of Research Area

This research paper is a focused effort in illuminating the practice of representing acoustic phenomena in musical composition through modelling. Through the examination of significant composers (past and present) in this field, the research identifies criteria needed for the plausible modelling of acoustic phenomena. These criteria are then applied to the author’s own compositions.

1.2.1 Acoustic Phenomena

An acoustic phenomenon may be defined as a physical phenomenon associated with the production or transmission of sound (Acoustic-Phenomenon, n.d.). A phenomenon, of any kind, is defined as an “observable occurrence” (Phenomenon, n.d.). Peter Blamey, in his writing *Sine Waves and Simple Acoustic Phenomena in Experimental Music - with Special Reference to the Work of La Monte Young and Alvin Lucier*, describes the presence and behaviour of acoustic phenomena in our every day life:

> Acoustic phenomena surround us. They are the natural physical actions and interactions of every sound event that occurs in the air or is transmitted through other solid, liquid or gaseous media. Therefore, the sound of any and all music is, of course, on at least a scientific level, comprised of a complex of different acoustic phenomena. (Blamey, 2008, p. 2)

When considering the characteristics of any “observable occurrence”, both the qualitative and quantitative values are the components that are under observation. Therefore, in this research paper an acoustic phenomenon is defined as: *An observable occurrence of the quantitative and qualitative values of sound as affected by its production or transmission.*

Examples of acoustic phenomena are sine waves, resonance, delay/echo, reverb, ring modulation, acoustic beating, phase shifting, synthesis, combination tones, transients and harmonics. These phenomena are observable in a variety of contexts, from everyday situations to their exploration in anechoic chambers and sound studios. Upon increased observation and understanding of these acoustic phenomena, composers of music began representing them in their musical works.
1.2.2 Representation through Modelling

Representation in music is a broad area of focus, as it spans almost the entire breadth of musical history. The commonly shared definition of *represent* is to “act as an embodiment of, to symbolize, to call up in the mind by description or portrayal or imagination, or to place a likeness of before the mind or senses.” In connection, a representation is an “image, likeness or reproduction of a thing” (Representation, n.d.).

The general use of representation in music is not under scrutiny in this research, rather a particular facet of representation; modelling. Unlike metaphor or other rhetorical devices, a model in science is a more rigorous type of representation that undergoes certain conditions to be plausible. This idea of modelling acoustic phenomena is not frequently found in musical texts but can be seen in the writings of musical commentators and composers such as Joshua Fineberg (2000a, 2000b, 2000c), François Rose (1996) and Julian Anderson (2000), for example. These writers comment on the use of a sound spectrum as a model in composition. Fineberg states that basing a composition on the relationship of frequencies evident in a sound demonstrates the way in which composers can use the “structure and perception of natural (environmental) and instrumental sounds [to provide] models for the way in which various frequencies are created and interact to form our auditory impressions” (Fineberg, 2000b, p. 82). This use of acoustic phenomena is also referenced by Peter Blamey, which is discussed in section 1.3.1.

The opinions of what a model is and the depth of understanding that can be undertaken with this topic are quite vast. This research does not seek to present an in-depth discussion of the concept of modelling, as it specifically deals with the study of models of acoustic phenomena and how these two components can be used to form a musical representation. For the purposes of this research, two significant seminal texts are used as references about modelling: *Models and Analogies in Science* by Mary Hesse (1963) and *Models and Metaphors* by Max Black (1962).

Max Black states that a model is a “symbolic representation of some real or imaginary original, subject to rules of interpretation for making accurate inferences from the relevant features of the model” (Black, 1962, p. 222). Any form of representation can be said to be either literal or metaphorical, or at least having a tendency towards one or the other. According to Black, when creating a model of a particular phenomenon these two methods are said to be either a *scale model* or an *analogue model* (respectively).
A scale model is the term used to describe a likeness of an object, system or process that has preserved the relative proportions of the original. Its purpose is to “reproduce, in a relatively manipulable or accessible embodiment, selected features of the original” (Black, 1962, p. 221).

When a model occurs in a different medium, it is referred to as an analogue model. This type of model is designed to reproduce the *structure* or series of *relationships* of the original in a new medium. Black states that the difference between these two varieties of models lies in the interpretation of them; while scale models imitate the original, analogue models are “guided by the more abstract aim of reproducing the *structure* of the original” (Black, 1962, p. 222).

The reason for creating models of any set of phenomena is to facilitate the application of observable characteristics or acquired knowledge into new arenas. Mary Hesse justifies the use of models in the furtherance of theories, stating that “without a model, it will be impossible to use a theory for one of the essential purposes we demand of it, namely to make predictions in new domains of phenomena” (Hesse, 1963, pp. 4-5). In this research, it is demonstrated that the creation of models takes equal precedence as it allows the extension of observable characteristics from one particular context into another.

Consider, for example, the analogy drawn between the properties of water and the properties of sound. These two areas are often paralleled in order to gain further understanding of one from the properties observed within the other. When a formula is created that explains the propagation of waves in water, this formula can then be transferred to the discussion of sound; the creation of this model, by shared analogy, allows more insight into the secondary field of knowledge. Hesse discusses this particular example:

> Now consider what happens when we make use of the known theory of water waves, and the analogies between them and sound, in order to construct a theory of sound... we construct one-to-one correspondences between the observable properties of sound... and those of water waves. (Hesse, 1963)

The juxtaposition of metaphorical and literal interpretations, as in the use of models, can be seen occurring in a wide variety of contexts. An example could be seen in the field of fine art: upon observing an external stimulus such as a flower vase, an artist could paint what they see, or express the connotations evoked from the external stimulus. From the one ‘muse’ (the flower vase) the results could be a “scale model” – a literal representation – in the form of a beautiful painting of a flower vase, or something more obscure – an “analogue model”, a
metaphorical representation in the form of shapes and colours that seek to represent the artist’s feelings or thoughts that are elicited by the flower vase.

This same juxtaposition is also observable in music. In his text *Classical Music, Why Bother* (2006) composer and author Joshua Fineberg (1969- ) uses a similar comparison – the metaphorical model observable in fine art – in the context of composing music that represents sound phenomena. Fineberg compares Gérard Grisey’s focus on internal sound as a model for music to the work of visual artists such as Chuck Close.

Chuck Close has made a career of painting realistic representative paintings where the picture is pixelated into individually visible shapes and colours. Each spot in the picture has a definite size and shape while still giving its overall colour and shading characteristics to the larger image, which becomes clear once you step back from the canvas to view it. (Fineberg, 2006, p. 115)

Fineberg is essentially describing a type of modelling that can be observed in fine art as well as music. For example, a composer could hear the calling of a bird, and use the inherent melody and pitch intervals that are apparent to create a melody that very much replicates the sound of the bird – or perhaps instead, the composer could be inspired by the beauty of the birdcall and seek to write a melody that captures the essence of it. Or yet again, a composer could even use the intervals of the birdcall to dictate the intervals of the rhythm of the melody.

In the first instance discussed above, in which the composer replicates the sound of the bird, we would say that the representation is a literal one. In the latter two instances, where the composer seeks to capture the essence of the birdcall or uses the pitch intervals evident to dictate rhythmic values we could say it is more of a metaphorical translation. Such is also the case with representing acoustic phenomena in musical composition – in can be achieved through either of two ways:

1. **Literal representation** – using the phenomenon’s parameters from its sound production context in a musical context (e.g. using a rhythmic device evident in a sound phenomenon to create a rhythmic device in a composition)

2. **Metaphorical representation** – using the phenomenon as a model to apply its parameters to a new context in music (e.g. using the interaction of a combination tone as a model to create a rhythmic device)
As with any type of model, these two distinctions can be referred to as *scale models* (1) and *analogue models* (2). In this research, these definitions are also further associated with terms that have been adapted from science and mathematics: *interpolation & extrapolation*.

**Interpolation**

The definition of interpolate is to “estimate (intermediate values) from surrounding known values” (Interpolate, n.d.). The term *interpolation* is rarely found in the context of music; however, it seems to be the most fitting term in describing the scale model type of musical representation. It could perhaps be said that this representational method is a type of *imitation*, as Black states that scale models “rely markedly upon identity: their aim is to imitate the original, except where the need for manipulability enforces a departure from sheer reproduction” (Black, 1962, p. 222).

Interpolating from an acoustic phenomenon’s characteristics is a way of representing the embodiment of the phenomenon in music, as literally as possible. To do this, an understanding of how the phenomenon operates is essential, which means identifying its key behavioural characteristics.

Interpolating means to use a known value (from data gathered) in a new context; in terms of representing acoustic phenomenon in musical composition, this means using a known behaviour or characteristic of the phenomenon in the context of music without changing the nature of the behaviour. For example, to extract the data from an acoustic phenomenon such as phase shifting and represent it in a musical composition through the process of interpolation would mean to observe a fundamental behaviour of the phenomenon and use it in a musical context (this analysis is conducted in the following chapter).

When scrutinising the behaviour of phase shifting, it could possibly be seen in a musical context as a device such as a mensuration canon, in which a passage of music is duplicated and played simultaneously with the original but at a different speed. So to represent phase shifting through *interpolation* would mean to use the rhythmic device apparent in its original (acoustic phenomenon) context in a musical context.

This type of representation can be seen in some of the uses of phase shifting in compositions by Steve Reich (1936-), an American composer often associated with the early beginnings of “minimalism” as a music style. It could be said that the majority of Steve Reich’s early work in
modelling acoustic phenomena involves the interpolation of their behavioural characteristics into a musical context. This is discussed further in the following chapter.

**Extrapolation**

To extrapolate is to “extend (a range of values or a curve) or calculate (unknown values etc.) by extension of trends in a known range of values etc.” (Extrapolate, n.d.). Extrapolation can be the term ascribed to the analogue model of representation, though the term may also be used with slightly different interpretations in a variety of contexts.

As with interpolation, an understanding of how the phenomenon operates is essential; the analogue model is based upon the structure or relationships evident within the original. Due to the extended process involved in extrapolating data (rather than the comparatively linear process of interpolating it), following the breakdown of the phenomenon’s operational characteristics an analogue model of its behaviour should be created in the form of a mathematical function or word flow. This model is most important in extrapolation; however it would still greatly assist in the understanding of how the phenomenon operates for the purpose of interpolating from its data as well.

The creation of a model, as in other applications of applied understanding, then allows an easy transition of a phenomenon’s particular behavioural characteristics from its acoustical context to a musical composition. This means that, for example, the behaviour of a rhythmic device in an acoustic phenomenon could be used to generate data for a melodic device in a musical composition.

In the example given above for interpolation – phase shifting – a rhythmic device observed in its sound context is transferred to a musical composition to generate data for another rhythmic device, which is an example of interpolation. If a composer wishes to extrapolate from the concept of phase shifting, they could perhaps then use the rhythmic device (in this case, essentially a mensuration canon) to create the intervals for a melody, or to dictate the changing pattern of the harmonic structure, or to dictate timbral variation. A metaphorical example of a type of phase shifting can be found in the musical device called an isorhythm, in which the pitch content gradually “phase shifts” with the accompanying rhythmic content. This rhythmic device has not typically been employed as a result of modelling phase shifting, but it can certainly be seen as paralleling its behaviour.
The possibilities within the realms of extrapolating data from acoustic phenomena into musical composition – as within other areas of art, science and philosophy – are seemingly much more expansive than when interpolating. This is shown through the chapters in this research and is also agreed upon by Max Black, who states in reference to analogue models that “identity of structure is compatible with the widest variety of content – hence the possibilities for construction of analogue models are endless” (Black, 1962, p. 222).

1.2.3 The Criteria for a Good Model

In order to evaluate its success, the use of a model requires an understanding of what constitutes a “good model”. Black states that while a poor or ill-fitting model may amount to “nothing more than a strained and artificial description of a domain sufficiently known otherwise”, a properly created model that is fitting for the context may assist in the observation of elements of a set of phenomena that would perhaps otherwise be overlooked; it allows us to “see new connections” (Black, 1962, p. 237). To understand the criteria for a good model, a few elements of terminology and concepts concerning models need to be addressed.

Hesse states that creating a model is effectively saying that one thing is analogous to another. In drawing this analogy, there are still components of the relationships observed in the original that are not evident in the new context to which the model may be applied. These elements may be referred to as the negative analogy, whereas the properties that are evident in both the original and the new are called the positive analogy. The properties of the model that are not yet known are referred to as the neutral analogy, and putting these terms together Hesse states that “when we consider a theory based on a model as an explanation for a set of phenomena, we are considering the positive and neutral analogies, not the negative analogy, which we already know we can discard” (Hesse, 1963, p. 111).

It can be said that an analogy may exist between two subjects by way of their common properties (Hesse, 1963, p. 64). An adequate analogue model should demonstrate the analogy of these common properties through a point-by-point correspondence in the embodiment of its relationships with those of the original. In reference to the analogue model, Black states that “every incidence of a relation in the original must be echoed by a corresponding incidence of a correlated relation” and that in order to conserve the “truth value” of the model there need to be rules for translating the terminology applicable to the model (Black, 1962, p. 222).
There is some leeway in the strictures of the creation of a model. Even with a scale model, Black says that there often needs to be some aspect of “unfaithfulness” in a model in order to adequately represent the original; in reference to scale models, Black states that “there is no such thing as a perfectly faithful model” (Black, 1962, p. 220).

The common properties of the model and the phenomena that it seeks to explain or justify must be observable similarities if, as Hesse directs, they are to “do the predictive job required of them” (Hesse, 1963, p. 77). These observable relations evident in and between a model and another context are described by Hesse as either causal relations or similarity relations. These two relations are further described as vertical relations and horizontal relations, respectively (Hesse, 1963, p. 66). Causal relations are those in which one observable property within a phenomenon is related to another by the necessary condition of the first; “the occurrence of one of the terms is a causal condition of the occurrence of the other” (Hesse, 1963, p. 86). The similarity relation is the connection between one element in a source to a similar element in another source.

Hesse provides an example of these relations observable between two different fields of occurrence: the properties of sound being used as a model for the properties of light.

**Table 1 - Causal and Similarity Relations between Properties of Sound and Properties of Light (Hesse, 1963, p. 66)**

<table>
<thead>
<tr>
<th>Causal Relations</th>
<th>Properties of Sound</th>
<th>Properties of Light</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Echoes</td>
<td>Reflection</td>
</tr>
<tr>
<td></td>
<td>Loudness</td>
<td>Brightness</td>
</tr>
<tr>
<td></td>
<td>Pitch</td>
<td>Colour</td>
</tr>
<tr>
<td></td>
<td>Detected by ear</td>
<td>Detected by eye</td>
</tr>
<tr>
<td></td>
<td>Propagated in air</td>
<td>Propagated in “aether”</td>
</tr>
</tbody>
</table>

The observance of echoes as a property of sound is related (vertically) to loudness, as the occurrence of the latter is a causal condition of the former. The connection drawn between the echoes observable in sound to the idea of reflection as a property of light is related (horizontally) through the similarity of one to the other. Once a similarity relation is made between elements of two phenomena, the causal relations of one are used to draw further connections of similarity between other subsequent properties.
Through the viewpoints expressed by Hesse and Black as well as those from this author, it can be said that a model may be deemed adequate in the event of it exhibiting the following criteria:

- A demonstration of observable similarity relations between the two subjects (Hesse, 1963, p. 77)
- A significant body of causal relations in the model in order to infer further similarity relations between the two subjects
- In the instance of a scale model: an imitative representation of the phenomena, except where the need for manipulation enforces a departure from sheer reproduction, which can be used for “reading off” properties of the original from the directly presented properties of the model (Black, 1962, p. 220 & 222)
- In the instance of a scale model: preserving the relative proportions between relevant magnitudes with minimal deviation; geometrical magnitudes in the original are still reproduced, though with a constant change of ratio (Black, 1962, p. 220 & 222)
- In the instance of an analogue model: preserving the structure or relationships of the original through point-by-point similarity relations, in which every incidence of a causal relation in the original must be echoed by a corresponding incidence of a correlated causal relation in the model (Black, 1962, p. 222)
- The simplest option for a model in producing hypotheses (Hesse, 1963, p. 115)
- Rules for translating the terminology applicable to the model to conserve its “truth value” (Black, 1962, p. 222)

These criteria can be seen as the elements that constitute a good model and are applied to the compositions of this author and other composers. Some composers perform this representative operation with intent, whilst others seem to inadvertently utilise musical devices that can be paralleled with the relational properties evident in acoustic phenomena. Other composers don’t seek to represent the phenomena at all, but rather present them in their natural acoustic state as part of a composition in order to enhance the piece or to provide a focal point.

1.2.4 Acoustic Phenomena as a Compositional Focus and External Stimuli

The use of sound to govern compositional practice has been present since the earliest beginnings of music; Fineberg states that “even the earliest Western treatises about music have used sound as the underpinning for their theoretical constructions – long before any
deep understanding of acoustical or psychoacoustical principles existed” (Fineberg, 2006, pp. 109-110). While this research doesn’t stretch back to the earliest points of musical history, it does examine the practices of composers such as those at the turn of the 20th Century whose compositions and ideas allude to this form of representation in their music.

In the 19th & 20th Century music saw the rise of Impressionist composers, whose compositional practices were named after the prevailing art form of the era. These composers were known for sonically capturing a moment in time within their music. Claude Debussy (1862-1918) and Maurice Ravel (1875-1937) were two prominent French composers in this area of musical composition. While the modelling of acoustic phenomena was not the issue in his music, Debussy’s musical compositions developed a focus on sound itself (Morgan, 1991). The musicologist and author Robert P. Morgan states: “A concern with sound for its own sake is especially apparent in Debussy’s orchestration, where a new level of finesse is achieved in the production of novel timbral effects” (Morgan, 1991, p. 46). Morgan also draws a thread between Debussy’s work and that of the symbolist writers of the late 19th century, such as Stéphane Mallarmé (1842-98), who dissolved traditional syntax to allow individual words to be appreciated more fully for their purely sonic values.

Debussy’s contemporary Ravel, another prominent figure in Impressionist composition, managed to produce at least one work that significantly advanced the idea of acoustic modelling. In his composition Boléro (1928) Ravel uses additive synthesis in the form of additive orchestration to create new timbres. In Electronic Music – Systems, Techniques and Controls, the prominent author on sound and music Allen Strange (1972) describes how in a particular section of Ravel’s piece he combines a horn, celeste, and two piccolos to produce a sound unlike any of the individual instruments used. Strange states that the tri-tonality evident in the score can actually be seen as a reinforcement of the harmonic series for each note of the melody, with the horn playing the fundamental tone, the celeste playing the first and third harmonics and the piccolos playing the second and fourth harmonics (Strange, 1972 p.5).

It can be said that Ravel modelled the orchestral configuration on the harmonic series, although it is unclear whether this is intentional or not. The works of Debussy have been noted by several sources (Hamilton & Mabury) to have had a strong influence on the spectral music movement that emerged in the 1970’s (spectral music is discussed in more detail later in this paper in regards to its contribution to the representation of acoustic phenomena in its
compositional practice). Andy Hamilton writes in the November 2003 issue of The Wire magazine that “French composers such as Debussy, Varèse, Messiaen and Boulez deployed colour intuitively, and spectralists seek to systematise this approach” (Hamilton, 2003, p. 44). Brett Mabury also connects the impressionist composers with the work of spectralists in his thesis paper An investigation into the spectral music idiom and its association with visual imagery, particularly that of film and video (2006). Mabury observes that “the musical thought process of Debussy and his contemporaries became an important source of inspiration and insight in the way spectral composers approach their art” (Mabury, 2006, p. 22).

Parallel to the works of Debussy and Ravel who were focusing on timbral considerations, experimentation was occurring in the field of science and music into the nature of sound and the behaviour of particular acoustic phenomena. The great German physicist and physiologist Herman Ludwig Ferdinand von Helmholtz (1821-1894) began experimentation into various principles of sound phenomena, which can be seen to have influenced the ways in which they were subsequently used by composers. In his text Trigonometric Delights (1998), Eli Maor presents a discussion of how Helmholtz used resonators in the form of small glass spheres of various sizes, each capable of enhancing one particular frequency in a compound tone, to demonstrate the existence of partial tones (Maor, 1998, p. 208). Maor provides a pictorial representation of what these devices would have looked like:

![Helmholtz resonator](image)

Helmholtz used these devices to perform a kind of Fourier analysis (Fourier’s theory discussed further under the topic of synthesis). Following the work of George Ohm (1789-1854), the German physicist and mathematician, Helmholtz discovered that the human ear worked in a similar way to Fourier’s theory, in that the ear could disassemble the combination of simple tones in a composite musical sound. He began experimentation into this phenomenon through a variety of apparatus, beginning with a bottomless bottle that had a membrane (made of a pig’s bladder) stretched across it – when Helmholtz would blow across the bottle the sounding
note would cause the membrane to vibrate in sympathy, which would in turn agitate a suspended wax and cord pendulum that was suspended in the bottle (Blamey, 2008 p.44).

Helmholtz then explored the visual representation of vibration by turning the bottle upside down, sprinkling sand on the membrane, and blowing across the bottle mouth. This system, which followed that of Ernst Florens Friedrich Chladni (1756-1827) (in which quartz dust was sprinkled on a sheet of glass, which was then stimulated using a violin bow), showed the sand moving to the places of the membrane that had the least vibrations – the ‘nodes’. Different notes would create different patterns of sand, as the vibrations would cause the membrane to sympathetically vibrate in different ways depending on the frequency of the note (Blamey, 2008). This scientific exploration was later used in musical compositions, such as Alvin Lucier’s *The Queen of the South* (1972), in which sand is sprinkled on a canvas and then excited by instruments sounding different tones, which moves the sand from the areas of vibration to the nodes (areas of least vibration) and creates visible patterns.

Following this, Helmholtz turned to the use of resonating cylinders to demonstrate Fourier’s theory – that any complex waveform is made up of a series of simple harmonic motions. Maor states that Helmholtz “demonstrated the existence of partial tones by using resonators – small glass spheres of various sizes, each capable of enhancing one particular frequency in a compound tone. A series of these resonators formed a primitive Fourier analyser analogous to the human ear” (Maor, 1998, p. 11).

Predating Helmholtz’s work by 1900 years were the concepts put forth by Marco V. Pollio Vitruvius (c.80 BC- 15 BC). Vitruvius was a Roman philosopher, scientist, architect and theatre designer of the 1st Century BC who was responsible for the architecture of a number of buildings during the reconstruction of Rome under Emperor Augustus. Architecture and sound are inextricably linked components – it seems apparent that throughout the ages, the design of buildings has, in many ways, been guided by the acoustic effects that can be created through the manipulation of the building’s various interior shapes and contours. Cathedrals, churches and other large-scale architectural structures exhibit significant acoustic qualities and concert halls seek to shape sound in a way that will enhance a musical performance without creating a biased sound.

Rob Godman (1964- ), a composer and sound designer from the UK, presents a discussion of Vitruvius in his text *The Enigma of the Vetruvian Resonating Vases and the Relevance of the Concept for Today* (2008). Godman states that during Vitruvius’ era, Greek theatres achieved
an impressive clarity of sound through the use of the curved seating arrangements. These arrangements formed “large horizontal reflecting surfaces” that ensured that the sounds waves travelling from the stage would hit the audience in a direct path, with no interruption or reflection (Godman, 2008 p.1). Vitruvius, in developing the architecture of Roman theatre of the time, felt that this quality could still be improved upon by use of his own special devices, stating that:

In theatres, also, are copper vases and these are placed in chambers under the rows of seats in accordance with mathematical reckoning. The Greeks call them Echeia. The differences of the sounds which arise are combined into musical symphonies or concords: the circle of seats being divided into fourths and fifths and the octave. Hence, if the delivery of the actor from the stage is adapted to these contrivances, when it reaches them, it becomes fuller, and reaches the audience with a richer and sweeter note. (cited in Godman, 2008, p. 1)

Vitruvius suggested that bronze vases were to be made of proportionate size to the theatre that would produce the notes of the harmonic series up to two octaves when struck. His idea was that the notes sung by the voices on the stage would then resonate within these chambers to produce an additional level of clarity in the sound; this sound would be particularly effective as it would be re-sounding the exact same pitches as those on stage (Godman, 2008, p. 3).

The placement of the vases was also an important consideration for Vitruvius. He stated that these vases need to be placed in a particular configuration to achieve the desired result. Once all the vases were set up, they would then sympathetically vibrate when they captured certain harmonics as sung on stage – this meant that when one of the singers performed a scale of notes perfectly in tune, the vases would ring and create the chord ensuing from the vocalist’s notes. It has also been considered that these vases were used to help singers that relied on their ear for maintaining pitch, as they could then hear the correct pitches being resonated by the vases whilst the incorrect pitches would be left silent (Godman, 2008, pp. 3-4). This is similar to the way timpani are tuned – the timpanist hums or sings the correct note into the timpani and if it is tuned properly it will resonate in sympathy, causing a re-sounding of the same pitch.

Following his research on Vitruvius, Rob Godman used the concepts of the ancient Greek architect in his own compositions. In his work Halo (2006), Godman uses an additive
synthesiser to emulate the Vitruvian resonating vases – this is achieved by using a microphone that detects transient dynamic changes through the patch [bonk~] in MaxMSP software, which then triggers the relevant pitches of the synthesiser (further discussed in the chapter on resonance).

The experimentation conducted into various acoustic phenomena by Vitruvius and Helmholtz paved the way for later composers who utilised these acoustic phenomena in their musical compositions. A brief introduction to some significant composers follows and will be developed in later chapters.

Alvin Lucier (1931- ) is an experimental music composer who repeatedly demonstrated the use of acoustics and their related phenomena in his musical compositions. His work *I am Sitting in a Room* (1969) was of particular influence to this research. In it he explores the use of resonance as a compositional highlight (Lucier, 2005). Another use of resonance can also be seen in Lucier’s *Nothing is Real* (1991). A number of Lucier’s works involve the use of scientific experiments and audio test equipment to give a demonstration of the principles of each phenomenon rather than a representation of them.

Karlheinz Stockhausen (1928-2007) was a contemporary of Lucier. Stockhausen was a German composer that both utilised and represented acoustic phenomena in his musical compositions; Allen Strange (1972) discusses how Stockhausen used amplitude modulation as a process of musical representation in his compositions, such as *Hymnen* (1967) (Strange, 1972 p.10). Stockhausen also utilised the phenomenon of ring modulation; an example of this can be seen in his work *Mantra* (1970), in which the pianists operate ring modulators and sine tone generators to modulate the sounds they produce on the piano.

The concept of sound as a compositional building block became the focus of composers following World War II, with the inception of *musique concrete*. This compositional style involved tape music based on “concrete” (natural) sounds, as opposed to electronically created sounds. These sounds were transformed through editing their length, playback speed and superimposing other audio fragments. Composers such as Pierre Schaeffer (1910-1995) and Pierre Henry (1927- ) began the experimentation into this field, with their combined piece *Symphonie pour un homm seul* (*Symphony for a Man Alone*, 1950), which involved the use of modified vocal sounds such as whistling, breathing, laughing and talking, as well as other sounds, such as footsteps. More recently, composers of this style like Helmut Lachenmann (1935) have been using an extensive palate of instrumental sounds as building blocks in their
compositions, such as the performing instrumentalist’s scrapes, breath sounds, nails on strings, guitar fingerboard fret-board squeaks and many more.

Another ramification of the experimentation conducted by Helmholtz in acoustic phenomena was the later use of synthesis. Helmholtz’s demonstration of the principle of synthesis allowed for the construction of the modern synthesiser, and then allowed composers such as Gérard Grisey to represent this acoustic phenomenon in his work from 1975 Partiels (Mabury, 2006, p. 77).

The representation of acoustic phenomena can be seen in the music of Steve Reich. His work involving the representation of acoustic phenomena in musical composition was one of the catalysts for this research project. Reich seems to be one of a select few composers that very clearly and forthrightly worked with this type of representational composition. The use of phase shifting as a compositional device was a primary influence on Reich’s early works. This application of the device can be seen paralleled in a number of compositional tools. For example, the medieval technique called isorhythm was a type of looping technique involving pitch and rhythm cycles.

The isorhythm is the unifying compositional device that first appeared in the motets (a variation of the “clausulae” vocal compositions that involved the addition of Latin or French words to the upper voice) of 14th-Century composers such as Philippe de Vitry (1291-1361). It is a device in which a repeating series of pitches (the “colour”) and a repeated rhythmic unit (the “talea”) of different lengths would often begin together and then converge after the repetition of their different lengths allowed – for example, a pitch series (loop) of six notes with a rhythmic unit (loop) of five notes would converge after five repetitions, or thirty notes (Grout & Palisca, 2001, pp. 84-85 & 98-100).

Modern interpretations of isorhythm can be seen in the work of French composer Olivier Messiaen (1908-1992). In the first movement of his piece Quartet for the End of Time (Quatuor pour la fin du temps, 1940) the piano part is comprised of a sequence of seventeen chords that are repeated in correlation with a rhythmic ostinato that is made up of twenty-nine durations. This isorhythm creates the effect similar to phasing which is also compounded by the superimposition of the cello playing a five-note pitch series against a fifteen-note rhythmic unit (Morgan, 1991, pp. 337-338).
Though not related to acoustic phenomena, the same cyclic loops evident in isorhythm can be heard in the early works of Steve Reich in his use of the acoustic phenomenon “phase shifting.” In these pieces the superimposition of two or more fragments of audio or melodic lines with slightly differing lengths create a “phasing” effect, as the parts begin together, move out of phase and then reconverge. Reich also associates his processes with that of strict musical canons, stating the process in phase shifting could be labelled an “infinite canon” (Reich, 1968).

Reich’s early tape pieces, *It’s Gonna Rain* (1965) and *Come Out* (1966) were the embodiment of the phenomenon phase shifting (two or more sound fragments sonically separating and then re-converging). Reich then took the behaviour observed within this phenomenon and used it within an instrumental context with his subsequent works *Melodica* (1966), *Piano Phase* (1967), *Violin Phase* (1967), *Drumming* (1970) and a myriad of other phase-focused works. This type of representation, which is the embodiment of the qualitative values of sound phenomena being transferred into a musical context, demonstrates the method of modelling acoustic phenomena for musical compositions.

More recently, composers have been utilising acoustic phenomena in their compositions through the use of technology. These composers work within the field of experimental music, computer music, avant-garde composition and sound manipulation. Carl Stone (1953-) works predominantly in the arena of live electronic music, and has used audio manipulation techniques such as time dilation as a compositional focal point. In his work *Shing Kee* (1986) he represents this through the gradual lengthening of a five-second audio sample over a period of around 17 minutes. This type of work is a demonstration of the use of time dilation and shows how highly involved sound fields of the past can now be encapsulated in a single synthesiser, allowing for the wide-spread use of acoustic phenomena. This kind of composition also exhibits the trend towards utilising sound phenomena through electronic manipulation techniques rather than the representation of the phenomena solely through instrumental performance techniques.
1.3 Defining the Research Area

This research project aims to examine the compositional processes used in representing acoustic phenomena in music. This research enquires into the modelling of acoustic phenomena over time, examines the frequent as well as the less common approaches and seeks to identify areas that are still yet to be explored.

The writing on this area of composition seems to be too disparate as contributors to the field rarely seem to draw the threads between the composers that all share this common principle. Also, it appears that few composers have actually used this principle in its fundamental form. Many proponents of this field utilise the acoustic phenomena in their compositions – through a variety of means – but very few seem to represent these phenomena, either explicitly or implicitly, by way of modelling their behavioural characteristics to generate a compositional practice in music.

On examination, the lack of clarity of this area could be assisted through redefinition and exploration, and there is also a need to assimilate the strands of compositional principles that are situated in related fields. There needs to be an argument put forward highlighting the importance of clarifying this area of music composition in addition to building on this body of knowledge. It is important to consider the value of the fundamental ideals when considering future progression in this field, as the furtherance of this research area needs to be achieved without letting the central principle become obscured.

This research is based upon pursuing these ideals, and will do so through a series of structured steps. It attempts to demonstrate ways in which acoustic phenomena can be used in new ways and to represent other phenomena that have had more limited use. This research also seeks to clarify the distinction between utilisation and representation. In the case of the latter, this will be further categorised as an interpolative or extrapolative modelling method that uses the fundamental characteristics of an acoustic phenomenon in application to musical form, texture, timbre, rhythm and pitch.

Fundamental to this research is the exploration of composers who have utilised or represented acoustic phenomena in their musical compositions, as well as the consequent composition of a number of new works that will demonstrate their use through pre-existing and non-existing (new) means. While this research can be seen as an exploration of representation within
music, it is specifically focused on modelling as the form of representation and on acoustic phenomena as the representational matter.

1.3.1 Existing Research

A body of research has been conducted that has very close ties to this area. Peter Blamey’s PhD thesis (2008) titled *Sine waves and Simple acoustic phenomena in experimental music: with special reference to the work of La Monte Young and Alvin Lucier* makes particular reference to the relationships between music and acoustics, with an emphasis on the work of La Monte Young, Alvin Lucier, acoustic phenomena and the sine wave. The thesis examines the use of these acoustic phenomena by avant-garde composers in their music and how their use have broadened the field of music and sound, and also examines the ways that Lucier and Young have explored and produced acoustic phenomena through their use of the sine wave.

Blamey’s focus on Young and Lucier can be summarised in one of his introductory statements: “In other words, what these two artists share is a practice that involves the realisation and elucidation of specific acoustic phenomena” (Blamey, 2008, p. 2). This summary makes it quite clear that Blamey’s work has an important relation to this research as they share a similar focus: the identification and exploration of the works of composers that have utilised acoustic phenomena in their compositions. Blamey provides a number of insights into the area that highlight some important points.

The title of this research thesis, *Acoustic Phenomena as a Model for Music Composition*, seems to be paralleled by Blamey’s statement of intention concerning his own thesis:

This thesis aims to detail the aesthetics of a compositional practice focused on producing these phenomena, and to examine the role that acoustic phenomena play in the account of perception (in the form of hearing), attention (in the form of listening), and of sound itself, that the works of Lucier and Young present. (Blamey, 2008)

It should be made clear, however, that there is a point of departure between this research and that of Blamey’s. Blamey’s text has a specific focus on the sine wave as well as the two composers, Lucier and Young, whereas this research attempts to outline a broader field of composers who have utilised this compositional ethic and explore a more diverse area of acoustic phenomena. It also strives to make clear areas that have yet to be explored or fully realised, and then to follow through into those areas by means of constructing compositions that demonstrate an incorporation of the phenomena’s characteristics into the music.
While Blamey does present a thorough discussion of the area it seems apparent that his focus – as is the case with a large body of the composers studied in this research – is more on the production of these phenomena in musical composition, and less so on the representation of them (which is the focus of this research).

### 1.4.4 The Contribution of Steve Reich to the Research

Before outlining the methodology used in this research it is first worth mentioning the contribution of Steve Reich to the development of the research ethic and the significance of it in relation to the development of the guiding principles behind the decision-making process that occurs at points in this thesis. This research, as it involves a creative process to produce the musical works, has had points at which decisions needed to be made. The a priori principles that were required to resolve these decisions were based on a number of factors. These factors stem from the importance placed on:

1. Representing acoustic phenomena as opposed to using them
2. Representation through instrumental techniques rather than electronic
3. Self-evident compositional techniques

This last value in particular stems from the ethic of Steve Reich. Reich’s musical works comprise a large proportion of this research’s inspiration and on-going stimulation, and as such a number of his ethical considerations towards the treatment of musical composition have also been shared and valued in this research. One of Reich’s primary principles of composition was that the processes used within were to be self-evident. In his essay *Music as a Gradual Process* (1968) Reich states: “What I’m interested in is a compositional process and a sounding music that are one and the same thing” (Reich, 1968).

This research also shares this ideal as the compositions created within are aimed at demonstrating a compositional principle. The demonstration of a compositional principle will not be effective if it is obscured by its own processes, as the listener will not be able to observe the technique without having the guiding principles explained. This means then that self-evidence in the production of the compositional techniques are a necessity, and as such this conclusion has been factored into the decisions made within the composition of each musical work.
1.4 Methodology

The data under examination in this research is not a series of quantifiable or statistical values which means that strict numerical analysis will not yield results. The nature of the material is that of musical compositions, philosophical principles, compositional and performance practices, academic viewpoints, previous scientific exploration of sound, and sound phenomena in their natural states.

Of these areas, the data selected for generating results and conclusions will be the observable practices of music composers and their principle ethics, musical compositions by other composers and musical compositions from this researcher. The compositions will be analysed through several mediums: through existing analysis of the compositions by other researchers, by analysis of the music scores and through auditory perception in listening to recordings of the works.

1.4.1 Method of Research

A survey of popular research methods revealed the following common approaches: descriptive, analytical, conceptual, empirical, applied and fundamental. Research approaches are also classified as either quantitative or qualitative. Upon examination of these different approaches, it was determined that a combination of these would be effective in application to this research.

The objectives of this research can be paralleled with the exploratory and descriptive study approaches. C.R. Kothari (2004) describes the objective of an exploratory or formulative research approach as gaining familiarity with a phenomenon or seeking to achieve new insights into it (Kothari, 2004, p. 2). This is essentially the objective of this research, although it could be said that it also has the objective of a descriptive research study, which seeks to accurately portray the characteristics of a particular individual, situation or group (Kothari, 2004, p. 2).

According to Kothari’s comparison of qualitative vs. quantitative research, this research seemingly utilises a longitudinal qualitative approach, as it is concerned with subjective assessment of attitudes, opinions and behaviour spanning several time periods (Kothari, 2004, pp. 4-5). However, it also has aspects of an inferential and simulation-based quantitative approach, as the research “forms a data base from which to infer characteristics or relationships of population” which in some cases occurs through “the construction of an
artificial environment within which relevant information and data can be gathered” (Kothari, 2004, p. 5). In terms of this research, this means the synergising of several composers’ compositional ethics in order to draw conclusions regarding the area as a whole, in addition to using acoustic phenomena in new compositions (simulation). Kothari states that the simulation approach “can also be useful in building models for understanding future conditions” (Kothari, 2004, p. 5), which is the goal behind the modelling of acoustic phenomena in this research.

The simple summary of these methods is the “case study” approach. According to Kothari, this can also be understood as clinical or diagnostic research, as the case study method reaches basic causal relations by going deep into the causes of things or events of interest. The case study method has been used in similar areas of research, such as the work conducted by Peter Blamey. In his exploration of acoustic phenomena Blamey presents in-depth case studies of the work of La Monte Young and Alvin Lucier. These case studies examine the works of the composers through several stages of their development and are then used to document the role acoustical theories have played, particularly in the conceptualisation and realisation of Lucier and Young’s musical works (Blamey, 2008, p. 19).

Brett Mabury’s paper from 2006 concerning “spectral music” can also be seen as using a case study approach, although he himself doesn’t mention the use of this method. His research involved a discussion of numerous composers’ works, which preceded the composition of his own work. His own composition, Moments, was used for further analysis and discussion into the relevance of the work according to the field of research.

The approach chosen for this research is that of a case study, due to the nature of the material under study in this research and the effectiveness of this method with the examples of research given above. Case studies are commonly used in research of a qualitative nature, as this type of research approach is often used to “penetrate situations in ways that are not always susceptible to numerical analysis” (Cohen, Manion, & Morrison, 2007, p. 253). The case study method is defined by Cohen, Manion & Morrison in Research Methods in Education as:

A specific instance that is frequently designed to illustrate a more general principle, it is ‘the study of an instance in action’... It provides a unique example of real people in real situations, enabling readers to understand ideas more clearly than simply presenting them with abstract theories or principles. (Cohen et al., 2007, p. 253)
According to Cohen, Manion and Morrison the case study approach to research should display certain characteristics:

- A rich and vivid description of events relevant to the case.
- A chronological narrative of events relevant to the case.
- The blend of a description of events and the analysis of them.
- A focus on individual proponents, or groups of proponents, with an aim to understand their perception of events.
- An elucidation of specific events that are relevant to the case.
- An integral involvement of the researcher in the case.
- A portrayal of the richness of the case in writing up the report.

(Cohen et al., 2007, p. 253)

The research design of this thesis will be structured around these principles of a case study.

### 1.4.2 Research Design

The case study approach is used in this research by studying instances in which composers have sought to represent acoustic phenomena in their musical compositions. Additional musical works have then been created based upon the ideas found in the various compositions, which in turn have been used as further case studies. The chapters of this research each present a different acoustic phenomenon and a subsequent discussion of the composers who are associated with their use, the modelling of the phenomenon and the musical works created by this author. This format allows the use of each phenomenon to be observed in comparison to others, which helps to draw conclusions about the viability of the phenomena as models.

The elucidation of how to represent acoustic phenomena in musical composition is the focus of this research. As previously stated, the two significant methods of approaching this type of representation is through *interpolation* and *extrapolation*. As the realisation of acoustic phenomena through musical composition is easiest to accomplish by identifying the underlying nature of each phenomenon, the creation of a model has taken place in this research for phenomena that can be subjected to *extrapolation* in addition to *interpolation*.

The research design for this paper was formulated to allow the case study approach to be utilised in each of the areas discussed above. This design entails the following procedure which is outlined in a number of sequential steps:
1. **Review of current and previous work in the field.** This involves research into a number of composers that have used, or sought to represent, acoustic phenomena in their musical compositions.

2. **Define research area.** In defining the research area, the inclusion of definitions of acoustic phenomenon and associated terms is also necessary for the ease of understanding the content. Outlining the research – its scope, parameters, and justification – is the focus of this current chapter.

3. **Identifying acoustic phenomena.** The scope of this research dictates that only a select few acoustic phenomena will be explored with any real depth, due to the numerous phenomena that exist. Each acoustic phenomenon is described according to peer review and personal observation.

4. **Experiment with the acoustic phenomena.** In some cases, experiments have been conducted with the acoustic phenomena to further elucidate their properties and to demonstrate the characteristics through first-hand observation. This has been achieved through physical experimentation – construction of live sound environments that allow an acoustic phenomenon to be observed – as well as digital exploration. The exploration through digital media has been achieved with the software programs Max MSP, Pro-Tools and Sound Forge. Some of this experimentation has been “behind-the-scenes” and therefore not catalogued.

5. **Demonstrate how the acoustic phenomena can be modelled for composition.** The creation of a model that represents each phenomenon’s behavioural characteristics has been performed for the majority of the phenomena. These ‘models’ take the form of a word-flow, picture diagram or algebraic function and include suggestions of how they could be used for musical applications.

6. **Compose a body of works representing a selection of acoustic phenomena.** Each acoustic phenomenon that was deemed to be most relevant to this research has then been represented and explored within a major musical composition or a minor compositional study.

7. **Analyse works and models.** The analysis of each work and the evaluation of the success of the models have been accomplished through personal observation using the pre-determined *Criteria for Evaluation*. As part of these criteria, each chapter states whether
the composition’s use of the acoustic phenomenon is either an extrapolation or interpolation of its behavioural characteristics.

### 1.4.3 Criteria for Evaluation to Develop Conclusions

In the case study format the overall conclusions of the research are drawn from an evaluation of each case. In this research, the evaluation occurs through the use of a set of pre-determined principles; the **criteria for evaluation**. These criteria take the form of a set of questions that have been referred to in the appraisal of the musical compositions, the process of modelling, the phenomena as models, the work of other composers and the research as a whole.

The primary aspect of the research subjected to evaluation is the modelling of phenomenon. This evaluation involves the determination of whether the models presented perform their function adequately and also whether the phenomena being used demonstrate the qualities of a good model. The set of criteria outlined for this follows the outline of what makes a good model discussed previously (in section 1.2.3: The Criteria for a Good Model). These criteria will be applied to each area of research in terms of their relevance to the subject under scrutiny.

<table>
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<tr>
<th>Table 2 - Criteria for Evaluation</th>
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<tr>
<td>Do the compositions demonstrate the application of a viable model through either interpolation or extrapolation?</td>
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<tr>
<td>Does the model facilitate the observation of similarity relations between the phenomenon and the composition? (Are the compositions self-evident?)</td>
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<tr>
<td>Is there a significant body of causal relations in the model in order to infer further similarity relations between the two subjects?</td>
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<tr>
<td>In a scale model: is it an imitative representation which can be used for “reading off” properties of the original?</td>
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<tr>
<td>In a scale model: does it preserve the relative proportions between relevant magnitudes with minimal deviation?</td>
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<tr>
<td>In an analogue model: does it preserve the structure or causal relationships of the original through point-by-point similarity relations?</td>
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<tr>
<td>Is the model in the simplest format, without it being an over-simplification?</td>
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<tr>
<td>Are there rules for translating the terminology applicable to the model to conserve its “truth value”?</td>
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<tr>
<td>Does the modelling in this context allow for the expansion of the field?</td>
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In addition, the principles that help guide the process of composition are those mentioned earlier, including (1) representing acoustic phenomena as opposed to using them, (2) representation through instrumental techniques rather than electronic and (3) self-evident compositional techniques. A final consideration specific to the compositions is whether or not they are strictly theoretical or performable – whilst this doesn’t impact the success of the models themselves, it will create implications for the research as a whole if the data leans toward a particular trend.

1.4.4 Compositions

A large part of this research is the experimentation with the acoustic phenomena observed within the compositional practices of other composers and other unused phenomena as well. The resultant forms of these experimentations are the musical compositions that demonstrate how acoustic phenomena can be represented in music in a number of ways. As previously mentioned, in addition to these compositions being a demonstration of this research principle they have also been referred back to the Criteria for Evaluation in order to generate the data for their analysis, which assists in drawing conclusions.

These may take the form of either small compositional studies or major musical compositions. These works have been created for the acoustic phenomena phase shifting, delay and resonance. The phase shifting representations are made up of a series of small compositional studies, delay has been used as a model for a major musical composition and resonance is represented in two major musical compositions. For the remaining phenomena, no works have been created to represent them; instead, theoretical outlines are given on how these compositions could be used in an extrapolative representation in music through the creation of a model.

Each of these compositions is provided in this research in two forms: the music score and sound recording. The scores and recordings for the works are provided as appendices, and each chapter that relates to the compositions will refer the reader to the relevant appendices. It is strongly recommended that readers of this research participate in an auditory and visual analysis of each composition, as it makes the principles developed within this research area more clearly evident – much more than words will allow.
1.4.5 Limitations to the Research

Breadth of History & Composers Encompassed

The composers who are examined in this research are those whose musical works span the ages from around 1950 to the present day, with the exception of references to a few turn-of-the-century advocates (Debussy, Ravel & Cowell). Other important contributors who pre-date this time frame will still be mentioned in order to give historical context, however less comprehensively than those who are within the primary time frame of focus.

The main focal point of this research will be the compositions and methods of Steve Reich, Alvin Lucier, La Monte Young and Gérard Grisey. These composers are deemed to have made substantial contribution to this field of research and their inclusion draws together the threads from a number of different musical sub-genres (minimalism, experimental music, spectral music and scientific experimentation with sound phenomena).

Scope of Research

This research is limited in its scope in regards to the number of ideas explored. The areas and ideas that are being explored could potentially lead a researcher to spend many more years cataloguing and extrapolating from, however only a handful of concepts are explored within this research due to time constraints. The final chapter of this research provides a few unexplored concepts that offer some interest, perhaps to be delved into at a later date.

Acoustic Phenomena Included in this Research

There are a vast number of recorded acoustic phenomena that exist in the natural environment and in the realm of electronically produced sound. As such, it is too big an area to be encompassed in one single piece of research, and only a selected few acoustic phenomena will be under study.

The main acoustic phenomena under study are phase shifting and resonance, and also delay, reverberation, waves (of sound, sine, the brain and the spheres of the Earth), synthesis, the harmonic series and sound spectra (encompassing harmonics, overtones and partials) and interference.

*Phase shifting* as an acoustic phenomenon has been included in the scope of this research as it seems to epitomise the earliest case of obvious representation – composers representing this phenomenon often made it quite clear what their intent was, which was not common in this field of study. Its use as a model in composition was also the catalyst for this research, as it
sparked the enquiry into what other phenomena had been utilised in this same way. The sound phenomena *resonance* and *delay* are included as they seem to be utilised in small ways by some composers, but largely unrepresented in this field of musical composition.

The phenomena *reverberation*, *waves*, *synthesis*, the *harmonic series* and *interference* were included in this research as part of an expose of composers who have modelled these phenomena in their musical works. They have also been included in this research to show how each of the phenomena can be modelled for further use in musical composition – the scope of the research, however, dictated that compositions were not created based upon these findings. Instead, the focus has been on what can be done with resonance and, to a lesser extent, phase shifting.

**Areas Excluded from this Research**

This research does not seek to present a comprehensive exposé of the *incorporation* of acoustic phenomena into music composition or performance. This field of inquiry contains many facets as composers and performers can be seen utilising acoustic phenomena in their music through multifarious methods.

This research focuses on *representing* acoustic phenomena through modelling their behaviour in musical works. The phenomena observed may be found in natural sound spaces, as well as the phenomena that are produced through the assistance of computers, synthesisers and sound manipulation technology. In the case of the musical works that are created as part of this research, the representation of the acoustic phenomena is achieved through purely instrumental techniques; the use of technology to produce the desired effect has been avoided where possible and explored in only limited means.

The exclusion of technology, except in speculative cases of theorising potential methods for modelling phenomena, does not limit the validity of this research area. The use of instrumental techniques to represent acoustic phenomena is an element that shows a point of departure of this research from other writings or compositional practices. This makes the research different to, for example, a discussion on the use of electronically created reverberation or computer-created sounds in live performances.

Some composers do share this value on the use of instrumental techniques as opposed to electronic. Fineberg composes works that often utilise computer generated sounds as an integral component of the music, but even he states his wariness of favouring music produced
solely through technology. In *Classical Music, Why Bother?* Fineberg states that “the homogenised sounds of even the best synthesizers give only very gross approximations of these [musical] attributes and no sense at all of the physicality or even possibility of the music,” and that acoustic sounds are much richer and more complex when compared to artificial sounds (Fineberg, 2006, p. 68 & 126).

The composers examined in this research are a combination of those who represent acoustic phenomena through instrumental techniques and those who use the phenomena in their compositions in their native sound form, through technology or otherwise. Allowing a circumventive foray into a slighter broader field of study allows for the clarification of a more refined scope in the primary area of research; however, the focus remains on determining which composers were able to represent the phenomena rather than use them.

### 1.4.6 Format and Organisation of Chapters

Each chapter of this research presents a focused discussion on a particular acoustic phenomenon. This involves a definition and discussion of the phenomenon, followed by a look at a selection of composers who have represented – or used– the phenomenon in their musical compositions. The final part of each chapter consists of the research developed from the information gathered – this culminates in a musical composition.

The chapter overview is as per the following:

1. Acoustic Phenomenon
   1.1 Definition
   1.2 Composers utilising this phenomenon in music
   1.3 Modelling the phenomenon
      1.3.1 Compositions created in this research (where applicable)
      1.3.2 Creation of a functional model

Chapter 1 is the introduction to this research, and the subsequent chapters 2-8 follow the above layout – a discussion of an acoustic phenomenon and its proponents. The final chapter (9) is a discussion of the findings and conclusions drawn relative to the thesis, and in particular has been related back to the *Criteria for Evaluation*.

The next chapter of this dissertation begins the exploration into acoustic phenomena – the definition of each phenomenon, their musical proponents and resulting compositions. This exposé commences with the phenomenon known as *phase shifting*. 
2.0 Phase Shifting

The observance of phase shifting, both as an acoustic phenomenon and its apparent use in music, was the catalyst for the commencement of this research. It has been used by a number of composers in musical compositions, particularly by Steve Reich in his early tape-loop pieces and his subsequent transference of the phenomenon into musical compositions.

Phase shifting can be classified as an acoustic phenomenon in accordance with the definition outlined in this research. It is an observable occurrence of the quantitative and qualitative values of sound, whether it is produced in a natural acoustic context or as an electronic manipulation of sound.

This phenomenon can be successfully used as a model for compositions; this is evidenced through its use by composers and through the out-workings of this research. It largely consists of temporal devices, which can then be easily modelled and used in the same way in music.

The causal relations evident in the acoustic context of the phenomenon are sufficient enough to allow numerous similarity relations to be drawn with its musical context through modelling, which means that phase shifting lends itself quite easily to becoming a scale model. It can also be used as an analogue model, which is discussed at the end of this chapter; here, a model is created that can be used to apply the phenomenon to a musical context through interpolative or extrapolative means.

Following a survey of composers and their musical compositions, this research explores what musical processes can be seen as representing this phenomenon, which results in the potential re-expansion of the concept. Firstly, a discussion will be conducted regarding the definitions of phase shifting in its many contexts.

2.1 Phase Shifting Defined

The term “phase” and, by relation, “phase shifting”, is used in a number of contexts. It can be used in reference to the interaction of a waveform in physics, the acoustic phenomenon as commonly heard in the superimposition of audio loops, as well as the use of this phenomenon as a music process.

2.1.1 Phase and Phase Shifting in the Physics of Sound

The terms “phase” and “phase shifting”, sometimes also known as “phasing” (such as in the text of Davis & Jones, 1989), are used in reference to the position of a waveform. There are
numerous texts that discuss the physics of sound (Backus, 1969; Berg & Stork, 2004; Howard & Angus, 2001; Moravcsik, 2002; Parker, 2009; Pierce, 1983; Pollard, 2002; Rossing, Moore, & Wheeler, 2001) and referring to these texts is helpful in understanding a discussion of phase shifting as the position of a waveform. The composer and sound consultant John Holland, in his text Sound Waves and their Properties in Surrounding Media, states that “phase shifts” are an occurrence within the interaction of two or more sound waves: “In addition, sound waves which interact with one another may produce sound interference phenomena, including stationary waves, phase shifts, and combination effects” (Holland, 1985).

In their text on recording, Huber & Runstein (2001) state that “phase is measured in degrees (°) and can be described as being the relative phase degree angle with another wave(s) over 360°” (Huber & Runstein, 2001, p. 31). This notion is easiest to consider with the sine wave, which is so named because its amplitude corresponds to a trigonometric sine function (Huber & Runstein, 2001). A sine wave is considered to start at 0° with an amplitude of zero, which then increases to its positive maximum when it reaches 90°, moves back to an amplitude of zero at 180°, increases to a maximum (in a negative direction) at 270°, and then reaches zero amplitude at 360° before beginning again.

This can be seen in the following diagram (figure 2) illustrating a sinusoidal waveform starting at zero amplitude and moving through an amplitude of 1, 0, -1, and 0 for the corresponding phase degrees of 90°, 180°, 270° and 360°. The x axis shows the phase position in degrees and the y axis shows amplitude (ranging from -1 to 1) at each given position.

![Figure 2 – A sinusoidal waveform.](image)

Whenever two or more waveforms occur at the same time in the production of a sound their amplitudes will be different at any point in time if they start their cyclic period at different times. These waveforms are referred to as “out-of-phase” with each other. The following examples show two waveforms of the same frequency and amplitude that are in different
phase positions in relation to each other. In the example on the left (in figure 3) the two waveforms are 90° out-of-phase with each other, and in the example on the right the two waveforms are 180° out-of-phase with each other.

Figure 3 – Two examples of out-of-phase waveforms.

Whenever two or more waveforms are sounded simultaneously their amplitudes are combined at each point in time. When the two waveforms are in-phase, their amplitudes are added together which creates a doubling of their amplitude. When the two waves are only slightly out-of-phase, such as in the example on the left in figure 3, they will constructively and destructively interfere with each other, creating varying amplitudes (Huber & Runstein, 2001, p. 31).

These occurrences can be seen in figure 4 in the diagrams from Huber & Runstein’s text demonstrating the effect of combining two waveforms that are either in-phase or slightly out-of-phase. On the left, two waveforms are in-phase so their combined amplitude doubles. On the right, two waveforms are slightly out-of-phase, which varies the resultant waveform’s amplitude:

Figure 4 – Waveform interaction & interference (Huber & Runstein, 2001, pp. 31-32).

Two waveforms are considered to be completely out-of-phase with each other when they have a phase difference of 180°, which, according to Rumsey & McCormick in their text Sound and Recording: An Introduction (2006), is when “the positive half-cycle of one coincides with negative half-cycle of the other” (Rumsey & McCormick, 2006, p. 8). When this occurs the two
waveforms cancel each other out as their amplitudes equate to zero when added together at each point. This occurrence is known as phase cancellation. This can be seen in the right hand illustration in figure 3, and a demonstration of this is also provided by Huber & Runstein:

![Waveform Illustration]

Figure 5 – Two waveforms 180° out-of-phase (Huber & Runstein, 2001, p. 32).

The concept of two or more waveforms constructively or destructively interfering with each other, such as in the second example (left hand diagram in figure 3), is referred to as interference, and results in acoustic beating. These elements of waveform interaction are discussed further in a later chapter, including the composers who have utilised these phenomena in their compositions.

Observing the effects of waveforms being in different phase positions in relation to each other allows phase shifting to be more clearly understood. Phase shift is the term used to describe the lead or lag in one waveform compared to another, which results from a time delay between the two waveforms (Huber & Runstein, 2001, p. 32). As an example, if a 500 Hz waveform, which completes a cycle every 0.002 seconds, was played simultaneously with another identical waveform that was delayed by 0.001 seconds it would then be out-of-phase by half a cycle, which is 180°.

The description of phase shifting so far has been in the context of two sine waves interacting with each other. On a larger scale, this same occurrence can be witnessed in complex waveforms made up of musical material, audio recordings of spoken phrases, or a variety of other sonic materials.

2.1.2 Phase Shifting as an Acoustic Phenomenon

Phase shifting, or phasing, can also occur when recording a single sound source with two or more microphones. If the two microphones are placed at different distances from the same sound source, a time delay will occur between the receipt of the two signals, which will then
Phase shifting can also be exhibited in a larger scale context. The phenomenon can be observed in the simultaneous playback of two sound sources; Steve Reich was one composer who initially noticed this effect occurring with a single sound source played back by two tape machines. Michael Nyman provides a definition of phase shifting as an acoustic phenomenon observable within the context of sound manipulation:

> Working solely with a single fragment of sound – a short pre-recorded spoken phrase, a short rhythmic or melodic pattern (usually a decoration of a modal harmony) – and submitting it to a process whereby it gradually and progressively moves out of phase with itself. (Nyman, 1999, pp. 152-153)

In this context, phase shifting can be seen an acoustic phenomenon that becomes audible when two copies of a single sound fragment are sounded simultaneously and one of the sounds is played slightly slower or faster than the other. According to the concept of phase shifting put forth in an acoustical context by Huber & Runstein, Rumsey & McCormick and Davis & Jones a phase shift need only be a delay in the placement of one of two sound sources. In the instance of phase shifting occurring within two loops of audio of slightly different speed being played back simultaneously the delay in placement of the second sound is a gradual process, as the two sounds gradually move out-of-phase with each other which creates the effect of them sonically separating. When the sound fragments are repeated and this process of phase shifting occurs the two sounds start together, and then begin a cycle of separation until they are 180° out of phase and then eventually become in phase again, reaching 360°.

Referring this concept back to the interaction of two or more sine waves, the process can be viewed in the following diagram. In the following figure (6) two sine waves of different speeds – the red wave two times faster than the blue – are played simultaneously. The two waves start together but due to their different speeds they gradually separate and the red wave completes two whole cycles in the space of one cycle of the blue.
Figure 6 – Demonstration of phase shifting in two sine waves.

The point at which the two waveforms reach 180° out-of-phase is indicated (circled) on the chart, which is where the two waveforms reach zero amplitude at the same time but coming from opposite poles (one from positive, one from negative). The convergence points are also indicated (with green stars), which are the points at which both waveforms reach zero amplitude coming from the same polar position (both from negative, in this case). In the context of the interaction of two waveforms, this is not the definition of phase shift as defined by authors such as Huber & Runstein, but is instead the appropriation and demonstration of the interaction of two complex sounds in what authors such as Michael Nyman define as phase shifting.

The latter context of this phenomenon was initially most commonly heard using tape loops, where one tape reel was played at a different rate than the other. It is unclear whether this manifestation of the phenomenon was apparent before its use in tape loops. Paul Hillier feels that this wasn’t the case; in his foreword of Steve Reich’s *Writings on Music 1965-2000* (2002) Hillier suggests that the phenomena “could hardly have been imagined until the invention of tape and the new opportunities it afforded for manipulating sound” (Hillier, in Reich, 2002, p. 5).

This acoustic phenomenon, in the context of two pieces of audio being played back simultaneously at different speeds (such as in the tape loops that Reich observed), can be visually demonstrated in the following diagram:
In this diagram, each box represents a particular sound and the series of boxes in a row represent the repetition of that sound fragment. “Sound B” is a slightly slower version of “Sound A”, which means that for every repetition of Sound B it becomes more out-of-phase with Sound A until it reaches the point of being 180° out-of-phase (indicated by the red dotted lines in the diagram). Following this, the two sounds keep repeating until they re-synchronise (indicated by the solid black lines in the diagram) and the process is repeated.

In this manifestation of phase shifting (either in tape loops or other forms of repeating audio) being 180° out-of-phase doesn’t create phase cancellation as it would if it were the interaction of two sine waves or two complex sound sources. As Rumsey & McCormick state, “phase is only a relevant concept in the case of continuous repetitive waveforms, and has little meaning in the case of impulsive or transient sounds where time difference is the more relevant quantity” (Rumsey & McCormick, 2006, p. 10). In this situation, the sound sources are complex waveforms on a large scale, such as a three-second length of audio, and the term 180° out-of-phase is used in more of a structural sense rather than a description of the interaction of the waveforms on a micro-scale.

When phase shifting is heard in an audio-loop manifestation, such as the tape loops of speech in Reich’s early works, it creates a range of acoustical effects. When listening to this phenomenon, the process can be heard to begin as a slightly ‘reverberant’ sound of a single sound source, similar to the early reflections of a natural or synthesised reverberation (early reflections discussed further in Reverberation & Delay). As the two sounds separate further, the effect increases and the sound then clarifies into two different sources that are rhythmically separate.

This effect is known as the *precedence effect*, sometimes known as the *Haas effect*. When a single sound is heard from two sources, a frequent example being a person’s voice arriving at a listener’s ears from both the person speaking and a loudspeaker amplifying the sound, if the two sounds are within 50 milliseconds of each other they are perceptually fused together;
both sounds are heard as one (Rumsey & McCormick, 2006, p. 34). If the sounds are spaced over 50 milliseconds apart they are audibly separable, with the second sound source being heard as an echo of the first. In the example of phase shifting being discussed, this process is heard with a progressively increasing distance between the sound sources, and then the process is reversed; the distance between the two sounds gradually decrease and they become audibly closer together until they reconverge and become a unified sound.

In this example the sounds would be considered to be 180° out-of-phase when the start point of one sound synchronises with the middle point of the other; for example, a four-second audio fragment synchronising with a duplicate sound at two seconds (half-way through the sound’s cycle). Paul Hillier also describes these effects apparent in listening to a tape-loop manifestation of phase shifting in a similar way:

> At first, only minute segments of time separate the two parts, creating the illusion of an acoustic echo. As the disparity between the two parts grows, the aural effect becomes more complex, and eventually this resolves into a clear imitative pattern, once the two parts have achieved a rational degree of separation of one beat or more. (cited in Reich, 2002)

In a way, phase shifting could be seen as having a direct relationship with the acoustic phenomenon ‘delay’. The process involved in phase shifting could be replicated by subjecting a sound source to a digital delay apparatus, in which the rate of delay used progressively increases. The echoed sound would perhaps start off as being a millisecond behind the transient, and then proceed to move further away from it. The delayed signal then gets sonically “pushed” further away from the initial transient until it synchronises with the repeated sound, at which point the sound and its delay become one sound again – and then the process repeats.

### 2.1.3 Phase Shifting as a Musical Process

Phase shifting has the potential to be used as a guide for musical processes, which can be in the form of a compositional tool or a performance technique. Phase shifting and its associated phenomena (phase, phase cancellation, phase position etc.) also have relevance to the manipulation of audio in a sound recording context, such as the phase inversion of a signal to nullify any instances of phase cancellation. In this discussion, however, the material discussed will be the ways on using the phenomenon in a musical process rather than sound manipulation.
One potential use of phase shifting in a performance setting can be observed in the inadvertent use of the technique by orchestral performers. As phase shift can be seen as the delay of one sound in relation to another, this can be seen in orchestral situations where one violin player may play slightly behind or in front of the beat of another. This creates a phase shift effect – not with a changing temporal value, such as in the phase shifting observable in the repeated sounds of audio loops that differ in length, but with a consistent delay. The effect creates what may colloquially be called a “fattening” of the sound, as the two sounds are not far enough apart to be audibly separable but create a broad texture in the orchestral setting, particularly when this effect occurs with a whole section of string players.

Phase shifting, when observed from a musical perspective, creates a number of effects that can be categorised into musical components. Compositionally, phase shifting may be used as a structural tool for rhythms. Rhythm is one of the driving features of this phenomenon, and is the primary characteristic that was seized upon by composers who utilised this phenomenon in their musical compositions. The rhythmic manifestation of the effects is akin to several musical techniques; the canon (as discussed in regards to Steve Reich), the isorhythm (as touched upon in the introduction) and the superimposition of multiple tempi (described in Conlon Nancarrow’s works as “tempo canons” by Kyle Gann, 1995). It can also be seen as a form of the rhythmic compositional tools polyrhythm and ‘beat displacement’. These latter two techniques can be seen as a parallel to phase shifting, as a secondary sound source – in these cases, a second rhythmic unit – is displaced by an increasing interval.

Consider this example of phase shifting in the context of two sound sources. If ‘Sound A’ is a crotchet played on every beat of a bar in 4/4 time, then ‘Sound B’ can be seen as starting with Sound A and then gradually being displaced a micro increment (in this example, a 16th-note of added value) for every occurrence. This displacement pushes the second sound further out of sync with the original sound, until the two notes are equally opposed – Sound B is, at this point, displaced by a quaver beat, placing it directly on the off-beat. As Sound B is then displaced further, it gets closer to the following crotchet beat of Sound A, and then eventually synchronises with this note. This musical example is illustrated in figure 8, in which the convergence points of the two sounds (A & B) are indicated by a red line and the point at which they reach 180°-out-of-phase is indicated with a green line.
Phase Shifting | Acoustic Phenomena as a Model for Music Composition

Figure 8 – Musical example (1) of phase shifting.

In figure 8, the point at which the two notes synchronise occurs every five beats. The crotchets of Sound A are the equivalent duration of four sixteenth notes (semiquavers), and the durations of Sound B are the equivalent duration of five sixteenth notes (a quarter note plus a sixteenth note). This means that the notes synchronise every \((5 \times 4)\) notes, which is every 20 (semi-quaver) notes.

If this same example were set in 5/4 time, the two sound sources would synchronise at the beginning of every bar. This is demonstrated in the next figure (9), with sound A & B being written alternatively as tied groupings of four and five semi-quavers, respectively:

Figure 9 – Musical example (2) of phase shifting.

At each stage during this example of phase shifting as a rhythmic device the listener can observe a variety of rhythms occurring, although never the same two in a row. This example shows phase shifting in a purely rhythmic light, as the harmonic content consists of only a single note. The effects that occur when the harmonic content is varied are wide-ranging; as in the tape-loop example described above, the harmonic interaction creates a constantly flowing and evolving harmonic texture.

Phase shifting, as well as having parallels with the acoustic phenomenon “delay” and the musical technique of beat displacement, could also be seen as having a connection to a strict musical canon. The type of canon that phase shifting would be equivalent to is one that has a variable and changing rhythmic interval between the two parts, as Reich points out (Reich, 2002, p. 139). Paul Hillier also backs up Reich’s view: “As Reich has said, phasing is essentially a
form of canon using irrational numbers” (Hillier in Reich, 2002, p. 5). As mentioned previously, it would be more apt to say that phasing is a particular type of canon; the mensuration canon.

Further discussion of the musical examples of this phenomenon and its parallels with other musical techniques will occur in the following chapters under the title of the different composers who have used this acoustic phenomenon.

### 2.2 Composers Representing Phase Shifting in Music

#### 2.2.1 Steve Reich

Steve Reich, one of the leading purveyors of the musical genre labelled ‘minimalism’, has provided a great deal of the inspiration for this research. Kyle Gann points to Reich as an advocate of the use of phase shifting in music:

This technique [phase shifting], of two identical phrases played at the same time but at slightly different tempos so as to go out of phase with each other, was most characteristic of Reich’s works of the 1960s and early 1970s: Piano Phase, Come Out, It’s Gonna Rain, and Drumming. (Gann, 2001 p.5)

Reich’s work in the examination and representation of ‘phase shifting’ in musical composition was a musical concept that sparked the initial motivation for research in this area. Through examining his works one can see the steady transition from experimentation with the acoustic phenomenon, such as in Come Out (1966), and It’s Gonna Rain (1965), to the representation of it in later works like Piano Phase (1967) & Violin Phase (1967). His works then seemed to settle into a pattern that created a very stylized sound that came to define the music we hear of Steve Reich. Reich’s musical compositions came to largely explore the idea of music as a ‘gradual process.’

Reich’s early works in tape loops were the first to demonstrate the acoustic phenomenon of phase shifting. In his early tape works and then his subsequent musical transference of the concept, Paul Hillier states that “a complete pattern is presented at the outset by two players (or tape machines). This is then juxtaposed against itself, as one player moves slightly ahead and therefore out of phase with the other” (Hillier in Reich, 2002, p. 4). This describes the process used in phase shifting, which is evident in the majority of Reich’s early works.
It’s Gonna Rain was Reich’s first tape loop piece, in which all the effects that are described in the previous paragraph on phase shifting can clearly be heard. In his interview with Jason Gross from Perfect Sound Forever: Online Music Magazine, Reich confirms that It’s Gonna Rain was the origin of his use of phase shifting: “Technically, it’s been said many times, the discovery of the phasing process was within that piece” (cited in Gross, 2000).

This particular phenomenon in the work came out by accident; Reich was intending to line up two fragments of audio (the spoken phrase “it’s gonna rain”) so that they were exactly 180° out of phase, but due to an uncalculated difference in the playback speeds of the two tape machines being used, one of the audio fragments was played back at a slightly different tempo which resulted in the phase shifting effect. Reich describes this process in his interview with Jason Gross:

I made identical loops and I thought I would line them up in a particular relationship... I put on headphones (which were stereo with each ear with a separate plug going into the two machines). By chance, two machines were lined up in unison. So what I heard was this unison sound sort of swimming in my head, spatially moving back and forth. It finally moved over to the left, which meant that the machine on the left was slightly faster passing in speed than the machine on the right. So the apparent phenomenon in your head is the sound moving to the left, moves down your left shoulder and then across the floor! Then after a while, it comes into an imitation and then finally after four or five minutes, you hear ‘it’s gonna... it’s gonna... rain... rain...’ (cited in Gross, 2000)

This effect, while unintended, was one that Reich was fascinated by, and as such the effect was kept as one of the primary focus points of the piece. The first half of the piece includes an “introduction”, which has a loop of the phrase in different lengths, and following this is a straight presentation of the phase shifting effect with the single repeated vocal phrase being played back by two sound sources slipping out of phase. This effect was so profound that Reich then decided to use the same technique in the second half of the piece but multiplied; Reich here uses 4 and then 8 different copies of the same sound fragment. This part of the piece also utilises a mosaic of vocal material, as bits of several vocal sentences are strung together.

The sound source for this piece is a speech by an African American preacher (Brother Walter), and due to the nature of his vocal characteristics – the vibrancy, extremes of pitch range and dynamic contrast – the tape-loop produces a wide range of effects. In the first half of the
piece, the two sound sources interact in a subtle way that increasingly shift and change. The
cyclic evolution is such that a barely perceptible difference in rhythm initially occurs, after
which it gradually creates new rhythmic features.

Figure 10 – Transcription of Reich’s It’s Gonna Rain (Schwarz, 1981, p. 388).

An approximate musical transcription of the vocalist’s speech pattern in It’s Gonna Rain,
provided by K. Robert Schwarz in Music as a Gradual Process Part I (1981) shows the perceived
melodic and rhythmic content. The melodic nature of the preacher’s voice creates a unique
melody that constantly changes – it’s almost like a serial composition that uses a pre-
determined set of notes that are then used over and over in different combinations. However,
it also seems more than this – a serial composition uses particular segments of the harmonic
spectrum (series of notes from 12-step or 24-step scales), whereas a composition utilising a
voice as the sound source includes the micro-tones that exist in between. When the vocal
layers split into 8 parts, the afore-mentioned effects are compounded and also expanded
upon; the main noticeable effect is that the vowel sounds of each word are initially extended,
as the 8 different voices start slightly after one another.

Reich went on to use this process again in Come Out, which exhibits the same effects as those
observable in It’s Gonna Rain. This piece also begins with two unison channels of the same
audio – a phrase spoken by Donald Hamm, shortened from “I had to open the bruise up to let
the bruise blood come out to show them” to “come out to show them.” One of the two audio
fragments are then allowed to phase forward by being played at a slightly different tempo and
later in the piece the channels are divided into four and then eight voices. The melodic content
of the spoken phrase has been approximated by Schwarz in a transcription:

Figure 11 – Transcription of Reich’s Come Out (Schwarz, 1981).
In the transcription it can be seen that the melodic quality of the spoken phrase differs from that of *It’s Gonna Rain*, which creates an entirely different sound when subjected to the phase shifting process. The effects that arose within these tape pieces occurred largely without any fore planning by Reich and also without any further assistance once the initial process had been set in place. According to Schwarz, Reich calls these effects the “resulting patterns” which involve melodic, harmonic and polyrhythmic combinations that progress on their own once the process of phase shifting has been implemented (Schwarz, 1981, p. 385).

When observing phase shifting in tape loops, such as in Steve Reich’s *Come Out*, the vocal sounds create their own unique texture. Each consonant of the short repeated phrase (“...come out to show them...”) interacts with the other repetition of the same sound, and in fact all the consonants of the phrase seem to somehow blend and mesh in a strangely inarticulate way. Whilst phase shifting in this form begins as described earlier (as in, sounding like a delay), it then transforms into something very hypnotic and trance-like, with a metronomic pulse being superimposed with another slightly faster pulse. The transient sounds of the vocal enunciations become a confusing stutter of staccato attacks that constantly shift and evolve, and the harmonic content is completely derived from the approximate pitch of the voice, which also shifts and evolves as different melodic points interact through synchronisation and syncopation.

Reich found the phenomenon of phase shifting to be particularly fascinating and useful as a compositional tool. In discussing his first observation of phase shifting in *It’s Gonna Rain*, Reich states “...I thought to myself, "This is unbelievable." Instead of a particular relationship, here is a whole way of making music, going from unison through all these contrapuntal relationships, all the way back to unison” (cited in Cott, 1996). Following his tape pieces, Reich then explored this concept of using phase shifting as a compositional tool in a musical context. This aspect of Reich’s musical development is what makes him a focus within this research area.

Other composers also developed a fascination for the effects created by Reich’s tape loop pieces. Brian Eno (1948- ), who is most well-known for his work with ambient music, found Reich’s *It’s Gonna Rain* of particular interest. Eno expresses his fascination with the effects created by the overlapping tape loops, stating that “since the material is common to both tapes, what you begin to notice are not the repeating parts but the sort of ephemeral interference patterns between them” (Eno in Tamm, 1995). His own work, *Music for Airports* (1979-1982), was an ambient tape loop piece that was a direct product of the influence
created by Reich’s work. *Music for Airports* involved cyclically structured phrases of seven recurring notes, performed by four tape players that are set in motion at random times, each playing long durations of a different cycle of phrases or events. The tape players were set so that they could play unattended, which allowed for the piece to stretch on for a great length of time (Tamm, 1995, p. 135).

Reich’s first representation of Phase Shifting as an acoustic phenomenon in a musical composition occurred in *Melodica* (1966), written for tape loops and melodica, which was closely followed by his first purely instrumental phase composition: *Piano Phase* (1967). In this work, Reich directly translates the manifestation of phase shifting in its tape loop context to music by applying the rhythmic features apparent in the former to the latter. Nyman directly connects *Come Out* as the predecessor of *Piano Phase*, as it uses the exact same technique but in a musical context. Nyman states that the process used in the context of a live instrumental piece instead allows it to be “coloured by human fallibility and adapted to musical, not spoken, sound…” (Nyman, 1999, p. 153).

Reich first conceptualised *Piano Phase* by making a tape loop of a small melodic riff (comprising of 12 notes) for piano and then playing the same ostinato along with it on a live piano. Reich allowed his live performance to slightly “phase forward” by playing at a slightly faster tempo, and after this was achieved it was next performed in a totally live setting by using two pianists to play the riff together. One player performs the pattern at a fixed tempo while the other player implements a slightly faster tempo, which creates the phasing effect of the two patterns. Once the second player’s ostinato reaches alignment with the first player’s, the process is paused and the two play the riff together but with the second player’s riff placed one note ahead of the first. This is shown in a brief segment of the piece in the figure (12) below.

![Figure 12 – Reich's Piano Phase (excerpt) m. 1-3 (Schwarz, 1981, p. 388).](image-url)
This excerpt of the piece shows the first three measures of the piece. In measure 1 only the first player performs the riff, and the second player commences in unison with the first at measure 2. The dotted lines between the measures denote the “phasing” that the second player is to implement, which is directed to occur over a period of around 20-30 seconds. At measure 3 the second player arrives one sixteenth ahead of the first player, but in rhythmic unison. This process then continues throughout the piece until the second player arrives back in rhythmic and melodic unison with the first, which occurs after twelve renditions of the cycle.

Upon transferring the characteristics observed in this acoustical phenomenon to a musical setting, Reich made the observation that the effect was much like that of a musical canon or round. Reich states that this technique was used firstly in Piano Phase, where “each phase position is just a short unison canon with a slightly different rhythmic interval” (Reich, 2002, p. 140), and this can be seen in the example of the piece above. He also observed the same technique being used in his early tape works It’s Gonna Rain and Come Out. Reich draws attention to the fact that by having one tape loop slipping slightly behind another identical loop what essentially is occurring is a unison canon in which the temporal interval between the two sounds is variable and constantly changes (cited in Cott, 1996).

In a general discussion of his early work in his Writings on Music, Reich provides a very simple statement of confirmation: “My work is based on canon” (Reich, 2002, p. 140). In his 1996 interview with Jonathan Cott, Reich reveals that this discovery was somewhat of a comfort as it meant that he was varying a musical concept that had pre-existed since at least the thirteenth century and also existed in forms of non-Western music. He states that “…to find that those techniques or related ones, occurred not only in your own culture’s music but also in the complicated polyphony of West African and Balinese music was even further encouragement” (cited in Cott, 1996).

Paul Hillier also confirms the relationship of Reich’s techniques to those used by earlier composers. Hillier notes that a particular hallmark of Reich’s work is a sense of shifting views of a single object, which he finds personified in the multitude of devices that Reich employs that can all essentially be seen as the one process: the canon. “This, more than any other single device, dominates Reich’s musical processes. Whether as strict imitation or in derivative forms (such as voice exchange or isorhythm), canonic devices permeate the texture of almost every one of Reich’s works” (cited in Reich, 2002, p. 5).
Reich’s next piece involving the instrumental setting of the acoustic phenomenon’s characteristics was *Violin Phase* (1967). The work was written for either four live violins or for one violin with three pre-recorded violin parts. The phasing in this piece involves each violin part being in a different phase position at each point in the piece; when playing the twelve beat rhythmic ostinato, violin 1 is four beats behind violin 2, which is eight beats behind violin 3. Violin 4, which is the live soloist if using the tape loop version of the piece, plays a gradually phasing version of the ostinato across the top of the other parts. Parts 1, 2, and 3 are shown in the following excerpt (measure 16) from the piece:

![Excerpt from *Violin Phase* by Steve Reich](image)

Figure 13 – *Violin Phase* (excerpt 1) by Steve Reich (Schwarz, 1981).

Violins 1, 2 and 3 are also written as a combined staff in the event of the piece being performed as a solo work with the tape loop. In this work, Reich stepped up the level of complexity by having four parts phasing against each other instead of two. Also, after using the process already in *Piano Phase* Reich was able to more predictably use the “resulting patterns” that were previously uncontrolled. This was partly achieved by the violin 4 part occasionally doubling these resultant patterns that appeared in the other parts as a consequence of their phasing content. The figure below shows the resulting patterns that emerge from the first three violin parts shown in the above figure.
The fourth violin is directed to play any of these resultant patterns in order to reinforce them. The blank bar (D) is Reich’s direction for the soloist to choose their own “resultant pattern” as a substitute for either of the first three (A, B & C).

A later composition of Reich’s that returned to the concept of phase shifting as a focal point was *Clapping Music* (1980). *Clapping Music* uses the same type of phase technique as used in *Piano Phase*, but rather than phasing through the micro-rhythms between the beats to reach the next synchronisation point it jumps from one quaver beat to the next. The piece is written for two players (who clap) and as with *Piano Phase* the first player maintains a constant 12-beat ostinato whilst the other player phases forward. This phasing is notated by shifting the whole rhythmic pattern backward by one quaver beat per every repetition.

Reich’s use of phase shifting as an acoustic phenomenon to represent in musical composition was an element to his style that he realised would affect his subsequent compositions. As K. Robert Schwarz noted, Reich realised that his transference of phasing from an electronic context to a musical one was “...indicative of a trend that was to occur several times in his
career – the exchange of ideas from electronic music to live performance” (Schwarz, 1981, pp. 386-387).

Following his study of West African drumming in Ghana and his subsequent composition of his seminal work *Drumming* (1971), Reich also set out on a course that no longer held electronic mediums in high regard for musical compositions. Reich states that “[*Drumming*] confirmed my intuition that acoustic instruments could be used to produce music that was genuinely richer in sound than that produced by electronic instruments” (cited in Schwarz, 1982, p. 233).

When looking at Reich’s body of works in light of this research focus, it may be said that the majority of his compositions interpolated data from acoustic phenomena in transference to a musical setting. The application of phase shifting to his compositions was as a structural device used for rhythm and musical form, and as the effects of phase shifting in its tape-loop context was also essentially a rhythmic phenomenon it’s justifiable to say that the representation was an interpolation. The same observable principles within the acoustical context were replicated as closely as possible in the musical context. Had Reich used the rhythmic feature of phase shifting to dictate the melodic content of the piece, for example, then it could be said that he had extrapolated the data.

This observation is by no means a criticism, however, as Reich can be seen as contributing one of the most significant body of works to this area of musical composition. Reich’s many-faceted use of phase shifting in a musical context was extensive and as such hardly leaves any room for other composers to explore unused avenues. Upon a re-clarification of what may be viewed as phase shifting, however, it seems that a significant number of other composers did in fact also pursue this same area.

### 2.2.2 Henry Cowell

Kyle Gann points to Henry Cowell as the origin of phase shifting being utilised in musical composition. Henry Cowell was an early ethnomusicologist who wrote the revolutionary text *New Musical Resources* in 1930, in which he presents numerous ideas that predate their later use. Kyle Gann (1995) titles Cowell the “twentieth century’s first great theorist of rhythm”, as he developed a new form of rhythmic notation developed from his ideas on how rhythm could be re-imagined.

Cowell had a great love of rhythm in all its forms, and in particular was very interested in how rhythms could be superimposed when derived from equal divisions of a common beat; an
example of this would be dividing a whole note into five, six and seven equal parts and then playing them all at the same time. This would result in a layering of three different tempos at the same time, in ratios to each other of 5:6:7 (Gann, 1995, p. 1). Cowell found the ensuing effect intriguing; he stated that “To hear a harmony of several different rhythms played together is fascinating, and gives a curious aesthetic pleasure unobtainable from any other source” (cited in Gann, 1995, p. 1).

Kyle Gann presents a discussion of Cowell in his text *The Music of Conlon Nancarrow*, as Nancarrow was inspired by Cowell’s works and took them to the extreme with his player-piano pieces (discussed in more detail in the next chapter). However it was in another of Gann’s texts – *Minimal Music, Maximal Impact* – that he actually suggests that the phase shifting technique used in the treatment of rhythmic devices by Reich (and others within the minimalist movement) had its antecedents in the work of Cowell – the superimposition of phrases of differing lengths.

Many of Cowell’s suggestions point to the possibility of phase-shifting, having regular rhythmic units go in and out of phase at the same time. And phase-shifting, of course, becomes the primary preoccupation of Reich’s work of the 1960s in Come Out, Piano Phase, Violin Phase, and It’s Gonna Rain. (Gann, 2001)

The connection that Kyle Gann draws between these two composers seems plausible. The technique of using phase shifting in composition by Reich is quite comparable to that of Cowell, as both of their compositional theories utilise a superimposition of rhythms that go in and out of phase with each other. After surveying other writings on the matter, it seems that Gann may have been one of the first to point out this connection.

Gann doesn’t seem to suggest that Cowell was aware of the acoustic phenomenon that shared a connection with his rhythmic devices, and it is doubtful that he would have been. Part of what this research seeks to do is to draw threads between different compositional fields of different ages and works of composers, and this connection between the representation of phase shifting by Steve Reich and the superimposition of rhythms suggested by Kyle Gann does just this. This connection then broadens the area under study when researching phase shifting.

If Kyle Gann can relate the idea of multiple layers of rhythms, as found in the work of Henry Cowell, to phase shifting (as seems quite rational), then the field of composers using phase shifting in their compositions expands exponentially. If phase shifting can be seen as analogous
to polyrhythm, then similarity relations can be drawn between the uses of polyrhythmic devices in any composition by any number of composers. In cases such as Cowell (and Conlon Nancarrow, discussed in the following chapter), the superimposition of more than two layers of rhythms could be determined as multi-tap phase shifting. The term multi-tap in this research has been coined from its context in digital delay, in which several echoes of differing temporal intervals are created from the one sound source. This concept can be seen in works such as Reich’s Violin Phase in which several layers of phase shifting occur simultaneously.

Cowell’s work became ever more fascinated with the concept of super-imposed rhythmic patterns, leading him to suggest to others the best ways to achieve them. Extraordinary means would be required to achieve them, as the playing of these rhythms by even highly polished performers would be extremely difficult. Cowell suggested using the “player piano” to achieve these rhythmic functions (Cowell, 1996 p.65), as the rhythms can simply be punched into the rolls and then be performed perfectly; “Why not hear music from player piano rolls on which have been punched holes giving the ratios of rhythms of the most exquisite subtlety?” (cited in Gann, 1995, p. 1)

This suggestion then seems to have inspired the majority of the work of Conlon Nancarrow, who took this concept as far as was physically capable with the medium of the player piano – which is much further than would be humanly possible.

2.2.3 Conlon Nancarrow

Conlon Nancarrow was an American composer who spent most of his life living in isolation in Mexico. His methods followed closely after Henry Cowell’s, whose text New Musical Resources came to define a great deal of his work. Like Cowell, Nancarrow was fascinated by the superimposition of different rhythms and tempi.

Nancarrow’s use of rhythmic devices can be seen as an earlier use of phase shifting that seemed to begin with Steve Reich. This is a consideration also put forth by Kyle Gann, who states that “many of his tempo canons for player piano feature melodies going in and out of phase with themselves, which is exactly what happens in Reich’s early phase pieces” (Gann, 2001). The extensively complex rhythms he created as a result of using this method lead him to take up Cowell’s suggestion and have them played by a player piano.

Kyle Gann’s text The Music of Conlon Nancarrow highlights the initial inspiration Cowell had on Nancarrow; Gann states that Conlon Nancarrow was inspired by Cowell’s idea (one that Cowell
himself never followed through with) and learned to produce his music independently of
performers by having his pieces played by a player piano. In 1948, Nancarrow bought a player
piano and used it to explore multifarious aspects of rhythmic superimposition and
simultaneous tempi (Gann, 1995, p. 2).

Gann discusses Nancarrow’s variety of style, rating him as equal to composers like Webern and
Ives; Gann states that the sheer volume of “diverse strategies” implemented by Nancarrow
compares to the significant work of Liszt and Busoni in their piano music (Gann, 1995, p. 2).

Nancarrow’s works can be seen as using phase shifting as their inspirational stimuli, however
upon listening to them it would be the farthest thing from a listener’s thoughts. Nancarrow’s
pieces are relatively brief but don’t sound that way, as at blisteringly fast speeds they cram in
more notes than a twenty-five minute Liszt sonata in under three minutes (Gann, 1995, p. 3).
For example, Nancarrow’s Study No. 36 (1969-1977) is less than five minutes long but has a
fifty-two page score that is “black with ink”, which requires intense listening from the audience
to hear everything that happens (Gann, 1995, p. 3).

Whilst the speed and intensity make aural analysis difficult, it’s still apparent that by following
Cowell’s footsteps Nancarrow commonly utilised phase shifting – in the form of tempo canons
and superimposed rhythmic layers – within his musical compositions. Nancarrow
demonstrates a plethora of equally intricate compositional techniques, some of which share
the role of portraying phase shifting – it seems that his isolation afforded him a concentration
on his chosen musical field unlike many other composers. The vastness of his ideas is quite
staggering, and the even more remarkable feature about them is the long, slow, painful hours
that would have been required to punch in the notes of his pieces into the piano roll – writing
that many notes in a score would be difficult enough! Kyle Gann helps demonstrate the weight
of Nancarrow’s compositional concepts, none of which he used more than once or twice:

1. A pitch row split into discrete segments (Study No. 1)
2. A pitch row using internal repetitions of a pitch cell (No. 4)
3. A texture built up from motives that repeat non-synchronously, i.e., out of
   phase (also involving every note on the piano without duplication) (No. 5)
4. An isorhythm (repeating rhythmic series) altered by systematic changes of
tempo (No. 6)
5. Different isorhythms played at once (No. 7)
6. A piece divided simultaneously into equal-length sections by texture changes,
   and into a different number of equal sections by melodic structure (No. 11)
7. Polyphonic use of isorhythm in which the color (pitch row) and talea (rhythmic
   row) are associated differently in each contrapuntal line (No. 20)
8. A canon in which the voices gradually reverse roles (No. 21)
9. A palindromic canon (No. 22)
10. A correspondence between tempo and register (nos. 23, 37)
11. Rhythmic canon in which the canonic voices have wildly disparate textures (No. 25)
12. Use of a 12-tones row as harmonic determinant for triadic music (No. 25)
13. Accelerating isorhythmic canon (No. 25)
14. A steady beat as a perceptual yardstick for changing tempos (nos. 27, 28)
15. A “scale” of tempos proportional to a pitch scale (Nos. 28, 37)
16. Interrupted (and resumed) acceleration (No. 29)
17. A tempo canon whose voices theoretically converge outside the canon’s time frame (No. 31)
18. Isomorphic transformation of a duration pattern to simulate a tempo canon (No. 33, Two Canons for Ursula)
19. Tempo changes within layered tempo contrasts (No. 34)
20. An entire movement played at the same time with itself at a different speed (No. 40)
21. An isorhythm accelerated by subtracting from the individual duration (no. 42)
22. Aleatory tempo canon (no. 44)
23. Use of Fibonacci durations to create the same rhythmic motive at different tempos (no. 45)
24. Irrational, un-notatable isorhythm (Nos. 45, 46, 47 – originally one work)
25. Structural acceleration within a tempo canon (no. 48)
26. Tempo canon in which voices are timed to converge not all at the same time (String Quartet No. 3)

(Gann, 1995, pp. 4-5)

The devices used in this list of note are the superimposition of isorhythms, tempo canons, an extended mensuration canon (example #20) and the use of a scale of tempos that correspond to pitch – this last example is a demonstration of the use of the phenomenon beating in composition, which is discussed further later. Added to this list are innovations that Nancarrow used often: irrational tempo relationships, glissandos with selected notes sustained, and tempo clashes at ratios of 4:5, 24:25, 60:61, etc.

From this vast number of techniques that Nancarrow has demonstrated in his compositions Gann surmises that a limitation of medium was a great benefit for Nancarrow’s work (Gann, 1995, p. 5). And it would seem this is a fair point, as clearly his limited medium has provided the environment in which he was able to produce so many compositional techniques and styles.

Gann touches on some important distinctions between two methods of rhythmic construction that arise within Nancarrow’s works. “Additive” rhythm, which was known to be used by Stravinsky amongst others, refers to the grouping of small durational units into irregular meter progressions such as 6/8, 5/8, 9/8, 7/8, 3/8 and so on. The method used by Cowell and
Schoenberg is the opposite – “divisive” rhythm, which is to take a larger unit (e.g. a whole note) and divide it simultaneously or successively into equal parts of various lengths (Gann, 1995, p. 7). Gann states that both types of rhythm are found in Nancarrow’s music, and that he was the only composer to “thoroughly synthesise the two opposing conceptions of rhythm” (Gann, 1995, p. 7).

Gann compares Nancarrow’s rhythmic concepts to Cowell’s, in relation to the latter’s limitations. The limitation Gann describes is the periodicity in Cowell’s rhythmic system, which meant that after every few beats all voices re-convene in a simultaneous attack. However, this sounds like a form of phase shifting – a number of parts separating (rhythmically) and then “re-convening.” So while it seems that a use of additive rhythm meant there would be less emphasis placed on synchronisation, Cowell’s “limitations” through the use of divisive rhythm meant that synchronisation was inevitable; Gann reiterates this point in saying that “the problem with divisive rhythm was its dependence on a too-predictable periodicity” (Gann, 1995, p. 8). Nancarrow avoided this by switching constantly between additive and divisive rhythms in his compositions, whilst Cowell stuck to divisive. This perhaps means then that an expression of phase shifting can best be represented by a use of divisive rhythmic system.

However, Nancarrow was able to avoid the “problems” of either rhythmic construct as he enjoyed the freedom of using both rhythmic systems. Gann poses that Nancarrow was able to preserve the “energetic, unpredictable feel of additive rhythms within the context of a tempo system related to the pitch relationships of the harmonic series” (Gann, 1995 p.8). Gann also suggest that he was the only composer was able to effectively integrate the micro-rhythms of one style with the macro-rhythms of the other, which solved the dilemma faced by the choice between the rhythm structures used by Schoenberg and those used by Stravinsky. This highly complicated combination of rhythms of course made the playing of them too difficult, which resulted in Nancarrow’s use of the player piano (Gann, 1995, p. 8).

A number of Nancarrow’s works can be seen utilising rhythmic devices that prophetically mirror the phase shifting technique used by Reich and others later in the century. Gann describes Nancarrow’s Study No. 21 (also known as Canon X, written between 1948 & 1960) and highlights the components that can be paralleled with Reich’s phase shifting:

...The famous “Canon X,” in which one voice starts slow and gradually speeds up while the other starts fast and slows down. The 51-note atonal row that governs all pitches in that piece is, by Nancarrow’s own admission, just an arbitrary prop for the tempo
effect. Replace that row with some more limited pitch set, and you could get exactly the same effect in a minimalist context. Several of Nancarrow’s studies flirt with minimalism or minimalist-style processes in one element or another. (Gann, 2001)

One of the hallmarks of Nancarrow’s works was his use of the tempo canon, which is the device used in the above description of Study No. 21 that achieves the effect of phase shifting. This device involves the transposition of the same melody played in different tempos, which are then played simultaneously (Gann, 1995). Gann explains that Nancarrow is not the first composer to use this technique, but is definitely the first to use so many different layers and different variations that result in the need to have a special terminology.

Inherent to the technique of the tempo canon are various events resulting from the canon’s structure to determine the overall form. The most central I have chosen to call the convergence point... the infinitesimal moment at which all lines have reached identical points in the material they are playing. In the late studies, from No. 24 on, this is usually either the climax or the end of a canon, though occasionally a convergence point will fall inaudibly on a rest, and in a few cases not until after the canon is over. Nancarrow’s ways of marking the convergence points exhibit astounding ingenuity. (Gann, 1995, p. 21)

This description by Gann highlights the similarities in technique to that of phase shifting – of course, in Nancarrow’s examples the phase shifting is what was earlier labelled in this research ‘multi-tap’ phase shifting, in which several layers are phase shifting at the same time. This similarity includes the fact that at some point, the different layers ‘converge’, or ‘synchronise’ – while Reich’s works involved these points by necessity due to the pre-disposed process set in place, it seems that Nancarrow celebrated these points in his works by marking them using a variety of musical techniques.

2.2.4 The Totalists

As stated previously, Henry Cowell’s love of layered rhythms is what seems to have pre-dated and perhaps driven the idea of phase shifting as it later manifested in the works of Steve Reich. Cowell’s idea then sparked Conlon Nancarrow’s interest in rhythmic layering, and then another group of composers that also seem to have emerged from this: the “Totalists”. While their works are not necessarily embodied by phase shifting or rhythmic layering, it does seem to take a large place in their compositional process.
Totalist compositions often used tempo canons in the same vein as Nancarrow, which can also be seen as phase shifting in musical composition through rhythmical layering systems. As stated earlier, while totalism as a compositional style does not include phase shifting as a prerequisite it does include a number of proponent composers that have utilised this technique. Kyle Gann, in *Minimal Music, Maximal Impact*, describes the key features of totalism:

Totalism is a style of great beat-related rhythmic complexity in a kind of harmonically limited, post-minimalist context. The rhythmic complexity can come from different tempos going on at once, repeating loops of different lengths, unsynchronized rhythmic cycles, shifts among beats of diverse durations, and so on. (Gann, 2001)

Gann states that Rhys Chatham was one of the first minimalists to enter into the new area of Totalism. Rhys Chatham (1952- ), having initially studied in the field of classical music, began to fuse the minimalist style with the rock genre. Chatham studied under La Monte Young, and his early works reflect the effect Young had on him – *Two Gongs* is a work that is 60 minutes of two different gongs being played in a variety of ways to create a psychedelic blend of sounds.

He then proceeded to reject the claim of Young’s influence on his works, and wrote *Out of Tune Guitar* – an obvious diversion from Young’s strict tuning systems (Oteri, 2008). Chatham’s works continued to involve the use of rock band instruments – electric guitar, bass & drums – in a blown-up scale; Chatham’s *Guitar Trio* uses three guitars in addition to the bass & drums to create a “continuum of overtones drawn from electric guitars playing one pitch” (Gann, 2001).

Chatham then stepped up his use of the electric guitar to create *Die Donnergötter* (for six guitars), followed by *An Angel Moves too Fast to See* in 1989, which is scored for 100 guitars (plus bass & drums.) In this work, Chatham demonstrates his use of phase shifting via polyrhythmic compositional devices. Gann describes his methods within this piece: “Chatham’s *An Angel Moves Too Fast to See* (1989) built musical structures and even melodies from repeating unequal phrase lengths” (Gann, 2001).

William Duckworth is another composer mentioned in Gann’s text *Minimal Music, Maximal Impact* in relation to using phase shifting as a compositional device. Gann links Duckworth’s technique to the same ones used by Cowell – the “links” and “phase-shifting phrases” (Gann, 2001). Duckworth’s piece *Time Curve Preludes* uses these techniques, as do other composers.
of the post-minimalist and totalist genre. The piece uses the Fibonacci series to create the phrase lengths, which are then shifted out of phase with each other.

The list of Totalist composers using tempo canons, which as discussed can be seen as a use of phase shifting, stretches on with a number of other proponents. Glenn Branca’s (1948-) composition *Sixth Symphony* (1988) utilised electric guitar chords recurring at different tempos, and in his *Tenth Symphony* (1994) he pursued tempo canons in much the same way as Cowell and Nancarrow (Gann, 2001).

Joshua Fried (1959-) uses tape and digital loops of different lengths, as well as triggered sounds, in a superimposition that is also reminiscent of phase shifting (Gann, 2001). David First’s (1953) compositional practice involved the use of sets of drones in which the rhythmic beat patterns mirrored the pitch relationships (Gann, 2001).

John Luther Adams (1953-) superimposed different phrase lengths in his works, such as *Dream of White on White* (1992) (Gann, 2001). Mikel Rouse (1957-) wrote a piece for rock quartet titled *Quick Thrust* (1984) that utilises a 12-tone melodic system, in which he also employs layered rhythms in ratios of three, to five, to eight (Gann, 2001).

While this may seem like a glossary of composers of the style, the inclusion of all these proponents of totalism should demonstrate that the art of using phase shifting is a widespread occurrence in the musical world, and in particular in styles like totalism. These composers may not necessarily have been intending to represent phase shifting in their compositions, however through a discussion of the work of Reich, Cowell and Nancarrow it can be seen that the use of polyrhythms, isorhythms, tempo canons and superimposed rhythms of a large variety are all sub-versions of phase shifting in a musical context. This brief overview of composers should highlight the fact that when phase shifting is defined in this way, the area of research becomes very broad indeed.

### 2.3 Modelling Phase Shifting – Compositional Studies and Theoretical Framework

**Development of Concepts and Preliminary Compositions in Phase Shifting**

A pivotal concept in this research developed when theorising how to use phase shifting as a compositional technique was the definition given by Reich as an “unusual canon” (cited in
Cott, 1996). As the use of the qualitative properties of phase shifting had been used in music quite significantly by Reich, there seemed to be little room for expansion upon the idea (except for perhaps using it in different musical settings) unless a composer were to take it a step further.

Reich drew the connection between the characteristics of phase shifting and the musical canon, but also connects it with non-Western styles of rhythmic devices such as the polyrhythms commonly used in West African drumming and Gamelan percussion of Balinese music.

Simply, Ghanaian drumming is constructed so that several repeating patterns, more or less in subdivisions of twelve, are superimposed so that their downbeats aren’t in the same place. Back in 1965, when I started the tape pieces and later the instrumental phase pieces, it was a similar situation: in the phase pieces, two or more players start off in unison but gradually player number two accelerates so that their downbeat is X number of beats ahead of the first player. (cited in Cott, 1996)

The connection drawn between phase shifting and polyrhythms was also noted in a discussion of the work of Cowell, Nancarrow and the totalist composers. This connection seems to pave the way for a significant re-expansion of the possibilities available for the use of phase shifting in musical compositions. For if every polyrhythm were to be viewed as phase shifting, then it would mean that this device has been used in music across an expanse of cultures and periods of musical history.

In order to demonstrate this principle, a number of short music composition “studies” were created that used the idea of what has been called in this research “multi-tap phase shifting”, meaning polyrhythmic and canonical devices. These compositions use “additive rhythm”, as was described earlier in a discussion of Cowell and Nancarrow – in fact, many of the following works can be seen as having a strong similarity to those used by these two composers. The use of layered rhythms of differing lengths is an idea put forth by Cowell and used extensively by Nancarrow in his tempo canons.
2.3.1 Phase Shifting Composition Study 1 – Clave

Appendices 1.1 and 2.1

The clave are commonly used as a rhythmic foundation in Latin American music as well as in African music. The clave seemed to be the natural choice of instrument to demonstrate a form of polyrhythm, due to its nature as a guiding rhythm instrument in several musical styles.

This very brief compositional study consists of layers of rhythms superimposed sequentially, where each rhythmic grouping uses a common denominator note value – the quaver or eighth note. The first rhythmic grouping is of 2 quavers (a crotchet), followed by groupings of 3, 4, 5, 6, 7, 8 and 9 quavers – each rhythm is not literally a series of 3, 6, or 8 quavers but rather single note that is the length of this grouping of notes e.g. 2 quavers = crotchet, 3 quavers = dotted crotchet, etc.

This composition demonstrates an example of additive rhythm, where different lengths of rhythmic units are used that share a common denominator – in this case, a quaver (eighth note). However, in this example the additive rhythm is vertical rather than horizontal – normally additive rhythm involves the use of several meters used sequentially but in this concept the different meters are superimposed to create multiple layers of phase shifting.

2.3.2 Phase Shifting Composition Study 2 – Drums

Appendices 1.2 & 2.2

This concept of viewing phase shifting as polyrhythms was then utilised in a brief compositional study for drum kit, the natural home for these kinds of rhythmic devices.

This kind of polyrhythmic device is not new to the drum kit, or to other percussion instruments; numerous examples are apparent even in popular and rock music genres. The point of composing this demonstrative work was to show the connection between this compositional/performance technique and that of the use of phase shifting in music.

This composition again involves different layers of note groupings, but as opposed to the sequential addition each rhythm’s entry used in the clave composition the rhythms all commence simultaneously at the beginning of the piece. The ride cymbal is used as a time keeping device by playing a constant quaver pattern, while the bass drum plays a grouping of three quavers (dotted crotchet), the snare drum plays a grouping of five quavers and the hi-hat (played with the foot pedal) plays a grouping of four quavers (minim).
In this example of phase shifting through the use of polyrhythms, the synchronisation points (or, as Gann calls them, “convergence points”) are also sonically marked and highlighted, much the same as Nancarrow’s synchronisation points. The marking in this piece is produced by playing the bell of the ride cymbal at the point where the rhythmic layers synchronise – this synchronisation is only deemed justifiable when it occurs at the beginning of the measure, which happens every fifteen bars (which is the point at which the piece repeats.) There is another convergence point, which occurs on beat three of bar eight (the “7.5th bar”, half of 15).

2.3.3 Phase Shifting Composition Study 3 – Breath

This compositional study utilises the medium of human breathing to demonstrate the same principles of “multi-tap phase shifting”. Like the clave composition, it consists of layers of rhythms superimposed sequentially, however the layers of rhythm are performed by the human breath. This means that each layer of rhythm has two different sounds – the “in” breath and the “out” breath.

Having this change in medium and also adding in another variable in the sound pattern creates a markedly different sonic effect – having a variation in sound means that the effects of the phasing rhythms are doubled, as there are more sound variations interacting throughout.
The layers in this piece had no common note value, instead the rhythms were derived from stepping up the tree of notes that exist within a bar of 4/4 time – beginning with minims, then proceeding from crotchets through crotchet triplets, quavers, quaver quintuplets, quaver triplets (sextuplets), septuplets, and semi-quavers.

The difference between this piece and the previous two (plus the following) is that the rhythms are an equal division of a single note (a whole note), which are then superimposed. The other three pieces all use the opposite technique, which is the superimposition of rhythms of different length that have a common denominator – e.g. a grouping of 3 quavers, 4 quavers, 5 quavers.

This rhythmic technique is in essence an even smaller scale use of phase shifting – all rhythms phase against each other and reunite in the space of one bar of 4/4 time. In the other pieces, the convergence points don’t occur for a much longer time.

2.3.4 Phase Shifting Composition Study 4 – Piano

Appendices 1.4 & 2.4

The initial concept for this compositional study was to have a number of simple piano phrases in different groupings of quavers (i.e. 3’s, 5’s, 7’s etc), played together.

The harmonic structure was designed so that the first notes of each phrase, when all played together, would form a consonant and harmonious chord like a C major 7. The rest of the notes would be inharmonious, so that the overall sound would be a jumbled chaos with an occasional shining moment when the patterns synchronised.

When composing the piece, however, some of these concepts were not fully realised, and so had to be changed. When the off-sounding notes began to be written for the first piano phrase, the researcher felt inclined to change them to more harmonious notes, and so the idea of an inharmonious chaos no longer seemed appropriate. The phrases were then continued in this same vein, with the following phrases being written with more consonant sounding melodies.
The piece was composed in C major, 4/4 time. It seemed that when writing each melody, however, that they all had a different tonal centre. The 5 note grouping seemed to be in an A pentatonic (Am7) or C major, the 3 note grouping pulled towards G major, the 7 note grouping uses notes from the E minor arpeggio and the 4 note grouping is firmly based in C major, using the tonic and dominant. The following figure shows the note groupings of 5, 3, 7 and 4 (respectively, from top left to bottom right):

When all phrases were written, it was calculated that they would synchronise after 420 bars (5 x 3 x 7 x 4), and then also realised that they would synchronise after every 105 bars. The start of every note grouping is accented, and occasionally 3 of 4 parts synchronise.

After completing each part, it was decided that the entries in the piece would be staggered to allow each to be heard more clearly. The piece begins with the 5-note phrase, the 3-note phrase begins at the point at which it would have synchronized with the 5-note phrase had they started together, and similarly the 7-note and 4-note phrase are brought in at the points at which they would have synchronized with their predeceasing phrases.
This piece seems reminiscent of Terry Riley’s seminal minimalistic work, *In C* (1964), in which multiple melodic units of differing lengths are superimposed at different points in the piece. Riley’s work was of course on a much grander scale – using fifty three melodic figures, instead of a mere four – however the similarities lie in the superimposition of differing phrase lengths and a tonality that is entirely derived from chance, when different notes of each part align at different points in time to form a number of harmonies. Riley also used melodic fragments that were comprised of the same rhythm and pitches but out of phase.

2.3.5 Conclusions Drawn from Phase Shifting Compositions

These brief compositional studies in phase shifting all use the concept of polyrhythms or micro-versions of tempo canons, such as those used by Nancarrow. These compositions were created in order to demonstrate the shared connection between the concepts of representing phase shifting in musical composition and the use of these other rhythmic devices.

This transference of data from the acoustical context of phase shifting demonstrates its use as a *scale model*. The causal relations evident in the phenomenon were connected to a musical context by similarity relations drawn from each point of its behaviour. These compositions demonstrate a viable application of an interpolative use of modelling as the rhythmic devices observed in acoustic context have been used as a rhythmic device in the musical setting also.

This modelling of phase shifting facilitates the observation of similarity relations between the phenomenon and the composition, as it utilises a scale model representation that can be used for reading off the properties of the phenomenon. The model has been used to draw connections between areas that were largely unconnected previously; what can be seen from these short compositional studies is that the realm of representing phase shifting in composition is a broad one. Drawing a connection between the use of isorhythms, polyrhythms, and canon variations with the use of phase shifting means that whenever any of these tools are observed within music it could be seen as a use of the acoustic phenomenon. This was one of the goals of this research: to demonstrate how composers have used this technique, whether intentionally or not.

All of these uses under study thus far can be seen as an interpolative data transfer, except perhaps for the use of the isorhythmic device by composers like Messiaen, which is touched upon in the introduction. The concept of extrapolating from phase shifting will be explored to give a theoretical framework so a composer could perhaps further utilise this type of representation in compositions.
2.3.6 Modelling Phase Shifting

The use of this phenomenon by other composers and the works created in this research are evidence enough of how it can be successfully used as a model for musical application. This concept can be explored even further through the use of a model.

The extrapolation of ideas from the characteristics of Phase Shifting to create a musical work is easiest to undertake through an understanding of how it functions on a base level. These behavioural characteristics can then be outlined in a model, which allows insight into new areas through the application of the model to a new (musical) context.

A pictorial demonstration of the behaviour of Phase Shifting was shown at the beginning of the chapter in figure 7. To translate this pictorial demonstration into a model with substitutional parts – such as a word flow diagram or algebraic function – one could look at phase shifting in the following way:

\[ X, \ X_r, \ X_r, \ X_r, \ X_r, \ X_r, \]
\[ X + \delta, \ [X + \delta]_r, \ [X + \delta]_r, \ [X + \delta]_r, \ [X + \delta]_r \]

\( X \) = subject matter
\( \delta \) = infinitesimal addition
\( X + \delta \) = subject matter plus infinitesimal addition
\( r \) = temporal repetition
\( X_r \) = representation of simultaneous presentation of both subjects
\( X + \delta \)

Now as a function, phase shifting could be musically represented in more abstract ways. This is achieved by replacing the elements of the function with different musical variables.

For example, rather than the subject matter being a sound fragment (as in *Come Out* or *It’s Gonna Rain*) or a musical phrase (as in *Piano Phase*), \( X \) could equal a particular frequency or note. The infinitesimal addition, rather than being a temporal lengthening of \( X \), could be an increase to the frequency of \( X \).

Let’s use the note A4 (440Hz) as subject \( X \), and replace \( \delta \) with an increase of 5Hz. If ‘\( r \)’ were to equal a repetition set at a rate of 60bpm and both subjects were to be presented (sounded) simultaneously, the following would be the result:
This use of phase shifting is similar to the method used by Conlon Nancarrow – equating the interval between two rhythms to that of two pitches. This is, however, just one simple example of the **extrapolation** of the properties of phase shifting into a musical composition – when you consider the realm of variables within music it is easy to see how this field of exploration becomes quite vast. Consider these musical variables:

- Pitch – intervallic relationships, key signature, harmonic flow
- Rhythm – meter, rhythmic intervals, pulse
- Timbre
- Dynamics
- Texture
- Performance technique
- Instrument choice

These variables form infinite possibilities when interchanged with the various parts of the function.

The model created can be referred back to the *criteria for evaluation* to determine if it demonstrates the properties of a good model. In light of this comparison, it can be seen as an adequate model through the following observations.

There are a significant number of causal relations evident in this model of phase shifting; the relations evident in phase shifting as an acoustic phenomenon are temporal, and these can be seen in use through scale modelling (interpolation) by other composers.

In this model, the temporal relations have been transferred into the model data, which then allows for the extrapolation (use of this model as an *analogue*) of the relations into new contexts. This extrapolation still maintains the structure of the relationships evident in the phenomenon through similarity relations, as it uses a point-by-point system of correlation with

<table>
<thead>
<tr>
<th>Repetition #:</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>...</th>
<th>88</th>
</tr>
</thead>
<tbody>
<tr>
<td>$f$ of Sound A (Hz)</td>
<td>440</td>
<td>440</td>
<td>440</td>
<td>440</td>
<td>...</td>
<td></td>
</tr>
<tr>
<td>$f$ of Sound B (Hz)</td>
<td>440</td>
<td>445</td>
<td>450</td>
<td>455</td>
<td>...</td>
<td>880</td>
</tr>
</tbody>
</table>
data in the model (represented by characters X & δ) with the temporal properties of the phenomenon.

This model is also the simplest format, as if it is simplified any further it would lose the “truth value”. The oversimplification of any phenomenon can often be seen in mathematical models, where according to Black the “drastic simplifications demanded for success of the mathematical analysis entail a serious risk of confusing accuracy of the mathematics with strength of empirical verification in the original field” (Black, 1962, p. 225). This oversimplification is avoided in this case, although by reducing a complicated physical occurrence to a mathematical equation it could be seen as getting dangerously close.

Also, in order to maintain this truth value, rules have been formulated through the terminology applicable – the terminology is in relation to musical variables that can be used to substitute components of the function. While the list provided is not exhaustive, it outlines the type of variables that can be used so that projected variables stay aligned with the focus of the intent.
3.0 Resonance

Resonance is an acoustic phenomenon that can be observed in many contexts. This phenomenon exhibits the qualitative and quantitative properties of sound in its production, even within the interaction of two simple harmonic motions.

It has also been the focus of musical compositions through its use as a compositional device by various composers. This phenomenon exhibits the potential to be an effective model; it displays sufficient causal relations in its acoustic context to draw similarity relations with a new subject area. The causal relations in resonance, as opposed to phase shifting, are not necessarily temporal. Although the relations form a conditional operation (in which an event occurs as a condition of another event occurring, which presents a temporal, linear function) the effects are not temporal in nature when viewed in the phenomenon, the model, or the musical setting. Perhaps it would be more accurate to state that the temporal relation is so rapid in occurrence that it almost negates the effect of any rhythmic property.

Despite its capacity for being an adequate model for musical composition, the primary use of this phenomenon seems to be the utilisation of its natural state as a focal point for compositions. The composers under study in this research (which present a reasonable sampling of current compositional practices) don’t seem to have modelled the phenomenon in order to apply its properties to music in an interpolative or extrapolative way.

In this chapter resonance is defined in its various manifestations and a survey of works by – and compositional practices of – other composers is conducted. Two major compositions were created for this phenomenon, which both demonstrate how it can be modelled for musical composition. At the end of the chapter a model is created that can then be used in application to musical works in an extrapolative or interpolative fashion.

3.1 Resonance Defined

This discussion refers to the acoustic phenomenon “resonance” in a number of forms; resonance and resonant frequency, acoustic resonant frequency known as standing waves, and sympathetic resonance.

3.1.1 Resonant Frequency

“Resonant frequency” usually refers to the inherent capability of an acoustic source to vibrate at a particular frequency, which results in an amplification of that particular frequency when
the acoustic source emits or is subject to that frequency. Resonance is defined as “the state of a system in which an abnormally large vibration is produced in response to an external stimulus, occurring when the frequency of the stimulus is the same, or nearly the same, as the natural vibration frequency of the system” (Resonance, n.d.).

Greated & Campbell provide an explanation of where and how resonance can occur:

Systems such as strings fixed at both ends and air columns which have certain definite frequencies at which they will vibrate are known as resonant systems. Resonance occurs because disturbances are reflected, e.g. from the fixed end of a string, such that the wave at any position is actually the superposition of waves travelling in more than one direction. (Greated & Campbell, 1987, p. 17)

Scott Stark, in Live Sound Reinforcement: A Comprehensive Guide to P.A. and Music Reinforcement Systems and Technology (2000), also provides an explanation of the phenomenon and details a few examples of its occurrence. Stark describes that in the instance of a vibrating tuning fork’s stem being “placed against an object with a larger surface capable of vibrating at the same frequency more air is set into motion, resulting in a louder sound” (Stark, 2000, p. 25). This also means that the kinetic energy given to the tuning fork is transferred to a sound-source that is more efficient, which is an example of resonance.

Stark also uses the example that Campbell & Greated cite, which concerns resonance within a cavity or air column – in particular, he states that this phenomenon is directly responsible for the inherent pitch of a bottle when you blow across the mouth of it. Stark states that “…The air in a tube or cavity of given internal dimensions and mouth size has its own preferred mode(s) of vibration” (Stark, 2000, p. 25).

### 3.1.2 Resonant Frequency of an Acoustic Space – Standing Waves

Resonant frequency may be observed in particular acoustic spaces in which the architectural components of the space (shape, size, materials etc.) lend themselves to the amplification of a particular frequency. In the context of an acoustic space, the phenomena responsible for these resonant frequencies are called “standing waves.”

Standing waves can occur when a sound hits a reflective boundary surface, or parallel surfaces (such as parallel walls). These surfaces cause the sound waves to travel back on their own initial trajectories, which interfere with their own amplitude (Huber & Runstein, 2001, pp. 72-73). The reflected waves positively and negatively reinforce the original sound, creating points
of maximum pressure crests (positive reinforcement) and minimum pressure troughs (negative reinforcement), which in turn create a “stationary pattern in the air, consisting of zones of low pressure (called nodes), alternating with zones of high pressure (called antinodes)” (Davis & Jones, 1989, p. 55).

Standing waves are also known as “room modes”, and result in certain frequencies being sounded in a particular acoustic space at a higher amplitude than others (Rumsey & McCormick, 2006, p. 22). For example, particular acoustic spaces may amplify frequencies of 440Hz from sound sources within that space more than other frequencies in the sound spectrum, whereas another acoustic space may favour the frequency 880Hz. Acoustic spaces are often architecturally designed to avoid these occurrences of this particular phenomenon, as they result in unwanted acoustical amplification of particular note values where performers generally prefer a flat, unbiased sound in the production of their music. According to Huber & Runstein, the construction of parallel surfaces in a studio environment that create standing waves is the all-time winner of the “avoid this at all possible costs award!” (Huber & Runstein, 2001, p. 72).

3.1.3 Sympathetic Resonance

“Sympathetic resonance” is the occurrence in which a sound source emits a particular frequency in sympathy with another sound source that is emitting the same frequency.

Sympathetic resonance can easily be observed in the case of a piano having its sustain pedal depressed whilst playing another instrument into the cavity of the piano – this results in the other instrument’s notes being resonated by the piano strings, which vibrate in sympathy with the second sound source. Allen Strange, in Electronic Music – Systems, Techniques and Controls (1972), uses a similar example.

Any vibrating source is capable of the excitation or generation of additional frequencies. The standard classroom method of observing this is to silently depress middle C4 on the piano and strike C3 an octave below. One will hear middle C ring out as it has been forced into vibration by the lower octave C3. This type of forced vibration is called sympathetic vibration and demonstrates that one vibrating system has the potential of generating other vibrations. (Strange, 1972, p. 14)

Helmholtz, the German scientist, held the view that sympathetic resonance could be seen in its most rudimental form in the function of the human hearing system. Helmholtz stated that the
regular vibrations that are emitted by any sound source are then transmitted to the eardrum through a shared “atmospheric vibration”, as the sounds heard are then sympathetically vibrated on the tympanum in the ear; “He defined this relationship between the object being heard and the apparatus of hearing as being one of ‘sympathetic vibration’ or resonance, which became the basis for his theory of hearing” (Blamey, 2008, p. 40).

Following Helmholtz’s perception of resonance as the function of the human auditory system, Blamey shares a similar perspective on resonance. His definition includes the function of the ear as one aspect of resonance, and also creates a more general picture of resonance being a phenomenon that occurs within and between structures; these structures can be “rooms, throats, ears walls or musical instruments.” Blamey also relates the term resonance back to its Latin origin, which means “to sound again”, and states that to utilise this as a phenomenon means to have musical sounds “sounded again” (Blamey, 2008, p. 41).

3.1.3 Resonance of Acoustic Instruments

Resonance is a phenomenon that is found throughout many facets of sound production and music. In designing the room of a recording studio, the layout and materials used are chosen in a way to ensure that standing waves are avoided. Resonance also plays a part in the construction of acoustic musical instruments. For example, when making stringed instruments the resonance of the instrument’s body plays a big part in its creation. Stark discusses these occurrences of resonance, stating that they appear everywhere – including in the vibrating physical bodies of musical instruments:

...The principle of resonance continually comes into play in sound reinforcement and in the behaviour of sound in general, occurring in rooms, speakers and speaker enclosures, acoustic horns, musical instruments and even electrical circuits. In fact, everything in the entire universe has one or more resonant frequencies, though not necessarily within the audio spectrum. (Stark, 2000 p.23-4)

The physical bodies that are used to produce the sound of an instrument are specifically crafted with its resonance in mind. When the physical bodies of the instruments vibrate they emphasise certain frequencies and attenuate others – this is called the instrument’s “formants” (Fineberg, 2000b, p. 87). An instrument’s formants are what allow the identification of the timbre, even in situations where the amplitude of the partials vary, such as in different registers or at different volumes.
3.2 Composers Representing Resonance in Music

3.2.1 Alvin Lucier

Alvin Lucier, the American experimental music composer, began explorations into sound and its many facets. His experimental compositions utilised a number of different acoustic phenomena as their guiding force including, in particular, resonance. Peter Blamey’s study on the use of acoustic phenomena is primarily focused on only two composers, one of which is Alvin Lucier. Blamey states that “...What these two artists share [La Monte Young & Alvin Lucier] is a practice that involves the realisation and elucidation of specific acoustic phenomena” (Blamey, 2008, p. 2).

Lucier’s works sit somewhere in between a science experiment and an avant-garde musical composition, as many of his works involve the exploration of sound, not just for the listener, but also for the performer of the work. In reference to his use of scientific experiments, Lucier states: “Scientific experiments have often given me ideas for pieces; sometimes I do little more than frame them in an artistic context” (Lucier, 1998, p. 8). What this statement reveals is a point that is raised in this discussion: Lucier often presents an illustrative utilisation of an acoustic phenomenon as a compositional focus in his works, rather than using instrumental techniques to represent them. There are a few exceptions to this, as some of his works do represent the phenomena in addition to utilising them, but also the idea of simply presenting a sonic phenomenon in its natural state could be justified through the mentality expressed by other composers such as Tristan Murail (1947-), who states that “posing sonic material, simply offering it to the listener’s hearing, is the primordial sonic gesture” (Murail, 2005b, p. 174).

His compositions often involve representing a variety of acoustic phenomena, as well as the ever-present sounds of life itself. Blamey’s study (2008) on Alvin Lucier and La Monte Young elucidates these composers’ efforts to make acoustic phenomena the focus of musical composition.

What distinguishes the work of Young and Lucier is the aim to utilise only one or two types of acoustic phenomena as the subject of a given work, and that the generation of specific kinds of phenomena, and the subsequent audition of those specific effects by an audience, comprise the very subject matter of a composition. These phenomena are often the result of the propagation of sound in space: acoustical beats, echoes, standing waves, diffraction effects and combination tones. As such they can be
categorised as basic or simple acoustic phenomena, which in turn are the foundation for an array of more complex acoustic interactions. (Blamey, 2008, p. 2)

Many of Lucier’s works seek to represent acoustic phenomena in musical form. Morgan describes Lucier’s *Hartford Memory Space* (1969), in which Lucier instructs the performers to go out into an urban, rural, hostile or benign environment and record the sound of that particular environment as exactly as possible. Lucier suggests some methods of achieving this, such as using sketches, tape recorders or just commit the soundscape to memory. The performance of the piece involves the players returning to the concert hall or other place of performance and duplicating (“as precisely as possible”) the soundscape that the performers have committed to memory using their own instrument or voice. The performers are instructed not to colour this representation with their own personal tastes, ideas, additions or to include improvisations (Morgan, 1991 p.453).

In a similar vein, Lucier’s *Shelter* (1967) also seeks to internalise the sounds of the outside world within the concert space. The audience for this piece are seated in a dark and quiet room, in which sensors are attached to the inside walls and fed through an amplifier and loudspeaker system. This allows the audience to hear the sounds – natural, man-made and planned performances – that are occurring outside the room. This piece effectively uses the room’s walls as filters for external sounds, transforming them with the structure of the new internal environment.

One of Lucier’s more famous pieces is *I am Sitting in a Room* (1969), in which he explores the potential use of resonant frequencies as the subject matter for a composition. The piece involves a recorded sample of human speech (in the original recording, Alvin Lucier’s voice narrating the steps taking place to construct the piece), which is played through speakers in a room. The sound is then re-recorded and then re-projected into the room in real-time, multiple times over and progressively added in. As each new sound is projected into the room, the resonant frequencies of the room are emphasised by the reinforcement of the multiple recordings whilst the voice is progressively “disintegrated” (this process is referred to as interference, or phase cancellation – which is first discussed in this research with reference to phase shifting, while here they are used in reference to the primary focus which is resonance).

Eventually, what remain are the reflections of the voice interacting with the shape of the room to bring out their inherent, or resonant, frequencies. This results in a wash of ethereal sound, as the resonant frequencies seem to manifest in pure sine tones that fluctuate independently
and interact in a constantly shifting process. The sound achieved seems to this researcher akin to a “soft pad” tone on a synthesiser, but intriguingly the sound source is actually the refractions of a voice.

Nyman draws a thread between Reich and Lucier through his observation that both composers use the idea of a “gradual musical process.” Nyman cites Lucier’s *Quasimodo the Great Lover* (1970) as using this concept, as well as *I am Sitting in a Room* (Nyman, 1999 p.107).

Lucier also created a piece that used sympathetic resonance – *Music for Piano with One or More Snare Drums* (1990), written for the pianist Hildegard Kleeb. In this work, the pianist performs a series of written notes in a free and overlapping pattern in a space in which several snare drums are positioned (Sholnick, 2000). The various snare drums resonate in sympathy to particular notes played by the piano – Sholnick writes: “The drums respond in various ways depending on the pitch of the piano tones, the resonant regions of the drums and their geographical locations in space” (Sholnick, 2000).

Among Lucier’s numerous other works that feature the acoustic phenomenon “resonance” is his piece titled *Music for Piano with Amplified Sonorous Vessels* (1991). This work utilises the phenomenon without relying heavily upon electric equipment (besides amplifying the vessels.) In this work, several small containers are placed inside and near a grand piano (but not touching the strings). The vessels include wine glasses, sea shells, clay pots and bamboo cups. The vessels are amplified by placing microphones into them, and then sent through an amplification system.

The chords and singular notes that are played on the piano resonate within the vessels, creating an eerie, spaced-out sound. The different shapes, sizes and materials of the vessels mean that the notes played on the piano don’t resonate the same way in each vessel, which is part of Lucier’s plan; where the fundamental tones played on the piano interact with the harmonically related partials and overtones, interference patterns are created which then beat at different speeds and create a variety of rhythmic patterns.

This manifestation of the acoustic phenomenon resonance was first used by the Greek philosopher and scientist Vitruvius in his concept of resonating vases. Vitruvius’ contribution to this area is discussed in the introduction of this research. Vitruvius’ resonating vases were the outworking of his idea on creating extra resonance within the theatre setting, for the benefit of the audience and to assist the performers with pitch identification.
Another piece of Lucier’s, *Nothing is Real* (1990), is an interesting adaptation of The Beatles’ hit *Strawberry Fields Forever* (1967). Aki Takahashi, a Japanese pianist, asked Lucier to write an arrangement of a Beatles song for her, and this song was chosen. Lucier’s arrangement is written for piano, amplified teapot, a recording device and a small sound system.

The actual melodic and harmonic content simply consists of fragments of the original song’s melody, played in different keys, registers, and at differing tempi; each phrase is also sustained, to form a note cluster at the conclusion of each. While this in itself creates quite a haunting atmosphere (particularly as the melody, whilst not having a voice to sing the lyrics, does give the connotations of the lyrics that the listener may be previously aware of in their new eerie context), Lucier then instructs the performer to record the performance and then play back the ensuing recording. The recording is directed to be played back through a teapot via a small speaker, which is then amplified with a microphone. The performer then raises and lowers the lid of the teapot at various intervals.

The teapot is utilised to allow different harmonics and resonant frequencies to be heard, which colours the piece in an entirely different way. Of course, the shape of the teapot used will determine the number of harmonics and quality of the resonance – one particular performer of the piece (Steven Speciale, 2010) provides useful insight into which type of teapot gives the best frequency response and allows the speaker to fit inside!

Here again a musical work by Lucier can be seen as harking back to the early proponent of the technique, Vitruvius. It can be seen from this repeated connection between a philosopher and architect of the 1st Century BC with an experimental music composer of the 20th Century that the concept of representing acoustic phenomena pre-dates any current work in the field, particularly in regards to the use of resonance. A similar technique has also been used more recently by John Beck, who wrote *Three Episodes for Timpani* (1980), in which a piano is placed behind the timpani with its sustain pedal depressed, so that the notes struck on the timpani resonate the piano strings in sympathy.

Amongst others of Lucier’s pieces that may feature resonance, the following list contains those that feature the phenomenon strongly enough for “resonance” to make the piece’s title:

- *Opera with Objects* (1997) for performer(s) with resonant objects, composed for Sam Ashley
- *I Remember* (1997) for chorus with resonant objects
Resonance held quite a prominent place in Lucier’s compositions, and characteristic of his work is the fact that he utilises the phenomenon in more of a demonstrative fashion. His works don’t necessarily represent the phenomenon as much as they utilise them. Lucier was very open about this fact, and that his pieces are often driven by a “scientific experiment” or “demonstrative” ethic of presentation. Lucier is upfront about the influence of scientific experiments and demonstrations of acoustical phenomena, claiming that he does “little more than frame them in an artistic context” (Lucier, 1998, p. 8).

This is not to say that his works don’t fit within the framework of this research, however; representing acoustic phenomena is on par with using them as a focus within a composition, as it shows the composer is still valuing sound as a source of interest within musical composition.

3.2.2 André Gonçalves

André Gonçalves is a contemporary Australian composer who has created a unique work involving the use of sympathetic resonance, titled Resonant Objects (2005). Gonçalves’ inspiration for the work stemmed from the concept of “bringing together sound and space through sympathetic vibrations, in which sound is used as a medium to excite space to be heard, in order to render audible it’s natural resonances - sound multiplying in and by space” (Gonçalves, 2009).

In this installation composition, six objects are suspended from the ceiling of the space in which it is to be performed – each object consists of a white globe that contains a microphone, speaker and electrical lamp. Each of the speakers are connected to a computer that triggers six different frequencies (one frequency for each of the speakers) – the frequencies used are the same as the natural frequencies of the globes, which then makes the objects resonate in sympathy.
The next chain in the process of this composition involves the sympathetic resonances being picked up by the microphones and sent to a “6-envelope follower system” that converts the amplitude of the audio signals to MIDI signals. These MIDI signals are then used to control dimmers for the lights, which means the light intensity of each globe is related to the sound’s amplitude (Gonçalves, 2009).

This piece provides an exploration of the acoustic phenomenon of resonance, and highlights its existence through invoking the physical properties of objects and enhancing them through the addition of visual components (the light dimmers.) The effect of the installation is described by Gonçalves himself in his abstract on the composition:

> The combination of these factors created a soundscape of resonant frequencies triggered by sine waves moving in tidal motion, a constant flow of never repeating patterns. With its very slow progression it mimicked the constant forward and backward motion of the sea. Likewise I always saw light as the sun’s reflection in the water, shining or dimming out according to the wave's direction. (Gonçalves, 2009)

This piece demonstrates how an acoustic phenomenon such as resonance can be explored through a soundscape style composition. It focuses more on the actual acoustic demonstration of the phenomenon, rather than a representation of it through music, much in the same approach as Alvin Lucier’s works. However, it does demonstrate the interest this phenomenon can create and the room for exploration in its involvement in musical compositions.

### 3.2.3 Rob Godman

Rob Godman authored the text on Vetruvian Resonating Vases, the details of which are included in the introduction of this research. Following the inspiration of Vitruvius, Godman composed a piece titled *Halo*, which was premiered in 2006. The work is written for piano, and uses an additive synthesiser to emulate the Vitruvian resonating vases – this is achieved by using a microphone that detects transient dynamic changes through the patch [bonk~] in MaxMSP, which then triggers the relevant pitches of the synthesiser (Godman, 2008).

The following figure is an illustration of the MaxMSP patch utilised by Godman in this piece.
Figure 19 – Godman’s “Vitruvian Harmonics Generator”.

This patch pictured in figure 19 registers the frequency of incoming audio signals and outputs a corresponding frequency through a synthesiser.

This piece is a good example of how composers can use technology to electronically reproduce a naturally occurring acoustic phenomenon, potentially to incorporate it into their musical compositions. Through this method Rob Godman effectively created a ‘virtual Vitruvian vase.’ This concept is not necessarily a type of musical modelling or representation of the phenomenon. Whilst the patch recreates the same effects as resonance through a model of its properties, the recreation is not a musical one but is instead an electronic sound manipulation that utilises a similar effect as the original phenomenon in a new context.
3.3 Modelling Resonance for Musical Compositions

Development of Concepts and Preliminary Compositions in Resonance

The experimentation with the concept of resonance in this research began with the exploration of resonant frequencies and room acoustics. As interpolating the data from any acoustic phenomenon is easier than extrapolating it, there is a larger number of compositions to reflect that – Resonance Concepts 1-2 and Resonance Compositions 1-2 are all utilising data that has been interpolated from the acoustic phenomenon ‘resonance’.

Resonance Concepts 1-2 both share a similarity to the work done by Alvin Lucier, Andre Gonçalves and Rob Godman. These composers each worked with resonance in their compositions by utilising them in their acoustical context within a musical piece. For example, Lucier’s Nothing Is Real involves using the resonance of a teapot to enhance the performance of his piece on piano. The concepts outlined here are of the same vein, as they utilise resonance in a musical composition.

Resonance compositions 1-2 however demonstrate a departure from this technique. In these works resonance is not utilised, but is represented through instrumental and scoring techniques. Both of these areas are involved within the confines of this research and have intriguing implications; however, the compositions that represent the acoustic phenomenon are the true focus of this research.

The first concept explored for a composition was to interpolate the data gathered from analysing the resonant frequencies of a room.

3.3.1 Resonance Concept 1 – Using the Resonant Frequency of a Room

The concept within this piece is to create a musical composition that is specifically tailored to a particular room’s acoustics. In order to do this, an acoustic space would be analysed (sonically) to determine its resonant frequencies. The resonant frequencies (translated as notes) would be used to highlight particular degrees of a scale on which the composition is based.

For example, if a room is found to have a resonant frequency value of 440Hz, which is the note A, the composition could potentially be written in the key of A to emphasise the tonic note of the scale.

This acoustic consideration would add another level of depth to the normal harmonic pull of a piece of music. For instance, a V-I cadence (perfect cadence) has an aural pull and a sense of
resolution due to the nature and evolution of western music and its listeners. If the tonic note, which is the “resolution,” is then resonated in the acoustic space in which it is being played, would it not then have an extra level of “pull” and “resolution”?

If, for example, this same piece of music modulated to another key it would normally begin a fresh tonal centre. If the original tonic (A) is still constantly being resonated, however, the modulation is kept relative to the initial key of the piece. Even if the tonal centre is different, the original tonal centre of “A” is still present, which could potentially change the perspective of the modulation to the listener.

Another example: keeping with the space that resonates the note A, if a piece were composed in the key of Eb then the scale degree being resonated would be the diminished fifth. This interval is often regarded as having an off-putting sound (being labelled the devil’s chord tends to stigmatise things a bit), so the entire piece would be “tainted” with the sound of this particular note resonating throughout.

This idea can be explored in many similar ways. Each degree of a scale has different aural connotations in relation to its tonic. For example, here is a list of scale degrees and their possible effect when amplified in a piece by resonance:

- Fifth – “floating,” “empty,” “hollow”
- Third – sweet or sad, depending on whether major or minor
- Major Seventh – slightly dissonant yet sweet
- Sixth, fourth, second – unresolved, pulling towards resolution notes

It seems that this type of room resonance composition has a parallel in “drone” compositions. Music played on a bagpipe, for example, exhibits the same considerations as the afore-mentioned compositions; the existence of the constant drone underneath the melody gives it a firm grounding in its tonal centre. This can be either very strengthening or very off-putting; if the tonal centre matches the key of the melody then it aids in the melody’s resolution, whereas if the tonal centre does not match the tonic of the melody then the whole melody is flavoured by this different tonal centre.

This same example can be seen in every day musical performance. When you next do the vacuuming, consider how the drone of the vacuum cleaner (with its particular tonal pull of whatever note it emits) effects the melody you are singing; if you are conscious of it, you can sing in the same key as the device and thus create a stability to your melody or, alternatively,
you can choose to sing in a different key and observe the effect the drone of the vacuum cleaner has on your melody’s tonal pull. When you are next in the bathroom humming to yourself, listen to the drone of the ceiling fan as you do and observe these same effects. The connection between resonance and drone music is discussed in more detail at the end of this chapter.

**Experiment – Resonance Concept 1**

In order to perceive the outcomes of this concept, an experiment is necessary. The following are the steps involved in the experiment:

1. **Analyse an acoustic space for its resonant frequencies.** After choosing an acoustic space (preferably one with some natural reverberation – a studio environment is not ideal), record an instrument or oscillator in a flat sound environment (studio). Then play the same instrument or oscillator in the chosen acoustic space and record. Play the original sound recording and the new together, with the phase reversed on one in order to cancel out the original sound, leaving just the additional frequencies resulting from the room’s acoustic response. Analyse this audio using a spectrum analyser (such as the one found in the Sound Forge audio software) to define which frequencies are being resonated.

2. **Compose a short piece, using the room’s primary resonant frequency as the key of the piece.** For example, if the room’s primary resonance (as there may be multiple harmonics present in the room) is ‘A’ (440Hz) then the piece will be composed in A.

3. **Perform the piece in the acoustic space.** This will provide verification that the resonance of the chosen note is present and an evaluation of the piece’s effectiveness. This may involve either a recording of the performance, or the presence of the audience, or simply a self-evaluation by the performer.

4. **Transpose the piece into another key, using the resonant frequency as a related degree of the scale.** For example, transpose into the key of D, in which A (440Hz) is the fifth degree of the scale.

5. **Repeat step 3.**

6. **Repeat step 4 with a different key, then step 3.**

7. **Repeat until sufficient data is produced.**

**Results – Resonance Concept 1**

This experiment was conducted using a bathroom as the resonant acoustic space, and a studio environment nearby to produce the ‘controlled’ variable. The instrument used to produce the notes was an alto saxophone.
According to the proposed steps listed above, the results were as follows:

1. **Analysis of an acoustic space for its resonant frequencies.**

An alto saxophone was used to play a chromatic scale of a 2 ½ octave range (full range of instrument – B♭3 to G6, which is G3 to E6 in concert pitch). *As an alto saxophone is an E♭ instrument, all notes sound three semitones higher than concert pitch instruments – this is reflected in the notes discussed in each example and diagram, as they will be shown in concert pitch.* Each rendition of the scale was done to a click track, with a note held for four counts followed by a rest for four counts. A large condenser microphone was placed in front of the instrument to record the scale into Pro Tools (music editing software). This was first done in a studio environment, using soft surfaces (such as a mattress and blankets) to absorb unwanted noise. This process was then repeated, using a bathroom as the chosen acoustic space.

The analysis was done in a more simple fashion than outlined above – upon visually comparing the two sets of waveforms, it was obvious which frequencies were more dominant in the bathroom. The diagram below shows the two waveforms aligned vertically – each “lump” represents a note being played on the saxophone, beginning with G3 on the left and ascending through each note of the chromatic scale until reaching E6 on the right. The top (blue) block shows the resultant waveforms from the bathroom, and the bottom (green) are the waveforms from the studio.

![Figure 20 – Comparison of waveforms recorded in a bathroom and studio.](image)

The dominant frequencies that were present in the bathroom are labelled on the diagram with the red lines – these notes were obviously louder in the bathroom due to the corresponding resonant frequencies of the room being stimulated. This can be seen in the diagram by the difference in size of the waveforms.
The results of this component of the experiment were as follows: the bathroom (acoustic space) has inherent resonant frequencies at the notes C#/Db4 & 5, G4 and C6. The following chart has been constructed to show the full list of notes and their frequencies in their different registers (0-8). The frequencies that were shown to be the resonant frequencies of the bathroom have been highlighted (in black).

Table 4 - List of Frequencies ($f$) & Corresponding Notes, Illustrating Frequencies Present in Room Analysis

<table>
<thead>
<tr>
<th>Register:</th>
<th>0</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td>f (Hz)</td>
<td></td>
</tr>
<tr>
<td>C</td>
<td>16.35</td>
<td>32.70</td>
<td>65.41</td>
<td>130.81</td>
<td>261.63</td>
<td>523.25</td>
<td>1,046.50</td>
<td>2,093.00</td>
<td>4,186.01</td>
</tr>
<tr>
<td>C#/Db</td>
<td>17.32</td>
<td>34.65</td>
<td>69.30</td>
<td>138.59</td>
<td>277.18</td>
<td>554.37</td>
<td>1,108.73</td>
<td>2,217.46</td>
<td>4,434.92</td>
</tr>
<tr>
<td>D</td>
<td>18.35</td>
<td>36.71</td>
<td>73.42</td>
<td>146.83</td>
<td>293.66</td>
<td>587.33</td>
<td>1,174.66</td>
<td>2,349.32</td>
<td>4,698.64</td>
</tr>
<tr>
<td>D#/Eb</td>
<td>19.45</td>
<td>38.89</td>
<td>77.78</td>
<td>155.56</td>
<td>311.13</td>
<td>622.25</td>
<td>1,244.51</td>
<td>2,489.02</td>
<td>4,978.03</td>
</tr>
<tr>
<td>E</td>
<td>20.60</td>
<td>41.20</td>
<td>82.41</td>
<td>164.83</td>
<td>329.63</td>
<td>659.26</td>
<td>1,318.51</td>
<td>2,637.02</td>
<td>5274.08</td>
</tr>
<tr>
<td>F</td>
<td>21.83</td>
<td>43.65</td>
<td>87.31</td>
<td>174.61</td>
<td>349.23</td>
<td>698.46</td>
<td>1,396.91</td>
<td>2,793.83</td>
<td>5587.66</td>
</tr>
<tr>
<td>F#/Gb</td>
<td>23.12</td>
<td>46.25</td>
<td>92.50</td>
<td>185.00</td>
<td>369.99</td>
<td>739.99</td>
<td>1,479.98</td>
<td>2,959.96</td>
<td>5919.92</td>
</tr>
<tr>
<td>G</td>
<td>24.50</td>
<td>49.00</td>
<td>98.00</td>
<td>196.00</td>
<td>392.00</td>
<td>783.99</td>
<td>1,567.98</td>
<td>3,135.96</td>
<td>6271.92</td>
</tr>
<tr>
<td>G#/Ab</td>
<td>25.96</td>
<td>51.91</td>
<td>103.83</td>
<td>207.65</td>
<td>415.30</td>
<td>830.61</td>
<td>1,661.22</td>
<td>3,322.44</td>
<td>6644.88</td>
</tr>
<tr>
<td>A</td>
<td>27.50</td>
<td>55.00</td>
<td>110.00</td>
<td>220.00</td>
<td>440.00</td>
<td>880.00</td>
<td>1,760.00</td>
<td>3,520.00</td>
<td>7040.00</td>
</tr>
<tr>
<td>A#/Bb</td>
<td>29.14</td>
<td>58.27</td>
<td>116.54</td>
<td>233.08</td>
<td>466.16</td>
<td>932.33</td>
<td>1,864.66</td>
<td>3,729.31</td>
<td>7458.62</td>
</tr>
<tr>
<td>B</td>
<td>30.87</td>
<td>61.74</td>
<td>123.47</td>
<td>246.94</td>
<td>493.88</td>
<td>987.77</td>
<td>1,975.53</td>
<td>3,951.07</td>
<td>7902.14</td>
</tr>
</tbody>
</table>

2. Composition of a short piece, using the room’s primary resonant frequency as the key of the piece

From the written results above, it was discovered that the notes C#, G and C resonated strongest within the acoustic space. As there were two C# notes in different registers that resonated, as opposed to a single G and C, it seems more pertinent to nominate C# as the primary resonant frequency.

A very short piece was composed for the alto saxophone, to be performed for the acoustic space. The piece was set in C# minor, as C# was the primary resonant note of the space. The melody uses the resonant notes in more abundance; primarily C#, but also C and G.
3. Performance of the piece in the acoustic space

The piece was performed in the acoustic space, and in this particular instance the decision was made to let the evaluation take the form of the performer’s perspective.

As predicted, the primary resonant frequencies of the space caused the corresponding notes of the piece to sound more reinforced and stronger. The C# in both the lower and middle registers felt stronger, so that moving to this note from another caused a strong tonal pull, and moving away from it to another note felt less stable. This was similar with the G, although not as noticeable in the top register. The C was also stronger, although not as much in the lower registers (the actual frequency that was recorded as resonating stronger was in the top register, C6).

To highlight these effects, below is the piece again but with the resonant notes circled. The different colours denote the strength of the note’s reinforcement; red is strongest, followed by green then blue. The tonic note of the scale is also indicated by a “T”.

Figure 21 – Piece composed for Resonance Concept 1 (C# minor).
Figure 22 – Piece composed for Resonance Concept 1 (with highlights).

By illustrating the notes of the piece in this way, it can be observed that the tonic notes of the scale are consistently reinforced by the resonant frequencies of the room. The “leading note” (a note that steps by a semitone; in this case, and often, up to the tonic) C/B♯ resonates in secondary prominence to the tonic, as does the C♯ (tonic) in the highest register. The flattened fifth degree of the scale, G, is reinforced the least by the acoustics of the room.

The effect of this reinforcement is that the piece has a very strong tonal pull toward the tonic of the scale, which creates a “solid” feel. This becomes only slightly unbalanced by the secondary reinforcement of the leading note and flattened fifth of the scale. Traditionally, the flattened fifth, (also known as the tri-tone, augmented fourth or diminished fifth) was commonly known as *diabolus in musica*, “The Devil in Music” (Cherubim, 2004), as it was deemed dissonant when compared with the usual modes and scales. The reinforcement of this note in this piece then colours it with a slight dissonance; while it’s not the strongest reinforcement of the tones in the scale, even the minimal reinforcement of the B♭ allows its musical effect to be enhanced.

4. **Transposition of the piece into another key, using the primary resonant frequency as a related degree of the scale**

To change the effects of the acoustic space on the composition, it was then transposed to a key that relates to (uses) the primary resonant frequency (C♯). It was also decided that the inclusion of the other resonant frequencies (G & C) in the relationship to the new key would be preferable, to increase the number of effects.
The key chosen was F harmonic minor. The following (shown in Table 5) are the notes of the scale, with the resonant frequencies of the acoustic space highlighted (in bold text) on the corresponding scale degrees.

Table 5 - Scale Degrees & Note Names of F Harm. Min. Scale Corresponding to Resonant Frequencies of Acoustic Space

<table>
<thead>
<tr>
<th>Scale Degree</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note Name</td>
<td>F</td>
<td>G</td>
<td>A&lt;sup&gt;♭&lt;/sup&gt;</td>
<td>Bb</td>
<td>C</td>
<td>D&lt;sup&gt;♭&lt;/sup&gt;</td>
<td>E</td>
</tr>
<tr>
<td>Description</td>
<td>Tonic</td>
<td>Supertonic</td>
<td>Minor Third (Mediant)</td>
<td>Subdominant</td>
<td>Dominant</td>
<td>Submediant</td>
<td>Leading Note</td>
</tr>
</tbody>
</table>

The following figure shows the piece transposed into the new key signature of F harmonic minor.

![Figure 23 – Piece for Resonance Concept 1 (F harm. min).](image)

5. Repetition of step 3 (Performance of piece with key modulation in acoustic space)

Following the performance of the piece in the new key of F minor, similar results to the first performance were observed but with the different scale degrees being resonated.

As the tonic of the piece was not being resonated, it created a marked difference in the performance; it didn’t feel as stable as the first performance. The second degree of the scale (G) being resonated, even at a lower level compared to the other resonant frequencies, had the effect of destabilising the tonal centre of the piece, as the tonic of the piece was no longer the strongest sound. In addition, having the fifth degree of the scale (C) being sonically reinforced created more of a “hollow” sound. Unfortunately, the effects of the primary
resonant frequency could not be established as this particular composition didn’t involve the note C♯, the sixth degree of the F harmonic minor scale.

To demonstrate the effects achieved, the piece is shown again in this figure (24) with the scale degrees highlighted (in descending order of strength, from green to blue) that correspond to the acoustic space’s resonant frequencies.

Figure 24 – Piece composed for Resonance Concept 1 (F harm. min, with highlights).

In this figure, it can be seen that the tonic (T) never aligns with a resonant frequency, which takes away from the stability of the piece when compared with the first performance.

**Conclusions – Resonance Concept 1**

From these results it can be seen that this concept generates some interesting possibilities. The use of an acoustic space to influence a composition creates a different spin on an existing connection between architecture and music. The designs of architecture have long been influenced by the music that was to be performed within the space – acousticians make a career out of designing a building’s interior shape to produce the desired acoustic effect. The interesting thing about this idea is that the process here is reversed – the music is designed with the acoustic space in mind in which the music is to be performed. The resonant frequencies of a room can successfully be utilised in dictating compositional features within a musical work. The composer can decide in what way the piece is to be enhanced by the space by manipulating the notes within the work – choosing a particular key or scale will create different acoustic effects, and the use of the notes within that scale can also dictate the involvement of the room’s resonant frequencies on the piece.
Compositions based on the resonant frequencies of a space can also expect a different result when using key modulations within the piece. Normally a modulation resets the tonal centre of the piece, but in the event of a musical work’s tonic being resonated by the acoustic space, a modulation to another key won’t have the same effect - the original key is being resonated any time the original tonic note is used in the new key.

This use of resonant frequencies would be classed as an interpolation of the data. In fact, this type of use of acoustic phenomena is more of a demonstration of its behaviour and the utilisation of that behaviour to enhance the music. This kind of representation has more links with Alvin Lucier’s works, as he too spends more time composing pieces that either demonstrate or utilise acoustic phenomena to enhance his musical compositions.

The over-all implications of the conclusions from these findings are discussed in more detail in Chapter 10.3.3.

3.3.2 Resonance Concept 2 - Using Resonators

Another concept for a piece using resonance is to create your own desired resonance using resonator tubes. This is different to simple reverberation, as only the desired frequencies would be resonated.

This can be achieved by using something as simple as pieces of poly pipe as resonators. The pipes can be cut to lengths that will cause particular notes to resonate over the others, following which a piece of music can be composed with these specific resonances in mind. A performance of this piece would involve playing into the resonators (much like a marimba).

Experiment – Resonance Concept 2

The following are the steps involved in the experiment designed to test the viability of this concept:

1. **Cut poly pipe into strategic lengths.** Cut into metre lengths of 3, 2, 1.5, 1, 0.75 and 0.5.
2. **Seal one end of each cylinder.**
3. **Gauge the cylinders’ resonance.** This can be achieved through either (i) striking the sealed end to produce sound, giving an approximate pitch through aural analysis, or (ii) direct either white noise or a sliding oscillator in one end, record the ensuing audio and analyse audio using spectrum analysis.
4. **Compose a piece using the resonant frequencies of the cylinders as key notes of the composition.**
5. **Perform piece on chosen instrument, playing into resonator cylinders.**

**Results – Resonance Concept 2**

This experiment was conducted using the previously outlined steps. According to the proposed steps listed above, the actions were as follows:

1. **Several pieces of poly pipe were purchased and cut into strategic lengths**

   As outlined above, the poly pipe lengths chosen were 3m, 2m, 1.5m, 1m, 75cm and 50cm.

2. **The ends of the poly pipe were sealed**

3. **The sealed ends were struck to gauge their resonant frequency**

   This rough method of evaluating the pipes’ resonant frequency turned out to be quite successful, which precluded the need to follow step 2(ii), which involved the analysis of the resonant frequency through the spectral analysis of white noise or oscillators projected down the cylinders.

   The cylinders were struck and compared with the notes of a (tuned) piano. In addition to this, a chromatic scale was played into the pipes with an alto saxophone which elicited the additional harmonics of the tubes. The results are demonstrated in the following table.

**Table 6 - Relationship of Pipe Length & Frequency Produced**

<table>
<thead>
<tr>
<th>Pipe Length (m)</th>
<th>3</th>
<th>2</th>
<th>1.5</th>
<th>1.0</th>
<th>0.75</th>
<th>0.5</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary Note Produced</td>
<td>A₂</td>
<td>E₃</td>
<td>A₃</td>
<td>E₄</td>
<td>A₄</td>
<td>E₅</td>
</tr>
<tr>
<td>Frequency</td>
<td>110</td>
<td>164.81</td>
<td>220</td>
<td>329.63</td>
<td>440</td>
<td>659.26</td>
</tr>
<tr>
<td>Secondary Overtone</td>
<td>E₃</td>
<td>B₃</td>
<td>E₄</td>
<td>B₄</td>
<td>E₅</td>
<td>B₅</td>
</tr>
</tbody>
</table>

By keeping the lengths of the pipes easily dividable integers it also then maintained a relationship between the frequencies produced. The longest pipe, 3m, produced the note A₂, with a secondary overtone of the fifth interval above that – an E₃. The next pipe, 2m, produced an E₃, which is a fifth higher than the 3m pipe. This pipe also produced an overtone of a fifth above, which is the note B₃. This cycle of fifths continues right up to the smallest pipe, the 50cm length, which produced a high E₅ with a fifth above that – B₅.

Conducting this simple experiment allows the principle of resonance to be seen, but also the interaction of harmonics and wavelength in the production of sound. The principle of the harmonic series, which is discussed further in a subsequent chapter, can be observed in this
procedure. The equal divisions of a tube create different wave patterns in the movement of air inside them that are musically related. As is discussed in the work of Henry Cowell, the relationship of musical intervals can be expressed in terms of their ratios. For example, the interval of a fourth can be expressed as the ratio 3:2 and the interval of a fourth can be expressed as 4:3. The interval of an octave is represented by the ratio 2:1, as the pitch of the fundamental tone is doubled to produce the same note an octave higher.

4. **Compose a piece using the resonant frequencies of the cylinders as key notes of the composition**

To compose a brief piece to use an example of how resonance can be incorporated into a musical work, only three of the six poly-pipe lengths were used: the .075m, 1m and 1.5m. These cylinders produced the notes E₄, A₄ and E₅, with the secondary overtones of B₄, E₅ and B₅. The piece was written in the key of E minor, which meant that the tonic note of the scale as well as the fourth and the fifth degrees would potentially resonate within the cylinders.

The musical material subjected to the experimentation with resonators is shown below (in concert pitch):

![Figure 25 – Piece composed for Resonance Concept 2 (E min.).](image)

5. **Perform piece on chosen instrument, playing into resonator cylinders.**

The piece was written for sax and was performed twice, recorded both times. The first playing was done with just the saxophone, and the second recording involved playing into the resonators. The waveforms of the two audio recordings can be compared in order to show the difference in amplitudes of the two versions on the notes that are resonated by the cylinders.
In the following figure, the first four bars of the piece are shown in both waveform and notation. The size difference between the reinforced notes is clearly observable, and these notes are highlighted in the corresponding music with their scale degree number.

![Figure 26 – Comparison of waveform with notation of composition.](image)

In this figure, the green waveform on the bottom is the result of the recording with the resonators and the blue waveform on top is from the recording with just the sax. The difference in amplitude of the waveforms can be observed to be correspondent to the highlighted notes, which happen to be the tonic (T) and fifth (5th) degrees of the scale.

This has ramifications that affect the perception of a piece such as this that is performed with the incorporation of selected resonators.

**Conclusions – Resonance Concept 2**

This type of compositional process has strong ties with the first Resonance Concept, and the results from the research of this second concept create further implications for the first.

On its own, however, this concept allows for a similar idea to the first to be achieved without changing the tonality of the piece or letting the acoustic considerations dictate the compositional process. As these resonator tubes are quite straightforward to construct, a composition can be made with as many chosen “resonant frequencies” in mind as necessary – the resonators can then be constructed according to which resonant frequencies have been chosen.

For example, if a composer wishes to compose a piece in which the tonic triad (first, third and fifth notes of the scale degrees in the key of the piece) were reinforced by resonant frequencies, then they need only choose a key and construct resonators that achieve this. Of course, the same effects will occur in the event of modulation within the piece that were
discussed in regards to the first concept, however this is the nature of the acoustic phenomenon and the effect it has on the musical work.

A way around this would be to create multiple sets of resonators; a different set for individual sections of the music. The unwanted resonators could be covered to stop them reinforcing those particular frequencies, and another set uncovered to allow them to resonate. This then creates a very flexible set-up for compositions, as the number of notes decreed to resonate is limited only by the space available to position the equivalent number of resonators in front of the performer.

**MaxMSP**

This same effect can be achieved electronically. The software application MaxMSP, for example, allows the user to create virtual circuit boards, and the outpouring of this availability is manifested often in the creation of sound modulators by performing musicians and composers. For the sake of this area of research, the program was explored and the following conclusions were drawn that relate to this particular area (using resonators to enhance a musical composition).

The program uses particular patches that can be connected together in a series of virtual circuit boards (as mentioned above). Through the combination of the patches “fiddle~” and “bonk~” (downloadable third party patches) representation of resonance can be achieved. In particular, this particular virtual circuit can be used to create electronic resonators.

The patch fiddle~ is an audio analysis patch, which has inputs for the desired audio. Fiddle~ then outputs the values of the audio’s frequency and MIDI number. By connecting this output (in form of an integer box) to the input of another audio production patch, it is effectively using one audio source to trigger another. The following diagram shows a representation of a Max patch that uses an adjustable oscillator to trigger another adjustable oscillator whenever a frequency of 440Hz or higher is emitted.
This patch represents a model of “resonance” – one thing triggering another to produce something else. Adding in a few more components, this patch can then be used as a resonator. When the initial oscillator is replaced by an audio input patch then the following diagram represents how the frequency of the audio input sets the value of the oscillator to then produce an identical note (with a synthesised sound).
By using one or more of these Max patches simultaneously in a live electronic setup, a musician could use artificially created resonant frequencies as part of their musical compositions.

This method has been implemented by the composer Rob Godman, which was discussed earlier. Godman used a MaxMSP patch to create a virtual Vitruvian Vase, where a particular frequency was registered and reiterated by a synthesiser. What Godman didn’t explore, however, was the combination of this process with the representation of a room’s resonant frequencies at the same time. Whichever of the previous two methods are used for this concept – either poly-pipe resonators or an electronic set of resonators – this approach then has implications on what can be achieved with the first concept discussed in this chapter (Resonance Concept 1).

This method of controlled resonance allows for another possibility with Resonance Concept 1. After composing a piece for a room (based on the room’s resonant frequency) the piece can be performed in another acoustic space. Without having to modify the piece for the new environment, the specifications of the initial acoustic space (on which the piece was based) can be simulated, using either poly-pipe resonators or a live electronic setup using MaxMSP.
This means that when composing a piece based on an acoustic space the composer has more freedom on where the piece can be performed. When performing a piece that is based on an acoustic space’s resonant frequencies, they have the choice of either:

1. Measuring the resonance of the new acoustic space, and transposing the piece into the new key (based on the new acoustic resonance), or:

2. Preparing resonators that replicate the resonance of the original space (i.e. if the original resonant frequency was 440Hz, prepare a cylinder that resonates at 440Hz and play the piece into the resonator, or the equivalent setup using MaxMSP.

Further conclusions regarding Resonance Concept 1 Resonance Concept 2 are discussed at the end of this chapter (3.3.7).

3.3.3 Resonance Concept 3 - Compositions Representing Sympathetic Resonance

The following two compositions are the more significant developments in this area of research and both utilise the same concept of representing sympathetic resonance. The data gathered for these compositions is in the nature of interpolation. As stated at the beginning of the chapter this use of the acoustic phenomenon “resonance” is where this research takes a step further than composers like Lucier, and uses the phenomenon as external stimuli for the representation of it in a musical composition rather than just utilising it in its acoustical context.

The concept involves a small ensemble of one instrument type – i.e. trumpet, voice, cello etc – in which a single instrument plays a melodic line, and every note played will be held by one of the other instruments in an accumulative fashion. This is demonstrated in the following figure, in which the upper case letters represent the melody, and the lower case letters represent the corresponding notes being sustained by the other instruments:

```
Aa Bab Cabc Dabcd Ebcde Fcdef
A = Melody
a = ‘resonance’
```

Figure 29 – Model of sympathetic resonance for compositions.

Each instrument only plays one of the sustained notes, and each note would be dynamically bevelled down until it dissipates entirely (as represented above).
Composing pieces like this have had interesting considerations and restrictions, such as the relation between the melody and number of accompanying instruments; for each note in the melody, an accompanying instrument is required, therefore if, for example, there are 12 notes within the space of 2 bars and each note will be sustained for the whole 2 bars then over 12 accompanying instruments will be required to be a part of the ensemble. The details of the compositional method—including limitations, challenges and points of interest—are included in the following chapters under the topic of each composition.

3.3.4 Resonance Composition 1 - String Resonance

Appendices 1.5 & 2.5

This is a composition that has been used to demonstrate the principles of this research project. It seeks to represent the acoustic phenomena “resonance”, and in particular “sympathetic resonance.”

As discussed in the preceding section of this research, this composition is based on the idea of representing resonance through the use of an ensemble comprised of one instrument type, in which a solo instrument’s notes are “resonated” by other the other instruments of the ensemble for a set temporal duration. The instrument chosen for this composition was the violin. The ensemble is made up of 13 violins and is accompanied by three percussionists playing the riq (an Arabic tambourine), doumbek (Arabic goblet drum) and large drum.

The violin was chosen to be used for this piece as stringed instruments have a natural inclination to resonate in sympathy with other sounds, as do percussion instruments, as opposed to brass or woodwind that don’t have this same feature. It seemed that a stringed instrument was the most logical choice to demonstrate this kind of resonance and of the instruments in the string family, the violin had the particular tone that favoured the style that was chosen for the piece.

In creating this piece there have been a number of considerations that affect the compositional process, such as logistics, scoring and shaping of the melodic phrases. A piece in which the number of instruments is determined by the number of notes in each melodic phrase requires these considerations.

In scoring the piece, there needed to be an accurate representation of the concept of resonance through the use of instrumental and scoring techniques. To do this, a set of
parameters was applied to the melody in order to create the accompanying parts. These parameters are:

- Tempo: 90bpm
- Delay: 1 sixteenth note
- Decay: 8 (crotchet) beats
- Sample length to be resonated: 1 crotchet beat
- Velocity: same as melody
- Key Signature: D minor

This means that a single note played by the solo violin will then be echoed by another violin, with an attack that starts a semi-quaver beat behind the original and held for duration of 8 crotchet beats. As such, the accompanying parts were all to be written a semi-quaver beat behind the note of the melody they are “resonating” and are held for approximately 8 beats, dynamically decaying across the whole 8 beats.

**Notes on Composing**

As a general rule in composing this piece, *approximation* is preferable to tedious, difficult and technically correct. If it was deemed that performing a certain part of the process to the letter meant destroying another aspect of the piece, the preference was to perform the process as close to the parameter’s stipulations as possible until it ceased to be of purpose or was detrimental to the overall effect of the piece. This compositional ethic was developed in order to maintain the integrity of the piece without making it horribly difficult & annoying to read and play.

According to the pre-set parameters, each note played by the accompanying violin needed to be written a semiquaver beat after the placement of the original note – for example, if the solo violin played a note on beat ‘1’ of the bar, then another violin would echo this note starting on the count ‘1e’. Writing the attack of each accompanying note a semi-quaver behind the melody note, while not difficult to do or play, seemed to create an over-complication when reading. When the desired effect of the accompanying parts was to just play a split-second behind the melody, reading a note that begins on the last quarter of the first note of a crotchet triplet, for example, seemed more complicated than necessary.

This meant that the most practical approach was to use an alternative scoring method. For this, several ideas were considered. Firstly, the use of a method involving writing the
accompanying parts with the same rhythm and point of attack as the melody, but directing the player to delay the attack according to their own judgment and a pre-designated delay time (e.g. a sixteenth note). Another alternative was to write the accompanying parts with the same rhythm and point of attack as the melody and to conduct the accompanying strings (or have them led by a member of the accompanying ensemble) a sixteenth beat behind the melody.

Another method considered was to allow each accompanying player to respond aurally to a particular note (or perhaps several) that is played by the lead performer, meaning not using music at all. This would have added another element to the performance and composition of the piece, as the player’s aural ability dictates the performance’s success.

A middle ground between these ideas would have been to give each accompanying player a copy of the melody and then stipulate which note they are to resonate. Each player then just follows the score and plays their pre-designated note when it appears in the melody and holds it for the correct length according to the parameters. Visual aids could be employed through the notation of the player’s copies of the melody, such as using a different coloured note head for their notes, or larger size.

It was decided that the most straight-forward method was the first suggestion – writing the notes with the same attack time as the original and directing the players to delay their attack by a sixteenth note. This was also combined with the visual aids of the last suggestion, which involved every player’s part including a copy of the solo violin’s music with their corresponding notes enlarged. This method allows the players to see their notes in an easy to read presentation and to also see which notes of the melody their music corresponds to which allows them to achieve the desired rhythmic placement and dynamic level without too much difficulty.

The key signature chosen for the piece was D minor. Originally, the piece was intended to be in a major key to have an uplifting and joyous feel, however it seemed fitting to have a contrasting sound to another of the major works (Delay Composition 1 – Marimba Delay) as the marimba piece had already utilised the major key.

Instead, a distinctive Eastern feel was used, with a pull towards the Arabic “wailing” style of the Middle Eastern Violin. This was achieved through the use of an ambiguous mode based around D minor that sits somewhere in between a Phrygian, the more typical Eastern sounding
scale that includes the flat second and major third, a Dorian, and an Aeolian. The scale used for the piece is shown below:

Table 7 - Scale Used in Resonance Composition 1 - String Resonance

<table>
<thead>
<tr>
<th>Scale Degree</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>D</td>
<td>E♭</td>
<td>F♯</td>
<td>G</td>
<td>A</td>
<td>B♭</td>
<td>C</td>
</tr>
</tbody>
</table>

The inspiration for this particular sound came from music played on the rabab, the bowed psaltery, and the Middle Eastern violin.

Writing for violin was a new area of composition for this researcher. Until this point, compositions had only been written for instruments that were familiar, such as marimba, drum kit, saxophone and guitar. This meant that a little background research into the functioning of the instrument was involved, as it was important to be sure of the instrument’s limitations, capabilities, range and techniques before composing. The main component that needed to be clear was the range of the instrument and the names of the strings – where “double stops” (playing two strings at once) were to be used, it was necessary to know which pairs of note would be possible to play.

The following diagrams were sourced from Wikipedia, as only a basic demonstration of the violin’s properties was necessary and using a publically accessible internet archive seems acceptable for this situation. These diagrams demonstrate the range, string names and first position fingerings of the violin.

![Figure 30 - Names of violin strings](Violin, n.d.)

![Figure 31 - First position fingerings of violin](Violin, n.d.)

![Figure 32 - Range of violin strings](Violin, n.d.)
These diagrams were consistently referred back to during the composition of the piece in order to achieve accuracy in the instrumental scoring.

There were several points of similarity and of departure with the composition of this work compared to that of Delay Composition 1 – Marimba Delay. An enjoyable aspect of this compositional process was that the counter-melodies (or “resonated” notes) could be written on the accompanying strings’ staff and then left there for the final product. This is quite a contrast to the process involved with Marimba Delay, in which each part had to then be amalgamated into one, with all the rhythms joined together to form a cohesive piece of music.

A point of similarity between the two pieces was that it became common practice in both to write no more than a few bars of a melody at a time before applying the resonance parameters to it. Although the results were predictable, it felt necessary to confirm the effect that would be achieved through aural analysis before moving onto the next section of melody.

A key compositional feature that arises when representing resonance in music is what can be called “chord sweeping”. By its very nature, resonance leaves a trail of sound – a note is sounded and is then held by the acoustic space in which it is sounded. From experience a musician will be aware of the layering effect that can be achieved by playing a rapid succession of notes in a resonant space (or into the open lid of a piano with its sustain pedal depressed) to create a chord, as the resonating notes layer on top of each other. This phenomenon is referred to in this research as the “resonance tail”.

This same effect is created in the musical context of resonance. Following the creation of parameters for a representation of resonance in music, they then dictate that the notes played by an instrument are “held” by another instrument and when notes are played in rapid succession they create the same effect as in the acoustic context of resonance. This concept is explored in both resonance compositions in this research – String Resonance & Vocal Resonance.

As with Vocal Resonance, this composition utilised the “chord sweep” technique described above; in practice, this method involves creating short sections of melody that deliberately form a particular chord once the resonating notes are present. There was also experimentation with some deliberately harsh sounds, which is a contrast to Vocal Resonance in which most of the chords were quite harmonious with only a few clashes. In this piece full chromatic sweeps were used to form tonal clusters. This was achieved in a few ways: firstly, through notes placed chromatically in descending or ascending passages, and secondly
through glissandos. The stipulation of the parameters is that this piece will contain no smaller tone partials than a semi-tone, meaning no quarter tones or otherwise. This allowed for the existence of glissandos in the melody, where the accompanying violins would resonate only the chromatic notes – so while the melody player is sounding all notes including quarter tones, the accompanying violins only “grab” the notes that they are dictated to in the resonance parameters.

These blocks of sound are reminiscent of Penderecki’s *Threnody for the Victims of Hiroshima* (1960), in which he uses graphic notation to symbolise blocks of tone clusters, played by a host of stringed instruments resulting in a horrendously clashing sound. *String Resonance* utilises this same idea, but contrasts it with the Eastern sounding melody and harmonious chord blocks as well.

While chord sweeping was a device featured in this piece, the main purpose was to construct a melody that would maintain interest without being used as part of a harmonic structure. The piece begins with a particular melody, which then has the parameters of the resonance applied to it. Using this method, as opposed to constructing the melody with the “resonance tail” in mind, means that tonal clashes occur more frequently. These occurrences are multiplied, in both number and effect, because of the decision made in the compositional process to emulate a Middle Eastern sound. The typical Middle Eastern sound uses a major scale with a flattened 2\(^\text{nd}\) scale degree, a flattened 6\(^\text{th}\) and a flattened 7\(^\text{th}\) (like a Phrygian scale but with a major third). This particular scale sounds fine when on its own or harmonised with diatonic chords, however when the harmonies ensue from the resonance tail, the harmonies tend to clash somewhat.

As with most of the compositions that stem from this research, the original intent is often to create works that have severely clashing or atonal sounds with occasional moments of pure tonal harmony – however, this soon gives way to harmony that is more pleasing to the ear, as is the way with a great deal of Western music. In *String Resonance*, the piece starts with very discordant sounds in a deliberate attempt to firstly demonstrate that resonance is the key compositional tool and secondly to create an unexpectedly jarring and grating sound – the use of so many violins in an ensemble besides an orchestra warrants an exploration of the instrument’s group potential.

As the piece progresses, the harmony tends to get more consonant, as the compositional procedure took more of a focus on how to manipulate the resonance tail to create tonal
sounds. In the final sections of the piece, there are selections of chords that are deliberately used in a repetitive pattern (the C major, D major, E minor sequence in section G). What was discovered through the composition and auditioning of this particular section (G) was that the use of the chord pattern imitated something close to an involved counterpoint structure, in which the suspension, anticipation and resolution of notes occurred through the use of the resonating tones. These resonating tones are created through the application of the pre-set parameters to the melody, which means that there is no control over their placement and duration. The only control that can be had is through prior thought in how to manipulate these notes in order to create a desired effect. This section had the effect of chords slowly morphing into one another, rather than having a traditional change-over point. Often one chord (for example, the C major) would have all the notes of the previous chord (D major) sounding at the same time, resulting in jazz-reminiscent suspensions – in this example (in section G), the chord would be labelled a 13th (which includes the 7th, 9th and 11th).

Another compositional ethic was decided upon in regards to the use of grace notes; the style of the piece dictates that there are numerous grace notes attached to regular notes. It became important to consider whether or not these notes would be “resonated” — they are so fleeting and often softer than the main note, that it was not clear how they should be processed. To solve the problem, it was considered how the sound (of grace notes) would react in a room that was resonating a number of frequencies – in this situation it was deemed that the sound of the grace notes would be resonated but at a softer volume, as the sound doesn’t have quite enough time to “catch” and is softer than the other sounds being resonated. To represent this in the piece, the grace notes are written at a softer dynamic than the note it is attached to. It was decided that the dynamic level of the grace notes would be 3 steps lower, so if the main note was at ff, then the grace note would be played at mp.

The attack of the grace notes was another aspect that needed to be considered — the grace notes are all “acciaccatura”, so they are technically played before the main note rather than sharing the main note’s rhythmic value. Rather than complicating things by notating a rhythmically precise interpretation of where exactly the note would land, the “resonating” violins note was written on the same beat as the main note. As the grace notes are subservient to the main note, the reasoning was that by having them played softer it would give the impression that they had come in at a different time anyway.
Repeated notes were another point for consideration; if two or more notes of the same pitch are repeated consecutively, do they require to be resonated by separate violinists or just one? To assist in this decision, the problem was again related back to the behaviour of the acoustic phenomena in a sound context. In this situation, the repetition of a sound in a resonant acoustic space wouldn’t necessarily create a resounding of the space’s resonant frequency, but it would perhaps reinforce the sound. This was then translated into the scoring of the music by echoing repeated notes with separate violins.

Another compositional consideration was the use of extended durations in the melody line. In the event of a short rhythmic value being used, for example a quaver, the echo of the note by another violin occurs a sixteenth beat later than the transient which creates the effect of it almost appearing instantaneously. Where a longer note value is used, for example a minim or semibreve, the transient of the note cannot be used as the reference point off which the corresponding “resonating” violin gains its rhythmic placement, the note is being held for another few beats. In this situation, it seems that to adequately represent the acoustical context of resonance a further compositional parameter must be observed: if the melody note is held, then the resonated note won’t start decaying until the end of the held note. For example, if a semibreve is written, the “resonating” violin will also play a semibreve, and then decay for 7 beats after that – for a total of 11 beats playing time, 8 of which it dynamically decays over. The decay begins at the end of the note – a sixteenth beat after the last crotchet beat. For example, a crotchet is decayed for 8 beats, starting at almost the same time (delayed only a sixteenth note value), so a minim will be held for the first beat, then start decaying after the second beat for a total of 9 beats playing time, 8 of which decay.

In order to further pursue the possibilities of the use of resonance in musical composition, a number of parameter changes occur halfway through the piece. The piece begins in the key of D minor and in 4/4 time, with the decay time of each resonated note set to be 8 crotchet beats in length. It is also scored to have a solo violin (Violin 1) that is accompanied by twelve other violins. The parameter changes that occur are thus:

- Metric modulation: from 4/4 time to 6/8
- Tempo change: from \( \dot{\text{Z}} = 90 \text{bpm} \) to \( \dot{\text{Z}} = 180 \text{bpm} \) \( [\dot{\text{Z}}:=:\text{Z}] \)
- Resonance decay: from 8 to 12 (crotchet) beats, at new tempo

Another compositional change that occurs is the addition of one of the “accompanying” violins (Violin 13) to the role of melody. This means that both Violin 1 and Violin 13 have the melodic
roles, and as such both need to be “resonated” by the remaining accompanying violinists. This change was decided on in order to increase the layering possibilities of the resonant notes. By adding another melodic line it meant that additional harmonies could be created and more interesting interplay between the melodic line and the harmonic content could occur.

This change also sparked another compositional addition that was designed to enhance the stage performance of the piece. When composing the ensuing echoes of the melody in the accompanying violin parts, each note was allocated to the parts consecutively – for example, a crotchet played on beat 1 in the Violin 1 part would be allocated to Violin 2, a crotchet played on beat 2 in the Violin 1 part would be allocated to Violin 3, on beat 3 to part 4, beat 4 to part 5, etc. This is illustrated in these two examples (Figures 33 & 34) of the note allocation system used.

This system can be used in a visual way to enhance the performance of the piece. Firstly, the stage positioning of the performers will help illustrate the linear note allocation. This is achieved through the performers standing in a single line on the stage, starting with Violin 1 on
the left and finishing with Violin 13 on the right. This is illustrated in the following diagram (figure 35 – violinist image courtesy of www.123RF.com):

![Figure 35 – Illustration of stage layout in String Resonance](image)

This stage setup will allow for the layout of the musical score to be visually illustrated in the performance. An audience would be able to visualise the “resonated” notes appearing down the line of players, flowing from the melody player. A way to further enhance this was then added into the composition: the addition of sound-sensitive lights. These can be small LED or other lights that are activated by sound, and are attached to each violin. Violin 1 requires a red light, and all other accompanying violins use blue lights. This visually demonstrates that Violin 1 holds the melodic interest and shows each note that is “resonated” by the corresponding violin players 2-13. Here an example of the score is shown, bars 124-127. It can be seen how the parts flow down across the accompanying violins from the 1st violin.

This score can be shown side-on, to better correspond with the layout of the string players on the stage. In this next diagram the score is shown in this way, with the illustration of the string players matching the corresponding parts and a demonstration of the use of the sound sensitive lights.
Violin 13 will require both a red and blue light. The reason for this is that during the first half of the piece Violin 13 uses the blue light to represent the notes being resonated from Violin 1, and in the second half of the song (section F) it takes a melodic role and so switches over to a red light.

For example, below is a sample of the score (measures 124-128). It shows how the two melodic violin parts (1 & 13) have “resonating” notes that seem to advance on each other across the score/parts. This is again illustrated with the red and blue lights, this time showing a red light for Violin 13. The accompanying parts have notes and lights that are produced corresponding to both Violin 1 & 13 (left and right).
These scoring and staging considerations are used to facilitate the success of this piece as a model for resonance. By making the process self-evident, it demonstrates the piece working off a successful scale model that can be used to relate the properties of resonance to an audience through the techniques employed (musical devices, staging and scoring).

**Analysis and Conclusions**

The conclusions drawn from this piece are gathered from the act of composing and the analysis of the resulting composition. An audio file of the piece was created in order to analyse the piece in an auditory way (appendix 1.5).

In order to draw conclusions about this piece, it was referred back to the research method and criteria for evaluation (see section 1.4.3). The analysis of the piece involves determining if it represents a successful model.

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*Figure 37 – String Resonance excerpt (2), with illustration of sound-sensitive lights.*
The first point to be addressed is whether this composition demonstrates the application of a viable model through either interpolation or extrapolation. According to the definitions of what characterises these two methods of modelling, this piece can be seen as an interpolative model. The composition uses the parameters of resonance in exactly the same as it would be used in the context of sound production, but uses it in a musical context.

This piece draws similarity relations from the observable characteristics of resonance with the equivalent musical devices. The number of causal relations evident in resonance makes it a good model for representation in music and this piece adequately demonstrates this analogy by correlating these causal relations with the closely resembling behaviours in musical devices. This is a scale model that can be used to observe the properties of the original subject.

As this composition interpolates the data from the acoustical context into the musical context it uses a fairly clear translation of the phenomenon. The behaviour of resonance is imitated and translated into musical notation, performance techniques and staging considerations. In the case of extrapolation, this would perhaps have been a different outcome as it often occludes the process and original intention of the piece from the listener. This is reflected in most of the compositions within this research, as they all use the interpolation of data rather than the extrapolation of it – another reason for this is due to the ethic shared with Reich concerning the desire for self-evidence of the compositional process in music.

The modelling employed in this piece facilitates the connection of similarity relations between the phenomenon and the musical processes. The process used in this composition was found to be self-evident. The beginning of the piece in particular was deliberately constructed to allow the listener to observe the audible process being used. As stated, this is a scale model that allows for the “reading off” of properties of the original subject. This representation of resonance in musical composition should be obvious to any musician who has experienced the phenomena before, and even those who find it unfamiliar may still be able to relate the composition to a concept similar to resonance.

The scoring consideration employed to assist in making the modelling even clearer was the use of the lights and performer positioning in the stage performance of the piece. Having the players in a single line allows the audience to see how each note from Violin 1 (and also Violin 13 later in the piece) is echoed by the accompanying strings down the line and this is highlighted by the use of the sound sensitive lights.
A number of the relations evident in resonance as an acoustic phenomenon were used as a model. The re-sounding of a frequency was translated into the resounding of a note, and the temporal element of this was modelled as closely to the original as possible. The scoring indicates that each accompanying part is to be played slightly after the transient of the melodic line, which was an attempt at translating the embodied characteristic of resonance into the music; a note resonating in sympathy with a triggering source sound tends to occur a split second after the initial attack. These connections demonstrate the preservation of the relative proportions between the magnitudes of resonance with minimal deviation.

In terms of whether this piece is strictly theoretical or performable, an evaluation of the difficulty level of the piece dictates that the piece is performable. There are no instrumental techniques used that create a difficulty in the piece’s performance and all notes are written within the range and easy fingering access of the instrument. The accompanying parts are quite simple, as they are always holding single or double-stopped notes for the duration of eight beats.

The only potential difficulty is the placement of the notes in the accompanying string parts. Each part is directed to be played a sixteenth beat later than written, and in some cases the entries are already placed in difficult positions. For example, in the diagram below (bar 1 of the piece) Violin 1’s melody creates some awkwardly placed entries for the accompanying violins, and each of these entries is directed to be placed a sixteenth (semiquaver) beat later than written.

However, this difficulty is offset by the overall principle outlined in the performance notes of the piece. The stipulation is that while accuracy is important, achieving it is not to detract from the performance of the piece. Also, to assist performers with their rhythmic placement a guide staff of Violin 1 is provided on each accompanying piece of music (and a guide staff of Violin 13...
later in the piece, where applicable). These factors are in place to assist in the performance of
the piece and to ease any concerns performers have regarding the strict placement of
rhythmic values.

The modelling used in this context allows for the further expansion of the field. As can be seen
with Reich’s plethora of compositions stemming from the single acoustic phenomenon phase
shifting, this single phenomenon of resonance has multiple applications and musical settings.
Add to this the point that this representation of resonance is more focused on sympathetic
resonance rather than resonant frequencies, the latter of which is merely conceptualised in
the first half of this chapter \textit{(Resonance Concepts 1-2)}. The use of resonant frequencies
involves the use of selective frequencies, as an acoustic space would not “resonate” every
single note played as does the representation of “sympathetic resonance” which can be seen
in the playing of an instrument into an open piano that has its sustain pedal depressed.

The variables in any composition can play a large role in the creation of many new works, such
as the instrumentation used, size of ensemble, key signature, key modulation, time signature,
metric modulation, rhythmic tools used and ratio of melodic instruments to harmonic and
rhythmic. In this type of composition, the list of variables is expanded by the fact that a
particular process is used, each parameter of which can be changed at the outset of the piece
or used as a constantly changing variable throughout the piece. This means that the addition
of the resonance parameters – the duration/decay time of the resonated notes, the sample
length that is to be resonated, and the rhythmic placement (delay) of the echoed notes – a
piece such as this has vast room for variation and exploration.

For example, this concept of resonance could be combined with the concept explored in this
chapter of resonant frequencies. In this situation, a composer could analyse an acoustic space
for its resonant frequencies and then use instruments in an ensemble to “resonate” these
selected notes of a solo instrument. Conversely, the analysis of an acoustic space need not
occur at all – instead, the composer may choose to “resonate” selected notes of their own
choosing, whether this is based on an “imaginary acoustic space” or is simply manipulated in
order to enhance particular aspects of the composition.

\textbf{Summary of Conclusions}

In summary, this piece can be seen as an over-all success due to its acceptance of the Criteria
for Evaluation points with a positive response. The work successfully demonstrates the
application of a viable interpolative model of the acoustic phenomenon “resonance” in a
musical composition. The scale model used for this representation allows for the self-evidence of the process and the phenomenon being modelled.

The work is performable by any ensemble of musician with adequate technical proficiency and the concept has a great deal of room for further exploration by subsequent composers.

### 3.3.5 Resonance Composition 2 - *Vocal Resonance*

*Appendices 1.6 & 2.6*

This is a composition that has been used to demonstrate the principles of this research project. It seeks to represent the acoustic phenomena “resonance”, and in particular “sympathetic resonance.”

This concept was discussed in the preceding chapter, under the discussion of representing acoustic phenomena, and in particular interpolating from the data gathered from the phenomenon “resonance.” In this chapter, the realisation and composition of that concept is discussed.

**Inception**

The initial idea for the piece originated from explaining the difference between resonance and reverberation to someone: “Reverberation is comparable to speaking into a room and hearing the reflections of your voice coming back at you. Resonance, using the same example, is more akin speaking into a room and having another person with the exact same voice repeating exactly what you said but in long drawn out tones.”

This led to the idea of a composition similar to the string resonance piece, but using voices. The concept was that a choir of voices would be used to accompany a solo singer, which immediately would be different to the string composition as each voice is different (unlike strings, which can be made to sound very similar).

Also unlike the string composition, the accompanying singers would not only have particular notes to resonate at certain times according to the soloist’s rhythm, but also certain vowel sounds. For example, if the soloist were to sing the phrase “I went to the park,” the accompanying singers would resonate “I”, “eh”, “oo”, “er”, “ar”. This concept created interesting possibilities, as it seemed the combination of vowel sounds could potentially create a variety of overtones and harmonics.
Extra consideration was needed for the scoring for this, as each accompanying singer is required to sing a particular note, but also needs to sing a particular word or vowel sound in the case of the same note being sung with two or more different words. For example, if one singer were to “resonate” the note “A” and the vowel “oo” (as in from the word “to”) and the next repetition of the note “A” uses the word “the” then another singer will need to sing that note and vowel sound.

These considerations meant that a larger ensemble was required to perform the piece such as this, which for orchestration purposes didn’t make much difference as choirs are generally large enough ensembles to cater for a large number of parts.

**Composing Vocal Resonance**

What can be said in an anecdotal analysis is that what the most enjoyable element of composing this piece was that the resulting composition was enjoyable to hear, due to the fact that the composition largely used a pre-determined process. Because of this, as with other process-driven works, the composer can simply write a brief sample of music and then watch the processes unfold. Unexpected by-products (like Gann’s prognosis of “meta-music” in Reich’s work, or “resultant patterns”, which is discussed in a later chapter) can easily come about, and just like winding up an old-fashioned wind-up toy, once released you can soon see if the process you’ve put in place was successful or not. In this kind of composition, the composer can be as much a part of the audience as the listeners.

**Notes on Composition**

A summary of these notes on the compositional process involved in this piece are also reflected in the “Performance Notes” that are attached at the beginning of the piece’s score. As shown in the performance notes, the whole piece is designed with set parameters; these parameters are used to ‘process’ the melody, which creates the additional parts.

- Tempo = 100bpm
- Duration of held notes = 4 beats
- Time signature = 4/4

In addition to these parameters, several decisions were made prior to composing the piece. One such decision was that all notes would be chromatic according to the 12 part western scale (i.e. no quarter tones). This was decided so that in the event that the soloist performed a glissando, the accompanying parts would not “catch” the quarter tones in between; just like a glissando on a piano, they would only resonate the chromatic notes.
The first step was writing the piece with only a melody in mind. Following the construction of a portion of the melody, the lyric content was chosen to mirror the mood of the piece created from the melody so far. The text chosen was *The Dream* by Edgar Allan Poe, and after this was chosen as the writing of the remainder of the melody ensued with the aim of then matching the melody to the lyrics. Edgar Allan Poe poetry seemed a fitting lyrical partner for the melody that had already been written by this point; the minor key and drawn out phrases were melancholy enough to suit Poe’s dark lyricism. This programmatic style of composition can be observed in the major modulation in the second half of the piece to reflect the lyrical content.

The notation that was chosen for writing the accompanying parts was a simplified version of what it could have been. The attack of each accompanying part should have been placed just slightly after the initial melody; rather than trying to notate this though (which was attempted before in the process of the composition, using semiquaver rests, semiquavers coming in off the beat, etc. – which proved to be very messy and hard to read) the parts were simply written having the same entry points as the main melody. The composer’s notes section at the beginning of the piece is used to instruct each performer to bring in each note slightly after the initial note, which will require them to listen carefully to the soloist. This was the same method as was used in *String Resonance*.

In terms of scoring, having eight parts worked well; there were no problems with overlapping notes as long as there were no more than 8 crotchets beats per 2 bars. For example, if a quaver was added in a two bar sequence a minim would be required in lieu of two crotchets in order to not overload the notes. In assigning notes to each part, a system was utilised that prioritised parts that had not played for a while. If a part had just had a one bar rest and another part had a two beat rest, the next part would be assigned to the former.

Modifications were required for the melody once the process to create the secondary parts had been set in place. Certain notes had to be truncated (e.g. a pair of semiquavers became a single quaver) due to the fact that it would’ve required adding a whole other part just to play the notes. In some instances, the melody had to be changed even more so. In pieces like this, the composer will reach a point where a decision has to be made: do you retain the melody as is and add extra parts to accommodate, or do you change the melody to avoid adding extra parts and changing the orchestration? In most instances the latter choice was opted for. Several beats of rest were added in between phrases that were originally a continuous series of notes, which allowed the parts to be retained as is without adding new ones.
In writing the vowel sounds for the accompanying parts (Voices 2-8), a number of decisions were made concerning how they should be written. These decisions were made for ease of composition and for a more straightforward read for the players.

The first issue that needed to be addressed was that when writing the phonetic spelling of the vowel sounds for each answering part, should proper letters be used that formally dictate the appropriate vowel sounds (for example, using a detailed phonetic alphabet) or should the vowel sounds simply be written phonetically in whatever way would be deemed most understandable? Having no real knowledge of phonetics and linguistics and not wishing to divert into this area for time purposes the latter option was chosen.

Writing the vowel sounds correctly would require some knowledge of phonetics and linguistics – for example, delineating the vowel sound for the symbol á. The choice in the end was to just use phonetic spelling of vowel sounds (for example, “ow”, “oh”) and in some cases where the phonetic spelling was difficult to work out a system was employed in which the vowel was followed by a bracketed consonant, allowing the player to sound the vowel sound in context of the part-word written; for example, i(n), a(t), o(n).

The vowel sounds for each additional part were worked out by process. For each line, the note that was “resonating” was observed and the word attached to that note. The vowel sound from the word was then extracted and attached to the accompanying part with the phonetic spelling of the vowel. In the case of composite vowels (vowel sounds that transition within words, such as “out”; made up of “ow” and “ooh”) the initial or dominant sound was used. This decision was made to simplify the composition, however it allows for a future possibility of expanding upon the parameters of the piece – each vowel sound in a composite word could be resonated.

In discussion of potentially expanding upon the parameters of the concept of this piece, another element that was considered was the point of attack of the accompanying parts. At that point in the compositional process, the resonance of each note was being positioned in relation to the point of the attack of each note; meaning that an accompanying part was provided to resonate each note of the melody in its entirety. In the case of sampling (sound production), the parameters stipulate how long each “sample” is to be for – for example, it may take a 500 millisecond sample, and in the case of a delay it would repeat that 500 millisecond sample at a set interval. In this composition, this parameter has not been dictated – instead, a simpler form of composition was chosen in which single notes are resonated,
regardless of their length. To make for a more accurate representation of an acoustic phenomenon however, there really should have been a stipulation of a parameter for what duration is being resonated. For example, if it were stipulated that each crotchet beat value (and smaller) is to be resonated, then when the melody exhibited a minim it would require two accompanying parts, one for each crotchet division of the melody’s note – in the case of a semibreve, 4 parts; a breve, 8 parts etc.

The end of the song was written with what was referred to in String Resonance as “chord sweeping”. As discussed in reference to this other composition involving resonance, the melody in this technique creates the harmony. For the most part, the melody was written with a horizontal or linear melodic structure – meaning, a melody that sounded good on its own and allowed for easy harmonic additions. The last passage, however, was written with the motive to create certain chords, using the one vowel sound – chord sweeping. These chords are used with only one word at a time, and as such each additional performer is singing the same vowel sound.

The coordination of the lyrics with the melody worked out well; the last four words of the piece have contrasting vowel sounds: i(n), ooh, ay, ah. It seems fitting that the final vowel sound is “ah” – the piece is very hymn-like, so the “ah” could be taken as an unfinished “a-men”; a parody of a hymn then, that leaves you waiting for the “men”!

The chord sweeping mentioned could be seen as similar to waving a sparkler in the air – you can see a trail of light that fades after a short time, and if you move fast enough you can write words and letters that will stay for a moment. If you move slowly, you can’t see the letter you are trying to write. Similarly (for a more “pop culture” reference), it seems reminiscent of the futuristic bikes in the movie “Tron”; the bikes leave a trail behind that do not fade, and you must be mindful of the trail lest you run into it and die! In the same way, the note trails left behind the melody by the accompanying parts had to be carefully considered when creating the melody, as inharmonious clashes were too easy to create. For example, in this composition the key of D minor was used and, unintentionally, a very traditional harmonic structure was used – chord patterns like i, iv, V, i etc. During the composition process it seemed that the raised seventh (leading note), while musically correct, created a very harsh clash of harmony (although it didn’t seem so dissonant when the section modulated to a major key). Applied dominants (e.g. V of iv), which were also used in the piece, clashed too. To create a more harmonious piece, a modal key should really be used, or a natural minor.
The final consideration of the piece was the scoring and final notation of each part. The parts needed to be written in a way that would allow for legible reading and clear understanding of the intent behind the parts. In a similar vein to Steve Reich, processes were avoided that would not be easily observed, and instead obvious processes were utilised that could potentially be appreciated by any listener.

The scoring of the individual parts include their own part as well as the soloist, as the coordination of entries was deemed to be easiest when the performers could see the original word (rather than just the phonetic spelling), where their note is in relation to the rest of the phrase, what the rhythm is etc. Simply writing pairs of semibreves with some rests in between would not make for a stimulating read, and would likely be more confusing and degrade from the performance of the piece.

It was considered that using a special form of notation may assist in addition to this, by perhaps using different colours for the notes. In the individual written parts, the idea would be to highlight the corresponding note on the soloist’s line, so that each performer could essentially be reading the soloists part but with highlighted notes to illustrate which notes are theirs and what vowel sounds to use. The result was that a similar idea was but through enlarging the notes of the melody that each part would resonate, as well as using a stem-up-stem-down system – the notes of the melody that were to be resonated by a particular individual part had their stems going up, while the rest of the notes in the melody had their stems down.

**Recording**

This composition has been recorded to aurally demonstrate its inherent concepts. The recording differs from the concept in a few ways, however; primarily, the difference being that a single female vocalist performed each part in a multiple-take, over-dubbed recording. Originally, the concept was to have the piece performed by a choir, but for a demonstration of the principles it seemed that recording in this fashion achieved the desired result.

**Analysis & Conclusions**

The conclusions drawn from this piece are gathered from the act of composing and the analysis of the resulting composition. The recording of the piece was also used to assist in the assessment of the work.

The same evaluative process was used as with *String Resonance*, which is by referring back to the research method and criteria for evaluation. As with the other compositions created in this
research project the acoustic phenomenon has had its behaviour characteristics *interpolated* into the setting of a musical composition. The concept of “sympathetic resonance” has been utilised to create a set of compositional parameters, which were then applied to a melodic structure within an ensemble setting.

This demonstrates the modelling of the phenomenon through creating parallels of its observable causal relations with musical components. This piece can be seen as a viable model as it meets the criteria for a good model. It presents an imitative representation of the phenomenon that can be used for “reading off” its properties and it does so by preserving the relative proportions of the magnitudes evident in resonance. These magnitudes are the relations that exist between the sound sources, the resonant body and ensuing resonant frequency, all of which are represented in the relations of musical elements in this piece.

The composition seems to successfully represent the phenomenon of resonance, except perhaps for one detail. The vowel sounds being echoed back from the accompanying ensemble is, as discussed, not an accurate representation of the phenomenon’s manifestation in acoustics. Perhaps a better way of representing this phenomenon using a vocalist would be to have the ensemble comprised of another instrument other than the voice – for example, a solo vocalist that has an accompanying ensemble of violins.

As the choice to resonate the vowel sounds is not necessarily an accurate depiction of what would occur in the acoustical context of sympathetic resonance, this could also be seen as an extrapolation of the phenomenon’s properties. When imagining the interaction of the phenomenon in an acoustic context, if a vocalist were to sing into the open lid of a piano of course only the notes sung would resonate back, not the words or vowel sounds.

This could be seen as an extrapolation of the phenomenon’s behavioural characteristics into the context of a musical work. The replication of a sound’s timbral qualities is not an inherent component of resonance – that quality is more attributable to the phenomenon “reverb” or “delay”, where the whole sound is reflected and not just the frequency that the acoustic space intrinsically reinforces.

This piece need not be a strictly theoretical composition either. This piece is performable, as is demonstrated by the recording of the work. The recording presented a different framework for the performance of the piece, as it was sung by a single vocalist and each part was layered on top of one another. In an ensemble performance this would obviously be different as there
would be the consideration of working together with a group of vocalists to achieve the same result, but besides this the recording demonstrates that each individual part is easily performable.

The application of the model in this context demonstrates the room for expansion within this subject area. In fact, this composition can be seen as a demonstration of the expansion of this concept, as the concept was first used in *String Resonance*. *Vocal Resonance* illustrates that the substituting different variables into the model can create a substantially different work when representing the acoustic phenomenon “resonance.” The changes in compositional variables that have occurred from *String Resonance* to *Vocal Resonance* are the ensemble type (voice), resonance parameters, key signature and modulation. These variables are, of course, just one example of the changes that can be made to create an entirely different work.

The concept of re-sounding the vowel sounds is also an avenue that could be further explored. A variation could be to echo the consonants instead of the vowels - this would be an interesting and challenging exploration, as vowel sounds are more commonly sung on than consonants. For example, the common consonant sounds are produced either by the tongue, lip or teeth. The tongue sounds are D, L, N, R, T and Th, the teeth sounds are C, G, J, K, Q, S and Z, and sounds produced by the lips are B, M, and P. Singing on these sounds requires more attention to the way in which they are produced, and it’s possible that the effect would be more percussive, aggressive and “noisy” as the different consonants use a more constricted flow of air than vowels. Furthermore, a composer could even choose to use both consonant and vowel sounds: one each sung by two different singers, which would compact a single word into its primary qualities.

**Summary of Conclusions**

In summary, this piece can be seen as an over-all success due to its acceptance of the Criteria for Evaluation points with a positive response. Like *String Resonance*, the work successfully demonstrates the modelling of the acoustic phenomenon “resonance” in a musical composition.

As demonstrated through the recording of the work, the piece is performable. This area of composition is also interesting and allows room for expansion for future composers.

In addition, it can be seen in both *String Resonance* and *Vocal Resonance* that a parallel exists with the works of composers like Tristan Murail. Murail’s works *Mémoire- érosion* (1976) and
Ethers (1978) both feature the concept of solo instruments that provide an acoustic model for an ensemble, which is also the case in both *String Resonance* and *Vocal Resonance*. These compositions are discussed in more detail further in this research paper.

### 3.3.6 Creating a Functional Model of Resonance

Thus far, all the compositions and concepts developed concerning resonance have been using the *interpolation* of the data gathered from the phenomenon. Representing harmonic resonance through interpolative modelling in a composition is perhaps one of the more straightforward methods of using acoustic phenomena to generate a compositional practice.

As this is a large area of research, there has been little room to explore the possibilities of how one may *extrapolate* the data from resonance to use in music compositions. Extrapolating from a concept is much more complex than interpolating from it and as such the number of resulting composition ideas reflects this difficulty. However, as the extrapolation of data presents a significantly broader area it also generates further possibilities than with interpolation. As Black states, “the possibilities for construction of analogue models are endless” (Black, 1962, p. 222).

In order to formulate further methods of representing “resonance” in musical compositions it is necessary to create a model of the phenomenon’s underlying behavioural characteristics.

A model for resonance can be created through an understanding of the base nature of resonance in an acoustical context. To describe the nature of resonance, it can be said that a sound is “received” by an acoustic space, which in turn produces a corresponding frequency.

To demonstrate this in a diagram, resonance can be seen as follows:

```
Sound A (f = X) ➔ Acoustic Space (Y) ➔ Sound B (f = X ± Z)
```

This shows how a sound (Sound A) of a particular frequency (X) played in a resonant acoustic space can then produce another sound (Sound B) with the same frequency of the first sound, potentially with a reinforcement or occlusion of particular partials (X ± Z, where Z = partials). Simplistically, this function could be viewed as X ➔ Y ➔ Z; however this function doesn’t really assist in understanding the phenomenon.
To break this model down even further, resonance in its most basic form could be illustrated as:

![Diagram showing trigger, receptor, and product](image)

In this diagram, a Trigger (X) is received by a Receptor (Y) which then produces a Product (Z) as a result of receiving the trigger. Therefore, resonance could be defined as a conditional operation; an operation that occurs upon the condition of something else happening.

Considering resonance as a model allows a composer to imagine new ways of creatively representing it through musical composition. As seen in the example of sympathetic resonance, or even resonant frequencies (acoustics), harmonic production as a result of an initial trigger could be represented musically.

By viewing the phenomenon in this model format, the different components of the model can then be substituted for a variety of musical components. These variable musical components are numerous, and as listed in the previous chapter on phase shifting, they could be:

- **Pitch** – intervallic relationships, key signature, harmonic flow
- **Rhythm** – meter, rhythmic intervals, pulse
- **Timbre**
- **Dynamics**
- **Texture**
- **Performance technique**
- **Instrument choice**

For example, instead of a frequency triggering the production of another frequency:

- A rhythmic figure could trigger the production of another rhythmic figure
- A rhythmic figure could trigger the production of a harmonic pattern
- A certain frequency could trigger the production of a rhythmic pattern
- A frequency or rhythm could trigger a timbre variation
- A variation in timbre could trigger a melodic phrase
These variables create a vast number of possibilities within this one area of representative music. A few of these listed ideas have already been used by past composers, but not necessarily with the intent of representing acoustic phenomena.

This model, in referring back to the criteria for evaluation, lends itself to the creation of scale or analogue representations. If the values (the characters X, Y & Z) of the model are re-substituted for the original properties of the phenomenon it shows that the model represents the relations of the phenomenon adequately.

In viewing this model as an analogue, it can be said that it maintains a point-by-point correlation with the relations evident in the phenomenon. The causal relations evident in the phenomenon are the relations between the acoustic space, the trigger (a sound) and the resultant product. These relations are maintained through a point-by-point correlation with the components of the function (the characters X, Y & Z).

Whilst the model also may seem overly simplistic, the oversimplification of the phenomenon was avoided (as in the case of phase shifting). It was stated in the creation of the model how an even simpler function could be created, which according to Hesse’s criteria is desirable. However, Hesse would likely agree with Black that the truth value of the analogy needs to be maintained, which often dictates against oversimplification.

When using this model for the purposes of extrapolating the relational properties into a musical composition, an issue arises that is common amongst numerous forms of metaphorical representations. The extrapolation of the characteristics of an acoustic phenomenon such as resonance into a musical composition tends to exclude the listener from the process. When listening to a piece that generates rhythmic values from the harmonic series, for example, the audience won’t know from where the compositional data has arisen. This is the crux of Reich’s argument; that musical compositions should have self-evident processes that are as obvious to the audience as they are to the composer.

3.3.7 Resonance as a Drone – A New Direction in Tonality?

These conclusions are drawn from the experimentation, results and compositions of Resonance Concept 1 and Resonance Concept 2 (section 3.3.1 and 3.3.2).

The presence of drones in music dates back to at least the twelfth century, where in florid organum the lower voice “lost its original character as a definite tune, becoming instead a series of single-note ‘drones’…” (Grout & Palisca, 2001, p. 74). In La Monte Young’s works he
often used a sine wave generator to provide a consistent background drone, and Alvin Lucier
did the same in compositions such as *Music for Piano with Slow Sweep Pure Wave Oscillators*.

Drones of course also appear in numerous other musical contexts, such as the drone pipe of a
bagpipe, the drone strings on a five-string banjo, and eastern instruments such as the sitar,
sarod, sarangi and rudra veena. Composers like Haydn, Beethoven, Mahler and Berlioz were
also know to use a drone in certain sections of their works, usually using an interval of a fifth in
sections to evoke an archaic feel.

In her text *Listening Through the Noise* (2010) Joanna Demers discusses the effect of drones on
a listener, particularly within the context of electronic dance music. Demers points out that
drones impose a sensory deprivation that then enhances the listener’s other modes of
perception. Subtle fluctuations of timbre, pitch and rhythm are more apparent and take on
greater importance when compared with the static background of the drone. (Demers, 2010,
p. 93)

In this research, other than the drones mentioned above there appears another type of drone
– that of resonance. The early resonance experiments and compositions demonstrated ways of
using the resonant frequency of an acoustic space to create a musical composition – by
analysing the resonant frequency of a space, writing a composition that utilises this particular
frequency as the tonic key, and performing that composition in the space. This use of
resonance can be equated to the use of a drone in a piece, but also has a number of points of
departure from the average drone music.

As stated in the earlier discussion on this subject, having the tonic note consistently sounding
(as in drone music) changes the nature of the harmonic progression, melody and sense of
tonal pull within the piece. For example, moving the harmony of a piece to the dominant (fifth)
chord and then to the tonic (first) chord (a “perfect cadence”) has a strong sense of finality to
it – every note in the dominant chord seems to pull toward the closest notes in the tonic
chord, as our ears have been trained this way in Western musical culture. If a tonic drone is
present throughout the cadence, however, it feels more like the harmony has not progressed
and in fact has never left the tonic chord.

Similarly, if a resonant frequency is used as a different degree of the scale – for example, the
resonant frequency of the room is a B and the piece is set in E, making the “drone” frequency
the fifth degree of the scale – then the tonal centre won’t feel as stable, as it is constantly affected by the resonant frequency of the room.

This seems to be an additional factor that can be incorporated in the performance or compositional practice of music – the resonant frequency of a space, and how it affects the performance of any given piece of music. As well as pieces of music being written for a particular space, it would also mean that extra consideration would need to be given into what pieces of music are performed in a particular space, knowing that the resonant frequency of the room could subtly enhance or disturb the tonality of the piece. This could give rise to pieces being transposed for the space they are in, and perhaps even new spaces being created that are tailored to resonate particular frequencies over others – imagine having the “D# Concert Hall”, and the “Perfect Fifth Theatre in Eb”, where composers can choose the space that creates the desired effect for their chosen musical work.

The other consideration, for both pieces that are written for particular acoustic spaces and for works that are to be transposed for a space, is that a portable system could be utilised to ensure that whatever space is being performed in the desired effect will be reached. This can be achieved either acoustically or electronically, through the use of resonators or computer patching software that achieves the same end.

The first system would be a row of resonating tubes, much like the resonators used by Helmholtz in his experiment or the resonators on a marimba or xylophone. The resonators would be performed in front of to allow selective notes to be resonated in the cylinders, and with a little more thought even a series of resonators could be used that can switch from being open to closed (and therefore muted).

Along the same vein, an alternative could be in the form of piano strings that are placed close by the performer, suspended on a device. Failing this, a piano can also be used with only selective notes being held down (by another player or by weights), such as in particular pieces that use depressed piano notes for a similar effect (e.g. John Beck’s *Three Episodes for Timpani* (1980), the first movement of which features a piano placed behind the timpani with the sustain pedal depressed, so that each note struck on the timpani will resonate the corresponding piano string).

The second option is to use an electronic system to achieve a similar result. The program MaxMSP allows for simple acoustic systems to be created as an audio patch – a composer
could utilise a system such as this by creating a series of resonance patches that register an
selected incoming frequency and transmit the same frequency back out, using whatever tone
the composer desires. This method was used by Rob Godman in his piece *Halo* (2006) for piano
and electronic apparatus – it uses MaxMSP to detect the notes of the piano, utilising the patch
[bonk~], which then triggers the equivalent pitches on the synthesiser.

This may be a more practical way to achieve this result, however the use of physical resonators
always seems more impressive, as it demonstrates a physical acoustical principle without the
aid of electronics (as seems to be the plight of too many parts of modern music.)

Whichever option is chosen – the manipulation of a space through portable devices such as
resonators, the construction of a purpose-built acoustic space, or the transposition or creation
of musical works that allow it to utilise an existing space – this area can potentially impact the
direction of modern compositional and performance practices.
4.0 Reverberation and Delay

The two acoustic phenomena reverberation and delay are closely related occurrences of sound. Echo is also closely related to delay, and all three manifestations of these three similar phenomena can be found in acoustical context as well as simulated using digital processes. In each context, they can be classified as acoustic phenomena as they exhibit the qualitative and quantitative values associated with the production of sound.

Reverb and delay can be used in musical compositions as an acoustic enhancement to existing musical material or can potentially be used as a model to generate new musical data. The properties of these two phenomena exhibit temporal causal relations (one at a more rapid rate than the other) and similarity relations can be devised between these behaviours and musical techniques. Some of the composers under study have achieved this while others have incorporated the phenomena in their acoustical contexts into their musical works.

This research demonstrates the use of delay as model through the creation of a musical composition (Marimba Delay) and also through the construction of a functional model of delay that can be applied to music in order to generate further compositions in an extrapolative or interpolative fashion.

4.1 Reverberation and Delay Defined

There is often confusion as to the differentiation between delay, echo and reverberation and in particular between delay and echo. While they each exhibit similar behaviour, they are still distinguishable by different characteristics that make them definable.

These acoustic phenomena all involve the interaction of sound waves with their environment and in particular their reflection from surrounding surfaces. Holland states that “when a sound wave or part of a sound wave encounters an obstacle, it is obstructed by the obstacle, resulting in various alterations of the wave, including sound absorption, reflection, scattering, diffraction, and refraction” (Holland, 1985, p. 1). This discussion refines and clarifies what occurs in these instances that Holland mentions.

4.1.1 Reverberation

Stark states that sound in enclosed spaces tends to “behave somewhat like the waves created by a pebble dropped into a fish tank. If the walls, floor and ceiling are bare, the sound undergoes numerous reflections before it dies out” (Stark, 2000, p. 17). Reverb is related to
delay and echo; Huber & Runstein state that is in the same category as these two, as “it’s actually nothing more than a series of closely spaced echoes” (Huber & Runstein, 2001, p. 393).

It is these reflections that Stark refers to that result in reverberation – “a series of echoes so closely spaced they cannot be distinguished apart, sounding instead like a continuous decay following the initial sound.” (Stark, 2000, p. 17) Each reflection causes the sound waves to lose energy until it finally dissipates as it is absorbed into the surrounding reflective surfaces. The initial sound is heard first, followed closely by the reflections of the room. These are referred to as “early reflections”; the echoes from nearby surfaces that arrive after the direct sound is heard, within around 50 milliseconds (Rumsey & McCormick, 2006, p. 23).

Huber & Runstein clarify the three components of reverberation. They state that “in nature, acoustic reverb can be broken down into three subcomponents: direct signal, early reflections and reverberation” (Huber & Runstein, 2001, p. 393). The direct signal is audible first, as the sound wave travels straight from its source to the listener’s ear, followed by the early reflections discussed above. The third component is the last set of reflections of the sound to arrive and make up the signal’s characteristic reverberation. These final reflections are what reverberation primarily refers to, as in the discussion above; these thousands of random reflections are so closely spaced that the brain can’t differentiate between them, so they are discernible only as a single dense sound that decays over time (Huber & Runstein, 2001, p. 393).

The reverberation of a room is often used to describe a room’s acoustic properties; the early reflections of the sound are what initially allow the listener to perceive the size of a room, as the distance between them and the direct sound is related to the distance between the major surfaces in the room that cause the reflections (Rumsey & McCormick, 2006, p. 23). In music concert halls and other performance spaces a certain amount of reverberation is often desirable, as it tends to even out the sound of an instrument and smooths out pitch discrepancies.

In studio environments, however, soft, absorbent materials (such as carpet and foam) are used in abundance on the walls, ceiling and floor in an effort to reduce the reverberation present, as the sound reflections are not desirable when recording instruments. Having a “dry” sound (a flat, untainted recording of an instrument, as opposed to a reverberant sound) is desirable, so that if reverberation is desired on the end result of the recording the engineer is in control of
how much, rather than being at the mercy of the existing reverberation of the room in which the sound is recorded. This is where artificial reverb is often utilised, which is constructed using a series of delay lines with varying time and amplitude variables (Huber & Runstein, 2001, p. 393).

4.1.2 Acoustic Delay, or Echo

The acoustic phenomenon “delay” is often referred to as “echo”. These two terms are often confused, and it seems that a clear definition is rarely provided. The natural occurrence of the phenomenon to which both terms commonly refer to is where a reflection of sound arrives back at the listener an amount of time after its initial production. This is caused by the sound being interrupted by a surface that causes the sound to reflect; the Oxford dictionary states that echo, or delay, is “the repetition of a sound by the reflection of sound waves” (Echo, n.d.).

While this description is also true of “reverberation”, the difference is that reverberation has many reflections that are so close together that no single sound can be clearly perceived. In comparison to an echo, the multitude of echoes that arrive in very quick succession are diffused and deflected by objects and surfaces, resulting in some sounds returning to the listener later than others with a different timbre. An echo, or delay, on the other hand is when the reflected sound is at a sufficiently large enough interval to be perceived as a separate sound. Rumsey & McCormack state that “echoes may be considered as discrete reflections of sound arriving at the listener after about 50ms from the direct sound” (Rumsey & McCormick, 2006, p. 23). Each of these echoes are audibly perceived as separate arrivals; as discussed in regards to early reflections, the sounds reflected up to around 50ms are not distinguished by the brain as separate sounds.

Before clarifying the difference between echo and delay for the purposes of this research, a discussion of digital delay will be provided to allow for more ease of understanding.

4.1.3 Digital and Tape Delay

The acoustic phenomenon delay (or echo) was re-created initially using tape machines. Allen Strange describes how this can be achieved in *Electronic Music: Systems, Techniques and Controls* (1972). A sound is then recorded on magnetic tape and is the subjected to playback on several evenly spaced playback heads of the tape machine; as the recorded sound comes into contact with each playback head the sound is repeated (Strange, 1972, p. 89). This creates the effect of an acoustic echo, as each repetition sounds like a reflection of the sound.
Adding to this, Fineberg describes the “re-injection loop” that was also used in this tape-loop process of creating delay: “When the input is an open microphone, this is called a re-injection loop, since the play-back is re-injected into the recording through the mixing board: producing a proliferation of sound” (Fineberg, 2000b, p. 112).

Since this time, digital delay has been refined in order to produce the same results in a much easier way. Fineberg states that in the new digital format the tape-loop was replaced by digital delay and the re-injection loop was replaced by echo (Fineberg, 2000b, p. 112). Delay is replicated in a digital context by storing sampled audio directly into a computer’s RAM and then releasing the audio (reading out from stored memory) after a defined length in time, usually milliseconds or seconds (Huber & Runstein, 2001, p. 390).

Huber & Runstein provide a telling statement concerning digital delay that assists in defining the difference between it and echo: “...Different delay ranges can accomplish a wide range of effects... [such as] the use of digital delay for creating such sonic effects as doubling and echo” (Huber & Runstein, 2001, p. 390). Following this statement, it can be said that digital delay is used to produce a number of echoes of the original source sound. Delay allows the reproduction of the sonic phenomenon echo, and all the parameters of the delay can be set to vary the perceived outcome of the echo; the interval between delays, the number of echoes (decay time) and the length of audio that is to be sampled.

A component of digital delay that is also involved in creating echo is the feeding back of the delay’s output into the input to create the echoes. Repeated echoes can be created by feeding some of the delayed signal’s output back into itself, and varying the amount of feedback gain can then be used to control the level and number of echoes (Huber & Runstein, 2001, pp. 392-393).

Through this understanding, delay is defined in this research as the repetition of a sound at a set temporal interval, for a specified length of time. A sub-component of delay is echo, which delay recreates through the process outlined above. Delay is used to create a series of echoes that, with each repetition, usually dynamically decrease over the specified length of time until it stops, maintaining the same clarity of the original signal.
4.2 Composers Representing Reverberation or Delay in Music

As with other acoustic phenomena, delay and reverb can be involved in music through the utilisation of either phenomenon in its natural or electronically reproduced state, or through the representation of it through modelling. Fineberg suggests that the concept underlying these phenomena, which is “a sound copying itself to generate a sonic structure different from the original sound” (Fineberg, 2000b, pp. 112-113), has influenced a number of composers, and in particular spectral composers.

Allen Strange discusses how a composer could theoretically utilise delay in their music, by taking advantage of the possibilities discussed in regards to delay and feedback loops in tape machines and electronic set-ups;

As the composer becomes more and more familiar with delay and feedback loops, he will find that they can be a valuable tool for achieving sound modification. The mixture and superimposition of attack and decay transients will also serve to produce a certain amount of timbre modification if the levels are carefully controlled... Even with the use of only two channels, the maze of multiple attacks, decays, and temporal distortions will seem to fill the sound field with a cloud of sound in which the direction of origin is not apparent. (Strange, 1972, p. 93)

Alvin Lucier has achieved this concept as suggested by Strange, and has also utilised reverberation as a compositional highlight in some of his works.

4.2.1 Alvin Lucier

Lucier’s Use of Delay

Allen Strange, in a discussion of tape delay, suggests that a “composer could utilise even more complex configurations [of tape delay] which also take advantage of different tape speeds and varying distances between the heads” (Strange, 1972, p. 93). Alvin Lucier was one such composer who utilised these complex configurations in his musical compositions.

Lucier’s work The Only Talking Machine of Its Kind in the Whole World (1969) required the use of seven or eight tape machines that together would form a highly complicated multiple delay/loop station. This set-up would be constructed during the performance of the piece – the “performance” being an aural presentation by a person with a vocal idiosyncrasy: Lucier’s directive is “...for any stutterer, stammerer, lisper, person with faulty or halting speech,
regional dialect or foreign accent or any other anxious speaker who believes in the healing power of sound” (Nyman, 1999, p. 92).

The idea behind this work is that the use of the tape delay system would “recirculate” the person’s speech in order to ease any anxiety they may feel by public speaking (Nyman, 1999, p. 92). As is common in Lucier’s works, this piece gives the acoustic phenomenon “delay” a precedent, but mainly in the form of a demonstration – it can’t be said that this is a form of representation.

**Lucier’s Use of Reverberation**

Lucier’s composition *Vespers* (1968) represents the acoustic phenomenon reverberation through a literal demonstration of the phenomenon. It uses the reverberations present in a space as part of an echo-location system, like those used by bats and dolphins. The connection between reverb as an acoustic phenomenon and echolocation in this example is that the naturally occurring sound reflections of the environment (reverberations) are used to create a spatial awareness through the technique of echo-location.

In this work, Lucier seems to strive to educate the performers themselves in their understanding of acoustics – the performers are given echolocation devices, to be used to find the location of the centre of the room. The Sondols – hand-held devices used for echolocation – emit fast clicks that are normally used to help the blind find their way around. These are used to create an acoustic orientation of the performance space for each performer by the clicks being emitted and returned at different intervals, which allows for the size of the space to be monitored (Nyman, 1999, p. 91).

Nyman states that the performer’s task is to take “slow sound photographs of his surroundings” (Nyman, 1999, p. 91). It seems that this piece seeks to create a special focus on the acoustic space in which it is performed, which again shows Lucier’s emphasis on sound and its associated phenomena as the focal point for musical composition.

Alvin Lucier discussed the piece in an interview with Thomas Moore in 1983. Lucier describes how he initially received the inspiration for the use of echoes in his music from Monteverdi, who used it in his composition, also titled *Vespers* (although preceding Lucier’s composition by 373 years). Lucier noted how Monteverdi’s compositional techniques seemed to incorporate the use of the echo; “an oboe plays a theme, and then it’s echoed in another oboe, in a similar instrument” (cited in Moore, 1983).
Upon seeing this notated on a page of music, Lucier then expressed the desire to use echoes “as they occur in nature” (Moore, 1983). This led him to create Vespers, in which Lucier states:

... Players really make sounds that echo off ceilings, walls, floors. And that idea was still an imitation of something else, but a crude imitation of the sound-sending abilities of bats and animals that use sound in a real way, and not in a language – sounds not as language with symbolism, but real sounds that do something. (cited in Moore, 1983)

Lucier also composed a number of other pieces that used reflections (or reverberation) as a focal point, including Reflections of Sound from the Wall (1981), for audio oscillator and moving wall, and Music for Snare Drum, Pure Wave Oscillator, and One or More Reflective Surfaces (1990).

4.2.2 Nigel Westlake

Nigel Westlake (1958- ) is another composer who has used the acoustic phenomenon “delay” in a musical composition. Westlake has utilised the phenomenon in the form of a digital delay station for two of his pieces: Fabian Theory (1987) & The Hinchinbrook Riffs (2003).

Fabian Theory is named after the great Roman dictator Fabius Maximus, who was renowned for using delay tactics in warfare. The piece utilises a digital delay station that delays the sound by 566 milliseconds, to create a multiple layered & rhythmically complex sound (Westlake, 2012).

Similarly to Lucier’s use of delay, Westlake simply uses the phenomenon in his pieces by way of digital signal processing of live acoustic audio – he doesn’t represent the phenomenon. Delay pedals are common used for guitarists in this same fashion, in their typical array of effects pedals.

The resultant concept that flowed into this research from Westlake’s use of delay is to represent delay in musical composition without the assistance of electronic equipment – playing the “delayed” (or echoed) parts instrumentally instead of electronically. This was the idea for the composition created for this research, Marimba Delay, which is further discussed later in this chapter.

4.2.3 Tristan Murail

Tristan Murail is a composer who is most well-known for his work in pioneering the Spectral music movement alongside Gérard Grisey.
Julian Anderson discusses a particular work of Murail’s, *Me’moire-e’rosion* (1976), in which Murail creates a simulation of the re-injection loop discussed earlier. This piece was written for horn and nine instruments and demonstrates a connection to early forms of electronic music through the simulation of delay in its tape-loop context.

Anderson notices a certain periodicity in the nature of the re-injection loop, as a live sound is recorded in tape, passed through several tape-heads, played back after a number of time delays and then sent back to the first tape recorder in order to combine the results with the original sound. This process also gradually disintegrates the sounds, as it progressively deteriorates into noise through the same tape being repeatedly played and copied.

Murail represents this early form of electronically created delay through the use of instrumental and scoring techniques. To represent delay, Murail uses the horn as a soloist and the ensemble as a backing to provide the “delay”. Every sound played by the solo horn is copied by the rest of the ensemble, which at first replicate the notes with mechanical precision in order to convey the concept of electronically reproduced delay. By maintaining the regularity of the phrases the horn creates the musical effect in the ensemble of the music circling around in “ever-closer concentric canons” (Anderson, 1993, p. 321).

In his representation, Murail also aims to stay true to all the sonic anomalies associated with delay in its original tape-loop setting. When a tape is repeatedly copied a “hiss” ensues, as does a “blurred periodicity, and Murail incorporates these elements as highlights within the music” (Anderson, 1993, p. 321). The disintegration of sound into noise mentioned earlier becomes a part of the scoring technique of the piece, as the ensemble not only repeats the melody but distorts it like the tape machine manifestation of the phenomenon.

The horn melody utilises simple, consonant arpeggios and often only one note at a time, as in the beginning of the piece, but the ensuing “delay” from the ensemble’s imitation of the horn’s material turns this into a complicated blur of noise. Even the spaced out single notes in the opening of the piece are transformed by the ensemble into a sporadic sonic structure. The scoring of the ensemble parts use tremolo figures, rhythmic variations of the melody and drastic timbre changes to create this chaotic noise.

Anderson notes that the scoring of the work makes it seem as if the soloist is trying to fight against this process, particularly with the horn’s cadenza at the end of the piece (Anderson, 1993, p. 321). This cadenza of glissandi and melodic leaps using a harsh timbre just create
more chaotic noise. After the cadenza finishes and the ensemble’s imitations cease, the end of
the piece features the noise of the tapes turning in the machine, re-created by the ensemble
through breath exhalations into the instruments. Finally, the “tape-machine” is turned off with
a loud click simulated by the ensemble.

This piece demonstrates the foundational concept of this research, which is representing an
acoustic phenomenon in a musical composition. The method used is imitative in nature, which
suggests that it is an interpolation of the phenomenon’s data being transferred to a musical
setting. This can also be said to be a scale model, as it retains the properties of the original
source as closely as possible. The composition engagingly presents a representation of tape-
delay, and even more than that it represents the whole tape-machine, with all its associated
sounds; the tape hiss, the whirring of the tapes, and the in-harmonic sounds that ensue from
the disintegration of the source signal.

4.3 Modelling Reverberation and Delay for Musical
Compositions

Of the composers under study in this area of research, it seems that the phenomenon of delay
was represented more consistently than reverberation. Alvin Lucier seems to be the only
composer under study having used the latter of the two, and his use of the phenomenon was
not so much a representation as it was the utilisation of it.

Westlake and Lucier’s involvement with delay was also through the utilisation of the
phenomenon, rather than a representation. Tristan Murail presents a work that can be seen as
a representation of the phenomenon, as he uses instrumental techniques to transform the
processes observed within the tape-loop context of delay into musical structures. Murail’s
work demonstrates how a scale model can be created from an acoustic phenomenon such as
delay.

This type of representation, as demonstrated by Murail, has been explored within this
research. The following composition can be seen as a similar form of interpolative
representation to Murail’s.
4.3.1 Delay Composition 1 – Marimba Delay

This is a composition that has been used to demonstrate the principles of this research project. It seeks to represent the acoustic phenomenon “delay” by modelling its behaviour and transferring the relations evident to a musical context in order to generate a musical composition. In this chapter, the realisation and composition of this concept is discussed.

Inception

This piece was one of the earliest concepts in this research when considering putting an acoustic phenomenon into practice as a musical concept.

This concept was inspired by the existing work for marimba: Fabian Theory, by Nigel Westlake, which is discussed above. The premise for this composition stems from the concept of a delay station being modelled rather than actually being used.

In this composition the aim was to represent delay using instrumental and compositional techniques rather than sound manipulation systems. The expectation at the outset was that the work would be a rhythmically and dynamically complex piece performed by a single player. This was planned to demonstrate the performing musician’s mastery over the electronic systems that tend to be utilised in music, as well as the performer’s mastery of the instrument.

The piece was written for 4-mallet marimba, as this was the inspiration for the piece.

Composing Marimba Delay

This is another piece in the research area that required a set of pre-prepared “parameters” in order to compose the piece. Once the parameters of the piece were considered and set, it was only then a matter of applying the parameters to the subject material in order to create the piece.

The initially proposed steps to achieve the creation of this piece involved the following:

1. Deciding on the parameters of the delay (i.e. delay intervals, tempo and decay).
2. Composing a simple melody
3. Applying the parameters of the delay to the melody
4. Writing the new counter-melodies that arise from this
5. Compiling the parts in a score to be read and played by a single performer
Delay Parameters

According to the previous proposed steps, the first was to decide on the parameters of the “delay”. Upon consideration, the following parameters were decided upon:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tempo</td>
<td>( \dot{\text{crotchet}} = 90 \text{ bpm} )</td>
</tr>
<tr>
<td>Time Signature</td>
<td>4/4 time</td>
</tr>
<tr>
<td>Sample length (to be delayed)</td>
<td>1 (crotchet) beat</td>
</tr>
<tr>
<td>Delay interval</td>
<td>1 (crotchet) beat</td>
</tr>
<tr>
<td>Decay length</td>
<td>7 (crotchet) beats</td>
</tr>
<tr>
<td>Decay velocity</td>
<td>Initial velocity = original note, final velocity = pianissimo</td>
</tr>
</tbody>
</table>

The reasoning behind these proposed parameters is such: the tempo is set at 80bpm, as opposed to a faster tempo, to allow for rhythmically complex passages and dynamics to be more easily achieved and so that the delay can clearly be identified by listening audiences. The sample length and delay interval are set at the same parameter, again to make it easier to understand and play – if the sample length was set to a smaller interval, such as a 16th note, it would create larger spaces in the soundscape as there would be fewer notes being repeated. The decay was set to 7 beats so that in 4/4 timing a crotchet played at the beginning of a bar would be repeated for the duration of 2 bars, and then end. Part of the decay parameter is “decay velocity”, which refers to the dynamic decay of the note – this is set so that the initial note is the same volume as the melody note it is repeating, and the final repeated note is played as soft as possible, with the notes in between gradually decrescendo-ing from the first to the last dynamic level.

This process, as with others of its kind in this research project, is essentially a way of creating a type of canon structure in a microcosm of its melodies.

Melody

The next step in the process was to write a simple melody for the parameters to be applied to. This is obviously largely a creative process, as the subject material (melody) could be any number of styles, in a variety of keys and moods. The key of D major was chosen for the piece and a melody was created with the intention of being uplifting and joyful in nature. The melody was also intended to be spacious (not too many notes in a bar) that would allow the delayed notes to be heard clearly without being overcrowded.

It’s worth mentioning that the creation of a melody as part of this process creates a different set of circumstances than if this process were applied to pre-existing subject material. The
scope of this research doesn’t allow for this concept to be fully explored, but perhaps in further research it could be a worthwhile experimentation process. For example, a composer could create a set of delay parameters such as in this piece, then apply them to Bach’s *Prelude No. 1* and write the ensuing counter-melodies. This could potentially create a whole new context for the piece.

**Applying the Parameters to the Melody**

The next 2 steps, once the melody was formed, were to apply the parameters of the delay to the melody and write the ensuing counter-melodies. Firstly, a small section of melody was written and 4 bars of this was used to apply the parameters to as an experiment, to make sure that the rest of the melody written would work well with the parameters. The following is the melody and its ensuing counterparts once the parameters were applied (figures 39 & 40, respectively):

![Figure 39 – Melody from opening four bars of *Marimba Delay*](image)

![Figure 40 – Counter-melodies ensuing from delay parameters applied to melody.](image)

The creation of an amalgamated part was more challenging – there were several ways of writing it and it wasn’t clear which would be the easiest to understand. For the sake of this initial trial, the decision was made to simply write all three parts in separate layers on a single staff:

![Figure 41 – *Marimba Delay* with amalgamated parts.](image)
The reason for writing this way was to clearly show the melody line that was being delayed, so that the performer relied less upon the full written notation and more upon the parameters of the delay, which could make the reading and playing easier.

Notes on Composition

Composing this piece involved a number of issues, some that were predicted and others that were unforeseen. The melody started off being a simple creative process that required no restrictive thought or boundaries – after all, writing a simple melody in D major is a fairly straightforward task. However, as the piece progressed it became apparent that issues would arise once the parameters were applied to the melody. As a result, instead of writing the whole melody and then applying the parameters to it, the compositional method developed in a way that had the parameters applied to the melody every bar or before writing more. It seemed that this method was the most effective, as otherwise an entire melody could be written before it was discovered it didn’t function well musically with the parameters applied to it.

The restriction of this medium is that it is played on a marimba by a single player – this is different to, for example, a piano player who can sound ten notes at one time, as a 4-mallet marimba player can only sound four notes at once. There did exist the option of using 5 or 6 mallets, however as 4-mallet marimba style was the most familiar to this researcher and was also the medium that was nominated for the piece, it was preferable to stick with it (no pun intended.) Combining this medium with the set parameters that were applied to the melody meant that a number of restrictions were placed upon the writing of the melody. For example, the melody (including its counter-melodies) was limited to never having more than four notes at once (double stops) and also has limited range: the intervals of the combined melody and counter-melodies are restricted by the reach of the four mallets.

As such, it became increasingly common in writing the piece to compose a melody with the parameters in mind – the notes being selected for use in the melody were carefully considered, as it became easier to predict the effect they would create once the parameters were applied and the counter-melodies created. The style of in-composition analysis also developed as the piece progressed – rather than analysing a portion of the melody after several bars had been created and had the parameters applied to it, the melody would be evaluated during its construction (with the end-result in mind) and also directly after writing the melody and counter-melodies. It also became more common to alter the melodies significantly after
the process had been applied to it, as it was discovered that some melodies found just didn’t work well.

So in essence, the subject material was manipulated to fit the process that would be applied to it, knowing what the outcome would be from this manipulation. The process itself didn’t actually get changed however, just the material. For example, rhythm placements and durations were chosen that would allow the process to clearly be heard – the first section of the piece is predominantly made up of semibreves and minims, with the occasional pair of quavers to break up the “plodding” effect of having a constant crotchet pulse. The beginning and ending rhythms of a phrase were commonly placed on the first beat of the bar, so that after 2 bars the counter-melodies would be “clear.” This was done so that the next phrase could begin without possible interference. Later in the piece, the counter-melodies were manipulated more by using disjointed and syncopated rhythms – this allowed for the control over where the counter-melodies would be placed.

To evaluate the over-all effect creation of each section, the following criteria were used:

- **Does it achieve a “nice” sound?** A very subjective question, however personal taste was used to dictate the response to this, as well as avoiding chords or tonal clusters that were overly clashing without having a strong tonal pull or sense of a chord.

- **Is it interesting?** Again, a very subjective question, but the compositional process tended to revolve around this idea of maintaining interest. As such, in the case of particular sections lacking the intensity or the harmonic intrigue of others they were changed.

- **Is it performable?** Whilst this is not necessarily an important in regards to the success of the piece, it will determine its use. The work is written for 4-mallet marimba and a single player, so the logistics of the medium have to be considered. These considerations were based on the composer’s own abilities with this style, and as such the playability of the piece was checked by attempting it on the instrument. Certain passages are quite difficult, however they were left in the piece as they would certainly be achievable after a considerable effort in practice.

Another issue that arose from the writing of this piece was the notation style. In composing this piece, the function of applying the parameters to a written melody would be performed and then the counter-melodies would be written in separate lines. When compiling these parts
into the one line (to be read by the performer), a number of difficulties were discovered. Firstly, the original melody line was intended to be very clear and to stand out from the counter-melodies so that the performer could interpret the piece using a clear relationship to the melody. When putting all the parts together, however, it seemed that with certain types of notation the melody would be lost amongst the other notes. For example, the following is a melody line and then the same melody again with piano-style notation – all the notes on top of each other:

![Figure 42 – Marimba Delay; potential method for amalgamation (1).](image)

This was obviously very messy and unclear, so another method was attempted. In this method, all the layers were kept separate by stemming them all separately; however this method was also still very convoluted.

![Figure 43 – Marimba Delay; potential method for amalgamation (2).](image)

A third option was attempted which involved leaving the melody separate but grouping all the counter-melodies. Consider the same example from above, but in this fashion:

![Figure 44 – Marimba Delay; potential method for amalgamation (3).](image)

After consideration of these options and experimenting with trials of each, the final method of the aforementioned methods seemed the most practical and clearly understandable. All the counter-melodies were grouped together, and the melody was stemmed separately. Initially, the stem direction of the counter-melodies & melody were kept pointing away from each
other, in the same direction for the entire piece, however at times this didn’t seem practical. As a result, the stems still point away from each other but they swap directions depending on the notes, to avoid clashes and to make for easier reading. This can be seen in the following example:

![Figure 45 – Marimba Delay; note stems swapping directions.](image)

In addition to this method demonstrated above, to help the melody stand out most of the stems were extended so that they reached beyond the limit of the counter-melodies stems. Also, a ‘melody reduction’ was included on the final layout of the piece, so that the performer could see exactly what melody is being “delayed” in the counter-melodies. This was implemented to also ease any problems relating to understanding the rhythmic placement of the different layers of notes. Here’s an example of the melody reduction in the first few bars of the piece:

![Figure 46 – Excerpt of Marimba Delay, with melody reduction.](image)

In this example it can be seen how the melody reduction is written on top of the fully amalgamated part.

The other aspect of the composition that proved to be a challenge was the dynamics. According to the parameters set, each “delay” needed to be dynamically decreasing for its duration – starting at the same dynamic level as the initial note, and gradually getting softer to a \textit{ppp}. The challenges with this involved both the notation and the playing.

Firstly, the notation was difficult to realise, as there were multiple dynamic changes occurring at the same time for different lines of counter-melody. Trying to notate this in a clearly
understandable way required careful consideration, as conventional methods of dynamic indication proved to be indecipherable in this context - having four or more decrescendo markings layered on top of parallel melodic lines would have been too messy.

As such, the use of an experimental form of notation was considered for this piece, in which the size of the note head is relative to its dynamic volume. This way, individual notes in a melodic line could decrease in volume simply by exhibiting increasingly smaller note heads. The following (figure 47) is an example of this notational technique:

![Figure 47 – Marimba Delay excerpt with diminishing note heads.](image)

The reduction rate used in the above example is: 90%, 80%, 70%, 60%, 50%, 40%, and 30%.

As this method demonstrates the concurrent dynamic layers most effectively it was utilised throughout the piece. The end result differed slightly from the above example, as the last few notes of each phrase tended to be too small, and also the first repetition of the initial note needed to be the same size as it represents the same volume level. As such, the following reduction rate was used in the final piece: 100%, 95%, 85%, 75%, 65%, 55% & 45%. In the middle section of the piece where the delay interval is set to a dotted quaver and there are 10 repetitions (explained in following paragraphs), the reduction rate was modified to: 100%, 95%, 90%, 85%, 80%, 70%, 65%, 60%, 55% & 45%. These two reduction rates can be seen in the following two examples (respectively):

![Figure 48 – Marimba Delay reduction rates (%) of note heads (1).](image)

![Figure 49 – Marimba Delay reduction rates (%) of note heads (2).](image)

Around halfway through the piece a change was implemented – it seemed that whilst a number of areas were still left to be explored in terms of the set delay parameter, another
aspect of the piece’s potential could yet be approached. The piece begins with the delay being set at crotchet intervals which creates a steady and rhythmically “plodding” pulse. In order to liven things up, it was decided that the delay interval would be changed to a dotted quaver (1 ½ beats, or 3 semi-quavers). This instantly achieved an increased feeling of motion, as the delay had more repetitions arriving at a faster rate and it also allowed for more dynamic rhythms that were complex and interesting. Once this rhythmic device was instated, a pedal tonic note was maintained for a few bars in the melody in order to allow time for the new change in the temporal placement of the delay to be made fully aware.

At the end of the piece’s structure it returns to the original melody. Rather than reinstating the original tempo and delay rate, however, the feeling of motion is sustained by performing a metric modulation. The dotted quaver delay pulse translated into the crotchet pulse for the new tempo, creating the sense that nothing had changed. The new delay parameter was in fact the same as the original, but set at the new tempo.

One more parameter change was included; in the final few bars of the piece, it seemed like a longer fade out of the melody was required, so the delay parameter was altered to include a longer decay time. The progression of tempi and delay parameter changes for the entirety of the piece are illustrated in the following list.
A particular tempo consideration of interest is that between sections 2 & 3 (as outlined above) because the tempo is modified there is actually no change in the delay parameter – by playing at a tempo 1.3333 times faster than the previous, it essentially makes the dotted quaver delay interval the new crotchet delay interval. Rather than mapping this out in terms of milliseconds however, it seemed more prudent to keep all elements outlined in musical terms and in relationship with the current tempo – thus, the new tempo is written as \( \dot{=} = 120 \text{bpm} \), and the new delay interval one crotchet beat.

Although these changes in the parameters of the delay were not part of the initial outlined of the piece, the decision to utilise them seems justifiable. Their use enhances the piece in ways that would be unachievable if the parameters were not changed, and the idea of changing the parameters of a delay “pedal” is actually quite common – on stage, you will see many a guitarist riding the pots of their effects pedals as they play in order to create the desired effect.
Other changes that were not part of the original concept were the inclusion of different time
signatures. The piece starts in $4/4$ time, as was intended, but a few $2/4$ bars and a single $15/16$ bar
are also used. The reasoning behind their use is that the delayed notes (the counter-melodies)
often required space at the end of their phrase before implementing a new melody, because
otherwise the countermelodies would play over the top of the new melody and clash with the
new notes. Whilst this is just par for the course using this process, there were a few instances
where the choice between having overlapping lines that clashed, or allowing a few beats of
silence between phrases yielded poor results in either instance. In these cases, one-off time
signature changes to continue the momentum of the piece. For example, in two instances $2/4$
bars were used to avoid both clashing lines of melodies and alternative silence at the end of
phrase.

The $15/16$ bar was used for the transition between one delay parameter and the next, in
conjunction with an unusual melody (between section 2 and section 3 from above). The
purpose of these two tools is to create a seamless transition by way of setting up the new
rhythm whilst in the old tempo and delay setting – by placing the notes of the melody on
certain beats of the bar the counter-melodies create a single rhythm, which is that of a dotted
quaver. At this point in the piece the tempo modulates using the dotted quaver as the
common rhythmic device to pivot on, so by playing only dotted quavers in the first section and
then only crotchets in the second section it creates the feeling that the pulse has not changed.
The reason for the $15/16$ bar is that without it there would be a single $16$th-note gap between
the bars that disjoints the flow of the phrase. By removing that single $16$th-note from the time
signature of just that bar it removes the problem.

Logistically, there were issues involved in the construction of the piece also. It was a relatively
easy venture to write the melody of the piece (despite having to increasingly make changes
and re-write on the run); by contrast, the process involved in writing the counter-melodies was
quite challenging. Each counter melody had to be written out individually in a notation that
made sense, and often two or more of these were put onto one staff to save room in the score
(which involved writing in several “layers”). Once all the counter-melodies were written
separately, they had to be amalgamated into one readable staff – while some of the issues
involved with the notation have been explained above, the compositing of these lines into one
was still quite time consuming and required careful attention. Writing a single rhythmic phrase
of a repeated note (or series of notes) was marginally straight-forward; doing this 153 times
(for 153 bars) was a little tedious; doing that for 6 different staves of music was more than a
little tedious. Then the problem arose of trying to put all the rhythms into one, which was another huge challenge. To summarise – whilst this piece may look easy to write, it proved to be far from it. The creation of the melody took one tenth of the time it took to follow the procedure set forth by the initial steps, and then the tedious work of “fixing” the score that followed blew these other components out of the water time-wise.

It became increasingly evident throughout the act of composing that the piece seemed to lack a strong melodic line with the counter-melodies imposed on top. The many layers seemed to drown out the melody, making it appear more as a complicated mesh of rhythms rather than a single line of melody being repeated. To rectify this problem it was considered that the piece could be modified to be performed by two players instead of one – the first player could play only the melody (perhaps on a vibraphone for contrasting sounds), and the second player performing the accompanying parts; the “delay”. Or, alternatively, the second player could also perform the melody (as was the original intention of the piece) and use the second player to strengthen the melody.

This idea of altering the number of performers again arose under a different set of circumstances; the problem of the dynamics. Throughout the composition of the piece and in particular as the note-head sizes were changed to represent the dynamics it became increasingly apparent how complicated a work it was. It would be enough of a challenge playing this piece on marimba due to the complicated rhythms and the fact that the melody needs to be heard above all else; adding up to four different dynamic levels that need to be played at the same time, each of which is constantly decreasing, exponentially increased the difficulty level.

This issue was pre-empted before the composition began, but in the end this forecast solution didn’t apply. The method conceived to achieve the dynamics was to use four different types of mallets – each varying in hardness. The method proposed that the hardest mallet would be used for the melody line, and then the other mallets would use progressively softer mallets. When playing four-mallet marimba, the mallets are labelled from one to four from left to right. Using the method prescribed would mean that mallet 4 is of a hard density, 3 is medium-hard, 2 is medium-soft, 1 is soft. This potential method is illustrated in the picture below.
The idea behind this method of mallet variation was that mallet 4 would be playing the melody notes, and then mallets 3, 2, & 1 would play the counter-melodies one by one (in the order listed.) This way, the melody would come out strong, and the counter-melodies would get progressively softer.

The problem with this method was that the melody would always have to be the highest note, and in a 2-bar passage the melody could only really ascend, never descend; this is because logistically the highest mallet (mallet 4) can’t really be switched around to take the place of any other mallet. The only way to do that would be to flip your right hand over and play the mallets upside-down, which was not a feasible solution.

As the initial idea seemed to be defunct, the idea of how to best achieve the dynamics was reconsidered. The concept of using diminishing note heads to represent the dynamic decay was utilised, and additional solutions were devised to ensure that the piece was a feasible composition in terms of its performance.

**Results, Reflections and the Final Product**

This piece is no doubt a challenging work, as discussed in many ways in the previous paragraphs. The overwhelming feeling drawn from the process during this composition was that the dynamics would be a problem. Upon completion of the composition it seemed that perhaps the dynamics in the piece would be almost unplayable; with so many layers of dynamics, each operating at different volume levels at different times, it seemed an impossibility to achieve in addition to the complexity of the notes. If a different instrumentation were used, such as piano, there could possibly be more room for this dynamic complexity – however, on a marimba played with four mallets this was less likely.
As a result, a solution was devised to ensure the piece as playable in one form or another. This solution was also used to address a previous issue, regarding the aforementioned requirement of creating a stronger sense of melody. In order to solve these issues, the initial concept of using a single performer for the piece was revisited. The piece has been written in a number of ensemble configurations, so that if the dynamics in this piece were too unmanageable for only one player they could be achieved in another ensemble format. While this piece may be only theoretical (not feasibly performable) it still seems a worthwhile contribution to this research.

The piece is written in a number of different configurations. The piece can be played as a:

1. Solo – with dynamics
2. Solo – without dynamics
3. Duet – one person performing the melody, the other the accompaniment – with dynamics
4. Duet – as above, without dynamics
5. Trio – one melody player, two accompanists – with dynamics

The solo parts both have a melody reduction written into them, as is explained in the previous paragraphs. The “dynamics” listed next to the above listed items refers to the note-head reduction system of demonstrating the dynamics. The duet is scored to have the melody played by one player, and the combined counter-melodies played by the other. This is firstly to make the piece more playable, and secondly, in the event of the performers wishing to attempt the dynamics the execution will be easier and only utilised by a single player.

This concept is further expanded with the Trio – the melody is played by a single player, and the counter-melodies are split into two parts. The two counter-melody parts are scored so that they can be played on a single marimba, as gathering three marimbas is often a rare venture. This layout does result in a cosy fit for the performers though, as all the notes in the counter-melodies are quite close together – this is a result of the melody having small intervals in most places. There is only one version of this ensemble configuration, which includes the dynamics. It was felt that this configuration was the easiest to perform for each player, and as such has the best chance at being able to realise the written dynamics.

As this end result is a variation from the original concept, it should be justified. The justification for this compromise is that the main concept behind this piece is the idea of representing delay in a composition, using a marimba as a medium; this aspect has not changed. The aspect that has changed is the idea of it being performed by a single player, as it
seems this is not necessarily a feasible option. So rather than scrapping the whole idea (or simply labelling it as “theoretical only”) it seemed reasonable to make this compromise in order to maintain the integrity of the initial concept. In this way, the idea will still be expressed – through whatever method and ensemble configuration is best fit to do it.

Obviously another aspect that has been left out of several of the above options is the dynamics – while this is a more significant blow to the concept’s integrity, it seems that the “delay parameters” could be changed to suit this and still adequately represent the concept of “delay”. The parameters would be altered so that the delayed notes would not decrease in volume, but would maintain the same initial volume until the repetitions cease. This would still technically represent a “delay”, just not the typical embodiment of one.

Analysis & Conclusions

The conclusions that have been drawn from this piece are gathered from the act of composing and the analysis of the resulting composition. Audio files of the two solo pieces were also created – with and without dynamics – in order to analyse the piece in an auditory way.

The same evaluative process was used as with *String Resonance* and *Vocal Resonance*, which is by referring back to the research method and criteria for evaluation. This piece presents a viable model of the phenomenon delay. It uses a scale model for its representational method, as the composition uses the parameters of delay in exactly the same way as it would be used in the context of sound production, but uses it in a musical context. This has always been the purpose of this piece; it was designed to imitate, or replicate, the fundamental principles of “delay” without actually using electronic means (a delay station).

There are three primary causal relations evident in the properties of delay as an acoustic phenomenon; the resounding of a tone, the temporal interval between repetitions of the resounding and the decay rate of the sound’s amplitude. These three elements have been correlated with musical devices in order to represent the phenomenon musically. The resounding of a tone is literally represented by resounding a note of the melody and the temporal interval between repetitions is pre-set in the delay parameters to correspond to musical terminology (interval equates to a crotchet pulse at 90bpm). The amplitude decay is equated to dynamic levels and the rate at which this occurs is set to be over a length equal to eight crotchets at 90bpm).
These very literal parallels between acoustic properties and musical devices are designed to make a clear scale model out of the phenomenon. The manipulation of the musical data (creation of the melody) was also used to make the process behind the composition as evident as possible. The opening of the piece in particular was created in a way to immediately make the process as obvious as possible. While there is no guarantee that a listener would be able to identify the process, the fact that the process occurs all the way throughout the piece does help to make it more obvious, as well as the opening and closing statements of the piece which utilise minimal notes to make the delay more prominent.

The scale model method and these extra considerations on how to make the process more obvious allow the composition to facilitate the reading off properties of the phenomenon. This piece also approaches the modelling in a simple way without over-simplifying it and uses musical terminology that relates well to the original acoustic context of the phenomenon in order to maintain the truth value of the representation.

The ease of performance of this piece is difficult to ascertain as at the time of writing the resources were not available to have the piece performed. However, it’s likely that certain elements of the composition are “theoretical only” whilst others are more achievable. The dynamics, as previously stated, make the solo part extremely difficult and would be the main aspect of the composition that could potentially make it theoretical only. However, as this potential issue was pre-empted towards the latter stages of the compositional process the eventuality of the piece being un-performable was avoided by arranging different versions of the song. Whilst the solo part – even without dynamics – may be too difficult, the duet or trio will be playable as they are much easier. So the response to this point is that the piece will not be strictly theoretical, and will be able to be performed.

The modelling of this phenomenon could prove to allow a significant growth of this area of composition. The points that make this an area that has a great deal of room for growth is that the variable parts of the compositional structure have each only had one input explored. The variables referred to are:

- Instrument choice
- Playing style of instrument
- Ensemble size & arrangement
- Length of composition
- Key signature
• Time signature (whether static or changing)
• Delay parameters
• Number of delay parameter changes

Each of these variables has only had one input explored in this piece, which means there are a multitude of other possible combinations. To elaborate, the current values associated with these variables for the piece are:

• Marimba
• 4-mallet style
• 1 marimba (or 2 for alternate arrangements)
• Length: 6 minutes, 26 seconds
• D Major (with a few modulations to B & F# minor)
• 4/4 time (with modulation to 2/4 and 15/16)
• Delay parameters: 1 crotchet beat delay, decay of 8 beats
• 2 parameter changes

If different values were inputted into each of the same variables, one example could be:

• Piano, sousaphone & zither
• Trio
• 56 minutes
• B major, modulating to E major, then to C# minor
• 7/8, 4/4 and 12/8
• Delay parameters: 1 quaver beat delay, decay of 21 beats
• 3 parameter changes

These changes are also still just adjusting the current variables; aspects that were considered but not explored include:

• Turning the delay “off” & “on”
• Adjusting the delay parameters over a number of bars – i.e. changing a parameter from one beat to the next in a sliding scale
• “Passing” the delay to another player
• Having multiple instruments use delay
There seems to be a great many options with this concept. To take one step further back: as previously stated this concept, even with all the possible variations, is only half of what’s possible as it all involves interpolation rather than extrapolation. The possibilities are opened up even wider if extrapolation is used to input new data gathered from the source material, which is discussed further in the following section.

**Summary of Conclusions**

In summary, this piece seems to successfully demonstrate the principle of representing acoustic phenomena through the modelling of behavioural characteristics in the specific context of using delay. The piece met the majority of the Criteria for Evaluation points with a positive response, which means that the work demonstrates the use of a viable model.

### 4.3.2 Creating a Functional Model of Delay

A model for the acoustic phenomenon delay can be created through an understanding of its fundamental behavioural characteristics in an acoustical context. To describe the nature of delay, it can be said that a sound is “received” by an acoustic space or a digital delay unit, which in turn repetitively reproduces the original sound at a set temporal interval for a certain length of time. The reproduced sound dynamically decays over the length of the repetitions.

To demonstrate this in a flow diagram, delay can be seen as follows:

This representative flow diagram can then be systematically reduced to the following function:

\[
\text{Trigger (A)} \rightarrow \text{Receptor (B)} \rightarrow \left(\left[A, Y\right] - Z\right) X
\]

In this function, A= sound/trigger, B= receptor, Y= separation interval, Z= decay rate, X= repetitions.

To demonstrate the function of the model the components can be replaced with the data from the acoustical context of delay. For example: the note D is played by a trumpet (A) at a sound level of 60dB (though a microphone) into a digital delay station (B), which in turn reproduces the note D repetitively every 1 second (Y) for a duration of 10 repetitions/seconds (X). The note decays dynamically by 6dB (Z) every repetition until reaching 0dB.
We can then replace the components of the function with different musical material. One example would be: A flute plays the note A 440Hz (A), which is then “received” by a pianist (B). The pianist reproduces the note A and uses the “separation interval” as a pitch interval of a major second, the note B (Y), by playing both notes simultaneously. This is repeated 25 times (X) with each repetition dropping in pitch (decaying) by 5Hz (Z).

Another example could be: a rhythmic phrase is played on a violin (A). This is then “received” by a snare drummer who reproduces the rhythm 3 times (X) with no separation between repeats (Y=0). With each repetition, the note values “decay” by half their value (Z= ½) resulting in the following rhythm:

Figure 51 – Using model of delay to generate rhythmic data.

In the example above (figure 51), a rhythmic phrase (left) generates a new rhythmic phrase (right) according to the data inputted into the function (above).

These two examples can perhaps allude to a far greater potential with this method of composition. As was the case with the previously discussed compositions in this research, the interchanging of the various components of the phenomenon’s model/function potentially allows for a great many permutations that could result in a sizable repertoire of works.

However, as was evident in both String Resonance and Vocal Resonance, the use of extrapolation as a data extraction method in this area means that the process is obscured from the observer. This seems a typically inherent feature of extrapolation, as evidenced in these compositions and through the general observation of this method within other art forms.

In referring this model back to the criteria for evaluation, this model seems to present an adequate representation of delay. A point-by-point correlation is made between the primary characteristics of the acoustic phenomenon and the elements (characters) of the function. The values observed in the phenomenon are represented in the model; these values include the number of repetitions of the original sound (represented by the character X), the temporal interval between the repetitions (represented by Y) and the rate of decay of the repetitions (Z).
There are perhaps elements of the phenomenon delay that aren’t represented in the function. In discussion of phase shifting, the term “multi-tap” delay was used, which was pulled from its context of use in delay. A multi-tap delay creates numerous delay lines, so that various sets of delayed signal are occurring simultaneously (in discussion of phase shifting, a parallel is drawn between this concept and the use of polyrhythms, tempi canons and other musical devices).

The model given above doesn’t represent this manifestation of delay, but it could be easily manipulated to do so. For example, the third portion of the function could be multiplied for as many delay signals are desired:

In this example only two delay lines are presented, with the different variable denoted by $X_1$, $X_2$, but this trend could continue for as many delay lines are present. The number of delay lines used will also be dictated by the number of instruments used as value B, the “receptor”. If multiple delay lines are being used, logistically the receptor will need to be an ensemble that contains enough instruments to play every delay line.

When musical components are substituted for the elements in the original function that are closely related to the properties of the phenomenon, the model can be used as a scale or interpolative model. This could be done, for example, by substituting the function A with a note or phrase played by an instrument, substituting function B with an identical instrument to create the product, which is a number of temporal repetitions ($X$) separated perhaps by an interval of 60ms ($Y$) that each decay dynamically perhaps 3db ($Z$). In this example, all components of the function are of similar relation to the properties observed in the phenomenon, which allow the ensuing composition to be a “reading off” of the values of delay. According to Black (1962) this is what defines a successful scale model.

This model can also be used as an analogue model, as is demonstrated in the musical example given above. In this case, the causal relations evident in the phenomenon are transferred to the model, which can then be used to draw similarity relations to a musical setting. As was the case with the model of phase shifting, in order to preserve the “truth value” the terminology used is confined to common musical elements. These aren’t re-listed in this section, however a composer using this model will understand through the information presented in this research.
that the elements to be substituted for function values are of a similar variety. Even within this list, however, there exists a huge range of potential – as is outlined above.
5.0 Sound Waves and Sine Waves

The modelling of acoustic phenomena can be seen as an attempt to produce a focus on sound itself and to increase the perceived value of the pure sonic nature that is inherent in music, the environment and even the human body. This value is considered paramount to certain composers; this is because, as Edgar Varese states, “for us as musicians, sound is one of our best teachers. That is why we should observe it and study it: its lessons are inexhaustible” (cited in Anderson, 2000, p. 11).

The most basic concept of sound production is the sound wave, although it could be said that an even more basic form of sound is the sine wave, as multiple sine waves (simple harmonic motion) can be superimposed to form a sound wave (complex waveform). Sound waves are defined as acoustic phenomena in all the contexts in which they are manifest. The sine wave may be contentiously argued to be an acoustic phenomenon, even though it is primarily generated artificially. In this research, the sine wave is defined as an acoustic phenomenon; this aligns with the viewpoint expressed by Blamey, that it fills the role of an acoustic phenomenon due to the fact that it holds a dual position as the “most basic unit of sound in and of itself, and also the component parts of more complex and meaningful musical sounds” (Blamey, 2008, p. 5). Fineberg also adds that sine waves are an ideal medium into which to decompose sounds and an ideal building unit for constructing sounds, as they are the “only periodic wave-forms whose spectra contain only the frequencies of their oscillation” (Fineberg, 2000b, p. 84).

The sound wave makes a good model for musical compositions, although composers rarely use the sound wave itself as a model; more commonly, composers use the concept of sound as a guiding factor in their music. Their model of sound is used to steer the music away from becoming too entrenched in the rules and stipulations of the harmonic practices prevalent in their cultures. This is often done by opting for simplicity and beauty over convolution and contrived structures. Minimalism as a style certainly developed from this ideal, as these composers were interested in “bringing music back to a more elemental foundation, freeing it from the accumulated weight of Western conventions” (Morgan, 1991, p. 424) in order to bring the focus back to sound itself by “[cutting] down the area of sound-activity to an absolute (and absolutist) minimum” (Nyman, 1999, p. 119).
Following the definition of these phenomena and a discussion of the composers who have incorporated them into musical works, a model has been created for the purposes of representing the sine wave in a musical composition.

5.1 Definitions of Sound Waves and Sine Waves

Sine waves can be seen as the artificial production of simple harmonic motion, and also occur naturally in the striking of a tuning fork. The interaction of large-scale atmospheric conditions of the Earth, the planet’s spheres, also manifest in waveforms. These different manifestations of waveforms have all been represented by composers through a variety of different methods.

5.1.1 Sound Waves

Following Aristotle’s initial discovery that sounds are carried to our ears by movement of air, it was Leonardo Da Vinci who then discovered that sound moves in waves. Later, Ohm presented the idea that the notion of a “sound wave”, which were characterised as regularly fluctuating disturbances or vibration in the air, could be graphically represented as a complex periodic waveform (Blamey, 2008, p. 29).

Sound can be defined as an audible vibrational disturbance – and as demonstrated by numerous people throughout history, these vibrational disturbances that can be transmitted through air, gas, water or solids travel in the form of a wave. Holland, in *Sound Waves and their Properties in Surrounding Media*, says of sound waves:

> A sound wave is produced by a disturbance in the air or other medium. Normally, a sound wave consists of particles or molecules which vibrate within a narrow boundary. The vibrating particles interact with nearby particles producing various collisions which result in the propagation of the wave. The averaged vibrational motion of the displaced particles in a sound wave creates a uniform pressure disturbance which results in the vibration of the wave. The wave vibrations in a normal sound wave in air propagate outward in all directions, forming a succession of concentric spheres. A normal sound wave in air propagates at the average speed of sound, or 1100 feet per second. (Holland, 1985)

Sound has been a principal guiding factor for many composers. Of course, music is sound – or “organised sound”, as originally defined by Edgard Varese (Varese & Wen-chung, 1966, p. 18) and later by John Cage (Cage, 1968, p. 3) – but this is not necessarily a related issue in the
context of composers shaping their music. Often composers get caught up in sound as notes, rhythms, phrases, key signatures and all other elements described in musical terms. Other composers seek to define their music through the over-all sonic effect, and shape it as a mass of sound that can be manipulated to create a homogenous creation of acoustic properties.

Composers of the spectral music style (discussed in the chapter on synthesis) have developed this form of compositional ethic. Concerning his own methods, Tristan Murail states that his material is not “a musical note, nor even a sound, but the sensation (sentiment) created by that note or sound. The material is not, for example, the harmonic spectrum (an object), but the harmonicity of that spectrum” (Murail, 2005a, p. 149). Predating Murail’s sentiment, Claude Debussy can be seen focusing on the sonic properties of his works; Debussy achieved this through the elongation of harmonies, which allowed the component sounds to be more fully appreciated, rather than following any finely determined musical process.

This area of representation of acoustic phenomena is perhaps less tangible than others, as it becomes more of an abstract concept and guiding principle than a process that can be qualified or examined. It may perhaps be misleading to title this discussion “sound waves”, as the composers mentioned herein are not focusing so much on the concept of what a sound wave is or how it functions and how to represent it in a musical setting, but moreover they are focusing on how to create a unified sonic structure, using sound itself as a guiding example.

### 5.1.2 Sine Waves

Sine waves have been utilised in a number of compositions. Peter Blamey, whose study is focused on the use of this phenomenon by composers as half of its content, states that “unlike the naturally occurring acoustic phenomena, the sine wave represents a specific model of sound; one that is reductive, abstract, theoretical, and, in its technological manifestation, synthetic” (Blamey, 2008, p. 18). The following diagram gives an illustration of a basic sinusoidal waveform.
As stated previously, sine waves are so named because their amplitudes correspond to a trigonometric sine function. Sine waves are commonly thought of in regards to their use in sine wave generators and particular sounds produced by synths, and are in fact the embodiment of a pure tone – “simple harmonic motion.” Scott Stark describes how simple harmonic motion is evidenced in the sound of a tuning fork, which has been used for centuries in tuning musical instruments. The resulting waveform of this simple harmonic motion, using Ohm’s principle of graphically representing these sound phenomena, is a sine wave (Stark, 2000 p.23).

Eli Maor, in *Trigonometric Delights*, also states that “the simplest musical tone is a pure tone; it is produced by a sine wave” (Maor, 1998, p. 207). Added to this, Allen Strange’s text *Electronic Music* contains a thorough explanation of some basic concepts of the understanding of sound, including an overview of the four basic wave forms: the sine wave, sawtooth wave, triangle wave and square wave. This explanation is accompanied by an image of the four waveforms represented visually with their harmonic content.
Sine waves are also used to demonstrate auditory principles, such as beating and interference, and can also be used to synthesise complex waveforms (discussed further under synthesis). Some composers, such as Lucier and Young have also used sine waves for these purposes in their compositions.

5.2 Composers Representing Different Waveforms in Music

The waveform has been represented in music in a variety of different ways. Sound itself often becomes a focus in the musical works of composers, which can be seen in the work Debussy. Debussy draws out the sounds of his works to achieve a focus on the music’s sonic values through his stable treatment of tonality. He uses blocks of static harmony, that whilst being connected create a stability through a lack of movement – each piece of harmonic progression can be appreciated as a singular moment in time, with the quality of the sound being foremost rather than what follows or precedes it. This can be observed in works such as *La Mer* (1905) and *La Cathédrale Engloutie* (1910). Debussy’s parallel chords were used to similar effect, as they moved a singular sonority across different harmonic foundations and allowed the mixture of tones to be appreciated fully before moving onto a different set of harmonies. Debussy’s work can be seen as early form of representation of sound itself in music.

Even waveforms not typically associated with sound have been utilised by composers as stimuli for representing in musical compositions. These types of compositions represent inaudible acoustic phenomena. For example, in *Music for Solo Performer* (1965), Alvin Lucier’s stimuli for his composition are the alpha waves generated by a performer’s brain. EEG electrodes are attached to the performer’s forehead and the alpha waves detected are amplified, which creates an electrical signal used to move percussion instruments on the stage. Like many of Lucier’s works, *Music for Solo Performer* seeks to make audible that which is typically inaudible – the speakers that are part of the elaborate set-up amplify the sounds of the alpha waves in addition to using them as triggers for other instruments, and the alpha waves also trigger recordings of sped-up alpha rhythms. Michael Nyman says of Lucier that his works are designed to explore sonic environments, of which Lucier has chosen for his piece;

The waveforms inherent in the spheres of the Earth have also been used as a model for musical composition by composer John Holland. The four spheres of the Earth – the lithosphere, hydrosphere, biosphere and atmosphere – have inherent wave motion within
them. These manifest as tidal waves, earthquakes, low and high pressure systems, and even
down to micro scale waves, such as in the formation of a crystal, or the concentric circle waves
from a splash in a puddle.

In addition to the discussion of sine waves and sound waves the concept of representing the
waveforms of the spheres will also be included. These waveforms demonstrate the properties
associated with the production of sound and therefore can be defined as acoustic phenomena,
although their frequencies cycle at a rate too slow to be perceived by the human ear. This
obstacle is overcome in the compositional process, which will be discussed shortly. The sonic
occurrences inherent within the spheres of the Earth are also classified by John Holland as
acoustic phenomena, who subtitled his work *A Musical Simulation of Acoustic Phenomena.*
This is one of the rare occurrences within this research area in which a composer has used a
term analogous with *representation or modelling* with *acoustic phenomena* in discussion or
referencing of a musical composition.

The use of brain waves is excluded from this discussion as this phenomenon doesn’t fit the
criteria of a true acoustic phenomenon as it doesn’t display the qualitative or quantitative
properties associated with sound or its production in the most obvious way.

### 5.2.1 La Monte Young

La Monte Young is a composer that has contributed significantly to this field of research. Many
of Young’s compositional techniques are used to represent pure sound, and he’s also known
for using a variety of acoustic phenomena in his compositions as well. Young can be seen
representing both the sound wave and sine wave in his compositions through a variety of
means.

Young’s perspective of the Western music of his time was that it had come “full-circle”, and
perhaps also drew influence from Eastern music more than was generally considered. A great
deal of Young’s work would come to consist of long tones, drawn out sounds, and sound
installations. This idea of stasis is something he noted in the works of Western composers such
as Webern as well as composers from outside this tradition; Nyman quotes Young on this,
saying that “climax and directionality have been among the most important guiding factors,
whereas music before that time, from the chants, through organum and Machaut, used stasis
as a point of structure a little bit more the way Eastern musical systems have” (cited in Nyman,
Sound Waves

In discussion of Young’s work *The Tortoise, His Dreams and Journey* (1967), Nyman makes several statements concerning the connection between his compositional techniques and their intent. Even in this one piece, La Monte Young is endeavouring to achieve a realisation of sound and acoustic phenomena. Nyman points out that Young was particular about the use of special methods of vocal production and the use of amplification; “The singers produce throat tones and nose tones, of which the latter are much closer to a simple wave structure, having fewer harmonics than throat tones” (Nyman, 1999, p. 122). This focus on sound is apparent in a number of Young’s works.

An important point that Nyman makes concerning Young’s work in *The Tortoise, His Dreams and Journeys* is that he is attempting to create a work that is focused on sound itself. Nyman states that this piece is a genuine “timbre composition” and that it is an example of a composition that is based purely on sound through a timbre based on a fundamental tone and its component partials (Nyman, 1999, p. 122).

Along the same vein, Nyman states that Young’s treatment of his amplification systems are aimed at emphasising particular elements of sound. He describes how they are built or adapted to his special requirements and they are used not for the purpose of amplification but as a way of highlighting particular aspects of the sound. Nyman states that in Young’s works often the “extremely high level of amplification is necessitated by the acoustic phenomena Young wishes to bring about” (Nyman, 1999, p. 123).

Robert P. Morgan observes the same ideology in Young’s works. In his discussion of *X for Henry Flynt* (1960), *String Trio* and *Composition 1960 No. 7* (1960) he draws attention to Young’s desire to make the audience aware of the presence of acoustic by-products. This (to again use Gann’s) “meta-music” is only achievable through the sustaining of either a short repeated musical idea or single notes. Through the repetitions or extensions of limited sound material Young was able to make audible the seldom heard elements of sound; the “subtle acoustical events (including the minute variations resulting from human performance) that necessarily remain imperceptible when sounds appear as only passing instants within larger successions” (Morgan, 1991, p. 425). Morgan’s observation helps highlight the fact that La Monte Young seems to consistently try to draw the audience into the inner world of sound.
As well as highlighting Young’s desire to convey acoustic phenomena in his compositions, Nyman also mentions his “…continuous practical research into certain psycho-acoustical phenomena… the sustaining of a select band of sounds over extremely long durations; the introduction of constants (the drone for instance); the establishment of an unbroken continuity, which is entirely filled with sounds” (Nyman, 1999, p. 123). Some of these intentions were fully realised in his installation work titled *Dream House* (1974).

La Monte Young grew up as a Mormon in a log cabin and his humble beginnings proved to be an inspiration in his musical career. He first began experimenting with long tones on the saxophone. In an interview with William Duckworth (transcribed in *Talking Music - Conversations with John Cage, Philip Glass, Laurie Anderson, and Five Generations of American Experimental Composers*, 1995), Young speaks of his fascination with long tones. Young was inspired by the concept of long tones and the pitch relationships evident within them; he expressed that his interest in this kind of pitch phenomenon was akin to an interest in “vibration on a higher level” (cited in Duckworth, 1995, p. 218).

Young’s use of sustained tones can be seen as a representation of sound itself – from natural sounds (such as those produced by the wind in his log cabin) to the electric sounds of the humming telephone poles. His piece *Vision* (1959) contains long tones produced by classical instruments, which seems to be an attempt to represent the static sounds of nature, and his *Composition 1960 #7* – consisting of two simultaneously sounded notes (B and F#) that create a perfect fifth, “to be held for a long time” – are literally representing the purity of a sine tone by its prolonged use.

**Sine Waves**

La Monte Young’s use of the sine wave was a significant body work. It could be said that the beginnings of this use of the sine wave came about through the use of the hum of electricity in his compositions.

Young’s performing group the Theatre of Eternal Music used what they called the “turtle motor” – the motor from an aquarium filter attached to a contact microphone – as a constant drone in their pieces. As the power supply they consistently used for their amplification systems was set to 60Hz, Young set the turtle motor to a 120-cycle so that it would stay in tune with the hum produced by the power supply through the amplification systems. The group then also began tuning their instruments to the power supply hum, so that they would be in
tune with the drone of the motor and the power supply. Peter Blamey connects this method with the practice of representing a type of acoustic phenomenon:

This technique of incorporation not only yielded practical results; it also had important conceptual ramifications. Designating the system hum as the fundamental drone frequency had the combined effect of removing noise from their music and of ‘musicalising’ the sound of electrical hum, and therefore by extension the continual hum of the modern electrified environment...The ever-present hum of the wires from Young’s youth now resurfaced in the ever-present hum of the 60 Hz electronic drone. (Blamey, 2008, p. 130)

This is a clear demonstration of how Young utilised and represented an acoustic phenomenon – the hum of the power lines that he grew up with – into his musical composition. This concept was furthered with his involvement with more in-depth use of sine tone generators.

One of La Monte Young’s first uses of the sine wave in an installation was in Drift Study (1964). In this work, two or more oscillators were routed through amplifiers to loudspeakers. These Oscillators were then allowed to “drift” in pitch, which created a sense of the waves moving around the room (Lucier, 1998, p. 8).

_Dream House_ was a concept of Young’s in which a particular space would be utilised as a sine-tone installation – the first of these installations ended up being in his residence with his partner Zazeela. In this installation, sets of frequencies were played 24 hours a day and the effects on people – themselves as well as guests who would come join the sound experience – were catalogued and used as momentum for their next era of composition.

Peter Blamey connects _Dream House_ with _The Tortoise_, in that it “continued Young’s programme of ‘musicalising’ the sound of ambient electrical hum that he had pursued in The Tortoise, His Dreams and Journeys” (Blamey, 2008, p. 142). Before establishing the _Dream House_, Young began with public sound installations with varying numbers of sine tone generators: the Robert C. Scull Commission (1967), the Claes and Patty Oldenburg Commission (1967) and the Betty Freeman Commission (1967). The first of these used fourteen densely packed sine tone pitches, then second used only four, and the final installation used only two sine tone pitches fitted inside a plexiglass light box (Blamey, 2008, pp. 142-143).
Young often used the phrase “listening inside a sound” (Blamey, 2008), which seems to be the epitome of appreciation of sound for sound’s sake and created in him the drive to produce works that featured sound in its simplest forms – whether this took the form of just playing a sine wave tone, or representing a long tone with acoustic instruments. Peter Blamey states that for La Monte Young, “individual sounds were required to be lengthened in duration to the point of appearing continuous, and combinations of different sounds have to be sustained almost to the point of stasis, so that the harmonic content of the sound could be perceived” (Blamey, 2008, p. 111). This perspective helps confirm the view that Young is a significant contributor to the field or representing acoustic phenomena in music.

5.2.2 Alvin Lucier

Lucier was known for his use of the sine wave in his musical works, either as a constant tone to accompany other musical facets, or as an individual sonic entity to be focused on, or as a way of eliciting other acoustic phenomena in the sound space of his compositions. Lucier often used sine waves in the latter method, creating such phenomena as standing waves and interference patterns, which he placed in either performance pieces of sound installations (Lucier, 1998, p. 8).

*Seesaw* (1983) and *Crossings* (1982) were two such works that used sine waves to create beating patterns –these works are discussed further in the chapter in interference/beating. *Queen of the South*, discussed previously in the introduction, was a work of Lucier’s that initially used sine waves to resonate particular surfaces in order to create visual disturbance in particle substances sprinkled onto the surface.

Part of Lucier’s reasoning behind the use of sine waves was the purity of their sound; “Because of their purity, the crests and valleys of their waves can be perceived in the same way as one can see the nodes and anti-nodes on a vibrating violin string” (Lucier, 1998, p. 8).

One of Lucier’s many works involving sine tones was *Still and Moving Lines of Silence in Families of Hyperbolas* (1973-1974). In this piece, Lucier constructs a sound environment using sine wave oscillators, which were distributed to an array of loudspeakers positioned within the space. These oscillators established the sonic context within which the performances of the piece were to take place (Blamey, 2008, p. 219). This sonic context is created by the elicitation of standing waves through the refraction of the sine waves against reflective surfaces. Once
the sound environment is created, the performers are then asked to respond to the waves and reflections moving through the environment.

The piece was written for singers, players, dancers and unattended percussion. The dancers are instructed to discover the troughs of quiet sound that exist in the phenomenon of standing waves, by moving slowly and carefully, and then to follow them until they either met with a sound peak or an intersection with another quiet trough (Blamey, 2008, p. 222).

The singers and instrumentalists actually perform the task of representing the sound properties that exist in the created environment. The singers are instructed to produce long pure tones in unison (or close to) with the sine tones, which create variable patterns of rhythmic beating by their voices interacting at different tempi to the sine tones. The wind and brass instrumentalists are also instructed to produce long pure tones that are slightly off being unison with the sine waves, but they are also asked to “explore the relationship between their own pitches and that of the sine tones in order to spin crests and troughs of sound around the performance space” (Blamey, 2008, p. 223).

The “unattended percussion” take the form of snare drums on stands, which are used to highlight the existence of the acoustic beating and standing waves. This occurs through the sympathetic vibration of the snare drum skins in the event of a high amplitude sound trough, which in turn vibrates the snares on the underside of the drum. Interestingly, this occurrence in an orchestral setting is usually an annoyance to other players, and percussionists are frequently found running back and forth to turn the snares on and off – in this piece of Lucier’s, this phenomenon is highlighted in an unconventional way and is in turn used to highlight other existing acoustic phenomenon occurring in the performance.

This piece demonstrates Lucier’s propensity for utilising sine waves in a composition. In this work he uses the oscillators to also stimulate other acoustic phenomena such as beating; Lucier’s use of this phenomenon in particular is discussed in a later chapter. The use of the sine wave was quite common in a large number of Lucier’s works. To name a few: Kettles (1987) for five timpani and two pure wave oscillators; Q (1996), for quintet and pure wave oscillators; Wind Shadows (1994) for solo trombone with closely tuned pure wave oscillators; Music for Accordion/Soprano/Baritone with Slow Sweep Pure Wave Oscillators (1993); Septet (1985) for three winds, four strings and pure wave oscillators, plus many more. There are at least sixteen other works of Lucier’s that also use sine waves.
5.2.3 John Holland – Wave forms of the spheres

A fascinating perspective on the representation of acoustic phenomena in music has been taken to the extreme by an American composer by the name of John Holland, who worked with fellow composer Josh Caswell to create Voices of Earth – a Global Symphony (2004). This is a composition of epic proportions that seeks to represent the acoustic phenomena that is inherent in the global workings of the earth – from sounds within crystals, to atmospheric waves, seismic waves, cyclones, tidal waves, earthquakes and low pressure systems. Holland states that “sound waves propagate in a variety of media, unaffected by the limits of human hearing, and range from tiny micro-acoustic waves in a plasma, to large-scale galactic waves in the interstellar medium” (Holland, 1985). The presence of various sound waves in the surrounding media produces a continuum of interactive sound events which occur throughout the universe at various orders of magnitude and scale.

The remarkable thing in Holland’s choice in representing these types of acoustic phenomena is that most of these are beyond the range of human hearing. This meant that Holland and Caswell had to translate the phenomena into a perceptible range in order for them to be represented in music.

The work is sub-titled A Musical Simulation of Acoustic Phenomena, and is a computer controlled musical installation that runs for 24 hours – from sunset to sunset. This timeframe, although gargantuan in comparison with more tame musical works, is apparently set in order to illustrate a single moment in time (or, single rotation – a day) in relation to the Earth’s 1.6 billion years of rotation. As a preface to the work itself, Holland & Caswell describe the phenomena they are attempting to represent;

There are four categories of acoustic phenomena that are generated within the various layers or spheres of earth. These sounds vibrate and propagate through the air (atmosphere), liquid (hydrosphere), solid (lithosphere), and organic substance (biosphere). Within each category of phenomena, there are a variety of sounds that vibrate within the range of human hearing. When these sounds enter the ear and brain, we experience the magnificence and wonder of speech and music. (Holland, 1985)

Holland & Caswell reject the idea that sound is limited to something we hear or listen to, as there are numerous sounds that occur on the planet that exist outside the range of human
hearing. Included within this list are acoustic phenomena such as tidal motions, earthquakes and atmospheric low & high pressure systems. These are the types of acoustic phenomena which require transposition into the realm of audible frequency range for humans to hear; some of these frequencies range from millions of cycles per second to a single cycle within years, and as such they needed to be “transposed”.

The composers present around thirty different acoustic phenomena in the work, all of which needed to be transposed into audible hearing range, which generally meant a transposition of eleven octaves upward. As many of the phenomena vibrated at a range of frequencies, a fundamental frequency and high to low range was established in order to convert them into musical tones. The amplitudes of the acoustic phenomena were used to dictate the dynamic level of the notes (larger amplitude = louder dynamic.) This dynamic control was taken to extreme lengths; to represent the tides, a crescendo and decrescendo would last for a total of twelve hours, in keeping with the earth’s rotation. Even this pales in comparison with the “large-scale” phenomena they also included – the earth’s rotation, tides & light-dark periods were represented as continuous sounds that persisted throughout the entire work. On the opposite end, the work also includes “pulses” of small oscillations to represent things like circadian rhythms, brain waves and ocean waves. These are achieved by the inclusion of sustained sounds that are separated periods of silence (Holland, 1985).

This type of representation can be seen as an extrapolation. The majority of the data gathered that is used to stimulate the resulting musical composition does not clearly exhibit properties of either a temporal, melodic or timbral nature, and as such the transference of this relatively unclear data takes a great deal of metaphorical thinking.

5.2.4 Giacinto Scelsi

Giacinto Scelsi (1905-1988) was an Italian composer who is often cited as being a primary influence on the later emergence of Spectral music (Anderson, Fineberg and Mabury). Scelsi’s notable compositional ethic can be elucidated by his quote (from the liner notes for the CD Giacinto Scelsi, 1989) that “He who does not penetrate to the interior, to the heart of the sound, even though a perfect craftsman, a great technician, will never be a true artist, a true musician” (Fineberg, 2006, p. 129). From this bold expression it can be seen that Scelsi was particularly passionate about sound as a guiding principle in musical works.
Scelsi’s initial musical style began in a similar vein to the serial composers, using the twelve-tone technique. Following a breakdown, his compositional style changed drastically. He is reported to have spent countless hours each day during the recovery from his breakdown repeatedly playing a single note on a piano in an attempt to listen inside the sound (Hamilton, 2010). Fineberg states that this apparently brought Scelsi back to the “purely sensual relationship with sound he had enjoyed as a child” (Fineberg, 2006, p. 128).

Scelsi’s piece *Four Orchestral Pieces on a Single Note* (1959) is a hallmark composition that exemplifies his ethos of penetrating to the interior of a sound. This piece severely reduces the possible pitch content that the listener has little choice but to pick up on the subtle nuances of sound that are so frequently missed, such as the harmonics, acoustic beating and difference tones. Scelsi achieves this by using “microtonal and orchestrational fluctuations to colour the single note that dominates each movement” (Fineberg, 2006, p. 129).

Scelsi also achieved similar results with the technique of sustained textures through the use of gradually evolving processes. This is evident in his pieces *Fourth String Quartet* (1964) and *Anahit* (1965). *Fourth String Quartet* uses a steadily rising band of pitches that stretches over the entire eleven minute duration of the work, where the pitches ascend no more than an interval of a sixth during this time. *Anahit* is a concerto for violin with chamber orchestra that seeks to provide an interior examination of the sound of a violin. The drawn-out tones of the violin are echoed by eerie harmonic fluctuations of the orchestra that seem to create a severely elongated version of a violin note. The technique used is similar to the previously mentioned pieces, but the central note that is being coloured in in constant progression. The effect is a strangely haunting sound that may seem the equivalent of drastically slowing down the sound of a violin note, in order to see the inner workings of the sound.

Scelsi’s passionate affirmation of the need to use sound as a driving force in composition inspired the work of Spectral composers such as Tristan Murail. This compositional practice is truly a representation of sound itself in music.

### 5.2.5 Gérard Grisey– Sine Waves

Gérard Grisey, who will be discussed primarily for his use of “synthesis” in compositions, also touched upon the idea of representing the sinusoidal wave in his composition *Vortex Temporum* (1994-1996).
The opening of this piece sounds like an endless “vortex” that spirals away from the listener, undulating in a wave-like motion – it ebbs and flows, and gradually (sonically) recedes. This also seems to have an element of representing the acoustic phenomenon “delay”, as each spiral of sound repeats itself, gradually fading with each repetition. These wave sounds are performed by the arpeggiated patterns that the piano, clarinets, flutes and strings.

Grisey states in the piece’s program notes that the first section of the piece is split into three, each representing different aspects of the original wave sound. These three wave forms are grounded in acoustics: the sinusoidal wave, the square wave and the jagged wave. The square wave is represented by a dotted rhythm, and a piano solo is used to represent the jagged wave.

Of the composers discussed thus far in this section, Grisey’s work seems to be the first that represents the sine wave. Of course Lucier and Young both heavily employed sine waves in their compositions and installations, and also used them as a means to elicit other acoustic phenomena such as beating and standing waves. Neither of these two composers sought to represent the phenomenon as Grisey appears to in *Vortex Temporum*; this seems to be a typical ethos developed within the spectral music style, which is discussed further in a subsequent chapter.

### 5.3 Modelling the Waveform for Musical Compositions

Representing sound, or at least using it as a guiding principle to structure the musical process, has been the focus of composers such as Debussy, Scelsi and Grisey. Grisey also seems to represent the sine wave in his piece *Vortex Temporum*, and Holland’s extrapolative representation of the waveforms of the spheres shows great innovation.

Some of these waveforms could potentially be taken a step further in the representational process through the extrapolation of their data by creating a model. The representation of a sound wave is not necessarily a feasible concept when exploring the method of extrapolation, as sound waves of course vary widely in their content and structure. The waveforms of the spheres have been used in an extrapolative manner by Holland, whose processes seem to outline the way in which this could be achieved by further composers.
The representation of the sine wave is perhaps more straightforward. Although Grisey performed this task in his piece, it seems that this phenomenon could benefit the most from having its behaviour modelled for future use.

5.3.1 Creating a Functional Model of the Sine Wave

It’s possible that representing this waveform could be based primarily upon the mathematical values associated with the generation of a sine wave, or it could be based upon the perceived sonic effect of the sine wave itself. A sine wave is represented graphically (as shown in the beginning of this chapter) as a curved waveform that oscillates from a starting point of zero amplitude to the maximum positive amplitude, to a maximum negative amplitude and then back to zero. This represents the waveform’s amplitude oscillating.

A sine wave could be said to be, in essence, a phenomenon that undulates from an initial starting value to a maximum value, back to the initial starting value, to a minimum value, and then back to the initial starting value. This is illustrated in this basic word-flow diagram:

For example, if the starting value and value of each incremental increase and decrease were all 10, then the function would look like this:

This is a rather simple representation of a sine wave, but could be used to dictate a number of other musical variables. In the context of a sine wave, these values describe the behaviour of the amplitude; a simple transference of this into a musical context would be an interpolation, using the amplitude’s behaviour to dictate the dynamic level of a musical structure.

Alternative representations could be more extrapolative. For instance, the values to be used in the function could be temporal, such as rhythmic values or tempi. The values used could also be pitch related, such as the frequency of a pitch in a melody or a progression of a harmony.
The modelling of this phenomenon can actually be observed in a musical technique. The sine wave’s oscillation from one amplitude to another is referred to as “amplitude modulation”, which will be discussed further in relevance to its application in synthesis. Amplitude modulation is used as a method of vibrato, where the vibrato is achieved through a rapid fluctuation of the sound’s volume; that is, a modulation of the sound’s amplitude. This could be seen as a basic modelling of a sine wave, which consists of an amplitude modulation repetitively occurring in the wave motion at a particular frequency.

The sine wave makes an adequate model due its rather simplistic function observant in its properties. The model created above, based on this simple functionality, demonstrates a direct correlation between the observed behaviour of the phenomenon with each element of the function. It preserves the relative proportions between the relevant magnitudes and does so with minimal deviation. According to these criteria, this model serves its purposes well.

The drawback in the modelling of a phenomenon such as a sine wave is the simplicity of the behaviours presented. While the creation of a model in its simplest format is creditable, the discredit lies in the lack of causal relations in the properties of the phenomenon. Compared with a phenomenon such as phase shifting, there are minimal interactions within the characteristics of the sine wave that can be used to draw similarity relations with a model. The sine wave really only exhibits two elements: the modulation of its amplitude and the frequency at which this occurs. This could perhaps be the drawback to using a stimulus that is the result of a significant “magnification” on a particular subject; zooming in (metaphorically) on the inner functions of a thing can sometimes make it difficult to see exactly what it is that is being observed, or in this case, represented.
6.0 Harmonic Series and Sound Spectra

The harmonic series and harmonic spectra are related sonic occurrences that have been used as stimuli to represent in musical compositions. They both are associated with the production of sound and as such are defined as acoustic phenomena.

These two acoustic phenomena can be used successfully in generating models for musical compositions, which can be seen through the work of composers under study in this research. This area is quite broad, as often the method of representing the harmonic series, or the spectrum of a sound, is achieved in musical works through synthesis. Because of the size of this area, as well as the observable differences between modelling sound spectra and modelling synthesis, these two areas are divided.

The observable properties of a phenomenon such as the harmonic series exhibit a number of causal relations in its functioning. These relations can quite easily be outlined in a mathematical function, which is exactly what has occurred when the phenomenon was first discovered. Any phenomenon that so easily lends itself to the creation of a mathematical function demonstrating its underlying behaviour may just as easily be utilised as a model for which to apply to other contexts.

The work of a number of composers is scrutinised in this section, including the compositional practice and pieces of Debussy, Messiaen, Stockhausen, Per Nørgård and Steve Reich. The creation of a model is discussed following a survey of the works of these composers.

6.1 Harmonic Series and Sound Spectra Defined

These concepts are fundamental to the understanding of music and sound production, and included in the discussion of these concepts are the related terms harmonics, overtones and partials.

6.1.1 The Harmonic Series

The harmonic series (sometimes known as the overtone series) is a mathematical relationship found in sound phenomenon that is defined by an integer relation between a fundamental frequency of a sound and its constituent components. This relationship was discovered at least as early as the ancient Greeks, and was later identified through generating harmonics on a monochord and then through Helmholtz’s demonstration through sympathetic vibrations that
this numerical structure exists within the sound of instrumental notes (Fineberg, 2000b, pp. 85-86).

As with other numerical contexts, the easiest way of expressing simple relationships is in terms of integers or whole numbers, and in the case of the harmonic series the most common type of vibrations occur at whole number intervals to the fundamental tone (Strange, 1972, p. 14). Allen Strange states that “most sounds made by what we call ‘traditional’ musical instruments exhibit a spectrum containing frequencies related by whole numbers” (Strange, 1972, p. 14).

The “fundamental” refers to the lowest frequency evident in a pitch, the production of which generates the ensuing harmonic series. The terms harmonic, overtone and partial are often used confusingly in a description of the tones related to the fundamental, but even though they are often used to describe the same phenomenon each term relates to a slightly different aspect of these generated tones (Blamey, 2008, p. 28).

There are, however, what seem to be a set of common definitions for these terms that are largely agreed upon. According to Allen Strange and confirmed by Peter Blamey, “overtone” refers to any component of a sound that occurs above the fundamental, and an overtone is referred to as a “harmonic overtone” or just “harmonic” if it has an integral (or simple whole number) relationship to the fundamental. The term “partial” refers to the specific spectral content of a sound irrespective of harmonic or non-harmonic relationships (Strange, 1972, pp. 14-15).

Using these definitions, Strange then states that each of these tones can be given a numbered ranking according to their harmonic or partial number. In terms of harmonics, the fundamental can be called harmonic #1, and the next tone above called the harmonic #2, etc. This numbering system then corresponds with the formula that Fineberg presents that is used to determine the frequencies of the harmonics in relation to the frequency of the fundamental. Fineberg states that the equation used to work out the frequencies of the harmonic partials from a fundamental frequency is:

\[
\text{Frequency (of partial)} = \text{Rank#} \times \text{Fundamental (frequency of)}
\]

...where ranking is an integer defining the partial number and both fundamental & frequency are expressed in cycles per second (Hz). For example, if the fundamental is
440 Hz then the second partial is 2 x 440 or 880 Hz, the third partial 3 x 440 or 1320 Hz, the fourth is 4 x 440 or 1760 Hz, etc. (Fineberg, 2000b).

Following the data produced by this equation, Fineberg states that these frequencies can then be expressed as musical pitches (rounding the frequencies off to the nearest quarter-tone) (Fineberg, 2000b, p. 86). This is demonstrated in the following figure, which shows the fundamental as the note C₂ (65.4 Hz) and the harmonic series ensuing from the fundamental, rankings two through ten, with the frequency of each labelled above:

The harmonic rankings as well as the partial rankings are both provided, as in the case of the harmonic spectrum of a particular instrument’s sound these two elements aren’t necessarily the same.

### 6.1.2 Sound Spectra

Pitched sounds are formed by combinations of partials in a harmonic series that relate to the fundamental (the fundamental is primarily identifiable as the note’s pitch); this is referred to as the “sound spectra”, or “harmonic spectra” (Fineberg, 2000b, p. 86). As described previously, the sine wave is a form of simple harmonic motion that contains no harmonics at all, as its wave form exhibits only the fundamental tone.

Instruments of particular sounds contain only selected overtones; for example, a clarinet has a sound spectrum that contains only odd-numbered harmonics, which means it exhibits the fundamental and then the 3rd, the 5th, the 7th, the 9th (etc.) harmonics. In this situation, each harmonic present is labelled as a partial beginning from one; the first partial is the fundamental (1st harmonic), the second partial is the 3rd harmonic, the third partial is the 5th...
harmonic, etc. This is demonstrated in the following figure, showing the odd-numbered harmonics and their corresponding partial rankings.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>65.4</th>
<th>196.2</th>
<th>327.0</th>
<th>457.6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Harmonic #:</td>
<td>1</td>
<td>3</td>
<td>5</td>
<td>7</td>
</tr>
<tr>
<td>Partial #:</td>
<td>1</td>
<td>2</td>
<td>3</td>
<td>4</td>
</tr>
</tbody>
</table>

*Figure 55 – Musical representation of odd-numbered harmonics of the series.*

There are other components that create the particular sound of a pitched note as produced by an acoustic instrument. Each partial has a particular amplitude at which it is produced, and the differences between these (in addition to the partials rankings evident) in the sound spectrum are what help create the instrument’s particular timbre. The emphasis of different partials also varies in different registers, volumes and attacks. Eli Maor, in his text *Trigonometric Delights* (1998) describes how the harmonics of a sound at different amplitudes creates its own unique sound. Maor states that the harmonics are what lend a sound its characteristic “colour”; for example, the sound of a trumpet gains its particularly bright quality through an abundance of harmonics, whereas the comparatively mellow sound of a flute is due to a lack of harmonics (Maor, 1998, p. 207).

Maor provides with this description a useful chart, demonstrating the harmonics evident in a comparison of a trumpet (left) and a flute (right) in figure 56:

*Figure 56 – Acoustic spectrum of a trumpet and a flute (Maor 1998, p. 207).*

Most instruments have been designed to produce sonic spectra that are very closely modelled on a pure harmonic spectrum in order to create clarity of sound and exact pitch. Because of
the physical system used to create the sound, however, the sounds are never entirely harmonic as there are often elements of “noise”, which could be caused by the breath of a wind player, or the scraping of a bow on the string (Fineberg, 2000b, pp. 86-87).

The physical bodies that are used to produce the sound of an instrument, in addition to creating “noise”, also affect the different amplitudes of the partials. When the physical bodies of the instruments vibrate they emphasise certain frequencies and attenuate other – this is called the instrument’s “formants” (Fineberg, 2000b, p. 87). An instrument’s formants are what allow the identification of the timbre, even in situations where the amplitude of the partials vary, such as in different registers or at different volumes.

6.2 Composers Representing Harmonic Series and Sound Spectra

A number of composers have used the harmonic series as guiding model behind their compositions, and similarly have used the analysis of a sound spectrum to dictate compositional elements. For some composers this was achieved through synthesis, and this particular representational tool will be discussed further in the following chapter. This separation of discussion is due to the fact that synthesis can possibly be seen as an acoustic phenomenon in its own right, so even though it is used as a tool for translating sound spectra into musical compositions it can also be used by composers as another stimulus to represent.

The musical style called “spectral music” will be discussed in the following chapter. As may be guessed from the term “spectral,” the initial underlying principal of this style is the focus on sound spectra as a compositional highlight. Often the method of achieving this, as mentioned above, is through synthesis, which deserves its own exclusive discussion.

Some composers present musical works that have a deliberate focus on the overtones present in the music and utilise tools to emphasise these. Other composers use the harmonic series to generate other musical data for their works.

6.2.1 Claude Debussy

A focus on the harmonic series as a stimulus to assist in the construction of harmonic content within compositions appeared at least as early as the work of Debussy. Brett Mabury of the
West Australian Academy of Performing Arts presented a thesis titled *An investigation into the spectral music idiom and its association with visual imagery, particularly that of film and video* (2006). The important point Mabury raises here concerns Debussy, and the connection between impressionism and spectral music; that is, the use of the harmonic series in music.

Mabury cites Peter Platt’s research on Debussy (1995), who stated that Debussy “continually presents sonorities which correspond with the harmonic series and/or segments thereof” (Mabury, 2006, p. 6, quoting Platt 1995). This statement is also backed up by music critic and composer Kyle Gann in an article for the *Village Voice*. Gann compares the work of Tristan Murail in *Terra d’Ombre’s* (2003-2004) to Debussy, stating that “if you take the first 10 overtones you get a Debussyan ninth chord, and the horns played quite a few of these in *Terra d’Ombre’s* background” (Gann, 2004). Gann also quotes the composer Frederic Rzewski, although only through personal communication at a concert, who seems to agree with this sentiment: "Spectral music... always sounds like Debussy" (cited in Gann, 2004).

Mabury presents a musical figure originally created by Peter Platt, who has constructed the notation of the harmonic series stemming from Bb. These vertical chord formations reflect what Platt sees as “Debussyan Sonorites”:

![Figure 57 – “Debussyan Sonorites” formed by B♭ harmonic series (Mabury 2006, p. 7).](image)

Through this figure (57) it can be seen that Debussy’s use of harmony reflects the harmonic series. Peter Pratt, according to Mabury, also analysed a series of Debussy’s works including *Prélude a l’après-midi d’un faune* (1894) and *Trois Nocturnes* (1900) (Mabury, 2006).

This is not to say that Debussy deliberately performed this construction based on the harmonic series, as it was only recently beforehand that Helmholtz had made public the discoveries he had made concerning the harmonic series. However, through a study of Debussy’s musical thoughts and practices (DeVoto, 2004; Howat, 1983) it seems plausible that even without this
knowledge he may have created the effect of the harmonic series through his own aural perception of sound, as the concept itself had existed for quite some time.

6.2.2 Olivier Messiaen

Olivier Messiaen’s use of isorhythms was given as an example of a musical technique that can be seen as a form of phase shifting, and now other techniques of his can be viewed as using the harmonic series to assist in the direction of his music.

Some of Messiaen’s work with this phenomenon seems to be a precursor to the spectral music style. Fineberg states that “Olivier Messiaen was the composer who assisted most directly in the birth of the spectral movement” (Fineberg, 2006, p. 124). Julian Anderson (2000) also draws this same connection and discusses the techniques Messiaen used that were derived from the harmonic series.

Anderson finds a particular methodology evident in Messiaen’s text *Technique de mon Language Musical* (*The Techniques of my Musical Language*, 1942). Messiaen purportedly justifies certain uses of modal and harmonic procedures through “nature”, or the harmonic series; one such technique he defended was the use of the cadence stressing the augmented fourth, which was a common feature of Messiaen’s music (Anderson, 2000, p. 10).

The harmonic spectrum is also used by Messiaen to construct what he called a “chord of resonance”. The fourth to fifteenth harmonics of the harmonic series are superimposed in equal temperament in order to create a “two-octave eight-pitch aggregate” (Anderson, 2000, p. 10). Also on the same lines of resonance was Messiaen’s use of what he called “added resonance” and “inferior resonance”, both of which he also justified through the harmonic spectrum. Messiaen uses the spectrum to add dissonant upper pitches to a diatonic triad in the middle register (added resonance) and dissonant clusters of notes added to a higher diatonic triad (inferior resonance), which was often played at the bottom of the piano (Anderson, 2000, p. 10).

Messiaen’s use of inferior resonance can also be seen as representing the harmonic spectra he observed in various “in-harmonic” instruments. On this, Anderson states that “Messiaen also cites the complex timbres of metal percussion instruments, especially bells, gongs and tam-tams, as the most immediate analogy with this type of harmony” (Anderson, 2000, p. 11). This seems to be where Messiaen’s work starts paralleling the later work of spectral composers such as Grisey and Murail, who involved themselves with instrumental simulation (synthesis).
of spectra generated from non-harmonic sounds, such as bells and the striking of pie tins, the latter of which was featured in Philippe Hurel’s work Leçon de choses, 1993 (Fineberg, 2000c, p. 119).

A specific work of Messiaen’s that has direct parallels with the work of the spectral composers is his *Couleurs de la Cité Céleste* (1963). Through his interest in trying to achieve more resonance in particular harmonies, Messiaen transforms the timbre of a low trombone note played at a *fortissimo* by adding three clarinets playing a higher chord at a *piano* dynamic. Figure 58 shows this superimposition of the three clarinet notes on top of the low D♭ of the trombone:

![Figure 58 - Excerpt from Messiaen's *Couleurs de la Cité Céleste*](Mabury, 2006).

This occurs in several instances using slightly different sources for the clarinet’s notes in each. These higher pitches played by the clarinets are sometimes picked from the harmonic spectrum of the trombone’s note, which tends to create a fused and resonant sound that blends well with the trombone. In other instances, the pitches selected are foreign to the harmonic spectrum of the trombone, in which case they create a distorted timbre when combined with the trombone (Anderson, 2000, pp. 10-11).

This technique used in Messiaen’s *Couleurs de la Cité Céleste* can be observed later in the work of Gérard Grisey, who performed the same process in his piece *Partiels* (1975). This piece and the work of Grisey will be discussed in the following chapter under *Synthesis*.

### 6.2.3 Karlheinz Stockhausen

Karlheinz Stockhausen (1928-2007) has been an important figure in the arena of sound phenomena being associated with music. Robert P. Morgan’s text *Stockhausen’s Writings on Music* includes the bold statement that Stockhausen “occupies an unparalleled position in the contemporary musical world” (Morgan, 1975, p. 1), and it has been validated by other sources
(Anderson, 2000 & Fineberg, 2006) that Stockhausen has played a significant role in the pioneering of techniques, concepts and musical writings that influenced the trends of subsequent composers, particularly those of the spectral music genre.

Stockhausen was a German avant-garde composer who often worked in the field of electronic music and was a particular influence in the development of integral serialism in Western music, although his “inclination to think of music in terms of larger formal units rather than individual elements led Stockhausen away from strict serialism in the middle 1950s” (Morgan, 1991, p. 345). Allen Strange states that Stockhausen’s theories developed at this time were of particular importance for practitioners of electronic music (Strange, 1972, p. 9). Important components of Stockhausen’s musical theories were his consideration of the “vibrating world, the continuum, the spiral and the galaxy” (Coenen, 1994, p. 207).

Stockhausen viewed the whole universe as a constantly changing constellation of relationships, in which the only truly universal constant is the element of “vibration”. To Stockhausen, this means that every relationship can be expressed as proportions of vibrations, which in turn makes it possible to translate the universe into music. Stockhausen stated in an interview with Jonathan Cott (Stockhausen: Conversations with the Composer, 1974) that “every object in the world, down to the smallest atom, produces waves which can be transformed into acoustic waves” (Coenen, 1994, p. 206, citing Cott 1974, p. 75). This concept seems to be mirrored by composers like John Holland, as discussed in the previous chapter, who took this idea and ran with it on a large-scale work representing the spheres of the Earth.

Stockhausen’s view of vibrations as a universal constant was expressed in his view of structures within music. He observed that long durations of time (macro-time) contain musical forms, whereas shorter temporal units contain phrases or motifs and even smaller temporal divisions enter the arena of rhythm. When rhythms are increased to around 18 times a second, they can be perceived as pitch (micro-time), and continuing in this same thought process a higher pitch (or faster rhythm) can then help provide information about a sound’s timbre. This process of thinking culminates in the implied theory that “form, phrase, rhythm, pitch, and timbre are all the workings of a single system – vibrations or variations” (Strange, 1972, p. 9).

Julian Anderson comments on this ideal having antecedents in the work Henry Cowell: “He [Cowell] draws a parallel between the composition of timbres and rhythmic patterns; exactly the same premise, with similar rhythmical results, was to be a crucial building block nearly
forty years later in Stockhausen’s *Gruppen*” (Anderson, 2000, p. 9). This notion of vibration is also discussed in a quote from Seppo Heikinheimo (sourced from Allen Strange’s text):

The musical organisation is carried into the vibrational structures of the sound phenomena. The sound phenomena of a composition are an integral part of this organisation and are derived from the laws of structure: namely, that the texture of the material and structure of the work should form a unity; and the microtonal and macro-tonal form of the work have to be brought into a conformity that accords with the basic formal idea which every single composition has. (cited in Strange, 1972, p. 9)

Stockhausen used the harmonic series and various sound spectra in a few different ways for a number of his compositions. *Stimmung* (1967) is a 70-minute piece written for six vocalists and is built around singing different vowel sounds in a fixed harmonic spectrum (Fineberg, 2006, p. 123). A single harmonic spectrum built on a B♭ fundamental is used as the thematic material, and the vocalists filter this spectrum through constantly changing phonetic colouration, or “overtone chanting” (Anderson, 2000, p. 13). The changing of the vowel sounds is achieved through the singers’ alteration of tongue and lip positions. This process then emphasises individual partials of the harmonic spectrum, up to the 24th partial, making audible the hidden harmonics of the fundamental (Mabury, 2006, pp. 17-18).

In regard to Stockhausen’s use of this technique, Anderson states that “this is a remarkable instance of applying acoustic research to composition in a thoroughgoing and consistent way” (Anderson, 2000, p. 13). Anderson notes the analogous nature of the musical composition with the acoustic phenomenon, and in particular that “the large-scale harmonic vocabulary (a single harmonic spectrum) is identical with the small-scale detail (made up of the individual harmonic spectra projected by each of the six voices)” (Anderson, 2000, p. 13). This statement seems to be addressing the fact that Stockhausen is effectively modelling the phenomenon through correlating relations evident in the phenomena with similar relations in the musical composition.

In addition to his successful use of this acoustic phenomenon Stockhausen has also created a number of other works that involve the use of the related sound phenomena “amplitude modulation” and “ring modulation”. This phenomenon encompasses a large portion of his other works, some of which will be touched upon in a later chapter.
6.2.4 Per Nørgård

Per Nørgård (1932- ) is a Danish composer who has been cited as an early influence on spectral music through his use of the harmonic series in his works. His works, like those of Scelsi, explored different timbres through the use of slowly transforming textures and also utilised cluster forms and multi-layered melodic development techniques.

His works often showed a “fondness for harmonic spectra”, as Anderson puts it (Anderson, 2000, p. 14). In his work *Iris* (1967), several harmonic spectra are superimposed at different speeds. Even more significant was his work *Voyage into the Golden Screen* (1968). This piece is based entirely upon two different harmonic spectra that are a quarter-tone apart; on a G and an A♭ lowered by a quarter-tone (Anderson, 2000, p. 14). The piece is split into two movements, in which the two harmonic spectra are represented by the orchestra through just intonation. The two spectra gradually unfold against each other, and their clashing creates audible difference that resulting in acoustic beating.

At the end of the piece, the clashing of the two spectra culminates in large scale, blurred multiple octave, that also includes noise sounds such as buzzing from the harp, flutter-tonguing from the wind and brass players and varying bow pressures from the strings. These elements of noise are introduced in order to highlight the grainy timbre present from the acoustic beating.

While this contains only a brief discussion of Nørgård’s work, the few pieces mentioned highlight the representation that Nørgård executes of various harmonic spectra.

6.2.5 Steve Reich

The work of Scelsi and Nørgård in slowly drawn out sounds to illuminate certain acoustical by-product such as harmonics and overtones can perhaps be mirrored in the work of Steve Reich. Reich often stated that his compositions demonstrated the idea of “music as a gradual process” (Reich, 1968, 2002), and peer commentary put forth the same observations (Nyman, 1999 & Morgan, 1991). Reich often subjected his musical content to this gradual process in order to elicit the seldom heard overtones and harmonics that the combinations of notes would produce – in the same way that even Debussy had a “concern with sound for its own sake” (Morgan, 1991, p. 46), Reich also wished to make apparent that which would normally be overlooked: the qualities of sound itself.
In reviewing peer comments on Reich — such as those put forth by Michael Nyman (1999), Robert P. Morgan (1991), and Reich’s own self-examination (Reich, 1968, 2000) — this idea of music as a gradual process in his music came up a few times. Nyman points out that Reich uses repetition as a means to “realize his concept of ‘music as a gradual process’” (Nyman, 1999 p.151) and it is this gradual process — almost a slowing down of the music — that allows the base concepts of pure sound and acoustics to be heard in his compositions. The processes Reich put in place, however, were created to be audible to the listener, rather than being obscured by the effects they created. This can be seen as a reaction against the use of indeterminacy by composers such as John Cage (1912-1992), who at times used external elements such as I Ching (the Chinese book of oracles) and coin-tossing to determine the musical structure, such as in his piece Music of Changes (1951) (Morgan, 1991, p. 362).

Kyle Gann refers to these acoustic by-products as “meta-music”. Gann actually claims the term comes from Reich, although as there is no reference this is difficult to confirm:

> For a while in the ’70s it seemed that Steve Reich’s chief preoccupation was the unintended acoustic details that arose (or were perceived) as a side effect of strictly carried-out processes. These included soft melodies created by the overtones of played notes, which Reich referred to as “meta-music,” and even reinforced with notated instrumental melodies in such works as his Octet. (Gann, 2001 p.6)

This “meta-music” was allowed to occur because Reich uses processes to determine the direction of each piece. This means that once the process is set in motion he has no more need to “interfere” with the piece, and as such allow for “uncontrolled results,” or in Reich’s words “details of the sound moving out away from intentions, occurring for their own acoustic reasons” (cited in Nyman, 1999, p. 152).

Paul Hillier describes these events as “resultant patterns”, a term which has been briefly mentioned in discussion of Reich’s phase shifting. Hillier describes how these resultant patterns occur in phase shifting through the interlocking of two models of the same pattern, and how the change in the relationship between these models then changes the resultant patterns. Importantly, he notes that while initially these resultant patterns are unforeseen and are not manipulated in any way, later in his works Reich seeks to emphasise these by-products and control the material in a way to ensure that more are produced (cited in Reich, 2002).
Michael Nyman confirms this view of how Reich’s resultant patterns were initially unforeseen but eventually were musically manipulated in order to enhance them. Nyman states that some of these patterns may be simple musical results such as cross-rhythms and sub-melodies “which because of the phasing process change from one phase relationship to another” (Nyman, 1999, p. 155). Nyman comments on how Reich eventually starts doubling the resultant patterns, which may be considered “sound objects thrown up in the natural process but which have absolutely no existence separate from the flow of the constant rhythmic stream” (Nyman, 1999, p. 155).

One piece in particular that utilises the emphasis of these resultant patterns is Drumming (1971). This piece draws attention to the acoustic by-products caused by complex cross-rhythms which are achieved through simple rhythmic patterns combined in a hocket style (the technique where a melodic or rhythmic line is shared between instruments; while one rests the other plays). Reich creates a focus on timbral limitation in pieces like Drumming through repetition, which in turn leads the listener to focus on something else other than the obvious rhythms and pitches. The repetition of combination tones played on a glockenspiel can create a myriad of dissonant harmonics which are almost painful to hear, but this creates another texture to the music that Reich set out to achieve.

These same effects can of course be heard in Reich’s earlier tape pieces. The by-products that arose from these pieces were often the components of the recorded audio that weren’t intended to be there. In It’s Gonna Rain some pigeons can be heard flying off in the background, and the beating of their wings becomes an integral part of the resultant patterns emerging with the combination of all the other sounds. The verbal transients and consonant sounds from the preacher’s voice are also added into the mix that creates these patterns, and through the phase-shifting process the relationship between all these elements also constantly shifts and evolves. The effect could be seen as similar to a sound from an instrument that had a constantly changing number of overtones.

In effect, what Reich is doing is unintentionally producing these “psycho-acoustic” by-products when he implements his musical processes, and then emphasises them with additional composition. While he doesn’t necessarily seek to create a representation of these phenomena (meta-music, or resultant patterns) his compositional emphasis of them is in a similar vein.
All these occurrences of “meta-music” or “resultant patterns” demonstrate how Reich draws attention to the subtle nuances of sound. While these occurrences might not always necessarily be harmonics or overtones of the sound spectra present, they seem to be quite a similar phenomenon.

6.3 Modelling the Harmonic Series and Sound Spectra for Musical Compositions

It may be said that perhaps Reich’s use of resultant patterns is an extrapolative representation of harmonics and overtones, although again this was initially unintentional as only later did Reich start reinforcing these occurrences through deliberate compositional acts.

The work of Stockhausen and Nørgård could be seen as more of an interpolative representation of the phenomena. Debussy’s work is less definable, as the representation of the harmonic series evident in his works were not necessarily of his own voluntary making, as it seemed to be more a guiding principle in his compositional process.

6.3.1 Creating a Functional Model of the Harmonic Series

In order to create a model for this phenomenon to allow the extrapolation of its characteristics, it may be simple enough to use the existing function for working out the frequency of the partials of a fundamental tone.

This function, provided by Fineberg earlier in the discussion, is demonstrated again here:

\[
\text{Frequency (of partial)} = \text{Rank#} \times \text{Fundamental (frequency of)}
\]

This function can be used in an extrapolative way to create a different outcome than was originally intended. Here the function is re-structured:

\[
\begin{align*}
\text{Rank # (A)} \times f \text{ of fundamental (B)} &= f \text{ of partial (C)}
\end{align*}
\]

This creates the relatively simple formula \( A \times B = C \). Such a basic concept of a formula could of course be used in any number of ways. For example, the number of beats (A) in a given bar of a piece, which shall be 4, is multiplied by the pitch interval of the first two notes of that bar (B),
which shall be an interval of say a fifth (for example, from the C to a G). The result (C) is used to dictate the number of notes in the ensuing bar, which will be 20.

As seen from this example, this is perhaps too vague to be related back to the relationship between a fundamental tone and its constituent partials. Another method of creating a model for this phenomenon is to consider the underlying nature of the phenomenon without using any pre-existing functions. In this case, it could be said that a harmonic series or spectrum is a pitch that is made up of a series of other pitches, that each have a particular volume level.

This new function could be seen as follows:

\[
\text{Sound A} = \text{Sound B} \times W + \text{Sound C} \times X + \text{Sound D} \times Y + \text{Sound E} \times Z
\]

... where A, B, C, D, & E = frequency of pitch, and W, X, Y & Z = amplitudes.

The added complexity of this function would allow for more interesting and varied results and would also yield a representation that had closer ties to the original. For example, A, B, C, D, & E could be used to input a value of the same kind such as a rhythm, while W, X, Y & Z could be used to input a different value, such as a pitch. Alternatively, each component of the function (A, B, C, D, E, and W, X, Y & Z) could be used as different values from differing musical variables.

This model has numerous observable causal relations in its behaviour, which lends itself to the connection of as many similarity relations with a model. The number of relations evident creates the danger of making an overly-complicated model that no longer represents the original subject, but in this case it seems that the truth value of the subject is still adequately preserved. Conversely, the over-simplification of the model was avoided in the demonstrated outworking of the process as it seemed to lose all identity with the acoustic phenomenon it attempted to represent.

When substituting the elements of the function with similar elements that the characters (A, B, C, D, E, and W, X, Y & Z) represent, this model could be used for an *interpolative* representation. In this situation, it would be clear what the model is attempting to represent, in which it could be used for “reading off” properties of the original phenomenon. The number of similarity relations drawn between the phenomenon and the model also means that when
used as an \emph{analogue model} it retains all the observable relationships in the properties of the phenomenon.
7.0 Synthesis

This discussion of synthesis is very closely related to the previous section on harmonic series and sonic spectra. In representing acoustic phenomena in music, synthesis can be seen as the method used to translate data from the harmonic series or sound spectra. As mentioned at the beginning of the previous chapter, these two areas are differentiated between for the purposes of this research as often the technique of synthesis creates just as much of a musical focal point as does the spectra which it is representing.

Synthesis can be defined as an acoustic phenomenon on its own. While it can be used a method for modelling the harmonic spectrum of a sound, it also demonstrates the properties associated with sound production. Synthesis can be seen as the interaction of two or more sine waves (defined in more detail below), which pertains to the interaction of sound.

This phenomenon also makes a good model for compositions. As well as being used as a tool for composition by various composers, it can also be modelled in the form of a function for the purposes of interpolative or extrapolative transference of data. Synthesis involves the superimposition of sine waves, which means that numerous causal relations are at work in its underlying behaviour. This allows for similarity relations to be drawn between it and a musical subject, which is achieved through the modelling of the phenomenon.

Composers of the Spectral music movement, such as Gérard Grisey and Tristan Murail, are particularly under observation here as they demonstrate a significant contribution to this area of composition. Constructing a model of synthesis is discussed following a survey of the works of these composers and a definition of the phenomenon.

7.1 Synthesis Defined

Jean Baptiste Joseph Fourier (1768-1830), a French mathematician and physicist, developed a theory and had it published in 1822 in his text *Analytic Theory of Heat*. This theory posed that a periodic motion that is finite and continuous can be deconstructed into a series of simple harmonic motions – that is to say, complex sound waves can be mathematically broken down into their simple component waves (Blamey, 2008, p. 27). Helmholtz – as discussed in a previous chapter regarding his input into the field of science in music – then capitulated that the human ear is in fact capable of performing a Fourier analysis (as it became known), and demonstrated this through the use of various apparatus that he constructed.
If Fourier’s theorem is the pulling apart of a complex sound wave into its simple component waves, then it could be said that synthesis is the act of putting it back together again. After Helmholtz demonstrated Fourier’s theorem – deconstructing a complex waveform into its constituent simple harmonic motions, he then also did the reverse: “by combining different simple tones of various frequencies and amplitudes, he was able to imitate the sound of actual musical instruments, anticipating the modern electronic synthesizer” (Maor, 1998 p.11).

Campbell and Greated (1987) first discuss synthesers in terms of “additive synthesis,” the base concept on which synthesers are built. They define “additive synthesis” as follows:

The method of adding component sine waves to construct a complex wave of given spectral characteristics, which was employed on the early Hammond organs, is known as additive synthesis. Modern organs use the same idea, the only difference being that the component waves are produced by electronic oscillator circuits. Any periodic wave can be constructed from the sum of component sine waves having frequencies which are integer multiples of the fundamental f0, i.e. f0, 2f0,3f0, etc. The greater the number of component frequencies, the closer one can get to any required complex waveform. (Greated & Campbell, 1987)

The more commonly used synthesers however are based upon the principle of “subtractive synthesis”, in which complex waveforms are added together and the unnecessary spectral components are stripped away (subtracted, thus “subtractive synthesis”). This process of subtractive synthesis is what has been used as the basis for the more modern syntheser – early designs of the syntheser were only monophonic, but modern synthesers are polyphonic. (Greated & Campbell, 1987)

Following this explanation, it could be said that synthesers are the embodiment of an acoustic principle (combining sound waves) being translated into musical sound. Although this is not necessarily the focus of the research, understanding the concept of how synthesers work helps to inspire more ideas on the translation of sound into music.

Allen Strange states that the method of additive synthesis had first been used in orchestral arrangements and then been applied to electronic music. Strange discusses how if a composer wanted to create the sound of a tubular bell, the sound of which has non-harmonic overtones, sine-wave oscillators could be used. The non-harmonic partials could be discovered through analysis and then, through a matter of trial and error, “various amplitude relationships,
transient factors, and attack and decay characteristics would be experimented with... until the right combination is eventually discovered” (Strange, 1972, p. 7).

An important aspect of synthesis is the concept of “modulation”. Modulations result from the interaction of two waveforms, in which one sound modulates the other. In his text *Introduction to Computer Music* (2010) Nick Collins states that “modulation in signal processing refers to the control of an aspect of one signal by another. The controlled (modulated) signal is the carrier, and the controlling signal the modulator” (Collins, 2010 p.123).

In the process of synthesis, there are three common types of modulation that are used to manipulate the sound: amplitude modulation, frequency modulation and ring modulation.

Amplitude modulation (AM) is commonly known as the technique used in the creation of AM radio waves. This can also be found in music as amplitude vibrato, such as in the vibrato used when playing a flute. The partials found in harmonic spectra have amplitudes that are constantly changing, even though it may seem that they maintain a constant level. Fineberg states that this aspect “is modelled in many synthesis applications with jitters (random enveloped generators) or low frequency oscillators (less than 20 Hz) modulating the main amplitude generator” (Fineberg, 2000b, p. 95).

Frequency modulation (FM) synthesis is the more common type of synthesis utilised, particularly for synthesizers. In a simple musical form, frequency modulation can be seen in pitch vibrato, like that used by violins and other stringed instruments. The technique of frequency modulation was re-formulated for the context of synthesis by John Chowning (1934- ). In Chowning’s text *The Synthesis of Complex Audio Spectra by Means of Frequency Modulation* (1973) he discusses how frequency modulation (FM), while being understood in terms of how it applied to radio transmission, had not until that time been used significantly in the generation of audio spectra (Chowning, 1973, p. 1). His subsequent theories lead to the generation of the most successful synthesizer series in the history of electronic musical instruments – the Yamaha DX-7 (Chowning, 1973 & Fineberg, 2006).

In frequency modulation synthesis, a spectrum is created through the modulation of one signal wave (the carrier) by another (the modulator), which generates the summation and difference tones between the carrier and modulating frequencies (Rose, 1996, p. 30). By varying the carrier and modulator signals and the modulation index, a variety of different sounds can be synthesised (Chowning, 1973).
To work out the frequency of the sidebands (summation and difference tones) that result from frequency modulation, the following equation is used: frequency = carrier + and – (index x modulator) (Fineberg, 2000b, p. 97). In this equation, the index equals 0, then 1, then 2, progressing upward. The further up the modulation index goes the higher number of sidebands will occur. Fineberg presents an example of the resulting musical data produced by carrying out a frequency modulation of the note A (440 Hz, which is the carrier signal) with the modulator of 100 Hz, which is the note slightly above a G₂.

![Figure 59 – Musical example of frequency modulation (Fineberg, 2000b, p. 96).](image)

Ring modulation (RM) differs slightly from frequency modulation. Ring modulation is a form of modulation that can be used in the process of synthesis, like the frequency modulation synthesis that John Chowning pioneered. It was initially used as an analogue electro-acoustic treatment for complex sounds, where a sine wave generator was used to modulate a sound recorded by a microphone. The difference between RM and FM synthesis is that in RM there is no hierarchy within the relationship of what would normally be called the modulator and carrier; there are simply two sounds that are used to modulate each other and that are both still present in the resultant sound along with the sidebands. This is achieved through addition and subtraction of each note from the first spectrum with each note of the second spectrum, which can potentially create a large number of combinations; according to Fineberg, this means that “the number of partials generated will be two times the number of partials in the first spectrum multiplied by the number of partials in the second spectrum” (Fineberg, 2000b, p. 97).

Joshua Fineberg again provides a useful diagram illustrating an example of ring modulation in music. The example has a modulation of one spectrum (1) built on the note A, 440 Hz, with its first two partials and the second spectrum (2) built on D (approximately), 80 Hz, with its first three partials:
There are a great deal further areas involved with synthesis, and numerous acousticians and composers who have contributed to the development of synthesis in all its many manifestations. Composer Jean-Claude Risset (1938-) in particular was involved with the development of synthesis as a musical tool. Risset spent years with Max Mathews at Bell Laboratories developing the possibilities of synthesis, and later assembled a catalogue of computer-synthesis sounds as the results of his own work. He then worked at analysing, imitating and synthesising a range of instrumental tones, including brass, percussion and woodwind. His compositions reflected his work with this acoustic phenomenon, such as *Little Boy* (1968), *Mutations* (1969) and *Inharmonique* (1977), the latter of which explores the inharmonicity of synthetic bell sounds and transforms the initial sound into fluid sound textures using different envelopes (Risset, 2003, pp. 3-4).

Beyond additive, subtractive, AM, FM and RM synthesis, there are numerous other methods. These include formant wave synthesis, physical modelling and wave-guide synthesis, neural synthesis, linear predictive synthesis and granular synthesis. The latter of these was theorised by Denis Gabor and has been pursued by acousticians and composers such as Ian Xennakis, Trevor Wishart and Curtis Roads. Granular synthesis uses “grains” of sound, which are brief micro-acoustic events, each made up of a waveform shaped by an amplitude envelope, with their duration being barely perceptible by human hearing. Combining thousands of these grains over time allows for the creation of “animated sonic atmospheres” (Roads, 2001, p. 87).

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*Figure 60 – Musical example of ring modulation (Fineberg, 2000b, p. 97).*
For the purposes of this research paper, however, these additional areas of synthesis are not pursued due to the sufficient data provided by the synthesis methods already discussed. Even within this relatively smaller area of synthesis, there is a proliferation of composers using the techniques in modelling for musical compositions.

### 7.2 Composers Representing Synthesis in Musical Composition

This technique of recreating a complex waveform through the use of simple harmonic motion (pure tones/sine waves) has been represented in an orchestral setting by particular composers.

Allen Strange (1972) suggests a few of these composers who have used synthesis as a compositional tool. He states that Ravel and Hindemith, and more recently Ligeti and Kagel, have all utilised this technique in some fashion or another. Strange cites Mauricio Kagel’s *Music for Renaissance Instruments* (1966) as an example, in which “the composer is concerned with constructing various types of non-harmonic sounds” (Strange, 1972, p. 5).

However, Strange’s list isn’t all that conclusive. This could be because at the time of publishing, the most important work involving the use of synthesis as a compositional tool was only just occurring in the budding new musical style called “spectral music.” The work of Gérard Grisey and Tristan Murail, often considered the two pioneers of the style, seems to be of primary importance in developing this technique (Rose, 1996 & Anderson, 2000). Another proponent of the style is Joshua Fineberg, who has used instrumental synthesis in his pieces *Receuil de Pierres et de Sable* (2008) and *Streamlines* (1994). While the work of Kagel, Ligeti, Hindemith and Fineberg are by no means to be dismissed, this discussion will focus around Grisey and Murail.

Before a discussion of Grisey and Murail, it is still worth discussing the work of Ravel, as it could be said that the technique of using synthesis in a composition was used first by him. In the introduction of this research, Ravel’s use of synthesis in his work *Bolero* (1928) is discussed. Just as Claude Debussy seemed to predate the more modern composers that would be deemed as pioneering this concept of representing acoustic phenomena in musical compositions, Maurice Ravel, Debussy’s (almost) contemporary of the Impressionist music movement, seemed to be ahead of his time in terms of this compositional technique.
Allen Strange mentions Ravel’s use of additive synthesis, which is described by Joshua Fineberg as “orchestral synthesis”, that is, “the technique of using the interior structure of a sound as a model for a rich new orchestral object, transforming its micro-fluctuations into macro-forms” (Fineberg, 2006, p. 116). Orchestral synthesis was then furthered by later composers that created a new compositional school, called “spectral music”.

### 7.2.1 Spectral Music

Spectral music as a musical style has had a particular focus on representing acoustic phenomena in musical composition. In the journal entry *Introduction to the Pitch Organization of French Spectral Music* François Rose draws the conclusion that one of the primary features of spectral music has been its creation of new techniques through the use of representing acoustic phenomena: “With its transference of such phenomena as ring modulation, frequency modulation, combination tones, and filtering from the electronic to the acoustical domain, it presents new compositional ideas and generates new procedures like microphony and macrophony” (Rose, 1996, p. 36).

The earliest forms of this “transference of phenomena” occurred in a compositional method where decisions regarding each compositional step were dictated by the analysis of sound spectra. To understand the use of the word “spectrum”, consider Scott Hunter Stark’s discussion of synthesis: “The sounds we hear are normally made up of a mixture of sine-wave components which together can be referred to as the *spectrum* or energy distribution of any given sound” (Stark, 2000 p.25).

In order to analyse a sound spectrum, a Discrete Fourier Transform (DFT) is used, which is the application of Fourier’s theorem (described above). This application is then used to generate descriptive data from a sound spectrum which can then be visualised using a spectrogram. Initially, some spectral music compositions focused on representing the harmonic spectrum of a particular sound by using the data gained from the spectrograms. Gerard Grisey was one such composer, and his use of this method will be discussed in the following section.

Spectral music then evolved beyond this initial compositional process, as it became guided more by a set of general principles. These “broad aesthetic consequences”, rather than “specific stylistic ones” meant that spectral composers could potentially have very different styles of music. (Fineberg, 2006, p. 112)
In 2003 a conference was held in Istanbul, titled the *Istanbul Spectral Music Conference*. On the conference announcements section of the Music Theory Online website, a brief description of the musical style (author unknown) is presented as part of the write-up for the conference proceedings which reveals a few insights about the nature of spectral music:

> By focusing on the physical, perceptual, and aesthetic attributes of timbre, composers and performers have produced new pathways to the interior of sound while at the same time developing novel compositional languages that address one of the most elusive aspects of musical discourse. (*Spectral Music*, 2003)

This style of music seemed to originate in France in the early 1970’s, in particular from composers Gerard Grisey and Tristan Murail and was later developed at IRCAM with their *Ensemble l’Itinéraire*. Tristan Murail, in his lectures at the Darmstadt summer course, has described Spectral music as an aesthetic rather than a style, an attitude towards composition rather than a set of techniques – the attitude being that "music is ultimately sound evolving in time" (Fineberg, 2006, p. 112).

The term “spectral music” was originally coined by Hughes Dufourt in his article from 1979 titled *Musique Spectrale* (according to Cornicello, 2000), however some of the major practitioners of this compositional method considered the term misleading and an oversimplification, such as Grisey and Murail. In an interview with David Bündler on January 18th, 1996 in Los Angeles (published in the March, 1996 issue of *20th-Century Music*), Grisey (as do most composers who are “labelled”) rejected any significance of the title “spectral” music. His stance was that while the label may have applied to the original concept behind the music, the direction of the compositional ethic moved beyond its origins and as such left behind any relevance of the title to the techniques used. This feeling expressed by Grisey, amongst others, seems to have led to a redefinition of the term spectral music. One such definition came from the 2003 Istanbul Spectral Music Conference, where the term was redefined to include any music that focuses on timbre as a key element of structure or musical language (*Spectral Music*, 2003).

Brett Mabury’s thesis on spectral music (2006) draws a connection between the impressionist composers such as Debussy and spectral music, stating that Debussy’s focus on timbre was a precursor to the later trend developed in spectral music. Andy Hamilton also concurs with this argument (as quoted in the introduction of this research). Mabury states that while there are obvious factors that separate these two compositional schools (such as the different eras and
levels of technology) they share a common compositional mindset that lead them to explore areas of seldom used timbres and to producing exciting musical colours (Mabury, 2006, p. 22).

A number of other composers are listed as being precursors to the spectral music style. Fineberg cites Oliver Messiaen, György Ligeti and Giacinto Scelsi in this list, and Mabury also includes Edgard Varese, La Monte Young, Paul Hindemith, Frederich Cerha, Karlheinz Stockhausen and Per Norgaard (Mabury & Fineberg, 2006).

The techniques used in composing spectral music are wide and varied; however, they are only secondary to achieving the resultant composition – “the means of achieving a sonic end” (Fineberg, 2006, p. 112). Spectral music compositions are largely focused on timbral structures – often timbral effects that would normally be generated by computer processes are performed by live musicians on their instruments. This method of translating acoustic phenomena is shared within the exploration of musical works within this research also. By choosing this method it means that the timbral effects must be translated into new variations of the traditional music notation. Pitch, dynamics and temporal aspects are subject to these timbral considerations also; where micro gradations of pitch occur (quarter tones or smaller) the musical octave will be divided into twenty four or forty eight pitches instead of the traditional twelve, and similar processes applied to dynamics and tempo make for notational difficulties.

What this discussion of spectral music boils down to is this: it seems that the general principle guiding the processes in spectral music compositions is that sound is the stimuli for their musical composition. Instead of literally imitating the behaviour of a particular sound, the composers use it as a muse from which to springboard metaphorical connections and ideas. In reference to the criteria outlined in this research, it seems clear that spectral music is really a compositional school that extrapolates ideas from acoustic phenomena. This is further unpacked under the discussion of specific composers in the next few sections.

Joshua Fineberg clarifies the extrapolative representation used by spectral composers in his text (2006), by way of a comparison with the fine arts:

Sound is the model for spectral composers in the same way that light is the model for Impressionist painters, yet Monet did not simply paint luminous washes of colour. In Monet’s series of paintings of the Rouen cathedral at different times of day, it is clear that the proximate subject (the cathedral) is just a vehicle for communicating the real
content: light, shadow, and colour. This is what Grisey means: The real content of music is not mathematics, quantum physics, or even aesthetic philosophy, but sound, the way sound changes in time and the effects it produces in the human mind. (Fineberg, 2006, p. 113)

Fineberg’s analogy focuses on the concept that spectral music is really using sound as a model for musical compositions, often in an extrapolative way.

### 7.2.2 Gérard Grisey

Gérard Grisey (1946-98) was a composer who, until his passing on November 11th 1998, wrote in the spectral music style. This particular field of composition, as outlined above, has important ties to this research area, as it shares a number of similar idealisms; primarily, the extrapolation of data from acoustic phenomena into musical composition.

Grisey studied under, and with, such influential composers as Messiaen, Ligeti, Dutilleux, Xennakis and Stockhausen. Grisey also cites Varese as an influence on his work and on spectral music as an art form. In the interview with David Bündler, Grisey states that “Varese was thinking in that direction also. He was the grandfather of us all” (cited in Bündler, 1996). These influences on Grisey played a role in the development of his compositional practice, particularly the push towards the style of Spectralism.

Grisey’s works, along with Tristan Murail’s, inspired the conception of spectral of music as a compositional school in the 1970’s and his use of synthesis as a compositional process was evident in his early works (Fineberg, 2006, p. 112). Mabury’s research into this area (2006) has illuminated many of the central idealisms of the spectral music genre and also the work of Gérard Grisey, who held a prominent position in the furtherance of this compositional style.

Mabury first discusses Grisey’s use of ‘instrumental synthesis’ (or ‘instrumental additive synthesis’), a technique he labels as being one of the first and most influential to be used in spectral music – he also credits this term to Grisey. “In 1975, Grisey took this concept of additive synthesis and applied it to an instrumental setting with his composition Partiels” (Mabury, 2006, p. 77).

Although Grisey later moved away from the initial concept of spectral music, he describes the process that was originally used in his compositions. Grisey was concerned with the physical aspects of sounds, different spectrums and the quality of spectrums; “At the very beginning, I
started with real spectrums that I would analyse and then transform into external types of writings” (cited in Bündler, 1996). This can be seen in Grisey’s work *Partiels*.

Fineberg states that in *Partiels* Grisey is using the “musical technique of using the interior structure of a sound as a model for a rich new orchestral object,” which is called “orchestral synthesis” (Fineberg, 2006, p. 116). Both Mabury and Fineberg provide a discussion on this point in their deconstruction of *Partiels* and describe how Grisey actually achieved this concept of additive synthesis in music composition. In *Partiels*, which was written for eighteen instruments, Grisey firstly used a trombone playing a loud pedal of a low E2 (41.2 Hz) as his sonic source, and then analysed the sound using an electronic sonogram. This then created a graphical representation of the sound’s structure; its time, frequency and amplitude.

Fineberg provides a replication of the sonogram’s pictorial representation in his discussion of the work. In the diagram the component bands, which are known as partials, are all equidistant; this is because the sounds are harmonic, which means that the partials occur at intervals equal to multiples of the fundamental frequency (Fineberg, 2006, p. 116). In this figure the x-axis shows the time and the y-axis shows frequency in Hz.

![Figure 61 – Sonogram of a trombone’s low E, from Grisey’s *Partiels* (Fineberg, 2006, p. 17).](image)

The darkness of each of the lines in the diagram relates to the amplitude of each of the partials and fundamental tone. Through this, it can be observed that the fifth and ninth partials are the loudest, as opposed to the fundamental and lower partials as would normally be expected – these particular notes actually form dissonant intervals in relation to the fundamental, which give the trombone its particularly brassy timbre. It can also be seen that the lower partials enter first, followed by the higher partials later (Fineberg, 2006, pp. 116-117).
The next step in the process according to Mabury and Fineberg is to translate this information from “the domains of time, frequency and amplitude to more musical dimensions like pitch, dynamics and rhythm” (Fineberg, 2006, p. 117). To do this, approximations to the nearest value are used for pitch, dynamic levels and rhythmic placement and each note (frequency) receives a dynamic marking proportional to its amplitude value. Fineberg provides another replica of this musical transference in the form of a musical model that corresponds to the partials evident in the trombone’s note.

In this musical model (figure 60) of the partials evident in the trombone note, numbers are written above each note representing the partial markings. This musical model, in conjunction with a model of the rhythmic placement of each note’s entry, was then used to create the instrumental score (Fineberg, 2006, p. 118). Grisey achieved this by designating the notes and their dynamics markings evident in the sonic spectrum to what he deemed to be the right instrument; by choosing the correct combination of instruments, their timbres would blend to achieve the desired effect.

In scoring these elements, Grisey also chose to begin the piece with the original trombone note – played at a forte (also con sordino), gradually decrescendo-ing over the first three measures. Added to this is the implementation of the double bass reinforcing the trombone’s note, with a decreasing sforzando on the attack of each repetition of the note (three consecutive soundings, played over semibreves), from sffz, to sffz, to sfz. As the trombone fades out, the other instruments – cello, viola, violins, clarinet and piccolo – gradually enter. The order in which they enter (as just listed) is directly related to the entry of the partial each instrument is representing (in accordance to the partials emergence following the sounding of the fundamental tone in the original sonogram analysis). These elements can be seen in an example of the score below.
The effect that emerges is that the trombone’s note is sounded, reinforced by the double bass, and as it gradually fades out the *synthesised* version of the trombones note, as played by the other instruments of the orchestra, is cross-faded into audibility. As Fineberg puts it, the trombone’s note gives way to an “instrumentally synthesised imitation of itself” (Fineberg, 2006, p. 118).

An important element of Fineberg’s discussion of *Partiels* is that he draws attention to the extrapolative nature of this type of representation. In essence, he is saying that this is not an imitation, nor an interpolative representation, as it neither seeks to directly copy the original sound source or even present a strictly literal transfer of the sound properties into a musical context:
This instrumental timbre does not seek to present an indistinguishable copy of the original, but rather to generate an amplification and transfiguration of the trombone note. The listener can still sense the underlying trombone colour of the sound, while at the same time a doorway is opened up to a vast new domain of sound found within the original sound. Deploying these notes in a vastly stretched out imitation of a trombone does not really sound like a trombone; it sounds like something entirely new, while preserving a distinct trombone-colour in its overall presentation. (Fineberg, 2006, p. 118)

In light of this comment and that of others (Rose, Anderson, Mabury) it seems that much of the nature of Grisey’s work as a spectral composer utilises this form of representation (extrapolative). When Grisey uses the acoustic phenomenon synthesis as an external stimulus to represent in a musical composition, it seems that he uses it as a loose guide (besides what may be considered a rigorous procedure to get to the point of instrumental scoring). Fineberg again backs up this view point, saying that even though the literal sonic models used by Grisey and other spectral composers were very important in their early works, they never really represented the totality or even just a section of a composition, and that more complex models that are less direct (or literal) are now more common (Fineberg, 2006, pp. 120-121).

Grisey also may be seen as the primary composer who has utilised this phenomenon in a profoundly influential way. Although a previous discussion in this research outlines Ravel’s use of additive synthesis in the form of instrumental orchestration techniques, Fineberg still credits Grisey as being the first user of this technique (or at least, the most influential): “Partiels is one of the best-known and earliest examples of a composer using an instrumental analysis to create a harmonic and gestural model that is then realized by an instrumental ensemble” (Fineberg, 2006, p. 116). Fineberg’s praise of this piece continues by drawing a connection between it and the works of subsequent spectral composers. Fineberg states that this particular musical moment (in Partiels) impacted on spectral composers by making them aware of the potential of “sonic phenomena” (Fineberg, 2006, p. 118).

Although, as Fineberg says, the technique of using additive synthesis in Partiels inspired the works of many other spectral composers, the compositional ethic behind the core of Spectralism had since moved away from this first technique of using synthesis as a compositional tool. The concept that emerged as the core ethic behind spectral music is something that is very important to Grisey due to his personal experience with the musical
style. In the 1996 interview with David Bündler, Grisey was asked about his use of spectral principles or methodology in his compositions. In his response to this question he cuts to the core ethic behind spectralism: “It's an attitude. It considers sounds, not as dead objects that you can easily and arbitrarily permutate in all directions, but as being like living objects with a birth, lifetime and death” (cited in Bündler, 1996). Grisey also explains that there are no systems as in other types of 20th Century music, such as in serial music or tonal music.

Another component of the developing ethic that Grisey employed as a spectral composer was the desire to keep the processes used in the compositions evident for the listener. He states that a key element of spectral music, particularly during its early conception, was to try and find a better connection in music between the concept of the piece and the way it was to be received by the audience. Grisey says that he does not want to “put the listener behind a wall of information through which they cannot find their way” (cited in Bündler, 1996). Grisey does go on to state that the establishment of a very clear process early in a piece then allows him to distort the perception of the process later.

This statement seems reminiscent of Reich’s ethic; he also desired to have a compositional process that was evident to the listener, in response to the experimental music movement that unintentionally hid all its processes by complicated systems that the listener could not perceive (Reich, 1968). It’s possible that this ethic would prove a limitation on the composers, as they would constantly need to stop short of adding anything into their compositions that would increase its complexity level to obscurity. This ethic however, is part of what defined Minimalism, and also seems to, in part, drive the music of spectralism.

A growing element of Grisey’s compositional style throughout his career was his use of extended time and continuity, as well as music in compressed time. Grisey’s inspiration for music in compressed time stemmed from Nancarrow, and the use of extended time and continuity from Ligeti (cited in Bündler, 1996). Grisey stated in his interview with Bündler that the use of these elements of musical focus came to have more connection with the core of spectral music than analysing sound spectra (cited in Bündler, 1996). Grisey’s fascination for extended time and for continuity seems to demonstrate a compositional ethic that is driven by stasis, which may be a result of his observation of stasis in the work of Varese and perhaps even Debussy (Bündler, 1996 & Morgan, 1991).

A final element of Grisey’s ethic as a composer that is of interest is his rejection of electronic mediums as a method of conveying musical ideas. Interestingly, Grisey’s choice to use
Instruments over computers to generate music stems not from a desire to represent synthesis in his compositions using instrumental techniques, but rather from a lack of confidence in his use of technology as well not wanting to be constantly “chasing after” technology as it changes. Grisey comments that: “I see around me all composers running, literally running after new technology that's going to be better in a few years” (cited in Bündler, 1996). Grisey’s attitude is mirrored by other composers and music critics, such as Reich and Fineberg, who in seminal texts have both expressed a desire to avoid electronic mediums where possible (Schwarz, 1981 & Fineberg, 2006).

In summary, it seems that Gérard Grisey is an example of a composer who successfully used an extrapolative form of representation by his use of instrumental additive synthesis (or orchestral synthesis). The extrapolative nature of this representation is evident through the analysis of the composition and is also confirmed by peer review, such as that from Joshua Fineberg and François Rose. In his paper titled *Introduction to the Pitch Organisation of French Spectral Music* (1996), Rose points out that the use of the term “instrumental additive synthesis” to describe Grisey’s procedure in *Partiels* is really in a metaphorical sense, as the analogy between the model (synthesis) and its realisation (musical transference: *Partiels*) is more theoretical.

Rose goes on to explain that this metaphorical nature of the representation is largely due to the fact that in the natural state of synthesis, sine tones, which are manifestations of simple harmonic motion, are used together to create a complex waveform. In the use of synthesis that Grisey has implemented, acoustic instruments are used in combination to create the additive sound; this means complex waveforms, each with their own identity, are being added to create another even more complex waveform. Rose explains that this representation creates a new entity:

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Thus it should be clear that the idea is not to create an acoustical reproduction of an electronic sound, but rather to adapt an electronic procedure for acoustical instruments. Naturally, the result of this procedure, while deriving from physical models, no longer shares but replaces the characteristics of the modelled phenomenon. (Rose, 1996, p. 11)
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This opinion helps validate the idea that Grisey is in fact using an extrapolative form of representation in his music composition *Partiels*, as he seeks to translate synthesis into musical terms through an indirect, or metaphorical, method. Fineberg also backs up this argument in
his statement concerning Grisey’s method of using a close-up view of sound structures as a model for larger musical structures: “His idea was not to recreate the original sound (which he could have just played, after all), but to make something new that preserves the overall coherence that comes from being part of a unified acoustical structure at a larger level” (Fineberg, 2006, p. 115). This perspective on Grisey’s method seems to be mirrored quite easily in the definition of using extrapolation as a method of musical transference outlined earlier in this research.

7.2.3 Tristan Murail

Along with Gérard Grisey, Tristan Murail is considered to be the other leading figure in defining early Spectralism (Rose, 1996). Amongst other techniques, Murail also employed orchestral synthesis as a compositional method, which like Grisey demonstrates the representation of synthesis as an acoustic phenomenon in music.

Murai achieved this synthesis through slightly different means than Grisey. Rather than analysing a sound to generate a visual harmonic spectrum to then synthesise, Murail used the technique of frequency modulation (FM) synthesis. Murail used a simplified form of this in his composition *Gondwana* (1980) to construct the sound of a bell and brass sound, by using a series of carriers to be affected by the one single modulator, which created a set of five different modulation-based harmonies to be played by an orchestra (Fineberg, 2000c & 2006). The pitches chosen by Murail for the five carrier signals and one modulator can be musically illustrated:

![Figure 64 – The five carrier signals & one modulator signal used in Gondwana.](image)

The modulator signal is used to modulate these carrier signals to create the harmonies for *Gondwana*. François Rose demonstrates the effect of the modulator (low G#) on the first of the carrier signals (G) by musically notating the ensuing summation and difference tones, shown in the following figure.
The summation and difference tones, which are notated musically in the second measure of figure 63, were then allocated to different instruments of the orchestra. The allocation of the notes to particular instruments as well as their temporal placement also shows Murail’s focus on the sonic qualities that he is representing in portraying a bell. A bell naturally has a particular resonant frequency and to represent this, the clarinet is allocated a note from the bell’s spectrum that sits comfortably within its register (in the case of the first carrier signal, the note is an Ab6), as notes in a comfortable register for a clarinet create a sound that has more presence and is spectrally richer. Murail also characterises the bell’s natural phenomenon of having the higher partials fading out quickest by using an instrumental transfer – instruments that have a rich timbre in higher partials (for example, trumpets or horns) are faded out and replaced by clarinets, which have less higher partials (Rose, 1996, p. 32).

Murail also represents the attack of the bell’s sound by orchestrating the vibraphone, crotales, tubular bells and piano to play the corresponding notes. As some other instruments in the orchestration are scored to play microtones, this latter named group of instruments create harmonic friction by playing the equal-tempered equivalent of the other instruments’ microtones (Rose, 1996, p. 32).

Each carrier and modulator combination used creates a combination of summation and difference tones that are then used in the same way, which form the basis of each section. The effect created in Gondwana, when performed by the large orchestra for which it is written, is a “series of enormous, orchestrally synthesized bell sounds that are gradually transformed into...
an orchestrally synthesised brass sound” (Fineberg, 2006, p. 118). In addition to this description, Fineberg again provides the insight that the type of representation used in *Gondwana* – of frequency modulation synthesis through the method of orchestral synthesis – is extrapolative in nature. Fineberg states that:

...The idea is not to offer a realistic simulation; after all, Murail could have just included bells with his orchestra. The idea is to go inside of a bell sound and render audible the normally microscopic structures that make it beautiful. Moreover, by recreating a hybrid with bell-like properties, it is possible to gradually manipulate these structures and make musical objects less and less bell-like in gradual increments... (Fineberg, 2006, pp. 118-119)

This extrapolative representational method seems common in the work of both Grisey and Murail. It’s also worth noting that the earlier works of these composers (for example, Grisey’s *Partiels* and Murail’s *Gondwana*) are closer to a interpolation than an extrapolation in their nature, as their later works progressively move away from the stricter sense of using sonic spectrums as external stimuli and steadily use more abstract methods of creating a musical representation. The original stimulus becomes less and less distinct as their works progress, and as Grisey stated the compositional style becomes more of an attitude than a process or particular set of methods.

In a self-analysis of his own work, Tristan Murail states his material is not just a musical note or a sound, but is more so the sensation created by that note. Murail goes on to say that the material is not the harmonic spectrum but the harmonicity inherent within that spectrum and the “possibilities of transformation that it contains” (Murail, 2005a, pp. 149-150).

This sentiment seems to sum up the general progression of Murail’s compositional ethic. While the specific compositions of his that are under study adequately convey the concept of representing acoustic phenomena in musical composition through an *extrapolative* method, it seems that his musical development continued to become even more abstract, adopting a compositional practice that loosely utilises the phenomena as a general principle upon which to base his music.

**7.2.4 Karlheinz Stockhausen**

Ring modulation synthesis has been used by a number of composers, as a performance technique and as an acoustic phenomenon used in generating compositional structures. Julian
Anderson states that Paul Hindemith (1895-1963) presents one of the earliest examples of what is known as ring modulation harmony (Anderson, 2000, pp. 9-10).

Among the other techniques that Stockhausen became reputable for, his use of ring modulation in music became one his defining compositional practices. This use of an electronic technique was something that Stockhausen seemed quite well versed in. Stockhausen explains the different areas involved in the musical process and how they can be used:

> The division of the musical process into three independent areas (sound production, sound recording, and sound transformation) makes it possible to combine the possibilities of instrumental practice with those of electronic techniques. Any desired sound source (traditional instruments, sound events of whatever nature) can thus be integrated into a coherent sound composition. (cited in Morgan, 1991, pp. 473-474)

Fineberg notes that the use of ring modulation is evident in Stockhausen’s works, such as Mixtur (1964) and Mantra (1970), as does Mikrophonie I (1964), which utilises ring modulation as a key aspect of its composition and performance. Stockhausen describes how he created Mixtur for five orchestra groups, four sine-wave generators and four ring-modulators, in which the sounds from the four instrumental groups are picked up by microphones and then sent to the four ring modulators. The modulators are used to modulate the instrument’s timbre, rhythm, dynamic level and pitch by the process of using sine wave generators which are operated by the musicians as dictated by the score. These sounds are then projected through four sets of loudspeakers, and added into the mix is a fifth group of instrumentalists playing tam-tams and cymbals, which are amplified by contact microphones and also played through loudspeakers. (Stockhausen, 1964)

Mantra was written for two pianos and live electronics, in which a thirteen-note “mantra” (formula) is played by both pianists. One pianist plays the original series and the other plays an inversion, both players starting on the note A3. Each pianist is also in control of a ring modulator and a sine tone generator and uses the ring modulation technique to alter the sound of the piano. Of the thirteen notes in the mantra, the first and last are identical to the tone being emitted by the sine tone, creating a consonant sound. The other notes in the Mantra are designed to conflict with the sine tone to create dissonant sounds. This constant exchange of consonant and dissonant sounds creates a constant dichotomy of tension and relaxation, but Anderson states that eventually the sounds become predictable enough to allow the listener to hear inside the timbre being presented (Anderson, 2000, p. 14).
7.3  Modelling Synthesis for Musical Compositions

The representation of acoustic phenomena is detailed in this research as being interpolative or extrapolative (or scale model and analogue model). It seems through the study of Grisey and Murail that the extrapolation of the phenomenon synthesis has been well used. Some may argue that their use of the phenomenon may be seen as interpolative, and this may be a fair point of view – after all, the phenomenon’s behavioural characteristics are being translated reasonably directly from the acoustical context to that of the musical one.

However, the work of these composers then takes a turn for the extrapolative. This can be seen on a micro scale in the methods utilised in the compositions, and on a macro scale in the overall development of the musical genre, which became progressively more abstract. Gérard Grisey stated as much in his interview with David Bündler: “I am always trying to first establish the rules of the game – the process of the form – for the listener rather clearly – very often too clearly – in order later on to be able to distort it or to change directions” (Bündler, 1996).

The purpose of this research is to explore avenues that have perhaps not been furthered by the work of past and present composers, and to also pave the way for future composers who may wish to pursue this area. As such, a model of synthesis is still provided; if one thing has been learned from the example set by spectral music composers, it’s that there are virtually no limits to the musical content achievable when extrapolating from a sonic spectrum. A model of the phenomenon, regardless of the work already done in this field, can surely only assist further in the expansion of this type of compositional process.

7.3.1  Creating a Functional Model of Synthesis

To recapitulate the basic form of synthesis, it can be thought of as the superimposition of multiple simple harmonic motions in order to create a complex harmonic motion. Broken down further, this can be seen as several sine waves being sounded at once in order to create a simulation of an acoustic sound wave, such as a musical instrument.

In light of this, it can be said that synthesis equates to the addition of a sine wave of frequency $w$, a sine wave of frequency $x$, a sine wave of frequency $y$ and a sine wave of frequency $z$, with the possibility of the addition of more or less sine waves; a minimum of two and a maximum that sits within realistic proportion. The summation of these sine waves equals a synthesised complex waveform, with component frequencies $w, x, y & z$.

To demonstrate this is a function, it can be seen as:
... Where \( f \) = frequency, \( \text{sine} \) = generated sine wave, and \( w/x/y/z \) = value of frequency.

To simplify this function and at the same time make it more generally applicable, the common component (the sine) is removed:

\[
fw + fx + fy + fz = f (w, x, y, z)
\]

Applying this function back to its original context assists in understanding and verifying its validity. If the value of \( fw, fx, fy \) and \( fz \) were sine waves of frequencies 73.42(D2), 349.23 (F4), 207.65 (G#/Ab3) and 3,520 (A7) respectively, then the function would result in the following:

\[
\text{Sine } f_{73.42} + \text{Sine } f_{349.23} + \text{Sine } f_{207.65} + \text{Sine } f_{3520} = \text{Sine } f_{(73.42, 349.23, 207.65, 3520)}
\]

In this example, the end result of synthesising these individual sine waves of varying frequencies is a complex waveform made up of all initial frequencies (the partial number, corresponding amplitude or temporal placement are not included in the data, as discussed below.)

To replace the function with musical examples, firstly consider the use of the model as a structural device. If \( f \) were to refer to a structural concept such as a bar, and the designations \( w, x, y \) and \( z \) were particular bar numbers of a piece, then the result would be a section created from the musical material contained within each of the bars. To illustrate this, a musical excerpt is shown of a section (A) with eleven bars in it, each numbered.

![Figure 66 – Musical example #1 to be subjected to model of synthesis.](image)
Using the data from this section, the structure of the subsequent section could be generated using the synthesis model. If the bars 2, 7, 11 & 5 were chosen as data to replace \( w, x, y \) & \( z \), and the section \( A \) and subsequent section \( B \) replaces function \( f \), then the function would be thus:

\[
A2 + A7 + A11 + A5 = B (2, 7, 11, 5)
\]

Musically, this would create the following section of the piece (Section B):

![Musical example](image)

Figure 67 – Musical example #1 subjected to model of synthesis.

As with each model of a phenomenon, the musical elements that can be used to replace components of the function are wide and varied. The breadth of variables is only exceeded by the number of variations that they will produce when subjecting musical material to this model.

This model seeks to represent the observable relations evident in the phenomenon synthesis and it seems to successfully correlate these relations with the relations of the function’s elements. There are, however, more elements to this phenomenon when synthesising with a complex waveform as the model, such as the required variation in amplitude of the component sine waves, the temporal commencement of each wave and the decay of each. Other components excluded are the summation and difference tones. These considerations could be added into a model of synthesis, however it is the primary characteristics of the phenomenon that are of concern with this model. There are enough musical outcomes when using even this basic model to generate data for compositions that it doesn’t really warrant the creation of a more involved model.

A further consideration in this model of synthesis is this: when modelling synthesis, it’s possible to either model the process or the subject material. Gérard Grisey’s work *Partials* uses orchestral synthesis to adapt the behaviour of the subject material – the note played by the trombone – into an instrumental setting. Therefore, synthesis is the tool utilised to represent the harmonic spectrum of the trombone note, which means the subject material being represented is not synthesis, but the harmonic spectrum. In contrast, if the process itself were
to be represented this would mean copying the behaviour of synthesis, not the harmonic spectrum of a sound.

The model presented above, then, is a model of the process. This process being modelled doesn’t take into consideration the values of any particular sound spectrum, which in the example is reflected by the use of unrelated values – in a harmonic spectrum, integer values and unrelated values would both be used. This model could perhaps be seen as a purer way of representing synthesis itself.

When considering these two options, it could be said that the discussion of synthesis within this preceding chapter is really a discussion of the process used in order to represent harmonic spectra. The justification for the inclusion of this discussion in the topic of representing synthesis, however, would be that synthesis is the primary focal point of the works. In other works, the harmonic series is used to generate data using different tools.
8.0 Interference/Beating

These two phenomena are often used synonymously, however the definition outlined below shows that “beating” is the audible outcome of “interference”. Beating (the term is used interchangeably with interference in this chapter) can be defined as an acoustic phenomenon due to its observable properties that pertain to the transmission of sound, as it involves the interaction of two or more sound waves.

This research looks at beating as a phenomenon due to its use by a number of composers. Their use of the phenomenon ranges from the incorporation of the phenomenon in its natural state to a representation of it. In the case of the former, the latter is sometimes used concurrently to enhance the occurrence of the phenomenon being exhibited; meaning that beating is a focus of the composition and musical devices are simultaneously employed that represent the phenomenon in order to enhance its audible effects. This can be seen in the work of Alvin Lucier in particular.

Interference can also be seen being used as a model by a number of composers. The nature of beating lends itself to the creation of such a model due to the fact that the behaviour of the components can be reduced to a mathematical ratio. The causal relations evident, then, are rather simple in the phenomenon which means that the model of the behaviour is easily applicable to a variety of musical contexts by drawing similarity relations with these evident properties.

The utilisation of beating is observable in the works of the selected composers of this research, Alvin Lucier and La Monte Young. The modelling of the phenomenon can be seen being accomplished in the works of Cowell and Nancarrow, who relate the beating evident in two sound waves to the interaction of two different rhythms: polyrhythms. The parallel drawn between these two elements form the basis of an adequate model for the phenomenon and as such no additional model is created in this research. The discussion of the work of these composers is prefaced by the definition of the interference/beating.

8.1 Interference/Beating Defined

Interference is also commonly known as acoustic beating. Kinsler, Frey, Coppens & Sanders define beating as: “In the sounding of two pure tones of slightly different frequencies, this variation in amplitude results in a rhythmic pulsing of the loudness of the sound known as...
beating” (Kinsler, Frey, Coppens, & Sanders, 1976, p. 24). Alvin Lucier elaborates on this
definition, and describes acoustic beating (interference) as the occurrence in which “two or
more closely tuned tones are sounded, audible beats – bumps of sound that occur when the
waves coincide – are produced” (Lucier, 1998, p. 8).

The term “interference” is often used in discussion relating to “phase” and in particular refers
to “phase cancellation.” Scott Stark discusses this phenomenon in his text (2000):

> When two sine waves of the same frequency and amplitude are superimposed with
> both starting simultaneously at zero degrees (exactly in-phase), the result is a sine
> wave of twice their individual amplitude... If the same two waves are superimposed
> with one starting at zero degrees and the other at 180 degrees, their amplitude are
> then exactly opposite one another (180 degrees out-of-phase)... cancelling one
> another out completely. (Stark, 2000 p.27)

Stark’s description of the reinforcement and cancellation of sound due to the superimposition
of the waves in or out of phase is also known as “constructive and destructive interference.”
Constructive and destructive interference within the interaction of two or more sound waves
is the phenomenon that results in audible beating, the term more commonly used by
musicians than interference.

Campbell and Greated also describe “constructive interference”. They state that when two
sounds are produced (by two flutes in this example) that are exactly “in phase – that is, if a
peak of high pressure from [instrument] A arrives simultaneously with a peak from
[instrument] B – the net effect will be a high pressure peak of twice the amplitude due to each
flute separately.” This is an occurrence of constructive interference. The opposite effect occurs
when the two sounds arrive 180° out of phase; “As the wave from flute A tries to make the
pressure rise, the wave from B is trying to make it drop... The net effect is that the two waves
cancel each other out completely” (Greated & Campbell, 1987). This is an occurrence of
destructive interference.

When two notes are played simultaneously that are very close in pitch – as when tuning two
instruments, or two strings on a guitar for example – a combination of constructive and
destructive interference occur as the two sound waves cycle in and out of phase. This results in
the “audible beats” that Lucier describes above.
8.2 Composers Representing Interference/Beating in Music

Beating has been used in musical compositions, both as through its utilisation or a demonstration of the acoustic phenomenon (as in the works of Alvin Lucier) and also as a source from which to extrapolate data for use in the compositional process. The latter is used by Henry Cowell and Conlon Nancarrow, which will be discussed below.

8.2.1 Alvin Lucier

In Lucier’s journal article for the Leonardo Music Journal, *Origins of a Form: Acoustical Exploration, Science and Incessancy* (1998), Alvin Lucier discusses several works of his that utilise acoustic beating (interference) as a compositional feature. As with many of Lucier’s works, these compositions make strong use of an acoustic phenomenon that demonstrates Lucier’s passion for acoustics and sound related phenomena. This use of the phenomena though is not so much a representation as it is a demonstration.

Lucier states that he has used this acoustic phenomenon numerous times in compositions and in sound installations by using sine wave oscillators to create standing waves and interference patterns, which manifests in audible beating, between closely tuned waves. (Lucier, 1998, p. 8)

An example would be Lucier’s sound installation *Seesaw* (1983). In this installation the tone of a slowly sweeping oscillator is juxtaposed against a fixed tone and as it sweeps across the fixed tone it causes audible beating patterns. These patterns speed up and slow down as the sweeping wave moves towards, in tune with and then away from the fixed tone (respectively). Lucier encourages physical involvement to be a part of the sensation; “If one stands midway between the two loudspeakers, one can feel the waves moving across the space as the wave fronts collide” (Lucier, 1998, p. 8).

As discussed earlier in regards to the sine wave, Lucier also enjoyed the combination of acoustic instruments with the sine wave generator, as it enhanced the effect of the acoustic beating. Lucier states: “In several works I used musical instruments combined with sine waves, whose purity of sound provided optimal beating with the richer instrumental timbres” (Lucier, 1998, p. 8). A piece produced a year prior to *Seesaw* used a combination of instruments and a sine wave generator to demonstrate the same phenomenon. *Crossings* (1982) was written for a small orchestra and sine wave oscillator in which the sine wave slowly rises from a low C (32.7 Hz) to a high C (4186 Hz). The instruments of the orchestra are used to play long tones across the rising sine wave as it approaches their range – for example, the double basses...
would play the low C when the sine wave begins and would then cease playing the note when the wave reaches the lower range of the next instrument. This creates the desired effect of audible beating that changes speed; initially this speed is equal to the difference between the frequency of the two pitches (sine wave and instrument). Lucier states that after this, the beating patterns begin “...slowing down to zero and not beating when unison is reached and speeding up again as the wave rises above the sustained note. The speed of this gesture doubles with each higher octave” (Lucier, 1998, p. 8).

Lucier then uses a purely instrumental setting for his next use of this phenomenon in a composition. *Fideliotrio* (1987) was a piece for written viola, cello and piano that also utilises the concept of acoustic beating in a musical context. Using the central pitch of A (220 Hz), the strings slowly sweep up and down from a semitone below to a semitone above. Lucier then highlights the phenomena being used by orchestrating the piano to repeatedly play the note (A) “at time intervals proportional to the distances between the pitch of the piano tones and those of the sweeping strings” (Lucier, 1998, p. 8). This rhythmic placement is manifested in the note being played further apart when the two pitches (the sweeping strings and the piano’s note) are closer together, with a spacing of 12 and 13 seconds when the notes are unison, (“which represent the distance (in HZ) between the adjacent semitones Ab and Bb, respectively”), and a note repetition every second when the strings reach the apex of their sweeps (Lucier, 1998, p. 8).

In this piece it can said that Lucier is using an extrapolation of the phenomenon’s characteristics into the scoring of the music, in addition to using the phenomenon itself. The strings sweeping up and down against the static piano note are used to demonstrate the acoustic beating that occurs, however the rhythmic placement of the piano’s notes in the music is dictated by the difference ratio between the frequency of the strings and piano. The latter of these two techniques is really an extrapolative representation of the phenomenon. Lucier seems to reinforce this argument: “The piano part could be considered a code for the strings’ tunings” (Lucier, 1998). Lucier’s use of the word code may be seen as synonymous with the term model, which seems to be the method he has employed in this work.

A more recent work of Lucier’s also explores the use of acoustic beating. *Ever Present*, written in 2002, was a composition written for piano, flute, saxophone and sine tone generator that based its pitch on the shape of walking paths in a garden. The sine tone generator slowly oscillates its pitch in a gradual arc, and the instrumentalists play long tones across the rising
and descending sine tone. This causes beating patterns to occur, at a variety of speeds that are determined by the closeness of the two tunings (Lucier, 2007). This piece is a demonstration of an interpolative modelling, as the acoustic phenomenon (acoustic beating) is a feature of the piece without having its characteristics extrapolated from into other musical components.

8.2.2 Henry Cowell

While Alvin Lucier’s fascination with this phenomenon led him to use it as a highlight in his musical compositions, Cowell used his own fascination with the phenomenon to do something different. While beating occurs when two tones are of very close frequency, Cowell also observed the same phenomenon in pitch intervals by slowing them right down.

Following his early use of the technique that can be called phase shifting, Cowell looked at taking the idea of rhythmic relationships further. In reference to tempos, rhythmic ratios and the harmonic series, Kyle Gann (1995) discusses Cowell’s aim to “bring to rhythm the same structuring possibilities that had already been applied to pitch, in fact, to draw an analogy between the two” (Gann, 1995, p. 5).

This was then realised in the insight that pitch intervals and cross-rhythms are actually manifestations of the same phenomenon, which are differentiated only by speed. For example, the higher pitch in a purely-tuned interval of a perfect fifth has a vibration rate that is one and a half times more than the lower pitch. This can be illustrated in a ratio of 3:2 (Ratios are expressed in ascending order, in line with Gann, 1995, p. 5). According to Cowell, a standard polyrhythm such as a triplet over a duplet – three against two – is the transfer of the “perfect fifth” idea from the realm of pitch to rhythm.

As the vibrations of a tone are slowed down, the pitch becomes lower, and if the frequency descends lower than a threshold of about sixteen cycles per second, the vibrations are no longer heard as a pitch, but as a steady beat. Cowell had a machine invented for him that would keep two sirens tuned at a constant ratio as he slowed them down and sped them up, and he was delighted to hear proof that, as a perfect fifth became slow enough, it turned into a rhythm of three against two. (Gann, 1995, p. 5)

This is an example of how acoustic beating can be viewed in the relationships of pitch intervals. Cowell then used the beating apparent in these intervallic ratios to generate data for rhythm. This is a great demonstration of how a composer can extrapolate the principles of an
acoustic phenomenon into a musical composition. The nature of extrapolation is to use data from one area and apply it to another, which is exactly what Cowell has done by using pitch relationships to generate rhythmic relationships.

When viewing Cowell’s work in this, it becomes easier to see how other composers have followed in this same method. The superimposed rhythms or tempi can even be seen as a representation of acoustic beating, as they have a particular ratio to one another that could also be expressed in terms of the equivalent pitch relationship.

### 8.2.3 Conlon Nancarrow

Conlon Nancarrow also can be seen to have used a representation of acoustic beating in his musical works. Nancarrow began exploring the relationship between rhythm & pitch in the same vein as Cowell, with a piece that related all its rhythms to two different simultaneous tempos, 120 and 210. The two tempos are related in a ratio of 4:7, the same ratio of a purely-tuned minor seventh interval, C to a slightly flat Bb.

Nancarrow then began chronologically climbing the harmonic series, relating different combination tones to rhythm in respect to their frequency ratios. Kyle Gann, in his discussion of Conlon Nancarrow’s work (1995) outlines a number of these pitch intervals that can be expressed in ratio form: “Expressed as pitch, 3:4 gives the perfect fourth, 4:5 the major third, 3:5 the major sixth, and 12:15:20 a first-inversion minor triad, i.e., G B E” (Gann, 1995, pp. 6-7).

Gann also discloses details of the rhythmic ratios that Nancarrow used in his pieces in the form of tempo canons. Each rhythmic ratio relates to a particular pitch interval; some of these are simple 2-pitch intervals while others are chords or tone clusters. The ratio 5:6:7:8 (which is used in Study No. 32) is analogous to an E seventh chord (E, G, Bb & C), the ratio 17:18:19:20 (used in Study No. 36) is related to a tone cluster (containing a C#, D, D# & E), the ratios 24:25 and 60:61 (used in Studies Nos. 43 and 48, respectively) are related to closely spaced harmonics. Nancarrow even uses square root relationships of numbers and the mathematical symbol of pi in relation to their pitch (Gann, 1995, p. 7).

This method of layering tempi in a relationship that mirrors a particular pitch interval follows Cowell’s work in the same area. Since the time of Cowell the concept of rhythm related to pitch has become a well-known phenomenon, but this phenomenon can definitely be seen as an extrapolative representation in any case of its use. Both Nancarrow and Cowell can be seen
using an extrapolation of the behavioural characteristics of acoustic beating in their treatment of rhythms and tempi.

8.2.4 La Monte Young

La Monte Young’s use of sine waves, as discussed earlier, lead him to utilise them in an exploration of other acoustic phenomena; namely, combination tones and beating.

An example of his works that use this technique would be the Betty Freeman Commission (1967), which was in collaboration with Marian Zazeela. This work, mentioned earlier, used two speakers inside a plexiglass light box to emit two sine tones. These two sine tones were set at frequencies tuned an almost imperceptible ratio of 64:63, which was designed to produce the acoustic phenomenon of acoustic beating (Blamey, 2008, pp. 142-143).

Blamey relates this use of acoustic beating with sine tones to Young’s often quoted ideal of “listening inside a sound”:

Using a pair of sine tones in a close ratio such as 64:63 to create a sonic environment emphasises one particular facet of listening inside a sound. While sustain would continue to be enacted through duration and harmonic stasis, profusion of sound is provided not by the number of components used to generate the sound or by the number of frequencies present, but by the acoustical and psychoacoustical phenomena produced by the two frequencies in the form of beats and standing waves. (Blamey, 2008, pp. 143-144)

Of his other commissioned works, the Robert C. Scull Commission (1967) also utilised a similar technique. This work used an array of fourteen closely packed sine tones, where the ratio between the lowest tone and the highest tone was 7:9. What this equates to is twelve different pitches being crammed into the space of a musical major third. This densely packed interval of fourteen pitches created a complex series of interweaving sine tones that produced very closely packed acoustic beating.

This use of acoustic beating evident in Young’s work demonstrates an interpolative representation, as the sine wave generators are used to produce the effect without allowing for any re-interpretation of the phenomenon.
8.3 Modelling Interference/Beating for Musical Compositions

It seems that while Alvin Lucier and La Monte Young pursued the idea of *utilising* the acoustic phenomenon interference/beating it was the work of Henry Cowell and Conlon Nancarrow that demonstrated how the phenomenon’s characteristics could be *represented* through modelling.

The extrapolative concepts set forth by Cowell and Nancarrow seem to dominate one whole area of possibility within this form of representation. These composers used the ratios of pitch intervals to dictate the ratios of rhythms; this demonstrates a functional model of the phenomenon. Their model very literally takes the relational properties of beating and transfers them through a model to draw similarity relations with musical devices. Their model forms a cohesive set of correlations between the points of the phenomenon’s behavioural properties and the related elements of the musical components.

As this demonstrates a functioning model, it is not necessary in this research to produce another model of the phenomenon. Composers wishing to utilise this particular phenomenon can follow the example set by these two composers in utilising the ratios between two pitches to dictate the ratios between rhythms and tempi. This can be used as a scale model or an analogue model to create a varied array of musical effects.

It should be mentioned though that this ratio system can be used further than projected by its use by these composers. Dictating the relationship of a rhythm by the relationship of a pitch is one scenario, and there are certainly others that would apply. Whilst still utilising a pitch interval relationship as the model, the ratio evident could be used to determine structural processes, timbral possibilities, other forms of pitch relationships, dynamic levels, and any other conceivable element of music.
9.0 Conclusion

The conclusions drawn from this research are in reference to the findings evident in the case studies. These case studies were the works of composers who have utilised acoustic phenomena as either a compositional focus or as external stimuli to represent, in addition to the compositional studies, major compositions and outworking of experiments and theories related to acoustic phenomena. Discussions of the effectiveness of the processes used, the compositions created and the models constructed refer back to the criteria for evaluation.

Representation of Acoustic Phenomena through Modelling

In this research, there were certain trends that emerged as to which composers utilised acoustic phenomena or represented them using instrumental techniques and in the case of the latter, which used an interpolative representation and which used an extrapolative representation.

These trends can be seen in the following table (Table 9). The table lists the composers under study in this research and a ✓ is used to signify that composer has utilised the corresponding phenomenon, and the suffix of I or E denotes that the use of that phenomenon was a representation, either interpolative or extrapolative in nature.
Table 9 - Demonstration of Trends in Composers’ Use of Acoustic Phenomena

<table>
<thead>
<tr>
<th>Composers</th>
<th>Beating</th>
<th>Delay</th>
<th>Harmonic Series</th>
<th>Overtones</th>
<th>Phase Shifting</th>
<th>Resonance</th>
<th>Reverb</th>
<th>Sine Waves</th>
<th>Sound Waves</th>
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From this display of data and from the research behind it, a number of observations can be made. The majority of composers under study utilised acoustic phenomena, and secondary to this was the interpolative representation of them. Extrapolation was infrequently evident.

The acoustic phenomenon of phase shifting seemed to be primarily employed by Reich and the Totalists, and their use of this phenomenon was through an interpolation of the rhythmic device evident in the phenomenon into an imitative rhythmic device in the musical compositions.

Similarly, the works of Cowell and Nancarrow can be seen as representations of phase shifting through interpolative means. Although their intentions behind the pieces don’t seem to indicate an awareness of this connection, it still seems apparent from an objective perspective that there is a relationship between their use of tempo canons, polyrhythms and other such devices.

The extrapolation of the phenomenon phase shifting seemed to only appear in the use of the isorhythm, such as in the work of Messiaen. Messiaen’s work in this area was not given a full exploration as it seems that he did not claim to be representing this phenomenon. The reason that the number of composers using an extrapolative form of representation of this acoustic
phenomenon seems so limited in the table above is because Messiaen was only one of many examples of composers who have used isorhythms; this is not a technique exclusive to him. The conclusion drawn from these results is that the extrapolation of this particular phenomenon into musical works could benefit from some more substantial growth. This could be achieved through the use of the model provided.

The most significant body of work presented in the field representing acoustic phenomena was contributed by the composers of spectral music. Many of their proponents, such as Grisey and Murail, utilised an instrumental imitation of synthesis and other forms of representing the harmonic series or particular sound spectra. In addition to the specific techniques, the fact that as a compositional group they have sound as a guiding principle in shaping the music means that the majority of their works can be seen as an extrapolative representation of sound phenomena, as at least in some way an association with sound exists within each work.

The connections between groups of composers across the time span under research is only limited. Authors of musical texts and musical commentators frequently draw a connection between Spectral composers and their supposed predecessors, such as Stockhausen, Ligeti, Varese, Scelsi, Nørgård and Debussy. Others draw a connection between Cowell, Nancarrow, Reich, and the Totalist composers. Few connections are made between the proponents of each group.

This research has attempted to bridge this gap, as all the composers under study within this research – from both lists of composers – are connected by the shared principle of the utilisation and representation of acoustic phenomena. As demonstrated throughout the research, some of these composers successfully modelled acoustic phenomena for their music whilst a majority of other incorporated the phenomena into the compositions.

The models of acoustic phenomena created in this research seek to represent the underlying behaviour of each particular occurrence as aptly as possibly. Through the on-going analysis of each model in the chapters of this paper, it is generally apparent that the models achieve the desired aim of their representation. For the most part, the models were each able to draw substantial similarity relations between the subject and the relevant musical equivalents. The phenomena under study were all chosen due to the hypothesis that they would all make apt models for this type of representation and that they can be seen being used in this way by a number of composers.
While the models discussed in this thesis may not be the only possible variety available for each phenomenon, they will at least propagate the concept behind this type of compositional method. The possibility of the creation of new and more refined methods for the modelling of acoustic phenomena is a welcome invitation for any and all composers who find the same passion in this area of composition as this researcher.

The phenomena under study in this research have been explored through case studies of other composers modelling their behaviour and through additional case studies in the form of musical compositions generated by this researcher. Having analysed these case studies to the extent available within this research, conclusions regarding the validity of the phenomena chosen for use and the significant number of successes can now be drawn.

Of all the acoustic phenomena studied, there were few that did not meet the criteria for a good scale model; the sound wave was one example of this, as it lends itself more to the as an analogue model to provide a metaphorical framework for guiding compositional practice. Some phenomena were closely related to others which precluded their use as separate entities, such as reverberation, echo, the harmonic series and harmonic spectra. Other than these examples, the phenomena under study were shown to be of use as adequate models for this type of representation in music. This was demonstrated through the extensive use of various phenomena by a number of composers and the possibilities touched upon in this research through the composition of musical works and the theorising of models. In addition to the phenomena under study, a brief list of additional sonic occurrences was given in the final chapter that alludes to a far greater area of acoustic phenomena that could potentially be used in the same way as those included in this research.

This means that the body of phenomena available to model for music is certainly significant. It also can be seen as substantial enough to justify this research, as it allowed for a number of conclusions to be drawn regarding topics arising in this.

**Application of Models**

Further uses of the models in this research can be exploited through different applications. The purpose of creating models for each acoustic phenomenon under study in this research was so that composers could take these theoretical models and use them in application to the creation of musical compositions. This primarily allows the use of an extrapolative form of representation, as the values of each function of the models can be substituted for a variety of different musical components.
In addition to this, it’s possible that these models could be used in application to existing works. It stands to reason that if a composer were to generate new musical material for which to subject a particular model of a sound phenomenon to, the same process could also be used for existing musical material.

**Modelling through Instrumental Techniques and Electronic Mediums**

Representing acoustic phenomena involves representing their core values in musical composition; sometimes the acoustic phenomena being represented are produced electronically, rather than acoustically. The idea behind the compositional ethic in this research is to represent acoustic phenomena *instrumentally*, as a way of bringing music back to a focus on sound in an unrefined form and celebrating sound as a singular important entity in music. A number of composers successfully achieved this instrumental representation, while others opted to use electronic mediums to convey the acoustic phenomena. Of the composers discussed in this research, there are equitable numbers of proponents in the use of each method.

Reich’s compositions demonstrate a progression from using electronic means to instrumental technique in modelling phenomena. After reproducing the effect of phase shifting in its tape loop manifestation, he then represented this electronic phenomenon using acoustic instruments and continues to compose in this fashion. Grisey represented the partials in his composition through instrumental techniques; by comparison, Alvin Lucier put the spotlight on resonance through actually demonstrating it in his compositions, such as the resonance within a teapot in *Nothing is Real*. Reich represents the acoustic phenomenon of phase shifting in *Drumming* through playing the evident rhythmical function on a number of drums; La Monte Young celebrates the sound of a phone line humming by using a sine wave generator and his “turtle motor”.

The conclusions drawn thus far are directly relatable to the primary line of enquiry of the research and the *criteria for evaluation* posed at the beginning of the paper. Having addressed these points, it is worth discussing a number of other considerations that arose from the results of this research that aren’t necessarily directly related to the thesis or criteria posed. These conclusions present ramifications that may be of import to the research area.

**Process Music**

A lot of the musical works studied within this field of research – and also those created as a result of it – tend towards being “process driven music”. This was typified in works by Steve
Reich, whose “gradual musical process” style of composition meant that once a process was set in place, he had the restriction (or the luxury, depending on perspective) of being unable to interfere, and as such became part of the music’s audience.

Alvin Lucier lists a number of works that can be described as process driven, such as those by Gordon Mumma, Robert Ashley, James Tenney and Steve Reich, as they share a common compositional principle: “an action or process, set into motion and sustained throughout the course of the work, [that] produces unexpected and complex results” (Lucier, 1998, p. 11).

Certainly, with “multi-tap phase shifting” in rhythms (or canons) that was explored in the compositions of this research the musical material is subjected to a process; the rhythms are set in place and left to run themselves, indefinitely, until it is deemed long enough for the interaction of musical ideas to be observed. However, in pieces like “Marimba Delay” the process is somewhat different. Although the process parameters are the same, the musical material to which it is applied differ throughout, making a more complex interaction to observe. This is a more limitless area to be explored; when a set amount of musical material is used the process becomes more like a “controlled” experiment, whereas the expansion of the musical material then blows the “control” out of the water, as many variables enter into the equation.

In the case of this researcher’s pieces becoming process-driven music, they then could be seen as sharing characteristics with the minimalist style of musical composition, as well as that of (the broadly named) experimental music. Cage’s use of the I Ching to determine certain attributes of his compositions demonstrates a use of a process.

While this research doesn’t have the scope to identify and explore all types of process-driven music, it is important that it identifies how it differs with existing compositional ideas of this nature and justifies the validity of those differences. So, from working through this research, it seems that it adequately sits under the umbrella of process-driven music, even in cases where the musical subject material is more expansive than the traditional amount used by other types of process music.

Self-Evident Compositional Focus
An important stipulation within this research’s criteria for evaluation was that the ensuing compositions be self-evident; that is, the focus of each composition can be clearly identified by an audience. This ethic was put forth by Reich, and this researcher intends to follow the same
line of reasoning: “I am interested in perceptible processes. I want to be able to hear the process happening throughout the sounding music.” (Reich, 1968)

It could be said of representing acoustic phenomena however that even the most obvious process still requires an attentive audience to perceive the effects taking place. Even when a musical composition doesn’t seek to represent this particular stimulus a well-tuned audience can still be able to “listen inside the sound” as La Monte Young did. As Peter Blamey puts it: “…The use of simple acoustic phenomena as the subject of compositions equally required an attentive audience keyed into listening for aesthetic experience in sounds beyond the instrumental” (Blamey, 2008, p. 3).

Reich’s works all seem to stick with the ethic of aiming to make the process audible for the listener. In Come Out, It’s Gonna Rain, Violin Phase, Piano Phase, Tellehim, Proverb, Music for 18 Musicians and a number of other works he successfully transfers the ideas of the composition through the performers to the audience. One can hear immediately in Proverb that his phase-shifting “canon” is being used in the juxtaposed vocal lines, followed by a mensuration canon as he slows the parts down. Reich justifies this obvious presentation by comparing it to serial processes and experimental music;

John Cage has used processes and has certainly accepted their results, but the processes he used were compositional ones that could not be heard when the piece was performed... The compositional processes and the sounding music have no audible connection. Similarly in serial music, the series itself is seldom audible... What I’m interested in is a compositional process and a sounding music that are one and the same thing. (Reich, 1968)

Even with other composers studied in this research it seems that their compositional focus points are self-evident; Alvin Lucier makes his focal points extremely obvious by, at times, including the focus as a sound experiment or visual feature on a stage. Music for Piano and Oscillators is one of Lucier’s many works that include the use of the sine wave, as a celebration of its unique sound. The continuously oscillating sound behind the piano is clearly the focal point of the piece, as it colours the piano’s tonality in amazingly different ways.

Conlon Nancarrow’s use phase shifting in the form highly complex super-imposed rhythmic canons can be clearly heard, although it does tend to get a little unclear as to what exactly is happening due to the levels of complexity. Going further down the spectrum of clarity,
however, we come to Gerard Grisey’s works – whilst Partiels clearly demonstrates a representation of synthesis in musical composition, works like Talea that seek to represent an overtone cycle tend to lose their translucence. Further compositions of the spectral music movement also obscure their intention – this is due to the fact that within this research, it was found that this musical genre was one of the few to actually use extrapolation rather than interpolation as the method of representation of sound phenomena.

Similarly, all the compositions that stem from this research – Vocal Resonance, String Resonance, Marimba Delay – have interpolated data from their particular acoustic phenomenon, and as a result their musical focus is clearly evident. Inadvertently, the compositions within this research interpolated their data due to the a priori requirement of the process needing to be self-evident, which would also seem to be there were no compositions made that made use of extrapolation as a data-gathering tool.

The choice, then, seems to be simple – in composing musical compositions that seek to represent acoustic phenomena, a composer could either use an:

1. **Interpolative model** of the phenomenon’s behaviour and characteristics to transfer to musical devices, which allows the acoustic phenomenon to be clearly heard or represented from within the musical composition, or an

2. **Extrapolative model** of the phenomenon’s behaviour and characteristics to translate into musical terminology, which obscures the original focus of the musical composition – it doesn’t allow the acoustic phenomenon to be clearly represented.

While either option would still be a valid choice for a composer to use, they could not expect to have a clearly evident compositional focus whilst using extrapolation – just as an artist who paints “metaphorically” from an external stimulus can expect patrons to spend countless hours pondering over the hidden meaning behind the canvas.
10.0 Bibliography

10.1 Reference List

   http://www.thefreedictionary.com/acoustic+phenomenon


Blamey, P. J. (2008). Sine Waves and Simple Acoustic Phenomena in Experimental Music - with Special Reference to the Work of La Monte Young and Alvin Lucier. (Doctor of Philosophy), University of Western Sydney, Western Sydney.

   http://www.angelfire.com/music2/davidbundler/grisey.html


http://dictionary.reference.com/browse/resonance


http://alucier.web.wesleyan.edu/listen.html


10.2 List of Musical Works

Debussy, Claude

- *La Cathédrale Engloutie* (1910).
- *La Mer* (1905)
- *Prélude a l’après-midi d’un faune* (1894)
- *Trois Nocturnes* (1900)

Grisey, Gérard

- *Partiels* (1975)
- *Vortex Temporum* (1996)

Lucier, Alvin

- Crossings (1982)
- Fideliotrio (1987)
- I am Sitting in a Room (1969)
- Music for Piano with Amplified Sonorous Vessels (1991)
- Nothing Is Real (1990)
- Quasimodo the Great Lover (1970)
- Queen of the South, The (1972)
- Seesaw (1983)
- Vespers (1968)

Messiaen, Olivier

- *Couleurs de la Cité Céleste* (1963)

Murail, Tristan

- *Me’moire-e’rosion* (1976)
- *Gondwana* (1980)
Nancarrow, Conlon

- *Study No. 21/Canon X* (1948-1960)

Nørgård, Per

- *Voyage into the Golden Screen* (1968)

Ravel, Mauricio

- *Bolero* (1928)

Reich, Steve

- *Come Out* (1966)
- *Drumming* (1970)
- *It’s Gonna Rain* (1965)
- *Piano Phase* (1967)
- *Violin Phase* (1967)

Scelsi, Giacinto

- *Anahit* (1965)
- *Four Orchestral Pieces on a Single Note* (1959)
- *Fourth String Quartet* (1964)

Stockhausen, Karlheinz

- *Hymnen* (1967)
- *Mikrophonie I* (1964)
- *Mixtur* (1964)

Westlake, Nigel

- *Fabian Theory* (1987)
Young, La Monte

- Betty Freeman Commission (1967)
- Claes and Patty Oldenburg Commission (1967)
- Composition 1960 #7 (1960)
- Composition 1960 No. 7 (1960)
- Dream House (1974)
- Robert C. Scull Commission (1967)
- String Trio (1960)
- Tortoise, His Dreams and Journey, The (1967)
- Vision (1959)
- X for Henry Flynt (1960)
11.0 Appendices

The CD track listings outlined in the list of appendices refers to the appendices contained on the following audio CD (appendices 1.1-1.7).

The musical scores of the compositions, which include their performance notes and individual instrumental parts, form the appendices 2.1-2.7. These are included in the following pages.

Appendices 1.1-1.7

(Audio CD)
Appendices 2.1 – 2.7

Musical Scores

From page 254 to the end of the document.

Compositional Studies in Phase Shifting

2.1 Phase Shifting Compositional Study 1 – Clave
2.2 Phase Shifting Compositional Study 2 – Drums
2.3 Phase Shifting Compositional Study 3 – Breath
2.4 Phase Shifting Compositional Study 4 – Piano

Resonance Compositions

2.5 Resonance Composition 1 – “String Resonance”
2.6 Resonance Composition 1 – “Vocal Resonance”

Delay Composition

2.7 Music Score 7......... Delay Composition 1 – “Marimba Delay”
Phase Shifting for Clave

David Tracy

Repeat each section until all previous parts have synchronised.

To finish piece, either repeat in reverse order by stopping one part at a time (beginning with "9"), or wait until all previous parts have synchronised and perform a final note together.

©2009
Phase Shifting for Drums
A Study in Poly-Rhythm

David Tracy

Demonstrates a superimposed pattern of different rhythmic units. The rhythmic units are in lengths of 1, 3, 4 & 5 quaver groupings. The hi-hat plays the 1 (every quaver beat), the kick plays the 3, the hi-hat foot pedal plays the 4 and the snare plays the 5.

All groupings synchronise at the repeat of the piece. The study may be played as many times as desired.

©2009
Phase Shifting for Breath
For 8 Amplified Voices

Each part should attempt a slightly different breathing style than the previous. Styles include:
with puffed cheeks and pursed lips, open mouthed,
ragged & hoarse, high pitched (open mouth shape),
low pitched (round mouth shape) etc. Variations can be used to signal the end of a repeated section,
in which case vary the last few breaths of the bar
Phase Shifting for Piano
For one, two or four pianists on one piano

©2009

To end, wait until all previous parts are synchronised and finish on first note of next bar, or remove each part in reverse order, starting with the last, until none remain.
String Resonance

For 13 Violins with Sound-Sensitive Lights

and 3 Percussionists

David Tracy
Performance Notes

This work is written as a representation of the acoustic phenomenon “resonance”, using the medium of strings to convey it. Specifically, it represents “sympathetic resonance” – this is the phenomenon in which a sound source is resonated by another sound source, which in effect sustains the original sound for an extended duration.

To represent this, a solo violin line (Violin 1 - Solo) is “resonated” by another 12 violin parts (Violins 2-13 - Ensemble). Each note that Violin 1 plays is re-sounded by one other violin part. The Violin 13 part also joins Violin 1 as a soloist halfway through the piece.

Resonance Parameters
The parameters of the resonance at the beginning of the composition are as follows:

- Tempo: $\frac{\dot{\text{m}}}{\text{a}} = 90$bpm
- Duration of resonated notes: 8 beats (\(\omega\)) / 2 bars
- Sample length (to be resonated): 1 quaver beat
- Attack: Approximately 1 semi-quaver beat after transient

The delay parameters and tempo change throughout various sections of the piece. The resonance parameter changes are indicated by “resonance reset”, followed by the new length of the resonance decay.

Temporal Placement
Each note in Violins 2-13 is written to have an attack that corresponds with the start of a note in Violin 1’s part. To simplify notation the attack of each note is exactly the same as Violin 1, however to accurately represent “resonance” each entry should be delayed by approximately 1 semi-quaver. This does not need to be exact; the easiest way to achieve this delay is to wait for the note to be sounded by Violin 1 and then re-sound the corresponding note.

At the ensemble director’s discretion this delayed temporal placement may be removed in favour of a simplified performance, in which case all rhythms would be played exactly as written.
Dynamics
In order to adequately represent the acoustic phenomenon each accompanying part (Violins 2-13) should have the following dynamics applied to each of their notes, in addition to the dynamics written.

The 7 “resonating” violins (Violins 2-13) all have notes that last 8 beats, and these notes should be dynamically bevelled down – from a dynamic that matches the transient (note played by Violin 1) into near-silence. For example, if Violin 1 sounds a note at forte, then the note sung by the accompanying violin should start at forte then decrescendo to pianissimo possibile over a period of 8 beats.

Notation
The parts for Violins 2-13 have a reduction of Violin 1 (and in some cases, also Violin 13 at different points) included in their individual scores. The Violin 1 part supplied with each part has been provided to assist in the reading and understanding of the accompanying violin entries, duration and dynamic levels.

Staging of the Ensemble
The ideal stage layout for the ensemble is to have all players in a straight line facing the audience, possibly with a slight inward arc to facilitate ensemble interaction. This setup allows the “resonance” of each note played by Violin 1 (and Violin 13) to be visually demonstrated, as the scoring of the piece involves the notes being resonated consecutively by each part down the line.

Percussionists
The percussionists are required to play a doumbek, riq and a large tom. If a doumbek is not available, other varieties of hand drums may be substituted. Similarly, if a riq is not available a normal tambourine will be adequate. The large tom can be an animal-skin type drum struck with a mallet or can be a modern floor tom struck with a mallet. The percussionists may be positioned anywhere on stage in relation to the string players.
**Sound-Sensitive Lights**

To further enhance the performance, this piece is written with an option of utilizing sound-sensitive LED lights. These are widely available and fairly cheap to purchase, and ideally, the ensemble will need 12 blue lights and 2 red lights. Often each light comes with a switchable selection of colours, in which case the whole ensemble may use the same.

These lights should be attached to the body of the violin with the self-adhesive strip supplied with the lights, or to avoid damage to the instrument a clip may be used to attach to either the performer’s clothing or the front of the shoulder-rest. Violin 1 requires a red light and Violins 2-12 require blue lights, to indicate the function of the soloist and ensemble (respectively).

Violin 13 is an exception, as it requires both a blue and a red light. If using individually coloured lights, both can be attached at once or if using a switchable-coloured light then only one is required. Violin 13 begins the piece with a blue light switched on to indicate it is part of the “resonating” ensemble. In section E, Violin 13 becomes a soloist with Violin 1, at which point the blue light needs to be changed for a red (either switching the colour selection of the single light or turning the blue off and the red on if using two lights). This is indicated in the score and the individual part.
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

David Tracy
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin 1
with blue light - Soloist

\( \text{(d = c. 90)} \)

David Tracy

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String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists
David Tracy

Violin 2
with blue light - Ensemble
No Vibrato. Decay each note to pianissimo possible over length of note duration
\( \text{\( \text{d} = \text{c} \approx 90 \)} \)

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String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin 3
with blue light - Ensemble

No vibrato. Decay each note to pianissimo possible over length of note duration

($ = c. 90$)

© 2011
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin 5
with blue light - Ensemble

No vibrato. Decay each note to pianissimo possible over length of note duration

David Tracy

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String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

No vibrato. Decay each note to pianissimo possible over length of note duration

\( \text{\textcopyright 2011} \)
String Resonance
Violin 6

\[ \text{Larghetto (M.M. } \frac{d}{c} \text{ 60)} \]
\[ \text{Resonance reset } \approx 8 \text{ crochet beats} \]
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

David Tracy

Violin 7
With blue light - Ensemble
No vibrato. Decay each note to pianissimo possible over length of note duration
(a = c. 90)

© 2011
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin 8
with blue light - Ensemble

No vibrato. Decay each note to pianissimo possibile over length of note duration

David Tracy
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

No vibrato. Decay each note to pianissimo possible over length of note duration
($= c. 90$)

© 2011
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin II
with blue light - Ensemble

No vibrato. Decay each note to pianissimo possible over length of note duration

\( \text{\textit{c. 90}} \)

\( \text{\textit{pp}} \)

A

\( \text{\textit{mf}} \)

B

\( \text{\textit{f}} \)

\( \text{\textit{ff}} \)

© 2011
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

Violin 12
with blue light - Ensemble

David Tracy

No vibrato. Decay each note to pianissimo possible over length of note duration

A

B

© 2011
String Resonance
Violin 12

Resonance reset – decay of 12 beats (crotchet)
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists
David Tracy

Violin 13
With blue & red lights. Ensemble & Soloist.
No vibrato (in ensemble section). Decay each note to pianissimo possible over length of note duration

© 2011
String Resonance
For 13 Violins with Sound-Sensitive Lights & 3 Percussionists

David Tracy

© 2011
Performance Notes

This work is written as a representation of the acoustic phenomenon “resonance”, using the medium of voice to convey it. Specifically, it represents “sympathetic resonance” – this is the phenomenon in which a sound source is resonated by another sound source, which in effect sustains the original sound for an extended duration.

To represent this, a solo vocal line (Voice 1) is “resonated” by another 7 voice parts (Voices 2-8). Each note that Voice 1 sings is sung by one other voice part.

Resonance Parameters
The parameters of the resonance and composition are as follows:

- Tempo: \( \text{\textbf{\textit{\#}} = 100\text{bpm}} \)
- Duration of resonated notes: \( 8 \text{ beats (\textasciitilde}) / 2 \text{ bars} \)
- Sample length (to be resonated): \( 1 \text{ quaver beat} \)
- Attack: Approximately 1 semi-quaver beat after transient

Temporal Placement
Each note in Voices 2-8 is written to have an attack that corresponds with the start of a note in Voice 1’s part. To simplify notation the attack of each note is exactly the same as Voice 1, however to accurately represent “resonance” each entry should be delayed by approximately 1 semi-quaver. This does not need to be exact, and in fact the easiest way to achieve this delay is to wait for the note to be sung by Voice 1 and then vocalise the corresponding note.

Dynamics
No dynamics are written, however to adequately represent the acoustic phenomenon each accompanying part (Voices 2-8) have the following dynamics applied to each of their notes.

The 7 “resonating” voices (Voices 2-8) all have notes that last 8 beats, and these notes should be dynamically bevelled down – from a dynamic that matches the transient (note sung by Voice 1) into silence. For example, if Voice 1 sings a note at forte, then the note sung by the accompanying voice should start at forte then decrescendo to a pianissimo (as quiet as possible) over a period of 8 beats.

Notation
The parts for Voices 2-8 have Voice 1 included in their individual scores. The Voice 1 part supplied with each part has been modified to assist in the reading and understanding of the accompanying voice entries, duration and vowel sounds. Each note of Voice 1 that is being resonated by that particular part has been enlarged, and has its stem up – all other notes of the melody have stem down and of regular size.
Vowel Sounds

The accompanying voice parts only resonate the vowel sounds of each word in the melody. These vowel sounds are phonetically spelled out – for example, “oh” (as in the word “open”) and “Ay” (as in the word “day”). In some instances, where the vowel sounds were more ambiguous if spelled out (as in the word “in”, “of” and “world”) the full word is written, but with the consonants in brackets at the end. In this case, only the vowel sounds are to be vocalised – the consonants are only to assist in understanding the correct vowel sound.

Ensemble Considerations

The consideration of the constitution of the ensemble performing this piece is necessary. Ideally, the entire ensemble should be of the same gender – if the lead vocalist is female, it can sound odd having a male voice accompanying her when the accompaniment is intended to be a resonance of the original sound. However, a mixed ensemble can be effective too, as it lends an entirely different quality to the overall sound. In the end, the ensemble director can take the decision into their hands at their own discretion.

Lyrics

This composition features as its lyrical content the words of Edgar Allen Poe’s A Dream. These are supplied in their entirety below:

```
In visions of the dark night
I have dreamed of joy departed –
But a waking dream of life and light
Hath left me broken-hearted

Ah! What is not a dream by day
To him whose eyes are cast
On things around him with a ray
Turned back upon the past?

That holy dream – that holy dream
While all the world were chiding,
Hath cheered me as a lovely beam
A lonely spirit guiding

What though that light, thro’ storm and night,
So trembled from afar –
What could there be more purely bright
In Truth’s day-star?
```
Vocal Resonance

Voice 1: In visions of the dark night I've dreamed of joy depart
Voice 2: p i - (s) ah ce mf ah
Voice 3: o - (n) ah mf o - (f) mp oh
Voice 4: p o - (f) i
Voice 5: p ah, mf o - (y)
Voice 6: p I - er
Voice 7: p
Voice 8: p

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Vocal Resonance

Ah! What is not a dream by day
To him whose eye are cast on

things around him with a ray
turned back upon the path?

mf

mp

f

mf

mp

f

mf

mp

f

mf

mp

f

mf

mp

f

mf

mp
Vocal Resonance

That holly dream that holly dream While all the world were chi

a - (i) oh

mf or

oh

mf or

ce

mf or

ce

mf or

ce

mf or

f a - (i) mf l - (ii) l

ding

Hath cheered me as a love-ly be-am A lonely spirit gui

l - (hg)

f A

ce

f ce

ce

f ce

ce

f ce

f ce

f ce

f ce
Vocal Resonance

Voice 1

In visions of the dark night I've dreamed of joy departed
But a waking dream of life and light Hath left me broken hearted

Ah! What is not a dream by day To him whose eyes are cast
Things around him with a ray turned back upon the past?

That holy dream that holy dream While all the world were ringing
Hath cheered me as a lovely beam A lonely spirit guiding

What though that light, thro' storm and night, So trembled from afar

What could there be More purely bright

Day Star?

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Vocal Resonance

In visions of the dark night I've dreamed of joy departed

But a waking dream of life and light Hath left me broken hearted

Ah! What is not a dream by day To him whose eyes are cast

On things around him with a cry turned back upon the past?

That holy dream that holy dream While all the world were chiding Hath cheered me as a love-ly beam A
Vocal Resonance

lo
    nely    s
    pi
    ri
    t

    gu
    i
    n
    d
    i

    ing

    What

    though

    that

    light,

    thr

    o
    r'

    i

    ng

    st

    o

    m

    n

    and

    night.

    So

    trem

    bled

    from

    a

    far

    What

    could

    there

    be

    (rd)

    ah

    m

    b

    ly

    brigh

    t

    f

    I

    n

    m

    p

    ooh

    i

    n

    Tru

    th's

    Da

    y

    ooh

    s

    Star?

    f

    Ah
In visions of the dark night I've dreamed of joy departed
But a waking dream of life and light Hath left me broken hearted

Ah! What is not a dream by day To him whose eyes are cast
On things around him with a ray turned back upon the past?

That holy dream that holy dream While all the world were abiding
Hath cheered me as a lovely beam A
Vocal Resonance

lonely spirit guiding

What though that light, thro'

_ oh_ _ I_

storm and night, So trem-bled from a far What could there be_

mf _ _ (r) ah_

More pure-ly bright I _ _ n

mp _ _ (r) i _ _ (n)

_ Truth this_ Day

_ ooh_

Star?

f Ah
In visions of the dark night I've dreamed of joy departed.

But a waking dream of life and light Hath left me broken hearted.

Ah! What is not a dream by day To him whose eyes are cast

On things around him with a ray turned back upon the past?

That holy dream that holy dream While all the world were chiding

Hath cheered me as a lovely beam A
Vocal Resonance

David Tracy

Voice 5

Voice 1

In visions of the dark night I've dreamed of joy departed

Voice 5

mf
t

But a walking dream of life and light Hath left me broken hearted

ah

mf

Ah! What is not a dream by day To him whose eyes are cast

mf

On things around him with a ray turned back upon the past?

That holy dream that holy dream While all the world were chiding Hath cheered me as a lovely beam

mf

ee

©2011
In visions of the dark night I’ve dreamed of joy departed
But a waking dream of life and light Hath left me broken hearted
Ah! What is not a dream by day To him whose eyes are cast
On things around him with a ray turned back upon the past?
That holy dream that holy dream While all the world were chiding
Hath cheered me as a lovely beam A
Vocal Resonance

1. Lonely spirit guiding
   What though that light, thro'

2. storm and night,
   So trembled from afar
   What could there be

3. More purely bright
   I

4. Truth's Day
   (n)

5. Star?
   f
   Ah
In visions of the dark night I've dreamed of joy departed
But a waking dream of life and light Hath left me broken hearted
Ah! What is not a dream by day To him whose eyes are cast
On things around him with a ray turned back upon the past?
That holy dream that holy dream While all the world were chiding
Hath cheered me as a lovely beam A
Vocal Resonance

lonely spirit guiding

What though that light, thro'

storm and night,
So trembled from afar

What could there be

--- (II)

More purely bright

I'm

--- (n)

Truth's Day

Ooh

mf

Star?

Ah
Vocal Resonance

David Tracy

Voice 8

Voice 1

In visions of the dark night I've dreamed of joy departed

But a waking dream of life and light Hath left me broken-hearted

Ah! What is not a dream by day To him whose eyes are east

things around him with a ray turned back upon the past?

That holy dream that holy dream While all the world were chiding

Hath cheered me as a lovely beam A lonely spirit gui
Vocal Resonance

What though that light, thro’ storm and night, So trem-bled from a--far

(i) mf a-- (i) eh

What could there be More pure-ly bright

mp oeh

ll--nn

Tru-th's

i--(n) oeh

Day

Star?

mf Ay

f Ah
Marimba Delay
Solo for 4-mallet marimba

With alternate arrangements for Duet & Trio

David Tracy
Performance Notes

This work is written as a representation of the acoustic phenomenon “delay”, using the medium of marimba to convey it. Delay is an artificially created acoustic phenomenon which is typically found in the form of a guitar effects pedal, or used as an effect on voices & instruments in recordings. Delay is the repetition of a sound at a set temporal interval, for a specified length of time. The repetition usually dynamically decreases over the specified length of time until it stops, although it maintains the same clarity of the original signal.

To represent this, a melody line has had the following delay parameters applied to it, and then the ensuing ‘counter-melodies’ have been amalgamated into the one staff of music so that the entire piece can be played by a single player.

**Delay Parameters**
The parameters of the delay are as follows:

- **Tempo:** ________________  \( \text{\textup{\textbar}} \text{\textup{\textbar}} \text{\textup{\textbar}} \text{\textup{\textbar}} \text{\textup{\textbar}} = 90 \text{bpm} \)
- **Time Signature:** ___________ 4/4 time
- **Sample length (to be delayed):** ________________ 1 (crotchet) beat
- **Delay interval:** ________________ 1 (crotchet) beat
- **Decay length:** ________________ 7 (crotchet) beats
- **Decay velocity:** ________________ Initial velocity = original note, final velocity = pianissimo

**Dynamics**
The dynamics in this piece are written using a graphic notation. The primary dynamic markings are the normal ‘forte’, ‘mezzo-forte’ method of notation, however throughout the piece a system is used in which the size of the note heads represent the volume of the note.

Each line of “delay” progressively decreases in volume, and as several “delay” lines are running at the same time it means that several layers of dynamic alteration are occurring concurrently. This graphic notation was the easiest way to communicate these simultaneous layers of dynamic control.

For example, when a dynamic change would normally be written like this:

![Dynamic Change Example](image1.png)

In *Marimba Delay* the same dynamic change is written like this:

![Dynamic Change Example in Marimba Delay](image2.png)

As the delay parameters stipulate, the initial volume of the “delayed” notes are to be the same as the note in the melody line that they are repeating, and then decrease in volume over 8 crotchets beats. The last note is to be played as soft as possible, with the notes in between the first and the last decreasing in equal increments using these two guide points.
Notation

As this piece has a melody line that is being “delayed”, the melody is written as a separate entity on the single staff with the counter-melodies. The melody and counter-melodies have their stems pointing in different directions, however they do not always stay the same – for example, the melody notes have their stems up at some points and down at others, but for each instance the counter-melodies have the opposite. The melody is also written separately in terms of its rhythmic notation, so that it can be understood the way it was originally intended.

The melody then stands out from the accompaniment, like in this example:

![Music notation example]

Melody Reduction

In this solo, there are two staves written – the lower is the staff that is played, and the top is a reduction of the original melody. This was included to help understand the notation, placement and feel of the original melody whilst being crammed in amongst the counter-melodies. If there is any confusion in regards to where a melody note sits rhythmically, this reference will help.

Alternate Ensemble Arrangements

As this piece can be quite challenging to play with all the concurrent dynamic changes, included are a number of alternate ensemble arrangements. These arrangements make the playing of this piece easier, as there are either more players to execute the notes, no dynamics, or both. The solo arrangement also comes without the dynamic indications, as the piece can be played in this fashion instead.

The possible arrangements provided are:

1. **Solo for 4-mallet marimba**
2. **Solo for 4-mallet marimba** – with no dynamic indications
3. **Duet** – one person performing the melody, the other the accompaniment. Written on a single score, 2 staves
4. **Duet** – as above, no dynamic indications
5. **Trio** – one melody player, two accompanists – with dynamics. This includes the score as well as the individual parts 1-3.

Instruments & Mallets Required for Alternate Arrangements

The duet and trio arrangements both require the use of 2 marimbas – one for the melody player, and the other for the accompanying counter-melodies. The accompanying parts for the trio (parts 2 & 3) can be played on one marimba, although both players will be playing quite close together.

In the duet Part 2 requires the use of 4 mallets, whilst Part 1 only requires 2. Similarly, in the Trio Part 1 uses only 2 mallets, Parts 2 & 3 both use 4 mallets (or 3 if preferred.)
Marimba Delay
Solo for Four-Mallet Marimba

Full part & melody reduction
with dynamic indications

David Tracy
Marimba Delay
Marimba Delay
Duet for 2 Marimbas

Parts 1 & 2 with
dynamic indications

\[ \text{\textbf{Marimba Delay}} \]
Marimba Delay
Duet for Two Marimbas

Parts 1 & 2
(no dynamic indications)

\( \text{\textbf{ff}} \)