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An interactive music system based on the technology of the reacTable

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USE OF THESIS

The Use of Thesis statement is not included in this version of the thesis.

Abstract

The purpose of this dissertation is to investigate and document a research project undertaken in the designing, constructing and performing of an interactive music system. The project involved building a multi-user electro-acoustic music instrument with a tangible user interface, based on the technology of the *reacTable*. The main concept of the instrument was to integrate the ideas of 1) interpreting gestural movement into music, 2) multi-touch/multi-user technology, and 3) the exploration of timbre in computer music.

The dissertation discusses the definition, basics and essentials of interactive music systems and examines the past history and key features of the three main concepts, previously mentioned. The original instrument is observed in detail, including the design and construction of the table-shaped physical build, along with an in-depth look into the computer software (*ReacTIVision*, *Max MSP* and *Reason*) employed. The fundamentals and workings of the instrument – *sensing/processing/response, control and feedback*, and *mapping* – are described at length, examining how tangible objects are used to generate and control parameters of music, while its instrumental limitations are also mentioned. How the three main concepts relate to, and are expressed within, the instrument is also discussed.

An original piece of music, with an accompanying video, entitled *Piece for homemade reacTable*, composed and performed on the instrument has been created in support of this dissertation. It acts as a basic demonstration of how the interactive music system works, showcasing all the main concepts and how they are put in practice to create and perform new electronic music.

Declaration

I certify that this thesis does not, to the best of my knowledge and belief:

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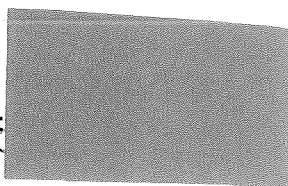
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CHAPTER 1

Interactive control and sound generation

In this first chapter, I will discuss the definition of interactive music systems given by various electronic music composers, at different points in time. Although earlier definitions may be outdated or incomplete, this allows for a greater spectrum of consideration of the issue, as it also gives the sense of the development of interactivity in electronic music. What classifies an interactive music system is discussed, as are the fundamentals, that is: 1) Sensing, processing, response, 2) Control and feedback, and 3) Mapping (Rowe, 1993). The principal concepts of motion control and multi-touch/multi-user interfaces are examined, which relate to the interface and 'playing' of new interactive music systems. The exploration of timbre in electronic computer music is investigated, while going into detail about three forms of sound synthesis: 1) Frequency modulation synthesis, 2) Additive synthesis, and 3) Subtractive synthesis. The idea of how timbre exploration is applied in interactive music is also mentioned.

1.1: Interactive music systems

1.1.1: Definition

Joel Chadabe coined the term *interactive composing* to describe 'a performance process wherein a performer shares control of the music by interacting with a musical instrument' (Chadabe, 1997, p. 293). The musical outcome from programmable interactive music systems is a result of the shared control of both the performer and the instrument's programming, where the interaction between the two creates the final musical response. Traditional roles of instrument, composer and performer are blurred in interactive composition. The performer can influence, affect and alter the

underlying compositional structures, while the instrument can take on performer like qualities, and the evolution of the instrument itself may form the basis of a composition (Chadabe, 1997; Drummond, 2009). As Chadabe pointed out, 'The instrument is the music. The composer is the performer' (Chadabe, 1997, p. 291).

In his book *Interactive music systems* (Rowe, 1993), Robert Rowe provides the following definition:

> Interactive computer music systems are those whose behaviour changes in response to musical input. Such responsiveness allows these systems to participate in live performances, of both notated and improvised music (Rowe, 1993, p. 1).

As opposed to Chadabe's view (that is, of a composer/performer interacting with a computer music system influencing each other with the musical outcome being a result of the shared control between them), Rowe's definition emphasises the response of the system; the effect the instruments programming has on the human performer is secondary. The definition is also confined to the ideas of musical input, improvisation, notated score and performance. Rowe's definition, however, should be considered in the context of when his book was written, that is of the early 1990s when most of the music software programming environments were MIDI based, and fixed around the musical ideas inherited from instrumental music (i.e., pitch, velocity and duration) (Drummond, 2009).

Todd Winkler, in his book *Composing Interactive Music* (Winkler, 1998), defines

interactive music systems in a similar way to Rowe. His approach is MIDI based, and

he focuses on the idea of a computer listening to, interpreting and responding to a live

human performance:

> Interactive music is defined here as a music composition or improvisation
>
> where the software interprets a live performance to affect music generated or
>
> modified by computers. Usually this involves a performer playing an
>
> instrument while a computer creates music that is in some way shaped by the
>
> performance (Winkler, 1998, p. 4).

As in Rowe's definition, Winkler restricts the focus of the types of input to be

interpreted to event-based parameters such as notes, dynamics, tempo, rhythm and

orchestration. There is no recognition of interactive music systems that are not driven

by instrumental performance (Drummond, 2009).

For the purpose of this dissertation, the definition provided by Sergi Jorda will be

used. In his doctoral thesis, he claims that interactive music systems are computer-

based, are interactive, and generate a musical output at performance time, under the

control of one or several performers. He adds that interactive music systems must be

'interactive' enough to affect and modify the performer(s) actions, thus provoking an

ongoing dialog between the performer(s) and the computer system (Sergi Jorda,

2005).

1.1.2: Classification

When it comes to classifying interactive music systems, the overall intention needs to be taken into account. For example, is the system intended as an installation to be performed by an audience, or rather by the creator, or multiple professional artists? (Drummond, 2009). Bongers (Bongers, 2000) classifies interactive music systems in three categories:

1. *Performer - System* (e.g., a musician playing an instrument)

   The most common interaction in the electronic arts is the interaction between performer and the system. This can be the musician playing an electronic instrument, a painter drawing with a stylus on an electronic tablet, or an architect operating a CAD (Computer Aided Design) program (Bongers, 2000, p. 46).

2. *System - Audience* (e.g., installation art)

   In the case of an installation work (or a CDROM or web site based work), one could say that the artist communicates to the audience *displaced in time*. Interaction between the work and the audience can take place in several ways or modalities. Usually a viewer pushes buttons or controls a mouse to select images on a screen, or the presence of a person in a room may influence parameters of an installation. The level of interactivity should challenge and engage the audience, but in practice ranges from straight-forward reactive to confusingly over-interactive (Bongers, 2000, p. 48).

3. *Performer - System – Audience* (encompasses works where the interactive

   system interacts with both performer and system)

   The performer communicates to the audience through the system, and the

   audience communicates with the performer by interacting with the system

   (Bongers, 2000, p. 49).

In his paper entitled *Understanding Interactive Systems* (Drummond, 2009), Jon

Drummond adds the following two classifications:

4. *Multiple performers with a single interactive system*; and

5. *Multiple systems interacting with each other and/or multiple performers*

Rowe proposes a different 'rough classification system' (Rowe, 1993) for interactive

music systems built on a combination of three dimensions:

(1) Score-driven vs. performance driven systems

   Score-driven systems have an embedded knowledge of the overall predefined

   compositional structure (Drummond, 2009). For example, they could use

   predetermined event collections, or stored music fragments, to match against

   music arriving at the input. Performance-driven scores, however, do not

   anticipate the realisation of any particular score, and have no pre-constructed

   knowledge of the compositional structure (Drummond, 2009; Rowe, 1993).

(2) Transformative, generative or sequenced response methods

Transformative methods take existing musical material and apply transformations to it to produce variants. For example, these could include transformative techniques such as inversion, retrograde, transposing, filtering, delay, re-synthesis, distortion and granulating (Drummond, 2009; Rowe, 1993).

Generative methods, like transformative, imply an underlying model of algorithmic processing and generation. The difference is, however, what source material there is will be elementary or fragmentary. For example, stored scales or duration sets (Drummond, 2009; Rowe, 1993).

Sequenced response in the playback of pre-recorded, or pre-constructed, music fragments that are stored in the system. Some aspects of these fragments may be varied, such as tempo and dynamics, typically in response to the performance input (Drummond, 2009; Rowe, 1993).

(3) Instrument vs. player paradigms

Instrument paradigm systems are designed to function in the same way as a traditional acoustic instrument. Performance gestures from a human player are analysed and processed, producing an output exceeding normal instrument response. In other words the response is predictable, direct and controlled (Drummond, 2009; Rowe, 1993).

Player paradigm systems try to construct an artificial player. The system

responds to human performance, but with a sense of independence

(Drummond, 2009; Rowe, 1993).

1.1.3: Fundamentals

1.1.3.1: Sensing, Processing, Response

Rowe organises the functionality of an interactive music system into three stages –

sensing, processing and response. The sensing stage collects real-time performance

data from controllers reading gestural information from the human performer. The

processing stage reads and interprets this information, where it is sent to the final

stage in the chain, the response stage. Here, the system, combined with a collection of

sound-producing devices, share in realising a musical output. According to Rowe, the

processing stage is the core of the system, executing the underlying algorithms and

determining the system's output (Drummond, 2009; Rowe, 1993).

1.1.3.2: Control and feedback

When examining the physical interaction between people and systems, Bongers

claims that interaction with a system involves both control and feedback. The flow of

control in an interactive system starts with the human performance gesture, leading to

the sonic response from the system and completing the cycle with the system's

feedback to the performer (Bongers, 2000; Drummond, 2009).

> Interaction between a human and a system is a two way process: *control* and
>
> *feedback*. The interaction takes place through an interface (or instrument)
>
> which translates real world actions into signals in the virtual domain of the

system. These are usually electric signals, often digital as in the case of a computer. The system is controlled by the user, and the system gives feedback to help the user to articulate the control, or feed-forward to actively guide the user. Feed forward is generated by the system to reveal information about its internal state (Bongers, 2000, p. 43).

Feedback is not only provided by the sonic outcome, as it can also be in a physical or visual form. When it comes to computer music systems, however, Bongers claims that due to the decoupling of the sound source and control surface, a lot of feedback from the process controlled was lost. Visual feedback and especially physical feedback are scarcely utilised in specifically designed electronic music instruments, compared to acoustic instruments (Bongers, 2000).

1.1.3.3: Mapping

Mapping, in terms of interactive music systems, is the connection between the outputs of a gestural controller and the inputs of a sound generator. The method is typically used to link performer actions to the generation and control of musical sounds and parameters. Relating to Rowe's *sensing, processing and response* stages, mapping would be the connecting of gestures to processing and processing to response (Drummond, 2009; Wanderley, 2001; Winkler, 1998).

There are four main mapping strategies that can be used in interactive music systems: *one-to-one*, which is the direct connection of an output to an input; *one-to-many*, which is the connection of a single output to multiple inputs; *many-to-one*, which is the connection of two or more outputs to control one input; and *many-to-many*, which

is a combination of the different mapping types (Drummond, 2009; Miranda & Wanderley, 2006).

1.2: Motion control and multi-touch/multi-user interfaces

1.2.1: Movement to music

In general, most traditional musical instruments are designed based on the human body and the physical nature of audio production that dictate the timbre, and pitch range, of the particular instrument. The efficiency of the interface largely determines controllability of, and interaction with, the instrument. Hence, body motion and gesture, directly and indirectly, contribute to various important factors of artistic performances (Ng, 2004).

The translation of human gesture and movement into computer data can be used in interactive music systems to generate music and affect aspects of the music produced. In *Composing interactive music* (Winkler, 1998), Winkler relates the human body to an acoustic instrument with similar limitations that can lend character to sound through idiomatic movements. With traditional instruments, different uses of weight, force, pressure, speed and range produce sounds that in some way reflect the effort and energy used to create it. Each part of the body has unique physical limitations that can lend insight into the selection of musical material. Thus, as Winkler puts it, 'a delicate curling of the fingers should produce a very different sonic result than a violent and dramatic leg kick' (Winkler, 1998, p. 319). He makes the point that physical parameters can be appropriately mapped to musical parameters. However, simple and obvious one-to-one relationships are not always musically satisfying, and it is up to the composer to interpret the computer data with software to produce

musically interesting results. By being aware of the underlying physics of movement, and instead of applying predictable musical correlations, it is possible to assign provocative and intriguing artistic effects, creating unique models of response. For example, more furious and strenuous activities could result in quieter sounds, while a small physical action, like the nod of a head, could set off an explosion of sound. Winkler sums up by adding that 'success for performers, as well as enjoyment for the audience, is tied to their ability to perceive relationships between movement and sound' (Winkler, 1998, p. 320).

Winkler considers how performers can shape and structure musical material through their physical gestures, and comments on how it is important to recognise not only what is being measured, but also how it is being measured. One method of measurement, using a MIDI foot pedal as an example, takes a set of numbers, often represented as MIDI continuous controller values between 0 and 127, to determine location over time within this predefined range. Other devices that may have less continuous reporting, like a computer keyboard, send out nonlinear discrete data that may represent predetermined trigger points. This data of numbers represents the location or body position of a performer over time within a predefined range, and software can interpret this information to create music based on location or position, or by movement relative to a previous location or position (Winkler, 1998).

1.2.2: Multi-touch/multi-user interfaces

Electronic multi-touch interfaces allow the recognition and calculation of multiple touch points at one time. The use of this technology permits greater human-computer interaction (Hoye & Kozak, 2010). There are various techniques that can be used to

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construct multi-touch surfaces. Without going into detail, these include *Resistance* based, *Capacitance* based, and *Surface Wave* touch surfaces. The most commonly used in *Do-It-Yourself* environments, however, is the optical based approach, which uses the concept of processing and filtering captured images on patterns, and generally incorporates cameras, infrared illumination, silicone compliant surfaces, projection screens, filters, and projectors (Schöning et al., 2008).

When it comes to the world of music, multi-touch technology is being used to build instruments that satisfy the performer's need to manipulate many simultaneous degrees of freedom in audio synthesis. Multi-touch sensors permit the performer fully bi-manual operation as well as chording gestures, offering the potential for great input expression. Such devices can also accommodate multiple performers, in the form of an interactive table for example, which creates the opportunity for duets, ensembles, and other collaborations using one instrument (Davidson & Han, 2006).

An example of a multi-touch product aimed at musicians is the *MTC Express Multi-touch Controller*, developed by *Tactex*. The *MTC Express* is designed as a pad that uses an internal web of fiber-optic strain gauges to sense multiple points of pressure applied to its surface, by multiple fingers or styluses, simultaneously. Thus, giving the user a three-dimensional control surface, where each sensed contact point provides data consisting of x, y, and pressure values, at a sampling rate of 200 Hz. With the *Studio Artist* software driver support, the *MTC Express* captures intuitive gestures made by the artist, and interprets them as control parameters. As the controller has an impressive temporal sampling rate (200Hz) and dynamic range in pressure, it can be

extremely useful for percussive control (Davidson & Han, 2006; Jones, 2001; Pacheco, 2000).

Various other instruments have been developed based on the ideas and technology of multi-touch. As Phillip Davidson and Jeffery Han explain in *Synthesis and Control on Large Scale Multi-Touch Sensing Displays*:

> Larger scale musical interfaces have also developed around the concept of the manipulation of trackable tangible assets, such as blocks or pucks. These tangible interfaces can accommodate more than one hand and/or more than one user (Davidson & Han, 2006, p. 217).

An example of an instrument in this new category is the *reacTable*. More on this instrument will be discussed in upcoming chapters, but basically, the *reacTable* is a tabletop instruments based on vision-based tracking of optical objects, known as *fiducials* (Davidson & Han, 2006).

1.3: Exploration of timbre in electronic computer music

The use of computers in the creating of music has expanded musical thought considerably when it comes to the composing of timbre. Digital tools present composers or sound designers with unprecedented levels of control over the evolution and combination of sonic events (Rowe, 1993). As Sergi Jorda declares:

> The most obvious advantage of the computer, in comparison to traditional instruments, lies in its ability to create an infinite sonic universe by means of a

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multitude of sound synthesis techniques: imitations and extensions of physical instruments, digital emulations of analogue synthesis methods, and inventions of new principles only attainable in the digital domain. Indeed the potential to explore *timbre* has been by far the most important aspect of computer music. (Sergi Jorda, 2005, p. 53).

1.3.1: Frequency modulation synthesis, additive synthesis and subtractive synthesis

There are many techniques used for digital music synthesis, including frequency modulation synthesis, additive synthesis, subtractive synthesis, granular synthesis and waveshaping. These techniques can be used to achieve rich, natural sounding timbres, reproducing sounds of acoustic instruments, or rather to explore new and different electronic timbres (Karplus & Strong, 1983). In this chapter, I will be discussing three forms of sound synthesis: frequency modulation synthesis, additive synthesis and subtractive synthesis.

1.3.1.1:   Frequency modulation synthesis

Frequency Modulation (FM) synthesis, discovered by John Chowning, can be used to produce a wide range of distinctive timbres that can be easily controlled. FM is the alteration or distortion of the frequency of an oscillator in accordance with the amplitude of a modulating signal (Dodge & Jerse, 1985). In other words, one waveform is used to modulate the frequency of another waveform. In the most basic and classic FM, both waveforms are sine waves, although alternative waves can be, and have been, used. The waveform applying the modulation is called the modulator, while the waveform being affected – the one we hear – is called the carrier. When a

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sine wave carrier is modulated by a sine wave modulator, for example, sinusoidal sidebands are created at frequencies equal to the carrier frequency plus and minus integer multiples of the modulator frequency (Aikin, 2002; Cook, 2002; Dodge & Jerse, 1985).

The ratio of carrier and modulator frequencies is an important variable in FM synthesis as it affects the timbre. Simple integer ratios will produce harmonic sounds while non-simple ratios will produce an inharmonic spectrum and thus, inharmonic, or dissonant, sounds. The amplitude of the modulator, called the modulation index is also an important variable that affects the timbre. The modulation index – the ratio of the maximum change in the carrier frequency divided by the modulation frequency – affects the volume of the sideband overtones, so the higher the modulation index, the more prominent the overtones will be, and thus the more complex the output signal becomes. By altering the amplitude of the modulator, sidebands can be introduced, diminish, disappear altogether, or even reappear with inverted phase (Brown, 2001; Cook, 2002; Reid, 2010).

1.3.1.2: Additive synthesis

Another form of sound synthesis used to create new and alternate timbres is additive synthesis. In *Signal processing aspects of computer music: A survey*, James Anderson Moorer describes additive synthesis as the production of a complex waveform by the summation of component parts, for instance, adding up the harmonics of a tone to produce a single sound (Moorer, 1977). This form of synthesis provides maximum flexibility in the types of timbre that can be synthesised. Using any number of

oscillators, any set of independent spectral components can be synthesised, and thus virtually any sound can be produced (Dodge & Jerse, 1985).

For example, the specific synthesis of a tone can be generated using a separate sinusoidal oscillator for each harmonic partial, with the appropriate amplitude and frequency functions applied to it. The output from each of the oscillators is added together to acquire the complete sound. Hence the name *additive synthesis* (Dodge & Jerse, 1985).

1.3.1.3: Subtractive synthesis

Subtractive synthesis is another method used in the generation of a signal that creates a desired acoustic sensation. In this form of sound synthesis, the algorithm begins with a complex tone and reduces the strength of selected frequencies in order to realise the desired spectrum. This is achieved by applying the technique of filtering to the sound source (Dodge & Jerse, 1985).

By rejecting unwanted elements in a signal, and thus shaping the sound spectrum, filters can vastly alter the timbre of a sound. Filters modify the amplitude and phase of each spectral component of a signal passing through it; however, they do not change the frequency of any signal or component. Different types of filters, with different cut-off frequency points, determine which frequencies are permitted to pass through. The various types include low-pass, high-pass, band-pass, and band-reject filters. As the names suggest, low-pass filters allow low frequencies to pass through, and be heard, while cutting off higher frequencies. High-pass filters are just the opposite; allowing higher frequencies to pass through while cutting off lower

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frequencies. A band-pass filter cuts both high and low frequencies, while midrange frequencies are not affected. Band-reject filters work in the opposite way, cutting off frequencies in a midrange band, letting the frequencies above and below through (Dodge & Jerse, 1985; Nordmark, 2007).

In classic Subtractive synthesis, noise and pulse generators are traditional sound sources, as they produce spectrally rich signals, and the technique has the greatest effect when applied to sources with rich spectra. Noise generators produce wide-band distributed spectra, while pulse generators produce periodic waveforms at specific frequencies that possess a great deal of energy in the harmonics. In saying this, any sound can be used as a source for subtractive synthesis (Dodge & Jerse, 1985).

1.3.2: Timbre exploration in interactive environments

Setting the idea of timbre exploration in an interactive music system environment, synthesis methods have variable parameters that can be shaped by a performer's input, imparting expressive control to the creation of specifically desired sounds. Continuous control of timbral parameters enables the performer, or 'player' of the interactive music system, to transform sound into an endless variety of permutations (Winkler, 1998).

In *Composing interactive music* (Winkler, 1998), although Winkler's discussions are primarily MIDI based, he recognises that when exploring timbre in interactive environments, the mapping of musical gestures onto the various parameters of signal processing is extremely important and must be carefully planned. He proposes that the output may be considered in two different ways; 'as an integrated component of an

instrument, capable of enhancing its timbral qualities, or as a generator of new musical material, producing variations and an accompaniment based on the original input' (Winkler, 1998, p. 249). Winkler gives examples that can be placed into these two categories; the example relating to the first category being a computer keyboard or mouse creating abstract 'soundscapes' fashioned from gestural input. The example relating to the second category is a performer using an acoustic instrument to trigger sampled sounds from everyday life. As mentioned previously, the established relationships between gestures and musical parameters in both cases are principal. He mentions how the 'composer is challenged to find musical gestures that serve the dual purpose of creating primary musical material and generating functions applicable to signal processing' (Winkler, 1998, p. 250).

1.4:   Summary

In this chapter I have discussed the definition, classification and fundamentals of interactive music systems. The three main concepts of motion control, multi-touch/multi-user interfaces and timbre exploration were also investigated. In the next chapter, I will examine, in great detail, an interactive music system I designed and constructed myself.

CHAPTER 2

Homemade interactive music system

In this chapter, I will firstly provide a basic description of the instrument, and how it is based on the technology of the *reacTable*, explaining the similarities and also the differences. The physical design of the instrument is discussed, looking into the measurements and component parts. I breakdown the working of the three computer software programs utilised in the instrument; *ReacTIVision*, *Max MSP* and *Reason*. I discuss the instrument as an interactive music system, describing the classification and its fundamentals. A main focus of this chapter is to provide the mapping of the instrument in great detail, looking into how each tangible object generates and controls parameters of sound. The limitations of the instrument are also discussed, as is the integration and employment of the three main concepts: 1) interpreting gestural movement into music, 2) multi-touch/multi-user technology, and 3) the exploration of timbre in computer music.

2.1:   The instrument in a nutshell

The interactive computer music system I have designed and constructed (see Figure 2.1) is in the form of an electronic instrument that incorporates multi-touch technology with a tabletop tangible user interface, based on the technology of the *reacTable* (S Jorda, Kaltenbrunner, Geiger, & Bencina, 2005). It can be played by a single performer, or by multiple performers.

Figure 2.1: Interactive music system

Like the *reacTable*, my instrument incorporates a clear tabletop with a camera placed

beneath, which constantly examines the table surface, tracking the nature, position

and orientation of the tangibles, or objects, that are placed, and moved around, on it.

The tangibles display visual symbols, called *fiducials* (see Figure 2.2), which are

recognised by the software. Each tangible is dedicated a function for generating or

manipulating/controlling a sound. Users interact by moving them around the tabletop,

changing their position, their orientations, or their faces (in the case of, say, a cube

object) (S Jorda, Kaltenbrunner, Geiger, & Alonso, 2006; S Jorda, et al., 2005).

Figure 2.2: Fiducial symbol

Here is where my instrument differs from the *reacTable*. The vision captured by the camera is sent to the open source software *ReacTIVision*, and then to *MAX/MSP*, which allows the instrument to work as a MIDI controller. This information is then sent to *Reason*, where the final mapping is completed to allow note on/off events (determined by a tangible being placed and displaced in the cameras vision), along with the x-position, y-position, and orientation of each tangible assigned to manipulate different parameters of music.

2.2: Instrument set-up and software

2.2.1: Basic physical design and build

As the instrument bares a tabletop interface, I found it rather appropriate that its entire physical structure – wooden frame – be based on the shape and design of a table (see Figure 2.3). The table stands 92cm high, at perfect mid-stomach height. As it is intended to be performed while standing up, this gives the performer a "birds-eye" view of the tabletop, while relieving them from having to bend or sit down to move the objects around. The dimensions of the tabletop interface – clear Perspex – are 46cm (length) x 37.6cm (width) (see Figure 2.4). This provides the performer with quite a large area (1729.6cm^2) to move the objects around. As part of the design, on either side of the interface are two 15cm x 46cm shelves intended for the objects to rest on.

Figure 2.3: Table design

Figure 2.4: Tabletop interface

A camera (see Figure 2.5) – with approx. dimensions of 84 x 67 x 57mm, and a video capture of 640 x 480 pixel – is placed 61cm directly beneath the tabletop, facing upwards in order to capture the vision of the objects being moved around. A problem I encountered, when it came to the image capturing, was that there needed to be a certain amount of light coming from above the tabletop, as well as from below. Achieving the top light was simple, as I would just turn on the light in the room (or whichever room the instrument was placed in); however, achieving the bottom light was not so straightforward. Lights could not simply be placed directly beneath the tabletop, side by side with the camera, as the reflection was too intense and would block the image of the object, or fiducial symbol rather, and thus be unrecognisable to the camera. This was the main reason I did not design and construct the instrument as a box instrument, with camera and lights inside, for the open wooden frame of the table design allows as much light in as possible. Even this light, however, was not enough for the camera to consistently recognise the fiducials. I overcame the bottom lighting problem by using two LED torches. The torches are place on either side of the table, on the same x-axis as the camera, however, roughly 25cm outside of being directly underneath the tabletop interface. They are then angled to shine on the bottom side of the Perspex. This allows the camera to constantly examine the interface, without any distracting light reflection.

Figure 2.5: Camera

More will be discussed in later chapters on the reasons behind the various shapes and colours of the objects in relation to the various sound generation/control categories they are placed in. However, for now I will simply give each objects' shape and size dimensions: The pitch generation/control cube is 7cm x 7cm x 7cm; the two flat rhythm generation/control objects are 7cm x 7cm; the six timbre generation/control rectangular prism objects (excluding the *Additive Synthesis* objects) are 7cm x 7cm x 2cm; and the three flat *Additive Synthesis* objects are 5cm x 5cm.

2.2.2: *ReacTIVision, Max MSP* and *Reason*

When it comes to the computer aspect of the instrument, three software programs are used in conjunction with each other in order for vision to be captured, analysed and then interpreted into sound, or in other words, for the instrument to function. The three computer software programs, which act as the "engine room" of the instrument, are *ReacTIVision* ("reacTIVision 1.4: a toolkit for tangible multi-touch surfaces," nd), *Max MSP* (Puckette, 2010) and *Reason* ("Reason," 2010). Without going into great technical detail, I will use this subchapter to explain the main functions of each program, focussing mainly on *ReacTIVision*.

ReacTIVision is the fundamental sensor component of my interactive music system. The software is a computer vision framework used for the tracking of the fiducial markers, displayed on the objects of the instrument. As its function is the analysing of visual information captured by the camera placed beneath the tabletop, *ReacTIVision* does not contain any sound components. Instead, Tangible User Interface Object (TUIO) messages are sent to a TUIO-enabled client application: in the case of my

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instrument, this is *Max MSP* ("reacTIVision 1.4: a toolkit for tangible multi-touch surfaces," nd).

The internal structures and workings of *ReacTIVision* can seem extremely complicated when going into precise detail. A basic explanation of the software is as follows: *ReacTIVision* tracks specially designed visual symbols, known as fiducial markers, in a real-time video stream. These symbols can be attached to any physical object to be tracked, which enables the table to be "played" like an instrument, by moving the objects around. The source image frame is first converted to a black and white image with an adaptive thresholding algorithm. This image is then segmented into a tree of alternating black and white regions (region adjacency graph). This graph is then searched for unique left heavy depth sequences encoded into the fiducial symbol. The found tree sequences are then matched to a dictionary to retrieve a unique ID number. The centre point and orientation of the fiducial marker are tracked efficiently, thanks to the specific design of the symbol. Open Sound Control (OSC) messages use the TUIO protocol to encode the fiducials's presence, location, orientation and identity, and pass on this data to the TUIO-enabled client application (M Kaltenbrunner, 2009; Martin Kaltenbrunner & Bencina, 2007; reacTIVision 1.4: a toolkit for tangible multi-touch surfaces," nd).

Max MSP acts as the client application in my instrument. Here, the fiducials' recognition, centre point and orientation information is processed and organised into four groups of numbers: note on/off (0 – 1), x-position (0 – 640), y-position (0 – 480) and angle (0 – 360) [The fiducials' recognition/derecognition relating to note on/off; centre point relating to x and y position; and orientation relating to angle]. Using

various techniques in *Max MSP*, I organised this information in a way that the zero

point was located at the bottom, left hand corner of the table. For example, moving an

object from left to right raises the value of the x-axis number, while moving an object

from bottom to top raises the value of the y-axis number. I also organised the

processing of information so that the value of the angle, or orientation, number rises

when an object is rotated clockwise. These sets of numbers are then scaled to 0 – 127

in order to be sent as MIDI information to the computer software program *Reason*.

Reason completes the process of interpreting object recognition and movement into

sound generation and control. To sum up, *ReacTIVision* has analysed vision of objects

and their placements, and sent this information to *Max MSP* where it has been

organised into sets of note on/off, x-position, y-position and orientation values and

finally sent to *Reason*. *Reason* is where the mapping of these values to parameters of

music occurs. Further detail on this issue will be discussed later in the chapter,

however, a quick example would be if the y-position value of an object was assigned

to the pitch shift parameter, therefore enabling the movement of this object from

bottom to top of the table interface to raise the pitch of the sound produced.

2.3:   Instrument as an electronic interactive music system

2.3.1: Instrument classification

My instrument may be classed in the *Performer - System* category of Bongers'

interactive music systems classification method if I alone myself were performing on

it. However, it could also be classed in the *Audience – System* category if it was

placed in an art installation environment (Bongers, 2000). The distinction can be

made when, as the designer of the instrument, I understand the relationships between

movements and sound previous to playing of the instrument, while in an installation

setting, audience members would gain understanding of the relationships *while*

playing the instrument.

2.3.2: Instrument Fundamentals

2.3.2.1:   Instrument sensing, processing, response

The *sensing*, *processing* and *response* stages of interactive music systems, proposed

by Rowe (Rowe, 1993), can be easily identified with in relation to my instrument. The

physical interaction between the human performer and the tangible objects – moving

them around the tabletop interface – is part of the *sensing* stage. Algorithms

performed by the computer softwares *ReacTIVision*, *Max MSP* and *Reason* form the

second, and most important stage: the *processing* stage. Finally, the musical output

from the computer, combined with a set of speakers are part of the concluding

response stage.

2.3.2.2:   Instrument control and feedback

The sonic outcome of my instrument is a major form of feedback, influencing the

musical control of the human performer. However, it is not the sole type of feedback.

Visual feedback also plays a key role in the sense that the performer is always looking

at the tabletop and at the objects he or she is moving around; placing one here and one

there, always with a complete view of which objects are present on the interface, and

what location they are in. This visual feedback undoubtedly influences the performer

in the moving around of objects, and therefore, what sounds are produced.

2.3.2.3: Instrument mapping

In terms of my instrument, I have employed multiple mapping strategies to establish relationships between the recognition/movement of different objects and the sounds produced. As the mapping is the most important aspect of the instrument (i.e., it determines what sounds the instrument makes, and how it is played), I will use this chapter to go into detail of the mapping used within the instrument, and give examples of how these mapping relationships can be utilised to create music.

[It should be noted that an alternative choice of mapping could completely change the instrument, and how it is used. For example, I could set up the mapping in a way that the placement of objects on the tabletop interface set off drum loops or pre-recorded bass line sample, and thus be used as a DJ instrument. This is not the case, however, but it is worth recognising that the technology does hold this potential.]

The tangible objects used to generate and control the sounds and effects of the instrument can be categorised into three groups: pitch generation/control, rhythm generation/control and timbre generation/control. The table below outlines the object categories, the function and fiducial number of each object, the note on/off (placement-on/placement-off the tabletop interface) functions, and the parameters of music controlled by the x-axis, y-axis and rotation of each.

Table 2.1: Tangible Object Function Table

Generation/Control, Colour, Shape, Size	Name/ Function	Fiducial No./ID	On/Off	x value	y value	ANG value
Pitch Brown Cube 7cm x 7cm x 7cm	SW Note C-2	0	Note on/off	Volume	Pitch shift	SW type
	SW Note C-1	1	Note on/off	Volume	Pitch shift	SW type
	SW Note C0	2	Note on/off	Volume	Pitch shift	SW type
	SW Note C1	3	Note on/off	Volume	Pitch shift	SW type
	SW Note C2	4	Note on/off	Volume	Pitch shift	SW type
	SW Note C3	5	Note on/off	Volume	Pitch shift	SW type
Rhythm Red Flat 7cm x 7cm	LFO to frequency cut-off	6	Effect on/off		LFO rate	LFO amount
	LFO2 to amplitude cut-off	7	Effect on/off		LFO2 rate	LFO2 amount
Timbre Green Rectangular Prism 7cm x 7cm x 2cm	Two-band para. EQ: A-Band	8	Effect on/off	A-band frequency	A-band gain	
	Two-band para. EQ: B-Band	9		B-band frequency	B-band gain	
	Digital reverb	10	Effect on/off			Dry/wet amount
	Scream 4 distortion	11	Effect on/off	Dist. type		Dist. amount
	Frequency modulation synthesis	12	Effect on/off		Mod. no.	FM amount
	Subtractive synthesis	13	Effect on/off		Res. amount	Freq. amount
Timbre Green Flat 5cm x 5cm	Additive Synthesis Osc 1	14	Effect on/off		Volume	Octave range
	Additive Synthesis Osc 2	15	Effect on/off		Volume	Octave range
	Additive Synthesis Osc 3	16	Effect on/off		Volume	Octave range

KEY

Generation/Control, Colour, Shape, Size	What concept of music the object relates to, the colour of the object, the shape of the object, and the size of the object.
Name/Function	The name of the fiducial (visual symbol) or object, and what musical aspect it generates/controls [NOTE: SW = Square wave]
Fiducial No./ID	The identity number of the fiducial recognised by the computer software
On/Off	What happens when the object is placed on the tabletop interface and recognised by the camera, and then removed and de-recognised
X-value	The parameter of music controlled by the x-axis of the object
Y-value	The parameter of music controlled by the y-axis of the object
ANG value	The parameter of music controlled by the angle, or orientation, of the object

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2.3.2.3.1: Pitch generation/control

Figure 2.6: Pitch cube object

The pitch produced by the instrument is generated and controlled by the pitch cube object (see Figure 2.6). Each of the six faces sets off a different pitch when placed on the tabletop, and in view of, and recognised by, the camera. The six pitches that can be produced, when the relative face is firstly recognised, are each the note of C, however, all are in different octave ranges. The notes are created by a square-wave tone generated by a single oscillator. When the cube object is removed from the tabletop, the note stops. It is possible to create chords of two and three notes using the pitch cube by angling it in a fashion so that the camera can see and recognise the two or three faces, generating the relative pitches simultaneously. Not all combinations of two or three notes are possible to create, only those that can be generated by fiducials on adjoining cube faces.

The x-axis of the cube object controls the master volume. The fiducials on each face are assigned, or mapped, to the same volume control. This is an example of a *many-to-one* mapping method. This means that if a pitch is currently being sounded, triggered by one of the faces of the pitch cube, and the face placed on the tabletop is changed, the current volume will be maintained.

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The y-axis of the cube object controls the pitch shift. Once again the fiducials on each face are assigned to the same musical parameter, this time being a pitch shift. The range of the pitch shift is seven semi-tones. Given a starting pitch of C, the highest the pitch can be shifted is to the G above, while the lowest is to the F below. The starting pitch will only be C if the cube object is placed in the middle of the y-axis. If the cube is at the top of the y-axis, and therefore producing a pitch-shifted G note, and the face is changed, the instrument will produce a pitch-shifted G note in the relative octave range.

2.3.2.3.2: Rhythm generation/control

The objects in the category of rhythm generation/control can be identified as red, flat objects (7cm x 7cm).

2.3.2.3.2.1: *LFO to Frequency cut-off*

Figure 2.7: *LFO to Frequency cut-off* object

The musical aspect of rhythm can be produced by the instrument by placing the red, flat object, entitled *LFO to Frequency cut-off* (see Figure 2.7) on the tabletop. A Low Frequency Oscillator (LFO), producing a sine wave, controls the frequency cut-off

point of the note, or pitch, being sounded. Removing the object from the interface switches the effect off.

While the x-axis of the object is not mapped to any parameter of music, the y-axis controls the rate, or speed, of the LFO. As the LFO produces a sine wave, it is the frequency measured in Hertz that is being altered. The minimum being 0.07 Hz, and the maximum being 99.6 Hz.

The rotation, or angle of the object controls the amount of how much the LFO affects the original note. A rhythmic pulsing effect is established if the LFO amount is low, while there is a more "wobble-like" effect if the LFO amount is higher.

2.3.2.3.2.2: *LFO2 to Amplitude cut-off*

Figure 2.8: *LFO2 to Amplitude cut-off* object

The second red, flat tangible, entitled *LFO2 to Amplitude cut-off* (see Figure 2.8) can also be used to produce musical rhythm. Here, a second Low Frequency Oscillator (LFO2), producing a square-wave is used to control the amplitude gain of the note, or pitch, being sounded. Removing the object from the interface switches the effect off.

Once again, the x-axis of the object is not assigned to any parameter, while the y-axis controls the rate of the LFO2. The frequency of the square-wave producing LFO2 is again being altered, with the same minimum and maximum values in Hertz (0.07Hz – 99.6Hz).

The rotation of the object controls the amount of how much the LFO2 affects the original note. If the amount is low, the amplitude gain, or volume, will not cut out completely. If the amount is at maximum value the amplitude gain *will* cut out completely, and because it is being altered by a square-wave, and therefore in a square-wave pattern, a rhythmic stuttering effect is created, alternating between full amplitude gain and zero gain.

2.3.2.3.2.3:   Using the pitch cube to generate/control rhythm

Another way to create rhythm is by using the pitch cube object. Because the note produced is generated by a square-wave, if the pitch is low enough (for example, set-off by fiducial 0 and at the lowest possible pitch shift), the waves are longer and therefore a rhythmic beating is created.

2.3.2.3.3:   Timbre generation/control

The objects in the category of timbre generation/control can be identified as green objects. Within this category, additionally, there are two subcategories: 1) The Additive synthesis objects, 2) The rest. The six non-Additive synthesis objects can be identified as larger rectangular prism shaped objects (7cm x 7cm x 2cm), while the three additive synthesis objects can be identified as smaller flat objects (5cm x 5cm).

2.3.2.3.3.1: *Two-Band Parametric EQ*

Figure 2.9: *Two-Band Para. EQ: A-Band* object

Figure 2.10: *Two-Band Para. EQ: B-Band* object

One way to create new and different timbres using the instrument is to work with the Two-Band Parametric EQ objects. This allows the player to emphasise certain frequencies while removing undesired ones, along with creating a range of effects in performance time, such as EQ sweeps. To make full use of this EQ effect, two fiducials, attached to two separate objects, are required: *Two-Band Para. EQ: A-Band* (see Figure 2.9) and *Two-Band Para. EQ: B-Band* (see Figure 2.10). The recognition of the first fiducial, or object, entitled *Two-Band Para. EQ: A-Band* switches the EQ on, while the removal, or de-recognition, switches it off. This means that even if the second EQ object, *Two-Band Para. EQ: B-Band*, is on the tabletop, in full view of the camera, and only the first EQ object is removed, the EQ will still be switched off. It also means that the second EQ object cannot be used to switch the EQ on in the first place.

The x-axis of the two objects controls the centre frequency points respectively (i.e., the x-axis of the first object controls the A-Band centre frequency, while the x-axis of the second object controls the B-Band centre frequency). This is the centre point of frequency that the player wishes to emphasise or remove. The range is 31 Hz to 16 kHz.

The y-axis of the two objects controls the amount of gain respectively (i.e. the y-axis of the first object controls the A-Band gain amount, while the y-axis of the second object controls the B-Band gain amount). The gain indicates how much the level of the selected frequency range should be raised or lowered. The gain range is ±18 dB. Because of the two bands, bass frequencies, for example, can be emphasised while treble frequencies can be removed simultaneously.

A parametric EQ uses independent parameters for centre frequency, gain amount (which have both been mapped to the x and y values of the objects) and Q, which is the width of the affected area around the set centre frequency. I have not set the instrument up in a way to control the Q, however, and have left it as a pre-set at a medium width (Nordmark, 2007).

2.3.2.3.3.2: *Digital Reverb*

Figure 2.11: *Digital Reverb* object

Although reverberation is traditionally used to create a space effect and simulate some kind of acoustic environment, I am using the effect primarily to contribute to changes in timbre. The recognition of the *Digital Reverb* object (see Figure 2.11) switches the reverb device on, as the de-recognition switches it off.

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The only parameter of the reverb open to manipulation is the dry/wet amount, controlled by the rotation of the object. This is the balance between the audio signal (dry) and the reverb effect (wet). The x and y axis' of the object do not control any parameter. The other parameters of the reverb device remain pre-set. These include the *algorithm* – represented by 'type of room' on device; *size* – emulated room size; *decay* – length of reverb effect; and *damp* – cuts off the high frequencies of the reverb.

2.3.2.3.3.3:   *Scream 4 Distortion*

Figure 2.12: *Scream 4 Distortion* object

Further alterations in timbre can be achieved with the use of the *Scream 4 Distortion* object (see Figure 2.12). As the name suggests, the placing of the object on the tabletop applies a distortion effect – provided by the *Scream 4 Distortion* device in *Reason* – to the audio signal, while the removing of the object terminates the effect. This allows the player to warp the original audio signal beyond recognition or, alternatively, produce more subtle musical effects.

The x-axis of the object controls the type of distortion applied. The 10 different types are presented in Table 2.2:

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Table 2.2:   Types of Distortion

TYPE	DISCRIPTION
Overdrive	Analog-type overdrive effect.
Distortion	Similar to Overdrive type. Denser, thicker distortion.
Fuzz	Bright and distorted sound
Tube	Tube distortion
Tape	Soft clipping distortion
Feedback	Combines distortion in a feedback loop
Modulate	Multiplies signal with a filtered and compressed version of itself, then adds distortion.
Warp	Distorts and multiplies incoming signal with itself
Digital	Reduces bit resolution and sample rate
Scream	Similar to fuzz. Bandpass filter with high resonance and gain settings placed before distortion stage.

The zero point on the x-axis (i.e. the leftmost of the table) produces the *Overdrive* effect, and as the object is moved further along the x-axis (i.e. to the right), the *Scream* effect is approached.

While the y-axis of the object is invalid, the angle, or rotation, controls the amount of distortion. While raising the amount of distortion, or damage, the master level may need to be lowered in order to maintain the same output level, and vice-versa.

2.3.2.3.3.4:   *Frequency Modulation Synthesis*

Figure 2.13: *Frequency Modulation Synthesis* object

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Another way to alter the timbre of the audio signal is by using the *Frequency Modulation Synthesis* object (see Figure 2.13). In order to achieve this, a second oscillator, called an *FM Pair Oscillator*, is activated. Once again, this is achieved by the recognition of the object by the camera, while the de-recognition deactivates it. This newly activated oscillator is made up of two pairing oscillators, hence the name. The first of the paired oscillator produces a sine wave, which acts as the carrier, and can be modulated by a second sine wave, known as the modulator, which is produced by the second paired oscillator. This is the basis for creating the frequency modulation effect. It should be pointed out, however, that the FM effect is *not* applied to the original square wave produced by the first main oscillator via the pitch cube. In saying this, the *FM Pair Oscillator* is layered with the original oscillator; so all other parameter manipulations (e.g., rhythm control, reverb, distortion, etc.) will apply to both.

While the x-axis of the object is not mapped to any parameter, the y-axis controls the modulator number, with the range 1 – 32. With the carrier number always set at 1, the frequency ratio of the two determines the basic frequency content, and thus, the timbre of the sound. As discussed in previous chapters, simple ratios produce 'nicer-sounding', more harmonic timbres than the dissonant sounding timbres produced by complex ratios.

The rotation of the object controls the FM amount. The amount determines how much the modulator sine wave, set at any modulator value from 1 to 32, affects the carrier sine wave. Changing the objects vertical position and orientation simultaneously creates very interesting sounds.

2.3.2.3.3.5:   *Subtractive Synthesis*

Figure 2.14: *Subtractive Synthesis* object

Another form of synthesis that can be used to explore further timbre possibilities is that of subtractive synthesis, which essentially is the method of removing harmonics. This can be achieved by placing the *Subtractive Synthesis* object (see Figure 2.14) on the table, and in turn activating a bandpass filter. As always, the removal of the object deactivates the filter.

The x-axis of the object is unmapped, while the y-axis controls the resonance. This determines the characteristic, or quality of the filter. As the filter is set to bandpass, the resonance setting adjusts the width of the band. When the resonance is raised, the band through frequencies pass becomes narrower.

The rotation of the object controls the filter cut-off frequency. Gradually changing the filter frequency is another way of producing the sweep effect, as mentioned when discussing the Two-Band Parametric EQ.

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2.3.2.3.3.6: *Additive Synthesis*

Figure 2.15: *Additive Synthesis Osc 1* object

Figure 2.16: *Additive Synthesis Osc 2* object

Figure 2.17: *Additive Synthesis Osc 3* object

The three *Additive Synthesis* objects can be used separately, or for a more effective

result, simultaneously, to form a complex tone. The placement of each object on the

tabletop interface switches on its own oscillator, and the removal of each switches the

relative oscillator off. The main function of the objects is to add overtones to the

original pitch, and thus create an additive synthesis effect. Each oscillator produces

the same note as is currently being generated (i.e., the note determined by the pitch

cube). This means that if the pitch cube is raised on its y-axis and thus the pitch of the

original oscillator's square wave rises, the pitches of the notes produced by the

Additive Synthesis oscillators will also rise in unison. Each oscillator produces a

different type of waveform; *Additive Synthesis Osc 1* (see Figure 2.15) produces a

sawtooth wave, *Additive Synthesis Osc 2* (see Figure 2.16) produces a square wave,

and *Additive Synthesis Osc 3* (see Figure 2.17) produces a sine wave. Because, technically, new sound layers are being added by the use of the objects, the rhythm generation/control objects do not apply to the tones produced by the additive synthesis objects, only the original square wave pitch.

While the x-axis of each of the three objects is unmapped, the y-axis of each controls the relative oscillator's volume. This is important in creating overtones, as the idea is to have the volume of the new tone at a level where it does not seem like an added layer, and therefore a chord, but rather part of the timbre of the original pitch – in this case the square wave produced by the pitch cube.

The rotation of each object controls the octave range of the tone produced, with the range being 0 – 9. If all three objects are generating the same pitch in different octave ranges, and at appropriate volume levels, this produces a much richer timbre.

2.3.2.3.3.7: Using the pitch cube to control timbre

The original timbre of the pitched notes produced by the pitch cube is created by a square-wave tone generated by a single oscillator. The type of square-wave can be altered by rotating the pitch cube. Once again, the fiducials on each face of the cube are mapped to this same parameter.

2.3.3: Limitations of the instrument

The first limitation as an instrument relates to the placement of the objects on the tabletop interface. As the objects must be placed on the surface of the table to carry out their assigned musical functions, it is not possible for two objects to be in the

same xy position. This means that certain combinations of sounds and effects are unachievable. This is a major reason behind the mapping strategy used, as most objects have musical parameters assigned to their orientation and only one of their axis' (x or y). This is because an object can be rotated on the tabletop interface without changing its xy location, and therefore, without interfering with other objects.

The second limitation as an instrument comes due to its use as a MIDI instrument, sending MIDI messages. The maximum range of values that MIDI can express is 0 – 127. This means that every parameter of music that each object controls is restricted to these 128 values. This is not a major limitation, compared to what was discussed in the last paragraph; however, it can become a problem if a player of the instrument requires ultra-specific values to thus create an ultra-specific sound.

2.4:   Three ideas, one instrument

The instrument is centred on three principal concepts: motion control, multi-touch/multi-user interfaces, and the idea of exploring timbre in electronic music. The first two ideas relate to the interface and "playing" of the instrument, while the last idea relates more to the music being created. I will briefly explain how each concept is employed in the instrument.

2.4.1: Instrument: Motion control

I employed the idea of motion control so that that the user interface must imitate the way people think, rather than get people to think the way computers do. Therefore, I created an interface based on already familiar human gestures (Holtzman, 1994).

The sounds produced by the instrument are generated and controlled exclusively by the recognition and tracking of the objects on the tabletop by the camera placed underneath. The objects can be moved all over the table, in any location, up and down, left and right, diagonally, in circular motions and rotated – that is, by familiar human gestures. To make the instrument effective and enjoyable to play – which overall is the main goal here – relationships between movement and sound are established. These relationships can be obvious, such as moving the pitch cube object in an upward motion on the table, raising the pitch of the generated sound; but also not so obvious, such as the rotation of a timbre object that determines the amount of frequency modulation synthesis applied to the generated sound, which may only be realised if you are experienced in electronic music and know exactly what you are doing, not just moving the objects around unknowingly.

2.4.2: Instrument: Multi-touch/multi-user interfaces

Although my instrument does not use traditional multi-touch technology, which is the sensing and tracking of finger-touch points, its interface is developed around the concept of the sensing and tracking of the multiple objects. There are twelve objects, all of which can be placed on the table together, allowing for all eligible parameters to be manipulated simultaneously, that is, if there are enough hands to move them all. In saying this, here is a potential example of how the instrument can be played by more than one player.

2.4.3: Instrument: Exploration of timbre

Although my instrument can generate and control all concepts of music, including pitch, duration, texture, structure, and dynamics and expressive techniques, I have set

it up in a way, through the use of multiple objects, to mainly concentrate on the exploration of the musical concept of timbre – creating vast possibilities of new and different sounds. Ironically, this is the opposite of traditional MIDI instruments, as MIDI can more simply represent pitch, amplitude and time through division of its 0 – 127 value range, however, has more trouble when it comes to timbre, as it is more structurally complex and varied, requiring multiple parameters to be described simultaneously in fine detail to be effective (Winkler, 1998). I do believe, however, that my instrument does achieve this goal of realising new and different timbres with great success.

2.5: Summary

In this chapter I have discussed an interactive music system based on the technology of the *reacTable* that I designed and constructed. The instrument's set-up, including its physical build and the computer software used, was examined, as was the classification and fundamentals of the instrument, focussed mainly on the mapping strategies employed. Limitations of the instrument and the integration of the three principal ideas of motion control, multi-touch/multi-user interfaces and timbre exploration were also discussed. In the next chapter, I will analyse an original work composed on, and for, the instrument.

CHAPTER 3

Original piece for interactive music system

As part of this thesis, I composed and recorded a piece of music for solo player on my interactive music system instrument entitled *Piece for homemade reactable* (Herrington, 2010) – found on the Data DVD, as a quicktime *.mov* file, located in the CD pocket on the back cover of the dissertation. Accompanying the music is a video, taken by the camera placed beneath the tabletop. This provides the observer with a view of the objects being placed on and moved around the interface (with each object's fiducial identification number being presented), allowing them to realise the relationships between the movement of objects and the sounds being produced. One can also relate to the fiducial function table, and the chapter on mapping, while watching/listening to the work to gain an even greater understanding of the relationships.

The work utilises all objects, and thus the manipulation of all eligible parameters of music, discussed in the mapping chapter. The music, however, does not include any spatial element, such as panning or chorus effects. Reverb is used, however, as mentioned earlier, this is used as a way to alter timbre. The reason behind this lack of spatial altering freedom is because the music produced is based more around the idea of timbre, rather than space.

The video image is a "glitchy", black and white representation of what the camera is seeing and thus the visual information it is capturing. The reason for this is because

the *ReacTIVision* software recognises and tracks the fiducials more efficiently when analysing this high contrast video image, than if analysing a standard video image.

3.1: Analysis

In this chapter, I will provide a basic (i.e., without going into specific values of frequencies, amplitudes, etc.) analysis of *Piece for homemade reactable*. It will be a look into the musical functions of the objects, and what they are achieving when placed, displaced and moved around the tabletop interface.

One must realise that, as the camera is placed beneath the table, the x-axis and rotation of each object are inverted. That is to say, if a player is performing the instrument, moving an object from left to right will increase the value of the parameter assigned to the x-axis of that object. The camera, however, will see this as a move from right to left. The same applies to the rotation of objects, as a player would rotate an object clockwise to increase the value of the parameter assigned to the orientation of that object. The camera, however, will see this as an anti-clockwise rotation. The player and camera both relate to the same parameter altering directions of the y-axis. That is to say, when a player moves an object from bottom to top, increasing the value of the parameter assigned to that object, the camera also sees the movement as bottom to top.

The following piece can be broken down into three sections:

1. Introduction of objects and the relationships of movement to sound

2. Exploration of timbre using the three main forms of sound synthesis
   (frequency modulation synthesis, subtractive synthesis and additive synthesis)

3. Improvisation involving many fiducials simultaneously

SECTION 1

At time 0.02 the pitch cube object (F3) is placed on the table, producing a square wave tone at a C1 pitch.

- As the object is raised in value to a near-maximum point on the x-axis (left to right from a players point of view, right to left from the cameras point of view) the volume is raised.

- The object is then rotated, altering the type of square wave.

- The object is then raised in value (bottom to top) to a near-maximum point on the y-axis, raising the pitch of the note produced.

At 0.20 the *Digital Reverb* object (F10) is placed on the table at a near-minimum point on its orientation, enabling a reverberation effect at a low dry/wet amount (mostly dry).

- The object is rotated and raised (clockwise from a players point of view, anti-clockwise from the cameras point of view) in value to a near-maximum point on its orientation, raising the dry/wet amount to mostly wet.

At 0.27 the face of the pitch cube object is altered (F5) to produce a square wave tone at a C3 pitch – the recently added reverb effect can be clearly noticed.

- The object face is quickly altered again (F1) to produce a square wave tone at a C-1 pitch.

- The object face is quickly altered again (F4) to produce a square wave tone at a C2 pitch.

- The object face is then changed back to the original starting pitch (F3) to produce a square wave tone at a C1 pitch.

- The object is rotated slightly, altering the type of square wave.

- The object is then angled in a way to show two faces of the cube to the camera (F3 and F5) enabling a chord of pitches C1 and C3.

- The object is then rested on the face (F5) producing the pitch C3.

At 0.47 the *Digital Reverb* object (F10) is rotated and lowered in value to a near-minimum point on its orientation, lowering the dry/wet amount.

At 0.55 the *LFO to Freq. cut-off* object (F6) is placed on the table at a minimum point on its y-axis, producing a low LFO rate, and at a near-maximum point on its orientation, affecting the original audio signal at a high amount.

- The object is raised to a near-maximum point on the y-axis, producing a higher (or faster) LFO rate as it is raised.

- The object is then rotated in a way to switch straight from maximum amount to minimum amount.

- The object is then lowered on its y-axis to a near-minimum value, while being rotated and raised in value to a near-maximum point on its orientation, resulting in the lowering of the LFO rate and the raising of the LFO amount simultaneously.

At 1.12 the *LFO2 to Amp. cut-off* object (F7) is placed on the table at an above-mid-range point on its y-axis, producing a mid-high LFO2 rate, and at a near-maximum point on its orientation, affecting the original audio signal at a high amount.

- The object is lowered in value to a below-mid-range point on its y-axis, and then back to its original position, lowering (or slowing down) the LFO2 rate, and then raising it again.

At 1.20 the *Digital Reverb* object (F10), currently at a low dry/wet amount, is rotated and raised in value to a near-maximum point on its orientation, raising the dry/wet amount.

At 1.25 the face of the pitch cube object is altered (F2) to produce a square wave tone at a C0 pitch.

At 1.27 the *LFO to Freq. cut-off* object (F6) is removed from the table, switching off the long wobble effect.

At 1.33 the *Scream 4 Distortion* object (F11) is placed on the table, enabling a distortion effect, at a near-minimum point on its x-axis, producing an *Overdrive* distortion type, and at a near-maximum point on its orientation, affecting the original audio signal at a high amount.

- The object is raised in value to an above-mid-range point on its x-axis, altering through multiple types of distortion types.
- The object is rotated and lowered in value to a near-minimum point on its orientation, lowering the distortion amount.

At 1.44 the *Digital Reverb* object (F10) is rotated in a way to switch straight from maximum dry/wet amount to minimum dry/wet amount.

At 1.46, the *Scream 4 Distortion* object (F11) is rotated and raised in value to a near-maximum point on its orientation, raising the distortion amount, while simultaneously raised slightly in value on its x-axis, switching to an alternative type of distortion, and then quickly back again.

At 1.55 the *Two-Band Para. EQ: A-Band* object (F8) is placed on the table, enabling an equaliser, at a near-minimum point on its x-axis and at a near-maximum point on its y-axis, thus highly emphasising the low (or bass) frequencies of the original audio signal.

At 1.58 the *Digital Reverb* object (F10), currently at a low dry/wet amount, is rotated and raised in value to an above-mid-range point on its orientation, raising the dry/wet amount.

At 2.02 the pitch cube object (F2), currently producing a square wave tone at a C0, is lowered in value to a below-mid-range point on its y-axis, lowering the pitch shift of the note.

At 2.08 the *Scream 4 Distortion* object (F11) is rotated in a way to switch straight from maximum distortion amount to minimum distortion amount.
- The object is then removed from the table.

At 2.12, the *LFO2 to Amp. cut-off* object (F7) is lowered in value to the minimum point on its y-axis, lowering the LFO2 rate, and then removed from the table, disabling the effect.

At 2.19, the *Two-Band Para. EQ: A-Band* object (F8) is removed from the table, disabling the equaliser.

At 2.22, the pitch cube (F2) is raised in value to an above-mid-range point on its y-axis, raising the pitch shift of the note.

At 2.23, the *Digital Reverb* object (F10), currently at a high dry/wet amount, is rotated and lowered in value to a near-minimum point on its orientation, lowering the dry/wet amount.

SECTION 2

At 2.30, the *Frequency Modulation Synthesis* object (F12) is placed on the table, enabling the frequency modulation effect (and thus a new layer of sound), at a near-minimum point on its y-axis, producing a modulator number value of 1, and at a near-maximum point on its orientation; meaning that the modulator sine wave is affecting the carrier sine wave at a near-maximum amount (FM amount).

- The object is rotated in a way to switch straight from maximum FM amount to minimum FM amount, and then rotated and raised in value to a near-maximum point on its orientation, raising the FM amount.

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- The object is raised in value to a near-maximum point on its y-axis, raising the modulator number value to 32.

- The object is rotated and lowered in value to a near-minimum point on its orientation, lowering the FM amount.

- The object is then lowered in value to a near-minimum point on its y-axis, lowering the modulator number value to 1, while simultaneously rotated and raised in value to a near-maximum point on its orientation, raising the FM amount.

- The object is then removed from the table, disabling the frequency modulation synthesis effect.

At 2.44 the *Digital Reverb* object (F10), currently at a low dry/wet amount, is rotated and raised in value to a near-maximum point on its orientation, raising the dry/wet amount.

At 2.50 the *Subtractive Synthesis* object (F13) is placed on the table, enabling the subtractive synthesis effect (or bandpass filter), at a near-minimum point on its y-axis, producing a low resonance amount, and at a near-minimum point on its orientation, producing a low filter cut-off frequency.

- The object is rotated and raised in value to a maximum point on its orientation, raising the filter cut-off frequency, then immediately rotated and lowered in value to a minimum point on its orientation, lowering the filter cut-off frequency.

- The object is raised in value to a near-maximum point on its y-axis, raising the resonance.

- The object is rotated and lowered in value to a minimum point on its orientation, lowering the filter cut-off frequency, then immediately rotated and raised in value to a near-maximum point on its orientation, raising the filter cut-off frequency.

At 3.00 the *Digital Reverb* object (F10) is rotated in a way to switch straight from maximum dry/wet amount to minimum dry/wet amount.

The Subtractive Synthesis object (F13) is rotated and raised in value to a maximum point on its orientation, raising the filter cut-off frequency, then immediately rotated and lowered in value to a minimum point on its orientation, lowering the filter cut-off frequency, while simultaneously being lowered in value to the minimum point on its y-axis, lowing the resonance – the difference between applying this technique with the reverb effect at a low amount compared to at a full amount is considerable, and makes for a very interesting contrast in sounds.
- The object is then removed from the table, disabling the subtractive synthesis effect.

At 3.10 the pitch cube (F2) is lowered in value to a below-mid-range point on its y-axis, lowering the pitch shift of the note.

At 3.15, the three *Additive Synthesis* objects (F14, F15 and F16) are placed on the table, enabling the additive synthesis effect, and thus adding three new oscillator-produced layers of sound.

[At this point in the piece, there is a cutting in/out of volume. This was due to the camera recognising/derecognising the pitch cube fiducial (F2). At 3.22 the pitch cube is raised in value to a mid-range point on its y-axis (into a location where it can be more easily recognised), raising the pitch shift of the note in order to counteract this problem.]

At 3.23 the three *Additive Synthesis* objects (F14, F15 and F16) are each moved into, and rested at, a different location on the table. That is to say, *Additive Synthesis Osc 1* (F14), producing a sawtooth wave, is positioned at a near-maximum point on its y-axis, producing a near-maximum oscillator volume, and at an above-mid-range point on its orientation, relating to the oscillators octave range (about 5 or 6); *Additive Synthesis Osc 2* (F15), producing a square wave, is positioned at a slightly-under-near-maximum point on its y-axis, producing a slightly-under-near-maximum oscillator volume, and at a highly-above-mid-range point on its orientation, relating to the oscillators octave range (about 7 or 8); and *Additive Synthesis Osc 3* (F16), producing a sine wave, is positioned at a maximum point on its y-axis, producing a maximum oscillator volume, and at a maximum point on its orientation, relating to the oscillators octave range (9).

At 3.32 the face of the pitch cube object is altered quickly between multiple faces (F4 to F1 to F2), producing each face's relative assigned pitches. It can be heard that, due to the additive synthesis technique applied, the timbre (and most notably the attack of each note) has changed significantly.

- The pitch cube (F2) is raised in value to a near-maximum point on its y-axis, raising the pitch shift of the note. The additive synthesis oscillator-produced tones can also be heard to rise.

SECTION 3

[This final section, beginning with the placement of the *Scream 4 Distortion* object (F11) on the tabletop at 3.34, is an improvisation utilising many objects simultaneously to create interesting sounds and music. I will not go into every movement and music result of each object as I have done for the previous two sections, although, I will discuss a few important moments of this final section.]

The first noteworthy moment occurs at 3.58 with the placement of the *LFO2 to Amp. cut-off* object (F7) on the tabletop. The object is placed on the table, enabling the effect, at a mid-range point on its y-axis, producing a mid-range LFO2 rate, and at a near-maximum point on its orientation, affecting the original audio signal at a high amount. The object is lowered in value to a low point on its y-axis, lowering the LFO2 rate. What is now happening is that the amplitude of the original audio signal is raising and lowering (almost to a volume of 0) in a square wave pattern. The amplitudes of the additive synthesis oscillator-produced tones, however, are not rising up and cutting down. The specific placement of the various objects used in this part of the piece enable this contrast to be clearly heard. When the three additive synthesis objects are removed at 4.07, one can only hear the original audio signal, with its amplitude still rising up and cutting down.

Another significant moment of this section occurs at 4.23 with the introduction of the *Two-Band Para. EQ: B-Band* object (F9) to emphasise the higher frequencies of the original audio signal. This object has not been previously used in the entire piece.

The final important moment in this section I will discuss comes at the end of the work at 5.48. The face of the pitch cube object is altered (F0) to produce a square wave tone at a C-2 pitch. The pitch cube is then lowered in value to the minimum point on its y-axis, lowering the pitch shift of the note to produce the lowest possible pitch the instrument can in fact generate. Because the oscillator is producing a square wave at such a low pitch, the waves are longer and a rhythmic beating is created. The pitch cube is then lowered in value to the minimum point on its x-axis, lowering the volume of the note. The object is then removed from the table, switching the note off, and thus bringing the piece to an end.

3.2: Summary

In this chapter, I analysed an original piece of music composed for and performed on the interactive music system I designed and constructed. There is also an accompanying video to the music, so one can clearly realise the relationships between the movement of objects and the sounds produced. In the next chapter I will provide a concluding statement about my Honours research project and dissertation.

CHAPTER 4

Conclusion

In conclusion, the interactive music system I have designed and constructed is intended to be played as a musical instrument, by one or multiple performers. The instrument can produce an almost unlimited range of potential timbres, achieved through gestural interpretation and the concept of multi-touch.

Each object has its own important function in the generation and control of sound, with its movement assigned to manipulate up to three parameters of music, on top of it's standard on/off function, relating to either pitches or effects. This new *reacTable*-based technology research is at the cutting edge of electronic music and is significantly advancing the understanding of multi-user instruments with tangible user interfaces. We are moving rapidly into an era of alternative means for command and communication, where electronic devices respond to touch and visual directions, such as finger swipe commands and recognised symbols employed in *iPads* and *iPods* (Hoye & Kozak, 2010). The field of music and performing arts also need to respond to modern and alternative ways of motion tracking, and their power in controlling the creative response. This *reacTable*-based technology applied in new electronic instruments – or controllers – such as my own, is achieving this, and contributing to music creation and performance of the times. Future research in this area could advance this investigation, for example, by developing a form of notation for the music produced by the instrument or exploring this type of technology in electronic computer-based commercial music such as *Hip-Hop* and *Dance* music.

The interactive computer music system I have created employs a more sound-based approach that embraces dynamic morphology as a foundation for evolving aesthetic and musical outcomes (Paine, 2002), which can be of high electronic-music standard. In addition to be used solely in a performance setting, the instrument can be potentially used in an art installation environment; as an instructional instrument, used as a teaching tool to demonstrate relationships between simple object movement and sound; or as a device used to purely construct certain musical timbres (which can then be saved as a patch and applied musically using other computer software), using a more hands-on approach by moving around multiple objects to achieve an outcome, rather than the one-point mouse approach employed in computer systems. I hope the instrument provides enjoyment for both experienced electronic musicians, and music novices, and presents a new and exciting way of composing, performing, and even thinking about electronic music.

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