Interaction and Interdependence:

Integrating Acoustic and Electronic Interfaces
for the Realtime Exploration of Sounds and Processes

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Master Thesis
Instruments and Interfaces
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The Netherlands
May 2015

ABSTRACT

This research aims to explore acoustic and electronic interfaces for the development of hybrid musical instruments. These instruments attempt to compliment the characteristics and capabilities of the two technologies whilst providing an interface for the realtime and improvisatory exploration, manipulation and organisation of sound. More specifically, the instruments produced during this research aim to increase the sensitivity and reciprocity of interactions between the performer and technology. In particular, multilayered, non-linear and interdependent mappings are implemented in order to provide more engaging and inspiring interactions with the instruments. Furthermore, this research attempts to find strategies for exploring and manipulating the predictability of the instrument's response and its influence on the performers decisions and actions.

ACKNOWLEDGMENTS

I would like to thank everyone who has assisted and inspired me throughout this research.

In particular, Kristina Andersen and Joel Ryan for their dedicated support, inspiration and guidance, along with the staff and artists at STEIM.

Kees Tazelaar, Johan van Kreij, Richard Barrett and the rest of the staff and students of Sonology for their inspiring and insightful lectures and discussions.

Finally, I would like to thank all my friends and family for their ongoing support, encouragement and enthusiasm throughout this process.

CONTENTS

INTRODUCTION

This thesis will attempt to document and describe my research process whilst exposing the various underlying concepts that have inspired and informed the work produced. More specifically, these concepts aim to support this practice based process and demonstrate their influence on the decisions that have been made. Furthermore, the insights that have emerged through this experience will be presented, demonstrating how the concepts discussed in the beginning chapters are explored through the practical work.

This research aims to explore and integrate the various characteristics and capabilities of acoustic and electronic technologies through the development of interfaces for realtime creation of music. In particular, these interfaces should facilitate the immediacy and spontaneity required for improvisational contexts whilst providing a more intuitive and detailed manipulation of the dynamics and subtleties of electronic sounds and processes.

In this way, the research aims to provide a tighter integration of acoustic and electronic technologies, increasing the communication between them whilst expanding the potential capabilities of the performer. Similarly, the resultant interfaces aim to incorporate the sensitivity and complexity of acoustic interactions with the increased precision, memory and automation abilities that electronic technologies provide.

More specifically, notions of mapping and interface design will be explored, investigating the significance of affordances and constraints in a musical context. This will also involve a discussion of the problems that arise during this process and propose possible strategies for solutions.

The first chapter will outline the methodological approach, background motivations and core interests that relate to this research. This will identify the key concepts and areas for exploration whilst providing a deeper understanding towards their influence on my musical practice. More specifically, it will discuss the increasing incorporation of realtime and improvisatory approaches within my work along with the interdependence that exists between performer, instrument and environment.

The second chapter will investigate technological developments and their influence on the creation of music. In particular, the affordances and constraints of various technologies will be discussed in relation to their sonic and interactive potential. This is not intended to be a comprehensive historical overview but rather highlight specific examples that are relevant to the ideas, tools and techniques that have been developed during this research.

The third chapter will focus on realtime interaction with computers within a musical context, identifying and exploring various related issues, approaches and strategies. In particular, concepts regarding indeterminacy, interfaces and mapping will be discussed.

The fourth and fifth chapters will present the practical work carried out during this research. More specifically, this will document the building process of the modified cello and digital modular instruments whilst discussing the relevant aims, insights and results.

CHAPTER ONE: Background and Approach

1.1 Materials, Movement and Momentum

My early approaches towards the creation of music focused around the combination of disparate materials through the use of recording and non-realtime editing processes. More specifically, this involved the dissection, transformation and recombination of various recorded events. This reconstruction process, involving an exploration of the symbolic significance and associative qualities of the sonic materials, results in a composite musical space guided by my decisions and actions. These previously isolated events, separated by time and space, become interwoven through this process, revealing a new perceptual space within the music. In this way, my own personal experience and reflections become embedded in the music, combining the internal and external perspectives provided by myself and the microphones.

The repeated and focussed listening that is necessary whilst working with recorded materials increased my awareness towards the richness and subtlety of detail present in acoustic sound. This led me to search for ways of reproducing and incorporating this level of detail within the electronic elements of my music. For example, the dense layering of parameter automation was used to accentuate and attenuate specific characteristics of the sound. Such tedious and time consuming processes have often resulted in the loss of perspective, creating unbalanced proportions within the macro level structure of the music.The prolonged focus and adjustment to micro level details can lead to small sections losing their relation to the larger structure and direction of the piece. This can be particularly problematic when blending materials and creating transitions.

In comparison to acoustic sounds that exist in the same space, the sounds combined through these editing processes often drew my attention towards the artificial nature of the composed space that arises in the music. The presence of these undesirable artefacts resulted in a lack of perceived continuity, distracting the focus of the listener and reducing the immersive quality of the music. More specifically, the juxtaposition of artificial and acoustic sounds highlights the differences between the perceived and physical spaces.This experiential rupturing of the perceived continuity shifted my focus towards the significance of its role within a musical context. Thus, revealing itself as an element for compositional consideration that can be manipulated to generate varying degrees of presence, distance and abstraction. In this way, the listener's focus is shifted between the perceived and physical spaces.

In addition to this, I developed an increasing interest in the physical properties that are perceived through listening. Sounds can give suggestions to the size, weight, texture and movement of its source based on our previous experiences gained through interactions with physical objects (Truax, 1984), (Smalley, 1997). Similarly, sound provides cues that facilitate our orientation within a space, informing us about the distances, shapes and sizes of the space and the objects within it. *“The sound wave arriving at the ear is the analogue of the current state of the physical environment, because as the wave travels, it is changed by each interaction with the environment.”* (Truax, 1984, p.15)

These experiences led me to develop strategies for creating movements and behaviours with the complexity, continuity and levels of detail that exist in the physical world. In particular, this resulted in a more performative use of the microphone, objects and environment during the recording process. Thus, allowing me to explore the sonic potential of these elements in a more expressive and intuitive manner. Here similarities can be drawn with the techniques used for the *Mikrophonie* pieces by Stockhausen (1989).

Although, my approach towards the editing and arrangement of the material was achieved using digital manipulations to the recorded sounds as opposed to the meticulous notation processes favoured by Stockhausen. In particular, I found that working with recording technologies in this way facilitated the reproduction of transient and emergent phenomena that may be difficult to reproduce in the physical domain. This approach is exemplified in my previous work *Matterreality* (Noisesinthenight, 2013).

In this way, the relative positions of the microphone and objects within the space can be used to accentuate and attenuate specific characteristics, creating natural movements and momentum in the sound. Furthermore, the use of microphones and other electronic technologies can facilitate an exploration of the internal structure of the objects, amplifying and exposing characteristics that would otherwise be impossible to hear. In particular, the use of multiple microphones provides multiple perspectives which can be blended in different combinations and sequences. Similarly, the resonant properties of objects and environments can be explored through the use of amplification and feedback.

Through experimentation with electronic transformation techniques I became aware of their influence on the perceived trajectories and physical properties of the sound's source. *“There is quite a difference in identification level between a statement which says of a texture, ‘It is stones falling’, a second which says, ‘It sounds like stones falling’, and a third which says, ‘It sounds as if it’s behaving like falling stones’. All three statements are extrinsic connections but in increasing stages of uncertainty and remoteness from reality.“* (Smalley, 1997, p. 110)

For example, the use of transposition and equalisation can alter the perceived size and shape of the sound's source. Such techniques can be used to create varying degrees of abstraction which can be used to build and break a sense of continuity. In this way, the aspects discussed above became the main areas for exploration in my work, building structures based around these properties and playing with the balance between recognition and imagination.

1.2 Improvisation, Interaction and Interdependence

The increasing incorporation of realtime processes in the creation of my work led me to an exploration of acoustic instruments and an interest in improvisation. In particular, I found the immediacy and unpredictability that is inherent to these approaches more engaging and inspiring than non-realtime interactions with the computer. This resulted in larger sections of recorded material being used, reducing the amount of time spent editing whilst retaining a focus on the macro scale structure.

Initially, the recorded material would consist of long improvisations that would be edited later, removing the sections that were either uninspiring. The selected material would then be arranged to form a loose structure which further layers could be built around. These sessions would often involve an exploration of the sonic potential of the objects, with the most inspiring and unexpected material emerging out of experimental intentions and accidental interactions.

Through repeated practice with acoustic instruments I achieved a certain degree of fluency which allowed me to create and organise these sections and structures in a more realtime and improvisatory manner. As a dialogue with the instrument developed, less time was spent searching for sounds and my focus shifted towards their combination and organisation. The immediacy gained by such an approach allows events to be structured as the music and time unfolds, using intuitive and qualitative judgements.

I found these interactions with acoustic instruments particularly engaging and was often reluctant to stop exploring new combinations and variations of the material. In this way, the shift in interaction modes that is required to start editing can be disruptive and undesirable, rupturing this continuous flow of ideas and actions.

The process of learning to control acoustic instruments increased my awareness of the inherent unpredictability and interdependence that exists when interacting with objects in the physical world. The large number of variables and their interdependence is difficult to predict, control and reproduce, giving the instrument the ability to surprise and suggest direction.

Such reciprocal interaction allows sections and structures to emerge in realtime through an exploration of the material, with the behaviours and responses of the materials guiding my decisions and actions. In addition to this, the ability to expressively articulate the subtle details of the sound led me to an awareness of the importance of dynamics, accents, syncopation and subtle shifts in timing. This is particularly prominent when improvising in group contexts were the interweaving of the different performer's intentions, perceptions and aesthetic preferences result in constant fluctuations of time and tempo that can be difficult to achieve using electronic instruments and non real-time processes. Furthermore, the act of improvising with different places and people presents different sets of expectations and possibilities, influencing the decisions that are made.

1.3 Affordances and Constraints

An increased awareness of the interdependence between actions, sounds and spaces has resulted in a more general interest in interaction itself. In particular, the significance of the affordances and constraints involved and their influence on the resultant sounds and their structuring became apparent.

Affordances are the perceived opportunities for interaction that result from the relationships between actor, object and environment. The term constraints refers to the limitations of these relationships, what the interaction does not afford. (Gibson, 1977) The perception of affordances is based on the accumulation of personal and cultural knowledge. Thus, allowing associations and predictions to be made regarding the affordances and constraints of a particular object or interface (Norman, 2004).

The development of musical instruments has emerged out of the combined capabilities of the human body, physical materials and their construction. Different combinations of performers and instruments provide a specific set of sonic possibilities and therefore influences the decisions of the composer. Similarly, instruments and their perceived affordances are often defined by the cultural context, aesthetic preferences and the technologies available. In this way, developments that have contributed to the evolution of musical instruments and styles are usually built upon the accumulation of perceived affordances, constraints and techniques..

This accumulation of associations and conventions allow predictions to be made regarding the response and behaviours of an interface. In this way, the affordances of an instrument are often not immediately obvious and usually require interaction and exploration. This process usually involves a continuous dialogue between the performer and instrument, in which the performer develops an awareness and understanding of their abilities and behaviours.

The acoustic environment in which the performance takes place also plays a significant role in the choices and actions made. For example, the reverberation time of the space can often influence decisions regarding the durations between sound events as the performer compensates for the acoustic response of the space. The influence of an environment's acoustic properties has resulted in the design and construction of spaces for the performance of specific styles of music. Similarly, different spaces have led to the development of different musical styles (Schafer, 1977).

The aspects discussed above have led towards an increased interested in their influence on my perception, experience and playing. Exploring their interdependence, affordances and constraints through realtime interaction with different materials, interfaces and environments.

The physical and cognitive effort exerted whilst playing an instrument can be controlled by the performer thus providing varying degrees of precision. *"The accidental can be exploited through the amount of control exercised over the instrument, from complete - producing exactly what the player dictates - to none at all - letting the instrument have its say"* (Bailey, 1992, p. 100).

This reciprocal relationship between the performer and instrument combined with the limits of the human body and the inherent unpredictability involved in it's interaction with the physical world contributes to the engaging qualities of acoustic instruments. Similarly, such techniques can be used to reduce the effects of habitualisation, allowing the instrument to surprise and inspire the performer, revealing new perspectives and possibilities. (Shklovsky, 1965)

These discoveries led me towards a more conscious consideration of the characteristics and capabilities of the performer, instrument and environment. More specifically, I began to explore how these aspects can be used to create open structures and frameworks within the music. Thus, exploring specific sets of conditions that arise from the instruments, objects and techniques that are used. In this way, the affordances and constraints produced by particular combinations of objects, places and people provides a parameter space of potential and influences the decisions made during a performance. In this way, the interactions that take place during the performance produce a particular trajectory through this space. Furthermore, the process of designing interfaces can be described as an *“exercise in realtime phenomenology”* where the act of their construction creates a different states of relationships between space and the body, perception and action (Rokeby, 2011).

The reduced possibilities that results from the use of time as a constraint focuses attention and encourages spontaneity and direction. For example, By specifying a specific duration for the piece before you start playing influences the introduction, transformation and transitions between materials. This is particularly advantageous when dealing with the seemingly infinite sonic potential of electronic instruments.

Similarly, the performer's current position in time and space restricts and reveals certain possibilities regarding the navigation of an interface. More specifically, the physical distances between various positions, parameters and interaction modes require time and effort to travel between. The limited degrees of freedom and spatial relationships that exist between the human body and instrument can obstruct certain movements. In this way, constraints allow for reflection, encouraging creativity, problem solving and engaging the imagination.

CHAPTER TWO: Acoustic and Digital Instruments

This chapter will discuss acoustic and electronic instruments, identifying and comparing their characteristics. More specifically, the affordances and constraints of these instruments will be explored along with their influence on the music that is made with them.

2.1 Acoustic

Perhaps the most significant characteristic of acoustic instruments is the direct coupling of sound and its interface. The two are inseparable with the sonic capabilities of a particular acoustic instrument as they are intrinsically intertwined with its physical properties. Acoustic instruments are usually categorized according to their materials and modes of interaction used (stringed, woodwind, brass, percussion).

These different groups of instruments also exhibit different modes of interaction with specific instruments becoming increasingly focused on expanding the articulation of particular parameters of sound. Technological advancements throughout history have expanded the capabilities of the performer along with the range of sounds and ways of articulating them. However, an increase in distance between sound and its causal action is apparent.

Furthermore, this facilitated the control of multiple sources whilst reducing the ability to manipulate the details. This is exemplified by the keyboard as it affords polyphonic control, focusing on the selection of multiple pitches and their amplitudes whilst restricting the continuous articulation of the individual notes. This increased distance has been accompanied by a lack of reciprocity and tactile feedback. As a result of this, the links between gesture and sound often become more difficult to perceive for both the audience and performer.

Through the use of mechanical and electrical technologies the energy required to produce sound with an instrument can be outsourced. This reduces the physical demands on the performer, facilitating continuous sound generation and expanding the abilities of the human performer. Thus, allowing the performer to focus on and manipulate other aspects of the music. Furthermore, these developments allow for speeds and precision beyond the capabilities of the human performer to be achieved.

2.2 Amplification

The introduction of amplification brought both technical and creative benefits allowing instruments of different acoustical loudness to play together. This manipulation of amplitude gave instruments a greater dynamic range, magnifying small sounds and increasing the distance the sound travels. This also empowers small groups and the individual performer allowing them to achieve the same sonic presence as an orchestra.

In addition to this, the transducers involved also influence the sound according to the physical properties of the materials they are constructed from. In this way, microphones and loudspeakers can be understood as an extension of the instrument. This is exemplified by the emergence of solid body electric guitars were the amplifier plays a more important role in defining the sonic identity of the instrument. Here the combination of the guitar and amplifier both contribute significantly to the character and capabilities of the instrument. Similarly, transducers can be seen as an instrument in themselves. The membranes of the speakers and the materials between them can be manipulated directly resulting in attenuations and amplifications of their spectral response.

Furthermore, the development of technologies for encoding and decoding information using electrical representations further separated sound from its source. In particular, the invention of radio transmission removed music from its context, increasing the distance it can travel, facilitating its distribution and reaching larger audiences. Thus transcending the spatial limitations previously imposed by acoustic sound and the human body.

2.3 Recording

Devices for the recording and playback of sound also had a significant impact on the production and distribution of music. The storing an recalling of electrical representations of sounds allowed for increased precision in their reproduction. Thus reducing the transient nature of sound and reliance on physical actions and human performers.

Such devices allowed for repeated and more detailed listening experiences, reducing and revealing the distortion/effects of memory through acts of comparative listening. In this way, the separation of sound from its initial production allows for different modes of listening, focusing on the perceptual properties of the sound itself.This is exemplified by the emergence of Musique Concréte and Electroacoustic traditions which aim to explore such possibilities within a musical context. (LaBelle, 2006, p. 25)

The representation of sound using a physical medium affords working with the material in a more direct manner. For example, the use of tape allows various techniques for manipulating the playback and temporal arrangement of sound. More specifically, the physical dissection and recombination of sounds using cutting tools makes possible sharp contrasts, juxtapositions and transitions that were previously unattainable within the domain of acoustic sound.

The performer is no longer creating the sound directly but is involved in manipulating, shaping and transforming pre-existing material. Thus allowing the performer to focus on larger scale structural considerations.

However, such transformations usually took place in the studio and required large amounts of time between actions and achieving the desired sound. This becomes problematic in live performance contexts. Furthermore, technical limitations and varied aesthetic preferences led to the emergence of various strategies for presenting such work in a performance context. These approaches exhibit varying degrees of reciprocity between the performer and technology.

In particular, the absence of human performers and detachment of sound from its source shifted focus towards the spatial qualities of sound. This led many composers to incorporate these aspects into their work, manipulating the movement and distribution of sound in the performance space. Furthermore, these techniques are often performed live by the composer, spatialising pre-arranged materials using multiple loudspeakers. (Zvonar, 1999)

In contrast to this approach, various performers chose to explore the medium's potential for realtime performative interaction. Thus exploiting the inherent physicality and malleability of the material. A prominent example of this is the emergence of turntablism and related practices. Here, the gestures and their influence on the sound are directly perceivable. The relatively limited capabilities but large capacity for expressive articulation allows for high levels of virtuosity to be attained. Furthermore, the use of multiple turntables allows the performer to control multiple sound sources simultaneously.

In addition to this, developments in the interfaces of electronic playback devices expanded the possibilities for performative interaction. Thus providing higher order controls and improving the capabilities for articulation (Collins, 2006, p. 50). Similarly the use of additional playback heads allows the performer to operate on multiple timescales, creating artificial echoes, delays and reverberation.

2.4 Computers

The introduction of computers have expanded our physical and cognitive abilities allowing us to perceive and operate on scales that transcend the temporal and spatial limitations of the human body. More specifically, the increased dexterity and precision provided by such technologies facilitates the manipulation of material on timescales that would be difficult to achieve manually using physical media. This is exemplified by the work of Roads (2006) and Wishart (1996) which features the implementation and development of granular techniques.

The increased computational power of these devices has resulted in a multitude of techniques for the generation and transformation of sound. In particular, this has expanded the possibilities for the analysis, exploration and articulation of timbral/spectral properties. For example, this allows for smooth interpolation and sharp juxtaposition of timbres that was previously unattainable using acoustic instruments.

These techniques are often based around conceptual models of physical phenomena, attempting to simulate/emulate and broaden our understanding of acoustic sound. However, more interesting results often emerge from exploring the boundaries of these models, revealing the points at which they break down. Thus achieving sounds that would otherwise be difficult or even impossible to produce through the interaction of objects in the physical world. This often reveals more about the characteristic of the computer and the limitations of our understandings of natural world.

As a result of these technological advancements, the manipulation of timbre has become an increasingly significant aspect in the creation of music. In addition to this, the ability to work on different time scales has enabled the construction of relationships between the sounds and their structuring. For example, Grisey's *Partiels* (1975) focuses around the development of larger scale structures derived from the characteristics of an individual sound event.

CHAPTER THREE: Interaction with Computers

The emergence of digital logic provided the ability to store and recall large amounts of information. Furthermore, increasing layers of symbolic abstraction were also developed allowing the user to interact in a more natural way, using textual and graphical representations (Dourish, 2004). These developments reduced the specificity of computers, increasing their flexibility and providing a more generalised solution. Thus lending itself to multitasking and the manipulation of multiple sounds and processes.

The multitude of parameters provided by electronic instruments proves problematic when attempting to manipulate them in a realtime and intuitive manner. In addition to this, the vast diversity in aesthetics and approach makes it difficult for commercially produced controllers to address this issue. The majority of these devices also fail to provide the subtlety of expression that is available when playing acoustic instruments, but instead favour more general purpose controls such as knobs and faders. Furthermore, such interfaces tend to inhibit the simultaneous manipulation of multiple parameters that correlates with the multidimensional nature of human gesture and movement. Thus making intuitive navigation and spontaneous action more difficult to achieve.

In addition to this, menus and modes are often implemented to reduce the size of the interface whilst retaining access to all the parameters. This allows multiple sound sources to be manipulated using a single interface with fewer physical parameters. However, the number of actions and time required to access the desired parameter are increased. Also, after switching between different modes the positions of the physical parameters do not correspond with the values in the software. In this way, the interface lacks immediacy, and restricts spontaneous action.

The increased accessibility of sensors and micro-controllers has led many practitioners to adopt strategies that address these problems. More specifically, such practices often involve the development of custom mappings, hardware and software devices. As a result of this, hardware interfaces and their mapping now play a significant role within electronic music creation and strongly influence the structure and form of the resultant music. Furthermore, trends towards miniaturisation have reduced size of devices increasing their portability and making them more viable as a performance instrument

These technological developments, combined with the diversity of individual approaches to such technologies, have created a lack of consensus regarding terminology and classification. This has led to the revision of the more traditional organological classifications of acoustic instruments. Paine (2010), Kvifte and Jensenius (2006) discuss such definitions along with issues regarding the incorporation of digital musical instruments.

3.1 Isolation of Parameters

The timbral parameters provided by acoustic instruments cannot be manipulated individually as they are interwoven with complexity of the natural world and the subtle nuance of human interaction/expression. The constant fluctuation of timbre produced by the large number of parameters and their entanglement contributes to the richness of acoustic sound.

In comparison, electronic techniques for generating and transforming sound can often exhibit a perceivable flatness or static quality. This is partly due to the increased precision and reproducibility that characterises such technologies.

The multitude of parameters provided by control interfaces and synthesis models are often isolated and difficult to control intuitively. They no longer rely on the physical properties of materials and lack the interdependence that is inherent to acoustic instruments.

The relationships between sounds and their parameters that produce this sense of complexity and movement can be reconstructed manually through the mapping and automation of parameters. However this can process can become tedious and time consuming when attempting to achieve a level of complexity and detail similar to that of acoustic sounds.

Similarly the combination of sounds recorded in separate spatial and temporal locations often requires manipulation to achieve a homogeneity similar to that of multiple sounds existing in the same space. Of course such juxtaposition of disparate material can also be viewed as advantageous and explored in a creative manner.

3.2 Predictability

The introduction of randomised and probabilistic processes can reduce the time taken to produce subtle nuances in the sound, facilitating a more streamlined workflow and the simultaneous control of multiple sounds and parameters. In this way, decisions regarding the precise manipulation of the details are made by the instrument allowing for an increased focus on larger scale structures.

In addition to this, such techniques can be used to create sounds and sequences that would otherwise not be found due to the aesthetic preferences and habitual practices of the human composer. However, these techniques can often result in the creation of arbitrary relationships that do not carry the meaningful interrelation of sounds and movements produced in the natural world.

Algorithmic and object-oriented approaches can be used to specify more precise and detailed relationships between objects and parameters. These relationships can exhibit a high degree of interdependence, allowing transformations to be made with relation to other objects and their past and present states (Vaggione, 2001).

More interestingly, such approaches can be used to describe and simulate interactions between objects in the physical world (Fry, 2001). More specifically, complex movements and behaviours can be defined using simple sets of rules and conditions. A prominent example of this is the simulation of swarming behaviour to control the parameters of a granular synthesis engine. (Davis & Karamanlis, 2007)

Here, the performer interacts with the system using higher level controls, facilitating the manipulation of many sounds and parameters simultaneously. The use of the computer to calculate the large number of lower level interactions between parameters reduces the precision of the performer's influence. Thus, resulting in probabilistic responses in which more generalised and ambiguous trajectories can be specified. In this way, such simulations can be used to provide a more engaging interface whilst encouraging emergent behaviour in the micro level details of the sound. Furthermore, such techniques can be used to generate micro and macro level structure simultaneously.

3.3 Mapping as Composition

No longer restricted by acoustical and physical constraints, the relationships between action and sound have become available as a dynamic parameter for manipulation. In addition to this, the mappings between the sound and the interface are not implicit and often involve careful consideration in order to create meaningful and engaging interactions.

The act of constructing these relationships creates a specific set of affordances and constraints regarding the interactions between materials, gestures and sounds. The decisions involved in this process are often guided by the abilities and aesthetic preferences of the composer/performer. Thus mapping can be understood as a compositional process, defining a field of sonic possibilities for the exploration and realisation of a piece (Doornbusch,2010) (Murray-Browne et al, 2011) (Magnusson, 2010).

3.4 Resolution and Reciprocity

The inherent latency induced by the linear, discrete nature of digital computation and its related conversion processes highlights latency as a significant factor. Thus influencing decisions regarding the transduction of gesture, requiring a compromise between speed and resolution to be made.

This is particularly important when considering the tight feedback loop between action and perception that is needed for the navigation and precise articulation of a musical instrument.This is exemplified by the sonic and tactile feedback required when playing acoustic instruments with a continuous control of pitch. The constant adjustments to the relative positions and pressures applied by the performer are guided by this feedback.

In comparison to that of digital computers, the seemingly infinite resolution of the physical world presents us with interfaces and details that exceed the control capabilities of the human body. The difficulty of reproducing specific conditions and the vast number of parameters involved results in a certain degree of unpredictability in the instruments response.

The precision of interactions can be increased through repetitive action and continuous dialogue with the instrument. However, complete control can never be attained and it is often more interesting to embrace the unpredictability involved in such interactions. Thus developing a deeper understanding of the instrument's behaviour , materials and sonic potential whilst increasing awareness and control over the influence of the human body.

Such reciprocal interaction along with the impossibility of complete control contributes to the engaging qualities of acoustic instruments, increasing their longevity and capacity for virtuosity. More specifically, a musical instrument should afford a wide range of behaviours that allow for the continual expansion of technique and provide the flexibility to adapt to different contexts.

Furthermore, Latency and delay can be understood as inherent characteristics of digital computation. The majority of signal analysis and processing techniques necessitate and exploit the storage of sound samples (Lippe, 1996). These delays naturally facilitate a call and response relationship between the performer and computer. However, the roles and influence of both can also be more dynamic, increasing the agency of the computer and the reciprocity present in the interaction.

3.5 Strategies for Re-entanglement

A discussion of mapping strategies is presented by Miranda and Wanderley (2006). Here, the various approaches are defined by their use of either Explicit or Model Based mapping techniques. Explicit mapping techniques involve predefined, fixed mappings whilst Model Based techniques involve adaptive, dynamic mappings. Furthermore, the terms 'one-to-one', 'one-to-many', 'many-to-one' and 'many-to-many' are used to describe the different types of parameter relationships.

Such complex mappings can be used to provide a more engaging interaction than explicit one-to one mappings (Hunt and Kirk, 2000). More specifically, they exhibit the interdependence and non-linearity of parameters that is characteristic of acoustic instruments. Thus allowing the performer to expressively manipulate multiple parameters simultaneously (Hunt, Wanderley and Kirk, 2000). Furthermore, these mappings reduce the spatial requirements of the interface,whilst improving the immediacy of interaction.

Similarly, the organisation and mapping of isolated parameters into multidimensional spaces facilitates intuitive navigation and continuous interpolation between specific states and timbres (Momeni and Wessel, 2003) (Brandtsegg, Saue and Johansen, 2011). Physical models, particle systems and other algorithmic procedures can also be used to explore this space. The use of such models can be used to create complex reactions and behaviours similar to those found in the natural world. Furthermore, these simulations exploit the preexisting knowledge and expectations of the performer, allowing for intuitive and semi-predictable interaction.

These intermediary layers of mapping provide spatial representations of the parameters allowing a more direct correlation to be made between gestures and sounds (Arfib, D. Couturier, J.M. Kessous, L and Verfaille, V., 2002). Thus complimenting the multidimensional nature of human interaction and facilitating the modularity of interfaces and sonic parameters.

Furthermore, analysis and feature extraction techniques can be used to derive qualitative data from the quantitative measurements of audio and control signals. In this way, patterns, direction and meaning can be detected and used to generate meaningful mappings and behaviours. In particular, the use of adaptive mappings can provide appropriate parameters and ranges with relation to materials and context (Tindale, 2007).

CHAPTER FOUR: The Modified Cello

4.1 Introduction

The cello was initially chosen as a sound source and interface for exploring the combination of acoustic and electronic sounds and processes. In particular, the wide sonic range and intuitive control over subtle transformations of the timbral/textural qualities of the sound makes the cello suitable for this purpose. More specifically, the physical size and fretless nature of the cello provide a high resolution and continuous control over pitch. Although, the large size of the cello also limits the speed of movements that can be made whilst playing the instrument and reduces its portability.

The bow acts functions as a lever, amplifying the gestures of the performers arm, allowing for a wide variety of detailed timbral manipulations to be made. Furthermore, an electric cello was chosen, reducing problems with acoustic feedback and allowing for modifications to be made to the body without drastically influencing the acoustic sound. However, the disadvantage of this is that there is a noticeable lack of warmth and depth in the sound when compared to its acoustic counterpart.

The most prominent issue that emerged from previous experiences combining electronic and acoustic instruments is the distances between the two. More specifically, these early attempts used commercially available interfaces consisting of various dials, switches and faders to manipulate software and transform the acoustic sound. As the cello usually requires two hands to fully articulate the sound, this creates a break in the flow of the performance, both cognitively and physically. This is particularly problematic whilst performing in group and improvisational contexts as it reduces the performer's ability to act and react in a spontaneous and intuitive manner.

Similarly, the constant interaction requires to generate acoustic sound makes it difficult to control the electronic interfaces at the same time. This can be overcome to a certain degree by using processes such as reverbs and delays to prolong the decay of the sound whilst changes are made to the electronic interface. In addition to this, the constant shifting of focus between the two interfaces increases the cognitive load of the performer. The performer must retain the current state of the software in their memory as it is not always immediately obvious from the hardware.

4.2 Aims and Initial Research

This project aims to provide a tighter integration between the acoustic and electronic interfaces, creating a hybrid instrument that provides a less disruptive transition between the manipulation of the two. This should facilitate the spontaneity and immediacy required for the creation of electronic and acoustic sounds in a realtime and improvisational context. Similarly, the resultant interface should provide a more expressive and intuitive control over the dynamics and subtleties of electronic sounds and processes.

Research into existing practitioners and related fields provided inspiration and technical information for this study. This also involved organising a meeting in which musicians and colleagues interested in the electronic augmentation of bowed instruments shared their experiences and ideas.

Previous work relating to the augmentation of the cello provides various examples from which inspiration can be drawn. In particular, the cello developed by CNMAT researchers in collaboration with Frances Marie Uitti (2006) provides an interesting example which uses various FSRs and switches in combination with a polyphonic pickup. The data from these components is primarily used to navigate through various presets and processing possibilities.

Furthermore, various solutions for augmenting the bow have been developed. A prominent example of which is the Kbow produced by Keith McMillan (2012). The bow uses an accelerometer, and various pressure sensors to gather gestural data from the bow in a non intrusive manner. Similarly, various artists have developed personal solutions such as the 'Fello' by Andi Otto (n.d) who uses an FSR and accelerometer mounted on the bow in combination with pedals and various commercial controllers.

4.3 Process and Potential

The development process for the cello mainly focused around developing an awareness of the intentions, characteristics and constraints of the cello, computer and myself. From this, technologies and techniques have been developed which augment these aspects and achieve a balance between physical and cognitive loads of the performer. More specifically, this involved an iterative process of playing, reflection, design and construction, with the sonic and physical experience of playing the instrument providing the main source of momentum. For this reason, the cello was also played in various group and solo contexts throughout the process. In addition to this a paper was written discussing the methodological approach taken to realise this project (Andersen and Gibson, 2015).

Initial explorations of the cello involved developing an understanding of the instrument's sonic and physical properties through the acts of playing, practice and reflection. This allowed the cello to reveal itself through different modes of interaction and helped build embodied and conceptual knowledge of the instrument's affordances and constraints. More specifically, this process focused around issues relating to the stability, balance and symmetry of the instrument and performer. From this, several areas around the neck and body of the cello were identified as being easily accessible whilst playing. This resulted in the incorporation of two panels, one on either side of the cello, for the placement of sensors.

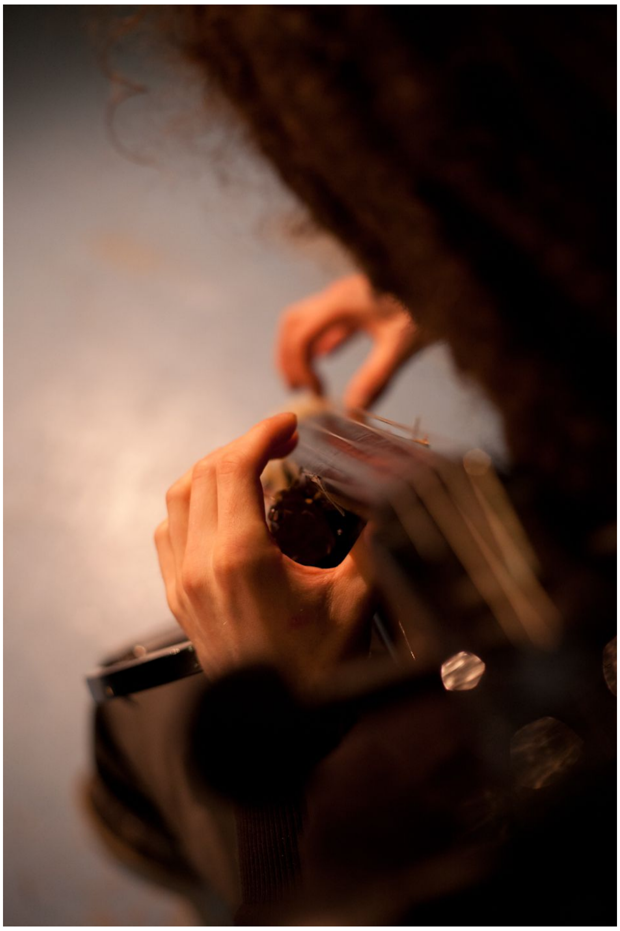


Fig. 1, Playing the cello

As a dialogue develops between the musician and instrument, various possible extensions and transformations to the sound and gestures are imagined. In this way, the act of playing acts as the main source of momentum and evaluation for the project and ensures an integration between the new and preexisting vocabulary of techniques.

4.4 Dissection and Analysis

The conceptual and physical deconstruction of the cello facilitated a deeper understanding of the functionality and interdependence of its various components. By reducing the cello to its fundamental components I became more aware of the physical processes that take place when interacting with the instrument. Through playing with these objects new possible directions emerge in which other aspects of the instrument's sound are emphasised and explored.

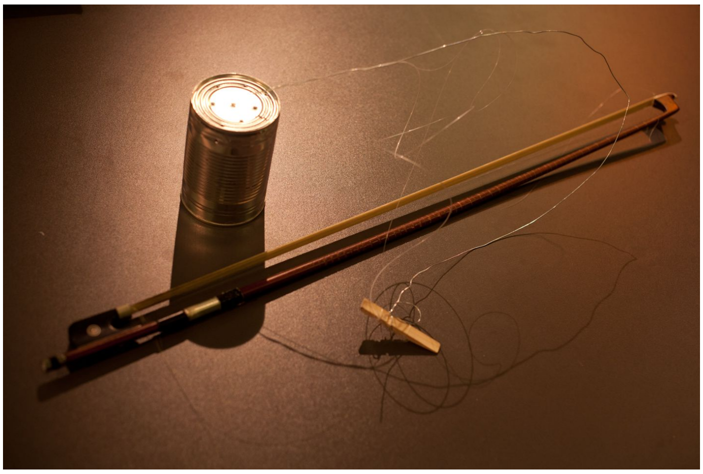


Fig. 2, The deconstructed cello

Similarly, various orientations between the cello and performer aimed to explore the instrument from different angles and reveal the extent that each constricts and frees the performers movements.

This process also focused around the following questions:
- How much can I remove/modify until I no longer perceive it as a cello?
- How much is the concept of a cello derived from its historical context?

Fig. 3, Exploring possible orientations and playing modes

Thus encouraging new perspectives regarding the cello, removing it from these references and revealing itself as an elaborate arrangement of sounding objects.

Fig. 4, Exploring possible modifications to the cello

From this, various techniques for expanding the acoustic capabilities of the instrument were developed. For example, experimentation with strings of different materials facilitated an understanding of their unique qualities and the subtleties of friction and pressure that are involved in the creation of different sounds. Similarly, notches were cut into the back of the bow, expanding the percussive and timbral possibilities of the instrument.

4.5 Sound Transformation

The software for this instrument was created using Max MSP and aims to provide simple processing that expands the timbral and pitch ranges of the cello whilst creating engaging mappings and interaction with the acoustic signal. In addition to this, delays and buffers expand the physical capabilities of the performer and allow rhythmic interplay between machine and human time domains. Such processes also facilitate iterative transformations and structures in which timbral modifications of the cello sound can be layered to create evolving textures of varying densities.

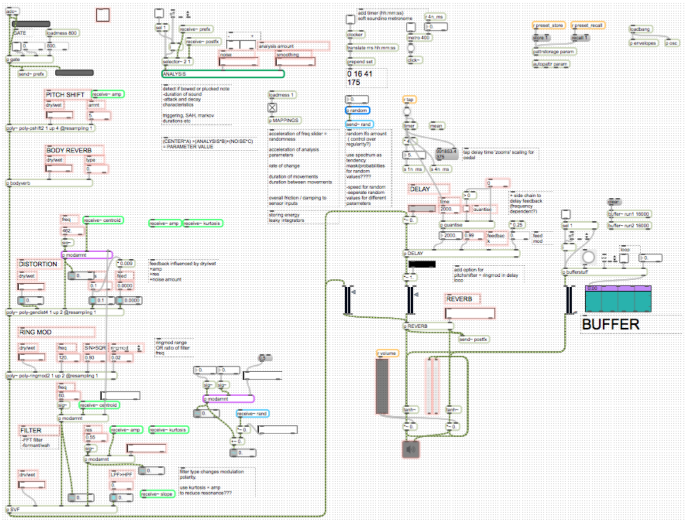


Fig. 5, Screenshot of the Max MSP patch

Furthermore, the use of delays and buffers to extend the sound reduces the need for the continuous interaction required to create acoustic sound, providing longer durations for the adjustment of other parameters. Similarly, the introduction of filtering, ring modulation and pitch shifting into the feedback loop of the delay produces a wide variety of interesting timbres. In addition to this, several techniques for creating acoustical manipulations to the cello sound were explored. This is exemplified by the notches cut into the back of the cello bow, allowing different rhythmic and timbral qualities to be explored. Pitch shifting was used to expand the pitch range of the cello, enabling the playing of notes in the viola and violin ranges. This is particularly useful when combined with the sampler as dense sections can be gradually layered. Additionally, ring modulation can be used to create numerous metallic, bell like timbres and is particularly effective with percussive sounds.

4.6 Expressing the Digital in the Tangible

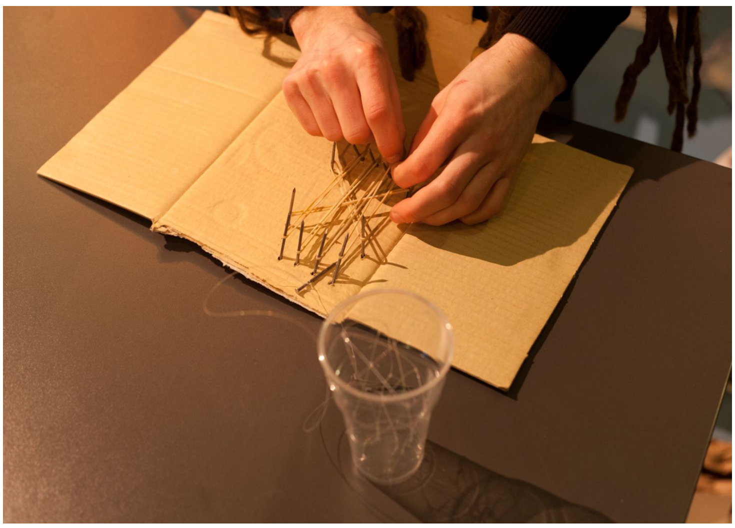


Fig. 6, Exploring notions of interdependence using physical materials

Further explorations involved the creation of simple sketches and models in order to further define the abstract conceptions of the desired extensions and transformations to be made to the software parameters.The sketches were completed with limited materials and time constraints, retaining a focus on ideas with less significance being attributed to the functionality and aesthetic qualities of the results. This process aimed to reduce the conceptual and perceptual gap between the digital and physical domains, providing direction and insight through tangible interaction.

Similarly, the creation of models using low cost and disposable materials facilitated a holistic thinking towards interaction with the digital elements of the instrument. The interdependence between individual parameters and their influence on the system as a whole was considered more thoroughly.

In particular, the elasticity, malleability and structure of interwoven rubber bands helped explore this concept and the ways in which it could be expanded and translated into mappings in the digital domain. In this way, these models helped focus on the experiential aspects of this interaction and generate techniques for navigating the complexity of the software through intuition and gesture.

4.7 Analysis and Mappings

The configuration of sensors and mappings was organised around the bimanual techniques of the cello and other acoustic instruments (left side of body = pitch/ timbre, right side = duration/ volume).
This was found to be particularly effective as it facilitates the incorporation of previously obtained gestures and techniques.

The use of sliders provides clear visual feedback and facilitates faster, less disruptive movements between specific parameter positions. The vertical placement of these sliders also act as fret markers, complimenting the fretting positions and gestures of the left hand.

In addition to this, a series of 3D printed dials of various sizes have been designed to be manipulated with the cello bow in the right hand. Similarly, the addition of two expression pedals extends this concept and provides a less disruptive control source.

Fig. 7, Temporary panels constructed from cardboard

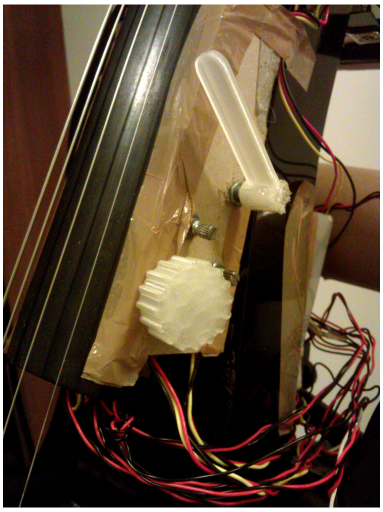


Fig. 8, Early versions of the enlarged dials

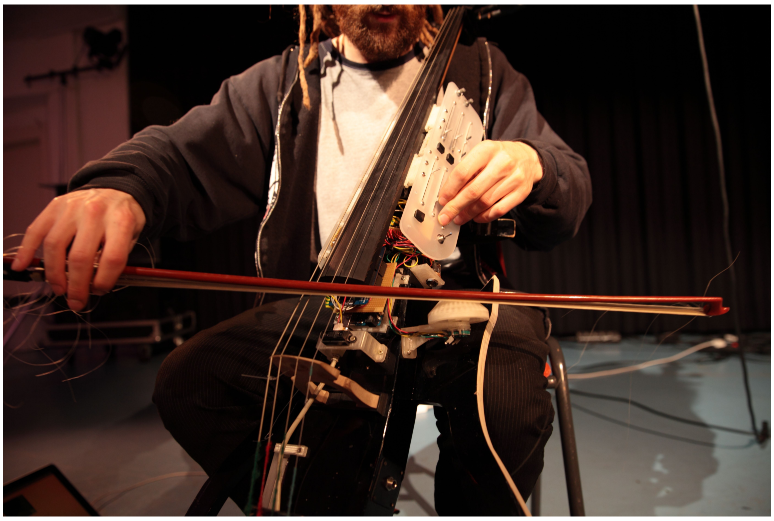


Fig. 9, Playing the final bowable dial

The mappings for this instrument facilitate a more reactive and engaging interaction with the software by providing perceptually significant and interdependent parameters. This allows the instrument to suggest possible direction for sonic exploration and encourages the tactile and serendipitous learning experience of acoustic instruments and improvised music.

In addition to this, audio analysis (Malt and Jourdan, n.d.) was used in an attempt to determine the intentions of the performer and generate complementary or contrasting reactions in the processing of the cello signal. In this way, the analysis acts as a modulation source, influencing the current parameter positions.

The signals generated by the analysis are also recorded into a buffer with the playback sample values being chosen randomly, producing a probabilistic output that is related to the previous input signal. These values can also be crossfaded with a noise generator, allowing this relatedness and the regularity of the values to manipulated. Furthermore, the input source for the analysis can be taken either pre or post processing, creating interesting fluctuations due to digital rounding and timbral changes of the decaying acoustic sound.

A prominent example of this is the use of pitch/centroid tracking to control the filter within the feedback loop of the distortion. This is most perceivable when the distortion is set to a high value/amount and the feedback loop produces dominant resonant tones which can then be controlled using a combination of the expression pedal and audio input (strings). The amplification of the acoustic signal increases the sensitivity to changes in the acoustic signal allowing extremely subtle movements on the strings and body of the instrument to destabilise the feedback and shift the dominant tone.

4.8 Reflections and Conclusions

The combination of acoustic input and DSP has resulted in the development of numerous new timbres and playing techniques. In particular, a wide range of percussive sounds can be created and manipulated using the ring modulation and short delay lines. Additionally, the combination of audio analysis and bowing techniques provides a detailed and subtle control over the processing and texture of the sound.

To conclude, the combination of acoustic input and DSP has resulted in the development of numerous new timbres and playing techniques. In particular, the combination of audio analysis and bowing techniques provides a detailed and subtle control over the processing and texture of the sound. In this way, I feel that the separation between the acoustic and electronic elements of the instrument have been reduced.

Overall, the cello provides good solution for tonal and percussive sounds, although the latter tend to be more difficult to play in patterns due to the traditional posture and size of the instrument. This can be compensated for by placing the cello on its back, freeing the performer and facilitating percussive techniques. Additionally, The strong historical, associative and physical qualities of the cello are focused around tonality and create certain expectations which I find inappropriate for certain contexts.

CHAPTER FIVE: The Hybrid Modular System

5.1 Introduction

The instrument consists of a series of modules with various inputs and outputs, allowing interactions to take place between them. The functionality of each module is defined in the software, using SuperCollider to generate a variety of different sounds and processes. In addition to this, a corresponding hardware interface provides tangible representations of the significant parameters and allows connections between the inputs and outputs for each module to be specified using physical patch cables. In this way, the instrument can be used to define and manipulate networks of relationships between various actions and reactions / gestures and sounds. In essence, providing an instrument with reconfigurable mappings and characteristics.

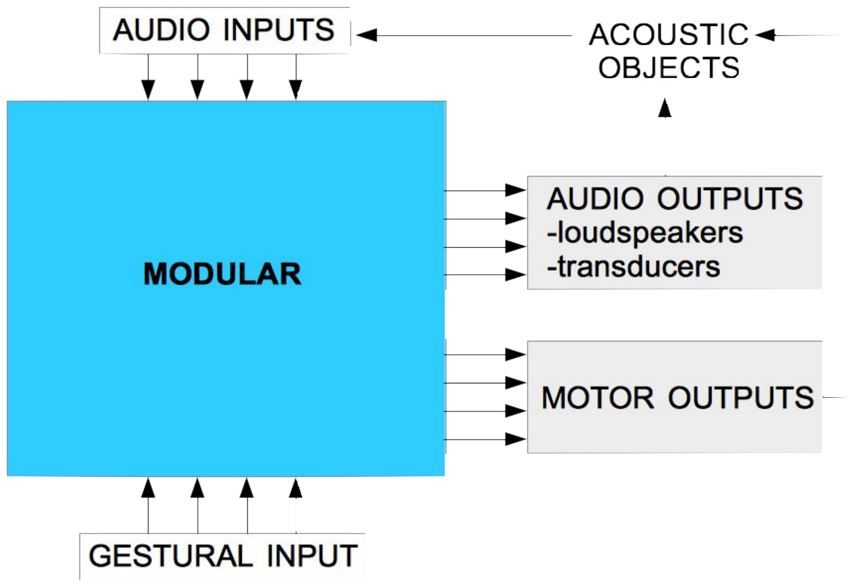


Fig. 10, Simplified diagram demonstrating modular's ability to interact with the physical world

<u>physical patch cables</u>
The use of physical patch cables provides a visual and tactile representation of these relationships, facilitating navigation of the interface through a combination of muscle memory and sensory feedback. The presence of physical patch points makes it easer to comprehend the possible connections, reducing the cognitive load and reliance on memory as the affordances are perceived more directly. Similarly, the delimited patching space provided by the hardware encourages a continuous flow of ideas and actions, reducing the often disruptive and tedious nature of mapping using the keyboard and mouse.

This can be particularly problematic when using software patching interfaces such as Max MSP where the seemingly infinite possibilities can become distracting, making it difficult to gain a sense of momentum and direction. In comparison to working with physical materials, where every action reveals and restricts future possibilities, the software is less influenced by the limitations inherent in physical interactions. These limitations allow for reflection and provide a more inspiring and engaging interaction.

In this way, the modular attempts to reduce the cognitive load required by the technical aspects of creating mappings, providing a more immediate and engaging interface whilst allowing the performer to focus on the sound and its structuring.

accessibility of parameters

However, the flexibility of software mapping (practically infinite cables and connections) is also retained as the functionality of the various hardware components and patch points can be changed in the software if necessary. This allows the relevant parameters to be made available during a performance whilst hiding the parameters that are used less frequently.

In this way, the modular system should provide a framework, allowing the system to expand and shrink in size as more or less functionality is required due to changes in aesthetics or the exploration of different materials and contexts. For example, it might be desirable to change the behaviour of the software to compliment the capabilities and qualities of certain sounds or instruments.

The shift in interaction modes required to make such changes to the system creates a separation between preparatory and performative processes. This encourages the development of muscle memory and virtuosity as the temptation to continually modify and expand the system is reduced. Furthermore, the software can easily be modified to allow communication with external devices. This expands the flexibility of the instrument facilitating integration with preexisting control interfaces and custom configurations of sensors and actuators.

patching as composition

The decisions regarding the mapping of the instrument, made by adding and removing patch cables, define a field of sonic and structural possibilities. In this way, the act of mapping resembles a compositional process, emphasising the design of interactions between sounds and processes, allowing specific sets of conditions and behaviours, along with their structural significance, to be created and explored.

More specifically, this interface facilitates the use of interdependent control structures and many to many mappings, allowing for interrelated micro level details and higher order, macro level controls to be constructed. These macro level controls are well suited to the multidimensional nature of gestural interaction, providing simultaneous control of multiple parameters.

memory and automation

The continual dialogue between performer and instrument allows for the building of intuitive knowledge, exploring nuances and constructing a cognitive map of the possibilities for use within a performance. Similarly, the ability storing of sounds, shapes and their mappings during this process allows for varying amounts of materials to be prepared before a performance. This helps to reduce the often transient nature of these phenomena by increasing the reproducibility of specific conditions and extending the capabilities of human memory.

Such techniques shift the performer's focus towards the larger scale structure of the piece as the details become automatic. For example, choices can be made about the order of patterns without having to consciously think about the pattern itself.

materials and context
In addition to this, musical structures can be created that are specific to the materials and environment used for a particular performance. This facilitates a bottom up approach to composition and opens up the possibility of augmenting the abilities, behaviours and qualities of the materials. In this way, the instrument should also exhibit a sensitivity towards the environmental conditions and intentions of the performer in order to produce meaningful and appropriate reactions and responses.

Transducers can be used to facilitate communication between acoustic and electronic phenomena. The use of analysis and feedback within the mappings of these elements can be used to create dynamic and complex systems of excitors and resonators with variable states of stability. This concept can be expanded further with the introduction of logic functions to create structural elements within the music, switching between states and triggering control signals in the software.

interdependence and unpredictability
The use of interdependent mappings and probabilistic functions allows the performer to define states of varying determinacy. During a performance transitions can be made between these states providing a control over the predictability of the instrument's response. Thus blurring the causality between events and actions, increasing the awareness of the performer and encouraging the exploration of new sounds and structures within the music.

This is particularly engaging when the influence of the performer and environment can be perceived but not fully understood. For example, acoustic instruments exhibit many layers of interrelated detail that are initially inaccessible to control. Increased precision can be gained, developing an awareness towards these details and then learning to control them through repetition and training of the body and mind.

In this way, the unpredictability of such probabilistic processes intrigue and invite further exploration of the system, the relationships between its components and the influence of the performer.

5.2 Initial Experiments

patchbay

This circuit was initially created using a breadboard to gain a deeper understanding of it's structure and the functionality of it's components. This process allowed me to analyse and explore the potential of the circuit, revealing ways it could be adapted for use with my instrument. More specifically, shift registers were used to expand the number of inputs and outputs available and various techniques for manipulating the connections in the analog domain were developed.

However, the small size of the interface became difficult to navigate especially when attempting to locate specific inputs and outputs. Similarly, patching new connections without disturbing the existing ones becomes problematic, especially when dealing with particularly dense patches. In addition to this, troubleshooting can also be an issue due to the general instability of the connections made using the breadboard.

For these reasons, the circuit was then transferred on to perfboard, adding headers to determine paths of communication between the various shift registers. This retains some of the flexibility of the breadboard circuit whilst allowing for the exploration of different configurations during the development process.

The patching interface uses a micro-controller to read and send information about the state of the patchbay to the computer using the serial protocol. This was achieved by sequentially sending a HIGH (5v) voltage out of each output pin and measuring the voltage levels for each of the inputs. This means that when a connection is made between an input and output, the voltage being sent from the output will be read in the input pin, returning a HIGH value.

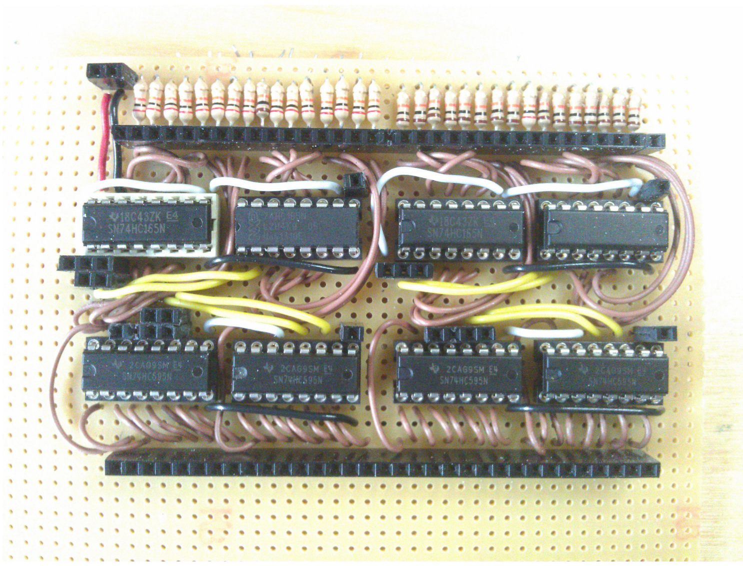


Fig. 11, first prototype for the patchbay circuit

This process is repeated continuously, at speeds fast enough to be imperceptible. However, due to the design of this device, adding more inputs and outputs drastically increases the time taken to read all of the pins. This became problematic when using large numbers of inputs and outputs as it introduced a significant delay between making a physical connection and its replication in the software. Additionally, this limits the expandability of the instrument as all of the inputs and outputs must be connected to the same circuit.

After measuring the time taken to execute different elements in the code, the continuous sending of serial data was identified as the main cause for the delay. This was improved by storing the data in a 2D array and making comparisons between past and current data. This allows the serial data to be sent only when changes to the patchbay have been detected. In this way, the amount of data being sent to the software was reduced by only sending the values that have changed rather than sending the entire array every time. The disadvantage to this is that it increases the chances of creating discontinuities between the software and hardware.

Ideally the circuit would use DAC/ADCs to transmit continuous data, allowing the actual control/audio data to be sent through the physical cables. This method would also facilitate the interaction with existing standards and systems such as control voltage and audio signals. However this approach is computationally and financially expensive due to the high resolution/sample rate and the large number of inputs and outputs that would be required. This approach is used by existing hardware modules that facilitate the interaction between the computer and devices that operate using control voltage (Gonçalves, 2011).

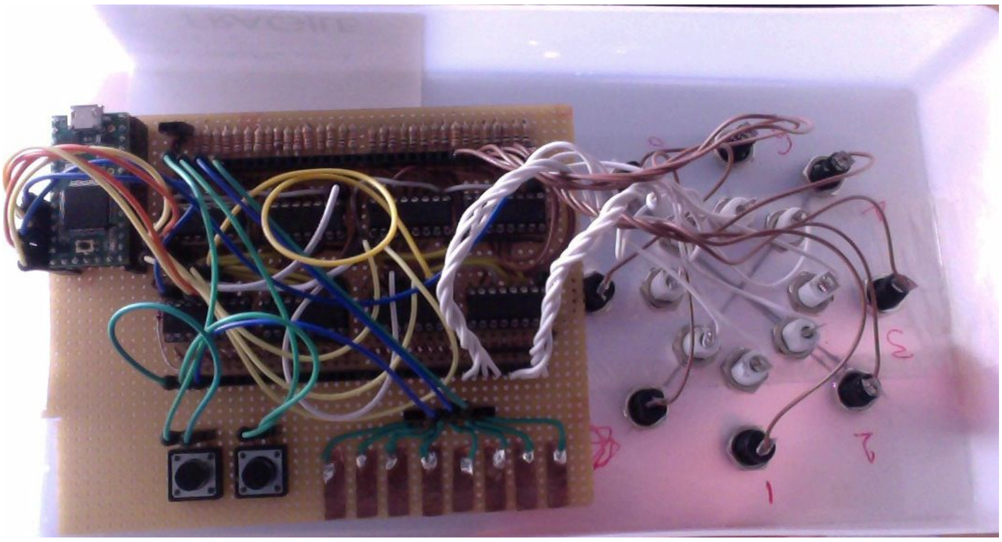


Fig. 12, patchbay circuit with added banana connectors, switches and touch contacts

The advantages gained from using digital signals to represent connections made in the software allows for more flexibility regarding the instruments response to interaction. More specifically, this allows the patching of multidimensional data such as gestures and groups of events. This is particularly advantageous when dealing with spatial trajectories and the simultaneous control of multiple sound parameters.

In addition to this, connections can be displaced through time, allowing for musically interesting transitions such as fading and timing offsets to be achieved when making or breaking connections. This allows the performer to think and act over different timescales, scheduling events in the future and facilitating the simultaneous control of multiple sounds and process.

The representational nature of the interface also allows for the storing and recalling of the instrument's state. This is particularly advantageous in a performance situation as it allows for fast transitions to be made between complex mapping networks that would otherwise be tedious, disruptive and time consuming to recreate manually using the physical cables.

However, the disadvantage to this is that the physical cables will no longer reflect the internal state of the software when previous states are recalled and could lead to confusion and hesitation. Here, the stored states can also be used as a reference point, restoring a familiar state and reorienting the performer when the patch gets too complex.

States can also be recorded automatically for every change made to the instrument, generating a set of instructions which allow for the analysis and reproduction of previous interactions and states. This process can be useful whilst generating material and ideas allowing states of interest to be stored for later exploration without disrupting the flow of ideas.

Furthermore, musical manipulations to these states and the transitions between them can be used to create contrast, direction and structure. New states can be derived from preexisting ones by adjusting and interpolating between specific sets of parameters. In this way, the parameters of multiple states can be analysed to generate ranges for probabilistic regeneration, providing control of the similarity and deviation of the new state from the originals. In addition to this, these states can be sequenced and modulated allowing for more global shifts and transformations to be made, influencing large groups or all of the parameters simultaneously.

patching between the physical and virtual

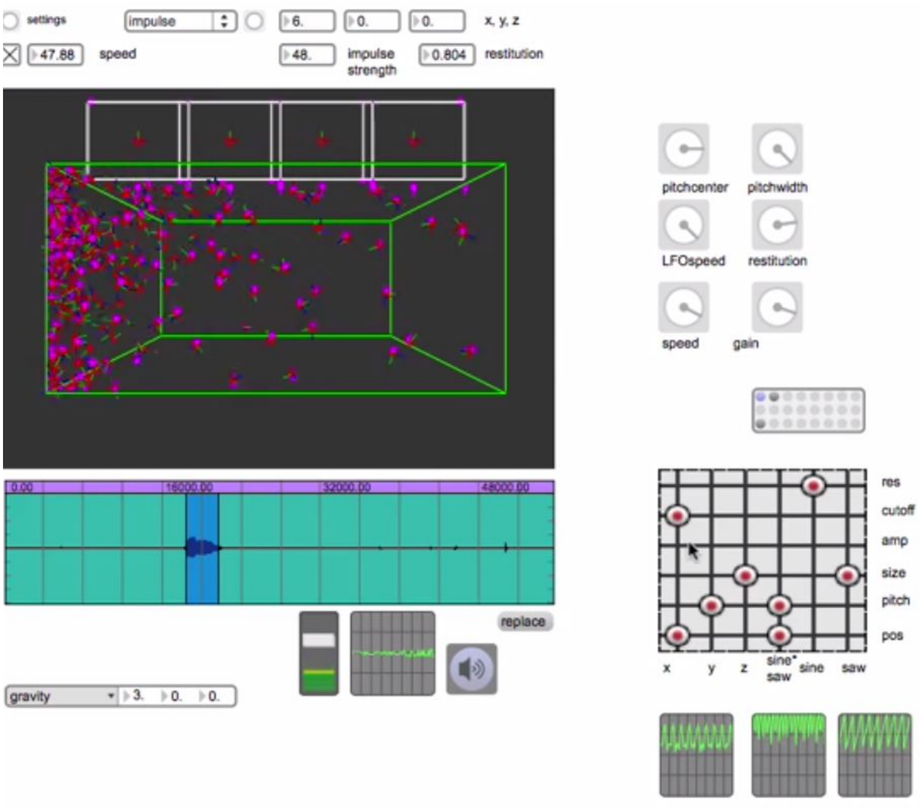


Fig. 13, Screenshot from the particle system Max MSP patch

This patch uses Max MSP to simulate the movement of particles in a virtual space. The positions of these particles are mapped to the speed and amplitudes of a granular synthesis engine. Because the simulation reacts in a way that is similar to the physical world it can be manipulated intuitively, building on existing knowledge and providing a more reciprocal interaction between the performer and sound. In addition to this, sensors can be used to influence the particles and parameters of the simulation with multidimensional gestures.

The patchbay circuit that was shown previously was used to specify the mappings between the movement of the particles and gestural control provided by the joystick. Connections made on the physical patchbay are represented in the software using the matrixctrl~ object. This object acts as a 6x6 matrix mixer which directs the streams of data. Later, automated control signals were added to the patch, allowing explorations of the balance and interactions between human and computer controlled gestures.

I found that this system was particularly useful for creating gradual evolving changes in sound. However fast and precise movements were more difficult to achieve. In this way, this interface could benefit from the combination of more direct mappings to the parameters of the grain generator.

CMOS modular synthesiser

After developing an understanding of the technical aspects required for building the patchbay circuit, the musical potential of this knowledge was explored. This involved the creation of a small modular synthesiser using various CMOS integrated circuits. More specifically, the 40106, 4040, 4051 and 4070 chips were used to create oscillators, dividers and sequencers. Initially, a breadboard was used to explore the functionality and musical qualities of these components, creating a simple circuit built around square waves and digital logic.

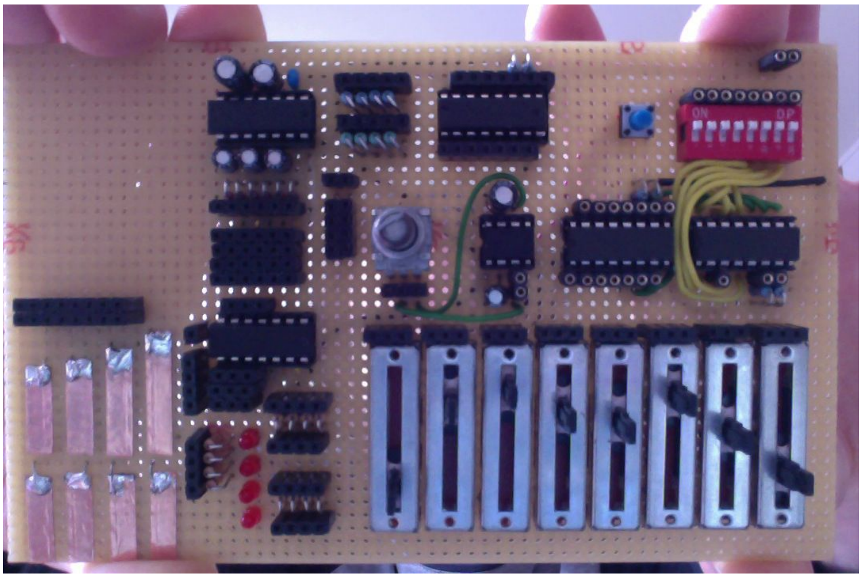


Fig. 14, CMOS modular synthesiser

Once a familiarity and understanding of the functionality and behaviour of the components had been developed, they were arranged and soldered to perfboard. This construction process was mainly improvised, with minimal preplanning, often returning to the breadboard to test and confirm ideas.

process

This process focused thinking around the spatial layout of the interface elements and their significance on how you approach and interact with the instrument. Proposing questions such as: What possibilities are revealed and hidden? and What direction and structures are suggested?

In particular, the grouping and positioning of the different components in relation to their functionality facilitates the creation of higher levels of structure and attempts to achieve a balance between specificity and generality.

For example, a 555 timer was placed next the step sequencer and prewired to a potentiometer for immediate control of its frequency. This in combination with its central position on the board suggests its use as a master clock with signal flow travelling from left to right across the board (to the sequencer). Similarly, the placement of the sliders at the bottom of the board provides a series of immediate patchable controls and distances them from the patch cables making them more easily accessible.

The use of fixed routings can simplify and reduce the amount of patching required for frequently used combinations of components, providing quick access to frequently used structures and mappings. Thus facilitating a less disruptive navigation of the instrument and an increased focus on the aesthetic considerations related to the sound.

The circuit was also constructed over a short timeframe, focusing on the exploration of ideas, providing immediate of results and reflection. In this way, the act of physically building the circuit encourages a continuous flow of thoughts and actions. In particular, musical interaction with the instrument throughout the building process led to many of the design choices being informed directly from the experience of playing. More specifically, This allowed me to listen to the instrument's expanding behaviours and capabilities whilst facilitating the imagining of new ones. This usually resulted in a desire to achieve a certain sound, or explore certain mapping structures. In this way, this process provided inspiration, momentum and direction throughout the development of the circuit and helped shape its sonic characteristics.

Similarly, this reciprocal relationship between building and playing also revealed more practical problems and desires that might have been previously overlooked. This resulted in the implementation of various elements that allow for the duplication and combination of signals. For example, two passive mixers allow for the polyphonic layering of sounds. Each mixer has four inputs and four duplicates of the mixed output. The use of two separate mixers allows for the creation of sub mixes and feedback. Several multipliers were also added for splitting and sending signals to multiple locations. These simple additions to the circuit are particularly advantageous as they facilitates the creation of more complex and interdependent mappings between the different components and their signals.

<u>physical and digital space</u>
The finite spatial restrictions involved whilst building on a single piece of perfboard provided a complementary contrast to the seemingly endless possibilities provided when working with digital space. In this way, the inherent limitations of the physical materials provide a conceptually graspable set of possibilities in which to work. However, care must be taken to achieve a balance between functionality and usability, In particular, the layout and spatial proximity of the various components should allow enough space to add and remove patch cables whilst not disturbing other connections. This was found to be problematic when creating large patches.

Similarly, a compromise must also be made between the desired functionality and the physical practicality of the building process. For example, the position of the input and output headers for ICs were initially placed next to the pins to simplify the soldering process. After experimenting with the circuit this was found to be problematic as the function of the headers can easily be confused. During the second half of the build, the headers were placed further away from the ICs and positioned in a way that provides a clearer signification of the patch point's functionality. However, positioning the patch points so that they are more suggestive of their function also increases the complexity of the building process as additional wires have to be connected between the IC pins and the headers.

Initially, a decision was made against the inclusion of visual feedback for the circuit as this would increase the complexity of it's construction and detract focus from the exploratory intentions of this process. After experimenting with the sonic capabilities of the circuit, four patchable LEDs were added for providing visual feedback. They can be routed to display specific outputs or visualise signal flow whilst patching. This decision to make the visual feedback optional helps retain the unpredictable nature of the instrument's interface whilst shifting focus toward the sound.

sound

In general, the circuit tends to produce rather static sounds and structures due to the constant electrical energy source. In this way, the circuit requires interaction in order to create change and movement in the sound. This tends to lead towards the creation of drones and a focus on manipulating the textural qualities of the sound.

The use of square waves and digital logic results in simple waveforms and I initially found myself mainly manipulating frequency of oscillators and sequences to create structures and transformations in the sound. After further exploration, more interesting and dynamic timbres were achieved using frequency modulation and feedback networks. From this, several modifications were made in order to create more movement and variation in the sound.

Firstly, four XOR gates were added to the circuit, creating an effect similar to ring modulation. The added harmonics produced by this chip create a range of rich textures and noisy sounds, especially when the four gates are chained sequentially. Their incorporation into feedback networks tends to create dense, rich layers of noise with more dynamic timbres.

interaction

By creating and removing layers of connections, this perceived density can be manipulated allowing the performer to create structural shifts and transitions within the music. More specifically, this can be achieved by controlling the relative harmonicity of the input signals to create movements between simple/smooth and complex/coarse texures. At lower frequencies this techniques also allows the performer to manipulate the sense of rhythmic stability.

In addition to this, the use of half connected wires and unstable ground connections amplifies the movement of cables and the performer's proximity to the circuit starts to influence the sound, creating a more engaging experience resulting in more dynamic sounds and timbres. In particular, I found that the noisy, unstable and unpredictable nature of this interface was particularly engaging and seemed to facilitate an uninterrupted flow of ideas whilst developing and interacting with the system.

In addition to this, the presence of loose wires, unstable grounding and the use of light dependent resistors within the circuit increased this feeling of interdependence and entanglement between myself, the circuit and its components. In this way, the unpredictability of the circuit is not arbitrary but intrinsically linked to it's environment. This gives the instrument a sense of presence and life as it reacts to the changes in its surroundings.

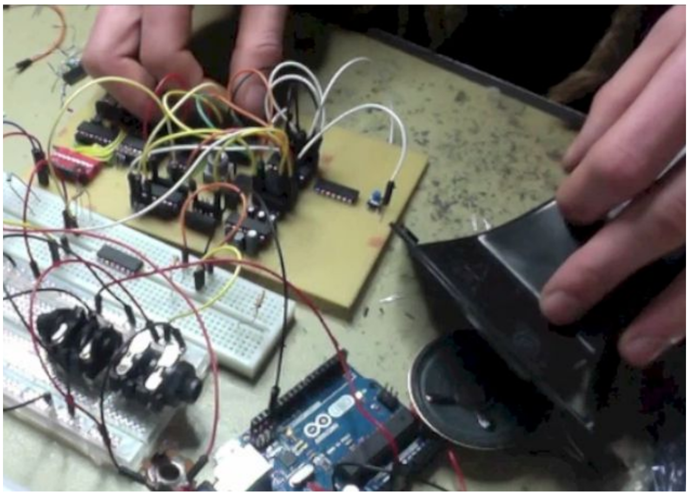


Fig. 15, Playing the CMOS modular synthesiser

Similarly, once a patch reaches a certain stage of complexity the significance of the individual connections, components and actions become difficult to navigate. At this point, the performer has to rely on intuition and awareness, reacting to changes spontaneously.

5.3 Transducers, Amplification and Resonance

Several amplifiers were built in order to bring electronic sounds into the acoustic domain, facilitating direct, physical manipulation of the sound and drawing attention to the spatial complexity and subtle textural details of acoustic sound sources. The sound is physically there in front of you and can be immediately manipulated intuitively with the body. In this way, the amplifier was found to be particularly effective when used in combination with SuperCollider.

Whilst exploring the capabilities of the circuit several found objects were used as resonators, each with their own harmonic properties. This was particularly interesting when combined with contact microphones and feedback, providing an extremely sensitive interface.

Fig. 16, Various found objects used as acoustic resonators

However, the generally unstable nature of this feedback system meant that I often found myself searching for *(and losing)* the resonant nodes of the objects and the thresholds of change between them. In addition to this, combinations of the different resonators can be stacked and sequenced, merging and contrasting the sound qualities of each. In this way, the collection of resonators provides a set of different intervals, determined by the physical properties and spatial positioning of the objects.

Smaller objects were also placed directly on the speaker membrane, creating a wide range of rattling, distorted sounds. Rich and complex spectral changes are often produced as the objects are moved around by the speaker. In particular, solder was used to create small structures which would dampen and accentuate different nodes of the speaker membrane. The use of these objects interferes with the natural modes of vibration of the speaker and resonators, producing rough sounding textures which complimented the pure tones produced by uninhibited feedback between the contact microphone and speaker.

Fig. 17, Exploring the interaction between the speaker and solder sculptures

Further experiments were carried out exploring various combinations of transducers, acoustic objects and DSP software. The patchbay circuit was placed in a temporary enclosure and used to control the routing of signals between these elements. Banana jacks were connected to several of the inputs and outputs of the patchbay facilitating the navigation of the interface due to their larger size and circular layout.

The placement of various objects between the input and output transducers allows the use the objects to filter the sound acoustically. By adjusting the positions of the transducers on the objects, different resonant modes can be excited. In this way, the transducer resembles a microscope, amplifying and revealing, allowing explorative listening to the internal structure and material qualities of the object.

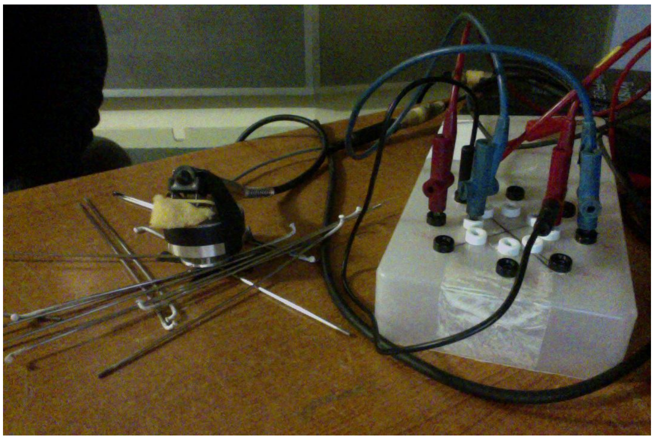


Fig. 18, Investigating the resonant structure of an arrangement of bicycle spokes

This process becomes particularly intriguing when exploring combinations and couplings between multiple objects, producing a wide range of rattling and buzzing sounds. More specifically, bicycle spokes were used to create temporary structures which could be explored by manipulating the distances, pressures and surface area between the transducers and objects. In addition to this, the constant vibration of these structures in combination with the multitude of parameters and forces involved leads to a certain unpredictability and constant evolution in the sound.

More specifically, these later experiments aimed to use the feedback tones created between transducers as raw sound material which can be dissected, processed and rearranged using SuperCollider. In addition to this, techniques for destabilising the dominant resonant tones were also explored, softening the threshold for sympathetic resonance and allowing the system to be controlled more easily.

The extreme sensitivity of this system to volume changes and the precision required to slowly introduce feedback tones can be particularly problematic and usually requires the performer to make constant adjustments to the volume. This was improved by the use of compression and a volume pedal with some mapping to soften the response curve.

A combination of delays and reverb were used to reduce the speed of the feedback process. This results in a more controllable system, providing the performer with more time to react to changes in the sound without losing the sensitivity provided by the acoustic interactions. Furthermore, this allows for a more detailed and contemplative listening to this iterative filtering process, as the initial sound slowly takes on the qualities of the object, creating a complex intermingling of objects and identities.

In addition to this, digital filters were added to the input signals, allowing the performer to selectively excite and explore the various different resonances of acoustic objects. The amplitude of the input signal was mapped as a modulation source for the filter's cutoff and resonance. This creates more complex interactions between the electronic and acoustic sounds, destabilising the dominant feedback tones as the resonant peak is always moving. This is particularly interesting as it produces short trails of feedback as the amplitude of a sound rises and decays.

The placement of additional loudspeakers on the same table as the amplified objects provided a richer and more engaging experience. This is partly due to the wider frequency range of these speakers but also because interactions with the objects and air surrounding the speakers start to influence the sound. Every subtle movement becomes amplified and significantly influences the sound. In this way, the amplification allows the exploration of sound as a physical phenomena, as vibrations in the surrounding space.

5.4 Software

The basic structure for the software is split into to two main functions. One creates and organises the busses and their connections whilst the other allows the modules, stored in external files, to be loaded, mapping the relevant inputs and outputs to the busses created by the patchbay. The use external files to store the modules increases the flexibility and organisation of the system, allowing for additions and modifications to be made easily, without having to navigate through the rest of the code.

In addition to this, the separation of the patchbay and module functions allows multiple instances to be used simultaneously and recursively. Thus providing control over multiple groups of modules and their connections.

The use of buffers within the software allows various sounds and shapes to be stored and manipulated. The data within these buffers can be recorded in realtime or generated using algorithmically. This process can take place in both realtime and non-realtime contexts, allowing varying amounts of preparation to take place before a performance. Similarly, the use of presets allows for mappings, materials and structural elements to be used as a starting point or loose structure for improvisation.

At various stages throughout the development process, the software was mapped to preexisting controllers in order to focus on the physical experience of playing and analyse the suitability of various decisions and predictions that had been made about the software and its mapping. This focused around developing an understanding of the spatial relationships between the bodies of the instrument and performer, whilst helping to determine the boundaries of what is physically possible whilst playing the instrument. Similarly, the limited number of physical parameters that were immediately available (without navigating through menus) helped to prioritise certain features and functionality.

In particular, decisions regarding the placement and distances between parameters were made to facilitate intuitive and fluid navigation of the instrument's interface. More specifically, parameters were grouped and arranged according to their function and frequency of use, reducing the distance between parameters that tend to be manipulated successively or simultaneously In this way, the resultant mapping and arrangement of parameters aims to allow the performer to react freely, encouraging simultaneity and multiplicity of actions. Thus allowing a more continuous flow of actions that are less influenced by the physical limitations of the human body and the short time frames available during a performance.

Modules

This section will discuss the functionality and potential of the various modules. A more detailed documentation of the parameters and modes for each module can be found in the appendix. This documentation resulted from an iterative process involving the identification of significant parameters and the grouping of similar functions. The resulting multi functionality reduces the spatial requirements of the interface whilst facilitating the exploration of different processes without altering the patch cables.

sampler

This module allows for the recording and playback of signals into buffers. The duration of the recording is defined using the record button. A trigger signal is sent from an output when the playhead reaches the end point of the buffer allowing the recorded section of material to be used as a basis for triggering and organising other events. This becomes particularly interesting when introducing techniques such as pulse dividers to create proportional relationships between events and durations.

In addition to this, the duration of the recording can be quantised using transients in the input signal or external triggers from other modules. This allows for the synchronisation of multiple sampler modules, increasing the possibilities for creating interdependent structures.

Furthermore, the recorded audio can be analysed and dissected into segments, providing perceptually relevant and intuitive control over the re-organisation of the material. These segments can then be played and manipulated using the voice module discussed below.

voice

This module mainly deals with the playback of buffered material. In comparison to the sampler module, the parameters provided focus around the manipulation of material with shorter durations. For example, this module allows for the use wavetable and granular playback techniques.

The voice module has three modes for external control, these allow for monophonic and polyphonic playback and the creation of grain clouds. These modes provide varying degrees of density and articulated control, fulfilling different roles within the music.External inputs are available for specifying the pitch centre value and its modulation, allowing the use of gestural interfaces and other modules as control sources. Similarly, modulation inputs for position and buffer selection are also provided.

This module also facilitates the creation of dynamic timbres through the interpolation and probabilistic selection of different sounds and waveforms. Parameters for centre position and deviation were used.This was found to be most effective when the various sound files are chosen and organised in a way that produces a desired range of timbres. Through this selection and organisation, a continuum between similarity and difference can be created, arranging the sounds according to perceptual qualities such as dark to light or low to high.

In addition to this, these buffers can be organised automatically by the software according to different characteristics such as duration, spectral centroid or timbral similarity.

envelope

This module acts as an envelope generator with parameters for manipulating and modulating its duration and the ratio between the attack and decay lengths. Parameters for curve and smoothing provide further possibilities for shaping the envelope. This is particularly useful when the envelope is triggered at audio rates as the resultant filtering allows the brightness and other timbral characteristics to be manipulated. In addition to this the module has three modes for defining the reaction to the modulation input. These modes allow for the envelope to be triggered once, sustained or continuously retriggered. A trigger is also sent out of a second output when the envelope reaches its end point allowing the chaining of events and sequences.

filter

The filter module allows spectral transformations to be made to a sound through the accentuation and attenuation of various frequencies. Different parameter states can be stored, recalled and interpolated allowing the sound to travel through the different filter spaces. The order in which these states are stored can be used to create a continuum which can be modulated using either trigger or continuous modulation signals. Trigger inputs can be used to step through these states sequentially or randomly. Continuous signals interpolate between the different states. There are also modulation inputs for frequency, resonance and blending between low pass and high pass filter modes. If no modulation input is specified then the amplitude of the filter's input signal can be used as the modulation source.

sequencer / delay

This module stores and recalls data in a buffer, acting as a sequencer or delay. The modulation inputs for recording and playback can be either discrete or continuous, allowing for sequential stepping or direct control of the playhead respectively. A modulation input is also available for reseting the playhead position, allowing for the creation less repetitive patterns by continuously changing the duration of the looped playback. Similarly, parameters are provided for manipulating the speed of playback, start position and feedback.

inputs and outputs

This module provides a series of undesignated inputs and outputs for use with external audio and gestural inputs. This general purpose module increases the flexibility of the system by allowing for future development and expansion.

Mapping Modules
These modules are focused around the mapping and transformation of signals, defining their routing and creating interactions between them.

analysis
The analysis module requires one input signal and gives outputs for onset detection, amplitude, noisiness and pitch centroid. This module can also detect and loop a series of onsets providing a simple method for the transplanting of rhythmic patterns between sounds. Parameters for adjusting the smoothing of these signals is also available, altering the immediacy of their response. These capabilities allow parameters to be mapped to the perceivable qualities of sound, increasing the possibilities for combining acoustic and electronic sounds and creating interdependent relationships between them.

switch
The switch module takes two input signals and uses a series of buttons to select which of four outputs the first input is sent to. This allows for the creation of diverging signal paths, sending signals to different inputs, spaces and processes. Fade times can also be specified for the opening and closing of each output, allowing for a variety of sharp and smooth transitions between destinations to be made. Several different modes allow incoming triggers at the second input to step through the selected outputs. The order in which the outputs are chosen can be either sequential, probabilistic or random. When using continuous signals as a modulation input the selected outputs can be interpolated according to the incoming value.

combine
This module provides various methods for combining two input signals facilitating the creation converging signal paths. More specifically, these two signals can be mixed, multiplied and crossfaded between. The mix or amplitude of the output signal is specified using either the slider or modulation input. If the external modulation source is selected then the slider scales the amplitude of the incoming modulation signal. In addition to this, several of these modules can be chained together, allowing for more flexible and interrelated routing and the creation of submixes and feedback.

compare
This module groups various different operations relating to the comparison of two input signals. From this, two outputs are generated, representing the relative state of the two inputs. In this way, conditions can be created in which moments of similarity and difference trigger events or alter signal paths.

The different modes of operation and their outputs are listed below:

	OUTPUT ONE:	OUTPUT TWO:
AND / NAND:	A & B	1 - (A & B)
OR / NOR:	A | B	1 - (A | B)
XOR / NXOR:	A xor B	1 - (A xor B)
COMPARE:	A > B	A < B
DIFFERENCE:	A – B	B - A
LOW / HIGH:	highest	lowest
CHANGE (amount of):	most	least

threshold

The threshold module provides three modes in which the amplitude of the input signal is modulated in relation to a *fixed* threshold. The first mode outputs a low or high value if the input signal is below or above the threshold respectively. The second mode gates the input signal if it's amplitude exceeds the threshold. The third mode acts as a compressor, reducing the amplitude of the input signal according to its distance above the threshold. In addition to this, an external modulation signal can be used to trigger the threshold facilitating the creation of interdependent interactions between the sounds.This can be used to generate a sense of movement, creating space in the spectrum as the sounds push and pull against each other.

div / gate / sah

This module groups similar functions that require two inputs and use triggers/clocks as a control source. These different functions allow the module to be used as a gate, pulse divider, sample and hold. There are also three different modes that determine the influence of external modulation signals. These modes are listed below:

trig**:** -modulation signal acts as external clock, providing single pulses
gate**:** -modulation signal gates an internal clock
tap**:** -modulation signal sets speed for internal clock

Similarly, a button on the front panel mirrors this behaviour, providing a more immediate source of control. In addition to this, a parameter for fade time facilitates the creation of envelopes and smooth transitions between speed changes.

mixer

This module acts as a matrix mixer with four inputs, four sends and a master section. The four sends allow signals to be patched back into other modules, facilitating the use of feedback. There is also a filter for each input and the master channel providing means for adjusting the spectral and dynamic balance between the different inputs. Furthermore, presets can be used to store, recall and interpolate between these different states of balance.

5.5 Hardware

Multiplexing

The modular system should be expandable, providing the flexibility for the system to grow and shrink in size, adapting to the current aesthetic and technical requirements of the performer. For this reason, a large number of inputs and outputs are required for the interface elements (sensors and LEDs). This was achieved using multiplexing techniques to expand the number of inputs and outputs available to and from the micro-controller.

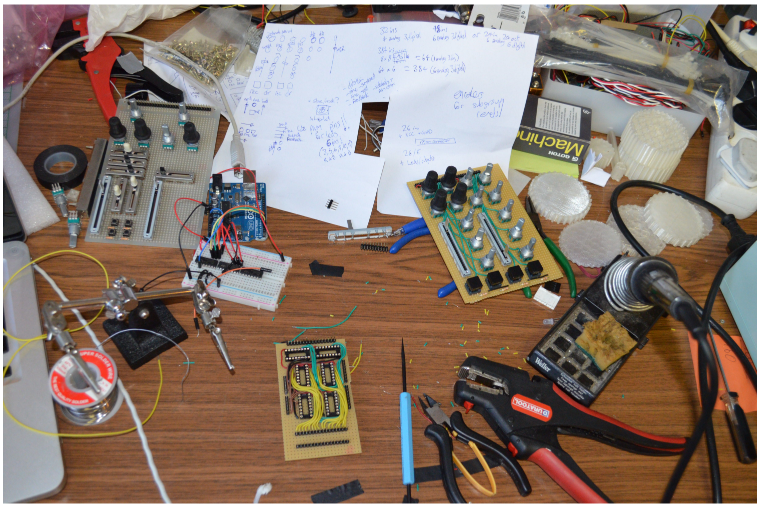
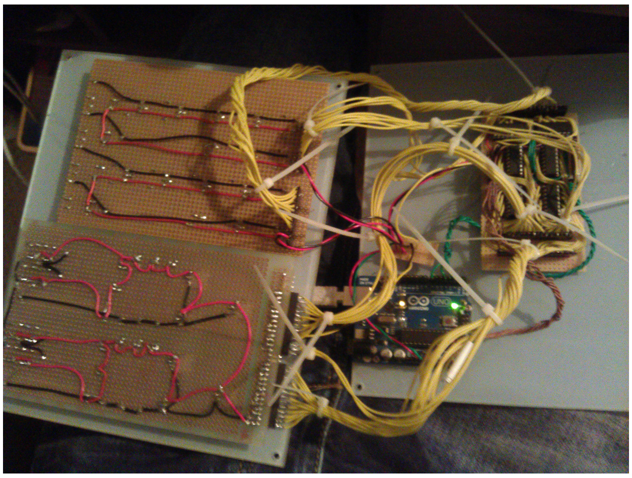


Fig. 19, Building the prototype for the mixer panel

Initially, the 74HC595 and 74HC165 integrated circuits were used to create a simple controller for exploring different layouts for the mixer channels. Additionally, this process acted as proof of concept for the structuring and layout of components for the modular. Allowing mistakes to be made and solutions to be developed for the later builds of the modular panels. For example, temporary cables had to be made due to time constraints. This resulted in various unstable connection and a realisation of the time required to solder the number of wires needed for the modular system.



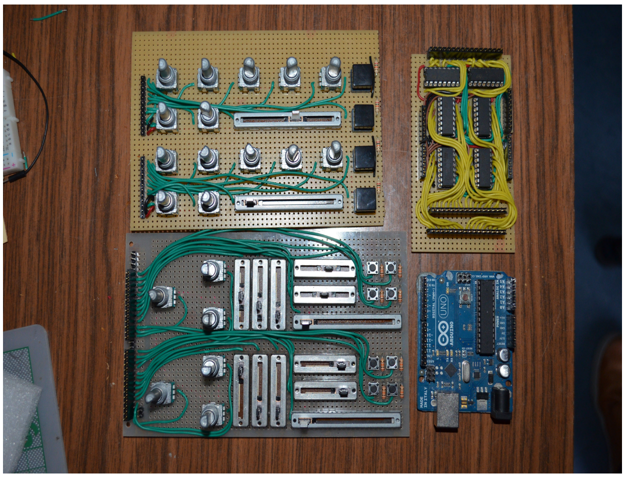


Fig. 20, Temporary cables connecting sensors to multiplexing circuit and Arduino micro-controller

Fig. 21, Finished circuits before making the connections between them

PCBs

For these reasons, PCB designs were created for the patchbay and multiplexing of inputs and outputs. This facilitates their reproduction and reduces the time consuming and repetitive nature of soldering the circuits by hand, allowing time to be spent on other aspects of the instrument. This is particularly useful if a board gets damaged and stops working then it can easily be replaced. Furthermore, the precise nature of machine manufacturing process reduces the chances of unstable connections. However, flexibility is lost making it more difficult to correct problems as the traces that connect the components cannot be altered as easily.

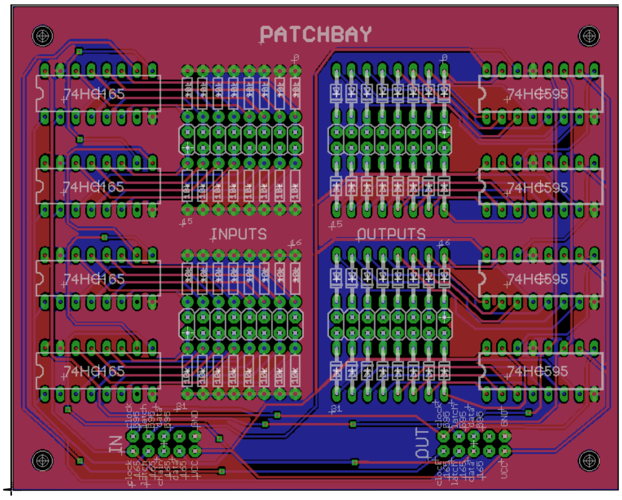
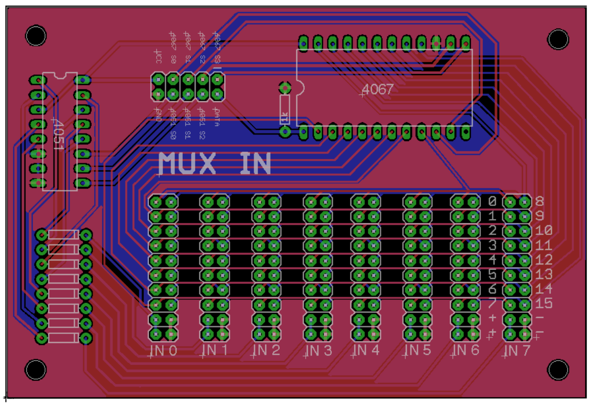


Fig. 22, PCB design for the patchbay circuit

The PCB designs aim to provide an interchangeable set of components for the internal structure of the instrument. This was achieved by using headers to allow the chaining of multiple boards together using ribbon cables to expand the system. Similar connections are also used to allow communication between the pcbs, panels and micro-controller.

This approach also allowed for decisions regarding power and routing to be made at a later stage of the development process. This is particularly advantageous as the power requirements will change in relation to the number of pcbs connected and the inputs and outputs that are being used.



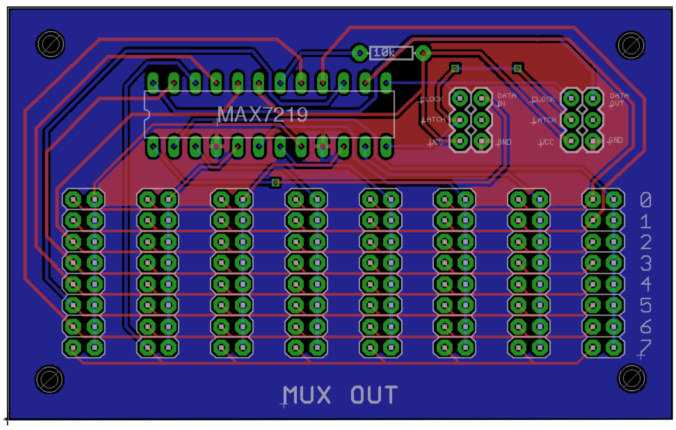


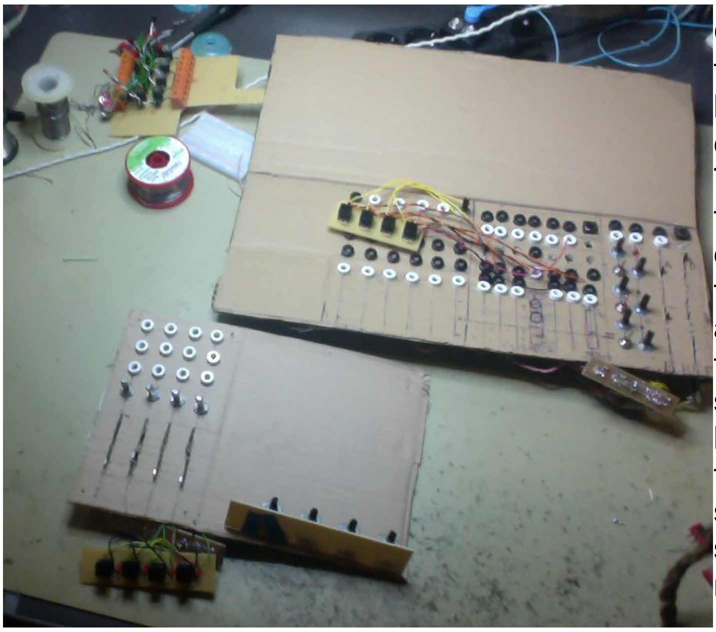
Fig. 23, PCB design for multiplexing outputs

Fig. 24, PCB design for multiplexing inputs

Panels

Initially, the layout for the panels were designed using pen and paper to identify the relevant parameters and their possible placements. This process revealed the possibility of using the same interface elements to control similar functions.

This led towards more generalised designs allowing for the same physical module to control multiple modules and modes in the software.This reduces the spatial requirements for the instrument whilst increases its flexibility, allowing for future modifications and extensions to be made to the software without needing to build new hardware. Although, the use of the pcbs and ribbon cables allows the panels to be easily removed and replaced if necessary.



Once decisions had been made about the significant parameters and their basic layout cardboard panels were created providing a tangible prototype. This allowed focus to be shifted towards the relationships between the components and my body. This helped to determine the required distances and adjacency of parameters. During this process it was decided that the size of the physical parameters should be related to their frequency of use and their perceptual significance on the sound. Furthermore, this temporary state allowed for modifications to be made quickly and easily.

Fig. 25, Building temporary panels using cardboard

Finally, more precise and durable designs were created using a combination of software and a laser cutter.

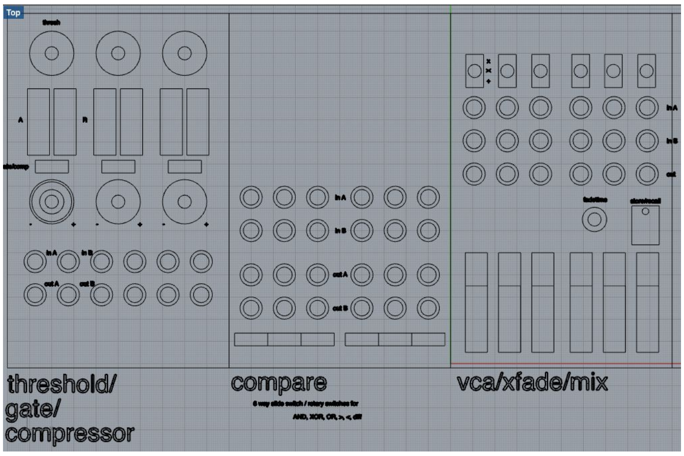


Fig. 26, Early panel designs for the laser cutter

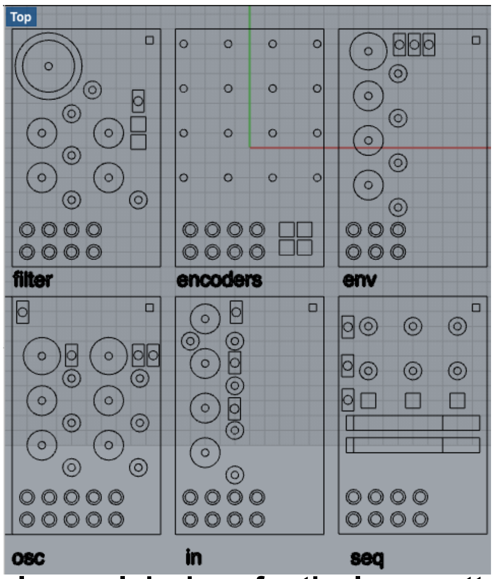


Fig. 27, Early panel designs for the laser cutter

REFLECTIONS AND CONCLUSIONS

Through the exploration of various approaches towards the realtime creation of music involving both electronic and acoustic technologies I have become increasingly aware of their characteristics and capabilities. More specifically, the affordances and constraints of the performer, instrument and environment influence the experience of interaction along with the possible sounds and structures that can be created. In this way, the research has attempted to develop a deeper understanding towards the use of these aspects as compositional parameters within a musical context.

The increased distance between sound and its causal action that often characterises electronic instruments reduces the influence of the instrument's physical properties on the sound and facilitates the control of multiple sounds and processes. However, this can also reduce their capacity for intuitive control and the articulation of subtle details and nuances in the sound. Similarly, the increased precision and reproducibility that electronic instruments afford can often result in a perceivable flatness in both the sound and interaction. In comparison to acoustic instruments, such technologies lack the inherent richness and unpredictability of interactions in the physical domain. Furthermore, the ambiguity and reciprocity that results from the interdependence between parameters, performer and environment is often what makes such interactions engaging.

The cello was chosen as a platform for exploring these concepts, expanding its sonic capabilities through the use of software. The integration of acoustic and digital interfaces reduces the physical and cognitive distance between the two, providing a less disruptive interaction whilst encouraging immediacy and intuitive actions.

The combination of analysis and interdependent mappings resulted in a more engaging playing experience and increased the reciprocity of the interactions with the instrument. Initially, these techniques were used in an attempt to determine the performer's intentions and create complimentary or contrasting reactions within the software. However due to the complex nature of this task (deriving meaning and context) combined with the primitive realisation of this idea made it much more interesting to explore the ambiguity generated by interactions in the physical domain and their translation into the digital. Similarly, the introduction of random and probabilistic functions providing control over the predictability of the instrument's response, allowing the performer to navigate through sections of varying determinacy and discontinuity.

The main disadvantage to the cello is that the reliance on the physical sound source limits the sonic range of the instrument. The range of sounds that can actually be articulated in an expressive manner is relatively small and focused. As a result of this, the resonances of the cello are often recognisable in the transformed sound. This could be improved by using more extreme processing techniques and incorporating additional transducers on different parts of the instrument's body. Thus providing a more diverse set of sounds and timbres.

Similarly, the large number of physical parameters that are required to achieve highly contrasting states makes it difficult to achieve fast and spontaneous transitions between them. This could be solved by adding extra switches for triggering changes to multiple parameters in the software. However, when these changes are triggered the positions of the hardware parameters no longer correlate to those in the software. For this reason, a decision was made against the use of this technique favouring consistency and immediacy.

In addition to this, the large size of the cello and the balance required whilst playing in the traditional position restricts the performer's movement. This, in combination with the inability to control multiple streams, sounds and processes led me to develop the hybrid modular instrument. Thus providing a more portable and flexible solution for the exploration of different combinations of sound and control sources.

The modular provides a flexible framework for the exploration and organisation of various different materials, gestures and contexts. The use of physical patch cables allows for multilayered, non-linear and interdependent mappings to be defined between these elements in an intuitive and immediate manner. Furthermore, these mappings can be used to create multidimensional and higher order controls that compliment gestural interaction.

Relationships and networks of varying complexity can be created between acoustic and electronic elements using transducers, analysis and feedback. Such techniques allow for structures to be created in relation to the materials being used, exploring their properties and internal structure through the use of various filtering and transformation processes.

In addition to this, the storing and recalling of sounds, shapes and mappings allows for varying amounts of preparation to take place before a performance. This stored information can also be generated during performance, recalling specific states of interest whilst creating sections and structure in a realtime and improvised manner.

The ability to specify which parameters are accessible from the hardware enables the creation of particular sets of affordances and constraints. Thus providing relevant parameters and functionality for the chosen materials and context. This separation between software and hardware interfaces reduces the temptation to continually alter the instrument's functionality and encouraging the exploration of the parameters that are immediately available. Similarly, the use of probabilistic functions, feedback and interdependent mappings can be used to experiment with the predictability of the instruments response, exploring and manipulating the perceived causality between action and perception.

The technical complications and the time constraints of this study limited the depth of understanding and fluency that could be achieved with regards to the sonic and performative potential of the hybrid modular. Therefore, future efforts will aim to further explore the capabilities of this instrument within different musical contexts.

Through this research I have developed tools and techniques that attempt to combine the ambiguity and sensitivity of acoustic interfaces with the precision and reproducibility of electronic technologies. More specifically, the affordances and constraints of the performer, instrument and environment provide a source of variety, engagement and inspiration. The resultant instruments implement computers and electronic technologies to enable a deeper exploration of acoustic phenomena by increasing the communication and reciprocity between the the two.

In addition to this, the construction of physical interfaces using sensors and micro-controllers reduces the conceptual and physical distances between the acoustic and electronic domains. Thus allowing for more immediate and intuitive interactions to take place. This process has highlighted the importance of the spatial configurations of the interface with relation to their functionality and the performer's body.
The process of designing and building these instruments was focused around the act of playing and their resultant sonic capabilities, using physical media and interaction to guide decisions and developments. In particular, it was found that extended periods of time spent interacting with the emerging instruments was necessary in order to adjust to the changes that have been made. More specifically, this process helped to clarify of the suitability of these changes.

In addition to this, the exploration of mapping as a compositional approach resulted in the use of multilayered, non-linear and interdependent mappings whilst attempting to provide variable control over the predictability of the instrument's response. This increased the reciprocity of interactions with the instrument, reducing the effects of habitualisation and inspiring new directions within the music.

Future directions for this research could involve experimentation with various combinations of acoustic object and sensors to provide a wider range of diverse sound sources and facilitate the development of new playing techniques. These hybrid interfaces should aim to further reduce the distance between acoustic and digital interfaces, allowing gestures to control both simultaneously and intuitively. Motors and other methods for allowing the computer to excite and manipulate acoustic sound sources could also be an interesting area to explore, especially when combined with transducers, analysis and feedback. Furthermore, the use of higher orders of analysis could be used to derive more meaningful information about the performer's intentions and the state of the instrument. For example, the various relationships between the parameters on the cello could be used to determine conditions that relate to different sound qualities and modes of playing.

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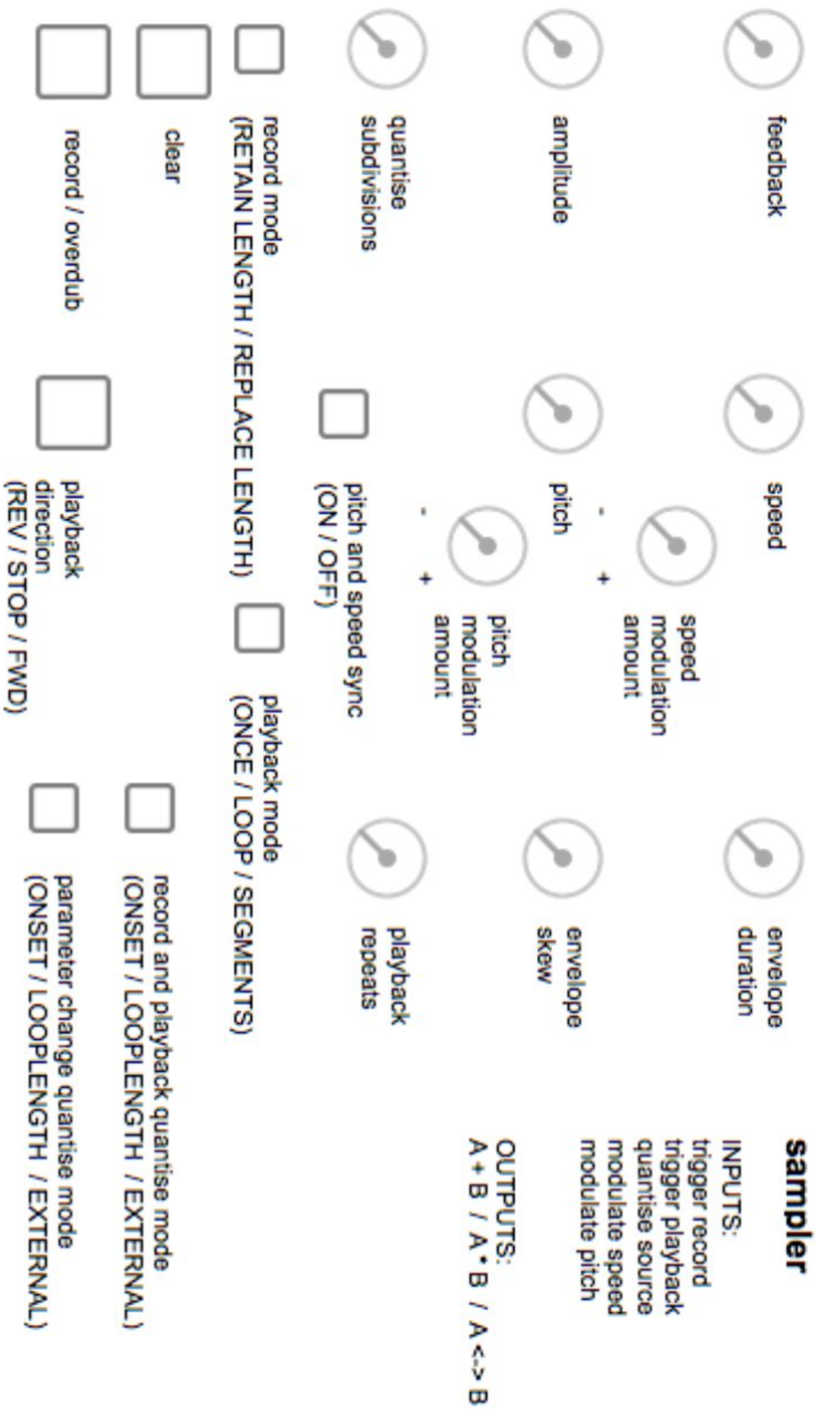
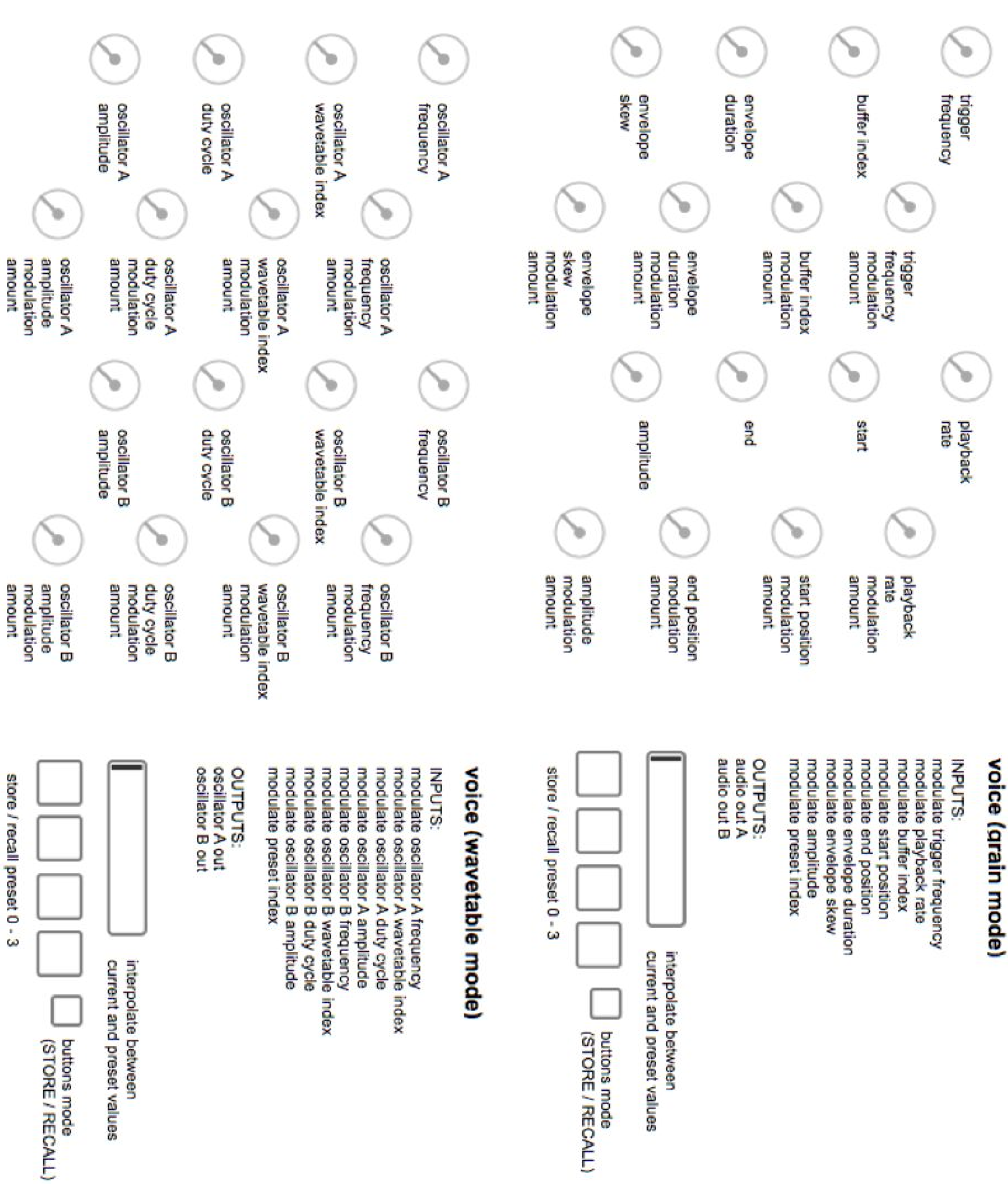
APPENDIX: module parameters and patch points

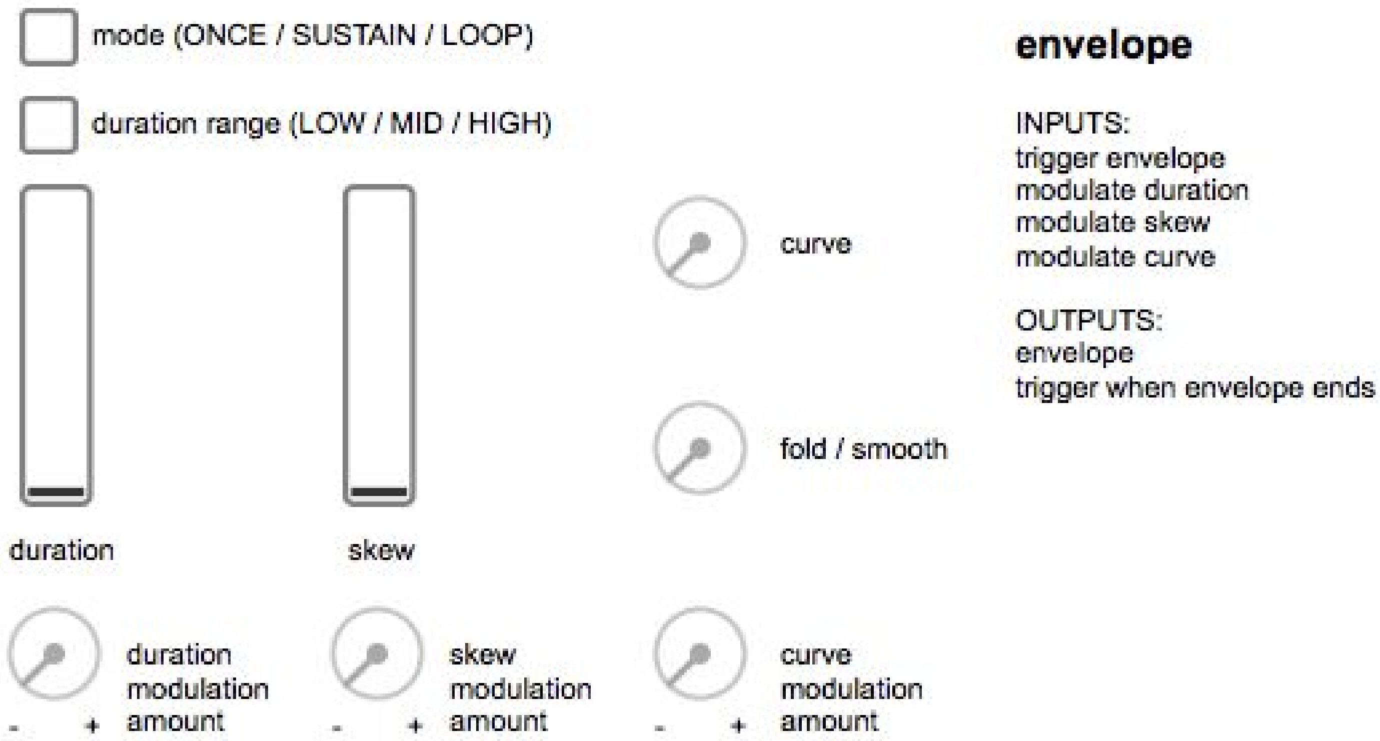
The following diagrams serve as a more legible version of the notes and sketches that were made during the design process for the hybrid modular instrument. This process involved the identification of significant parameters and the grouping of similar functionality. In particular, this process was focused around defining the possibilities for interaction between modules.These diagrams aim to document the results of this process, showing the decisions that were made regarding the various modes and mappings for each module.

Sampler

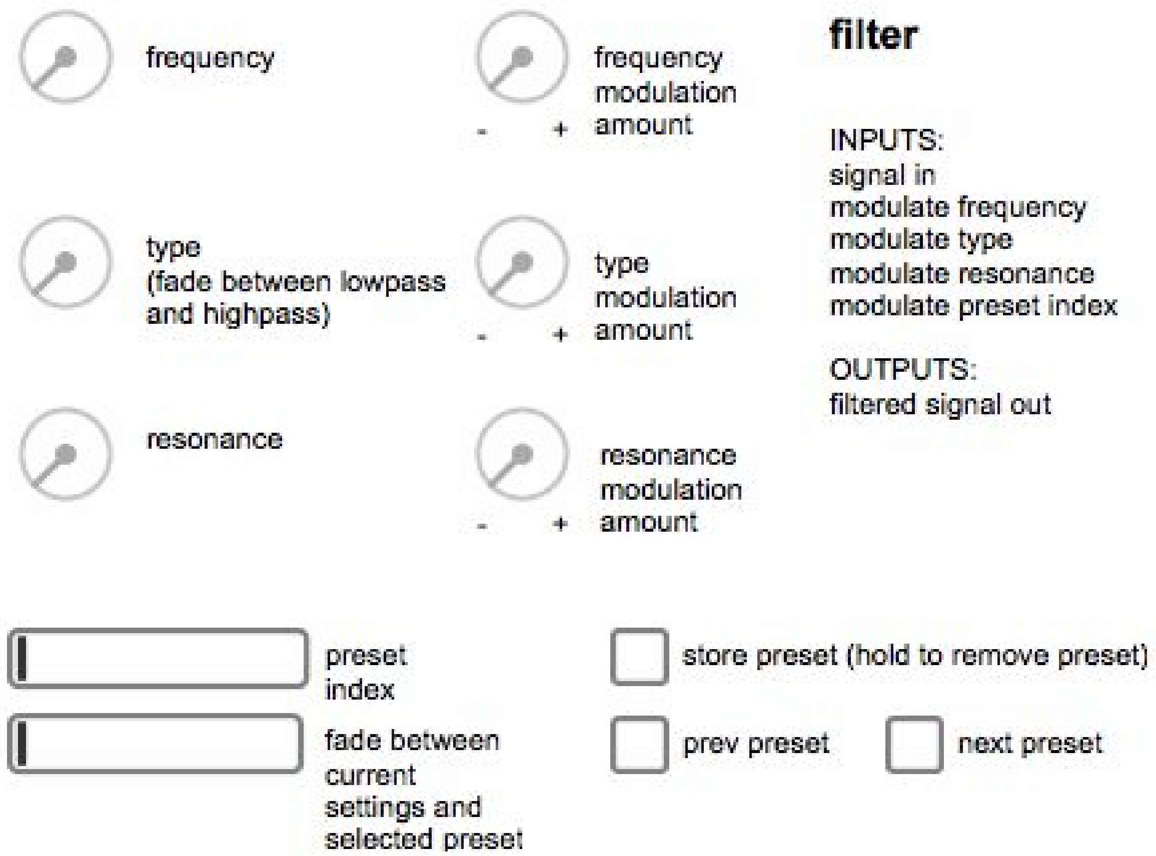



trigger frequency
trigger frequency modulation amount
buffer index
buffer index modulation amount
envelope duration
envelope duration modulation amount
envelope skew
envelope skew modulation amount
playback rate
playback rate modulation amount
start
start position modulation amount
end
end position modulation amount
amplitude
amplitude modulation amount
voice (grain mode)
INPUTS:
modulate trigger frequency
modulate playback rate
modulate buffer index
modulate start position
modulate end position
modulate envelope duration
modulate envelope skew
modulate amplitude
modulate preset index
OUTPUTS:
audio out A
audio out B
interpolate between current and preset values
buttons mode (STORE / RECALL)
store / recall preset 0 - 3
oscillator A frequency
oscillator A frequency modulation amount
oscillator A wavetable index
oscillator A wavetable index modulation amount
oscillator A duty cycle
oscillator A duty cycle modulation amount
oscillator A amplitude
oscillator A amplitude modulation amount
oscillator B frequency
oscillator B frequency modulation amount
oscillator B wavetable index
oscillator B wavetable index modulation amount
oscillator B duty cycle
oscillator B duty cycle modulation amount
oscillator B amplitude
oscillator B amplitude modulation amount
voice (wavetable mode)
INPUTS:
modulate oscillator A frequency
modulate oscillator A wavetable index
modulate oscillator A duty cycle
modulate oscillator A amplitude
modulate oscillator B frequency
modulate oscillator B wavetable index
modulate oscillator B duty cycle
modulate oscillator B amplitude
modulate preset index
OUTPUTS:
oscillator A out
oscillator B out
interpolate between current and preset values
buttons mode (STORE / RECALL)
store / recall preset 0 - 3

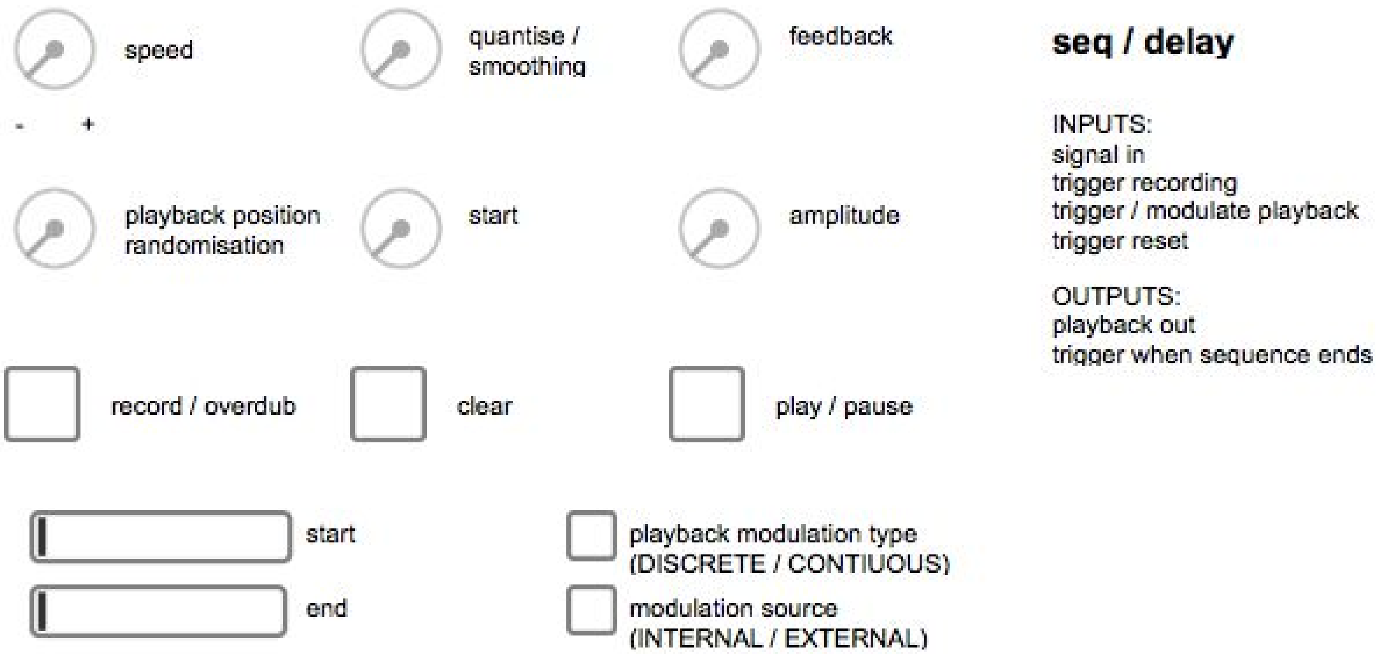
Envelope



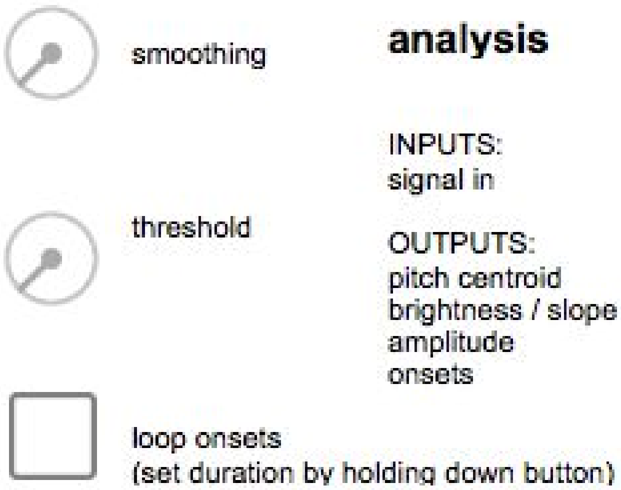
Filter



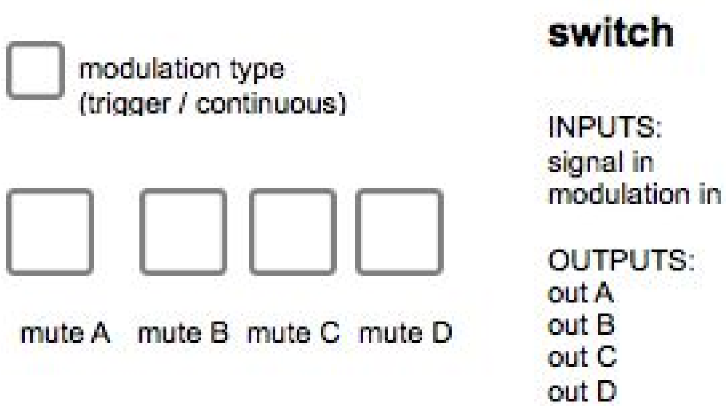
Sequencer / Delay



Analysis



Switch



Combine

combine

INPUTS:
signal in A
signal in B

OUTPUTS:
A + B / A * B / A <-> B

mix / amp

mode (MIX / VCA / XFADE)

Compare

compare

INPUTS:
signal in A
signal in B

OUTPUTS:
A & B / A | B / A xor B / A > B / A - B / highest / most changed
1 - (A & B) / 1 - (A | B) / 1 - (A xor B) / 1 - (A > B) / 1 - (A - B) / lowest / least changed

Threshold

mode (THRESH / GATE / COMP)

threshold / gate / comp

threshold

modulation
amount / ratio

- +

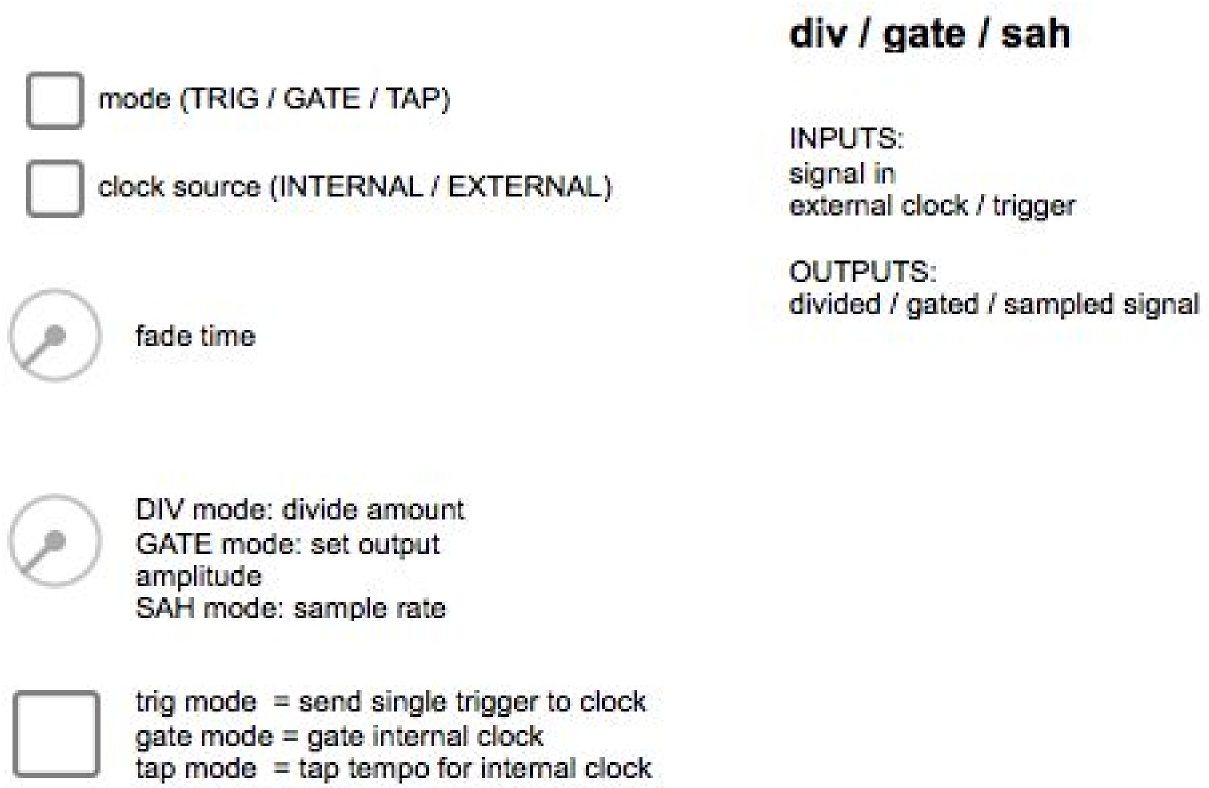
attack

decay

INPUTS:
signal in
threshold in

OUTPUTS:
above threshold / gated signal / compressed signal

Div / Gate / Sah


div / gate / sah
mode (TRIG / GATE / TAP)
clock source (INTERNAL / EXTERNAL)
INPUTS:
signal in
external clock / trigger
OUTPUTS:
divided / gated / sampled signal
fade time
DIV mode: divide amount
GATE mode: set output amplitude
SAH mode: sample rate
trig mode = send single trigger to clock
gate mode = gate internal clock
tap mode = tap tempo for internal clock