University of Birmingham

College of Arts and Law Graduate School

Music Department

Georgios Nikolopoulos, ID: [redacted]

Supervisor: Dr. Scott Wilson

Co-Supervisor: Dr. Christopher Haworth

Ph.D. research in Musical Composition

Theme:

A Collection of Musical Works Composed by the use of Computer Music Applications and Technology

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A thesis submitted to
The University of Birmingham
for the degree of
Doctor of Philosophy

2021

CONTENTS

1. Preface

This dissertation has been written to fulfill the graduation requirements of the PhD research in Musical Composition Program at the University of Birmingham. The whole project is based on interdisciplinary research and the music in this portfolio can be classed as electroacoustic music, drawing as it does on sound synthesis, physical modeling, real-time audio, interactive music and improvisation, ambisonics, and orchestral music for acoustic instruments with tape. The music works *Paradoxes* are inspired by postmodern philosophy, and particularly the work of Gilles Deleuze *The Logic of Sense* (1969). The introduction discusses the basic concepts of the *Deleuzean* philosophy related to my music work and illustrates some of my compositional influences, their thought and some of my own concepts as well. The *Six Paradoxes* analysis follows an analytical listening approach with spectrograms and offers some insight to my compositional process. The *Machine Learning Study* focuses on interactive music and improvisation, while *Formulations* is a fixed media work with reactive visuals. The next composition, *Turmoil,* consists of two versions, one is the electroacoustic version and the other one is music composed for acoustic instruments and tape exploring further the idea of the continuous change between real and unreal sounds. Lastly, my fixed media acousmatic work *Fossils* deals with the idea of structural duality and is based on interactive inputs. The portfolio totals approximately 95 minutes (table 1.1).

All of the music works presented henceforth have been realized with the valuable access to the electroacoustic music studios of the University of Birmingham. A version of the *Paradox I – III* was selected for the International Computer Music Conference (ICMC) 2017. During my research time another two works, which are not included in this portfolio, were selected and premiered at the ICMC 2016 and the Forum Wallis Ars Electronica 2018.

On the last chapter of my thesis there is an extensive list of my music compositions, projects, examples and other related to my research materials that can be found on the accompanying digital storage media (USB flash drive) or the following link: https://beardatashare.bham.ac.uk/getlink/fiQzc8SkTjDQo96C9d8gvToy/

Finally, I would like to thank my head supervisor and co-supervisor for their time, guidance and support during researching, composing and writing all these years (2016 -2020).

List of Works	Duration
Paradoxes I – III	9 min., 41 sec.
Paradox IV	10 min., 32 sec.
Paradox V	12 min., 44 sec.
Paradox VI	9 min., 39 sec.
Machine Learning Study (ML, Arduino and Sensors)	9 min., 51 sec.
Formulations (audiovisual)	10 min., 06 sec.
Turmoil I (electroacoustic version)	11 min., 20 sec.
Turmoil II (for Strings, Percussion and Tape)	10 min., 31 sec.
Fossils (fixed media acousmatic work)	9 min., 11 sec.

Table 1.1: List of Musical Compositions

2. Introduction

2.1 A Meta-Deleuzian Approach to Musical Composition

Paradox I – VI is a series of electroacoustic music compositions that attempt to examine the relations, if any, between real life, art and philosophy. The Machine Learning Study, Formulations, Turmoil I and II, and Fossils are individual musical compositions which incorporate a variety of compositional methods and thoughts. The conceptual influences that were decisive for the creation of these music works derived initially from Deleuze's work *The Logic of Sense* (1969) and secondarily his work *Difference and Repetition* (1968). *The Logic of Sense* presents a series of paradoxes exploring the relation between sense and non-sense, the bidirectional sense of paradoxes and introduces Deleuze's philosophical theory with regard to event and becoming (Deleuze 1990). Although within the context of the Deleuzean theory, a music stylistic approach closer to composers of New York school and experimental music such as John Cage is to be expected, efforts have been made to keep a specific distance (more details about the concept of distance below) from the particular movement they represent regarding techniques and aesthetics as well. Similarly, my approach to musical composition is a long way from being influenced by Deleuze's political theory. Using Deleuze's philosophical theory and the paradoxes, which were not originated by him, is not an attempt to legitimize any of my musical compositions but it works more as a reference and basis for the construction of a prime musical idea that is used for my *Six Paradoxes*. Undeniably, there are many composers who have been inspired by Deleuze and have conceptualized his philosophical ideas into their musical compositions in unique ways such as Antoine Bonnet and Pascale Critton.

However, in my case, the possible interrelationship between music composition and Deleuze's ideas – specifically, the ideas of distance as a plurality of distances, and time as a plurality of times – has been sought on a basis of an artistic expression combined with audio programming and digital sound processing experimentation. Additionally, I have been particularly interested in applying Deleuze's ideas about of paradox to the paradox of the creation process itself, particularly in the sense that "creative practices must be both non-deterministic and somehow controllable or predictable" (Thornton 2015). From the point of view of a composer, who is experimenting with sound utilizing creative coding, the concept of duality plays a constructive role in my work. The *Six Paradoxes* and the rest of my musical works capitalize on the dualities of deterministic and non-deterministic, synthetic and concrete sounds, slow and fast paced sonic figures or events, distant and proximate sound sources in the stereo field or space (in the case of a multichannel setup). There are extensive influences of composers such as the pioneers of computer music John Chowning (1934) and particularly his works Stria (1988) and Turenas (1988), and Jean-Claude Risset (1938 – 2016) and his works Mutations (1969), Sud (1985) and Elementa (2001) in terms of how they use textural layers of computer synthesized sounds. Furthermore, Curtis Road's (1951) work *Point Line Cloud* (1999 -2003) was another important influence in terms of the computer synthesis techniques he used for these microsonic music works and aesthetics as well. The *Point Line Cloud* deals with the multidimensionality of musical time where sonic particles, metallic and submerged soundscapes are of primary importance (Robindoré 2004).

For the next chapters I follow a twofold model for the analysis of my musical compositions. The first is a listening analysis approach which is based on the listening experience and is supported by spectrograms. The second model is a more objective approach which analyzes the compositional approaches and materials being used for my music works. Regarding my acousmatic thought process I can specify that it has been fundamentally influenced by Pierre Schaeffer's (1910 – 1995) four quadrants model where: 1) the perception of a sound event, 2) listening to the sound with a focus on the physical objects and actions that produce it, 3) sound's spectral and morphological characteristics and 4) our responses to organized sounds within a musical context are the four basic distinct modes (Young 1996).

Recognition of sound sources and "reality" in electroacoustic music was a central reference point for composers such as Trevor Wishart (1946) and Denis Smalley (1946). Trevor Wishart particularly pointed to a distinction between recognition of objects and recognition of spatial contexts consisting of combinations of real-unreal objects and real-unreal spaces (Young 1996). While I was developing my own compositional thought I extended all these concepts and created an acousmatic space of thought that is three-dimensional and could be depicted in the following figure (Fig. 2.1). The X axis refers to sonic patterns and figures where minus is the place of irregularity and randomness and the positive side is the place of regularity and periodicity. The Y axis refers to real and unreal sound sources or objects where the negative side is the space of unreal sounds and the positive side is the place of real and easily recognizable sounds. Finally, the Z axis illustrates the real-unreal (positive to negative) space which can be significantly altered with the various computer music techniques that are available today.

Another important difference between my compositional influences and my own approach is that I don't focus on the concept of transitions or transformations in the short run inside the three-dimensional plane, but I mainly follow the idea of a constant movement in the conceptual space. Apart from the duality of things, bi-directionality is an aspect of my thought closely interconnected with Deleuze's philosophy that explores the structural basis of language through the paradoxes. That means in the three-dimensional conceptual space shown in the figure below

(Fig. 2.1) we don't have a static point or points but we have a constant movement in the search of a transcendent point where real and unreal unite. Indeed, this concept is an extension of preexisting ones such as the reality-abstraction continuum compositional idea which has been already used in electroacoustic music and is more apparent in Bernard Parmegiani's work *La Création du Monde* (1982 – 84) (Young 1996).

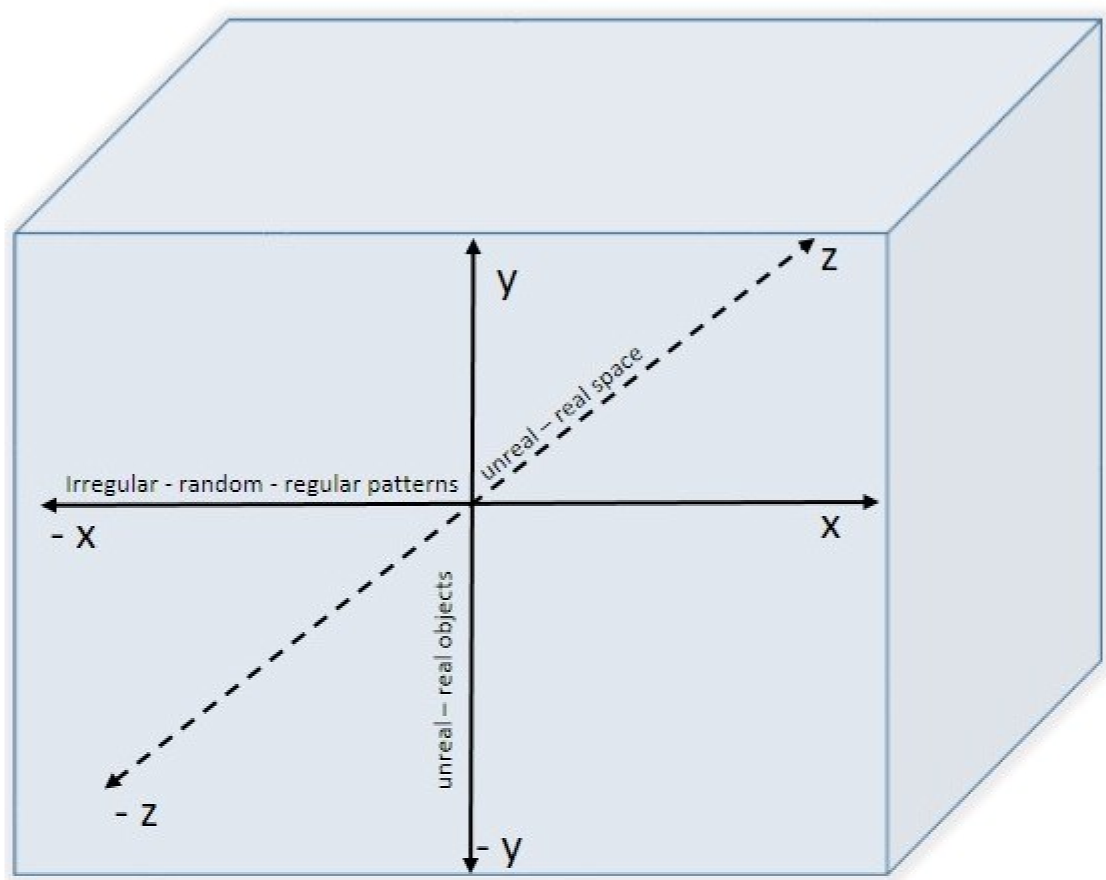

Fig. 2.1: The three-dimensional plane that brings together the basic concepts of my compositional process.

In electroacoustic music works that make use of real world sounds the element of metaphor and the associations each real sound source can create are of great importance in the ways we perceive these musical compositions. Adkins in his article *Metaphor, Abstraction and Temporality* *in Electroacoustic Music* goes a step beyond and makes a distinction between musical metaphor and metonymy; "Both Michael Bridger and Denis Smalley have further identified the tendency of listeners to electronic sound to search for an implied physical source that produced the sound and therefore derive mental imagery that is directly associative rather than metaphorical as in traditional instrumental musical expressivity." (Adkins 2010, 2). Inevitably, the concepts of metaphor, metonymy and extrinsic connections embrace very well the concept of distance. As the composer Smalley has pointed out, these sounds are remote surrogates (Adkins 2010). Consequently, in my mind, an artwork of electroacoustic music composed of these sounds is to some extent estranged from the environment that these sounds were initially placed or found. Therefore, thinking again in my three-dimensional plane, I could say that the distance between us and an artwork becomes greater as we deal with more recognizable sounds or things (representation), while the distance is reduced when we move to the unreal, negative axis.

The initial concept of distance, which is based on Deleuze's theory, refers to a constructive distance that provides the means for a better understanding of the past by giving new insights and different readings of the already existed or created, and paves new ways to explore the present and the future. While the modern societies have crossed the age of the mass mechanical reproduction of the artworks (Benjamin 1968) and are well in the age of information characterized by an "information overload" (Gergen 2000), art couldn't remain unchanged. In the context of this new environment, the constructive distance as I mean it comes to play an important role in the creation process of my music work from a practical and aesthetic point of view. Distance has the meaning of looking at the whole at first, the general structure of the things, before focusing on any details. For example, by conceiving the whole structure of my *Paradoxes* and their basic elements helped me to eliminate any minor distractions while I was experimenting with sound and audio programming techniques. By distance I also mean the difference between the subject of observation, which is the sound itself, and the observer or listener, who has the potential of different perspective views over a zoom-in and out process. A constructive distance offers a deeper and wider understanding of the past and subsequently the history. It is pertinent to mention here that although I never used traditional harmony, rhythm, counterpoint and instrumentation in my musical compositions, all these fundamental elements of music are not neglected and are still present in each of my works. Furthermore, the distance could be conceptualized as the *Deleuzean virtual*, which is “the condition of genesis of real experience” (Smith and Protevi 2020). In that sense, any creation process contains the stage of distance, which is the step just before the actualization of the composer’s initial idea.

In *becoming*, I have been interested in the idea of “at least two systems come together to form an emergent system or assemblage” (Smith and Protevi 2020). The newly formed system will have “its own ontological status” (Smith and Protevi 2020), although it will be composed of different entities. Furthermore, *becoming* represents constant change and “consists of an unlimited movement” (Lundborg 2009, 3).The dimension of sense as the boundary between propositions and things (Deleuze 1990) is another important point in Deleuze’s philosophy and a basis of the development of his paradoxes. Deleuze’s paradoxes in *The Logic of Sense* have a mixture of references to analytical philosophy, the Stoics and Lewis Carroll, and all of them derive from the structure of sense. Deleuze argues that the focus should be on the genesis of sense or truth instead of its conditions. Similarly, focus should be given on the genesis of sound instead of its conditions, the main ingredients or elements of music. Thus white noise can better express the idea of sound because it contains all the frequencies and “relations destined to be actualized” (Smith and Protevi

2020). We can extract any single frequency or combinations of frequencies we want from the spectrum of white noise.

My commentary isn't going to focus on a definition of noise music, but on the usage of noise itself as a structural element. As Ikeshiro puts it, "defining noise as a genre of music, which is itself open to debate, is as arduous a task as defining music. The issue is compounded by the fact that the delineated category appears to be contradictory by satisfying the conditions of both noise and music." (Ikeshiro 2011). Being based on the two contradictory statements of a paradox, this series of music compositions (Paradoxes I – VI) aims bring out the contradiction between different degrees of noise spectra and their extracted components, or musical instrument sounds as derivatives of noise. The two entities coexist creating a system that is characterized by heterogeneity. The sixth paradox with its counterpoint texture between noise-synthetic sounds and physically modeled acoustic instruments presents this approach in the most apparent ways.

Another concept which is of interest here is the concept of events and how Deleuze distinguishes between bodies and events. Deleuze follows the Stoics "whereas bodies and their qualities are material items which can exist in the present, events are immaterial and unpresentable" (Cook 1998, 28). According to the Stoics' philosophy, events have "a quite different relation to language" (Cook 1998, 28) than corporeal things do. The exploration of a similar correlation of sound with language can be misleading because we cannot simply extract the sound from states of things. Nevertheless, we could probably extract sonic events from states of sound. The sonic events, released from traditional forms and practices of musical composition, have the potential to involve multiple processes of change and transformation without having a predetermined beginning and end. A sonic event can follow various directions or can be distinguished by lack of directions and has the potential to escape "the idea of a predetermined goal and stays open to different forms of creations and transformations" (Lundborg 2009, 4). Hence, in the appropriate musical context and environment, the sonic events can be repeated as many times without remaining the same. "Ideas about what has happened might be repeated an endless number of times. But these repetitions can neither be cemented, nor result in a static conception of one and the same "thing" " (Lundborg 2009, 6). There is a clear distinction from musical minimalism though, because here we don't have an exploitation of minimal musical materials, but rather the engagement of a complexity of minimal musical materials. It could be described as a symbiotic system of heterogeneous musical materials. The infinitesimal moment of these sonic events in time eludes the present, "because every instant subdivides the present into past and future" (Lundborg 2009, 2). On the other hand, events can be perceived as a succession of instants, where past and future can be seen as parts of the present. Of particular importance is the double structure of events that are the basis of a different approach to the creation of the sonic events as extractions from the states of sound. "Chronos" and "Aion" are the two main conceptions of time according to the Stoics, "Whereas Chronos consists of the present as the constitutive element of time, Aion can be said to escape the present and only let movements of past and future remain." (Lundborg 2009, 2).

There is no a single interpretation of the sonic events that appear in my works and how they occur over time, however, they open possibilities for multiple 'listenings' or interpretations. They don't have or produce directions, rhythms and melodies in a musical sense, but they are based on the *Deleuzean* concepts of difference and repetition. Deleuze "-posits a repetition, no longer subject to identity and sameness, but rather to difference and variation, and which, he suggests, is best exemplified in Nietzsche's notion of the eternal return" (Campbell 2013, 9).

In the following chapters an analysis of my music works will be presented explaining more about their main ideas, concepts and techniques used for their realization.

2.2 Compositional Vocabulary

The purpose of this section is to facilitate our understanding of the compositional vocabulary descended from Deleuze's philosophical ideas and how this vocabulary is related to my musical compositions.

Dualities. This term is used in my works to indicate two contrasting elements which coexist; the deterministic and non-deterministic, the synthetic and concrete sounds, slow and fast paced sonic figures or events, distant and proximate sound sources in the stereo field or space (in the case of a multichannel setup). It can also refer to contrasting structural sections of a musical composition.

Becoming. This term represents the constant change which is characterized by bi-directionality, the oscillating movement between real and unreal, realistic and imaginary, recognizable and abstract. As a structural element has been used in a way that connects past and present; by altering the properties of the past and putting them in a new context, we alter also the properties of the present and that particular assemblage is what I have defined as *becoming* in the environment of sounds.

Boundary. It is the reference point between the two contrasting entities of real and abstract where very few things (or almost nothing) happen.

Sonic Events. This term is used to describe sonic figures that carry out multiple processes of change and transformation in long run without having a predetermined beginning and end. These sonic figures incorporate the *becoming* feature.

Fragmented Events. This term refers to the *sonic events* discussed before with the difference that *fragmented events* have undergone a process which makes them irregular with a much shorter duration than the initial sonic events.

Fragmented Sum. It refers to a situation of a sonic dissolution where materials from the past will be processed mainly by various types of granulators and then will be placed together in layers in order to from a new section.

3. Six Paradoxes

3.1 Paradox I – III

Paradoxes I – III are the first three interrelated and interconnected sections of an electroacoustic music composition that encompasses six paradoxes. Paradox I - III consists of three stages: 1) *Contradiction*, where the overlapping layers and a "static synthetic voice" contradict each other, 2) *Collision*, where the coherence of the previous status starts breaking into fragments leading to a more abstract situation and 3) *Dissolution*, where the sound material from the first two parts disappears into itself. Eventually, there is a small 'coda' which returns us back to a transformation of the first introductory part.

The most important structural elements of this series of music compositions are the spectrum of the sound and the *Deleuzean* events. For all of the six paradoxes, I chose not to use systematic harmony, rhythmic patterns and structures in order to achieve a cohesive whole in combination with a high level of unpredictability and non-standardization. The spectrogram of the first part shown in the picture below (Fig. 3.1) is the contradiction between overlapping layers, the static synthetic voice and other events that although they seemingly repeat themselves, are never the same.

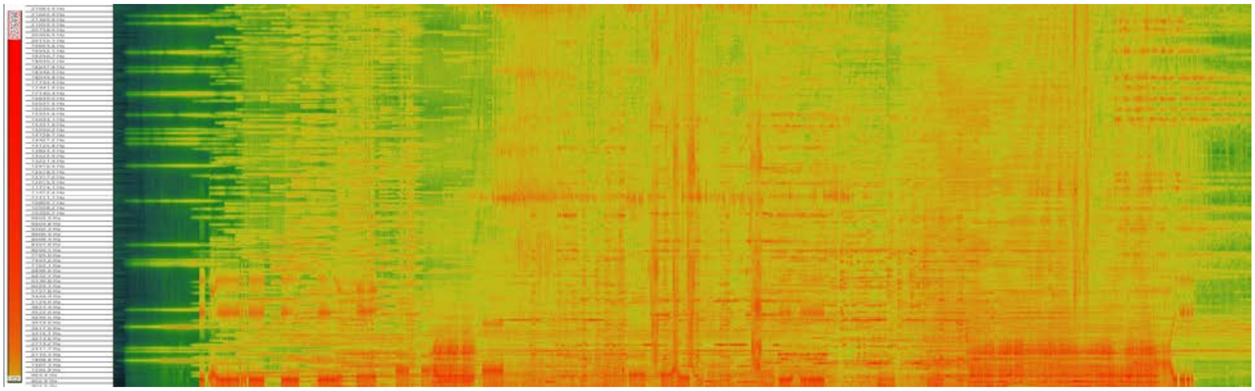

Fig. 3.1: Paradox I: Contradiction (approximately 4 minutes).

There is no before, present and after and especially there is no a sense of direction that the traditional tools of tonic harmony can ensure through the use of harmonic progressions and rhythmical figures. However, since music is by its nature an art that evolves mainly in time, it is particularly challenging to completely erase or shrink the differentiation between before and after (Deleuze 1990). The synthetic vowels pronounced on a static pitch and the general slowness of the piece come to play exactly this role. There is nothing to be said (by the synthetic voice) and there is nothing to be expected. Either listening to the next vocal fragment or to a previous one will put listeners in an identical situation. It is the same without repeating the same. Random vowel orders and random fragmented patterns don't come from anywhere and are not headed to anywhere as well. On the other hand, the overlapping layers spread out on a wide range of frequencies and have a movement characterized by slowness. The direction these overlapping layers have is determined somehow by the intensity of the sound, the degree of resonance and a variable spectrum density which is extracted by the white noise (FFT processing). The escalation of tension towards the end of the first part comes to intensify the contradiction and creates a 'bridge' to the next part.

A spectrogram of the next part shown below (Fig. 3.2) depicts the fragmented events that due to their inharmonic spectrum sound similar to metallic objects. These metallic objects are getting different dimensions every time and each of them have unique properties. However, the assemblage of different metallic objects creates a new unity, which is characterized by heterogeneity, and alters their properties in their entirety.

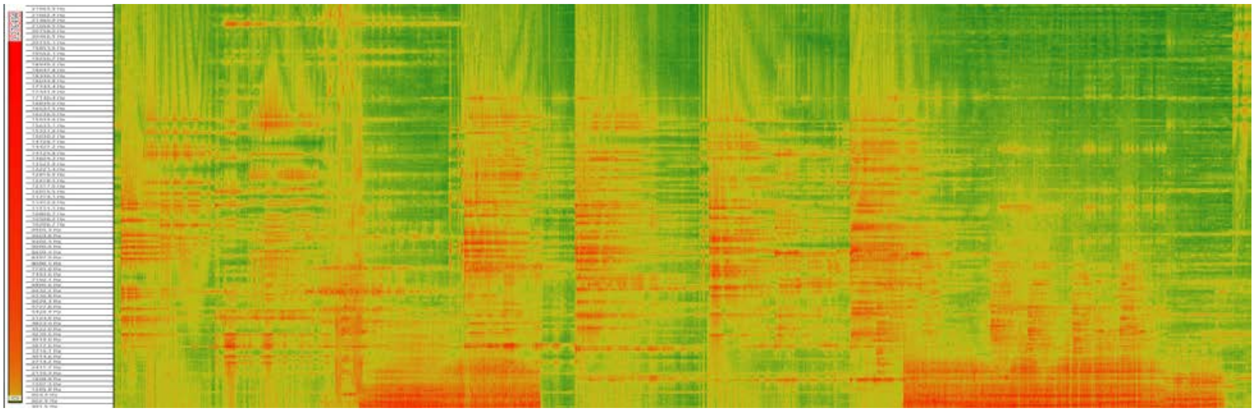

Fig. 3.2: Paradox II: *Collision* (approximately 2 minutes).

The subtitle, *collision*, doesn't necessarily indicate that the current part represents superficially a particular situation neither is the result of the foregoing contradictory events. The main characteristic of this second part is that the inharmonicity starts increasing due to the appearance of fragmented metallic sounds, while the spectral variance starts decreasing. This is illustrated in the next figure (Fig. 3.3), where we can also see how loudness, spectral variance and inharmonicity are closely related to each other throughout the whole duration of the piece. Another characteristic is the irregularity of the fragmented events and their contrast to low frequency passages.

Moving on to the second section, the overlapping layers of the first part now are splitting into pieces and become irregularly bouncing fragments. The physical models of these metallic sound events change density over time, they always become something different, and their properties are changing but simultaneously remain the same irregularly bouncing metallic bodies. These fragmented events are drawn into each other forming high-pitched long tails that are audible along this part and have also been used as a 'transition' to the next section.

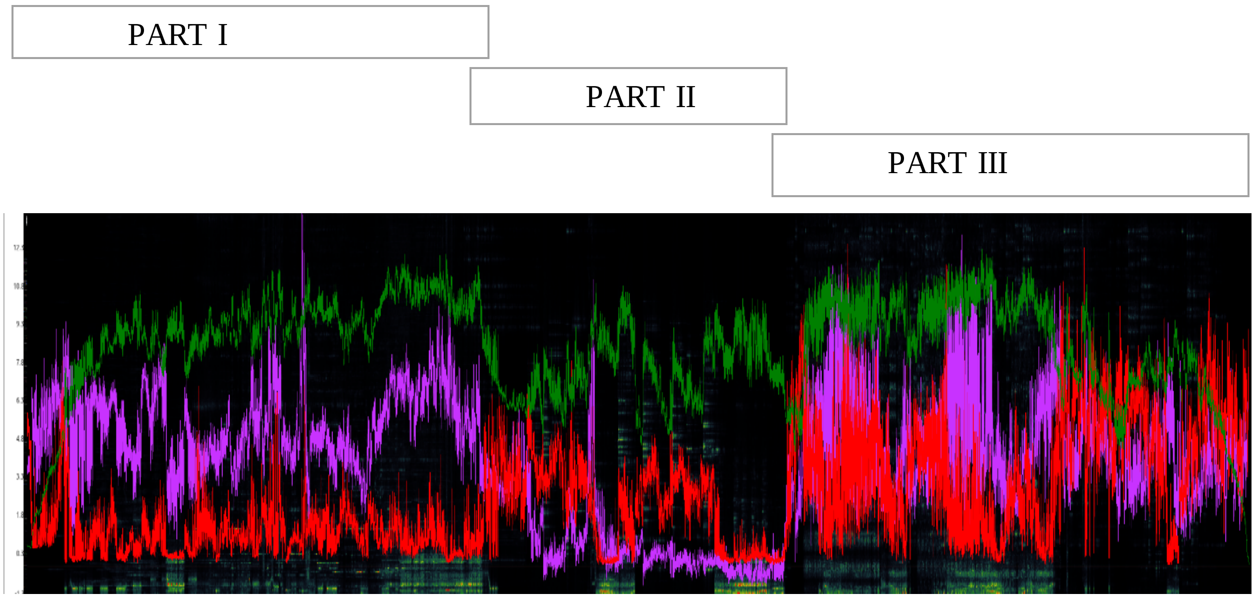

Fig. 3.3: Paradox I – III. ■ : Loudness ■: Spectral Variance ■: Inharmonicity

In this last section the degree of fragmentation increases gradually until we reach a total dissolution of the same sound material used before. The sound material disappears into itself and and submerges in its fragmented materials through a multifaceted granulation processing. The overlapping layers now have become granulated layers and that is depicted on the spectrogram of the next page (Fig. 3.4). Inharmonicity, spectral variance and loudness reach their limits and as shown in the third figure above, they are merged to a greater extent than in the first two parts. The main characteristic of this part is the density of the sonic events, which this time occupy a wide range of the sound spectrum. There is no direction in the sense that musical phrases or harmonic structures succeed each other either in a random or deterministic way. There is no actual repetition of the musical entities or events, however, there is an unlimited movement related to the idea of *becoming* as mentioned in the previous introductory chapter. The sonic event described here is characterized by multidimensionality and escapes “the idea of a predetermined goal and stays open to different forms of creations and transformations” (Lundborg 2009, 4). The presence of this contradiction aims to stretch the boundaries between becoming and unbecoming, past and future and has an ultimate goal to dismiss the present (Deleuze 1990).

The coda section of the last minute returns us back to the overlapping layers and the static synthetic voice of the first part. However, since the use of the term coda is not appropriate within this context it would be better to describe it as a fragmented sum of the previous sections. The descriptive words -fragmented sum- can illustrate better the conclusion where the stutter effect is applied on the static synthetic voice , while at the same time the diluted high frequency layers, which are the debris of the metallic bouncing sounds of the second part, are fading out.

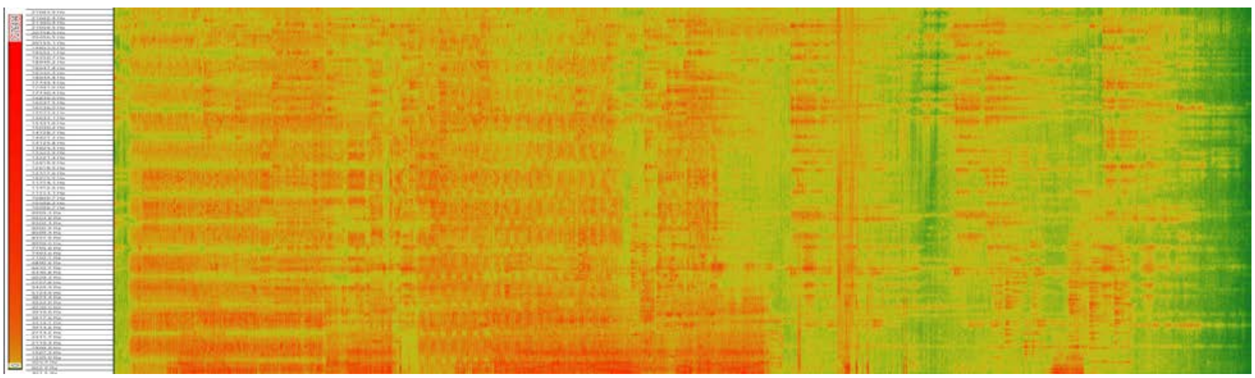

Fig. 3.4: Paradox III: *Dissolution* (about 2 minutes) and 'Coda' (about 1 minute)

All the spectrograms illustrated in this chapter were realized with the Sonic Visualizer software, the Queen Mary University of London Adaptive spectrogram plugin and Jamie Bullock's libxtract plugins (Loudness, Inharmonicity, Spectral Variance).

Technical Analysis: Paradoxes I - III

In this chapter emphasis will be given to the analysis of some of the basic technical aspects related to my music work. We will examine vocal synthesis, Brownian noise, FFT processing, sample and hold, resonators, which are some of the techniques used for the realization of this composition and are the basis of my next compositions as well.

The low-pitched vocal events of the first part derive from a previous vocal synthesis project that I had made using the Max/MSP visual programming environment. In this project, I had synthesized the physical model of a soprano voice that could sing the folk song Daisy Bell (Bicycle Built for Two). A short demonstration of my vocal synthesis projects can be found on the digital storage device in the video examples section. The first one (Ex.01 Voice Sythesis Soprano (no lyrics).mp4) demonstrates the singing voice of a mezzo-soprano without text, the second (Ex.02 Voice Synthesis Choir (no lyrics).mov) presents a choir singing a J.S.Bach's choral without text, and the third example (Ex.03 Daisy Bell with lyrics.mp4) is the Daisy Bell project, where a synthesized soprano voice sings the lyrics of the song accompanied by a Moog synthesizer which is based on the FAUST library examples (GRAME 2019). In fact, any song could be used, however the particular choice was made for historical reasons; more specifically because that was the first song sung by a computer (Curtis 1995). This project can be also extended to a speech synthesis project although text and linguistic analysis is currently out of the scope of my artistic activities and interests. The balance between intelligibility and aesthetic quality was my main concern, since the soprano needed to sing with as much comprehensible text as possible. At this point, I would also like to quote an example of a synthesized voice singing the Daisy Bell song coming from the examples included in the Festival Speech Synthesis System (Ex.04 Festival Speech Daisy Bell.mp4) and point out the trade-off between intelligibility and aesthetic quality befitting a song.

Although there are plenty of new approaches for speech and vocal synthesis, my project was based on (and partly extended) a classic technique derived from Rabiner's (Rabiner 1978), and Oppenheim's theories (Oppenheim 1975). As the result comes out to be satisfactory, there are not significant disadvantages to doing so and I have also included below for reference other singing voice synthesis projects that use different and more recent, advanced synthesis techniques by Perry Cook (Cook "Singing Synthesis") and Xavier Rodet (Rodet 2012).

The physical model for singing synthesis that was created and finally realized using Max/MSP can be simplified in the following scheme:

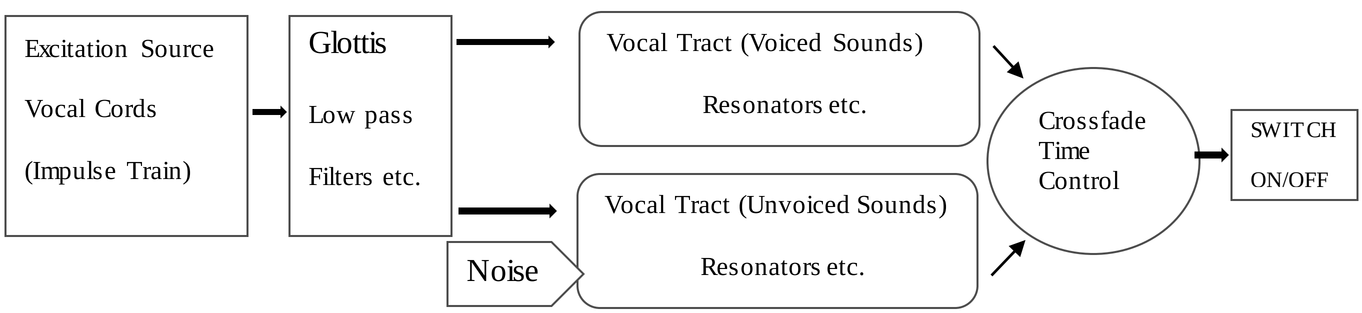

Fig. 3.5: A preliminary plan for singing synthesis

It might not be necessary to discuss the filters and their transfer functions used for the project, since all this information can be found in any Digital Signal Processing book. However, it might be useful to briefly discuss the vocal tract stage and observe one of its properties which is to vary its parameters almost every 10 milliseconds (Rabiner 1978).

In order to approximate this physical function of the vocal tract, a vibrato with an irregular adjustable rate and depth has been applied to the excitation source. The vibrato section comes from another academic project where I had translated the Pure Data examples of Miller Puckette’s textbook *Theory and Techniques of Electronic Music* into the Max/MSP environment, involving also the development of a library with external objects that could make these examples portable with precision. The external objects library and some abstractions I developed are included in the storage media device as well (Nikolopoulos Pd4Max, 2020).

The excitation source used in my composition is an impulse train (Oppenheim 1975):

$$\sum \delta(t - kT)$$

However, this particular impulse train is band limited and constructed using a variation of a second-order Thiran allpass filter (Num 2009) in gen~, which supports sample by sample signal processing inside Max/MSP. The following recursive equation gives us the filter mentioned above:

$$y(n) = a_2x(n) + a_1x(n - 1) + x(n - 2) - a_1y(n - 1) - a_2y(n - 2)$$

Next, we will examine a basic FFT processing used for the overlapping layers of the first part. The sound source used for the FFT processing is a Brownian noise generator, the vb.brown~ Max/MSP object (Böhm n.d.), with variable step size which allows us to change its spectrum in real-time providing a greater flexibility in spectrum manipulation. A Brownian noise spectrum in its initial stage is characterized by an almost exponential decay as shown in the figure below (Fig. 3.6). Simply enough, the Brownian noise power spectrum falls off with frequency $1 / f^2$ where f is frequency in Hertz (Moore 1990). Therefore, the output of the FFT processing algorithm that operates on our Brownian noise generator has an output characterized by a limited and more controllable inharmonicity.

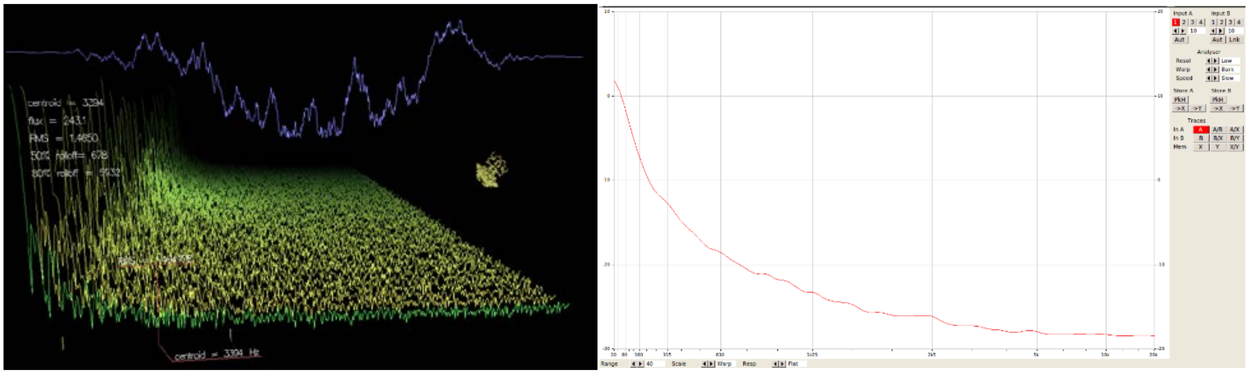

Fig. 3.6: Brownian noise analysis with sndpeek real-time 3D audio visualization (left) and JAPA audio spectrum analyzer (right).

The basic FFT processing which operates on a Brownian noise is a variation of a Pure Data example by Miller Puckette where the FFT acts as an equalizer (Puckette 2006). We write and store data to a buffer and then the bin index reads up to the Nyquist frequency (half the FFT frame size) inside the FFT processing object. In Max/MSP there is the pfft~ object for FFT processing that utilizes the STFT algorithm. The bin index retrieves then the data stored previously in our buffer and operates on the amplitude of our incoming signal. Analyzing the theory of this simple FFT processing technique, which is also the basis for various other techniques used for the rest part of my research project, we can express it and understand it better with the following formulas.

First let's get started with the well-known Fourier Transform formula (Lechner 2014, 168):

$$X(\omega) = \int_{-\infty}^{\infty} x(t)e^{-j\omega t}\,dt$$

Where x(t) is our continuous input signal multiplied by a phasor and X(ω) is the output as a function of frequency. Since we are dealing with digital signals and samples, the previous formula will be modified in discrete time as follows (Smith 2002, 2):

$$X_k = \sum_{n=0}^{N-1} x_n\, e^{-jkn2\pi/N}$$

And since Max/MSP utilizes the Short Time Fourier Transform (STFT) algorithm, we get (Curtis 1997, 49):

$$STFT\left(x_{(n)}, k, t\right) = \frac{1}{N} \sum_{n=-\infty}^{\infty} w(n-t)x(n)\, e^{-j2\pi kn/N}$$

Where w(n) is the window function(STFT provides also overlapping windows), x(n) is our signal which is multiplied by the clockwise phasor, 1/N is a necessary normalization, and k = 1, 2, 3,…. up to N – 1. From the above we get the real and imaginary parts as follows:

$$Re\left\{\left(x_{(n)}, k, t\right)\right\} + jIm\left\{\left(x_{(n)}, k, t\right)\right\}$$

Where Real and Imaginary parts respectively are:

$$Re\left\{\left(x_{(n)}, k, t\right)\right\} = \frac{2}{N} \sum_{n=-\infty}^{\infty} w(n-t)x(n)\cos(\frac{2\pi\kappa n}{N})$$

$$Im\left\{\left(x_{(n)}, k, t\right)\right\} = \frac{-2}{N} \sum_{n=-\infty}^{\infty} w(n-t)x(n)\sin(\frac{2\pi\kappa n}{N})$$

Then our FFT processing that operates on the Real part in discrete time for arbitrary length becomes:

$$\frac{2}{N} \sum_{n=-\infty}^{\infty} w(n-t)x(n)\, A_i \cos\left(\frac{2\pi\kappa n}{N}\right) - \frac{2}{N} \sum_{n=-\infty}^{\infty} w(n-t)x(n)\sin\left(\frac{2\pi\kappa n}{N}\right)$$

Where A is the stored amplitude retrieved from the buffer mentioned before and i is the bin frequency index that goes up to the Nyquist frequency.

The weighing of the amplitudes follows a Max/MSP standard random distribution technique. Furthermore, there is a density control that controls the number of partials that will be generated each time. The higher the density factor is, the noisier the spectrum that results.

The most characteristic musical figures of the second part are the irregularly bouncing metallic objects. The physical models were made using Brownian noise, sample and hold and resonant bandpass filters and they were based on an example from the book *Electronic Music and Sound Design vol.2,* from the section Interlude C (Cipriani,Giri 2014) that I have used as reference. A Brownian noise object with a very steep exponential decay goes through a sample and hold object whose trigger input is controlled by a random signal generator. The frequency of this random signal generator is then controlled by a ramp generator which has a function similar to an accelerando that follows a ritardando. The sample and hold signal goes through an envelope that also has an exponential decay and controls the amplitude. Eventually, the signal passes through a group (poly~ object) of resonant bandpass filters with different frequencies and resonance times for each one. Different instances of this model have been used with various timing properties in order to make a counterpoint structure without the traditional meaning. This kind of counterpoint technique has been used here with an intention to create a sense of assemblage with objects fading in and out with a shift in timing of their development.

A similar use of counterpoint techniques will be continued in the third part. Various granular synthesis techniques have been combined in a voice against voice way. The dissolution and lack of direction are more obvious in the third part due to the following reasons: a) Counterpoint is harmonically independent, b) The granulated very short bursts don’t allow us to think about the pitch.

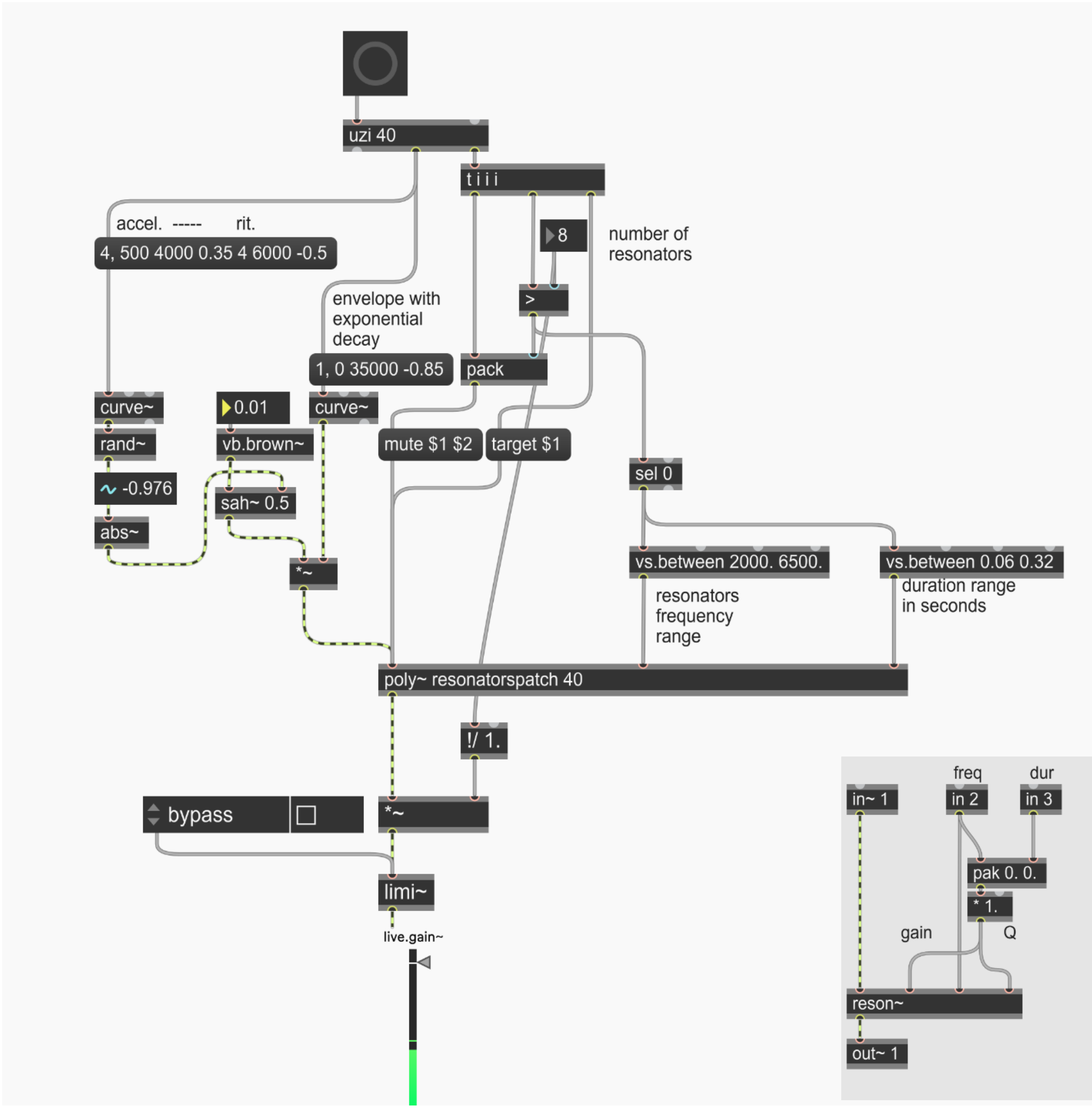

Fig. 3.7: The physical model of the irregularly bouncing metallic objects and the subpatch with the resonant bandpass filters (bottom right).

3.2 Paradox IV

The fourth Paradox is an electroacoustic music work that is based solely on physical models, sound synthesis and real time audio. While real time audio performance provides a great flexibility on specific things such as the structure of the piece and a large scale of parameters that can be altered, there are disadvantages that should be considered before deciding the one over another way. One disadvantage is the CPU usage that can rise unexpectedly when we need more demanding signal processing tasks or when we need to choose a smaller buffer size for a more accurate performance. The other disadvantage is that mixing the live output using master bus equalizers, compressors, multiband compressors, limiters, might not work the same well as in a fixed audio work due to the unpredictability of the improvisation.

Paradox IV deals with two distinct entities and explores ways that these two things can coexist. The first entity encompasses the musical sounds as a result of physical models and the second one encompasses the non-musical sounds as a result of the application of various filtering methods on different kinds of noise generators. The choice of using physical models is not an attempt to necessarily mimic sounds from nature or musical instruments such as wind chimes, strings pizzicati etc. Their pure purpose is to stimulate the listeners' perception through concrete pitched sounds and their familiar timbres. On the other hand, there are non-musical sounds including but not limited to modeled environmental sounds, which attempt to create an unfamiliar environment coming in contrast to the pitched sounds that are moving randomly, however, they are organized in distinct frequency range layers. The relation between musical and non-musical sounds has been explored in some way in ambient music (Adkins, Cummings 2019). Nevertheless, my approaches, methods and outcomes are totally different from the concepts in ambient music.

My music compositions feature computer-generated noise incorporating traditional and unconventional techniques as well. For example, I use physical models of traditional musical instruments or other synthetic sounds in combination with a wide variety of processing techniques which are based on noise. Noise is a structural element in my works through its contrast with other pure musical elements such as random string pizzicati or metallic sounds with concrete pitch. The variable balance between noise and musical sounds is a characteristic of many of my works, apart from my *Six Paradoxes*, defined by the duality between abstract and concrete.

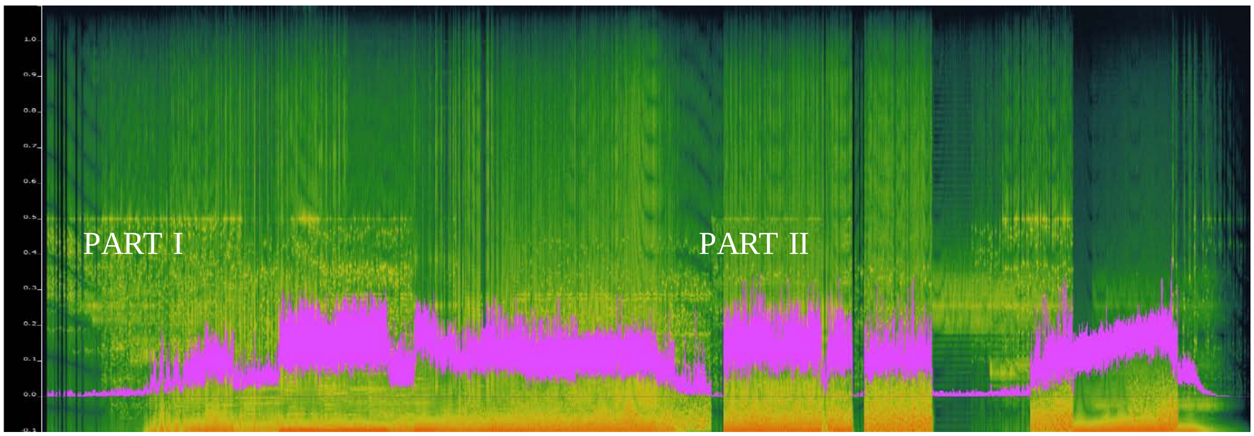

Fig. 3.8: A spectral analysis of Paradox IV with the purple line indicating the course of the level of the sound in RMS (Root Mean Square) values.

The structure of Paradox IV consists of two distinct parts whose basic difference can be easily observed in the spectrogram above (Fig. 3.8). As the figure shows, there is also an RMS analysis with a reference line below the purple line. The first part has the characteristic of a continuum of contrasting events and extends to the moment where the purple line of our RMS analysis almost touches the reference line, zero. In fact, the second part starts when the continuum abruptly pauses and the silence begins. However, something distorts the silence and that heralds the change in our form. Abrupt pauses happen a few times in the second part alternating either with pitched sounds (such as pizzicati) or with sounds shaped by various kinds of noise filtering such as state-variable filters etc.

As we can understand and presumably perceive by listening, there are contrasts inside the entire form itself. These contrasts and contradictions, as structural elements, come to substitute the lack of direction, an impression that the absence of melodic and harmonic passages can give (Meyer 1994). That seems also to be one way whereby we are able to develop larger forms and preserve at the same time a relative coherence for new compositions, since our perception of listening has been influenced by an education system with a huge traditional background and has been shaped by stimuli of our everyday life in a specific and usual way. Whether we like it or not, we are exposed to melodies, harmonies and rhythmic patterns in our daily activities. That is to say, there aren't any issues in the material itself, yet it's an issue about the way our listening skills have been cultivated to perceive sound and music in general.

Technical Analysis: Paradox IV

In this section I would like to provide some ideas on multichannel mixing and working with ambisonics. Ambisonics is the modeling of the waves that sound sources radiate and for that purpose spherical harmonics are used. The ICST ambisonics library facilitates the decomposition of the sound waves, the encoding and decoding process with the use of a few externals objects (Schacher and Kocher, 2006). Furthermore, the ambipanning~ object, which utilizes equivalent panning algorithms bypassing B-format, offers an alternative solution to the ambiencode~ and ambidecode~ Max objects. Ideally, the speakers need to be evenly spaced and for a sixteen or twenty loudspeaker multichannel setup, a third order Furse-Malham or a fifth order N3D or SN3D has given satisfactory results so far. The picture below shows how the speakers are positioned (right) with the speakers 13 to 20 being elevated.

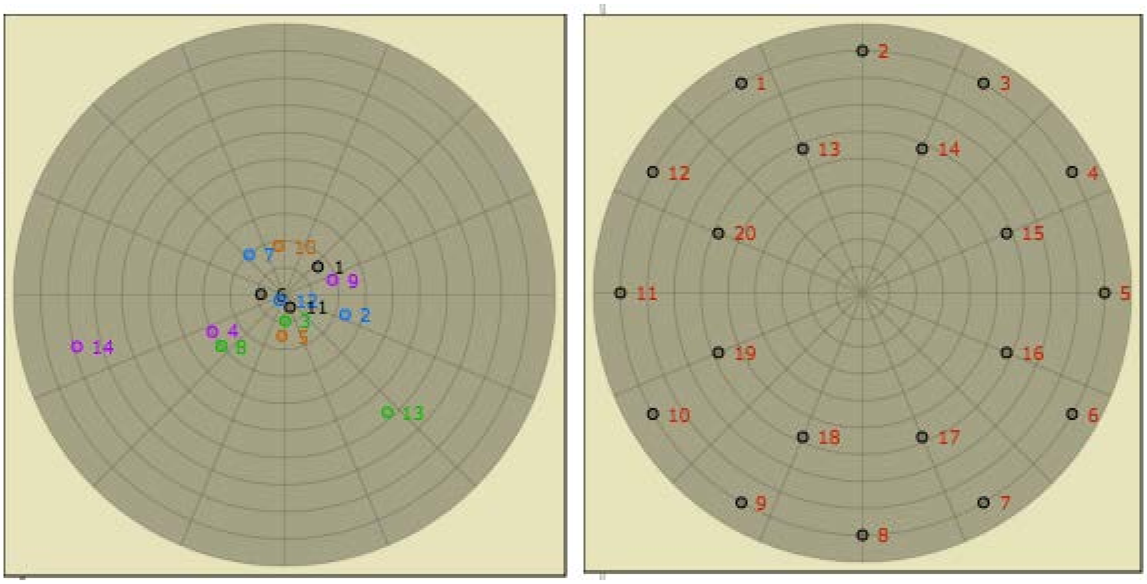

Fig. 3.9: Left side are the moving sources and right side are the speakers setup.

Since sound mass and diffusion were more important than sound localization for the present situation, I extended the stereo mix I had already to an ambisonics mix treating each channel of my individual sound sources differently if the initial source was stereo and differently if it was mono. Furthermore, I preserved the original stereo mix volume level (including the programmed volume automation) of the individual sources and therefore I kept the ambisonic distance parameter in one stable position unchanged. The trajectories of the sound sources were programmed to follow a variety of patterns. I used signal objects such as Brownian noise object, the cycle~ (cosine) and the rand~ objects, smoothed them and then scale them to represent degrees and control the azimuth and elevation parameters. Next figure depicts a part of this programmed motion (Fig. 3.10).

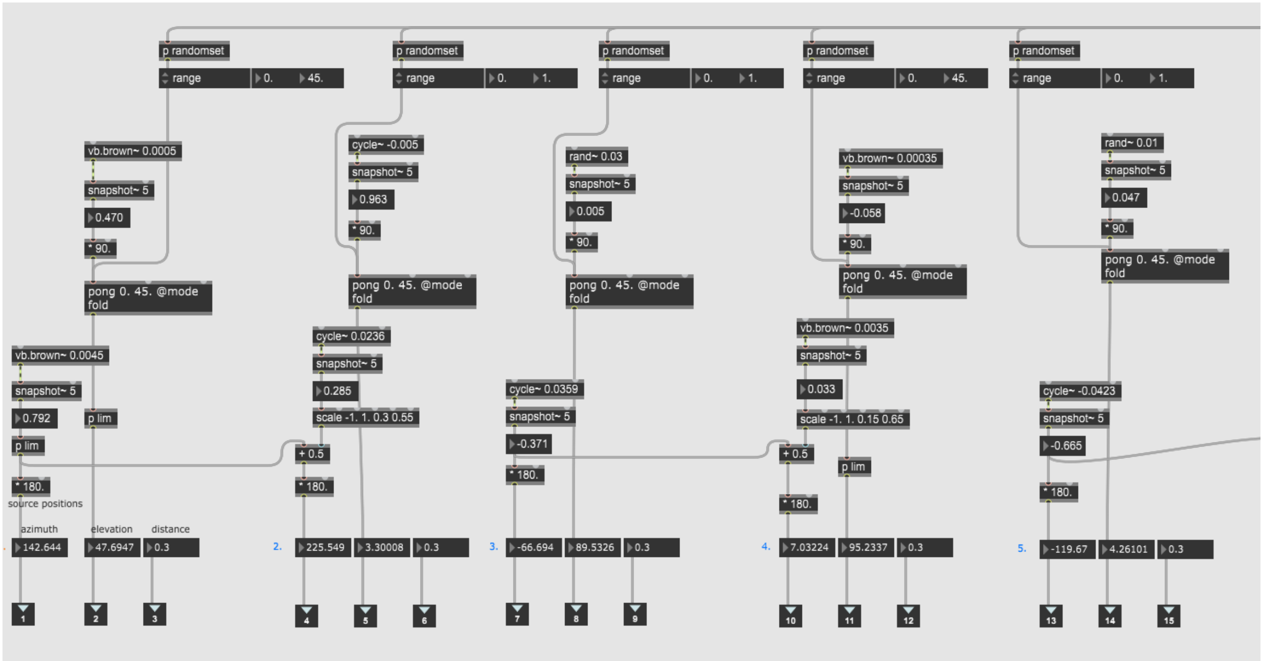

Fig. 3.10: Signal objects converted to float numbers, smoothed and scaled to represent degrees.

In order to achieve a greater flexibility in terms of shaping the sound in real time, three more Max/MSP objects have been used between the ambisonics decoder and the master output volume. One equalizer, one compressor and one limiter, objects with multichannel capabilities that can change dynamically their parameters and control the overall sound. I developed these three multichannel Max/MSP externals after some time of experimentation with signal processing and open source plugins on Linux operating systems (Nikolopoulos Linux4Max, 2020). The radiumcomp~ object is the radium compressor (Matheussen 2019) by K. Matheussen and the fil4~ and dplimit~ objects (Gareus 2019) are Robin Gareus' plugins which in fact are the updated versions plugins that were initially developed by Fons Adriaensen (Adriaensen 2019). A moderate use of a parametric equalizer and dynamics processing before the master volume output could improve and unify the final multichannel mix (Fig. 3.11).

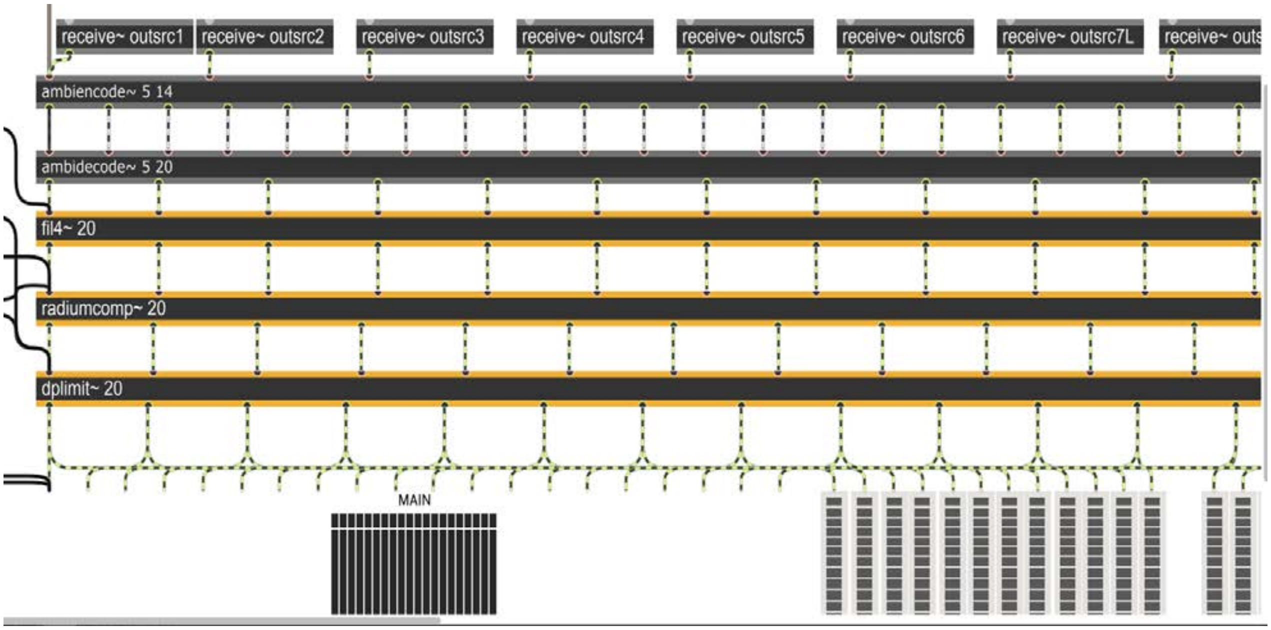

Fig. 3.11: An example of ambisonics setup with a multichannel filtering and dynamics processing on the master bus using my max externals library linux4max.

All of my Paradoxes follow a similar procedure regarding ambisonics except the last one where not so much sound source movement is necessary. For the last paradox, I experimented with the multichannel (mc.) objects in order to diffuse the sound and as we can see from the next picture (Fig. 3.12) the process is very simple. I split the left channel to the left semicircle and the right channel to the right one, duplicated each channel half times the amount of the total number of the available speakers, scaled their volume down and with the mc.mixdown~ object I diffused the sound to each side with a different amount of deviation (deviation message distributes the given attributes in a random order every time we click on the message box).

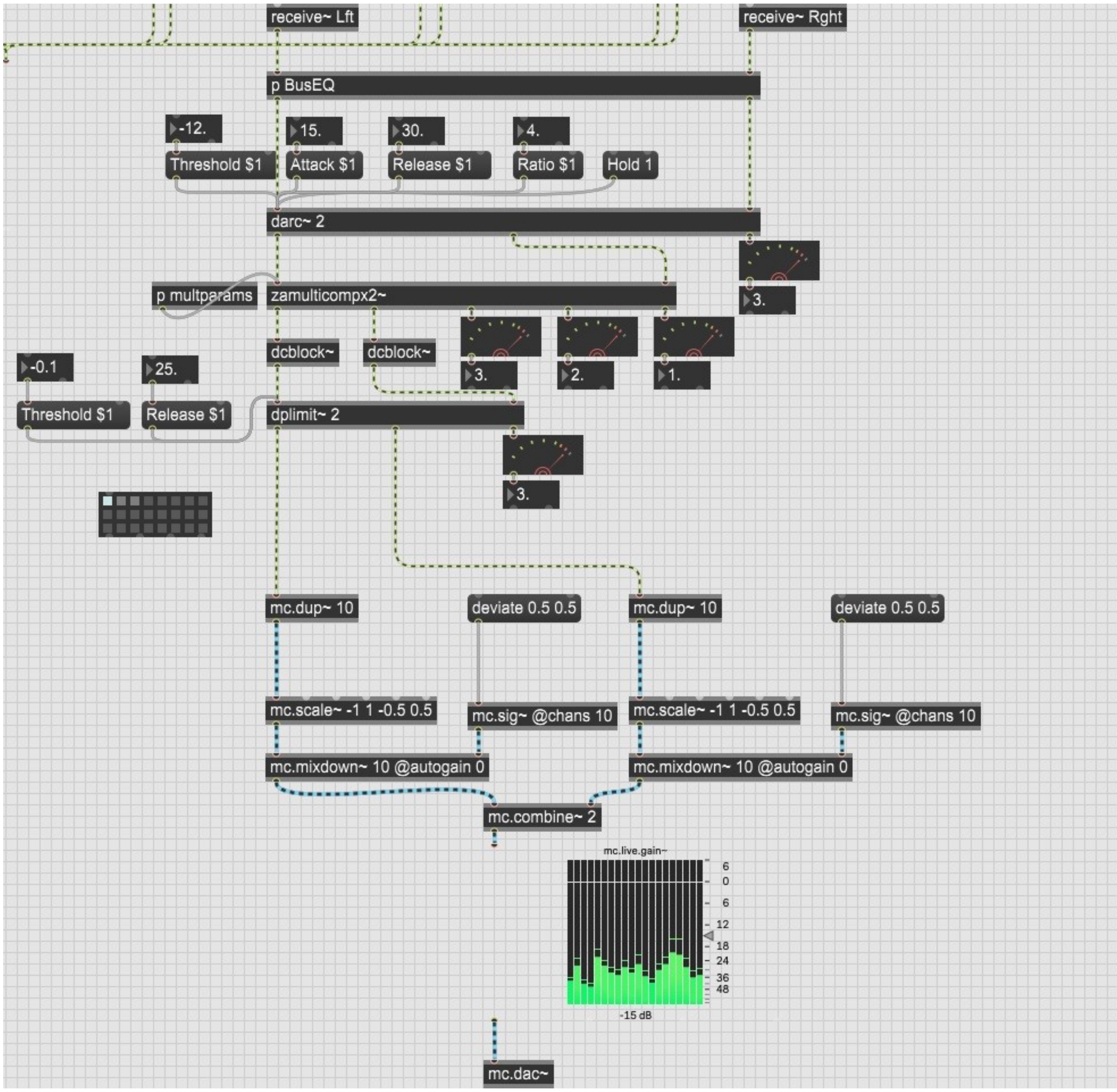

Fig. 3.12: Another way to diffuse a stereo sound with the Max/MSP multichannel (mc.) objects (used for Paradox VI).

3.3 Paradox V

Paradox V is an electroacoustic music work that combines nonlinear synthesis, physical models, and digital signal processing on sound field recordings. It deals with the idea of 'becoming' and the contradiction between continuum and abruptness or emptiness. Consequently, the *Deleuzean* idea of 'becoming' had motivated me to compose an assemblage of divergent musical and non-musical events that would head gradually to a sound energy peak. The next figure (Fig. 3.13) shows a melodic range spectrogram with the energy of sound (RMS) of the first section. This section consists of three basic sound characteristics: a) non-linear sound synthesis that results to a kind of mechanical continuum of sound, b) moments of a variable filtering applied on noise generators and c) extreme or moderate time stretching that is applied on sound materials.

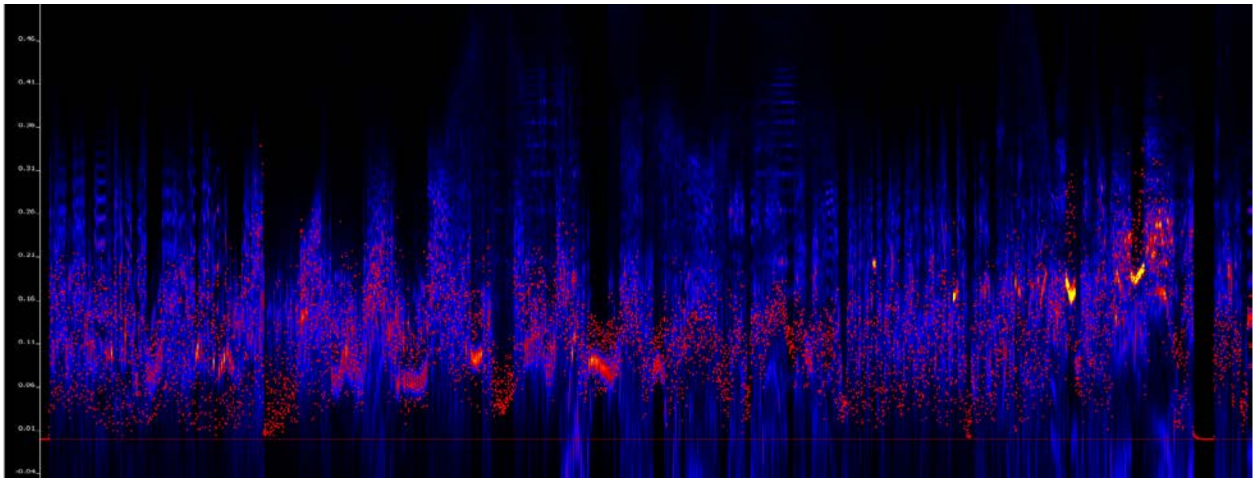

Fig. 3.13: Melodic range spectrogram of the first three minutes of the fifth paradox. The red points indicate the sound energy in RMS (Root Mean Square).

One feature that occurs throughout the whole piece is the reference to the past and specifically to the previous Paradox works. There are moments when some very short sound material from Paradox IV has been processed using time-stretching processing and then combined with the new events in order to support a sound energy escalation for example. Furthermore, there are also variations of physical models of string pizzicati that begin after a moment of abruptness, functioning also as a reference to the past. By altering the properties of the past and putting them in a new context, we alter also the properties of the present and that particular assemblage is what I have defined as 'becoming' in the environment of sounds.

In the next section, which is illustrated the figure on the next page (Fig. 3.14) the time-stretched layers have a more pronounced presence along with high-pitched brief figures. At the same time, a new material, which derives from the application of signal processing on audio recordings of sharpening metallic objects sounds, makes its appearance with a sparser and slower pace than the rest events. These new sounds will be dominant taking various forms and shapes for the third and last section, where a long sonic continuum comes to give a particular emphasis on the properties alterations and the 'becoming' process of each of the events used in the Paradox V.

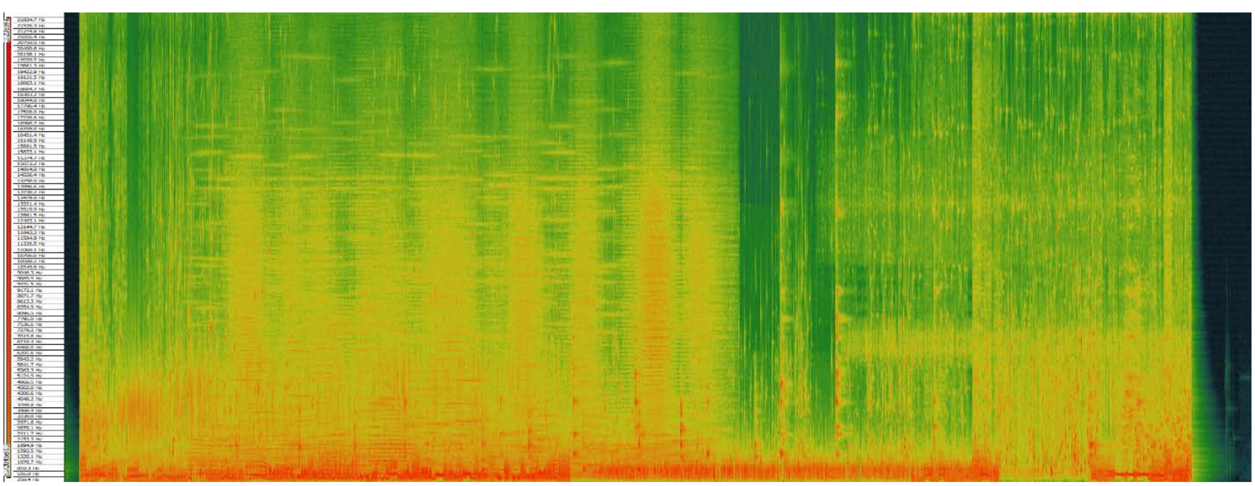

Fig. 3.14: Spectrogram of the second section (approximately two minutes) with prolonged time stretching events.

3.4 Paradox VI

Paradox VI, the last, is an electroacoustic music composition that deals, more than any other piece in this series of paradoxes, with the contradictions between musical and non-musical sounds. The Paradox VI is based on physical models of acoustic instruments and an algorithm which can produce a wide variety of sounds such as pitched percussive sounds and sounds that have the phonological aspects of a real language or emotional speech. However, the latter ones, although they give the impression of a real language, are incomprehensible due to exogenous factors such as radio interference, or filtering because of the existence of physical objects between source and listener. There is a technical explanation of this algorithm that I have developed below.

One of the main characteristics of the sixth paradox is the slow pace of the piece and the relatively long tails played by the acoustic instruments that aim to put the listener in a situation of deep thinking and provide plenty of time to perceive and interpret the sounds through 'multiple listenings'. The acoustic instruments used for this work are violins, celli, piano and bass clarinet, while their physical models are Max/MSP externals I made using the source code from FAUST programming language examples (these objects are not just automatic exports from FAUST for a greater efficiency and flexibility) (GRAME 2019). These objects are a part of my 'Linux for Max library' (Nikolopoulos Linux4Max, 2020) I had developed and used in some of my pieces, including objects for sound synthesis and dynamic processing. The library linux4max has been included in the accompanying materials that can be found on the USB flash drive.

The structure of Paradox VI is distinguished in a single and unified form (Fig. 3.15). Inside this unified form, there are three important distinct situations or events that take place: a) non-musical sounds alone b) musical with non-musical sounds together where the non-musical sounds tend to dominate and c) musical with non-musical sounds together where the musical sounds tend to dominate.

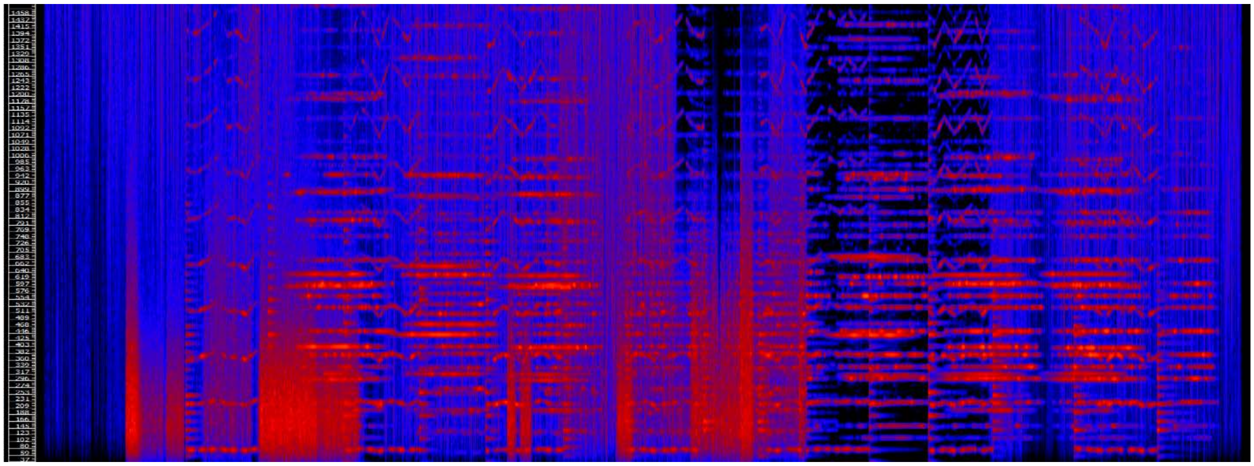

Fig. 3.15: A melodic range analysis that shows an overview of the unified form of Paradox VI.

Furthermore, there is a contrast between musical and non-musical sounds that is well pronounced. The acoustic instruments preserve a slow pace for the whole duration of the piece, while tiny percussive figures or other noisy spectrum events have a more active participation. Using physical models to compose an ensemble of acoustic instruments has specific advantages in terms of flexibility and sound organization. It allows us to experiment in real-time with different combinations of these instruments, preserve particular entries of them whenever we want, and change their entry order while preserving the harmonic structure and coherence of the piece. The next figure (Fig. 3.16) shows the basic harmonic structure that spreads during the whole music work. In addition, physical models allow us to have a greater control in terms of pitch or other technical restrictions. For example, the double arrows next to violin notes in the score indicate a deviation in pitch that is no more than a quarter of tone and the amount of deviation is specified by a particular random distribution. Some of these microtonal deviations are technically impossible to be played by a violin without scordatura (see last measure, second violin).

**Every instrument sounds as it is written*

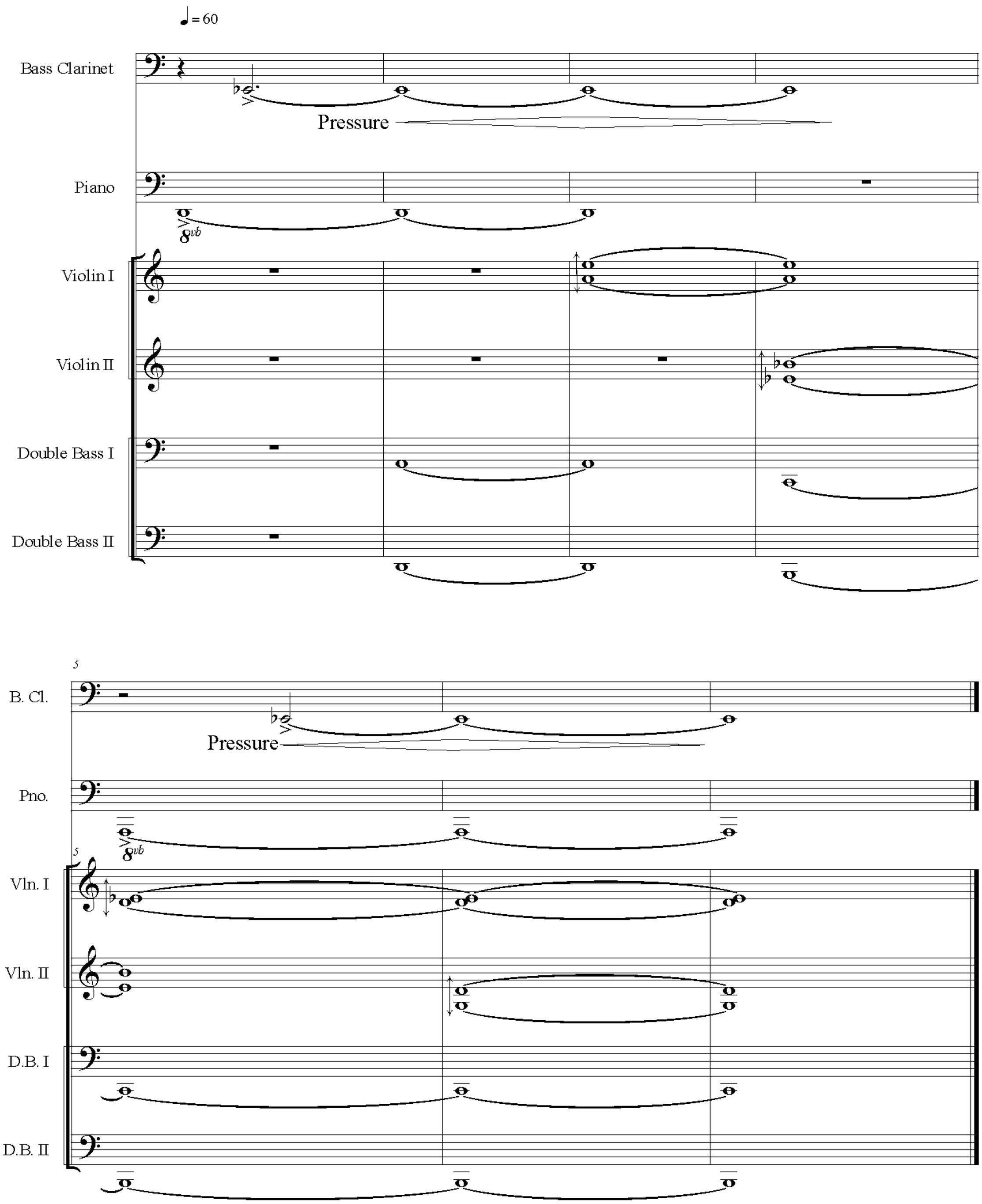

Fig. 3.16: The music score example shows the instrumentation used for the last Paradox and gives an overview of the harmonic structure that appears in different orders. The double direction arrows indicate a random pitch deviation that doesn't exceed a quarter of tone.

Technical Analysis: Paradox VI

In this chapter we will be looking at the development of an algorithm used for creating a variety of percussive sounds and modeling of real speeches or voices, which can't be understood due to exogenous factors as mentioned before. For the purpose of our analysis, the particular algorithm could be split into two sections shown in the next two figures (Fig. 3.18 and Fig. 3.19). A combination of fourth degree polynomials, phase/frequency modulation (Chowning 1973), sample and hold processing and logical operations has been used. As we will see later, the polynomials function as triggering inputs for the sample and hold processing, as phase modulators to another oscillator and as amplitude modulators to the algorithm's final amplitude control output.

The first section (Fig. 3.18) that is based on the polynomial could be expressed mathematically as follows: $x = pm^4 + pm^3 + pm^2 + pm - \mathrm{a}$

$$if\ x > 0\ then\ x\ \ else\ 0$$

Where 'a' is an offset value, which in combination with a gate object filters out the negative values and controls the amount or rate of the positive values passing through. The 'pm' indicates a phase modulation instance that consists of an oscillator and a random generator signal. It could be expressed mathematically as: $pm = \cos(f_1 t\ + (rand(f_2))^2)$

Expressing the same idea with a diagram we could simply depict it as:

× Degree polynomial	➞	Offset & Gate	➞	Only positive values pass and the offset variable × controls the rate, from slow to fast (starting from a constant value equals to the polynomial degree and moving downwards).

Fig. 3.17: Diagram representation

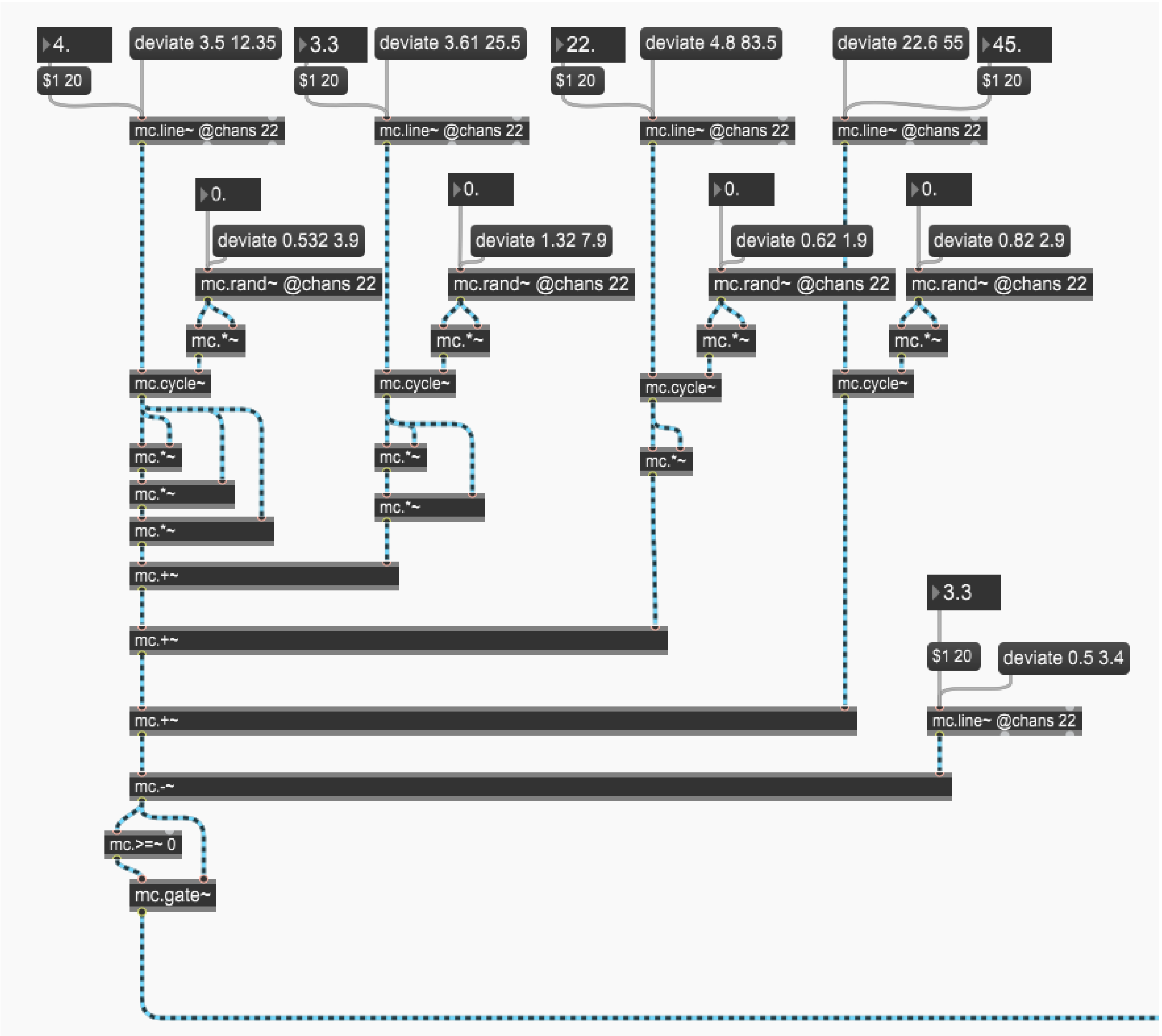

Fig. 3.18: The first section consists of the polynomial structure, the offset and the gate that filters out negative values.

The basic structure of the second section of the algorithm could be expressed mathematically as:

$$x = P\cos(n_1 t + \cos(n_2 t + P))$$

Where P is the output of the polynomial from the previous section, which controls the amplitude of the main oscillator and the phase of a secondary oscillator, n_1 is the noise generator after a sample and hold processing and logical operations, which controls the frequency of the main oscillator and n_2 is another noise generator passing through sample and hold processing and logical operations, which controls the frequency of the secondary oscillator. The later one consequently controls the phase of the main oscillator. Summarizing, the basic body of the algorithm is a cosine wave oscillator that modulates the phase of another cosine wave oscillator. Both oscillators' parameters will be controlled by the polynomial and other intermediate processing such as sample and hold, logical operations and filtering.

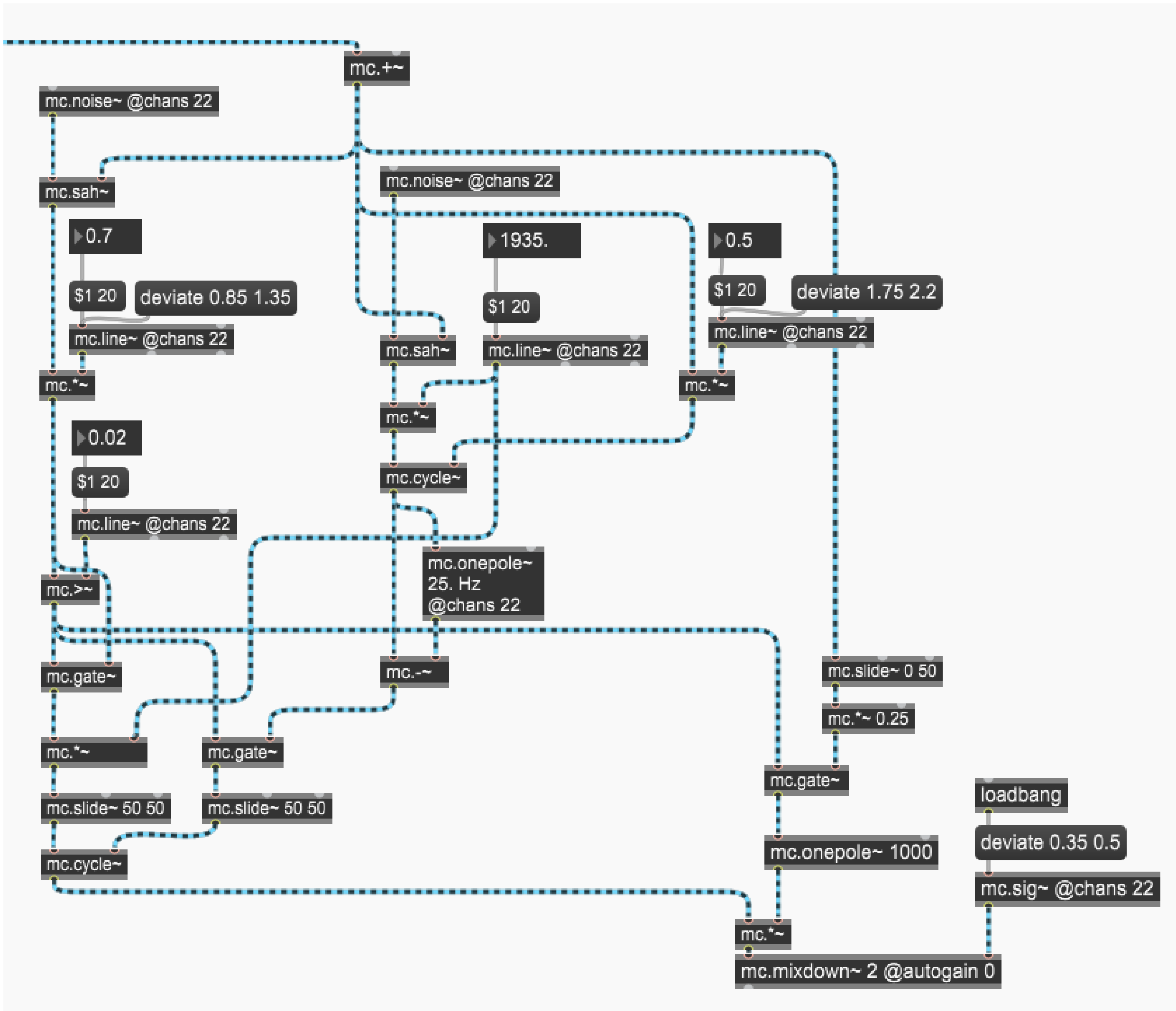

Fig. 3.19: The second section consists of sample and hold processing, logical operations, a double phase modulation, and amplitude modulation.

Furthermore, as it is shown in the figures above, the standard Max/MSP objects have been replaced with their multichannel counterparts (mc.), which is a new feature providing a flexible way to create polyphonic algorithmic patches with an individual parameter control for each voice separately. Although it is beyond the scope of this writing to explain the multichannel objects in Max/MSP, the deviate message is one of the messages that can be used to spread different parameter values over the multiple instances.

In addition, a probably better understanding of the second section of the algorithm analyzed before could be provided by the diagram that appears in figure 3.20. The box with the letter P indicates the output from the first section described above (polynomial signal output), while the asterisks generally indicate an amplitude control. The amplitude weights are variables that dedicate the quality and timbre of the produced sound. Something that has not been graphed is some filtering between these connections that aims to smooth and polish the signal processing in particular parts of the algorithm.

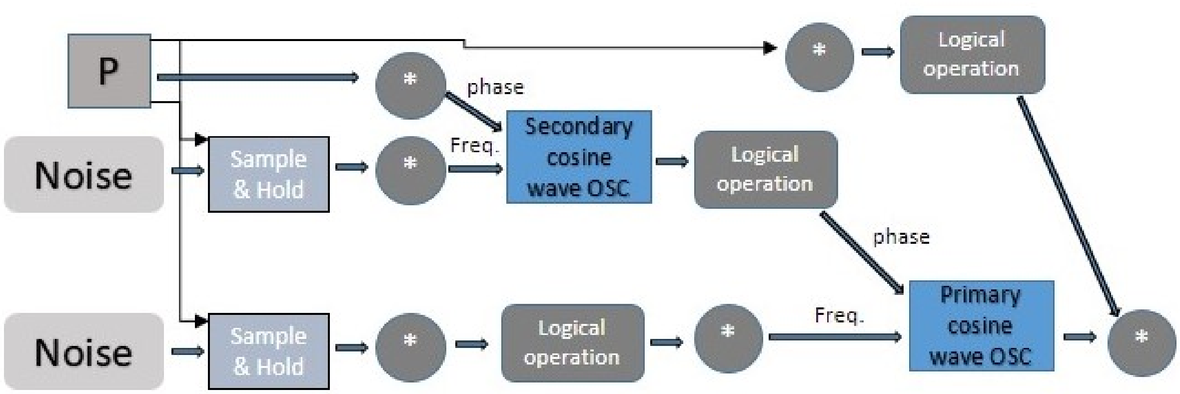

Fig. 3.20: A diagram representation of the second section of the algorithm. Wherever a multiplication sign receives only one arrow (signal) as input indicates the existence of a variable value (not depicted) which is changeable.

4. A Machine Learning Study

Machine Learning Study is an electroacoustic music composition that explores real time audio processing and improvisation using an Arduino board with a set of sensors that capture the motion of the palm and react accordingly in numerous of ways. The main concept of the piece deals with the idea of the 'continuum of moments' and *becoming* (see page 11). The concept of 'continuum of moments' is an effort to introduce a kind of release from the conventional ways of listening that are interconnected with focus on tonality, spectrum, any kind of harmonic and rhythmic progression and direction, which are all subjective qualities. The 'continuum of moments' is not a system of sounds that evolves linearly in time, but it is a system of sounds that 'acts'. The psychical actions, which create the input data received by the sensors, are being translated into sonic actions during the improvisation time. This composition involves improvisation and to some extend is connected to the thoughts discussed in the introduction regarding Deleuze's philosophy and Paradoxes; "creative practices must be both non-deterministic and somehow controllable or predictable" (Thornton 2015).

After using the descriptive term 'continuum of moments', it might be worth mentioning at this point Karlheinz Stockhausen's (1928 – 2007) work *Momente* (1962-69) for solo soprano, four mixed choirs, and thirteen instrumentalists and his *Kontakte* (1958 - 60) for piano, percussion and electronic sounds (an electroacoustic version also exists). The marked similarity to my work is the idea of a sequence of rapid, self-contained sonic events that are not related to each other. However, the idea of a non-linear evolution of sound is realized in a completely different way in my Machine Learning Study work. Although the gestures that define the quality of sound are uncontrollable, the sequence of changes (triggered by the distance sensor) are predetermined. Each sequence could be seen as a container with ‘continuum of moments’ inside. Thus, there is a concrete macro-progression which is however flexible in timing and contradicts to the ‘continuum of moments’ that contains; a duality that I have used for creative and structural purposes.

Machine Learning Study is based on a procedural audio environment that has been used also in video games, interactive arts, multimedia applications on the web etc. As it has been specified by Andy Farnell “Procedural audio is non-linear, often synthetic sound, created in real time according to a set of programmatic rules and live input” (Farnell 2007, 1). Before proceeding with any further details of my music work, I would like to demonstrate another project I have made using Unity and Max/MSP programming languages and the OSC (Open Sound Control) protocol for the communication between these two languages. The fifth video example (Ex.05 Procedural Audio.mp4) in the accompanying materials shows my video game project, which makes use of procedural audio techniques and is actually based on one of the Unity examples, Roll a Ball (Unity 2020).

Although the improvisation is an important aspect of this work, there is a basic structure that was designed beforehand. The figure below shows a general diagram of the structure and the time when the different ‘moments’ are introduced. Since we are dealing with improvisation, the timeline diagram could be only approximately presented for the purpose of our analysis. Although the general structure exists, it’s very flexible and every change occurs after a specific gesture by the performer when approaching the hand close to the distance sensor. Once the distance sensor detects an object ahead, then a new ‘moment’ will be introduced or a distinct change will happen. The Arduino board, as it has been constructed for my piece, could almost simulate the brain part of a robot that can move and have a basic understanding of the space that it is moving in. On the other hand, the performer’s gestures are the external stimuli that change the ‘actions’ according to the data received by the various kinds of sensors. Talking about gestures, it might be worth mentioning that there are no bizarre movements or gestures involved, but only discreet ones.

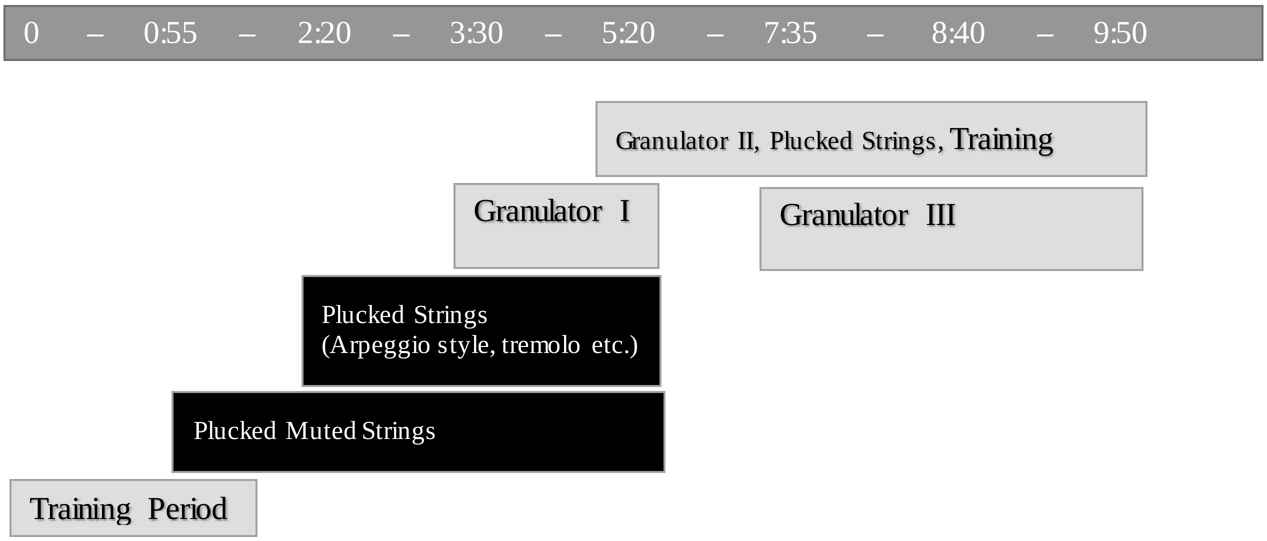

Fig. 4.1: Machine Learning Study: The first upper rectangle indicates the different sections in time (minutes:seconds) and the boxes below give an overview of the content of each section.

There are three kinds of movements used for the performance: a) vertical movements of the palms of two hands that affect the main amplitude control through a light photoresistor, b) a slight left or right rotation of the wrists that affects the plucked strings' rate and pitch, and granulation parameters through two pyroelectric sensors and c) a short gesture of the one hand which passes in front of the distance sensor thereby changing some of the main mix settings such as introducing new sounds, changing their volume levels, or interrupting, pausing other sounds.

After working and experimenting with interactive music systems, machine learning and sound, a question may arise: What can we learn from that? I personally believe that an interdisciplinary research project, which involves technology and art can help us to understand more things about ourselves, our expressions and the ways we are thinking. More specifically, I found that we can learn more flexible ways of thinking. Additionally, different performing ways involve physical actions, listening and decisions making. It's an active participation that doesn't require following stiff rules in a similar fashion to the way we play an acoustic musical instrument and motivates us to explore more about our physical actions in space and time, and the ways these actions could be interpreted or interconnected with other fields except music and sound. Furthermore, an interdisciplinary study helps us to think in multiple layers and probably discover things that we couldn't discover before. However, on the opposite side, it could also reveal our weak points and even ignorance about fields that we don't know very well, because it's perhaps impossible to infiltrate in depth in each field that exists in a potential multidimensional study. In this case, prioritizing or deciding where to focus more, is an important task that could solve many related issues.

Technical Analysis: Machine Learning Study

A general idea of the basic workflow of my Machine Learning project can be viewed in the scheme bellow (Fig. 4.2). Arduino board with sensors and Max/MSP have been used for the realization.

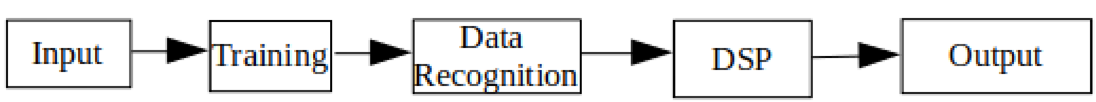

Fig. 4.2: Basic workflow of the piece.

The sound parameters that will mainly be affected after the training period are the spectrum, rhythm and dynamics. After training, the data are stored and used later on in the pattern recognition stage. The interaction between the performer and the computer will be based on this pattern recognition that will be crucial for the different formations of sound through Digital Signal Processing, electroacoustic music techniques etc. One kind of input will be used; Arduino board with sensors such as two pyroelectric (passive) infrared sensors (PIR), one light sensor (photoresistor), and one distance sensor. The next photo (Fig. 4.3) shows the board with its all sensors, resistors and connections I have used for the project (Fitzgeralnd and Shiloh 2013).

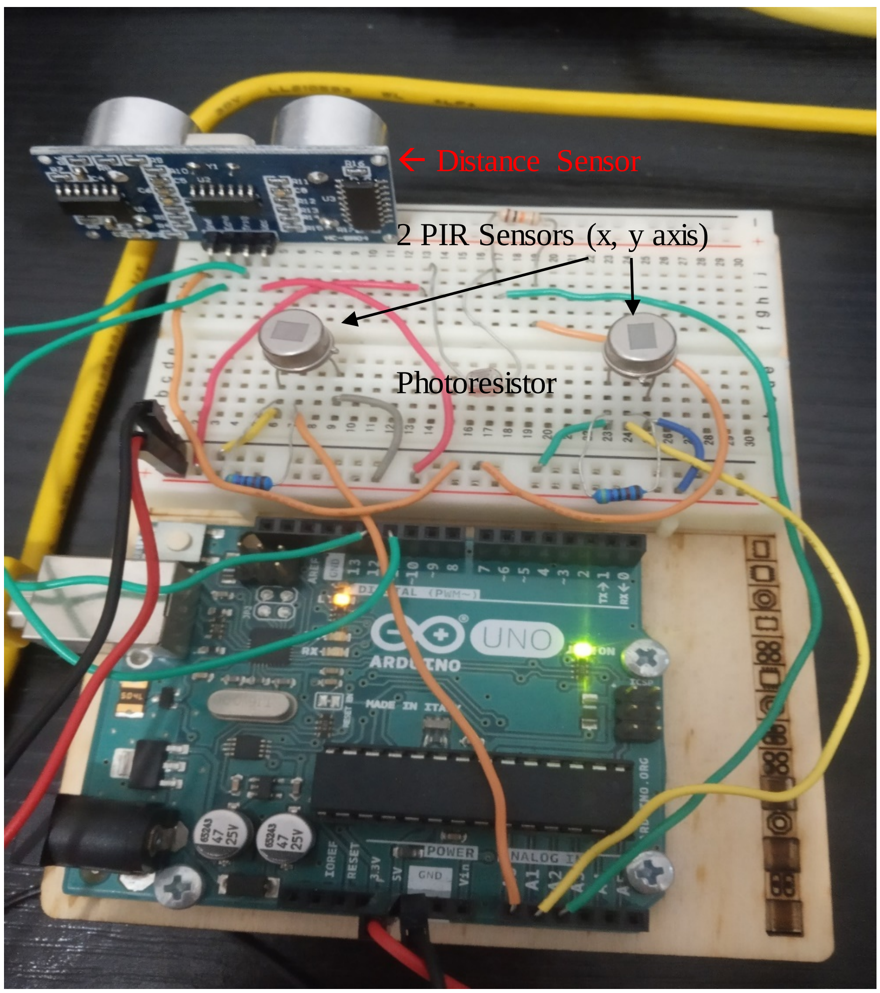

Fig. 4.3: The Arduino board with the sensors used for the project.

This project is made of six distinct Max/MSP patches and each one has specific computing tasks to carry: 1) Arduino input (data from sensors) and some programming for sending (send objects) parameters automation over time for all the other patches, 2) Stochastic noise objects and synthetic sounds (mainly for the training period time), 3) Modeling of a harp instrument with various rhythmic patterns capabilities that can change in real-time, 4) Machine Learning patch that uses self-organizing map (SOM), 5) Machine Learning patch that uses Adaptive Resonance Theory Neural Network and Multi-Layer Perceptron Neural Network, 6) Final mix patch with volume track control, equalization and dynamics processing for the Master output.

A calibration period of a few seconds using my external object *calibrator* is necessary before the training time starts, so that the sensors can adapt to the new space conditions. A short video demonstration of the PIR sensors can be viewed on the next video example (Ex.06 Arduino & PIR sensors.mp4) where I use the Vector Synthesis patch from the book *Designing Sound* (Farnell 2010, 279) as a test tone. A PIR sensor "measures infrared (IR) radiation from objects in its field of view. All objects emit some low level radiation."(Passive infrared sensor 2018). These infrared detectors are used to detect human body movements in a specific range, which is depended on the sensor's spec. Their "output is usually simple digital signal" (Passive infrared sensor 2018), however, I have used their analog signal output for my machine learning study (Arduino Forum 2011).

Using a short gesture in front of the distance sensor the training time begins and the Max/MSP Machine Learning library starts receiving the input data. For both Machine Learning patches I utilized the ml.star library by Benjamin D. Smith (Smith and Garnett 2012). The first Machine Learning patch applies the technique of self-organizing map (ml.som Max object) which "provides unsupervised clustering and classification, mapping high-dimensional input data onto a two-dimensional output space, preserving the topological relationships between the input data items as faithfully as possible" (Smith and Garnett 2012, 1). In my *Machine Learning Study* the input data will be the data received from the two PIR sensors (x and y) and the ml.som object will produce their projection onto a two-dimensional plane. During the learning period, once the ml.som object receives an input data, a search is performed in order to detect and adapt the most similar node and its neighbors to represent the new input with the highest precision possible (Smith and Garnett 2012, 1). As we can see from the image below the clustering representation helps us to visually understand the moment when the training period can be turned off. After the training period, the input data from the PIR sensors trigger lists of the Neural Network's weights which will be extracted from the object's (ml.som) left output and will be sent to control the following parameters of a granulator: 1) starting time, 2) size, 3) pitch and 4) panning of the grains.

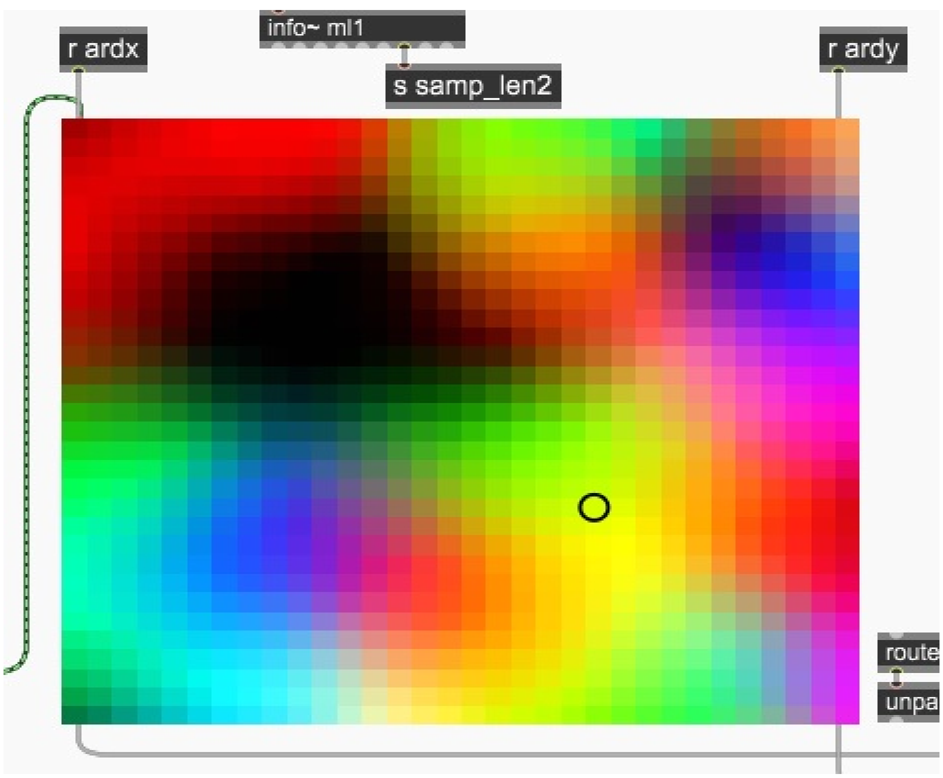

Fig. 4.4: When the colors start getting smoother the training period can be over. The black circle indicates the position determined by the PIR sensors (x and y).

The second Machine Learning patch makes use of a different approach which is called Adaptive Resonance Theory (ART). The training method is similar to the previous one with the self-organizing method (SOM), however, the nodes in ART don't influence one another during training. "While the SOM maps feature space in a continuous fashion across the map (i.e. intermediary points between nodes in the SOM network could be interpolated) the ART encodes a continuous area of feature space into each node" (Smith and Garett 2012, 2). The Adaptive Resonance Theory method (ml.art object) is used in conjunction with a Multi-Layer Perceptron Neural Network, the ml.mlp object, which is actually the external object that does the training and outputs the lists of data to be used to control another granular synthesis instance and the parameters of a stochastic sound synthesis object, dynstoch~ (Lyon 2012).

In Machine Learning, the perceptron is the smallest, very basic and important unit. An image example of a perceptron is shown in the diagram below (Amini 11:00):

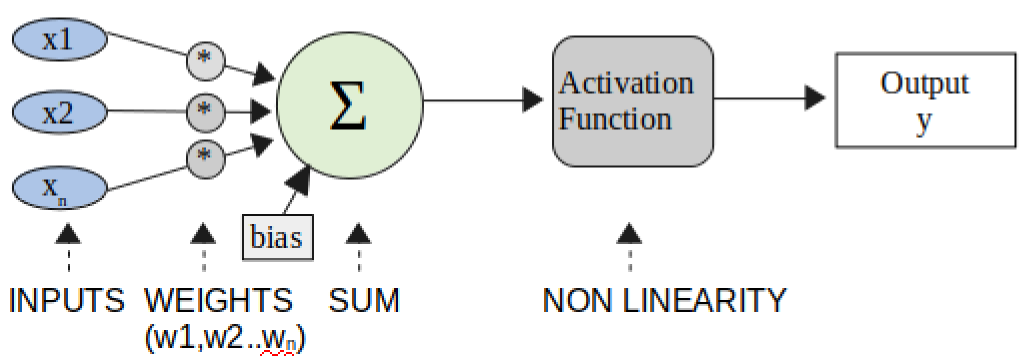

Fig. 4.5: The Perceptron, the starting point for building a neural network

The diagram above can also be expressed mathematically as (Amini 11:00):

$$y = \varphi \left(w_0 + \sum_{i=1}^{n} x_i w_i \right)$$

Where y is the output, φ is the activation function or threshold or non-linear function that introduces the necessary non-linearity, w_0 is the bias and x_i w_i are the inputs (x_i) multiplied by the weights (w_i).

The next step is to combine the inputs with many perceptrons together in order to form a basic two hidden layers neural network, which could be depicted in the figure below (Haykin 2008, 124):

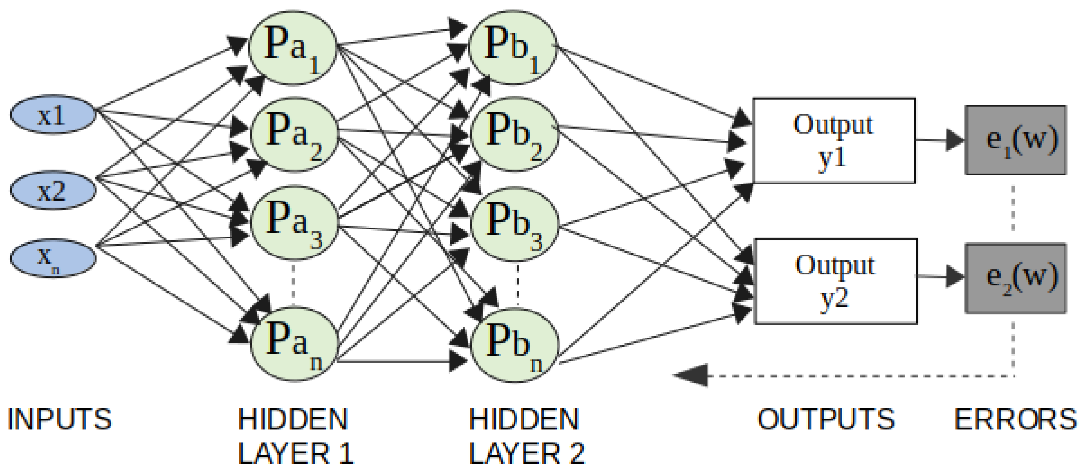

Fig. 4.6: The two hidden layers network with backpropagation training method.

Training and testing have a significant importance in Machine Learning. Backpropagation is the most efficient method for the training of a multilayer neural network, where two kinds of signals might be in our interest: 1) Function Signals and 2) Error Signals. Function signals are calculated as a function of the inputs and weights applied to our neural network. We will need to use the actual and predicted outputs in order to find out the error that will be used in the backpropagation process later on (Haykin 2008). There are different kinds of loss functions for calculating the errors, however, it is beyond the scope of this writing to analyze them extensively.

What was challenging while live mixing an interactive performance was to preserve a consistent master output with an appropriate dynamic range and balance of the spectrum of the sound. For this purpose, I have used the library of Max/MSP objects (Nikolopoulos Linux4Max, 2020) which I have developed from open source high quality plugins. The Master bus output consists of a 4 band parametric equalizer, the fil4~ (Gareus 2019), a compressor for reducing moderately the dynamic range, the darc~ (Gareus 2019), a multiband compressor for reducing the dynamic ranges of specific frequency ranges, the zamulticomp2~ (Zammit 2019), and a limiter for avoiding clippings and audio distortion, the dplimit~ (Gareus 2019).

5. Formulations. An Audiovisual Work

Formulations is an electroacoustic music piece that deals with the composition, properties and formation process of glass. The work doesn't attempt to describe the formation process of glass in a specific order, but it has been nevertheless inspired by its particular formation stages such as heating, cooling, crystallization etc. Three elements are the basis for this composition: 1) different combinations of prepared piano clusters, individual piano notes and their signal processed derivatives, 2) various glass sound recordings that appear mainly processed and 3) physical modeled fire sounds. Generally speaking, the techniques used for the piece include a large variety of granular synthesis methods that become less or more dense during the evolution of the work.

The transformation or transition from the liquid state to the solid state takes place over a variety of temperatures, which is the glass transformation range. The glass is formed after a slow rate cooling process that causes the material to shrink (Varshneya 2016). This is exactly the property that could justify the slow pace way that the work begins with, after the characteristic rhythmic figure and chord of the opening played by the prepared piano. Glass processed recordings evolve slowly in time, while the prepared piano figures and chords preserve their sparse appearance. The rhythmic activity starts increasing after the first minute. As we can see from the next figure, which displays the first two minutes and forty-five seconds of the piece, the tempo and beat tracking in the second half depicts more sharp changes than the first half of the piece.

Another conceptual element, which is closely related to my work, is the classic definition of glass by the physicist W.H.Zachariasen. Regarding its atomic structure, glass is a "three-dimensional network of atoms forming a solid that lacks periodicity or ordered pattern" (Varshneya 2016). Periodicity and generally any kind of proportion have been restricted either in terms of the form or in terms of harmonic, melodic and rhythmic figures. Although there are some sonic events played by the prepared piano that are repeated in different sections of the piece, they will be structured within a different context (see page 11 about sonic events). Rapid or sudden changes in form and inner musical elements of the piece correspond to the crystallization stage and are inspired by the kinetic theory "that any substance can be brought to glass if it is cooled rapidly enough" (Varshneya 2016).

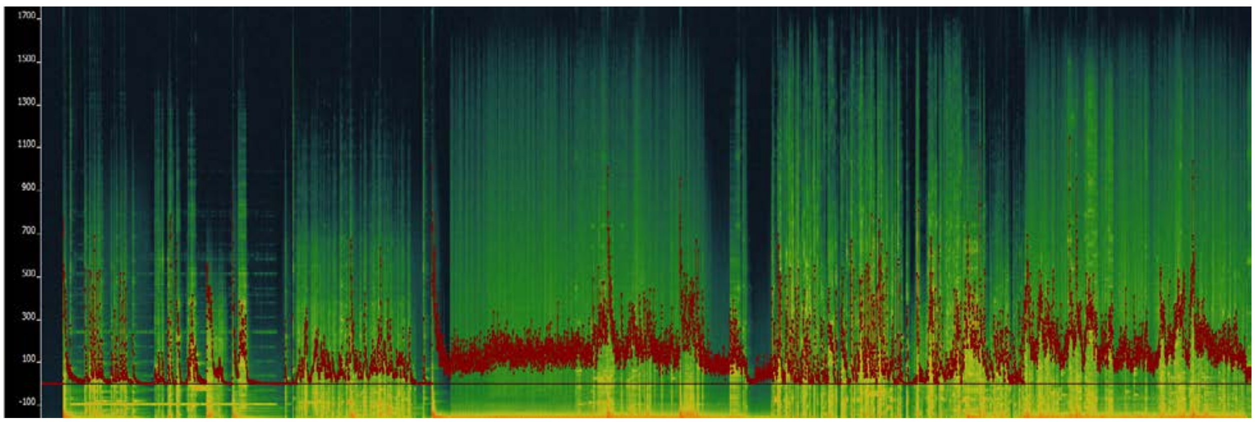

Fig. 5.1: Formulations (first 2 minutes and 45 seconds): Spectrogram with a Tempo and Beat tracking analysis (plugin by Queen Mary, University of London).

Another structural idea, which derives from the general properties of the glass, are the alternations between the states of relaxation and tightness of this composition's structure. When the molten liquid is cooled through the transition range and changes from liquid to a solid, then its structure relaxes (Varshneya 2016). Observing the random atomic order of the glass, the atoms are packed less densely than in other solids such as in the crystals (Varshneya 2016). We will notice by listening to the work that there are moments of relaxation when very few things happen and a simplicity is achieved by using less dense structures. On the other hand, even when the structure gets denser there is always ample space between the different layers that allows them to 'breath'.

Furthermore, there is a correspondence between the states of relaxation and tightness in the form of glass and the less or more dense clusters played by the prepared piano. The pitches used for the cluster formations are not as important as the 'weight' of these particular clusters. The main functionality that all the vertical chords have is to create less or more dense moments. The choice of the prepared piano could perfectly serve this purpose, because the mixture between a large range of pitched and percussive sounds offers more control possibilities over the moments of density. In addition, these rhythmic figures don't serve any macrostructural concept and for the most of part they appear for a very short time. They have only been used in order to rapidly increase the energy in some particular sections of the piece and not to drive us to a specific direction or a passage to a new situation. There are parts that have a more pointillistic structure, and other parts with more rhythmic or static structure through an excessive sound stretching. Nevertheless, all sound materials are heading together in a direction where they are merged into each other and they lose their initial substance in a similar way that the molten liquids do.

Glass fracture, is another element that has been used in specific moments of the work in order to increase the tension spontaneously. Sounds of glass fracture and fragmentation belong to the same category with the prepared piano active figures, because both groups aim to increase the energy of sound rapidly. Eventually, after almost every rapid increase in tension of sound follows the relaxation in structure.

At this point, the concept of constant flux (discussed in the introduction) between unreal and real were used with the intention to provide a degree of cohesion between the visual and audio materials. As for the audio part itself, there is a continuous movement between two merged entities: a) the distinguishable sounds (short figures played by the piano, glass sound recordings etc.) and b) the undistinguishable sounds as a result of various forms of signal processing. The visuals most of time are reactive to sound; however, this is only a surface characteristic that mainly serves as a marker or orientation between the different parts and less as a structural element. The continuous movement from the complete representation to the complete abstraction (see *becoming* page 11) and vice versa is the most prominent interrelationship between visuals and sound, and is happening in a way that we find ourselves crossing the property lines of these two 'worlds' either tending towards one direction or another.

The material properties of glass - and other non-musical materials as well - are indeed a familiar point of sonic interest in electroacoustic music since the time the electronics technology were introduced in musical composition practice. For instance, Pierre Schaeffer (1910-95), the pioneer of musique concrète, is one of the precursors who introduced the manipulation of recorded sounds through the philosophy of organized sound. With regard to my work *Formulations*, Curtis Roads (1951), as a composer and author, was the most important influence. From an aesthetic point of view, sound material, transformation and organization are the pillars of my music compositions. As Curtis Roads writes in his book *Composing Electronic Music* "Just as the molecular properties of mud, thatch, wood, stone, steel, glass, and concrete determine the architectural structures that one can construct with them, sonic morphology inevitably shapes the higher layers of musical structure" (Roads 2015, 17).

Although the visual part contains clear references to fire element, multidimensional glass structure, heating and cooling process, it doesn't strictly follow the idea of representation, nor a technique of a complete abstraction. The conception of the relation between sound and image follows the idea of the *nuances of the intermediate* (Fig. 5.2). The idea of *intermediate* or *boundary* (see page 11) can be understood as a discrete line between two different 'worlds'. The 'world' from the one side reflects the complete representation, while the 'world' from the other side reflects the complete abstraction. The distinct separation of these two systems occurs in space. The degree of blending of these two 'worlds' mirrors the concept of *nuances or hues,* which happen in time. The idea of the constant movement and variation of distance in the three-dimensional plane (figure 2.1, page 6) as discussed in the introductory chapter before, is applied here visually and aurally as well. However, although audio and visual parts are well synchronized throughout the whole piece, in a certain degree this audiovisual work didn't succeed in achieving its primary goals of conveying this exactly constant movement between real and unreal, the easily recognizable and the abstract.

One possible explanation could be the fact that composing for sound and image we need to consider how their combination influences and alters the relation between our aural and visual perceptions. According to Michel Chion in his book *AUDIO-VISION* "We never see the same thing when we also hear; we don't hear the same thing when we see as well" (Chion 1994, XXVi). I personally think that although the sound part in my work *Formulations* worked itself well within its compositional context, it couldn't have an impact on the listener's perception and reveal an image that we would not otherwise see. Probably a good strategy for the empowerment of the relation between sound and image could be the deployment of methods such as the *masking* and *forced marriage* methods. The deconstruction of the synchronism through *masking* might be an efficient way to influence the listener's perception by "moving alternatively from a diegetic state to a non-diegetic state" (Torelló and Duran 2014, 113). Following the *masking method* I could potentially mask the sound sources while creating focus on the visual elements and the opposite in different parts of this audiovisual work. Furthermore, using the *forced marriage method* in some other parts I could change the characteristics of sound noticeably while keeping the visual elements uninterrupted; "This distorts and alters the original relationship between the audiovisual elements and exposes the random relationship that sometimes exists between the two elements within audiovisual language, opening up a space for creativity in the analysis" (Torelló and Duran 2014, 115). Although these methods could improve significantly the relationship between sound and image, in the framework of my current audiovisual work there is a concern whether all of these methods and strategies could keep the character of *Formulations* unchanged without introducing a musical dramaturgy which is often better suited for films.

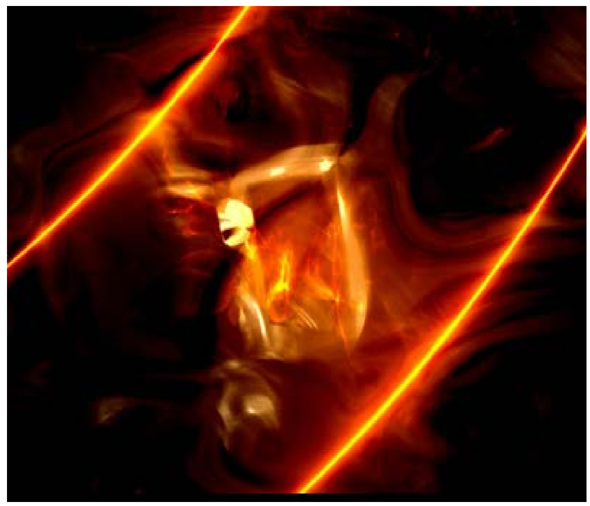

Fig. 5.2: Two of the visual instants that represent the idea of the *nuances of the intermediate*.

Technical Analysis: Formulations

Formulations is a fixed medium audio music work that also has an audiovisual version. A versatile set of tools and methods have been used during the composition process of the piece. From samples and audio recordings to Max/MSP/Jitter, STK (Cook and Scavone 2019), Paulstretch (Nasca 2019), Cecilia (Bélanger 2019) and a Digital Audio Workstation (DAW). All the individual prepared piano samples were organized in samplers and cleaned up with filters before can be used with MIDI input. Libraries for Max/MSP such as MUBU and Catart , both developed by IRCAM, have been used in order to process glass sound recordings with granulators and a variety of randomness. Paulstretch and Cecilia have been used to process and stretch the sound further, while a DAW was very helpful in organizing all the small pieces of sound and processing them in dozens of different tracks.

The method followed for the composition of this piece was not a linear process. Different parts were made in different times and composed with other parts in a non-linear fashion. The step by step ordinary compositional method was out of my interest, since creating a sense of irregular arrangements was my primary goal. After the editing, the project ended up with around sixty tracks for mixing (figure 5.3 shows only a part of this work). For the mixing part (equalization, dynamic processing etc.), I worked in Mixbus 32C, which is a DAW based on Ardour with the important difference that Mixbus has integrated an exact emulation of the analog circuits of the Harrison 32C parametric four-band equalizer for each channel, and additional stereo summing buses. The mixing process consists of moderate regulations in volume (faders), equalization, compression and eventually summing many tracks together into the analog style summing buses, forming groups that are necessary for the final mix adjustments.

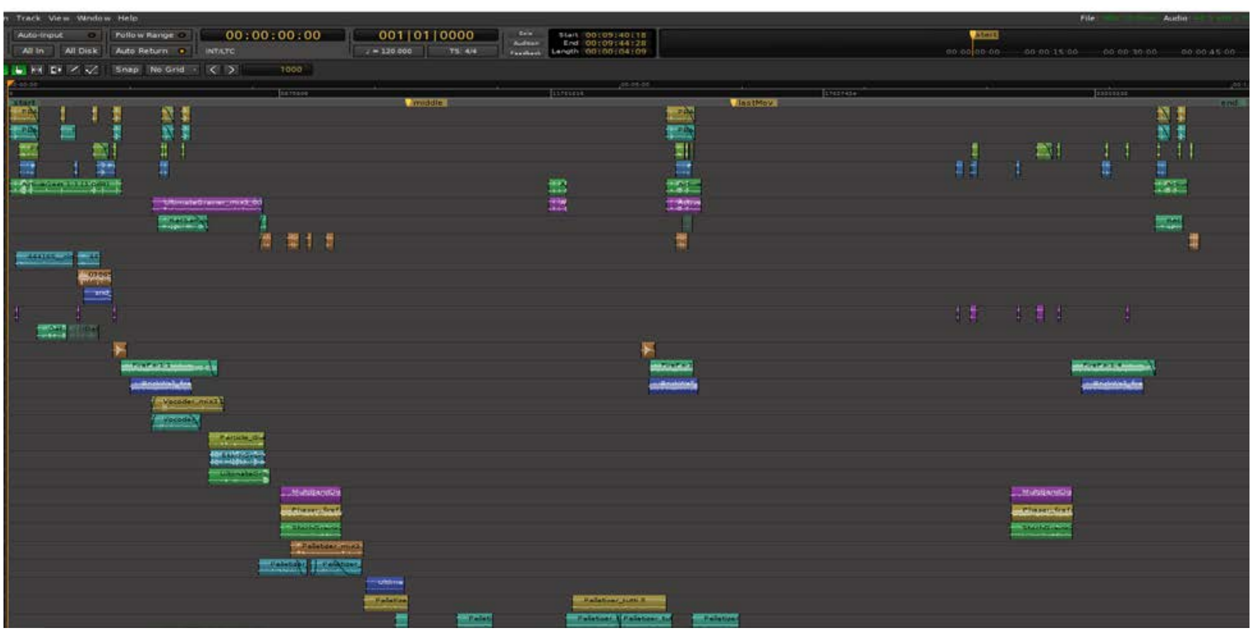

Fig. 5.3: A few of the total sixty tracks in the editor of Mixbus 32C.

Figure 5.4 provides an overview of the analog style channel strips and their summing buses. The summing buses are also an exact emulation of the buses that can be found in Harrison analog consoles and they differ a lot from the auxiliary buses which are widely used in many other Digital Audio Workstations. The approximately sixty audio tracks have been grouped in seven separated sub-mixes in order to apply some extra moderate dynamic processing and volume balance control for each category of sound. These groups (sub-mixes) consist of the prepared piano instrument sounds, glass recordings, sounds of fire and four distinct granulator groups. The prepared piano samples come from two different resources (Cuckoo 2013 and Werner 2011) and I have edited, cleaned up and organized them into multisampler instruments in order to use them for this work.

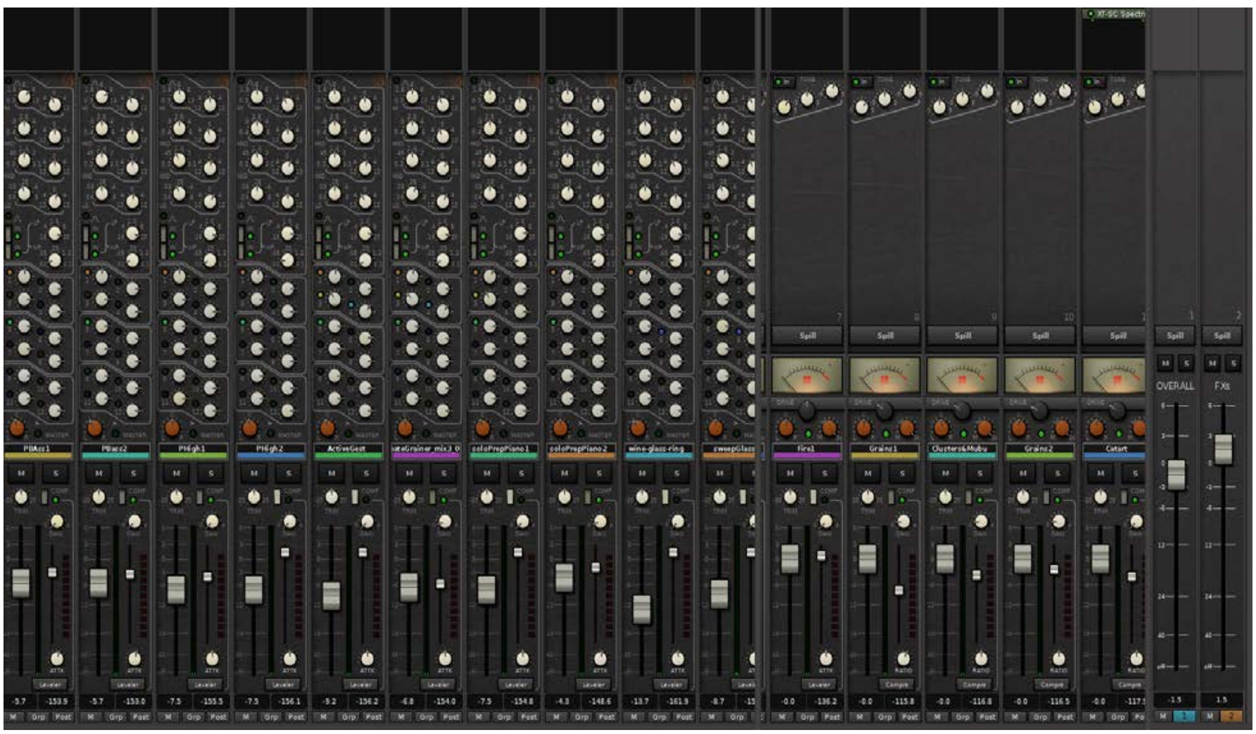

Fig. 5.4: A part of the mixer in Mixbus 32C with channel strips (left side) and summing buses (on the right) in an analog mixing style console.

6. Turmoil I and II

Both of the music composition versions, Turmoil I and II, are an attempt to approach the philosophy of the unexpected and foreknowledge in our daily life from the perspective of sound. It is based on the idea that the future doesn't exist and therefore is non-actual. Thinking about the past and present differs from thinking about what is expected or predicted to happen (Currie 2013). Turmoil I and II are an effort to attribute the idea of provisionality of the future and how the expectations take different forms and paths with the passage of time. Both works contain the concept of surprise which in musical terms is perceived as an abrupt differentiation in terms of the sound energy and instrumentation. The foreseeable or expected is very different from the tense future which is "something that lies ahead and yet which is already complete, not what will happen, but what will have happened" (Currie 2013, 1). This concept is closely related to the philosophy discussed in the introductory chapter and the *Six Paradoxes*. Nevertheless, it belongs to another category of paradoxes, the epistemic paradoxes. They are riddles which contain conflicting answers and while they are evolving in time these paradoxes lead us to make corrections of our errors. For instance, foreknowledge and freedom are two conflicting statements that govern the whole spectrum of our everyday life and creativity. "Prior knowledge of an action seems incompatible with it being a free action." (Sorensen 2018).

Therefore, it is still a question if and how the element of surprise can be actualized in music and generally in art through the repeated listening of a music work. That happens because our memory and intention are inevitably involved in the analytical listening and artistic creation process. Therefore, I use the terms 'surprise' and 'unexpected' metaphorically and not literally. Eventually, these terms acquire mainly a structural value for the piece. There are two versions, Turmoil I is a pure electroacoustic version and Turmoil II is music for a string orchestra, percussion and tape. Although Turmoil I and II are distinct versions, there is an important point that brings both versions into a common conceptual framework. That is the dialectic relationship between the real and unreal instrument sounds (see *becoming* page 11), which determines the interaction between these two contrasting elements. Considering again my three-dimensional plane (Fig. 2.1, page 6) in the introductory section, these elements belong to the Y axis. The use of a real string orchestra and percussion in the second version comes to intensify this particular dialectic relationship between real and unreal sounds where the tape contains recognizable and extremely processed sounds, while the orchestra exhibits a wide range of playing techniques moving also from real to unreal sounds, and vice versa.

The electroacoustic version includes unaltered acoustic instrument sounds such as pipe organ, various percussion instruments and audio recordings (such as baby piano toys), whose forms have become indistinguishable after some signal processing techniques have been applied to them. The pipe organ has a commanding presence in two parts of the piece. The first is the low end opening and the second comes two minutes later as a variation of the first. The low frequency tones, made with Aeolus and OpenMusic software, are intended to raise questions about our foreknowledge and the conflict between freedom and foreknowledge; the active figures, which come after the pipe's organ continuum, are expressing the 'unexpected' events. There is an audible conflict between the pipe organ's sounds and the processed active figures, a conflict that evolves in time without signs of resolution.

The form of the work consists of two distinct sections. The first section extends up to the sixth minute and twenty seconds, when the percussion instruments appear for the first time 'unexpectedly'. The first section seems to have a periodic structure with the continuum layered sounds succeeding the active figures described before and vice versa. As the figure below depicts, the melodic range spectrogram and spectral power analysis show the moments when these alternations occur. We can also see that the continuum parts occupy a fairly prolonged period of time. The purpose of that is to prepare the listener for the next 'unexpected' event. The third continuum (also shown in figure 6.1) lasts almost two minutes and has the intention to boost the sense of the 'unexpected' event once the percussive transients hit on the beginning of the second section (06:25).

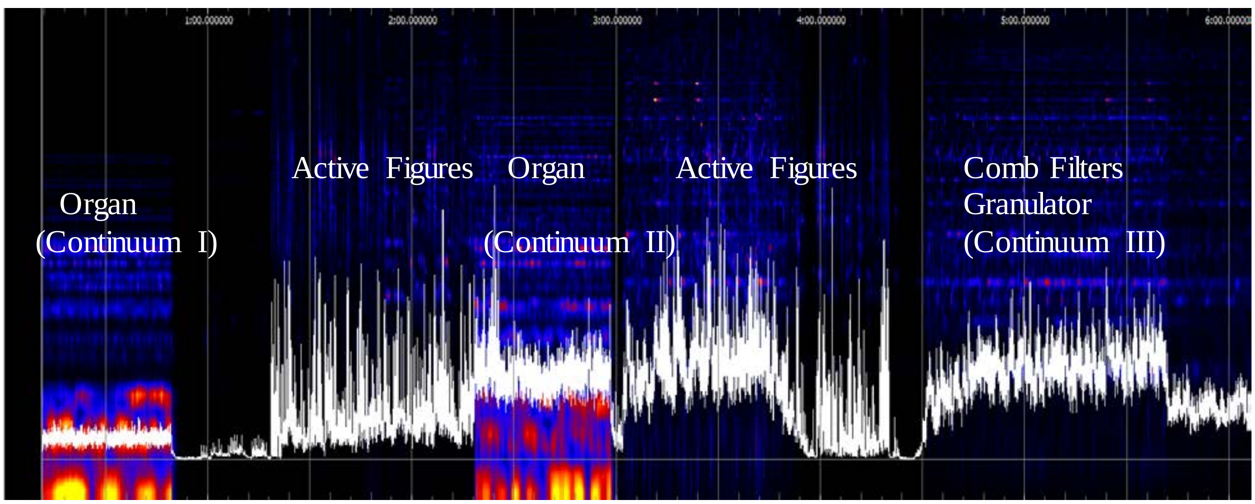

Fig. 6.1: Spectrogram and Spectral power analysis of the first six minutes.

Turmoil II, the second version of this composition, involves acoustic instruments; string orchestra, percussion and tape. The string orchestra's instrumentation follows a specific direction from the singularity of the beginning to the multiple divisions later on. The main elements from the side of the acoustic instruments are the very fast pizzicati, bowing techniques that produce a range of noise from whispering sounds to higher volumes of noise, and groups of strings carrying microtonal clusters. The whispering sounds category contains both the various right-hand bow pressure techniques and the half harmonics techniques (left hand) as well. The very fast string pizzicati form fugal passages in stretto (Fig. 6.2) that don't allow the listener to perceive them as melodic or rhythmic structures due to their speed and density. The only pitched sounds that will be audible by the vibraphone will appear in particular parts of this work and they have been used only structurally in order to stress the appearance of 'unexpected' events later. The most parts of the work utilize bow techniques with a varying range in pressure or other techniques that result mainly to whispering or masses of undefined pitched sounds. Therefore, it would be undermine our understanding for the piece to rely only on a MIDI mock up, no matter the quality of samples used.

I would like to mention here some of the musical precursors that influenced my work and the music score writing. Regarding the bowing and generally other string playing techniques influences can be found in composers such as Gérard Grisey (1946-1998) and especially his work *Partiels* (1975) which is based on the sonogram analysis of a single tone. His innovative use of the bow pressure for the production of sounds rich in overtones inspired me to think how these different qualities of sound could become a structural element in my work. Another important influence was Helmut Lachenmann's (1936) work *Air* (1968 - 69) for large orchestra with percussion solo. The latter was an extensive source of ideas about writing for string orchestra and experimenting with non-traditional playing techniques.

With regard to microtonal clusters I extensively use, there are two works that I would like to list here. The first one is the composition *Gruppen* (1955 -57) for three orchestras by Karlheinz Stockhausen (1928 – 2007). The way Stockhausen used micro-intervals and sonic masses in his work had motivated me to think of the idea of a gradually increasing strings section division incorporating groups of microtonal clusters. György Ligeti (1923-2006) and especially his work

Clocks and Clouds (1972-1973) for twelve voice female choir and orchestra had also a significant impact on my music work. Although this work was less influential in terms of harmony and melody, I have been mainly interested in the ways Ligeti introduced the micro-intervals to his harmonic and melodic structures.

For the synchronization purposes between Orchestra and Tape, a separate player for the tape, perhaps using a DAW, is necessary. The tape player is intended to make use of the numbered tracks and follow the conductor according to the music score sheet. In that way, the tape player becomes an integrated part of the orchestra and the conductor is not responsible for an additional tape-following task.

Fig. 6.2: The fugato opening of the strings. The synchronization among the string players is not important here as long as they start and end the phrase according to the notation.

7. Fossils

Fossils (2019) is an electroacoustic music work that deals with the history of life on earth and the evolutionary significance of once-living organisms' remains and traces. The word fossil comes from the Latin word *fossus*, meaning "having been dug up" (Evers 2013). The main musical ideas derive from the basic distinction of the fossils in two categories, the microfossils and the macrofossils.

Small sonic fractions dominate almost for the largest portion of the piece and they vary in terms of spectrum range, intensity and speed. The texture and the form of the piece follow a structure of simplicity. There is a type of *monophonic* musical texture that occasionally integrates an additional layer of *macro figures* which oppose the constant movement of the *micro figures*. The whole piece could be characterized by symmetry and duality (as explained in the introduction and page 11). The symmetry refers to the general structure of the work with its distinct compressions and rarefactions and duality refers to the opposite elements of micro and macro figures. The micro figures could be further classified according to their frequency range and the density of their presence.

For instance, we can see on the next figure (Fig. 7.1) that the micro figures focus on a mid-frequency range, while afterwards (Fig. 7.2) they cover a wider range from mid to high fundamental frequencies. There is no transition from one phase to another but an evolution which consists of various contrasting motivic phrases. The gradual change is occurring within a framework of randomness and at the same time has a deterministic order upon which the form is built. The speed of change of these instantaneous micro figures gives an impression that there are considerable repetitions. As we will see next, the cyclic structure of the piece integrates the element of repetition as well.

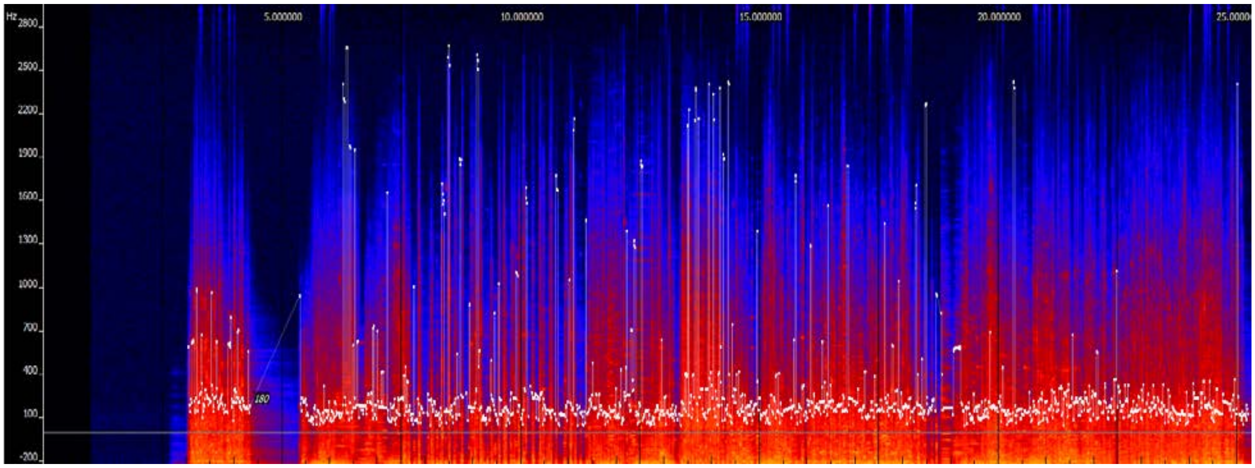

Fig. 7.1: The first 25 seconds with focus on mid frequency range.

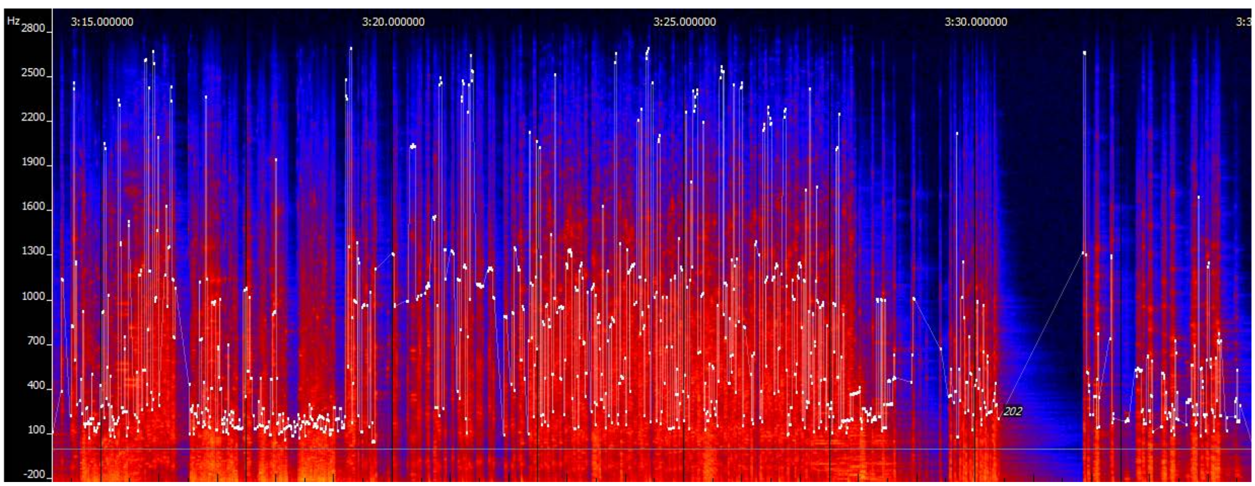

Fig. 7.2: A different part of the piece where there is a spread to a wider frequency spectrum (mid to high frequencies).

Additionally, other parts of the piece (Fig. 7.3) appear to focus on a higher frequency range with a less dense texture unlike other parts (Fig. 7.4) that focus mainly on mid to low frequencies and contain more active *micro figures*. The stretched sonic layers or *macro figures* appear occasionally and will be unified with the instantaneous figures as the soft sediments and fossils do during the fossilization period. In figure 7.4 we can see a certain escalation of density of the *micro figures* and the spectrum exhibits a particular shift to a noisy spectrum. However, it would be incorrect to describe it as noise because the low to mid fundamental frequencies are still standing out.

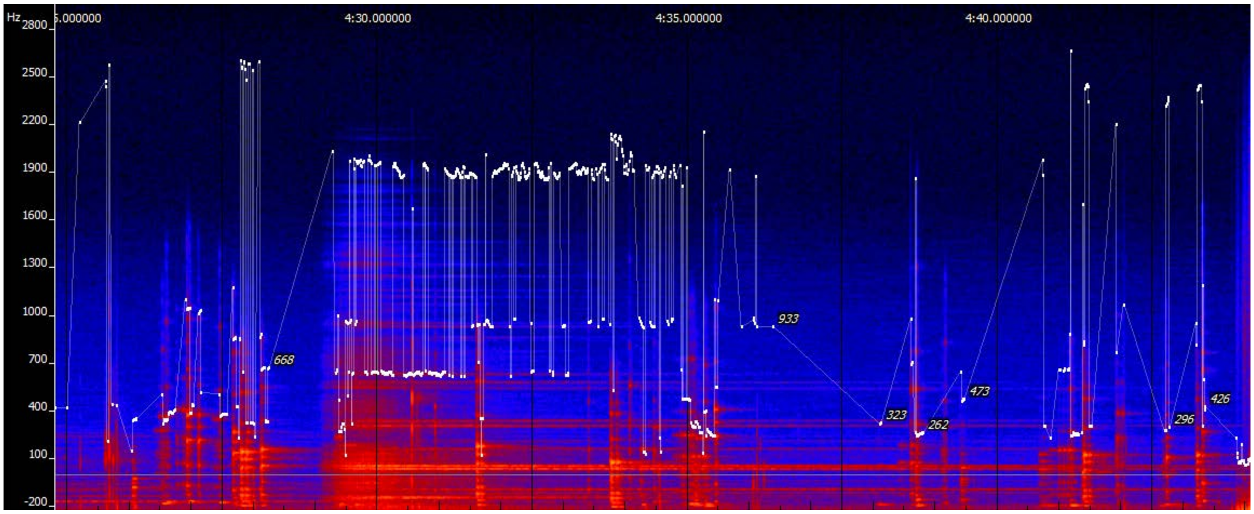

Fig. 7.3: A less dense part of the piece where high frequency figures coexist with the stretched sonic layers.

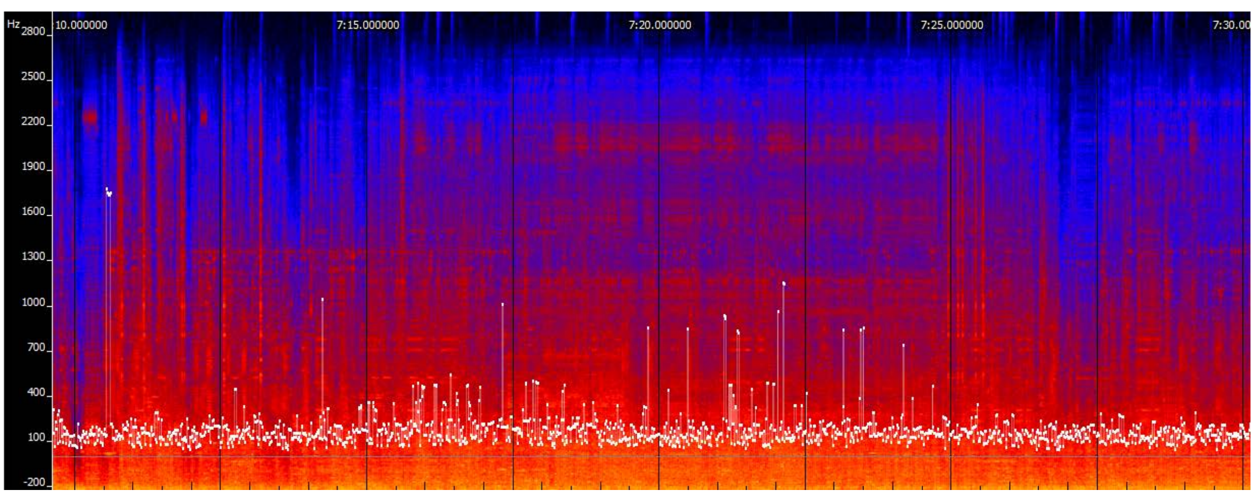

Fig. 7.4: A shift to a noisy spectrum with a focus on low-mid frequency range.

With regard to the form of this work, we could describe its structure as composed of concentric circles whose perimeters correspond to the different lengths of the parts that make up the whole structure. Their common center correlates with the initial idea of the *micro* and *macro figures* and their areas with their frequency spectrum, density and repetition degree.

The entire piece consists of six sections as we can see in figure 7.5 and each of them takes the form of a duality; The *micro figures* are the instantaneous variations that come in contrast to their subsequent *macro figures*. Although the musical materials used to express the duality are identical, each section diverges from others in a fundamental way, which is closely related to the concentric circles scheme mentioned in the previous paragraph. The opposing musical gestures (*micro* and *macro figures*), metaphorically speaking, are the product of input data, gestural composition and interaction between human input and a computer, which is programmed to respond to the received data and control a number of audio processing parameters.

1st	2nd	3rd	4th	5th	6th
0:00 – 0:42	0:42 – 1:42	1:42 – 4:44	4:44 – 5:58	5:58 – 8:10	8:10 – 9:11

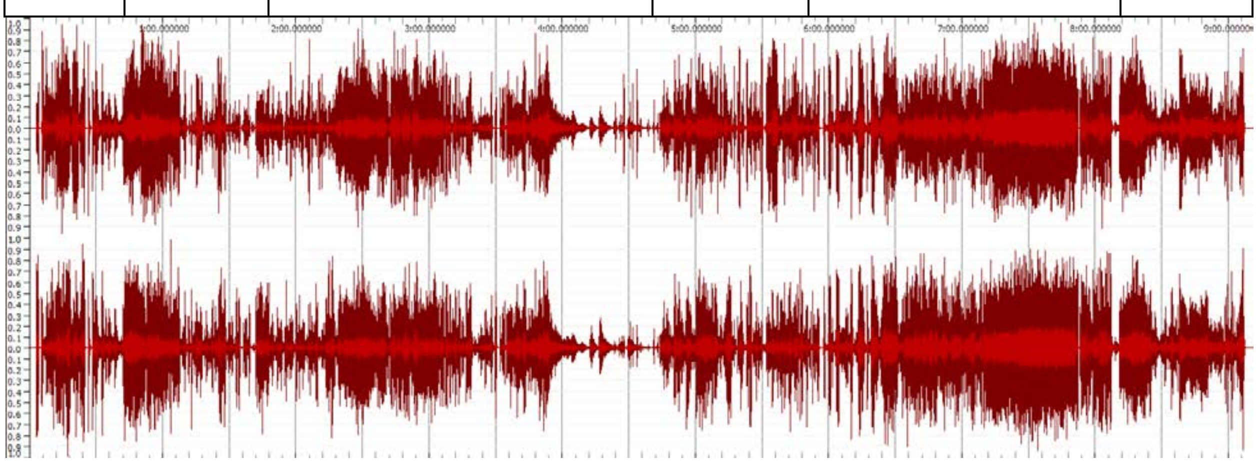

Fig. 7.5: The sections and their lengths that make up the work Fossils.

One method I incorporated to my work *fossils* for creating an interactive environment is the use of images imported to the software IanniX. "IanniX is a real-time "graphical open source sequencer for digital art" ("What is IanniX? | IanniX."), based on Xenakis's *UPIC* (Bourotte 2012)." (Scordato 2017, 2). The real time data will be received in the Max/MSP where unsupervised machine learning will process the data further. In Max/MSP I followed a similar approach like in my *Machine Learning Study* using Adaptive Resonance Theory Neural Network and the ml.star library (Smith and Garnett 2012). For the communication between IanniX and Max/MSP I used the OSC and User Datagram Protocol (UDP). The picture of fossils (Jurassic Coast Trust n.d.) we can see below (Fig. 7.6), is one example I used in order to produce the real-time input data for Max/MSP. Since the trajectories are not very clear together with the picture, the next figure (Fig. 7.7) shows clear the most important elements I utilized in Iannix software: 1) Cursors, 2) Curves and 3) Triggers.

Fig. 7.6: A picture of fossils imported to IanniX software as a background picture.

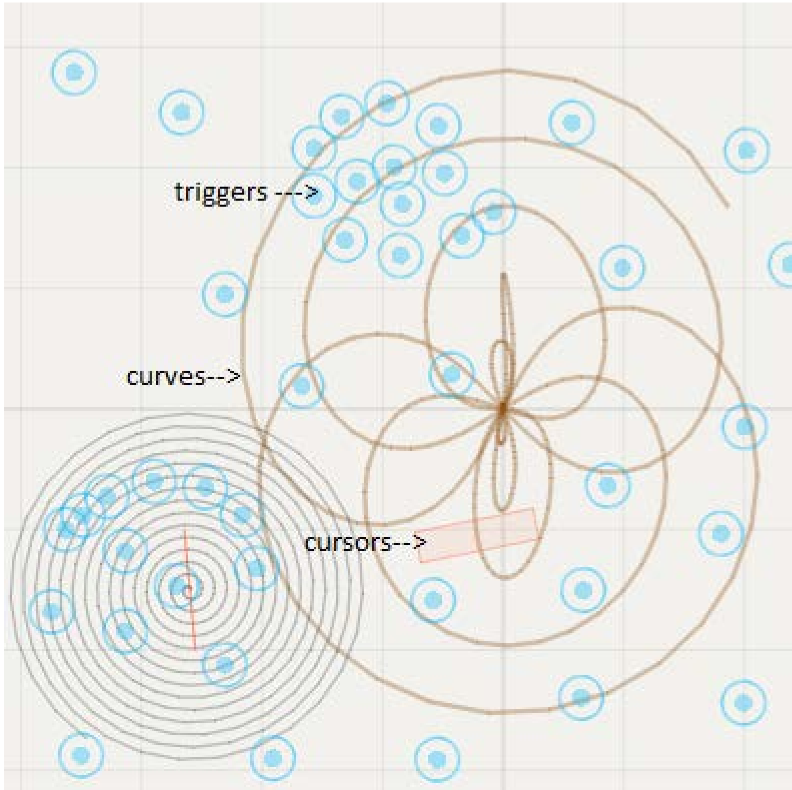

Fig. 7.7: A provisional plane of cursors, curves and triggers.

In fact, the elements that actually output data are the cursors and triggers. Cursors are time-based elements that can move along the trajectories of curves in different ways such as one run on curve, loop on curve, loop roundtrip on curve and many others. Cursors can have different lengths and widths. Triggers output midi messages every time a cursor passes by (Scordato 2016). I have experimented with different types of curves, cursors, speeds and motions. The patch example in the next figure (Fig. 7.8) shows how the lists of data have been routed and sorted in order to get the x and y axis we need. Furthermore, the patch shows just one way of using interchangeably cursors in random orders. The midi notes (triggers) will be converted to frequencies and then will be used as a pitch deviation for the granular synthesis in the next stage.

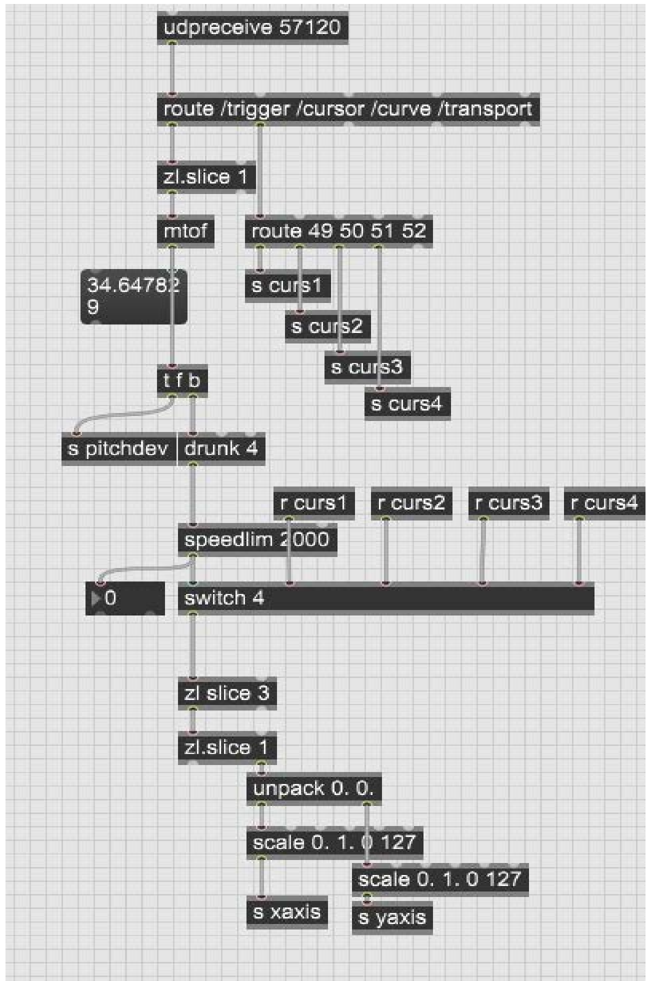

Fig. 7.8: The input data from IanniX to Max/MSP.

Once we retrieve the x and y axis data, we will use them for the machine learning stage (Fig. 7.9) in a similar way described in the *Machine Learning Study* using the ml.art and ml.mlp objects (see page 46). Contrary to the *Machine Learning Study* work, *Fossils* don't involve real-time audio and improvisation. Different audio takes have been recorded and with a Digital Audio Workstation these audio tracks have been edited, processed further and eventually mixed.

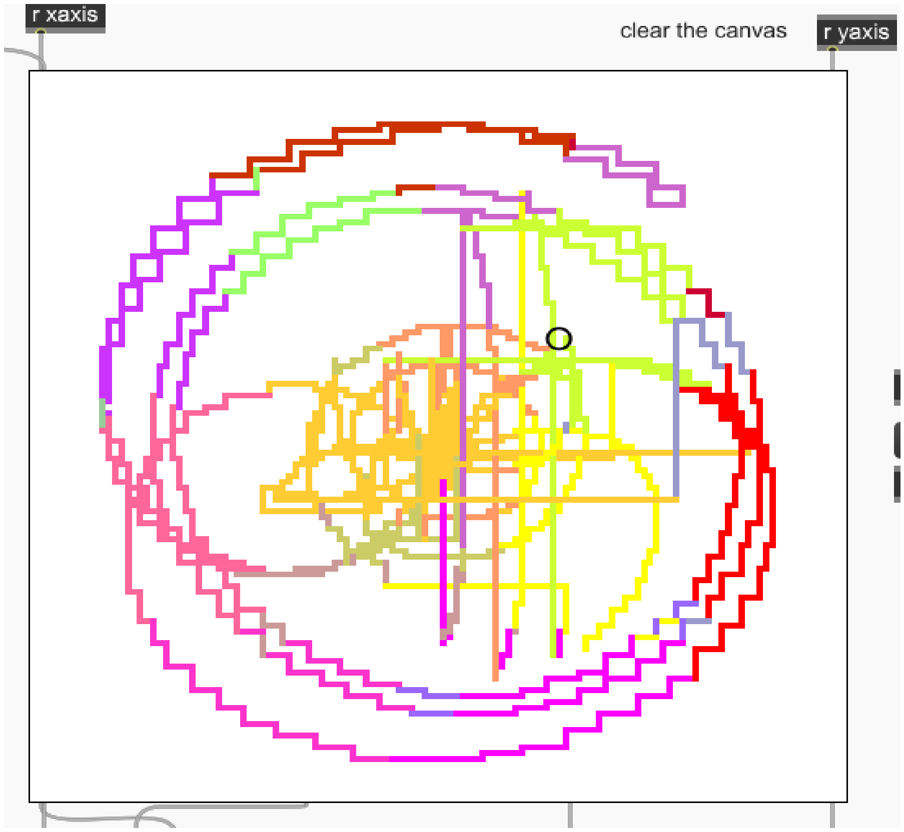

Fig. 7.9: Drawing on ML canvas with input data received from IanniX.

8. Conclusion

The present portfolio of musical compositions is the outcome of research which combines different academic disciplines together with a strong focus on music and computer music applications and technology. Intellectual curiosity and creativity that has involved unlimited experimentation were important motivational factors for the completion of this research. I have drawn ideas from Deleuze's philosophy in order to compose the *Six Paradoxes* and his work *The Logic of Sense* was in the epicenter of my study regarding the paradoxes. All the *Six Paradoxes* make use of ambisonics and audio programming with Max/MSP. I have experimented with Arduino, sensors and machine learning libraries to create music that is based on real-time audio and improvisation. I have combined computer generated graphics with electroacoustic music for composing the piece *Formulations*. I have experimented with music for string orchestra and various string playing techniques in combination with electroacoustic music (tape) for my works *Turmoil I and II*. I have composed computer-generated music emphasizing on sound synthesis and also fixed media electroacoustic music works with popular music influences.

At this point, I would like to refer to the disciplinary context of computer music as described by Richard Moore on his book *The Elements of Computer Music,* where he states that while the "most academic disciplines are disciplines of thought", computer music "is strongly interdisciplinary" (Moore 1990, 23). Especially nowadays with the enormous advancement of music technology, I solidly believe that an interdisciplinary approach to musical composition, and other artistic fields as well, is a very important and necessary aspect of our better understanding of the past, present and future of our artistic activity. Indeed, there are specific downsides dealing with a wide range of scientific topics, however, the positive elements definitely prevail over the negative ones. That is possible to happen in art related fields because I think, art has the unique power to combine expression and science in a way that science turns into a medium of expression as well. In other words, an interdisciplinary research and use of music technology in musical composition, and by extension to any other field of art, aims to broaden the horizons of expressive possibilities instead of restricting or automating the creative process.

A potential future research project will be probably related to a deeper study of Digital Signal Processing, audio algorithms and Fast Fourier Transformation operations, and the development of relevant external objects (library) for Max/MSP. It will also include sound analysis and synthesis of musical acoustic instruments, and thinking of ways to use them in new spectral oriented compositions. Sound design, sonic morphology, creation of computer-generated sounds and algorithms will be still the central point of my future study.

Interactive environments will be also an important part of my potential research area. It will have a purpose to design various types of interactivity between humans and computers by the use of sensors, video tracking and acoustic instruments, and also between different software environments that will be able to interact each other after being programmed by the usage of communication protocols. Projects of interactive media could have artistic and educational goals as well, because they can cover a large spectrum of learning by incorporating interactive experiences into the educational practice.

Furthermore, I would like to examine further the ways in which sound and creation with sound has impact on humans and the real world, not only physiologically but also in terms of perception. Layers of perception, their interconnections and how our perception is influenced by the cultural contexts will be probably an interesting area of investigation through my musical composition process and thought.

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Contents of the Digital Storage Media

Link: https://beardatashare.bham.ac.uk/getlink/fiQzc8SkTjDQo96C9d8gvToy/

Audio files: Stereo, format: PCM, Sampling Rate: 44.1 kHz, Bit depth: 24 bits

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1) Paradox I – III.wav, 2) Paradox IV.wav, 3) Paradox V.wav, 4) Paradox VI.wav

5) Machine Learning Study.wav, 6) Formulations(audio only).wav, 7) Turmoil I.wav

8) Turmoil II(Strings and Tape).wav, 9) Fossils.wav

Video file: Video Format: MPEG-4, Frame Rate: 30 FPS, Resolution: 1920x1080

Audio Format: AAC LC, Sampling Rate: 44.1 kHz, Compression Mode: Lossy

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Formulations(audiovisual).mp4

Video Examples:

Ex.01 Voice Sythesis Soprano (no lyrics).mp4

Ex.02 Voice Synthesis Choir(no lyrics).mov

Ex.03 Daisy Bell with lyrics.mp4

Ex.04 Festival Speech Daisy Bell.mp4

Ex.05 Procedural Audio.mp4

Ex.06 Arduino & PIR sensors.mp4

Max/MSP externals libraries (for Windows 64 bit operating systems and Max/MSP 6 or greater)

Linux4Max: 36 objects for dynamic processing, physical modeling, help files, source code.

Pd4Max: 40 objects from Pure Data, 3 abstractions from Pure Data, help files, source code.

Music Score (pdf format):

Turmoil II for Strings, Percussion and Tape