ISTANBUL TECHNICAL UNIVERSITY ★ INSTITUTE OF SOCIAL SCIENCES

CREATING SOUND MASS USING LIVE SOUND PROCESSING AND FEEDBACK WITH SYMPATHETIC VIBRATING STRINGS

Ph.D. THESIS

Gökhan DENEÇ

Department of Music

Programme: Music

Thesis Advisor: Doç. Dr. Can KARADOĞAN

SEPTEMBER 2015

İSTANBUL TEKNİK ÜNİVERSİTESİ ★ SOSYAL BİLİMLER ENSTİTÜSÜ

CANLI SES İŞLEME VE GERİBESLEME KULLANILARAK SEMPATİK TİTREŞEN TELLER İLE SOUND MASS TARZI SESLER YARATMAK

DOKTORA TEZİ

GÖKHAN DENEÇ
(409042004)

Ana Bilim Dalı: Müzik

Programı: Müzik

Tez Danışmanı: Doç. Dr. Can KARADOĞAN

EYLÜL 2015

Gökhan Deneç, a Ph.D. student of ITU Institute of Social Sciences student ID 409042004 successfully defended the dissertation entitled “CREATING SOUND MASS USING LIVE SOUND PROCESSING AND FEEDBACK WITH SYMPATHETIC VIBRATING STRINGS”, which he prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

Thesis Advisor:	**Doç. Dr. Can KARADOĞAN** Istanbul Technical University	....
Jury Members:	**Doç. Dr. Tolga Zafer ÖZDEMİR** Istanbul Bilgi University	....
	Yrd. Doç. Dr. Ozan BAYSAL Istanbul Technical University	....
	Yrd. Doç. Dr. Eray ALTINBÜKEN Istanbul Technical University	....
	Yrd. Doç. Dr. Burak TAMER Bahçeşehir University	....

Date of Submission : July 07, 2015
Date of Defence: September 15, 2015

To Şükoş'cuğum and Hasso'cuğum,
for making the impossible simply possible

FOREWORD

I would like start with thanking to my advisor Doç. Dr. Can KARADOĞAN, and the members of my jury, Doç. Dr. Tolga Zafer ÖZDEMİR and Yrd. Doç. Dr. Ozan BAYSAL, for their support throughout my research.

I would like to present my gratitude to Prof. Dr. Teoman TURGUT and Dr. T. Burak KAYNAK from Boğaziçi University Physics Department for their invaluable support and for letting me make use of their faculty resources, offices and studying environment.

If Lec. Tunç BUYRUKLAR didn't lend a helping hand to provide an old piano for my experiments and my colleague Taylan ÖZDEMİR didn't share his office for my study needs, I would not be able to finish this research. I must also express my gratitude to Prof. Şehvar BEŞİROĞLU for her full support and reserving one study room at MIAM for my research.

Also, I would like to thank to the following professors who had spared their precious time, whom I had attended their classes and had taken benefit of their guidance (in chronological order):

Doç. Dr. İpek Yada AKPINAR	İTÜ	Architecture
Doç. Dr. Alper MARAL	YTÜ	Music & Performing Arts
Doç. Dr. A. Taylan CEMGİL	BOUN	Computer Engineering
Yrd. Doç. Dr. Serdar ALTOK	BOUN	Mathematics
Doç. Dr. Burak GÜREL	BOUN	Mathematics
Prof. Dr. Hakan DELİÇ	BOUN	Electrical & Electronics Engineering
Doç. Dr. Burak ACAR	BOUN	Electrical & Electronic Engineering
Arch. Sertaç ÖZTÜRK	İTÜ	Industrial Product Design

Special thanks to following people and companies for support my research:
MASS (*Mehmet Akkaş Sound Systems*), Kağıthane – amplifier provider,
FERAY Alaşım San. Tic. Ltd. Şti.,Davutpaşa,– material provider,
Çağdaş AYSEV – For providing a silent, homey study environment,
Erhan SULUDERE (Senkop Müzik, Tünel), providing know-how on coil winding,
Starbucks Co. Etiler Shop & Staff – For providing a cozy study environment,
Karaköy Elektronikçiler Çarşısı, Karaköy Hırdavatçılar Çarşısı, çarşı esnafı.

And I reserve my warmest feelings to someone who is my source of inspiration, joy and happiness, who had always been my companion, to my lovely wife Dr. Evren Deneç who made my life meaningful. Last but not least, I give my warmest hugs to my lovely daughters Ege and Çağla, who simply turned my world to heaven.

May 2015

Gökhan Deneç
Sound Engineer

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ABBREVIATIONS

AC	**:** Alternating Current
ADSR	**:** Attack, Decay, Sustain and Release (Envelope Parameters)
AWG	**:** American Wire Gauge
BOUN	**:** Boğaziçi University, Istanbul
CCRMA	**:** Center for Computer Research in Music and Acoustics in Stanford University, Stanford
dB	**:** Decibel
dBFS	**:** Decibel Full Scale
DC	**:** Direct Current
G.o.T.	**:** Glossary of Terms
GUI	**:** Graphical User Interface
İTÜ	**:** Istanbul Technical University, Istanbul
K-S	**:** Karplus-Strong Algorithm
Pd	: Pure Data
SPL	: Sound Pressure Level
YTÜ	**:** Yıldız Technical University, Istanbul

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CREATING SOUND MASS USING LIVE PROCESSING AND FEEDBACK WITH SYMPATHETIC VIBRATING STRINGS

SUMMARY

From the beginning of the 20th century, music authorities had started indicating the need for exploring the physical properties of sound in compositional studies. Besides melody/harmony and rhythm, timbre which had not been considered as a significant musical property was seriously taken into account in musical composition.

Timbre was considered as a prominent feature in electronic music, as the instruments as well as the performing and composing techniques of this genre has matured; its unique nature bonded with timbre. Together with the increase in the search for new timbres, different approaches were established regarding the authentic source of sound material used.

Three fundamental sound sources of electronic music could be mentioned when the composition in the literature are examined:

1. Synthetic (artificial) sounds: Musics organized with sounds produced from various equipment which can oscillate waveforms that do not exist in nature; whose properties like frequency, timbre and envelope are designed.
2. Sampled (recorded) sounds: Musics organized by replaying the sounds whose authentic vibrations are captured previously and then used as authentic versions or be manipulated with various techniques.
3. Feedback: Musics organized by the resonances that occur when sound outputs are *fed back* in the sound inputs.

In a historical context, the first and the second methods were initiated and represented by the German school of Electronic Music (*elektronische musik*) and the French school Solid Music (*musique concrète)* approaches consecutively. However, the third method being feedback, cannot be explicitly related with a foundation or a group. It found its place in avant-garde compositions and performances.

This thesis is grounded on the synthesis of three musical concepts which seem to be unrelated, namely feedback, live sound processing and Sound Mass style.

Feedback is a phenomenon well known by artists, performers or technicians operating in the live sound business. It is caused by the controlled or uncontrolled electronic or acoustic connection of the input and the output of a sound system.

Live sound processing practices have taken the stage with experimental quests and creative use of timbre processing equipment. Any arbitrary acoustic sound or the sound of an instrument played is sealed off from its source and is manipulated by electronic means in order to achieve different musical patterns.

Stylistic features of Sound Mass composition involve the prevailing concept of 'continuum', blurring of the macro rhythmtic structure and its distribution into sub-

particles, withdrawal of the conventional consonance/dissonance in terms of intervals and numerous micro-movements that gather and form complex sound textures. Prominent composers of this style are mostly European textural composers and could be exemplified with *Atmospheres* of György Ligeti (1961) and *Threnody to the Victims of Hiroshima* of Krzystof Penderecki (1959).

Starting point of this thesis is to build a live sound processing setup that can initiate itself with the use of feedback, being an unwanted and avoided notion in the conventional industrial practice. This idea has grown mature with the inspiration gained from the associated with the Sound Mass style. Main motivation of the study is to obtain similar 'sound outcomes' to Sound Mass style, by realizing timbral organizations and gestures with algorithmic foundations.

The aim of this work is to excite acoustic sound from a musical instrument by bringing live sound processing methods together with feedback. The piano is selected as the base instrument, when the sound range and physical properties are considered. Piano strings are triggered/actuated with a setup working with the principle of 'a vibrating object causes vibrations in another if they possess common frequencies or overtones'; in other words sympathetic vibrations. Acoustic sound is then processed and with use of feedback, the results are sound textures that resemble the Sound Mass compositions are acquired.

For this purpose, a unique interaction setup working with electo-magnetism is developed. The sounds initiated with this setup are processed by algorithmic simulations developed in the software environment. This procedure is handled with Pure Data programming routines.

The sound processing method and setup explained above is developed between 2012 and 2015, focusing on the internalized absence of both the composer and the performer. It proposes a mechanism of generating sound organizations, handled by the instrument on its own by listening to itself. It is possible to expand the timbral varieties of the musical instruments to an extent where conventional performance techniques cannot reach. This setup can be adapted to various instruments with vibrating metal components and it can be programmed for other musical purposes as well. With a compositional approach, it can reveal the potential sound regions of an instrument which has not been explored yet.

Audio and visual examples from various stages of the setup and Pure Data patches are available with the DVD provided.

CANLI SES İŞLEME VE GERİBESLEME KULLANILARAK SEMPATİK TİTREŞEN TELLER İLE SOUND MASS TARZI SESLER YARATMAK

ÖZET

20. yüzyılın başından itibaren müzikal üretimde melodi/armoni ve ritim dışında sesin fiziksel özellikleri öne çıkmaya başlamış ve o ana kadar müzikal üretimin özerk bir bileşeni olarak değerlendirilmemiş olan *tını* olgusu, müzikal bir çalışma alanı olarak gündeme gelmiştir.

Elektronik müziğin doğası gereği, bu tarzın çalgıları, icra ve bestecilik yaklaşımları ortaya çıktığı andan itibaren tınının sese dair başat bir özellik olduğu kabul edilmiş ve spektral içerik üretim süreçlerinde ele alınmıştır. Tınısal arayış arttıkça, ses malzemesinin hangi kaynaktan elde edildiği üzerine farklı yaklaşımlar sergilenmiştir.

Elektronik müzik literatürü incelendiğinde, elektronik müzikte temel olarak üç farklı ses üretecinden/üretim metodundan bahsedilebilir:

1. Sentetik (yapay) sesler: Doğada olmayan tipte titreşimler üretebilen, birbirinden farklı tasarımları olsa da benzer çalışan salınım kaynaklarıyla (osilatör) oluşturulmuş ve sesin titreşim sıklığı (frekans), tınısı (doğuşkan içeriği), zaman içindeki genlik değişimi (zarf) gibi özelliklerinin tasarlanması ile kurgulanmış müzikler.
2. Örneklenmiş (kayıtlı) sesler: Özgün tınısı kayıt altına alınmış ses örneklerinden, aslına benzer ya da çeşitli tekniklerle dönüştürülmüş kopyalarının ilgi ve bağlam farkı ile kurgulanıp çalınmasıyla oluşmuş müzikler.
3. Ses çıkışının ses girdisini geribeslemesiyle ortaya çıkan seslerin yapıtaşı olan müzikler.

Birinci ve ikinci yöntemler tarihsel olarak Alman okulunun Elektronik Müzik (*elektronische musik)* ve Fransız okulunun Somut Müzik (*musique concrète)* yaklaşımları ile filizlenmiş ve çeşitlenmiştir. Üçüncü yöntem olan geribesleme ise, tarihsel bağlamda bir kurum ya da ekip ile ilişkilenmemekle birlikte, daha çok öncül (*avant-garde*) besteci ve icracıların yapıt ve performanslarında kendine yer bulmuştur.

Bu tez çalışması birbirinden bağımsız görünen üç müzikal kavramın bir araya gelip sentezlenmesi ile temellenmiştir.

Geribesleme, canlı müzik sektöründe faaliyet gösteren sanatçı, icracı ya da teknisyenlerin aşina olduğu ve sıkça karşılaştığı bir olgudur. Ekipmanın ses girişi ile çıkışı arasında kurulan kontrollü ya da kontrolsüz, elektronik ya da akustik bağlantı ile meydana gelir. Kimi zaman istenmeyen bir gürültü olarak değerlendirilen geribesleme kaynaklı sesin varlığı, bu tez çalışmasının kalbine yerleştirilmiş ve tasarıda devinimin birincil kaynağı olarak yer almıştır. Herhangi bir icracı veya akustik sistemi harekete geçiren bir uyaran yok iken, ses düzeneğinden ilk sesi üretmek ve devamında oluşacak sesleri yine geribeslemenin kontrollü değişimleriyle elde etmek düşünülmüş, bu olgu geliştirilen mekanizmanın temeline yerleştirilmiştir.

Canlı ses işleme pratikleri, deneysel arayışlarla tınıya etki eden araçların yaratıcı kullanımı sonucu ortaya çıkmıştır. Herhangi bir akustik ses ya da bir çalgının icra esnasındaki sesleri kaydedilir, gerekirse özgün bağlamından soyutlanarak, elektronik yordamlar yardımıyla başkalaştırılarak farklı müzikal kurgular için yapıtaşına dönüştürülür.

Sound Mass bestecilik tarzı, süreklilik kavramının öne çıktığı, müzikal tartıyı dağıtık alt-parçacıklara bölen, geleneksel uyumlu/uyumsuz ses ilişkisine bağlı kalmayan, mikro ölçekte çok sayıda devinimin bir araya gelerek karmaşık ses örgüleri oluşturduğu bestecilik tarzıdır. Bu tarzın önde gelen temsilcileri Avrupalı 'dokusal' (*textural*) besteciler olup, tipik örnekleri olarak György Ligeti'nin *Atmospheres* (1961) ve Krzysztof Penderecki'nin *Threnody to the Victims of Hiroshima* (1959) eserleri sayılabilir.

Tezin çıkış noktasını, sektörel pratikte istenmeyen, sakınılan geribesleme olgusunu kullanarak kendi kendini tetikleyen canlı ses işleme düzeneği ile müzikal yapılar üretme fikri oluşturur. Sound Mass tarzı müzikal eserlerden alınan ilhamla bu fikir olgunlaşmıştır. Bu eserlerin tınısal yapılarının algoritmik olarak kurulmasıyla benzer 'ses sonuçlarına' ulaşılabilmek, tezin çıkış noktasıdır.

Tez çalışmasının amacı, geribesleme olgusu ve canlı ses işleme yöntemlerini bir arada kullanarak akustik bir çalgıdan tınılar elde etmektir. Ses genişliği ve fiziksel özellikleri göz önüne alındığında, çalışmada kullanılacak çalgı olarak piyano seçilmiştir. Titreşen bir ögenin ortak frekanslara (ya da doğuşkanlara) sahip olan diğer ögeleri titreştirmesi kuralıyla, diğer bir deyişle sempatik titreşimler prensibiyle çalışan düzenekte piyano telleri uyarıcı/tetikleyici bir ses sinyali ile harekete geçirilmiş; bu hareket sonucunda oluşan seslerin işlenmesi ve geribeslemesi sonucunda da Sound Mass tarzı ses örgüleri elde edilmiştir.

Bu amaçla özgün bir elektro-manyetik etkileşim düzeneği geliştirilmiştir. Bu düzenek, elektro-manyetik indüksiyon prensibiyle, manyetik etkileşen (metal alaşımlı) cisimleri harekete geçirebilmektedir. Elektro-manyetik indüksiyon prensibini uygulamak için elektrik akımı ve mıknatıs birlikteliğinden üretilen, elektrik akım değişimini manyetik alan değişimlerine çeviren dönüştürücüler (*tranducer*) tasarlanmıştır. Bu dönüştürücüler, piyano telleri üzerine yerleştirilerek tellerin istenilen sıklıkta titreşmesi sağlanır.

Düzenek tarafından üretilen ses, yazılım ortamında geliştirilmiş algoritmik simülasyonlar tarafından işlenir. Bu aşama, Pure Data ortamında programlanmış yordamlardan oluşmaktadır. Bu bağlamda, farklı kaynak-ses tiplerinden farklı tınılar elde edilebilmekte ve piyanodan alışılageldik tını dışında- yaylı çalgılardan klavsene, kilise orguna kadar - farklı tipte sesler üretilebilmektedir. Bu tez çalışmasında birden fazla ve farklı tipte ses üreteçlerinden oluşturulan uyarıcı sinyaller piyanoya gönderilmektedir.

Tüm teller aynı anda titreştirildiğinden, tellere gönderilen herhangi bir frekans/nota değeri, hangi telin harmonik yapısında mevcutsa, sempatik titreşim prensibi gereği o telleri de titreştirerek tüm piyano üzerinde zengin ve karmaşık bir tını yapısı oluşturur.

Yukarıda tariflenen ve 2012-2015 yılları arasında geliştirilen ses işleme yöntem ve düzeneği icracının ve bestecinin yokluğunu içselleştirerek çalgının ses potansiyeline odaklanır. Çalgının kendisini dinleyerek akustik ses organizasyonları üreten bir mekanizma önerir. Bu yöntem ile çalgıdan geleneksel icra teknikleri dışında tınılar elde etmek mümkündür. Bu düzenek, metal titreşene sahip diğer çalgılara adapte edilerek uygulanabileceği gibi farklı müzikal amaçlar için programlanabilir ve istenen ses organizasyonlarını üretebilir. Bestecilik potansiyeli bakımından çalgıların henüz keşfedilmemiş ses bölgelerine ulaşarak bu tınıları açığa çıkarabilir.

Çalışmaların farklı safhalarının işitsel/görsel örneklendiği bir DVD de, tez ile beraber sunulmuştur.

1. INTRODUCTION

The term *experimental music* describes the unusual or avant-garde music independent of any musical genre (Cox & Warner ed., 2005, p.207). The terms *avant-garde* and *experimental* is often used to categorize radical composers and their works yet “avant-garde remains more than a slogan than definition and experimental music is ill-defined and the concept it is used to define is vague” (Nicholls, 1998, p. 517). Avant-garde music can be viewed as occupying an extreme position within the tradition, while experimental music falls completely outside of tradition (p.518). In a broader sense, the term experimental music characterizes different composers and eras. Composers like Charles Ives, Pierre Schaeffer, Meredith Monk, Edgar Varese, Henry Cowell, Morton Feldman, John Cage etc. are attributed as ‘experimental composers’ from different perspectives. A particular definition of *experimental* is given by John Cage as “an action which the outcome cannot be foreseen” (1961, p. 69). Cage’s definition of the ‘experimental action’ is the contextual basis of this thesis and will be referred as a methodological approach, regardless of the musical genre.

1.1 Objectives

Following this Cageian approach, the experimental composer pushes the music beyond the limit of tradition; first, by disconnecting the intention of the composer from the musical process and second, by shattering the deterministic design methods of the musical outcome. From this perspective, the *convergence* between the *maker* and his/her *opus* once widely accepted and naturally conceived, had been forced to diverge both in the nature of the *making* process and the cause-and-effect-relation between the creator and his/her creation.

The most basic method of experimental composition is to design a set of initial conditions – technical, sonic, conceptual, verbal, social etc. – and then to leave them to unfold, more or less on their own. In this essence, if the reference is towards an autonomous world of evolving sounds, rather than the one that is composed, then the cancellation of intention might bring the following idea to mind: can wind chimes be considered as the precursors of experimental music practices? Morton Feldman explains:

> My past experience was not to 'meddle' with the material, but use my concentration as a guide to what might transpire. I mentioned this to Stockhausen once when he asked me what my secret was. "I don't push the sounds around". Stockhausen mulled this over, and asked "Not even a bit?" (Cox &Warner ed., 2005, p.205).

According to Feldman, the experimental composer/performer tries 'not to meddle with the material', not 'to push the sounds around'. During his prepared piano works, Cage abandoned this submissive refraining from 'meddling the material' for a moment, especially in the performance practice and altered the nature of the performance medium of the instrument. This was referred as experimental at the time, too. As he stated, his professor Henry Cowell's (new at the time) methods such as reaching inside the piano and manipulating the strings were direct influences to his idea. He meddled with the material and pushed the sounds around quite a bit in this approach. This time the concept of *experimental* was based on not only the compositional approach and the performance process, but also towards the conventional timbre expectancy of instruments. Most of the time, unexpected, surprising timbral richness lies in the potentials of acoustic instruments, waiting for to be mined to meet the daylight.

Coming back to the previous idea of 'setting the environment for an experimental music outcome', is it possible to push this matter to another level by dissecting both the composer and the performer from their performing medium - *the instrument*? Considering Cage's words: "composing is one thing, performing is another, listening is a third, what can they have to do with one another" (p.15), can we quest for ways to provoke the instrument to 'push its own sounds around' by *listening* to itself? Evoke sounds by not composing, not performing but by listening and resonating with the self-sound?

In the field of medicine, there is a phenomenon called Magnetic Resonance Imaging (MRI). MRI is applied with massive circular magnets. These clinical magnets are spun around the body of the patient, creating a strong magnetic field around the area to be imaged. Atoms in different layers of tissue and organs resonate and respond with different vibration characteristics meanwhile the sensors capture and evaluate these signals. Hence the anatomy of the structure is demonstrated within the captured images, for further diagnosis, etc. Deriving from this idea, is there a way of resonating the sound potential in an instrument, which is initially silent and at rest; by another source with a matching spectrum?

Live processing is a popular and wide spread musical approach realized by altering the sound properties of an instrument during performance and in turn affected sound becomes a unique 'voice'. Hence it is possible to metamorphose the sound, re-assign the compositional relation and function. This method, based on a large variety of tools and techniques, is a transformation process that captures the original performance. By manipulating sonic attributes, it gives the sound a new identity by changing the previous form and state into new ones (Cox &Warner ed., 2005, p.205). If the *experimental* essence of this transformation process is to be considered it is possible to design certain initial conditions that lead to certain alterations of sound. With the previously mentioned questions in mind, can a live processing technique be realized so that it would take control of the transformation itself, without a controller of its performance? Going further, can it lead to creating some kind of virtual performing practice - if it can be called a performance?

If such a system could be realized, the final material might not be *foreseen* precisely but by considering the various possibilities it may as well be planned beforehand. Furthermore, it can be controlled and be shaped interactively as the corresponding outcome is heard. Such a setup can be realized by algorithmic means, conceptually it is possible to direct the computer to follow some predefined paths and methods. In addition, if non-deterministic methods would be combined with this concept, it would lead to an example of always-changing, non-repeating outcome, a prototypical performance. Thus a live processing setup as such would be a realization of the Cageian 'experimental', as it would have the potential of exploring various sonic outcomes when proper stochastic methods are applied.

This thesis offers a live sound processing method, that the output could be only 'foreseen' as the collection of string resonances. Most of the choices and musical decisions will be generated and realized in stochastic algorithms. The musical texture of the processing outcome will be based on 'generated' layers of resonances and the physical interaction between them. As these sound elements evoke and evolve, each with their own time-span and sonority, they will create dense sound clusters. The 'foreseen' outcome of this experimental setup is a reminiscent of the Sound-Mass composition style, particularly of the Sound-Mass composers from the early 1960's.

Including the idea of the *instrument listening to itself*, or feedback in technical terms, this study proposes a setup consisting of hardware and software systems that generate algorithmic sounds on its own, by using the principles of *sympathetic vibration* [available in G.o.T.] and feedback manipulation.

1.2 Inspiration

A sound engineer operates using his/her ears as the gateway of incoming information. Having gathered valuable experience in numerous pieces of hardware and software that act upon the sound characteristics, the sound engineer's ears are tuned in and trained by being exposed to every kind of sound manipulation. So one can capture processing remarks, recognize added sound effects and categorize sound morphing methods instantly just by hearing the sound material. As this professional habit develops in time, there are times that one is deceived by the judgment of the ear; to a sound that has no effects or processing (*dry* in engineering terms) a sound engineer can insistently respond otherwise. One may be thinking of recognizing a certain processing method – in reality there might be none.

Mastering engineer Bob Katz exemplifies such an adverse effect of this habit in his highly acclaimed book *Mastering Audio.* During his mastering sessions, he often realizes editing problems where no one noticed before. His ears catch these editing errors at once and he sends the audio material back to editing/mixing. He is so used to the characteristic sound of these editing problems, when listening to live performances he ironically says: "I got so paranoid in time that sometimes I think I hear editing problems in live concerts!" (Katz, 2002, p.47).

A listener unfamiliar of the Sound Mass genre – such as the writer of this thesis, who has an intrinsic reverse-engineering approach – might have associated Penderecki's raw energy of instruments and Ligeti's cloud-like sonorities with the masses of granular synthesis and random pitch generation algorithms. The writer had paralleled the vast variety of sound bodies and gestures with the outcome of familiar sound processing methods such as granular synthesis, real-time transposing algorithms, feedback systems, randomized variations of digital sound processing parameters, etc. An intuitive momentary reaction of a sound engineer, not derived from examining the score but from hearing the actual performances of the compositions – has been the main inspiration and motivation to construct such a programmable setup that can explore the new (at the time) discovered sound world.

In short, the main inspiration of this thesis comes from the *Sound Mass* [available in G.o.T.] composers, particularly of Ligeti and Penderecki and their compositions dated back to late 50's and early 60's.

1.3 Contents and Structure

This study consists of a final report of a research. The framework of this research consists of the musical outcomes of feedback manipulation. The initial aim was to explore and evaluate methods and approaches to create musical sonic structures by evoking, controlling and diverting the feedback of a sound, using the potentials of a typical live performance scenario. For this purpose, a fertile situation was sought within the interaction of the feedback in a sound reinforcement system and its physical influence on an instrument.

The general structure of this thesis is as follows: in the next section, Chapter 2, dwells on the details of the research and development stage, explaining how the method is developed and how the setup is produced. Chapter 3 further explains the setup through the stylistic features of sound mass composition; focusing on how these features are modeled and achieved via the software. Last section, Chapter 4 is the conclusion, encapsulating the contributions and possible future expansions of this thesis.

This study is done in between 2012 June and 2015 May. Visual materials of the past experiences and actual setup are placed in the Appendix section, accompanied by audio examples and Pure Data software codes which can be found on the DVD that is provided with the thesis. Also in the Appendix, there is a collection of keywords under the topic *Glossary of Terms*. Keywords explained in this section are indicated with a 'available in the G.o.T.' indicator throughout the thesis.

2. METHOD

The aim of the project was to find ways for triggering sound on an instrument which can produce musical sounds via exciting its vibrating elements. In pursuit for such methods, a setup was built to create sympathetic resonances. The main motivation of the study was to test the idea that if feedback could be controlled to match with the harmonics on an instrument, for example; the strings on the piano or the guitar; then it could as well produce sound and be manipulated for musical and timbral variety. Such a setup would allow the manipulation of certain spectral regions of the instrument. That would enable the compositional design of the sound. In a broader sense, extended compositional approaches for complex sonic textures could be achieved.

It should be stated that at the time of this study, works of Per Bloland from Oberlin Conservatory of Music on acoustical actuation of the piano in 2001, his collaboration with Edgar Berdahl and Steven Backer from CCRMA of Stanford University in 2005 (Bloland, 2010), and their concept of *Electromagnetically Prepared Piano* (Berdahl, Backer, Smith, 2005) nor the works of Andrew McPherson and Youngmoo Kim from Drexel University and their concept of *Electromagnetic Instrument Actuation* were not examined (McPherson and Kim, 2010; McPherson, 2012). Nevertheless, the discovery of such related works had been a major boost to understand and get into details and connect the missing parts in the puzzle. However, this discovery took place at the very late phases of this study.

The method of this thesis is a combination of generative software and its realization on an instrument, using a custom-built hardware. The method is developed in a presumptive approach; the path leading to the proposed method was traced intuitionally. The development stages were based mainly on experimentation. The evaluation of the experiment results led to further improvements. The concept and signal processing were developed in the software environment; also a compatible hardware setup was developed to realize the concept. Photographs from various experiments and earlier stages are available in the Appendix section.

2.1 Research and Development Stage

At the early stages of research and development, it became evident that an instrument with a wide sound and harmonic range would be appropriate as the basis of this setup. Furthermore, it should be possible to physically reach to the vibrating components of the instrument from outside the instrument, because the primary aim was to trigger those vibrating elements by exposing some kind of triggering acoustic force. To achieve sympathetic vibrations, a stringed instrument would be proper to work on. The longer the vibrating element, the easier it would be to evoke vibrations and precisely spot the harmonic plots (*antinodes* – the points that do not vibrate on a string) on it. Considering such requirements, the piano is thought to be an appropriate selection for the setup.

2.1.1 Piano as a sympathetic vibrating instrument

In a typical piano there are 216 to 250 piano strings[1], which are tuned to the frequencies roughly between 30Hz - 4000Hz, covering a range of 8 octaves. As a percussive instrument, when any piano key is pressed the hammer underneath the keyboard hits a set of strings corresponding to the same note - either two or three strings – results to sound louder than a single string instrument. These multiple strings are not tuned exactly to the same frequency. By being slightly out of tune the string couples preserve their energy rather than consuming and diminishing the amount of transfer via the sympathetic vibrations. This phenomenon is known as *beating*.

In a tuned piano, the strings are under tons of tension. To handle this huge amount of force, the strings are attached across a metal frame, which is usually made from cast iron. The iron frame is attached on top of a wooden soundboard that gives its distinctive quality of the piano timbre. When a key is pressed, vibrations of the strings are transferred to the metal frame via the bridge, from the frame to the soundboard and from the strings and the soundboard to the air; thus sound emanates from the piano.

As a characteristic of the instrument, piano sound has a long decay time. To take control of such long decay, the instrument has the *sustain pedal* that controls the

[1] This quality depends on the piano design.

damper mechanism placed on top of the strings. This long, naturally sustained sound is achieved by a set of design features of the piano; such as the use of multiple strings for each note, the low damping factor of the interaction between high-tensioned strings and the frame that result in slower energy transfer from the strings to the frame, and from the frame to the soundboard.

This exceptional feature leads to a longer sustain of the played notes, which is taken into account and will be used in advantage of the proposed setup in this thesis work. In order to have the resonances occur freely and longer in time, the damper mechanism of the piano is removed in this setup.

2.1.2 Experimenting with initial idea of acoustic triggering

The first idea of triggering the piano strings was by means of acoustic exposure. In the early stages of the study, a prototype consisting of multiple loudspeakers and microphones were used. The mechanism was planned as following: First, both the microphones and the speakers were placed inside the piano facing towards the strings. Sound coming from the microphones would be amplified, initially it would be ambient noise and then it would be played back through the speakers; in an attempt to create a feedback loop. Thus, the strings would be exposed to acoustic energy and supposedly they would vibrate as they react to the triggering signal. By means of sympathetic vibrations, tuned strings those were harmonically related with the feedback signal would get into motion. Through this process, the sound coming from the piano would be continuously recorded and processed in order to manipulate the feedback and then played back into the instrument again. The audio processing mentioned here was meant to achieve a crucial task, firstly controlling the instantly occurred feedback, and then manipulating and leading it to different potential feedback regions on different frequencies (of different strings). By doing so, the sound would continue to emerge with controlled feedback and would be carried to other pitches enabling continuous flow of occurring harmonics. These harmonics would build sound blocks on various pitches as the feedback moved on and explored different regions of the piano.

This is analogous to surfing on top of the ocean wave, as if gaining control of the initial exposure of a force in order to make use of it; or in the martial art of Aikido, that aims to redirect the momentum of the opponent's moves instead of consuming one's own energy.

In numerous experiments, despite the efforts to run a self-controlled sound mechanism, the aim could not be reached. This method had a design flaw: the sound coming from the speakers were directly picked up by the microphones and the intended processing method was being locked into a static feedback frequency much before being able to trigger the strings. Each time the occurring feedback overcame the processed sound, which made it impossible to control the sound.

As a result, it is decided that triggering the strings by acoustic force is not suitable and efficient for this purpose. Photographs of these experiments are available in Appendix B. An alternative solution was developed as explained in the following section.

2.2 Proposed Hardware Setup: Electro-Magnetic Exciter

It became evident that acoustic power was not effective enough to set a piano string in motion in this case. When the aim of vibrating another body without touching but with applied force was considered, sound-producing principles of some of the earliest known electronic instruments, such as Telharmonium, Musical Telegraph as well as latter successors – like the Hammond Organ or the electric guitar come to mind. These instruments have a common triggering principle, all are based on *Electromagnetic Induction* [available in Appendix subsection; Glossary of Terms] comes to mind. This method has been an inspiration after encountering problems with previous acoustic excitation methods. In research for an alternative method, studying and experimenting with electromagnetism had been fruitful.

2.2.1 Historical background

In 1820, a Danish physicist namely H. C. Oersted discovered that a steady electric current created a *magnetic field* [available in G.o.T.] around itself (Url-1).

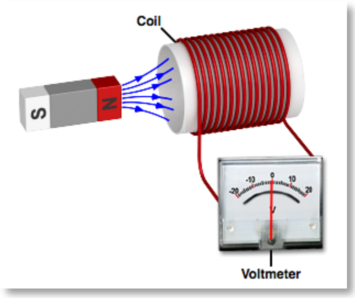

Figure 2.1 : Due to electromagnetic induction, the motion of the magnetic substance through a coil generates electric voltage, (Url-1).

Following this in 1831, another scientist in England, namely M. Faraday discovered electromagnetic induction, that is when a conductor is exposed to a varying magnetic field, it produces an electric current across it; proving electricity that could be generated by magnetism (Url-2).

Among his contemporaries, electricity and magnetism were thought to be two different forces until J. C. Maxwell, a Scottish mathematical physician, explained the interaction between the two forces as stated in his electromagnetic theory, in 1873. He proved that the magnetic and electric fields interact, as changes in electrical field results in a change in the magnetic field and vice versa; because the positive and negative charges – in both electrical or magnetic fields – are regulated by the same force (Url-3).

2.2.2 Electromagnetic transducer

Coming back to the discussion in the previous topic, as the aim of the study is to trigger strings avoiding any unwanted sound in the acoustical space, electromagnetic induction is chosen as an appropriate method for triggering string resonances. For this to happen, an *electromagnetic transducer* [available in G.o.T] would be needed.

The most popular example of this kind of transducers is the E-bow. It is a hand-held, battery powered electronic device for playing the electric guitar, invented by Greg Heet in 1969. Instead of having the strings hit by the fingers or a pick, they are moved by the electromagnetic field created by the device, producing a sound reminiscent of using a bow on the strings. There are two coils inside the e-bow, one input and one driver coil. The signal from the input coil is amplified and sent back via the driver coil. This signal path results in a change in the magnetic field and causes the string to vibrate naturally in its first mode: fundamental frequency. This vibration is captured by the input coil thus a *feedback* [available in G.o.T.] loop is generated (Url-4).

Figure 2.2 : The E-bow (Url-5).

If the suggested setup were to be built with e-bows, it would not be possible to reach the desired musical results. The setup developed throughout this study aimed to create sounds from piano strings with sympathetic vibrations. By building blocks of these produced sounds it proposes a way of building sound clouds. Yet in the case of E-bow, only the fundamental frequency of the exposed string (or as a an option, an octave higher) is produced with magnetic feedback, and some additional harmonic content is naturally evoked when the sting vibrates. In the proposed method, it is presumed to be possible to trigger any other harmonics of a string, not only the fundamental.

Also the proposed design aims to send the same triggering signal not only to one string but all of the strings on the piano. It is not the case with the E-bow as it is projected to one single string only. But as a foresight, it can be said that if a group of conventional e-bow's were applied to the piano, as one e-bow for each set of the same note - and if the e-bows were optimized for the piano strings[2] so that it would produce enough power to move the heavy, high-tensioned piano strings - then the sonic outcome would be the sum of every fundamental frequency of all the strings, according to the notes that they are tuned; like the result of pressing all of the piano keys.

[2] The writer had experimented with an e-bow along the piano strings and came to a conclusion that it does not produce the same efficiency for all the strings, there are some strings effected especially in the middle region but note all of them. Other unaffected strings are thicker, heavier for the e-bow to affect in the bass region and for the treble strings they cannot vibrate alone with magnetic feedback since they are under great tension, not loose enough for an e-bow.

A typical electric guitar pickup is worth mentioning at this moment, because the prototype developed during the evolution of the suggested setup were primarily influenced from the interaction of the string and the pickup on an electric guitar. As electric guitar pickups, too, work on the electromagnetic induction principle, the motivation was to produce a setup where one would function in reverse direction; not picking up the sound but causing it.

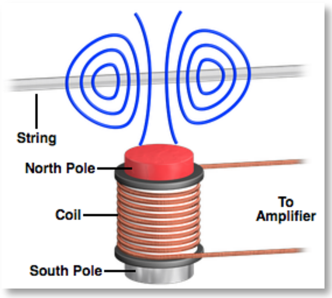

Figure 2.3 : Typical electric guitar pick-up. The red substance is the north pole of the cylindrical magnet sitting inside the coil. Curves in blue indicate the magnetic field created around the coil. When the set in motion, the string oscillations cause a change the potential of the magnetic field around the coil and this change creates a current, at the same time a voltage difference between the two edges of the coil wire (Url-6).

Around a magnetic core, typically a magnet, electrical wire is wound to create a coil as shown in Figure 2.3. Then the coil is placed next to a metal string. When the string is put into vibration, it vibrates in three dimensions. As the string moves closer to and further from the magnet, it interacts with the magnetic field around the coil. As the potential changes in the magnetic field occur, electric current is generated along the coil. This electric signal is then sent to an amplifier and the current is modified to e large enough to move the speaker cone of loudspeaker.

The magnitude (amplitude) of the generated electric current depends on the following properties:

- type of the magnet - meaning the strength and the direction of the magnetic field,
- type of the coil – meaning, A) the thickness of the electric wire and B) the number of spins in the coil.
- the distance between the coil and the string,

The electromagnetic transducer prototype developed for this suggested setup is realized after a series of heavy experimentation. The starting point was to use a typical electric guitar pickup connected reversely, in terms of function. By building an array of electromagnetic transducers, it would be possible to trigger the piano strings. Specifications and technical details of the transducer prototype will be explained in the following sections.

2.2.3 Developing the transducer prototype

The electromagnetic transducers used in this study are developed after a prototype, as this was developed as a hand-wound coil around a magnetic core. The components of this prototype would be explained in the following subsections.

Component: magnet

Rigorous research and several experiments were conducted to determine the kind of magnet that would be used. It was sorted out that cylindrical magnets made from *Neodymium* [available in G.o.T.] material are used for such magnetic transducer applications. Neodymium cylindrical magnets come in different sizes – in height and diameter.

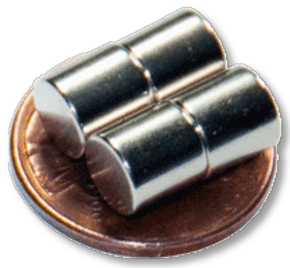

Figure 2.4 : Typical neodymium magnets, compared with the size of a dime (Url-7).

As a principle, the bigger the magnet is, the stronger the magnetic field it has around it. One restriction for the selection of the magnet size was the thickness of the piano string. In order to place the transducers in order to arrange an array, a number of issues should be addressed in terms of putting them side-by-side. They should evenly be distributed along the strings to cover the instrument and each magnet should correspond to a number of strings that should be same along the whole range. After experimenting with different sizes, cylinder magnets in size of 2 centimeters in height and 1 centimeter in diameter worked well as a starting point. Magnets with 2cm height and 1 cm width would fit to cover any string on the piano, bass or treble.

Component: wire

The wire used in the coil has an isolation layer around so when spun on itself, in order to make a coil, unwanted shortcuts and short-circuits are avoided. The amount of the voltage that any coil can produce depends on the physical properties of the copper wire, such as the diameter and number of turns (spins) around the core. Due to the nature of the material, all electric wires produce a certain amount of resistance to the current that is passing through them. This internal resistance increases as the length of the wire increases. At the same time, this resistance depends on the wire diameter. It would increase as the diameter decreases. Resistance is a phenomenon under *Direct Current* (D.C.) [available in G.o.T.], where impedance is under *Alternating Current* (A.C.) [available in G.o.T.]. The amount of wire affects the coil's impedance, as longer wire means higher impedance. In such applications where coils and electric wires are used, wire diameters are specified with a standard called *American Wire Gauge* (AWG) [available in G.o.T.].

Wire diameter

Most common types of coils used in electric guitar pickups are wound with #42 or #43 AWG, these number indications correspond to 0.063cm and 0.055cm in diameter. Such thin wires are spun thousands of times (typically 4,000 to 8,000) around a magnetic core. Coils wound with these types of wires typically have 4,000-12,000 ohms of impedance.

Table 2.1 : Comparison of typical electric guitar pick-up wires and the wire used in the proposed setup (data acquired from Url-8).

AWG	Diameter (in)	Diameter (mm)	Resistance (Ohm/meter)
43	0.0022	0.0559	7.030
42	0.0025	0.0635	4.907
24	0.0201	0.5105	0.084

The transducer prototype developed for this work should meet certain requirements: First, it should produce a higher amount of voltage than a typical electric guitar pickup because the strings of the instrument are different. To vibrate piano strings that are much thicker than the guitar, coils must have higher voltage potentials. To produce higher currents/voltages in pickups, stronger magnets should be used in the core. Increasing the wire spin number would be another idea, but there is a trade-off here: as more wire is wound, the length of the wire will increase which will also increase the resistance of the wire; being the impedance of the wire, under AC. The role of the impedance factor will be discussed later in this chapter.

Component: electromagnetic coil = magnet + wire

After numerous experiments to assemble a supporting body for the wounded wire (at the same time, considering the magnet core in the center), it came to a solution where a type of plastic reel that originally is used in reels of soldering wire can make a perfect housing for such a need, as seen in Figure 2.5.

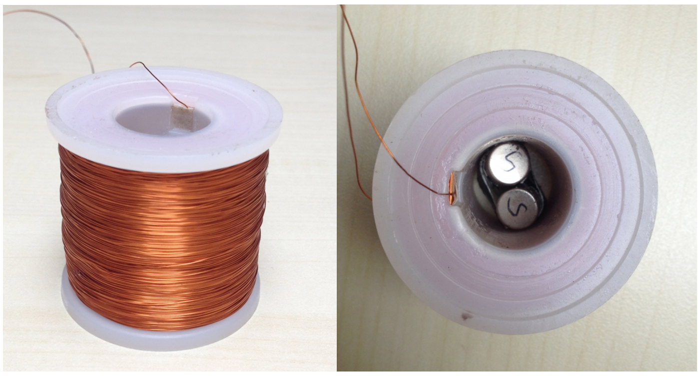

Figure 2.5 : Hand-wound prototype used in the proposed setup (left). Coil and the magnets – electromagnet prototype as seen from top (right).

In this body, the opening in the center was wide enough to hold two magnets attached side by side. The array idea was modified accordingly as each transducer would hold two magnets and this augmentation would have an effect of enhancing the strength of the magnetic field. However, in order to align the direction of the magnetic field properly, same poles must be placed facing the same direction; as if creating a bigger magnet. Otherwise the polar opposition will weaken the magnetic field created by magnets. Yet the same poles repel each other, each magnet pair is fixed tight by gaffer tape. This support avoids the magnets to repel each other in this position of "similar poles sitting together", Figure 2.5. In the figure, aligned south poles are seen from the top as the north poles are facing downwards, which will be directed to the piano strings.

Coming back to the previous discussion on impedance, it is worth explaining this feature as it gains importance when multiple transducers are connected together, they exhibit different behaviors under different connection types.

Nominal impedance

The impedance of one single coil is an important constraint of this design that will affect the overall impedance of the coil array. The proposed setup is designed so that the coils will be driven as though they were loudspeakers - an amplified audio signal would be send to them. Conceptually, coils were to be connected side-by-side and this collection would to be connected to the output of an audio amplifier[3]. This dictates that the array should not exhibit large impedances. Coming back to the main motivation, the electric guitar pickup, these types of pickups exhibit high impedances, typically between 4,000-12,000 ohms; because they were intended to be used in the input stage of an electrical circuit. Speaking of the output (load) impedance of a typical audio power amplifier is in the range of 8-16 ohms[4], and most amplifiers can give power to an output of 4 ohms minimum. If a speaker was to be connected to an amplifier output, it should meet this impedance requirement. This requirement was followed as a guideline for the actual impedance calculations both for a single coil and the array.

To meet this 'low impedance-high voltage' requirement, it would be appropriate to choose a wire diameter that is not so thin (to decrease the impedance) and wind it in larger spin numbers (to increase the voltage) around the coil. This said, AWG #42-43 wires that are used in electric guitar pickups can not be used in this setup, because they are extremely thin and they exhibit greater impedances; beside the fact that physically it would be enormously difficult to handle such thin wire to wind without breaking.

At two extremes, lowest and the highest piano strings would be hard to vibrate; considering the past experiments with the e-bow, and this might require changes in the standards of the coil design. Lowest strings in the bass region are extremely heavy and highest strings are under great tension with smaller lengths, making it difficult to vibrate with an electromagnet.

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[3] With the motivation to run a transducer in reverse fashion, meaning in order to function a transducer as a signal output, the desired triggering signal should be amplified first and then sent to the transducers as if they were loudspeakers in a conventional sound system setup. This is why a power-amplifier is used in this setup.

[4] The given quantities are the typical speaker impedances used in a loudspeaker-power amplifier connection.

The maximum number of prototype transducers over the strings is then calculated by measuring the width of the piano and considering the physical possible plots of placing them. It was measured that 20 coils would fit properly across the desired sound range[5]. It is calculated as if the nominal impedance (the final impedance exhibited by the desired number of coils together) can be finalized between 8-16 ohms, electrically it would be a proper to connect to the output of the power amplifier. Considering 20 coils are connected together, the connection type should be parallel, and each transducer should exhibit 160-320 ohms of impedance to match the nominal requirement (calculation details explained in Table 2.2).

A prototype with 200 ohms of impedances was planned, as that would result in 10 ohms of nominal impedance (due to the calculation in Table 2.2) when 20 of them were used together. As stated earlier, the aim was to keep the impedance of all coils in the safe region of 8-16 ohms for the output of an audio amplifier.

Table 2.2 : Impedance calculations in series and parallel connections (impedance is shown with the letter Z; Z_T is the total impedance)

In serial connections,	$Z_T = Z_1 + Z_2$
In parallel,	$1 / Z_T = 1 / Z_1 + 1 / Z_2$

After a series of experiments with wires of different diameters the final decision was to use AWG #24 (diameter: 0.5mm) as an appropriate size to handle the large spin numbers with low impedances. Having almost 5000 spins of AWG #24 wire on the plastic reel (5124 spins, to be exact), 200 ohms of resistance values under DC was observed. Afterwards the prototype was measured under AC, impedance deviations under different frequencies were so negligible that this prototype is set to be the example of design for the manufacturing of other coils. It could be accepted that each coil in the array has 200 ohms of impedance.

Later when the hardware setup is carried to an actual grand piano, because the iron-cast frame differed in shape, narrower space allowed the use of lesser number of electromagnets. This time 18 coils could be placed over the strings. 18 coils

[5] This number is calculated during the research and development stage, when an upright piano was being used as the experimenting medium. In order to simulate the grand piano, the upright piano was rotated, laid back and placed horizontally.

connected parallel, resulted in 11.1 ohms of nominal impedance, which was still in the presumed safe region of 8-16 ohms.

The brand and model of the audio amplifier used in the setup is a Lab Gruppen 2000C power amplifier. Technical specifications indicate that it can produce 2000 W peak and 1000 W continuous power when the output load impedance is 8 ohms.

As a result, the final electromagnet prototype was wound with #24 AWG - 0.5 mm diameter insulated copper wire, having 5124 spins of wire in each coil. This resulted in a resistance of 200 ohms under DC.

Figure 2.6 : Hardware of proposed setup: electromagnets aligned and placed on the piano, each hung 2mm over the strings, as seen from close up (*ABOVE*), from the top (*BELOW*).

Replicas of the prototype are produced. 18 electromagnets are connected in parallel and hung over the grand piano. They are arranged in two rows, upper row that is closer to the keys carry 10 transducers side-by-side. Behind it another row is arranged carrying 8 transducers. The rows are metal bars, sitting on the iron cast frame of the piano, standing on rubber supports. Each transducer is hold with a clamp and these clamps are hung on the bars with tight and thick rubber strips. To adjust the distance between the strings and the coils, rubber blocks differing between 0.8 cm and 2.1 cm in height are placed between the holder bars and the transducers. Rubber strips stretched and held each coil tight and the distances between each coil and the facing set of string is fixed to 2 mm. The final setup can be seen in Figure 2.6.

2.3 Proposed Software Setup – Signal Generation

Sound synthesis, exciter signal generation and level controls are realized in the software section of the proposed setup. The software blocks are designed and implemented in Pure Data programming environment. Signal flow in the synthesis and processing algorithms are illustrated as an abstraction in Figure 2.7.

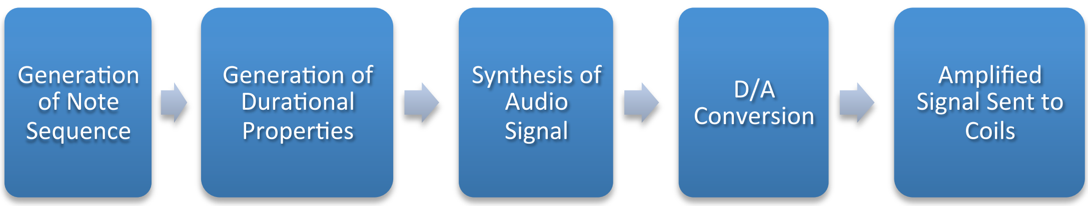

Figure 2.7 : Simplified illustration of data flow in Pure Data patches.

The main function of the software is to produce triggering signals for the piano strings and to send them to the instrument in an appropriate level. Main motivation being the Sound Mass style, the software prepares blocks of notes - sound clouds - to trigger the matching harmonics on the piano strings. With this perspective, to increase the amount of the excited harmonics, triggering signal is designed in the form of crowded collection of notes, all sent at the same time to the piano. With the principle of sympathetic vibrations, triggering signal will produce vibrations in the piano.

Functional overview

Considering the functions of the software, a simple definition of the flow can be made as the following:

- Generation of a collection of notes, that will construct one tone cluster as the triggering source,
- Generation of durational properties for each tone cluster,
- Synthesizing the clusters as audio signals,
- Sending signals to the transducers in the piano.

Actual sound level in the piano is monitored and with this sound information software evaluates further steps, this generation sequence in Figure 2.7 is repeated automatically.

Tonal organization

Staying within the borders of Henry Cowell's theory (Cowell, *New Musical Sources*, second ed. 1999), triggering signals are generated in the form of tone clusters. This generation is made with collection of notes organized with intervals of seconds. This is the default setting in the software however, there are deviations from of the theoretical limitations as expansions of musicality, they are available in the software as optional functions. This reference of theory and its implementation in the software is broadly explained in the 3rd chapter.

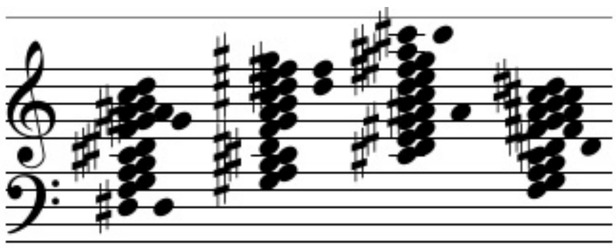

Figure 2.8 : Example of four consecutive tone clusters generated by the proposed software. To indicate the extreme intensity of the close neighborhood notes, software is modified for this example and a 16-note cluster template is used. Note durations, dynamics and envelopes are arbitrary and not indicated in this notation.

The software is capable of producing a vast majority of tone clusters because the base note selection and the interval relations are assigned randomly. To give an example, four consecutive generated clusters with the software are shown in Figure 2.9. In this example, the software is augmented to generate 16-note clusters in order to give example of the concept of dense sonic bodies that can be created with the software, also to show its extremity, the irrelevance of the human-like performance capabilities. Conceptually, dense sonic textures are generated to imitate the Sound Mass-style sound clouds and these abstract tonalities with close neighborhoods create sharp resonance peaks – a fertile climate for the feedback to occur.

Stochastic nature

Most prominent factor in the proposed setup is its stochastic nature. Generation processes are dependent on control variables and these variables are non-deterministic, as a design feature. For example; for each process that needs to produce a state of decision, a probability distribution is defined within the bounds of physical or stylistic constraints. Inside these probabilistic borders, one single state among others is selected randomly. By assigning percentage values for parameters called *weight* in the probabilistic process, the ‘tendency’ for each state is defined beforehand. These particular stages of decision-making processes are shown and explained in the scheme below.

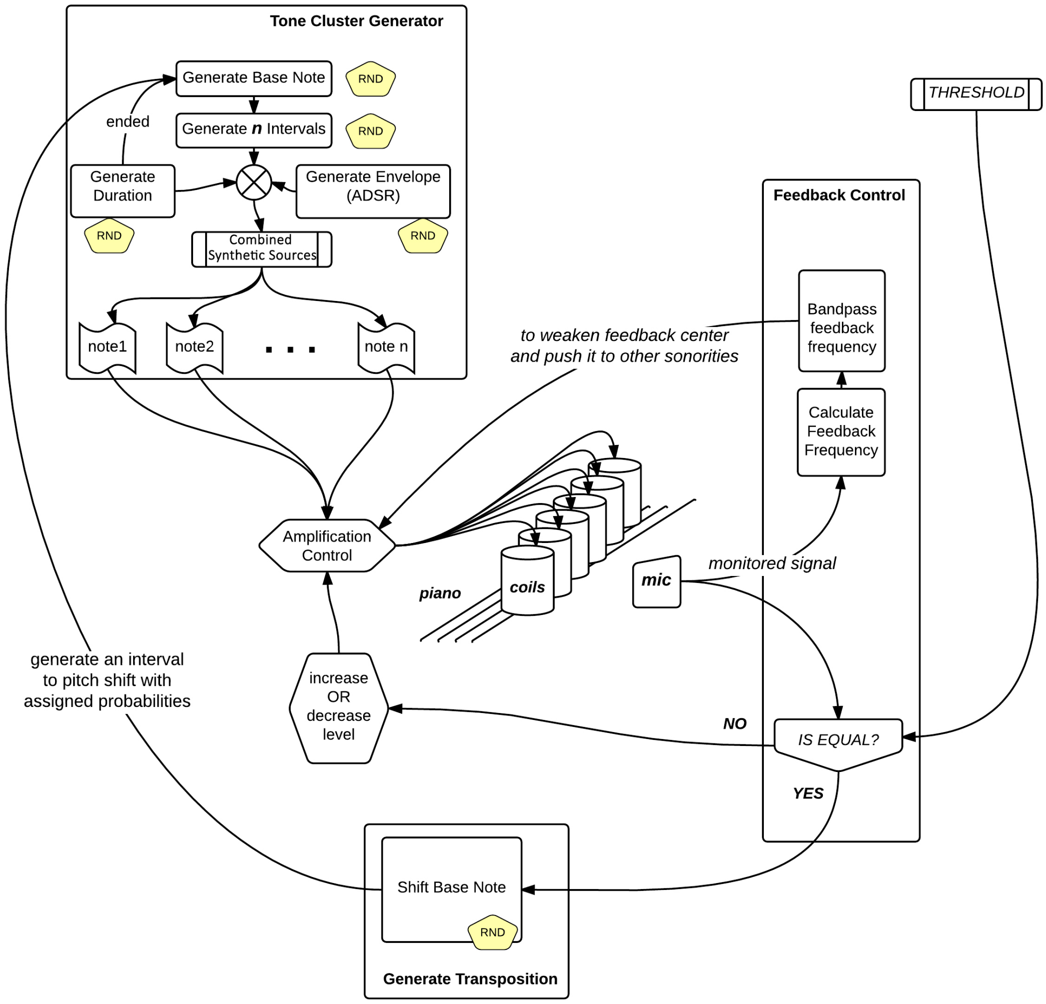

Figure 2.9 : Functional overview of the software setup.
The *RND* labels indicated the random value generation stages.

Conceptual data flow of the software is shown in Figure 2.9. The whole process is designed to have autonomy; as the software starts playing with the initial 'start' command, until it is stopped manually it keeps running on its own. The functional blocks will be explained in detail in the following paragraphs.

Loudness setting and adjustment

Functional control of the software relies on one important parameter, the *Threshold,* Threshold is the magnitude of the 'desired' acoustic sound level in the instrument, assigned by the user. It is what causes the start and the following movements. Although it has a default value in the beginning, it can be manually adjusted. It indicates the actual loudness level that is to be matched by the setup. Sound coming from the piano is monitored continuously. Until incoming sound level matches the level indicated by the threshold value, the software will increase the gain of the signal sent to the piano. If the loudness level is beyond the threshold value, the software will diminish the overall gain in order to match the threshold.

There is a feedback path in the design and it is active at all times. As the signal going into the piano increases, occurring feedback will increase, too. This feedback is then manipulated for more musical results. The detail of this stage regarding the feedback and how it is handled will be explained in the end of this chapter.

Software GUI

The main control window of the software can be seen with all the controls above in Figure 2.10. The components are indicated in colored boxes and these units control and direct the process. The details of each unit will be presented in the next chapter; however general overview of the software will be explained here.

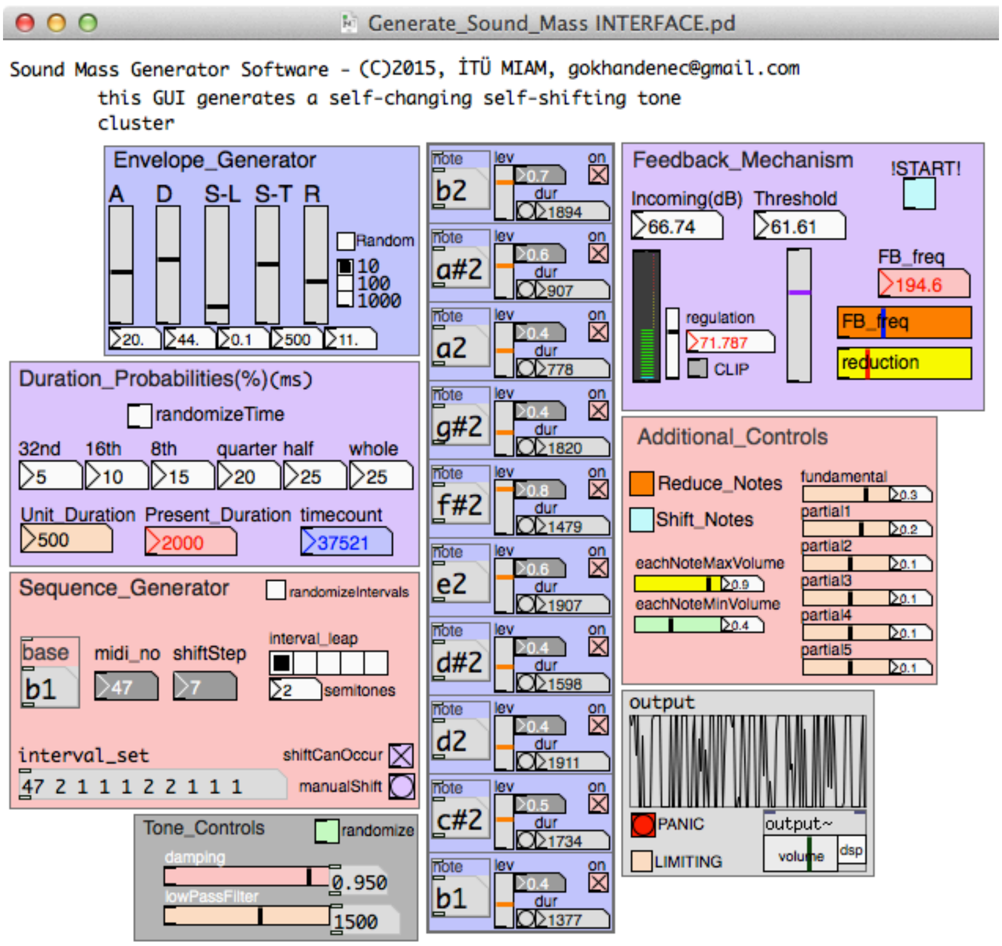

Figure 2.10 : The main window of the sound mass generator GUI.

This is the proposed software setup that controls the hardware on the piano. What is seen here in the GUI is the generation process of a tone cluster, as an example a 10-

note tone cluster being constructed. Number of notes in a cluster can be changed to a desired amount, however this needs to be done by going back into the programming stage and it is available as an option in the proposed design. A 10-note cluster is a part of the design concept; as the human performer has 10 fingers to play the piano, further cause and the details of this selection will be explained later in the Chapter 3. As the most significant part is the tall block in the middle, it is the triggering signal being the tone cluster. On left hand side, the generator functions for the construction of the cluster notes and durational properties can be seen. On the right hand side, the functions related to the creation and analysis of audio takes place. *Sequence Generator* generates the inner tonal architecture of the cluster. *Durational Properties* controls the tempo and the relative note duration types. *Envelope Generator* shapes the generated sound in time domain, *Tone Controls* shapes the sound in frequency domain, *Feedback Mechanism* keeps track of the incoming and outgoing signals as *Additional Controls* components enhances the sonic palette.

Sound level control

These blocks create the feedback and instantaneously monitor the incoming signal and its spectral components. The level comparison of the threshold value and the incoming signal strength is significant here. Until the two levels will be matched, the software will adjust the signal level being sent to the piano. If this has been achieved, the software will prepare a change in the triggering signal. This is the state where desired sound level has acquired from the piano. Tonal changes would be prepared as a pitch-shift for the base note and a new tone cluster to be built to shift to other regions. This movement will affect the occurrence, overall level and the harmonic content of the feedback. Other functions of the software and detailed explanations will take place, as the individual components seen on Figure 2.9 will be examined in detail in the next chapter.

Building a tone cluster

As stated earlier, the long block seen in the middle is the notational representation of the generated tone cluster. The note on lowest row is significant here, it is the generated *base note*; in Figure 2.9 it is B1 (B in the 1st octave). Other properties of this note block will be explained in detail.

Going back to Figure 2.9, on the left hand side, a collection of numbers consisting of *1*'s and *2*'s can be seen. It is the list of the corresponding intervals of the randomly generated tone cluster and note that there are 9 intervals in a 10-note block, by nature of the process. 1's and 2's indicate the interval leap amounts, either minor 2^{nd} or major 2^{nd}. This collection of numbers indicates a prescription of a tone cluster of 10 notes.

Notion of rhythm

Time management of the generated sounds are handled in a certain way that mimics note values; like the quarter or the 8^{th} note for example. Each generated tone cluster is assigned with a generated duration. This feature mimics as if the notes are actual notes played from an arbitrary score and the outcome is to be heard as a musical event, rather than a collection of randomized notes and durations.

Each note value has its own generation probability and this can be set in the *Durational Probabilities* section. These note durations are calculated as the integer multiples of the smallest duration possible, the *Unit Duration*. It is like the *Pulse* in music, used for creating a certain rhythmic structure. Unit duration is an atom in milliseconds and the value can be manually edited. Other rhythmic note values are acquired as integer multiples when this atom duration is doubled, tripled etc. The generated duration of each cluster is shown in the *Present Duration* number box.

Timbral organization

The selection of the synthetic sound source is developed as a result of experimentation. Experimented with both individual and combinations of basic synthetic waveforms (sine, triangle, square etc.), also with continuous-frequency collections in the form of filtered noise, it came to a conclusion that the proposed electromagnetic transducer setup is a highly versatile tool that it can produce acoustic outcomes with huge timbral varieties. The strings were exposed to different sources of material; single frequency waves, combination of basic wave types, sound samples, groups of discrete harmonics - tonally related and/or unrelated - in varying amplitudes, and also sculptured noise, each time the sonic outcome was drastically different. As the wave type had become richer in higher harmonics, high frequency response had increased accordingly. In the wake of McPherson's spectral evaluations, the usage of noise made the initial transient of the sound sharper as the high frequency component was rich and present. A combination of periodic waves and noise is used as the synthetic source, also a number of fine tunings were done after studying McPherson's spectral comparison of acoustic piano note and triggered sound, in Figure 2.11 (McPherson, 2010).

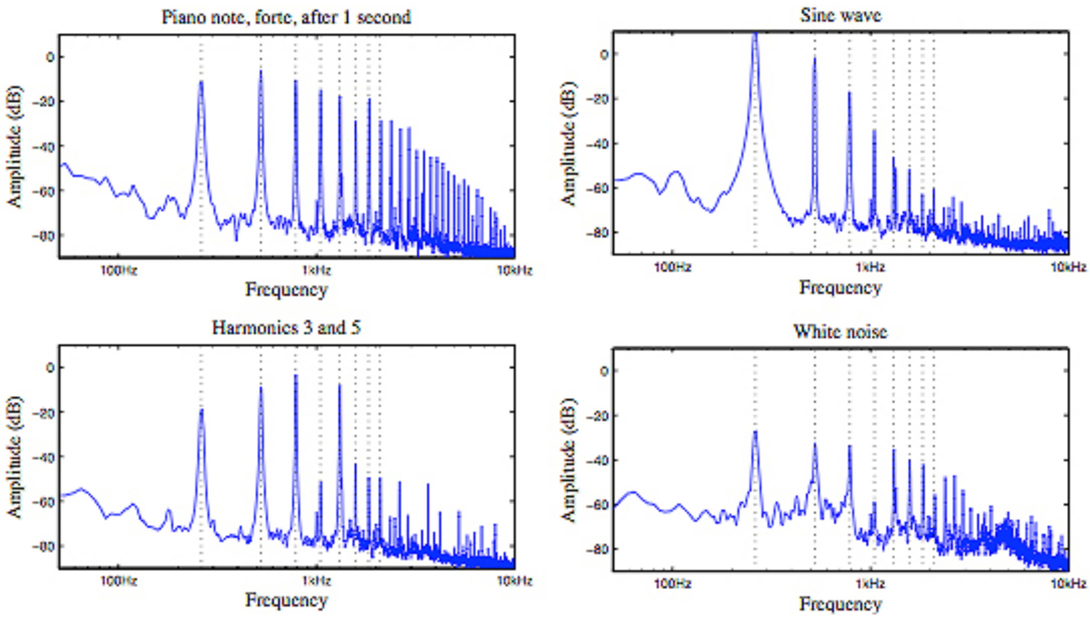

Figure 2.11 : Spectra for actuation with several waveforms compared with a standard piano note, C4 (McPherson, 2010, Figure 5).

According to McPherson paper, comparative analyses are made among frequency spectrums of the actual piano note and the actuated signals. In Figure 2.11, on the upper left, the spectrum of the actual piano note C4 is shown. The dashed vertical lines correspond to first 8 harmonics of the string. It can be seen that the first 8 harmonics are quite prominent and the later ones exhibit a linear gradual decrease in amplitude. On the upper right, the spectrum when actuated with a single sine wave; being the only fundamental frequency, is shown. Naturally, the fundamental is emphasized, higher 7 harmonics are present but it is examined that they lose energy rapidly. Above these, in the higher frequency region only weak components take place without any prominence. On the lower left, the spectrum when actuated with the 3rd and the 5th harmonic is shown; naturally the 3rd and the 5th harmonics are the highest in amplitude. Their natural vibrations evoked a considerable amount of the fundamental and the 2nd harmonic, too. It can be clearly seen that the 4th harmonic fell short but interestingly, the rest being the higher harmonics show a bit of resemblance of the actual piano note. The high frequency content is not lost rapidly as in sine wave spectrum. On the lower right, the spectrum when actuated with white noise is shown. The first eight harmonics are present with a few deviations, the 4th harmonic is surprisingly weak, and the overall amplitudes of the harmonics are proportionally smaller as if they are scaled down. However the high frequency energy is remarkable and kept present in white noise actuation.

As a guideline, it can be said that if the noise source will be amplified enough and with the addition of sine components, the timbral characteristics of the piano note can be imitated. Periodic waves can strengthen the desired harmonic (or harmonics), at the same time usage of noise gives a boost to the higher frequency components. Even if timbral imitation to the original piano sound is not intended, by combining periodic components, and the addition of noise, both in varying amplitudes, immediate changes in timbral quality can be acquired.

It is decided that in this software setup, all the notes in the tone clusters will be created with likely combination. Each note is created with a combination of by *Karplus-Strong* algorithm [available in G.o.T.] and first 6 harmonics with *additive synthesis* [available in G.o.T.] to strengthen the first order of harmonic components.

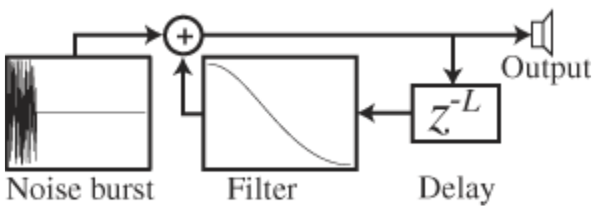

Figure 2.12 : Illustration of Karplus-Strong algorithm (Karplus and Strong, 1983).

Further manipulation of the harmonic content in terms of level and duration made possible with advanced options, those will be explained in the 3rd chapter.

As the strings are put to vibration with the produced tone cluster, the sympathetic vibrations occur in the piano strings. Each cluster excites more notes than the amount of notes in the cluster, because harmonics correspond to and match in more than a single string. For example, any arbitrary note that is played into the piano evokes reflections from multiple strings if it matches the harmonic spectrum of that set of strings. The removal of the damper pedal also increases the effect of this principle.

Envelope

The *Envelope Generator* affects the triggering synthesized sound. In this section, there are conventional settings for attack, decay, sustain and release properties of an envelope. These A, D, S_L, S_T, R (*attack, decay, sustain level, sustain time, release*) input boxes can be edited and quantities can be assigned from the sliders, or as an alternative option they can be automatically generated within certain boundaries.

Tone Control section in the GUI is linked to the K-S algorithm parameters. There are two parameters: *Damping* is the amplitude of the feedback signal in the algorithm. In the K-S algorithm, the feedback amplitude should always be less than the original signal; this is how K-S feedback is kept under control. This parameter gives color to the synthesized string sound, controlling the sustain time of the modelled string. *LowPass Filter* sets the filter frequency that is affecting the feedback. This eliminates the excessive high frequency content of the modelled string, which is especially more prominent in the initial transient part.

2.3.1 Concept of the monitored signal and the feedback component

Acoustic sound in the piano is constantly monitored. The incoming signal form the piano is monitored via the microphones placed inside the piano. As mentioned in earlier in Research and Development section, contact microphones are used for monitoring the vibrations in the piano strings. It was tested and decided that air-pressure microphones cause inconsistency and won't work efficiently for such an application because they pick up the sound in the environment and periphery. By their nature, contact microphones stick to the surface and they pick up only the sound from the instrument body; suppressing the noise in the environment. The proposed setup has a contact microphone under the soundboard of the piano. The microphone is a replica of an AKG C-411 model.

Figure 2.13 : Example for an air-pressure microphone; AKG C-451, left and a contact microphone; AKG C-411, right (Url-9 and Url-10). Sizes are not proportional.

The acoustic sound emanated is monitored and evaluated; this is how the software controls the actuator signal strength. Incoming signal level is instantaneously compared with the *Threshold* level.

Sound level management

The *Threshold* is actually the desired level of the sound level created in the piano. It is set by the user also have a default level. Until the level indicated by the *Threshold* is matched, the software automatically adjusts the level of the actuator signal that is sent to the piano. The threshold is a scaled control, as could be seen in the screenshot. The number in this control corresponds to a *decibel* [available in G.o.T.] value, as the signal levels in Pd are expressed in decibels[6].

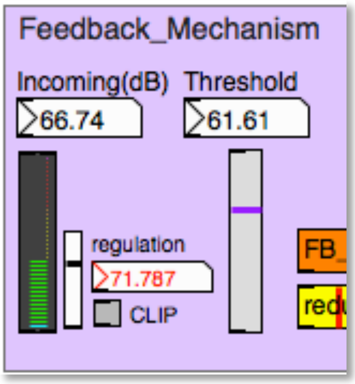

Figure 2.14 : The threshold setting.

If the sound level monitored from the piano is less than the set threshold amount, the software automatically increases the triggering signal strength up to an arbitrary point, until it will meet the desired amount. This is done with positive feedback. If the incoming sound level exceeds the desired amount than the triggering signal strength is reduced with negative feedback, again until an arbitrary level where it meets the given threshold. This is the first function of the sound captured by the contact microphone. The gain levels can be seen in the number box named,

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[6] Decibel levels in Pure Data are arbitrary, does not relate to real-world Sound Pressure Level (SPL) calculation.
In Pd, amplitude levels are rendered in proportion to the absolute maximum level available in the sound system – that is specified as 0 dB FS [available in G.o.T.]; decibel fullscale.
Any audio object in Pd outputs audio samples values between -1 and +1.
Amplitude levels though, are expressed in absolute-value numbers and are calculated between 0 (zero) and 1 (one) - as the maximum amplitude is indicated by and normalized to 1.
As a preference in Pd, maximum audio level coefficient 1 corresponds to 100 dB RMS amplitude level in the digital sound system.

regulation. This is the 'regulated', evaluated level of amplification. This value is a coefficient, does not relate to the dB unit.

Figure 2.15 : Illustration of how the signal strength is calculated. Threshold is set manually. Volume adjustment is done automatically.

Another function of the signal captured from the microphone is to cause acoustic feedback. In order to avoid excess rumble from the wind and the environment noise, the properly filtered version of the input signal is sent back to the piano. This creates a new layer of sustained sound, different than the actuator signal. The actuator signal acts as if it is setting the potential regions beforehand for the feedback to occur. The feedback level and frequency is manipulated with the details mentioned in the Chapter 3.

Shifting the tone cluster

If the monitored signal level matches exactly the indicated level in the *Threshold*, it will be a call for a change. The software assumes that there is enough sound in the piano, matching with the desired level. Considering the free path of feedback, this acts as a control over the feedback. The tone cluster generator runs and produces new set of parameters that will lead to creation of a new cluster. This is as if following an arbitrary score, a transition is made from one block of notes to another. A new cluster will be constructed with the same stages as described above; first a new base note will be selected, then the intervals and then time and envelope patterns. The result will be played back as the signal is sent to the piano. It is crucial to note here, any transition will wait for the permission to be implemented – all clusters will be played back with their assigned durations (note values that are generated at 'birth' by the *Durational* Properties). After the generated duration has passed, it is only then a transition can occur.

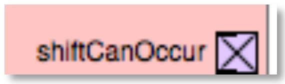

Figure 2.16 : The indication of a possible transition, in the *Sequence Generator*. This indicates that the generated duration of the cluster has passed and if the threshold is matched, a shift can occur.

The transposition amount of the base note is calculated following its own weighted random distribution. In summary, this pitch-shifting motion can be described as:

a. If the incoming signal level matches the threshold level, that means the level is not lower nor higher than the desired level, enough sound power on the piano is achieved. There should be a change in sound parameters in order to avoid the feedback before getting uncontrollable. This denotes a change in the actuator signal, which is constantly leading the path to the feedback.

b. A pitch-shift for the 'base-note' comes into play at this stage. First, the direction of the 'transposition' would be decided – the movement to either higher or lower notes. This is defined clearly in the software, as a %50-%50 probability.

c. Second, the width of the shift will be in terms of intervals. There is a set of constraints defined for the probability measures for each interval leap within an octave. This table is developed with subjective evaluation of the writer and contains fixed values (Table 2.3). If desired it can be re-defined by going back into the programing stage, however this feature is not accessible on the GUI.

Pitch-shift leap amount

In Table 2.3, all possible intervals are considered within an octave range and compared by the writer and a subjective evaluation is made to assign probability values of possible pitch shifts. Interval leaps bigger than an octave is not considered, as it would be an abrupt jump and a distraction to the natural flow of the sound. Using these weights, randomized pitch-shifts are calculated.

Table 2.3 : Probabilities defined for transposition intervals with the writer's comments regarding stylistic results.

Interval	Probability %	Stylistic Comment
Minor 2nd (m2)	2	Lacking feel of change
Major 2nd (M2)	4	Lacking feel of change
Minor 3rd (m3)	6	Not bad
Major 3rd (M3)	7	Not bad
Tritone	8	Acceptable
Perfect 4th (P4)	9	Acceptable
Perfect 5th (P5)	10	Acceptable
Minor 6th (m6)	12	Sounds good
Major 6th (M6)	13	Sounds good
Minor 7th (m7)	14	Favoured
Major 7th (M7)	15	Favoured
TOTAL	*100*	

It is worth mentioning that only the destination note will be generated as the new base note of the next tone cluster. The pitch-shift algorithm generates only the base notes, as the rest of the cluster notes will be generated with random intervals, built on consecutive notes.

Although the values in the list are the writer's personal comments on how these leaps sound musically, with the fact that transpositions affect only the base note, each new cluster will consist of new intervals groups. The subjective results here are evaluated in relation to the 'imitative' pitch-shift concept. As new upper voices will be generated each time, even the generated tone clusters on the same base note will consequently have different internal structure and each cluster will sound differently. This feature enables the sonority and transition process to be 'unforeseen', never-the-less probability collections are derived from subjective results.

This 'transposition' triggering mechanism and the augmentation of sound level has direct consequences both the actuator signal and the feedback level. Using this self-refreshing design, the sound produced by the setup can trigger itself and keep the progression of the 'generated music' autonomously. This feature is one of the mechanisms that help keeping the feedback level under control.

Hence the constant build up of the feedback frequencies is avoided. This is a critical design feature: feedback levels can rise to a limit and by cluster pitch-shifts feedback is loosened and the potential resonances are carried to new regions.

2.4 Constraints and Bottle-necks

In summary, the hardware and the software components generate a bunch of desired signals and gets into interaction with the piano hence create acoustic sounds. This schematic design has certain constraints and bounds specified by several significant conditions those directly affect the outcome and the efficiency of the whole setup.

Indoor acoustics and interaction with feedback

Room acoustics have direct relevance to the occurring feedback. Indoor spaces constantly dictate the potential regions for feedback with their room modes and resonant frequencies. Also nearby surfaces and objects that reflect and/or focus the sound such as concave, corner-like structures affect the occurrence of feedback. Especially the walls and the floor can create strong standing waves around and under the piano that for some notes they could couple acoustically with the strings and get into interaction with each other. It is examined that the feedback mechanism is limited in variety with the room acoustics.

This setup had been developed in a typical study room where it was not acoustically treated. To diminish the interaction of the piano resonance and the room modes, the instrument was moved around in the room and left in a position where it's less likely to couple with the room. However, the efforts receive a partially return as the physical outcome, since the sound is created in an encapsulated physical place, the setup is bounded by sound behavior in that space. This has a direct leading effect on the occurring feedback frequencies.

Tuning and temperament issues on the piano

As the details will be explained in the following chapters, the actual frequency numbers of the notes those will be played are automatically calculated and assigned in the PureData software environment[7]. The default way of the tuning is the *equal temperament* and this setting is not altered during the programming stage for this work.

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[7] As a programming detail, in Pure Data patches of the proposed work, notes are generated as MIDI notes and their frequency values are evaluated automatically with the [mtof] object – stands for 'MIDI to frequency'. So no particular evaluation is considered in terms of temperament system since Pd works in equal temperament by default.

However, there are two major problematic issues concerning the actual tuning of the piano. With the principle of sympathetic vibrations, it is possible to vibrate some other body if the source vibration corresponds to, in other words if the two frequencies are matched *exactly*. If these two values differ in slight amount, the amount of vibration exerted on the target is greatly lessened. Furthermore, as the frequency difference widens, the efficiency is dropped critically and acoustic response from the target body cannot be obtained. This is a strict limiting condition for the proposed setup and the causes for this will be explained in detail later, but the piano strings should be tuned to their corresponding frequencies in accordance of a tuning system. It may be the case that many pianos that are inside out reach are tuned acceptable, some rather well if not perfect, but the assumption here is that it must not be totally out of tune. Otherwise the source and the target frequencies will not be matched and no force would be transmitted to the target body. Since it is not easy to tune a piano like a guitar for example, this initial condition of the piano holds a major significance.

Another issue is the temperament system used in tuning the piano. In equal temperament the distance from one pitch to its nearest neighbor is considered the same, and this interval ratio is repeated to generate all other pitches. Musically this might be bringing the allowance of transposition to any desired keys without the risk of sounding dissonant, but acoustically the pitch frequencies are augmented from their natural harmonics. In a piano tuned with equal temperament, the proposed setup would produce broader resonance, more tension and acoustical beating.

However, in just intonation the tuning system are evaluated from the harmonic series where any vibrating object in nature exhibits the same spectral pattern. In this tuning system, pitches are calculated as the ratios of frequencies resulting in a natural compatibility with the harmonic series. The intervals sound pure and harmonious. The downside of this is, compositionally same intervals on different keys sound differently in terms of consonance and dissonance. This gives the key-color to the composition. If such a piano, tuned with just intonation were used with the proposed setup, with the principle of sympathetic vibrations more acoustic energy could be evoked from the vibrating strings; since pitches would be tied to each other within their harmonic relationships and resulting sounds would be more harmonious.

3. SOUND PROCESSING WITHIN THE SOUND MASS APPROACH

Primary aim of the study is to trigger acoustic sound from an instrument and in turn process it. The nature of the setup is designed to able to run on its own and primary function is to create musical patterns in reference to Sound Mass compositions. In order to achieve this goal, organization of sound and its attributes are specifically designed in synthesis and processing stages. The sound capturing and processing stage is realized with Pure Data environment. In order to clarify the stylistic decisions made in the software and relate them to the musical concept of Sound Mass, first the compositional approach will be explained by mentioning its characteristic attributes. Later, technical explanation of the software with functional and conceptual analysis will be given.

3.1 Stylistic Features of Sound Mass

The term *tone cluster*, the atom of sound mass style, was postulated in 1912 by the American composer Henry Cowell (1897-1965) and adopted in his own works, at first on the piano whose keys were depressed with the whole hand and lower arm. In his own words:

> "The tone-cluster is simply a group of two or more minor seconds; that is, it is a cluster of three or more tones, each a half step from its neighbor, sounded simultaneously" (Cowell, *Harmonic Development of Music*, 1921. Higgins, 2002, p. 282).

A cluster causes all the chromatic notes to sound simultaneously within a prescribed compass. The ear cannot perceive any particular pitch (except perhaps the higher and the lowest note) since the pile of notes includes an overlay of harmonic upper partials as well as their fundamental dissonances (Swinger, 1989).

György Ligeti (1923-2006) had foreseen that a new sensation for musical form was emerging at his time. As he concluded in one of his essays that: "what is necessary is to try and achieve a compositional design for the process of change". In *Apparitions* (1959) and *Atmosphères* (1960), Ligeti created his trademark namely the 'Ligeti sound' (Steinitz, 2003). His musical textures of 'continuous' essence; based on a fluid approach of musical time using stationary sounds, constructed as clusters that create dense structures with extremes of registers and dynamics. This evoked the sense of continuity and unfolding; like a musical entity without beginning and an end. (Toop, 1999).

Later he coined the term '*micropolyphony'* [available in G.o.T.] and explained this texture as:

> "The dense counterpoint in which one can no longer hear the individual voices, but is simply aware of changing degrees of activity" (Toop, 1999, p.70).

Another contemporary composer Krzystof Penderecki (b. 1933) had a similar approach in *Anaklasis* (1960) and elaborates this approach in *Therenody for the Victims of Hiroshima* (1960) with its screaming clusters and whole array of unorthodox performance techniques (Toop, 1999) the piece ricochets from pitched sound to noise. Although Penderecki and Ligeti became associated in public mind, their technical approach was significantly different: Penderecki emphasized broad washes, generalized clusters, glissandi and noise effects, while Ligeti a far more intricate micropolyhonic web in which every part is individually shaped.

Karlheinz Stockhausen (1928-2007) shared Ligeti's approach in the idea of 'moment form' where each moment of a piece stands by itself, yet related to all other moments. He explores this concept in his works *Gruppen* (1955), *Kontakte* (1958) and *Carré* (1959).

According to Iannis Xenakis (1922-2001), 'mass' effect in music draws parallels with the natural world as swarm of bees, flocks of birds, clouds billowing across the sky. He explored this vision of massed sonority and aggregated individual pitches to create differentiated sonic 'characters' in his works *Metastasis* (1953) and *Pithoprakta* (1956). In one letter from 1962, after analyzing the score of *Pithoprakta* Ligeti wrote to Xenakis:

> "There are musical textures and new sonorities in this work which serve more adequately than any other music as typical examples of the technique of global composition and of the destruction of the individuality of voices…" (Steinitz, 2003).

As the compositions of these composers imply, predominant stylistic feature of sound mass is the gathering of individual pitches as the building blocks to form a larger whole. A cloud-like structure of sound forms where the discrete components fuse into each other; at the same time metrical rhythm - rhythmic periodicity and precision - is loosened, expanded in time with an intricate *beatlessness*, despite the complexity of the motion.

3.2 Software Design and Concept

Henry Cowell's theory is the initial inspiration for building the sonic structure in the software stage. His theory defines a term as the combination of notes sounding together, called a *tone cluster* as Cowell coined the word; is built from intervals of seconds. Conventional compositional approach regards the intervals of seconds as dissonance, but he offers a new understanding from his historical perspective:

> "Ancient instruments were not nearly so rich in over-tones as our modern ones, and it is perhaps for this reason that only the simple major triad, formed by some of the lower reaches of the overtone series, was formerly regarded as a *natural* chord. On present-day instruments the higher overtones, as well as lower ones, are so easily heard that the ear cannot help being aware, when a single tone is played, of sounds which would formerly have been called discords." (Cowell, *New Musical Sources*, second ed. 1999, p.4).

According to him, tone combinations should have a sonic foundation. This foundation is the interval relationships in the tone combinations and the interval types should be present in the overtone series. Like the major triad intervals built with thirds, he states that half-step intervals can be used and in tone combinations they can form minor and major second intervals (p. 117).

In the following sections, the developed generative software model will be presented. A user interface is provided for controlling the functions and monitoring (as seen in Figure 2.9, p.22), following chapters will explain each component seen in the GUI. Conceptually, generative controlling functions are designed for sonority generation, amplitude properties, timbral properties and temporal properties. There are detailed explanations of each function with their design features, there are provided in the next sections in this chapter.

3.2.1 Sonority generation

To build the sonority, a generation algorithm runs in the software making use of randomness; within certain boundaries. This algorithm, assuming a 10-note tone cluster is to be constructed, for example; runs the following tasks:

1. A base note would be selected, as the fundamental, lowest note of the cluster. It is the foundation, which also will gain additional importance in the pitch-shifting decisions that will be explained in detail later.
2. Remaining nine notes in the cluster are selected by generating successive intervals; independently randomized distance relationships of new notes in relation to the previous ones.

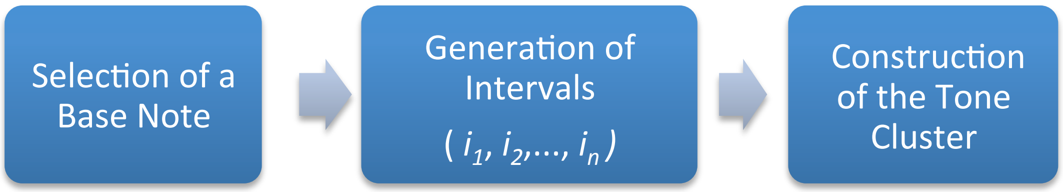

Figure 3.1 : Illustration of how the tone clusters are built with a selection of a base note and (n-1) individual intervals to make up a n-note cluster. All decisions are made randomly.

When the software is run for the first time, initial selection of the base note is random – it can be any note. But for the consecutive tone clusters, following base notes are the results of the decisions and selections in the *pitch shifting* mechanism; that shifts the base note to another note and builds a new tone cluster, details will be explained in the next section.

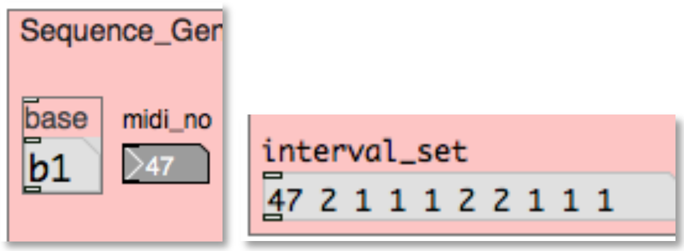

Figure 3.2 : Generated base note example; in the *Sequence Generator* component. Corresponding MIDI note number and the generated note interval sequence can be seen. 1's and 2's are the number of semitone intervals; first member of the sequence is the base note MIDI number.

The distance relationships between each note are defined as leaps in terms of intervals. For example, if a cluster of *n* notes is to be created, in the first step the base, $note_0$ is generated. Following this base, a list of *n-1* intervals will be generated as: i_1, i_2, …, $i_{(n-1)}$; respectively. Each generated interval amount will be added starting from the $note_0$ hence the cluster will be constructed by the notes $note_1$, $note_2$, …, $note_{(n-1)}$.

To keep the concept in the borders of Cowell's theory, default interval choices consist of 2 alternatives: it can either be a minor-second (*m2*) interval, or a major-second (*M2*) interval. For this reason, the basic sound construction idea in the design of the software relies on building tone clusters with major and minor second intervals. However, successive intervals of seconds are to be used in order to stick with the theory in the initial stage, it is observed that if the tone cluster is diversified and manipulated with larger intervals, the outcome was richer in terms of musical variety. Thus, interval types (2, 3, 4 and 5 semitones) are made available as a selective option in the software.

Interval selection option

By default, the construction of tone clusters is based on seconds but other interval types can be be assigned by this option. It is worth noting that by selecting an interval type, it means the maximum number for interval randomization is specified. For example; if interval type of 5 semitones is selected for cluster generation, each consecutive note will be built with either a minor 2nd, or a major 2nd, or a minor 3rd, or a perfect 4th interval relationship, regarding the previous note.

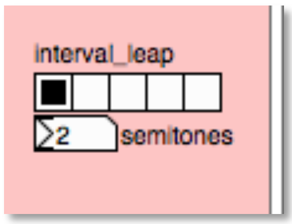

Figure 3.3 : Interval leap amount, in the Sequence Generator module. This affects the neighborhood relations of the notes in the generated clusters. Available values are: 2, 3, 4, 5 or 6 semitones, corresponding to the intervals of minor 2nd, a major 2nd, minor 3rd, a perfect 4th or a tritone.

With the basic fact that the most crowded note group that can be produced with two hands – using both the fingers and the palms, a design limitation was identified as the following: when intervals of seconds are considered, it is assumed that a maximum of 10 notes could be performed with two hands. This limitation does not cover the technique of playing with the arms and elbows that Cowell was famous of. The idea of playing with the palm, arm and elbow is left out intentionally. It would be hard to focus on precision of the intervals of seconds, besides the software already has a non-deterministic way of decision-making, the idea of playing with the arms or elbows overlaps with this nature and exaggerates the sense of randomness. The significance of each note then would be questioned and it could lead the nondeterministic, unforeseen approach of the design to unspecific, obscured path. Inspired from the nature of two hands, number of notes, ten, is both a choice and a constraint of the design.

The number of notes in clusters can be set to any number desired; but as a design feature it is not made available in the software option. This change though can be realized by going back to the programming stage. There are optional manipulations regarding the note number in the clusters and sonority construction, which will be explained in the next section.

Note block

Considering the main GUI window in Figure 2.9 (p. 21) in the 2nd chapter, the vertical purple block in the middle section represents the generated tone cluster. Lowest note is the base note, as the block goes higher other members of the cluster can be seen. There is more than a typical note representation in this collection of data, both conceptual and functional. So it is worth to mention the major properties and the indicators of each note-representing block, as one individual element is seen in Figure 3.4.

Figure 3.4 : A block representing one note and its properties, in the GUI. Information could be monitored on this block are the *Note Name*, *Amplitude Level*, *Active/Passive State* (on/off), and *Duration of Transition* to a new amplitude level.

In this single note component, some important information could be monitored:

i) *Note Name*: The name of the note, B1 in the example, is the generated note. This block is taken from the lowest note in a cluster, meaning it is the base note of the cluster. Higher notes are generated by the *Sequence Generator* component, which will be explained in detail.

ii) *Level* and the *Active/Passive State* of the note: It is a design feature that each note in the cluster has their own amplitude values. This feature carries the conventional dynamics notation in music to a further stage. Taking advantage of the signal processing software; if one decides to play several notes at a time, as the performer is not a human being but a computer, individual dynamics can be assigned to each note. Conventions in Pd are followed here; amplitude levels can take values between 0 and 1. Minimum and a maximum amplitude levels can be controlled in the *Additional Controls* component, individual levels are generated between these two boundaries.

Individual note amplitudes change dynamically in time, this enables the composition of the generated sound-clouds to vary in time with ever-changing resonances. The vertical slider in the block shows the amplitude level and the corresponding number, the duration number indicates the transition time from one amplitude level to another, controlled automatically and changes can be monitored in the level controls.

There is another powerful but optional feature to enhance the musical variety: notes in the 10-note cluster decide to be sounded or be muted during the performing stage. This allows the creation of different sonorities from the same generated note sequence; for example. Also by changing the intensity of the clusters the feedback gravitational center is affected. Instead of playing notes in close neighborhoods, leaps in the note sequences allow the feedback to be weakened and its level is kept under control in an easier fashion. This option can be activated or deactivated from the *Additional Controls* component; the indication of a note being played or being eliminated in the cluster representation is the toggle box on the upper-right hand side of each note block.

Further manipulation of sonority

Optional features are developed to avoid static tone clusters playing one after another. Some of the features are mentioned in the previous block but *Additional Controls* component will be explained in detail here.

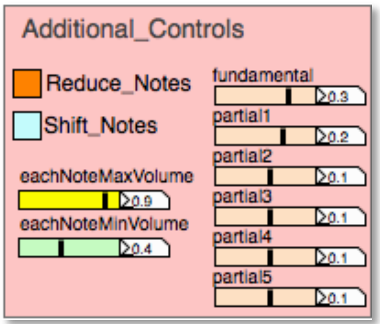

Figure 3.5 : *Additional Controls* component. Optional controls of Note Reduction, Note Shifting, Note Amplitude Levels and Tone Shaping of the oscillators are available here.

a. Default feature of individual amplitude assignment for each note: Each note in the cluster can have their individual amplitudes, like the dynamic variable: *touché*. But this is more complex than simple dynamic indication (like ***ppp*** of ***fff***) because all the cluster-notes are generated with their individual amplitudes. This result is not easy to achieve when the standard performance of the piano is considered. This enhances the sonic intensity and changes the variety of the string vibration. This has direct effect on the feedback mechanism, as the sound-color picked up from the piano changes the potential frequencies for feedback saturate and spread along the strings/partials.

   The minimum and maximum note amplitude levels (namely: *eachNoteMaxVolume* and *eachNoteMinVolume*) can be set in this component and individual levels are generated within these levels.

b. *Reduce Notes*: A 'reduced' version of the tone cluster can be built as an option. With this feature, the routine, constant ringing of full 10-note block would be broken into portions with leaps. As the notes in the tone cluster are reduced, a random selection of notes would be muted when the others on this 10-note block are voiced. This has a broader outcome than the previous feature; it changes the composition of the *tones* in the cluster; in extreme situations it even abandons the tone-cluster concept. It is analogous to the uncut jewel – with this option the rough sound cloud gains a more elegant shape as the lines of the motion are softened and the harmonic texture is morphed. Again, by the fact that feedback is also put into play, this option strongly colors the feedback. It tames the feedback by loosening the tonal texture, this way it avoids it to reach to the amplitudes beyond control.

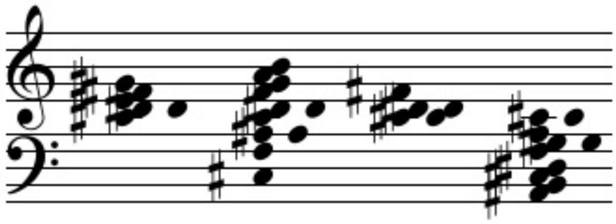

Figure 3.6 : An example for 'reduced' version of the tone-cluster; four successively generated 'reduced' clusters by the software. In this specific generation, instead of 10 notes a 16-note cluster was used to help pointing out the varieties in the software, and it was opted to reduce notes in the cluster. As could seen in the example, the generated four blocks consists of 6, 11, 5 and 10 notes consecutively.

c. *Shift Notes*: An unfamiliar *glissando* is produced when this option is selected. When selected, each note decides to shift or not; frequency-wise. This shift feature is an imitated glissando and realized by a specific percent rise or drop in the frequency domain. This randomly generated feature dissects the initially generated 'tonality' of the cluster and passes over in the realm of scales of micro-tunings. Because of its random nature in calculation of the shifting frequency leaps, the emerged sounds trespasses through a complex level of dissonance in terms of tonality and harmony. Upon hearing, the majority of the listeners' response was the resemblance of this movement with the characteristic Penderecki slide.

d. Each note sound is realized by a typical Karplus-Strong algorithm and additive synthesis of first 6 harmonics; the fundamental plus 5 partials. First 6 harmonics are the combinations of sine waves and their individual amplitudes can be controlled as desired in this component.

These are the major functions that shape the sonic palette of sound generated by the proposed software setup. In the following sections, details of the inner structure of cluster intervals and time domain controls will be explained.

Organization of intervals

In Figure 3.7, *Sequence Generator* module in the interface block, responsible for building sonorities with generated tone clusters can be seen. The major functions and information seen on the block are as follows:

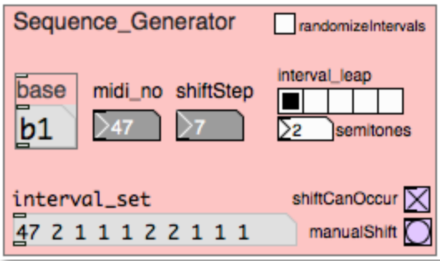

Figure 3.7 : *Sequence Generator* module, optional adjustments and the available information that can be monitored.

i) *Base note*: This component generates the base note; B in the 1st octave in this example. Corresponding MIDI note number is 47, can be seen in the number box.

ii) *Interval set*: A collection of intervals is used to generate the following notes of the clusters. Other than the base note, 9 intervals are randomly generated - the ceiling value of *Leap Amount* selection taken into the account. This information block is carried to other software components as the general prescription of the cluster. In the example, 47 (B1) and numbers 1 and 2's indicating the minor and major second intervals are listed. This sequence will produce the tone cluster: B1, C#2, D2, D#2, E2, F#2, G#2, A2, A#2, B2. Note that the first element in the sequence is the base note.

iii) *Shift Step*: When the software decides to move the tone cluster at specific times (the details are explained in section 3.2.3), this is the generated interval leap amount from the previous base note. This leap amount is a generated value and calculated with the probability data on Table 2.3, p.29.

iv) *Interval Leap*: Option for selecting the maximum semitone value for each interval to be generated. Available options are 2, 3, 4, 5 and 6 semitones; corresponding minor 2nd, a major 2nd, minor 3rd, perfect 4th and the tritone.

v) *Shift Can Occur* indication: Indication of shift availability - will be explained in detail but worth mentioning here. The software assigns rhythmic note values to each cluster. Also it keeps track of the duration of each cluster being played. If the duration properties are fulfilled, meaning if any cluster is sounded at least as long as its assigned note value, then it gives the pitch-shifting mechanism permission that it can produce a new cluster. If the rhythmic note value had not been processed though, permission is not given for a new cluster. This state can be tracked with the *Shift Can Occur* toggle box.

vi) *Manual Shift*: If a shift to a new tone cluster is desired, this button can be pressed any time during operation. This overrides whatever the software is doing at that certain time and forces it to pitch shift the base note to a new location and generate a new interval set immediately afterwards.

vii) Randomized function: Major components in the software are designed with a randomization option. This selection randomizes the interval leap amount in each new constructed cluster and adds variety to the sonic outcome.

3.2.2 Generation of durational properties

After the construction of the tone cluster, the durational properties are generated. There are two types of controls in terms of duration. First, the conventional *Attack*, *Decay*, *Sustain*, *Release* time parameters are set (in milliseconds). *Sustain Level* and *Sustain Time* can be set separately.

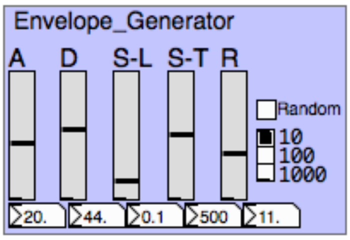

Figure 3.8 : Generation of envelope properties, the change of sound levels in time by the *Envelope Generator* section.

As the overall time for any cluster is generated, actual set (or generated) durations of attack, decay and sustain adjustments happen within this overall duration of the block. However, release occurs after the playing time of a generated cluster is finished, just like mimicking the release of a note that had been played on the keyboard. All parameters can take values of 1 – 999 milliseconds. The sustain-level parameter is generated between the levels of 0 and 1 in terms of amplitude.

Optionally, envelope variables can be generated randomly within the indicated bounds. When the randomization option is selected, each parameter takes a value that is randomized between three selectable settings: between 1-10 milliseconds, between 1-100 milliseconds or between 1-1000 milliseconds.

Tempo, rhythmic signature and note values

Duration Probabilities section is the provider of the rhythmic structure. Any cluster in general, can take the values of either a whole note, a half note, a quarter note, a eighth note, sixteenth note and a thirty-second note. Here what is meant with the note value is arbitrary. In convention, note values are directly related to the tempo, rhythmic signature and the duration of a single measure under that signature. Here and different values are in relation not with an arbitrary tempo. Instead of a beats-per-minute concept, rhythmic variety is derived from the actual note durations. The reference of note duration calculations is an atom, namely *Unit Duration*; that is in milliseconds. *Unit Duration* is the conceptual duration of a 32^{nd} note. Each cluster takes a note value generated by the specified probability distribution, each are the multiples of this Unit Duration. Each note values that are played at any time are indicated in the *Present Duration* number box; in milliseconds.

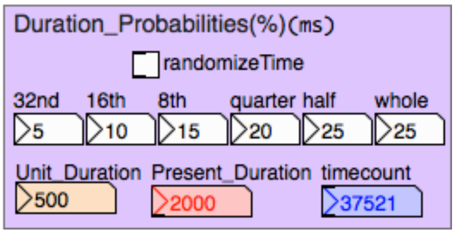

Figure 3.9 : Generation of the total duration of a cluster and the note values with weighted randomization. Rhythmic varieties of the notes are calculated from the *Unit Duration*.

When each cluster is generated, its individual duration is calculated and assigned to it. This calculation is straightforward but the operation has an exception:

a. The special value called *Unit Duration* dictates the time for the shortest, fastest note (a 32^{nd} note in this design). This number indicates the shortest possible rhythmic pulse in the sonic production of the setup. Hence, the overall rhythmic structure is dependent to the ‘durational atom’. The rest of the note durations are derived from this unit. In the software design, possible note values are 32^{nd}, 16^{th}, 8^{th}, quarter, half, and lastly whole note.

b. In *Duration Probabilities* component, the probabilities of the note values are assigned. In contrast to the dominant nature of the software, these values are not randomized. There is a set of values initially assigned but they can be edited and changed anytime. The sum of all the percentage values should be 100, and the software is automatically controlling this limit. Also, it is required to edit the probability values starting from left to right (from the fastest to the slowest note).

c. *Timecount* data represents the time that had passed since the beginning of that individual cluster being played. It is worth noting that, as could be seen from the example; there are times where the notes are actually played longer than their calculated duration values. This happens when the monitored incoming signal from the piano is not as strong as desired. The mechanism will increase the gain of the signal going in the piano to match the *Threshold* level, in the *Feedback Mechanism* component.

This exception will be better explained in section 3.2.3, where the details of the movement and pitch-shift controls are explained in detail. However, it is worth mentioning this exception during presenting the durational conception of the software.

Exceptions in duration processing algorithm

The desired incoming sound level, meaning the sound of the piano that is momentarily monitored in the software is set with the *Threshold* control as stated earlier. If the monitored signal level reaches the level indicated by the *Threshold*, it means the desired sound level is achieved. As a design feature, the software will go in a state of looking for a possibility to generate a new cluster. This tendency intends to keep the feedback level between the controlled bounds; before the feedback gets uncontrolled terms of level, the setup will generate a new tone cluster and then play it on the piano, moving the matching feedback frequencies to new harmonic centers.

Although this instance is explained in the next section 3.2.3, there is one strict parameter that this shift is tied to: it will only take this action if the cluster had been played back long enough; meaning only if the specified note duration (that was generated by *Durational Properties*) had been matched. If a note (or a cluster in general) is played back long enough, than *Durational Properties* will send a positive signal to the shifting mechanism that it can produce a new cluster, when the Threshold level is reached.

The *Present Time* indicates the duration of the notes being played and the *timeCount* constantly indicate the actual played back duration of that instance. If the rhythmic note value is has matched its desired duration but the *Threshold* had not been reached yet, there would be times that clusters can be played back longer than their conceptual assigned note values. At these exceptional times when the software waits for the enough sound intensity to build up in the piano, *timeCount* will overshoot the *presentDuration*.

This non-exact nature in the duration concept can be observed momentarily in *Present Time* and the real time counter, *timecount*. This exception however, allows the software to generate unexpected, abrupt rhythmic varieties. This somehow *loosening* in the durational strictness channels the software to meet with another stylistic feature of Sound Mass: sense of fluidness as the rhythmic structure disintegrates and dissolves.

Modifications of timbral quality

The concept of control parameters changing in a self-dictated but unpredicted manner creates rich, complex and fluid passages of rhythm, also of timbres, too. This unique feature enhances the number and the interaction of the actuated harmonics in the piano and the result can be evaluated as:

a. The sound properties of the actuator signal and the outcome it produces from the piano, especially for long sustain-times, vary unpredictably but somehow musically pleasing. The ADSR properties act upon only for the actuator signal but the response of the piano is most of the time different. The actuator signal might get into the decay phase but because the system works on feedback, the physical outcome might be different. Acoustic decay of the piano occurs independently from the release duration in the software. Removal of the damper mechanism also helps this long sustain, less energy-loss feature. Although the actuator signal changes properly, because the phase relationships between the actuator signal and the piano strings are not aligned, this difference creates musically complex but rather pleasing level-changes and changes in resonant frequencies.

b. As short durational passages occur successively, the new produced notes interfere with the previous ones; where the previous harmonic content is still present and in vibration along the piano strings. This causes successively generated notes to fuse into each other, an unplanned coincidence. This is not a result of meticulous compositional design but it is the nature of this experimental process. This leads to regions of dense intersections to happen and harmonic collections in sound clouds become rather complex and imitating the primitive examples of the phenomena namely *micropolyphony*.

c. As the feedback chain is active as a system characteristic, feedback interacts with the actual vibrations of the strings. As feedback grows, the software will force the feedback to stay inside the boundaries of controlled levels; implementation details are explained in the following section. As the software manipulates the feedback, dominating frequencies will differ in the actuator signal but the piano would be ringing with these frequencies. Hence new interactions on different notes will occur on different string partials, with the principle of sympathetic vibration. This also imitates the previously mentioned pseudo-micropolyphic gestures.

3.2.3 Generating movement and pitch-shift

As the sound intensity of the piano is increased enough; as the incoming signal is matched with the indicated level in the threshold, the software assumes that a certain level of stress has been presented to the strings and gets into the tendency of changing the sonic distribution. This tendency would be realized only if the *Durational Properties* component gives permission, in other words only if the generated cluster had been sounded by at least its assigned duration.

This *shifting* motion is designed to imitate the fluidness of consecutive notes being played as if in an arbitrary score. When the software will get into the state of preparing a new cluster, it generates a measure of leap considering the aesthetical opinions of the values in Table 2.3 (p.29). This selection is generated with the provided possibility distribution table, and the *Sequence Generator* module generates a single jump as a base note pitch-shifting. This shift is immediately followed by the generation of next notes in the cluster, prescribed as a list of consecutive intervals as discussed earlier. Hence, a new cluster is generated on a new note with new interval relations.

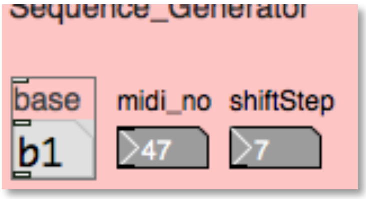

Figure 3.10 : Generated pitch shift amount, in semitones, shown in *ShiftStep* number bow. Corresponding destination destination note, and the MIDI number can be monitored in the *Sequence Generator* component.

When this leap amount is generated, it is indicated in the *Sequence Generator* component. The shift amount is shown in semitones – negative values indicate the movement towards lower notes. Next to this, the destination note is shown, both by its name and the corresponding MIDI number.

3.3 Feedback Control

Feedback Mechanism component controls and monitors highly important parameters in regard the rest of the software. This section is the audio command center of the software. Major functions of this block are;

- to handle incoming audio signal,
- to measured both in amplitude and frequency domains
- to scale the amplitude properly to match the levels set in *Threshold* indicator,
- to detect the feedback frequency,
- to process incoming signal by tracking the feedback strength and execute appropriate filtering to keep the feedback in control,
- and to send the processed signal back into the piano.

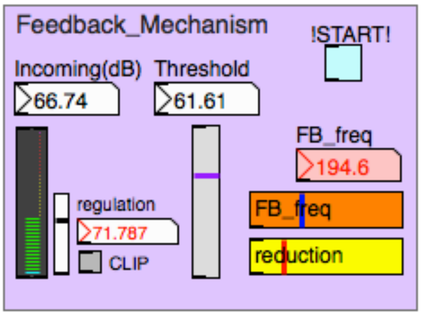

Figure 3.11 : Feedback Mechanism component, the heart of audio computing and signal processing in the proposed software. The most important setting in the concept, *Threshold* can be set here.

This component gives a beginning cue for all other components to run by the *START* command, manually selected in the toggle box. Most important parameter affecting signal level handling, the *Threshold* setting, can be adjusted from this component. *Threshold* is manually set arbitrary value that is continuously compared in number versus the incoming signal level in decibels[8].

Incoming signal strength is momentarily monitored, and indicated on the vertical VU meter. The momentary gain level is indicated in *regulation* box. This level is the amplification level for the incoming signal to match the levels dictated by the *Threshold*. If the incoming signal is weaker than the *Threshold*, this component will increase the gain of the outgoing signal. If the incoming signal is stronger than the *Threshold*, outgoing signal amplitude will be reduced.

This section also handles detection tasks for the dominant frequency in the occurring feedback and then applies a reduction by tracing these frequencies and filters them in an equalization function. Momentarily feedback frequency and the reduction level is also shown in this section.

As the actuator signal, the generated tone cluster is sent to the electromagnets acoustic sound that emanates from the piano is picked up by the contact microphone;

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[8] It is worth noting again here, that decibel levels in Pure Data are arbitrary, does not relate to real-world Sound Pressure Level (SPL) calculation.
In Pd, amplitude levels are rendered in proportion to the absolute maximum level available in the sound system – that is specified as 0 dB FS [available in G.o.T.]; decibel fullscale.
Any audio object in Pd outputs audio samples values between -1 and +1.
Amplitude levels though, are expressed in absolute-value numbers and are calculated between 0 (zero) and 1 (one) - as the maximum amplitude is indicated by and normalized to 1.
As a preference in Pd, maximum audio level coefficient 1 corresponds to 100 dB RMS amplitude level in the digital sound system.

that is attached under the soundboard. This specific place is evaluated and decided after numerous experiments for microphone positioning.

Incoming audio information is used and evaluated as the following:

a. The feedback must be kept under control by avoiding its eventual rise - but only beyond a certain level. Otherwise, some portion of the the acoustic sound caused by the setup would be the feedback. The *Threshold* level is significant at this stage. If the acoustic sound level in the piano matches the threshold this means a change must occur in order to distribute the feedback frequencies to a different region on the instrument. If a change is not made, the feedback will grow in an uncontrolled manner. This state triggers the pitch-shifting mechanism as explained above. This triggering however, would wait until the note durations assigned by the *Durational Properties* component.

b. Before the pitch-shift occurs, there is another method for the diffusion of feedback. From the monitored signal, the dominant feedback frequency is determined by processing the signal. This frequency is being filtered out with a *notch filter* [available on G.o.T.] implementation in Pd. This filtering mechanism is the main control in frequency domain that helps the feedback to occur freely, but if the levels get out of control, then forces it to diminish.

As an exception, sometimes the room modes resonate on the piano strings. This is not directly controllable in software but a standing wave in the room would rise to feedback sometimes. Most of the time, this standing wave is harmonically related with the actuator signal. That being said, the room and the placement in the room gains direct importance in this kind of setup.

As the feedback gets controlled as explained above, it accompanies to the actuator signals, creating an extra layer of frequencies correlated to the tone clusters. These extra layers of accompaniment become more prominent if the note values in the *Durational Properties* component are lengthened. This can be realized by assigning longer values to the *Unit Duration*. In short note durations, feedback might not find enough time to occur and dominate, though.

In summary, it can be said that as a self-generative live processing setup, there is a significant difference from other conventional live sound applications. Conventional live processing setups are designed and are run under the composers' direct influence. Here there is the inevitable disconnection from the composer's intention. Instead, this intention is substituted by stochastic decision-making process. The outcome *cannot be foreseen*, it can only be predicted forehand. The proposed software and hardware combination in this setup is an example of going beyond the *avant-garde* and falling into the *experimental* realm, by its nature; in connection with Cageian ideas presented in the introductory chapter.

4. CONCLUSION

The proposed work in this thesis consists of a live sound processing setup. The aim is to create resonances using sympathetic vibration principle with feedback manipulation. The selected medium is the piano. The method of the study is realized by developing a piece of software that is controlling a custom hardware: an actuator signal generation process followed by its transmission into the instrument via an electomagnetic transducer setup. As a result, acoustic sound is emanated from the piano. Primary aesthetic reference of the created sonic textures is the Sound Mass compositional approach.

When the writer of this thesis, who is primarily a sound engineer, had experienced listening to Sound Mass compositions of particularly György Ligeti's and other composers like Penderecki, Bartok, Xenakis and Scelsi, compositions in varying forms of orchestral and/or choir works to electronic music, had triggered the involuntary reflex of reverse engineering of the sound material. This idea of creating an autonomous setup on a musical instrument that will produce sonic structures like Sound Mass compositions gave birth to such subjective thinking and design concept.

The proposed setup is a package of both software and hardware, it produces sound sculptures from generative scripts with the form of 'imitated' sound blocks and gestures used in Sound Mass compositional approach. The generative process in software focuses on building tone clusters and manipulates the sound further by making variations. Actuator hardware setup sends audio signals to the electromagnets, which vibrate the piano strings without touching the strings or pressing the keyboard. The source of the resonances in string actuation is further augmented with the presence of a feedback loop. The actuator signal acts as a precursor to the feedback, the density of the feedback acts as a control parameter for the variations of the built sound forms. Hardware setup is constructed as a collection of electromagnetic transducers, software algorithms are programmed in Pure Data environment.

The hardware component was configured and offered as a solution to the need for a transducer type; that will not interact with the acoustic environment. Furthermore, this mechanism can be considered as a standalone tool for sound production from the piano. By proper alterations, this concept could be adapted to any instrument with metal vibrating parts. The software is designed for generating a particular musical outcome in terms of style, being Sound Mass, however it can be altered and customized in various ways for various musical outcomes as desired.

4.1 Comparison with Previous Work

It should be stated that at the time of development of the actuator setup, the previous works and studies of Per Bloland, Steven Backer, Edgar Berdahl and Robert Hamilton (CCRMA of Stanford University), Dan Overholt (Aalborg University) and Andrew McPherson (Center for Digital Music of Drexel University) on electromagnetic actuation of the piano were yet not known, but discovered in the later stages of this thesis work. The mentioned works had brought a new idea with their studies as they combine art, technology and interactive digital media (McPherson, 2012).

However, their works differ from the proposed setup with a major significance, their designs, too, were developed for expanding the timbral capabilities of the instrument but from a performance perspective. They aimed to pronounce one single note at a time. Their design concept is based on triggering an individual harmonic or a single frequency with a similar electromagnetic setup, but as if it was played form the keyboard of the piano. Different than the aforementioned designs, the proposed setup in this work aims to create rich resonances and sound bodies by sending one type of actuator signal to the whole range of the piano strings; to all of the strings at the same time. This approach favors a clash of sympathetic vibrations and harmonically related partials on any string are set to vibrate. These vibrations gather and build the world of sound that is aimed within the scope of this thesis work. Previous designs also proposed new methods for sound production, as a new medium of practice and composition. However, the proposed hardware setup proposed here is not the primary outcome of the work, but is a major component, an important tool to realize the aim of the thesis. It was the outcome of a development process, in search for a specific tool for triggering the piano strings *silently*. Electromagnetic actuation was developed as a gateway to overcome the problems in acoustical actuation of the strings. The selection of the piano was not made for its performance characteristics, or its timbral properties; it was considered as a proper medium for string vibration when its wide range of sonority is considered.

In this approach, the piano has been a medium of sound design tool rather than the performance instrument. The software is an interface that is capable of producing numerous types of synthetic waves and has total control over the timbral and durational properties. This tool enables to explore a new sound world that was previously unknown or unpredicted, further extending the sound palette that can be obtained from a piano; or simply put - from any acoustic instrument that has the potential of electromagnetic interaction. It can be stated that this tool is a kind of a hybrid synthesizer that is run and controlled digitally but the outcome is acoustic sound.

4.2 General Features of the Proposed Setup

A prominent feature of this setup is the dissection of the performer and the instrument. With its algorithmic nature, proposed method focuses on the algorithmic compositional domain, rather the performance stage. It can exceed a real-world performer, because of the performance limitations of the human nature.

The non-deterministic nature is a common feature in all generation stages. All generative processes work on a predefined or manually assigned probability distribution and the progression of decision-making is unpredicted.

It should be mentioned that the initial idea was to vibrate the strings acoustically, with sound directed from loudspeakers. After numerous experiments, it was understood that acoustic triggering method would fell short to actualize such goal. Within the search of a suitable 'transducer medium', alternative string excitation methods were sought and this electromagnetic actuator setup was developed as a solution. Mainly, it is inspired from the electric guitar pickup; as if it was connected and run in reverse direction of function; as early experiments were realized with electric guitar pickups and guitar strings.

The musical ideas are built and composed in the software component. The proposed design is made to achieve specific types of sound textures, in reference to the Sound

Mass composition style. In brief, the software blocks generate tone clusters. By producing durational and tonal variations and alterations, they are intended to become the clouds of sound from an arbitrary composition. The main goal was to separate the performer and the composer from the sound producing process and leave the piano on itself. When any string on the piano is vibrated, by sympathetic vibrations it would vibrate other strings, if they have common members in their harmonic series.

As a control component in the setup, a contact microphone is placed inside the piano to receive the acoustic information on the instrument. This is how the software can 'listen' to its acoustic result and decide to make changes accordingly. Although this software is modeled for a specific purpose, however, it can also be used for different sonic intentions. Only constraint would be the software to be augmented.

In a broader sense, any signal can be sent to the actuator setup and the piano strings would vibrate as the string partials correspond accordingly. The proposed setup can be divided into two as the physical hardware and the conceptual software; the following sections discuss the aforementioned design features with their potentials and the future possibilities of this setup.

4.3 Future Experimentation and Improvements

Physical and electrical properties of the proposed electromagnetic transducers demand further study as they were developed primarily by experimentation. Physical and electrical properties of the coils can be improved as some physical properties could not be quantified yet. The margin of error in most of the stages has not been identified and the efficiency of the setup can be increased by precise electrical and mechanical design. These and the following topics could be studied further and the system can be fine-tuned:

- Frequency response of the coils – the frequency responses of the electromagnetic transducer prototype could be studied and quantified, also the electrical sensitivity of the transducer. The internal resonance of the transducers should be calculated and taken into consideration for any sonic design scenario since they contribute the acoustic sound emanating from the piano, on the other hand directly affects the electrical efficiency.

- Amplification accustomed for the actuator signal – if the transducer response to different audio frequencies is known, then such custom, well-tailored amplification stages can be designed regarding the electrical properties of the transducers,
- Cooling design– because they are exposed to high electrical power, the coils overheat if they run for prolonged periods of operation. There could be an optimization of the physical coil properties, power consumption levels and the control of the duration on the normal state of operation.
- Phase monitoring and manipulation – if the phase information of any individual frequency on any string could be detected, it will allow the electromagnetic setup to explore new world of sounds. With the prototype setup, even if short envelope parameters (ADSR) are assigned, short attacks and momentary sounds are hard to be realized. It is mainly because the damper pedal of the piano was removed beforehand - this allows the strings to vibrate freely for much longer sustain times. The proposed setup however did not intend to create such short duration sounds by its design, since the stylistic aims were in the opposite direction. Especially sounds with very short attack and release times would be hard to create with the present setup. But if phase detection feature would be added, the transient response of the system and moreover the decay characteristics of the strings would be drastically improved. Sounds with short durations even without any sustain time could be added to the sonic palette of the proposed setup. Also, in the software, musically more pleasing and varied results can be achieved in addition of abrupt musical gestures.
- Abandoning integral actuator cluster design - the one-duration-for-each-cluster concept can be abandoned and instead controls for micro durational events can be generated – for example; notes in the sound blocks could be individually realized with the interplay of their unique tonal and durational parameters. This would be realizing another perspective of the Sound Mass style.

4.4 Further Application Possibilities

The adaption of a magnetic resonator setup to a grand piano opens up new creative opportunities for especially algorithmic composers and sound designers. The principle of electromagnetic induction can be applied to all metal string instruments to bring the strings into vibration; for example; the electric guitar, acoustic guitar, mandolin, banjo, tanbur etc. The coil type and application setup can be adapted accordingly to the new medium. Similar works that were already implemented are the Haptic Drum, the Feedback Resonance Guitar, Electromagnetically Prepared Piano, the Overtone Fiddle (Overholt, Berdahl, Hamilton, 2011), the EMvibe - Electromagnetically Actuated Vibraphone (Britt, Snyder, McPherson, 2012) and the Electromagnetically Sustained Rhodes Piano (Shear, Wright, 2012).

In the proposed setup, electromagnetic setup is run with the same sound source. One kind of sound was designed as an actuator for all the strings and it was sent to all of the coils at the same time. As a different approach in design, any number of target strings, or in general vibrating parts, can be actuated with their own generated signals, as well.

The transducer setup can be used as a tool for extending the timbral properties of another instrument. By sending another instrument's sound to the electromagnets the result will be the addition of the original instrument timbre with the piano strings' reflections. For this purpose an audio interface could be used; the external instrument can be connected the inputs and the coil setup can be connected to the outputs as well.

As a future projection, this setup can be improved as a standalone music box. For example, a closed-box design can be developed to play any kind of music on the piano. By further improving the design and resolve the dependencies to a computer and a sound interface, an all-in-a-box version can be redesigned, manufactured and be presented as a commercial tool where users can load any desired music on the device (in the format of MIDI files, for example) and assemble the device on the piano and play music.

4.5 Musical Expansion Ideas

The setup was designed with an experimental approach, in Cageian terms as discussed in the first chapter, the Introduction section. The main motivation was the dissection of the conventional composition and the performer relation. Algorithmic compositions can be easily realized on this setup and when combined with the ability to monitor the acoustic sound emanated from the instrument, any kind of algorithmic composition can be performed with the setup; without any performer. Any arbitrary selection of music can be played back on to the piano with the transducers and could be heard as a piano performance.

In order to test the transducers during development, piano strings were actuated with conventional waveforms such as sine waves, triangle, square etc. It is seen that by changing the waveform and the durational behavior (envelope), a wide palette of timbral colors can be achieved with the piano strings. The timbral resemblance could be varied from a bowed instrument to a harpsichord, for example.

In the same work of composition, different instruments can be mimicked with the same setup. By changing the actuator signal type, timbres from the piano string can be changed from one another in the same composition, with the help of programming.

The composer and performer can be dissolved into the same slot, when adapted in this concept. Musical design is realized within the software domain, as the software controls the sonic structure, the musical outcome would differ as desired.

The software can be re-designed with the aim of capturing different sound forms from the piano, either with the combination a performer or without one. Such an example is the work of composer Per Bloland, *Elsewhere is a Negative Mirror* for piano with electromagnets, commissioned by The Society for Electro-Acoustic Music in the United States (SEAMUS) and The American Society of Composers, Authors and Publishers (ASCAP) in 2005 (Url-11).

The sympathetic vibrating piano realized with the electromagnetic actuation is a powerful tool for sonic artists, composers and performers, emphasizing its open-ended nature by the software control and the possibility of adaptation to any desired medium.

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APPENDICES

APPENDIX A: Glossary of Terms
APPENDIX B: Photos of Experiments - Research and Development
APPENDIX C: Photos of Proposed Setup – Research and Development

APPENDIX A - Glossary of terms

Sources:
[1] Rane Audio Reference *http://www.rane.com/digi-dic.html*
[2] Wikipedia *http://www.wikipedia.com*

Additive Synthesis: Additive synthesis is a sound synthesis technique that creates timbre by adding sine waves together. The timbre of musical instruments can be considered in the light of Fourier theory to consist of multiple harmonic or inharmonic partials or overtones. Each partial is a sine wave of different frequency and amplitude that swells and decays over time. [2]

Alternating Current (AC): An electric current that reverses direction at regularly recurring intervals of time. [1]

American Wire Gauge (AWG): A specification for non-ferrous (e.g., copper, aluminum, gold, silver, etc.) wire diameter. [Note, for example, that this means that 14 gauge galvanized steel wire & 14 gauge cooper wire have different diameters.] Also known as Brown and Sharp (B&S) wire gauge, after J.R. Brown who devised the system in 1857. [1]

dB (decibel): Equal to one-tenth of a bel. [After Alexander Graham Bell.]
1. A measuring system first used in telephony (Martin, W.H., "DeciBel -- the new name for the transmission unit. *Bell System Tech. J.* January, 1929), where signal loss is a *logarithmic* function of the cable length.
2. The preferred method and term for representing the *ratio* of different audio levels. It is a mathematical shorthand that uses *logarithms* (a shortcut using the powers of 10 to represent the actual number) to reduce the size of the number. For example, instead of saying the dynamic range is 32,000 to 1, we say it is 90 dB [*the answer in dB equals 20 log x/y, where x and y are the different signal levels*]. Being a ratio, *decibels have no units*. Everything is relative. Since it is relative, then it must be relative to some *0 dB reference point*. To distinguish between reference points a suffix letter is added as dBu, dBV, dBm, dBA, etc. [1]

dB FS: A digital audio reference level equal to "Full Scale." Used in specifying A/D and D/A audio data converters. Full scale refers to the maximum *peak* voltage level possible before "digital clipping," or digital overload of the data converter. The Full Scale value is fixed by the internal data converter design, and varies from model to model. [1]

Direct Current (DC): An electric current that flows in one direction. [1]

Electromagnetism: The study of the electromagnetic force which is a type of physical interaction that occurs between electrically charged particles. The electromagnetic force usually shows electromagnetic fields such as electric fields, magnetic fields and light. [2]

Electromagnetic Coil: Formed when a conductor is wound around a core, to form an inductor. [2]

Electromagnetic Induction: The generation of an electromotive force (*voltage*) and current in a circuit or material by a changing magnetic field linking with that circuit or material. Electricity and magnetism are kinfolk and form the foundation of audio transducers found at both ends of any audio chain: *dynamic microphones* and *loudspeakers with voice coils*. The principle is beautifully simple: if you pass a coil of wire through a magnetic field, electricity is generated within the coil (*dynamic microphone*), and if you pass electricity through a coil of wire (*voice coil*), a magnetic field is generated. *Move a magnet, create a voltage; apply a voltage, create a magnet*. This is the essence of all electromechanical objects. [1]

Feedback: 1. (Acoustic feedback) The phenomenon where the sound from a loudspeaker is picked up by the microphone feeding it, and re-amplified out the same loudspeaker only to return to the same microphone to be re-amplified again, forming an acoustic loop. Each time the signal becomes larger until the system runs away and rings or feeds back on itself producing the all-too-common scream or squeal found in sound systems. These buildups occur at particular frequencies called *feedback frequencies.* [1]

2. (Electronic feedback) The return of a portion of the output of a process or system to the input, especially when used to maintain performance or to control a system or process. [1]

Impedance (resistance): A measure of the complex resistive and reactive attributes of a component in an alternating-current (AC) circuit. Impedance is what restricts current flow in an AC electrical circuit; impedance is not relevant to DC circuits. In DC circuits, resistors limit current flow (because of their resistance). In AC circuits, inductors and capacitors similarly limit the AC current flow, but this is now because of their inductive or capacitive reactance. Impedance is like resistance but it is more. Impedance is the sum of a circuit, or device's resistance AND reactance. Reactance is measured in ohms (like resistance and impedance) but is frequency-dependent. Think of impedance as the complete or total current limiting ohms of the circuit. Since AC circuits involve phase shift -- i.e., the voltage and current are rarely in phase due to the storage effects of capacitors and inductors, the reactance is termed "complex," that is there is a "real" part (resistive) and an "imaginary" part (the phase shifting resistance part). To summarize: resistance has no phase shift; reactance (capacitors & inductors in AC circuits) includes phase shift; and impedance, is the sum of resistance and reactance. [1]

Karplus-Strong String Synthesis: A method of physical modelling synthesis that loops a short waveform through a filtered delay line to simulate the sound of a hammered or plucked string or some types of percussion. [2]

Magnetic Field: The electric field surrounding any current-carrying conductor. A condition found in the region around a magnet or an electric current, characterized by the existence of a detectable magnetic force at every point in the region and by the existence of magnetic poles. [1]

Micropolyphony: Micropolyphony is a kind of polyphonic musical texture developed by György Ligeti and then imitated by some other twentieth-century composers, which consists of many lines of dense canons moving at different tempos or rhythms, thus resulting in tone clusters vertically. [2]

MIDI: (Musical Instrument Digital Interface) Industry standard bus and protocol for interconnection and control of musical instruments. First launched in 1983, now generalized and expanded to include signal processing and lighting control. [1]

Neodymium: Popular rare-earth metal used to make superior magnets for loudspeakers and microphones. Neodymium iron boron magnets have a more linear frequency response, are more powerful and smaller, with higher output levels than conventional iron magnets. First used by Electro-Voice in the late '70s. [1]

Notch Filter: A special type of cut-only equalizer used to attenuate (only, no boosting provisions exist) a narrow band of frequencies. Three controls: frequency, bandwidth and depth, determine the notch. Simplified units provide only a frequency control, with bandwidth and depth fixed internally. Used most often in acoustic feedback control to eliminate a small band of frequencies where the system wants to howl (feedback). [1]

Pure Data: Pure Data (a.k.a Pd) is an open source visual programming language, enabling musicians, visual artists, performers, researchers, and developers to create software graphically, without writing lines of code. Pd is used to process and generate sound, video, 2D/3D graphics, and interface sensors, input devices, and MIDI. [2]

Resistance (resistor): Circuit symbol: R

1. An element within a circuit that has specified resistance value designed to restrict the flow of current.

2. A device with the primary purpose of introducing resistance into an electric circuit. (A resistor as used in electric circuits for purposes of operation, protection, or control, commonly consists of an aggregation of units. Resistors, as commonly supplied, consist of wire, metal, ribbon, cast metal or carbon compounds supported by or embedded in an insulating medium. The insulating medium may enclose and support the resistance material as in the case of the porcelain tube type or the insulation may be provided only at the points of support as in the case of heavy duty ribbon or cast iron grids mounted in metal frames.) [1]

Sound Mass: Sound mass style is where the compositional texture is formed by dense sonic blocks build masses like void, gaseous clouds, without any identifiable melodies and rhythms emerging but without any occurrence of stagnation. Pitch, timbre and texture fuse together animating the space. [2]

Sympathetic Vibration: A vibration produced in one body by the vibrations of exactly the same period in a neighboring body. [2]
Transducer: A device, such as a microphone, or loudspeaker, that converts input energy of one form into output energy of another. [1]

APPENDIX B - Photos from experiments during research and development stages

Figure B.1 : Jan 8, 2013, ITU MIAM, Room No. 201. Equipment: one Mackie SRM 450 speaker and one AKG C414 large diaphragm condenser microphone in cardioid mode. Both are controlled by Pd patches those control the feedback and process of the captured piano sound. Speaker position: opposite of keyboard, next to the bridge.

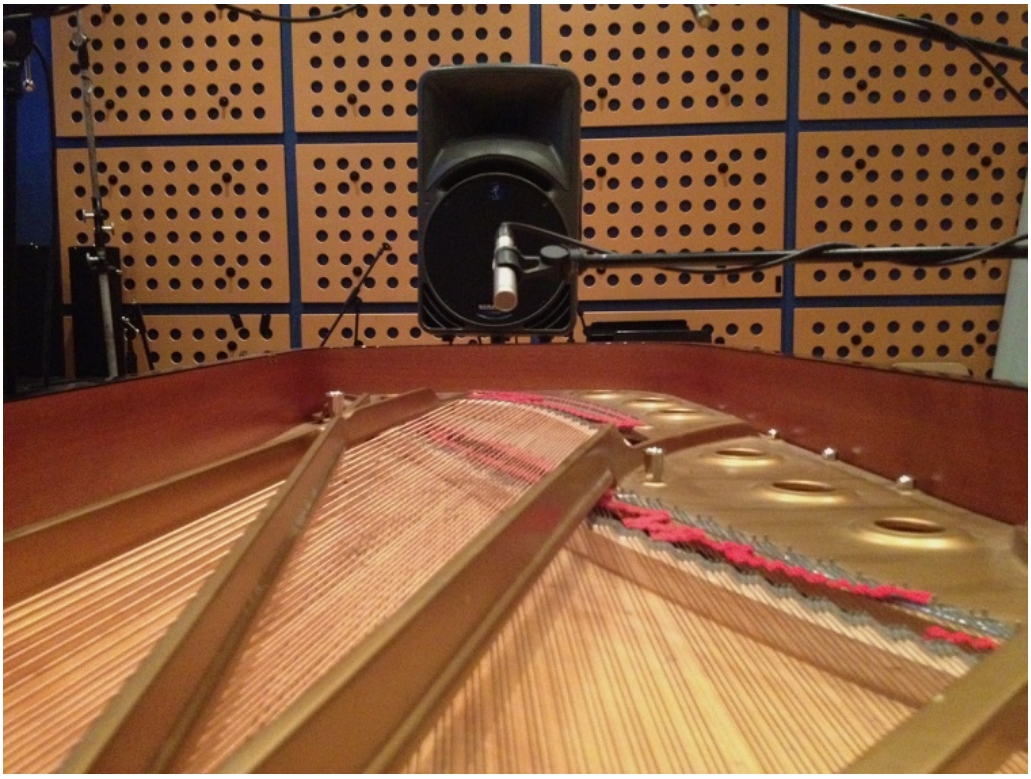

Figure B.2 : Jan 28, 2013, ITU MIAM Main Studio. A Mackie SRM 450 loudspeaker, one Neumann KM184 small diaphragm cardioid condenser in close position and one DPA 4006 small diaphragm omni-directional microphone above.

Figure B.3 : Jan 28, 2013, ITU MIAM Main Studio. Experimenting with one Mackie SRM 450 loudspeaker, one Neumann KM184 small diaphragm cardioid condenser microphone in close position and one DPA 4006 small diaphragm omni-directional microphone above. All connected to an RME sound interface and controlled Pd feedback patches.

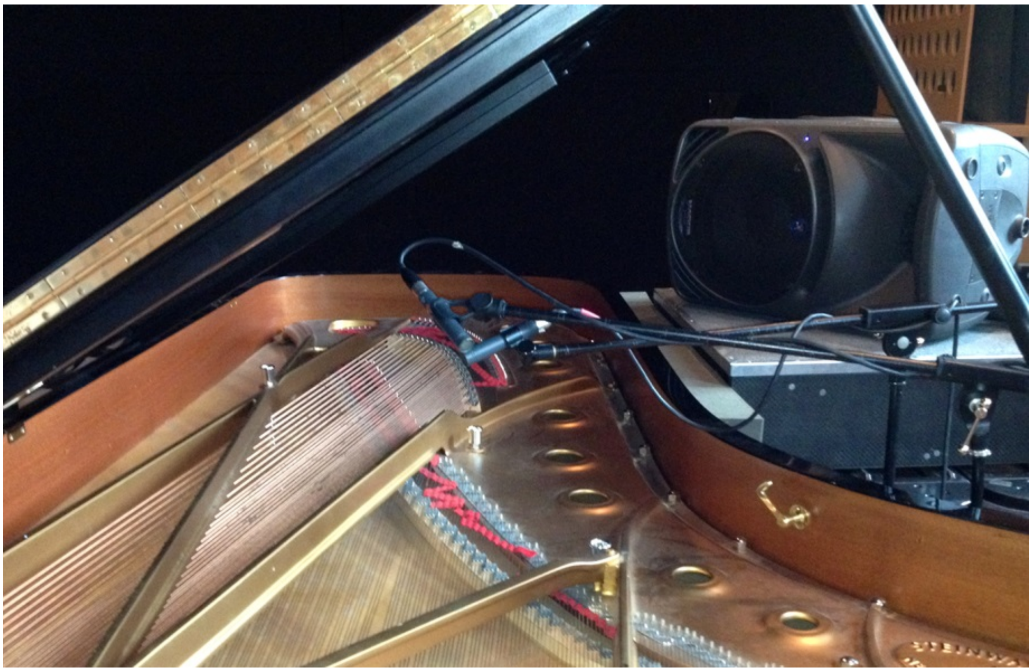

Figure B.4 : Feb 4, 2013, ITU MIAM Main Studio. Used equipment: one Mackie SRM 450 loudspeaker, two Neumann KM184 small diaphragm cardioid condenser microphones in X-Y position.

Figure B.5 : May 9, 2013, ITU TMDK Room No. 103. A Mackie SRM 450 loudspeaker faced towards the strings.

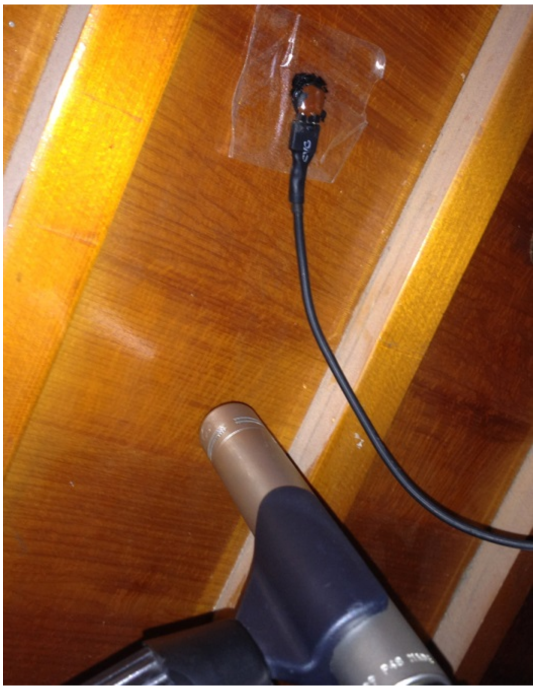

Figure B.6 : May 9, 2013, ITU TMDK Room No. 103. One small diaphragm cardioid Rode NT5 and one contact microphone positioned behind the upright piano.

Figure B.7 : May 9, 2013, ITU TMDK Room No. 103, whole setup. Microphones and the speaker are connected to a TC Electronics sound interface, feedback and sound manipulation controlled by Pd patches.

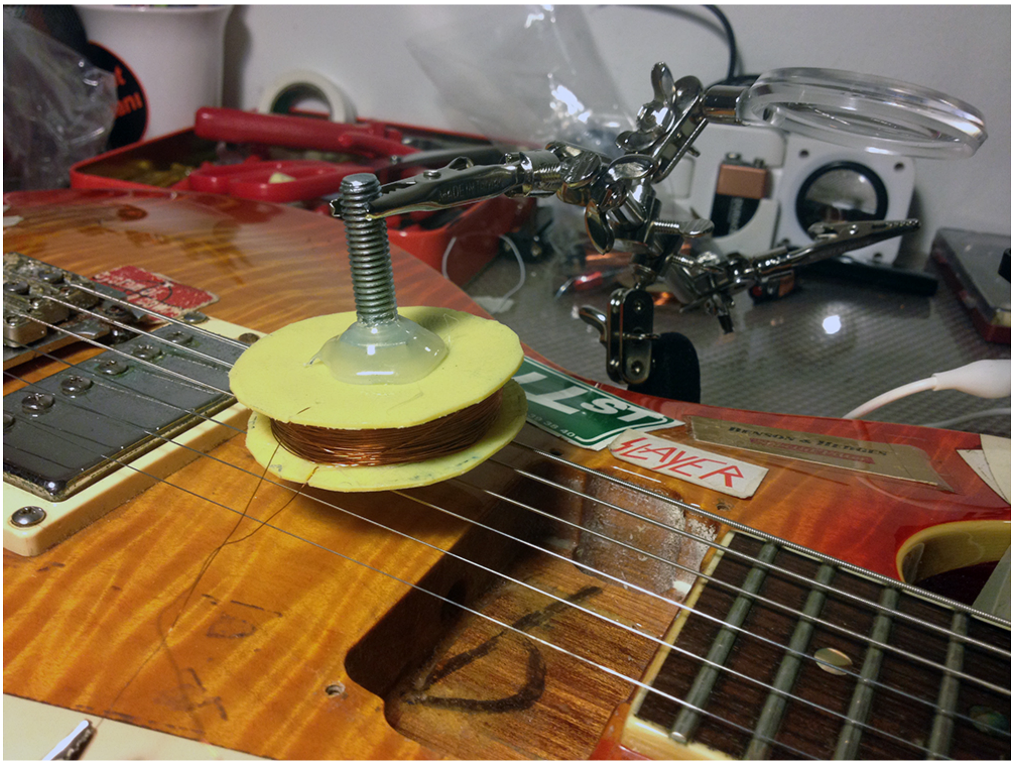

Figure C.1 : Nov 28, 2013, study room. Experimenting on the prototype transducer. First time that sound from a string could be heard. Electric guitar is used as the base instrument. A cylindrical ferromagnet glued between two plastic wheels, wire wound around. Handmade amplifier of 1 watt is being used (*ABOVE*).
Same from a different angle (*BELOW*).

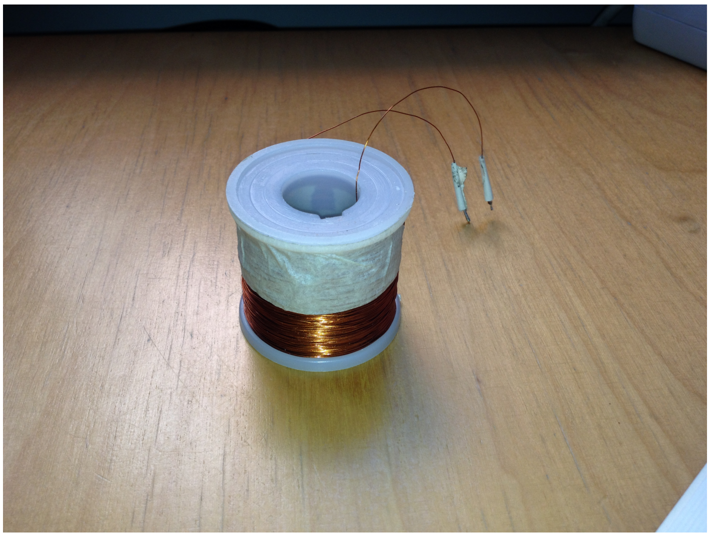

Figure C.2 : Mar 06, 2014, study room. Finished prototype. This type of plastic reel is selected to hold the magnets steady in the core. 20 plastic reels to develop clones of the original prototype are acquired from Feray Alaşım Ltd., Davutpaşa, Istanbul.

Figure C.3 : Jun 03, 2014, ITU TMDK Room No. 103. Actual version of electromagnets assembled for the upright piano, with the hammer mechanism of the keys removed. 20 coils connected in parallel configuration.

Figure C.4 : Jun 16 2014, ITU TMDK Room No. 103. Latest version of electromagnets, 13 of them connected in parallel. Experimenting on an upright piano, which is laid-back to imitate a grand piano (*ABOVE*).
Same, from a different angle (*BELOW*).

Figure C.5 : Audio amplifier used in the setup, a Lab Gruppen 2000C. Acquired from MASS Ltd., Mehmet AKKAŞ, Kağıthane, Istanbul.

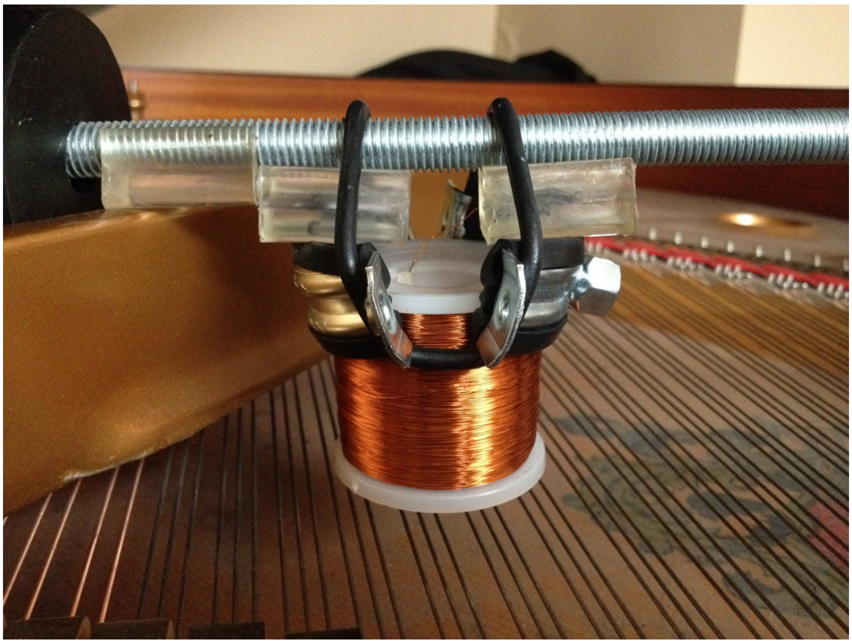

Figure C.6 : May 28, 2015, ITU MIAM, Room 120. Experiments for hanging a single electromagnet. Materials (metal bar, natural gas pipe clamp, rubber strip, plastic supports, etc.) provided from Hırdavatçılar Çarşısı, Karaköy, Istanbul.

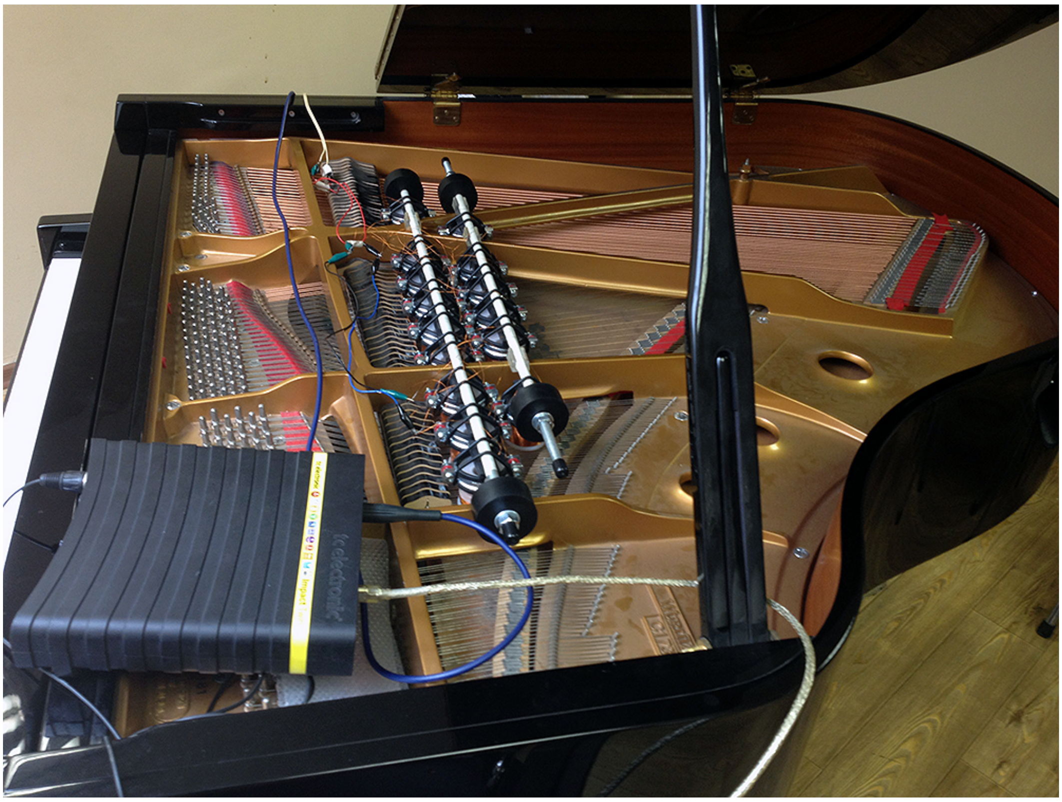

Figure C.7 : May 28, 2015, ITU MIAM, Room 120. Proposed setup as seen from the top. The blackbox on the left corner is the sound interface. The array of 18 electromagnets are hung on two metal bars, physically covering 4 octaves. Actual sound range extends the physical coverage.

ÖZGEÇMİŞ

Ad Soyad: Gökhan DENEÇ

Doğum Yeri ve Tarihi: İstanbul, 06/03/1977

Adres:

E-Posta: gokhandenec@gmail.com

Lisans: İTÜ Fen-Edebiyat Fakültesi Matematik Mühendisliği

Yüksek Lisans : İTÜ Dr. Erol Üçer Müzik İleri Araştırmalar Merkezi (MIAM) Müzik

Mesleki Deneyim ve Ödüller:

Gökhan Deneç 2003'ten bu yana çeşitli albüm prodüksiyonu, film/dizi/reklam filmi post-prodüksiyonu ve farklı türlerdeki çok sayıda canlı konser organizasyonlarında profesyonel ses mühendisi olarak yer almıştır.

İstanbul Teknik Üniversitesi Türk Müziği Devlet Konservatuarı Müzik Teknolojileri, Yıldız Teknik Üniversitesi Sanat ve Tasarım Fakültesi İşitsel Tasarım ve Bahçeşehir Üniversitesi Ses Teknolojileri Yüksek Lisans bölümlerinde ses tasarımı ve ses sistemleri üzerine ders vermektedir.

Yayın ve Patent Listesi:

TEZDEN TÜRETİLEN YAYINLAR/SUNUMLAR