ISTANBUL TECHNICAL UNIVERSITY ★ INSTITUTE OF SOCIAL SCIENCES

INTERACTIVE SOUND DESIGN
FFT SOUND SYNTHESIS ENGINE MODEL PROPOSAL

Ph.D. THESIS

Yahya Burak TAMER

Department of Music

Music Programme

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İSTANBUL TEKNİK ÜNİVERSİTESİ ★ SOSYAL BİLİMLER ENSTİTÜSÜ

ETKİLEŞİMLİ SES TASARIMI
FFT SES SENTEZLEME MOTORU MODELİ

DOKTORA TEZİ

Yahya Burak TAMER
(409052005)

Müzik Bölümü

Müzik Programı

Tez Danışmanı: Prof. Dr. Cihat AŞKIN

EKİM 2012

Yahya Burak TAMER, a **Ph.D.** student of **ITU Institute of Social Sciences** student ID **409052005** successfully defended the dissertation entitled **“INTERACTIVE SOUND DESIGN - FFT SOUND SYNTHESIS ENGINE MODEL PROPOSAL”**, which he prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

Thesis Advisor:	**Prof. Dr. Cihat AŞKIN** Istanbul Technical University	…………………..
Jury Members:	**Doç. Dr. Kıvılcım YILDIZ ŞENÜRKMEZ** Mimar Sinan Fine Arts University State Conservatory	…………………..
	Yrd. Doç. Dr. İlke BORAN Mimar Sinan Fine Arts University State Conservatory	…………………..
	Yrd. Doç. Dr. Can KARADOĞAN Istanbul Technical University	…………………..
	Yrd. Doç. Dr. Ozan BAYSAL Istanbul Technical University	…………………..

Date of Submission: 28 May 2012
Date of Defense: 17 October 2012

FOREWORD

I would like to thank my advisor Cihat Aşkın and the remaining members of my jury committee; Şehvar Beşiroğlu, Kıvılcım Yıldız Şenürkmez, İlke Boran, Can Karadoğan and Ozan Baysal.

When I completed my undergraduate study on Electronics and Communication Engineering in 1997, I got into research for a graduate study where I could connect my engineering skills with music and sound design disciplines. Finding out about MIAM (thanks to Gökhan Deneç) and studying music theory, history, sound design, engineering and composing has improved my conceptual perception as well as analytical skills. I am grateful since I have been a part of this synergy where open-minded musicians could share their views and interact freely. I would like to thank all of my instructors in MIAM and Yıldız Technical University for sharing their knowledge on arts and sciences, their input throughout the years has made this thesis possible.

I would like to express my appreciation to Fatoş Adiloğlu and Güven Çatak for their continuous support throughout the thesis writing process. I am also indebted to my colleagues in Bahçeşehir University as interacting with each one of them and their professions has widened my perspective on sound arts.

During my years in MIAM, I was lucky enough to be able to explore other visions on music, sound and life while interacting with fellow students. Out of this very prosperous set of relationships one in particular influenced a world of sound, creation and honest friendship. I would like to thank Barkın Engin for that.

My dear friend Tuba Yapıcı did not leave me alone during the defense, I should express that her support was very valuable.

Finally, I would like to thank my family and friends.

October 2012 Yahya Burak TAMER

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ABBREVIATIONS

AD	**:** Analog to Digital
ADSR	**:** Attack Decay Sustain Release
BPM	**:** Beat Per Minute
CD	**:** Compact Disc
DAC	**:** Digital Analog Conversion
DAW	**:** Digital Audio Workstation
DI	**:** Direct Injection
EQ	**:** Equalization
EZDAC	**:** Easy DAC
FFT	**:** Fast Fourier Transform
LED	**:** Light Emitting Diode
MIDI	: Musical Instrument Digital Interface
RAM	**:** Random Access Memory
VST	**:** Virtual Studio Technology

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SUMMARY

The following thesis provides an insight into the history of electronic musical instruments and their effects on composers and performers. Early 20^{th} century designs have been selected as the focus of this study since the instruments introduced during this period have set the standards for both technical and aesthetic terms. *Telharmonium* (1897) introduced the concept of additive synthesis, which forms the basis for all hardware and software synthesizers of today. The concept of live stream music was achieved via telephone network in 1906. *Theremin* introduced unique controlling mechanisms while *Ondes Martenot* improved the design by enriching the timbre and making the instrument blend in with the traditional orchestra. *Trautonium* introduced the technique of subtractive synthesis as well as sequencing. The period from 1900 to 1935 has determined the development of electronic musical instruments and the way they are experimented by composers and performers, thus has become the focus of this study.

Having provided a technical and aesthetic basis on the introduction of electronic musical instruments, the thesis progresses to propose and apply a model for a new electronic music instrument titled 'the instrument with no sound'. After the very first electronic musical instruments, various designs have been presented but very few of them won recognition and survived. It should be noted that the very first electronic music instruments actually did survive, *Theremin* is still popular today; *Ondes Martenot* is used in orchestras as well as modern recordings. *Telharmonium* did not survive due to its massive size and outdated technology but the additive synthesis technique certainly did. Live stream music that was first introduced to play *Telharmonium* concerts is a key concept in music distribution today, embedded in social media structures. *Trautonium* did not survive in its physical form either as it was not mass-produced, but the subtractive synthesis technique is amongst the most important sound design techniques in today's modern synthesizer.

My approach to propose a new electronic music instrument design was to combine the engineering input with the musician input in order to form a composite approach to design. As the technology of electronic musical instruments advanced, the instruments became far too complex to get into instant interaction with. However, a successful design reveals itself easily. Even a novice player can play simple melodies on a piano, yet as you advance, the design unfolds to offer new possibilities of musical expression and control. This being the first principle of my design; the second is the consideration of the power of electronic sound design techniques to create unique sonorities. Combining the simplicity of use with the wide range of acoustic / electronic / electro-acoustic timbres was the aim of this new instrument design. Another concern was the individualization of the instrument.

Two guitar players with equal levels of experience can play the same partition under the same circumstances, yet the result will never be the same as each one of them has a certain style that we may call their individual sounds. The instrument's level of expression is so well designed that it allows certain details of touch, hold etc. to be revealed, hence the individual sounds.

'The instrument with no sound' explores to reinforce these principles to create a new electronic music instrument. As the name implies, the instrument has no sound when you first start using it. This is a deliberate design decision. As for control surface, I have chosen a traditional keyboard for the sake of simplicity, but any controller surface can be implemented to the algorithm, the model I have developed for this thesis can as well be considered a prototype. In order to play the instrument, the player is required to record a 2 second sound sample. Using the Fast Fourier Transform technique, the algorithm extracts the timbre from this sound sample and distributes it along the keys of the traditional keyboard. Having completed this process, the player can instantly start playing the polyphonic instrument with the timbre that has just been determined. The sound source could be anything; it could be a musical sound or the sound of any daily object, voice, ambience sound scape etc. Therefore the player experiences and creates his/her own selection of sound colors, thus the process increases awareness of sound information through moments of our daily life. Alongside the frequency domain processing that is used to extract the sonority of the samples recorded, the instrument has basic time domain processing functions such as an envelope so that after recording the sound, the player can adjust the ADSR envelope. In other words it is possible to create a sustained tone as well as a staccato sound.

Described above is the basic operation mode of the new instrument. Frequency domain processing with FFT allows us to 'convolve' two sounds. This creates a composite new sound out of the two (or more) sound samples convolved. When one of the tones is a basic periodic waveform (such as a sawtooth wave) and the other human voice, this process becomes the well-known technique titled *Vocoder,* which is a single state of the convolution technique. When two sounds are convolved, the FFT algorithm multiplies the re-synthesized frequency bands of each sound source so that only the common frequency bands live to reinforce each other while the others are attenuated. When the instrument is used in the convolution mode; the player now chooses two sound sources to experiment with. After these sound sources are recorded as 2-second samples, they are convolved into a single composite sound. This technique is useful for creating unique timbres that is not possible to acquire via the natural world. Once the convolution process is over, the same procedure in the basic mode applies; the player can directly play the newly formed sonority via the keys of the traditional keyboard.

The instrument has been designed in the object oriented software language titled 'Max MSP'.

ETKİLEŞİMLİ SES TASARIMI – FFT SES SENTEZLEME MOTORU MODELİ

ÖZET

Bu tez erken dönem elektronik müzik enstrümanları tarihi ve bu dönemin besteci ve icracılar üzerindeki etkilerini incelemektedir. 20. Yüzyılın başlarında tasarlanan enstrümanların çalışmanın odağı olarak seçilmesinin sebebi bu dönemde gerçekleştirilen örneklerin hem teknik hem de estetik açılardan takip edecek buluşlar için temelleri belirlemiş olmasıdır. *Telharmonium* (1897) *eklemeli sentezleme* yöntemini tanıtan ilk enstrümandır ve bu teknik günümüz yazılım ve donanım temelli *synthesizer*'larının esas prensibini oluşturmaktadır. *Live stream* müzik yayını ilk defa yine bu dönemde (1906) telefon şebekesi üzerinden gerçekleştirilmiştir. *Theremin* özgün bir kontrol mekanizması tanıtmış, *Ondes Martenot* ise tınıyı zenginleştirip enstrümanı geleneksel orkestranın içerisine dahil edecek biçimde konumlandırarak tasarımı ileriye taşımıştır. Yine bu dönemden *Trautonium* isimli enstrüman *çıkarmalı sentezleme* ve *sekanslama* gibi teknikleri ilk defa kullanmıştır. 1900-1935 arası dönemde geliştirilen enstrümanlar takip edecek elektronik müzik enstrümanlarının gelişiminde ve bu enstrümanların besteci ve icracılar tarafından değerlendirilmesinde belirleyici olmuş, dolayısıyla bu çalışmanın odağı olarak seçilmiştir. 21. Yüzyıl itibarı ile, dijital teknolojinin hem kullanıcı tarafından erişilebilirliği hem de işlemci gücünün üstel artışı gibi etkenler göz önünde bulundurulduğunda elektronik ses sentezleme için çok önemli buluşlar sunulduğu da bir gerçektir. Bu tezin enstrüman modeli önerisi bölümünde uygulamaya geçmeden önce hem güncel elektronik müzik enstrüman ve ses işleme prosedürleri incelenmiş, hem de bu güncel tasarımların tezin girişindeki ilk elektronik müzik enstrümanları ve kullandıkları sentezleme modelleriyle organik bağlantıları sunulmuştur.

Elektronik müzik enstrümanlarının başlangıcına dair teknik ve estetik bir altyapı belirlemenin ardından tez 'sesi olmayan enstrüman' isimli yeni bir elektronik müzik enstrüman modeli önermek ve uygulamak üzere devam etmektedir. İlk elektronik müzik enstrümanlarının ardından türlü tasarım sunulmuş ama çok azı kabul görmüş ve uzun vadede sürdürülebilmiştir. Göz önünde bulundurulmalıdır ki ilk elektronik müzik enstrümanları kabul görmüştür ve halen aktif şekilde kullanılmaktadır. *Theremin* günümüzde halen popüler bir enstrümandır; *Ondes Martenot* klasik orkestralarda ve modern kayıtlarda kullanılmaktadır. *Telharmonium* kullanımdan kalkmış teknolojisi ve büyük boyutu nedeniyle kendi formunda varlığını sürdürememiş olsa da *eklemeli sentezleme* tekniği varlığını kesinlikle korumuştur. İlk defa *Telharmonium* konserlerini telefon hatları üzerinden abone dinleyicilere aktarmak üzere kullanılan *Live Stream* müzik yayını bugün müzik yayınında sosyal medyanın temelini teşkil etmektedir. *Trautonium* da toplu üretime geçilmemesi sebebiyle kendi fiziksel formunda varlığını koruyamamış olsa da *çıkarmalı sentezleme* ve *sekanslama* teknikleri modern *synthesizer*'da kullanılan en önemli ses tasarımı teknikleri arasında yer almaktadır.

Yeni bir elektronik müzik enstrümanı önermede benim yaklaşımım mühendislik bakış açısı ile müzisyen bakış açısını karma bir yaklaşım oluşturmak üzere birleştirmekti. Elektronik müzik enstrümanlarının teknolojisi ilerledikçe, enstrümanlar anında etkileşime girilemeyecek derecede karmaşık bir yapı edinmeye başladılar, oysa ki başarılı bir tasarım kendini rahatlıkla çözülen bir düğüm gibi ele vermelidir. Yeni başlayan bir kişi bile piyanoda basit melodileri kısa sürede çalabilir hale gelecektir, fakat seviyesi ilerledikçe enstrüman tasarımı da katman katman açılarak derinliğini ortaya koyar; müzikal ifade ve kontrol için yeni olasılıklarını açığa vurur, benim tasarımımın da birinci prensibi budur. İkincisi ise doğal yollarla ya da akustik enstrümanlarla elde edilemeyecek özgün tınılar yaratmak üzere elektronik ses tasarımı tekniklerinin uygulanmasını göz önünde bulundurmaktır. Kullanım basitliğini geniş yelpazedeki akustik / elektronik / elektro-akustik tınılar ile birleştirmek bu yeni enstrüman tasarımının çıkış noktasını teşkil etmektedir.

Bir başka kaygı ise enstrümanın kişiselleştirilebilmesi idi. Geleneksel enstrümanlardan bir örnek ele alacak olursak; aynı tecrübe seviyelerine sahip iki gitarist aynı partisyonu aynı şartlar altında çalabilir fakat her biri onların kişisel sesleri diyebileceğimiz belirli tarzları nedeniyle sonuç asla aynı olmayacaktır. Enstrümanın ifade seviyesi o kadar iyi belirlenmiştir ki dokunma, tutma gibi ufak detayların bile açığa çıkarak kişisel seslere dönüşmesine izin vermektedir.

'Sesi olmayan enstrüman' bu prensipleri pekiştirmeyi araştıran yeni bir elektronik müzik enstrümanı yaratmayı hedefleme sonucunda ortaya çıkmıştır. İsminden de anlaşılabildiği gibi ilk kullanıma başlandığında enstrümanın sesi yoktur. Bu kasıtlı bir tasarım kararıdır. Bu prototip için kontrol yüzeyi olarak basitliğini de göz önünde bulundurarak geleneksel piyano klavyesini seçtim ama herhangi bir kontrol yüzeyi bu algoritmaya uygulanabilir. Enstrümanı çalmak için kullanıcıdan 2 saniyelik bir ses örneği kaydetmesi istenmektedir. Algoritma *Hızlı Fourier Dönüşümü* tekniğini kullanarak bu ses örneğinin tınısını çıkarır ve geleneksel klavyenin tuşlarına dağıtır. Bu süreci tamamlayan kullanıcı polifonik enstrümanı belirlenen tını ile anında çalmaya başlayabilir. Tınıyı belirleyen ses kaynağı her şey olabilir; müzikal bir ses, günlük kullanıma ait bir nesnenin sesi, insan sesi, ses dokularının oluşturduğu bir ambiyans sesi vs. gibi. Dolayısıyla her kullanıcı kendi ses renklerini deneyimler ve yaratır, bu nedenle süreç günlük hayatımız içerisindeki akustik bilgiye karşı olan farkındalığımızı da arttırır. Kaydedilen ses örneklerinden tınıyı çıkarmak için kullanılan frekans düzleminde işleme tekniği yanısıra enstrüman temel zaman düzlemi ses işleme tekniklerini de barındırmaktadır. Uygulanan *zarf* ile kullanıcı ADSR zarfını belirleyebilir; başka bir deyişle sönümsüz / sürekli ya da staccato / kısa zamanlı tonlar çalmak ve varyasyonlarını tayin etmek mümkündür.

Yukarıda tanımlanan, enstrümanın temel çalışma modudur. FFT ile frekans düzleminde ses işleme tekniği iki sesi 'katlama'mıza imkan vermektedir. Bu teknik, katlanan iki (ya da daha fazla) ses örneğinden karma yeni bir ses yaratmamızı sağlar. Tonlardan biri temel periyodik bir dalga (testere dişi ses dalgası gibi) diğeri ise insan sesi olduğunda bu süreç bilinen *Vocoder* tekniği olarak tanımlanmaktadır. *Vocoder* genel anlamda ses katlamanın özel bir hali, dolayısıyla bir alt kümesidir. İki ses katlandığında FFT algoritması her bir ses kaynağının yeniden sentezlenen frekans bantlarını birbirleriyle çarpar; dolayısıyla sadece ortak frekans bantları birbirlerini kuvvetlendirir ve bileşik seste varlığını korur, diğer bantlar ise bileşik seste bulunamazlar. Enstrüman katlama modunda kullanıldığında kullanıcı bu sefer denemeler yapacağı iki ses kaynağı seçer.

Bu ses kaynakları 2 saniyelik ses örnekleri olarak kaydedildikten sonra tek bir bileşik sese katlanırlar. Bu teknik doğal yollarla elde edilemeyen özgün ses renkleri yaratmak için idealdir. Ses katlama işlemi tamamlandığında temel çalışma modundaki prosedür geçerlidir, yani kullanıcı yeni oluşturulmuş bileşik sesi geleneksel polifonik klavyenin tuşları aracılığı ile çalabilir.

Enstrüman 'Max MSP' isimli obje temelli yazılım dili kullanılarak tasarlanmıştır.

1. INTRODUCTION

1.1 Purpose of the Thesis

The first part of this thesis explores the early twentieth century electronic music instruments and sound synthesis techniques. The study intends to research and compare the outcome of the interaction of these innovations with the composers. The research includes technical elements due to the capabilities of these instruments thus related issues of performance as well as development of compositional techniques. After achieving this insight into electronic sound design and perception of these techniques by composers and performers, the study progresses on to proposing a model for a new electronic music instrument design.

Concerning the interdisciplinary roles that it will be occupying in the near future of music, sound engineering, digital arts and interactive design; the discipline of digital audio signal processing is still in its infancy. My aim in this study is to achieve a multidisciplinary approach to digital signal processing techniques in terms of their design and implementation on music, sound engineering and interaction design. The introduction of the early twentieth century electronic music instruments has set most of the standards for the theory of sound design techniques. As the technology advanced, the electronic music instruments became more versatile. Considering their use in today's music production, software and hardware electronic musical instruments play a significant role. The instruments of the twenty-first century are much easier to access, present a variety of opportunities for synthesizing and processing sound, are superior concerning the issues of transportation, maintenance and economics. Obviously, the technical superiority is a positive development, but if we are to consider both pros and cons of modern electronic music instruments, their technical complexity forms an obstacle when considered in terms of instant interaction that translates musical creativity into sound. Modern electronic music instruments have become technically very demanding. In order to interact and play an instrument properly, one has to master each technical element of the workflow as well as individual parameters and the level of their interaction with the sound output. During this process lies the danger of losing musical creative focus and being too much associated with the technical possibilities that the device can offer. Another concern would be that most of the modern instruments are presented to the player filled with libraries of sound. These ‘presets’ are obviously meant to present the possibilities of the instruments as well as serving as templates for further modifications, but often times the players get lost in these vast quantities of preset banks and parameter combinations, while achieving a musical idea in mind instantly becomes a goal far too hard to accomplish.

Taking into consideration this certain gap between the engineer input and artist input along with the examples viewed in the history of electronic music instruments, the model for a new electronic music instrument design proposed in this thesis aims to built a composite approach that includes engineer’s view as well as artist’s view.

1.2 Background

Electronic music instruments built in the early twentieth century have introduced composers to a variety of new means of musical expression, thus enabling new aesthetic forms to be presented. The search for new dimensions in composition in the late nineteenth century introduced first by chromaticism, later followed by the twelve-tone technique in the twentieth century coincides with the period. Olivier Messiaen, Arthur Honegger, Darius Milhaud, Pierre Boulez, Tristan Murail, Paul Hindemith used instruments such as *Ondes Martenot*, *Trautonium* and *Theremin* in their compositions. The use of these instruments has shaped the revolution of music as we continue to encounter it today.

Hermann von Helmholtz’s book “On the Sensations of a Tone” (1862) has influenced musicologists of the twentieth century. The *Helmholtz Resonator* proved that a tone could be identified as a combination of musical pitches and irregular frequency components. Ferruccio Busoni’s book “The Sketch of a New Aesthetic of Music” (1911) mentions the electronic music instrument *Telharmonium* and encourages the twentieth century composers to open their music to all sound. The *Futurist* movement, introduced by Luigi Russolo with his Futurist Manifesto “The Art of Noise” (1913) abandoned the use of traditional instruments while embracing the use of any sound source as musical material (Davies, 1990). These works constitute some of the early sources of inspiration for the creation of new instruments and the thirst for the research in sound synthesis. The studies also indicate that while working on the development of new techniques of musical composition, the composers of the period were also interested in achieving new means of musical expression through the use of new timbres that do not coincide with those of traditional musical instruments.

The musical technologies of the early twentieth century (from 1900 to 1935) have a significant impact on the perception and development of electronic music instruments and their use in composition and performance. If one is to propose a new electronic instrument model, it is crucial to analyze the technical and aesthetic content of this period.

Figure 1.1: The Musical Telegraph by Elisha Gray, left. Carbon Arc street lamp of the Victorian Britain, right (Crab, 2005, p. 14).

1.3 Before the Early Instruments

The very first electronic music instruments known are*: Clavecin Electrique, Musical Telegraph* and *The Singing Arc.Clavecin Electrique* (Electric Harpsichord) was invented in 1759 by Jean-Baptiste de La Borde in Paris, France. The instrument was controlled by a keyboard. It was based on electrostatic principles; the sound source mechanism consisted of bells that were struck by clappers charged with static electricity. *Musical Telegraph* was invented by Elisha Gray in 1874. The first version of the instrument was a single note oscillator. The mechanism of the instrument comprised small reeds whose vibrations were created and transmitted over a telephone line by electromagnets.

Later, Gray developed a two-octave version of the musical telegraph which was polyphonic, thus predated the introduction of the first portable electric organ by sixty years. *The Singing Arc* was invented by William Duddell in 1899.Duddell invented the singing arc mistakenly while he was working on removing the hum noise of the Carbon Arc Lamp which was used for street lighting before the invention of the electric light bulb. During his experiments, Duddell found out that by varying the voltage applied to the lamp (using another circuit system) he could control the frequency of the humming tone produced, which is basically a demonstration of the frequency modulationtechnique. He later attached a keyboard to his device as a control interface.

2. EARLY 20TH CENTURY ELECTRONIC MUSIC INSTRUMENTS

2.1 Objectives

This section explores the early twentieth century instruments such as *Telharmonium, Theremin, Ondes Martenot* and *Trautonium*. Each instrument is studied in terms of its technical elements, historical aspects, repertoire and certain issues related to performance and composition. By looking into the most innovative designs of the century, the section forms a basis for the further research that studies the impact of musical technologies on composition and performance as well as the model proposed for a new electronic music instrument design.

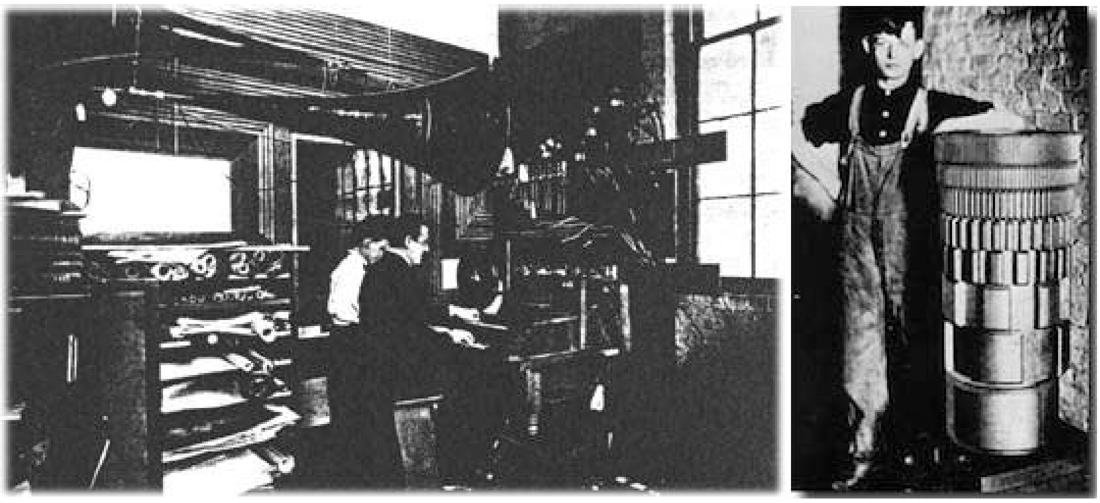

Figure 2.1: The Telharmonium controller keyboard, left. Rotating tone wheels from the Telharmonium, right (Chadabe, 1997, p. 5).

2.2 The Telharmonium (Dynamophone) by Thaddeus Cahill, 1897

2.2.1 Historical aspects of the Telharmonium

The first Telharmonium was built in Washington D.C., in 1900, in order to gain financial support. It was a small prototype version. In 1906 the Telharmonium was completed and demonstrated in Holyoke. This version used 145 tone wheels, five octaves and two touch sensitive keyboards. It included all ‘stops’ and ‘expression devices’ that are used to vary tone color, introduce vibrato and control crescendo and diminuendo of sounds. The entire instrument weighted about two hundred tons, occupying massive space. Only a single note (with six partials) used approximately two meters of shaft and five octaves needed ten meters of height. The musician was placed in a small room in the same building, with the control mechanisms, which were the touch sensitive keyboard, expression pedal for shaping the envelope of sound and stops for varying tone color.

In 1906 Telharmonium was disassembled and moved to New York, to a building across the Metropolitan Opera House in midtown Manhattan. The huge machinery was installed in the basement. The venue was called *Telharmonic Hall*. Cahill developed the instrument further by applying additional wiring and switches so that each tone wheel could be used for more than one note, enriching the timbral capacity. He also added a third keyboard to the instrument so that different voices could be played at the same time. Concerts in New York began in 1906, leading hotels and restaurants became subscribers of Telharmonium music as well as several wealthy clients who took the service directly into their homes. The public concerts increased from two to four performances a day.

Despite the public interest, some telephone users complained about Telharmonium interference in their conversations. Eventually the New York Telephone Company cancelled the agreement. There were other technical problems due to the power regulation of the massive synthesizer. Cahill’s new wiring and switches that provided additional harmonics caused the power supply to be overloaded, so music gradually became quieter as more notes were being played at the same time, in other words chords were quieter than single notes.

In 1906, Lee De Forest patented the *Audion*, which used the vacuum tube technology.

Thus transistors were introduced to the world of electric circuit design, preparing to reveal certain possibilities of sound generation which were to be explored extensively in the following years. He collaborated with Cahill to transmit Telharmonium music via wireless technology, but this did not turn into a permanent agreement due to factors such as the navy signal interference, the commercial dependability of this new technology etc. It is claimed that this decision by the New York Electric Music Company led to the demise of Telharmonium. In 1908, New York Electric Music Company collapsed; the Telharmonic Hall was locked up. Cahill moved the instrument back to Holyoke and built a third Telharmonium. The improved version was installed back in New York again, but this time did not meet the public enthusiasm it once had. In 1914, the New York Cahill Telharmonic Company declared bankruptcy.

Telharmonium was an ambitious project considering the technological circumstances of the time. It was innovative; being the first electronic music instrument containing the principles of additive sound synthesis. Cahill foresaw the potential of electronic music as a form of media that could be transmitted over a telephone network, decades before the introduction of wireless systems or radio broadcasting (Cox, 2010). Despite all the breakthroughs achieved by the instrument, it is not possible to say that Telharmonium was used to create new aesthetics or forms. It was influential, as Busoni has mentioned the invention in his *Sketch of a New Aesthetic of Music*. But the music played on the Telharmonium was Rossini, Bach, Chopin, Schumann, Beethoven, Schubert etc.

2.2.2 The technical principles of Telharmonium

Telharmonium uses sound generating technique of the rotating tone wheel, and the live electronic music is transmitted over telephone wires (Laurens Hammond has later used the tone wheel technique for his famous electronic organ in 1935).

Cahill’s patent in 1896 describes the instrument in full detail; the opening paragraph of it even uses the word ‘synthesizing’ to describe the way Telharmonium uses individual tones to create composite sounds. The tone wheels were mounted on pitch shafts, or axles, when rotated by the movement of the shaft, the tone wheels got into a rapid on-and-off contact with the metal brushes that were actually a part of an electrical circuit. The grooves in the tone wheel created the electrical oscillation for the desired frequency. Each tone wheel could produce a single pure sine tone. The instrument contained twelve pitch shafts, each for creating one note of the chromatic scale. Cahill had to cut correct size and number of grooves in the surface of each tone wheel, so that it could generate a ‘ground-tone’ which was identified by him as the first partial. Cahill added as much as five more tone wheels to provide overtones for each note of the scale. The pitch shafts were rotated in unison by a single motor, thus eliminating phase and tuning problems within the tone wheels. Each of them contained groups of tone wheels corresponding to different octaves of a single note. There were seven octaves in the device. The patent described that the first five octaves used six partials, the sixth used four, and seventh two. Cahill designed this according the fact that at higher frequencies musical sounds have fewer overtones. The Telharmonium had a pressure sensitive keyboard due to the coil in the circuit closing system. When depressed each key on the keyboard closed a circuit, thus activating the tone wheels corresponding to that note. The tones being played were mixed in a transformer circuit. By using creative mixing and filtering, Cahill was able to imitate sounds of acoustic instruments such as oboe, cello and French horn (Lee, 2000). The output of the sound was achieved by telephone receivers with large paper horns, through telephone wires. The thin diaphragms of the receivers provided better bass response. Cahill’s patent even included a preferred design of an electromagnetic loudspeaker with a wooden soundboard, but this unique forerunner of modern loudspeaker was never realized. Cahill’s design was no less than a complete electronic music synthesizer, with all stages of tone generation, dynamics control, mixing, amplification and keeping the system in tune.

2.3 Theremin by Lev Sergeyevich Termen, 1920

2.3.1 Historical aspects of the Theremin

Lev Termen invented the Theremin (*Aetherphon, Thereminvox* were its initial names) in Russia, 1920. The first composer ever to write for Theremin was Andrei F. Paschtschenko. His *A Symphonic Mystery for Theremin and Orchestra* was premiered in May, 1924 by the Leningrad Philharmonic (Holmes, 1985). Termen embarked a European tour in 1927. His performances in Berlin, Frankfurt, London and Paris were met with enthusiasm by the audiences. At the Paris Opera police forces were called to keep order among the crowds who were there to see Termen's demonstration of the new instrument. Theremin got to be well known in America after 1927. Termen signed a licensing agreement with the RCA Company to manufacture and market a commercial version of the instrument. Demonstration concerts at the Metropolitan Opera house were followed by performances with the New York Philharmonic. Termen and his new students performed with the New York Philharmonic Symphony Orchestra on August 27, 1928. Racmaninoff's *Vocalise* and Liszt's *Hungarian Rhapsody #1* were performed (Chadabe, 1997). The RCA Theremin was introduced in 1929. This version had a range of three and a half octaves. Joseph Schillinger wrote *Airphonic Suite for RCA Theremin and Orchestra* in 1929 to promote the RCA Theremin.Though met by public interest, only five hundred instruments were sold due to the fact that it was really easy to play the instrument and understand its principle at a first glance, but it was quite difficult to master since it required precise body control and great physical discipline. Due to the fact that Theremin contains no physical reference like a fingerboard, frets, keys etc., its effective control required perfect pitch and precise control over finger and hand motions. As it would be expected, building expressiveness by articulating with the left hand in addition to the pitch control mechanism occurring at the same timewas even harder.

During the first demonstrations of the instrument, the repertoire was filled with programmatic solo parts that could have been played easily on a violin or cello. The first and most well known virtuoso of the Theremin has been Clara Rockmore. She has played conventional music on Theremin, performed classical music recitals consisting of adaptations of string parts in works by Racmaninhoff, Stravinsky, Ravel and Tchaikovsky (Smirnov, 2010). In 1932 at one of Termen's concerts, Clara Rockmore was asked by Termen to perform on his *Terpistone,* the experimental dance platform that enabled the dancer to play a melody while dancing so that a perfect synchronization of sound and motion could be achieved. As he puts it in his own words, Termen asked this from Clara Rockmore since "none of the dancers who tried it could carry a tune" (Chadabe, 1997, p.9). Clara Rockmore performed her first concert at New York's Town Hall in October 30, 1934, accompanied by her sister Nadia on the piano. She could play trills and pitch leaps with great accuracy. Her articulation could work both on the flowing passages and staccato ones. Robert Moog describes Clara Rockmore's playing technique:

> She uses finger pattern movements in coordination with the wrist and arm to 'catch' pitches. So, when playing an arpeggio, she would start with the right hand titled back, with withdrawn fingers. To play the next note she would move her hand forward from the wrist, keeping her arm motionless. The third note would be played by extending the finger this time, and the forth by extending other fingers while turning the wrist sideways to bring the fingers closer to the pitch antenna. She would then continue the arpeggio by moving her arm forward and titling the wrist back again, so that the succession of movements can be repeated. During the right hand movements, she would use the left hand on the loop antenna to continuously articulate the notes. By shooting the fingers down and withdrawing them rapidly she could silence the tone for very short periods of time during the right hand movements from one pitch to another.(Holmes, 1985, p.52)

Composers such as John Cage complained about the use of the unconventional instrument for conventional classical music, and stated that the thereminists shielded the public from new sound experiences by the way they used the instrument to play in resemblance with the sound of violin or cello. While Clara Rockmore was working on the virtuosity of the Theremin playing technique, Lucie Bigelow Rosen was interested in exploring new musical possibilities of the instrument. She commissioned several composers to write original works for Theremin. Bohuslav Martinu's *Fantasia for Theremin, Oboe, Piano and Strings* explores the outer ranges of the instrument's pitches, dynamics and timbres. Characteristic long melodic lines that both blend and contrast the timbre of the Theremin with oboe and strings is another feature of the piece. Rosen premiered the work at Town Hall in New York, in November 1945. Rosen even wrote a manual for Theremin, including technical notes for maintenance and troubleshooting tips.

During his time in New York, living in the house provided by the Rosen family, Lev Termen continued to work on his inventions. His instruments from this period include: Rhythmicon, the Keyboard Theremin and the Terpistone.

Rhythmicon was a complex rhythm machine, an early form of drum machine using photoelectric principles and a keyboard. It was Henry Cowell's musical ideas that triggered the invention of the Rhythmicon; in other words Cowell specified the input and output phases of the instrument and given that information Termen designed the circuits to realize Cowell's idea. Rhythmicon was the first electronic rhythm machine. It was a keyboard instrument based on the Theremin, using the same type of sound generation technique - heterodyning vacuum tube oscillators. The seventeen key polyphonic keyboard produced a single note repeated in a periodic rhythm for as long as it was held down, the rhythmic content being generated from rotating disks which interrupted light beams that triggered photo-electric cells. The working principle of the instrument depended on light beams cast over photoelectric circuitry to transform the frequency of the beams to pitch and rhythm. When a key on the keyboard was depressed, it produced a pitched rhythm. It was possible to play multiple notes and rhythms by depressing more than one key at a time. The seventeenth key of the keyboard added an extra beat in the middle of each bar. The transposable keyboard was tuned to an unusual pitch based on the rhythmic speed of the sequences and the basic pitch and tempo could be adjusted by means of levers.

Cowell wrote two works for the Rythmicon, *Rythmicana* and *Music for Violin and Rythmicon* (a computer simulation of this work was reproduced in 1972). Cowell lost interest in the machine, transferring his interest to ethnic music. After Cowell, the machines were used for psychological research and one example (non working) of the machine survives at the Smithsonian Institute. The Rhythmicon was rediscovered twenty-five years after its creation by the producer Joe Meek (creator of the hit single *Telstar* in 1961). He discovered the instrument abandoned in a New York pawnbroker. Meek brought it back to his home studio in London where it was used on several recordings. This Rhythmicon was used to provide music and sound effects for various movies in the fifties and sixties, including: *The Rains of Ranchipur*, *Battle Beneath the Earth*, *Powell and Pressburgers*, *They're a Weird Mob*, *Dr Strangelove*, and the sixties animated TV series *Torchy*, *The Battery Boy*(Crab, 2005, p. 52). Tangerine Dream also used some sequences from the Rhythmicon on their album 'Rubicon'.

The Keyboard Theremin was a primitive synthesizer (a bank of tone generators controlled by a traditional organ keyboard) designed to emulate other instruments.

Martin Taubmann with his *Electronde* in 1933 also created a variation of the original Theremin, changing pitch/dynamics and timbre controlling principles instead of the space-controlled technique that required a certain level of precision, thus making it easier to perform. Terpistone was a small space-controlled dance platform upon which the foot movements of a dancer would trigger sounds from a Theremin. The 'musical floor' allowed the dancer to control pitch and volume by body position. A bank of colored lights was mounted on the wall behind and each light was activated by its corresponding pitch.

Edgar Varese asked Termen to develop an instrument for his piece *Ecuatorialfor a Small Ensemble*. The instrumentation consisted of baritone voice, organ, brasses and percussion. To this mix, Varese wanted to add an electronic instrument with a pitch range that exceeded the high C on the normal piano by an octave and a fifth. Termen developed the Fingerboard Theremin which uses the same beat frequency principle as the regular Theremin. The instrument is played upright, the left hand on a cylindrical fingerboard slides up and down to determine pitch while the right hand controlled the dynamics with a lever. When he revised the work in 1961, Varese substituted two Ondes Martenots in place of the Fingerboard Theremins since the instrument was no longer available.

Another important invention by Termen during this period is the Electronic Harmonium. It was developed by Lev Termen and Sergei Rzhevkin in1926. The instrument had 1200 divisions per octave, and was designed for studies in melody and harmony. Illuminovox was invented by Lev Termen in 1926. The instrument used an electro-optical projector with rotating discs to produce sound. Another instrument by Termen belonging to the ten-year period between 1920 and 1930 is the keyboard electronic tympani. One of the most ambitious concerts of the 1930's was the 1932 Carnegie Hall concert, at which he presented a sixteen-piece Theremin electrical symphony.

In 1930, the great depression took hold and Termen's financial situation got worse. There are a number of explanations regarding his departure from America, such as that he was a KGB (Russian Committee for State Security) agent and escaped from the States or he was captured and returned back to Russia under Soviet arrest. He was not heard from again for almost thirty years. Back in his motherland he was put into scientific researches and invented an electronic surveillance device; the wireless bug. Termen left a Theremin for Lucie Rosen as a departure gift. This traveling model had an angled top that enabled a built in music stand and a neon tube which allowed the performer to visually preview the pitch through a small hole in top of the cabinet. In 1991, at the age of 95, Lev Termen came back to the United States for a visit. His daughter Natasha performed Rachmaninoff's *Vocalise* on one of her father's Theremins, accompanied by Max Mathews playing his Radio Baton. During his reunion with old friends in this last visit, Termen recalls a performance in Moscow in 1921 and his encounter with Lenin, as he demonstrated the instrument by playing Glinka's *The Lark*. Lenin played the melody, starting with the assistance of Termen holding and positioning his hands, later going on to finish the tune by himself. "It is not so often that a head of state tries out the latest electronic music instrument and, yet more exceptionally, plays it well"(Chadabe, 1997: p.8).

Theremin became widely known for its use in film scores later in the period. The soundtrack score of Miklos Rozsa for Alfred Hitchcock's *Spellbound* won the Academy Award. The Theremin in the score was performed by Dr. Samuel J. Hoffman. Robert Whitsell built a specialized Theremin for Paul Tanner, a trombonist from Holywood. This version of the instrument was later named 'Electro-Theremin'. The difference in operation was the control mechanism. Consisting of an oscillator and an amplifier circuit; the Electro-Theremin was played mechanically by a sliding handle. The sliding handle was mounted on a fifteen-inch strip of paper which had a keyboard image on it so that the corresponding pitches to the slider position may be viewed. The loudness was controlled simply by the volume dial on the amplifier. Another distinction from the original Theremin was its sound, as electro-Theremin produced pure sine waves with no side bands or harmonics added. Tanner performed in the album *Music for Heavenly Bodies*. The instrumentation consisted of orchestra and Theremin, conducted by Andre Montero and arranged by Warren Baker.

After this first performance, Whitsell made some adjustments to the design, specifically to improve the manual articulation of notes. Tanner used his instrument to create sound effects for several Warner Bros movies and ABC television shows of the late '50s and early '60s, as well as CBS and NBC television networks. Some of the movies include *The Giant Gila Monster* and *Straight Jacket*. The instrument was used for sound effects for the TV shows *I Love Lucy*, *My favorite Martian*, *Dark Shadows* and *Lost in Space*. Tanner got his most famous electro-Theremin job when

Brian Wilson of the Beach Boys asked him to join their 1966 recording sessions. From the album *Pet Sounds* Tanner played in two pieces, *I just wasn't made for these times* and the single *Good Vibrations*. Later Robert Moog constructed a ribbon controlled transistor oscillator for the band to take on tour with them, so they used this instrument to perform the partitions of the electro-Theremin on live shows. The instrument was played by sliding the finger along the ribbon controller that could be marked at the places of the desired pitches due to the partitions. It also had a volume control (Holmes, 1985).

Although Theremin became known in the USA as a home instrument and featured in many film soundtracks of the 1940-50s and appeared in several pop records of the 1960s it never overcame it's novelty appeal and was used for effect rather than as a serious instrument; most recordings employ the Theremin as a substitute string instrument rather than exploiting the microtonal and pitch characteristics of the it. Theremin continues to be a popular instrument for performance and composing even today. American composer and Theremin player Eric Ross wrote more than fifteen works for Theremin since 1982. Jazz trumpeter and thereminist Youseff Yancy plays Theremin since 1960s and often teams up with Ross. Another important performer of the Theremin is Lydia Kavina, the granddaughter of Termen's first cousin. Kavina released the album *Music from the Ether: Original works for the Theremin* in 1999, which consisted only of works composed specifically for Theremin.

Theremin's design inspired several early electronic music instruments depending on the heterodyning circuit technique. Jorg Mager's *Sphaerophon* (1926) –an improved version of the early *Electrophon* with added keyboard- was designed to play quartertones. The monophonic instrument was controlled with a keyboard. In 1931 Winifred Wagner (Richard Wagner's daughter in law) commissioned Mager to produce electric bell sounds for the production of the opera *Parsifal.* Mager developed a polyphonic version of the instrument that could play chromatic scale in 1935; it was named *Partiturophon.* The instrument was basically a five-voice *Sphaerophon* with three to five keyboards. It allowed the player to play four (or five) voices at once, one voice per keyboard. Since the polyphony came from separate manuals, the keys were constructed to be narrower and shorter than the regular organ or piano keys.

2.3.2 The technical principles of Theremin

Theremin is the first space-controlled instrument. It is monophonic and its performance technique is suitable for solo instrumental playing. Sliding tones or effects such as vibrato is easy to achieve technically, the sound (sine tone) is continuous as long as the hand is in the vicinity of the vertical antenna.

Figure 2.2: Lev Termen playing the Theremin (Crab, 2005, p. 31).

Theremin operates on a modulation principle called *beat frequency oscillation*, or *heterodyning*. The technique uses two vacuum tubes as oscillators that generate frequencies above the human hearing range. The difference of these electrical signals (beat frequency) provides a signal that is in the human hearing range. One of the vacuum tubes generate a fixed frequency while the frequency created by the other one can be altered by moving the performer's hand in the vicinity of the vertical antenna. The pitch is controlled by the back and forth movements of the performer's hand in the electromagnetic field of the vertical antenna; the closer the hand to the antenna, the higher the pitch. There is another loop antenna positioned horizontally to control the loudness of sound and shape its envelope. Bringing the hand down close to the antenna silences the sound while taking it upwards made it louder. Some models also included a foot pedal to control dynamics. The original Theremin is said to have had a range of five octaves (Chadabe, 1997).

To investigate the structure a bit closer; the circuitry consists of two sections: one for supplying power to the electromagnetic fields, the other for tone producing, which is known as the *beat frequency oscillator*. The oscillators operate well above the human hearing range. One of them operates at 170 kHz fixed frequency, while the other in the range of 168-170 kHz. The upper limit of the human hearing range cannot exceed 20 kHz. The variable oscillator is connected to the vertical pitch antenna through a large inductor. The pitch antenna has a small capacitance to the ground. The antenna and the inductor form a series resonant circuit whose resonant frequency lies in the 168-170 kHz range.

When a very small amount of capacitance (as small as one picofarad) is added by the hand near the antenna, the resonance frequency of the circuitry is altered, thus generating the desired pitch. When the performer brings her hand closer to the pitch antenna, the resonance frequency drops while the fixed oscillator frequency remains constant. A third circuitry within the structure, called detector or mixer, combines the two signals and extracts the difference, a frequency of 0 to 2 kHz that lies in the human hearing range (Roads, 2004).

The volume control circuitry operates in a similar manner. A series resonant antenna circuit is connected to a high frequency oscillator. The high frequency energy flowing through the volume antenna is used to heat the filament of a vacuum tube that is in the amplifying stage of the instruments. Therefore as the left hand approaches the loop antenna the circuitry is detuned, less energy is outputted to heat the tube, resulting in a lower volume. To be able to perform Clara Rockmore’s articulation technique, this design introduces a serious disadvantage, since the filament of the amplifying tube requires time to heat up or cool off, therefore limiting the rapidity of the articulation. Later models of Termen’s instrument use a volume antenna circuitry with a faster response.

The sound of Theremin was very close to that of a pure sine tone, but with enough side bands to add depth and body to the tone. As it was the first space-controlled instrument, the performance of the Theremin introduced a high degree of theatricality.

The moving hands of the performer in the air and the ‘untouched’ instrumentmystified the audiences.

The sloped surface of the Theremin serves as a convenient music stand. Vertical antenna controls the pitch while the horizontal loop antenna for controls dynamics. Tuning knobs and control switches are located in the lower part of the front of the cabinet.

To play the instrument, the performer stands in front of the Theremin, a little left off center. The feet are spread slightly to keep the body a still as possible. When the instrument is properly tuned, the pitch goes lower than two octaves below middle C when the player’s hand is back at her shoulder, to approximately two and a half octave above middle C when the hand is almost touching the antenna. Maximum loudness is achieved when the left hand is removed from the antenna. Silence occurs when the hand is at rest on the antenna. The two antennas respond to all body movements, therefore it is very important for the performer to have firm control over body and head motions as well as hands and arms. The ability to stand motionless is absolutely necessary. To play partitions of rapid arpeggios *aerial fingering* technique is required, as described in the previous section. *Aural feedback correction* is another technique used by the thereminist Clara Rockmore. The placement of the loudspeaker is extremely important for the realization of this technique. Unlike the fingerboard of a violin or the keys of a piano, there is no physical connection with the instrument in the Theremin, so the performer simply trims the pitch after the first attempt, meaning the fine tuning of the intonation comes right after. Therefore it is crucial for the player to be able to hear the output of the instrument clearly, as this is the case for all electronic music instruments that have no acoustic output. Clara Rockmore uses an open back speaker cabinet placed behind her, directed towards the audience in order to realize this control technique.

2.4 The Futurist Movement and the Introduction of the Audion Piano

Before proceeding to next important instrument of the period, it is crucial to mention the Futurist Manifesto and the invention of the vacuum tube oscillator in order to gain an understanding of the dynamics of the period.

The Futurist Movement is generally associated with Luigi Russolo as well as the mechanical instrument *Intonarumori*. However it was first introduced by Francesco Pratella in 1910 with his *Manifesto for Futurist Musicians*.

In his *Technical Manifesto of Futurist Music*, Pratella suggests that composers should “master all expressive, technical and dynamic elements of instrumentation and regard the orchestra as a sonorous universe in a state of constant mobility, integrated by an effective fusion of all its constituent parts” (Manning, 1985, p. 4). The Futurist movement; like the introduction of the early electronic music instruments, have influenced what was to come later, an aesthetic breakthrough of traditional forms and vehicles of expression.

Although not electronic (acoustic generators), the *noise-intoners* built by the painter/musician Luigi Russolo and the Futurist Manifesto was a primary source of inspiration for composers such as Edgar Varese, Pierre Schaefer and John Cage. The *Intonarumori* was basically a solid rectangular box (of varying sizes) operated with a crank for evoking the noise and a lever for adjusting the pitch. A horn was attached for amplification and projection of sound. The Intonarumori were used to realize sounds that were listed in the *Art of Noises* by Russolo, such as roars, whistles, whispers, screeches, percussive noises and voices of animals and humans. Russolo later invented new instruments, *Rumorarmonio* – Noise-Harmonium (1922) which put several of his devices under the control of a piano-style keyboard. *Enharmonic Piano* (1931) was another invention by Russolo.

Lee De Forest invented the triode electronic valve, thus introduced the vacuum tube technology in 1906. This technology dominated the design of electronic music instruments until the introduction of the semi-conductor transistor in the 1960s.

The immediate application of the triode valve was in the radio technology, wireless transmission. Forest noticed that it was possible to create audible frequencies with the valve, using a technique called *heterodyning.*As it was used in Theremin, heterodyning is a beat frequency technique.

The difference of two high frequency signals generated by the triodes results in a lower frequency in the audible range.

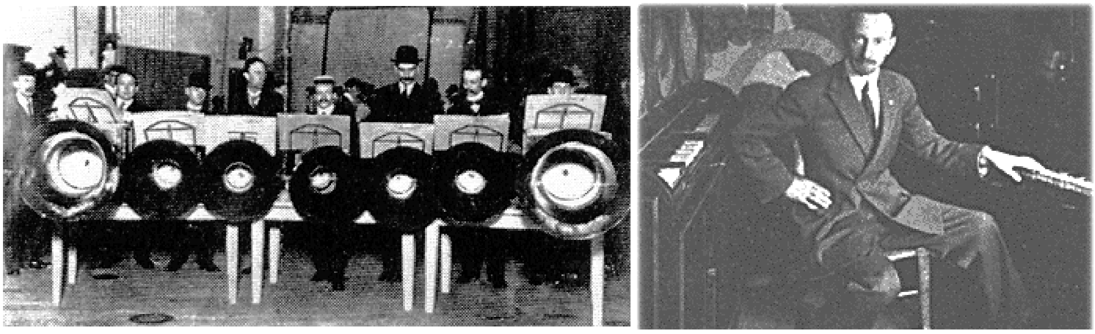

Figure 2.3: The first public concert of the Intonarumori, 1914, left. Russolo and two of his ‘Rumorarmonio’, right (Holmes, 1985, p. 39).

The Audion Piano is the first vacuum tube instrument, invented by Lee De Forest in 1915. The instrument was monophonic, using a single triode per octave, thus it was possible to play one note at a time within an octave. The output was sent to a set of speakers placed around the room for dimensional effect. The instrument is also the first to apply heterodyning oscillator system and body capacitance to control pitch and timbre. Forest remarks “In fact the pitch of the notes can be changed by merely putting the finger on certain parts of the circuit. In this way very weird and beautiful effects can easily be obtained” (Crab, 2005, p. 27).

2.5 The Ondes Martenot by Maurice Martenot, 1928

2.5.1 Historical aspects of the Ondes Martenot

Originally called the *Ondes Musicales* (musical waves), the instrument was designed by Maurice Martenot. Martenot’s intention was to invent an electronic music instrument that could join the ranks of traditional symphonic instruments and be the focus of works written by the leading composers. To achieve this aim, Martenot analyzed the factors that prevented Theremin to become widely accepted by musicians and composers. The main factors were; due to its ambitious design Theremin did not look like any traditional music instrument, and when it came to technique, it was very hard to master.

Ondes Martenot uses the same principles for sound generation with Theremin, but its control mechanism depends on entirely different principles, which may be traced in the traditional instruments. Therefore, the instrument looked ‘at home’ in the orchestra. It was the size of a small, upright keyboard instrument such as the clavichord; its wooden cabinet and matching loudspeakers were pleasing to the conventional eye.

Martenot carefully packaged the introduction of his instrument by commissioning an orchestral work to spotlight its musical qualities, as he wanted his instrument to find a place among the traditional symphonic music instruments from the start. He believed that the power of Theremin's entrance to the musical world was diminished since the instrument was judged as a scientific curiosity in the beginning, and only after this wave passed it started to get accepted as a serious musical instrument among musicians and composers.

Ondes Martenot was first introduced to the public in Paris. Martenot himself played the solo part in the world premiere of Dimitri Levidis's *Symphonic Poem for Solo Ondes Musicales and Orchestra* in May 1928. The piece included microtonal elements such as quarter and eight tones, so the impact of the instrument's entrance to the public consciousness was dramatic. The Ondes Martenot was not a difficult instrument to learn, thus it appealed to the musician more than the precise body control required to play the Theremin. After this successful premiere in Paris, a European tour followed. The conductor Leopold Stokowski brought Martenot to the United States to perform the Levidis work with the Philadelphia Orchestra in December 1930. A world tour followed, at the Exposition Internationale de Paris of 1937, there were demonstration concerts by Ondes Martenot ensembles of up to twelve musicians. In 1960, the Paris conservatory offered classes in Ondes Martenot performance. A formalized training program and school for the instrument was established under the direction of Martenot.

Though its fame and success among composers and musicians, Ondes Martenot never really achieved mainstream status. Martenot was a musician with ideals, not a commercially minded person. He never tried to industrialize his instrument, as he produced them in his atelier, at a rate of approximately three per year. Laurens Hammond, on the other hand, was a commercially minded person, as he developed the first commercially successful electronic music instrument. In 1935, he produced the Hammond Electronic Organ and it quickly achieved mainstream status, so that for many years people said 'Hammond' when referring to an electronic organ (Holmes, 1985).

During the 1930's, well-known composers such as Darius Milhaud, Arthur Honegger and Olivier Messiaen wrote works for the Ondes Martenot. Marcel Landowski's *Jean de la peur* uses the instrument to create effects as an atmospheric accompaniment to the orchestra. Messianen's *Turangalila Symphonie* (1948) and *Le Merle Noir* (1951) use the instrument's ability to create sounds such as bell sounds or birdsongs. Messiaenfurther contributed to the instrument's repertory with his earlier work *Trois petites liturgies de la Présence divine* (1943-44), in which the instrument provides a shifting drone accompaniment to women's voices, piano, strings and percussion; and his compositional summa, *Saint François d'Assise* (1975- 83), where the Ondes Martenot features in three of the nearly four-hour work's eight tableaux. The Ondes was also used effectively as an ensemble instrument in Milhaud's *Suite for Martenot and Piano* (1933) and Jacques Charpentier's *Lalita for Ondes Martenot and Percussion*. More than 300 composers have contributed to the repertoire, containing approximately 100 chamber works, 50 operas, 100 symphonic works and ballets, 500 scores for theatre and film.

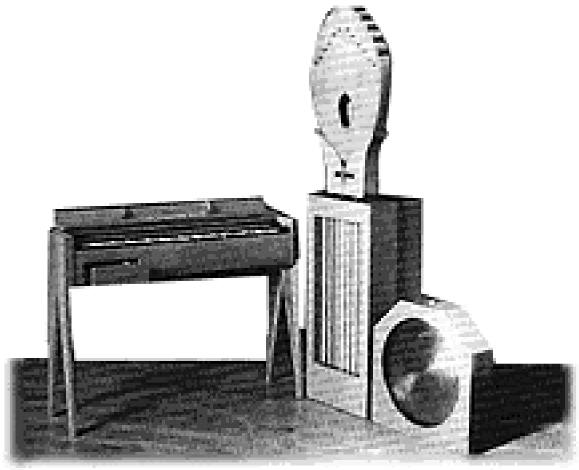

Figure 2.4: The Ondes Martenot, concert version (Manning, 1985, p. 152).

Like Theremin, the Ondes Martenot has been associated with several virtuosi performers. Martenot's sister Ginette Martenot was the first one. The best-known Ondes Martenot performer is Jeanne Loriod. She has dedicated her career to the mastery of the instrument and the documentation of its written repertory. Loriod performed all of Messiaen's works for Ondes Martenot, and she recorded the *Turangalila Symphony* six times. Though the Ondes Martenot partition in this work was written for Martenot's sister Ginette, it was Loriod who popularized it.

She was the sister of Yvonne Loriod, the second wife of Oliver Messiaen.In 1974, she founded an ensemble (sextet) to perform Messianen's first work for the instrument, *Fete des Belles Eaux for sextet of Ondes Martenot*. In 1970, She started teaching at various conservatories and finally succeeded Martenot himself in 1970. She published a definitive work on the instrument, a three-volume magnum opus named *Technique de l'onde electronique type martenot* in 1987. She performed in Maurice Jarre's (fellow Ondes Martenot player) film scores for *Lawrance of Arabia* (1962) and *Mad Max* (1985). It is said that shortly before her sudden death in 2001, she was to perform with the British pop-rock group *Radiohead*, which may have been a new introduction for this early twentieth century classical music instrument (Crab, 2005).

2.5.2 The technical principles of the Ondes Martenot

The Ondes Martenot uses the same *beat frequency oscillator* technique as the Theremin yet it was designed as a keyboard instrument. The Ondes Martenot is monophonic, thus it is restricted to play melodies.

The original instrument played by Martenot at its premiere in 1928 controlled pitch by a ribbon controller. A metal ring that was moved laterally using the index finger of the right hand produced pitch. The ring was attached to a metal wire that adjusted a variable capacitor on the ribbon and thus changed the frequency of the tone over a seven-octave range. The ribbon was superimposed over a picture of a keyboard (which actually becomes a keyboard in the later versions) so that the corresponding notes on the chromatic scale could be viewed. Sliding the ring to the left played lower notes, sliding to the right played higher notes.

In 1932, Martenot added an organ style keyboard to the instrument. The instrument could be played using either the keyboard or the finger slide control. The ring in this version used a metal ribbon in place of the wire, and the ribbon was placed in front of the keyboard, so that notes corresponded to the position of the ring. The surface of the ribbon was also marked by small metal bumps corresponding to the notes on the scale.

The special feature of the keyboard on Ondes Martenot was that fluctuations in pitch for vibrato effect could be achieved when depressing a note, by moving the key laterally.

The left hand controlled volume with a pressure sensitive key. This was unique, since when the key was fully released, no sound was heard. So the left hand operated as an expressive dynamics controller for this monophonic device. Gradually depressing the pressure sensitive key resulted in volume increase. A knee lever was also applied, so that the foot could take on the dynamics control when the left hand operated on the small bank of keys near the pressure sensitive key to select timbre and filters to alter the sound. A lever underneath the keyboard was added, which could be controlled by the upward push of the right knee, resulting in continuous changes of timbre.

The output of sound in the Ondes Martenot was another important issue. Martenot designed four basic loudspeakers as diffusers to project the sound of his invention. They were called: *Haut-parleur* (loud speaker), *Resonance*, *Metallique* and *Palm*. *Haut-parleur* was a standard loudspeaker and was the loudest of the four varieties. *Resonance* was a speaker for creating reverberation. The design had an upright wooden cabinet and a standard speaker cone, but the front of the box was sealed with vertically oriented plastic strips. They would produce resonating sonorities that were fresh at the time. The *Metallique* was shorter and produced sound by the means of a gong. The signal of the tone was run through a transducer directly into the gong, using the sympathetic vibrations of the body to create audible pitch. This metallic sounding speaker resonated often to produce ring modulation type of effects. *Palm* had a resonating body shaped like an upside down cello.

Twelve strings were attached to the front and back of the speaker. The electrical tone signals were transformed through a transducer and played to the strings which vibrated to reproduce pitch. The combination of the cello-like resonating body and the vibrating string produced ‘eerie’ bowed string sounds. Later when the instrument was in use, some technical problems due to the usage of four speakers were addressed, as connecting all speakers diminished the output considerably, or the three effective speakers other than the *Haut-parleur* became almost inaudible in certain situations.

The selection of speakers was controlled by the left hand, using the switches on the panel that also housed the pressure sensitive key. These speaker selections, combined with the filter control gave the musician and composer an extra-ordinary range of sonic possibilities. The volume control of the left hand allowed the manual shaping of the envelope of the sound.

The Ondes Martenot could play the twelve notes of the chromatic scale (with the keyborad) and everything in between (using the ring slide control), so it was possible play microtonal music with the instrument. Pictures of various microtonal scales could be placed parallel to the ribbon controller, so that the performer could play the pitches accurately and with ease.

2.6 The Trautonium and Mixturtrautonium by Dr. Friedrich Trautwein, 1928

2.6.1 Historical aspects of the Trautonium

The Trautonium was developed in Germany between 1928 and 1930. The early evolution of the instrument was born from the collaboration of Trautwein and the composer Paul Hindemith. Oskar Sala was a composition student of Hindemith at the time.

Dr. Friedrich Trautwein and Paul Hindemith met at the experimental radio station in Berlin Academy of Music in 1930. Trautwein's earlier attempt to design an electronic organ was refused due to the lack of funding. When Trautwein met Hindemith, he decided to exclude the idea of a keyboard and design a string controller instead, as he was inspired by the viola virtuoso composer who evaluated the electronic instrument design issue from the perspective of string instruments, instead of an organ.

The first concert of the Trautonium, was named the *Electric Concert*and it was given at the Berlin Academy of Music in 1930, featuring the premiere of *7 Trio pieces for Three Trautonien* by Paul Hindemith. Paul Hindemith had agreed to write music for the instrument if Trautwein agreed to build three of them by June 1930. The instruments were played by Hindemith himself, Oskar Sala and a piano instructor from the academy. The concert was such a success that the German electronics firm Telefunken which produced the neon-tube oscillators that are used in the instrument decided to manufacture and market the Trautonium for home use. The model produced included a single fingerboard and a single pedal. This commercializing attempt was not successful though, as only a hundred were built, and even less could be sold between 1932 and 1935.

Hindemith composed more works for the instrument, most notably the *Concertino for Trautonium and String Orchestra* in 1931.

It was Oskar Sala who has been most associated with the instrument over the years both as a performer and musician. During the World War II, Trautwein's research on Trautonium was not banned and he was not forced to leave the country in self-exile like many other artists at the time had to, due to his close relationships with the Nazis. Trautweinmanaged to demonstrate this experimental project as a conservative and harmless attempt, thus was left alone. After the war, Trautwein continued to work on his instrument but he was already years behind Sala in terms of engineering skills. In 1952, he built a *Monochord* for the Electronic Music Studio of West German Radio in Cologne which was a specialized instrument based on the same technology. By the time of Trautwein's death in 1956, Oskar Sala had took on the mission develop Trautonium further as an engineer, inventor, composer and performer. Despite the ongoing developments in Cologne, the establishment of the electronic music studio of Herbert Eimert and Karlheinz Stockhausen, Sala kept on working with Trautonium alone and did not prefer joining other mediums of expression for electronic music.

During the 1960's, Sala formed his own studio and took on commissions for stage, screen and television. In 1961, he collaborated with composer Remi Gassman to produce the score for the George Balanchine ballet *Electronics*. Remi Gassman's comments from the time reveal an obvious distaste for most of the electronic music being produced by his contemporaries. He considers the Studio Trautonium (Mixtur Trautonium) an electronic instrument that enables the production of music without having to sacrifice all the values of the traditional perspective. He states that considering the latest improvements made by Sala, the Trautoninum incorporates the complete resources of the electronic sound studio as well. Within this instrument and the ballet *Electronics*; electronic sound, the virtuoso possibilities of the instrument due to performance, and further manipulation stages of the electronic sound studio are bound together for the first time (Holmes, 1985, p. 72). He dislikes the 'pure' sound of Theremin and Ondes Martenot (that do not contain enough overtones according to his taste), as well as the 'dehumanized' effect of Musique Concrete and the 'tonal equations' of the German school of Elektronische Musik and the Serialist Movement.

In 1962, Lejaren Hiller visited Sala at his studio at MARS Film in Berlin. According to his impressions he implies:

“Sala is convinced of the necessity of performing music to achieve the results he wants. He improvises much of his music for films directly on the instruments while watching film proofs…” (Chadabe, 1997, p. 12).

After this ballet project, Sala was asked by Alfred Hitchcock to produce a totally electronic score for his 1963 film *The Birds*. Sala created the score using Mixturtrautonium and magnetic tape. Even the sounds of the birds were crested using the instrument. Sala has completed over six hundred works, which have been stored on magnetic tapes in his studio. The German brand Doepfer Musikelectronik has recently worked with Sala to produce a semiconductor version of the Mixturtrautonium as well as modules for re-creating the ‘subharmonic’ filters and other controls associated with the actual analog instrument. Ambient/electronic composer Pete Namlook continues to write music for the Trautonium.

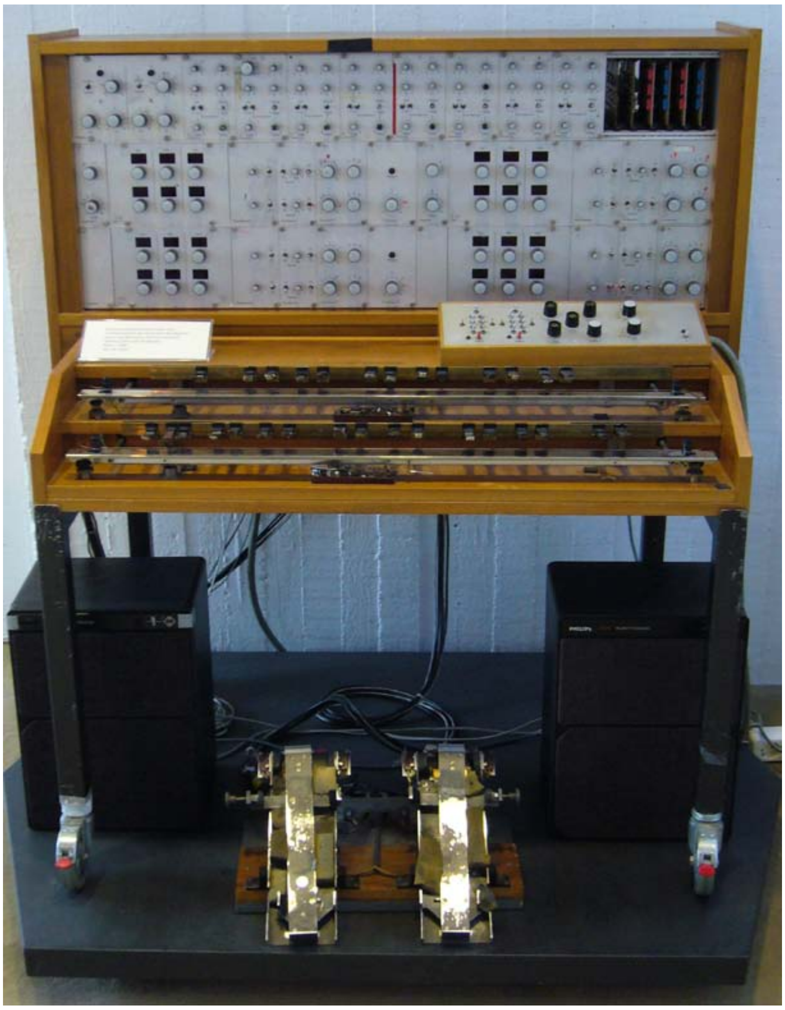

Figure 2.5: The Mixturtrautonium (photo taken at the Berlin Musical Instrument Museum).

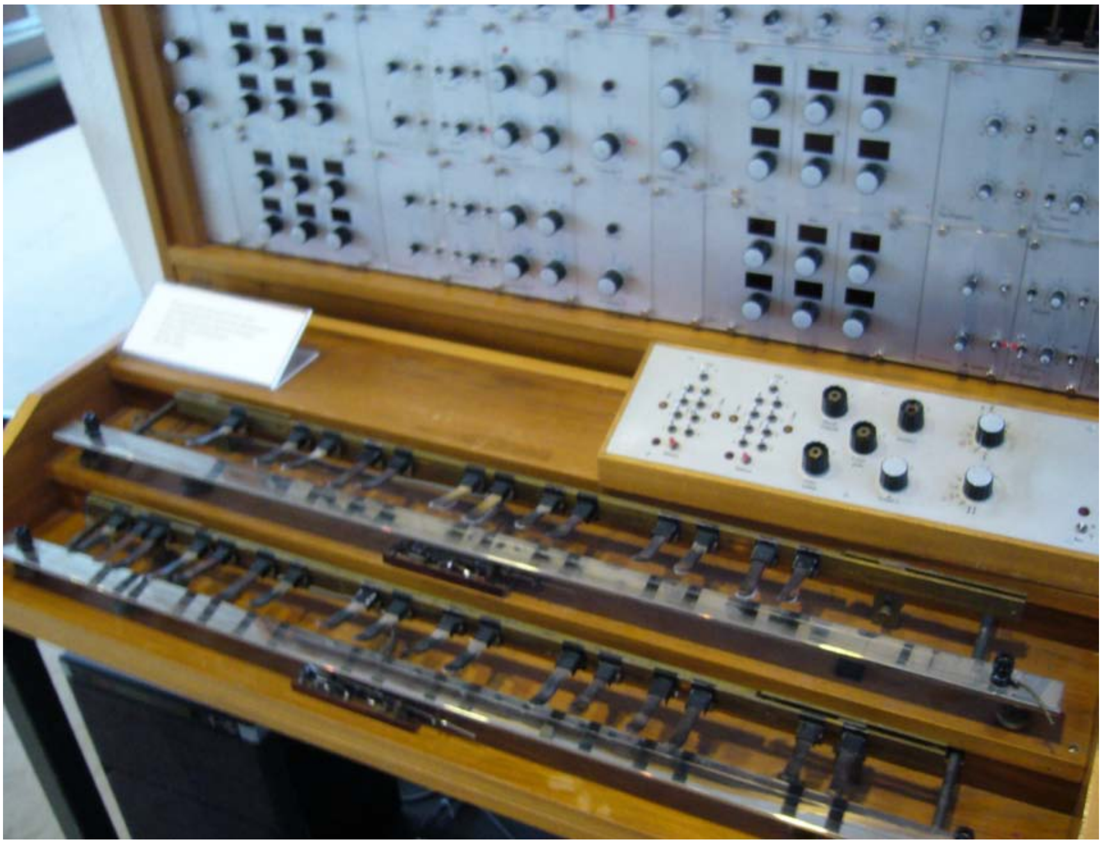

Figure 2.6: The Mixturtrautonium keyboard with leather covered tongues and the subhoarmonic filters panel (photo taken at the Berlin Musical Instrument Museum).

2.6.2 The technical principles of the Trautonium

The instrument comprises “neon-tube sawtooth tone generators with resonant filters to emphasize formants” (Crab, 2005, p. 47).

Trautwein barrowed the principles of *Hellertion* by Bruno Helberger and Peter Lertes for developing the Trautonium. Built in 1928, the Hellertion used the neon-tube oscillators for sound generation and was played by pressing a leather covered metal ribbon against a resistance plate to change the pitch. The earliest version of Hellertion was monophonic, later version (demonstrated in 1936) included four separate monophonic fingering ribbons to allow polyphony.

Trautonium is considered an electronic string instrument, yet it is not an electronic version of cello, viola or any other string instrument. The string constitutes the controller interface which is a wire pressed by the finger to create sound. The instrument has a fingerboard in the form of a metal plate. The wire is stretched a few centimeters above this metal plate. To produce a tone the player simply presses the finger on the wire.

When the wire touches the surface of the fingerboard (the metal plate), the circuit is closed and a current is sent to the neon tube oscillators. The instrument is monophonic and spans three octaves. The pitch goes up from left to right along the fingerboard. A foot pedal is employed to control the volume. The positions of notes of the chromatic scale are marked on the fingerboard, to give reference to the musician.

In 1934 Trautwein added a second fingerboard to his instrument so that two notes could be played at once. Another feature to this version was the addition of the ‘tongue’. Tongues were metal strips covered in leather (nonconductive material) that were mounted on a rail installed few centimeters above and running parallel to each of the two resistor wires. The tongues could be slid to any position along the length of the wire, thus creating a ‘preset’ opportunity for the performer, as it became possible to set the positions of the tongues according to the repeating pitches used in the composition that is to be performed. Pressing a tongue was like pressing a key; the wire is pushed downwards so that it contacts the metal plate.

This feature introduced ease of performance. Unlike with a vibrating string of a traditional string instrument, the gradation of the electrical string manual is linear (instead of exponential) so that all octaves have the same finger range.

The sound producing circuitry of Trautonium is different than that of Theremin and Ondes Martenot, as they both used the beat frequency oscillator technique. The sound of the Trautonium is a sawtooth waveform that is rich in harmonic sidebands, created by the neon tube oscillators. Trautonium has an audio oscillator at exact pitch rather than a beat-frequency oscillator at difference pitch. This distinguished its sound from Theremin and Ondes Martenot, which was carried further by the addition of a set of filters, controlled with rotary dials. With the use of the filters, it was possible to adjust the balance of the fundamental and the harmonic sidebands in relation. This was a forerunner of subtractive synthesis technique which is basically the careful manipulation of sidebands to produce timbral changes. This unique form of subtractive synthesis produced a tone that was distinctive and unusual when compared to the usual heterodyning valve instruments of the 1920-30s. The opportunities for shaping sound was not limited to only one set of controls, addition of more sawtooth waveform oscillators and filters to fine tune harmonics were presented in order to achieve a wide palette of tone color variations.

Control of dynamics was not arranged in the early versions of the instrument; all of the notes had sharp attack that could not be controlled. In the later versions, Trautwein devised a circuitry to make the fingerboard touch sensitive. He used mercury filled resistors beneath the wire mechanism, so that the harder it was pressed, the louder the sound produced would be.

In 1952, Oskar Sala made improvements to the instrument and called the new version *Mixturtrautonium*. The primary improvement was the expansion of harmonics available for the tones and improved controls. 'Mixtur' defined the combination of four sub-harmonics for a given master frequency. The Mixturtrautonium had two fingerboards, thus two oscillators allowed two notes to be played at the same time. The circuitry was designed to allow up to three mixtures of harmonics for each of the two fingerboards, twelve harmonics for each fingerboard, which makes a total of twenty-four for the two manuals.

The harmonic mixtures were controlled by two foot pedals (one pedal for each fingerboard) and side switches. The player triggered notes with the left and right hands on the fingerboard, controlled loudness and harmonic mixture with the both feet. Other improvements by Sala were the addition of a reverberation unit, a white noise generator and a power regulator to produce rhythmic sequences.

3. MID 20TH CENTURY ELECTRONIC MUSIC INSTRUMENTS

3.1 Objectives

The main purpose of this chapter is to explore the important electronic musical instruments of the mid twentieth century, focusing on the period from 1935 to 1950. Some of the transitional instruments of the early twentieth century will be examined too in order to achieve a unified approach to the topic. A classification according to operating principles of electronic music instruments will be suggested for further analysis.

3.2 The Electronic Organ by Laurens Hammond, 1935

The electronic organ by Laurens Hammond operates electromechanically, using the tone-wheel principle that was first introduced by Thaddeus Cahill with his Telharmonium. The Hammond electronic organ uses ninety-one metal tone-wheels, each about the size of a coin. All of these tone wheels are placed on and rotated by a common rotating shaft. Therefore, the sound generation stage of the instrument is handled electromechanically by the tone wheels. Hammond used vacuum tubes in other stages of the instrument such as: power control, amplification and sound mixing. The advantage of the vacuum transistor technology that was available to Hammond made it possible for him to house the instrument in a small cabinet, whereas this required an entire basement and approximately two hundred tons of material for Cahill back in the 1900s.

Hammond designed the Electronic Organ to mimic the functions of a pipe organ; the instrument had sliding tone filters reminiscent of organ stops that were used to remove partials from sound.

The design was stable; it stayed in tune. It had an instantly recognizable sound that was regarded as warm by the audiences as well as the musicians. The purpose in engineering of this organ was not to produce sounds that were unheard of, it was simply meant to simulate a pipe organ and serve conventional tastes, with the ease of performance and maintenance issues.

“In order to succeed, the step between the old and the new should not be too large” (Braun, 2000, p. 12). This instrument can be considered as the first commercially successful electronic music instrument. Approximately five thousand units were sold by 1940, more than one third went into the churches. Its mechanical simplicity made the instrument suitable for mass production. The Hammond model B3 which was introduced in 1950’s remains one of the most popular electronic organs among rock, jazz and rhythm-and-blues musicians. The Rangertone Organ was introduced in 1931, this instrument also used Cahill’s tone-wheel principle for sound generation but it never become a commercial success.

In 1939, Hammond organization introduced additional models that depended on vacuum-tube technology for sound generation: The monophonic *Solovox* and the polyphonic *Novachord*. The Solovox was a soloing instrument that was used in combination with a piano or another organ. It was basically a monophonic vacuum tube oscillator instrument with a divide-down circuitry. The Novachord (comprised 169 vacuum tubes, divide-down synthesis and formant filters) was a much more ambitious design compared to the original Hammond organ. It had complex attack decay characteristics, sustain controls, tone color controls, percussive sound options and a six-octave keyboard instead of five.

The instrument generated sound electronically, using twelve vacuum tube transistor oscillators to generate the upper octave of the keyboard. A circuitry of additional vacuum tubes divided these high frequencies in order to produce the tones for the lower octaves, giving a six-octave range using the frequency division technique. The Novachord was one of the first electronic instruments to use this technique which was later to become a standard in electronic keyboard instruments. The tone controls of the instrument included presets such as*deep, brilliant, full, normal* and *small* as well as vibrato presets such as *strong* or *soft* and envelope presets such as *bass* or *percussion* (these presets made it possible for the instrument to mimic orchestral sounds, making the instrument the forerunner of later synthesizers). Due to its interior design, which is the circuitry with more than a hundred vacuum tubes, the instrument proved to be unstable and thus its manufacture ended before the end of World War II.

In May 1939 *The Novachord Orchestra* lead by Ferde Grofé performed daily at the Ford stand at the New York World Fair with four Novachords and a Hammond Organ. The instrument was used in Adrian Cracraft's *All Electronic Orchestra*, it was also featured in several film scores such as Hans Eisler's *Kammersinfonie* in 1940.Due to the instability of its multiple tube oscillators and demanding playing technique the instrument lost popularity; The Novachord was discontinued in 1942. A Hammond employee comments:

> The Novachord made beautiful music if played well, but it was not well adapted either to an organist's style or a pianist's style. Thus it required development of a specific style, which not many musicians were prepared to do. It also had technical problems, requiring frequency adjustments to keep it operating because the frequency dividers and electronic components before the war were not nearly as good as those available in later years. The Hammond Organ Company could have revived it after the war, and could have made it better in light of available technology at the time, but sales had been disappointing and so it was not considered a good commercial product. (Crab, 2005, p. 65)

There were several other electronic organ designs using the vacuum tube technology for sound generation. Some of these along with the other important early electronic music instruments of the period are examined in the following section.

Figure 3.1: The Hammond Electronic Organ (Holmes, 1985, p. 75).

3.3 Other Electronic Instrument Designs of the Early and Mid 20th Century

Pianorad was invented by Hugo Gernsback in 1926. This polyphonic instrument was based on vacuum tube oscillators. It had 25 single LC oscillators for every key of its two-octave keyboard giving the instrument full polyphony. The oscillators produced virtually pure sine tones. Each one of the twenty-five oscillators had its own independent speaker mounted in a large loudspeaker horn on top of the keyboard. The whole ensemble was housed in a housing resembling a harmonium. *Dynaphone* was invented by Rene Bertrand in 1928. The instrument used a multi-vibrator oscillator for sound generation. Dynaphone was a portable, monophonic non-keyboard, dial operated vacuum tube oscillator instrument. The instrument was semi-circular in shape with a diameter of 30 centimeters. It was built by the support and collaboration of Edgar Varese. The first public demonstration of the instrument was a performance of Ernest Fromaigeat's *Variations Caractéristiques* for six Dynophones in 1928. Later the instrument was featured in *Roses de Metal*, a ballet by the composer Arthur Honegger.

Hellertion was invented by B. Helberger and P. Lertes in 1929. The forerunner of Trautonium, the monophonic instrument used a vacuum tube oscillator with feedback and continuous linear controllers. The Hellertion was developed collaboratively by Peter Lertes, an electrical engineer in Leipzig, and Bruno Helberger who was a well-known pianist of his time. The Hellertion was one of the first electronic instruments to use a fingerboard (continuous controller) instead of a keyboard manual. The fingerboard was a flat metal resistance strip covered in leather which when pressed completed a circuit.

Depending on where the strip is pressed, a different resistance in the circuit is created alternations in the voltage that was sent to the oscillator and therefore produced different pitches. The force of the pressure controlled the volume of the output signal. The fingerboard was marked to help the performer find the correct pitch on the strip and had a range of approximately five octaves. The original instrument had just one fingerboard strip which was later increased to four and then on the following models six that were aligned horizontally (in parallel) at the height of a piano keyboard. The four and six strip models allowed four and six voice polyphony when the strips could be played simultaneously with fingers and thumbs.

The Hellertion was occasionally used in concerts as an addition to the piano, the melody being played with one hand on the Hellertion and the accompaniment with the other hand on the piano. A microtonal version of the instrument was produced in 1931; it was tuned to 10 divisions of an octave (Crab, 2005).

Givelet – Coupleaux Organ was invented by J. Givelet and E. Coupleaux in 1930. The instrument used seven hundred vacuum tubes for automated additive synthesis. The oscillators were controlled by paper tape. *Clavier à Lampes* (1927),*Orgue des Ondes* (1929) and*Piano Radio-Electrique* (1929) were earlier instruments by Givelet and Coupleaux. The Givelet combined the principles of the Pianola with those of electronic sound generation so that the instrument could be controlled via a pre-punched tape. This ability to program the production of sound is the forerunner of the use of computers that introducemusical programming.

Pitch, volume, attack / envelope, tremolo and timbre could be controlled by cutting and splicing paper rolls. Like the *Wave Organ*, the five-octave Givelet was polyphonic. The technique of using punched paper 'programs' was not explored until fifteen years later in the 1950's with the RCA Synthesizer.

Givelets and Coupleaux's instrument was designed to be a commercial and cheap replacement for pipe organs and utilize the ability for 'silent recording'. The Givelets were installed in churches around France and at a broadcasting radio station in Paris, but the instrument eventually could not compete with the commercially successful Hammond Organ.

Rangertone Organ was invented by R. Ranger in 1931. It was one of the early tone wheel organs. Similar to the Hammond, the Rangertone had its pitch stability controlled by tuning forks; therefore it was possible to change the temperament by rearranging the tuning of the forks. Timbre was controlled by buttons placed on the right of the keyboard and/or by switching between six different amplifier/speaker combinations, which had tremolo and tonal quality selections. The original version was a huge machine with more than 150 valves. A portable single-keyboard model was built for concert performances.

Welte Licht-Ton Orgel was invented by Edwin Welte in 1936. It was an electromechanical instrument using electro-optical tone generators as photoelectric transducers.

The instrument's sound generation unit consisted of 12 glass disks which were printed with 18 different looped waveforms in concentric rings.

The glass tone wheel-disks were rotated over a series of photoelectric cells, filtering a light beam that controlled the timbre and pitch. The resulting combinations of tones gave three different timbres for all the octave registers for each note on the keyboard.

Parallel Bandpass Vocoder was invented by H. Dudley at the Bell Laboratories in 1939. *The Vocoder* (Voice Operated Recorder) was a composite device consisting of an analyzer and an artificial voice that was synthesized. The analyzer detected energy levels of successive sound samples measured over the entire audio frequency spectrum via a series of narrow band filters. The results could be viewed graphically as a function of frequency against time as it is in a spectrum analyzer. The synthesizer reversed the process by gathering the data from the analyzer and feeding the results to a feedback network of filters that are driven by a noise generator to produce audible sounds.Werner Meyer-Eppler(the director of Phonetics at Bonn University) recognized the relevance of the machine to electronic music after Dudley visited the University in 1948 and used the Vocoder as a basis for his future writings which in turn became the inspiration for the German Electronische Musik movement.

Univox was produced by the Univox Company in 1940. The instrument used vacuum tube sawtooth generators with a diode waveform shaper circuit for sound generation. The Univox keyboard had a unique double contact system under the key which allowed basic control over the note shape. This means striking the key harder caused an impulse generator make a shorter decay thus creating a staccato effect, and striking the key softly gave a long decay of up to two seconds. A vibrato oscillator was provided to modulate the output and to retrigger the vacuum tube to create mandolin type repeated notes. The Univox had a front panel of fifteen switches to further control the timbre of the instrument including three vibrato controls, a modulation control and an overall knee operated volume control. It had an external amplifier and a ten-inch speaker unit. The Univox was noted for the realism in producing string and reed tones such as clarinet and saxophone.

Ondioline was invented by Georges Jenny in 1941. It was a monophonic vacuum tube instrument which consisted of a single oscillator and a small eight-octave touch sensitive keyboard

Table 3.1: Transitional electronic musical instruments of the early 20th century.

Instrument's Name	**Year of Invention**	**Inventor(s)**	**Technical Principles**	**Other**
Electrophon	1921	Jorg Mager	Heterodyne tone generator with a filter	Electronic Instrument.
Neo-Bechstein Grand Piano	1931	W. Nerst	Piano with electromagnetic transducers instead of a sound board	Electroacoustic Instrument.
Emicon	1932	N. Langer and Hahnagyi	Gas discharge tube oscillator	A monophonic vacuum tube oscillator instrument controlled with a standard keyboard. Able to produce tones similar to a cello, saxophone, oboe, trumpet, mandolin, guitar and bagpipe.
Everett Orgatron	1935	F. A. Hoschke and B. Miessner	Amplified vibrating brass reeds combined with electromagnetic pickups	Created under the company titled *Wurlitzer*.
Photona	1935	Ivan Eremeef	Sound generation by photoelectric means via 12 electro optical tone generators	Developed at WCAU Radio in Philadelphia, USA.

Table 3.1 (continued): Transitional electronic musical instruments of the early 20th century.

Syntronic Organ	1935	Ivan Eremeef and L. Stokowski	An electro-optical tone generator based instrument	Able to produce one-hour of continuous variationvia an optically generated tone using films of tone-wheels.
Electrone	1935	John Compton	Electrostatic rotary generators	Based on the design by L. Bourn.
Warbo Formant Organ	1937	Harold Bode and C. Warnke	Partially polyphonic four-voice keyboard instrument with 2 filters and key assigned dynamic envelope wave shaping	The instrument's features were used in the postwar *Melochord*.
Oscillion	1937	W. Swann and W. Danforth	Gas-discharge tube oscillator	French Horn and Bass Clarinet simulation.
Melodium	1938	Harold Bode (developed with the assistance of Oskar Vierling, inventor of the *Grosstonorgel*)	Monophonic instrument with a touch sensitive keyboard	The instrument was used extensively for film music and light music during the 1940s.

The keyboard was switchable through six octaves and tunable via an octave transposer. It was possible to create complex waveforms via a series of filters and the sound could be shaped with the use of a touch wire, affecting the attack with a vertical finger movement or adding glissando or modulation by a horizontal movement. Its keyboard was mounted on springs for vibrato. The overall volume of the machine was controlled by a knee lever.The Ondioline became a popular instrument in Europe, used widely in film and theatre music as well as in light music and cabaret. The instrument was marketed in Germany under the name*Pianoline* and in The Netherlands as the *Orcheline* and made a notable appearance during the Brussels World Fair (1958) when it was played on top of the *Atomium* building. A microtonal version of the instrument was built for the composer Jean-Etienne Marie during the 1960s consisting of a four-octave keyboard which could be tuned to a variety of microtonal systems.

Hanert Electrical Orchestra wasinvented by J. Hanert in 1945. This synthesizer was an instrument for composition and synthesis of electronic music similar to the later RCA Synthesizer and other programmable performance machines. Instead of using punch paper tape like the RCA Synthesizer, the Hanert Synthesizer had a mechanical scanner head that moved over a two-metertable covered byforty centimeters paper cards.

The paper cards held the characteristics of the sound (pitch, duration, timbre and volume)stored in the form of graphite marks that were 'read' by direct electrical contact of the scanning head. The instrument was referred to as an *Apparatus for Automatic Production of Music* (Crab, 2005, p. 76). The sound generating section of the instrument occupied a whole room and consisted of a bank of vacuum tube oscillators, a random frequency generator (to produce white noise spectral characteristics for percussive sounds) and wave shaping circuits. Automations such as speeding up (accelerando) and slowing down the music could be controlled by altering the speed and direction of the scanning head.Hanert's unique system allowed a great deal of flexibility in composition and synthesis, marks could be added to the cards simply by using a pencil and the cards could be arranged in any order allowing variations and multiple combinations in the composition.

Joergensen Clavioline was invented by M. Constant Martin in 1947. The Clavioline (monophonic, three octave keyboard) was designed to be a light portable keyboard.

It was aimed at pop musicians of the time and became one of the most popular electronic instruments during the 1950s. It was a monophonic, portable, battery powered keyboard instrument. The first version of the instrument appeared in 1947 and was originally designed by M. Constant Martin in 1947 at his factory in Versailles, France. The Clavioline consisted of two units: the keyboard with the controllable sound unit and a carrying case box fitted with an amplifier and speaker.

By using an octave transposer switch the single oscillator could be set within a range of five octaves (which becomessix in the Bode version). The keyboard unit had eighteen switches (twenty-two in the Selmer version) for controlling timbre (via a high pass filter and a low pass filter), octave range and attack plus two controls for vibrato speed and intensity. The overall volume was controlled by a knee lever. Martin produced a two voice polyphonic model of the Clavioline in1949 shaped like a small grand but this*duophonic* model never went into production.

The Clavioline made brass and string sounds which were considered very natural at the time and was widely used throughout 1950s and 60s by pop musicians such as the Beatles, Joe Meek's *the Tornadoes* (on *Telstar*) and by experimental the jazz musician Sun Ra (Holmes, 1997, p. 75).*Electronic Sackbut*(voltage controlled synthesizer with pitch, waveform and formant controllers) was invented Hugh Le Caine in 1948.

The keyboard section of the instrument was tailored for rapid execution of scales and arpeggios. As in the keyboard of Ondes Martenot, it was possible to move the keys laterally to produce vibrato. The differing feature was that the extent of pitch change in any direction produced by this lateral pressure may be made as much as an octave either way. Thus, it was possible to create glissandos, smooth slides from one note to another. The pitches that are not on the equal temperament chromatic scale could be produced by the lateral movement. It was possible to produce long slides, gradual glissandos by varying the pitch control placed behind the keyboard. The control of dynamics was achieved by the pressure sensitive keyboard. When the player used gradual pressure, a violin-like attack resulted while thesudden strike to the key resultedin a sharp attack. Ways to alter the timbre of the sound was presented as a device to generate an effect similar to the buzzing produced by a trumpeter was provided. Another mechanism introduced a breath tone, reminiscent of flute. The effects were introduced in small amounts to create a natural expressiveness.

This approach also preventedthe monotonous purity of electronic tone (Chadabe, 1997, p. 13).

Free Music Machine(electronic oscillators and continuous electronic control) was invented by an Australian composer named Percy Grainger in 1948. Grainger created the perspective which he named as *Free Music*. In 1938, he wrote:

> … it seems to be absurd to live in an age of flying and yet not be able to execute tonal glides and curves … Free music demands non-human performance … should pass direct from the imagination of the composer to the ear of the listener by way of delicately controlled musical machines. Too long has music been subject to the limitations of the human hand … that is why I write my Free Music for Theremins, the most perfect tonal instruments I know.(Crab, 2005, p. 79)

Grainger decided to develop his own Free Music Machine, in 1944 he met Burnett Cross who was a scientist and they began a collaboration to build it.

As described by Grainger, the Free Music Machine had to be able to play any pitch within its range, free of the limitation of quarter tones, eight tones etc. the machine had to be able to go from one pitch to the other by a controlled glide as well as a leap.

The machine had to able to perform complex rhythms accurately. The machine had to be workable by the composer himself, without the aid of additional engineers or assistants. The final version of the Free Music Machine was finished in the mid 1950s. It read separate graphs for pitch and volume. Light was passed through the graphs to photocells which controlled the frequency of oscillators.

There were eight oscillators; durations and complex rhythms were realized by calculating relationships between the length of a line in the graph and its speed through the photocell apparatus.

Free Music Machine represents the orchestral scoring approach to electronic instrument design which actually has its roots in the Pianola(The Player Piano – late nineteenth, early twentieth century). Music is treated as information, the perforated paper rolls stored data. The composer Conlon Nancarrow later used this instrument to write new music scores directly in the paper rolls (Focke, 2011).

Finally, the first electric guitar (solid body construction with electromagnetic pickups) –*TheElectric Guitar Prototype* was invented by Les Paul in 1927. Guitar amplification started out due to the guitarists' demand for their solos to be heard through the sound of big bands.

As the electric guitar became popular, its expressive varieties became accepted and a new aesthetic was formed through the new invention. Distorted sound from overdriven amplifiers; the feedback noises soon became a part of the music. This concept of reconceptualization of a former deficiency turned it into a virtue, proving that musical instruments are not always finished with the design process, but they can be redesigned into use by the musicians, as certain aspects that are not considered as part of musical aesthetics during the design can be discovered by the musicians, thus making them necessary.

3.4 Classification of Sound Generation Mechanisms of Electronic Music Instruments:

There are 3 types:

i. Electroacoustic Instruments

ii. Electromechanical Instruments

iii. Electronic Instruments

Electrophone has been the general term that describes all instruments generating sound via electronic means,whether this is achieved in combination with other techniques or by only electronic sound sources. In 1914, Eric von Hornbostel and Curt Sachs published this first systematic classification system for musical instruments known as *The Hornbostel-Sachs classification system* (Davies and Braun, 2002, p. 43).

i. Electroacoustic Instruments

The passive electroacoustic oscillator normally consists only of a vibrating object that is positioned close to one or more electrical coils, in between a light source and a photoelectric cell or in direct contact with a piezoelectric crystal as transducers transforming acoustic vibration into analogous electric current.

Most electroacoustic instruments closely resemble their acoustic ancestors, such as pianos, harmoniums, reed organs, guitars, bowed string instruments. However, it is crucial to clarify the subtle distinction that the amplified result is not produced in an identical manner to the pure acoustic sound, but only one that is parallel to it. Thus the amplified electroacoustic instrument becomes a hybrid; introduces new timbral possibilities as well as new playing techniques, aesthetical values that offer the presentation of new playing styles or musical genres.

Electromagnetic sound transducers; pickups in other words are one category within this classification.Another is the *photoelectric instrument* in which the movements of a vibrating sound source masks a beam of light that is received by a photocell in order the generate a relevant current (that varies according to the resistance value) to produce sound. The sound source is acoustic again, but the means of transducing is achieved by photoelectric circuitry. Rarely used electrostatic transducers are also a part of this category.

This type of transducer consists of a fixed electrode and a movable electrode charged electrostatically in opposite polarity. Motion of the movable electrode changes capacitance between the electrodes, thereby varies the voltage. This type is also known as the condenser transducer. The final type of transducer is based on the piezoelectric ceramic crystal, which is the only one that requires direct physical contact with the vibrating acoustic source, since it is based on the ability of the crystal to generate a relevant voltage when a certain amount of stress is applied to it.

ii. Electromechanical Instruments

Tone-wheel technique for sound generation is listed in this category. The rotating cylinder called the tone wheel contains waveforms inscribed on its rim that affect the value of current in the transducer, which may be electromagnetic, electrostatic or photoelectric. The RCA synthesizer would be listed in this category instead of the electroacoustic category, since the tuning fork oscillators that are excited by the electromagnetic pickups function as stable frequency oscillators, not acoustic oscillators that are controlled by the expressive nuances of the musician. Telharmonium, Hammond electric organ are listed in this category.

iii. Electronic Instruments

In electronic instruments, sound generation is fully electronic, containing no mechanical moving parts. The electronic technology introduced vacuum tube triodes (transistors) which later in the century evolved to semiconductor transistors, integrated circuits and finally VLSI (Very Large Scale Integrated Circuits).

The waveforms range from pure sine tones to random noise generators. The early members of this category such as Theremin and Ondes Martenot use the beat frequency oscillation technique, their circuitries contain vacuum tubes. The Trautonium uses vacuum tubes to generate saw-tooth waves. The dividing technique later used in most of the electronic organs and synthesizers also use the vacuum tube transistors as oscillators of the high octaves. The frequency is then divided by additional circuits to supply the necessary current for lower pitches.Subcategories may apply under this headingsuch as monophonic, partially polyphonic and polyphonic instruments. Any keyboard mechanism can be regarded as a remote control device. Finally, this category contains voltage controlled synthesizers and the MIDI protocol.

4. THREE CONJUNCT VIEWS ON THE EVOLUTION OF EARLY ELECTRONIC MUSIC

If one is to evaluate the musical outcome of the period, it is crucial to examine the three most important instruments of the early 20th century and their repertoire; considering how one invention and its effects caused the following invention and thus musical direction to evolve. The following section explores each of the three selected instruments relating them to performance practices. Pieces composed for the instruments will be examined; the study will give the opportunity to compare these three instruments (and the sound synthesis techniques that they employ) with each other.

4.1. Trautonium

Paul Hindemithcomposed*Langsames Stuck und Rondo for Trautonium* (Slow Piece for Orchestra And Rondo for Trautonium) in 1935.

This composition lays out a variety of Trautonium techniques, displaying the ranges of the sound and articulation that can be achieved by the instrument. Basically, the composition consists of three sections. The first section is slow in tempo; it contains sustained chords (tones) accompanying melodic lines. The timbre of the long tones and their distinctive envelope characteristics are features of Trautonium. The music is polyphonic; one instrument is capable of playing two notes at the same time. The dynamic range of the instrument is another feature displayed in the composition along with the varying timbre of tones. The second section is rhythmic, fast in tempoand demonstrates the instrument's ability to produce staccato notes with fast attacks and long tones of extreme vibrato or glissandos in conjunction with each other. The third section is similar in texture to the first.

The reason for the expressive dynamic character of the Trautonium is the fact that the instrument is played by pressing a metal string above a metal bar. Therefore the selection of pitch and dynamic articulation are combined together in the same instant.

This feature displays the traditional side of the instrument, as the mechanism principle is basically the same in traditional instruments with acoustic sound generators.

While playing a guitar, two hands are on the same string, one controlling primarily dynamics and the other pitch; both interact to shape the resulting sound. Piano has the same principle, although the keyboard is a control mechanism of a larger number of strings, the struck note and determination of dynamics is achieved at the same physical spot and instant. This principle does not apply to the Theremin, where the pitch is selected by the hand in the vertical antenna vicinity, and the dynamics by the other hand in the vicinity of the loop antenna. Therefore the selection of pitch and its dynamic value are arranged by the performer independent of each other.

The timbre of Trautonium is controlled by a set of filters; the neon tube oscillators and the manipulation of rich side bands produce unique subtractive synthesis. Maybe the ‘weakest’ point of Theremin would be considered its timbre. It produces a sine wave fundamental with sidebands; the timbres of tones do not vary in time. In other words, except for some basic controls on the front panel, the performer does not alter the timbre of the constant tone produced. Ondes Martenot uses the same principle for tone generation as Theremin, but it uses a set of (four) speakers, providing timbral combinations to enrich the sound (Khan, 1999).

Theremin cannot produce fast staccato sounds. Ondes Martenot uses a ribbon and a keyboard controller to achieve both long tones (with extreme vibrato or wide glissando) and staccato sounds, but rapid combinations of these two techniques cannot be achieved despite the ability of the keyboard to produce vibrato due to lateral movement; besides, the instrument is monophonic. It is possible to achieve this with the Trautonium, as the second section of this composition points out. The staccato partitions can suddenly evolve into sustained tones with extreme vibrato/glissando characteristics; together with ‘naturally assigned’ dynamics for both.

Using his Trautonium, Oskar Sala composed music for the Alfred Hitchcock’s movie *The Birds*. The bird sounds of the film were also achieved by the instrument.

The piece named *Concertando Rubato from Elektronische Tanzuite* by Oskar Sala, was released in the CD compilation ‘OHM, The Early Gurus of Electronic Music’.

It comprises live Tratonium partitions playing staccato melodies, accompanied by taped rhythmic Trautonium partitions.

4.2. Ondes Martenot

Olivier Messiaen wrote Oraison in 1937. A recording of its performance can be listened to at the compilation release titled *OHM – The Early Gurus of Electronic Music*.

The piece is written for an ensemble of six Ondes Martenots. The performance on the OHM disc is by Ensemble d'Ondes de Montreal. Messiaen later turned *Oraison* into *Praise to the Eternity of Jesus* section in his *Quartet for the End of Time*. The piece has a wide dynamic range, including extremely quiet pianissimo passages. Due to the four speakers offering a range of spectral possibilities, the timbre of the instrument includes a certain amount of variation. This presents a 'natural' sound approach, as opposed to the timbre of the Theremin, which is sometimes referred to as 'monotonous' by listeners. The instrument is capable of producing low frequency tones as demonstrated in this piece, as well as high frequency tones as demonstrated in *Ecuatorial* by Edgar Varese. The piece originally had two fingerboard Theremins in place of two Ondes Martenots. But Messiaen revised the instrumentation since fingerboard Theremins were not available anymore, and their inventor was back in Russia. The two Ondes Martenots in the composition play at the exit of the transition of dense and loud sections tothequiet sections and the high tones of the Ondes Martenots that provide a timbral effect. Since the loud section is suddenly resolved to this silent texture, this high-pitched electronic sound is focused. Beside these high drone tones, the Ondes Martenots play glissandos, even play in unison with the baritone voice at some moments.

In Messiaen's *Trois Petites Liturgies* the Ondes Martenot underlines the melodic lines of the women choir as well as some glissando fills that combine sections. Messiaen's *Turangalila Symphony*contains solo Ondes Martenot partitions in some of its movements. In the second movement titled*Chant d'amour* (Love song), the music is based on an alteration between a fast and loud theme dominated by the trumpets and a soft and gentle theme for the strings and Ondes Martenot. In the sixth movement titled *Jardin du Sommeil d'amour* (Garden of Love's Sleep), the 'love theme' is introduced in full.

The theme is played by the strings and Ondes Martenot. Other orchestral color effects and the birdsong played by the piano accompany the first full representation of the theme by the strings and Ondes Martenot (Mimaroğlu, 1991).

Theremin is still being produced and performed today. Ondes Martenot and Trautonium have not been commercially successful like Theremin. The aim in the creation of Ondes Martenot was to place the instrument in the orchestra along with other traditional instruments which was achieved and maintained by the repertoire created. Although its superior capabilities, Trautonium never became a commercially successful instrument. A number of composers wrote works for the instrument, but the main performer and composer of the instrument was also the creator of Mixturtrautonium, Oskar Sala. He wrote several pieces for the instrument as well as film scores.

4.3. Theremin

Joseph Schillinger wrote *Mouvement électrique et pathétique* in 1932.

The piece demonstrates the frequency range of the instrument, the Theremin starts playing notes of low register combined with rapid glissando melodies, later on moving upwards in register. This piece is a good example for examining a variety of unique articulations that can be achieved with the instrument, primarily related with vibrato and sweeping tones. As we also hear in Schillinger's *Melody*in 1929, the Theremin partitions written consist of continuous tones with vibrato and glissando some of which can be performed on a traditional bowed string instrument such as a cello or violin. But the partitions also include some extreme vibrato and glissando in terms of range and rapidness that are unique to the Theremin.

Clara Rockmore, the first and most well known virtuoso of the instrument plays a classical repertoire. Besides the concerts she has performed, her two released recordings of the Theremin including works by Achron, Rachmaninoff, Stravinsky, Tchaikovsky, Ravel, Fuliehan, Dvorak, Schubert, Chopin, Bach as well as popular tunes by George Gershwin, Avery Robinson, Manuel Ponceand Louis Louiguy. In other words, it is possible to say that Clara Rockmore followed a strictly traditional way to create her career as a thereminist. She even refused to play the Theremin in film scores.

This was due to the ‘spooky’ and ‘weird’ effectsrequested by the Hollywood composers, as she did not want to the instrument to be pushed into this unserious direction, she did not want to be a representative of this approach.

Lydia Kavina on the other hand, performed classical music repertoire like Clara Rockmore. She has released recordings of Debussy’s *Claire De Lune* (arrangedFor Theremin & Piano) and Bach’s *Air on a G String* (arrangedFor Theremin and Orchestra). She has also performed works originally written for Theremin, such as Schillinger’s *Mouvement electriue et pathetique* and *Melody,* Friedrich Wilckens’s *Dance in the Moon* and Isidor Achron’s *Improvisation*.

She played Bohuslav Martinu’s *Fantasia for Theremin, Oboe, Piano and Strings*. This piece explores the outer ranges of the instrument’s pitches and dynamics. Characteristic long melodic lines that both blend and contrast the timbre of the Theremin with oboe and strings is another feature of the piece. At two moments during the fifteen-minute piece, the Theremin plays high to low note slide with a very short portamento time, thus fast glissando. It was another Theremin virtuoso, Lucie Bigelow Rosen who premiered the work at Town Hall in New York. All these works performed by Lydia Kavina (released in the album titled *Music from the Ether*) are originally written for the instrument, yet within a certain range, these pieces apply the traditional practices of melody and harmony, in other words they do not step into the modern direction of music with electronic instruments or means like the other parallel ongoing evolvements in the period. However, although all of them have been composed in the 1990s (except for Percy Grainger’s *Free Music #1* in 1936), there are works that were written for Theremin, using the instrument to create music not based on traditional practices (Wishart, 1996).

Percy Grainger’s *Free Music #1*is one example of this from the period. Grainger’s score consists of drawn lines for each Theremin on a scaled paper. One of the lines represents pitch while the other the dynamics. The piece is for four Theremins, so there are four lines for pitches and four for dynamics, eight in total. The pieces composed later in the century are Lydia Kavina’s *Suite for Theremin and Piano* in1989, Lydia Kavina’s *In Whims of the Wind for Soprano, Theremin and Piano* in 1994, Jorge Antunes’s *Mixolydia for Theremin and Electronic Tape* in 1995 and Vladimir Komarov’s *Voice of the Theremin for Theremin and Electronic Tape* in 1996 (Adlington, 2009).

The second movement of Kavina's *Suite* consists of pitch articulations that can only be achieved on Theremin, a varying range of fast and slow glissandos of wide or narrow pitch intervals. *In Whims of the Wind* displays the close relationship between the voice and Theremin, how one of them can mimic or reinforce the other, in terms of melodic lines as well as other expressive gestures. The two combine at certain moments in the piece; mimicking each other and creating textures, whereas at other parts they move onto opposite directions, demonstrating their distinctive outcomes and therefore reinforcing each other musically, this time from a distance. Antune's *Mixolydia* has been written for Theremin and electronic tape. The piece takes advantage of the Theremin's ability to control a variety of glissandos, large frequency range and accompanies the instrument with electronic tape partitions. Sudden jumps, large and fast leaps in pitch, gradual rises, exploration of the extreme ranges of the pitch of the instrument frequently are some features of the Theremin partition, which also is useful in evaluating the composer's intention in creating a work dedicated to the instrument. The electronic tape partition comprises percussive parts with rich timbral textures, in order to contrast with the continuous tone of the Theremin; these partitions come at the end of long Theremin 'solos', often used to punctuate the end of one section. There are however, electronic partitions of drone characteristics (again with varying timbral color and dynamic articulation) that accompany the Theremin. Vladimir Komarov's work *Voice of the Theremin* incorporates the inventor's voice and a rendition of Glinka's *The Lark*, which Theremin had performed for Lenin to demonstrate the instrument (Young, 2002).

A unique approach is held in this piece, along with the usual varieties of glissandos, the Theremin is used to create accompanying sounds, similar to birdsongs, from a high register range. Lev Termen's voice develops into a rhythmic texture after the middle of the piece, processed by electronic means. Theremin partitions enrich the texture by creating bird effects as well as playing the lead melodic line on top.

Theremin has also been used in the popular music scene. Beach Boys used the instrument in their hit single *Good Vibrations* and *I just wasn't made for these times* in their album *Pet Sounds* released in 1966. Good Vibrationscontain a Theremin partition playing a counter melody to the vocal line during the chorus section of the piece. *I just wasn't made for these times* contains a short solo section of the instrument occurring towards the end of the piece.

Lydia Kavina has collaborated recently with the *Messer Chups*, and experimental band from St. Petersburg, Russia. She is featured on a recording of the band titled *Lo-fi Woman*. The piece comprises conventional melodic lines for Theremin, as well as effect-like extreme glissando partitions. The piece ends with the Theremin, playing the bass partition of the piece using discrete pitches, in other words without sliding between notes.

Theremin is featured on several film scores including: *Spellbound* by Miklos Rozsa, a film by Alfred Hitchcock and *Ed Wood* by Howard Shore, a film directed by Tim Burton. The performances have been achieved by Lydia Kavina. Kavina also performed in Howard Shore's score for the movie *Existenz* as well as for *Spellbound*, *The Day the Earth Stood Still* and *The Lost Weekend*. Considering the score for *Spellbound*, the Theremin is mainly used as a solo instrument to play the main Theme along with the orchestra. Theremin takes this task not only as a solo instrument, but for some sections it accompanies the main theme in the orchestra, behaving like an instrument section within the orchestra instead of a solo instrument. The highly rated and relatively wide ranged vibrato effect of the instrument is used at certain sections to create the so-called 'spooky' effect requested by the Hollywood producers. The timbre of the instrument comprises a sine tone fundamental and sidebands to add some thickness; this contrasting timbre to the rest of the orchestra made up of acoustic instrument created the desired effect for the Hollywood producers.

In *Ed Wood* Theremin solos ornament the main themes. In general, considering the TV series *Dark Shadows*, *Dr. Strangelove* and *Lost in Space*, the Theremin is employed for the creation of this effect, as the plot of these series coincide with it.

It is a fact that as an instrument, Theremin derives its power from the theatrical aspect of its performance, combined with the unique expressiveness of the air control. The instrument was perceived as a scientific curiosity by the public when it was first introduced. After Theremin got accepted as a serious instrument, some listeners commented on the timbre of it as being 'tiring' or 'monotonous'. It is obvious that Theremin's power comes from its uniquely designed control mechanism and the visual aspect of this issue during performance (Demers, 2010). Except for a few controls placed on the front panel of the instrument, the performer cannot alter the timbre.

The pitch and dynamic controls operate independent of each other unlike the operation of traditional acoustic instruments. Therefore it is hard to achieve rapid partitions without sweeping between tones, since the left hand must ‘draw’ an envelope for each note of the rapid partition in order to achieve ‘discrete’ sounds without glissando. This limitation pushes the instrument to perform partitions of melodic lines ornamented with glissandos, therefore its continuous tone becomes an effective representative of the instrument for people, hence some comments to it on being dull. This feature is regarded as a weakness of the instrument by some listeners while according to some it is a powerful aspect since it distinguishes the sound of the instrument from other members of the orchestra that are not capable of producing this pure tone. Hollywood producers, as mentioned above, have used the instrument as a soloist in film scores, associating this ‘unearthliness’ sound with plots including outer space orunnatural horror etc. (Cohen, 2009).

If we consider the place of these instruments today, it is clear that only Theremin has been able to survive. The Ondes Martenot is still being used, but mainly for the performances for Messiaen’s *Turangalila Symphony*. The Trautonium was used in several recordings and film scores, but mainly by its single performer / composer and, we can say the second creator; Oskar Sala. The Ondes Martenot and

Trautonium never gained commercial success like Theremin, which is still being produced and sold by companies today (though there are followers of these instruments too, the German analog modular electronic musical instrument company *Doepfer* has published a schematic for building a modern version of the Trautonium using the brand’s modular electronics). There are a number of virtuoso performers of Theremin all over the world. As these facts suggest, although it has not been an inseparable part of the orchestra, the Theremin still survives and its practice goes on.

5. FFT SOUND SYNTHESIS ENGINE MODEL PROPOSAL

5.1 Objectives

This section works on building a new electronic music instrument model using the technology of today. Certainly software programming has been one of the strongest technologies in the twenty-first century. Software networks in and around our daily life continues to merge and emerge from within a variety of contexts, music programming or in other words synthesizing sound with the aid of programming digital networks has become the foremost technique of designing electronic sound. This section studies the FFT (Fast Fourier Transform) technique. A FFT model is programmed via the object oriented programming language Max MSP. The model can work as a basic Vocoder or perform the convolution of any other two signals other than human voice and a synthetic tone. The convolution of two signals creates a composite sound out of the two sounds used as input sources. The study proceeds to create a 'spectrum freezer' model using complex numbers for calculation in the frequency domain where the FFT data is processed. The spectrum freezer can extract the sonority of any two-second sound sample played through. A spectral morphing block diagram is provided to visualize the idea behind the creation of 'the instrument with no sound'. This instrument is basically a polyphonic FFT synthesizer with convolution. This new electronic instrument design model presents possibilities for encouraging the player to create his/her unique set of sounds for individualization and boost our awareness of sound in any musical instrument or non-musical element which can be any object or ambience that occupies space in or accompanies our daily life.

5.2 The Convolution of Two Signals

FFT operates in the frequency domain. This method of sound processing largely differs from the usual time domain processing perspective. The time domain processing methods are useful when the time parameter of the signal is not adjusted drastically and on purpose. Processes such as equalization, compressing audio, delay networks, reverb algorithms, additive & subtractive synthesis and many more operate in the time domain where real numbers are valid. Frequency domain processing however, operates with complex numbers. This process requires a conversion between the Cartesian coordinates and Polar coordinates (Boulanger, 2000).

Using frequency domain processing to manipulate audio is commonly used in noise reduction & crossover algorithms for phase linear filters, time compression and expansion, spectrum analyzers for audio utilities and the well-known ‘vocoding’ technique which creates a synthetically aided vocal sound.

Looking into the phases of a FFT process, it is crucial to realize that the sound generated, even though the source can be acoustic; is digital since it is digitally re-synthesized. According to the mathematician Fourier who discovered the *Fourier Series*; any signal (or sound signal in particular) can be re-synthesized by adding necessary amounts of pure tones (sine waves) with the correct frequency, amplitude and phase values. When Fourier’s theorem is applied to sound signals, as it is relevant in the additive synthesis technique, theoretically it is possible to synthesize any sound by adding the correct amount of pure tones together. One would think that if this is the case, then it should be possible to synthesize any acoustic instrument from computer and this brings us to the fact that the theory cannot fully be applied in practical terms since these sounds may require calculations for millions of pure tones with varying amplitudes, phases and frequencies over time. This means that even a small moment of acoustic sound information can take more processing power than the computers of today can handle (Roads, 1995). The continuous analysis phase as well as the real time re-synthesizing is not implemented in any of the digital instruments of today; so although possible in theory, acoustic instruments cannot be fully modeled by digital means. Even with the complete modeling done, a proper controller interface would have to be developed for each instrument to mimic the performance issues.

FFT cannot be used for complete acoustic modeling yet still it is a powerful technique for sound synthesis. Considering the realization terms of the Fourier Transform theory in engineering, it is possible to adjust some of the parameters to get the highest resolution from a Fourier Transform. One of these would be time, taking a ‘slice’ of audio and analyzing that sample with enough frequency bins and crossovers to prevent the ‘smearing’ effect of the spectrum results in an accurately analyzed sample that is ready for further processing in the frequency domain, even though the process is not continuous. As the speed of playback goes to the smallest extreme it is possible to stretch the duration of this ‘slice’ of audio to the infinity. This method provides us a way to extract the timbre of an instant from the FFT transform, as it will be the essential principle in the design of ‘the instrument with no sound’. This idea will be fully explored and applied in the following sections, but now let’s take a look at the convolution of two signals.

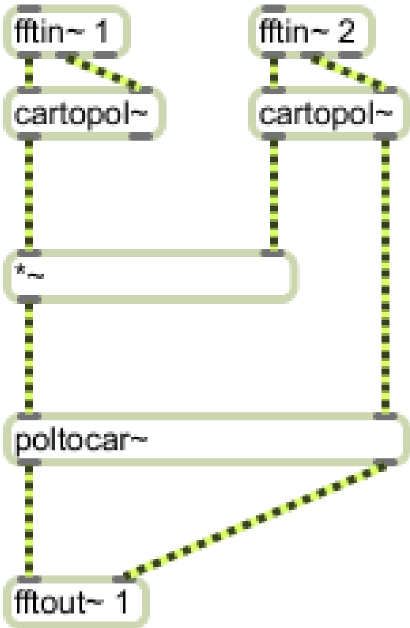

Figure 5.1: The convolution of two signals.

Figure 5.1 shows a Max MSP patch that realizes the convolution of two signals. This is the core of the convolution algorithm. The fftin~1 and fftin~2 objects take the input signals and convert them into complex numbers as coordinates in the Cartesian coordinates system with real and imaginary parts. The default windowing function is ‘Hanning’. Other windowing functions such as Square, Triangle, Hamming and Blackman (with overlap of four or more bins) are available if specified as an argument in the object (Cipriani, 2010).

The second stage converts the Cartesian coordinates into Polar coordinates in order to achieve frequency multiplication. During this process, the frequency bands that exist in both of the sounds reinforce each other while the bands that are not in common are attenuated.

Only the amplitude data of the samples are multiplied and the phase data is gathered from the second sample. This allows the combination of the harmonic content of the two sounds be 'played' by the spectral envelope of the second sound. Naturally, the success of this type of effect depends heavily on the choice of the two sounds used.

The third phase converts the Polar coordinates into Cartesian data, the fftout~1 object performs a reverse Fourier Transform in order to convert the frequency domain data of the composite sound into time domain data that will be fed to the output of the parent patch.

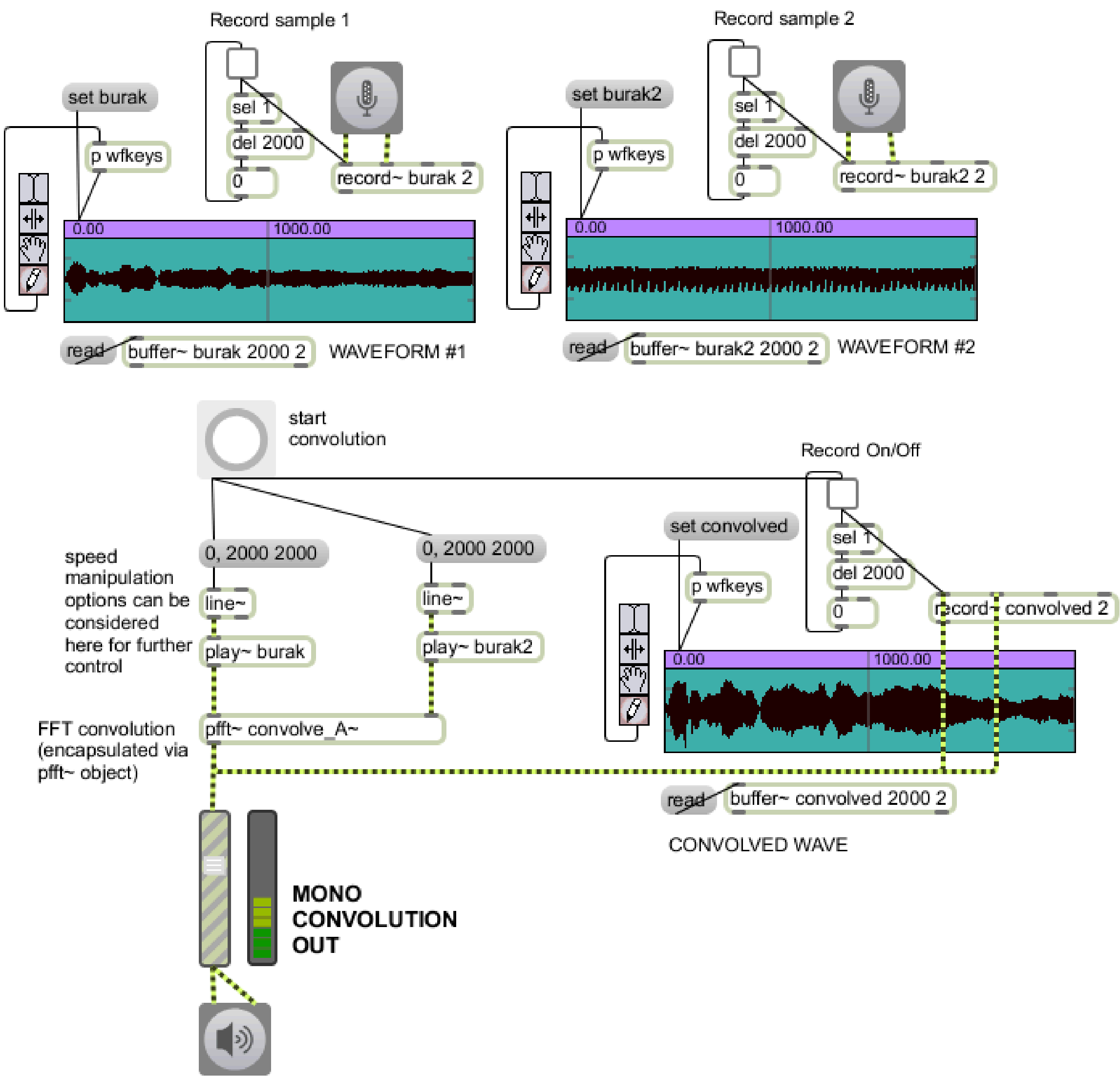

Figure 5.2:A mono convolution algorithm realized via Max MSP.

Figure 5.2 presents a mono convolution algorithm. The upper side of the patch has two sections that are identical. These sections are for capturing live audio via an input on the computer.

This can be a microphone signal through a preamplifier and an AD converter or a line level input through a DI box and an AD converter. The toggle switches on the top of each waveform display and start recording into the buffer, which is the RAM of the computer. The sample recorded into the buffer lasts for two seconds as implied as an argument in the buffer object. The sample is stereo and displayed in the waveform object. Some tools to take selections, zoom in and out of the display and manually alter sample values are provided, but are not the focus on this patch yet. After recording the two sounds that are to be convolved, the user clicks the ‘start convolution’ button in order to apply convolution and record the composite sound data into a new buffer zone in the RAM.

The pfft~ object encapsulates the basic convolution algorithm shown in Figure 5.1. The line objects drive the play-head to read through the to samples in real time (it should be noted that this part of the patch can have variations due to the multiple playback speed envelopes that can be applied. For the sake of focusing on the convolution aspect, I will not go further into exploring this control that can serve as a musical element). The pfft~ object performs FFT as well as the convolution of these sounds. The re-synthesized and convolved composite sound is processed back into the time domain by a reverse FFT. This signal is sent to the final fader and meter where its level can be adjusted appropriately according to the data displayed on the meter. The EZDAC (easy DAC) object is a stereo DA converter. So the mono signal is fed to the stereo outputs and thus occur as sound from our speakers while it is possible to view the waveform of the composite sound in the third waveform display object.

The stereo convolution patch shown in Figure 5.3 applies the same principles as the mono algorithm. The two stereo signals recorded are convolved into one composite sound while the timing of the frequency bins are not manipulated. Since the convolution is to be applied in stereo this time, each of the pfft~ convolve_A~ objects encapsulate a basic convolution algorithm. The left channels of the audios’ (sesA and sesB) are convolved with each other and the same principle is applied for the right channels. Like it is the case with the previous mono implementation, the level of the convolved audio can be adjusted by the aid of the stereo fader and meters. The waveform of the composite sound can be viewed in the third waveform display object.

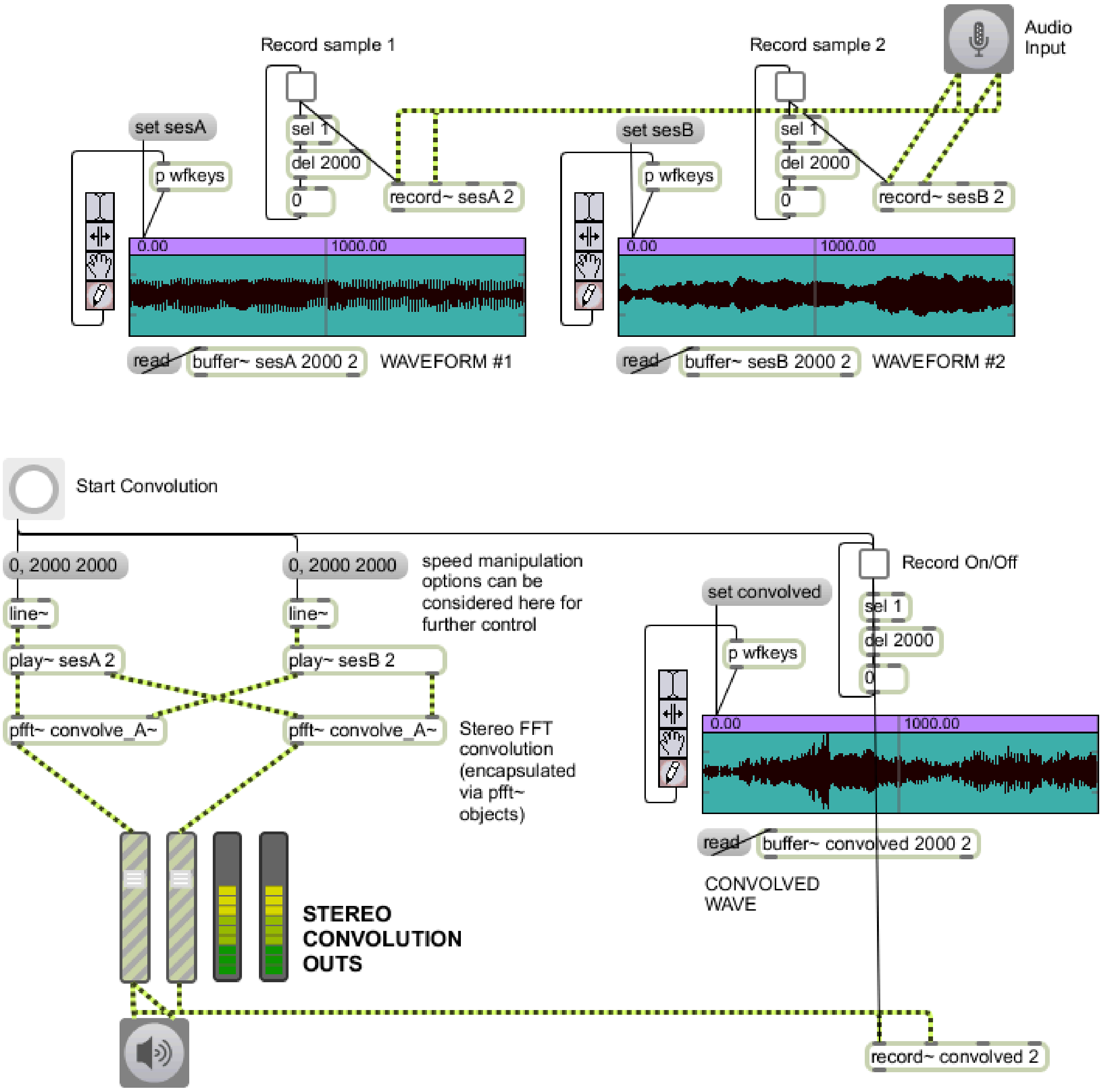

Figure 5.3:A stereo convolution algorithm realized via Max MSP.

5.3 A Specific State of Convolution: The Vocoder

The Vocoder has been developed at the Bell Labs, the research division of AT&T. It was never the intention to create a musical instrument; rather the aim was to reduce the cost of long distance calls when the Vocoder was being invented. Wendy Carlos used the instrument on the soundtrack of*A Clockwork Orange*in 1971. Carlos used the Vocoder to play an interpretation of the fourth movement of Beethoven's Ninth Symphony, thus introduced the instrument to the public. The German electronic music band Kraftwerk used the instrument in their works. The Vocoder was not widely used since it was an expensive technology and could only be employed in indoor music studio environments (Miller, 2008).

Digital technology allows us to re-invent certain procedures such as the Vocoder, we can re-create the Vocoder since it is a specific state of the convolution algorithm studied in the previous section.

The classic Vocoder uses two inputs to operate. One of the inputs is the human speech, singing or any signal that carries the musical information. The second inlet is the synthesized tone. This tone is generated by a tube generator and can be played via a traditional keyboard polyphonically. The convolution process creates a composite vocal sound with a pitched and synthetic voice since the synthesized tone's spectral envelope 'plays' the voice signal's harmonic content.

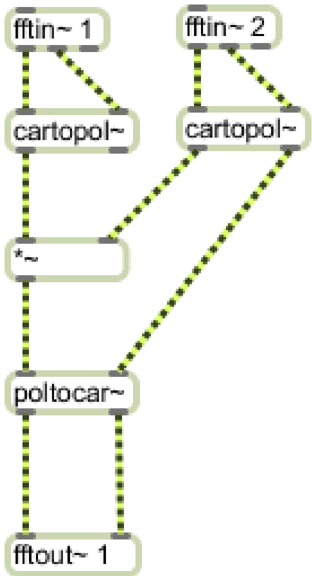

Figure 5.4:A convolution engine. The second signal's phase draws the spectral envelope of the composite sound.

Figure 5.4 displays the convolution algorithm used for this digital Vocoder instrument. The input one at fftin~1 is the human voice. Notice that the phase input is driven by the second input at fftin~2 which is the synthesized tone, therefore the phase of human voice is ignored in the Vocoder algorithm.

Figure 5.5 displays the digitally implemented Vocoder with two-voice polyphony. The pfft~ basic_convolve object encapsulates the Fourier Transform and reverse transform in Figure 5.4. For input one, microphone input is provided as well as the hard disk sample player.Input two is a sawtooth generator for each voice of the polyphony. The input signal is also fed to a control phase that triggers the noise generator.

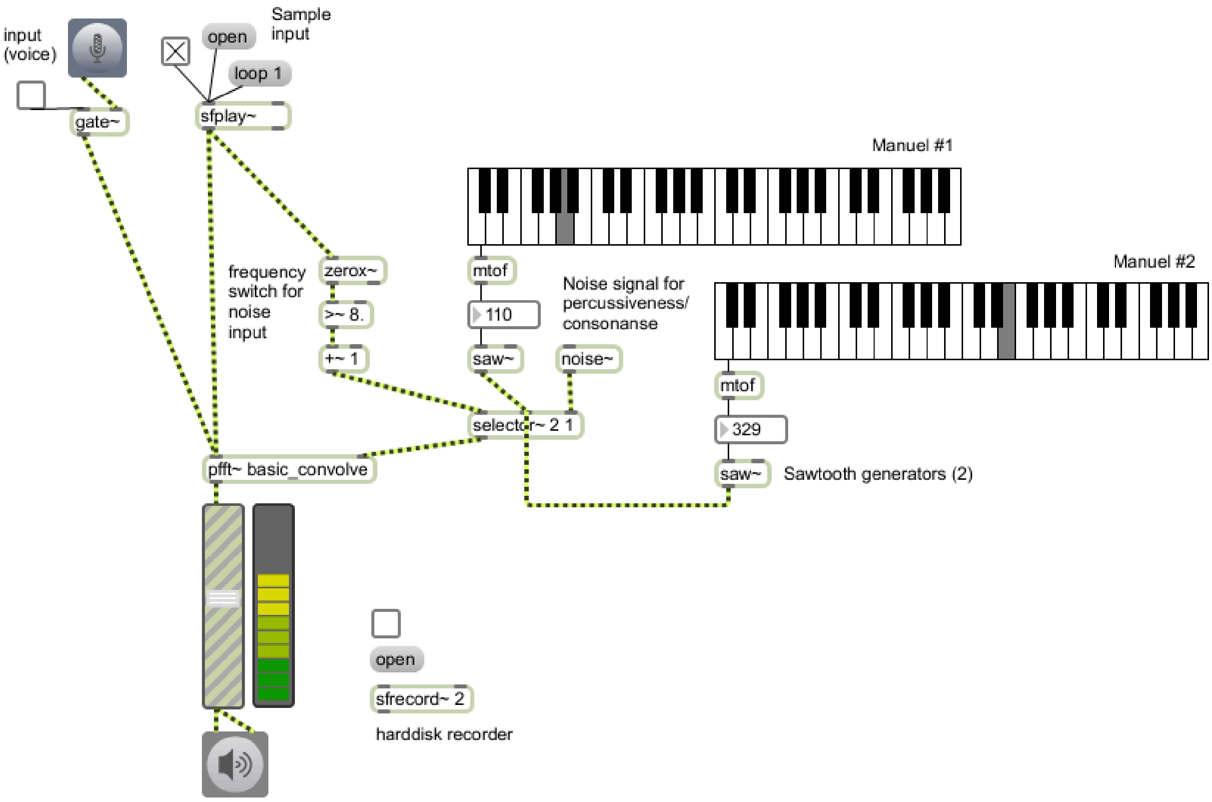

Figure 5.5:A mono, 2-voice polyphonic digital Vocoder.

The zerox~ object counts the number of zero-crossings in the signal, so that when this number exceeds eight, the noise generator is activated. This ‘frequency switch’ allows the noise signal to pass to the output during the consonants in the signal as this helps to conserve the percussive values of the vocal / speech input.

The sawtooth tones and human voice input are convolved together in the encapsulated convolution algorithm. The gain fader and meter allow us to adjust dynamics properly. A hard disk recorder is employed to record performances.

5.4 Freezing the Spectrum using FFT Techniques

The Vocoder certainly presents a very prolific area of the FFT technique. Another strong feature of the FFT technique is the time compression expansion algorithms. By reducing the playback speed to zero it is possible to achieve an infinite time expansion that freezes the spectral content and stretches it to an endless envelope, which we will later shape in following section with the instrument with no sound.

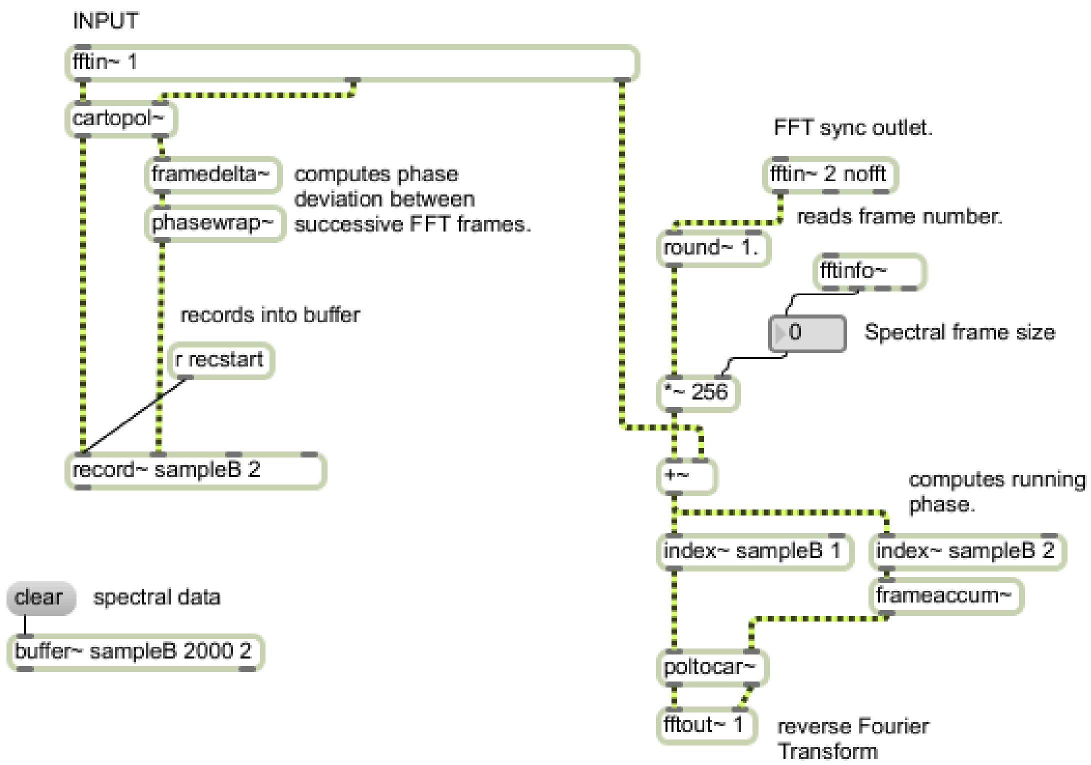

Figure 5.6:A FFT Engine. The spectral data is recorded into a 2-second buffer.

Figure 5.6 displays the FFT engine algorithm used in this section's spectrum freezer patch. The real and imaginary coordinates are converted to amplitude and phase information via the cartopol~ object. The output data is recorded into the RAM via a buffer object. fftin~ 2 nofft object does not perform FFT synthesis, however it calculates analysis information for the transform. This section reads the frame number and the spectral frame size to compute the running phase of the convolved signal (Sack, 2003).

Figure 5.7 shows the full spectrum freezer patch. The pfft~ fftengine~ object encapsulates the FFT procedure described above. This patch has two inputs provided; it can play files from disk.

It can either play two second files (or any two seconds within an audio file) or a two second sample can be recorded from the microphone input.

After recording the sample, using the toggle button with the label ‘3’, the contents of the buffer has to be played back to the pfft~ patch in order to record into buffer again, this time as spectral data.

The phasor~ at the right side of the patch is used as the play-head of the file playback mechanism. The value ‘1’ provides forward playback with the actual speed. Reducing the speed will expand the time while increasing will compress it. Therefore when the value ‘0’ is applied to the phasor~ input we can ‘freeze’ the playback process at a certain ‘slice’ of audio where the re-synthesized harmonic content of the sound file is available for further manipulation since it is in the frequency domain now. This infinitely stretched slice can be perceived as timbre extracted from one instant of the sound sample. At this state we hear the frozen sonority as a drone sound with no envelope scaling variations in dynamics.

The waveform display object visualizes the data stored in buffer and allows us to select certain moments along the time axis. By changing the audio slices, variations of the timbre that is extracted from the sample can be achieved. The output of the playback phasor~ section drives the fftin~ 2 input which is the analyzed spectral data that computes running phase.

With this patch, it is possible to freeze an instant in audio and extract the timbre of this audio slice in a fully re-synthesized manner that makes it ready for further processes in the frequency domain. The output section allows us to mix the frozen spectrum slice with the unprocessed original sound for further exploration.

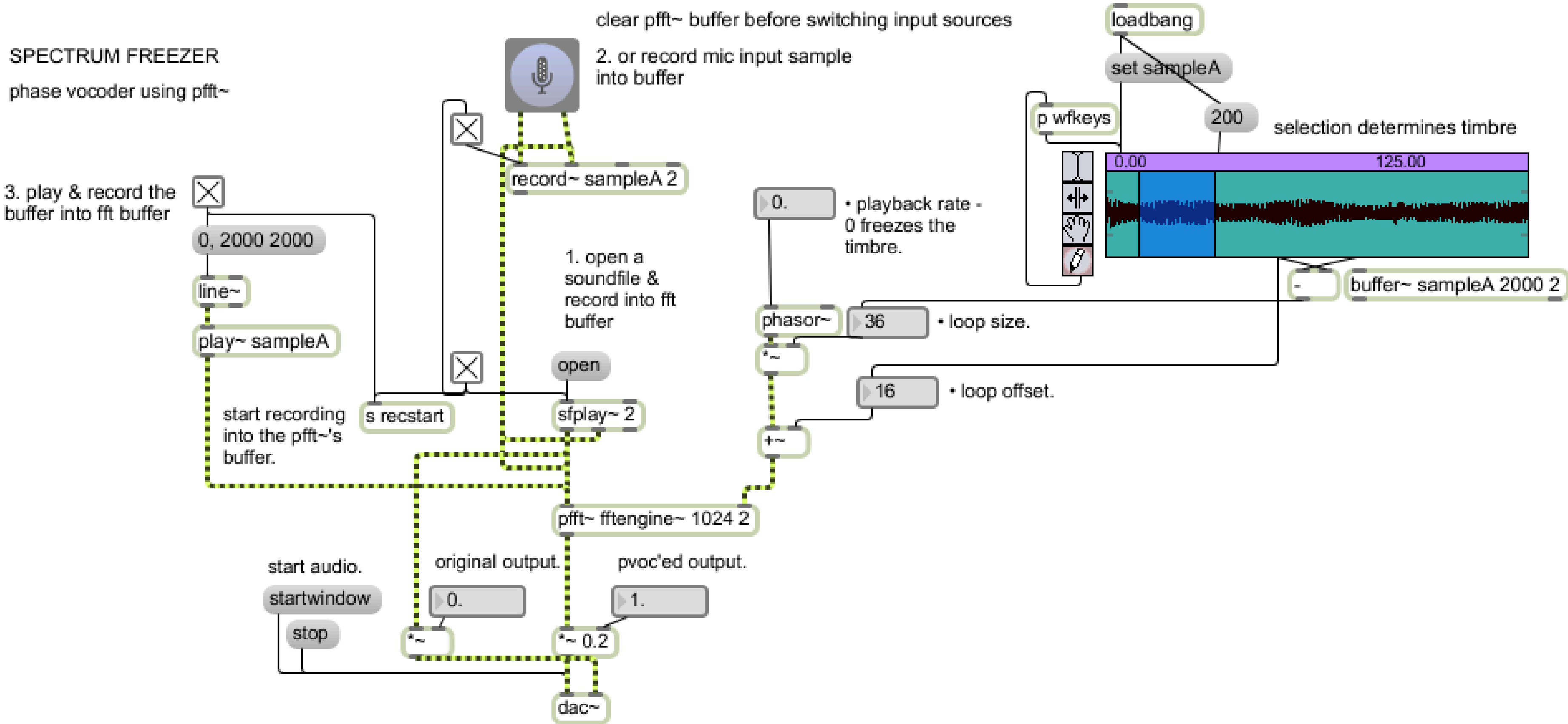

Figure 5.7: The Spectrum Freezer.

5.5 The Instrument with No Sound

The Instrument with No Sound explores the spectrum freezing algorithm as a sound generation module of a polyphonic FFT synthesizer with convolution. The following sections investigate the building stages such as the block diagram, the encapsulated FFT sound generator and the polyphonic FFT synthesizer with convolution, as well as the idea behind the instrument.

5.5.1 The perspective

Considering the role of electronic music instruments on today's music production, it is possible to say that there are a wide variety of ranges that these electronic sounds are used. In the same level of diversity, the principles of the instruments vary in these uses though some principles still remain in common. It is crucial to note that alongside this 'mainstream' set of instruments, there are some which have been accepted but are not widely used due to factors such as economics or the requirement of technical knowledge in order to 'play' them. Certainly digital software synthesizers hold a strong position due to their advantages such as storage, maintenance, mobility and stability (Wilson, 2002). The down side is that since these instruments do not generate sound like their analog ancestors, they tend to use algorithms that imitate the analog sound generation of the vacuum tube technology in order to produce satisfying sonic results. This process of imitation faces certain obstacles such as the limitations of CPU power that results in the loss of harmonic content resolution. Another challenge in programming is in the analysis and application of the way the analog instruments 'behave' into the digital software. Since the circuit parts are made up of actual elements that hold and transmit electrical voltage and current, they somehow resemble acoustics instruments in the fact that both are made up of natural vibrating material whereas the digitized circuits of the digital technology isolate these stages in the circuitry into the binary system in order to achieve absolute stable results. This prevents the digital circuitry from being unpredictable in certain ways; a quality that we may refer as to being like a living organism, breathing and changing in an everlasting way. The control mechanism is another issue that the digital instruments, in our case now the digital software synthesizer, has to consider. Software synthesizers of today use MIDI keyboard controllers that transmit pitch, the velocity information and the after touch. These may seem well enough but it should be noted that even with the addition of the pitch bend wheel expressive control is limited in the instrument. Ribbon controllers such as the *R2M MIDI Ribbon Controller* by the German company Doepfer Musikelektronikor the *VMeter USB MIDI Controller Touch Strip* are often employed for glissando performances. Obviously adding portamento to a regular synthesizer keyboard controller is another option but does not introduce as much control as the dedicated ribbon keyboard. Some of the important representatives of sequencing approach to MIDI controller design are the *Launch Pad* by *Novation* and the *Trigger Finger* by *M-Audio*. The former uses a LED colored matrix for arrangement and controller data control while the latter uses a pressure sensitive matrix and separate controllers as fader and knobs optimized for the same tasks. The latter can also be played as a percussive instrument due to its versatile design.

Reactable is one of the important electronic music instruments of the twenty-first century. This instrument belongs to the side category that stands beside the vast number of hardware or software (and analog or digital) synthesizers. The instrument is structured as a table equipped with modular devices that the user can interact with through putting them on the table, turning them around and placing them on the table while managing the interaction between the modules. Even though this workflow and design is very influential, this kind of instrument can be classified as a ‘sequencing’ interface. Reactable allows us to program certain oscillators and/or loop players driven by sequencing control interfaces. It is also possible to feed these partitions into time domain processing modules such as depth processors or spectral drives for further sonic manipulation. There are global objects provided which can alter the tonality and tempo of all the objects on the table. The most influential feature of this instrument is that is reveals certain possibilities while interacting with various modules on the table; a sequencer that is meant to control a pure tone generator can suddenly be placed near and thus related to a time domain effect unit which allows the sequencer to manipulate a certain parameter of the new unit as well, according to the notation data that it applies to the pure tone. This feature claims the Reactable a ‘live dynamic network’ which can reveal possibilities of a system that was not intended by the user in the first place, thus triggers creativity. Apart from this innovative approach Reactablefunctions just like a sound studio from the

1940s(Sexton, 2007). Along the modules we have software signal generators (vacuum tube generators) or loop players (magnetic tape players) and sequencing modules to play them (this could refer to either the performances or the recordings of the devices as well as punched paper devices). At the output we can apply delays, reverbs and overdrive effects, just like it would happen in a traditional sound studio. The futuristic interface of the instrument is certainly an important issue since it visualizes the signal network and inspires the user so that once achieved, the whole network can be viewed or listened to, performed or re-thought for variations. The physical versions of the Reactable ranges from hardware table version to software multi-touch iPhone and iPad applications (Rogers, 2002).

The *Continuum* by *Haken Audio* remains one of the top instruments of twenty-first century. This instrument employs a special hardware controller that comprises a traditional keyboard, horizontal and vertical ribbon controllers all placed in a single housing. The instrument uses the *EaganMatrix* system for sound generation, which is a digital synthesizer inspired by the classic synthesizers such as the *ARP 2500* or the *EMS Synthi 100*. Due to its highly sophisticated design that takes advantage of modern technology, the *Continuum* is a highly expressive instrument. Yet this electronic music instrument remains in the side category of instruments that accompany the mainstream synthesizer movement; the reason being its inaccessibility due to its high cost. The Continuum controller that is presented without the sound generation module offers a promising ground for sound synthesis experiments yet the economic considerations remain similar.

The conventional approach applies in all of the well-known high-end industry standard digital audio workstations. *Avid Pro Tools HD* and *Ableton Live* both employ multi-track audio recording, editing and mixing options as well as MIDI tracks and selection of MIDI synthesizer instruments along with the opportunity to load in third party VST instruments and effects. The sound studio techniques and the electronic music instruments that these softwares employ still apply the same procedures of sound generation and manipulation that has been researched and introduced since the invention of the Telharmoniumin the early twentieth century. Audio editing techniques are more efficient and much easier to employ compared to the magnetic tape period. This is a stage where we can clearly benefit from the absoluteness of the digital system that describes a discrete digital audio sample as the smallest unit in time (which equals to approximately 22.7 microseconds in 44.1 kHz sampling rate). As for the sound synthesis techniques, alongside Additive and Subtractive synthesis, Granular synthesis has been introduced. Granular synthesis takes an audio file or an audio input and gathers ‘grains’ from this input. The user specifies the length and pitch variations of the grains as well as the amplitude variations and the playback speed. This procedure may or may not employ the FFT technique although most of its professional applications do so (Puckette, 2007). The idea behind building ‘the instrument with no sound’ was to explore the gap between the musician and the engineer point of views. The technological and in general informational dynamics of the twenty-first century we live in are evolving with a very fast pace. Everyday we encounter new applications that were only possible in theory put in use since the engineering terms that allow us to realize them are actually available now. This advance in technology has contributed to electronic instrument design, however the most common electronic instrument of the day, the modern synthesizer has become a challenge for musicians to operate. The modern software synthesizer approach presents presets and parameters to the musician to work with. The presets hold a large number of pre-adjusted sounds that either imitate acoustic instruments or instrument sections and are named accordingly, or voice electronic sounds that can play any register or expressive quality within the sonic possibilities of an orchestra. The parameters are there for further adjusting the presets, so that the user can individualize the sounds or make them appropriate for the partition that is being worked on. The problem this introduces is that the chance of instant interaction is further buried into these multi-layered procedures. (Bolter, 2003). This decreases the chance of expressive connection with the instrument too.A musical idea in mind needs to be transformed into a voice instantly, without shifting the focus to other certain technical procedures of the instrument that is being used. The gap between the engineer and the musician perspectives introduces the lack of instant interaction in the modern day digital synthesizer. Each of the presets on the instrument could be viewed as a basis for further individualization that the user has to go through, but at this stage we face another obstacle. Each digital instrument has its own interface and set of parameters that differ in effect, so one has to master every parameter interaction in order to work on the timbre of the instrument. This process puts a lot of time and constructive stages between the player and the music that is to be performed (Noble, 2009).

The Instrument with No Sound works to propose a new approach considering the issues discussed above. Instant user interaction is considered one of the crucial aspects of this instrument design, as well as providing a timbre with acoustic properties when desired. Considering the city structure, development of the society, and all kinds of digital media that surround us; we are exposed to a vast amount of continuous acoustic data through our daily life. However, due to this data overload, only a few of this sonic information is interpreted and perceived in our consciousness (Licht, 2007). The procedure introduced with the new instrument seeks to reinforce our perception of everyday sound world. The sources of these sounds can be anything: a string of an acoustic guitar ringing, an ambience city soundscape, the sound of a paper being crumpled up etc. Every sound that we encounter in our daily life contains acoustic data that we perceive and interpret in order to extract physical qualities and metaphoric resemblances of the phenomena. The Instrument with no Sound leads the user to study and analyze this acoustic data surrounding us; therefore it triggers creativity through our physical medium by increasing awareness to sonic information.

Since the user interacts with the sounds around him, individualization of the instrument is achieved at the same stage with the instant interaction. As the name suggests, this instrument has no sound when the player first starts using it. Only after a brief stage of individualization the user is allowed to experience the sound of the instrument. These stages of individualization and interaction coincide and occur instantly to leave the player alone with the sonority and the musical idea. The user is allowed to record his experiences to build his individual preset library. Since the timbre of the instrument is determined by the sound defined by the user, it is possible to use acoustic sound sources as well as electronic ones.

5.5.2 Overview of the technical principles

This section discusses the stages of the new instrument model providing a block diagram of the design in Figure 5.8. The block diagram displays the phases of design and certain possibilities that each of these stages can offer.The timbre of the instrument is derived from the audio input stage which can accept two seconds of high resolution (24 bits bit depth and 44.1 kHz sample rate) audio either as live input from a microphone or a line level feed.

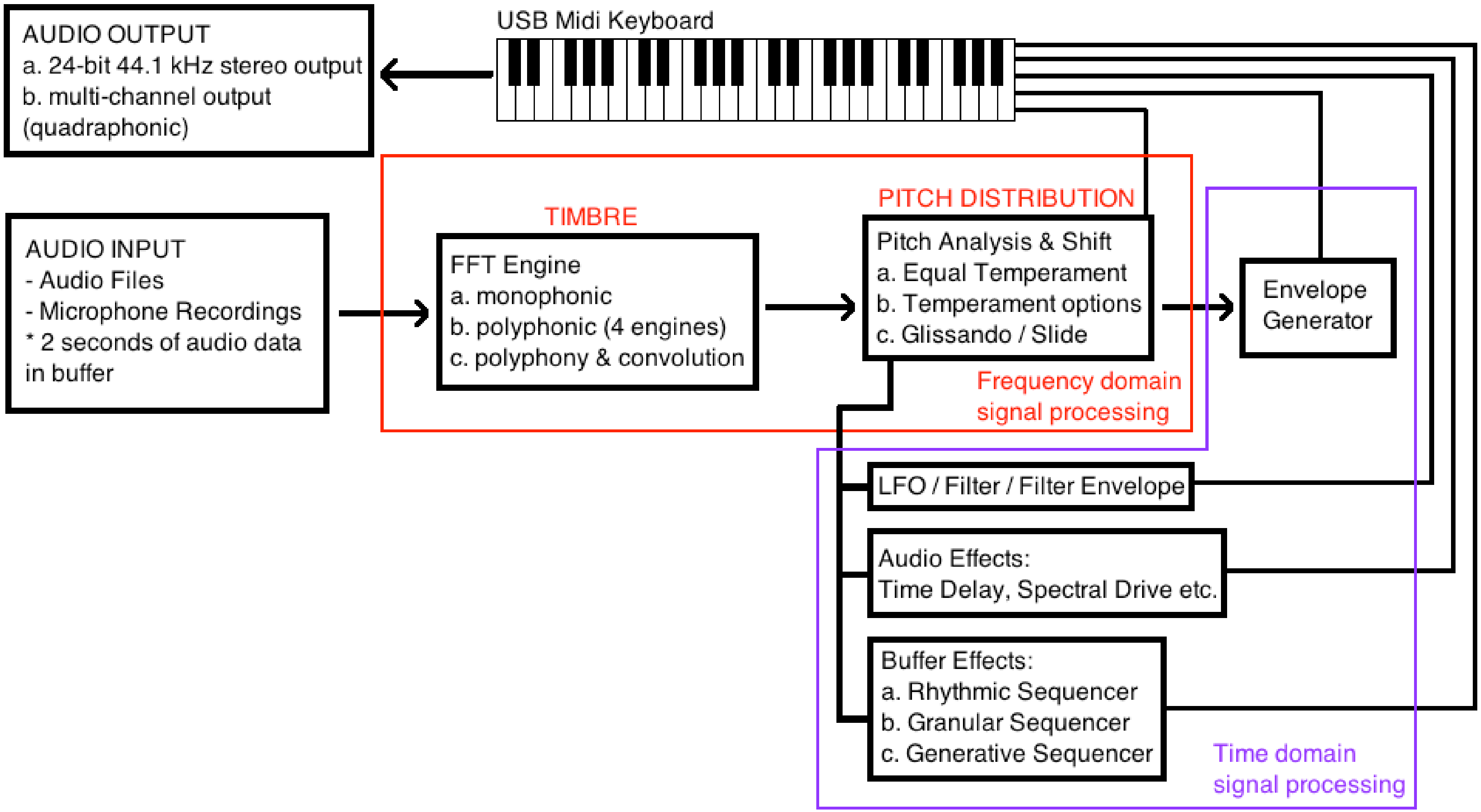

Figure 5.8: Block Diagram of the Spectral Morphing Algorithm.

The level in can be replaced by an audio file sample to be read from hard disk directly. The timbre extraction algorithm works on the FFT basis, it can be monophonic, polyphonic (4 voices or more) and polyphony combined with convolution. Each of these engines is realized as working models and will be investigated in the following sections thoroughly.

A traditional keyboard controller is provided as the control surface of the instrument. The pitch of each tone is determined by the pitch distribution stage. The pitch of the incoming re-synthesized audio from the FFT engine is analyzed and transposed accordingly along the keys of the keyboard. At this stage, modes of equal temperament or other microtonal scales are available as well as achieving portamento or glissandos. The following sections provide and apply the equation and algorithm for the equal temperament scale. However applying the microtonal scales or modes to this algorithm is just a matter of applying the relevant tonal scale information in the equation. Playing glissandos directly is only possible via a ribbon controller as discussed in the previous section, so a version of the instrument with portamento will be presented in the following section.

So far, we have discussed the frequency domain processing stages of the instrument which constructs the core of this design. After the timbre extraction and pitch distribution, an envelope generator applies an ADSR envelope to the sound that has been transformed back into the time domain. The envelope generator provides us a graphical interface where we can mimic the dynamic qualities of acoustic instruments or sounds as well as create unique dynamic variations. The length of the envelope is also determined in this phase.

The re-synthesized, pitched and enveloped sound then proceeds to further time domain processing (incorporation of these stages are optional) for depth and/or spectral data processing. Creative time domain processes such as rhythm sequencers may apply at this stage. The sound then proceeds to the DA converter and is outputted as stereo high-resolution continuous audio. At this stage a multi-channel version of the instrument may be applied, such as a quadraphonic system for spatial effects. For the sake of simplicity, the models presented in the following chapter use the stereo output. This design can serve to create unique sound textures as well as to mimic acoustic instruments. However, the success in sonority is highly dependent on the sound samples performed. Therefore, the creativity and perception of the user is in charge. The stronger reflection of the algorithm comes in when the user searches for new dimensions in sound design. As for acoustic instruments, the instrument transforms the acoustic experience into a sort of hybrid electro-acoustic experience.

5.5.3 Realization of the FFT sound generation module via Max MSP

This section studies and proposes a model for the FFT Synthesis Engine that is to be encapsulated for the polyphonic control interface, the parent patch.

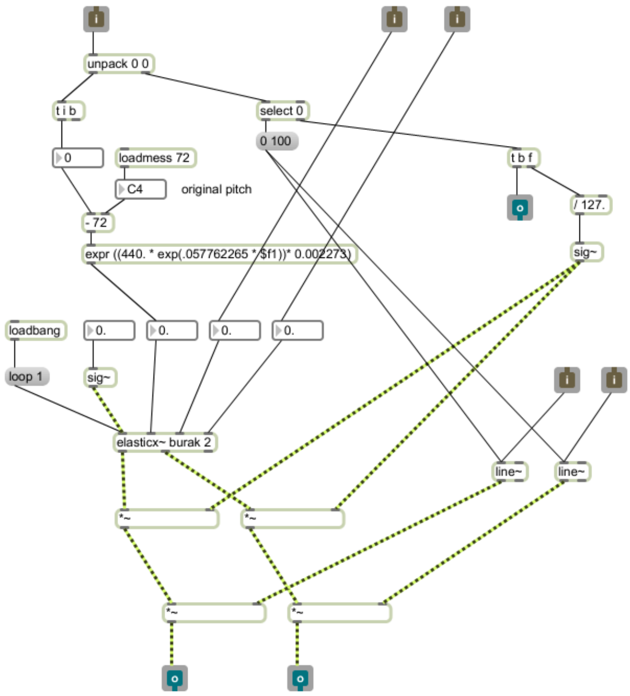

Figure 5.9:FFT Engine (first voice), Spectral Freezer algorithm encapsulated, envelope generator inputs to parent patch.

Figure 5.9 displays the first voice of the polyphony. This patch encapsulates the spectral freezing algorithm and is encapsulated as the first sound engine of the instrument.

First input splits the midi note and midi velocity data into two signals. The midi note of the signal is biased originally to the note C4; by transposing certain intervals, this note in each of the sound engines can be arranged for each voice of the parent patch in order to reinforce certain musical modes. The other equal temperament mode would leave the transposition ‘unbiased’ to a certain tonal center, in this case all of the note parameters would be kept same and be altered in the same amount of intervals to get the desired pitch register (Miranda, 2002). The application of this model will be studied beginning with the latter method since it is easier to implement than the former which creates a more complex polyphonic system. The output of the tuning parameter feeds into an equation that converts the scaled midi note values to the ranges of the pitch input of the encapsulated spectrum freezer. Note that the playback speed input is kept at zero to freeze the spectrum portion that is being read. The other two inputs of the encapsulation are the loop selection minimum and maximum times in milliseconds. These inputs are fed by the waveform display object in the parent controller patch. The stereo outputs of the encapsulated spectrum freezer go through two stages of amplitude scaling. In the first phase the amplitude is scaled by the note on velocity value, then the second phase modulates the dynamics over time as it transmits scaled amplitude values of the generated envelope. The first input's second unpacked data is the velocity of the signal thatscales output in the first phase. The velocity zero is set to a 100-millisecond release to avoid clicks when a note off occurs. The rest of the velocity scale triggers the main envelope dialog in the parent patch form the third outlet object.

The encapsulation has five inputs and three outputs. The first input takes the midi signal in, second and third selects the timbre portion from the buffer, five and six input the envelope scaling. The first and the second outputs are the stereo outs of the spectral freezing algorithm. The thirdoutput triggers the envelope in the main patcher.

Figure 5.10 displays the second, third and fourth voices of the first model of the instrument with four-voice polyphony. In this encapsulation the envelope generator is placed within. This allows the user to determine envelope variations for each voice. For the sake of simplicity during the introduction phase of the instrument, all four voices will have the same envelope that is to be dumped to these voices via the controller parent patch. Input four now sets the duration of the envelope and input five is being used to dump envelopes to this encapsulation. There are two outputs which are the stereo outs of the spectral freezing algorithm. In the main controller patch these two encapsulation models, first voice and the second, third, fourth voices, will be used to form the polyphony of the instrument.

Figure 5.11 displays the first voice encapsulation of the instrument with additional portamento time and master tune inputs. A number box dialog in the main controller patch controls the duration of the ramping up and down pitch slides in milliseconds.

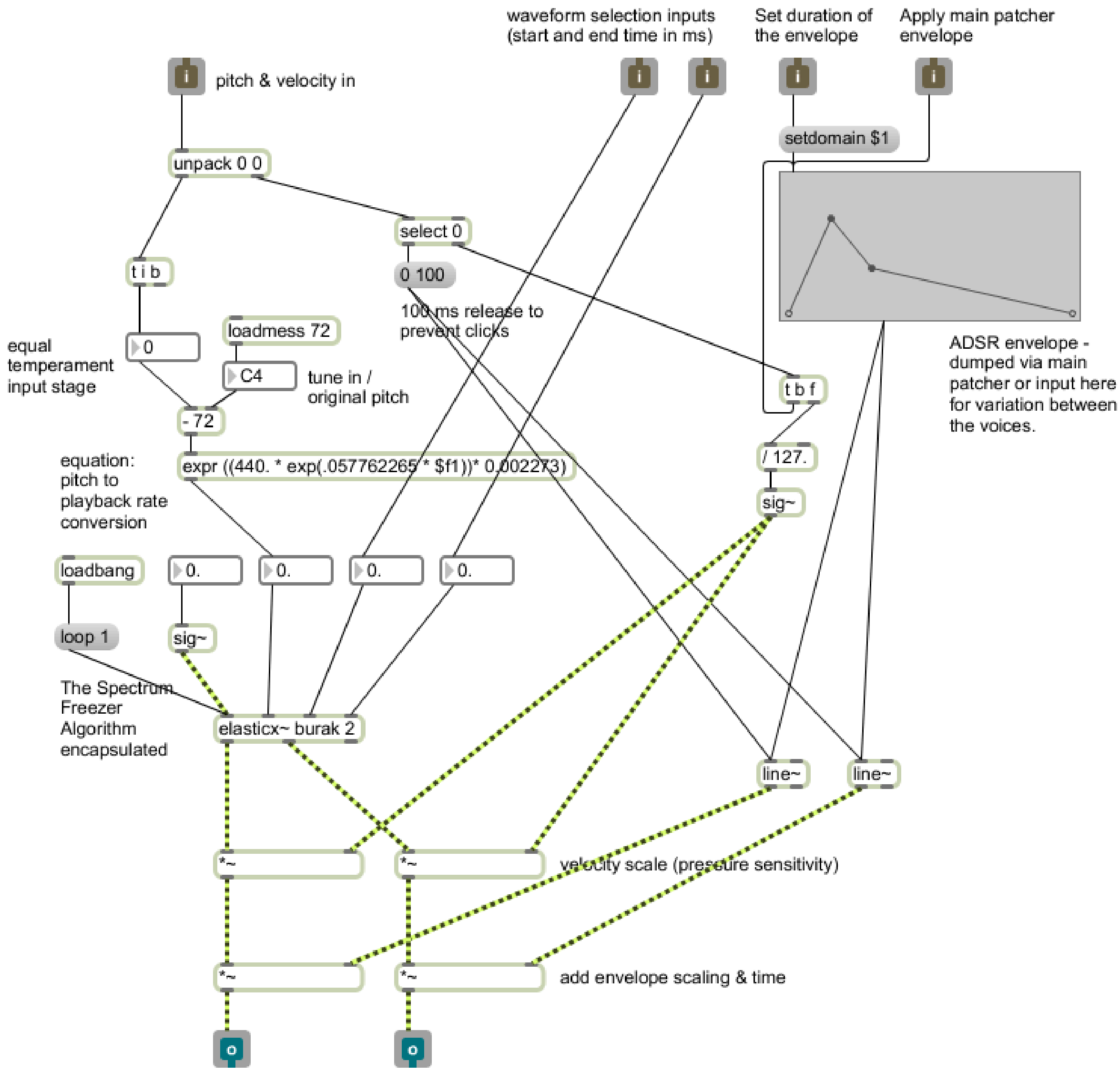

Figure 5.10:FFT Engine (second, third and fourth voices), Spectral Freezer algorithm encapsulated, envelope generator dumps from parent patch.

The master tune dialog in the main patch controls the tuning bias of each of the voices. In the following section, this input will be provided both in notes of the equal temperament system and in a continuous microtonal axis.

Figure 5.12 displays the second, third and fourth voices of the encapsulation with portamento time and master tune inputs. The previous procedure in the first voice applies; input six accepts the pitch ramp time in milliseconds while the seventh input takes the master tuning frequency as in either equal temperament biasing or microtonal transposition values. These voices are to be used with the first voice introduced in Figure 5.11 in the parent patch. Some of the possible parameter implementations of these encapsulations offer portamento time.

This portamento time occurs with randomized (within determined limits) variations.This arrangement could create a certain organic feel by introducing a certain amount of unpredictability (Ballista, 1992).

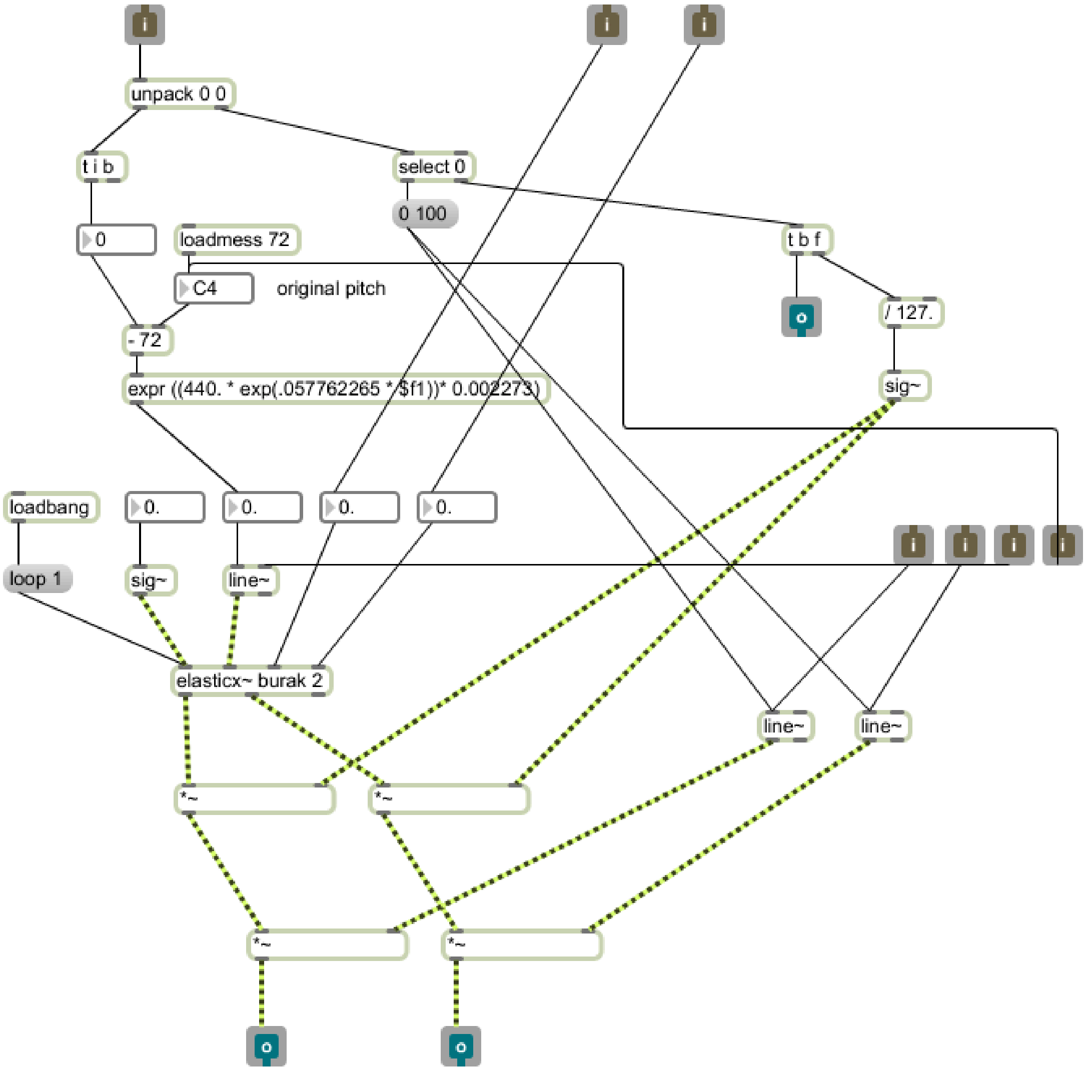

Figure 5.11:FFT Engine (first voice) with global portamento and master tune inputs.

Another implementation that could extend the using range of this instrument presents us the use of microtonal systems. The design of the instrument so far covers the equal temperament, when the keys of the traditional keyboard controller are pressed notes with equal temperament intervals come out no matter what master frequency they are set to. This is due to the distribution equation used in the encapsulated patch.Defining optional equation stages here that would produce various microtonal scales is an idea for further development of the instrument.The final improvement that is to be explored further is a controller stage improvement.

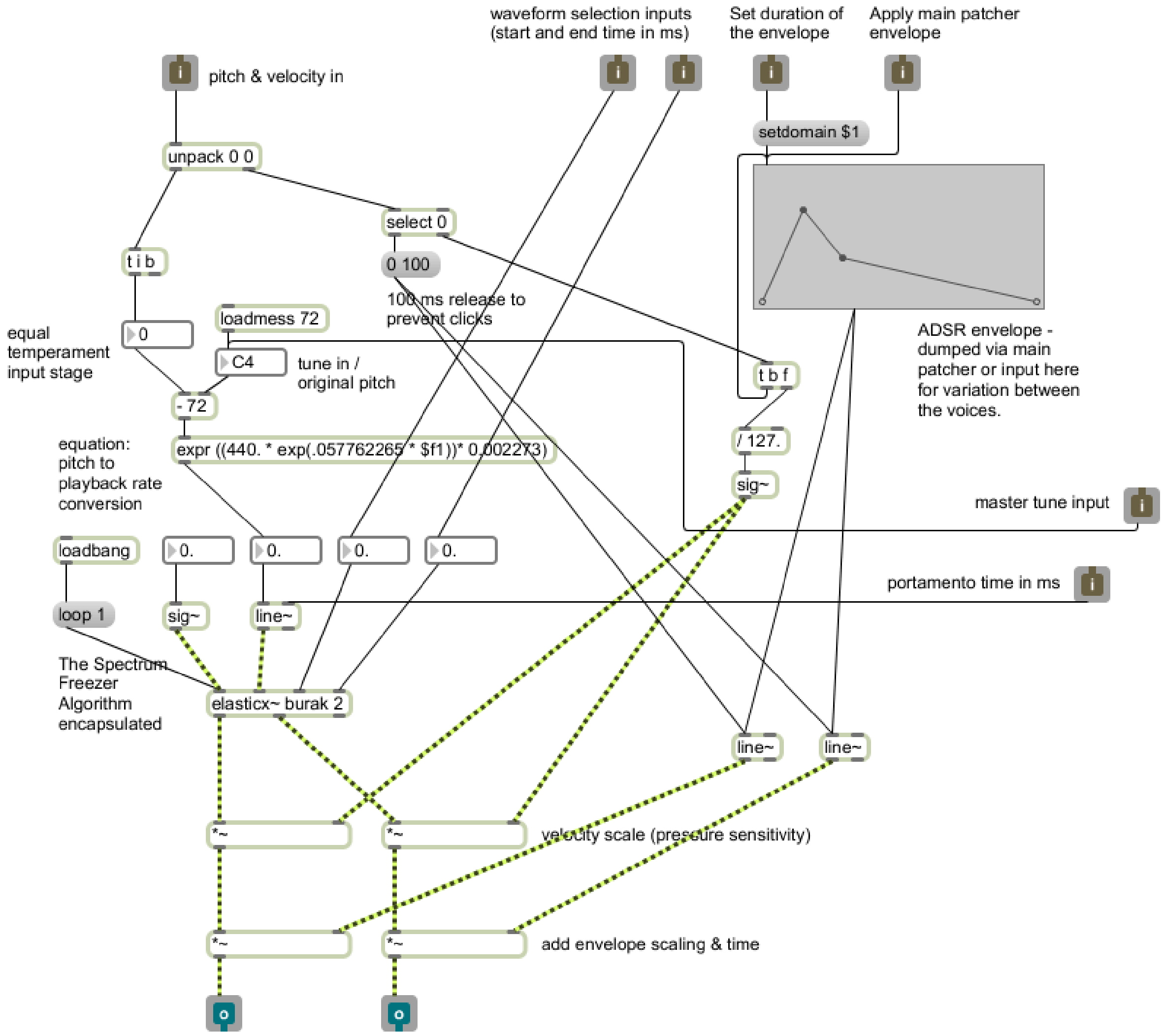

Figure 5.12:FFT Engine (second, third and fourth voices) with global portamento and master tune inputs.

The final improvement that is to be explored further is a controller stage improvement. The current configuration can produce notes of the equal temperament scale and portamento ramps in between them if desired. It is not possible to play glissandos unless an appropriate ramp time is specified for the portamento input. In combination with the traditional midi keyboard controller, it is possible to use a midi ribbon controller (such as the Doepfer R2M) to play glissandos without depending on the portamento time.

5.5.4 Realization of the FFT synthesizer with four-voice polyphony and convolution

This section works on driving and controlling the encapsulated FFT engines as voices of the instrument.

The core engine is made up of the spectrum freezer algorithm. Each voice of the main patch is an encapsulated controller. The main patch acts as a global controller for all voices and the user interface. Therefore the structure comprises two levels of encapsulation to house the sound generation units.

Since introducing polyphonic control requires careful management of each voice, it should be tested in certain phases of development. The first step will be a four-voice polyphonic FFT sound generator device (Figure 5.13).

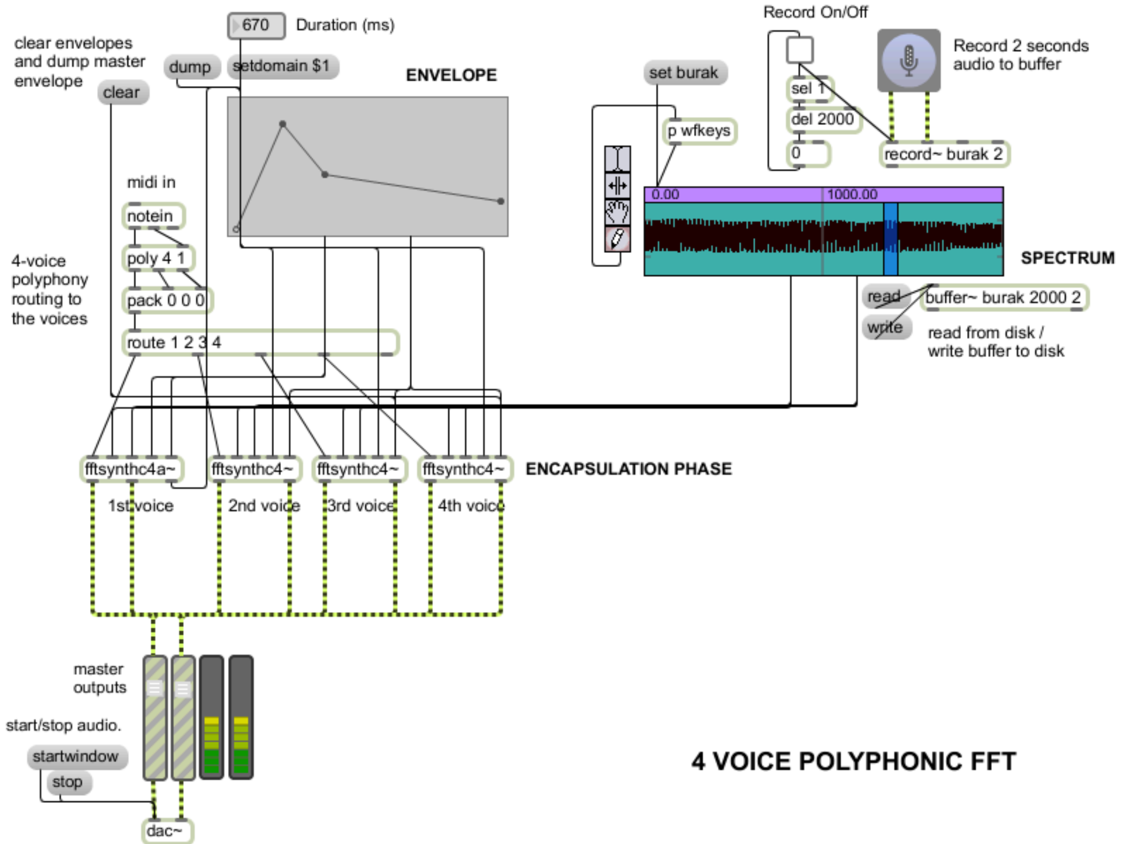

Figure 5.13:4-voice polyphonic FFT sound generation.

The MIDI keyboard has been selected as the physical input of the instrument. The MIDI data protocol is widely used in digital audio workstation software, hence it is considered as an important input to test and compare the instrument's output. There are many controller interface design possibilities that can be considered for further exploration, some of these approaches will be discussed in the conclusion section. This study uses the MIDI controller data as input to the instrument to set the basic functionality of the algorithm. Direct physical interaction as well as computer programming of MIDI data through DAWs constitute the two main controlling interfaces of the instrument (Önen, 2011).

The note in object takes the midi data from the controller or DAW midi notation. The outputs of notein are the pitch as midi note number and velocity as numbers from 0 to 127. 0 represents no sound while 127 means full dynamics. Poly object takes these inputs and sends each of them as outputs with voice numbers. The patch in Figure 5.13 has four voices, the poly object sends out voice numbers attached to pitch and velocity value pairs so that the polyphonic input of the player can be distributed along the sound generators. The pack object packs these number pairs as strings of three and the route object applies the distribution of the input controller data along the sound generation controller encapsulations. Each output of the route object is sent to one voice in order to achieve polyphony. This states that the number of the outputs of the route object determines the polyphony, which is four in our case in the first phase.

Route distributes the pitch and velocity pairs into the first input of the FFT engines that has been described in the previous section as first, second, third and fourth voices. The second and the third inputs of the encapsulation are fed by the start and end times of the selected portion in the waveforms object, in milliseconds. These inputs proceed to the core engine within the encapsulation to set the portion of the buffer that is to be resynthesized.

The fourth input separates the first voice from the other ones since it does not encapsulate its envelope that is being used as an interface object as well as a global envelope controller. Therefore the third output of the first voice sends a trigger message to its envelope located in the main patcher. The second output of the global envelope sends all of the points of the function in line format and is received in the encapsulation by a line object to be passed onto the second input scaling stage of the sound generator. The fourth input of the remaining sound generators is driven by the global duration parameter. This number box on the presentation display sets the length of the envelope generator in milliseconds.

The fifth input is fed by two separate commands. The third output of the global envelope generator sends the function as a list when it receives the dump message. Therefore once the global envelope has been set in the main patcher, via the dump message it has to be sent to each individual voice’s envelope in order to apply the global envelope to other voices of the polyphony. A clear message that deletes the ADSR function in theenvelope generators of voices.

This feature has been supplied as an additional input to the fifth input of thesevoices to fix accurate refreshing of envelopes before global dumping takes place.

The stereo output of each sound generator is sent to the gain sliders that constitute the master gain scaling phase of the instrument. The waveform section of the instrument suggests a picture slider object that has four modes for display and sample based manipulation. The top tool is for determining the selection of audio. The second one adjusts length of the portion while dragging up and down with the mouse. The hand tool zooms into audio waveform when dragged towards the right corner and down. This is useful when selecting very short durations of audio material. Dragging towards the left corner and up zooms out of the display. The audio input is recorded into the buffer when the toggle button is pressed. After recording two seconds of audio into buffer the toggle button is reset. The buffer can read an audio file from the hard disk, and write its recorded content as an audio file to the disk.

The flow of the user interaction proceeds as such: The user begins by recording two seconds of audio into buffer. This can be any sound; a musical element, a concrete sound object or an ambient soundscape etc. Another choice presented here is the ability to read into buffer from disk. So, the user can read any file from disk without using the microphone input. This may serve to load in previously recorded audio as well as present the opportunity to cycle through any recorded sound or music to gather timbres from. Having loaded the sound into buffer with either one of these methods, the user then selects the portion of audio that the spectrum is to be derived from.

Setting the duration of the envelope is the other step; using the function generator interface the user determines the ADSR envelope that is desired for the timbre selected. To transfer the global envelope to each voice, clear and dump messages are sent consecutively so that each individual envelope will be reset first, and instantly updated with the main envelope via the dump message. The final step would be adjusting proper levels for the configuration via the stereo gain sliders and their meters. The user can now play the instrument via a MIDI controller keyboard or send MIDI messages to the instrument via DAWs or other hardware devices.

Figure 5.14 displays the second stage of the development of the polyphonic instrument.

This version adds the convolution algorithm introduced in the previous section to the polyphonic FFT sound generation unit. The version also works on some improvements for the global controller design due to user interaction and extra control for expressiveness. The version suggests a new algorithm that eliminates the use of two consecutive commands while applying the global envelope to every voice of the polyphony. The clear message triggers a bang message that is to be delivered to the dump message. In order to prevent conflicting bang messages, the signal is delayed 200 milliseconds so that the clear messages clears the content of the function objects of each voice before the dump message transfers them the main patcher envelope (Farnell, 2010).

A new global controller stage is introduced in order to control the master tuning of the instrument. The reference tone (C4) that sets the tuning bias of each individual voice is controlled by a master parameter which is C4 by default. Therefore, if the user does not interact with this parameter, the timbre extracted stays with no transposition. The tuner section suggests a fine-tuning option, so if the user changes the tonal bias to lower or higher registers to tune the derived timbre to the desired register, he can then fine tune the tonality by using non-whole numbers in the second number box below the master tune note input. The number box on the right displays the midi note number of the master tune setting; this display is useful when the user needs to get back to microtonal intervals smaller than a minor second as it displays the scaling as midi note number (Wilde, 2004). Notice that the number of inputs on each of the sound generators has increased to seven in this version. The sixth input is the portamento time.

This input is fed to the line object that is located in the second stage of encapsulation of the sound generator voices. The line object creates ramps and the global parameter sets the duration of these ramps in milliseconds. For the sake of simplicity, this version proposes the same portamento time for every voice of the polyphony. Further exploration may be applied to the algorithm of this setting in order to randomize (within certain limitations) the portamento time setting of each voice for organic behavior of the instrument as this process will introduce a certain amount of unpredictability that simulates the infinite parameters of acoustic, electro acoustic and analog electronic instruments. Notice that the voice number output of the poly objectis sent to a large bang input.

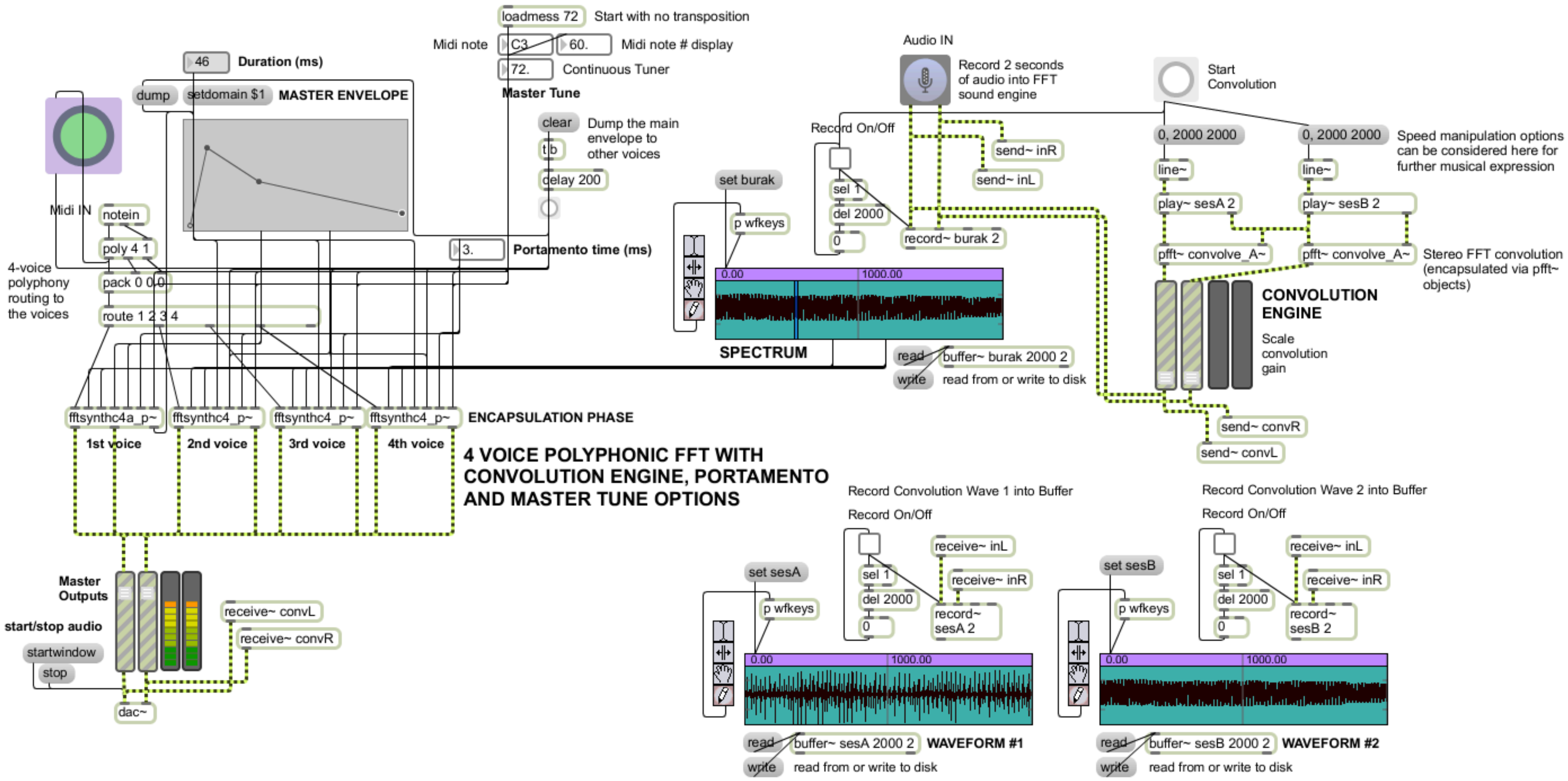

Figure 5.14:4-voice polyphonic FFT sound generation with convolution. The parameter settings suggest a staccato envelope with a very brief pitch ramp.

This procedure serves two purposes: The large bang displays every midi note-on message that the instrument receives, so it may be considered as a crucial data visualization. Its second function is its use for triggering the pitch ramp of every voice. Every midi action triggers the voice number message output of the poly object, therefore the bang display visualizes the midi input while the output of the bang realizes the portamento time for each new note-on message.

Due to these additional controls, the flow of the user interaction is altered. The user starts with the same basic sound input via the microphone or the hard disk. When the timeline selection and the ADSR envelope along with its duration are set, the instrument is ready to be used. The output gain is scaled using the provided stereo slider and meters. The user now can set parameters of the master tuning and portamento time. The timeline selection on the waveform display lets us to use different portions of audio for timbre extraction. This action results in variations of the tone color, resembling the filter stage of time based additive synthesis algorithms. The master tuner transposes the sound generators to the desired range; this setting is crucial to reinforce the performance of the interaction due to the fact that the user can record sounds from any register and / or may seek to play sounds from any register (Russ, 2009). The portamento input enables the player a certain expressive quality reminiscent of glissandos. However in order to play proper glissandos using this technique, the portamento time would have to be scaled according to the rhythmic values in the notation. The addition of a ribbon controller enables the user to play glissandos independent of the portamento time parameter, as discussed in section 5.5.1.There is a convolution section introduced to the polyphonic instrument in this version. This section enables the user to record and convolve sounds, then play the composite timbre on the polyphonic keyboard. The user may either use one sound or convolution of two sounds for a direct or composite timbre extraction. This alters the flow of the interaction with the instrument as the user can experiment with both techniques, read each file from disk or record new sounds into buffer while writing the ones to be saved to the disk. The convolution section operates on a similar procedure. The user either records or reads sound onto buffer, these are labeled as waveform one and waveform two.

After completing this process, sending a trigger signal with the bang button of the convolution algorithm convolves the two sounds into one composite sound.

This sound data is written to the primary buffer that the spectrum is derived from. The stereo faders at the convolution stage allow us to scale the gain of the convolution. Therefore it is possible to boost the gain of the low-level convolution signal recorded into the primary buffer, or vice versa, reduce the gain in order to prevent clipping of the primary buffer data. The convolved signal is then ready to be played, the user can interact with the global envelope, duration, portamento and master tune options to further shape the sound.

5.5.5 Realization of the FFT synthesizer with eight-voice polyphony and convolution

The eight-voice polyphony has been implemented to the instrument using the modular approach that has been applied so far. Each voice represents a spectrum freezer algorithm encapsulated within its individual controller mechanism that translates messages from the global controller algorithm. Therefore, in order to achieve eight voice of polyphony, we simply add four more voices to the previous version. This addition extends the number of notes that can be played simultaneously to eight. The first voice can be considered as a transitional voice to the main patch and its controllers since it acts as an amplitude scaler for the others. After the first voice has been planted and ready to transmit its control signals to the other voices, sound generators can be added as a parallel network to increase the number of voices of the polyphony. Note that the arguments for the poly object has been changed to eight and route object outputs are increased accordingly in order to distribute midi messages to the voices.

Figure 5.15 displays the realization of the eight-voice polyphonic FFT sound generation module. The parameter settings suggest a 281 milliseconds envelope with 24 milliseconds of pitch ramp time. The right end of the envelope has not been ended in the zero amplitude value; therefore as long as the key of the keyboard is held down, sound generators continue to transmit output. The ADSR envelope occurs in the stated duration while after the key has been released the zero velocity message fades the sound out in twenty milliseconds due to the argument of the line object operating within the control algorithms of each voice. The sound is tuned to an octave lower than the original voice at the master tuning section. An improvement has been made to this control level of the interface.

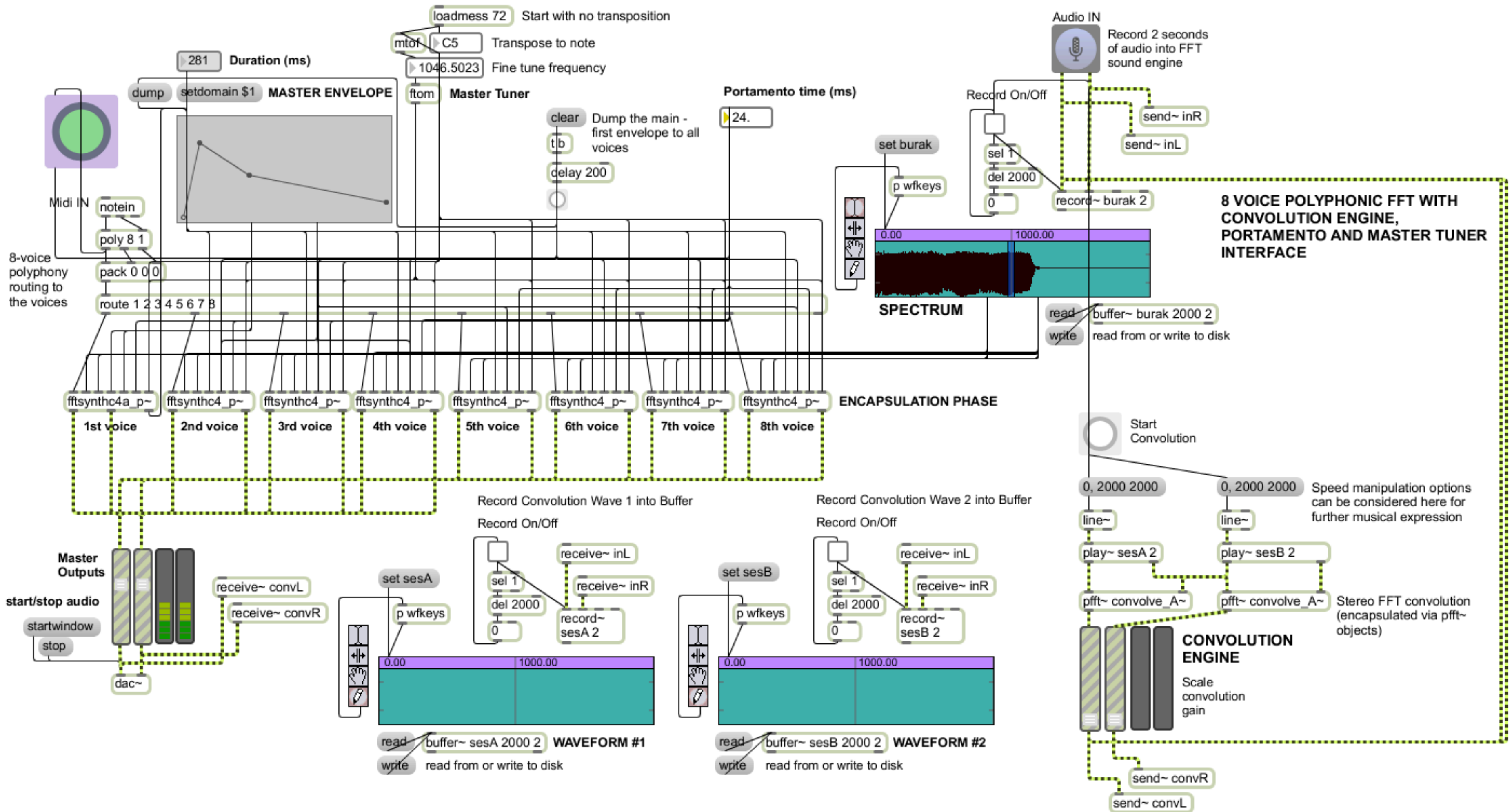

Figure 5.15: 8-voice polyphonic FFT sound generation (the convolution unit is idle in this example).

The tuning section’s default value is no transposition. The user is asked to transpose in steps of equal temperament to find the right register and sound color for the partition to be performed. The lower box represents the fundamental frequency of the selected note in units of Hertz; therefore it is possible to fine-tune the determined pitch here with microtonal adjustments by dragging onto the numbers. This continuous tuning action brings up the question of the tuning concept as a vital aspect of this instrument. Since the instrument’s design perspective depends on instant and natural interaction as well as organic and expressive sonorities, tuning the instrument to the appropriate register will be included in the interaction flow. This step reinforces the concept of individualization of the instrument. Like every acoustic instrument has to be first set to the appropriate tuning, this instrument has to be tuned after the reading the sound data into buffer (Winkler, 1998).

Since the instrument accepts any sound as material to extract its sonority from, the results of samples with dense low frequency energy due to the default setting differ from thosewithhigher frequency. The master tune is useful since it offers the possibility to explore any sound material; the user can transpose in octaves to set the correct register.

The microtonal tuning option lets us individualize the sound further. With this parameter the sound generators are openly presented to the player so that they can be considered as a physical electro-acoustic generator device of a traditional instrument. There is one more control stage that the registers can be transposed at. This lies in the midi keyboard controller octave shift functionality. Considering these two stages of tonal shifters, it is possible to say that the master tune stage sets the center of the register to be performed while the octave shift on the keyboard can be used to expand the selected range.

Note that the arguments set in Figure 5.15 do not display convolution. In this example, the sound, which is a high-pitched sustained vocal sample, is recorded directly to the buffer using a condenser microphone and a signal chain of an analog preamp followed by a DAC. However, it is possible to use the built-in on-board microphone to collect acoustic sounds into the buffer.

Figure 5.16 displays the same instrument with an active convolution engine.In this approach, the user either records or reads sound data into the two buffers.

Once this step is complete, the start convolution button convolves these two sounds into the primary timbre buffer. The gain scaling stage at the output of the convolution unit offers proper leveling of the composite sound data while it is being recorded into the main buffer. This example suggests an approximately two seconds envelope duration that is suitable for playing in legato as the release of the ADSR has not been terminated in order to sustain. The master tuning of the instrument has been set to the E4; therefore the timbre is transposed downwards a major third from its original tonality.

The composite sound data in the buffer features characteristics of both sounds from the convolution engine. However, in order to acquire musical sounds using this approach, the user should either narrow down the harmonic content of the result to match certain sonic expectations or study the contents of the two sounds to be complementary and therefore suitable for convolution (Wilson 2011). This process can be considered as a wide area of experimentation as well as realizing versions of traditional approaches to convolution such as the vocoding technique.

5.5.6 Presentation of the instrument and the interaction model

This section presents a programmer's interface for the instrument which can serve as a basis for the actual user interface (Figure 5.17). The intended user interaction to the instrument will be evaluated in micro and macro scales.

The first step of the user will be determination of the timbre. This may occur on two levels with choices under them. The first level would be recording directly into the provided buffer for timbre extraction. The user simply clicks the microphone icon on the upper left corner to activate the microphone input and hits the record toggle button to start recording.Instead of recording into the buffer, the user may choose to read previously recorded files from disk which constitute the second choice of the non-convolved level. In this case the user clicks on the read button under the spectrum waveform display to open up the browser dialog to read from disk. The file to be read may be a sample from any recording on the drive. At this point the user is ready to proceed to sound generation directly but we will first evaluate the second stage of timbre input.

The second level starts the interaction at the convolution engine section.The user now interacts with two buffers and is required to record two sounds.

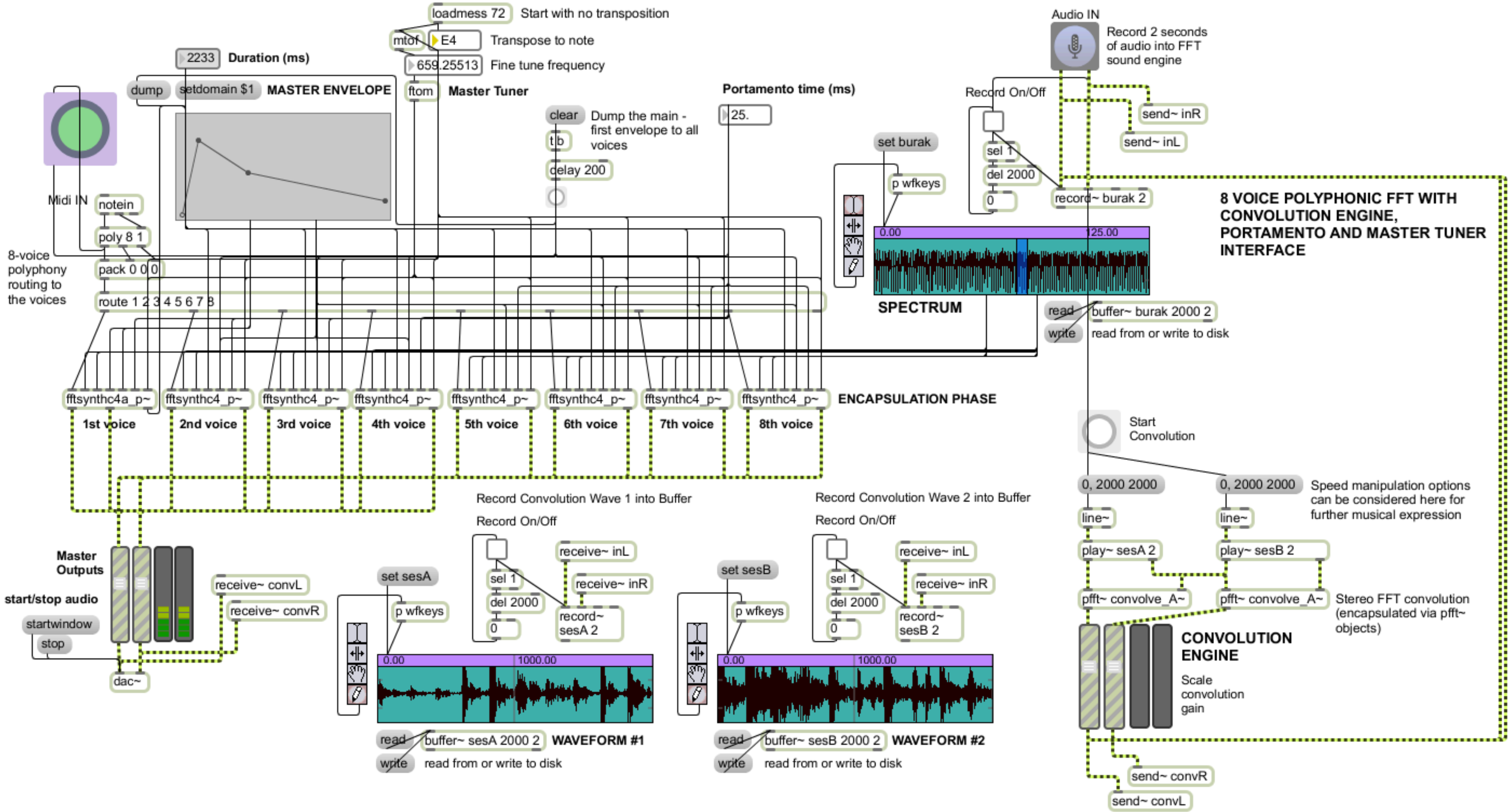

Figure 5.16:8-voice polyphonic FFT sound generation with an active convolution unit.

This action can be achieved by clicking the toggle buttons placed on top of the waveform displays. New choices reveal themselves at this phase since the user can either record or read files from disk to fill the contents of the buffers. The user experiments at this stage either with recording directly into both of the buffers, recording into one of them and reading a file for the other or reading files for both of the buffers. After deciding the two sounds in the buffers, the user starts the convolution by clicking on the convolution button. It is necessary to set appropriate levels for the convolved signal to pass through using the faders and meters. The main spectral waveform display provides adequate visualization to check for waveform dynamics at this phase. Once the convolution signal has been recorded into the buffer as sound data, the user is ready to proceed to the next phase.

It should be noted that the convolution process includes two levels of transformation from the time domain to the frequency domain. The time domain sound data which is read from the two buffers is transformed into the frequency domain for the convolution. They are transformed back to the time domain by a reverse FFT process in order to be played back to the main spectral buffer as an input. Once this composite audio data is recorded into the spectral buffer, it is going to be transformed into the frequency domain for spectral freezing. As this constitutes the heart of the instrument, the sound generation is transformed back to the time domain again to be fed out to the speakers via the sound outputs of the computer.

The user selects the portion of audio that is going to be used for the timbre extraction. The provided tools for adjusting selection length and sample values are considered here as inputs to the waveform display interface. It should be noted that before manipulating any of the other controls in the interface, it is possible to start playing with the default values of the envelope and tuning settings. This gives the opportunity of instant interaction with the timbre so that the rest of the arguments could be shaped accordingly and while sound is present. The user may go back to change the portion of buffer that is to be re-synthesized and may even go back to the first step to rerecord or reselect sound. Changing the sound data selection often acts as a filter determined due to the sonic variations introduced in the sound data.

When the user is satisfied with the timbre selection, sound shaping options are available for further expression. The length of the ADSR envelope can be set with the duration parameter located in the master envelope module.

The ADSR is set in the function object embedded in the interface; every click introduces a new breakpoint in the envelope. The user has to drag to move the points.Shift clicking on the points removes them. In order to apply the determined ADRS envelope to all voices, the user is required to click the clear message in the master envelope module.

Portamento time is another parameter to be set located in the master envelope box. Envelopes with shorter durations tend to sound percussive when they are complemented with fast attack times. Longer durations combined with slower attack times are appropriate for playing legato partitions. This legato setting can be combined with intermediate portamento times to simulate sliding sustained tones.

The percussive sounding envelopes complement with very brief portamento times to create subtle organic performance effects whereas longer durations of portamento can introduce fluctuated pitch control as if a vibrato with a high depth and low frequency (in the case of large interval leaps in the partition performed) is being introduced to the sound regularly.

The master tuner section contains two arguments: the master transposition value (set to C4 by default) and the fine frequency tuner. The user may either leave the tuning untouched and just shift in octaves to center the proper register of the current sound or manipulate the tuning by transposing certain amounts in equal temperament intervals. The frequency of the note is displayed at the bottom number box simultaneously with the note symbol display. If the user decides to manipulate the pitch, it is possible to shift the tuning microtonally with the frequency tuner.

When the user plays notes from the midi keyboard, the bang button under the microphone input starts to blink to indicate that the midi messages are being transmitted. If the user has more than one midi device connected to the computer, it is possible to select the controller for the instrument by double clicking on the note in box located in the inputs section. This action opens up a list where the user can change controllers (Fry, 2008). The master level of the instrument can be set using the stereo gain faders and meters. Dragging the left fader equals the levels of both faders as it also drags the right channel which it is linked to.

When left and right channels need separate gain levels, setting the left one first and manipulating the right one should be the interaction route taken by theuser, as the right fader does not control the left channel.

When the user is satisfied with interaction and the timbre created, it is possible to save the sound data in the buffer(s) by clicking on the write boxes located within each buffer / waveform display box. The user can either write the spectral buffer data, the convolution buffers or all of them. Constituting a library of this kind encourages the user to experiment with the spectral information. The user can simply recreate the experience by loading in the proper files into the buffers or may try to convolve the main spectral data and vice versa where either parts of the convolution data can be readinto the main spectral buffer. The stop button in the start / stop audio box bypasses the instrument to make it inactive.

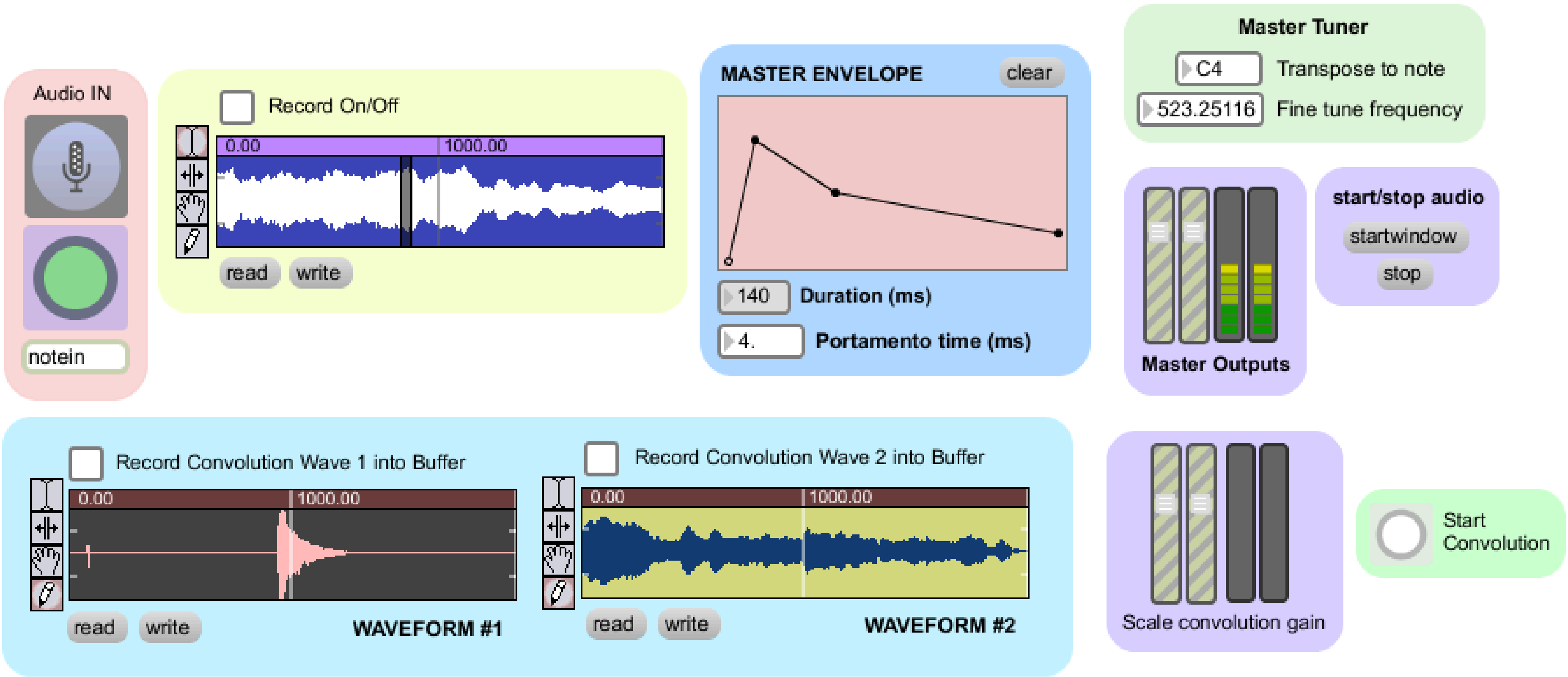

Figure 5.17:8-voice polyphonic FFT sound generation in presentation mode.

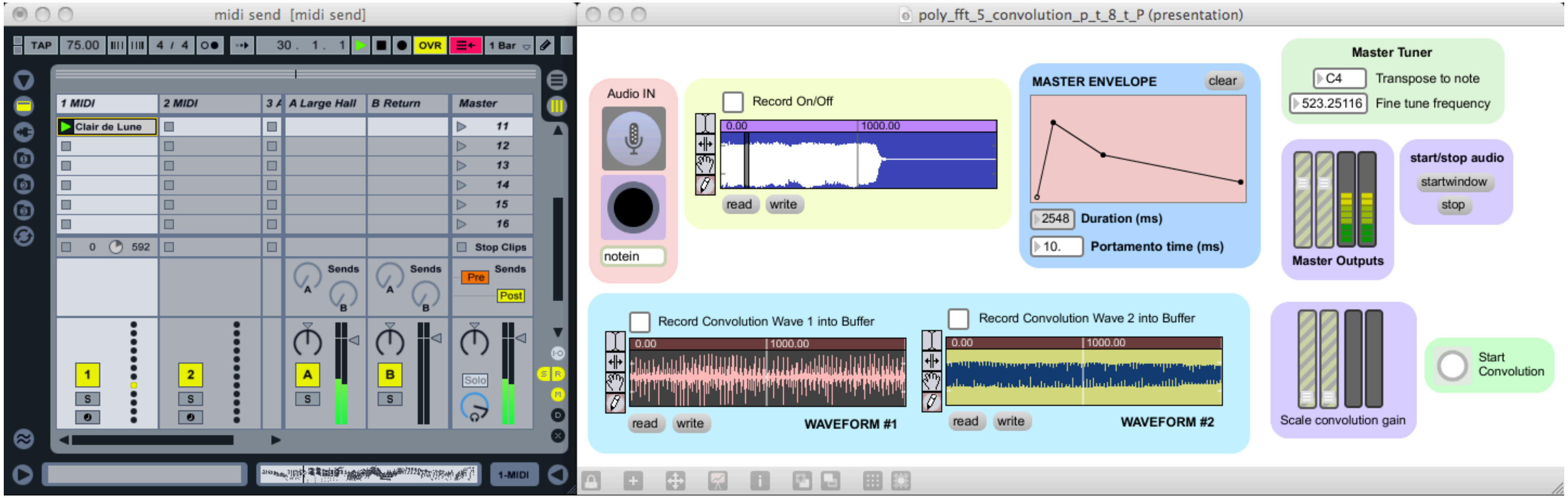

Figure 5.18:8-voice polyphonic FFT played by MIDI signals sent to Max MSP via a DAW (Ableton Live).

6. CONCLUSIONS

Theaim of this study was to propose a model for a new electronic musical instrument design based on the research that explores the early twentieth century instruments and their reflections on composition and performance throughout the twentieth and twenty-first centuries. Studying the technical principles of the early instruments have provided the knowledge of the basic electronic sound synthesis techniques such as electronic sound generation, additive synthesis, subtractive synthesis and time domain processing. The research approaches the electronic sound phenomena from various points of view.

When we consider the actual meaning of the word ‘instrument’ we encounter that it is a device or tool that has been optimized to handle a certain task. The case of musical instruments is similar; musical instruments are machines that can produce a certain range of sounds in their own characteristics,they are designed to address and complement our perception of sound. Our hearing mechanism and the way we interpret audio information (which is transformed to electric neuron pulses in our brain)determines the guidelines of musical instrument design (Creeber, 2009). Acoustic instruments provide the player with a control mechanism that allows varying levels of musical expression and is suitable for mastering as reflexes due to frequent interaction in long term. The control mechanism provides the expressive control of pitch and rhythm, yet the level of expression that can be achieved reveals itself as the player progresses in mastering the instrument’s technique. Acoustic instruments have acoustic sound generators (that are in direct interaction with the player and can reveal certain expressive qualities at the instant of sound generation, determined by the player) and their dedicated acoustic amplifiers, therefore the whole process of sound generation and control is perceived as organic; the carefully designed machine has gained acceptance as a proper tool for musical expression.

Electronic musical instruments for that matter; have been regarded as outsiders by certain portions of public (including audiences as well as traditional musicians) when they were first introduced in the early twentieth century.

Traditional acoustic instruments represent a school of music; the eighteenth century western classical music discipline presents strict rules in that sense about harmony and form of the musical pieces. These boundaries are guidelines to compose and / or perform music in the genre of the well-established classical period. It is not hard to realize that when the electronic music instruments were introduced they struck attention and were considered as a threat to the traditional disciplines of music. It should be noted that the period was also incorporatedresearches for new dimensions in musical composition such as chromaticism, the atonal movement and serialism. Certainly, the interaction of music with technology is inevitable. The designs that we accept as musical instruments today are nothing more than very well designed machines that let us control pitch, rhythm and timbre in order to accomplish musical expression. A piano has the strings as the source of vibration, the soundboard as an acoustic amplifier. The player plays through a control surface called the keyboard that simply activates hammers which hit the strings when the keys are pressed. Hybrid instrument designs such as the electroacoustic instruments have been introduced as well which have been investigated through this study.

Electronic musical instruments operate in coherence to the principles of acoustic and electroacoustic instruments. The sound generation occurs electronically (either as electric currents in analog domain or binary numbers in the digital domain) and is controlled by a control mechanism presented to the user for interaction and mastery of technique. Today’s most common electronic music instrument is the digital synthesizer due to its easy access and versatility. The digital synthesizer finds its place in music production for simulation and reinforcement of acoustic and electroacoustic instruments as well as creating pure electronic sounds. However it is a fact that electronic synthesizers can never fully simulate the acoustic or electroacoustic instruments.

The application of Fourier’s theorem is a matter of engineering terms, this means that the infinite number of pure tones and their modulations in time will be achieved in the future when we consider the doubling of the CPU power every year and the anticipated introduction of the quantum computers in the 2020s. Although in the near future when the necessary number of calculations can be handled their sound may be fully synthesized via a computer, the control mechanisms that allow direct manipulation of the sound source for expressiveness will still not be exact.

Therefore we can say that the ability to imitate acoustic instruments is important but this does not reflect the true musical potential of electronic sound synthesis (Greenberg, 2007). Today advanced technology and communication of the twenty-first century has introduced a faster pace of life with information running in from various sources continuously. This may seem and is advantageous; it is much easier and faster to gather information via online sources now compared to ten years ago. The disadvantage of this technological advance is the fact that it is still hard to gather complete and detailed information for any type of research as information pollution exists which can be misleading.

The digitals synthesizer concept may be considered analogous to this analysis; the synthesizer is accepted as a musical instrument however it is misinterpreted by the public due to its vast amount of implementations that do not offer the depth of a real instrument design and is tailored for entertainment or simulation purposes. The polyphonic FFT instrument design model that this thesis proposes challenges these issues this research has brought up and questioned. First of all, it is a process and is not necessarily applicable for acoustic instrument simulation. This can obviously be done to a certain extent, yet the instrument design already suggests that once the acoustic sound data has been passed onto the buffer, it becomes the source material for the timbre of the instrument. This timbre can be shaped further by the provided frequency and time domain processing but at this stage it is obvious to the user that the timbre is isolated and transformed into a new interactive structure.

At this point it is important to discuss the possible development issues of the model proposed. So far the model can apply FFT to the sound data in buffer, introducing granular synthesis to this approach will extent the ability of the instrument to create unique sound textures that may be appropriate for use in contemporary music as well as sound design purposes. In order to introduce granular synthesis to this instrument, the playback speed of the encapsulated FFT sound generation has to be set to numbers larger than zero which in our case is a parameter that freezes the spectrum. Achieving granular synthesis will require longer sound data in buffers, so it is convenient to increase the duration of the buffers to ten seconds or more. Variations for playback speeds may be introduced, controlled by a global playback speed parameter and a BPM argument to align timbre variation rhythmically to the tempo of the partition performed.

These variations may still include the case of spectral freezing so that some of the voices will be playing the extracted timbre while others play a larger portion with varying playback speeds that are in accordance with each other. This procedure will be introducing more reference to the original sound data that modulates in time, thus the implementation will have to be carefully adjusted. The procedure will result in realistic sound textures that can actually refer to the action taken in the sound data. One specific parameter arrangement of this improvement will occur when the playback speed is set too high while the portion of audio selected remains as a short duration. This will introduce a repeated resynthesized sound whose grains (particles of sound data with approximately two-three millisecond durations) will be resonating within each other to output a new texture with the timbre determined. Obviously, like the playback speed variations have to be arranged to the tempo via an algorithm that can introduce rhythmic note values, the durations of the audio selections has to be tuned in order to maintain the design's stability.

Another crucial improvement to the design that claims the behavior of the algorithm more organic for the human perception would be adding some variations to parameters such as the portamento time, release time for note off messages, audio data selection and sinusoidal envelopes to playback speed. These variations should be randomized procedures and therefore applied to each voice separately.

When the global portamento time is set, the algorithm introduced in the encapsulations of each voice interprets the time in milliseconds by adding or removing a random duration that is shorter than two milliseconds.

The procedure will change with every use of each voice and therefore will be perceived as a continuous behavior. Same procedure may apply to release time which is set to twenty milliseconds for our model. Randomized patterns may add or subtract durations up to four milliseconds to distribute variations to the performance aspect. This process should be applied to each voice separately and will be renewed with every note off message. The data selection in the buffer constitutes the heart of this model as it determines the timbre of the instrument. Varying the buffer selection for each voice in terms of time would be considered as another way to reinforce playability. Basically the principles of the algorithm will be similar but the calibration of randomized patterns will have to be adapted. An appropriate setting for this procedure would be using randomized patterns.

This can be achieved in the start time parameter of the waveform display object modulating up to ten milliseconds. It should be noted that this improvement might produce unpredictable results as it strongly depends on the sound data in the buffer.

Finally, the algorithm of the instrument may be repurposed to recreate the occurrence in the sound data by combining the frequency and time domain techniques. This procedure can be considered as an addition or variation to the playback speed options introduced earlier in this chapter yet it incorporates a different perspective and is therefore subject to research. In this approach the attack, decay, sustain and release portions of the audio are preserved. Obviously the selection in the buffer will count and it will be possible to overwrite or reinforce the envelope with the master envelope module. This approach sets the playback speed to one so that the resynthesized audio is played back with its real timing. However, when the playback header reaches the end of the selected data in the buffer, a control algorithm updates the playback speed to zero so that the spectrum is frozen. Appropriate master envelope setting should be applied at this stage to take the benefit of this approach. The result of this procedure is a naturalistic approach where the time based action on the recorded audio is preserved by the aid of time domain processing while it is still possible to repurpose the sound as the timbre of the instrument since when the real time playback ends the sound does not cut out.

At this point the control algorithm activates the spectrum freezer so that the sound can keep sustaining. The whole process occurs in the frequency domain however the attack and decay portion of the audio are resynthesized as if they were in the time domain and the sustain portion can be as long as it is desired since the spectrum freezer algorithm takes on once the time domain playback ends.

If the scaled pitch data is driven by a sinusoidal signal, then it will be possible to introduce certain amounts of vibrato to the instrument. However the control mechanism which is the midi keyboard does not allow lateral movement that could be interpreted for expressing vibrato. This feature should be considered for further research on controller design.The model proposed and realized so far in this thesis produces successful results when sustained tones are used as sound input data. This category can be reduced to sounds caused by longitudinal vibrations as these sources (such as the wind instruments) produce sustaining tones with variations in dynamics and timbre.

Transverse vibrations on the other hand can produce satisfying results in this version of the model yet the sound color transformed may depart from the original source. This procedure can be useful when considered for sound design purposes however when realistic implementation is desired, the algorithm improvements described in this chapter has to be appropriately applied in order to maintain time domain features alongside the timbre extracted.

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APPENDICES

APPENDIX A: GLOSSARY

ADSR Envelope: ADSR stands for the attack, decay, sustain and release portions of the dynamic envelope of a sound wave. This physical parameter determines the dynamic quality over time of the sound produced.

Amplifier: Theamplifier term in audio terminology has a range of applications. Whether in the analog or the digital domain, the amplifier increases the gain of the input signals while it may impose certain variations of a filter on the input spectrum depending on its non-linear boosting transform curve.

Bang: The most common message type in the object oriented audio softwares such as Max MSP or Pure Data. Bang messages can trigger systems in the control rate or route messages in DSP networks. Its common implementation is in the form of sending a ‘one’ message.

Convolution: The multiplication of two signals in the frequency domain. The frequency bands that are common in both of the inputs remain in the convolved signal while the non-intersecting portions of the spectrum are cancelled.

Object Oriented: An object oriented programming language does not use line-by-line text like the traditional coding applications. Rather, this approach treats the algorithm as a network and comprises previously coded modules with a certain number of input and outputs that form the patches.

Phasor: Physically, this term refers to the two parameters of a spectral component other then the time dependent frequency which are the phase and amplitude.

Portamento: This is a musical term that indicates a slide between two pitches. In sound synthesis terminology, the term refers to the time (driven bay a linear ramp, in milliseconds) that it will take to go from the previously played voice to the new voice that is being played at the moment.

Spectrum: The spectrum of an audio signal derives its information from the frequency domain. The spectrum display presents the amplitude value of every frequency component along the frequen**cy** axis. When time as the third axis is added, a three dimensional display occurs.

Synthesizer: Basically a synthesizer comprises electronic sound oscillators that generate basic waveforms. The synthesizers are built in modular approach and time domain processing units such as envelopes, modulation, LFO, filter networks, delay networks etc. are suggested in a various routing combinations.

Timbre: This term refers to the color of a sound. Due to its spectral content and how each frequency band in this content evolves in time, sound waves leave imprints in our acoustic memory. Perception of sounds is highly affected by these imprints that we refer to as the timbre of a sound.

Tone Wheel: These devices were first used in the early electronic music instruments. Each tone wheel carried an imprint that is the analog of the actual partial it represented. Even with the introduction of the tube transistor technology, mechanical tone wheels were still continued to be used in instruments such as the Hammond Organ.

Triode: It is an electronic amplification device. They were used as vacuum tubes in consumer electronics as well as electronic musical instruments during the early and mid twentieth century until a wide range of their use were replaced by the semiconductor transistor.

Vibrato: It is a musical term.Basically it is the oscillation in pitch introduced by the player for musical expression.

Voltage Controlled: A voltage-controlled oscillator produces oscillations whose frequency can be controlled with a varying voltage. This procedure constitutes the foundation of the analog sound synthesis. The input may be driven by modulating signals to achieve frequency or phase modulation.

White Noise: When all the frequency components in the spectrum of a sound wave have equal gain, the resulting sound is called the white noise. When the high frequency content of this noise is rolled off, the sound resembles daily background noise as the roll off simulates the loss of high frequency content due to reflections and the broad spectral energy the noise sound of unorganized sonic environments.

APPENDIX B.1: Selected Theremin Repertoire

Andrei F. Paschtschenko – *Symphonic Mystery for Theremin and Orchestra*, 1924.
Joseph Schillinger – *Melody for Theremin and Piano*, 1929.
Joseph Schillinger: *Airphonic Suite for RCA Theremin and Orchestra*, 1929.
Joseph Schillinger – *Mouvement Electrique et Pathetique for Theremin and Piano*,1932.
Friedrich Wilckens – *Dance in the Moon for Theremin and Piano*, 1933.
Edgar Varese – *Ecuatorial*, 1934.
Percy Grainger – *Free Music #1for four Theremins*, 1936.
Anis Fuleihan – *Concerto for Theremin*, 1942.
Bohuslav Martinu – *Fantasia for Theremin, Oboe, Piano and Strings*, 1944.
Isidor Achron – *Improvisation for Theremin and Piano*,1945.
Lydia Kavina – *1. Andante, 2. Moderato, 3. Lento for Theremin and Piano*, 1989.
Lydia Kavina – *In Whims of the Wind for Soprano, Theremin and Piano*, 1994.
Jorge Antunes – *Mixolydia* for *Theremin and Electronic Tape*, 1995.
Vladimir Komarov – *Voice of the Theremin for Theremin and electronic Tape*, 1996.

Theremin in Motion Picture Soundtracks:
Spellbound (directed by Alfred Hitchcock, music composed by Miklos Rozsa), 1945.
The Lost Weekend,1945.
Lady in the Dark,1946.
The Fountainhead,1949.
Rocketship X-M, 1950.
The Thing, 1951.
The Day the Earth Stood Still, 1951.
The Ten Commandments, 1956.
Billy the Kid vs. Dracula, 1966.
The Giant Gila Monster(The electro-Theremin performed by Paul Tanner), 1959.
Straight Jacket (The electro-Theremin performed by Paul Tanner), 1964.
Ed Wood(directed by Tim Burton, music composed by Howard Shore, Theremin performed by Lydia Kavina), 1994.

Electro-Theremin in Popular Music:
Dr. Samuel J. Hoffman– *Music out on the Moon*,1947.
Dr. Samuel J. Hoffman– *Perfume Set to Music*, 1948.
Dr. Samuel J. Hoffman– *Music for Piece of Mind*, 1950.
Warren Baker – *Music for Heavenly Bodies*, 1958.
Beach Boys – *Good Vibrations*,1966.
Beach Boys – *I Just Wasn't Made For These Times*,1966.
Led Zeppelin– *Whole Lotta Love*, 1969.
Clara Rockmore – *The Art of Theremin*, 1987.
Pixies – *Velouria*, 1990.
Portishead– *Humming*, 1997.
Lydia Kavina – *Music from the Ether: Original works for the Theremin*, 1999.
Kurstins– *Gymnopedie*, 2000.
Tom Waits – *Blood Money,* 2002.
Clara Rockmore – *The Lost Theremin Album*, 2006.

APPENDIX B.2: Selected Ondes Martenot Repertoire

Maurice Ravel –*Quartet for Strings in F major: 1st movement, Moderato très doux (Ondes Martenot versions authorized by Ravel)*, 1903.
Dimitri Levidis – *Symphonic Poem for Solo Ondes Musicales and Orchestra*, 1928.
Darius Milhaud– *Suite for Martenot and Piano*, 1933.
Edgard Varese – *Ecuatorial for two Ondes Martenots, Choir and Ensemble*, 1934.
Arthur Honegger – *Jeanne au Bucher for Ondes Martenot, Orchestra and Choir,* 1935.
Olivier Messiaen – *Fete des Belles Eaux for sextet of Ondes Martenot*, 1937.
Andre Jolivet– *Danse Incantatoire for Two Ondes Martenots*, 1937.
Olivier Messiaen– *Oraison*, 1937.
Charles Koechlin – *Second symphony opus 196 for Orchestra and Ondes Martenot*, 1939.
Pierre Boulez – *Quatuor pour quatre*, 1946.
Andre Jolivet – *Concerto for Ondes Martenot and Orchestra*, 1947.
Georges Auric – *Les Parents terribles for Ondes Martenot and Orchestra*, 1948.
Marcel Landowski – *Jean de la peur – Symphony No.1*, 1949.
Oliver Messianen – *Turangalila Symphonie*, 1948.
Oliver Messiaen –*Le Merle Noir*, 1951.
Marcel Landowski – *Concerto for Ondes Martenot and Orchestra*, 1954.
Jacques Charpentier – *Concerto for Ondes Martenot and Orchestra*, 1962.
Henri Dutilleux – *Trois tableaux symphoniques for Orchestra and Ondes Martenot*, 1965.
Giacinto Scelsi – *Uaxuctum*, 1966.
Jacques Charpentier – *Lalita for Ondes Martenot and Percussion*, 1968.
Roger Calmel – *Stabat Mater for Ondes Martenot and Orchestra*, 1970.
Sylvano Bussotti – *Due voci for Ondes Martenot, Soprano and Orchestra*, 1970.
Henri Sauguet – *Symphonie no.4 for Orchestra and Ondes Martenot*, 1971.
Jacques Chailley –*Le Cimetière Marin*, 1979.
Toshi Ichiyanagi– *Troposphere*, duet for Ondes Martenot and Marimba, 1990.

Ondes Martenot in Motion Picture Soundtracks:
Lawrance of Arabia (Ondes Martenot performed by Maurice Jarre), 1962.
Mad Max(written for full orchestra, a chorus, four grand pianos, a pipe organ, digeridoo, fujara, a battery of exotic percussion and three Ondes Martenots), 1985
Jesus of Nazareth(Ondes Martenot performed by Maurice Jarre), 1977.
The Bride(Ondes Martenot performed by Maurice Jarre), 1985.

APPENDIX B.3: Selected Trautonium Repertoire

Paul Hindemith – *7 Trio pieces for Three Trautonien*, 1930.
Paul Hindemith – *Concertino for Trautonium and String Orchestra*, 1931.
Paul Hindemith – *Langsames Stuck und Rondo for Trautonium(Slow Piece for Orchestra and Rondo for Trautonium)*, 1935.
Harald Genzmer – *Konzert für Trautonium und Orchester (Concerto for Trautonium and Orchestra)*, 1939.
Harald Genzmer– *Konzert für Mixtur-Trautonium und großes Orchester (Concerto for Mixtur-Trautonium and Large Orchestra)*, 1952.
Oskar Sala– *Concertando Rubato from Elektronische Tanzuit*, 1955.
Remi Gassman, Oskar Sala, Geroge Balanchine*Electronics(Ballet)*, 1961.
Remi Gassmann– *Electronics*, 1962.
Oskar Sala– *Five Improvisations On Magnetic Tape*, 1962.
Oskar Sala– *Subharmonische Mixturen*, 1963.
Harald Genzmer– *Cantata for Soprano & Electronic Sounds*, *Suite De Danses for electronic instruments,*1969.
Oskar Sala – *Electronic Virtuosity For Selected Sound*, 1969.
Oskar Sala – *Suite Für Mixtur-Trautonium Und Elektronisches Schlagwerk*, 1970.
Oskar Sala– *Konzertante Musik Für Mixtur-Trautonium Und Elektronisches Orchester*, 1970.
Oskar Sala– *Musique Stéréo for Electronic Orchestra in five parts*, 1972.
Oskar Sala – *Fantasie-Suite In Drei Sätzen Für Mixturtrautonium Solo*, 1977.
Oskar Sala– *Elektronische Tanzsuite*, 1977.
Oskar Sala – *Impressionen (Electronic Impressions)*, 1978.
Oskar Sala– *Electronic Kaleidoscope* (*a collection of soundtracks for short films and television)*, 1983.
Oskar Sala – *Electronic Kaleidoscope,* 1983.

Trautonium in Motion Picture Soundtracks:
Dein Horoskop - Dein Schicksal, 1955.
Schneeweißchen und Rosenrot, 1955.
Forschung und Leben - Schöpfung ohne Ende, 1956.
The Birds (directed by Alfred Hitchcock, music composed by Oskar Sala), 1963.
Der Wuerger von Schloss Dartmore / The Strangler of Castle Dartmore (music composed by Oskar Sala), 1963.
Der Fluch der gelben Schlange, 1963.
Die Vögel, 1963.
Der Würger von Schloß Blackmore, 1963.
Die Todesstrahlen des Dr. Mabuse, 1964.
Make Love Not War - Die Liebesgeschichte unserer Zeit, 1967.
Unterwegs nach Kathmandu, 1971.
Gestern war heute noch morgen - Planet Erde, 1991.
Das letzte U-Boot, 1992.

CURRICULUM VITAE

Name Surname: Yahya Burak Tamer

Place and Date of Birth: Istanbul, 1979.

E-Mail: tamerburak@gmail.com

B.S.: Yıldız Technical University (1997-2001)

M.A.: Istanbul Technical University (2002-2005)

PhD: Istanbul Technical University (2005-2012)

PUBLICATIONS/PRESENTATIONS ON THE THESIS

- **Tamer, B.,** Aşkın, C. Etkileşimli Ses Tasarımı: Çok Sesli FFT Elektronik Ses Sentezleyici Modeli. İTÜ Dergisi (In Press).