NEW TRENDS ON PHYSICAL MODELING OF MUSICAL INSTRUMENTS

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Abstract

This thesis introduces the topic of the physical modeling for musical instruments and its current trends in the commercial and academic world, emphasizing on the necessity of the emulation of musical instruments' *performance articulation. T*he physical modeling instrument topic has lead to the commercial implementation. Therefore a great opportunity to participate can be seen for the Music Technology Research at National Chiao Tung University. The academic resources and the proximity with the technology industry of Taiwan R.O.C may facilitate the development of an algorithm or set of algorithms to emulate musical instruments. Instruments like the Pipa (琵琶) and the Guqin (古琴) could be a start point for a research because generic algorithms for plucked instruments have been developed. Simulations of some western and eastern instruments have been successfully implemented; we can mention the guitar, the mandolin and the sitar. Therefore, by learning the techniques of physical modeling the Music Technology Research Team of National Chiao Tung University can propose a methodology and accomplish this goal.

On the other side, we mention the importance of a research on interfaces to control the parameters of instruments. In the case of Pipa and Guqin both of them are plucked instruments however they do not share the same performing articulations. Therefore, specialized controllers for these instruments should be designed.

This thesis contains, comparisons on "Classic" synthesis and physical modeling synthesis, a summary of physical modeling techniques of acoustic instruments, interface and parameter controllers trends and experimentation with the Karplus-Strong Algorithm for plucked string instruments.

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1. Introduction

1.1 Music Instruments Evolution and the search for new sounds.

The music instruments evolution from its very beginning demonstrate the necessity of new sounds in music. Detailed knowledge of acoustics and the materials of which the instruments are made of is needed to develop new instruments and to expand the possibilities of new sounds. A very good example tracing back the history is the pipe organ, which had changed from a water powered hydraulic in the 3rd century BC[1] to a developed of pneumatic, electric, and electro-pneumatic organs in the late nineteenth century. The sophistication of modern pipe organs in terms of timbre shaping can also exemplify a historical search of new sounds in musical instruments. This is done by the organ stops. In some models, stops can produce percussion sounds and some effects, for example the "Nightingale" which is a stop that blows wind through a whistle submerged in a small pool of water, creating the sound of a bird warbling[2], and the "Éffet d'orage", a thunder sound in some French organs of the nineteenth century[3].

One characteristic of musical instruments is the expression control. The performer is trained and gives a lot of importance to this matter. The expression is fundamental for the composer as well. Modern instruments have a developed written code to denote the different kinds of expression in the music staff; often composers even select their own words to describe the expression needed to certain musical piece or section of it. A musical instrument offers a wide range of expression. Going back to our example of the pipe organ, we can find also many kinds of expression pedals to control the volume.

1.2 Summary of the contents

In this thesis work there will be emphasis in the musical instruments problem of design and construction in order have a clear idea of what is being

[1] http://en.wikipedia.org/wiki/Pipe_organ
[2] ibid.
[3] ibid.

looked after when designing a virtual instrument with physical modeling techniques. These models have been developed thanks to advance in computer technology and its ability to simulate jet engine turbulence or create a weather forecast[4. In other words, simulation of “multi-domain physical systems, such as those with mechanical, hydraulic, and electrical components[5].” Applications like Simulink and Simscape from Mathworks can be used for a variety of automotive, aerospace, defense, and industrial-equipment applications[6].

In Chapter 2 there will be a review of what has been done in physical modeling instruments for commercialization, departing from the development of sound synthesizers. Also, we will analyze why these instruments may be the next step in musical instrument development. Chapter 3 talks about the techniques that make possible to simulate the acoustic characteristics of instruments in the digital domain. Later, we will review the acoustics of the vibrating string with two ends. Then, the focus will be on the waveguide synthesis, reviewing the Karplus-Strong Algorithm for plucked string instruments with an overview of an implementation in Csound for a plucked bass and flute simulation.

Chapter 4 will be dedicated to the problem of the articulation in physical modeled instruments. Also, there will be a brief analysis of the controllers available in the market and the need of these controllers to enhance the expression of music.

Experimental results on Chapter 5 contain two experiments carried out for this paper. One is a modification of the Karplus-Strong Algorithm from the STK Percolate series of physical modeled instruments using Max/MSP. The second implementation is performed using Csound, which is a sound rendering application. Here we made other modifications of a plucked bass and create a cello like sound.

Chapter 6 shows the conclusions of this research, mentioning the importance of knowing the advantages and disadvantages of the waveguide

[4]Chafe, Chris.
[5] Simulink web page: http://www.mathworks.com/products/simscape
[6] ibid.

synthesis for instrument simulation. Also points out the need of acquiring knowledge in modal synthesis, which has been researched deeply to develop sounds from 3D shapes. Also, we encourage the development of new applications to model acoustic instruments.

Later on Chapter 7, we mention the potential a new research project on traditional instruments like Pipa (琵琶) and or Guqin (古琴) that have not been developed.

1.3 Trends on physical modeling instruments in the commercial world

Taking to the extreme the sound of traditional acoustic musical instruments is being expanded, and the creation of realistic synthetic sounds is touching borders like never. There are many challenges to go trough, as for example the standardization of these designs, to let many composers to have access to a powerful instrument and compose pieces for posterity or on the other extreme open source tendencies for design to let researchers and others to design their own versions of instruments. On this issue, the main electronic music instruments manufacturers like Yamaha, Korg, Roland and others may have a word on the future. Also, there has been a very big production of plug-ins that works with PM techniques to produce sound. Not necessarily only simulation of real music instruments, but for the synthesis of other sounds described by physical systems.

Plug-ins and virtual instruments provide a wide range of options in software synthesis, which can make us say that there is an identifiable trend but on the other hand the use of sensors and external controllers open the door for more control on the sound in the real time synthesis world.

There are many applications to work on real time, for our purpose on Chapter 4 Dynamic Change of Articulation there will be also information on how to deal with Max/MSP external objects with STK (Synthesis ToolKit) and the use of digital filters to control the sound going out of a plucked instrument using the Karplus-Strong Algorithm.

1.4 The acoustic domain

There is another problem that is very important to deal with which is the production and propagation of sound. Electronic instruments have had a reputation of producing non-realistic sounds. There is still a great deal of improvement of the sound they emit. In this present work there will be a proposal for a solution, which consists to let the sound go out from the resonant box in case of the guitar instead of emitting the sound from a pair of speakers separated from the instrument. In this way we think that the performer will feel these sounds resonating along with his body as in an acoustic instrument.

Another problem to mention in physical modeling instruments is that, in the example of the guitar, the definition of the timbre not only comes from the simulation of the dynamics of the string, but also from the resonant box (guitar body) and other parts of it. An instrument is constructed of several parts of different materials and even the characteristics of the material alter the spectra. The study of the acoustics of the instrument comes from the coupling and interaction of all these components. This complexity must be considered as a true vibrational behavior that depends on the transient[7]. The study of musical instruments should not been seen as a static one, because the music played by them is not like that. A static vision would be the one that described the instruments based on their eigenfrequencies or their frequency dependent properties concerning the dissipation of the sound[8].

An approach to musical instruments apart of the string's transverse and longitudinal movement acoustics is basing it on the study and synthesis of the initial transient. The transient in acoustics is the "rapidly changing initial attack or

[7]Bader, Rolf.
[8] ibid.

final decay portions of an otherwise steady sound" [9]. In the case of the acoustic guitar, which is a plucked instrument, not only the transient analysis, but also the guitar body modes and discrete instruments models with finite-element analysis have been applied[10].

Apart of the research mentioned above the physical modeling of instruments have second stage which is the conversion from the analog to digital.

1.5 The digital domain.

It is not until the Karplus-Strong algorithm and the additions by Julius O. Smith and Jaffe and the improvement of DSP hardware and native processors in the late 1980's that the implementation of these models was possible. The method called digital waveguides consumes even more resources because of the need to integrate digital filtering and non-linear elements[11]. However, implementations have been developed on the Yamaha VL-1 synthesizer and other hardware and software synthesizers. This opens the door to develop more complex models in the future justifying the need of constant research on this area.

[9] Hall, Donald.
[10] Smith III, Julius O.
[11] Ibid.

2. Traditional Sound Synthesis and Physical Modeled Instruments.

2.1 Historical background of Sound Synthesis

As mention in Chapter 1 the search of new sounds in musical instruments has been a constant in history. This development involves the participation of inventors with technical knowledge in acoustics and also musicians. This relation still happens in the modern academic research centers and in the industry of musical instruments.

The design and construction of instruments is a profession. In our case, as researchers we follow the same path. As far as concerns the theory behind musical instruments and its refinement is a specialized task. The physical modeled instruments are not the exception. The knowledge on mathematics, acoustics and digital signal processing is required to understand and develop these instruments. But, before we get to this part, we will focus briefly on the development of the electronic instruments and the computer music, also in what happens in the industry.

2.1.1 Analog Synthesizers

We can considered as very important the appearance of the "Musical Telegraph" 1876 by Elisha Gray who accidentally discovered that he could control sound from a self vibrating electromagnetic circuit and in doing so he invented a basic single note oscillator. Oscillators are being used to produce sounds in electronic instruments; this is not the case of the physical modeled instrument.

Skipping a few years ahead we should mention the RCA Mark II Sound Synthesizer (nicknamed Victor), which was the first programmable electronic music synthesizer. This was part of the equipment at the Columbia-Princeton

Electronic Music Center by the 1950's. Famous composers like Milton Babbit programmed music using this device.

The RCA consists of many electronic sound generations assisted by a sequencer. This attracted many composers who wanted to have more flexibility on the generation of sounds, which formerly was based on the manipulation of separated recorded sounds in tape and the splice the together to create a musical piece.

The RCA's sequencers use a paper tape reader to provide instructions to the synthesizer playback. Then an output sound is generated and then compared against to the score. This process could be repeated until getting the desired results[12]. This machine represents a milestone in electronic instruments, providing more complex possibilities to generate sound and compose music.

The size of the Mark II is the one of a big room full of machines, also, definitely not affordable for individual musicians. The next generation of synthesizers would mark the trend in size and price for these instruments. The next member of the family was the Buchla and the Moog synthesizer.

The Buchla Modular synthesizer created by Don Buchla in the early 1960's is considered one of the first commercial synthesizers. The Buchla synthesizers contain oscillation, filtering, sample and hold to have an effect on the pitch and timbre, amplitude and spatial location of the sound[13] facilitating the process of creating new sounds and therefore new instruments in one single device. These methods marked the way music making a new era in musical instruments.

Later the Minimoog Model D created in 1971 Robert Moog and his company Moog Music. This was the first widely available, portable and relatively affordable synthesizer. By this time synthesizers were monophonic, in other words only one note at the same time can be played. Later Polyphonic synthesizers as the Polymoog, Prophet-5, Coupland Digital Music Synthesizer

[12] http://en.wikipedia.org/wiki/RCA_Mark_II_Sound_Synthesizer
[13] http://en.wikipedia.org/wiki/Buchla

appeared[14], increasing the capacity of these instruments and attracting more musicians. All the machines mentioned above are of the family of the analog synthesizers, which was the preview of the next revolution on electronic music instruments, the digitally produced sounds.

2.1.2 Computer sound synthesis and digital synthesizers

Computer synthesis of sound appeared thanks mainly to Max Mathews. On his paper called "The computer as a musical instrument" [15. Based on the Bell laboratories, Mathews attracted the attention of composers like John Chowning and Jean Claude-Risset and Vladimir Ussachevsky. A new generation of sound production started thanks to the development of computer music program called Music V by Mathews and Joan Miller[16].

These advances will point to the portable digital synthesizers that will change things forever in the electronic music. Not only from the technical point of view but also from the manufacturing which happens in a different place, Japan.

Yamaha Corporation, a company founded in 1887 started manufacturing reed organs and pianos is currently the biggest musical instruments producer in the world. Yamaha leaded the revolution of the digital synthesizer [17]. The synthesizer "YAMAHA DX7" works on FM synthesis techniques developed by John Chowning at Stanford University in 1967.

In 1973 Stanford licensed the discovery to Yamaha, with whom Chowning worked in the development of a synthesizers and electronic organs. This patent was Stanford's most lucrative patent at one time, eclipsing many in electronics, computer science, and biotechnology's patents[18].

The DX7 starts the revolution of digital synthesizers selling thousands of units. This same technology is also in use in computer sound cards as in many

[14] http://en.wikipedia.org/wiki/Synthesizer
[15] Risset Jean-Claude.
[16] ibid.
[17] http://en.wikipedia.org/wiki/Yamaha
[18] http://en.wikipedia.org/wiki/Synthesizer

other devices such as a new product line YMF795-EZ 24SSOPAPL-2 FM sound synthesizer for automobile[19] developed by Yamaha.

This led us to mention who are the main players in the electronic music instruments business Yamaha, Korg and Roland. These corporations contribute to the development of new products for the market. The product line and the company profiles of these enterprises is not part of the scope of this investigation. The next step is to mention the birth and development of the physical modeled instruments.

2.1.3 The imitation of real instruments through sound synthesis and physical modeling.

In terms of sound synthesis there has been always an intention to imitate real instruments. For example Chowning and Jean-Claude Risset had and extensive research on the instrument timbre design like sounds using FM Synthesis and waveshaping. Their approach is based on the comparison of a produced sound with the real instrument and analyzed by a spectrogram in order to estimate the envelopes of the several harmonics of a sampled instrument sound and not from a physical modeled approach. In the spring of 1969 Risset put together his work on music synthesis in a document called "An Introductory Catalog of Computer-synthesized Sounds". In this document the French researcher-composer intention is to establish a point of departure for developing timbres and sonic processes[20].

Later in 1979 McIntyre and Woodhouse (1979) proposed a computational model for vibrating strings based on physical reasoning. This model developed to be applied on bowed strings simulations. [21] In 1983 Kevin Karplus and Alex Strong took a one of these simulations that has a realistic tone and started to work on improvements of it. These extensions are detailed in the paper called

[19] http://www.yamaha.co.jp/english/product/lsi/prod/sgl/index.html
[20] Risset Jean-Claude.
[21] David A. Jaffe and Julius O. Smith.

Extensions of the Karplus-Strong Plucked-String Algorithm. On this modeled "the availability of multiplies, allows several modifications and extensions that increase it usefulness and flexibility" [22].

"Music synthesis based on physical modeling is currently the highest quality imitation of music instruments because is possible of having the same control parameters as a real instrument" [23].

There has been a constant develop on these techniques. More details on the schemes of physical modeling synthesis will be found in Chapter 3.

There have been many applications available for musicians such as Modalys from IRCAM and Perry Cook's C++ STK (Sound Tool Kit) software with a complete documentation on his book "Real Sound Synthesis for Interactive Applications" show some practical examples to implement different models, some of them based on modal synthesis.

On the other side, Yamaha developed the VL-1 Synthesizer based on the patents from Stanford's CCRMA in 1994 on digital waveguides. Since then, many other commercial devices and Virtual instruments VST have been produced in the commercial world.

Composers like Hans Tutschku, Paul Lansky, Claude Cadoz, Andrew Schloss, Matthew Burtner, David Jaffe, Achim Bornhoeft, Juan Reyes, Chris Burns and Torsten Belschner have composed pieces experimenting with the physical modeling instruments. More details on the paper called Physical modeling and composition by Chris Chafe 2003[24].

2.2 Comparison between traditional models and Physical Modeling.

2.2.1 Reviewing the traditional sound synthesis techniques.

[22] Ibid.
[23] Smith III, Julius O.
[24] Chafe, Chris.

Below a classification of sound synthesis:

- Frequency Domain
  - Linear
    - Additive Synthesis
    - Subtractive Synthesis
  - Non Linear
    - Amplitude Modulation
      - Ring Modulation
      - Waveshaping
    - Frequency Modulation
- Time Domain
  - (Non-Linear)
    - Granular Synthesis

2.2.1.1 Granular Synthesis,

Another method is the *Granular Synthesis,* which is a Non-Linear Time-Domain technique. Proposed by Gabor (1947) who said that signals could be created from the combination of short sonic grains[25]. The duration of the grains is usually between 10 to 60 milliseconds. In this category oscillators can be used to generate sounds. To implement this, grains (elementary sounds in computer music) are organized into frames and its parameters are updated in time. There are two axes in each frame (Amplitude and Frequency). When these frames are played a set of grains with varying frequency and density is heard[26]. Another way to work with grains is by scattering them within a mask[27]. The amount of data used in granular synthesis is considered as a disadvantage when it comes to its implementation. Also its output does not come from analyzing natural sounds as in the additive.

[25] Gabor, D.
[26] Rajmil Fischman.
[27] Giovanni de Poli.

2.2.1.2 Additive Synthesis

In the case of *additive synthesis* the sounds are generated by an addition of sinusoidal oscillators where the amplitude and sometimes the frequency is time varying. Additive synthesis can be based on a Fourier analysis of a real sound of musical instruments. This analysis assumes that every sound is a sum of sines or cosines. The mathematical expression is:

$$s(t)=\sum_{i=1} A_i \sin(2\pi f_i t + \theta) \qquad (2.1)$$

Where A is the amplitude, $2\pi f$ is the angular frequency (ω) and θ is the relative phase angle. This angle is determined by the time of origin of the signal. For example $\sin(\omega t)$ and $\cos(\omega t)$ are shifted. Their difference relies on the fixed phase angle $\pi/2$.

$$\cos(\omega t) = \sin(\omega t) + \pi/2 \qquad (2.2)$$

This way the energy of each partial is determined an imitated using the oscillator and modifying its amplitude envelop then added up together. The problem arises with the big amount of data and precision needed to create convincing sounds because that the more sinusoids added up the more complex the sound can be.

2.2.1.3 Subtractive Synthesis

Subtractive Synthesis is a complex signal rich in harmonics like white noise of a pulse of trains, which passes through one or more Digital Filters.

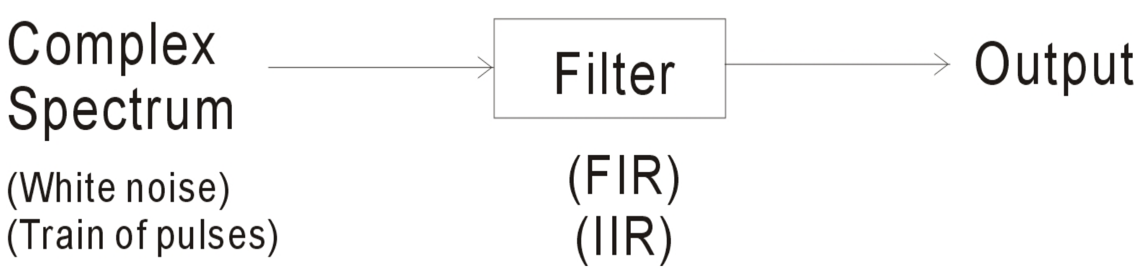

Figure 1: Subtractive synthesis

The manipulation of the parameters of the filters result in desired possibilities of sound by eliminating or emphasizing certain frequencies using the FIR or IIR filters.

Subtractive methods are effective in the simulation of musical instruments, especially wind instruments where the excitation contains a lot of harmonics. Also resonance of the instruments can be obtained with the use of resonance filters. Another application of these techniques is the simulation of the human voice used in speech research by changing the resonance of the filters according to the vocal tract[28].

2.2.1.4 Amplitude Modulation

Amplitude Modulation is the multiplication of two signals.

$$s(t) = x_1(t) \cdot x_2(t) \qquad (2.3)$$

$$s(t) = \sin(2\pi f_m t)\,\sin(2\pi f_c t) \qquad (2.4)$$

Where f_m is the modulator which modifies the amplitude of the carrier f_c. Amplitude modulation would be to modify the amplitude of a signal by another signal using an oscillator instead of a constant value. This modulation produces two additional frequencies known as sidebands, forming harmonic or non-harmonic sounds. [29]

2.2.1.5 Waveshaping

On the waveshaping synthesis a transfer function distorts the incoming waveform into a different shape.

[28] Smith, Julius O.
[29] http://www.sonicspot.com/guide/synthesistypes.html

Figure 2: Waveshaping Synthesis

This new shape depends three things:

- Shape of the incoming waves. (Its amplitude is called waveshaping index)
- The transfer function
- The amplitude of the incoming signal.

Depending of the value of the amplitude index and on the transfer function the result will be a *distortion* creating in some cases a complex spectrum.

2.2.1.6 Frequency Modulation

The last method to comment is the Frequency Modulation. It allows the production of spectral complexity by using the modulator frequency to modify the carrier frequency with low computational effort. It most simple form is:

$$s(t) = \sin\,[2\pi f_c t + I\sin(2\pi f_m t)] \qquad (2.5)$$

$$s(t) = \sum_{k=-x}^{+x} j_k(I)\sin\left|\, 2\pi(f_c + K\, f_m\, t \,\right| \qquad (2.6)$$

Where I is the modulation index, which controls the dynamic spectra. [30]

The list below mentions some synthesis techniques. There are also many extensions or variations of them not included here. For example, Frequency

[30] de Poli, Giovanni.

Modulation Techniques have some extensions like the Non-sinusoidal carrier method, the Compound Modulation, Nested or Complex Modulation, Two-Input and Non-linear Transformation[31].

•Subtractive synthesis
•Additive synthesis
•Granular synthesis
•Wavetable synthesis
•Frequency modulation synthesis (FM)
•Phase distortion synthesis
•Physical modelling synthesis
•Sample-based synthesis
•Subharmonic synthesis. [32]

2.2.2 Physical modeling techniques vs. Traditional Techniques.

S. Petrausch from the University of Erlangen-Nuremberg says in that Traditional methods are tools to model sound. They are used to create new sounds, but have several disadvantages in imitating sounds of real acoustic instruments[33].

On the other hand Julius O. Smith states in his paper "Physical Modeling using Digital Waveguides" that physical modeling instruments "promise the best quality of highest quality" in the imitation of real instruments. "Because the artificial instrument can have the same control parameters as the real instrument, expressivity control is unbounded". For example in a plucked string model many parameters as the stiffness, the position of the pluck and others can be simulated. Or in the flute model the *air pressure* is also imitated and can be controlled by the performer.

[31] ibid
[32] http://en.wikipedia.org/wiki/Synthesizer
[33] S. Petrausch
http://www-nt.e-technik.uni-erlangen.de/lms/research/projects/soundsynthesis/index.php?lang=eng

Also, the dimensions and shape of the body of the instrument can be modified, and eventually the material of which the instrument is constructed as well. These simulations are not possible when using the traditional sound synthesis because their formulation does not coincide with the instruments physics.

Another advantage over the traditional techniques is that the control of the parameters can point to simulate sounds not possible for real instruments, such as manipulating the decay of a pluck string to have duration of one minute or more. Or even in the flute model can be modified to imitate a flute with extreme proportions. For example, composer Paul Lansky has been using Perry Cook's flute physical model to create a flute that is 20 feet long and with a diameter or 3 feet producing interesting sounds. This emulation can be heard piece "Things She Carried" (1996) waveguide, STK Flute[34].

2.3 Justification to use Physical Modeling.

These and other reasons justify the use of physical modeling. In the section of Future Works of this thesis will summarize the future of musical instruments based on these synthesis techniques and show how people around the world is having much more interest to have in their hands more developed applications. Meta instruments controlled by sensors or simply new devices emulating real or imaginary instruments happen to be a trend. Gary Scavone in his piece Pipe Dream (2003), performs an algorithm with a MIDI wind controller called the Pipe. This device uses a set of sensors including buttons, potentiometers and accelerometers that respond to breath and finger pressure and tilt[35].

On the other side, with physical modeling synthesis hybrid of instruments are possible, for example imaging a guitar that has the parameters of a flute multiplying the possibilities of sounds.

[34]Chafe, Chris. Case Studies of Physical Models in Music Composition. Center for Computer Research in Music and Acoustics. Stanford University.
[35] ibid.

3. Physical Modeling Techniques.

3.1 Physical modeling approaches and schemes

Based on Castagne and Cadoz[36] there are 5 major types of physical modeling-based schemes: Traditional approach, Mass interaction Modular Scheme, Waveguides, Modal scheme, Non-Linear Source/Filter and Dynamic Non-Linear and Black-Boxes Approaches.

3.1.1 Traditional approach.

This scheme is proposed under the classic acoustic theory, used primarily by physics specialist to describe the behavior of the systems constructing a continuous model, which is later, discretized by numerical analysis. Bader (2005) for example works on “a new paradigm in instrument acoustics” which is “based on time-dependent transient analysis and simulation of complete musical instruments” [37]. At a later stage, he uses a computer model to demonstrate the transient synthesis. This software called Musical Transient-Modeling Software MTMS simulates the Classical Guitar and allows users to change the parameters of various settings of this instrument and hear their result.

A model based on the traditional approach helps physics, instrument makers, teachers and instrumentalists to understand the mechanics behind a musical instrument rather than produce music.

3.1.2 Mass interaction Modular Scheme

Mass interaction schemes (Cadoz 1979) [38] are used to models objects like strings, plates, wind resonators and others trying to imitate them. It works by

[36] Castagne, Nicolas; Cadoz, Claude.
[37] Bader, Rolf. Computational Mechanics of the Classical Guitar. Springer 2005.
[38] Cadoz C.

joining together masses and physical and linear and non-linear interactions. This approach is adequate to create music by a musician. [39]

3.1.3 Digital Waveguides

This approach seems to be most popular one. Also is being used in the manufacturing of the commercial synthesizer Yamaha VL-1. Waveguides simulate oscillating objects, which are later passed through resonators.

Proposed by Julius O. Smith. This method consists on a solution of the wave equation in a general way to get "traveling waves in the medium interior" [40]. These waves are simulated in the waveguide model, summed together produce a physical output. In lossless cases, digital delay lines simulate traveling waves between two points. In the linear case the simulation is carried by multiply free delay lines saving computational power[41].

The disadvantage is that it has no "non-physical based control of the transfer function coefficients" [42]. In addition, "it does not seem very efficient when the goal is to let a musician handle the scheme at a basic level" [43].

3.1.4 Modal Scheme

Also known as the spectral approach. Using oscillators with coupling data to model a mode of the structure. It is less efficient than the Digital Waveguides. It works simulating oscillating objects connected to linear resonators. It is similar to the additive synthesis. There is a software implementation called Modalys developed at IRCAM. "Modalys is used for creating virtual instruments based on simple physical objects such as strings, metal plates, tubes, membranes, plectra,

[39] ibid Castagne, Nicolas; Cadoz, Claude.
[40]Smith III. Julius O.
[41] ibid.
[42] S. Petrausch
http://www-nt.e-technik.uni-erlangen.de/lms/research/projects/soundsynthesis/index.php?lang=eng
[43] ibid Castagne, Nicolas; Cadoz, Claude.

reeds and hammers." [44] Recent versions can be controlled in Max/MSP environment.

3.1.5 Non-Linear Source/Filter and Dynamic Non-Linear and Black-Boxes Approaches.

This model is fully based on physical modeling approach. Its algorithms are based on signal source and a numeric filter coupled by a non-linear retroaction function[45]. It has good precision for certain uses as in wind instruments.

3.2 Some Physical Modeling Implementations (PerColate and Csound)

The waveguide model will be discussed in detail in order to have a basic approach for musician-users. As seen in the previous section, various schemes can be approached to simulate instruments. We intend just give a quick overview on them. More details on these models can be found in Castagne and Cadoz article on the evaluation of PM schemes.

Various software programs, as mentioned before, are utilized to implement PM instruments, we will mention the two applications used in our research PerColate and Csound.

3.2.1 PerColate

PeRColate external objects for Max/MSP, which is an open source code developed by Dan Trueman at Princeton University and R. Luke DuBois at the

[44] Modalys
[45] Rodet X, Depalle P, Fleury G & Lazarus F: "Modèles de signaux et modèles physiques d'instruments, études et comparaisons" – Colloque Modèles Physiques, Création Proc. of the 6th Int. Conference on Digital Audio Effects (DAFX-03), London, UK, September 8-11, 2003 Musicale et Ordinateurs – Grenoble 1990 – Edition de la Maison des Sciences de l'Homme – Paris, 1994.

Computer Music Center, Columbia University. The PeRColate has a port of STK (Sound Synthesis Tool Developed by Perry R. Cook).

STK consists of an easy to use library of synthesis and signal processing functions in C programming language that can be wired together to create conventional and unusual instruments.

On the other side STK includes several synthesis objects like

- Physical Modeling
- Modal Synthesis
- PhISM class instruments (Physically Informed Sonic Modeling to simulate percussion instruments).

The code for these instruments can serve as foundations for creating new instruments as well and can be used to teach elementary and advanced synthesis techniques. [46]

3.2.2 Csound and waveguide instruments

Csound is an application for rendering signal processing developed by Professor Barry Vercoe. Csound opcodes are built-in subroutines to perform different tasks. These tasks are oscillators, filters and table-generating functions. Csound also has several opcodes coded by users and available within the program. In other words, new tasks can be programmed using C and then call then in the Csound compiler. For example, many opcodes for waveguide physical modeling have been included in to work on:

- pluck
- repluck
- wgbow
- wgbowedbar

[46] Percolate Web Page.

- wgbrass
- wgclar
- wgflute
- wgpluck
- wgpluck2
- wguide1
- wguide2

These opcodes contain parameters to control the waveguide, for more information see *Csound* manual. Their code can be seen and even modify using *C Programming Language*. In this presentation we will discuss the waveguide models proposed by Hans Michelson in the Csound Book[47]. The use of Csound facilitates the learning of sound synthesis from a rendering approach. For live synthesis Max/MSP, Pd and Supercollider are commonly used.

3.3 Physical Modeling Basics (Acoustics and Digital Waveguides)

3.3.1 String Instruments

There are three ways to excite a string. Plucking, bowing and striking. The following table mention a few string instruments classified based on it excitation:

Plucked String	**Bowed String**	**Struck**
Guitar	Violin	Piano
Pipa (琵琶)	Cello	Sheng
Guqin (古琴)	Erhu (二胡)	Yangqin (揚琴)

[47] Michelson, Hans.

Guzheng (古箏)		Cimbalom
Mandolin		
Sitar		
Oud		
Lute		

In order to have an idea of their physical behaviors during the vibration period we must understand the natural modes and the mixture of them.

3.3.2 The Ideal string

The strings of an instrument are tight between two ends. The thickness and tension of these strings determine their pitch. Modes can be observed by analyzing the strings from a transverse point of view. The modes are particular *standing waves* pattern that move in simple harmonic motion at the frequency. It has no resistance to bending or zero stiffness [48]. Then the restoring force make the string return to the equilibrium state.

During the *excitation* the nodes are generated. Nodes are a point in a standing wave at which a physical variable remains constant in spite of its time variation at the neighboring points[49].

A vibrating string consists of two *traveling waves*. One goes to the right and the other one to the left. Adding these two waves produce the standing wave of the third natural mode.

Modes have the shape of sine waves with the nodes evenly spaced between the two ends of the string.

[48] Hall, Donald.
[49] Ibid. p 461

In the pluck string system the position of the pluck is also important. The excitation of any mode is proportional to how much motion that mode has at the plucking point. For instance, when plucking in the middle of the string would produce a strong fundamental component in the sound favoring modes 2, 4 and 6. Plucking at one fourth of the string modes 2, 6, and 10 will have more intensity[50].

Another thing to consider is the propagation of the sound of the string, which is facilitated by the body of the instrument. In order to study the string motion, the electric guitar offers the easiest way. The *pickup* detects the sound only of the string without any interference of the body's reflections.

The *waveguide model* for plucked strings simulates the motion of the string using delay lines. On a later stage we must add resonance filters to emulate the body of the instrument.

Displacement of the string has to be considered in the model as well. This model is based on the wave equation of the ideal string:

$$y(x,t) = \sum_{k=1}^{\infty} c_k e^{jk\omega_0 t} \sin(k\pi/L) \qquad (3.1)$$

Where "y" is the transverse *displacement* of the string respect to its resting position. And x =0 and x=L represent the both ends of the string. **Figure:**

[50] Ibid.

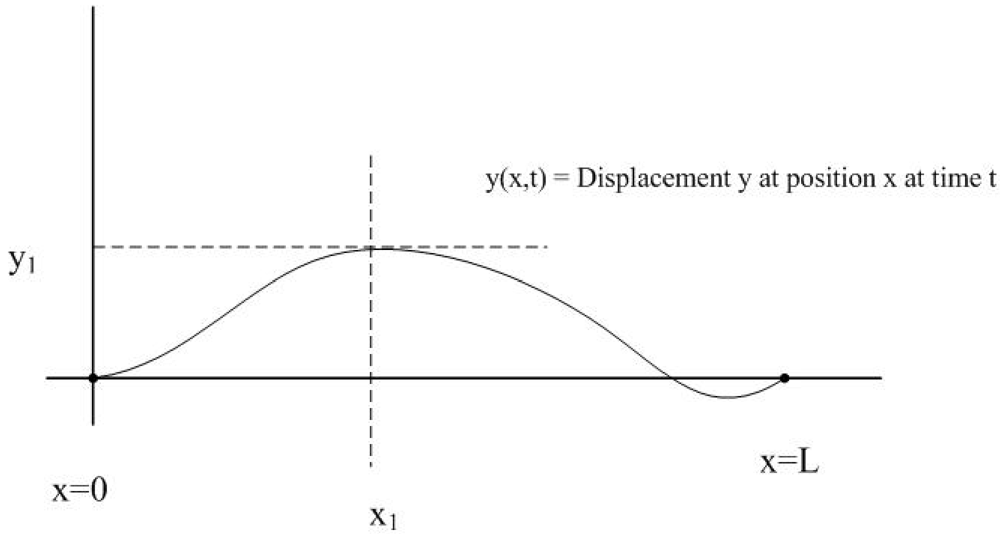

Figure 3: Displacement on a vibrating string

We depart from the condition that the displacement is zero at x=0 and x=L

The same equation can be applied in the vibrating column of air model that can be used for clarinet model and pipe organ simulation; this can be accomplished by substituting air pressure deviation (ξ) for string displacement.

3.3.3 Karplus-Strong Algorithm for plucked strings

The Karplus-Strong Algorithm (Karplus and Strong 1983) with the extensions of Jaffe and Smith (Jaffe and Smith 1983) works with the general solution of the dynamics of reflections of waves in a vibrating string fixed at two ends or wave-fronts in vibrating columns of air:

$$y(x, t) = f_r(t - x/c) - f_\ell(t + x/c) \qquad (3.2)$$

Where c is the speed of the traveling wave, and t is the time. Since this is a function with two variables. To know the value of either x or t, we will need to make one of them a fixed value. On the other hand, the function y= $f_\ell(t + x/c)$

denotes the wave traveling to the left and its counterpart will be the wave traveling to the right $y = f_r(t - x/c)$.

To simulate this vibration we need to sample the traveling waves. So we substitute in the equation the time and have number of samples. In other words sample the traveling wave amplitude at intervals of T seconds with a sampling rate of $f_s = 44.1$ kHz for CD quality sound.

$T = 1/f_s$.

Delay lines corresponding to the traveling waves accomplish this simulation. Then we substitute:

$x \rightarrow x_m = mX$

$t \rightarrow t_n = nT$

Where X is the spatial sampling interval, which is the distance sound propagates in one temporal sampling T[51].

Then obtain from the continuous traveling equation[52] :

$$y(t_n,x_m) = y_r(t_n - x_m/c) + y_l(t_n + x_m/c) \quad (3.3)$$

$$= y_r(nT - mX/c) + y_l(nT + mX/c) \quad (3.4)$$

$$= y_r[(n - m)T] + y_l[(n + m)T] \quad (3.5)$$

Factorizing T we get:

Right traveling wave component propagating to the right: $y^+(n)$ defined as $\mathbf{y_r(nT)}$

Left traveling wave component propagating to the left $y^-(n)$ defined as $\mathbf{y_l(nT)}$.

Then $y_r[(n - m)T] = y^+(n - m)$ is the output to an *m*-sample delay line whose input is $y^+(n)$. This is delaying the signal by *m* samples.

On the other side, $y_l[(n + m)T] = y^-(n + m)$ is the input to an m-sample whose output is $y^-(n)$. So by adding *m* to the time *n* the signal is advancing one *m*-sample.

[51] Smith III. Julius O. S
[52] ibid.

The resulting waveform will look like this[53]:

$$y(t_n,x_m) = y^{+}(n - m) + y^{-}(n + m) \tag{3.6}$$

3.3.3.1 The real vibrating string

The energy loss is caused by the air and internal friction within the string. The loss varies with the frequency in a complex way. The right and left traveling waves decay exponentially in their respective direction. Loss increases with the frequency. In the frequency-dependent loss model distributed factors between samples together become a *resultant loss factor.*

To emulate the decay of harmonics, Smith and Jaffe propose that the partial (when the harmonics are not uniformly spaced) should enhance the possibilities of *inharmonicity*. Therefore a partial at frequency f Hz circulates in the loop of the equation. In each one of the periods through the loop should be attenuated equally to the loop-amplitude response given by the loop gain.

In the digital waveguide implementation an average in the form of a low-pass filter allows the higher frequencies to be attenuated before the low ones, emulating the decay of the real strings. Smith and Jaffe suggest methods to alter the decay by making it short or by stretching it[54].

3.3.3.2 Tuning the waveguide plucked string instrument.

Tuning the waveguide instrument can be as difficult as in real instrument. Because the delay line length should be an integer number cause problems on tuning. To solve the problem the loop length should be written in terms of the phase delay. Following, there is a need to include an all-pass filter that has a small delay that does not interfere with the overall loop gain[55]. Tuning all the

[53] ibid.
[54] David A. Jaffe and Julius O. Smith.
[55] ibid.

harmonics with this filter is not possible, plotting the phase delay we can observe that the upper harmonics show higher relative delay than the fundamental [56].

3.3.2.1 Summary

The Pluck String filters result as shown in the figure, a burst of white noise rich in harmonics. Then a comb filter (Feedback filter) resonate the partials and a Low-pass filter to simulate the decay. Also after the LP filter, an all pass filter to keep the instrument in tune. (FIGURA).

3.3 Waveguide bass (A Csound Implementation)

Mickelson[57] propose an easy implementation of the waveguide model using the Karplus-Strong algorithm in the Csound book.

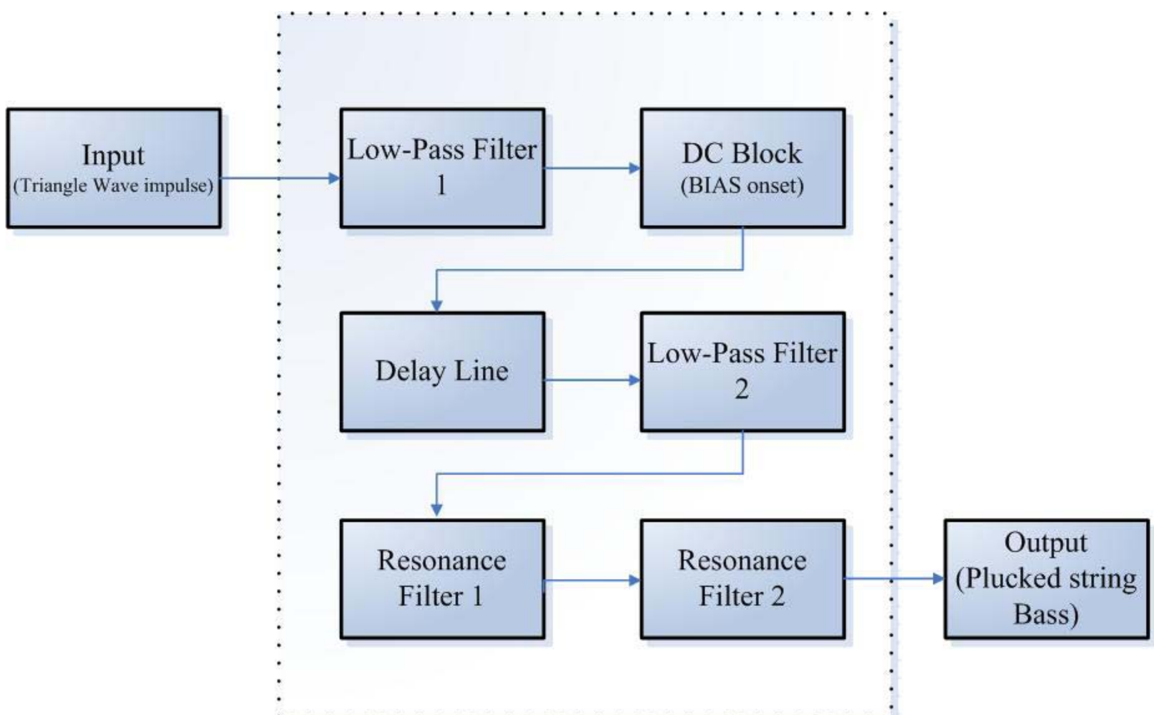

Figure 4: Waveguide Bass (Michelson, Hans)

In this model the excitation is done by a filtered triangular wave. The linseg (Line-segment shows the shape of the initial burst). Then the output is passed through a Low-Pass Filter (`tone`) The Low-pass filter in Csound (`tone`) implements a first-order recursive low-pass filter in which the variable khp (in

---

[56] Steiglitz, Ken.
[57] Michelson, Hans

Herz) determines the response curve's half -power point. Half power is defined as peak power / root 2 (see Csound manual). In this case the value for khp is 200Hz.

```
; INITIALIZATIONS
ifqc      =        cpspch(p5)
ipluck    =        1/ifqc*p6

kenvstr  linseg   0,ipluck/4,-p4/2,ipluck/2,p4/2,ipluck/4,0,p3-ipluck,0
aenvstr   =        kenvstr
ainput    tone     aenvstr,200
```

Changing the parameters of the p6 variable can change different pluck styles.

Then a DC blocker filter keeps the BIAS onset of the signal.

```
ablock2   init     0
ablock3   init     0
ablock2   =        afeedbk-ablock3+.99*ablock2
ablock3   =        afeedbk
ablock    =        ablock2
```

To tune the instrument the output of the DC blocker is passed through a delay line with 1/frequency of length with a subtraction of 15 samples to bring it into tune. The delay opcode in Csound takes the input and apply an "idlt" which is the requested delay time in seconds. The space required for n seconds of delay is 4n * (sample rate bytes).

```
adline    delay    ablock+ainput,1/ifqc-15/sr
afiltr    tone     adline,400
```

Then the resonance is added to simulate the body of a bass:

```
abody1    reson    afiltr, 110, 40
abody1    =        abody1/5000
abody2    reson    afiltr, 70, 20
abody2    =        abody2/50000
```

This resonance filter in Csound (`reson`) is a second-order filter in which kcf controls the center frequency, or Hertz position of the peak response (the value of kcf above is 110Hz and 70Hz respectively) , and kbw controls its bandwidth (the Herz difference between the upper and lower half -power points) The values for kbw are 40Hz and 20Hz).

These are some samples of the sound, se how clearly the decay is being successfully simulated[58].

3.4 Slide flute

The following instrument has been taking from the Perry Cook's Slide Flute. The input of the system in this case is an airflow emulated by a noise burst.

Figure 5: Slite Flute (Perry Cook)

Two delay lines represent the feedback of this scheme. The first one is for the "embouchure" of the air jet and the other one for the flute bore. The embouchure delay must be equal to ½ of the length of the flute bore.

```
atemp1    delayr    1/ifqc/2
ax        deltapi   afqc/2
          delayw    asum2
```

Then a cubic equation $x-x^3$ simulates the interaction of the embouchure and the flute bore.

```
ax        delay asum2, 1/ifqc/2; Embouchure delay
apoly      =    ax - ax * ax * ax   ; Cubic equation
```

[58] ibid

A low-pass filter simulates the reflections of the flute bore at low frequencies. Then it is delayed and feedback before the embouchure delay. Changing the length of the bore delay line changes the pitch:

```
afqc =    1/ifqc-asum1/20000-9/sr+ifqc/12000000
```

As in the plucked-string model we also need to tune the flute including and interpolating delay tap:

```
atemp2    delayr    1/ifqc
aflute1   deltapi   afqc
delayw    avalue
```

Breath pressure can vary the pitch in a slight way. This allows a vibrato effect in the instrument. We just need to add to the delay tap a term based on the pressure:

```
atemp     delayr    1/ifqc
aflute1   deltapi   1/ifqc-12/sr+asum1/20000
delayw    avalue
```

Overblowing techniques can be added as well, but extra tuning would be required.

To have more examples on waveguide models refer to Mikelson's article in the Csound book. The Clarinet model and a waveguide drum is included as well. [59]

3.4 Conclusion

Implementing the waveguide model, the sound designer will need to have knowledge on Acoustics and Filter design. For example, in order to solve the comb filter to generate the resonance, the designer should have knowledge about the partials and their respective energy during the excitation.

[59] ibid.

Digital Waveguides is a very conventional scheme, utilized by many users in spite of its limitations. It is the most used model and even it has been implemented in commercial synthesizers.

4. Dynamic Change of timbre.

4.1 Articulation in Physical modeling instruments.

4.1.1 Acoustic Viability

The idea of simulating a real musical instrument relies also on the attention for the articulation taken by the designer. This concept is called *Acoustic Viability* (see Beck) [60]. This view gives importance to the expressivity and the timbre of the synthetic instrument. A good design based on these factors will result more interesting for the performer and for the listener.

To accomplish this goal there is a need of understanding the physics of the musical instruments in terms of their articulation. For example, a loud sound from an instrument is richer in harmonics than a soft one. A timbre shifting also happens between the registers of the instruments. This reality applies to several acoustic instruments like the clarinet whose three different registers differ in timbre from one to the other. Similar characteristic is heard on the Flute, Violin, Piano and others.

Therefore, the dynamics and pitch cause the timbre to become complex instead of static. These variations are part of the timbral character of each instrument [61]. Instrument designers must take in count these properties in order to create the conditions that respond to these two variables.

"Analysis and re-synthesis techniques (Linear-predictive coding and phase vocoding), Karplus-Strong Algorithm, FOF synthesis, granular synthesis and physical modeling respond to the analysis methods in order to imitate the behavior of the instruments" [62].

[60] Beck, Stephen David. Designing Acoustically Viable Instruments in Csound. The Csound Book. p.155.
[61] ibid.
[62] ibid.

4.1.1.2 Analysis and Synthesis.

Time-frequency representations of the sounds should be used because musical instruments output non-stationary, and also because the evolution of a sound as a function of time is important from a perceptive point of view[63].

The time-frequency analysis has two parts, the modulus and the phase representations. Spectral line estimation methods and matched analysis methods help to associate one amplitude and one frequency modulation law to each spectral component of the sound. [64]

As mentioned before signal generation models and physical modeling can do the synthesis of sound.

4.1.2 Resonation

Resonance refers to the difference of timbre in every register of the instrument. It demonstrates the importance of the resonation body. Resonators amplify certain frequencies. In the design of real instruments, the selection of materials will make a big difference on the timbre as well. For example, the difference of the timbre of a guitar made of different kinds of wood. Their material characteristic are also involved: homogenous and heterogeneous materials and thickness.

Experiments on the resonation of a guitar plate and violin plate can be referenced for further information (See Bader 2005 for resonance in guitar plates[65] and for Janson 2002 for violin plates[66]). To measure a resonator we need to take in count its frequency response (vibration sensitivity). The response curve gives a picture of each resonance.

Things to take in count for a resonator:

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[63] Ystad, Sø lvi; Guillemain, Philippe; Kronland-Martinet, Richard.
[64] ibid.
[65] Bader, Rolf.
[66] Jansson, Erik.

- Mass (weight),
- Stiffness (spring)
- Friction.

A resonator can be made of a vibrating area, like in the top plate of a guitar.

Maximum vibration happens at the antinodes and the sections with no vibrations are called nodes. Also the decay or reverberation should be measured as well.

In the waveguide physical synthesis the resonance simulation is done by resonance filters.

4.1.2 Amplitude and timbre

The energy applied during the excitation is very important as well. The more the amplitude is, the more the richness of the harmonic spectra.

4.1.3 Performing Articulation

Another problem to take in count when modeling an instrument is to analyze its performing articulation. Some instruments share the same articulations and others are exclusive to certain.

In other words, the technique that a violin player uses to slur notes will be completely different from the technique used by a trumpet player, and a pianist and a vocalist will do different things to make a melody sound legato. In fact, the violinist will have some articulations available (such as pizzicato, or "plucked") that a trumpet does not have[67].

Some common articulations are:

- Staccato

[67] Schmidt-Jones, Catherine. **Ariculation.**

- Slur/ ties
- Legato
- Accents
- Portamento
- Marcato [67]

The behavior of all these articulations means a great challenge for instrument designers. These details can make the physical modeled instrument more similar to the real instrument. Also these articulations in the ideal PM instrument should be attached to the instrument by means of controllers. This is discussion appears in section 4.2.

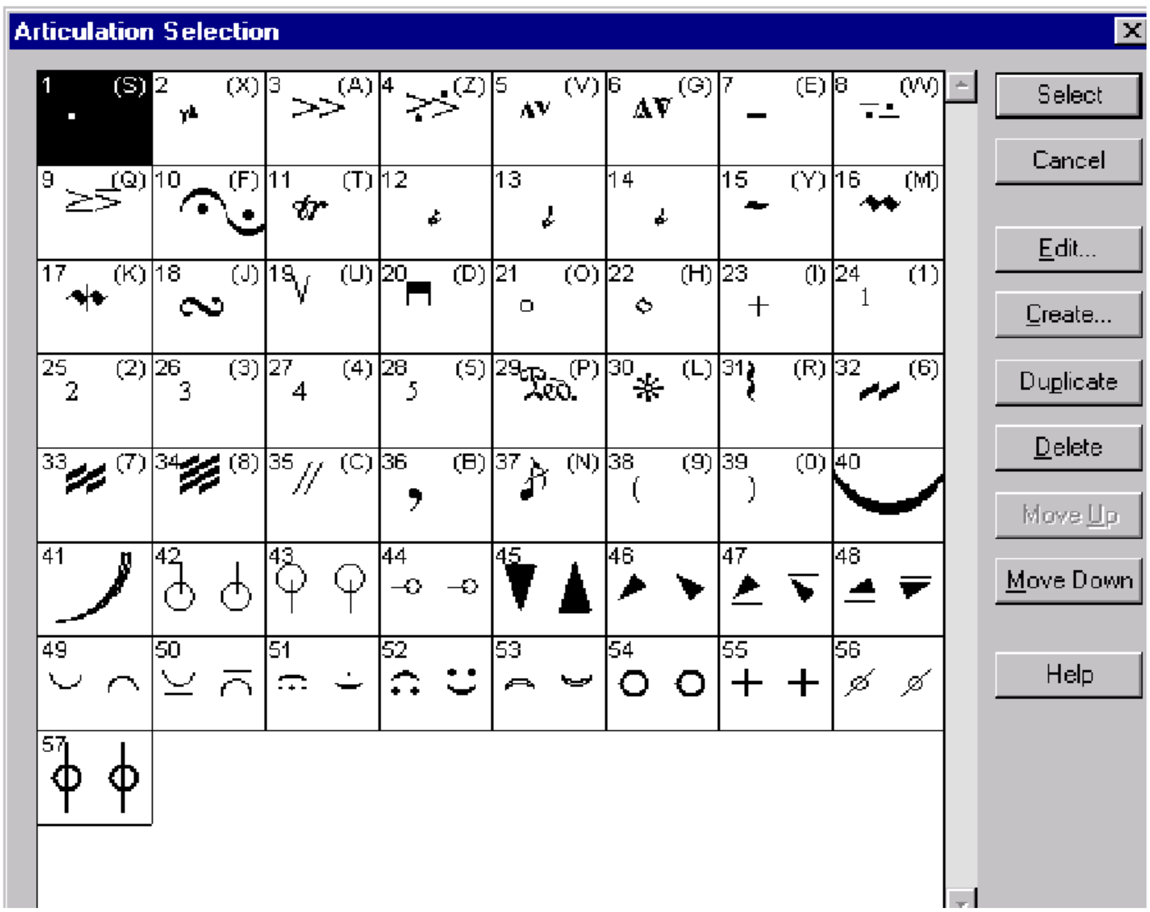

Figure 6: Performance Articulation graphic representation. (Finale Sofwtare 2005 (Make Music! Inc.)

Articulation for different instruments also has a special notation. Above is a chart taken from the software Finale 2005. Note all the different types of articulation proving that a complete simulation of an instrument should have many possibilities.

4.2 Controllers and Sensor for Articulation Control

As discussed in the previous section, a complete instrument would be the one, which can control the articulation in a convenient way. The evolution of the instruments should accompany with interfaces that can provide certain *ergonomic* characteristics and be user friendly. The recent development on sensors and interfaces is compiled in the paper called "Current Trends in Electronic Music Interfaces" by Joseph A. Paradiso and Sile O'Modhrain. Here they comment about the evolution and the current state of the musical interfaces. PM instruments should have present in their design the use of controllers in order to have control over the expressivity of the instrument.

"Acoustic musical instruments have settled into canonical forms, taking centuries, if not millennia, to evolve their balance between sound production, ergonomics, playability, potential for expression, and aesthetic design" [68].

The development of the musical interfaces in electronic music evolves so fast the one musician does not come to the point of mastering one tool and therefore do not become virtuoso. As in the traditional instruments their design had change slightly during the years leading to depurate performance techniques, compose musical pieces and create learning methods for musicians.

After the birth of MIDI we have seen the start of a new instrument made of a controller connected to a general computer that directly generates complex real-time audio without the need for an external synthesizer. This configuration grants the controller to be connected into "an arbitrary number of synthesis

[68] A. Paradiso, Joseph and O'Modhrain, Sile

parameters while still reserving sufficient cycles to build sophisticated and intelligent gesture-to-audio mapping algorithms” [69].

The construction of musical interfaces has been done by experimenting new ways of applying different kinds of technology. Several combinations of sensors and other controllers are possible. Many types of sensors in the market have been used as controllers for virtual instruments. Such as gravity, light, temperature, infrared, potentiometers and even air pressure sensors. “Musical interface construction is more art than science” Says Perry Cook on his paper called “Principles for Designing Computer Music Controllers” [70]. Here he presents specific cases in which creativity and technology get together to find innovative ways to design controllers.

4.2.1 Hybrid instruments

Hybrid instruments are the ones that integrate traditional musical instruments with sensors. One case is the instrument proposed by Ystad, Guillemain and Kronland-Martinet. This team is working on the design of a hybrid model for a flute. In which they simulate a resonator using the physical modeling waveguides and wave shaping coupled with real instruments. They used magnetic sensors for finger position and a microphone at the embouchure level to detect pressure variations. The sensors function is to create sounds transformations.

Another example of a hybrid instrument is the electric guitar with a MIDI controller. This is a standalone MIDI interface to transform the sound of the guitar into other instruments. Various choices of MIDI instrument patches and effects is possible.

[69] ibid
[70] Cook, Perry.

4.2.3 The future of sensors and controllers.

The future of sensors is going to the direction of precision and speed processing. Also the commercialization of these devices will reach the public soon. A sign of this is the release of the Nintendo Wii, which works precise infrared sensor for gaming controlling. On the other side the MIDI Bluetooth is hitting the market giving the possibility of wireless interfacing.

In the academic research interfaces reach the neuroscience and artificial intelligence. Going to the *Brain-Computer Music Interface* in which the music engine module is a "set of generative music rules, each of which produce a musical bar, or measure. The system works as follows: every time it has to produce a bar, it checks the power spectrum of the EEG at that moment and activates rules associated with the most prominent EEG rhythm in the signal." [71]

The system called BCMI-Piano is a BCI computer-oriented system maps the music with the EEG signals opens more possibilities of expression by performers. The intention of the system is also to train the user to control the EEG in order to solve problems of migraine or attention deficit and create conditions in their environment.

[71] Miranda, Eduardo; Brouse, Andrew.

5. Experimental Results

5.1 Dynamic change of timbre in live synthesis with Max/MSP

The problem in question is to **change the timbre of a synthesized instrument while it is played**. This can be accomplished using programs such as Max/MSP (Cycling74). To create this timbre change in **real time** we propose the use of digital filters. Because "spectrum shaping by filtering suggests interesting ways to change signals while keeping control of meaningful parameters like the bandwidth and center frequency of the overall spectrum shape" [72]. **Figure 7** shows a first approach for timbre changes using physical modeling (Karplus Strong Algorithm waveguide synthesis) and a proposal to implement a biquadratic filter to change the timbre in real time. On a later stage, the biquad filter is replaced with a cascade of biquad filters (Figure 8)

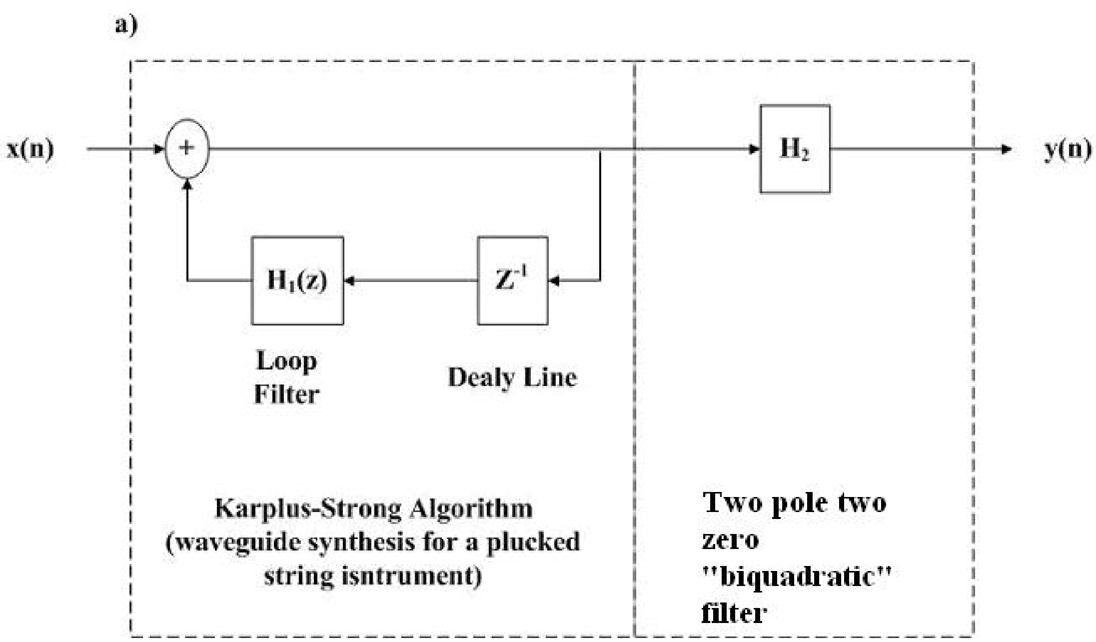

Figure 7: Two pole two zero filter implementation

[72] Steiglitz, Ken.

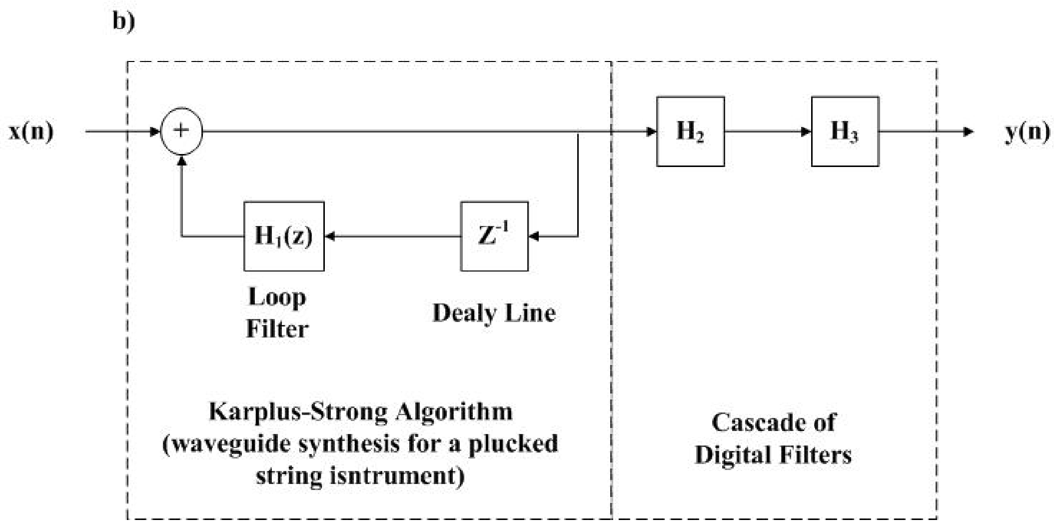

Figure 8: Cascade of filters implementation

Digital waveguides models are applied to specific musical instruments and considered as *computational physical models,* they consist of delay lines, digital filters and sometimes non-linear elements (Smith, 2006) [73].

5.1.1 Platform

To implement the Karplus-Strong model for plucked string we used the PeRColate external objects for Max/MSP, which is an open source code developed by Dan Trueman at Princeton University and R. Luke DuBois at the Computer Music Center, Columbia University. The PeRColate has a port of ***STK*** (Sound Synthesis Tool Developed by Perry R. Cook).

5.1.2 Additions to the plucked~ external object.

We can make the system to become unstable provoking longer and richer sound with more texture to work in the later filter stage. *In terms of poles and*

[73] Smith ,Julius O.

zeros, an irreducible filter transfer function is stable if and only if all the poles are inside the unit circle in the plane[74] *(Smith 2006)*.

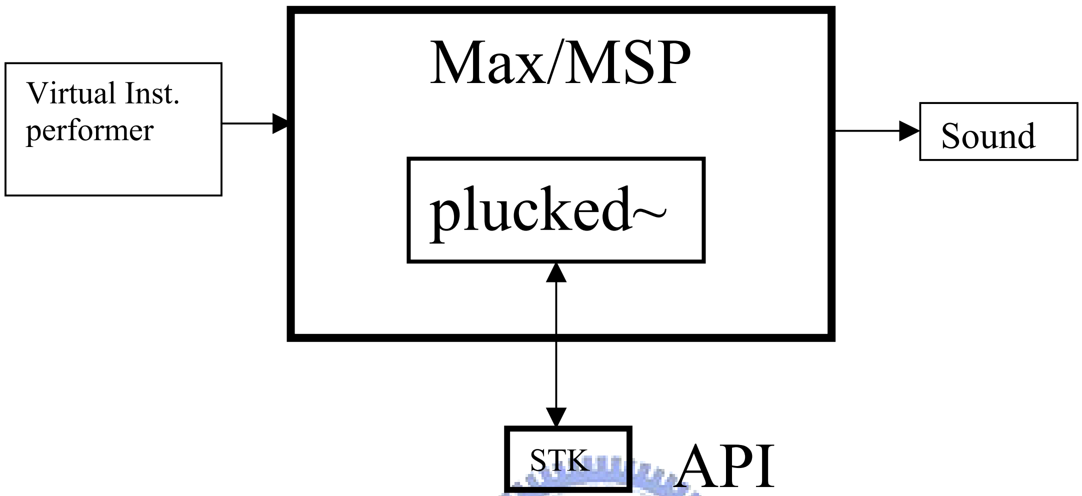

Figure 9: Max/MSP external object called plucked~ with Karplus-Strong Algorithm inside linked with the STK API.

[74] Smith, Julius O.

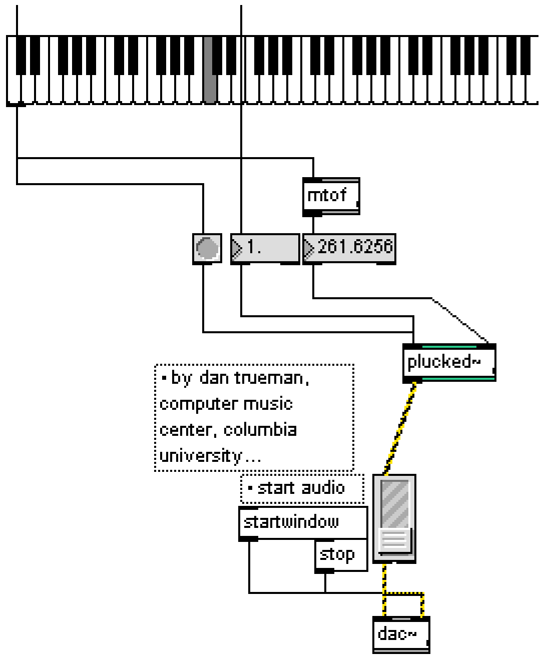

Figure10: The plucked external object and patch.

5.1.3 Biquad Filters Implementation

We have done two different patches to be added to the output of the MSP plucked~ object.

The first addition is a biquad~ filter. The interface is shown in **Figure 11**, Modifications can be made in real time.

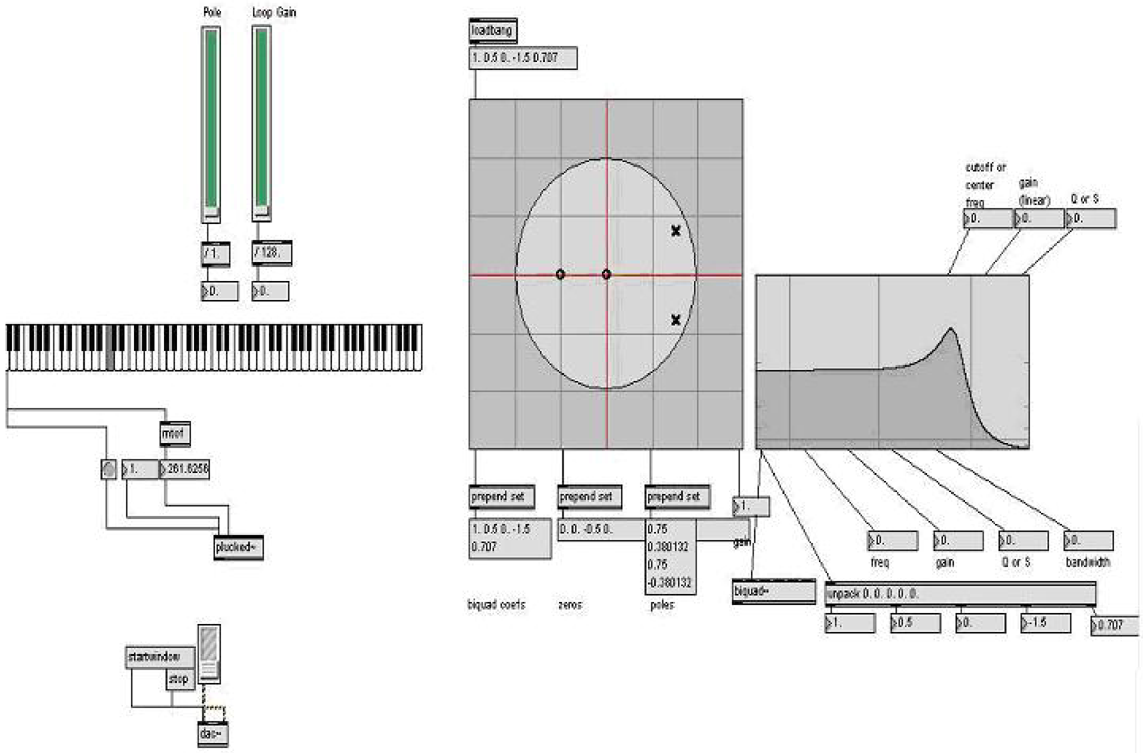

Figure 11: Biquad filter to modify the output of the plucked~ external object.

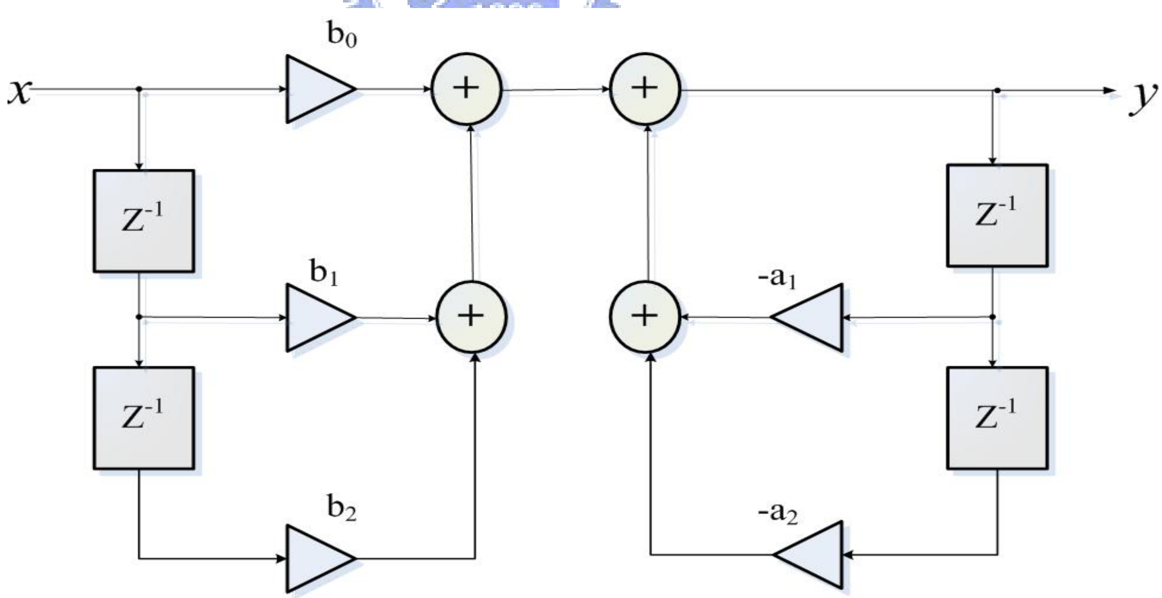

Figure 12: Block diagram of a biquadratic filter. Max/MSP setting.

http://ccrma.stanford.edu/~jos/filters/Direct_Form_I.html

Above a biquad~ filter diagram from Max/MSP software. The biquad~ of biquadratic filter is a two-pole two zero filter. The difference equation for the filter implementation based on the block diagram is:

$$y[n] = b_0*x(n) + b_1*x(x-1) + b_2*x(x-2) - a_1*y(n-1) - a_2*y(n-2) \quad (5.1)$$

The correspondent transfer function will be:

$$\frac{Y}{X} = \frac{b_0+b_1z^{-1}+b_2z^{-2}}{1+a_1z^{-1}+a_2z^{-2}} \quad (5.2)$$

With the biquadratic filter we can make a resonator. A common setting for the zeros when using biquad~ as a resonator is to place one at z = 1 and the other z = -1 (half the sampling rate) For example b_1=0 and b_2= -1 (Smith, J.).

$$B(z) = 1 - b_2z^{-2} = (1 – z^{-1})(1 + z^{-1}) \quad (5.3)$$

The advantage of using Max/MSP version 4.5 (Cycling 74), is that we can modify the parameters of the biquad filter using a graphical interface. Using the computer mouse or, in an ideal scenario, using a controller we can change the values of the parameters in real time. The values can also being limited in order to maintain the filter in stable conditions, without letting the poles or the zeros to go out of the unit circle.

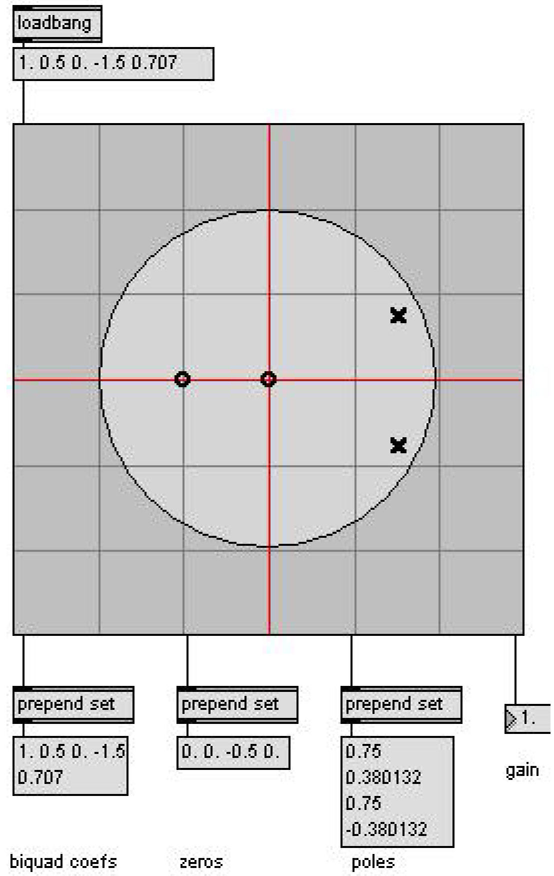

Figure 13: z-plane from Max/MSP 4.5 interface.

The graphical interface of the unit circle can be expanded to work from 2 zeros and poles up to 24.

The next addition will be a set of **biquads in cascade**. Figure 11. The cascade~ object in Max/MSP allows the user to customize the desired number of filters in cascade.

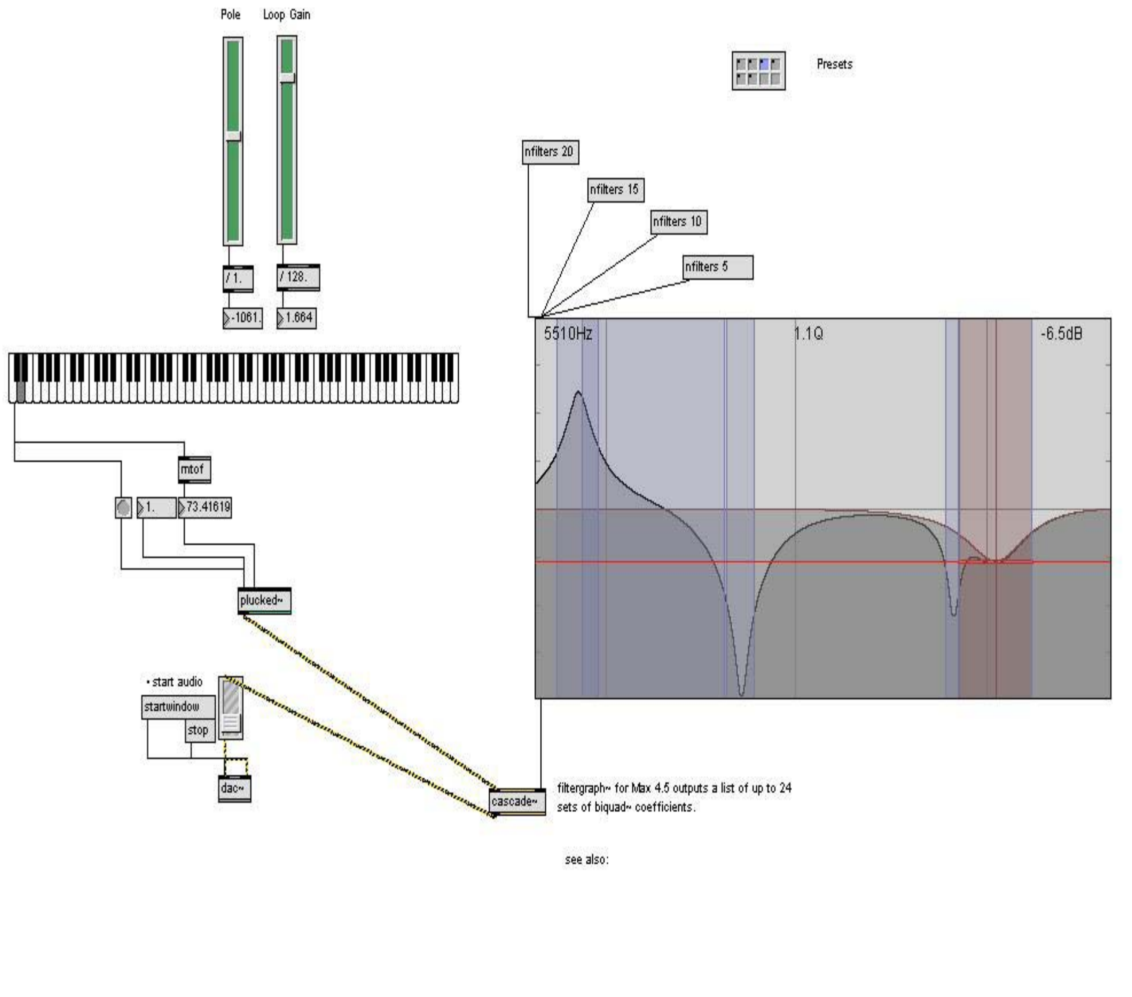

Figure 14: Cascade filter implementation.

The number of biquads in cascade can be modified up to 24. Figure 15 shows how to change these parameters by pressing the parameter “nfilter”.

The general form of the difference equation for biquad filters in cascade will be[75]:

$$y(n) = b_0x(n) + b_1x(x-1) + \ldots + b_Mx(n-M) - a_1y(n-1) - \ldots - a_Ny(n-N) \tag{5.4}$$

$$y(n) = \sum_{i=0}^{M} b_i x(n-i) - \sum_{j=1}^{N} a_j y(n-j) \tag{5.5}$$

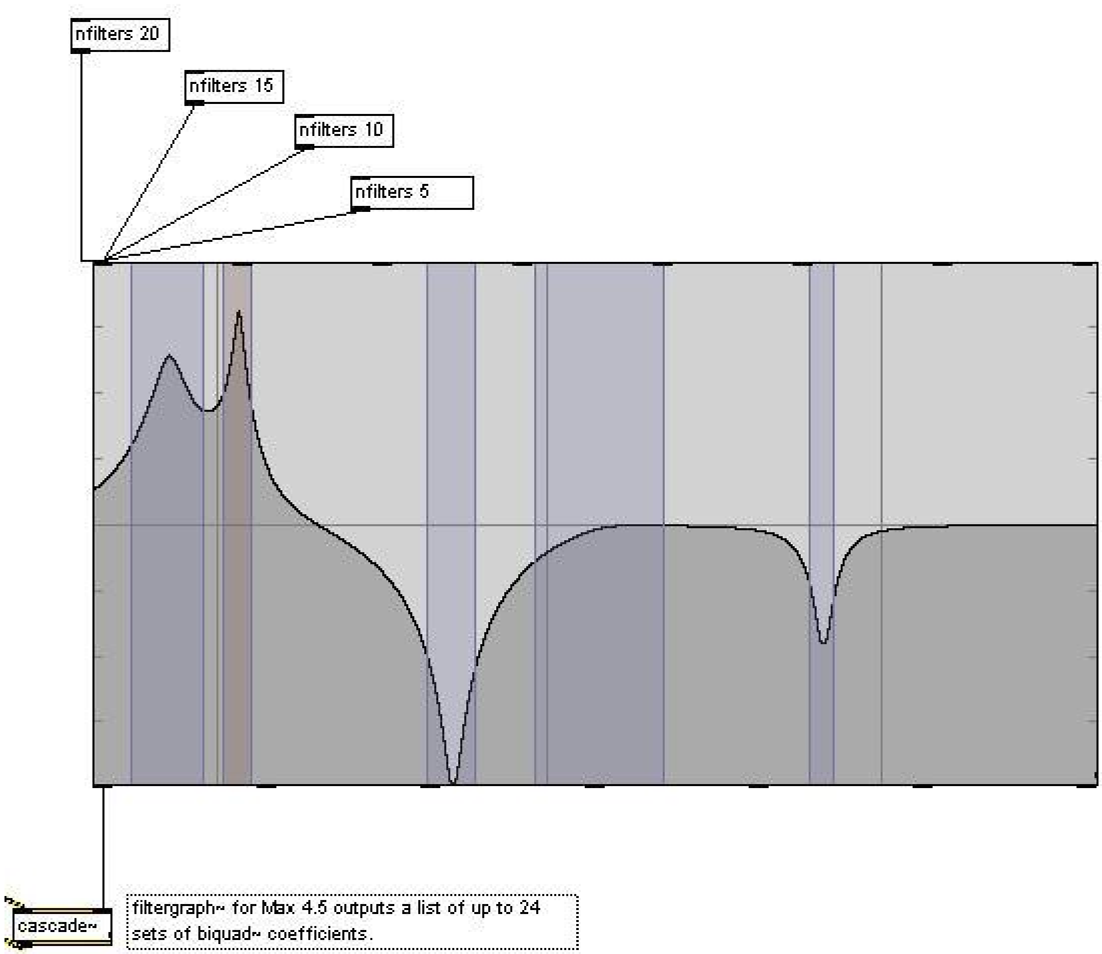

Figure 15: "nfilter" parameters change the number of biquad filters in cascade.

After setting the desired parameters, using the mouse, the user can save this parameter in a preset (figure). During the performance of the instrument, the

75 ``Introduction to Digital Filters with Audio Applications", by Julius O. Smith III, (August 2006 Edition).

user would be able to switch from one preset to the other. In this way the instruments possibilities of timbre increase.

Figure 16: Filters allows the user to design a sound using filtering shaping then the presets in Max/MSP call each one these settings, allowing the instrument to have more variety of timbres.

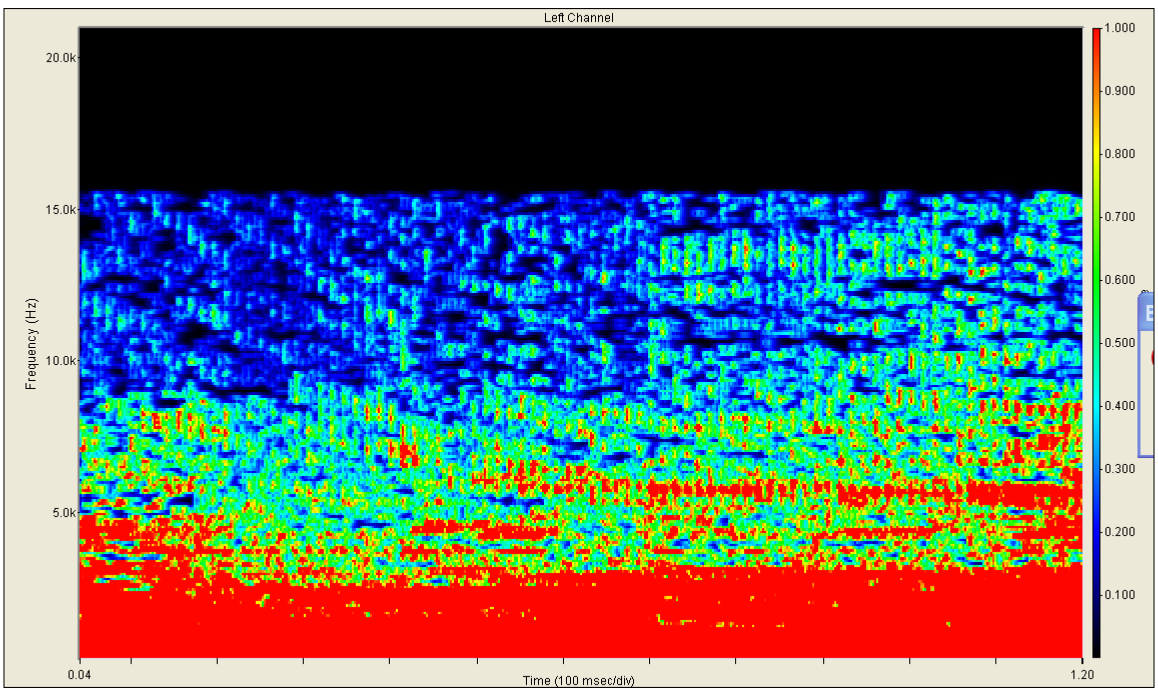

Figure 17: Sound sample1: shows texture and dynamic change of timbre.

The implementation of this system is shown in **Figure 14**. With the help of the ‘preset object (in Max/MSP)’ the user can set a specific timbre for any desired note. Also, allows playing different notes with timbre change within the same instrument. On the other side, if desired, the user is able to change the timbre of a single note while playing it.

The spectrogram above in **Figure 18** shows how the spectrum is modified when the user change the parameters of the filter in real time.

5.2 Digital Waveguide Implementation on Csound. (A new instrument).

5.2.1 Csound Waveguide

The intention of this paper is to implement a dynamic change of timbre. This experimentation has been developed in Csound programming language for sound rendering. Commonly used in sound synthesis and instrument design.

The implementation of this model is based on the bass waveguide by Hanz Michelson included in the Csound book which was presented in chapter 3.

The dynamic change of timbre is based on a dynamic change of the filter's parameters in time.

First, I did a separation of each one of the sections to see how they sound like:

- Impulse (Triangle wave)

```
kenvstr      linseg       0,ipluck/4,-p4/2,ipluck/2,p4/2,ipluck/4,0,p3-
ipluck,0

aenvstr   =         kenvstr
ainput    tone      aenvstr,200
```

As seen in the spectrogram the triangle wave does not present frequencies higher than 5kHz. This impulse remains in the low frequencies range to generate a bass sound.

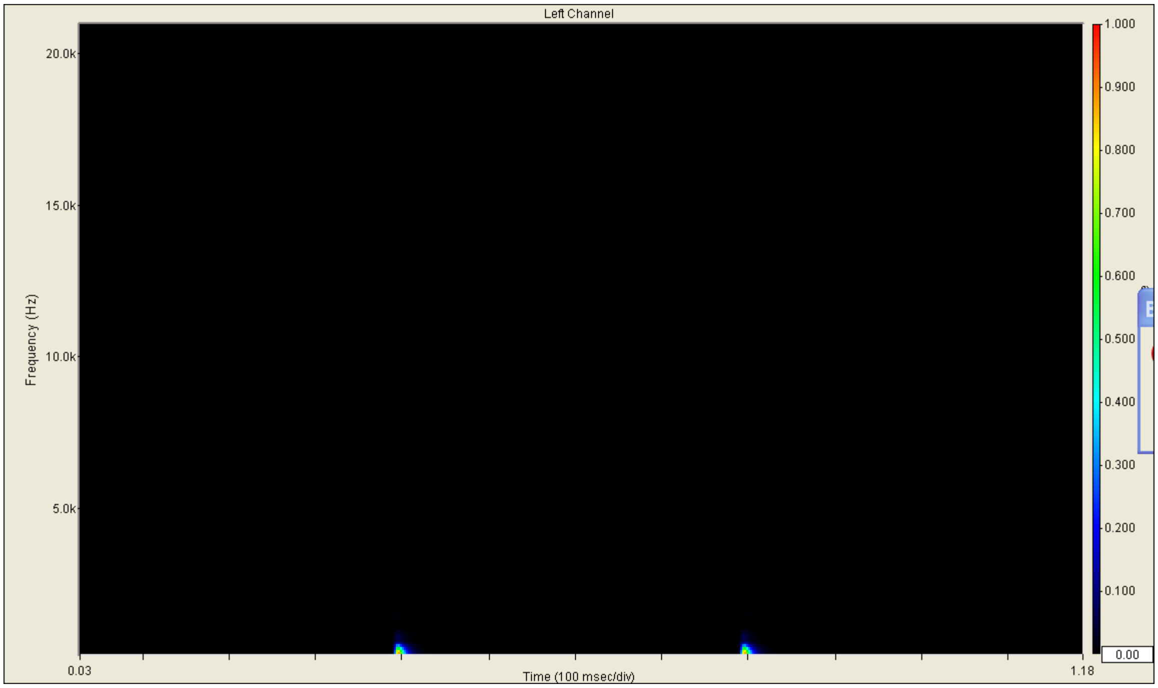

Figure 18: Triangle wave impulse.

- The next step is to pass the triangle wave through the first delay and low pass filter `tone`. (See details of tone parameters)

```
; THIS ENVELOPE LOADS THE STRING WITH A TRIANGLE WAVE
kenvstr   linseg    0,ipluck/4,-p4/2,ipluck/2,p4/2,ipluck/4,0,p3-
ipluck,0

aenvstr   =         kenvstr
ainput    tone      aenvstr,200

; DC BLOCKER
ablock2   =         afeedbk-ablock3+.99*ablock2
ablock3   =         afeedbk
ablock    =         ablock2

; DELAY LINE WITH FILTERED FEEDBACK
adline    delay     ablock+ainput,1/ifqc-15/sr

afiltr    tone      adline,400
```

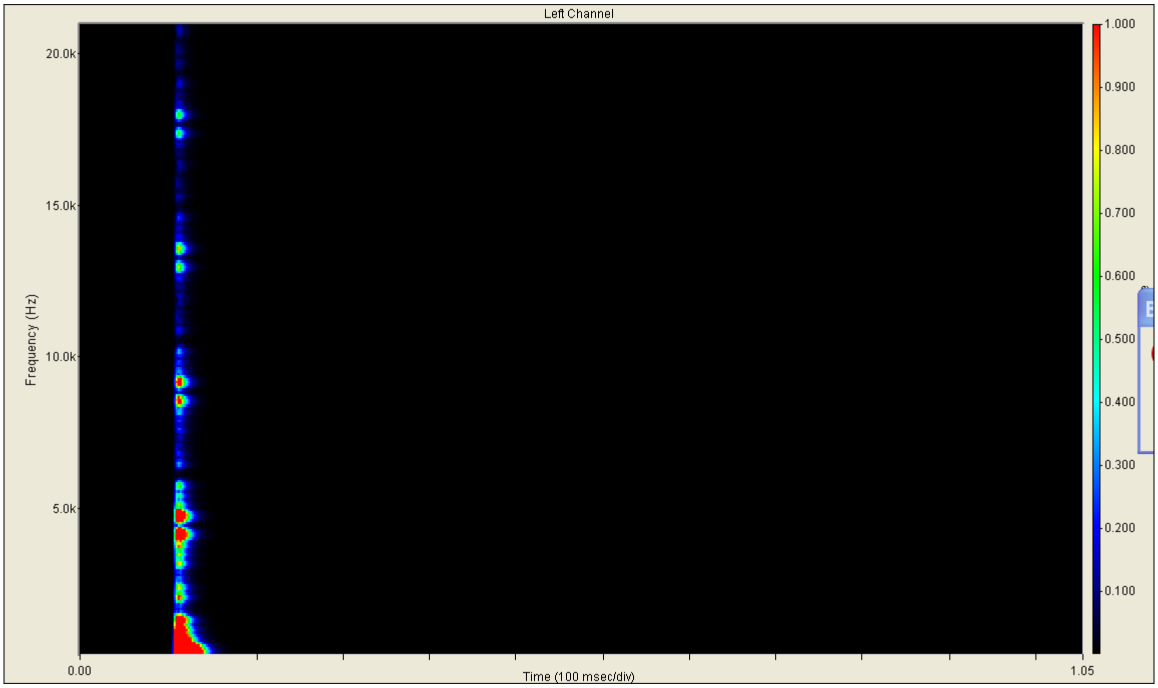

Figure 19: The spectrogram above shows the impulse after being delayed `1/f-15/sr`.

The value ifqc denotes the fundamental frequency and then subtract 15 samples to bring it into tune.

- Then the resonance filter resonates the partials in order to imitate the Bass resonance.

```
; RESONANCE OF THE BODY
abody1     reson      afiltr, 110, 40
abody1     =          abody1/5000
abody2     reson      afiltr, 70, 20
abody2     =          abody2/50000
```

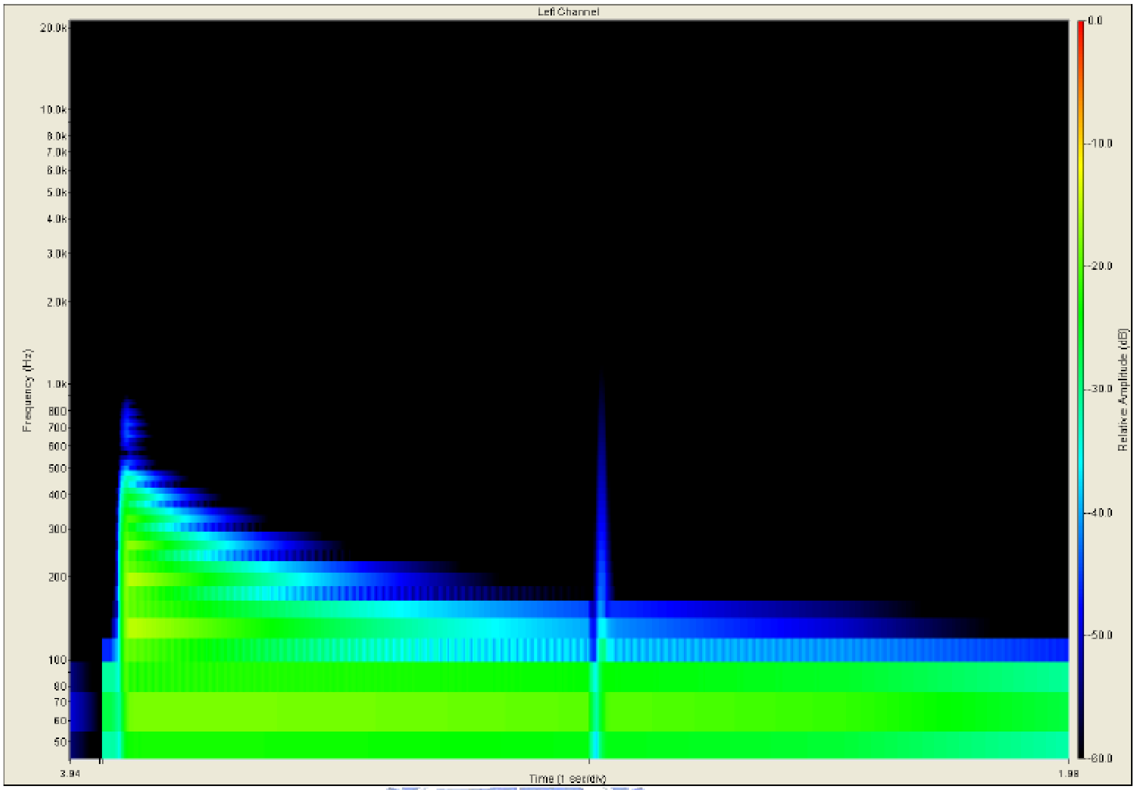

Figure 20: Waveguide bass: in the spectrogram below displays the characteristic decay of the plucked string, high frequencies attenuated before the low frequencies.

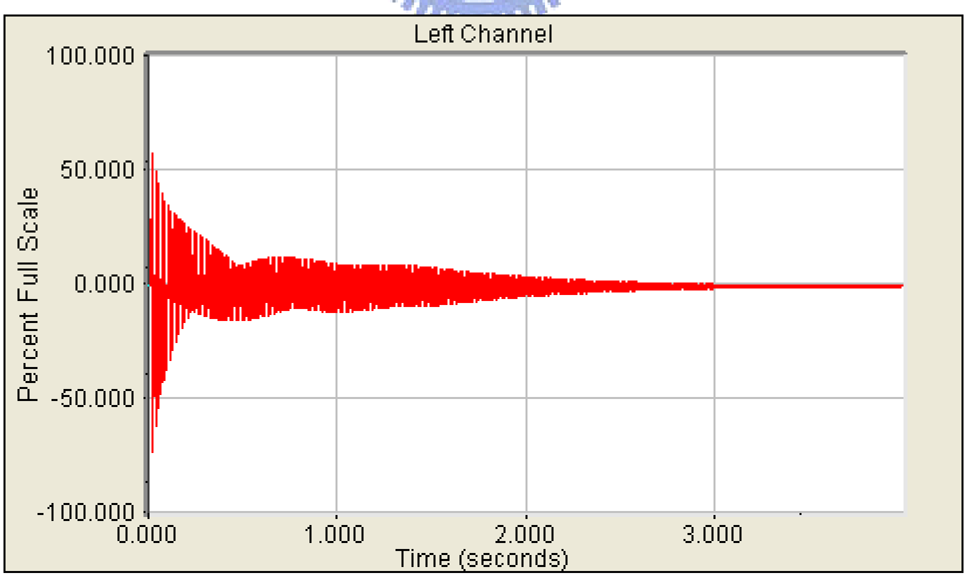

Figure 21: Time series graph of waveguide bass.

5.2.2 Modifications.

1. First instead of having a triangle wave as an impulse I placed a white noise impulse. Also the white noise has an amplitude envelope that I called `kampenv`. Next, I created an envelope called `kfilter1` to modify the frequency response of the LP filter in time.

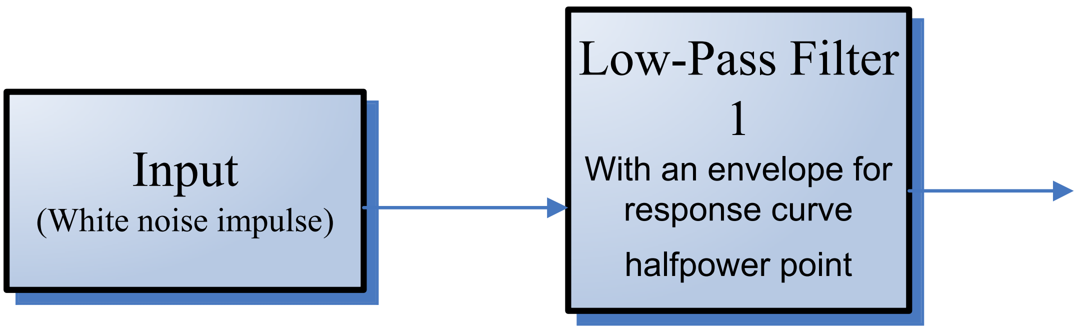

Figure 22: White noise input and Lowpass filter

Envelope of LP filter 1 (`kfilter1`)

Time	0	t/2	t
Frequency Response	50	1000	100

The envelope kfilter1 draws a straight line between each one of the values of the frequency response. The duration (t) of the note is determined by the user.

```
; Noise input and kampenv
kampenv linseg  p4/2,p3/2,0
kfilter1 linseg 50, p3/2, 1000, p3, 100

aenvstr    noise    kampenv, 0.5

ainput     tone       aenvstr,kfilter1
```

2. Then the DC remains the same (this is to center the sound in the BIAS). Also the delay's parameters remain the same and the next LP Filter apply the same envelope `kfilter1` as in the previous filter. The user can also change this envelop value.

```
; DC BLOCKER
ablock2    =        afeedbk-ablock3+.99*ablock2
ablock3    =        afeedbk
ablock     =        ablock2

; DELAY LINE WITH FILTERED FEEDBACK
```

```
adline     delay     ablock+ainput,1/ifqc-15/sr
afiltr     tone      adline,kfilter1
```

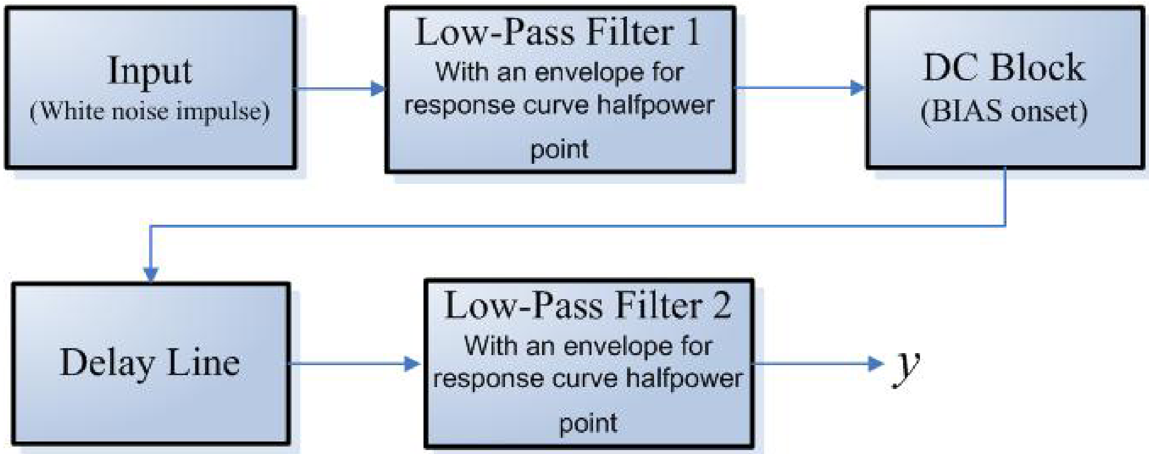

Figure 23: Instrument after delay line is implemented

3. We have created exponential envelopes for the resonance filters: one linear envelope for k-rate center frequency: (kfilter2). Exponential change for the k-rate bandwidth in Herz (kopen1 and kopen2).

```
; RESONANCE OF THE BODY
kopen1 expon 100, p3, 40
kopen2 expon 5, p3, 40
kfilter2 linseg 50, p3/3, 1000, p3/3, 100, p3/3, 700


abody1     reson     afiltr, kfilter2, kopen1
abody1     =         abody1/5000
abody2     reson     afiltr, kfilter2, kopen2
abody2     =         abody2/50000
```

So the new diagram will look as follows in figure 24. The resulting sound is similar to the cello sound.

The values of the envelopes were selected arbitrarily. The musician can select desired parameters in order to create the textures he needs.

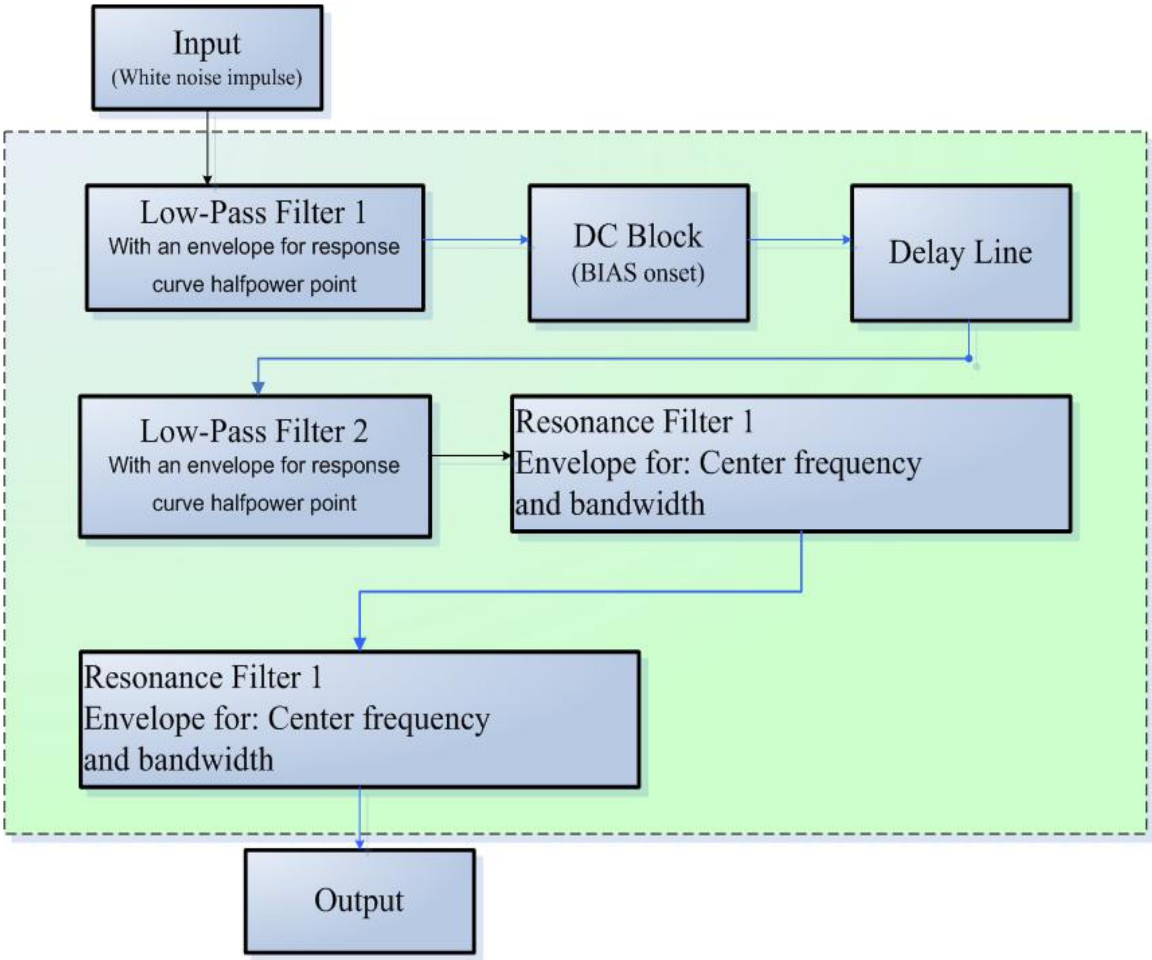

Figure 24: Resulting block diagram after implementing the dynamic change of filter parameters.

Then the complete .csd file will look like this (Also check the spectrogram at Figure 25)

```
<CsoundSynthesizer>

<CsOptions>
</CsOptions>

<CsInstruments>

; just playing around

; BASS PHYSICAL MODEL

sr          =          44100
kr          =          4410
ksmps       =          10
nchnls      =          1
```

```
          instr     1

; INITIALIZATIONS
ifqc      =         cpspch(p5)
ipluck    =         1/ifqc*p6
kcount    init      0
adline    init      0
ablock2   init      0
ablock3   init      0

afiltr    init      0
afeedbk   init      0

koutenv   linseg    0,.01,1,p3-.11,1,.1,0 ; OUTPUT ENVELOPE
kfltenv   linseg    0, 1.5, 1, 1.5, 0

; THIS ENVELOPE LOADS THE STRING WITH A TRIANGLE WAVE
kampenv linseg  p4/2,p3/2,0
kfilter1 linseg 50, p3/2, 1000, p/2, 100

aenvstr   noise     kampenv, 0.5

ainput    tone      aenvstr,kfilter1

; DC BLOCKER
ablock2   =         afeedbk-ablock3+.99*ablock2
ablock3   =         afeedbk
ablock    =         ablock2

; DELAY LINE WITH FILTERED FEEDBACK
adline    delay     ablock+ainput,1/ifqc-15/sr
afiltr    tone      adline,kfilter1

; RESONANCE OF THE BODY
kopen1 expon 100, p3, 40
kopen2 expon 5, p3, 40
kfilter2 linseg 50, p3/2, 1000, p3/2, 100


abody1    reson     afiltr, kfilter2, kopen1
abody1    =         abody1/5000
abody2    reson     afiltr, kfilter2, kopen2
abody2    =         abody2/50000

afeedbk   =         afiltr

aout      =         afeedbk
          out          50*koutenv*(aout  +  kfltenv*(abody1  +
abody2))

          endin
```

```
</CsInstruments>

<CsScore>

; PLUCKED BASS
;   START DUR     AMP          PITCH   PLUCKDUR
i 1   0    16     2000         6.00      .25

e


</CsScore>

</CsoundSynthesizer>
```

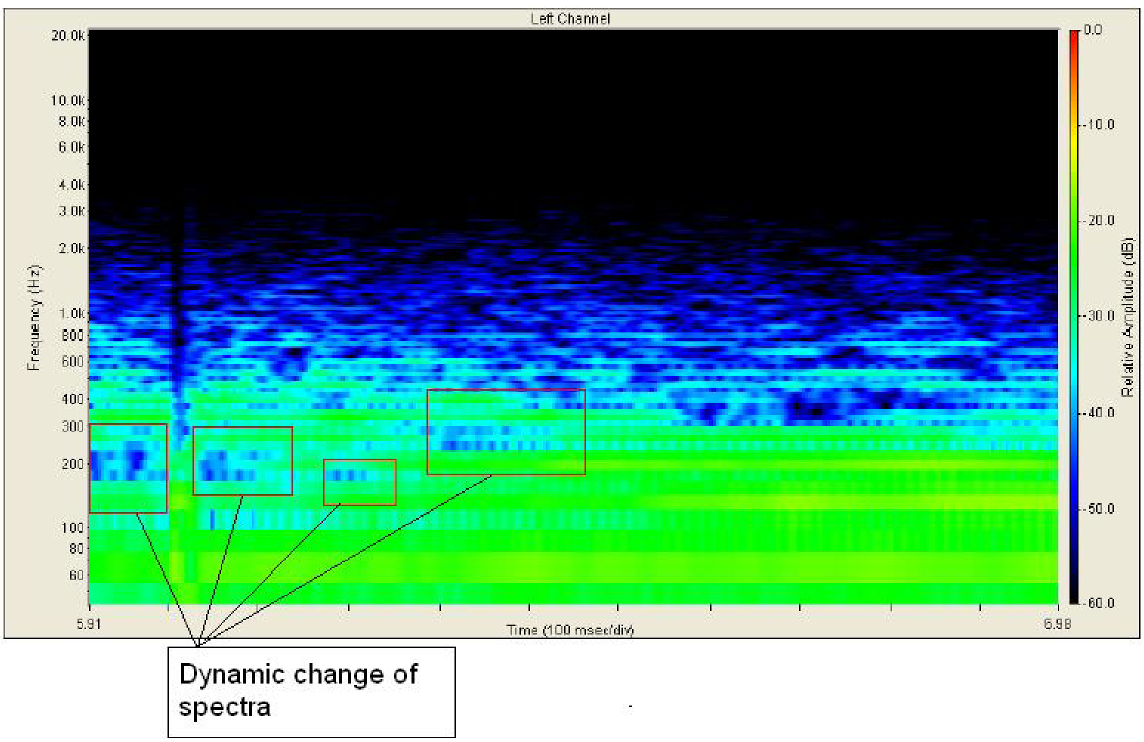

Figure 25: The spectrogram below show a section of the resultant sound, is richer in tone because of the white noise input. Also note some dynamic change of spectra caused by the filter envelopes:

The texture of this sound is similar to the one of the bowed string. Random input in this case is provided by white noise. Other possibilities should be explored.

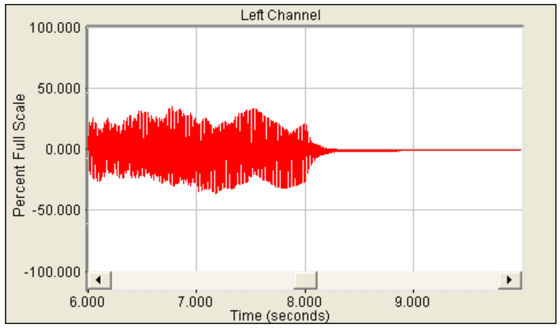

Figure 26: Time series graph of bowed-string like sound

More modifications will be implemented to the model to keep finding new possibilities for *Articulation*. The creation of the envelopes facilitates the dynamic change of timbre in the instrument making it more interesting. We will use other random input other that white noise. Also, another challenge would be to create different sets of frequency envelope.

6. Conclusions

The world of physical modeling techniques for sound synthesis is spreading in the research community. Due for being out of scope, some models were not covered by this presentation. Other models besides the Digital Waveguides include the Finite difference integration, functional transformation methods and modal synthesis and others. For a complete description and more in depth analysis see Trautmann and Rabenstein (2003).

We consider the digital waveguides to be a trend in physical modeling, because it has been spread out with a lot of background and documentation. Specially, that the fact of having a commercial implementation put in evidence the willing of musical instrument manufacturers in develop products if this type.

However, physical modeling is also pointing to the modal synthesis, which can generate models in 3 dimensions and applied in musical situations. "Modal analysis determines the natural frequencies and mode shapes of structures and provides a means for determining the resulting oscillation of a system under the action of various forcing functions (Heand Fu 2001)." [76]

To follow up on the research, there must be a multidisciplinary group consisting of people from different backgrounds, acoustics, digital signal processing, musicians and others. Also, it is recommendable to show the output sounds to ear-trained people and see what the perception of the instrument is.

Also, techniques to analyze spectra are important to use. The analysis provides a visual perception of what is going on inside the sound. The FFT and DFT analysis was not included in depth in this research.

Another important thing to mention is that the applications that we use to generate sound (MSP and Csound) are applications very traditional and didactical. It is also, important to have a deep study in Matlab and also try to develop our own applications for generation of sound from physical modeling. An

[76]Bruyns, Cynthia.

example mentioned before, the MTMS software for guitar simulation has been developed by the same author (Bader) in a windows environment. Similar applications should be developed for further research.

We hope that the present work catches the attention of other students in order to continue the learning of this topic. Also, recommend having an advanced course in sound synthesis to cover the physical modeling techniques.

On the other side, another team in parallel should be focused on the development of *controllers and sensors* to later be used to control the physical modeled instruments. The integration of these devices can be done by developing external objects for Max/MSP that contain the code the of the instrument plus the conditions to attached the sensors and controllers.

A good background in MIDI and sensors is needed and also guidance from the professor to collaborate in a joint project. The documentation available is good enough to start to thinking about a project that includes the development of physical modeling by one team and a second team in charge of developing the controllers. A suggestion would be to start to implement prototypes using the existing models of digital waveguides and start from there.

Taiwan is good place to develop such an instrument like this. The infrastructure and human resource of NCTU provides a suitable environment to develop such a project. For further research, would be recommendable to find a person at the Management Institute to see the possibilities of creating a patent and commercialize an instrument. Or even, keep finding an opportunity the Music Technology staff to contribute to the scientific community of the Audio and Music Technology.

7. Future Works

A future work to study in physical modeling would be the inclusion of traditional instruments to be modeled. For example, the guitar, mandolin and sitar, just to mention a few have been modeled. Instruments such as the Pipa (琵琶) and the Guqin (古琴) are examples than can be modeled either by using the waveguide model or any other scheme that approach the physical modeling.

In the case of the Guqin (古琴), the possibilities of articulation are endless that even the plucked string model would need to have many adaptations in order to have a convincing emulation.

For this objective there must be a thorough study of the physical characteristics of the instrument. Placing microphones and other sensors in the instrument to study its resonance. Also, there should be an analysis of the sounds of the different articulations to study their harmonic content. See Figure 27, 28 and 29 for Guqin spectrograms.

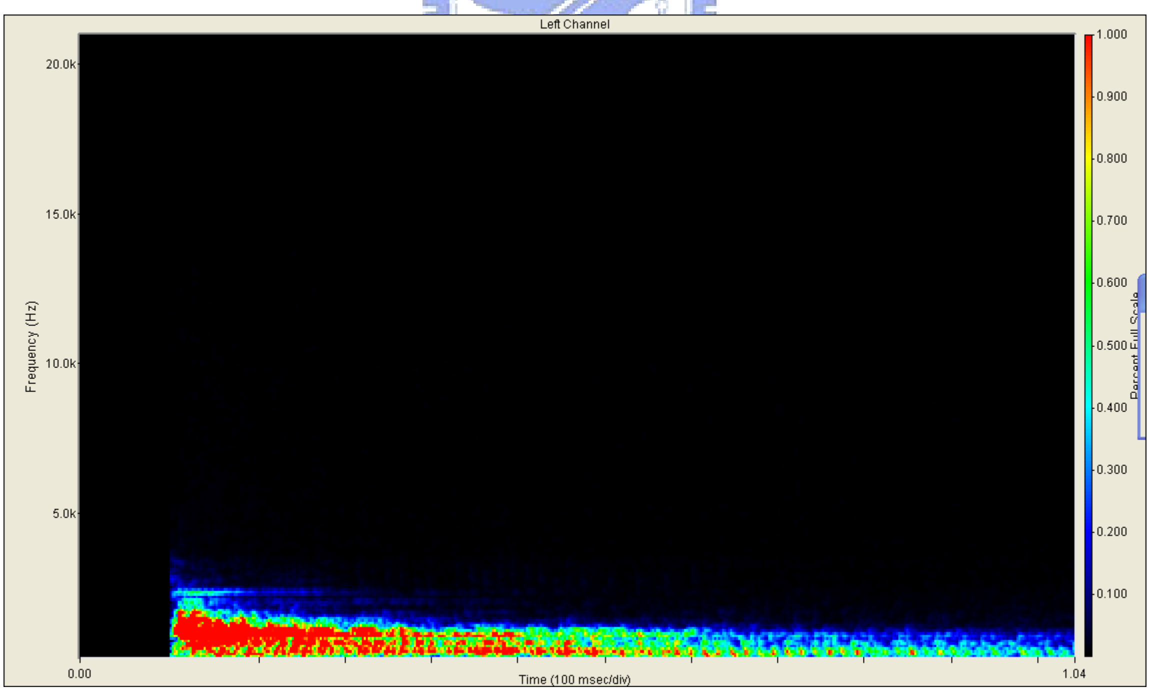

Figure 27: Guqin Open string

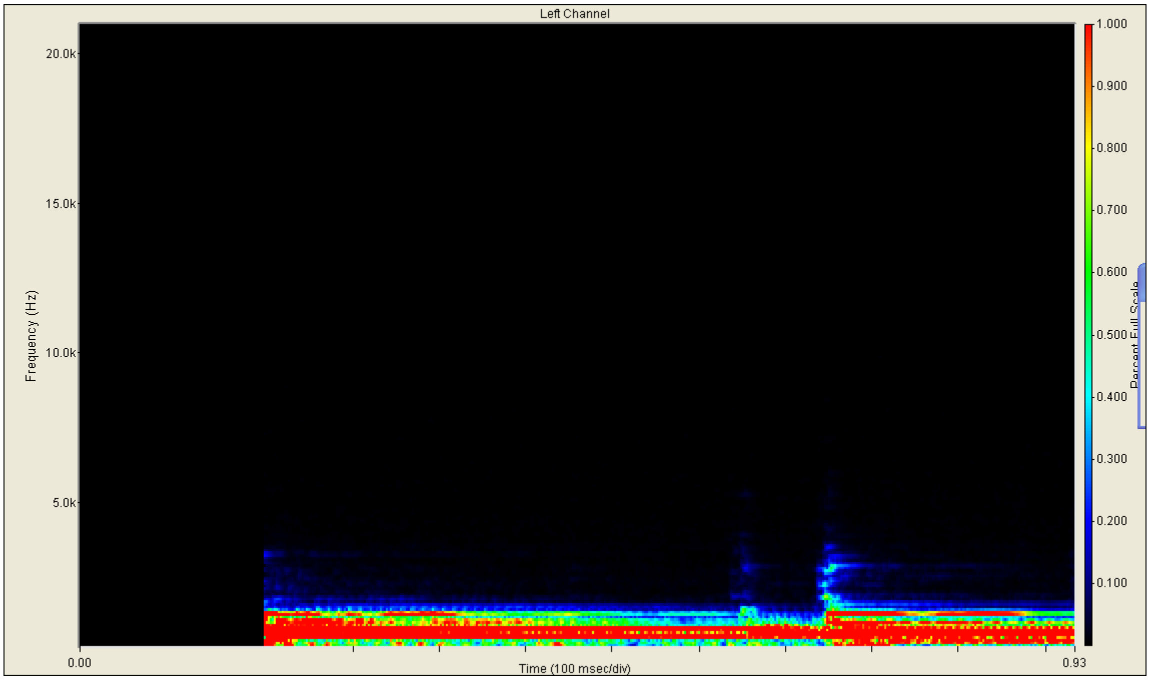

Figure 28: Guqin overtones.

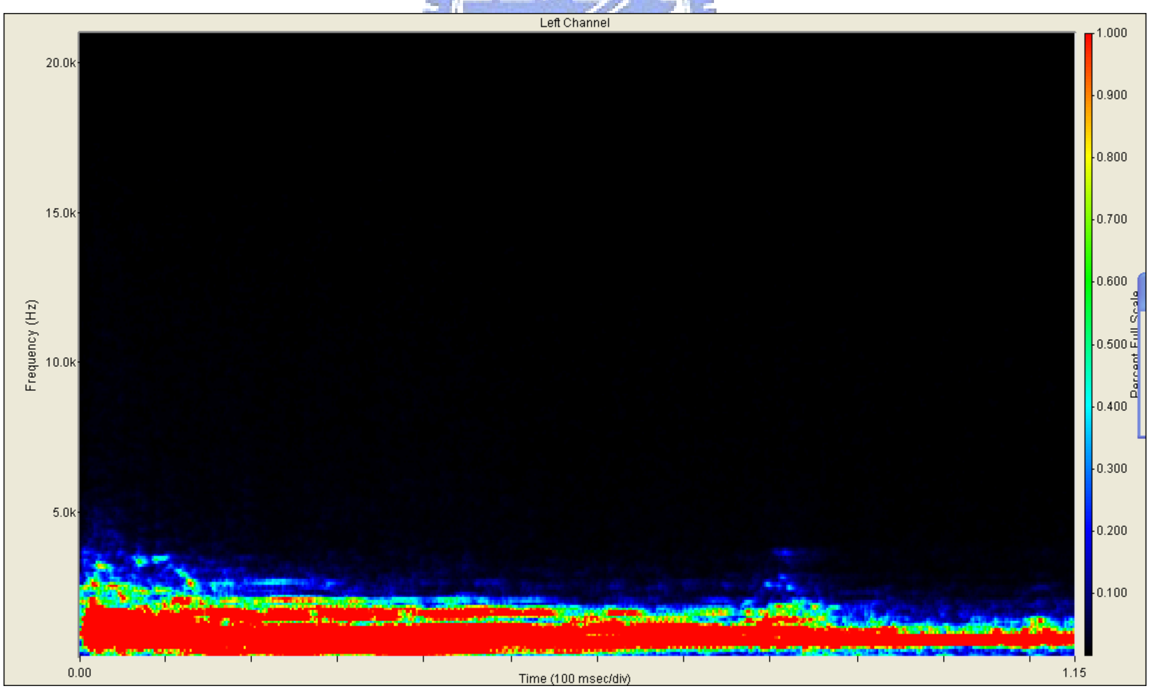

Figure 29: Guqin Plucked.

There are plans for the creation hybrid virtual Guqin (古琴), in which sensors may be placed under the strings to transform the excitations. Also, we are considering having additional controllers. For example, an EEG reader to map musical parameters. And this will mean being at the state-of-the-art in approaches of a hybrid instrument.

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