Digital audio processing system for live electronics performance based on a hexaphonic pickup guitar

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Abstract

This paper presents an ongoing project for designing a digital audio processing system based on the acoustic input from a hexaphonic guitar pickup. The artistic aim for the project is to use this system as an interface for live electronics performance. The article discusses compositional, aesthetic and technical ideas which have been implemented in the design. It describes also the steps which have been taken in the process of building the hardware and the software components of the system. Finally, it traces possible future development of the instrument.

Autor Keywords

Augmented Instrument, Acoustic-Digital System, Real-Time Digital Processing, Spatial Live Electronics

1.Introduction

The present article is an overview of ongoing artistic and technological research. The aim of this research has been to explore the sonic capabilities of a hybrid acoustic - digital system, one that is designed to be applied as a performable and compositional tool. A fundamental condition, which has determined and directed the building of the system was the use of a hexaphonic guitar pickup as an initial device. This pickup captures and introduces the instrumental sound as multichannel input of the digital audio processing system. The environment of Max/MSP has been used for building the above mentioned digital processing network designed to function as set of modules. Each module represents a complex audio processing of the incoming acoustic signal. Furthermore, a variety of transformations and sonic textures have been achieved by incorporating these modules in different ways. Additionally, compositional strategies along with mathematical procedures have been used to determine the behavior of the system. As a result a significant part of the decision making during a performance is held by the computer. On the other hand , the behavior of the system, accordingly the sound output determines the next step that the performer will take by listening to this output. The complex behavior representing a chain of choices/decisions, made by the performer and the machine, operates in tandem to determine the entire artistic output. Such an inter-dependance references notions of ***interactive composing*** (Joel Chadabe) , ***composed interaction*** (Di Scipio), ***second order interaction, second order cybernetics*** .

The presence of a multichannel output naturally leads to thoughts about its spatial representation. This in turn raises the issue about different approaches for multichannel mapping and how they have been considered in connection to the overall goal of this artistic project.

The *Feedback concept* — which has much in common with *Interactive composing* — is also implemented by two distinct approaches, applying both to the way the output is being routed back to the system.

2.Initial conditions

Several circumstances directed my attention towards this project.

On one hand, my work as a composer led me to a moment of need, where I wanted to find a seminal way to combine instrumental and electronic sounds in my music. Contrarily, in order to break the boundaries of the fixed and predefined instructions of musical scores (or fixed media), I sought to find a more flexible, improvisatory approach towards the organization of sound. Furthermore, as a performing musician I also felt a need to enrich my capabilities of expression and to embed my passion about electronics in my performance practice.

The above prompted research into possible pre-existing implementations of the hexaphonic pickup. However, this led me to predominantly encounter examples of conventional guitar music — aside from say only a few more creative approaches (such as the one of the composer Christopher Trapani or the researcher and instrument builder Otso Lähdeoja). Still, the majority of approaches with which I came across were focused on directions very different than the one I was imagining to develop. The absence of such a system together with the idea about the capabilities of the hexaphonic pickup such as: individual processing of each string; creation of polyphonic, poly processed textures; spatial mapping; , therefore provided another strong impulse for me to delve even more deeply into this area of research.

Last, but not least, another initial condition was the affordable price of the project. Since this project has been entirely a self-funded initiative, economy of price has been of a vital importance. Moreover, the affordability of this system is important also from the perspective of sharing it with other enthusiastic users.

3.Design

In this section I present in details the components of the project and the steps which were taken in order to accomplish it.

3.1 Hardware components

Guitar

The guitar I have found to be most appropriate for this project is the *Yamaha SLG130NW (Fig. 1).*

Figure 1. Yamaha SLG130NW

SLG stands for *silent guitar* and *N* for *nylon.* One of the reasons to choose this particular guitar was also because I already owned one; also the simple and clean design which allowed me to mount the hexaphonic pickup on the body much more easier then mounting it to any other kind of acoustic or electric guitar (without interrupting of its basic functionality); the "silent" aspect (the most important reason in terms of sound). This is important especially when hard digital processing, for instance pitch shifting is applied (as it might be a case with my system). Personally speaking, I find it confusing to hear the acoustic sound of the guitar during a performance together with the processed one and this is something which would happen, when I would perform with an acoustic guitar for example. After some time of experimenting with the "silent" guitar and observing I concluded that I could utilize "silent" input in order to provide the perceptual phenomenon that the processed output was actually the natural sound of the instrument.

Strings
I normally I rely on nylon strings in my performance practice (default for this model guitar), however, in this case they were replaced with a special ***steel core/nylon flat wound*** strings in order to excite the electromagnetic pickup and still keep the feeling of classical guitar strings by touch to which I am used to.

Hexaphonic pickup
There are not many hexaphonic pickups on the market and most of them are electromagnetic, intended for use on electric guitars, ruling out nylon-string classical guitars. The Ubertar electromagnetic hexa pickup was chosen for this project because it seemed to be the best possible option. Nevertheless, this pickup still suffers from few disadvantages - such as noise, crosstalk and very low signal level discussed later. But first, the exact nature of a hexaphonic pickup needs to be explained.
"*……Passive electromagnetic. Coils and magnets. No batteries. No active electronics. No preamps. ….A hexaphonic guitar pickup (sometimes called a divided pickup) is a pickup with six outputs (one for each string of the guitar)…..*" [1] (*Fig. 2).*

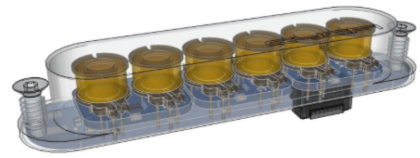

Figure 2. Hexaphonic pickup

Despite the promoted qualities presented in the above quotation [1], a preamp was still needed in order to receive at least a decent sound energy level at the input of the digital system and this was built and integrated in a later stage of the project (see below).

3.2 Design of the physical elements of the project

Mounting of the pickup on the body of the guitar was the next procedure needing to be done. This required a little modification of the body of the guitar.
It was important to choose carefully the correct position of the pickup. Bearing in mind at first place the gradual transition of the quality of sound according to the distance of the bridge and then the comfort of playing as factors that would be crucial in my decision , the position was set as it is shown on (*Fig. 3)*

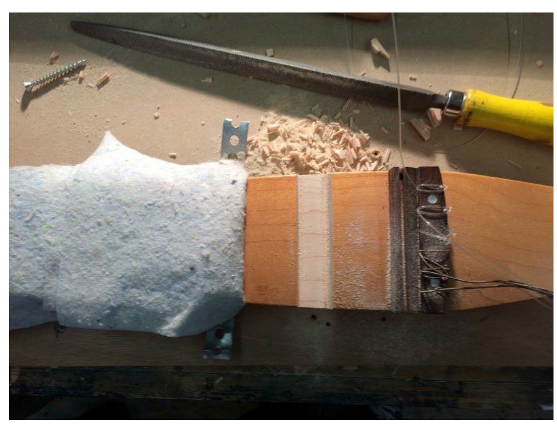

Figure 3. Modifying the wooden "body" of the guitar

Hexaphonic Preamp

The initial excitement of the successful craftwork was soon replaced by a disappointment about the very low sound energy level produced by the passive electromagnetic coils of the hexaphonic pickup. Building a 6 channel preamp turned to be an unexpected, but completely necessary step in the realization of the project. A proposed by Lex van den Broek (The Institute of Sonology) amplification circuit with a ratio of ≈1/25 was tested and based on the results of the test a prototype version of a 6 ch. preamp was built. *(Fig.4)*

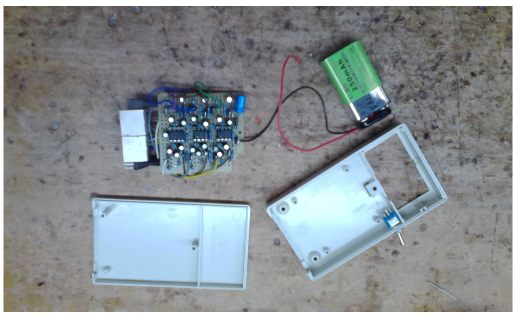

Figure 4. A 6 I/O preamp using 3 dual OpAmps and powered by a 9V battery.

Additionally, under the supervision and help of Lex I was able to assemble and design it in a compact way, which together with its cabling does not affect the quality of performance and the flexibility of the instrument itself. *(Fig. 5)*

Figure 5. The complete design of the hexaphonic preamp.

3.3 Design of the software toolkit. Composing processes.

The intellectual activity commonly known as programming or patching (in this case with Max/MSP), is something I prefer to call *composing a process.* This description, I find more closely corresponds to the way I have approached digital processing during the development of the system.

I tried to implement in it compositional strategies and principles which I find very powerful and which I have also applied in various ways in some of my previous works. One such principle is the *interdependency* or *interrelationship* of the elements in a system (this presumes that a musical composition is a complex system, which is manifested in time) whose strong existence we may find in the aesthetics and music of Anton Webern and the Second Viennese School in general. (Variants of such compositional thinking might be found as well in examples from different styles and periods of the history of music).

Ring modulation

The first module which I composed and in which I applied the *principle of interdependency* was a ring modulator with 6 sine/triangle oscillators, each of them modulating the sound of one independent input (string). *Ring modulation* was the first processing experimentation I have tried once I had the hardware stage of the project finished. Multiplying the sound of the guitar with oscillators with constant frequencies had little in the of sonic potential, largely because I one gets bored soon listening to the output always containing the same frequency, but to make the frequency of each oscillator dynamically change over time in a manner that emphasize the relation between all the 6 inputs seemed to be a worthwhile task.

Specifically, a change which will be driven by the process of sound generation itself in a causal relationship.

I achieved it by applying frequency analysis of each incoming acoustic sound and sending the estimation to an oscillator , but to one which is ring modulating with a sound from another channel, avoiding a situation in which a sound is multiplied with its own fundamental frequency. To organize that behavior for all the 6 inputs and further evolve the interdependency and interrelationship between the 6 incoming signals I builded 6 I/O matrix. In this matrix an incoming signal (for instance the signal of input 1 — meaning 1st string of the guitar) has a list of 6 options for ring modulation: with the frequency estimation of any of the 5 remaining incoming signals (freq. est. of the sound produced by 2nd to 6th string) or none (when 0 is sent, no change will apply - a sound is ring modulating with the previously received frequency). This six possibilities are alternating in random order until they run out and another 6 randomly ordered options are set.

I need to clarify here an important general behavior of my system, just mentioned above: every new generation of sound is also used as a trigger (often triggering a ***bang*** object) for a number of processes. One of those is the run through the list of 6 possibilities. 6 sound generations will make the system go through the entire list and start the process again. Additionally, a MIDI controlled multiplication/division factor is added for each oscillating frequency that has been sent out of the matrix. Thereby is possible to apply a frequency range tendency to the ring modulation process.

To summarize shortly: at this point the ring modulation module of my system is completed as principles of causality, limited randomness (6 possibilities in arbitrary order) and interrelationship between the 6 inputs have been implemented in an attempt to achieve an artistically worth output.

Delay Lines

Use of delay or delay lines have been a basic audio processing technique since the early days of electronic music. This fact may creates an impression of triviality, which might be avoided until certain extend by a very original and creative approach. In any case implementation of delay lines to my audio processing system seemed inevitable in order to achieve a complex processing.

Excluding the idea of fixed delay times and the very conventional mechanical output I started experimenting with variable delay time (meaning each next repetition will differ in length wit the previous one) and variable feedback level. The conclusion was that the output was not so convincing, differing just a little from the conventional approach. So , I abandoned totally the idea of having repetition of the original sound and went to combining pitch shifting and delay in a commonly functioning system in order to have a room for invention and creativity. I implemented a compositional strategy, which seemed to produce a convincing output and has the advantage to be related with the one implemented in the ring modulation module previously.

I composed a process in which an incoming sound is first delayed with a random time value varying from 512 samples (a single vector size) up to almost 3 seconds. It is then transposed to an interval randomly chosen out of 24 quarter - tone values covering a range of one octave. Only after that the sound is outputted. Delay time probability tendency (relatively short, relatively long) is controlled by MIDI.

The same interval of transposition is not repeated until all the 24 values have got used. The order is random within those 24. By default all the possible values are placed within a half octave (a tritone) above and half octave below the original pitch. Therefore a MIDI control for a range offset is added.

Quarter-tone scaling for the pitch shifter was chosen after experimenting with few other possibilities like a half-tone and a six-tone ratios. The first I found too trivial, the other — hardly noticeable. 24 quarter tone random order chooses in one octave is an idea that refers to the 12 tone paradigm and the compositional strategy of creating interrelationships one more time. As mentioned before, MIDI controlled range offset is important part of the expression as well.

The entire architecture of a single channel delay line currently consist of 4 delay + pitch-shift units. That means that an incoming sound will be delayed, transposed and outputted, at the same time sent to the second module — one more time delayed, transposed and outputted, but also sent to the third and so on.Once it reaches the end of the chain, it is fed back as an input. The level of the feedback can be adjusted by assigned MIDI control.

Triggering for any of the processes is again provided by the generation of acoustic sound.

In short, limited randomness (delay time within certain boundaries, dynamically changing time tendency, random order of the 24 transposition values) together with a composition strategy (24 different quarter tone values of transposition in an octave, none is repeated before all of them run out — reminiscence of the 12 tone technique) have been used to accomplish a delay lines module and to shape its behavior.

Granulator

The real - time granular synthesis module implemented in my system is based on an example developed by Johan van Kreij at his *Real-time processes with Max/MSP* classes from 2015/2016. Its architecture provide ability for dynamic control over number of parameters, whose output determines the entire behavior of the granulator. My contribution was to design this dynamic control in a way that will produce artistically worth output.

This was entirely done by an obsessive process of experimentation, observation, conclusion and adjustment of bunch of random numbers generation processes. Max objects as ***random, decide, drunk, select*** were implemented in a network which provides an output in desired ranges. This *limited randomness* is used to control parameters such as: transposition, duration, time offset of the grain. Sound generation provides triggers for the random numbers processes as well as in the previously discussed modules.

Feedback

The feedback paradigm is another very widely spread notion and sound processing technique as well. With enormous amount and variety of implementations, some of which became classics of the electroacoustic music, it is a good soil for achieving stronger expression of an audio processing system. At the current state of development of my project two different approaches towards implementation of feedback have been used. First is the very simple one of routing the digital output back as an input (before to reach the speakers, so no physical feedback). Since it is a simple linear circuit at the moment more work is intended to be done on it. The second I find interesting and promising for future extension — the mix of 6 outputs is sent to a small amplifier, then the amplified signal to a coil which the performer holds in his hand. By moving it very close to the hexaphonic pickup (acting as a receiver of the electromagnetic signal produced by the coil) the performer is able to send the mixed output to any of the 6 inputs (coils of the hexaphonic pickup) and thus distribute it in the space as well (in the case of having 6 speakers surround output).

4.Future development

As it was said in the beginning this is an ongoing project which may be considered as relatively complete only in a sense that in its current state it holds the necessary quality and amount of expression for an interesting performance. Such was already done ones, in order to test the system in a real performative situation. However, much more could be realized in order to achieve even more expressivity and interesting artistic output.

I distinct two directions in which I am working currently.

One is to program or to compose more processes and implement them in the already functioning system. Filtering and wave shaping are two processes in which I currently try to implement the composition strategies already described above.

The other is the "physical" or object related one including all kinds of preparation of the guitar or using objects directly affecting the electromagnetic properties of the hex pickup (E-bow, feed backing coil and a tea spoon are examples of objects that have been tested successfully).

It is important to mention also the technical and physical limitations as a determinative factor for the capacity of the system. Limitations as a CPU power and a number of MIDI controls possible to be handled by a performer playing an instrument at the same time draw a frame in which any future development needs to fit.