SpeakerDrum

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ABSTRACT

SpeakerDrum is an instrument composed of multiple Dual Voice Coil (DVC) speakers where two coils are used to drive the same membrane. However, in this case, one of them is used as a microphone which is then used by the performer as an input interface of percussive gestures. Of course, this leads to potential feedback, but with enough control, a compelling exploration of resonance haptic feedback and sound embodiment is possible.

Author Keywords

Speakers, Dual Voice Coil, Karplus-strong, feedback

1. INTRODUCTION

Since the invention and popularization of sound amplification through electronics, an interesting phenomenon took place in the perception of musical instruments and musical practice: The musical gestures used by musicians to produce sound were physically separated from the sound source that effectively produced the sound. In this sense, many instruments were enriched and acquired an incredible new range of possibilities. A clear example of this is the electric guitar in which even though the instrument is generally consider independent of the amplifier used with it, the musical experience is virtually impossible without an appropriate enhancer to expose the shy vibrations of an electronic instrument. In contrast, acoustical instruments have an intrinsic entanglement between the sound generation method and the sound source where a chaotic feedback relationship is nurtured and exploited to achieve expressiveness.

Whereas in the electric guitar the string vibration is affected by the different techniques the performer uses to pluck the string and by the way the standing wave in the string itself is building up in an enclosed resonant system, it is not, in general, heavily affected by the sound produced by plucking the string after it has been amplified by a guitar cabinet. Moreover, it is common practice in recording studios to physically separate the guitar player from the amplifier in order to push the limits of the amplifier and achieve textures not bearable without proper isolation [7]. In contrast, piano players are very keen about when is it appropriate or not to use the sustain pedal. In general, they try to avoid its use if the purpose is only to produce a *legato* articulation as it is often considered bad technique to use the sustain pedal to hide the lack of ability to achieve such articulation [1]. This, from an acoustic perspective, helps the performer to avoid contaminating the sound spectrum by over using the sustain pedal. Otherwise, raising the dampers allows strings to sympathetically oscillate as different keys are played producing a chaotic combination of pitches with undesired effects. The former is not a rule by any means; however, it shows how entangled the musical gestures and the produced sounds can be thanks to the inherent connection between the mechanical excitation and the sound itself. An important question emerges in the era of modern digital techniques: Is this separation a limitation of digital instruments? or is there a way to reintegrate the mechanical excitation and the sound itself in a cohesive system to achieve a broader range of expression within a digital musical instrument?

2. APPLIED PRINCIPLES

2.1 Physical Modeling Synthesis

It did not take too long for computer musicians to explore the possibilities of digital physical models by simulating their dynamical behavior to achieve a greater sense of realism. In this process, an additional layer of expressiveness was also achieved: the interaction with such digital simulations allowed the performer to affect such systems in very realistic ways. Additionally, given the resonant nature of physical models, the chaotic nature of acoustical instruments was revived. In this chaotic world, plucking a string repeatedly does not produce an entirely predictable outcome, in the sense that the state of the system determines how it is going to react and produce the plucks that are still ahead.

Nevertheless, achieving convincing ways to interact with such systems was still problematic. Products like the Yamaha VL1 where incredible windows to interacting with physical models; unfortunately windows with a hard glass that impede the interpreter to transmit musical gestures to these systems effectively [6]. Since then, new explorations of creating controllers capable of dealing with musical gestures in natural ways have been developed to drive such systems. Very often, piezo-electric microphones are used in conjunction to digitize striking or percussive gestures with great success [5]. However, the acoustic physical model is enclosed in the digital domain and therefore it is, very often, only reached by the interpreter through its input; the sound production element is still a speaker which has no inherent relationship with the mechanic excitation.

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2.2 DVC Speakers

In an attempt to reconnect the sound generation method and the sound production method, an interesting element seems to provide a reasonable interface for such connection. DVC (Dual Voice Coil) speakers are a particular kind of speakers built around a single membrane with two concentric coils where both of them drive the cone without sharing an electrical connection inside the speaker. They are often used in speaker design to achieve flexibility in impedance matching from an amplifier to a speaker as connecting the coils in series or parallel gives different impedance values. Additionally, some speaker manufacturers use DSP to feed different signals to each coil in an attempt to optimize their sum to produce a better performance of the speaker [2].

In this case, DVC speakers are understood as two electrodynamical transducers that are inherently connected by a mechanical bridge. Therefore, using one of the coils as an input and the other one as an output in the scope of physical modeling techniques would produce a single point of interaction where the performer has access to interacting with the system directly. In addition, the mechanical excitation and the sound itself are entangled reestablishing a coupled system where new explorations of expression can easily arise. Finally, this being bot an input and output point, the interaction is no longer unidirectional as the performer has a physical feedback of the instrument he/she is playing; therefore, providing haptic feedback while allowing a very organic interaction with the instrument.

2.3 Karplus-Strong like algorithms

The Karplus-Strong algorithm is well known as a simple physical model of a vibrating string [8]. By emulating the propagation of energy through a string modeled as a delay line and then feeding back the output into such delay line, it is possible to create a virtual resonating system. The damping factor, together with the delay length are then used to model the decay rate of the energy in the system. Moreover, including other elements in the process (filtering, source selection), an immense range of sounds can be achieved. The flexibility of the Karplus-Strong algorithm resides on the way it is bringing the concept of resonance into the digital realm; however, this is also a critical point for the algorithm. In the acoustic world resonance can be exploited in very diverse ways thanks to the nature of passive systems: given that the resonances produced can never have a positive net gain as no energy is being produced by the system, a natural protection against instability arises from the physical laws that govern the instrument. However, in the digital world, energy is a concept governed by ones and zeros where a simple miscalculation can lead to immediate unstable systems and, consequently, to uncontrollable disaster.

In addition, when this algorithm is interfaced with reality by means of electronics, several active gain stages appear introducing not only the possibility but the likelihood of instability. By proposing the use of DVC speakers in such configuration it is easy to foresee that parameters such as the gain values of the preamplifier used in the digitalization of one of the coils and the amplification factor of the amplifier used to drive the speaker will be crucial in the control and character of the system.

3. DESCRIPTION AND CONSTRUCTION

3.1 Physical Interface

The Speaker Drum was conceived as a percussive instrument composed by five 6.5” DVC speakers laid out in an hexagonal truncated pyramid. Each of the speakers is located on each of the faces of the pyramid leaving one face available for connections with the digital system. It was built from 12mm plywood following the sketch presented in *Figure 1*

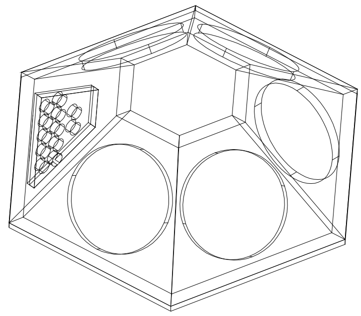

Figure 1: 3D Model of the Speaker Drum

Additionally, the top face is equipped with a layer of elastic fabric that can be pressed by the performer to cause changes in the timbral properties of the instrument. Such fabric is digitized with infra-red depth sensors allowing the performer to control a multidimensional mesh to change multiple parameters of the digital engine in real-time.

3.2 Electrical Connections and Signal Flow

From the electrical point of view, using a speaker as a microphone causes an impedance and level mismatch that needs to be solved by a simple resistor network. Such a network needs to address to some degree an attenuation of the signal as well as a correct coupling with the preamplifer used prior to the digitization of the signal; however, it also needs to be simple enough to provide an easy implementation. *Figure 2* shows the implementation of this resistor network. On the other hand, the digital system will return the signal

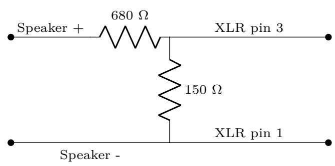

Figure 2: Level Attenuator

through a conventional digital to analog converter (DAC) but in order to drive the speaker needs to go through a power amplifier. The full schematic of such system is presented in *Figure 3*. Finally, the controllable mesh on top of the Speaker Drum is achieved through three infrared sensors attached to a micro-controller (Teensy 2.3) that is then connected through universal serial bus (USB) sending musical instrument digital interface (MIDI) messages to the computer driving the digital model. *Figure 4* shows this setup.

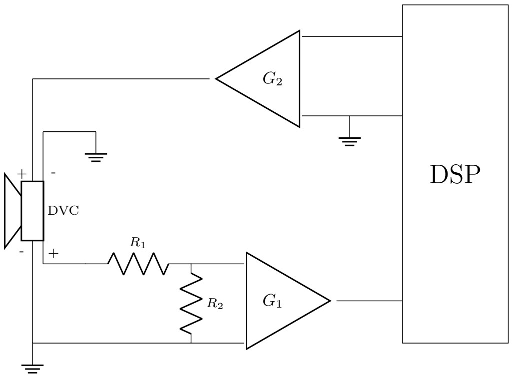

Figure 3: Full Schematic of the Analog Section

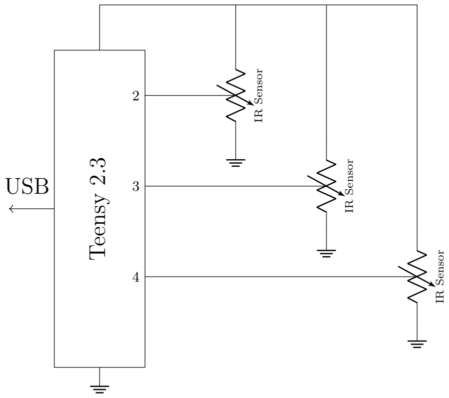

Figure 4: Schematic of the Teensy control section

All of the previous mentioned elements need to fit inside the enclosure previously described and interfaced with the digital system through the connection panel. *Figure 5* shows the connection panel and *Figure 6* shows a close up of the infra-red sensors to digitize the elastic mesh. And *Figure 7* shows the mesh itself in action.

Figure 5: Control Panel

3.3 Digital Model

The digital engine running the Speaker Drum is, to some extent, based on a Karplus-Strong algorithm as it uses feedback to drive the dynamics of the system; nevertheless, it differs in how the output signal is generated. Additionally, in this implementation some limitations arise from having the DVC Speaker is in the middle of the delay line.

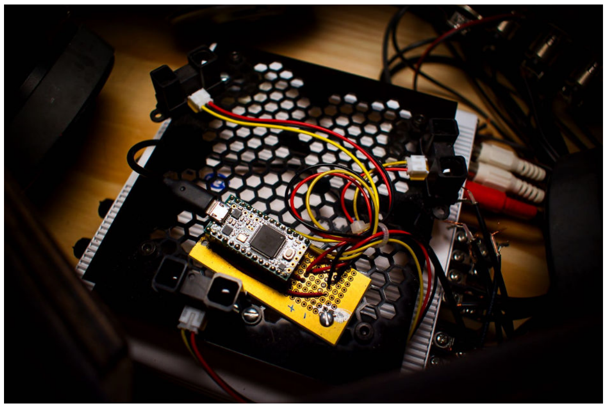

Figure 6: Infra-red sensors to digitalize the elastic mesh

Figure 7: Elastic mesh being pressed

3.3.1 Signal Coloration

Given that the speaker is in the middle of the delay line, the spectrum of the produced signal is immediately constrained to the frequency response of the speaker. In general DVC implementation of speakers are found in subwoofers, and accordingly, the high frequencies are impossible to sustain in the current implementation. Additionally, the transfer function between the coil used as a speaker and the coil used as microphone will determine an inherent coloration of the produced signal. Although, we often consider that speakers should have a relatively flat response, in this case, this is merely a constraint but not a weakness.

As with many acoustical instruments, the resonant system has its own coloration which will determine its timbre and will make the instrument unique. Moreover, due to the latency constrains of an ADC-DAC roundtrip the delay lines will have a minimum length determining the highest possible pitch in a feedback scheme. Finally, notice that not only the higher and lower ranges of the system are constrained by the usable bandwidth of the speaker; the flatness of the passband will also affect the usability of certain pitches. In that case, the impedance curve of the speaker will show peaks which will be more likely to cause resonances; yet again, that is also how acoustical instruments behave. For example, a wind instrument cannot play notes that it was not designed to play as it cannot sustain resonance - very much like the Speaker Drum - at certain frequencies due to its acoustical impedance curve.

3.3.2 Overall level

As stated before, the idea of the Speaker Drum raises some concern over the potential feedback that may arise from various amplification stages involved in the system and in fact this is a real limitation of the instrument. Given that the controls for amplification are all in the path of the feedback of the delay line, it becomes impossible to amplify the speaker drum beyond a certain point. Even though currently the level achieved is enough for the performer to play the instrument and hear a clear signal coming out of the speakers, the only way to achieve a louder signal is to strike the speakers harder. Just as an acoustic instrument, if more level is required, then more energy needs to be fed into the system. this will also be a limitation that the performer might try to cross with very likely the same results as an acoustic string that has been struck too hard to survive.

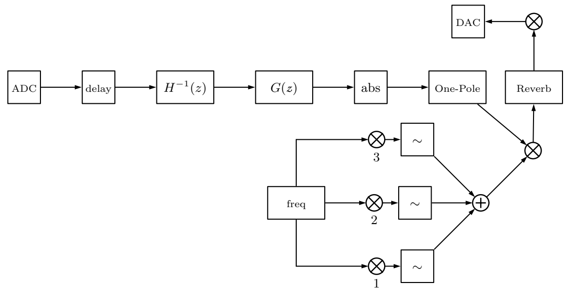

Figure 8: Signal Flow of the DSP Section

Taking into consideration the limitations of the Speaker Drum, it is now possible to show what the actual digital model is and how it behaves. *Figure 8* is a flow diagram of what happens in the system from the digital perspective. This was implemented in Faust (a functional programming language for digital signal processing (DSP) development [3]) with a simple code that exploits the parallelism of each speaker module to efficiently write the whole signal flow. Nonetheless, the configuration is not entirely easy, as all the parallel structures are initialized with exactly the same values. Due to this, and trying to keep in mind the requirement of low latency for the whole system, a ChucK (a strongly timed programming language [4]) script was used to control Faust through Open Sound Control (OSC) without affecting at all the latency performance of the system. Additionally ChucK is also used as a preprocessor of the MIDI data produced by the teensy from the upper mesh which is then used to control some parameters of the Faust sketch in real-time.

3.3.3 The Speaker Drum System

Figure 9 is a diagram of the whole signal flow of the speaker drum: From left to right it is possible to see that the input of the system is the membrane of the DVC speaker, which goes into a small resistor network to control the impedance missmatch and the excessive level. This then goes to a preamplifier and subsequently into an ADC converter to enter into the digital realm. In the digital realm the signal goes through two filters: an inverse filter to compensate excessive deviations on the transfer function from the speaker to the microphone in the feedback loop and a secondary filter used to eliminate undesirable resonant frequencies. Then the signal goes into the delay line, which determines to a great extent the set of sustainable pitches in that branch. After this, the signal goes through a full wave rectifier (implemented through a simple absolute value function) and into a one-pole filter which tracks the overall amplitude of the signal. The output of this filter is then used as the envelope control, through simple multiplication with three different oscillators configured as three harmonics of a base frequency. The relative level of this three oscillators is determined by the pressure applied to the top elastic mesh which is digitized with the infra-red oscillators shown on the right side of the diagram. A small amount of reverb is added to the signal which then goes to the output of an audio interface through its DAC into a power amplifier to finally drive the DVC speaker. This, in turn, feeds its signal into the other coil of the speaker closing the loop. Is important to notice that the loop is actually open in the digital realm and it is only closed through the mechanical coupling of the two coils in the cone of the speaker.

Additionally, it should be clear that even though the general algorithm resembles some qualities of a Karplus-Strong algorithm, it is by no means equivalent in many ways. Without loosing its resonant properties, the presented structure allows to produce specific pitches (defined by the frequencies of the oscillators and the delay lengths) with some control over the timber and envelope through the manipulation of the different gain stages, the oscillators and specially through the manipulation of the pole position in the one-pole filter. This enables a rich combination of possibilities that surprisingly fall within the same sonic realm. The end result is a cohesive sound similar to that of an acoustic instrument: a guitar may acquire many different colorations depending on how its played, tuned or even setup; nevertheless, it will always sound very much like a guitar. *Figure 10* shows the completed Speaker Drum system.

3.3.4 Fine Tuning

Given that the sound produced by each speaker is intended to be different from each other, the configuration of each Faust branch should be done in a different way. Each set of oscillators for each branch is meant to produce a different pitch which, in turn must work in conjunction with some delay length to support the intended spectrum. This is because even though the pitch has been determined by the oscillators, not only the speaker's impedance curve sets limitations on which frequencies can be sustained through a feedback loop, but also the delay length will determine which frequencies and harmonics can be efficiently stored in the oscillatory system. This is because the time relationship between the initial pulse and the produced waveform is critical to avoid destructive interference in the speaker itself which would result in an over-damped system. However, if the time relationship provides a delay length that can enclose an exact or close to exact period of the produced waveform, then the feedback loop creates a resonant system even though the wave is not propagating entirely across the delay line as an oscillating wave (due to the rectification process and the one pole filter).

This creates an interesting relationship between the digital luthier and the instrument were a fine tuning of the parameters of the instrument is required to design the sound produced by each speaker. Additionally, this fine tuning is then further affected by the fact that multiple speakers are all oscillating in the same enclosure; all of them are in a resonant state very close to producing feedback and all of them are prone or susceptible to become unstable if the gain staging is not properly handled. Nevertheless, whenever it is, a beautiful cohesive instrument arrises from the deep connection of the acoustical and digital realms which

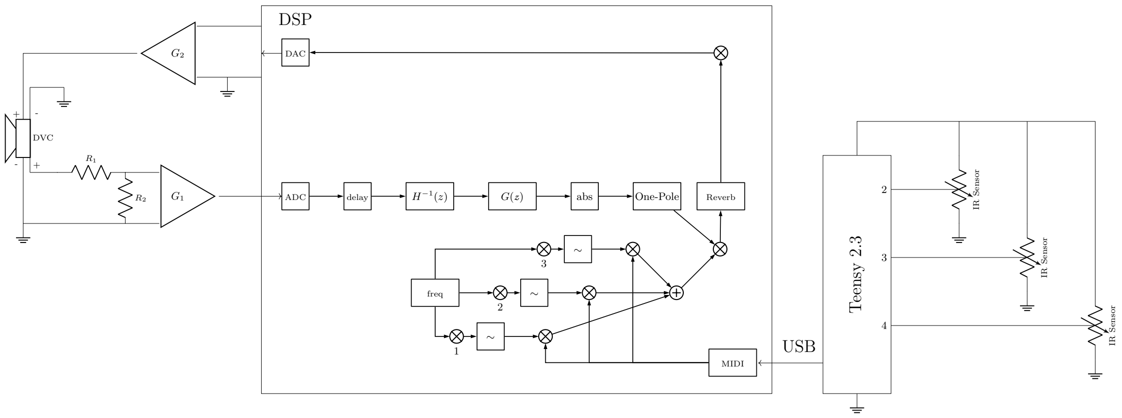

Figure 9: Signal Flow of the Speaker Drum

Figure 10: Speaker Drum Close Up

in all its complexity is incredibly simple in its interaction. Moreover, it is not only simple but organic and intuitive in how different interactions may affect it. For example, after striking one of the speakers the performer can easily stop the vibrations by touching the speaker and really interact with the sound production engine that is running in the digital world.

Table 1 shows a detailed configuration of the parameters of each branch and the intended pitches to be produced.

Midi Note	Pitch Class	tau (ms)	delay (smpl)
50	D3	20	47
53	F3	10.5	60
55	G3	2.5	54
57	A3	4.7	73
60	C4	30	48

Table 1: Fine Tuning of Speaker Drum

4. CONCLUSIONS

The speaker drum is an attempt to reconcile a very special relationship present in acoustic instruments and lost not only in digital instruments but even in electronic instruments such as the electrical guitar. When this relationship between the mechanical excitation and the sound itself is recovered, the corresponding bilateral effects between them are saved; an entire set of expressive possibilities is unlocked when the performer, with his/her musical gesture, can interact with these relationships acquiring greater control over what the instrument is doing at any point.

In this particular case, it was very interesting to see how reconfiguring this relationship by bending the definition of a DVC speaker and taking advantage of the physics behind it, reconstituted so many properties of an acoustical instrument which sometimes created limitations (such as the limit on the possible amplification of the system); nevertheless creating also a very synergic relationship where timber exploration regained relevance in the relationship between the performer and its instrument. Moreover the project brings the attention over the advantages of making hybrid resonant systems between the acoustic world and the digital world to exploit the best parts of both worlds and reconcile the unicity of a musical instrument.

Finally, the DVC speaker served the purpose, in this process, to be the bridge connecting output to input to create a resonant system, but several improvements and developments suggest themselves in this regard. Certainly the DVC speaker is not the only way to achieve this, but hopefully it served the purpose of being a clear higher order concept of how it is possible to create this hybrid analog-digital resonant type of system. At the end, the Speaker Drum became a compelling exploration of resonance, haptic feedback and sound embodiment in itself.

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