HYBRID AUDIO DEVICES:

"Analog concepts" in a digital age.

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Contents

1 INTRODUCTION

Although digital technology offers flexibility, sonic possibilities, and routing capabilities unavailable in most analog audio devices, many purists argue that analog circuitry still possesses properties that digital mimicry is incapable of replicating. Companies and other research interests continue to develop improved methods for digitally modeling electronic components, the audible phenomena which result from their behavioral properties, and the complex variables involved as separate systems interact. Regardless of how close digital modeling has come, analog methods of signal manipulation and physical music making still have a certain natural appeal. The result is an expansion in the area of rethinking dedicated digital, or pure analog systems exclusively and finding an optimal balance by hybridizing the best features of both platforms.

Along with the argument of audio system format comes a host of related questions of both a practical and philosophical nature which deal with modes of operation in relation to tools; in this case the discussion centers predominantly around sound generating and modifying tools, both acoustic and electronic. Aesthetic issues as well as those of physicality are important from the standpoint of performance usage, as well as the audience viewpoint of how a performer or musician interacts with these tools, and the achieved (or perceived) results. Many commercial devices such as Xbox Kinect, Nintendo Wii, and Apple's touch screen 'I' devices are finding market success by engaging the user through physical action. Developers, composers, and performers alike are not only attempting to appropriate these technologies, but also find ways to specialize them for musical applications.

The focus of this paper will begin with a general discussion of digital modeling. The importance of physicality in implementation will be discussed in relation to the analog counterparts, as well as the philosophies and benefits of each type of system. Digital versus analog, as well as an early commercially successful hybrid system, will be discussed in order to see the concepts of the argument at work. Finally, in order to consider the theory of hybrid system functionality and operation, I will investigate a few digital control problems in the context of a simple analog guitar effects pedal. The major focus in terms of actual audio will be on the manipulation and creation of live sounds, rather than recorded sound.

2 DIGITAL CONCEPTS

2.1 DIGITAL MODELING

All digital signal processing is based on the concept of sampling. Whether it be measuring the changes in temperature over the course of a day, or the changes in amplitude of an electrical signal over the course of a minute, the capability to analyze and compare these values comes from sampling. In audio applications, sampling is the process by which a continuous signal is measured and converted into discrete values at a defined time interval. The values are quantized, or truncated to the number of significant figures that the system is capable of handling (based on the number of bits). When stored, these values can later be accessed for manipulation and use; the storage period is based on the application. In many audio digital signal processing (DSP) applications where minimal input to output time is required, this storage period is on the order of micro, or even nanoseconds.

Digital modeling is the attempt to numerically represent the behavior of a system and how it reacts to a set of given inputs in the analog domain. What digital modeling also offers is the possibility of creating systems that are not possible, or could never physically exist in the real world. As an example, Miller Puckette describes a type of reverberation in a room where the walls amplify reflecting sounds, defying the conservation of energy.[1, pg.196] This is obviously impossible, however artificial means or modeling could be used to achieve this effect.

Intertwined with digital modeling is the idea of physical modeling, which attempts to precisely replicate a process through mathematics.[2, pg.265] In many applications physical modeling is used to model 'mechanico-acoustic' processes, but it can also involve the physics of electronic components as well. It may sound outrageous, but using these techniques it is possible to predict the behavior and thus the sound of a flute thirty feet long, a speaker that is 1/100th of an inch wide, or an amplifier that uses five thousand vacuum tubes in series. All of these things probably would not be physically attempted, with the aforementioned speaker as the possible exception, however digital techniques are limited only by the power of the imagination and ingenuity of the programmer.

Digital modeling essentially follows one of two modes of operation: macro or micro. In macro model, the behavior of an entire system is represented formulaically. In a micro model, smaller divisions of the signal path are represented as individual systems; additional realism can be accomplished by injecting a representation of the interaction between these systems. Using mathematical equivalences, micro models and formulas could be reduced in order to provide a single macro representation. The trade-off of macro representation is the sacrifice of easy manipulation of the control of single variables, versus the power of individual manipulation the micro level offers.

So what is the point of digital modeling if these devices already exist? There are a number of benefits to using DSP techniques which model analog devices. These include cost factor, availability, portability, and the possibility of simultaneously using multiple iterations of the numeric representation. Because the device is actually a set of discrete values, it can be cloned, transported around the world via the internet, or carried in one's pocket on a thumb drive given sufficient storage space.

The concept is that traditional hardware which requires expensive components in order to function properly can be replaced. Based on this reduced cost the tool will be available to a wider user base. The properties of many devices can be numerically summed into a single device, thus reducing the physical space required to store them, or as mentioned, transport them. In some applications dozens (if not hundreds on some of today's powerful computers) of the same device could be in use simultaneously with only a single numeric representation, whereas in the physical world, it would require the given quantity of the actual device. In situations where certain components are becoming scarce, digital modeling can attempt to preserve the heritage of their sound and properties.[3, pg.1]

The modeling community has generally relied on simplified and memoryless filtering processes where computing power is limited. In cases where computational power is available, wave digital filtering methods or directly solving non-linear state space systems using Kirchoff variables are preferable. [3, pgs.1-2] In certain cases, models have been derived using manufacturers' data sheets rather than concepts involved in fundamental physics.[4, pg.2] Commercial releases of modeling devices, both soft and hardware, are centered around a cost balance between sales and performance. Cost generally wins over performance and as a result musicians often find that modeled devices lack elements of realism. For this reason, coupled with the proven designs of many analog systems, many 'models' are kept from consistent, widespread professional use.

2.2 THE QUESTION WHICH REFUSES TO DISAPPEAR: DIGITAL OR ANALOG?

"Audiophiles have argued this question since the widespread release of compact discs in the 1980's, and there is no sign of the argument abating."

- Mark Ballora in "Essentials of Music Technology"[5]

Format wars are nothing new, and the debates surrounding them tend to be highly subjective and often times commercially driven. The classic example, and really the first format war of the electrical age was the battle for who would electrify the United States. Thomas Edison was a proponent of utilizing direct current, while George Westinghouse was proposing to use alternating current (AC) in order to distribute power throughout the country.[1] [6, pg.53] Edison went so far as to electrocute a series of animals in public demonstrations of how 'unsafe' AC was - including an Elephant.[7, pg.141] Luckily, no one is harming animals over the digital versus analog debate, but it is still a prominent issue weaved into many aspects of music technology.

There are a number of key points that surround the argument. Because it is based on sampling, digital is at best an approximation of a continuous analog signal. To many, analog sounds have more character, are warmer, richer, or more interesting than their 'sterile' digital counterparts. The error caused by digital quantization results in varying types of (undesired) distortion, or wave shaping; these can produce different harmonics, which in turn affect the listeners perception of timbre.[5, pgs.103-104]

From a functional standpoint, digital and analog systems have very different power supplies. All electronics function optimally with power supplies that can be considered clean. In the case of audio equipment, this means the voltage the unit was designed for, coupled with a current free of electromagnetic interference, and other forms of noise. Many (most) digital logic applications are currently based around one of three standard direct current systems, versus a slew of various voltages that other analog systems implement. Most digital circuits operate with 0's and 1's, represented by high and low states, or voltage and ground. These circuits tend to be intolerant of variations in either supply voltage or ground, and do not operate as designed when power supply values drift. Analog circuits have a tendency to be more tolerant.

Digital does have its clear advantages over analog methods in many applications. Digital has lower background noise. The discrete numbers which represent the signal are more reliably stored and are easier to manipulate.[5, pg.104] Error checking can be included in digital systems so that if information is lost, it can be recovered with little to no deviation from the original. Digital media, unlike tape or vinyl, does not

[1] Westinghouse' AC was a product of research done by Nikola Tesla; years later Tesla would propose a system of 'free' energy.

degrade with playback, meaning it could be read out a billion times, and it will be the same playback all billion times. Copies of digital media are in fact clones, bit-by-bit identical twins of the original. An analog copy might be compared to a set of fraternal twins who have striking similarities, but are not 'genetically equivalent.' In digital methods, a billion copies of the billionth copy of a digital representation could be made, and every resulting copy would be identical to the original.[2]

A great deal of the digital versus analog debate stems from psychoacoustics, and also market driven tactics of commercial interests. Regardless of the format or purpose of a product, most companies will emphasize that the consumer can hear the difference between their equipment and another manufacturer, and how their current unit is an improvement on the existing technology. The psychoacoustics portion of the debate deals with the connection between physical sound waves and how they are perceived by the listener.[8, 173]

One of the heavy points of contention, and probably the long time heart of much of the argument is the concept of sampling rate. The sampling rate is the speed at which a digital representations discrete values are measured and stored. In order to effectively represent a given signal, the rate at which it is sampled must be twice the signals highest frequency content. The upper frequency limit a particular sampling rate is able to handle is referred to as the Nyquist frequency, after the father (one of the fathers) of the sampling theorem, Henry Nyquist. [9, pg.22] The core of the argument comes from the fact that most human hearing only extends into a region around 20kHz. In theory, a sampling rate of 40kHz would be sufficient to represent most signals perceivable by the human ear.

Although it may seem counterintuitive, there is more to sound than hearing. Based on interpretations of human perception, scientific experiments confirm the effect of sounds above 22kHz as having impact on the listener from both physiological and subjective standpoints.[2, pg.31] In other words, higher sampling rates, provided they are reproduced on proper equipment, can make a difference in the listener's overall perception of realism in a given audio event. Additional tactile stimulation received during performances, and in playback, can also play an important role in reception of musical content.[10, pg.2] This is a key defense in the argument for both higher sampling rates, and/or the use of analog equipment in order to mitigate digital frequency bandwidth limitations.

The secondary defense for higher sampling rates is attached to the discussion of problems caused by necessary anti-aliasing low-pass filters used when a digital signal is converted back to the analog domain. Often times filters with sharp cutoffs, such as the 'brick-wall', incur phase delays at given portions of the spectrum and create sharp resonant notches, or gains, near the cutoff frequency. According to Chamberlain, it is often advisable to have a sampling frequency which exceeds Nyquist by more than two and a half times the maximum frequency which one wishes to sample. This helps to mitigate the phase and resonance problems of higher order filtering processes.[11, pg.111] By having a higher sampling rate, a lower order and thus simpler filter with enough roll-off can be used to attenuate frequencies gently over a longer spectral range.

In order for digital to totally replace analog in signal manipulation, ultra-high sampling frequencies would be ideal, however they are also increasingly expensive, both computationally and monetarily speaking with every step-up in performance. As with digital modeling, increasing the load puts strain on processing power, or forces the switch to more powerful and therefore expensive hardware. Although personal computing has come a long way, many audio applications are still resource intensive operations. This is particularly true in cases where a 'real-time' response is desired for a real world audio signal input. As will be discussed in

[2]The exception to this is the Minidisc, but the number of plays it has is so high that this is almost an arbitrary factor.

portions of the following sections, a balance generally needs to be achieved between the amount of power required to process a signal, and the acceptable time difference of processed output from input.

3 TACTILITY

"Our bodies are important in many ways. Most of those goals I spoke about,the ones we attempt to solve using our intelligence - have to do with our bodies: protecting them with fuel, making them attractive, making them feel good, providing for their myriad needs, not to mention desires."

-Ray Kurzweil in "The Age of Spiritual Machines"[12]

3.1 WHAT IS A MUSICAL INSTRUMENT?

"I've always believed any instrument, electronic or mechanical, should be hard to play. Overcoming resistance in the control of musical materials hone the human sensibility in contact with them and makes music performance and, by implication, composition the high arts that they are."

- John Eaton in "This is an Instrument!" [13]

In 1994, Bob Moog and Herb Deutsch were called before a court as expert witnesses in an import tariff dispute between the United States Government and the Japanese company Casio. The products in question were Casio devices, and the question posed was should they be considered machines or musical instruments? [14, pg.306] Moog's response was that musical instruments should have real-time control and that the Casio instruments lacked this. Because they contained an amplifier and speakers, the courts reached a verdict that a higher tariff could be charged for a musical instrument. According to Chadabe, an electronic musical instrument can look like anything, whether it be modules in a rack, a computer, a lot of little gray boxes, or "like a violin, or for that matter, like virtually anything."[15, pg.215] The point he is making is that performers will select the tools they use based on the style, and musical position they want to occupy. From sticks as percussion instruments to the modern day appropriation of playback devices, like the tape machine, turntable, and Ipod, human beings have utilized various sound tools based on function, availability, and desired results for millennia. With increased computing power, the concept of real-time and interactivity have come to the fore of many peoples musical conscious in examining the nature of electronic instruments. Traditional methods for computer control, such as the mouse and keyboard, do not necessarily provide a level of flexibility commensurate with the power of many software applications. There are also still aesthetic issues with the laptop as a musical instrument, where audiences feel it is in "violation of the codes of musical performance." [16, pg.101]

Interaction with an instrument, as well as instrument designs are manifestations of the goals that Kurzweil brings up. They represent problems which require strategies for dealing with the human desire for expression. Like many things the definition of what a musical instrument 'is' can be rather subjective. What will be made clear is that an instrument is what a user feels comfortable working with in order to achieve their artistic vision. On the same token, performance aesthetics also dictate what an instrument should be, defined by what type of visual information should be offered to an audience member in conjunction with auditory stimuli to create a universally meaningful experience.

3.2 THE PERFORMER/COMPOSER PERSPECTIVE: USING THE TOOLS

"The best user interfaces are both easy to work with and, not coincidentally, incorporate deeper knowledge of musical structure and music-making strategies...as systems become smarter there is a temptation to let them impose their knowledge."

- Curtis Roads in "The Computer Music Tutorial"[2]

Instrumental gesture is defined as a method of operation specific to the given gestural channel, and the physical interaction and dynamics can be mastered by the subject.[17, pg.10] Feedback to the performer can be offered in a number of ways including primary feedback which acts upon sight, hearing, or touch, while secondary feedback is the sound directly produced by a given instrument. Passive feedback is provided through the physical characteristics of a system and active feedback is produced by the system in response to a user action.[17, pg.11]

Because of the type of responsiveness that humans naturally experience through the five senses, instruments which utilize some sort of physical interface seem to be commonly preferred over software approximations among many musicians. Ergonomics is the study of how objects are designed in relation to human proportion and gesture.[2, pg.642] Advocates of computer and electronic music have long been exploring ways to ergonomically employ physicality as input for the control of these applications. Systems can also offer users kinesthetic (haptic) and/or visual feedback. Control surfaces used for interfacing with software, whether musical keyboards or groups of knobs and faders, are a prime example of intuitive interfacing. "In any well-designed physical computing application, the flow of activity between the person and the computer should follow the same comfortable flow of a good conversation." [18, pg.181]

The user interface of any tool is going to define its controllability, and it can be stated that the interplay between musician and instrument is a very significant aspect of musical expression itself.[19, pg.2] Although software and other digital applications often offer an increased number of variables, accessing them in a clear and natural manner is not always easy. The functionality of any system is a result of the creativity and foresight of the designer, and these limitations transcend the designer shaping the users experience. Considerations of physicality, or lack thereof, can be a limiting factor in ease of use. In the days when digital was not an option, devices offered one control for each variable in a system. This gives the user a clearly defined structure for input and control, based on an advantageous one to one ratio.

An issue with usability surrounding many digital, and especially software based applications, is the concept of latency. In situations where performers are used to the responsiveness of acoustic instruments, latency can become a significant barrier in their ability, or desire, to utilize the given tool. Even the lowest of lag times from input to output can be noticed by the performer. This concept is known as temporal precision, and is necessary in capturing the nuances of performance which contribute to emotional meaning and content.[2, pg.643] Specialized hardware generally mitigates the issue of latency but not in all cases. Latency inherent in a system can also dictate its usage; for instance pitch to MIDI conversion units, as advanced as many have become, can still have trouble tracking low frequencies. Realizing this limitation many users may choose to either play higher notes, or keep the notes in the lower register longer and less florid.

Using a keyboard and mouse also seems counterintuitive to the types of workflow that are associated with analog hardware. Although it is arguable that computer applications shouldn't be used for the same types of work, it's unarguable that they commonly are. The fact that many commercially produced software plug-ins, synthesizers, and digital audio workstations are designed to resemble their analog counterparts proves this point. According to O'Sullivan/Igoe, "Computers should take whatever physical form suits our needs for computing."[18, pg.xvii] In this instance, users needs are comfort and familiarity. The newest prospect in intuitive controllability comes in the form of capacitive multi-touch screens as seen in devices such as the Apple Ipad, and Hewlett Packard's 'TouchSmart' series of computers. The future may see additional user friendliness applied in the form of virtual reality. An extension of this concept is augmented reality where computer generated images are layered on top of the physical world through some form of visual or heads up display (HUD). [20, pg.166] Although virtual and augmented realities are being released now, they aren't yet mature and widely available technologies. The only limitation of any of these systems is what can be effectively represented in the field of their viewing area.

3.3 THE AUDIENCE PERSPECTIVE: VIEWING MUSICAL PERFORMANCE

"The question of gesture is crucial in music. It lies in the intersection of two axes: one that binds together an observable action (the gesture of the instrumentalist) and a mental representation (the fictive movements evoked by sound forms), and another one that establishes a link between the interpreter (that produces the gestures) and the listener (who guesses, symbolizes, and transforms them on an imaginary plane)."

- Francois Delalande

Inputs and controls are important for the performer, but equally important is the correlation of gestures to a resulting sound for the audience. Removing correlation from gesture and sound can diminish the ability to convey emotional content. This is not only from physical gestures as they occur, but also when audience members anticipate events based on the physical activity of the performer.[21, pg.192] Visual areas of the brain that deal with identifying, or extracting correlation and rules, have developed through evolution to be directly linked to emotional limbic structures.[22, pg.84] It is hard to ignore the physical interaction of a performer and an instrument, regardless of what that instrument is. In many instances, it is also difficult for audience members to find coherence watching a performer whose physical action does not reflect the nature of the sound being generated.

The earliest electronic music was performed resembling the traditional aesthetics of previous centuries. Before digital technologies, MIDI, and even other analog electronics, instruments such as the Theremin, Ondes Martinot, Croix Sonore, Ondioline, and others, utilized the natural movements of the human body as the source of their inputs. [2, pg.622] Although the Theremin is unique in that no physical contact with the device is necessary, there is an easily observed link between action and sound. After World War II, the technology boom resulted in a wide variety of newly available consumer items, but among the post war developments was tape. Musique Concrete and other electronic forms materialized from behind studio doors carried on these reels. A new performance aesthetic emerged - one devoid of performers. If the concept of 'static' loudspeakers are not considered a 'real' source, then this is similar to the concept in cinema called Diegetic sound. This is where the source is neither visible to the observer, nor is it contained within the story space.[22, pg.101] The concept radically altered, or at least challenged, the nature of what constituted a performance. Although total control had been achieved for the composer, something was missing for many audience members.

For this reason, the tape and related concepts like modern day laptops, have been strongly challenged media. Where was the performance if no one was on the stage? Why should people come to a concert hall when they could purchase the recording and listen at home? What is that person doing behind that computer? Composers who were not interested in the tape aesthetic, began to experiment with building electronic instruments and called upon the skills of engineers to solve the technical problems which were the barriers to their desired results. According to Gordon Mumma, many composers completely abandoned the tape media in favor of attempting to work with live electronic musical instruments, and especially in conjunction with acoustic instruments. [23, pg.292]

A new generation of electronic instruments emerged from the collaboration of composers and musicians with individuals such as Dr. Robert Moog, Don Buchla, and Peter Zinovieff. In the late 1960's and early 1970's composers also began exploring the possibilities of utilizing live instruments and parameters of the live instruments' sound in order to control electronics. In 1974 and 1975 Morton Subotnick composed Two Butterflies and Before the Butterfly.[15, pg.217] These works used muted violins as the control sources for voices on tape, in what Subotnick effectively called 'ghost' pieces.[15, pg.217] Although the sound of the violins themselves was not apparent, the physical gestures of the performers could be correlated (at least somewhat) to parameters of the tape playback.

There is still a sensitive balance between performers use of tools in an artistic manner, and the inclusion of an audience. "Electronic music, even when designed to be interactive, can lack performance interest and is frequently musically unsophisticated."[19, pg.1] Although electronic media in general have represented a paradigm shift in musical thinking, it may be beneficial to look at the successes of acoustic music and allow some of its aesthetic principles, both musical and performance, to influence future artistic decisions.

3.4 DEVELOPING A RELATIONSHIP WITH AN INSTRUMENT

"If I built a synthesizer, it would be fairly unpredictable - that's one of the things I feel is missing with synthesizers - a personality."

-Brian Eno in "Analog Days" [14]

Whether referring to a Stradivarius violin, or the pickups of a Fender Stratocaster electric guitar, musicians often discuss the 'sound' of an instrument. This sound is a general term referring to the makeup which defines what the audible qualities of an instrument are. Our individual aesthetic preferences and intellectual curiosities may define why each of us are drawn to certain sounds, but what is clear is that over the course of history particular instrument builders have created tools with attributes that both performers and audience members find highly desirable. Although attempts are made at scientifically dissecting what gives particular instruments these powers, part of the intrigue lies in the mystery and the circumstances surrounding their production.

The force of gravity exerted by these instruments has the ability to draw in and hold the interest of performers and musicians so that over time they can achieve the mastery that John Eaton referred to in the previous section. Although this mastery is traditionally associated with acoustic instruments, the question must be asked, is it possible to develop this type of relationship and mastery with an electronic or software instrument? According to Suzanne Ciani, she was "in love" with her Buchla 200. [14, pg.155] When the RCA Mark II Synthesizer at the Columbia-Princeton Electronic Music studio was vandalized beyond repair, composer Milton Babbitt abandoned computer music composition. Composer and performer Andrew Schloss has continued to upgrade and perform with Max Mathews radio baton in an instrument he calls the radio drum for over two decades.[15, pgs.231-234]

It is clearly possible to develop similar attachment to both acoustic and electronically based methods of working, and the type of workflow or performance aesthetic that the given instrument provides. An electronic instrument can be mastered in the same way that any performer gets to know the specific mechanics of certain acoustic instruments in their collection. As was discussed previously, it is easy to see how the layout of a tool affects this workflow.

What is questionable, particularly with Moore's Law and its sub-variants in mind, is whether or not in a rapidly changing technological field certain tools exist for long enough to even be mastered? With considerations of ever changing platforms in hardware formats, operating systems, and lack of backwards compatibility in many cases, can technological tools survive long enough in order to have someone master them? Many software packages are now reaching mature states where their expanded functionality has evolved significantly over subsequent generations or versions. What happens to the time invested users of these software packages when a company decides to discontinue this software or it fades from the market? A similar question can be asked of any electronic instrument - what happens when it breaks and replacement components are no longer available due to obsolescence?

Because of this possibility many technological applications, and particularly software, represent a relationship with a strange and possibly infinitely steep learning curve. Although the user's creativity and modes of thinking could be continually challenged by the use of new systems, it still begs the question of the ability to master and form a meaningful relationship with the tool. Music, as a living art, provides its practitioners with continual opportunities for evolution, but evolving when the tools continue to change in conjunction with philosophies and musical materials has a very different effect. A violinist dealing with these challenges occupies a different position than one whose main instrument is the laptop. The violin will not change very much, while the laptop and its subsequent function will continue to rapidly shift, drawing attention from the mastery of the focal point in harnessing the musical materials - the tool itself.

4 CLOSING ARGUMENTS

4.1 AFTER ALL OF THIS...SO WHY DIGITAL?

"Human interaction with computers, as with any technology, is a creative partnership with potential for meaning making that has become an important idea for musicians."

-Steve Dillon in "Meaningful Engagement With Music Technology" [32]

Although the argument between analog and digital continues, there are some clear cases where digital technology has advantages and possibilities that analog could never offer. With a newer generation of computers, programmers, and forward thinkers, a fresh ensemble of audio processes have emerged. Processes such as convolution are much easier to implement in the digital domain and have opened the flood gate to new sonic possibilities. Effects based around the Fourier Transform, such as the phase vocoder, sinusoidal modeling, and tools for musical analysis have the potential to be more powerful and flexible than any previous analog method.

Quite simply, digital should at a minimum be used for the things that analog is incapable of. With this in mind it is easy to see one significant answer to "why digital?" A major advantage of digital implementations is the ability to recall data. Whether this is in the form of user presets or patches, or for use in machine learning applications, no analog application has a memory capable of random access the way that digital systems do.

In a discussion of memory and cognition, composer and electronic music pioneer Morton Subotnick discussed his early process of keeping track of how he achieved sounds. In order to be able to approximate or recreate sounds in the event he would want to recreate them, Subotnick had to document all of the settings of his Buchla 200 and accompanying devices by hand, in a notebook. [24] This very scientific approach led to the ability to recreate sounds, but the amount of time it would consume is astounding when considering the digital capability to store thousands of variables in one push of a button or keystroke. The thing that his notebook also contained were his thought processes, something that isn't currently a feature of most preset storage applications.

Digital methods of signal manipulation and creation have the potential to be as exciting as their analog and acoustic counterparts for not only the users, but also audience members. Drawing from the concepts set forth in the previous four sections, it should be possible to provide the designer, user, and audience member all with a satisfying experience. Unfortunately, in an age of technology, users are often mesmerized by the technology rather than the manner of expression it can be used for. We have reversed form over function. Technology is not a replacement for artistry, and technical prowess is not equivalent to artistic merit. Unfortunately, in a society ever impressed by speed, capacity, size, and gimmicks, it is difficult for many to separate the two.

4.2 SO WHY ANALOG *OR* DIGITAL...WHY NOT BOTH?

"To me, it's (the Rhodes Chroma) the best of the analog synthesizers in terms of the way it sounds and how flexible it is. It's sensitive, and it feels good. What you do is what you get. In other words, when I want a certain sound, my finger will do something, and I get what my finger does. I hear what I expect to hear. There are not many instruments, and on the ones I've had contact with, I may not always get back what I hear in my head when I try to execute it."

-Herbie Hancock in "Keyboard Magazine", 1983[25]

What Herbie Hancock described is the essence of the answer to the question, "why not both?" The Rhodes Chroma offered him everything he seemed to want, and based on previous discussion, the things that give any type of tool its character. The unit, originally boasting a three to five-thousand dollar price tag, was one of the first commercially successful hybrid synthesizers, and one of earliest keyboard instruments to successfully implement velocity sensitive keying. [26, pg.184] The Chroma's (non-keyboard) physical interface was a series of membrane switches with corresponding LED indicators accompanied by two and eight digit displays. In order to preserve the tactile feel of traditional keyboard switching, the manufacturers implemented a solenoid thumper which would tap the panel, letting the user know even in loud situations, that the switching action had been executed.[11, pgs.715-716]

All oscillators, VCOs, VCAs, and traditional synthesis components were implemented with analog circuitry. Envelopes were generated via software and converted using digital to analog techniques.[11, pg.716] Control of the entire Chroma was accomplished using only an 8-bit microprocessor, and limited amounts of random access and EEPROM memory. Given the available technology at the time the level of sophistication and control achieved is astounding. At the time, the tool represented the best of everything previously discussed - at least for Herbie Hancock it did. As with all types of development, the Chroma represents an arrival point.

During the 1960's, significant developments in voltage controlled synthesis techniques allowed for varying voltages to control any number of parameters designed into a system. [17, pg.221] In the earliest days of using computers and digital sound generation, methods for creating pieces which processed sound in 'real-time' or a live manner were being codified. Specifically in the late 1970's and early 1980's, digital devices were being employed in order to control analog systems. [17, pg.222] Although these hybrid systems have become more elaborate and/or compact, particularly with the introduction of miniaturized surface mount technology, they continue to employ the same basic methods of applying varying voltages in order to control parameters. Commercial and research interests continue to invest resources in the race to find the perfect digital models of analog; what is generally achieved can be referred to as acceptable. Despite a great deal of focus on digital models, a number of companies and independent instrument designers continue to build hybrid devices which offer the appeal of both worlds. Moog Music sells a line of digitally controlled synthesizers with a completely pure analog audio path. (http://www.moogmusic.com/products) Solid State Logic offers a number of hybrid mixing consoles. (http://www.solid-state-logic.com/music/consoles.asp) Line 6 corporation is now producing an amplifier called the DT50, which is a digitally controlled pure valve amplifier in both the pre and power stages. (http://line6.com/dt50)

What is being realized by some, is that only working digitally with computers is not always an entirely rewarding activity.[19, pg.2] The result of the products being offered is the kind of user satisfaction that is obviously desired by many consumers, coupled with the properties that these same consumers have, for lack of a better term, 'fallen in love with.' Although it is arguable that some of these devices are part of niche markets, if individuals were not interested in these types of devices, and proving it by purchasing them, (intelligent) companies would not continue manufacturing them for extended periods of time.

5 THE PROJECT: A MICROCONTROLLED ANALOG DISTORTION PEDAL

Although there are hosts of methods for producing varying voltages and controlling analog circuitry through digital means I attempted to explore some of them for use in a project which will be fully realized in the near future. I wanted to begin with something familiar and experiment knowing that the first portion of functionality was already in place - the analog circuitry itself. The project that I have decided to start with is one that I successfully constructed twice in the past, and that is 'Project no. 24, Tube Sound Fuzz' from Craig Anderton's book Electronic Projects For Musicians.[27, pgs.170-173]

The project is actually perfect for a number of reasons. The circuit itself is simple, but it presents a variety of unique design problems. The unit itself only has two potentiometers, or variable resistors, which control parameters of gain (i.e. drive/fuzz), and output volume. One potentiometer is linear, while the other is logarithmically tapered, potentially offering two different challenges in code architecture when the microcontroller comes into play. There are also two switches, one with an on-off state, and the other with a continuously-on selectable 'A/B' state. All of the parts utilized can be obtained in a through-hole format, and therefore easily swapped and tested on a solder-less breadboard.

In order to make the actual attempts at controlling the unit, I had to select a control platform. From a cost standpoint, there are many inexpensive microcontrollers, but being able to procure free samples from Microchip Corporation, the amount of code for the device family floating around on the internet, and the ability to program in a C compiler (versus assembly language) made the PIC seem like the right choice. Coupled with the fact that I already owned an EEPROM burner capable of programming many PICs, the decision was relatively easy. For the purposes of testing code, I utilized Microchip's free design suite MPLAB.

Stepping backwards, what is a microcontroller? How is that different from a CPU or a microcomputer? Most general purpose computing applications are based on the 'Von Neumann' architecture, named for John

Von Neumann and share common features in their general architecture.[28, pg.1] The term microprocessor generally refers to the implementation of the central processor unit functions of a computer in a single, large-scale integrated (LSI) circuit.[28, pg.1]A microcomputer, is a computer built using a microprocessor and a few other components for the memory along with the input and output (I/O). [28, pg.1] The microcontroller is a microcomputer with its memory and I/O integrated into a single chip [28, pg.2]For many applications, microcontrollers are a relatively simple, affordable, and practical solution in system design.

The first problem to approach was the simplest : switching applications. In the design there are two switches - one for bypass and a second for high versus low gain settings. Simple switching is easily accomplished on the PIC microcontroller since it is only capable of creating two discrete (binary) states at any output pin, on or off, equal to 0 volts and 5 volts (logic level). Looking at effectiveness and noise specifications as dictated by Douglas Self, I decided to use relays as my switching mechanisms over other methods like CMOS Analog gates. According to Self, "relays give total galvanic isolation between control and signal, zero contact distortion, and in audio terms have virtually unlimited signal-handling capability." [29, pg.397]

In both cases double pole, double throw (DPDT) logic level relays were selected. By using a DPDT as the bypass switch, what is generally referred to as a 'true effect bypass' is achieved where there is no additional loading on devices which follow in the signal chain when the unit is off. For the gain switch, only one half of the contacts were utilized, in essence having the DPDT act as a single pole, double throw switch. In programming, two input pins on the PIC act as input sensors; when either of these pins receive a voltage, they output a voltage from the software designated control pin which activates the relays to place the pole in its appropriate position.

Switches were a simple solution because the control for input that they were replaced with was essentially the same thing as the original (only cheaper). Potentiometers represent a different type of challenge that is threefold. The first problem faced is what component is utilized in the circuit to accept microcontroller input to create a variable resistance? Second, what is used to replace the physical input for control? Third, depending on the type of input utilized, how is feedback given to the user to let them know the current setting of the control? In all three cases there are a number of effective answers that become a balance of cost versus ease of implementation.

The simplest and most direct replacement for a potentiometer would be a motorized potentiometer. There is no additional design in terms of the audio portion of the circuit, it adds no additional noise or distortion, manual control is always available, and if the memory is non-volatile, it holds its position when power is removed. [29, pg.239] By having a traditional knob with a tick mark to indicate position, the interface that many users are used to would not be altered. Other viable alternatives are a number of specialized IC's available which handle volume (gain) applications, but also a new wave of digitally based potentiometers being released by manufacturers such as National Semiconductor and Analog Devices.

There are three distinct parameters the selection of a digital potentiometer must be based upon: the value, the taper, and the resolution. The value represents the maximum resistance the device is capable of. The taper has to do with how the potentiometers resistance changes and some potentiometers are tapered linearly while others, particularly those made for audio use, are logarithmic. The resolution deals with the number of discrete steps between the minimum and maximum values. Finding a device with all of the appropriate values, and in an IC capable of handling the operating voltage of the audio circuit are the major drawbacks of this solution. Although interfacing with the microcontroller is a fairly easy task, without redesign this is not a viable control solution for not only this project, but for many circuits.

Although it was earlier stated that output pins on the PIC are only capable of low and high states, there is at least one exception to this rule - pulse width modulation(PWM). Many PICs offer a 10-bit PWM resolution which provides 1024 discrete steps between 0 and 5 volts. This allows for a wide enough range to accomplish many tasks based on voltage controlled resistance (VCR). PWM uses very high frequency square waves which alternate only between pure on and off states. The length of the pulse, or the duty cycle, determines the effective voltage being output and this creates a voltage level generally referred to as a 'pseudo-analog voltage.' [18, pg.112] In many cases a capacitor is utilized as a simple low-pass filter in order to smooth discontinuity from the pulse signal.

The first method with VCR, is to utilize the field-effect transistor (FET) as a voltage controlled resistor. The FET itself is a voltage controlled device, although it can also be referred to as a current driven device.[6, pg.82] Where traditional transistors are referred to in terms of their base, collector, and emitter, the FET is described by its source, gate, and drain. In FETs, the current across the drain and source is directly controlled by the voltage applied to the gate pin.[30, pg.1] Applying more voltage to the gate allows an increased amount of current to flow from the source to the drain, or vice versa depending on how the transistor is biased. By decreasing the gate voltage, the resistance of the device is increased, thus it is fairly clear to see how this could be substituted for a potentiometer.[3]

The second option under consideration for the VCR methods, are opto-isolator methods, principally a component known as the 'Vactrol.' A key to successfully using the digital control in a system such as this, is to keep the control portion isolated since stray voltages and external loading can cause unwanted disturbances in the performance of the audio path. The opto-coupler, and opto-isolator are often used to accomplish communication 'in isolation', especially in data transmission. As a precaution, the MIDI protocol requires an opto-isolator in the path of MIDI input circuits to protect against damage from an improperly transmitting device.[18, pg.303] Vactrols themselves have found common use in many audio applications from gain control in tube amps and compressors, to novel effects like vibrato and tremolo. The opto-coupler consists of two basic parts, the LED (light emitting diode) which transmits light and a receiving photocell. The source and sensor are often packaged into a single integrated circuit.[31, pg.45] The photocell varies its output accordingly with the intensity of input it receives from the LED. The Vactrol is essentially this type of packaged emitter and photo-resistor which comes in a variety of values, but always the same 4 lead package, and is specifically designed for varying resistance. Due to its isolated sections, the Vactrol is a relatively easy component to implement. Ultimately there are trade-offs with any part : Vactrols are only currently available through a handful of distributors, some of whom require large quantity orders, and the roll-off of their resistance values can behave non-linearly.

With these options, aside from the secondary feedback which is the resulting sound, how does a user know where in the range of values the settings lie? How is the input generated? In terms of the feedback, this is totally up to the designer, and options range from using PWM to simulate various intensities in an LED, all the way to displaying values on a liquid crystal display. I experimented with 7 segment LEDs, which are capable of producing numbers 0 through 9, as well as A through G and a decimal point by illuminating combinations of the individuals segments. For input, common solutions include a simple set of up and down buttons, as well as traditional knob style interfacing like rotary encoders or jog wheels.

To complete this project there would have to be two power supplies: the 9 volt audio supply, and the 5 volt logic supply - or at least a 5 volt regulator slaving off of the 9 volt. When experimenting on the bench I have a unit that is capable of outputting two voltage supplies simultaneously. In order for the unit to be truly effective, some type of memory storage and recall would also need to be implemented. Although I researched serial methods of data transmission and storage, as well as prototyped a MIDI input/output, full

[3]Because of the limited number of FET's I had on hand to test with, I was unable have this method be fully functional on the bench.

implementation was beyond the time constraints available prior to this paper. Microcontrollers can be used for numerous other types of variable control in analog circuits. They can output waveforms via wavetable look-up in order to drive oscillators and modulation effects. They can be used to read in information from the analog world and convert them into digital control signals through comparators. As with any of the other tools mentioned throughout this paper, the function is left to ingenuity of the designer, and eventually the user. The newest generation of microcontrollers include dedicated DSPs, offering a host of new possibilities in modestly priced yet flexible digital and hybrid systems.

6 CONCLUSION:MOVING FORWARD

"...making music with computers can promote the types of isolated activity that are not necessarily in-line with social values and interactions that we normally associate with music."

-Steve Dillon in "Meaningful Engagement With Music Technology"[32]

In today's world, devices tend to be multipurpose. Cell phones have cameras, games, MP3 playback capability and a number of other design features that many people won't ever know are contained on the device. General purpose computers tend to have parallels. According to De Man, "Convergence of technologies in itself is a way to explore new frontiers and applications needed to cope with the grand challenges of the 21st century."[33, pg.41] Although convergence of technologies will provide flexibility and increase sales by giving the consumer options, a certain amount of divergence or specialization of technologies will probably lead to the most thorough innovations. According to Hoadley, "it may be well advisable to develop strategies for the deployment of faster but more expensive interfaces for specific interactions."[19, pg.12]

Examining advancements and product development in the audio industry, there seems to be an increasing interest in specialized hardware. With such systems, the user can adjust certain parameters but the basic task remains fixed. [34, pg.585] With multi-tasking being the focal point of so much of modern technology and activity, removing the ability to switch tasks may seem like a limitation, but the advantages of effectively designed dedicated hardware cannot be denied. Obviously in many cases, neither can the cost, but this is where the technological trends come into the favor of users.

In recent years the cost of dedicated DSP chips has plunged to a point where dedicated digital or hybrid audio solutions should be considered over general purpose computers - and now more than ever they are. Although many systems are still linked with the computer as a control point, or for ease of use with graphical user input, much of the processing power has been placed in outboard devices reducing issues of latency and extending computational resources. From the KYMA system to Pro Tools HD, and in a number of other commercial devices, this concept is being applied.

What has been presented in this paper are ideas of physicality, the importance of gesture and basic concepts surrounding performance and general aesthetics as relating to music in the age of technology. The exciting array of possibilities that exist, coupled with the affordability of many new technologies, allows for the possibility of developing audio hardware which may facilitate the removal of general use computers in these applications. The main prospect here is that performers and composers may be offered simple audio hardware (although possibly software driven) which is flexible, ergonomic, and easy to use.

Although dedicated hardware seems outmoded in some ways, with current technologies there are a host of possibilities for the creation of hybrid devices based around dedicated DSP's and microcontrollers - the result, an equally large host of new possibilities for composer and performer alike. The concepts outlined here by no means represent any absolute, however regardless of the stance one takes on the issues presented, it is important to consider them as decisions are made technically, artistically, and aesthetically.

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