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The Acoustics of Tesla Coils

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The Acoustics of Tesla Coils

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Abstract:

This project is an exploration into the acoustic qualities of Tesla Coils and the physics of sound generation. It includes the construction of a Solid State Tesla Coil capable of replicating the audio production properties of a conventional speaker, as well as a unique musical interface designed to transform the coil into a non conventional musical instrument.

Acknowledgements

Although this project was primarily a solo effort, there are a number of people without whom none of this would be possible. I would first like to thank my advisors Professor Germano Iannachionne, and Professor Frederick Bianchi, both of whom never failed to provide the materials or support necessary to keep the project going. I would also like to thank Mr. Roger Steele for all his advice on building the coils, and the use of many of his resources. Additionally I would like to thank the other members of the faculty who assisted me, specifically Mrs. Jackie Malone and Mr. Fred Hudson. Finally I would like to thank my friends Tom McDonald, who provided one of the schematics for the coil, Sean Levesque, and James Montgomery, all of whom provided invaluable help while building the circuit, particularly within the realm of ECE.

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Executive Summary

Tesla Coils have always been of interest to the scientific community, but it is rarely that they are embraced into more artistic areas. This project was an attempt to display the unique traits of this little known technology in a primarily musical setting. In order to understand this technology and its place in the ever developing electronic music aesthetic, we must understand how this aesthetic developed, and the role that technology played in its creation. We must also understand how and why the Tesla coil operates as it does, and the unique connection this has to acoustics. The final result of this project will be a fully functional Tesla coil capable of replicating acoustic frequencies up to and beyond 22 kHz. This coil will be controlled by a unique musical interface that implements a series of bend and turn sensors to produce MIDI messages that will be converted into audio signals. In this way, the coil will become a unique performance instrument that exemplifies the relation of acoustical and electrical physics to music.

Electronic music effectively began with the work of Edgard Varese and several of his predecessors and successors in the musical world. Unfortunately for Varese and the pioneers that came before him, the technology at the time was not developed enough to support their desires for more unique sounds. It was not until the invention of more advanced recording devices and electronic instruments, that Varese and other musicians, composers, and engineers, such as Pierre Schaeffer, could truly discover and experiment with new sounds. As the technology advanced, so did the musicians, and more and more unique styles and pieces began to develop. Synthesizers were developed, which created unique sounds that were slowly adopted by mainstream musicians. While these synthesizers began as bulky analog devices, they soon shrank in size, as more portable technology was developed, and digital then later software synthesizers began to

arise. This technology revolutionized the Music industry as musical recording and performance began to rely more and more on the advancement of technology.

Along with the rise of electronic music came the advancement of the Tesla Coil, primarily through the work of both amateur and professional Tesla Coil enthusiasts. Musical Tesla coils began to develop in these circles, but few ever became mainstream performance tools. However, a number of large brand speaker designers, used the technology to create plasma tweeters for various sound systems. Yet, there was still little interest in this technology in the music world.

As the technology developed three general forms of Tesla Coil emerged. The first being the classic Spark gap Tesla Coil that used a spark gap as part of the primary RLC circuit to create the high voltage build up in the secondary. Secondly, there is the Solid State Tesla Coil design, which uses a network of solid state transistors to create the proper current flow in the primary coil to induce a voltage in the secondary. Because the frequency and duty cycle in the primary circuit of this model can be shifted, it is a far superior design to use for audio modulation. The third type is the Dual Resonant Solid State Tesla Coil which simply includes a capacitor bank in addition to the solid state transistors.

The construction of the coil began with two different designs chosen for their various strengths and weaknesses. After a series of issues arose with the first design, the second design was adopted for its simplicity, despite the reduced output. The coil was tuned to a resonant frequency of approximately 129.7 kHz, well above the 44 kHz required to reproduce the entire range of human hearing. Although the range of human hearing only extends to about 22 kHz it is necessary for the Tesla coil to spark at at least twice that frequency to avoid any audible distortions. In the end the coil was able to reproduce these frequencies quite well, and the audio output only truly suffered from a lack of adequate volume.

The musical interface worked well in every respect except that the interface for the sensors, that were originally meant to be used, malfunctioned. This prevented any MIDI output from reaching the Max/MSP program that was designed to transform these messages into audio signals. As a result a MIDI USB keyboard was used in place of the sensors to provide MIDI data during the tests.

Overall, the Coil performed as it was expected, given the power requirements it had. However, a number of future developments were considered to improve both the output and sound quality of a future design. In the end the Coil accomplished its goal of creating polyphonic music from the corona discharge of a high voltage coil, in an accurate replica of a conventional speaker driver.

1. Introduction

For years the Tesla Coil has been a topic of interest for professional engineers and hobbyists alike. The Tesla Coil was invented in 1891 by Nikola Tesla during his experimentation with high frequency phenomena. Since then, the design behind Tesla Coils has been adapted for many uses including high frequency lighting, the production of solid nitrogen compounds, high power radio transmission, and even the production of music.

This project is designed to be an exploration into the acoustic qualities of this unique technology by constructing a coil capable of producing clear polyphonic sound. The result of this project will be a fully functional Solid State Tesla Coil (SSTC) capable of modulating the sound of its corona discharge into music. The musical interface for the coil will be created using the Max/MSP program, and a series of bend sensors that will have varying effects on many attributes of the sound including the pitch, duration, and volume. The goal of this project is to create a musical instrument with a unique interesting interface that displays some of the physical aspects of how sound is created.

Over the past century, huge advancements have been made in the technology used in music synthesis and production. With this new technology has come a new musical aesthetic known as electronic music. Electronic musical began with the works of Edgard Varese, Pierre Schaeffer, Ferrucio Busoni, and the other pioneers who saw the potential in the newly developing technology. The experimentation with electronic and electromechanical instruments began in the early 1900's with the Telharmonium, Theremin and the Hammond Organ. Over the years these instruments have been refined and expanded to produce many of the instruments and synthesizers that exist today.

While these instruments and their descendants are unique in the sounds they create and the methods for creating them, many of them still rely on loudspeakers, or what primitive versions of loudspeakers existed at the time, to create audio. Such as the telephone receivers amplified by acoustical horns often used with the Telharmonium. Tesla Coils, however, provide their own physical means of producing sound through the electrical arcs they generate. In effect, Tesla coils are creating music not only through electronic means, but are literally generating sound by passing electricity through the air.

Tesla coils can be used in this respect as performance tools, one popular example of this being, the group ArcAttack who use a pair of large Tesla Coils combined with other automated instruments to perform live concerts. However in this case, and in most others, the Tesla Coils are almost exclusively large, and therefore heavily distorted. Tesla coils can also be built to produce a much more high quality sound by increasing their resonant frequency. In these cases, the coil can be used to produce extremely clear sounds without any distortion, particularly in the higher frequency ranges. Thus, Tesla Coils can be built to act as very high quality tweeters that rival even the best sound systems.

For all this Tesla Coils are rarely associated with electronic music and there is little experimentation being done with their unique creation of sound. This is primarily because of the hazards and difficulty inherent in creating these devices. Tesla coils could become a huge part of electronic music, particularly in live performances which are becoming more and more common for electronic musicians. However, first they must be demystified in the eyes of musicians who shy away from them because of their highly technical and potentially dangerous nature. This project will remove some of the mystery surrounding this device and help to bridge the gap between musicians and engineers as music technology continues to grow.

The next few chapters will provide a more detailed background into the history and function of Tesla Coils and electronic music. Section 2.1 will look at when and how electronic music first developed and how it has influenced our musical world today, including the development of computer programs designed for music production. Next section 2.2 will describe the history of Tesla coils, and how they developed into their modern state. Section 2.3 will describe the state of the art in Tesla coil technology and use, particularly in musical performance. Section 2.4 provides a description of how modern loudspeakers operate and describes the advantages and disadvantages of using a plasma arc over a conventional speaker driver. Section 2.5 will describe how a Tesla coil operates, and why SSTC's are typically used for audio modulation. Section 2.5 will also look at how Pulse Width Modulation works and why it is useful, and often required, for audio modulation in Tesla Coils.

Chapter 3 will begin with a detailed analysis of the design and construction for the primary driver circuit and the primary coil. Section 3.2 will then cover the design and construction of the secondary coil, along with the calculations for the secondary coil's resonant frequency. Section 3.3 will describe the program generated with Max to produce audio input for the coil and how the bend sensors were used to manipulate the audio. Finally Chapter 4 will be an assessment of the performance of the Coil and musical interface, and recommendations for future improvements.

2. Literature Review

Since its development in 1891, the Tesla Coil and its variants have been implemented in numerous areas for many different purposes. The design has been used in professional applications of many modern electronic systems such as high frequency lighting, radio transmissions, and wireless energy transfer. It is also common in more amateur applications by Tesla Coil hobbyists, sometimes called “Coilers” and other Electrical Engineers. However, for the purposes of this project the most relevant application of the Tesla Coil is its use as an alternative method of sound generation.

Section 2.1 will begin with a history of the electronic music aesthetic, and how these developments influenced the technology that is present today. This section will cover the major electronic instruments and synthesizers from the first Theremin and electronic organs to the modern equivalents, as well as the recent developments in musically oriented software. Section 2.2 will provide a brief history of the Tesla Coil, how and why it was developed, and its early uses. It will then cover the various uses of the Tesla Coil and the impact its design has had on modern electronics. That will be followed by section 2.3 which gives a close look at how the Tesla Coil has been applied in musical settings, both as an alternative sound system and as a performance tool. Section 2.4 gives a brief discussion of modern loudspeaker design and how it relates to Tesla coils. This section then describes how a coil produces sound, particularly tonal sound, and then discusses the side effects produced during operation. Finally, Section 2.5 will give an in depth explanation of the theory behind the sound modulation of Tesla Coils, including the involvement of Pulse Width Modulation (PWM), and a close look at several existing Coil designs.

2.1 A History of Electronic and Synthesized Music

Electronic Music is essentially any music performed through an electrical means, whether it is through electrically powered musical instruments or through purely electronic technology. In many cases the term Electronic Music is typically reserved for the purely technological sources such as the synthesizer and Theremin. Likewise, electronically powered instruments such as the Hammond Organ and Electric Guitar are typically referred to as electromechanical instruments. Electronic music has developed significantly in a very short amount of time compared to other musical styles, as it has only existed, and truly developed into a musical style, in the last century.

2.1.1 The First Electronic Instruments

In 1897, Thaddeus Cahill invented the Telharmonium which became the first significant electronic musical instrument. The instrument was capable of reproducing respectable music of the time, such as Bach and Chopin, but was bulky and difficult to operate. Nevertheless, there quickly arose a desire for existing composers to implement the new technology into their work. This served as a pathway to ease the integration of the new technology into the musical world. As electronic instruments became more and more popular, they became more and more developed and refined. The Telharmonium was soon discarded, and new instruments arose such as the Theremin and Croix Sonore.

The Telharmonium was, in many ways, the predecessor of all electronic instruments. The music was transmitted to listeners via telephone lines, and sometimes even performed in large concert halls. The later versions of the instrument were so large that they filled entire rooms, and were often housed beneath the concert hall in which they were performed. (Williston, 2000). This was also one of the first instances of sound produced through electromagnetic impulse on a paper cone; a design that would eventually develop into modern loudspeakers.

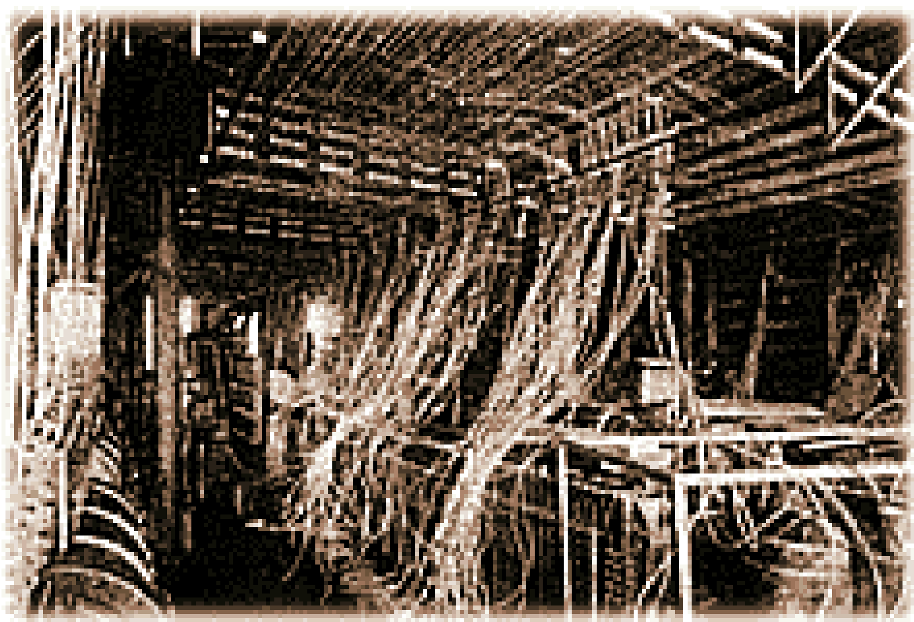

Figure 1: The inside of the Telharmonium. The figure standing in the left of the image shows its tremendous size ((History of Electronic Music: The demise of the Telharmonium)

The Telharmonium quickly lost popularity however due to its large power consumption and the possibility of crosstalk over telephone channels. Despite this, the instrument was revolutionary in that it functioned much like an organ with multiple stops that allowed it to produce a polyphonic sound of varying timbre. (Williston, 2000)

Then in the 1920's a number of instruments were developed that redefined the way electronic music was produced. The first was the Theremin, invented by Leon Theremin in 1921. The Theremin was revolutionary in that its dual antenna design removed the need for the performer to actually touch the instrument. Instead the performer could change the pitch and volume by varying the position of their hands relative to the two antennae.

Figure 2: Leon Theremin posing with the Theremin, the left loop controls volume while the right loop controls pitch.

The Theremin gained popularity for its uniquely eerie constant tone sound, and was implemented in many Science Fictions movies at the time. Beginning in the 1940's it was also integrated into popular music. The Theremin's hands-off interface made it a difficult instrument to master. However, the unique sound and playing style had a prime impact on the style and development of electronic music. (Termen, 2007)

Other instruments that produced a Theremin like sound that arose within the same time period were the "Croix Sonore", or Sonorous Cross, and the Ondes-Martenot. The Croix Sonore was developed in 1929 by Michel Billaudot and relied on the capacitance between the antennae and the performer's body, much like the Theremin (Cross Sound, 2010).The Ondes-Martenot produced a very similar sound but did not incorporate the hands off style of playing. It was also later expanded to include more timbrel sounds. (Bloch, 2004) While production of the instrument stopped for a time, a new project called the Ondea arose in 1997 that was based on the Ondes-Martenot. Then in 2008 another instrument officially called the Martenot was developed by Jean-Loup Dierstein.

These instruments were the forerunners of electronic instruments, and while they were very popular among composers of classical, pop, and film music during their time, they also had a large role in the general development of the electronic music aesthetic.

2.1.2 The Development of Electronic Music as a Style

The rise of electronic music cannot be solely attributed to the actions of individuals, for it takes many to accept an idea and develop it into a global style. However for any great change to occur there have to be instigators, and one such instigator of electronic music was Feruccio Busoni. In 1907 Busoni published *Sketch of a New Esthetic of Music*, which detailed his thoughts on the newly developing electronic sources of music and their future in the music world. In his work he states his opinion of structured musical styles, saying "We apply laws made for maturity to a child that knows nothing of responsibility…. They [mankind] disavow the mission of this child; they hang weights upon it. This buoyant creature must walk decently, like anybody else." (Busoni, 1962) Busoni goes on to talk about the idea of absolute music and how it cannot be achieved through rigorous application of forms and structures. He is also quoted as proclaiming the necessity of electronic instruments in the development of music. His most famous statement and one which stuck with his student Edgard Varese throughout his life was simply, “Music is born free; and to win freedom is its destiny.” (Busoni, 1962) (Snyder, Ferruccio Busoni)

Another notable figure who preceded, and heavily influenced, Varese was Luigi Russolo. Russolo was a futurist composer whose manifesto, *The Art of Noises* (Russolo, 1913), had a profound impact on the development of musical aesthetics. In his manifesto, Russolo describes the evolution of sound, and how it must break away from the limitations placed upon it by early civilizations. In his manifesto, Russolo states that when the exceptional noises of hurricanes, earthquakes, and the like, are removed, nature is predominantly silent. The discovery of sound was seen by ancient peoples as a great spiritual development, attributed to godly powers, and remained a mystery to most. Thus sound was made distinct from the noise of life. Early civilizations took this and broke it into discreet intervals that were to be used. Thus, music became structured, and therefore limited. Russolo then describes the development of harmony and the chord or “complete sound”. What started as pure sounds grew more complex, starting with the triads and becoming more dissonant. This music became more and more polyphonic to compete with the “multiplication of machinery” that dulled the emotional impact of pure sounds.

This is where Russolo brings up the advancement of technology. Russolo claims that orchestras can be broken down into four types of instruments, bowed strings, brass wind instruments, wood wind instruments, and percussion. However, through technology musicians can extend beyond these instruments and manipulate sounds and noise. In Russolo’s eyes traditional music had becomes so mundane that nothing new could come from it, and that audiences were always left “waiting all the while for the extraordinary sensation that never comes.” (Russolo, 1913)

Russolo then goes on to explain the aspects of noise, its harmonic and rhythmic nature, and the six families it can be sorted into. The six families that Russolo describes are:

1. Roars, Thunder, Explosions, Rumbles, Booms, Crashes, Splashes
2. Whistles, Hisses, Snorts
3. Whispers, Murmurs, Mumbles, Grumbles, Gurgles
4. Screeches, Creaks, Rustles, Buzzes, Crackles, Scrapes
5. Percussion noises from hitting wood, metal, skin, stone, etc.
6. Voices of animals and men (usually not speaking or singing)

Russolo concludes by stating that it is up to futurist musicians to bring these noises into the world of music. They must closely observe the world to determine the specific aspects of noises that allow them to be used compositionally and harmonically, without compromising their complex nature. And perhaps most importantly, they must find ways to distinguish and recreate these sounds so that all sounds can be composed into a master orchestra of noises.

Perhaps the most notable pioneer of electronic music and one who is sometimes referred to as the father of electronic music was Edgard Varese. Varese was an engineer who was later trained as a classical composer, and was influenced heavily by Debussy. In 1918 Varese broke from European styles and moved onto more abstract pieces. His work *Hyperprism*, featured a number of percussion instruments and a "Lion's Roar" (an improvised instrument made with a rope pulled through a tube), which caused a riot during its first performance, and ended with half the crowd leaving in an uproar. Yet Varese's piece was later performed by the renowned composer Leopold Stokowski and numerous lesser conductors. Varese's other popular work, *Ionisation*, was the first use of a siren, accompanied by 37 percussion instruments, as a musical device. It was not long before Varese became frustrated with the traditions of orchestrated music and published his manifesto *The Liberation of Sound* (Snyder, Edgard Varese: The Father of Elecronic Music) In this manifesto Varese lamented the absence of the technology capable of creating the types of sounds he desired. In his manifesto Varese sought an instrument that could produce any range and denomination of pitch and timing. Varese states that many accused him of attempting to destroy traditional music, yet he claims, “Our new liberating medium - the electronic - is not meant to replace the old musical instruments which composers, including myself, will continue to use. Electronics is an additive, not a destructive factor in the art and science of music. It is because new instruments have been constantly added to the old ones that Western music has such a rich and varied patrimony.” (Varese, 1936) Despite this, Varese’s ideas earned him the distrust of the majority of the musical world. After WWII the technology he so desperately wanted finally arrived and he began experimenting with new sounds. His “*Poem Electronique*” which debuted at the Brussels World Fair in 1958 marked one of his largest successful performances and opened the eyes of the world to this new style of music.

Another great leap in the technology and style of electronic music began in 1946 when Pierre Schaeffer began his “research into noises”, at the Club d’Essai de la Radiodiffusion-Television Francaise. Schaeffer was not a trained musician, but had been working as a radio engineer when he began a revolutionary technique in sound manipulation that would soon come to be known as “musique concréte”, or concréte (real) music. Schaeffer began by recording fragments of sounds with phonographs, whether they be musical or not, and combining them into collages. In 1948, Schaeffer broadcast his first public pieces of musique concréte, labeled “noise etudes”. They were entitled “Etude aux Chemins de Fer” or “Study of Railroads”, “Etude Aux tourniquets”, “Etude aux casseroles”, “Etudes pour piano” (actually two pieces) and “Etude pour orchestra”. These Etudes helped introduce the use of sampled sounds as compositional material, particularly “Study of Railroads” which implemented the sounds of locomotives. In “Study of Railroads”, Shaeffer isolated rhythmic leitmotivs and, through mixing, created both musical and dramatic sequences. These dramatic sequences were considered to be unmusical until Schaeffer used spectral transposition to alter specific envelopes of sound and create musical sequences. However, like many of the other unconventional pieces at the time, this was met with a great degree of opposition.

In 1950 Schaeffer collaborated with Pierre Henry to produce “Symphonie pour un home seul” which was a 12 movement piece featuring the sounds of the human body. A year later Schaeffer began experimenting with magnetic tape recorders, which revolutionized the way musique concréte was produced. The magnetic tape recorder removed the need for multiple phonographs to provide various samples, and allowed sounds to be cut, spliced, and transformed with ease; or at least with ease relative to the time. (Ankeny)

Musique concréte was more than just a fancy method of organizing sampled sounds. It was a completely new approach to music. In *Machine Songs V*, Carlos Palombini describes musique concréte as an inversion of the traditional approach to composing. Instead of mentally conceiving a piece, copying it down in notation, and then having it performed, musique concréte cannot be conceived prior to its performance, it is conceived by experimentation and compiled into its final form by the composer. It was due to its experimental nature that musique concréte tended to sound less like a new form of music and more like just an experiment. In Schaeffer’s mind it “lacked a theoretical grounding”, and required some sort of method and criteria to classify the infinite sounds available to sample. The answer to this came in part in 1951 with Schaeffer and Henry’s piece Orphee. From this, Schaeffer created the notion of a pseudo instrument, or sounds and families of sounds that could fill the roles of orchestral instruments. Despite this, musique concréte was eventually assimilated into the German elektonische Muisk, (electronic music) genre. For years Schaeffer would struggle with the method of musique concrete, until finally in 1958 he established the Groupe de Recherches Musicales which moved on from the topic of musique concréte into a more general form of musical experimentation. Nevertheless, Schaeffer's work was paramount in the development of electronic and non-traditional music, and had a profound influence on composers after his time. (Palombini, 1993)

2.1.3 The Development of Modern Electronic Instruments and Musical Software.

While the advancement of the electronic music aesthetic is credited mainly with the composers who developed the ideas and styles for it, it is important to note that many of these composers were limited at first by the technology available to them. Without the advancement of technology in musical instrumentation and production the ideals of these composers could never have been realized. In this respect, few developments are as notable as the invention of the music synthesizer.

The first synthesizer is credited as the invention of Elisha Gray, who created it during his attempts to develop a working telephone. A battle he eventually lost to Alexander Graham Bell. Around 1876 he made his first breakthrough with an electric oscillator, a bathtub, and his own hand. He found this combination could produce a vibration in the bathtub by using his hand as an amplifier for the electric signal. He then performed a similar experiment using a metal plate and the body of a violin. Eventually Gray's experiments led to the development of the first multi tonal synthesizer that used a series of eight keys laid out in the manner of a keyboard. While Gray's invention did not become mainstream in its own right, his discovery that music could be transmitted along steel reeds through telephone wire became an important stepping stone that other inventors could build from. (Pioneers of Electro-Acoustics)

For years afterward a number of variations of the synthesizer popped up, but it was not until 1964 that a truly commercial synthesizer was produced. At that time, Bob Moog, a former engineer for the RCA Mark II synthesizer which like many synthesizers at the time filled nearly an entire room, developed the first synthesizer that was commercially viable in the music industry. In 1967 the Monkees featured the first Moog synthesizer in their album, Pisces, Aquarius, Capricorn, and Jones. This marked the first instance of a synthesizer being used in a popular album. In 1970 Bob Moog developed the Minimoog, the first portable synthesizer, and his company began to gain popularity until more and more musicians began to use synthesizers in their work.

In1978 the first digital synthesizer, the Prophet-5, was developed by Sequential Circuits. Unlike analog synthesizers, these new digital synthesizers were capable of producing polyphonic sound and were able to store sounds on their microprocessors. One of the more popular and affordable models of digital synthesizer was the Yamaha DX-7 which was used by a number of artists including the Beastie Boys, Nine Inch Nails, Depeche Mode, Madonna, and The Cure. While these types of synthesizers became very popular with many musicians, a good number of artists continued to use analog models which allowed then to make real time changes to the sound. (Baker)

Regardless of whether or not they were preferred by musicians, digital synthesizers had a profound impact on the entire music world especially with the introduction of MIDI. MIDI stands for Musical Instrument Digital Interface and was revolutionary in that it does not transmit an actual audio signal. Instead MIDI data contains a number of different messages that correspond to various musical properties, such as pitch, velocity (or volume), duration, and any spectral effects such as pitch bend and modulation. In truth, MIDI messages are simply values ranging from 0 to 127, until they are interpreted in some fashion by a synthesizer or other interface. In the early days of MIDI devices, there were many issues with transmitting this data from one machine to another due to the lack of an industry standard. This meant that when a sound was composed on a machine by one manufacturer then transferred onto another there was a good chance that the sound would be completely different. However, in 1991 the General MIDI standard solved many of these issues by providing cross manufacturer industry standards that allowed musicians to compose on one device and then transfer to another with a reasonable expectation of how it will sound. (Tutorial: History of MIDI)

The creation of MIDI preceded the development of computer software synthesizers, which first appeared in the 1990's, by nearly a decade. Yet in many ways software synthesizers were a step above analog and digital synthesizers, even with MIDI. This was because they did not rely on any hardware and could effectively reproduce the sound of any synthesizer. With various Plug-ins, VST's, and emulators a software synthesizer could provide more sounds and effects than any single synthesizer in the market, and could be easily controlled using a computer keyboard. (Tutorial: History of MIDI)

While MIDI and software synthesizers were major breakthroughs in the world of music technology, they are by no means state of the art. Technology has grown rapidly throughout the past few decades and led to the development of audio programming languages. Audio programming languages are computer programming languages specifically designed for sound production or synthesis. Csound, one of the first audio programming languages, was created in 1985 and is primarily a text based language. Csound has been under constant development over the years and is currently a powerful tool for audio production and synthesis as well as live sound performance. (Clemens)

Around the same time Csound came out, Miller Puckette, who had collaborated on the project, came out with a program called Max which, unlike Csound, uses a graphical user interface. This means that instead of using lines of text as code, the user maps a program out using graphical objects which are connected by patch cords. An example can be seen in Figure 3.

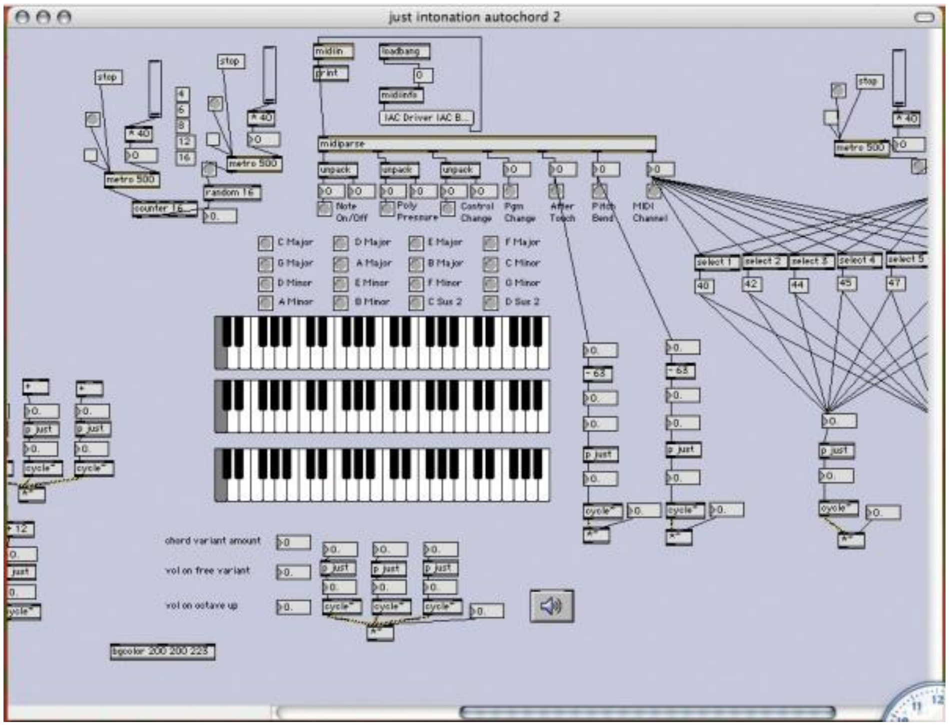

Figure 3: a sample patch in Max, (Matmos)

The unique advantage of graphical interface languages like Max is that they are very good at portraying the program structure in a way that the user can easily understand. In fact, Max has been described as the lingua franca for interactive music performance software (Lossius, 2006). Max operates through a series of objects that act as discrete programs linked together by patch cords that pass messages from the outlets of one object to the inlets of another. The types of messages that Max supports are based on six basic data types which are: int, float, list, symbol, bang, and signal. Signal data is used exclusively with the Max Signal Processing

(MSP) extension. Another unique aspect of Max's graphic object oriented design is that users can design their own objects for a specific purpose and transfer then to other programs. Essentially the user can create layers of programs within an overarching patch. Max patches can also be bundled into stand alone programs that are often incorporated into other audio production software. (What is MAX, 2011)

Supercollider, another major audio programming language, is a high level programming language engineered mainly for real time audio synthesis and algorithmic composition. Supercollider also plays a major role in acoustic research, and is popular for interactive programming. In addition to its C based text style programming language, Supercollider supports cross a platform graphical interface which is shown in Figure 4. Most importantly however, Supercollider's dynamic programming allows the code to be quickly modified and executed on the fly. This allows for the programmer to edit code mid performance to generate different effects.

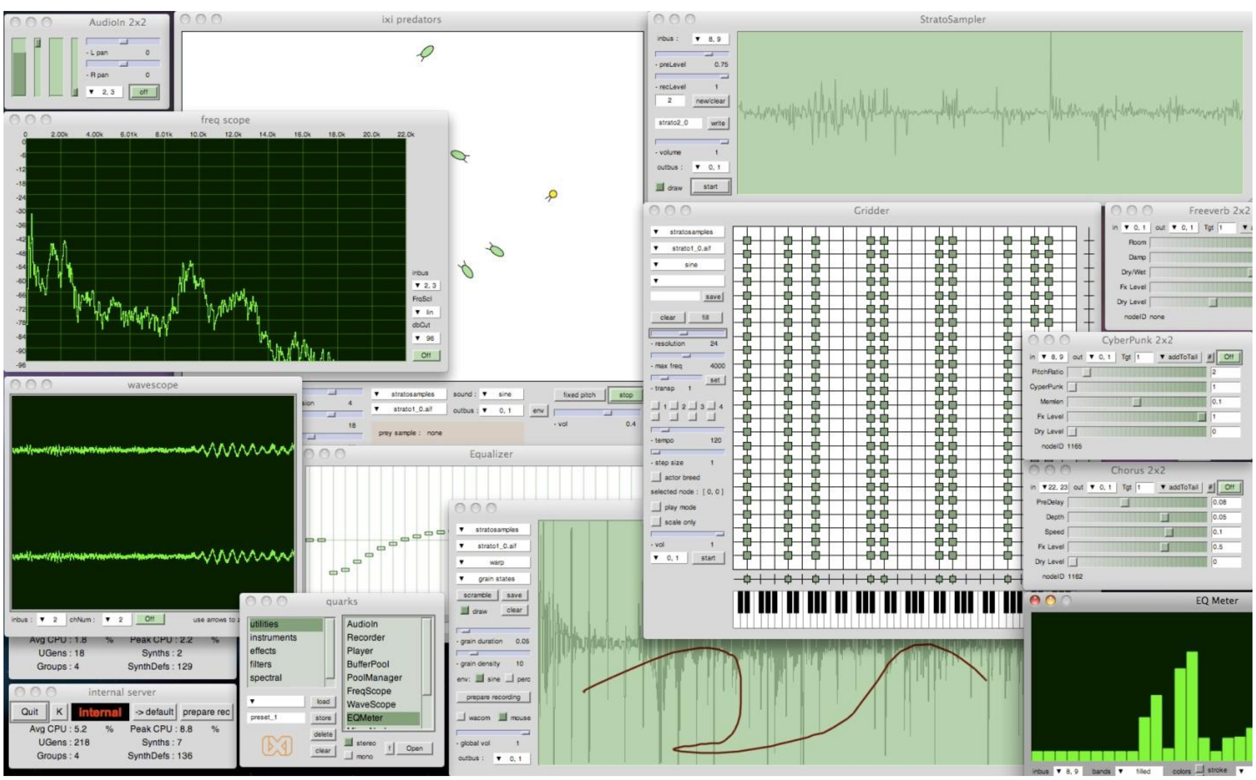

Figure 4: A sample display of supercollider's graphic interface (Supercollider)

This idea of real time live programming is perhaps one of the most significant developments in electronic music. Live programming brings the programmer into the spotlight of the performance. Where once all computer and electronic music was pre generated in advance, now the musician can influence the music as it was being displayed. (Supercollider)

All the while other musicians were coming up with more and more unique ways to perform concerts using innovative instruments and interfaces, such as the laser harp invented by Bernard Szajner and made famous by Jean Michel Jarre. More recently a new group of musicians named ArcAttack has arisen with a unique performance that incorporates an enormous Tesla Coil to produce synthesized musical tones.

So it was that electronic music began in the minds of musicians and composers well before the technology for such things existed and developed quickly through their ideas and innovations. Through the exploration of non-traditional sounds and sound generation a number of unique musical instruments such as the synthesizer and styles such as musique concréte were born. Then as technology developed even further synthesizers became digital, and then available in software form and more audio production software arose to the point where the once passive computer musicians now had an active role in performing their music. Now computer and electronic music is as common as traditional musical instrumentation and performance.

2.1.4 The Tesla Coil as an Electronic Instrument

In this world of rapidly developing electronic techniques and hardware, the Tesla Coil has been largely ignored by electronic music enthusiasts, despite the possibilities it presents. Electronic music is founded on the desire to encompass the full spectrum of sound available in the world, and the Tesla coil presents a great opportunity to experiment with a large variety of sounds resulting from the unique design of each coil.

Tesla coils in their simplest form can be seen as percussion instruments. The spark generated by the arc of electricity causes a loud pop or bang that can be repeated with any desired frequency. If designed correctly, this frequency can be increased beyond the range of discrete beats, to the point where the rapid sparking of the coil actually produces a tone. A properly designed and tuned coil can hit any range of notes, well beyond that of human hearing. This allows for infinite possibilities in notes and scales. However, this itself is nothing new and was accomplished years ago with the Theremin and other constant tone instruments.

What makes the Tesla Coil unique is that it is one of the very few electronic instruments that produce the sound themselves, without the use of a loudspeaker. The Tesla coil is itself the loudspeaker, generating the variation in air pressure through the sudden heating and cooling, and thus the rapid expansion and contraction, of the air around the spark. More so, a Tesla Coil can be designed to provide varying levels of distortion, creating an even broader spectrum of possible sounds. The most common forms of Tesla Coils built for performances are large and have a sound related to early synthesizers and are often very heavily distorted. The most well known group to implement musical Tesla Coils is Ark Attack, a group of engineers and musicians who were the first to develop Tesla coils specifically designed for musical production. In these cases the pitch of the Tesla Coil is controlled through an outside MIDI controller and sound synthesizer that feeds the frequency to the coil.

However, the Tesla Coil is also unique in that it can be modulated with any type of electrical signal, including any type of audio signal. This means it can be substituted as a loudspeaker to play sounds from any source. If designed to have a low resonant frequency the Tesla Coil will sound more distorted. Alternatively Coils can be tuned at very high resonant frequencies, which can provide nearly flawless sound reproduction for certain frequencies. This has led to Tesla coils and variations of the design, to be used as high quality tweeters for sound systems.

Thus, in addition to providing a unique and interesting means of musical performance with a distinct sound and visual effect, the Tesla coil could revolutionize the loudspeaker industry. The sudden rush of interest into electronic music has come hand in hand with an expansion of music technology, primarily through the continued development of technology like this. The Tesla coil literally creates sound through the manipulation of electricity in the air, and it may be that in the future this technique could be expanded further into the electronic music aesthetic.

2.2 The Development of the Tesla coil and Singing Arc

The Tesla Coil itself has been around for over a century yet it has never truly been a commercial device. The reason for this is the dangers inherent in working with high voltages, which explains why the majority of Tesla Coils among the general public are built by professionals or Tesla Coil enthusiasts. However the possibilities of a Tesla Coil as a unique and innovative performance device are astounding, and in time there is a good chance that this type of device will come further into the spotlight.

The Tesla Coil was developed in 1891 by Nikola Tesla as a tool to conduct various high voltage, high current experiments. One of the major uses for Tesla's early coils was high frequency lighting. Tesla conducted numerous experiments with fluorescent and incandescent lamps as well as high frequency arc lighting. This research led him to develop the first high efficiency high frequency lighting ballasts. Without this discovery modern Metal Halide Lamps would not be possible. (Twenty First Century Books, 2011)

One of Tesla's earliest discoveries using electrical resonance was that it is possible to eliminate one of the conductors used to carry current form a power supply to an electrical load. Through 'electrostatic induction' or 'capacitive coupling' a circuit can be completed using a metal plate connected to one of the high voltage leads of the power supply and another plate to the load. This led to his development of the carbon button lamp and a single wire electrical motor. Later he also found it was possible to complete the circuit through the ground by increasing the distance between the plates. Further work with a single wire incandescent lamp led to Tesla's discovery that vacuum tubes can produce x-rays through the process of Bremsstrahlung. By studying these X-rays, Tesla was one of the first to find the hazards of X-ray exposure.

An operating Tesla Coil tends to produce a fair amount of Ozone. Tesla used this to develop a device designed specifically to cause a reaction between oxygen and nitrogen, leading to the eventual production of solid nitrogen compounds form atmospheric nitrogen.

Some of the most important experiments Tesla conducted with his coils where those with wireless telegraphy and telephony. By replacing high frequency alternators with his resonant transformer, he was able to produce radio waves significantly more powerful than previously possible. Tesla continued to experiment with wireless communications and developed the basis for many aspects of our modern telecommunications systems, as well as other modern electrical systems. Even one of the first particle accelerator designs featured a Tesla Coil as its source for high voltage. (Twenty First Century Books, 2011)

Tesla did not perform many experiments concerning audio applications for the Tesla Coil. The first exhibited instance of audio modulation of plasma was in 1900 by British physicist and electrical engineer, William Duddell. Duddell found that by varying the voltage supplied to a Carbon Arc Lamp (a lamp typically used to provide street lighting before the invention of the electric light bulb), he could change the pitch of the humming produced by the arc. Duddell attached a keyboard to the lamp and was able to produce audible tones, thus creating the first electronic instrument. Unfortunately this instrument became little more than a novelty and Duddell did not patent it. (William Du Bois Duddell)

2.3 Modern Applications

Today, Tesla's original coil designs, and close adaptations of them, are most often built by hobbyists and electrical engineers for private projects. However the original designs were adapted and refined and can be considered the predecessor to modern flyback transformers, and the ignition system in internal combustion engines. Although these devices do not use resonance, they store energy through an inductive "kick" much like the Tesla Coil stores energy. Tesla Coil designs are often used in high voltage labs for experimentation, and low power coils are occasionally used as high voltage sources for Kirlian photography.

While there are many instances of Tesla Coils or Tesla Coil descendants in many areas of modern technology, the main topic of this section concerns the design and use of Tesla Coils as a musical device on a professional and amateur level.

2.3.1 Professional use of Musical Tesla Coils and the 'Plasma Speaker'

The first instance of a musical Tesla Coil was a performance by the group Arc Attack. Arc Attack formed in 2005 and built the first musical Tesla Coils with the help of Steve Ward, an electrical engineer from Illinois. Since then, they have become a popular performance group and one of the most well known examples of musicians using Tesla Coils. The group implements two large custom engineered and built Tesla Coils that produce sounds similar to those of early synthesizers. This is augmented by a robotic drum set and live instrumental performances by the crew. (Arc Attack)

Another famous example of a musical Tesla Coil is the performance by Steve Ward at Duckon 16, an annual science fiction convention held in the Chicago area. In this performance Steve Ward also used a pair of large Tesla coils that again produced sounds similar to old synthesizers. (Ward, 2007) While there are numerous examples of performances using large, heavily distorted coils, the majority of experimentation and innovation in the area comes from personal projects by electrical engineers and professionally crafted speaker designs by corporations.

Plasma Speaker is a term used to refer to a loudspeaker that produces sound via the expansion and contraction of air caused by the manipulation of a plasma arc or flame. The varying temperature of the arc causes rapid expansions and contractions in the air, producing sound waves. Many claim that plasma tweeters provide a far better quality sound than conventional speaker because of the lack of weight in the driver. In tweeter design, one of the main qualities to incorporate is a lightweight dome, and in plasma speakers the dome is essentially mass less, leading to less distortion and higher transient response. While this is great news for tweeter design, plasma arcs are not efficient at moving large quantities of air, making them less suitable for lower frequencies.

2.4 Conventional Speaker and Sound System Design

In order to fully understand how unique Plasma Speakers are in the way in which they produce sound, it is necessary to look at how conventional speaker drivers operate. In a conventional loudspeaker, sound is generated by the oscillations of a paper, plastic, or metal cone in the surrounding air. These oscillations produce pressure waves in the air that we interpret as sound. The cone is driven by a coil of wire, called the voice coil, attached to an extension of the cone, called the "former". The voice coil is suspended inside a permanent magnet so that it lies centered between the magnet pole pieces and the front plate of the driver. The ends of the voice coil are connected to the crossover network which is mounted on the speaker binding posts on the rear of the enclosure. The voice coil is kept centered in the gap by a "spider" attached to the frame of the driver, and a dust cap mounted at the center of the cone prevents air from entering from the front of the speaker. In low and mid range speakers there is a rubber surround connecting the outer edge of the cone to the frame which allows for more flexible motion of the cone. Figure 5 displays the typical conventional speaker driver design.

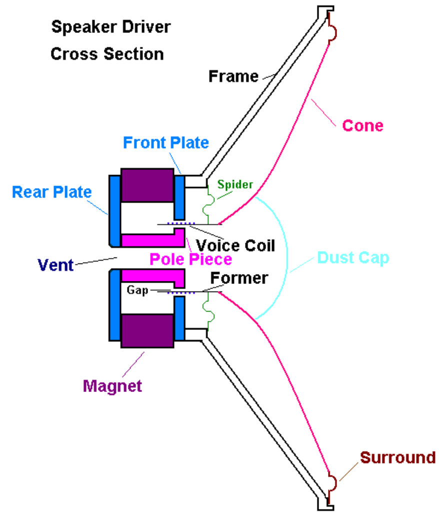

Figure 5: A conventional speaker driver (Everything You Wanted to Know About Speakers, 1998)

In order to produce oscillations in the voice coil, and thus in the cone, the voice coil is fed an electrical audio signal. When this current is introduced to the voice coil it becomes an electromagnet with variable poles. As the poles switch, the coil is either attracted or repelled by the permanent magnet, repelled if the poles match, and attracted if the poles are opposite. These poles switch in time with the oscillations of the electrical signal, which causes the same oscillations in the voice coil and the cone. Thus, the electrical signal is converted to into mechanical oscillations, which is converted into air pressure waves, which are interpreted by our ears as sound.

An important component of speaker design is the crossover network. The crossover network is designed to split the audio signal into groups of frequencies which are sent to the appropriate drivers. Capacitors are used to limit low frequencies and pass high frequencies to the tweeters (high pass), while inductors are used to limit high frequencies and pass the low frequencies to the subwoofers (low pass). (Everything You Wanted to Know About Speakers, 1998)

In addition to the standard cone style speakers, there is a type of driver that employs a thin foil or flat membrane. These are called ribbon speakers and use either a thin metallic foil ribbon or non metallic membrane connected to foil. These membranes are suspended between permanent magnets, much like the voice coil in cone speakers, and oscillate when a current is applied. In electrostatic speakers the membrane is coated in powdered graphite which is connected to a positive charge of several thousand volts. The membrane is flanked by perforated sheets of metal through which the audio signal is sent, causing them to attract or repel the membrane as the signal fluctuates. These types of speakers are exceptional at producing mid and high range frequencies but tend to suffer in the low ranges. Figure 4 shows the basic design of electrostatic speakers.

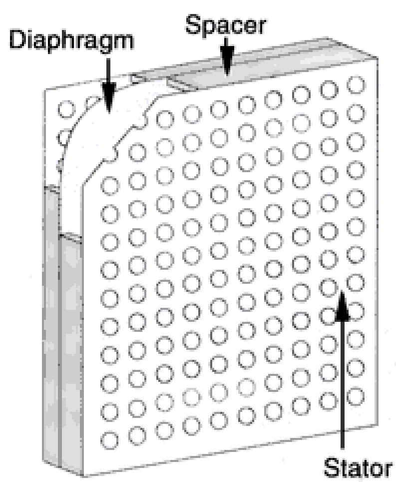

Figure 6: Electrostatic ribbon speaker design   (Audio File)

Another important aspect to consider when comparing speakers is the impedance of the speaker. The impedance of the speaker is the amount of resistance the signal from the amplifier encounters while passing through the speaker. In most designs the impedance is nominally

8ohms but in some cases it can be 6 or 4 ohms. Nominally means that the average resistance is 8ohms, however since the resistance can vary with frequency, the range of impedance could be from 3 to 20 ohms. Impedance has a significant effect on the current drain from amplifiers since, according to Ohm’s law: V=IR, (Voltage =Current x Resistance) and the Power Function: P = VI (Power = Voltage x Current), a speaker with an impedance of 4ohms will draw twice as much current at a given voltage as a speaker with an impedance of 8 ohms. This means that at any given power, the voltage drop for a 4 ohm speaker will be divided by a factor of 1.414 while the current will be multiplied by a factor of 1.414 when compared to an 8 ohm speaker. This means that the amplifier for a 4 ohm speaker will have to provide a significantly increased current at higher volumes.

Theses are just some of the factors to consider in speaker design, all of which are very important. However, the main topic that will be in question here is the effect of the cone material on sound. There is one major choice that determines what sort of material is used in driver cone, and that choice is between uniform motion, or rigidity and self damping. Other issues that often arise are cavity resonance and magnetic non-linearities, all of which are interesting to consider when using a plasma speaker.

The rigidity of a driver corresponds to the accuracy of the translation of the signal from the voice coil to the cone. Basically, a higher rigidity means a flatter response, fast pulse rise time, low IM distortion, and a more transparent sound. These are all important and good attributes to have, however the more rigid the cone is, the stronger its resonances become. This can cause certain frequencies to sound stronger and longer. This is partly the result of the poor coupling between a rigid body and the surrounding air. This poor connection means that the air does not act as a very strong damping force on the cone and the cone will ring for a long time. This is a problem for loudspeakers which are required to produce a lot of different frequencies rapidly. The solution to this is to introduce more damping either through amplifier damping or the intrinsic qualities of the cone.

Ideally the amplifier would, by acting through the voice coil, stop the cone completely rather than leaving it to ring. In reality the coil only dampens a portion of the cone and other methods of damping are required. Often rubber surrounds are placed partway down the cone to add some damping. In this case a lot of attention has to be paid to the damping effect of the spider and surround materials. Even with the best Kevlar, carbon-fiber, and aluminum cones, there is always one high Q peak somewhere within the 3 to 5 kHz range. This is unfortunately right around where the ear is the most sensitive which makes it difficult to deal with, even with a sharp crossover or notch filter. An alternative is to use a highly lossy (soft) material, usually polypropylene in most modern speakers. In this case the cone damps itself, however lossy materials tend to have strange hysteresis modes which lead to IM distortion.

Another issue that typically arises is cavity resonance. Cavity resonance is the result of high-Q peaks forming in the small spaces between the dust cap and pole piece of the magnet. In most cases these peaks are high frequency and directional. This often misattributes them to a problem with the tweeters rather than the mid range speakers, which is where they typically occur. The other major issue of magnetic non-linearity results from a varying of the inductance of the iron core pole piece of the magnet. As the coil moves, the inductance of the iron core varies which causes the roll off frequency to constantly shift. This can also result in significant IM and FM sidebands throughout the entire frequency spectrum when very deep bass is played, which results in a blurriness to the sound.

It can be seen that there are a number of problems inherent in modern speakers that inevitably lead to compromises depending on what sort of sound is pursued. This holds true for plasma speakers as well. However there are some attributes of plasma speakers that make them unique and in many ways better than conventional speakers.

Plasma speakers are unique in that they have absolutely no resonance, and the best speakers have an accurate pulse and frequency response up to 100 kHz. This is because they are effectively mass less, meaning that they have an infinitesimally small transient response. The driver actually has a mass equal to that of the surrounding air, which means the acoustic coupling is 1:1. The drawbacks of conventional plasma speakers is that they are often inefficient as they require a very high voltage to operate, and either produce ozone (for speakers that use ionized air) or require a constant, somewhat expensive fuel (in the case of helium based speakers). An alternative to this is flame speakers which would use a combustible material to create a flame that could then be modulated like any plasma. However, this type of design has not been explored to any great degree. (Olson, 2001)

2.5 Tesla Coil Designs

The following section describes the three main Tesla Coil designs and their functionality as musical tools. The first design to be examined is the conventional or Spark Gap driver. This is the simplest design, but is less suitable for audio modulation than the others. Next we will cover the Solid State Tesla Coil (SSTC), which is the most common design used for audio modulation.

The main difference between the two designs is the replacement of the spark gap with solid state switches. Thirdly, this section will look at the Dual Resonant Solid State Tesla Coil (DRSSTC) design and highlight its advantages and disadvantages over the SSTC. The fourth section will describe the secondary coil and the process of finding its resonant frequency. Finally we will examine the process of pulse width modulation (PWM).

2.5.1 Spark Gap Tesla Coil

A classical spark gap Tesla Coil includes two main stages of voltage increase. The first being a conventional iron core transformer, and the second being the air core transformer formed by the resonant coils. The driver circuit for a spark gap Tesla Coil consists of the iron core transformer, a capacitor bank, a spark gap, and the resonant air core transformer. Figure 7 displays a schematic for a spark gap Tesla Coil.

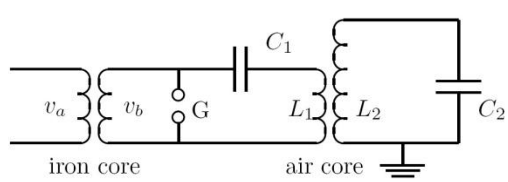

Figure 7: The Classical Tesla Coil (Johnson, 2009)

This circuit consists of the driver, which includes the iron core transformer, the spark gap G, capacitor bank C_1, and primary coil L_1. The schematic also shows the secondary coil circuit which consists of the secondary coil L_2 a ground, and the combined capacitance of the windings of L_2 and the top load of the coil. The capacitor bank of the driver is typically a low loss, high voltage capacitor that is used to build up charge before the spark gap activates. The spark gap itself acts as a switch that closes when enough voltage has been built up. Typically, the spark gap simply consists of two metal spheres separated by a small air gap. When the gap is not sparking the primary coil acts as a short and the capacitor is being charged by the iron core transformer. This is shown in Figure 8.

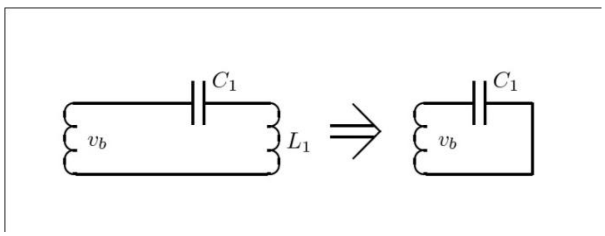

Figure 8: C1 Being Charged With Spark Gap Open (Johnson, 2009)

While the Spark gap is not conducting, the iron core transformer is increasing the AC voltage input, causing the capacitor bank to build up charge. The primary coil acts as an inductor, opposing the change of current and building up energy in the form of a magnetic field. When the spark gap activates it allows the circuit to oscillate, effectively becoming an RCL Oscillator with the spark gap as the main source of resistance. The circuit will oscillate at a frequency determined by C_1, L_1, L_2, and C_2.

When the gap is active, the complete circuit diagram for the driver circuit and secondary coil can be shown by Figure 9.

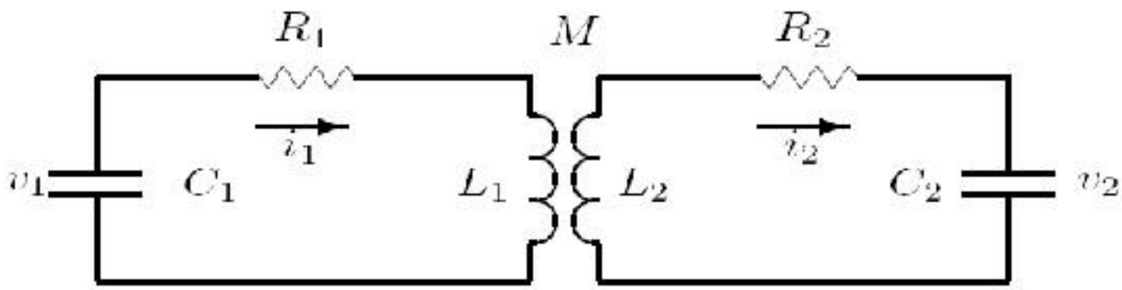

Figure 9: Lumped circuit model of a Tesla coil with active arc. (Johnson, 2009)

During this time, the energy stored in C1 is dispersed throughout C_1, C_2, L_1, L_2, and M where M is the mutual inductance of the Primary and Secondary circuits and R1, and R2 are their respective resistances. If designed correctly, all of the energy in the Capacitor bank can be transferred to the secondary coil within a certain time t. This means that at time t there is no voltage across C_1 and no current across L_1. If the gap is opened at this point then there is no way for the energy to be transferred back to C_1. This causes the Secondary to act as a separate RCL circuit that oscillates with a frequency determined by C_2 and L_2. If a proper tuning of the two circuits is found, it is possible to build very large voltages in the Secondary, leading to large discharges. (Johnson, 2009)11

2.5.2 Solid State Tesla Coil

A Solid State Tesla Coil operates differently from a classical spark gap coil in that it implements bi-polar junction transistors (BJTs), metal-oxide semiconductor field effect transistors (MOSFETS), or some other form of solid state device to create oscillations. Figure 10 shows a simple Tesla Coil driver circuit using two of these switches.

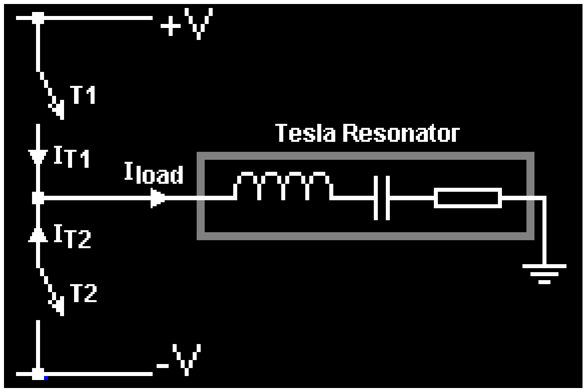

Figure 10: Sample Solid State Tesla Coil Driver Circuit (Burnett, 2001)

This design creates a switching cycle during which the two switches alternate between off and on while a sinusoidal wave current is passed through the system. In the first stage the first switch is on while the second is off, causing current to flow into the load through T1. Then the first switch is turned off while the second is turned on, causing the current to flow out of the load through T2. During this cycle the sinusoidal load current passes through zero at two points, halfway through the cycle when switch one turns off and switch two turns on, and at the end of the cycle when they switch back. It is very important that the switch occurs at these points in order to reduce the switching losses and any voltage spikes or unwanted ringing. This also helps to improve the load sharing between parallel switches, and reduces the amount of avalanche stress on series switches. The driver can also be built with a variable circuit using a timer circuit or a pulse width modulation (PWM) controller. These types of controllers will be covered in later sections.

The advantage of a Solid State Tesla Coil over a classical spark gap Tesla Coil is that it is easier to modulate the frequency using PWM controllers or timer circuits. Also, the spark gap on a classical Tesla coil is very loud, sometimes louder than the discharge from the secondary coil, and it can produce intense UV light which is harmful to the eyes. (Burnett, 2001)

2.5.3 Dual Resonant Solid State Tesla Coil.

A Dual Resonant Solid State Tesla Coil (DRSSTC) operates much like a conventional spark gap Tesla Coil in that it has a similar corona discharge and implements a capacitor bank. However, instead of the spark gap, a DRSSTC implements a half-bridge of MOSFETS or IGBTs. The combination of the capacitor bank and solid state switches, both of which are resonators, gives the DRSSTC its name. This combination leads to better control over the length appearance, and sound of the spark than a classical Tesla Coil, which means that like a SSTC a DRSSTC can be audio modulated. One main difference between SSTCs and DRSSTCs is that SSTC can operate safely in steady state without much danger, while DRSSTC that are driven for extended periods at resonant frequency run the risk of blowing the IGBTs or causing overvoltage of the primary capacitor. Thus, more precautions must be made with a DRSST in order to achieve longevity. Although audio modulation is possible with a DRSSTC, this project will focus on the SSTC design as it will be easier to produce with the available resources and time.

2.5.4 Secondary Coil Design

The secondary coil of the Tesla Coil is where the voltage is built up before being release through corona discharge. In simplest terms it can be modeled as a circuit formed by a capacitor in series with a resistor and inductor. This is shown in Figure 11.

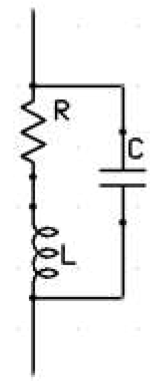

Figure 11: Sample Schematic of Secondary Coil of a Tesla Coil (Johnson, 2009)

The resistance is created by the large amount of wire used, typically hundreds or thousands of turns, while the inductance is that of a single layer of tightly wound coil, as shown in Equation 1.

$$L = \frac{\mu N^2 A}{\ell}$$

Equation 1: Inductance of a Solenoid (MacDonald, 2009)

In this equation, N stands for the number of turns in the coil, which is often large for secondary coils. A stands for the cross sectional area of the coil, and ℓ stands for the overall length of the coil. This is not to be confused with the length of the wire. ℓ is effectively the height of the coil for most coil designs. The final component, μ is the permeability of free space, a physical constant equal to $4\pi*10^{-7}$ H/m

The capacitance of the coil is more difficult to determine. While the capacitance of this simplified example can be modeled as the voltage difference between the top and bottom of the coil, most coils include a sphere or toroid at the top of the coil which changes the image of the secondary coil as shown in Figure 12.

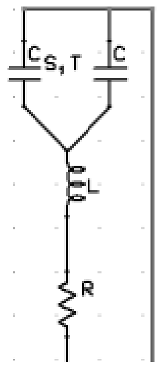

Figure 12: Equivalent Circuit Representation of the Secondary Coil with a Capacitive Load (Johnson, 2009)

In these cases, the capacitance used to find the resonant frequency of the Tesla Coil, must include the capacitance of the coil as well as the capacitance of the top load with the ground. To find the capacitance of a spherical top load we begin with Equation 2.

$$C = \frac{4\pi\epsilon}{\frac{1}{a} - \frac{1}{b}}$$

Equation 2: Capacitance of Spherical Capacitor

Figure 13: Spherical Capacitor (Johnson, 2009)

Equation 2 shows the capacitance between two spheres, with one sphere placed inside the other. Figure 13 shows the physical model of this. If b, the radius of the outer sphere, is made larger the capacitance will drop, eventually reaching a minimum as b→∞. At this point the equation for the Capacitance of the sphere will be Equation 3.

$$C_\infty = 4\pi\epsilon a$$

Equation 3: Capacitance of Spherical Capacitor as b→∞

Another type of top load typically used in Tesla coils is the toroid. The formula for finding capacitance of a toroid is slightly more complex than that of the sphere, as shown by Equations 4 and 5.

$$C_S = \frac{1.8(D-d)}{\ln\left(\frac{8(D-d)}{d}\right)} \quad \left(\frac{d}{D} < 0.25\right) \ (4)$$

Equation 4: Capacitance of a Toroid for d/D < 0.25

$$C_S = 0.37D + 0.28d \qquad \left(\frac{d}{D} > 0.25\right) (5)$$

Equation 5: Capacitance of a Toroid for d/D > 0.25

In these equations D stands for the toroid major diameter, from outside to the outside, while d is the toroid minor diameter, the thickness of the ring. This is shown by Figure 14.

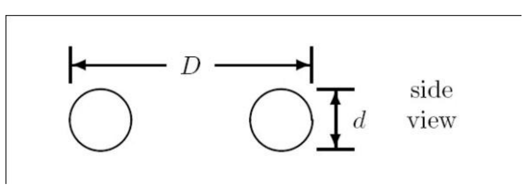

Figure 14: Toroid Dimensions (Johnson, 2009)

In both equations the capacitance is measured in pF and the diameters are measured in cm. To determine which equation is used it is necessary to find the ratio between the two diameters. If the minor diameter is greater than 25% of the major diameter then equation 5 is used. If the minor diameter is less than 25% of the major diameter then equation 4 is used. These equations are empirical, and because capacitors have a tolerance of around 20% these equations will be appropriate.

To find the total capacitance of the Tesla coil, we must find the capacitance of the coil itself. Medhurst developed a number of empirical equations to calculate this.

$$C_m = HD\ pF$$

Equation 6: Capacitance of a cylindrical coil of wire (Lux, 1998)

Equation 6 is the simplest expression for this capacitance, and shows that the capacitance in the coil is equal to the diameter of the coil, D, times a factor H. H is determined by the ratio of D to the length of the coil ℓ. For ℓ/D =2 H =0.51, while for ℓ/D=5 , H = 0.81. For ℓ/D from 2 to 5 this relationship is linear. Equation 7gives H for any value of ℓ/D between 2 and 8.

$$H = 0.100976\frac{\ell}{D} + 0.30963$$

Equation 7: Formula for Medhurst Constant H, for values between 2 and 8 (Johnson, 2009)

In order to find the total capacitance of the Tesla coil, the individual capacitances of the top load and the coil must be combined. Since they are connected in parallel the capacitances will simply be added. However, there are a few factors that influence the resulting capacitance. When the top load and coil are combined, shielding occurs, causing the total capacitance to decrease, yet when both are brought near the ground, the capacitance increases. Since these factors oppose each other, the capacitance is typically within 20% of the calculated sum. (Johnson, 2009)

Once the total capacitance and inductance are found it is possible to find the resonant frequency of the coil using a second order differential equation.

$$v(t) = L\frac{d^2q}{dt} + R\frac{dq}{dt} + \frac{1}{C}q(t)$$

Equation 8: Differential Equation for Voltage across a Capacitor (Blinder)

This equation can be solved to obtain:

$$F_r = \frac{1}{2\pi\sqrt{LC}}$$

Equation 9: The resonant frequency of a single coil.

Where F_r stands for the resonant frequency of the coil, L stands for the inductance, and C stands for the total capacitance. Finding the exact resonant frequency of the secondary coil is essential for most spark gap designs. However for Solid State coil designs it is possible to sweep through a large range of frequencies so less precision is needed.

2.5.5 Pulse Width Modulation

Although Tesla Coils can be constructed in many ways, most Solid State Coils implement a technique known as Pulse Width Modulation (PWM) to properly control the flow of current through the coils. In simple terms, Pulse Width Modulation is the control of the “on” and “off” time of a voltage pulse. This pulse is then applied to some sort of MOSFET or BJT gate which regulates the flow of current in another part of the circuit.

The simplest way to generate a modulated signal is by using a differential comparator, and in fact this is what most PWM controller components, such as the TL494 used in this design, use to generate the signal. In this configuration the differential comparator generates either a high or low signal based on the relationship between the two input signals. In most cases a triangle or saw tooth wave is fed into the – input while an audio, or other signal is fed into the + input. When the audio signal is at a higher voltage than the triangle wave, the comparator will output a high signal. Likewise, when the audio signal is a lower voltage than the triangle wave the comparator will output a low signal. This relationship is shown in Figure 15.

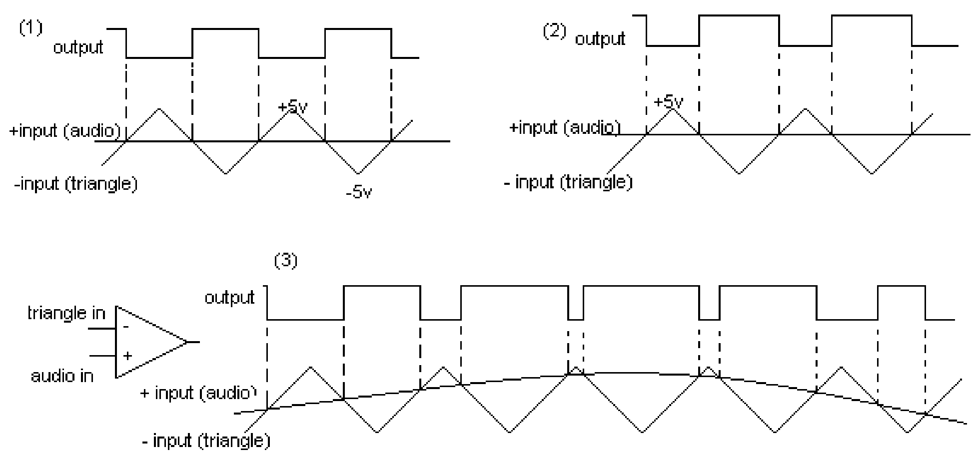

Figure 15: Example of pulse width modulation using a triangle wave and audio signal (Cloutier)

As we can see in Figure 15, the relative duration of high and low signals is determined by the voltage of the + input or audio signal. In the case of a 10 volt peak to peak triangle wave, an audio signal of a constant voltage of 0v will create a pulse that is 50% high (on) and 50% low (off) as seen in Figure 15(1). In terms of duty cycle, or the percentage of "on" time, this is 50%.

If the audio signal has a higher voltage, as shown in Figure 15 (2), its voltage will be higher than the triangle wave for a longer period of time, and thus the duration of the high signal from the comparator will be longer. As the audio signal fluctuates rapidly, the duration of the high signal output will change. This variable duration is what we call pulse duration, or width, modulation.

This output signal is then amplified and sent to the MOSFET or other switching device, which only allows current to pass when it receives a high signal from the PWM. For Solid State Tesla Coils and plasma speakers this switching translates to the sparks produced from the secondary coil. These sparks occur so rapidly that they replicate the fluctuations of a speaker driver and cause high frequency pressure waves in the air.

Thus through pulse width modulation, an audio signal is transformed into a series of discrete high and low signals that vary with the fluctuations of the audio voltage. These signals then determine the flow of current through the primary coil, which in turn controls the flow of current in the secondary coil. The flow of current in the secondary coil determines the voltage buildup and the resulting sparks from the coil, which reflect the fluctuations, and in turn the sound, of the audio signal. (Cloutier)

3. Methodology

3.1 Primary Coil and Circuit Design

The majority of the design and construction process revolved around finding an appropriate design for the primary circuit of the coil. The coil would need to be driven at a frequency of at least 44 kHz to be able to produce all of the frequencies that can be heard by the human ear. Since a higher frequency would produce a clearer sound, it would be preferable to have a coil and driver tuned to at least 100 kHz. Due to the lack of time and expertise, it was found that the best course of action would be to find an existing design. A number of designs were examined and two were selected to be attempted, for various reasons.

The first design was created by Richie Burnett, a Tesla Coil enthusiast and lead engineer at a university in England. This design uses four sub charging circuits in what is called a full H bridge design. The primary is run off of a 240v main power supply and driven at approximately 350 kHz. Although 350 kHz would be perfect, the driving frequency would ultimately depend on the design of the secondary coil. Regardless, because both designs use solid state transistors the driving frequency can be adjusted to match whatever resonant frequency the secondary coil requires. This circuit design is very intricate and requires that all of the components operate flawlessly, which is often more difficult to achieve than it may seem. However, because such a high voltage is being fed into the primary the resulting sparks will be very large. Figures 16 and 17 show the Control and Power Circuits for this design respectively.

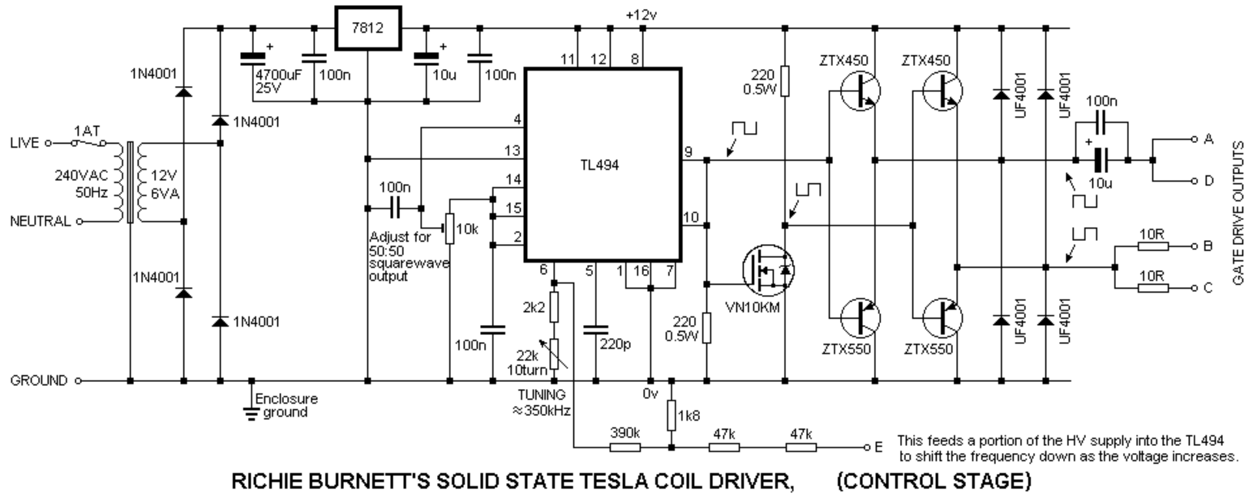

Figure 16: Solid State Tesla Coil Control Circuit (Burnett, 2001)

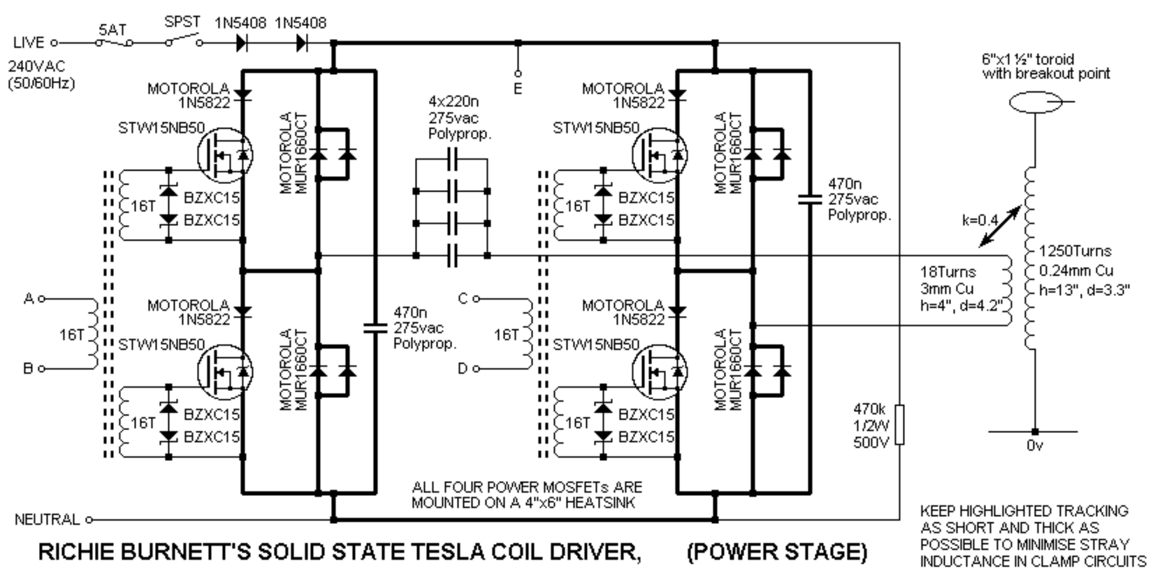

Figure 17: Solid State Tesla Coil Power Circuit (Burnett, 2001)

Figure 17 shows the four sub charging circuits centered on the STW15NB50's which are N channel MOSFETS. These MOSFETS receive a square wave from the control circuit via the small 16 turn 1:1:1 transformers. This square wave is inverted for the MOSFETS on the right of the schematic. This means that the two pairs of circuits will be switched out of phase, causing the current to flow only through one pair of MOSFETS at a time. This is essential for the design to function and, if done successfully, will increase the amount of current flowing through the primary coil as the current is “swung” back and forth by the circuits. The Control circuit is a typical PWM design with a few adjustments. The VN10KM component is a special N channel enhancement mode MOS transistor. Its basic purpose in this design is to change the phase of the square wave so that the second output is 180 degrees out of phase with the first. The rest of the additions to the control circuit are designed to step down the 240 v AC to the 12v DC that the TL494 requires.

As mentioned before this design is very intricate and all of the components must be working properly. That being said some of the parts used in the design are now obsolete and can no longer be purchased. Thus, it was necessary to use substitute parts for many of the components in the design. This led to some interesting effects that resulted in the circuit drawing too much current. As a result the fuse in the power circuit would consistently blow and on numerous occasions, a number of the components would short out and break. After numerous tests it was decided that the best course of action would be to use the alternative circuit which had been proven to work.

The second circuit design that was chosen was a simpler design that used only 24V DC to power the primary coil. It had been used in a previous project and was known to provide results with the components listed in the design. The circuit was originally modified from a plasma speaker schematic provided by HV Labs.

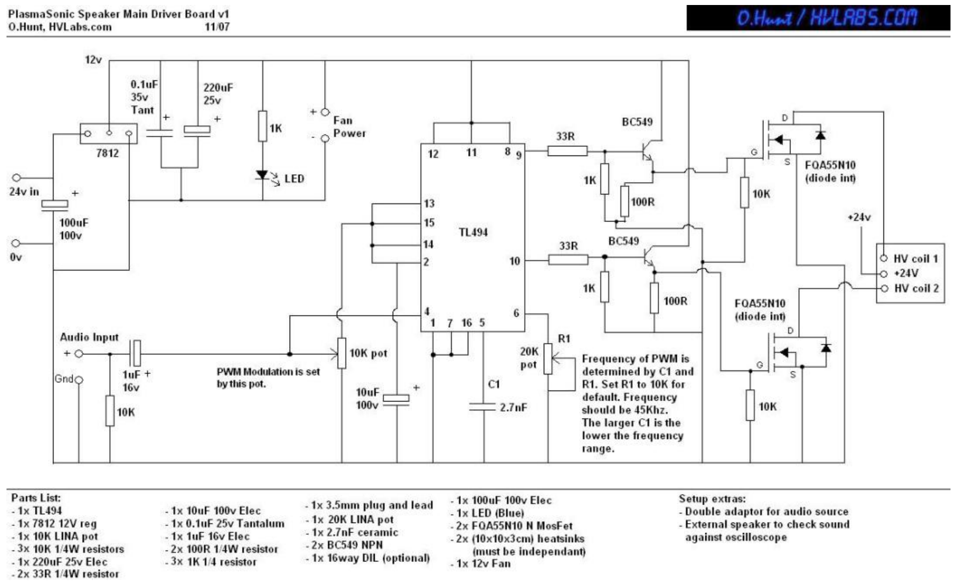

Figure 18: Unmodified Plasma Speaker Schematic (Hunt, 2008)

Unlike the original circuit, the modified version called for a separate DC voltage source to power the TL494 PWM controller. This removed the need for the 7812 voltage regulator shown in the schematic above. Additionally, in order to add more protection to the MOSFETS, a pair of power diodes where added to give the current an alternative path around the MOSFETS in case there was a problem with the switching times. The modified schematic can be seen in Figure 19.

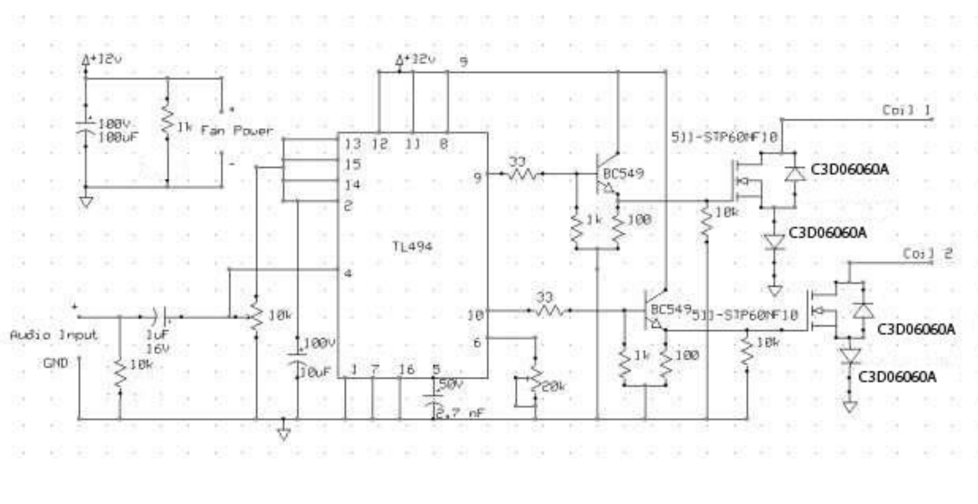

Figure 19: Modified Tesla Coil Schematic (MacDonald, 2009)

This design is different from the previous one in that in runs off of DC instead of AC. As can be seen in the schematic above, there are two connections to the coil, just as there were in the first design. These leads connect to a center tapped primary coil which is fed 24v DC from the center tap. One half of the coil is wrapped clockwise while the other is wrapped counterclockwise. This is shown in Figure 20.

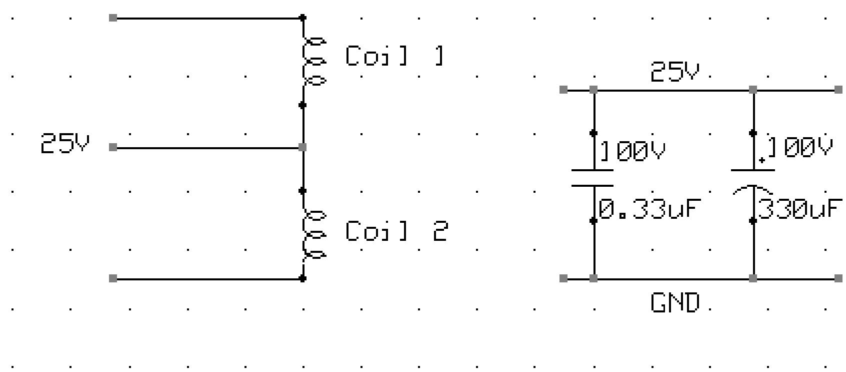

Figure 20: (a) The Center Tapped Primary Coil. (b) The Filter for the Primary Coil (MacDonald, 2009)

The AC needed to induce the appropriate magnetic field on the secondary coil comes from the switching MOSFETS. The MOSFETS switch out of phase so that one is off while the other is on. This creates and alternating current in the primary coil by limiting which direction the current can flow. The actual coil used for this project was two lengths of 16 AWG wire wrapped around the base of the secondary coil 4 times each in opposite directions. The wires were twisted together at the top to form the center tap and the other ends were attached to the MOSFETS. In order to achieve the best coupling between the Primary and Secondary coils, the coil was mounted on the bottom edge of the secondary coil and insulated from it with Kapton film and electrical tape.

Since the PWM controller and surrounding components only required 12v, the majority of the driver circuit could be mounted on a breadboard. This eliminated the need for complicated and time consuming soldering on most of the components. Figure 21 shows the breadboard supporting the PWM circuit.

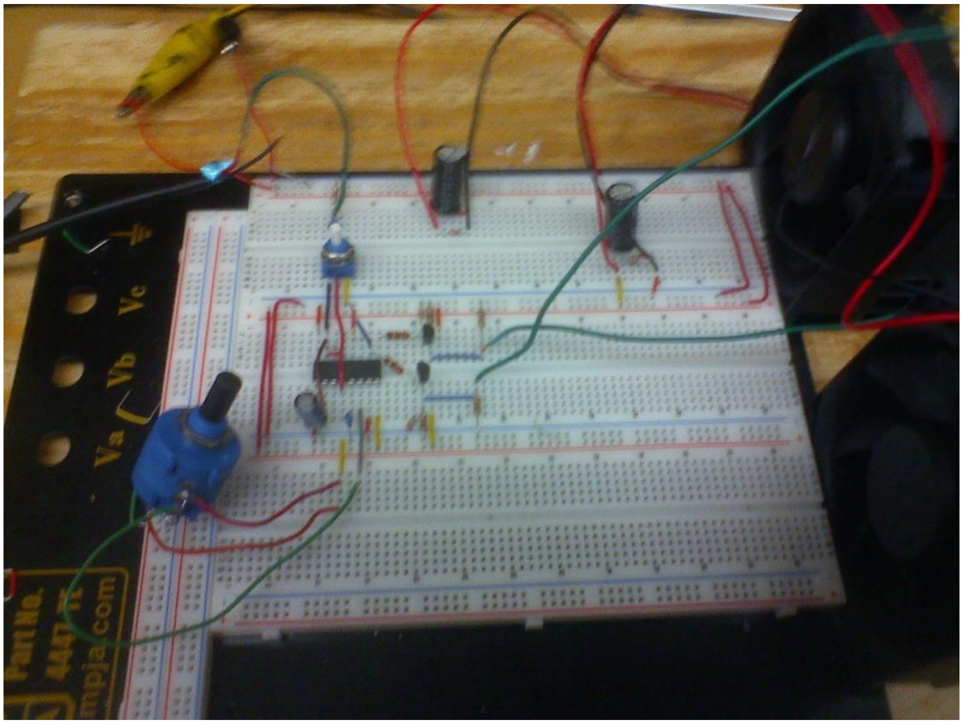

Figure 21: PWM circuit mounted on a common breadboard

However, due to the high voltages running through the MOSFETS and power diodes, these components would need to be connected by something that could handle higher voltages. Since the components handling this voltage were likely to heat up significantly during operation, both MOSFETS and all four power diodes were mounted on heat sinks and cooled by small 12v fans powered from the same source as the PWM circuit. The easiest way to mount the components was to solder the appropriate connections using lengths of wire. Then a thermal compound was used to establish a thermal connection with the heat sink and the components were taped in place. Figure 22 and 23 show the mounted MOSFETS and power diodes, and the Primary Coil mounted on the base of the secondary.

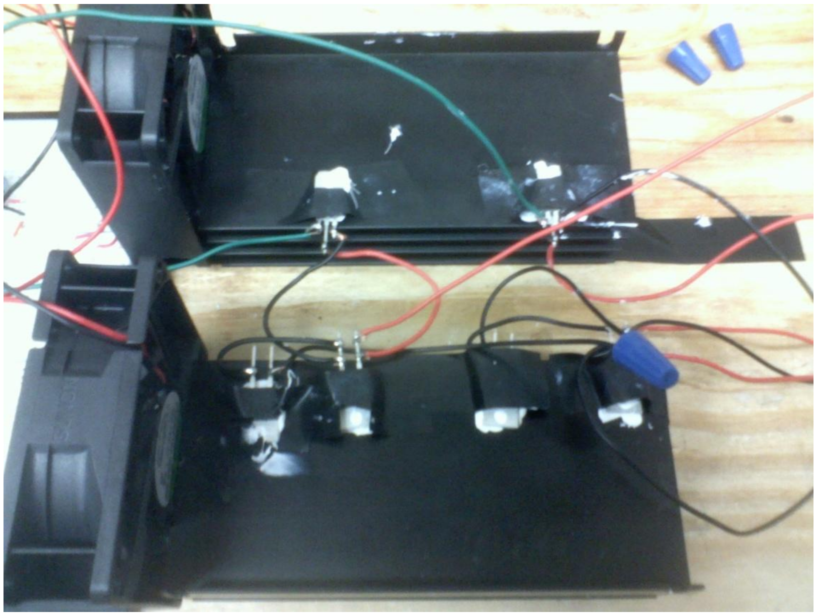

Figure 22: MOSFETS and power diodes mounted on two heat sinks.

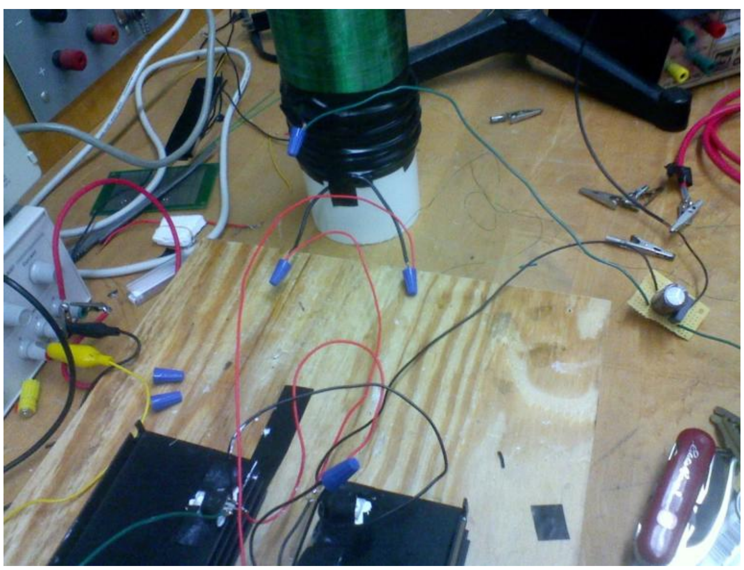

Figure 23: Primary coil mounted on the base of the secondary.

The components used in the circuit are slightly modified from the ones used in the original schematic since some of the parts were obsolete and no longer sold. Table 1 shows a list of the materials used in the circuit.

Table 1		
Part number	**Type of Component**	**Quantity**
C3D06060A	Power diode	4
TL494	PWM Controller	1
BC546	BJT Transistor	2
511- STP60NF10	Power MOSFET	2
Heat Sink	Heat dissipation	2
Thermal Compound	Heat Transfer	1 bottle
12V fan	Heat Dissipation	2
2.7 nf, 50V	Capacitor	1
0.33 uf, 100V	Capacitor	1
1.0 uf, 16V	Capacitor	1
10 uf, 100V	Capacitor	1
100 uf, 100V	Capacitor	1
330 uf, 100V	Capacitor	1
2200 uf, 10V	Capacitor	1
0.1 uf, 35V	Capacitor	1
20 kΩ	Potentiometer	1
10 kΩ	Potentiometer	1
10 kΩ	Resistor	3
1 kΩ	Resistor	3
100 Ω	Resistor	2
33 Ω	Resistor	2

Table 1: List of materials

3.2 Secondary Coil Design

During the process of building the primary circuit, a number of different secondary coils were tried in search of the best results. There was a series of coils left over in the physics department from past experiments, but for the initial design, a custom coil was built. The coil was wound around a piece of PVC pipe that had an outside diameter of 3.5 inches. The pipe was attached to a lathe and spun slowly while the wire was kept taught so it would wind tightly and neatly. The coil was made using 30AWG wire, and wound approximately 1250 times around the pipe, leading to a total coil length of about 13 inches. The other two coils available were pre wound, also on PVC pipe with an outer diameter of 3.5 inches. One coil was wound with 30 AWG wire and the other with 22 AWG wire, and both had a final coil length of approximately 1 meter.

The difference in wire gauge had a profound impact on the number of turns in the coil, and therefore on the resonant frequency and amount of power build up in the coil. Thus it was necessary to calculate the resonant frequency of each coil in order to determine the appropriate frequency to drive the primary circuit. This was done using the equation for the resonant frequency of a coil as shown in Section 2.5 which is:

$$F_r = \frac{1}{2\pi\sqrt{LC}}$$

Equation 10: Resonant Frequency of single coil

To find the resonant frequency of a coil using this equation we must know L, the inductance, and C, the internal capacitance, of the coil.

To find C we can use the Medhurst formula for finding the capacitance of a cylindrical air core coil. In section 2.5 we had the equation:

$$C_m = HD\ pF$$

Equation 11: Formula for Medhurst Capacitance

Where H is:

$$H = 0.100976\frac{\ell}{D} + 0.30963$$

Equation 12: Equation for Medhurst Constant for 2< ℓ/D < 8

for values of ℓ/D between 2 and 8. To find out if this equation is appropriate we must simply calculate the ratio of our coil length over the coil diameter. Which in this case is:

1m/ 0.0889m = 11.24.

Equation 13: Calculation of ℓ/D

Since this value is greater than 8 we must use an alternative formula for H which is:

$$H = \frac{0.28\frac{\ell}{D}}{\ln\left(1 + \frac{\ell}{2D}\right)} + \left(\frac{\frac{\ell}{D} + 2}{12\pi}\right)^2 + \frac{1.672}{4\pi\frac{\ell}{D} + 1} - 0.36$$

Equation 14: Formula for Medhurst Constant (Jermanis)

Plugging in the values for ℓ and D we get:

$$H = \frac{0.28\frac{1}{D * 0.0889}}{\ln\left(1 + \frac{1}{2 * 0.0889}\right)} + \left(\frac{\frac{1}{0.0889} + 2}{12\pi}\right)^2 + \frac{1.672}{4\pi\frac{1}{0.0889} + 1} - 0.36 = 1.44$$

Equation 15: Calculation of Medhurst Constant

So for our two coils of length 1m our capacitance will be:

$$C_m = 1.44 * 8.89cm = 12.8\,pF$$

Equation 16: Calculation of Medhurst Capacitance

The next step is to find the inductance of the coil which is given by the equation:

$$L = \frac{\mu N^2 A}{\ell}$$

Equation 17: Equation for inductance of a solenoid

Where N is the number of turns, A is the cross sectional area of the solenoid, ℓ is the length of the coil, and μ is the permeability of free space. The resonant frequency for all the coils was calculated but for simplicity's sake we will only show the calculations for the coil that was used, which was the 1m 30 AWG coil. For this coil the number of turns, after making adjustment for the insulation on the wire, came out to be approximately 3921 turns. Plugging this into our equation for the inductance of the coil we get:

$$L = \frac{4\pi * 10^{-7} * 3921^2 * \pi * 0.04445^2}{1} = 0.1199H$$

Equation 18: Calculation of inductance of the secondary coil

Now that we have both the inductance and capacitance of the coil we can find the resonant frequency:

$$F_r = \frac{1}{2\pi\sqrt{0.1199 * 12.8 * 10^{-12}}} = 128.471\,kHz$$

Equation 19: Calculation of resonant frequency of the secondary coil.

Note this resonant frequency may change slightly depending on the type of load placed on top of the coil. This equation gives the resonant frequency of the coil without any modifications.

To find the resonant frequencies of the other two coils we simply use the same equations with the appropriate dimensions and number of turns. The resonant frequency calculated for the long, 1m coil wound with 22 AWG wire is approximately 345 kHz, while the resonant frequency for the shorter 13inch coil wound with 30 AWG wire was approximately 335.7 kHz. The reason the taller 30 AWG coil was chosen is because it had the greatest number of turns and would therefore have the greatest voltage increase when coupled with the primary coil.

During operation the true resonant frequency of the coil was found to be approximately 129.7 kHz which is remarkably close to the calculated frequency. At this frequency the spark was steady and completely silent when no audio signal was added. Currently the power supply available is limited to 16v at approximately 3 A. This means that the sound is lower than expected but it is still audible.

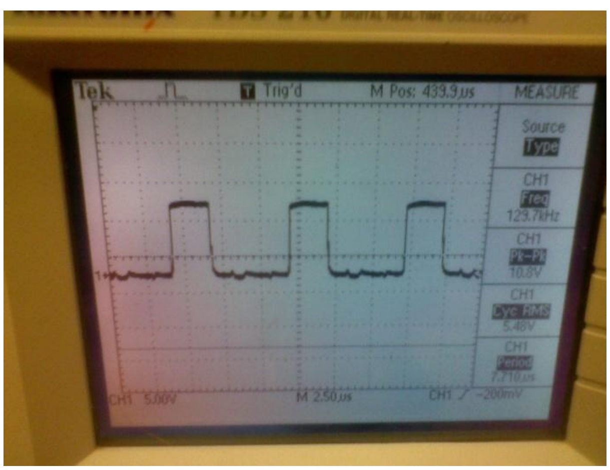

Figure 24: Square wave generated by PWM controller during operation. The frequency is approximately 129.7kHz

3.3 Designing the Max program and Musical Interface

The final portion of the project involved creating a non conventional musical interface that could be added to the coil to create a unique musical instrument. The type of instrument depended heavily on the materials available. In this case, these were a series of sensors that output MIDI information that could be processed using various audio programs. The sensors available were four bend sensors, that output MIDI values from 0 to 127 depending on to what degree they were bent. The other sensors available were a simple turn sensor that output values according to its rotation, and a motion sensor that would output a stream of values whenever it detected a moving object.

The original plan for the interface revolved primarily on creating something that was biological in nature, and would reflect some aspect human motion, beyond simply playing a note on a keyboard or other instrument. The bend sensors would be ideal for mapping the motion joints and so the idea arose of creating a glove with the bend sensors implanted in select fingers. While there are numerous joints on the human body that may have worked for this purpose, the hands and fingers provided a centralized location that could include all of the sensors and still provide fine control for each. Originally the turn sensor and the motion sensor were not going to be part of the interface. However, during the programming process it was discovered that the turn sensor could prove to be a useful tool for changing settings in the program without sacrificing the use of one of the bend sensors. The motion sensor was not included because it did not allow as much control over the output values as the other sensors.

In order to determine how best to place the sensors so they could interact easily with the program and each other, it was necessary to determine what purpose each would serve. For that, the program needed to be created. The best option for this was to create a patch in the Max/MSP program which could easily interface with the MIDI messages from the sensors and transform them into signal messages. These would then be output from the computer as an audio signal into the PWM circuit. The patch can be seen in Figures 25 and 26, note that Figure 26 shows the sub patch that is programmed into the main patch shown in Figure 25.

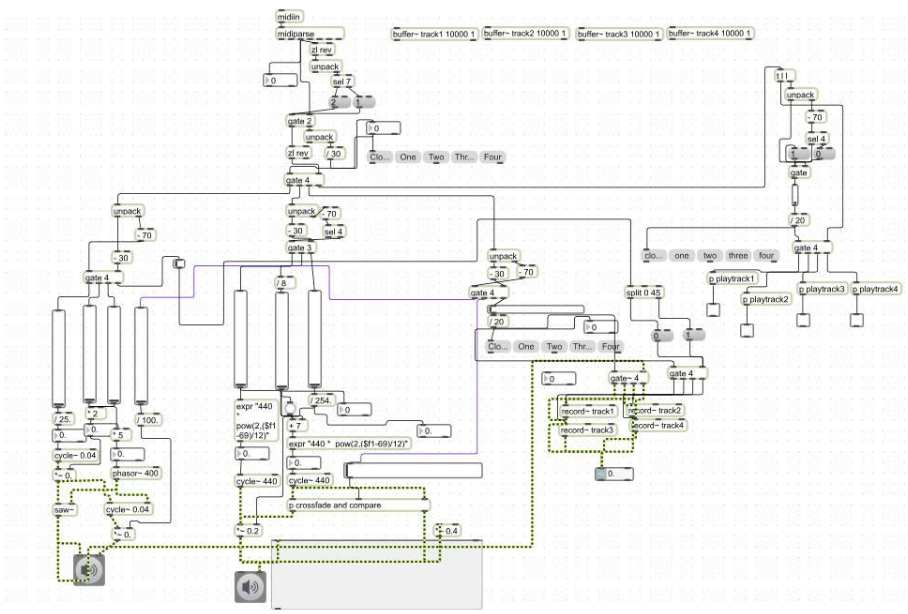

Figure 25: Max Patch for Music Interface: the leftmost section is the beat generator, the second section is the tone generator, the third section is the recording channel selector and additional effects, and the last section is the loop controls.

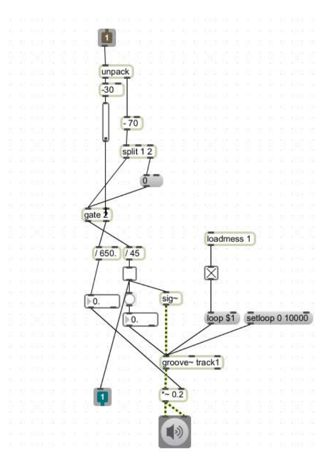

Figure 26: Playback object for switching on/off recorded loops.

The prime concern with the program and the interface was how to create something that could be considered musical, and work well when limited to the higher frequencies. The limit on frequency is due to the coil’s limited capabilities of reproducing low frequency sounds. The end result of the program was a series of signal generator objects that could interact in various ways to produce interesting combinations of oscillations. These produced beats of varying length and frequency as well as a signal that was shifted by discreet frequencies to replicate traditional musical notes, with the added effect of optional harmonics up to one octave above. A recording function was also added that could be used to create short loops that the performer could improvise over.

The first section we will analyze will be the beat generator. This section is comprised of a saw object which is modulated by a cycle object and a phazor object. The phasor object determines the frequency, or the effective pitch, of the primary tone which is then modulated by the cycle object. The frequency of the cycle object is scaled to be significantly less than that off the phasor which results in a steady beat that can be modulated in pitch by one sensor and frequency and duration by another. A third sensor controls the volume of the saw signal so that the beat can be dynamic in volume as well as pitch and tempo, or even removed entirely. During this setting the fourth sensor controls the recording function. When the fourth sensor is bent to a certain extent, the record function begins recording the signal from the saw object into the selected buffer object. The signal is recorded until the sensor is released or the buffer reaches its limit which is currently set to ten seconds. There are currently four buffer objects capable of recording signals from the beat generator and each can be played in any combination with the others.

The next section is the tone generator which takes the MIDI data from the sensor and converts it into a pure tone using the cycle object. Unlike the beat sensor, which produces continuous tones, a formula is applied to the data going to this cycle object that converts the values into discrete musical notes. This is done using the equation:

$$F = 440 * 2^{\frac{N-69}{12}}$$

Equation 20: Conversion of MIDI notes into note frequency. (Signal Parameters in MSP, 2010)

Where N is the MIDI value corresponding to pitch, assuming that the MIDI value 69 corresponds to the A above middle C which has a frequency of 440 Hz. This tone can be accentuated by a secondary tone that can range anywhere from the tonic of the base tone to the octave above. This allows for a simple method of creating two tone harmonies or dissonant pairs. This can create interesting effects if the secondary tone is placed at the second or third of the base tone and the base tone is then shifted rapidly in pitch. Once again the third sensor in this section controls the volume of these tones and, like the beat generator, they can be removed entirely if desired. An additional effect in this section is the ability to modulate the volume of the secondary tone separately. This allows the performer to select a base tone, and then remove it so that the secondary tone can be modulated freely within a set octave range. This effectively creates a fine tuned control over pitch within a limited range.

The third section controls the additional effects of the first two sections and allows the performer to select which buffer object he would like to record with. In this section, the first sensor controls the volume of an additional effect in the beat generator. The added effect is an oscillating pure tone that changes in pitch and tempo with the beat produced by the beat generator. One interesting effect of this tone is a strong bass beat that occurs when its volume is increased. This will not be very discernable with the plasma arc, however if a bass crossover, and subwoofer, is added to the device it can be used to accentuate the rest of the performance.

In this section, the second sensor controls volume of the secondary tone of the second section. If the volume of the base and secondary tones is still up when this secondary volume is increased it will boost the volume of the secondary tone, and often cause it to clip, which creates some interesting effects. If the volume for the second section is turned down then the secondary tone will be isolated and the performer will have the fine tuned pitch control mentioned earlier.

The fourth sensor in this section determines which of the four buffer objects the signal from the beat generator will be recorded to when the recording begins. The sensor is scaled so that when it is unbent the first buffer object is selected, and the fourth buffer object is selected when it is fully bent. The recording can only be started while the beat generator section (section 1) is being modified; it cannot be done from any other section.

The fourth and final section is the loop control. In this section, the first sensor determines whether or not a loop starts or stops. If the first sensor is bent to a certain extend the selected loop will begin and continue looping until the sensor is released. If the sensor is bent to nearly its maximum value the loop will play at twice the normal speed. The second sensor determines the volume at which the loop will play, which can be determined for each individual loop. This allows for the performer to set up multiple loops to create rhythmic effects. In the Max/MSP program if the output volume of any signal exceeds a value of 1 the signal will clip. To counter this, the volume of the loops and most other effects in the patch are limited to 0.2, meaning that 5 different signals can be sent to each of the two channels without the risk of clipping. The fourth sensor in this section determines which buffer object is being accessed. Just like in the third section this sensor is scaled so that it selects the first buffer object when unbent and the fourth buffer object when completely bent.

The purpose of the turn sensor is to determine which section is being accessed. There are five choices that are available to the performer; sections 1-4 and closed. If the closed section is selected then the data from the bend sensors will not be transmitted to any of the other sections and the output will remain how it was.

After, and in part at the same time, each sensors function in the patch was decided; a design for how the sensors would be integrated with the glove was formed. The first three sensors were placed in the thumb, pointer finger, and middle finger of the right hand, while the fourth sensor was placed in the thumb of the left hand. The turn sensor was placed in the back of the right hand glove where it could be easily manipulated by the left hand. This setup allows a great deal of control of the first three sensors, which are the primary controls for the sound generation. By placing the fourth sensor in the left hand the sound and recording controls were separated. This made it easier for the performer to focus on controlling the music with one hand and recording with the other. Additionally by placing the fourth sensor in the thumb, the performer could easily manipulate the turn sensor on the right hand without accidentally altering the recording settings.

4. Analysis and Future Improvements.

4.1 Analysis

Figure 25 shows the corona discharge from the coil during operation. When the spark gap was set to about an inch in length, the spark produced was stable and nearly silent while no audio signal was added. Under these conditions, the circuit was powered with 24.0 +/- 0.1V and consistently drew around 7.2 +/- 0.1 A.

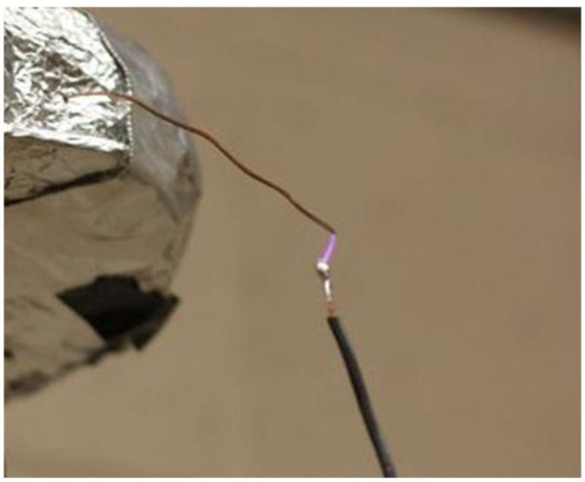

Figure 27: Active spark gap from secondary coil.

When an audio signal was added there was no noticeable distinction in the motion of the spark, however the audio could be heard quite clearly, if rather quietly. Two different power supplies were used in the final testing. The first could provide a maximum of 30V at 3As. This provided enough power to produce a spark and audible music. However, the current cap limited the voltage to about 16V. Since the amount of power being provided was limited, the size of the spark and thus the volume of the audio output were reduced. The second power supply had a much better range of 70V and 8 A. Using this power supply the circuit could be brought up to the recommended 24V while running at approximately 7.2 A. Under these conditions the audio output was significantly louder, yet still not particularly loud compared to speakers. There was also some danger of overheating the components, and several MOSFETS and power diodes were destroyed due to problems with heatsinking.

Since the coils act as a step up transformer, the voltage buildup in the secondary coil will be a factor of the voltage applied to the primary coil. For example in a step up transformer with a ration of 1:10, where there are ten turns in the secondary for every turn in the primary, the voltage in the secondary will be roughly ten times the voltage of the primary. For this coil, the ratio of turns between the primary and secondary coil is about 4:3921 or about 1:980. Thus, if the primary coil is given approximately 24V, the resulting voltage buildup in the secondary will be about 23.5kV. The base dielectric strength of air is about $3x10^6$ V/m or 3kV/mm. This value can vary depending on the shape and size of the electrodes and the temperature and pressure of the surrounding air. For a spark gap of 0.5 inches or 12.7mm the dielectric strength of the air is calculated to be 38.1kV. This is significantly more than our 23.5kV but remember that the shape of the electrodes changes this value. Specifically, the dielectric strength is reduced significantly if the electrodes have sharp points, as they did in this case. This allows for a voltage breakout at 23.5kV, or most likely less due to imperfect coupling, even at a distance of 12.7mm.

Overall the performance of the coil was acceptable, but not ideal for its intended purpose. The audio output was far too quiet to be a proper performance instrument. However, it could be boosted in a variety of ways which will be discussed in the next section. For now we will look at the operation of the musical interface.

The musical interface portion of the project was intended to use the bend and turn sensors, integrated into a glove, to provide MIDI data that the Max/MSP program would use to generate audio signals. Unfortunately an issue with the sensor interface prevented the output of the data from the sensors from reaching the program. As it was soon apparent that the interface would have to be replaced, an alternative method was sought until the problem was resolved. The sensors were replaced by a series of control knobs on a USB MIDI keyboard. These knobs output the same range of values as the sensors and worked very well in testing the Max program.

Initially the program only had a number of bugs, primarily related to combining the various messages from the signal generators into the two channels. This was remedied by lowering the maximum volume of each signal output to just under 0.2. If a composite signal ever reached an amplitude of 1 or greater it would clip. This meant that with these new settings up to five signals could be combined each channel without the risk of unwanted noise. However, this reduced amplitude meant that the audio output from the coil was also reduced, further limiting the already low volume level.

While the program and the coil both worked as expected, there were a number of things that could have been improved to produce more impressive results. If the spark length could be increased, the volume output would be increased as well making the device more viable as a performance instrument. The audio quality was quite good, but also could have been improved by increasing the resonant frequency of the secondary coil. The Max program worked as it was designed, however the use of the MIDI keyboard removed most of the biological interaction that was desired. The interface could also have benefitted from a greater number of sensors so that more joints could be used as controls.

4.2 Future Developments

Two options are available to increase the length of the sparks and loudness of the audio. Either the amount of power being applied to the coil could be increased or the number of turns in the secondary coil could be increased. If the voltage applied to the primary is increased, the resulting voltage buildup in the secondary will increase by the ratio of turns. This can lead to a rather significant increase in output voltage with only a small increase in the initial voltage. The problem with this is, the increase in voltage will require a larger current, and the power applied to the primary circuit will increase. This can be dangerous if the components are not properly heatsinked since there is a good chance they will heat up faster. This can also be dangerous if the components are not rated for the increased power. Also, because the voltage in the primary and secondary will be greater, there is a greater chance of arcing between the two, which will require additional insulation to prevent. Most primary circuits are built with a particular voltage in mind and it can be complicated to rework the circuit to accommodate higher voltages.

Alternatively, or additionally, to increasing the voltage applied to the primary, the number of turns in the secondary could be increased. This would increase the ratio of secondary to primary turns and thus increase the amount of voltage gain. The problem with this approach is the effect it will have on the resonant frequency of the secondary coil. The secondary coil must be carefully designed to provide a high enough voltage gain to achieve the desired sparks, yet still have a resonant frequency high enough to produce all of the audible frequencies. It is also important to be aware of the range of frequencies to which the primary coil can be tuned. If the resonant frequency of the secondary coil is outside this range it will be impossible to couple the two coils together.

Improving the musical interface beyond its current state could be as simple as integrating the gloves and sensors that were originally planned to be a part of the design. Beyond that, more sensors could be added to provide a greater range of effects and more precise control. Likewise, the types of sensors could also be modified. There are a number of sensors available that measure more interesting features of the human body such as heart rate and voltages on the skin that could be used to produce interesting musical effects that would be more difficult for the performer to control. This begins to get into the area of subconscious musical performance, in effect, a performance that is modified by the natural reactions of the body to environmental effects.

5. Conclusion

The goal of this project was extend my knowledge of electrical engineering and acoustics and shed some light on the technical and artistic nature of Tesla coils, while attempting to create a unique and interesting musical instrument. The coil that was created was capable of producing audible music, and the musical interface was limited only by the lack of properly functioning equipment. While there are a number of improvements that could be made, the project served its initial purpose in creating a coil capable of acting as an audio source and illuminating the finer points of creating such a coil. In addition to providing an example of an interesting form of musical performance that is largely ignored.

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