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Multi Channel Audio Environment

Charles Whitfield Bleakney
Worcester Polytechnic Institute

Christopher William Egan
Worcester Polytechnic Institute

Rayce Coleman Stipanovich
Worcester Polytechnic Institute

Richard Theodore Eberheim
Worcester Polytechnic Institute

Selim Enis Tanriverdi
Worcester Polytechnic Institute

See next page for additional authors

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Author

Charles Whitfield Bleakney, Christopher William Egan, Rayce Coleman Stipanovich, Richard Theodore Eberheim, Selim Enis Tanriverdi, and William Alexander Bacon



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**Multichannel Audio Production
FB 1406**

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Submitted to the faculty of
Worcester Polytechnic Institute
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Degree of Bachelor of Science

Student Authors:

Charles W. Bleakney IV
Richard Eberheim
Christopher Egan
Rayce Stipanovich
Selim Tanriverdi

Project Advisors:

Frederick Bianchi

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Abstract

This project is dedicated to creating an interactive and immersive three-dimensional surround sound interface. Different from most 5.1 or 7.1 systems, this immersive environment combines sound wave mechanics with a 40 point source system to localize a sound in a user controlled 3D environment. An interactive and real-time system like this can be used to experiment with psychoacoustics, and to simulate complex environments where many sounds are coming from many different places.

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1. Introduction

Consumer interest in audio products has increased as new technologies are developed. Consumer markets have reflected a continued interest in these products as revenues in this industry have been reliable since 2001. Interest in this market is fueled by development of higher quality components, such as sound bars and subwoofers, and the rising popularity and accessibility of 5.1 and 7.1 channel audio systems. Despite the consistent interest in audio equipment, systems larger than 7.1 are generally uncommon and can be very costly. While this is most likely the result of film and music generally being optimized for these smaller channel systems, the reliance on 5.1 or 7.1 channels limits the creativity of sounds that are being produced.

This project seeks to expand beyond a traditional audio experience through the creation of a massive multi-channel sound system. The system, upon its completion, will contain up to 40.1 discrete channels in which audio can be played through. Each channel will be able to play sounds that are independent of the other channels in the system allowing for a greater amount of creativity to be expressed within an audio project. The system will be capable of interfacing with a single computer using a custom designed digital signal processing software to allow easy, individual control of each channel.

In addition to creating a fully functional 40.1 channel sound system, the project also seeks to control each channel in such a way that the sound appears to be coming from a specific point in 3-dimensional space. While this can be partially achieved on smaller systems, using all 40 channels allows for a much higher degree of sound stage manipulation, allowing for a much

more immersive and convincing experience than can currently be created. This functionality can be achieved by modifying the current DSP software to create 3D sound imaging.

Implementing a system capable of producing 3D sound imaging not only creates a different and novel experience of listening to music and audio but also opens new avenues of exploring how sound is perceived in the environment. Sound perception is a multistage process starting with sound entering the outer ear and ending with the brain processing the captured audio signals. The audio processing that occurs in the brain gives rise to different psychoacoustic effects, such as being able to localize a sound's origin. While humans are generally able to perceive a sound's location when only a couple of sound sources are present, their ability to do so is affected when additional sources are added. Since no other audio system of this magnitude currently exists, it presents new avenues to experiment with the human perception of sound.

2. Background

2.1 Existing Multichannel Systems

Commercially available audio systems have grown from units that contain a single channel to the massive 22.2 channel systems that exist today. The earliest attempt at creating a surround sound system was created in the nineteen forties. It was comprised of three separate audio channels broken up into the center, right and left channels. The three channels were played through multiple speakers creating the first surround sound system. This technology was developed for theatrical applications, meant to provide moviegoers with an immersive audio experience.

The development of a more expansive audio system would not take place until many years later. Movie technology continued to advance the audio field leading to the creation of a five channel system in 1976. The five channel system not only contained the three channels already widely used in audio devices, but also included two additional channels, known as rear right and rear left. A little less than ten years later another development occurred when a sixth channel was added which was designated to handle low frequency sounds or bass. This would mark the creation of the 5.1 channel system which still remains popular.

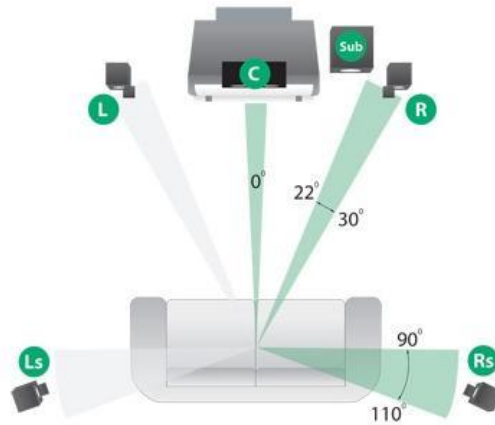


Figure 1: 5.1 Channel System

Since the development of the 5.1 channel system, many more advanced systems have been created. Standard audio systems will contain 1-11 channels, intended for higher frequency playback. In addition to the high frequency channels, it is common to have either 1 or 2 additional channels devoted to low frequency effects. These systems complement the earlier 5.1 channel system, filling in the gaps between audio sources with the additional audio channels. Larger systems, such as the 22.2 channel surround sound system, are present but not as common as smaller ones. This system follows the same paradigm held by the standard system speaker placement but adds a dimension of height. The speakers in this system are organized into three layers, with the middle layer comprising of ten channels, the top layer seven and the bottom layer five, including the two low frequency channels.

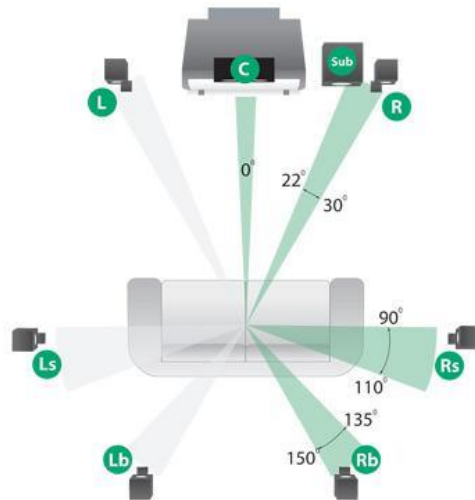


Figure 2: 7.1 Channel System

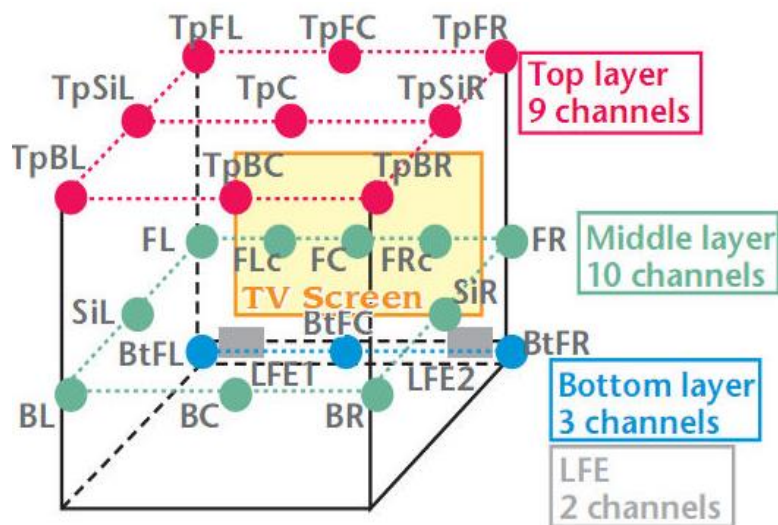


Figure 3: 22.2 Channel System

Even though commercially available audio systems are limited in how many discrete audio channels they provide, larger audio systems have been created for the purpose of providing concert goers a unique experience. Notable examples of such systems include the Poème électronique. This eight minute electronic music performance made its debut in 1958 as part of the Brussels World Fair. The performance included the use of 450 channels of audio, 350 of which could be used simultaneously. Each speaker could be controlled by an array of rotary

phones, making the speakers controls as much a part of the performance as the music being played through them.

2.2 Human Sound Perception

The human ear has a remarkable ability to filter and analyze sound. The ear is such a sophisticated sensory instrument that it can localize a sounds origin and perform complex analysis of the sounds being heard. Much of this processing occurs directly in the ear, so that the final signal sent to the brain is not simply an analog signal, but instead a complex binary protocol. This signal not only transmits the sound itself, but also encodes localization and other data about the sound (Alton 43).

One capability of the ear is the ability to gauge the size of the sound environment. Technically known as the “Precedence Effect” it is most simply explained as “echo.” The human ear goes far beyond the basic concept of echo. The process for determining the size of a sound space is affected by both the time delay, and the intensity of the reflected sound. When the time delay is between 5 and 35 milliseconds, the reflected sound must be about 10 dB greater than the original sound to be heard as an echo. This is due to the fact that the sound is reflected from many different locations and instead of processing each echo as a separate sound parcel, the ear combines and adds the sounds, increasing the sensation of a small space. When the delay is greater than 50 msec the echoes are perceived as individual echoes, with a separate and distinct origin (Alton 75). The following figure demonstrates this property visually.

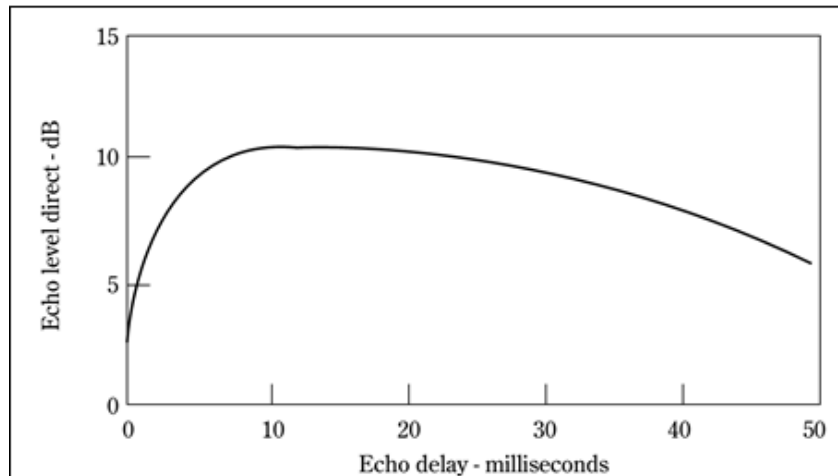


Figure 4: Haas Effect

The functions of the ear most immediately valuable in this research pertain to localization. Localization is the ability to determine information about the source of a sound such as its spatial relation, to and the distance from, the observer. The ability of human ears to distinguish relative position of a sound's origin stems mostly from the presence of two ears. By comparing the sounds received in each ear it is possible to localize a sound horizontally, left to right. This method takes advantage of the of the fact that the human head provides, in essence, an “auditory shadow” for the ear farthest from the sound's source (Alton 68). The effect is that one ear hears the sound louder that the other, and by comparing the relative levels of the two sounds it can be determined where the sound has come from in the left-to-right axis.

Problems do exist with this method of localization, as lower frequency sounds are able to penetrate much farther than higher frequency sounds, thus rendering this method mostly useless for sounds below the 1000 Hz range. Below 1000 Hz the human ear struggles to distinguish the differences in loudness of the sound from one ear to another. This calls for another tactic entirely, and coincidentally, the width of the human head just so happens to be roughly the

wavelength of a 1000 Hz sound wave. This anatomical property of the human head means that below 1000 Hz the ears are able to employ a phase shifting algorithm where one ear can compare the location of the wave with the other ear. By knowing the relative wave position as well as the frequency, it is possible for the ear to determine the left-to-right location of the sound as a function of the relative positions of the wave heard at each ear. Through seamless integration of these systems, the human ears are capable of determining the origin of a sound in the left-to-right axis (Alton 68).

To determine the relative location of a sound's origin in the vertical axis as well as front to back, the ears employ a completely different tactic. The human ear is shaped in such a way that it generates a constantly changing transform function on the sound, collected as the sound is played from different locations in the vertical axis (Mehrgardt and Mellert 1967). This transform function allows the processing mechanisms in the human ear to look for patterns in the incoming sound. These patterns, appearing as a transform function on the spectrum of sound received, correspond to a location where the sound is received from. This information is parsed out by the analysis mechanisms in the ear. The following figures demonstrate the behavior of this transfer function as a sound source is rotated vertically around a person from the front and centered location (Mehrgardt and Mellert 1973).

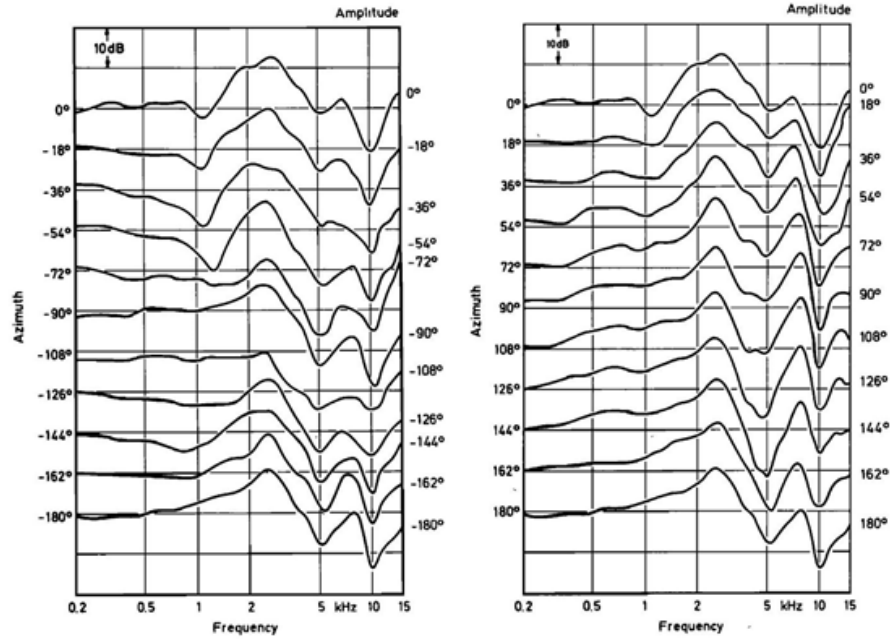


Figure 5: Amplitude of Transfer Functions

Having gained a working knowledge of these mechanisms, it is possible to use the advanced capabilities of this multi-channel sound system to enhance the capabilities of much less complex systems, artificially altering the sound played to simulate varying sound environments.

2.3 Determining Audio Quality

Determining high quality audio is not always a straightforward process. Since the sound is perceived rather than being measured, audio quality still remains largely subjective. While equipment, and measuring techniques do exist to aid in this endeavor, people perceive sound in such a complicated way that using data alone to determine audio quality often falls short. Even though audio quality is still fairly subjective, some major parameters have been defined that can intuitively provide good insight in determining sound quality.

One of the most basic and well known parameter used to determine high quality audio is referred to as noise. Noise is most easily heard when a sound system is turned up while not playing any audio, and is often described as a hissing sound. This parameter is generally expressed as a signal to noise ratio, where a ratio of 100 decibels would describe a system where the audio level is 100 decibels higher than the level of noise produced. Every component in an audio system adds some degree of noise, although higher end systems will seek to reduce the amount of noise produced by using superior components and topologies.

Frequency response is the range of frequencies that can be produced by an audio system. It also describes how uniformly frequencies are reproduced, from the lower frequencies to the higher ones. Most systems range from around 20Hz to just below 20kHz as this coincides with the range of human hearing. An audio system with good frequency response will be able to preserve the loudness level between various frequencies, while not over emphasizing any, being as “true” to the original source as possible.

Distortion occurs when a new frequency, not present in the original audio source, is produced. Two main types of distortion exist, harmonic and intermodulation. Harmonic distortion adds new frequencies that are musically related to the original source. Since they produce related frequencies musicians, such as electric guitar players, sometimes use them to gain a more dynamic sound. Intermodulation distortion also adds new frequencies; however they are not related to the audio source and thus sound more unpleasant. Just like noise, all components add a small amount of distortion. While some distortion is always present, audio designers strive to reduce distortion to such levels that it cannot be heard.

Sound imperfections are not always the result of the audio equipment used, but often a result of the room parameters. Glass windows, hardwood or tiled floors, and sound reflective walls all diminish audio quality through two separate phenomena. Standing waves occur when there are parallel reflecting surfaces present. These waves distort bass and midrange frequencies by canceling out some frequencies while making others more powerful. This creates pockets of sound in the environment in which low frequencies can either be under or overrepresented. Flutter echo is caused by the same scenario as standing waves, but only affects higher range sounds. Higher frequency sound waves reflect off of surfaces in a similar manner to light and add harshness to frequencies above 500 Hz. Both of these phenomena can be fairly easily reduced by adding sound absorbing panels to the audio environment.

3 Methodology

3.1 Project Goals

The goal of this project was to improve the existing multi-channel surround system by building a versatile and robust foundation for an interactive sound room, capable of being used across the WPI engineering disciplines. In acclimating ourselves to the existing research on multi-channel audio systems, and interfaces, the project will further explore areas in which this system could be improved upon. The existing room will be surveyed for potentially useful equipment, and improvements will be made when necessary. An interface for the audio system will be developed that is containing advanced matrix abilities, intuitive usability, and appropriate physical signal path peripherals including, audio decoding, wiring, signal distribution, amplification and speaker placement/mounting. Through use of extensive testing and documentation, we will be able to help future users and project teams move forward, without repeating stumbling blocks that we encounter along the way. Once the system is functional, we would like to test the psychoacoustic effects on our perception of sound, through manipulation of the interface controls.

Our project objectives are as follows:

1. Assess existing equipment, and research functionality and expandability of the system.
2. Develop an interface for the computer that will allow for manipulation of the sound stage, through point-source control of a dynamic output matrix, with at least 40 discrete channels.
3. Select appropriate amplification units to amplify each channel individually.
4. Define shortfalls of existing equipment and how to better address any vulnerability
5. Provide evidence for the potential use of the system in testing psychoacoustic effects.

6. Keep the cost of the project within the allocated budget, between one and two thousand dollars.

3.2 Project Timeline

To successfully achieve the projects goals, each individual goal was divided into several subtasks. These tasks ranged from researching and assembling audio equipment, to organizing, and writing segments of the project report. This subdivision allowed for the goals to be understood in greater depth, while also making it easier to come up with a strategy for accomplishing each one. It became clear that the most effective way to achieve the project goals was to assign each member to a set of individual tasks. In addition to these assignments a timeframe was established for each task in order to keep the group on track throughout the duration of the project. The division and timeframe of these tasks have been visually represented in a task chart, as seen below.

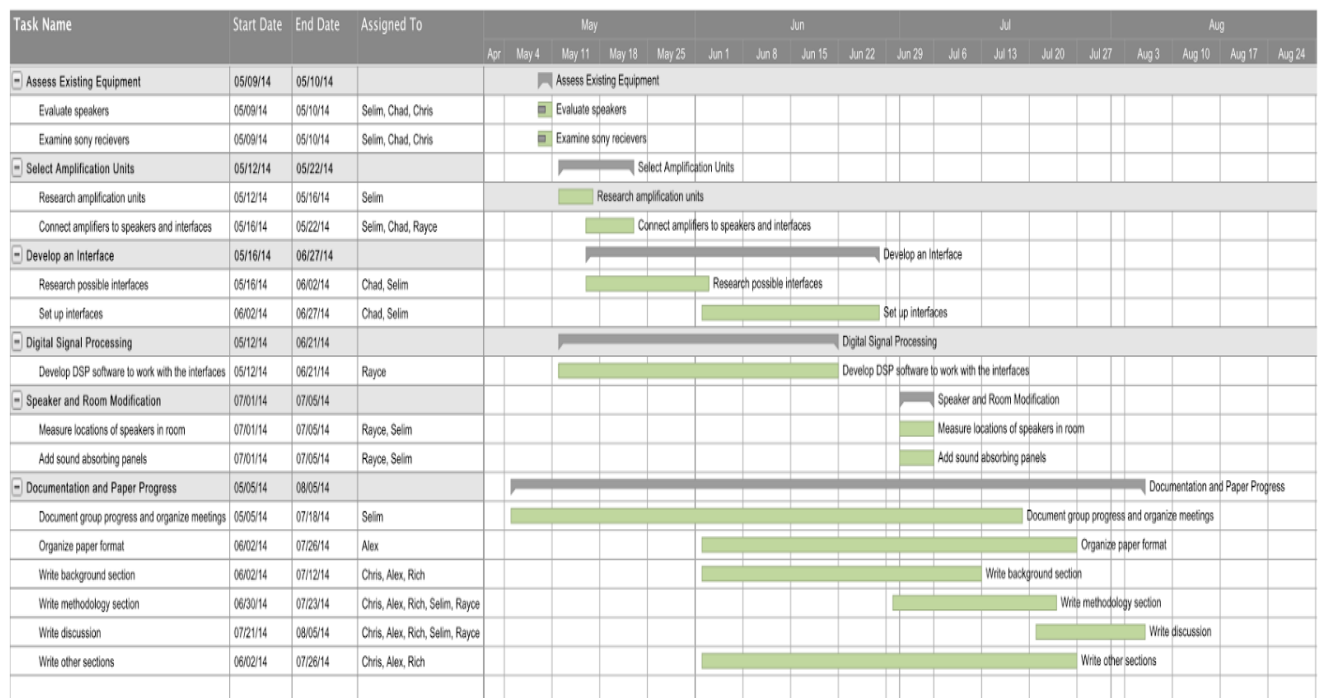


Figure 6: Project Timeline

3.3 Initial Inspection of Current System

Our initial plans for implementing this audio system involved using the hardware currently present to us. At the projects beginning, forty speakers were present in the lab, which were connected to eight 5.1 sony surround sound receivers. The receivers took input from either a coaxial, or optical SPDIF input, which was processed using Dolby DTS encoding to create 5.1 surround sound. It was reasoned that a USB to SPDIF adaptor could be used to communicate between the software and the receivers, however the encoding used by the receivers proved to be problematic.



Figure 7: 5.1 Sony Surround Sound Receivers

While the receivers worked to deliver sound to all the speakers (Sony Corporation), they did not allow for the use of 40 discrete channels in real time. The Dolby DTS encodes six channels, 5 main channels and a subwoofer, into a two channel encoding, which is then sent through the SPDIF connections. Using this encoding system, it would only be possible to have a large scale 5.1 channel system in real time. Previous ideas for the system planned to use a DVD encoding to stream the Dolby DTS to the receivers. This would require burning 8 sets of DVDs to be burned ahead of time. Using separate DVD readers, the time alignment would be off, and

the system would need 4 people present to start the playback. All play buttons would need to be pressed at the same time using 2 hands per person. Furthermore the encoding system used is proprietary and requires licensing by Dolby to be used on the receivers. Dolby offers the Dolby Multichannel Digital Encoder DP569, which costs \$5,400 (Dolby Digital), eliminating the opportunity to encode the signal ourselves. Fortunately the speakers that came with these systems, currently present in the lab, could be driven by other means.

After deciding that the receivers would not work for the purposes of this project, the speakers were examined in greater depth. The speaker selection is comprised of twenty-eight SS-MSP75 (Sony Corporation) front/surround full-range speakers, seven SS-CNP75 (Sony Corporation) center channel, four SS-MSP2 (Sony Corporation front/surround, and one SS-CNP2 (Sony Corporation) center for a total of forty full range 100 watt speakers. Five of these speakers come from a different set than the other thirty-five since one of the receivers it came with was different than the other seven. Despite being from a different system, these five speakers only vary slightly from the others and, given that they fall within the same parameters, should not cause any appreciable difference in tonal quality. It was also noted that while the center speakers had a different shape than the full range speakers, the same drivers are used in each, effectively performing the same as the matching full range speakers.

3.4 Interfacing and Amplification

Given that the receivers we were initially presented with for this project, were incapable of fulfilling the projects goals, a new solution was needed to interface between the speakers and the software. This process would be completed in two stages. The first stage is an interface stage. This stage would be responsible for using hardware that could communicate with the

software to successfully output 40 discrete audio channels. These outputs would lead into the second stage, the amplification stage. In order to have the 40 signals from the interfaces be capable of producing audio on the speakers, each individual channel would need to be amplified separately. This called for a amplification unit that has either 40 separate amplification channels, or a conjunction of smaller units with less channels, but could be used simultaneously so 40 channels could be achieved.

3.4.1 Interfacing

The interfaces chosen for this project would have to be able to connect to a computer, communicate with custom software, and output 40-channels of audio determined by the software. Many options for the interfaces were available including the M-Audio Profire 610, Rednet 3 system, Dante Ethernet based interfaces, and USB Digital to Analog Converters. Each system had a wide range in quality and cost, and this was used to determine the most appropriate interface system.

The M-Audio Profire 610 (InMusic Brands Inc.) interface was the first system experimented and researched in depth. The Profire contains 8 individual analog outputs and utilizes firewire connectivity. The system also supports the ability to daisy chain devices using the firewire connections which, was reasoned, could be used to connect multiple Profire interfaces together. Four Profire systems were available on site, allowing testing of its daisy chaining abilities to be quickly performed.



Figure 8: M-Audio Profire 610 Interface

Difficulties arose in the attempt to use Profire interfaces as was initially intended. The systems proved to be difficult to connect to the Macintosh computers as a connection to the device could only be established during the computers boot time. A connection was eventually established allowing for 8 audio channels to be successfully outputted to the amplifiers; however, other problems arose when trying to connect two systems at once using a daisy chaining arrangement. Upon further inspection of the Profire 610 user manuals, it was concluded that only one device could be connected to a computer at a time using the firewire connections. Alternative uses for this interface were considered, including using a Thunderbolt to firewire adapter (Apple Inc). These adapters have a high bandwidth and connect directly to the computer's PCI bus. This led to realization that the Profire 610 interface cannot be daisy chained because it contains no onboard memory, instead using the connected computers hardware for interfacing.

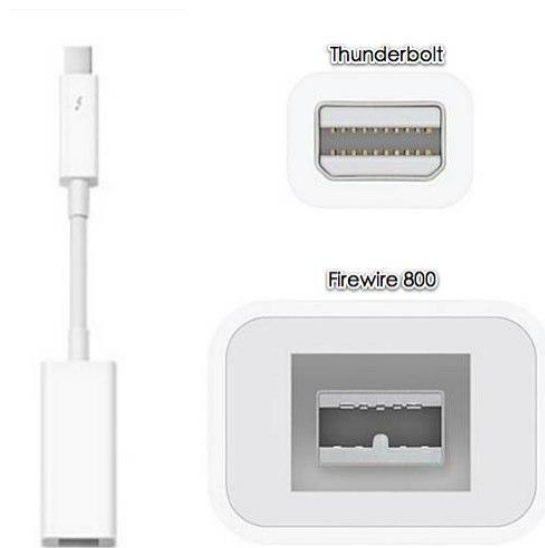


Figure 9: Thunderbolt to Firewire Adaptor

A similar device, the M-Audio Profire 2626 (InMusic Brands Inc), was examined as a potential interface. This interface, in addition to the eight analog outputs, contained SPDIF/ADAT outputs on the back which could be used to achieve a maximum of twenty-six channels on one unit. Unfortunately this interface contained the same problem the other Profire model had, being that it could only connect one at a time to the computer.



Figure 10: M-Audio Profire 2626 Interface

The Rednet 3 system (FocusRite) has support for up to 32 audio channels by utilizing all its AES/EBU, ADAT Optical and S/PDIF outputs. Further inspection of the system showed that the ADAT/SPDIF outputs were only capable of true multi-channel audio through the use of

Dolby DTS encoding which, as previously described in relation to the Sony receivers, would not be possible for the purposes of this project. Although the other outputs could be used separately, without the encoding, multiple Rednet units would be required to achieve the number of needed channels. Given the budget constraints associated with needing two units, the Rednet 3 was determined to be a poor fit for this project.



Figure 11: Rednet 3 Interface

Dante Ethernet based interfaces are very high end audio systems for a widely transformable audio interface. Uses of these interfaces range from recording to matrix switching, and are capable of nearly any audio needs. Ethernet communication allow for devices to be separated by great distances while still maintaining very high bandwidth and low latency communications. It is also possible to control this units from anywhere along the Ethernet cable routing, and can plug into a proprietary PCI card. In order to use this interface to create the forty channel system, multiple units would be necessary as well as a control system. Although this interface, from a technical perspective, would provide the highest quality audio and be capable of delivering forty discrete channels, the cost of a single unit exceeds two-thousand dollars and would not work under this project's budget.

Previously researched interfaces communicate with the computer using, firewire, SPDIF/ADAT and Ethernet connectivity among other connections. However one commonly used bus had not been utilized or researched, the USB. Being that the USB is the simplest and most common serial bus available it showed great potential to be used in this project. Low cost USB Digital to Analog converters were found online that output 16-bit analog stereo through 3.5 millimeter stereo jacks. Each USB DAC requires 12 Mbps of bandwidth. According to USB 2.0 protocol up to 480 Mbps can be supported, theoretically allowing 127 devices to be connected simultaneously. Using this information it was hypothesized that twenty of these devices could be connected to a single USB host, enabling forty channels of audio to be outputted.

Although the USB specification states that up to 127 devices can be connected to one master root hub, the reality is that the system will run out of bandwidth far before the max device limit is reached. On any given computer system, there are many internal USB connections, which are used for devices such as Wi-Fi, webcams, brightness controllers, keyboards and track pads among others. The iMac being used in the lab contains several internal USB dependent devices, thus limiting the availability of potential external slave devices. Specifications from apple showed that four devices could be connected through each of the USB ports on the iMac.



Figure 12: Syba USB Audio Interface

While the number of slave devices allowed by the iMac were limited, it was possible to continue to use the USB ports to gain additional output channels. The M-AUDIO Fast Track C400 USB interface contains multiple outputs, and could be used alongside the Syba USB interfaces. The Fast Track interface contains one USB B input, and 6 outputs, four of which were 1/4" mono TRS outputs. These TRS outputs offered fairly easy implementation in gaining additional audio channels. The interface was capable of outputting four audio channels at 26 bits with the potential for a fifth channel if the SPDIF output was enabled.



Figure 13: Fast Track C400 USB Interface

The Syba USB audio interface is a very simple Digital to Analog Converter (DAC) chip that talks to the master device with serial. The computer sends high bandwidth signals along the USB device, and that digital information is output as an analog signal at 16-bits of resolution. The voltage range of a DAC is described by the Full Scale Range (FSR). And the exact resolution per bit of this device is as follows.

$$Resolution = \frac{FSR}{2^{16}}$$

The other systems in this design, such as the Profire 610 and and Fast Track C400 have 24 bit DACs inside them. This offers a greater resolution, one which has 256 times more resolution for the same Full Scale Range.

The Profire 610, Fast Track C400 and the Syba audio interfaces were compiled into a large aggregate device. The iMac's 3.5mm stereo headphone jack was also used to interface to the amplifiers, as it is built into the computer's hardware. This allowed for experimentation with the total bandwidth capabilities of Ableton Live, the iMac and the data buses for the interfaces. We are able to get 16 channels of amplification, with 20 possible analog outs using coaxial SPDIF output devices from the Profire 610 and Fast Track C400. The 16 channels were a good proof of concept in terms of the interfaces, and demonstration of the system. Given a more expanded budget we would be confident that the system would work well beyond 40 channels of audio.

3.4.2 Amplification

Amplifications options were explored extensively to find the most appropriate way to amplify the forty signals from the interface stage. Common amplification systems, such as car

audio systems, were explored first with the intent of keeping costs down. These systems require a twelve volt power supply, generally from an automotive battery, and have four outputs, one for each of the car doors. Although relatively inexpensive compared to other audio amplifiers, the lack of an internal power supply would make them difficult to implement. No single common twelve volt supply could be used to power the necessary number of units, meaning that each unit would need its own power supply. Given the added costs of the power supplies and the risk of power supply noise interference, new options were explored.

To mitigate the risks associated with power supply noise, and possible faults within each supply, more isolated units, with built in power supplies, were researched. The Niles 12-channel amplifier and the Pyle 8-channel amplifier were two of the top products found for the purpose of this project. Each system had an internal power supply and had discrete audio inputs for each of its channels. While the Niles amplifier would make for a more compact system, the price was determined to be excessive and the Pyle system was selected instead.

The Pyle 8-channel system is capable of delivering up 1000 watts to each of its individual channels. While the systems power is somewhat excessive for the 100 watt speakers, it fell within budget constraints and allows for further expansion of the system in future projects. The system receives 120VAC, which can easily be connected to a standard wall outlet, several of which are present in the lab. The Pyle contained RCA inputs and screw terminal speaker outputs. Since the speaker wire was already in conduit along the walls of the lab, it was hooked up to the terminal connectors.



Figure 14: Pyle 8-Channel Amplifier (Back)



Figure 15: Pyle 8-Channel Amplifier (Front)

3.5 Digital Signal Processing

The Digital Signal Processing for the project was performed in a Java environment in MAX/MSP. This specific environment was chosen as it is well suited for signal processing and also easily integrates with most Digital Audio Workstations, namely Ableton Live 9. This integration provided extremely low-latencies, as well as a native user interface that resided in the Digital Audio Workshop, allowing for a streamlined work environment. As the MSP software was designed for the DSP operations required for the project, it was selected for implementation.



Figure 16: Ableton Plugin Interface

The primary governing principle behind the system was to use a combination of delays, attenuators, and all pass filters to affect each individual speaker in the room. By combining these techniques, shaped wavefronts could be constructed through the summation of individual point sources. This method can physically reproduce the wavefronts that occur in other physical spaces, unlike the two traditional point sources found in most stereo systems. These ultimately sound more natural to the listener, as both ears experience the sound in a time and phase-aligned context. As mentioned earlier, the Precedence Effect provides for a more natural localization of sound sources in the stereo field. In addition to wavefront synthesis, the software also possesses

the ability to add room impulse response convolutions to the output signal. This can accurately model reverberations or secondary reflections from reverberant surfaces, even if they do not physically exist in the room. A space larger than the room can be virtualized, enhancing immersion to the listener.

These specific methods are achieved in a multi-step process. Initially, the distances are calculated from the ideal point source to the positions of the speakers in Cartesian space. These distances are then sent to the Digital Signal Processing plugins. As each output channel has a separate processor instance, and each output channel drives a single speaker, every speaker in the array receives different distance compensation. This is achieved through DSP with the implementation of group delays, all-pass filters, and attenuators. Using a combination of relative location and driver orientation information, the software calculates volumes and time-delays for every speaker in the system such that their output constructively sums to generate a wavefront that can be physically perceived to originate from the desired location.

3.6 Speaker Mapping

In any audio system placement and positioning of speakers plays a large role in the systems overall quality of sound playback. The placement of the speakers greatly affects tonal accuracy, staging, and imaging, as well as realistic sound reproduction. In an ideal environment a high quality audio system is specifically designed to eliminate audio nuisances such as reverberations and standing waves. Considering the room in which project's audio system is located has different features that are less than ideal, such as windows, a drop ceiling and sound reflective wall, extra measures were taken to compensate for these interferences.

The most basic measure taken to improve audio quality and hinder sound reverberation was the addition of sound dampening foam panels to the room. The panels work to absorb and deaden sound, helping to improve the amount of sound wave reflections in the room. Standing waves can also be reduced this way. To improve the system's abilities of sound imaging and realistic sound reproduction, the speaker distances were measured in Cartesian coordinates with respect to the center of the room, the positive axis pointing towards the southwest corner of the ceiling. Velcro patches on the back of each speaker were used as references to the center of each speaker.



Figure 17: Speakers with Sound Dampening Foam

The coordinates gathered from the measurements were used by the DSP software to create virtual representation of the environment, so that sound imaging could be more accurately performed. The initial measurements were recorded in inches; however the DSP software requires metric units. An excel document was used to easily convert the coordinates into units used by the software.

3.7 Assembling the System

Now that the interface and amplification stages had been researched thoroughly and a number of appropriate components had been selected, the multichannel sound system could begin to take form. The completed Digital Signal Processing software used multiple ports on the iMac to communicate with the interface stage. The outputs of these interfaces were converted to RCA male leads so they could be connected to the amplifiers. The two amplifiers outputted sixteen channels to the speakers through the speaker cables running throughout the room. A basic overview of this process, along with the cables used to connect the components, is represented below.

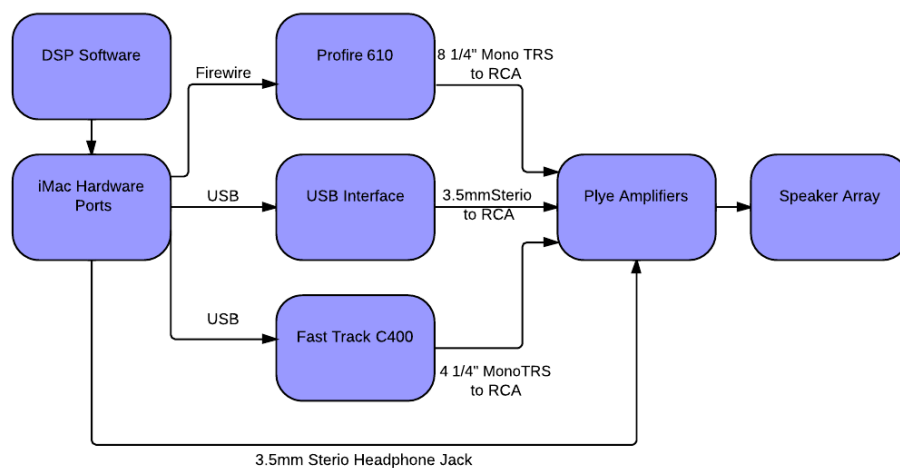


Figure 18: System Block Diagram of Multi-Channel System

The components themselves were assembled inside of a medium sized audio rack. The two 8-channel Pyle amplifiers were located on the bottom of this rack, one on top of the other. Since these two units were the heaviest of all the components, this created a low center of gravity in the rack, making it very stable. Two of the interfaces, the M-Audio Profire 610 and the Fast Track C400 USB interface were located on the top shelf of the audio rack. By placing these

units at this distance from the amplifiers, heat could be easily dissipated and wires could be run from components neatly without them bunching up.



Figure 19: Profire 610 (top left) and Fast Track C400 Interfaces(top right) and Pyle Amplifiers in Audio Rack (bottom)

The Profire 610 interface was connected to the iMac through a USB to firewire cable. The interface has 8 mono tip ring sleeve outputs which were connected to the amplifier by TRS to RCA conversion cables. TRS to RCA cables were also used to connect the Fast Track C400 interface's outputs to the amplifier, while its inputs were connected to the computer using USB A to USB B cable. The Syba USB interface plugged directly into the computer so there was no need for an extra cable. This unit was connected to the amplifier through a 3.5mm to RCA cable.



Figure 20: Connections to iMac including (from the left) 3.5mm headphone jack, USB to Profire 610, Syba USB, and USB/Firewire to Fast Track

The Pyle 8-channel amplifiers were connected to the speakers, located around the room, using the speaker wire that was present at the projects beginning. All sixteen channels of the amplifiers were utilized allowing for sixteen speakers to be used. The speakers in use were selected based on their location in the room so that the fullest sound possible could be achieved. The interfaces were controlled by the digital signal processing software, which utilized a custom made two-dimensional slider which determined how much power was distributed among the speakers.

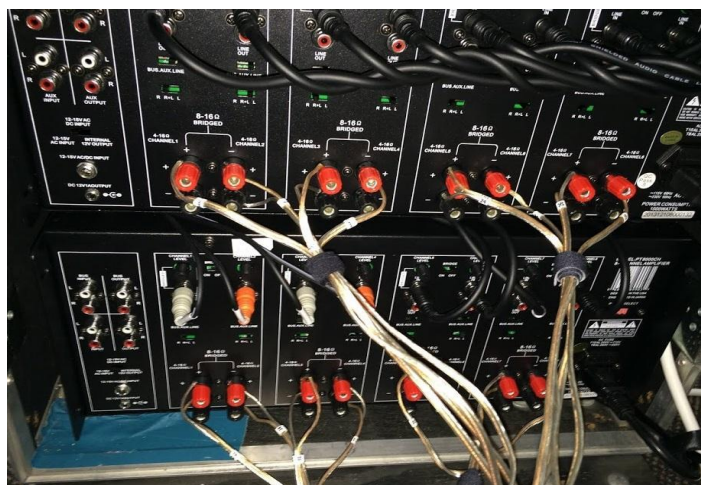


Figure 21: Backside of Pyle Amplifiers Connecting to Speakers

4. Discussion

Setting the level across multiple devices and interfaces can be very difficult when differing devices are used to perform the same task. In the case of this system, the signal path is outputted from the application through 4 different devices, some having physical controls on the devices, some without. These devices also used an assortment of connections to carry the signal from the computer and the devices. After being converted from a digital to analog signal in the devices, the means of output is standardized as a uniform length of analog RCA cable, which is driven into two identical 8-channel amplifiers and output to reasonably identical speakers.

The digital to analog conversion of the headphone jack was performed on the computer's sound board, while the other 14 channels are output digitally. Next, a small USB headphone/microphone device, plugged directly into the computer, converts the signal and sends it to the amps via an analog cable; this device has minimal controls, with only an individual gain control via the computer. An M-Audio C400 is sent the signal over a USB cable, which has manual gain controls on the device, in addition to sample rate, bit depth and gain controls via the computer. Lastly, the M-Audio ProFire 610 is connected to the computer via a firewire cable, then the device outputs analog as the other devices. The ProFire also has physical gain controls, in addition to bit depth, sample rate and further gain control via the computer.

To reduce the possibility of errors, a starting point was set for the sound system and documented all the various settings outside of the Ableton interface, so that a baseline could be repeated by following users. Inside the Ableton interface, there is a single slider for each output channel that is used to adjust the gains of each channel in order to standardize the individual speaker sound levels for the final usage. This method of system signal path, and control allows a

lot of room for accidental errors to occur. Gain alone is easy to change by any one of the individual interfaces or physical knobs. This leads to a system that is incorrectly set to produce a seamless soundstage, instead having hot spots where a particular channel has been increased, or dead spots where its gain has been reduced. The differing digital signal paths can cause the signal to have lags in timing, potentially leading to unintended echoes in the system due to differences in time individual interfaces take to process audio. Fortunately there are some features built into the Apple computer's "Audio Midi Setup" application, wherein the "Aggregate Device" constructed of all our outputs, offers "drift correction" that corrects timing issues caused by the devices that are not able to be connected to the "master clock" of the chosen master device.

Further degradation of the sound stage can occur when each different device adds its own fingerprint on the output sound, when "noise" is introduced in each stage of conversion and signal path. Though this is minimal in this particular system, it is worth noting that in a system with higher precision, and lower sound-to-noise ratio speakers, the differing hiss of individual channels could lead to a notable distraction in low volume testing and experimentation.

Attempts were made to use interfaces that could be daisy chained but, due to the limits of firewire and thunderbolt, it was not possible to daisy chain firewire devices. USB protocol technically does allow many devices to be connected at once, however the bandwidth becomes limiting very quickly. In reality many USB Root Hubs can only support up to four to five devices at a time. The host and any hubs used in the system do count as a device. An iMac was able to support a USB hub, and two USB interfaces at a time with ease, once a third device was added,

all the software halted. This was due to limitations at the root hub level of the iMac not being able to support the three USB interfaces, plus USBH hub, in addition to the root hub itself. The total of five devices reached the limit of the iMac and all software became unresponsive.

The shape and structures of the room proved to have a noticeable effect on the audio being played. The most notable way the environment of the room changed the audio was the presence of flutter echo when playing higher frequencies. The echo was reduced greatly with the addition of the foam paneling, however due to the rooms rectangular shape, could not be reduced completely. Other room features such as the radiator and the glass windows would vibrate when listening to lower frequencies at higher volumes.

Once the system was completely assembled, it was capable of driving sixteen speakers. The computer interface made it fairly easy to control which speakers were being driven at any time. Using this interface it was possible to test how well the system is able to replicate sound originating from a single point. To demonstrate the systems overall ability to localize a sound, the interface was used to make a sound “travel” around the room. The transitions of the traveling sound were not easily noticeable as it moved from speaker to speaker and it was fairly obvious which speaker the sound was coming from. Sound was then played from one location in the room to see how well the system can play audio that was clearly identifiable as coming from one source. Determining the sounds origin also proved to be relatively easy, however, due to reverberations in the room, the sound appeared to becoming from the other sources although very faintly.

5. Recommendation

5.1 Potential System Uses

The primary value of this system lies in its potential for sound localization research. Because this system has the ability to create both point sources of sounds and a general region of sounds origin, it would be possible to test the sound processing abilities of the human ear.

One frequently studied, yet poorly understood ability of the human sense of sound is speech recognition. It is known is that the human ear is the best speech recognition system (Fastl and Zwicker 361). Beyond this, the exact mechanics of how humans are so adept at being able to filter out speech, and process it, are not well documented. Tests performed with this system could be used to gain further insight into the mechanisms of speech recognition.

Tests similar to those used to analyze different types of cochlear implants could be adapted to test human speech recognition in new ways (Fastl and Zwicker 354-356). These tests all contain at least two auditory elements, background noise, and one or more instances of human speech. In the tests mentioned above, both CCITT and Fastl types of noise were used for the background elements and sentences were used as the speech component, for ease of testing (Fastl and Zwicker 355). In tests performed on this system, similar components would likely be used. An interesting example would be to modify the location of the origin of the speech around the test subject to see how recognition ability varies based on the origin of the speech.

This system also has the ability to vary the size of a sound source anywhere between a point source and a fully encompassing sound. Varying both the relative size and relative origin of the speech and noise opens up the possibility for conducting a number of very interesting

tests. For instance, if both the noise and speech are made to be very small how does varying the distance between their origins impact the ability of the subject to distinguish the speech? In the natural world, speech is almost always a point source sound. Whether or not the point source nature of speech is important for speech recognition could be studied by varying the size of the speech origin. If speech emanating from many directions is harder to decipher, it would point to the idea that the localization of speech to a single point is important for effective speech recognition. Further tests could be performed with multiple speech sources to quantify the importance of localization in speech recognition. A potential application of this research could be in “turn-by-turn” navigation systems. Currently, most systems simply play the navigation instructions through the car’s speakers indiscriminately. It is possible that by altering the origin of the instructions, they could be made much clearer for the driver. In addition it could be studied how making a sounds origin, relevant to the instructions being given (a direction to turn left could come from the left side of the driver), could impact ease of comprehension.

The most immediate use of the system is creating a three dimensional sound space for use in a virtual reality system. Beyond that the 40 channel system can be used for research in understanding how the ear processes sound, and how audio systems with far fewer channels can simulate much larger systems. The human auditory system uses, in essence, a stereo microphone to gather all sound information. Because of this fact, all other information determined about the sound must be obtained by encoding that information into the received sound. This encoding is performed by the outer ear and ear canal. The shape of these structures alters the sound wavefront in a regularly varying fashion based on properties of the sound such as location of origin and amplification of certain frequencies. This information can be used to enable a stereo

system to be capable of the same spatial dynamics as a large many channel surround sound system. The fact that the human ear is only a stereo system makes it possible to simulate any sound environment with only two channels, the only requirement is to understand how the other components of the ear interact with the incoming sound. To perform these experiments it would require either microphones to be embedded in the ears of a human subject, or to create an anatomically correct model of a human head and ears (Mehrgardt and Mellert 1967).

This research on the mechanical performance of the human ear can be extended to enhance the performance of various hearing augmentation devices. Nearly 70,000 people in the United States, and 219,000 worldwide rely on cochlear implants for hearing. (American Speech-Language-Hearing Association) Currently these devices are capable of very basic integration with the human auditory system. Nearly all of the additional information encoded into the sound, especially localization information, is lost. By comparing the performance of the human ear in tests, such as those suggested above, with a cochlear implant or hearing aid it would be possible to enhance the function of these devices significantly. Not only could the sound processing algorithms of these devices be improved, but other factors, such as the placement of the microphone could be improved to provide the wearer with the best possible restoration of their hearing functions.

Another interesting psychoacoustic property which could be examined are binaural beats. The phenomenon of binaural beats is described as the “fluctuation that is perceived when two relatively low frequency tones differing slightly in frequency are presented independently to the two ears.” (Grose, Buss, and Hall 40) Currently all understanding of binaural beats depends on the differing frequencies being played between the two ears. The many channel nature of this system could allow testing of this phenomena in other axes. The differing frequencies could be

varied from a left to right alignment to a front to back alignment to observe the orientations where this phenomena is able to occur. The angle between the two frequencies could also be varied to study the minimum angle between the sounds for the beat frequency to be heard. Because all previous studies involving binaural beats have been solely based on stereo systems, it could also be researched if beat frequencies produced by differing frequencies played from more than two sources are also possible.

5.2 System Improvements

Although the multi-channel audio system was successful in creating a large scale audio system; improvements could be made to increase sound quality and, most importantly, the number of channels. An increase of discrete channels could be achieved through utilization of either the Dante Ethernet or Rednet 3 interfaces. These professional systems can process sound and distribute it to many different locations simultaneously. Furthermore a single computer can be used with several PCI ethernet controllers, allowing for every aspect of the system to be controlled through the control station. Using only ethernet based connections not only allows for higher bandwidth, but also eliminates the need to use additional ports. Since the ports used for this project have different latencies, this would improve latency, and improve sound quality significantly. While these units exceeded the projects budget, future adoption would allow for higher bandwidth connections, greater user control, and an increased number of usable channels.

To mitigate problems associated with peripherals not being detected by the software, a proprietary communication protocol could be adopted. Such protocol would replace the current drivers for the various ports with custom designed drivers, written specifically for

communication between the software and the interfaces. An implementation of this protocol would reduce the reduce connection based issues, making the system easier to use.

Other improvements to the system could be achieved by modifying, or changing the systems location. Ideally, the best environment for this audio system would be a square room. This would simplify the digital signal processing as there would be little difference between speakers on the front and back walls or those on the left and right walls. It would also reduce negative harmonics created in the room. Having the speakers be evenly spaced would simplify the math of the wave mechanics, further decreasing the load on the digital signal processing.

6. Conclusions

The goal presented for this project was to create a 40 channel surround sound system that would provide a platform for unique musical creations and psychoacoustic experimentation. The system created is capable of producing 16 sixteen discrete audio channels, with a potential for expansion to 20 channels, while still using the current hardware. A custom MAX digital signal processing software provides precision controls of each channel, allowing for system to be easily developed for. While the number of usable channels was less than the desired amount, methods were presented to increase the available audio channels.

Using the Rednet 3 or Dante Ethernet it would be possible to meet the desired number of channels, all in a more compact system, using a smaller number of hardware ports. These interfaces would allow audio development to be performed with greater ease, as each channel would not need to be balanced individually. However, an increased budget would be required to use these interfaces as each individual unit exceeds this projects allotted funds.

Potential uses of this multi-channel audio system were suggested. The primary usage for the system will most likely be for developing music for WPI classes. Other uses range from testing of psychoacoustic topics, such as localization, to researching human sound perception. A further understanding of such topics could lead to the development of more advanced audio recording and assisted hearing devices.

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8. Appendices

A. User Manual and Troubleshooting Guide

Introduction:

This manual was created with the intent of guiding users in the startup, settings and usage of the complete surround sound system, Installation details will be provided as well in case a new computer must be used, or the initial settings would like to be confirmed.

System Overview:

The system is configured using a computer interface to control the sound in a manner that allows the user to control the perceived point source of an audio input, with respect to a listener's position. The signal is controlled within the program and output to 16 discrete channels of audio, meaning that each speaker can be controlled independently of each other.

The output of the application is fed, digitally, into an array of devices that can convert the digital signal to an analog output. This analog signal is then fed into each of the amplifier inputs, there the signal is amplified to be audible through the speakers inside the room. There are a number of amplitude controls in the system, both manual knobs, and electronic programming which could cause individual channels to vary in amplitude from the others. Modification of these settings should be avoided unless you are certain that you know what you are doing.

The interface of this system is on the iMac closest to the system control rack. To log into the computer, enter "lab" as the user, and no password. The installed programs, and custom designed interfaces, will be where the manipulation of the audio source occurs. The interface is programmed with the existing 16 speaker coordinates, but the others can be added if this system is expanded in the future.

Speaker locations are measured using the lower speakers on the longer walls as xy-plane, and each speaker is measured from a position at the center of the room using Cartesian coordinates. The length of the room is 4.07 m by 3.13 m wide. All the speakers in the room are labeled on the wall at the speaker location. There are 40 speakers, though we are only using 16 in the current iteration.

Applications and Settings:

All the required interfaces are already installed on the project computer, however the core interface that was designed for this project was created using ‘Appleton Live 9’, and ‘Max 6’. These versions or newer must be used to run the interface as it is designed currently. We will also be using ‘Audio Midi Setup’, allowing us to control the multiple inputs as a single ‘Aggregate Device’.

Opening and checking the application settings:

Start by logging into the computer, using ‘lab’ as the user, there is no password.

Open “Audio Midi Setup”

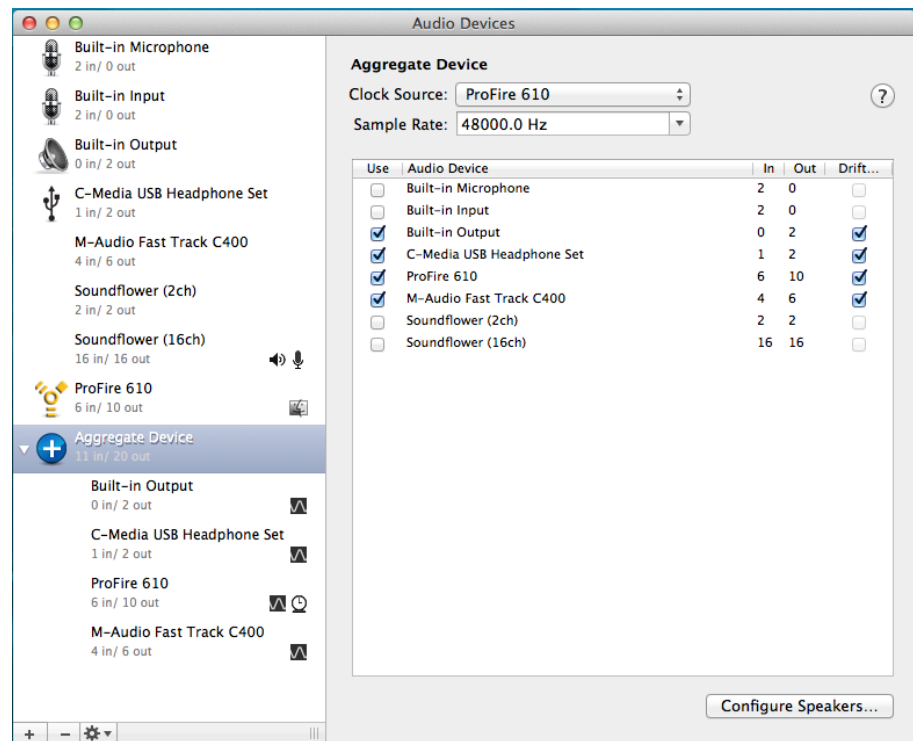


When this application is opened, you will click on the ‘Aggregate Device’ in the left column, expanding it to view the contents. Only the ‘Built-in Output’, ‘C-Media USB Headphone Set’, ‘ProFire 610’ and ‘M-Audio Fast Track C400’ should be checked ‘Use’

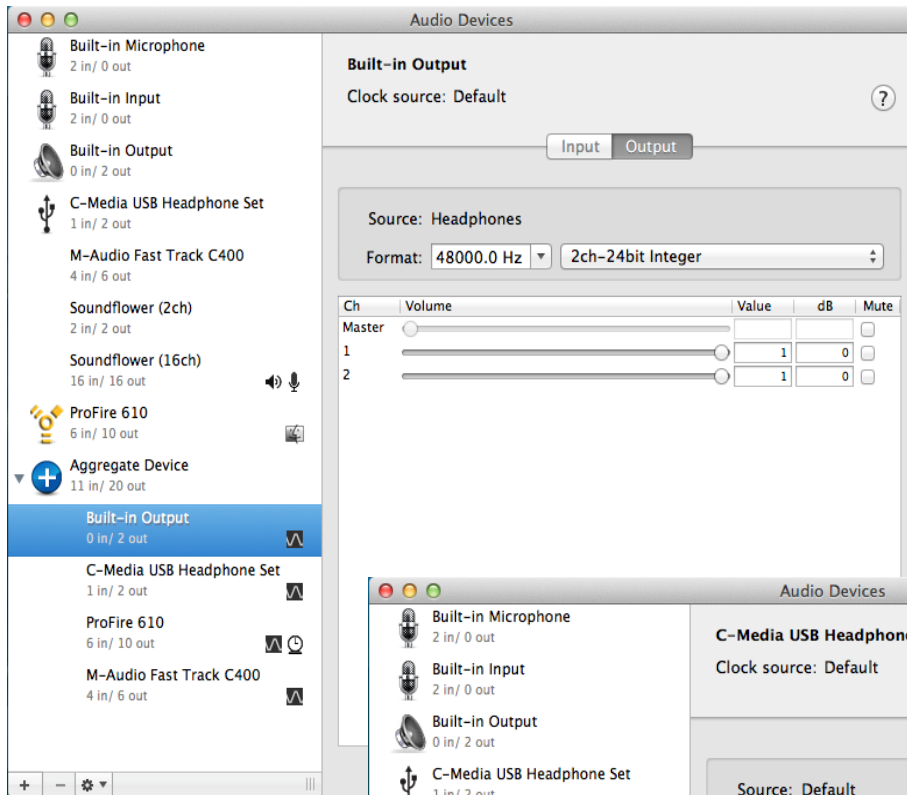
Next, you will see a ‘Drift Correction’ box, and it should be checked for all **used** devices. This will allow for the computer to automatically time-align the different devices within this aggregate device. If there is skipping or choppy audio signal being output, this should be checked.

The ProFire 610 should be the ‘master clock’ source, denoted by a small clock symbol next to it inside the aggregate device.

Sample rates should be 48,000 Hz for all devices, and the bit rate should be 24 bit for all devices except the ‘C-Media USB Headphone Set’ which is not configurable, and is static at 16 bits.

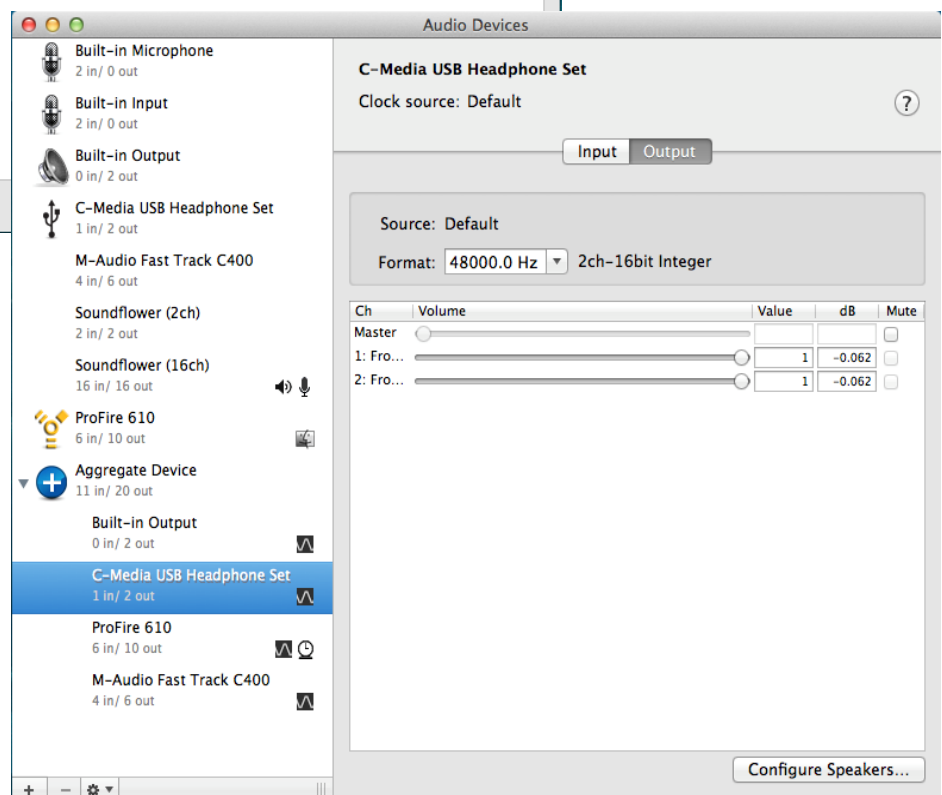


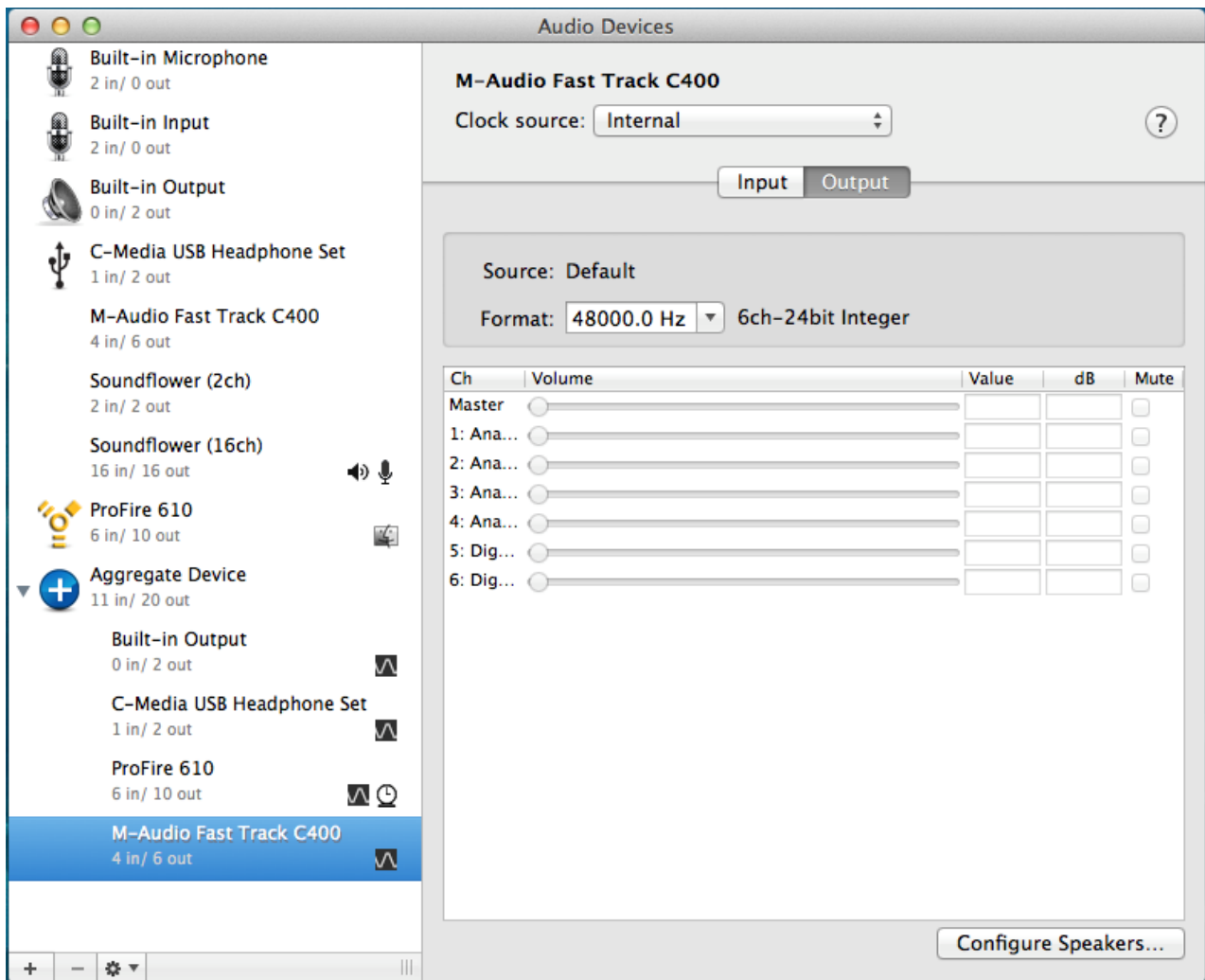
Next, check the gain settings for each of the devices within the Aggregate Device to confirm they match the values the system was tuned to.



The built-in output should have both channels all the way up, at 0 dB gain.

‘The C-Media USB Headphone Set’ slider should be at max, and will display a value of -0.062 dB gain.





The ProFire 610 and C400 gains are locked within the aggregate device, so these sliders should be grayed-out.

Open “Ableton Live 9 Suite”



This is the main interface you will be using, it includes a number of programmed modules that run in the background, as well as a live workspace that can be saved.

You will take your program file, “IQP Processing.als” file and open it within this program. If you click on the Ableton interface to bring it forward, you can also click on the File drop down menu, and ‘Open Recent Set’, which will give you a list of “Live Sets” that can be opened. These are called live sets, because the files can be saved with all the live data manipulation, like a snapshot of the system at that moment. The Processing modules are not affected by this process, which allows the user to modify multiple iterations of a file without affecting the core processing.

It should also be noted that the GUI layout of the screen is saved as well, so you should revert to your primary control screens before saving.

For the purposes of this interface, you will not need to open Max, however the modules in the interface use this program, so you will see it running in the tool bar.

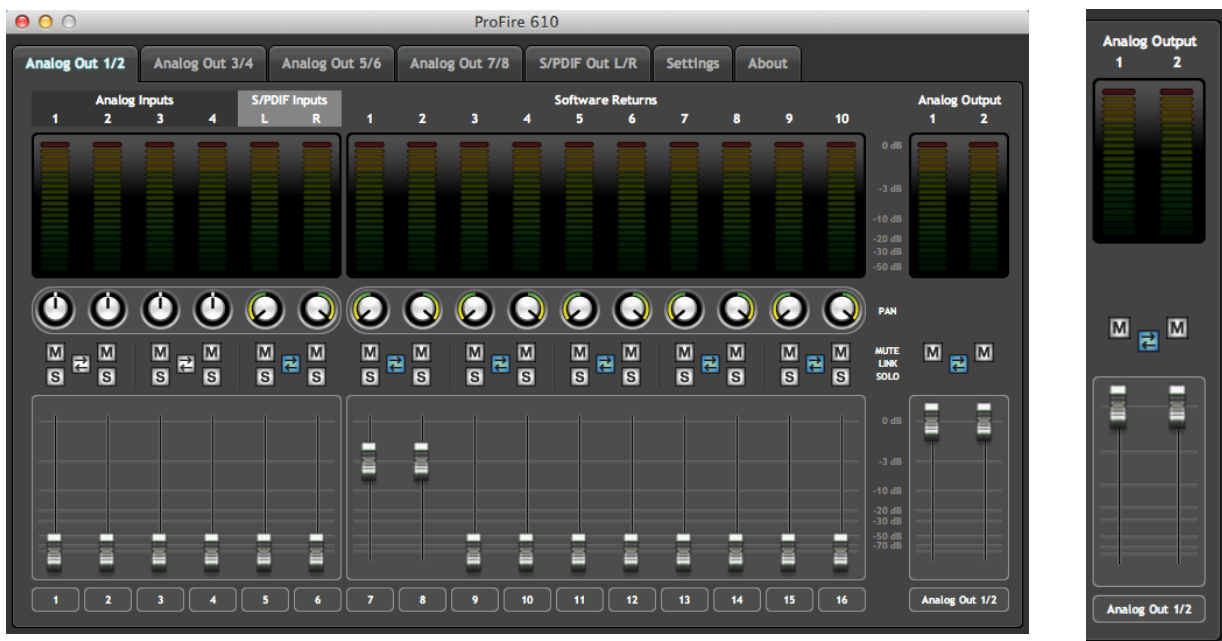
Open “M-Audio_ProFire”



This is the interface that controls the ProFire 610 settings, though the Ableton application takes control of many of these settings automatically. It is good practice to check this occasionally to be sure the 610 is setup correctly to be used in the system.

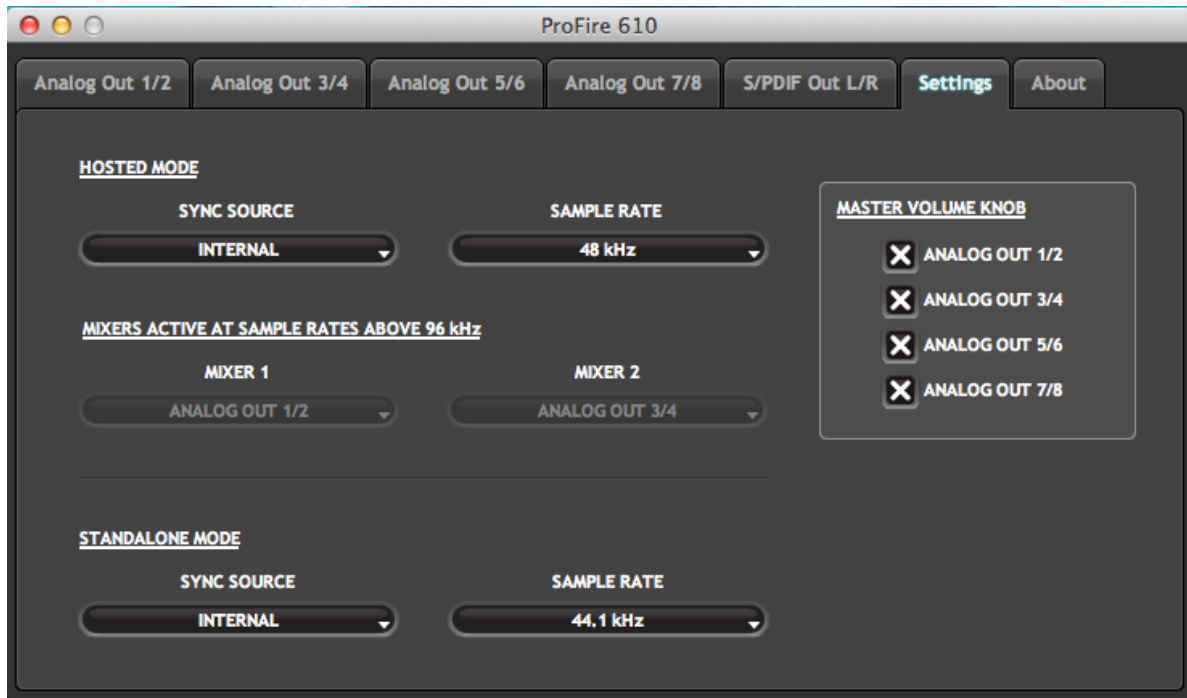
Each pair of analog outputs has its own ‘Software Return’ gain control. These should be on -2.8 dB, or the second gray line from the top of the slider.

Adjust the ‘Analog Out’ slider at the far right to 0dB, for each channel group tab.



Click on the Settings tab, and confirm that the Sync Source is “Internal”. The Sample rate is set within the ‘Aggregate Device’, but should show 48 kHz.

All boxes in the ‘Master Volume Knob’ area should have X-marks in them. If you are having trouble with any particular channel running at full gain, this setting can be checked to potentially resolve the problem.



Open “M-Audio Fast Track C400 Control Panel”

Like the ProFire, there is an interface for this device, and it has a number of settings that are configurable. We will be checking the gain for each of the analog outputs.

Choose the tab for Analog channel pairs, then confirm the slider is at 0 dB. Move only the sliders corresponding to the channels that match the tab you have chosen. The Master slider at the far right should also be set to 0 dB.



Physical Equipment Settings:

Pyle PT8000CH Amplifier:

Turn on power by pressing the button on the front of the amp, inset button position is ON.

On the back of the unit (All settings are same for each of the 8 channel groups):

The gain knob on each channel should be fully clockwise, in the max gain position. The 'Bridge' Switch should be in the 'off' position, and each 'Bus.Aux.Line' switch in right most position.

Each group of channels, Channels 1 and 2 for instance, should have the left input selection switch towards the left, in the 'R' position, and the right switch in the right-most position, the 'L' position.

Speaker wires are connected below, on the output posts.



M-Audio ProFire 610:

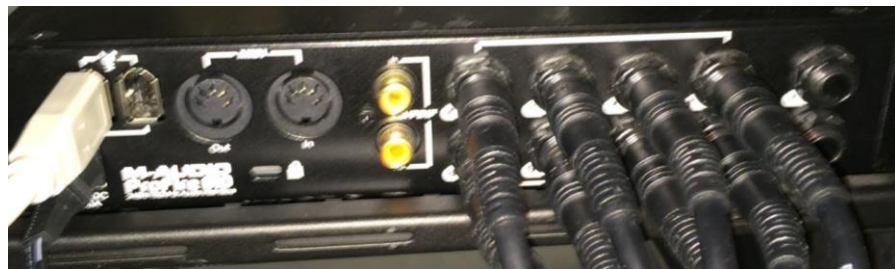
This unit is primarily controlled via the computer interface, but the Power button must be pressed into its innermost position, a blue light should illuminate within 15 seconds, showing it is powered up and ready. The ‘Master Volume’ knob should also be turned to max gain, in the most clockwise position.

On the back:

This unit receives a digital signal from the computer, and decodes it into Analog signals. The Digital input is via a 6-pin, Firewire 400 cable; the other end of the cable connects to the computer via a Firewire 400 to 9-pin Firewire 800 adapter.

The Analog outputs are RCA’s, and are routed down to the amplifier inputs.

This device also receives power from an external power supply. In this case a “wall-wart”, that connects below the Firewire connector.



M-Audio Fast Track C400

Another Digital to Analog signal device, the controls are located on the top/front of the device. Rotate the 'Volume' knob to the max gain, most-clockwise position. Then confirm that both the 'A' and 'B' buttons are lit, confirming all 4 channels will be outputting to the amplifiers. This device receives both the Digital signal, and its power via a USB cable connected to the computer.



Sabrent USB Line Device:

This is a simple device with no physical controls. The device has an input and output port, of which we are only using the output. The output port is signified by the green colored ring around the $\frac{1}{8}$ " (3.5mm) stereo audio port.



Apple Computer:

This is the primary interface device used in the system; it runs a number of programs that allow for the manipulation of the outputs. The photo below shows the cables that are plugged into the back ports of the computer. From left to right;

1. The onboard audio jack is used to output 2 channels, via a 1/8" audio jack.
2. A USB cable connects the M-Audio C400 device.
3. The Sabrent USB Audio device plugs into a USB port, outputs to a 1/8" audio jack.
4. Apple Keyboard(mouse plugs into the keyboard)
5. Firewire 800 Port on the computer, with Firewire 800 to Firewire 400 adapter, and a Firewire 400 cable that connects to the M-Audio ProFire 610.



Speakers:

A collection of 8 home AV surround sound systems were assembled for this project, each set having a receiver and 5 speakers. Of these speakers, 4 were identical, and the fifth was a center channel. The center channel uses the same driver as the others, but has a differing form factor. For our uses, and given the limits of the rest of the system, the acoustic properties of the differing speaker enclosure should be considered negligible.

The speakers are suspended from the ceiling via strings, in addition to being attached to the wall with Velcro. A limitation was placed on this room not being permanently modified, with no holes or other attachments that would cause damage, and any modifications to the room must follow the same rules, unless otherwise specified by a WPI staff member.



All speaker locations are numbered on the wall, or cable track, and a matching label is placed on the wires on the rack end.

A standard has been used, in which the silver colored lead of the wire is attached to the negative terminals (Black), and the copper colored is attached to the positive terminals (Red). If this standard is not followed, a speaker could become wired out of phase with the others and can cause the speakers to cancel each other, and distort the overall sound image.



Using the Interface; Ableton Live 9 Suite

Getting familiar with the interface

The Ableton application is a very powerful tool for audio processing, though the interface you will be using has been created to make the specific processing tasks of this system as simple and intuitive as possible.



Adding and Playing a Source

At the top of the interface, you have the main source control, in which you can drag-and-drop audio sources to be played through the system. A Track can be chosen and played by clicking on the gray triangle to the left of the track, it will illuminate green when the file is actively playing



Main Volume Control

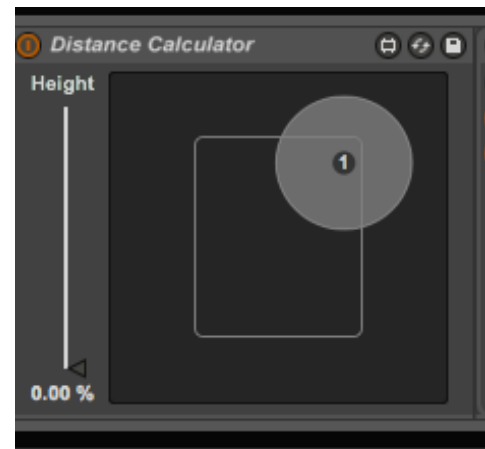
In the Bottom left of the window is a box named 'Master' in which you will find a dial named 'Gain' that controls the master volume of the system. This is your primary volume control that will attenuate all channels simultaneously. Click on the knob and move the cursor to adjust total gain.



Distance Calculator

This is the module in the program that calculates the time delay and the volume of each speaker in the system, based on a grouping of equations vary these parameters using the distance the speaker is from the center of the room, and its location with respect to the node being manipulated in the visual interface. The node is the opaque gray circle in the middle of the module that can be dragged around the box, manipulating the system's time alignment and gain.

On the left side of this module is a vertical slider, labeled 'Height' which controls the location of the node in the Z axis.

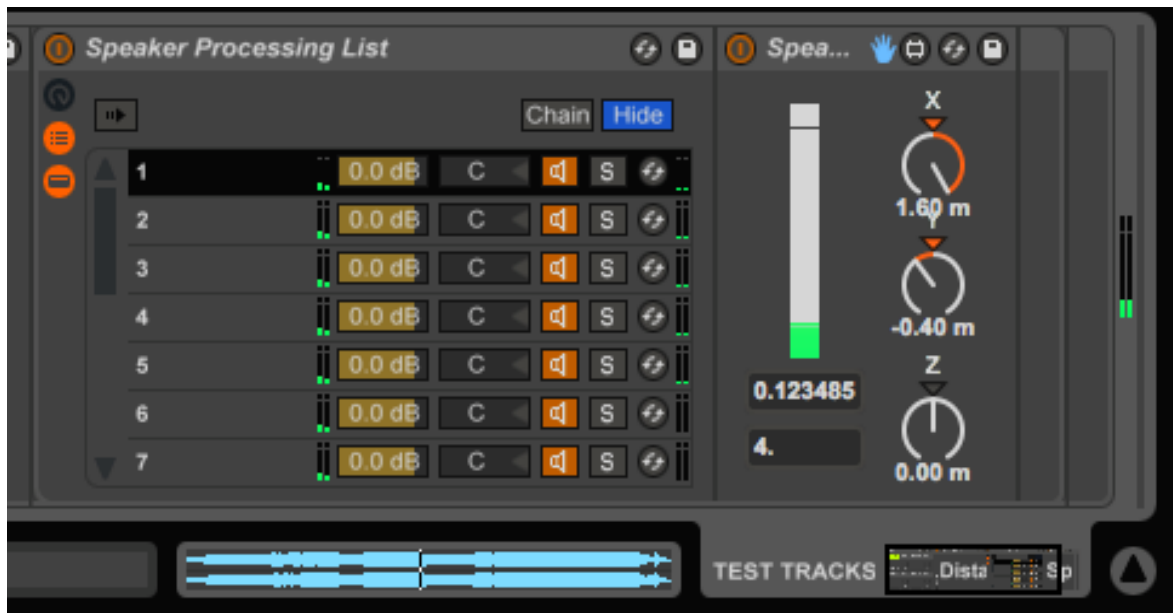


Speaker Processing List

This is another part of the module that calculates the alignment and gain of each speaker. Each item in this scrolling list, is a channel of the system, hence we have 16 items here. Each item can be clicked on, and the distance information for each speaker can be checked or manipulated using the dial. Each knob turns from negative to positive, and the number displayed is the coordinate for that part of the Cartesian coordinate x, y and z. The distance is displayed in meters from the origin of the room, a (0,0,0) coordinate in the geographical center of the xy-plane, and aligned on the lower row of speakers along the long walls.

This module can be expanded for future expansion to a greater number of channels; the dials were added to the module to allow the coordinates to be added and manipulated in real

time, and saved in the live set. This presents a possible problem whereby a coordinate is accidentally changed, which would affect the time alignment and volume of an individual speaker, as well as how/where it responds on the ‘Distance Calculator’ visual interface.



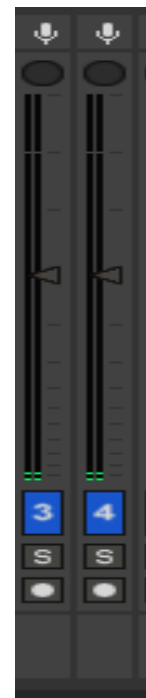
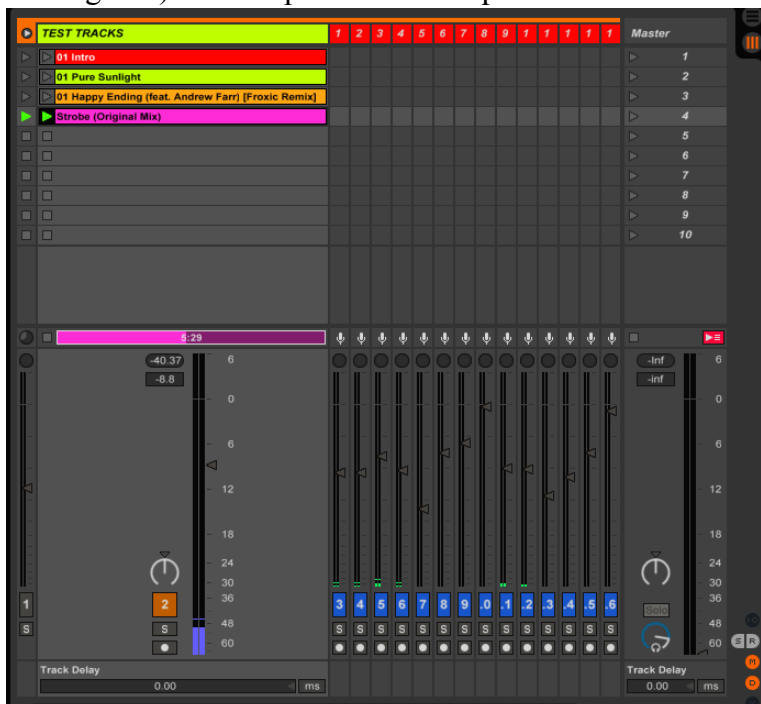
Adjusting the interfaces

Though we would love to live in a perfect world where no adjustments to the interface will ever be needed, inevitably they actually will need to be tweaked for refinement, system changes, upgrades, etc. This is where you can get a general clue as to where simple adjustments can be made to fine tune the system parameters. Core processing module modification will not be discussed here, as this is intended for the proficient end user, and intimate knowledge of Ableton and Max are neither assumed nor implied.

Individual Channel Gain

Each channel's gain can be independently controlled through a number of the interfaces along the signal path, but for the sanity of all, please only make adjustments via the main channel gain controls. This allows the gain control to be stored in an individual 'Live Set' without causing another person's work to be manipulated by an attempt to resolve a problem.

If the fine-tuning is done to improve the base file, the original Live Set (Currently "IQP Processing.als") can be updated so as to provide a better starting point for any new projects.



Adjust the gain by moving the gray pointer on the vertical bar.

Troubleshooting:

Equipment not powering on:

If you experience a single or group of components will not turn on, first check the wall plug to be sure it is plugged into the outlet.

Next, there is a power strip inside the cabinet, just above the amplifiers that may trip, or be turned off manually- check this switch and be sure both amplifiers and the power supply for the ProFire 610 are plugged in.

The C400 device is powered via USB from the computer, so if this device is not powered on, check the cable at both ends to be sure it is plugged in, and the computer must also be powered on for this device to receive power via the USB.

If there is no power to the equipment rack, or computers, the circuit may have blown and needs to be reset. Please call campus maintenance to have the circuit reset. The circuit breaker panel is located across from the entry door to this lab.

Sound Source is choppy, skipping, or artifacted:

This can be caused by a number of settings, please first open the “Audio Midi Setup” application and check the settings given in the “Opening and checking the application settings” section of this manual match those, the most likely culprit would be the “Drift Correction” has accidentally been unchecked.

Next, in the same “Audio Midi Setup”, check that the Sample Rates and Bit Rates match the required values.

Lastly, if the problem is still present, the Max and Ableton applications can be closed and reopened. Restarting the computer may also help in this case.

Echo in system, time alignment is incorrect.

See “Sound source is choppy, skipping, artifacted” and check “Audio Midi Settings”

Next, in the Ableton application interface, go to the “Speaker Processing List” module and select a channel. Use the module to the right of this, “Speaker”, to confirm the X,Y and Z parameters match those listed in the ”IQP Speaker Positions meters.xlsx” document.

The Interface portion of this manual can be used to further explain the different modules, uses and locations within the Ableton interface.

Channel Gain Incorrect

The gain for individual channels should only be modified within the Ableton interface, and only after all of the manual gain knobs on all devices and individual devices within the “Audio Midi Settings” Aggregate device have been checked. Be sure they have not been accidentally changed.

Check the settings given in the “Opening and checking the application settings” section of this manual to confirm that each device within the “Aggregate Device” has the correct gain settings.

Additionally, the ProFire 610 and Fast Track C400 interfaces can be opened and checked to see that they match the parameters given in the settings portion of this manual.