Colour The Sound: Analog Audio Processing Web Project

Written Defense Thesis

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Colour the Sound is an online interactive audio project with the attempt to give digital access to vintage analog tube equipment over the Internet. A physical space that houses analog equipment is 'virtualized' using standard and readily available digital components in order to allow any user from the “Internet Cloud” to upload digital audio content, have it automatically converted to analog, have remote control of the device from their computer over any network, and then process their audio through an analog device and receive their audio back in the same digital form. Currently while there are many advantages to a digital recording environment, specifically stemming from the cost efficiency and usability of a digital environment, there still exist several specific reasons that analog equipment is still used, specifically pertaining to their natural sound quality and a specific sought after “sound aesthetic” that they provide. While certain DSP effects exist to emulate the sound of these devices, either to great or undesirable effect, they are still imitations of a real process, and still exist in a digital domain. Digital is effectively always an imitation, as it is an artificial sampling of a real event, and it is impossible to ever truly create a perfect recreation of an analog effect digitally. Still, to use these devices along with digital recording environments pose many problems: they are significantly more expensive to use especially for occasional use, they pose the issue of converting a digital recording to analog and back again, which can be a difficult process, and lead to poor results if done improperly, they are not as easy to use as digital devices and they must exist as a physical unit and therefore are not portable and accessible everywhere. The purpose of this Media Production project was to attempt to make access to various vintage analog equipment through a digital means and try and eliminate many of the issues involved with using this analog equipment in a digital recording environment.

While this is new to the audio industry and audio production realm, many devices and procedures are utilizing the Internet for digital remote control and access, from electronic surgeries being preformed by doctors 1000's of miles away from their patients, to remotely controlled webcams and different installation spaces and promotional advertisements. Therefore, many devices and pre-existing components were available to create this project, but they have not been used before for this purpose.

Today, the large majority of audio recordings are done completely in a digital setting, while in the past recordings were done completely in an analog recording method, whether on magnetic tape or vinyl records. With analog based recordings, sound waves are converted into varying voltages and currents when recorded or processed through analog solid state and tube amplification systems, where varying degrees of sonic information can be present, as many as the cycles per second, or hertz, the specific system allows for. The term analog is used because the recording medium is created in a maaner that is analogous to the vibrations within the air pressure of the sound that is being recorded. The compressions and expansions of air molecules that make up a sound wave are turned into a voltage that is directly proportional to these events. This way analog systems most accurately represent a real sound wave as “sound is naturally an analog signal. An analog signal is continuous, meaning that there are no breaks or interruptions. One moment flows into the next.” (Strickland 3)

Within a digital system, these voltages and currents are converted into a sequence of numbers, represented in binary within a computer system by an analog to digital converter, or ADC, and back into digital with a DAC. (Zolzer 31) In order to translate an analog signal into a digital form, the voltage of an analog signal must be sampled at varying intervals in time and given a specific level of resolution at that instance in time. This “digitization of a sampled signal with continuous amplitude is called quantization,” (Zolzer 13) and refers to the processing of a varying analog electrical signal into digitized bits. There is no current practical way to convert sound energy directly into a digital signal, and thus most current digital systems convert sound into an analog voltage based system, and then into digital with an ADC. The two most important differences between analog and digital is that while the first is a continuous signal in time, the latter is discrete in time. The difference is that while an analog recording is a continuous variation no matter how small of a time period you consider, digital recordings have distinct parts that have division points between them. (Reference.com)

While the digital processing offers many advantages over an analog system, such as extremely low noise and digital error correction, a digital signal will always be an inferior representation of a true sound event as “digital signals are not continuous. They use specific values to represent information. In the case of sound, that means representing a sound wave as a series of values that represent pitch and volume over the length of the recording.” (Strickland 4) Digital recordings can miss subtle nuances within the recording that exist in an analog system, which is why digital sampling is now far exceeding the range of human hearing, with most professional studios recording with a sample rate of 192 kHz and resolution of 24-bits. While 44kH/z should be sufficient for the range of human hearing, as our hearing is limited to around 25 kHz, due to issues presented by the Nyquist sampling theorem (Zolzer 11) and certain psychoacoustic properties occurring above our perceived hearing range, that much greater sample rates are required to attempt to mimic the continuous nature of real life sound. The concept that as long as you are sampling at a rate that is similar to the levels that human ears can hear has some issues associated with it. The problem is also that the human ear can distinguish the differences in quality between the original sound and the digital synthesis of that sound being played back as a recording. (O'Neal) There is a lot of details going on within an analog signal that cannot be replicated by a digital recording even at high sample rates. (O'Neal)

Besides for the purpose of experimentation with remote Internet control devices and as an art installation for manipulating digital sounds, there are some specific reasons why one would want to use analog processing audio gear, specifically vacuum tube equipment in conjunction with digital audio. This is due to the sound aesthetic that is associated with the processing of audio through analog equipment, a human preference for sound signals that are reproduced through analog systems over completely digital ones. While digital systems are slowly becoming becoming better at emulating this continuously varying analog signal with higher sample rates and resolutions, and while there is a claim that it is impossible for one to discern the difference between a digital and analog recordings when sampled above 192 kHz at a resolution of 24 bits, there are still many subtle psychoacoustic events that occur with an analog system that have been eliminated with a completely digital environment. There exists within sound recording and reproduction particular aesthetic qualities that are desirable and are created by analog sound recording systems. Some of the terms used by audiophiles to describe these tonal qualities or timbre of the recording are “full, warm and airy.” (Strickland 6) While they represent a somewhat indescribable phenomenon that exist within analog recordings, these terms can be tied to some technical understanding. A warm sound is typically associated with a recording that effortlessly reproduces low frequencies, even below the human hearing range, and an “airy sound means that the music reproduced gives the listener the impression that all the instruments are in a spacious environment,” and is typically created by a very responsive and accurate high frequency range. (Strickland 6)

At the same time, while audio engineers use the metaphor of “warm” to associate with analog recordings, the term of “cold” is applied to digital recordings. These terms apply from the opinion that “analog music seems more lifelike, inviting and 'warm',” while “ digital music sounds more artificial, offputting and 'cold” (Gozzi) due to many factors in the recording process and sound processing. Bernie Grundman, a world renowned mastering engineer who owns Bernie Grundman Mastering, calls digital a 'skinny sound' due to the fact that “Digital always has a thinness about it” (Iverson) while Analog has size, and is usually what he uses for mastering in his studio.

While from an engineering standpoint transistors and digital recording are more accurate than a vacuum tube, but from an artistic perspective within music, many people seem to appreciate the qualities that a vacuum tube makes. In an article by audio engineer Miles O'Neal, playing a song he wrote through both a transistor based guitar amp and tube guitar amplifier illicited completely different results from the listener, the latter having a more profound and enjoyable effect on them. (O'Neal) This is also the same reason why folk-rock musicians The White Stripes record all of their music onto analog tape machines through tube-based audio equipment, to impart the same dynamics of the 60's and 70's music they are emulating. This is not merely just a niche oriented opinion on which recording technique is more favorable. A study was done where people were given blindfold tests of both analog and digital recordings, and people were told to say which they preferred. The conclusion was that about half of the people preferred analog recordings from vinyl records, and half preferred the same recording from a CD. (Gozzi) One of the biggest reasons for the aesthetic appreciation of analog recordings is simply because that it is what most people are accustomed to hearing, as all-digital recordings have only been around for the past decade, (Audio Master Class) but there are also some technical reasons for this association.

Ironically, much of the noise issues with analog that have been eliminated in a digital system as described above have actually come to be perceived as desirable by many from a music aesthetic point of view. People have become accustomed to the natural imperfections of analog recordings, so that some audio enthusiasts like Dub producer Adrian Sherwood and audio journalist Harry Pearson to proclaim that “LP's are decisively more musical...CD's drain the soul from music. The emotional involvement disappears.” (Bookey) This assertation is due to the fact that the 'analog sound' being described as 'warm' and 'fat' “is more a product of analog format inaccuracies than anything else” (Reference.com) While analog recording is in fact a more accurate reproduction of sound, from the point of view of a sound aesthetic, it is also the type of imperfections and noise associated with analog recording production that we have become to associate as having warmth and fatness to the sound. Distortion deriving from the overdriving of a piece of audio equipment, typical in rock and pop music, is very important to this sound. While the digital overdriving of equipment creates unwanted tones, known as 'clipping' and digital distortion, “tube amplifiers give a more pleasant distortion and compression to musical signals than transistors” (Earlevel) and digital signals. This is due to “the role of harmonics in the perception of tones” (Plomp 3) and the many overtones that exist within a pleasant musical recording. (Plomp 4) Natural musical instruments resonate and give off harmonic tones that are even multiples of the original root frequency or fundamental tone that musical device is producing. (Murphy) Human ears have become accustomed to the sound of these natural devices from an artistic point of view on what makes a 'pleasing' sound. It can be said that “our ears are built for even-order harmonics," which means that each harmonic is one octave or multiples of one octave higher than the fundamental, and are thus musically related to the fundamental. Analog tube based devices give off pleasing overdrive and distortion within music because they create distorted tones that are naturally even-order harmonics, giving a more pleasing and harmonious distorted sound. Conversely, digital distortion creates odd-order harmonics within music, like thirds, fifths and sevenths from the fundamental tone, and are associated as being dissonant in nature and having an unpleasant sound. (Murphy)

Another characteristic of analog recordings which is associated with a warmer sound is modulation noise. Modulation noise is the noise that changes in level as the signal level changes. (Audio Master Class) Irregularities in the speed of analog tape travel, known as 'scrape flutter' and Barkausen noise create harmonics within the sound that are not audible on digital recordings. A 1kHz recording on digital will produce a very accurate 1kHz tone, but on analog tape will consist of 1kHz plus two ranges of other frequencies that are ever changing due to random variations. This modulation noise thus accounts for a 'thickening' of the signal, creating the typical 'fat' sound associated with analog. This is not new knowledge though, as for years engineers artificially increased the amount of modulation noise on recordings by unbalancing roller heads of their analog tape reels, creating a greater range of frequencies and a thicker sound. (Audio Master Class) So, it is not only the accuracy of analog in terms of an ever-varying wave form that technically makes it more realistic compared to digital recordings, it is its imperfections as well which have become their own aesthetic quality which has been ingrained in public appreciation of music.

Thus, ever since recordings have shifted from an analog recording to digital recording domain, many "sound engineers are working on ways to 'warm up' digital sound," (Gozzi) by not only increasing sample rates to better reflect a true audio signal, but by striving to "inject that same distortion" (Audio Master Class) and noise typical in analog recordings. While many mixing studios have moved to an almost digital environment, even though some analog processing devices can be found still lying in racks, most mastering studios still utilize many analog and vacuum tube devices in order to give the digital recordings the 'warm' aesthetic that many people are accustomed to hearing in popular music. This processing of digital sound through an analog device to impart it with the analog qualities found in these devices is known as 'colouring' the sound, to give it a specific tonality and emotion not present in digital recording. When an audio signal is passed through physical elements such as vacuum tubes, they will impact the audio signal in some way, even if this audio equipment is set to bypass, the routing of the sound signal through these devices has an affect on it. Depending on your goal and preference, this can be a benefit or detriment, as it does lessen the technical sound quality of the signal, as it will change its characteristics from the original recording, but it also can create tonal qualities which are associated to being 'fat' and 'warm'. (Abbot)

Although there are many digital simulations that exist of analog and tube processing technologies, and even digital signal processors that attempt to mimic the sound of analog tape and tube compressors, “most of them only simulate the distortion characteristics of analog,” and not all of the complex psychoacoustic properties that occur when these devices are used. (Audio Master Class) This does not mean that one should have to forsake digital's many convinces though, “as we can always use other means, such as tube compressors, to fatten the sound if needed,” (Earlevel) and still record using digital technologies.

In order to attempt to create a service or device which would give a compromise between the simplicity and accessibility of digital, and still offer the usage of analog tube based equipment, the concept of an on line project which would allow users to upload, control and process audio through an analog device entirely through digital means was created. The project for this masters was simply a prototype or proof of concept for a larger scale conception from which the idea came about. While the initial idea from which this project stemmed from was for a large commercial website business plan that would give users access to a large collection of vintage analog equipment on line for either a per-use fee or advertising revenue, this project is an experiment to determine the effectiveness and feasibility of processing audio in this way, as well as an on line “installation” for people to manipulate, control and process digital sound analogously, while still staying in a digital realm. For the scaled down project only one vintage device would be used, and while the commercial business plan offered to create a site that would have user specific logins with certain times of the device being purchased by users, on line storage for audio files, and savable presets for each device, the experimental prototype for this project would simply allow users to upload and process files on a first come, first server basis, with a simple HTML web interface.

The minimum requirements for the prototype required that a user be able to upload an audio sample in high quality to a web server, then be able to control the device remotely over the Internet while being given a preview of their audio clip in real-time so that they have a response for their actions and know what affect their control over the device is doing to the audio signal. This is known as a Human Interface Control or user input, and a feedback or output of the effect. A simple example would be a mouse controller on a computer and the cursor being graphically displayed by a computer monitor. In order to provide this user feedback, not only would a low quality real-time sample of the audio signal be available for the user, but also a webcam video broadcast of the device's VU meter to show an even more accurate visual representation of what is happening to the sound. Finally, a high quality downloadable file of the processed audio would then need to be available for the user to 're-implement' the recording back into their digital project or to use in whatever way.

The purpose of the project was to make the experience as easy and simple as possible for the end user, versus giving many additional options and services. Thus, rather than using some form of email or FTP based uploading and downloading process, users can upload and download their audio files entirely on the HTML website. To do this, a PHP upload and download script creates an embedded uploader and downloader directly within the webpage, and also limits the file size to 100 Megabytes, file type to uncompressed Microsoft Wavetable Audio (.wav format, an industry standard) and sample rate and resolution to 96khz/24-bit. As discussed later, this was chosen over using a compressed lossy audio format since while a compressed format would offer much smaller file size and quicker transfer, it would lower audio quality to the point of negating the purpose of using analog tube audio equipment.

In order to minimize lag and make hosting and creating the project as simple and problem free as possible, the website is split up into two portions. The informative introduction and explanation of the project which is hosted on Ryerson's STW network server, while the portion of the site where the user can upload and process their audio files is hosted directly on a computer server within the physical 'installation space' of the site in the Rogers Communication Centre at Ryerson University. This computer also performs every other aspect of virtualizing the analog audio gear: storing the uploaded audio files to be processed, providing remote control over the device, hosting the audio and video webcam streaming, processing the audio signal from digital to analog and back again, and finally providing a downloadable digital link for the user. By performing this entirely on one computer, it greatly simplifies the need for complex communication between two computer servers over different networks and keeps lag within the system to a minimum, while being much more affordable. While the transition from the 'main site' to the 'processing site' is fairly seamless, users are actually being redirected from the Ryerson Web hosting server to directly connecting with the server computer housed within the installation space.

Specific programming languages and devices were chosen to complete this project due to their intended use and closest relationship to what was required for the project to work, as well as their affordablitiy and ease of use. As many open-source solutions were attempted to be used, but proprietary software was needed in certain situations. Since this is the first time a project like this has been attempted, there was not always an obvious answer, device or code that would work perfectly, and a modified solution had to be made. The computer server runs on Windows Server 2003 (http://www.microsoft.com/windowsserver2003/default.mspx) and hosts the 'processing' portion of the website using Apache, as well as PHP 5.2 for downloading and uploading and a MySQL table database.

An all in one open-source solution known as AppServ (http://www.appservnetwork.com/) was used which easily integrated all of these components into one program. A PHP script was created which allows users to upload files to a specific folder on the server computer, and programmed to check that the filesize (<100 Megabytes) and format (.wav). The audio file is than timestamped, which means that it is renamed to the time and day that it was uploaded.

In order to process the sound from analog to digital automatically and back again, MaxMSP, "an interactive graphical programming environment for music, audio, and media." (http://www.cycling74.com/products/mmjoverview) was used. Within this framework, a Max program was created that checks the folder on the computer where users upload their files, and pre-loads the latest timestamped wav audio file. This way, the latest user to use the system automatically gets control over only their audio file that was uploaded. The MaxMSP code also configures and connects to the DAC/ADC of the system, so that the system can convert the sound from digital to analog and through the analog device and then record it back into digital. A firewire based Motu 896 (www.motu.com) was used due to its high quality conversion of audio and maximum sample rate and resolution of 96khz/24-bit, so that there would be as little signal degradation as possible.

In order to allow the user to preview and playback their audio file remotely over the internet, a system had to be created to let them start and stop audio within MaxMSP on the server computer, but from their remote location. While many different open-source methods were considered, from writing a JAVA (www.java.com) applet to exist on the website and communicate with MaxMSP through the OSC (Open Sound Control) Communications protocol (http://opensoundcontrol.org/) , as well as a Java server that allows for Flash frontends to communicate with programming languages known as Flosc, (http://benchun.net/flosc/) a third-party MaxMSP object known as Flashserver (http://www.nullmedium.de/dev/flashserver/) which lets Flash movies communicate directly to MaxMSP through an XML Socket was used for its simplicity. This way a Flash SWF movie with buttons to control the audio of MaxMSP is embedded on the server computers website, providing a frontend for a remote user. The SWF is programmed to create a connection to the flashserver object in MaxMSP over a specific UDP port, and then each button is programmed in Flash to send XML packets to a specific udprecieve object in MaxMSP, which then trigger the playback, stopping and recording of the loaded audio file in MaxMSP.

In order to control the analog device remotely, RC servos and servo controllers had to be setup to work remotely so that one could manipulate the device over the internet. The specific device, an ADL-1000 analog tube compressor had only two knobs for gain and peak reduction thus simplifying how many servos were required for this prototype. Two USB Phidget servo microcontrollers (http://www.phidgets.com/products.php?category=11&product_id=1000) were used along with two HiTec HS-322HD Standard Servo motors. (http://www.hitecrcd.com/) The Phidget Microcontrollers allow for a Webservice to be used which over a specific TCP port allow for them to be addressed over a network. By creating a Flash based frontend and hosting it on the web server, users can control the devices specifically up to one degree in realtime by dragging a slider in Flash with their mouse.

Finally, in order to provide audio and visual feedback of the device while being controlled, a Broadcam Audio and Video Streaming Server (http://www.nchsoftware.com/broadcam/index.html) was set up with a Microsoft Lifecam to show the VU meter of the compressor. This particular system was used as it offered both a low quality, highly compatible Motion JPEG video stream as well as a high quality Windows Media Player 10 video stream. The audio stream quality of the Windows Media Stream was chosen to be 128 kb/s WMA, which offers fairly realistic sounding results, but creates roughly a 5 second delay between when the user changes the knob using a servo, and the result being heard, while the low quality JPG stream is almost instantaneous on high bandwidth networks.

One of the most difficult decisions of the project was specifically what piece of analog tube based audio processing gear to use for this prototype website. One of the main reasons to offer this equipment on line in such a way is due to its rarity, expense and inaccessibility to most recording engineers, so to in a sense greatly simplify its availability for use in a digital context. While this makes the process easier for the end user, acquiring a piece of hardware within the projects budget proved to be extremely difficult. After consulting with both Henry Warwick and Brian Moncarz, both Radio and Television Arts professors at Ryerson University specializing in audio recording as well as Ben Swarbick, an audio engineer in Toronto and on line Internet polls, the most wanted and recommended piece of gear for the project was an all-tube analog compressor. My own on line poll found that 67% of all people in an audio engineering field found that analog compressors still surpass their software and digital counterparts, and another on line poll found that 71% of all audio production enthusiasts preferred hardware compressors over software compressors (Homerecording)

After researching, it became clear that the Universal Audio LA2A compressor would be a good fit for the project due to its relative simplicity requiring less control remotely, as well as its popularity and all vacuum tube design. In the end for the prototype project, the Anthony Demaria Labs ADL-1000 was used due to it being a complete replica of the LA2A compressor and significantly more affordable while still proving the same conclusions for this audio experiment.

While the use of analog sound equipment being digitized over the Internet has never been done before in quite this way, there are several different websites and uses of robotics and microcontroller functions over long distance communication that have been experimented with before. Both the transfer and processing of digital sound through physical spaces as well as the manipulation of servo motors through internet applications has been done both with audio websites Silophone and Tank-FX, and also controllable video webcam applications and other robotics usages. One of the most popularized usages of these technologies is several different laser surgeries, such as “RoboLase” where “Internet-based laser scissor-and-tweezer technology” is used for “for physicians to perform medical procedures from distant locations.”(American Telemedicine Association)

Some Internet based interactive examples similar to this project also utilize the same programming and applications, but to do either very different or somewhat similar tasks. For example, Flashserver, the flash-movie based front-end and XML communication Object for MaxMSP, has been used in other installations. For example, A project called Sale Away by Staalplaat Soundsystem used flashserver as a graphical user interface for “Archy,” their mobile-phone robot which users could control at the European Media Art Festival in Osnabrueck, 2004. It was designed to actively engage passers-by otherwise absorbed in looking at things for sale, at themselves and at others. Sale Away invited people to play and play with a complex mechanical sound orchestra installed in a shop front using their mobile phones. (Osnabrueck) While these are all examples of either newly emerging technologies within closed medical communities and art installations, one of the most commercial and popular uses of controllable robotics devices on the Internet are controllable webcams. These advanced webcam devices allow users to either gain a determined amount of individual time or have share access to control the zoom, rotation and yaw of the camera, using flash or java based interfaces. (www.mst.edu/)

Two websites that largely influential on the concept of this project were Tank-FX and Silophone, which record and playback uploaded audio content through physical spaces. Tank-FX was set up in an old ferro-concrete water tower in Oberhausen Germany and is described by the creators as a “non-virtual effects device” (Cremers) and Silophone is described as an interactive “instrument” where users are able to upload and play sounds within an abandoned Grain Silo in Montreal Quebec. While the former has been created as a practical device for recording engineers to upload and process audio through a physical tank for reverberation processing and the latter as an art installation, they both use similar technologies to upload, playback, record and download sound samples back to the user. The creators of Tank-FX were using the space themselves to record musicians in, and thought up the possibility of giving worldwide access through the Internet to the space. By connecting two stereo speaker monitors and two condenser microphones at opposite ends of the tank up to a Unix webserver, people are able to upload FLAC audio files and have them simultaneously playback and record the sound through the tank, and be given back to the user as a download.

Many of the elements used within these projects are utilized with the Colour The Sound experiment. While Tank-FX used a Unix webserver with PHP to allow for high quality samples to be uploaded and downloaded, it also used a simple command-line driven audio playback script known as Ecasound (ecasound.com) in order to automate the playback and recording of the audio within the tank. Only one person can record in the tank at a time, and users uploads are cued until the tank is free to use, with a one-minute file size restriction. Silophone differs in that it allows multiple users to playback sound samples simultaneously, so that "anyone can use it and listen to it at any time." (Clements) Like Tank-FX, "microphones and loudspeakers have been installed inside four of the storage cylinders." (Dolan) and users can either upload sounds on line through the Internet portal, by calling in with a cellphone or from a sonic observatory placed near the silo. Sounds are all listened two back in realtime through Realmedia RealProducer audio script on the site, and can be listened to by anyone at anytime.

Since this project is one of the first to utilize this equipment in this way, some issues were encountered through the process of creating the site. These include issues with the microcontrollers, web hosting and automation of the audio processing. While there were other methods that were investigated, such as using an Arduino (http://www.arduino.cc/) microcontroller and using Processing and Java to create a controller for the device, the Phidget solution was a lot simpler as well as used flash Actionscript, which I was much more familiar with. Also, I initially attempted to use a Continuous Motion Servo to control the ADL-1000's knobs, which proved to be the wrong kind of servo motor, as it operated more like a traditional motor, spinning continuously in one direction forever without any sort of feedback positioning. Thus, the Phidget servo controller and non-continuous servo were opted to be used instead. One of the most major issues was getting every device to 'talk' to each other over an Internetwork of switches and severs. Since the server for the physical installation of the device is located in the Ryerson building, it must be networked through a series of switches and logical layers before being visible to the world wide web. Getting the HTTP and PHP server to broadcast out and host the video and .swf files onto a users computer, and then open up multiple ports for data to be sent and returned to these interfaces over the network proved to be an extremely difficult process. Ports had to be opened for the HTTP webserver, webcam, servo controllers and Max/MSP audio controller, all simultaneously. Prior to even being visible on the Internet, getting the flashserver frontend for Max and the Actionscript based servo controller Phidget web front end to communicate with each device proved difficult enough on a Local Network, but there are even more issues once the data must pass through Internet Service Providers and multiple gateways and switches on the Internet. A final issue was encountered through creating the automation of the loading and playback of audio through Max/MSP, since the folderwatch object was not designed to monitor a folder and also select the latest created file. A solution was created by timestamping the audio file coming in through PHP, and having Max/MSP automatically load the file with the largest number, which due to timestamping is always the most recent file. Currently the most recent issue that is being encountered is that the upload of a large file in PHP over the network causes the webcam server to go off line, which is currently in the process of being fixed.

Finally, while the project does work as intended as a beta prototype, certain elements were not implemented as of yet due to time constraint and their lesser importance within the context of the project. Both a queuing system for uploading files, which would notify you when the device is available with an e-mail, as well as storage of the final processed audio file for later download were not implemented into the project.

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