ADVANCEMENTS IN ANALOG AUDIO RECORDING

THROUGH PULSE WIDTH MODULATED OPTICAL RECORDING

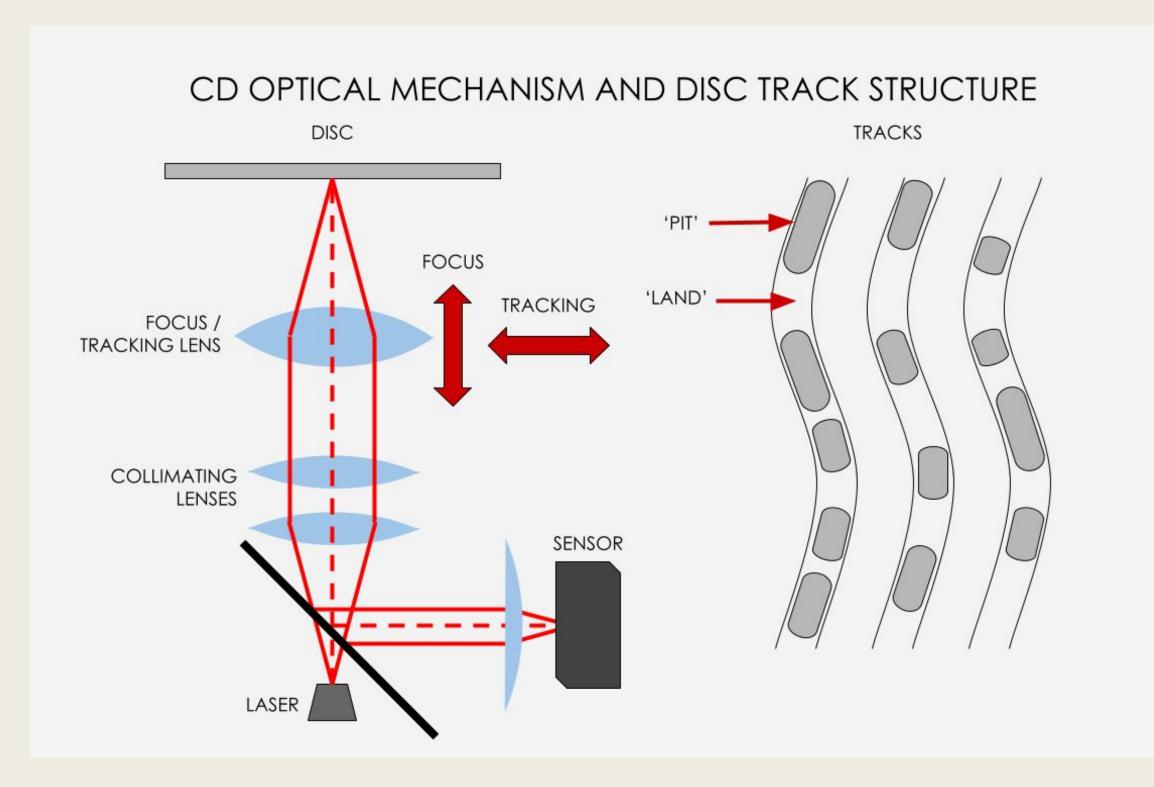
RESEARCH

CDs use a system of 'pits' and 'lands' that act as 1s and 0s in their conventional digital playback and recording systems. Pre-recorded CDs use physical height differences in the reflective layer, while recordable CDs use a dye that changes color when hit with a recording laser. They both have the same effect of creating sections where light reflects, and places where it doesn't.

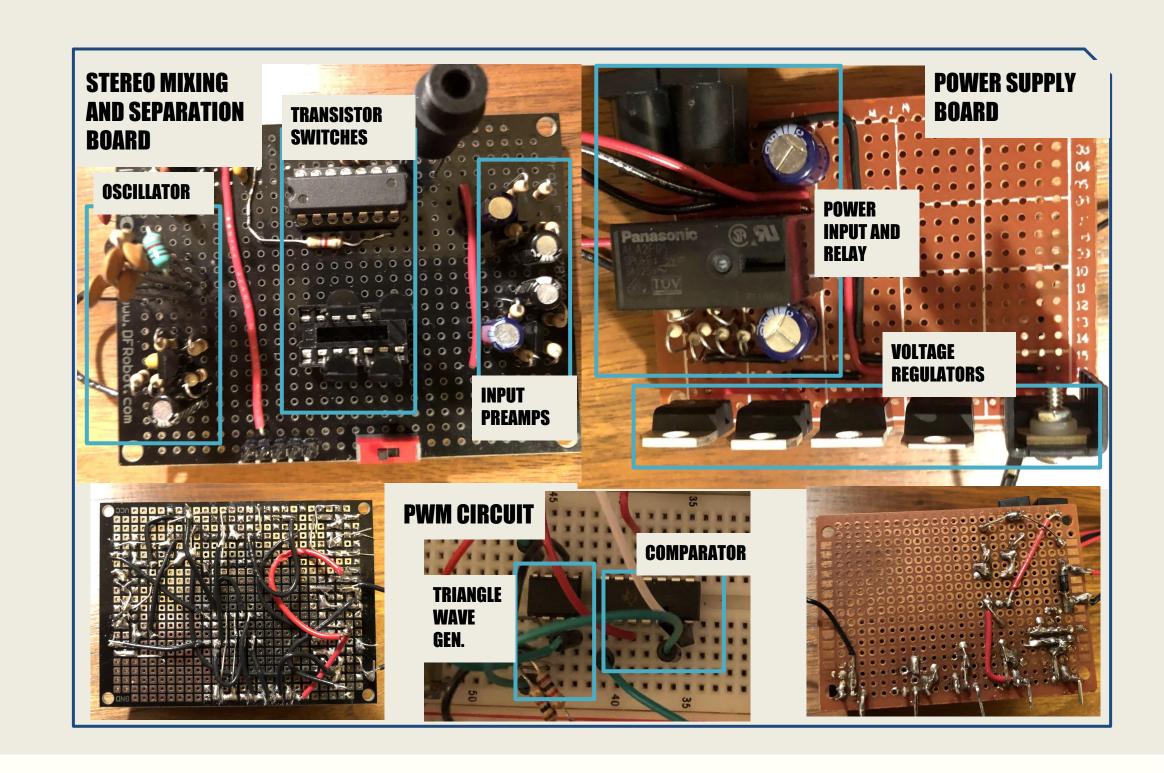
For keeping the laser focused on the right track at the right speed, there are four different systems simultaneously running. The first is the spindle motor, which spins the disc at a constant linear velocity, meaning the speed of the track passing the laser is always kept constant while the rotational speed of the disc is adjusted to compensate for this. The second and third systems work in tandem inside the laser mechanism. The laser beam is split off into two side beams separate from the main beam that land just to the left and right of the track. When the laser is offset from the data track, the side beam begins sensing the track and the lens assembly is shifted to the side to correct for this. The shape of the laser beam also acts as feedback for how in-focus the disc is. The lens moves up and down to compensate for this and achieve optimal focus. The final system acts as a coarse adjustment for the sideways tracking adjustment. When the lens assembly has reached its mechanical limit, the entire assembly is shifted to the side by a motor to re-center the lens.

A class-d amplifier operates by comparing an incoming signal to a triangle wave, either outputting a 'high' signal if the signal is higher than the triangle wave, or a 'low' when the signal is lower than the triangle wave. The frequency of the triangle wave is generally a few hundred kilohertz so as to be out of the limit of human hearing and thus inaudible. This 'compared' signal acts as a pulse-width modulated representation of the original signal using only 'high' and 'low' signals. The original wave can be reconstituted by passing the PWM signal through a low-pass filter that only lets the audible portion of the signal through.

Because there is only a single signal being recorded onto the CD, the resulting recording will be monaural. However, stereo recording can be accomplished by multiplexing the left and right channels onto the recording signal. This is accomplished during recording by rapidly switching between the left and right input signals at a frequency high enough to be out of the range of human hearing. During playback, the signal played back from the disc is rapidly switched between the left and right outputs at exactly the same frequency and timing as it was during the recording process. This process allows multiple signals to be recorded simultaneously onto a single channel, at the cost of slight distortion.



CONSTRUCTION



RECORD PLAYBACK INPUT PREAMPS STEREO PULSE-WIDTH OPTICAL DISC SENSOR OUTPUT **STEREO** OUTPUT LOW PASS FILTER AND RECORD MULTIPLEXING MODULATION MECHANISM PREAMP SEPARATION AMPLIFIERS AND COMPARATOR LEVEL CONTROL **SWITCH** SWITCH PLAYBACK LEVEL CONTROL OUT SIGNAL SIGNAL OUT IN SWITCH LASER IN SENSOR OUT SWITCH STEREO MULTIPLEXING OUT ANALOG CD **STEREO** PHASE ADJUSTMENT TRIANGLE WAVE TRACKING, FOCUS, SEPARATION **GENERATOR** AND MECHANISM RECORDER BLOCK FREQUENCY CONTROL **GENERATOR** RLRL DIAGRAM PULSE-WIDTH MODULATION CONVERSION **AUDIO INPUT** PWM OUTPUT BY NIKOLAS FAULKNER

SQUARE WAVE

TRIANGLE WAVE

INPUT

ABSTRACT

Over the past four decades, audio technology has largely shifted away from analog and towards digital formats. However, this trend has not enjoyed unanimous support, and there is still a growing market demand for analog audio recording and playback formats that is no longer being satisfied. The purpose of this project is to create a new analog audio recording and playback format by modifying modern hardware designed for digital use, namely, the Compact Disc.

After thorough research, a basic block diagram and theory of operation for a device that could record and play back an analog audio signal from a compact disc was developed. The central concept was to feed the incoming signal into a circuit similar in operation to a class-d amplifier in order to create a pulse-width modulated representation of the signal that could then be recorded onto the disc. During playback, the signal could be read off of the disc and directly fed through a low-pass filter to retrieve the original analog audio without any digital processing. Individual circuit 'blocks' were constructed that each performed portions of the operations of the device. Their performance was measured in terms of frequency response and other metrics, and were found to operate adequately for use in an audio recording application.

Although attempts to create a fully integrated device for recording and playback were unsuccessful due to time constraints, the individual circuits were found to operate, showing that this technology could become reality with continued development.

BACKGROUND

There is an increased market demand for analog audio products, and that market is currently growing. In 2020, the sales revenue of analog vinyl records surpassed that of digital Compact Discs (CDs) for the first time since 1986, according to the RIAA. Almost all analog recording and playback technology is from over 30 years ago, with little or no research or advancement being done on the topic since then, in favor of digital audio.

The two dominant formats of analog audio recording, magnetic tape and vinyl records, were first developed in the early 20th century, over 70 years ago. Optical recording, especially onto discs, is a much newer format where most serious development started in the late 1960s. Further development into analog optical recording ceased in the late 1970s in favor of further research into digital optical recording. This led to the development of the digital CD and its release in 1982. With less than a decade of development, there is likely still much more advancement possible in the field of analog optical recording.

GOAL

Can analog audio recording and playback formats be improved upon, and can modern commercially available components be used to achieve it? This project is to create a device utilizing commercially available low-cost modern components and recording media to develop a new method for analog recording.

The final product of the project should be a self-contained device that can both record and play back analog recordings on a medium not previously utilized for analog audio recording. The device should be able to record a usable range of the human hearing spectrum (20-20,000 Hz), with minimal distortion. The target for frequency response should be at least 150-10,000 Hz. Stereo separation should be noticeable to the listener.

RESULTS

Experiments on and development of the pulse width modulation circuitry and stereo multiplexing circuitry has been conducted. The pulse-width modulation circuitry was created and tested by building a class-D amplifier out of discrete components and was tested to ensure that the frequency range was acceptable. It was then modified into the pulse-width modulation circuitry that was used in the final device by being split into two parts, with the PWM section of the amplifier being part of the recording circuitry and the low pass filter and output stage of the amplifier being part of the playback circuitry. The PWM circuitry was tested with a frequency generator and oscillator to observe its behavior at various frequencies as part of the testing stage.

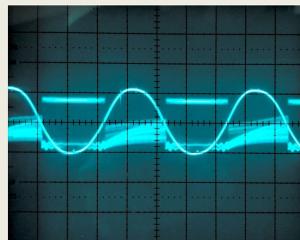
The CD recording mechanism was planned to be made by modifying an optical drive from a PC. A variety of commercially available CD recording drives were experimented with, but unfortunately due to time constraints, a method for injecting signals into the recording laser has not been found yet, and thus recording onto the disc has not yet been accomplished.

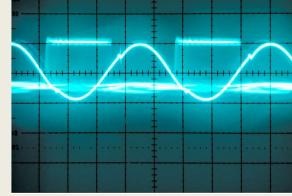
The stereo multiplexing circuitry consists of four or five components: an electronic switch that is rapidly toggled between the two input signals so that only one channel at a time is being outputted onto the single recording signal, a second switch that is responsible for splitting the multiplexed single-channel signal back into two discrete channels again for playback, a crystal oscillator that generates the high-frequency signal necessary to ensure the record and playback switches are operated at exactly the same frequency, and a phase control circuit that adjusts the phase of the switching signal during playback to ensure the playback is in phase with the recording, and the correct signals are sent to the right and left channel outputs. The switches and oscillator were both developed and tested.

Separate power supply circuitry was also developed, consisting of two laptop 'power bricks,' voltage regulators, a relay for switching the power, and miscellaneous support components.

DATA AND CONCLUSION

Pulse-width modulation conversion results:



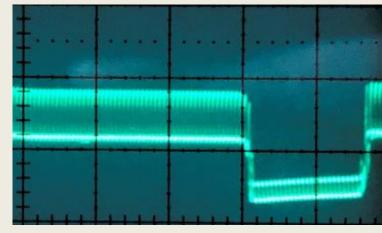


Performance at 1KHz

Performance at 100KHz

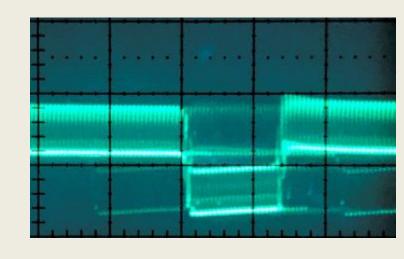
The curved sine wave is the input signal, and the square wave superimposed over it is the output of the pulse-width modulation circuitry. It is a pulse-width modulated representation of the sine wave. In both cases, the pulse-width modulated output switches from positive to negative as the input sine wave crosses the zero voltage line. The amplitude of the signal is relatively consistent across the frequency range, but interestingly, there seems to be some increased interference or distortion at the lower frequencies that is less noticeable at the higher frequencies.

Stereo separation circuit results:



Left channel output from: <- Left signal





The images show the output from the left output channel of the stereo separation circuit, after combining a stereo signal onto a single channel and then separating it again. The stereo test signal had a 1Khz tone on one channel at a time. The signal leakage from the left channel onto the right channel is likely due to imperfect timing of the switching allowing the left channel signal to be partially outputted onto the right channel, and vise-versa. Much of this issue likely stems from the distorted waveform that the oscillator was outputting into the switching circuitry. This distortion would have caused the 'on' time for each channel to not be perfectly equal to each other, and the uneven slope of the signal could have led to situations where the switching field-effect transistors were both partially activated at the same time, allowing the signals to mix together.

When the signal was tested with a signal tracer to hear the outputted audio signals, the correct signals were dominant in each channel, with a faint 'ghost' of the other channel mixed in. There was also some audible distortion in the audio, which was likely partially due to the switching signal that was being introduced into the audio output.

While the circuitry must be improved upon, these results are promising because they show that the various components of the device work as expected, meaning that it is likely feasible to create a completed analog CD recorder device.