Standalone Modular Audio Processing System

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Introduction

Modern audio production tools often vary in size and scope, and it is unclear for many beginners where to start - popular digital music production suites can vary greatly in price, and those desiring to work in an analog environment will quickly find themselves overwhelmed with the wide range of (potentially costly) options available to attain the effects a user may desire, not to mention each option creating that effect in subtly different yet noticeable ways. The goal of this project is to create a single platform for prospective users to connect an instrument to and have a good range of desirable effects available out of the box, while maintaining both an affordable price and the ability for more experienced and/or adventurous users to route their audio signal in and out of the device at various points, should they wish to combine the effects provided with the product with other tools that may already be available. This platform will combine select popular and recognizable effects with novel digital signal processing techniques that provide new options to manipulate incoming audio and create variety in the output signal.

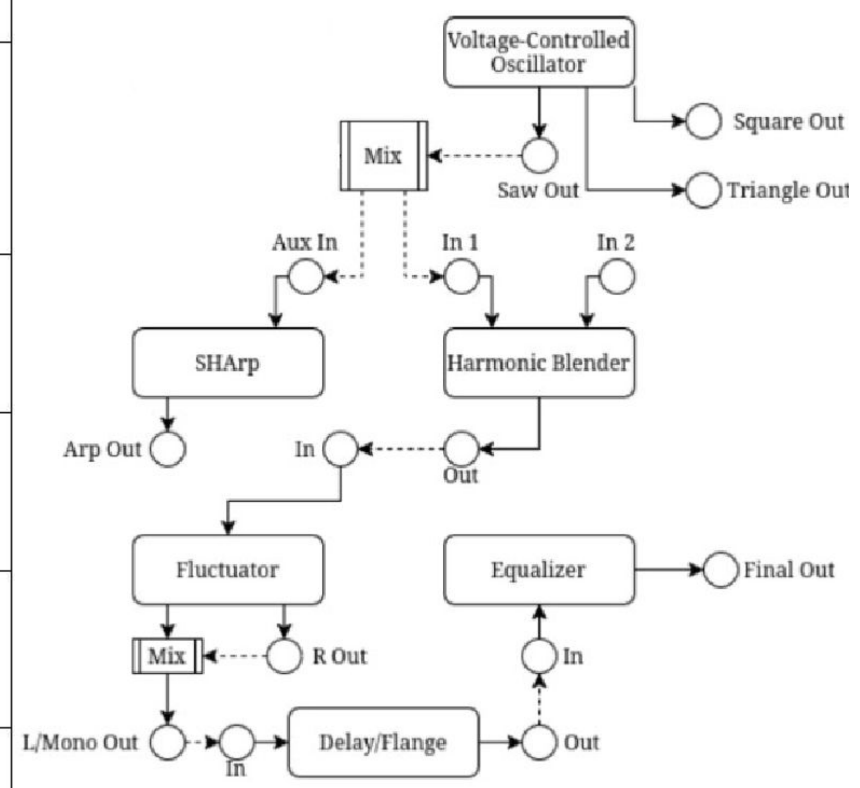
Executive Summary

The aim of this project is to develop a versatile digital signal processing tool that enables users, such as musicians and sound engineers, to creatively manipulate audio signals and sounds. The device will modify audio signals using techniques like filtering, delay, sample-and-hold, and advanced spectral analysis. The product will consist of two primary components: an embedded digital signal processing microcontroller and a set of analog signal processing systems working alongside the DSP controller. The device is also planned to be partially compatible with the Eurorack modular instrument standard to allow a user great control over signal routing/processing through the unit, using a base of commonly used connections and controls that many users will likely already be somewhat familiar with. Supporting internal components will include operational amplifiers for buffering, filtering, mixing, and modulation, analog-to-digital and digital-to-analog converters for handling audio input and output, and potentiometers/switches for user control over function parameters. The system will also include an on-board controllable simple sound wave generator to allow for introductory exploration and mixing of sounds without the need for an external sound source.

Scope

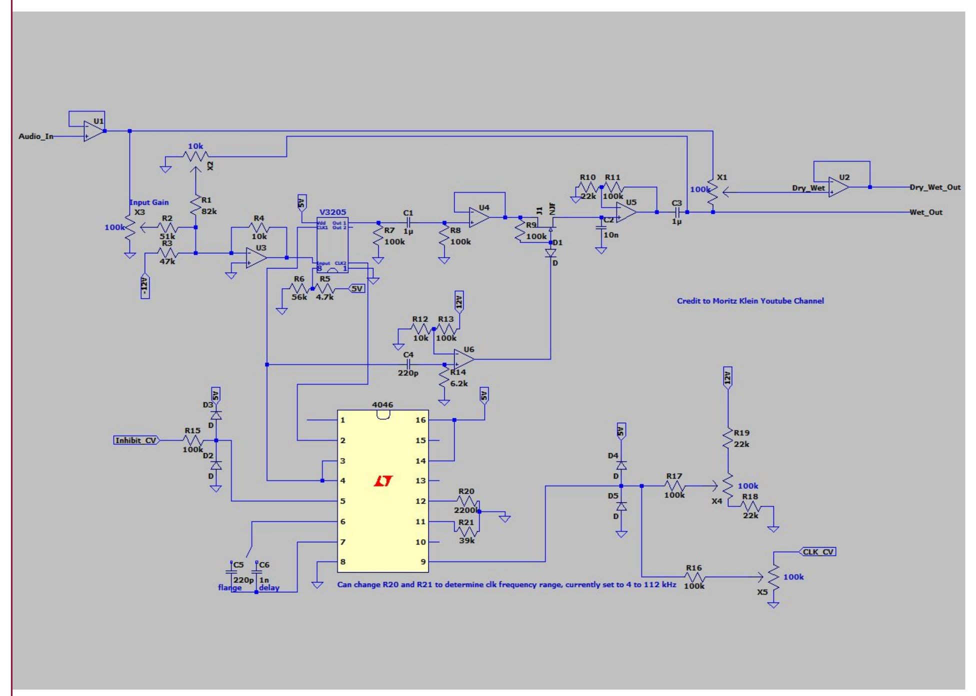
Function	Method	Description
Voltage-Controlled Oscillator (VCO)	Analog	Generates three simple waveforms whose frequency/pitch can be adjusted by the user.
"Harmonic Blender"	Digital	Converts an input signal into its frequency-domain representation, and allows the user to split the representation into two sets of either (a) even and odd harmonics, (b) low- and high-frequency components, or (c) low- and high-amplitude components. The user can then control the presence of each set and how the signal is recombined and output.
"SHArp" (Sample-and-Hold Arpeggiator)	Digital	Monitors the input signal. The user can record the signal for a brief, adjustable length and store it as a "sample." Up to five samples may be stored at the same time; the user can then play these samples back sequentially in a loop, emulating an arpeggiator.
"Fluctuator"	Digital	Takes an input signal and duplicates it, applying minor changes (such as slight distortion or flanging) in distinct ways to each duplicate. The two altered signals are then output separately, allowing for playback in stereo, re-mixing down to mono, or re-routing of the individual signals.
Delay/Flange	Analog	Utilizes a bucket-brigade circuit to capture the input signal and play the signal at the output after a short delay (20-200 ms, user-controllable). The real-time signal is also passed through to the output, and the mixing of the input and delayed signal is user-controllable.
Five-Band Equalizer	Analog	Routes the input signal through five band-pass filters that each handle a specific frequency range. Each band has its own attenuation control, allowing a user to draw out audio in some frequency ranges while suppressing others.

List of all planned functions, with descriptions of each function's operation

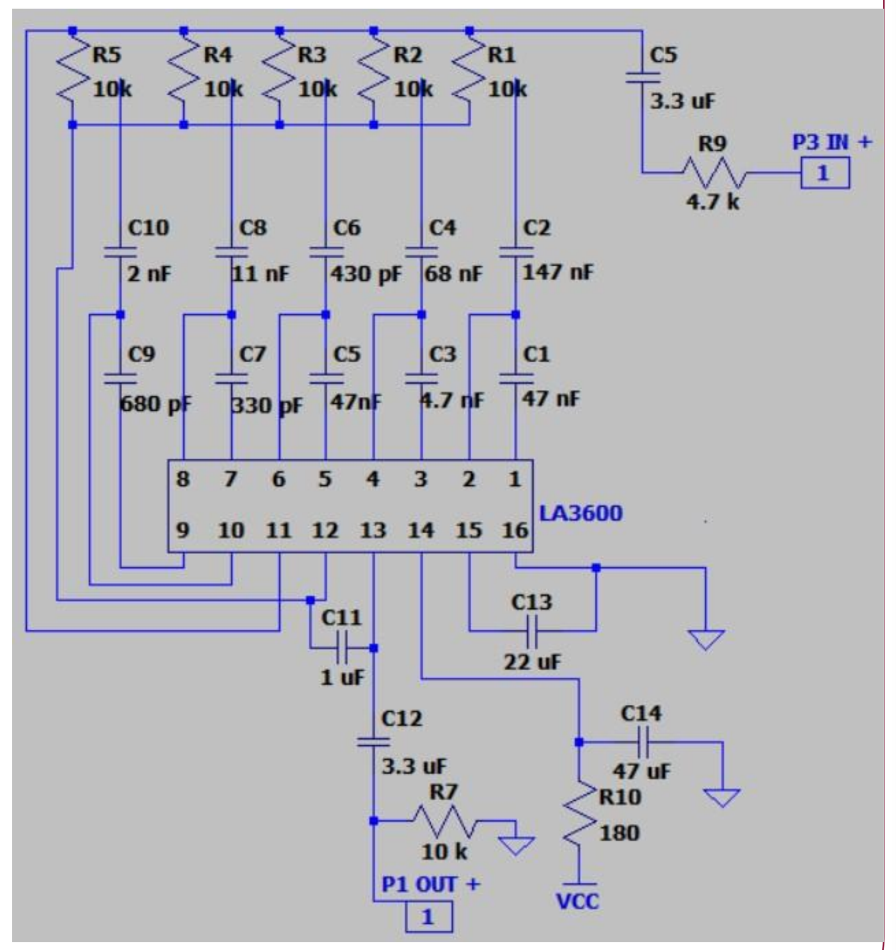


Block diagram of system design

Analog

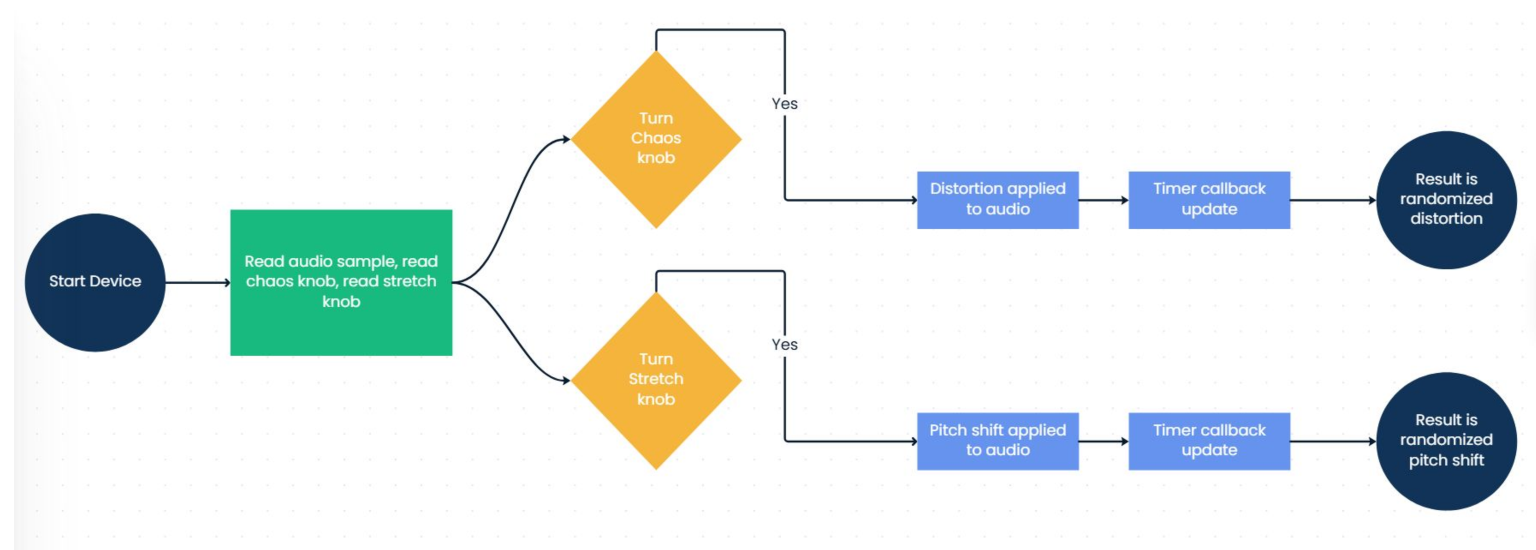


Delay circuit schematic

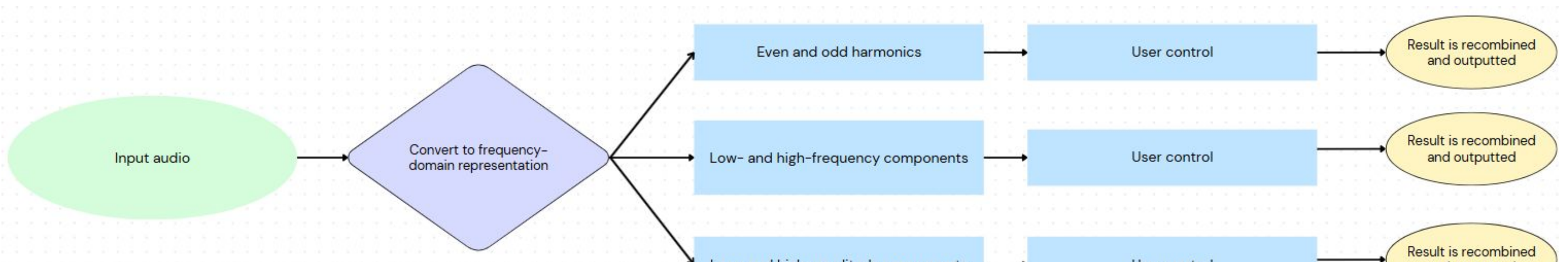


Equalizer circuit schematic

Digital



Fluctuator block diagram



Harmonic blender block diagram

Testing

The team has tested the analog and digital components of the project independently. Each excel on their own, with some small errors along the way – such as a faulty IC chips for the delay and equalizer circuit or frying boards. Putting together the circuits and making this work harmoniously with the code was cumbersome, even more so was fitting it all under the chassis and adding the faceplate. Testing results in this project will be demonstrated with audio.

Conclusion

The aim is to develop a versatile product that allows real-time audio manipulation and is compatible with modular Eurorack systems through careful planning, design, and testing. By utilizing the STM32F407 microcontroller for advanced DSP functions alongside analog components for simpler signal processing, this device bridges the gap between digital precision and the physical nature of analog equipment. The team was able to accomplish this by designing and making a prototype for any user who wishes to creatively manipulate audio signals. In the end, our team has gained professional experience and successfully completed the project.