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Daisy Dub: a modular and updateable real-time audio effect for music production and performance

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ABSTRACT

This paper presents the development of a versatile and modular real-time audio effect unit called the Daisy Dub for music producers, performers and DJs. The device utilises the state of the art Daisy Seed development board by Electrosmith and comes with a custom-made PCB and hardware case. It features a range of real-time audio effects, with an emphasis on creative delays and includes a range of modulation effects and filters in its feedback path. In addition, the unit is compact and portable, with an interactive graphical interface, four knobs and two arcade buttons for performance control, an encoder for menu diving, and an OLED screen for spectrum analysis. It features quadrophonic audio processing in real-time and is portable and powered by USB Type-C. DSP is developed with Gen by Cycling '74 and took inspiration from state of the art hardware audio delay units and modular real-time audio effect processors. A series of DSP and hardware evaluation tests were performed along with two usability tests, from an early cardboard prototype to the final manufactured device, to evaluate the effectiveness and usability. Our research demonstrates the feasibility and potential of creating a versatile and modular real-time audio effect for music production and live performance. The Daisy Dub offers a modern take on contemporary real-time audio effects, emphasising a delay effect, and we believe it contributes to the field of audio effects and music-making in general.

1. INTRODUCTION

In recent years, the rapid advancement of microprocessor technology has greatly impacted the audio processing and sound design field [1–12]. The use of microprocessors in musical devices has enabled a new level of modularity, flexibility, and real-time audio effects. The development of compact and portable devices, such as the Daisy Seed microprocessor by Electrosmith [8], has made it possible for musicians and sound designers to create high-quality audio in a wide range of settings, from music production to live performance. This research paper focuses on the further development of the Daisy Dub [6]. The original Daisy Dub paper submitted to the 2nd NordicSMC conference by Rasmus Kjærbo showcased a prototype built using a cardboard box, the Daisy Seed development board, and components connected to a breadboard. The report outlines the efforts made in creating the initial version of the Daisy Dub and conducting its initial user testing. The report ends with the findings from that user test as a basis for improvements for the next iteration. The first task performed was evaluating and testing these improvements. The key differences compared to the final version include the material (cardboard vs robust casing), construction (breadboard vs custom-made PCB), aesthetics (rudimentary vs refined design), user experience enhancements, and improved stability and durability. The cardboard box prototype served as a proof-of-concept for functionality but lacked the functionality and reliability of the final version. This research paper will examine the technical and design challenges encountered in further developing the Daisy Dub (see fig. 1), including using the Electrosmith Daisy Seed platform and Cycling '74 Max Gen software, as well as exploring various audio delay units and their design principles. The research results demonstrate the feasibility and potential of creating a versatile and modular real-time audio effect unit for music production and live performance.

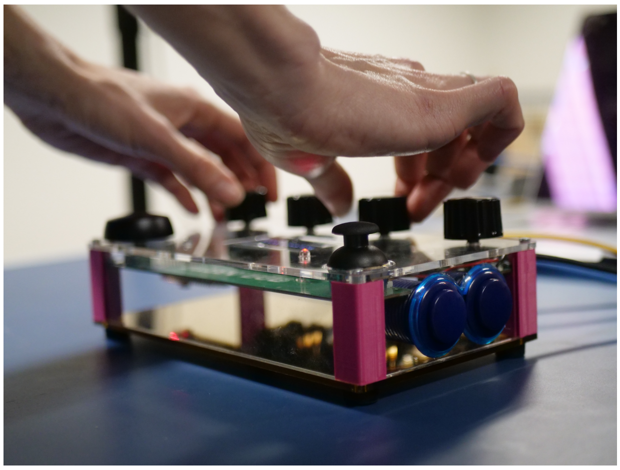


Figure 1: Daisy Dub prototype.

The music industry is constantly evolving, and there is a demand for new and innovative audio effects to help artists and producers create unique and compelling sounds [13]. One potential solution to this demand is developing a modular and updateable real-time audio effect. Music producers and DJs often rely on various audio effects and processing tools. The ability to easily update and customize an audio effect could make it more suitable for changing applications and environments. Another potential benefit is leveraging technological advances and digital signal-processing techniques. As these techniques evolve, new and improved audio effects will likely become possible.

2. BACKGROUND

Modular real-time audio effects have been a staple in music production and performance for decades, providing a means for musicians to manipulate and shape the sound of their music in real-time [2, 3, 12–14]. These effects can be hardware-based, such as pedals and rack-mounted units [15, 16], or software-based, such as digital audio workstations (DAWs) and plug-ins. One key feature of modular real-time audio effects is their ability to be customized and configured to meet the user's specific needs. This can be done through patch cables, which allow users to create complex signal-processing chains by connecting different modules. In recent years, there has been a resurgence of interest in modular real-time audio effects, with several companies producing hardware units and software plug-ins that allow users to create customized effects. These units often utilize digital signal-processing technology, allowing greater flexibility and higher performance levels than traditional analogue effects.

2.1 Delay Units for music production and performance

Delay is a fundamental audio effect in music production and performance [2, 3, 13]. It creates an echo-like effect by repeating a signal at a specified time interval and can be used to create various sounds, from subtle ambience to more extreme effects. A wide range of delay units are available on the market, ranging from simple analogue pedals to more advanced digital units with a wide range of features and capabilities. These units can be used in various settings, including recording studios, live performances, and DJing.

3. STATE OF THE ART

3.1 Comparison of Roland SDE-2500 and Strymon El Capistan

The choice of Roland SDE-2500 and Strymon El Capistan as reference hardware was selected based on their reputation and widespread use in the audio industry, especially among music producers and performers of dub and reggae music in the studio and on stage. Both the SDE-2500 and El Capistan are renowned for their high-quality sound and versatile delay effects, making them suitable benchmarks for evaluating the performance of the Daisy Dub.

Format: The Roland SDE-2500 is a rackmount processor, while the Strymon El Capistan is a stompbox-style pedal. The SDE-2500 is suited for studio use, while the El Capistan is more portable for live performances. **Digital Signal Processing**: Both the SDE-2500 and El Capistan use DSP for their delay and effects algorithms. The SDE-2500 has a 15-bit linear PCM digital audio processor with a maximum delay time of 750 ms. The El Capistan uses a 520MHz ARM processor with 24-bit 96kHz A/D and D/A converters, offering a 20-second maximum delay time. **Modulation**: Both devices offer delay time modulation, with the El Capistan providing more extensive and nuanced modulation capabilities, allowing for a wide range of effects. **Control**: The SDE-2500 features traditional knobs, buttons, and an LCD for control. The El Capistan has a primary control layout with secondary knob functions, offering a unique approach to control. **Connectivity**: Both devices have stereo input and output. The SDE-2500 has separate send and return for external effects, and a footswitch input. The El Capistan includes an expression pedal input for continuous control. **Memory**: Only the SDE-2500 has memory functionality with 64 locations for storing and recalling delay settings. **Size, weight, and compatibility**: The SDE-2500 is a rackmount processor, larger and heavier than the El Capistan, a stompbox-style pedal, providing portability. Both devices are compatible with various audio equipment. **Sound quality**: Both devices offer high-quality sound with clarity and detail. The SDE-2500 uses a 16-bit digital audio processor, while the El Capistan utilizes a 24-bit processor for potentially higher quality and more detailed sound. **Ease of use**: The SDE-2500 and El Capistan are designed with ease of use in mind. The SDE-2500 features a simple layout and preset modes. The El Capistan has a compact interface, tap tempo input, and comprehensive controls for versatile delay effects.

4. IMPLEMENTATION

4.1 Software design

The software framework for the Daisy Dub project is primarily written in Gen by Cycling '74, an add-on for their visual programming language, Max. Gen allows for creating custom digital signal processing algorithms and is an intuitive way to design new audio effects (see fig. 3). The visual interface makes it easy to understand how signals are routed and processed, and the real-time preview allows for rapid iteration and fine-tuning of the algorithm. Integrating the gen code with the Daisy Dub hardware gives users a powerful and flexible platform for creating custom audio effects (see fig. 6). The Oopsy package from Electrosmith is another key tool for implementing custom algorithms on the Daisy Dub hardware. It is a Max package that allows the creation and flashing of custom firmware. With Oopsy, users can easily create new effects using gen code and deploy them to the hardware streamlined and efficient. The package includes a library of pre-built objects and examples to help jumpstart the development process, which we have further tweaked to match our custom hard-

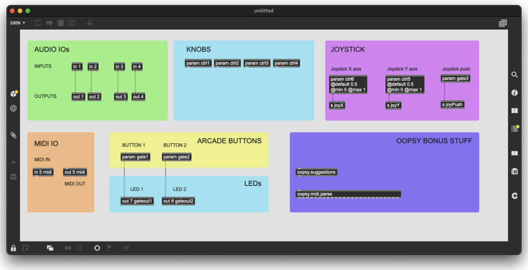


Figure 2: Oopsy Daisy Dub clean template.

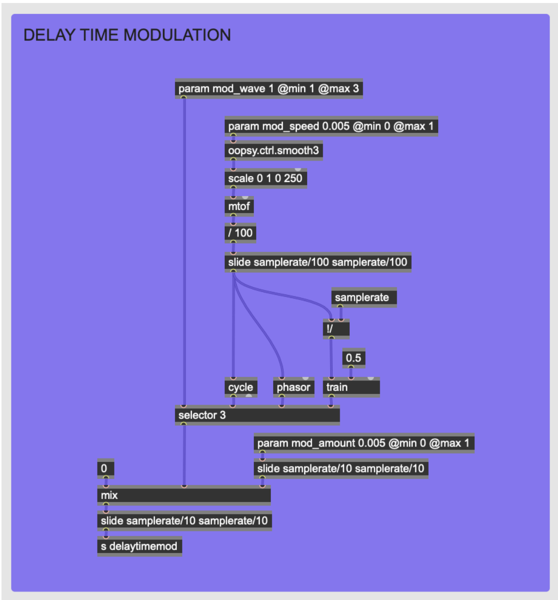


Figure 3: Delay time modulation section of the gen patch.

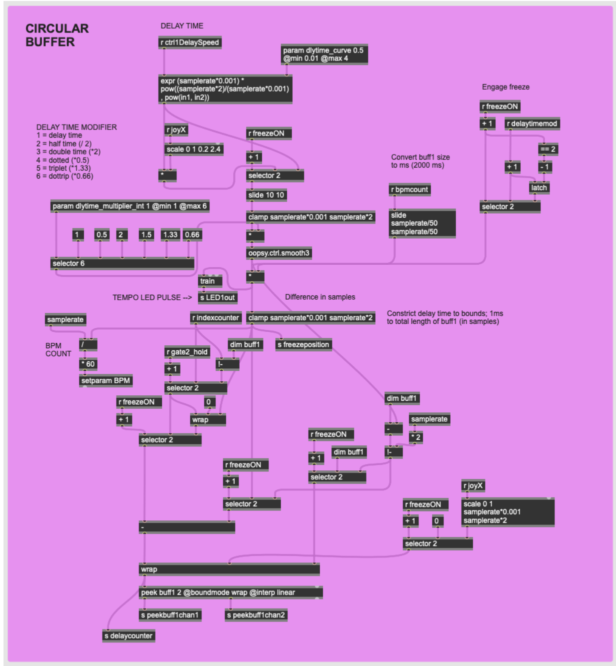


Figure 4: Circular buffer delay in gen.

ware (see fig. 2). Integrating gen and Oopsy provides a complete and flexible solution for developing custom effects for the Daisy Dub. It is possible with further software tweaks, such as custom JSON mapping of IOs and OLED display hacks. The visual programming interface of gen allows for rapid prototyping and fine-tuning of the effect algorithm. At the same time, Oopsy provides a robust and reliable way to deploy the custom effect to the hardware.

4.2 Daisy Seed by Electrosmith

The Daisy Seed platform is a 32-bit open-source audio development board with a strong focus on audio processing and the ability to handle real-time processing. It is programmable in C++, Arduino, Max/MSP Gen, Pure Data, and Rust, leaving plenty of options for both paid programming environments and open-source. By utilizing an open-source platform, developers can customize and tweak the firmware as needed rather than being limited by proprietary restrictions. The Daisy Seed platform features a powerful ARM Cortex-M7 MCU, running at 480MHz, capable of handling high-resolution audio processing with minimal latency. This makes it ideal for real-time audio effects such as Daisy Dub, which requires rapid processing to produce seamless, natural-sounding effects and delays. It includes a 24-bit analogue-to-digital converter (ADC) and a 24-bit digital-to-analogue converter (DAC) at 96kHz (AC-Coupled).

4.3 Audio algorithms

The Digital Signal Processing (DSP) of Daisy Dub is centred around a circular buffer delay (see fig. 4), allowing for the creation of delay and echo effects in real-time. It works by storing a certain amount of audio samples in a circular buffer and continuing reading and writing to this buffer at different points in the audio signal, overwriting the oldest data with new data as it becomes available. Using a circular buffer for delay effects allows for low-latency processing, as the delay time can be changed per sample. This makes it well-suited for live performance settings where delay times may need to be adjusted quickly and accurately, especially when exploring creative uses of aggressive feedback, self-oscillation, and feedback-path signal processing. Adding in parameters to manipulate both the delay time and feedback amount allows for creative exploration of sounds, which can be used to create delay, echo, or self-oscillation. Between writing and reading the sample, a series of effects are serially connected; a state variable filter (SVF) (see fig. 7), waveshaping and a phaser. Implementing such in the feedback path of a delay can greatly enhance the creative potential of the effect (see figure 6). Dealing with a complex setup of effects and routing adds many control parameters to the system. To maintain the simplistic and performative design of Daisy Dub, all non-essential parameters are accessible via a menu on the OLED screen (see fig. 5). The screen doubles as a spectrum visualizer, mostly for eye candy and engagement, but also to inspect the audio inputs and outputs. The control scheme for the dub delay patch is seen in figure 9.

Figure 5: Menu system with user parameters accessible via OLED and encoder.

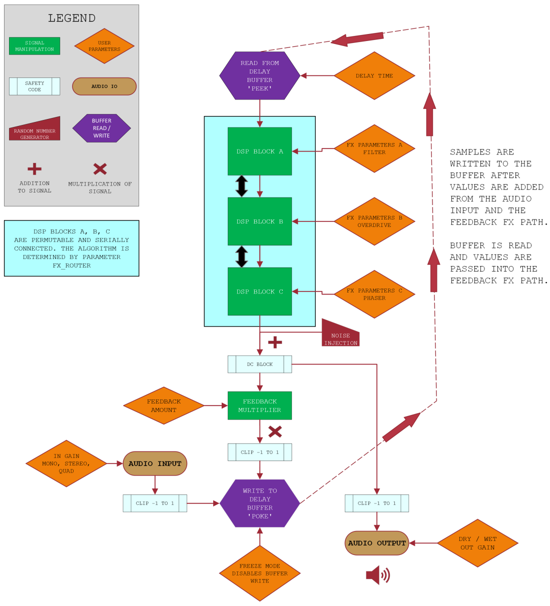


Figure 6: Daisy Dub DSP flowchart for a single audio channel. The circuit is copied for the other input channels. The effect blocks A, B, C are serially connected and interchangeable via a parameter (fx_router), changing the order of the effects; filter, waveshaping, phaser.

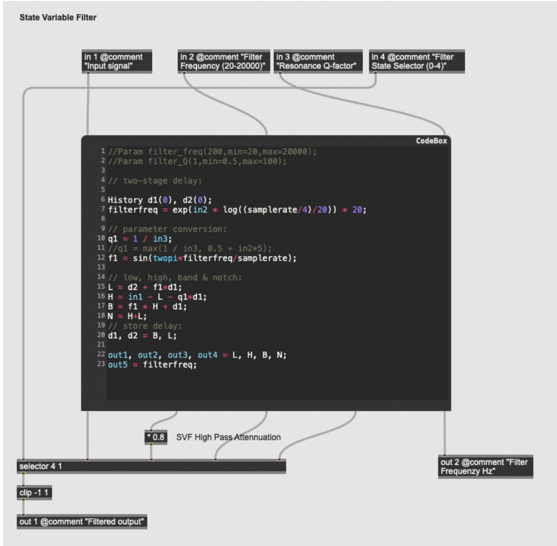


Figure 7: State Variable Filter gen code.

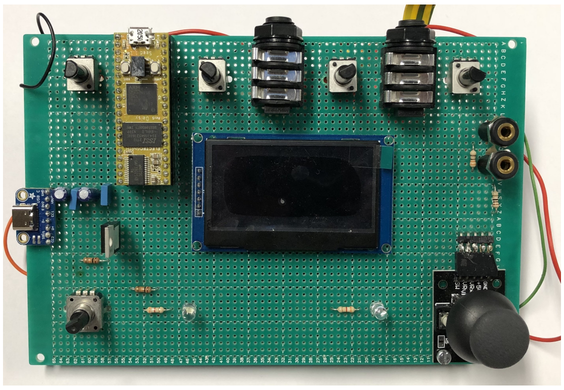


Figure 8: Daisy Dub prototype, testing of ergonomics and evaluating analogue circuits.

4.4 Electrical design

This section will describe the work and the troubles encountered while implementing certain hardware aspects of the Daisy Dub. Components and circuits integrated into the device were tested on breadboards and prototyping boards. Figure 8 shows one of the prototypes. The components and circuits used required 3.3V and 5V, and an estimate of the current needed for the device is 150mA.

4.4.1 USB Type-C port

The decision to power the Daisy Dub device through a USB Type-C port was motivated by the need to conform to contemporary technology standards and enhance user convenience by providing a standardized port. USB Type-C supports power delivery up to 48V/240W and rapid data transfer. During the process of integrating the desired port, it was discovered that the data pins (D- and D+) exposed on the Daisy Seed development board are OTG (On-The-Go) pins which only allow it to be used as a USB host and not as a USB device, which was the intention for the Daisy Dub. Considering these limitations, the developed prototype incorporates a USB Type-C port for power delivery and utilizes the onboard micro USB port for data communication. This configuration should suffice until a more permanent solution is achievable since the onboard connection is only necessary when flashing firmware updates to the Daisy Seed board. Since the device needs to be powered by 5V, a simple resistor method is used where an upstream-facing port must connect a valid pull-down resistor to ground to both CC1 and CC2 pins [17]. The value of the resistors must be 5.1kΩ for the port to provide 5V and up to 3A.

4.4.2 PCM3060 audio codec

The Daisy Seed hardware platform provides only two audio input and output channels, necessitating an external audio codec to add full quadrophonic audio capabilities. Texas Instrument's PCM3060 audio codec is selected as the preferred codec, a 24-bit asynchronous stereo codec with a 96/192kHz sampling rate. A 2-wire I2C communication protocol facilitates the codec's configuration and control, while clocks and audio signals are transferred via the Serial Audio Interface (SAI).

4.4.3 User interface

A diagram of the Daisy Dub user interface is seen in figure 9. It includes four potentiometers, a clickable rotary encoder, a clickable XY-axis thumb joystick and two arcade buttons connected to the analogue and digital input pins on the Daisy Seed board. Furthermore, it includes two LEDs (one for delay tempo blinks and the other for state changes). A 2.42" OLED display with an SSD1309 driver rests in the middle of the device. The LEDs are connected to digital pins on the Daisy Seed board while the OLED display communicates with the board over Serial Peripheral Interface (SPI) bus protocol. Scaling and mapping of each control parameter is done in the software. An example is delay time which is scaled with the following equation:
$expr(samplerate * 0.001) * pow((samplerate * 2)/(samplerate * 0.001), pow(in1, in2))$.

The potentiometers are linear, and the resistance values are translated to 32-bit floating point numbers between 0 and 1 in gen.

4.5 Custom-made analogue circuits

To enhance the device's audio capabilities, an external codec, the PCM3060, was necessary. This enabled the device with two extra audio inputs and outputs. Several analogue circuits were designed to ensure compatibility between the Daisy Seed microcontroller and the PCM3060 codec to guarantee high audio quality. These circuits consisted of a bias generator, a DSE (differential to single-ended) conversion circuit, and an input buffer, all working together in a chain to provide clean and accurate audio signals.

Bias generator

A bias generator is a device that separates an input voltage into two equal halves and distributes them as two biases (see fig. 10). This enables the DC offsetting of an input signal, allowing for manipulation of the signal's voltage level. Since a dual-operational amplifier was used, the cir-

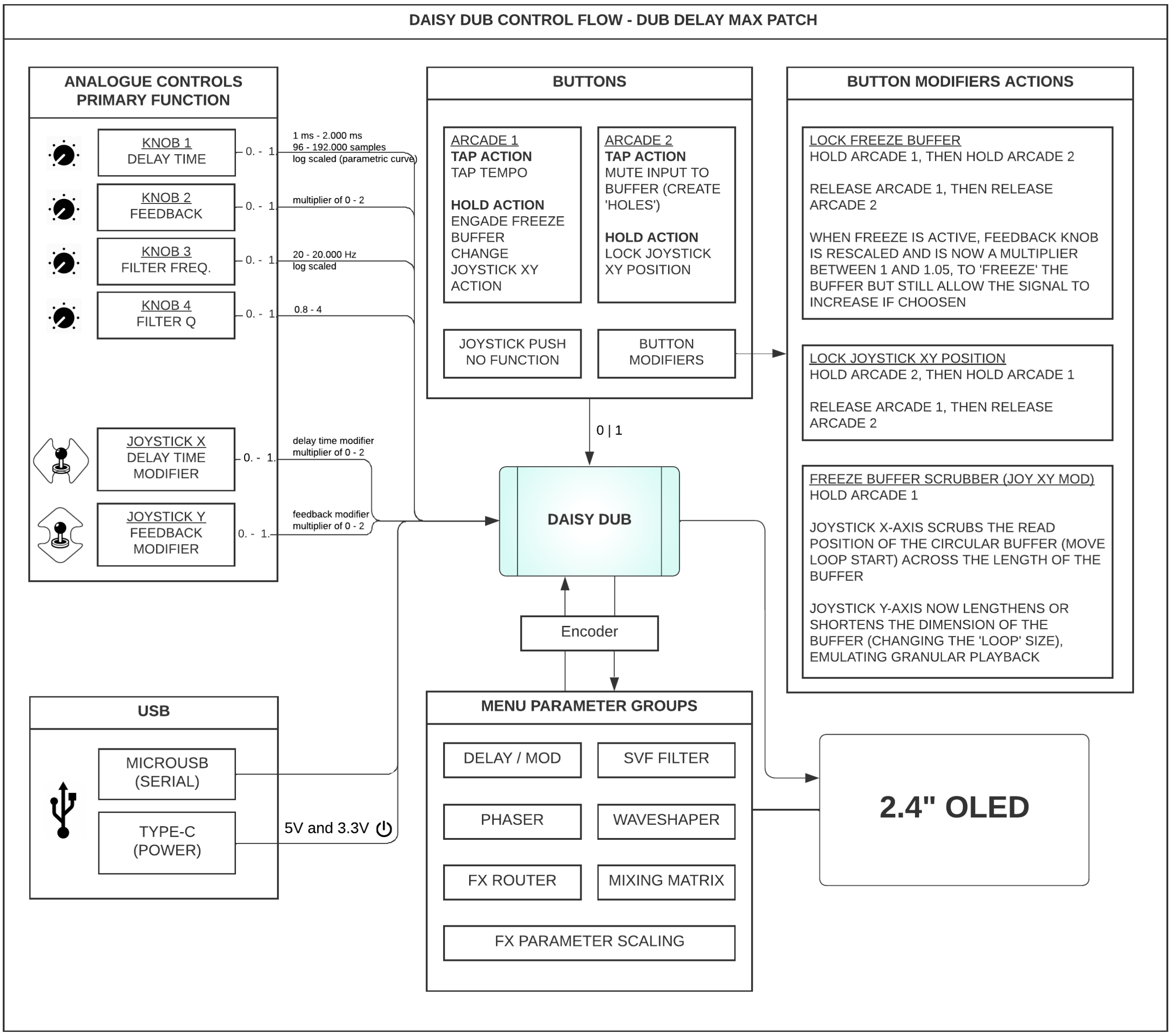


Figure 9: Daisy Dub delay patch - control flow diagram with scaling and features.

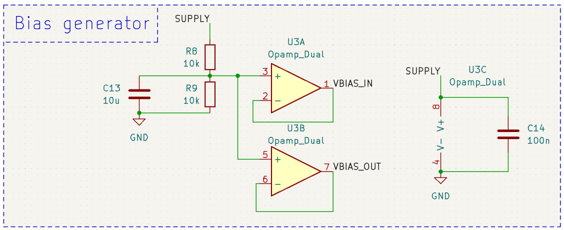


Figure 10: Bias generator electrical diagram.

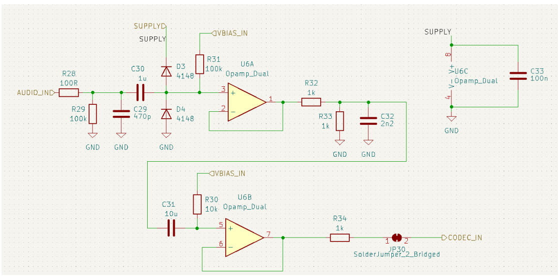


Figure 11: Audio input buffer electrical diagram.

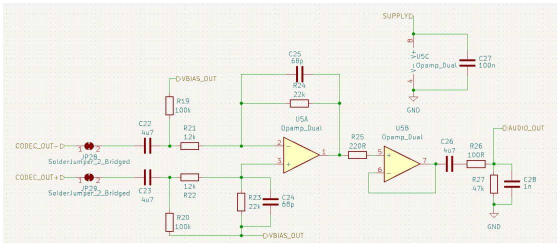


Figure 12: Daisy Seed differential to single-ended (DSE) conversion.

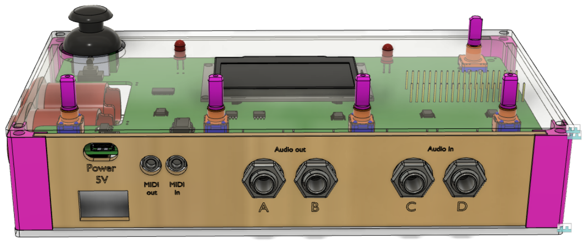


Figure 13: Daisy Dub backside with connectors.

cuit has two offsets to increase channel separation. Ideally, there should be a bias voltage for each input and output.

Input buffer

An input buffer is a crucial component in an audio signal chain (see fig. 11). Its primary function is to isolate the circuits in terms of impedance. Additionally, the input buffer helps to eliminate any DC offset that may be present in the input signal. This is accomplished by filtering the signal and creating a buffer that separates the input from any DC offset that may be present. The result is a clean, stable audio signal that is ready for processing by the codec.

Daisy Seed input buffer

Similar to the previous input buffer, this buffer is placed between the input jack and the Daisy Seed onboard ADC. It filters the incoming signal to reduce potential noise and remove any DC offset to maintain the signal quality transmitted to the Daisy Seed.

Codec DSE conversion

This circuit converts a differential signal, consisting of a positive and a negative component, into a single-ended signal by combining the positive and negative components into a single signal. The circuit eliminates any common-mode component that may be present in the differential signal, ensuring that only the differential component of the signal is present in the single-ended output.

Daisy Seed DSE conversion

This circuit converts the output signal from the Daisy Seed's onboard DAC into a single-ended output by combining it with one of the bias outputs (see fig. 12).

4.6 Case design

The case design is based on the cardboard box from the initial paper [6]. Modelling was done using Fusion360 and encompasses custom-made 3D printed corner profiles which double as a PCB rest and the schematics for laser cutting the case top plate, bottom plate and side. The backside accommodates the additional audio inputs and outputs, feeding the single-ended signals from the Daisy Seed and the PCM3060 codec to the audio jacks. Two separate audio channels are routed to either channel of a stereo jack connector, two stereo jack connectors per audio input and output section. See figure 13.

As the Daisy Dub is intended to be a DIY project, the case should be simple to assemble and also simple to swap out materials for the case if that is desired. With this in mind, it was decided that the sides of the case should be all straight, as this would be a lot easier for most people to source rather than curve material in any way to build the case. This decision means that the corner connecting pieces have to be able to accommodate all of the sides, as well as provide a place to secure the PCB.

5. METHODOLOGY

5.1 Usability testing

The study combined qualitative and quantitative data collection methods. The approach consisted of a usability test and a questionnaire to evaluate the device's performance in real-world settings and gain generalizable insights into user satisfaction and ease of use. The study included a standardized 10-point Likert-scale System Usability Scale (SUS) with a maximum score of 100 to assess the perceived usability of the device [18].

5.2 Methods

The usability test was conducted by two researchers, one acting as the interviewer and the other taking notes. The participants' ages ranged from 25 to 40, and they all had experience using electronic musical instruments and devices. The tests took place at the electronic music school and artist community, Rumkraft. After a few minutes of familiarizing themselves with the device, the participants were asked to complete a set of tasks using the Daisy Dub in combination with a regular two-octave MIDI keyboard in Ableton Live. The tasks evaluated the device's ability to manipulate audio in real-time and playfully enhance interaction and improvisation. The tasks included changing and adjusting different parameters of the audio algorithm: the delay time, feedback amount, filter type and cutoff frequency, and more. Participants were asked to provide verbal feedback during the test and complete a post-test questionnaire. No more than five people were tested as more testing would be a poor return on investment [19]. Testing five people should uncover most of our usability issues without superfluous testing. The questionnaire consisted of open-ended and closed-ended questions and lasted about 20 minutes per participant. After the participants completed their questionnaires on a computer, the authors conducted a series of qualitative questions to evaluate the performance and operability of the device. The participants were encouraged to speak freely, and the researcher not interviewing took detailed notes while the other researcher asked follow-up questions to gain a deeper understanding of their responses. They were asked if the device reacted as expected, how well they could change a desired parameter, how much they felt in control of the sound, their proficiency in modifying desired parameters, perceived control over the output, and the overall sound quality. Furthermore, participants were prompted to identify any interactions that they found inconvenient and their likelihood of using a product like this.

5.3 Results

Overall, the usability test yielded good results and gave great feedback. The average SUS score of five participants was 79.5, considered an above-average score and can be interpreted as a grade of a B+ [20]. The responses to the questionnaire indicated the need for a visual upgrade of the menu. Issues such as confusing names, an insufficient separation between variables, and difficulty in detecting that the menu extended beyond a certain point were reported by subjects. A visual upgrade of the menu is necessary to create a clearer separation between controls. Subjects also expressed a need for easier control over different features, with the unit's overall dry/wet control being the most frequently mentioned. They also suggested making it easier to manipulate the delay time, for instance, by halving it or putting it into triplet or dotted notes of the current position. These adjustments are considered valuable, and work will be done to expand the range of controllable parameters and to find a solution for altering the currently selected delay time. Subjects reported difficulty in accurately dialling in the delay time due to the extensive range of the control. While this can result in a less quantized sound, a solution will be sought to retain this while allowing for finer adjustments. Finally, subjects suggested adding a bypass control in the form of a button that would allow the signal to pass through the delay only when held. This would allow for a faster bypassing, mainly when using the delay as a send effect. Subjects raised most of these concerns during the exit interview, and their difficulties navigating the menu were evident throughout the testing.

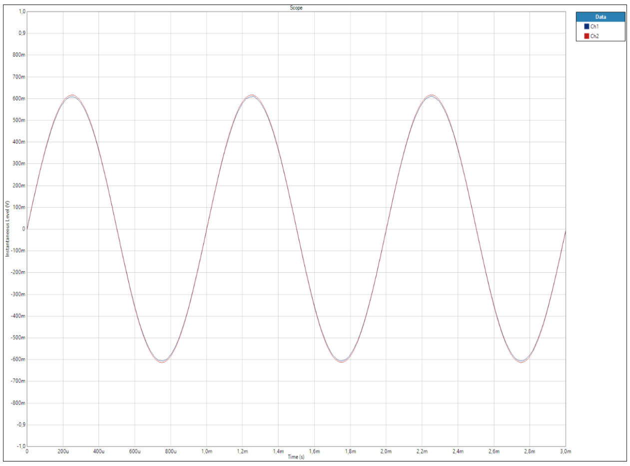
The audio capabilities and evaluation were performed at Sound Hub Denmark. The results are in figure 14. We observe a decent signal-to-noise ratio and correct creation of a pure 1kHz sine wave tone.

6. DISCUSSION AND CONCLUSION

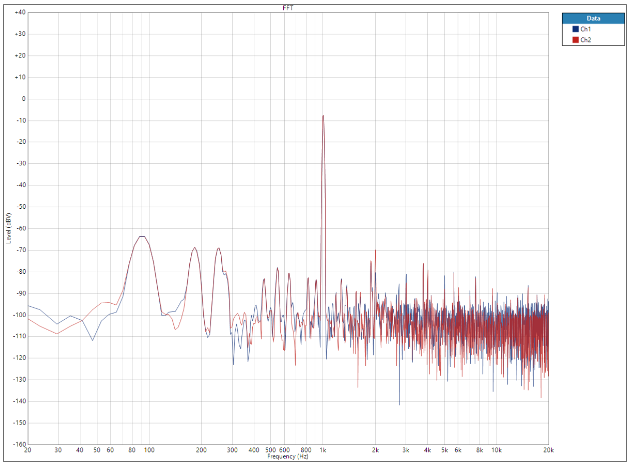
This project aimed to develop a customizable and updateable performance-oriented real-time audio effect ready for production. The project was guided by the state of the art analysis and a usability test to ensure that the resulting product would meet users' needs. The usability test results indicated that several areas of the device could be improved to enhance the user experience. The most pressing issue identified was a need for a visual overhaul of the menu, as participants had comments or complaints about different aspects of the menu. The analysis showed that there is a demand for audio effects that are customizable and updateable, as these features allow users to adapt the effect to their specific needs and keep it up-to-date with the latest technology. Additionally, the analysis revealed that users highly value performance-oriented audio effects, enabling musicians and producers to achieve their desired sound in real-time. The ease of use and intuitive interfaces are important for audio effects, as users need to quickly and easily access and control the parameters that matter to them. The findings from the usability test confirmed these trends, as several participants highlighted the importance of easy control over the audio effect parameters. The results of the usability test and the state of the art analysis provided valuable guidance for developing the audio effect, helping to ensure that the final product will meet the user's needs.

We will be offering the device as an assemble-yourself kit catering to electronics enthusiasts interested in exploring the inner workings of audio effects and gaining hands-on experience with electronics. Furthermore, it would also provide a valuable learning opportunity for individuals pursuing STEAM education, as they could apply their knowledge of electronics and audio processing to the construction and operation of the device. This will include a detailed manual with the kit allowing individuals to understand each component and its purpose while also serving as a practical guide to assembly. The manual will also highlight important aspects of audio processing and real-time audio effects, providing a valuable educational experience, including the PCB and components as an assemble-yourself kit has the potential to provide valuable educational experiences for electronics enthusiasts and individuals pursuing STEAM education while also providing a unique and customizable real-time audio effect for performance and production purposes.

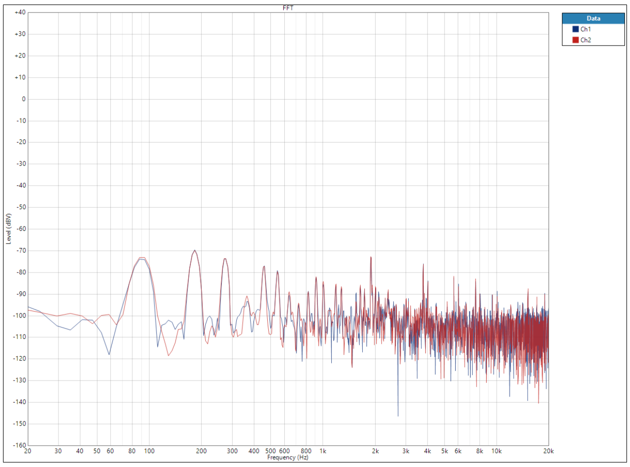
Acknowledgments



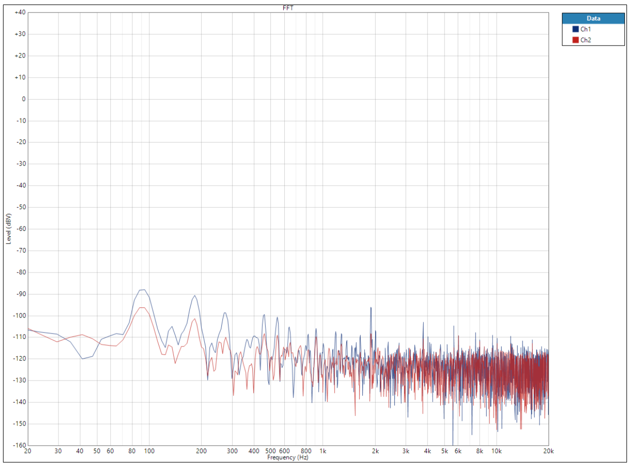
(a) Clean DSP algorithm with PCB input 1 routed directly to PCB output 1. Similar routing for input and output 2. Scope visualization of 1kHz pure sine wave fed to PCB audio inputs 1 and 2 and measuring the resulting output from PCB audio channels 1 and 2 (Daisy output A).



(b) Clean DSP algorithm with PCB input 1 routed directly to PCB output 1. Similar routing for input and output 2. FFT visualization of 1kHz pure sine wave fed to PCB audio inputs 1 and 2 and measuring the resulting output from PCB audio channels 1 and 2 (Daisy output A).



(c) Self-noise measurements with PCB input 1 and 2 shorted directly to ground. Clean DSP algorithm with PCB input 1 routed directly to PCB output 1. Similar routing for input and output 2. FFT measurements of PCB output 1 and 2.



(d) Self-noise measurements with PCB input 1 and 2 shorted directly to ground. DSP algorithm bypasses input 1 and input 2 (Daisy input C), so the measurement is independent of the input circuits and AD conversion. Output 1 and 2 outputs zeroes. FFT measurements of PCB output 1 and 2 (Daisy output A) FFT channel 1 and 2.

Figure 14: PCB noise measurements were done with an Audio Precision APx517 acoustic audio analyzer at Sound Hub Denmark. FFT and scope visualisations presented. All DSP code is run at 96.000 Hz sample rate with Oopsy BOOST ON and FASTMATH OFF. Pass-thru patch with no FX.

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