The G9 Audio Processor

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https://github.com/stevecorey/g9

***Abstract*—In a recording studio, fully analog processors are well-loved for how they can alter the recorded audio. These audio processors have characteristics that change sound to feel thicker, warmer, or airier among other effects. While the latency of analog processors is minimal, their controls are manual, making them difficult to use in a computer-based environment. Digital signal processors (DSPs) provide easier control as well as the ability to save and restore the state of those controls, however DSPs greatly increase the latency of the system. To utilize the best of both types of signal processors, we are creating an all-analog processing device under digital control such that every parameter the user can control is storable, automatable, and recallable. This connection between analog processing and digital control can be made by replacing manually operated potentiometers with digitally controlled potentiometer ICs. The user interface will run on a separate computer, sending parameters to the analog processor. It will have modules for equalization, dynamics, harmonic generation, and ambience to cover the needs of most audio engineering tasks and will give sound engineers the ability to easily use presets and remotely control the device.**

***Index Terms*—analog, audio, digital, filter, reverb, signal processing.**

I. Introduction

RECORDING engineers use a wide variety of audio signal processors in recording studios to achieve the sounds they desire. Digital processors offer a flexible and economical method to achieve virtually any effect imaginable. The problem with digital signal processing is the signals latency: digital processing requires converting an analog signal to a digital bitstream.

If two digital processors have different latencies, using them in parallel can result in the severe comb filtering caused when summing two signals with a different delay. Depending on the amount of processing desired, the final latency of a serial chain of digital processors can be in the hundreds of milliseconds. That might not sound like a lot, but musicians and audio engineers are sensitive to even the slightest delays. In order to keep latency to a minimum, an all analog signal path for the audio must be maintained, but this means giving up digital control.

Analog audio signal processing has extremely low latency on the order of nanoseconds, and it is well-loved by recording engineers for the character of the sound that can be obtained. Vacuum tube circuits in particular are loved for the distortion effect they produce when the voltage of the signal is amplified beyond the circuits linear zone. Integrating analog signal processors into a modern digital recording studios workflow becomes difficult because of the manual controls for each processor. During a recording session, the parameters of a recording studios processing chain need to be saved and recalled at a moments notice. The G9 project takes a hybrid analog/digital approach. It will maintain an all analog audio signal path, but the adjustable processing parameters will be digitally controlled. Using the strengths of each method, the G9 project will overcome the weaknesses of the other.

II. Background

In the past, there have been some audio processors with all-analog audio signal paths under some sort of digital control. A few notable examples are as follows.

A. Harrison Series 10

The Harrison Series 10 was the first mixing console to have digital control over its analog signal path. It included EQs and compressors on each of its mixing channels, but did not include reverb. It was a very expensive system, costing over $500,000 in 1985 for the base system and is no longer in production [1].

B. Total Recall

The company Neve made a digital storage and recall system for their analog signal processors called "Total Recall." It worked by storing the position of each control in a computer. Then to "recall" the parameters, the computer would display the stored position of each control and the user would manually adjust the control until it matched what was on the computer's display [2].

C. Flying Faders ™

Systems for using motors to control the knobs and sliders on a processing device were developed beginning in 1973 [3]. Some companies built in motor control from the beginning of the processor's lifecycle, and other companies built add-on systems that could retrofit existing processors with motor control. The term "Flying Faders™" is applied to this sort of system, and the term is now trademarked by Martin Sound [4].

D. Hybrid Synthesizers

Outside the realm of processing already existing audio, there are analog synthesizers that create audio with analog circuits. Synthesizers that have digital control over analog sound generators are known as hybrid synths, and are currently quite popular among electronic musicians for their sound qualities and ease of control. Figure 1 shows an example of a hybrid synth. Many of these synths also offer processing options of already existing audio.

Hybrid synths demonstrate the success of the digital-control-over-analog-audio-circuitry approach. They are readily integrated into computer-based recording studios. The G9 processor takes this hybrid approach to sound generation and applies it to sound processing.

Fig. 1. Arturia Microbrute analog synthesizer with digital controls [5].

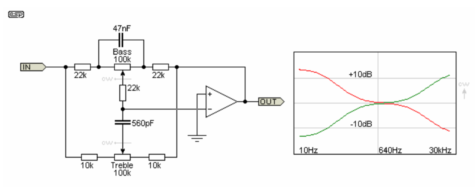

Fig. 2. Baxandall filter circuit and frequency response [6].

III. Project Scope

Because most recording engineers use equalization (EQ), amplitude dynamic range compression (compressor), and ambience (reverb), the G9 will have each of those modules.

A. EQ

The EQ in the G9 will have high and low shelving filters based on a Baxandall circuit; Figure 2 shows the circuit and its frequency response. It will also have a parametric midrange boost/cut with adjustable frequency.

B. Compressor

The compressor will be based on Teletronix LA-2A vacuum tube leveling amplifier. This circuit has two main elements that give it its characteristic sound. First, a gain element is controlled by a combination of an electroluminescent panel and a photo-resistor. The non-linear response curve of that combination results in a very pleasing sound of the gain reduction. Second, tubes distort the signal in a musical way when the signal level pushes them out of their linear operating zone. Figure 3 shows the operating principle of the compressor.

Drip Electronics has developed an updated version of the LA-2A compressor and sells a PCB of their design as well as a PCB for the power supply of the compressor. We already have these PCBs (figures 4 and 5) and now need to acquire the parts and assemble the unit.

C. Reverb

Reverb options are limited in the analog domain. The favored option is to place a loudspeaker in one part of a reverberant chamber and place a microphone in another part. When sound is played through the loudspeaker it is reverberated by the chamber and picked up by the microphone.

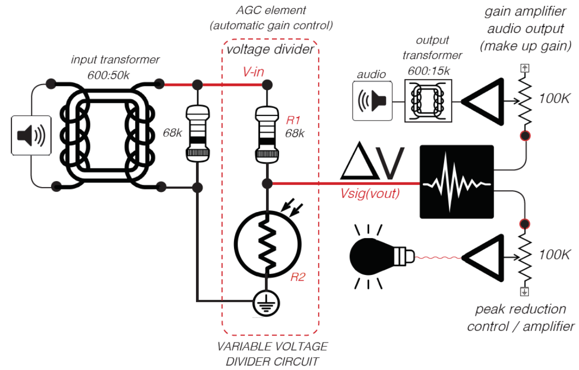

Fig. 3. Operating principle of the compressor's optical attenuator [7].

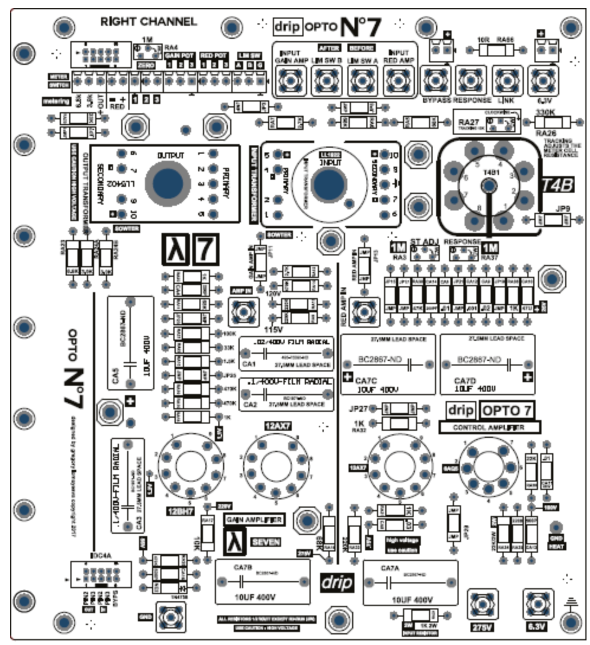

Fig. 4. PCB of the compressor circuit [7].

Reverb of this sort is not feasibly placed inside an enclosure for use in a studio.

The sound of reverb is generally divided into two parts: early reflections and late reflections [8]. Early reflections are heard as discrete echos, as when sound bounces off the walls of a chamber. Late reflections are heard after the early reflections as the early reflections build up and fuse into a wash of sound. This part of the reverb is also known as the tail. The time it takes for the tail to fade by 60 dB below the initial sound level is specified as the reverb time. The G9 will combine

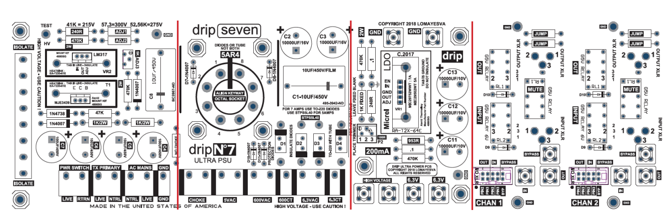

Fig. 5. PCB of the compressor's power supply [7].

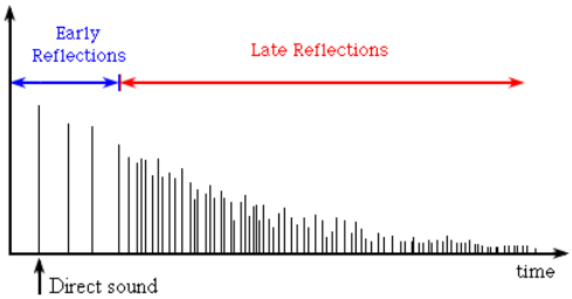

Fig. 6. Impulse response of a typical reverb signal [9].

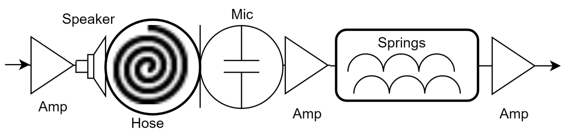

Fig. 7. Diagram of the reverb section

two methods to achieve a simulation of the chamber's early reflections and reverb tail.

A hose delay will be used for the early reflections. An amplifier will drive the audio signal through a balanced armature driver, like those used by in-ear monitors. The driver will be coupled to one end of a thin hose or tube. A microphone will be coupled to the other end. Sound played at one end will travel through the hose and be picked up by an electret microphone at the other end. Feedback can be used for sound reflections at a multiple of the delay time; multiple hose rigs can be used for specific reflections.

An amplifier will take the signal from the microphone and drive it through a pair of springs to generate the dense, diffuse reflections that happen in a reverberant chamber after the sharp early reflections decay into a wash of sound. Figure 7 shows a diagram of the reverb section of the G9. Amplifier schematics for the reverb are shown in figure 8.

D. Digital Control

A graphical user interface (GUI) will run on a separate computer to send control signals to the G9's processing blocks. The computer will maintain the state of the system and be able to store and recall the state at any time. Figure 9 shows a mockup of the G9's GUI.

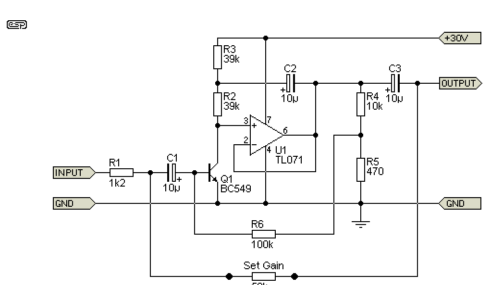

Fig. 8. Reverb amplifiers [10].

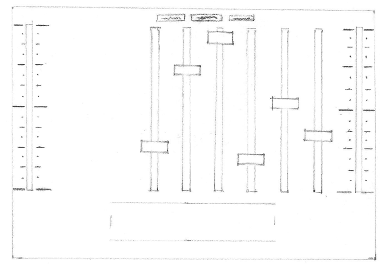

Fig. 9. Mockup of the G9's GUI.

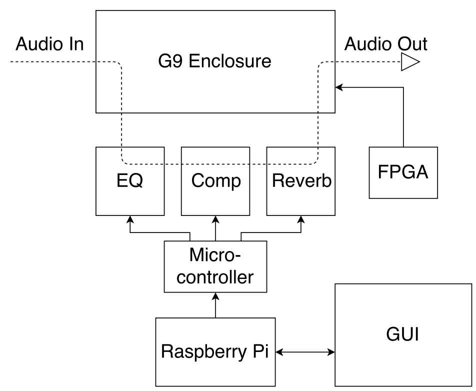

Fig. 10. Block diagram of the G9.

IV. Design Approach

The G9 will utilize digital potentiometer ICs to control the parameters of its signal processors. These ICs will be communicated to through I^2C or another serial protocol. An FPGA will provide processing for a graphical display separate from the GUI. The graphical display will show information about the sound level and frequency spectrum of the output audio signal itself. The GUI will have slider controls and show the state of the G9. A Raspberry Pi and will act as the CPU and take control signals from the GUI and route them to the appropriate digital potentiometers. Figure 10 shows a block diagram of the G9.

Figure 11 shows a mockup of the G9's enclosure. It will have LED meters showing input and output signal levels, an LED meter showing the compressor's gain reduction, and an LED meter showing the reverb's wet/dry signal balance. It will also have two rotary encoders to adjust the signal level going into the processor and the signal level coming out of the processor. The rotary encoders will allow for the changes to be saved with the GUI as well as for the state to be loaded easily.

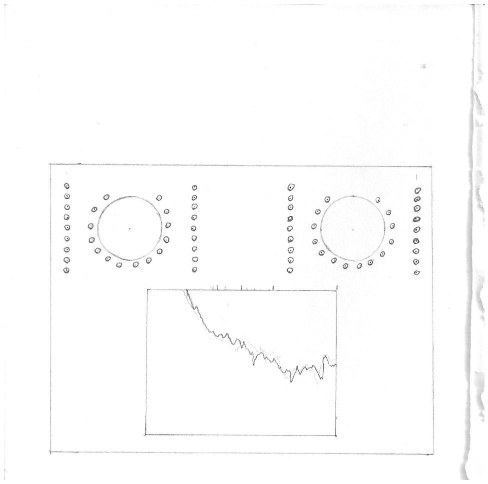

Fig. 11. Mockup of the G9's enclosure.

V. TASKS

This is a list of the parts the project that will need to be completed.

- EQ circuit
- Compressor circuit
- Reverb circuit
    1) Speaker into tube
    2) Microphone out of tube, piezo driver into spring
    3) Pickup out of spring to be mixed into main signal
- Digital control of the processors
- GUI
- Control parameters from GUI to CPU
- Control parameters from CPU to control signal generator
- LED meters on the G9 enclosure
- Rotary encoders and LEDs on the G9 enclosure
- FPGA analysis of the audio signal for display
- FPGA display driver

VI. TESTING PLAN

The G9 is highly modular. Each piece can be tested on its own to ensure it functions correctly before attempting to combine all the elements together. GUI, CPU, FPGA, display, EQ, compressor, reverb will all be tested on their own. When each component is functioning correctly on its own, then the pieces will be connected together one-by-one and tested.

VII. DEMO

A base demo of the G9 will consist of a sound playback device being plugged into the G9, and the output of the G9 will be plugged into a sound system. Each of the three sound processors will be demonstrated in turn by adjusting their parameters one by one on the GUI and listening to how the sound changes on the playback system. A more detailed demo will consist of a more in-depth presentation of how the GUI, control system, analog processors, and information display all work together.

VIII. CONCLUSION

Digital audio signal processors have processing latencies on the order of milliseconds. When multiple processors are used in recording studios, their interacting latencies can cause comb filtering or lengthy delays in the signals being processed. Analog audio processors have latencies on the order of nanoseconds and also are often preferred for the character of the sounds they produce. The problem with analog processors is that their controls are often manually operated potentiometers and switches. Some companies in the past have created products that take a hybrid analog/digital approach, but currently no product is available that maintains a fully analog audio signal path incorporating the main processing blocks desired by sound engineers. The G9 processor has the three main processing blocks of EQ, compressor, and reverb, and takes advantage of the low latency and sound quality of analog processing with the convenience and repeatability of digital control.

REFERENCES

[1] Wikipedia contributors. Harrison audio consoles — Wikipedia, the free encyclopedia. [Online]. Available: https://en.wikipedia.org/wiki/Harrison_Audio_Consoles

[2] H. Robjohns. Neve 8803 dual-channel equalizer. [Online]. Available: https://www.soundonsound.com/reviews/neve-8803#para2

[3] An abbreviated history of api. [Online]. Available: http://apiaudio.com/about.php

[4] J. Martinson. Flying faders: The untold story. [Online]. Available: http://www.martinsound.com/ar_fly01.htm

[5] Arturia. Microbrute. [Online]. Available: https://www.arturia.com/products/hardware-synths/microbrute

[6] R. Elliott. Designing with opamps - part 2. [Online]. Available: http://sound.whsites.net/dwopa2.htm

[7] G. Lomayesva. Opto 7. [Online]. Available: https://www.dripelectronics.com/opto-seven/

[8] VOCAL Technologies, Ltd. Early reflections. [Online]. Available: https://www.vocal.com/dereverberation/early-reflections/

[9] Reverb. [Online]. Available: http://manual.audacityteam.org/man/reverb.html

[10] R. Elliott. Low noise microphone preamplifier. [Online]. Available: http://sound.whsites.net/project13.htm