G12 PedalVision

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***Abstract* — G12 PedalVision is an instrument multi effects system designed to be an alternate to the fully digital boards utilizing digital signal processing, and the pricey single analog effects on the market. Specifically the G12 PedalVision was designed to have the more desirable aspects of both analog and digital effects while offering versatility and expandability, while maintaining low weight and ease of portability. In addition to the multiple audio effects, the G12 PedalVision will have an led feedback matrix display that will allow for the audio signal to be translated to into a multitude of colors and blinking patterns depending on the current settings.**

***Index Terms* — Audio, digital signal processing, instrument, feedback, led, matrix**

I. Introduction

When it comes to music an individual's sound is everything. Musicians are constantly searching for the "best" sound they can find, whether that be with regards to a specific tone quality or cool sounding effect to enhance what is being played. Sometimes people want something to enhance a show's visual aspect and want the visual part of their performance to match their audio portion. With this project idea we aim to kill two birds with one stone by allowing the user an assortment of sounds, both digital and analog, and a light display that will react to the electrical signal created by the instrument playing.

As it stands currently, many musicians crave the "warm" sound of analog circuitry but must purchase each analog effect unit separately, thus creating a large foot print that the musician has to worry about with regards to mobility. On top of having to purchase each effect separately the price of each box is high in comparison to the cost of the components that make up the effect unit. The only alternative to paying these high prices is to switch to digital effects. On top of the reduced price of the digital effects, they offer more customizability in that they can be programmed to recall settings instantly. Although the cost of a digital effects unit with multiple sounds is relatively cheap as compared with the analog unit, many musicians feel that the digital effects are not as "organic" sounding or have a "fake" sound to them; therefore, the digital effects are seen as inferior.

With this project we aim to combine the best of both worlds by including both the "warm" tones of analog effects and the versatility of digital effects into one unit. This combination will help reduce the overall footprint of the multi effects unit and save the save the user money since they will only need to buy one unit as opposed to a different unit for each effect they want.

In addition to the audio effects, this project aims to enhance the user's visual performance as well. The multi effects board will be able to interface with an LED display, allowing the user multiple modes of operation for the display. The user could use the LED display to determine of the current pitch being played matches the musical pitch standard A440 (440 Hz), the musical note A above middle C used as a general standard for musical pitch, in order to ensure pleasant listening experience. The LED display could also strobe or change color depending on what the output of the user's instrument indicates.

Ultimately, the group's objective is to create a product that will have the capability to satisfy the modern musician's desire for analog effects, programmability and versatility of digital effects, and the visual appeal of the attachable LED display in order to create the ultimate performance tool.

II. Goals and objectives

With this project being an all-in-one unit we want to meet the needs of many different musicians and their playing styles. Many musicians want the classic sound quality of analog effects. Many players believe that the analog effects produce a warmer more "organic" tone. But since these are analog effects they are not easy to change. With this in mind, some musicians instead want the versatility of the digital effects. Although they do not produce the exact sound of the input, with today's technology the reproduced signal is getting closer to being reconstructed more accurately, making the digital effects sound closer to the analog effects. Due to this divide the project will have the best of both worlds by combining the analog and digital effects. The analog effects will be have effects such as overdrive and distortion which do not need to be changed frequently as they play more of a tone conditioning role in a musician's desired sound. While the digital effects, such as delay and reverberation, will be constantly changed depending on the tempo of the song and how much of the effects the musician wants to hear.

While performing the musician needs to put on a great show, with this the visual aspect comes into play. By incorporating a multi-functional LED display the performer can tune their instrument exactly by using the LED to indicate if the pitch of the instrument matches tuning standard of 440Hz. The Display can also be used as a fun display to help add dimension to the performance. A system block diagram is shown in Figure 1 below. With this product the musician will have the ability to produce classic sounds with modern technology, excellent versatility due to the digital effects, and an eye catching display to enhance their performances.

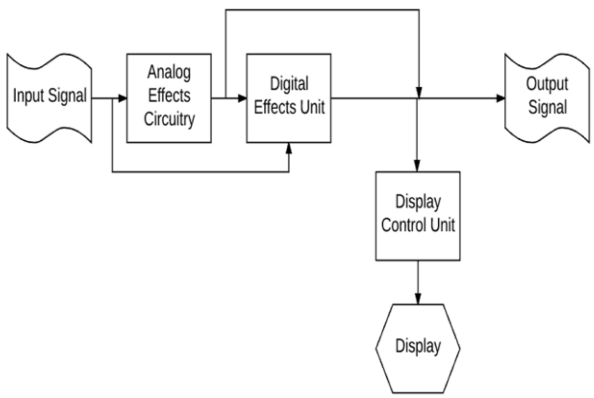

Fig. 1. System block diagram

III. SPECIFICATIONS

The design specifications the G12 PedalVision needed to adhere to were as follows:

Total System	
Description	**Specification**
Total Weight	Less than 30lbs
Total Size	Less than 15 $in3$
Price	Less than $500 total cost
Analog Effects	
Input Impedance	Minimum 500k
Output Impedance	10k
Bypass Frequency Response	20Hz-20KHz
Digital Effects	
Sampling Rate	Minimum 40kHz sampling rate
ADC/DAC Bit Resolution	Minimum 12 bit resolution
External Memory	512 Kbyte minimum
LED Feedback Matrix	
Number of LEDs	12
ADC Bit Resolution	12 bits
ADC Maximum Sample Rate	1MSpS

The G12 PedalVision will have the capability to create soft overdrive clipping, hard distortion clipping, many equalization filtering options and a compression/limiting circuit all through analog circuitry. The unit will also have the capability to create delay/echo effect, reverberation, and flange through the digital controller of the effects unit. A user programmable option will be available in order to allow for more flexibility and for possible expansions of the system. The addition of an LED display will also be incorporated which will be able to read the input provided by the effects output of from the output of the instrument itself in order to determine what action it will take with regards to what to display. All effects will be able to be toggled on and off through foot switches. Settings for the digital and analog effects can be adjusted by hand through knobs, while the digital effects and LED control will also be able to be adjusted through a connection through USB.

IV. ANALOG EFFECTS

For this project we will have multiple analog audio effects. The effects that will be included in this design are overdrive, distortion, compression, and equalization filter network. On top of the aforementioned effects, an adjustable input stage will be implemented as well.

Although we could have implemented many different types of effects, we decided to include these four effects since they are the most common effects used by the modern musician. Another reason we decided to implement these effects using analog circuitry is that they are more of a “set and forget” effect. The reason for this is that these effects deal more with the frequency aspect of the musician’s sound which is not changed frequently between songs unlike other effects like delay and reverb. With this in mind, we do not need to implement these digitally to allow for quick modification. Another reason to implement these effects using analog circuitry is that the output waveform will be exact as opposed to a digital implementation which will not have the same signal reconstructed exactly due to the effect of sampling and digital level approximations.

In order to activate each effect, there will be a toggle system for each effect that the user will be able to manipulate. This system will allow the user to have a single or multiple effects active at the same time through the use of multiple footswitches, one per effect to be exact. The foot switching system is crucial to this project due to the fact that the user may need to change effects quickly while playing his/her instrument and cannot use their hand to toggle the desired effect. The values of the many parameters such as gain, drive, tone, threshold, and volume will be controlled through many knobs the user will be able to adjust by hand on the effects unit. Having the ability to change the parameter values and toggling ability will allow for better user control of each of the analog effect.

a) Input Buffer

Although the input buffer may not seem like a must have effect in a signal chain it plays a very vital role in keeping your signal true to what you are playing. The purpose of the input buffer is to replicate what it sees at the input at the output of the circuit. This basic function allows the more accurately replicated signal to pass through the rest of the circuits in the signal processing chain. When implemented correctly, the input buffer will increase the high frequency response of the signal chain due to the capacitance of instrument cables, long lengths of the instrument cable, and other effects that may be present further down the signal chain. The key to a good input buffer is a high input impedance and a low output impedance.

b) Compression

Much like overdrive and distortion, compression is used to condition a musician’s sound in order to get a more desirable sound. But unlike the previous two effects, overdrive and distortion, compression in the traditional sense is supposed to be transparent. This effect detects the amplitude of the incoming signal and either reduces or increases the amplitude of the output signal in order to get more of a uniform amplitude for the output signal. One side effect of compression is the creation of more sustain due to the amplification of lower amplitude signals and the reduction of higher amplitude signals. Due to this change in amplitude of the input signal the effect creates the illusion of a very long steady pitch when in actuality you input signal is not very stead or as sustained. The way a compressor works is by amplifying small signals and attenuating large signals. This is done my having a system that can sample the output signal of the circuit and either amplifying or attenuate the signal.

c) Overdrive and Distortion

Since their creation, overdrive and distortion effects can be heard all over contemporary music. Overdrive is created when a signal begins to clip, thus flattening the peaks of the incoming signal. In order to create this effect in the 1950’s and 1960’s when guitar amplifiers were starting to become popular in popular music, the musician had to turn up the volume of their amplifier to very high levels in order to create this clipping effect. Although this did work, it was not pleasing to the ear to play at such loud volumes. Due to this problem, different ways of clipping one’s signal were created later on. These newer methods of clipping were done through the use of diodes clipping the signal or creating multiple gain stages with lower rail voltages. The other method of creating this effect is it to manipulate the operating point of the load line for transistor circuits. Although the two effects function very similarly, for the sake of this product overdrive will be considered as soft clipping while distortion will be considered as hard clipping. Soft vs Hard clipping is a descriptive of how flat the signal peaks become when view at the output.

d) Tone Stack

Equalization is an effect that can greatly shape an individual’s sound due to the fact that it is a network of filters that the user can adjust to their liking. The equalization pedal works by boosting or cutting certain frequencies or a frequency ranges in order to further shape the tone of the output signal. This method of boosting and cutting is done by passing the musician’s input signal through the desired filter. The various frequency curves can be created by adjusting the amount of attenuation around the cut off frequencies designed in the tone stack. To the large overall attenuation of traditional tone stacks, this section will be followed by an amplifier in order to get back some of the volume that was lost during the filtering process.

e) External Effects

Although the user will be able to accomplish many different effects with the G12 PedalVision’s analog and digital effects, they may want to add one of their own effects that they love or think would work well with this product. In order to for this ability to expand on this product there will be a capability to allow the user to decide if the output put of the analog effects will go directly to the next stages of the G12 PedalVision (such as the digital effects, LED module, or the output of the entire system) or to an external effects unit, then back into the G12 Pedal Vision for further processing or direct to the output. Since all of the DC biasing added to the signal within the G12 PedalVision will be removed before proceeding to the next stages, this added feature will not damage any external effects. In order to ensure a good quality sound a second input buffer will be used. The function of this second input buffer will work much like the input buffer at the very beginning of the G12 PedalVision's signal chain. This will provide a high input impedance and a low output impedance. This will help with any potential "tone sucking" some other pedals on the markets may exhibit on the user's tone quality.

V. DIGITAL EFFECTS

There are a few digital effects that our team is going to emulate for this project. There are many effects that we could have implemented using digital signal processing (DSP), however we decided to stick to only time based effects over frequency based effects. Because of this key feature of these effects, it makes sense to implement them digitally as opposed to with analog. Using DSP, it will allow for easier manipulation by the user.

We also would like to create an interface that will be used to communicate with the microcontroller and allow the user to change the settings of the digital effects, as well as turn them off if desired. In order to toggle effects the user should be able to easily cycle between each effect and use a foot switch in order to turn the digital effects on or off. The reason for the footswitch is to allow the user of this multi-effects pedal will need to be able to turn these effects on or off while on stage playing. Historically the way this is done is by having a footswitch on each separate pedal that can be pressed in order to enable or disable them. Since this product will include each effect in a single pedal enclosure, we will set up an interface to change the values of each effect, as well as a way to toggle each effect on or off.

Having the ability to change values and toggle will allow for better user control of each of the digital effects. On a normal effects pedal, the values for each parameter are set by turning a knob until the desired settings are made without knowing the exact values. With the ability to see the values change settings, the user will be able to choose more exact values for each effect. An example of this would be where the user will be able to choose the exact value for the potentiometer setting the feedback control of the delay effect. Having this detailed control of each effect will allow for the user to get the exact sound they are trying to create for their music using the multiple controls.

Digital effects have become much more popular in recent effects pedals. The output sound as a result is very similar to that of an analog effect pedal, but there are slight modifications to it before and after the DSP. In order to use DSP in signal modification, an analog to digital converter (ADC) must be used to take the analog input signal from the guitar, and turn it into something that a microcontroller or DSP chip can understand. This conversion is a vital part of digital effects pedals, and must be understood and done correctly before any code can be used effectively.

The time based digital effects that are implemented include delay/echo, reverb, and flanger. These effects all deal with a modification of the output to allow the original signal to be delayed slightly.

a) Delay/Echo

The delay effect, also known as echo, is a fairly simple effect. The best way to explain how it sounds is to compare it to an actual echo that might be produced by speaking in a large room, or cave. After the initial sound is produced there are recurring copies of the same sound that gradually get softer and softer. It also could seem like the sound is being repeated but every repetition it is further away from the point of origin. This effect is produced by the sound waves bouncing off of a wall or something similar and traveling back to the original location moments later. This produces more and more repetitions the more times it is able to bounce off of the walls. The sound originates from a source, then bounces off walls and other objects before reaching the receiver at a lesser intensity than it was originally sent out as.

This is a natural phenomenon that occurs any time a sound wave hits an object. The same idea of the effect is used in echolocation, where a signal is sent out and the reflected wave is measured. This allows data to be gathered about the object that is reflecting the sound. This process can also allow us to understand how to modify the sound digitally to make it sound as if the repetitions of sound are occurring from an object close by or further away. [1]

This effect is very simple and does not take very much computation. In order to create this effect a simple delay is needed. In order to create this delay a delay line function can be used. This is shown below.

- $y(n) = x(n) + d * x\,(n - M)$

The function takes in the original signal x(n), adds a delayed version of the signal to itself at a lesser intensity, and outputs as y(n). This function can be modified in order to create some of the other delay based effects.

b) Flanger

Flanger is an effect used in signal processing in order to create a unique output sound. The input sound is modified to create a sweeping effect. This effect has been used in signal processing since the mid 1900's and was originally discovered and used by Les Paul. Since computers were far from advanced at this time in history, the effect had to be created in a much different way than it is created today.

Originally the effect was achieved by using two tape machines that would play the same audio. One of these tape machines would be physically touched to slightly delay its signal. These two outputs were then mixed together equally. Because of the small delay in one of the machines, the result of the output ends up with a sweeping sound. In order to stop the flanger effect, the other tape would also be slowed down by the same amount, in order to match the signals once again.

As technology advanced, so did the process of creating the flanger effect. This effect was implemented with analog before digital was used. In the analog circuit the signal is taken in like normal, then mixed with a delayed version of the signal. This method is still very prominent today in the market for effects pedals. However, for the purpose of this project, the digital implementation was chosen for this specific effect due to the fact that it is a time-based effect.

DSP became more popular for newer applications of the flanger effect. In order to achieve this through DSP, a feedforward comb filter is used. This filter is able to perform the same operation as the original way of creating the flanger effect. In the comb filter the input signal is taken down two paths. One path takes the original signal directly through to the output, while the other path goes through a delay and a depth operation. The equation can be represented by the function below. [2]

- $y(n) = x(n) + d * x(n - M(x))$.

This function is very similar to the delay/echo effect, however the M value, which is the delay time, is modulated based on x. This modulation of the amount of delay is what causes the unique output sound.

c) Reverberation

Reverb is an effect used to create a sound that appears as if the recording is being done in a large room. This is a natural effect that occurs when listening to any audio in a room that does not absorb the sound waves. As the sound travels from the source it is able to bounce off walls in all directions before actually reaching the target. Along with the direct sound that is coming from the source, the sound waves that bounce off of the walls ultimately make it to the target, but create a echo type effect. These bouncing signals are called early reflections. These reflections of the original sound occur somewhere between 5 and 20 milliseconds after the sound is produced.

The reverberation method more commonly used today, especially in guitar pedals is the digital signal processing method. The way this process works is by taking a converted analog signal, then taking the samples and modifying them to be passed to the DAC. In order to modify the signals, multiple delay lines are used in order to create the early reflections. Several delay lines are used at the same time but at slightly different sample times to mimic multiple signals coming from different directions from the walls. This series of delays fade out over time, allowing the sound to appear as if they are decaying. [3]

VI. Digital Effects Hardware

The hardware used for digital effects in the G12 PedalVision is based on the Hoxton OWL digital board, which is an open source DSP audio effects unit. This board includes a microcontroller, external RAM chip, and an audio codec. This hardware was checked against the G12 PedalVision's digital effects specifications. The microcontroller chip used for the digital signal processing is an ARM Cortex M4 chip. This allows for enough power to be able to process the input signal. The RAM chip being used has enough space to allow for the code to store samples of the input signal to be used at a later time. This storing of the samples is what creates the delayed time. Without this chip, it would be difficult to create the effects since the on-chip memory would not be enough to buffer the amount of samples necessary. The audio codec that is on the board will allow for high precision analog to digital conversion. Along with these main components, the input and output buffers to the chips needed to be implemented on a separate board. The design of these filters is based on the Hoxton OWL pedal schematics. Modifications to these circuits were made based on testing of the setup in order to meet personal requirements of sound.

The firmware used on the chip is developed by Hoxton OWL and is under an open source Gnu General Public License. The reason for this choice was to allow for users to not be limited to only the effects that are provided with the G12 PedalVision. By using this firmware, we are able to develop the code for creating each effect. We are also able to load effects that are available open source through the Hoxton Owl website.

An interface is also developed in order to allow for quick and simple user interaction with the system an Arduino Uno using an ATMega-328p chip. This board allows for easy integration to a 20x4 LCD display through i2c communication. This display will show users exact values for the potentiometer inputs, to provide a precise modification of the sound.

VII. LED FEEDBACK MATRIX

The LED matrix feedback display is an array of individually addressable LED lights that are arranged in a circular pattern and toggled on and off to different colors and intensities dependent on the frequency and amplitude of the incoming signal. The LED Matrix feedback display is responsible for using the input signal and sampling it to obtain the frequency and amplitude of the signal in order to output the proper display. This matrix does not add any value to the actual guitar effects pedal section of the project. It does not offer any type of filtering or distortion to the input signal. The input signal is merely used as a reference to determine the LED settings to be displayed. What the display does offer is the ability for the end-user to have a visual feedback of the notes that he or she is playing and their affected output after any type of effects are applied to the signal. This may not be important to all end-users due to the fact that it doesn't enhance the sound quality or playback effect, but to the sit at home and play in their room with the lights low user, it can add a feeling of being in sync with your music. Allowing them to turn their amplifier up loud to feel the music, of course hear the music, and now see the music like never before.

This is the effect that we are going for, adding that new dimension to music playing that the average user hasn't been able to experience. This being said, we realize that those other end-users that don't desire this encapsulation of the music need to get some usefulness out of the LED matrix feedback display. We have accomplished this by offering a tuning feature that is built into the unit. This feature will not only allow users to tune their instruments quickly and efficiently, but it will serve as a conversation peace to show other musicians. Having a unique design for the tuning feature gives it that one of a kind feel and look that will make it worthwhile to share with others.

With each note mapping to a particular color the user will begin the process of learning how their playing changes the display color. After a very short while the user will begin to anticipate a particular pattern of colors due the set of notes that they are playing. This display will handle individual notes very well but chords that are commonly played on the guitar will have a uniqueness to them. Depending on the intensity each of the strings making up the particular chord played has, the output color will vary due to how the output waveform that the system is sampling is being read. This system being designed in this manner will allow the user to notice when they are inconsistent with their playing, allowing them to make changes if they see necessary.

The chip used to help accomplish the goals set fourth for the LED Matrix Feedback Display was the TLC 5955. This robust LED driver is very powerful, yet other than the datasheet, has very little documentation on it from other designers using the chip. This added a layer of complication and opportunity for growth. With this obstacle in front of us, we developed an entire library to be used with the chip to allow for easy programming and reconfiguration. This was necessary as the chip had two 769-bit registers that required updating in order to change the color or intensity of the LEDs.

VIII. POWER SUPPLY

Something that is very crucial to a project , but that is often overlooked is power. For this project, a power supply was designed to run the analog PCB, digital PCB and the PCB for the LED display. Dropping down the voltage and supplying the correct amount of current were two of the main concerns for this part of the project.

a) Design Approach 1

Initially the power supply was supposed to be a part of an all inclusive PCB. The supply would have dropped down the voltage to 9 V for the analog and digital effects, while also supplying the 5 V to the LED display.

The flaw in this design was it would result in a larger board and the power would be very close to the designs implementing the analog and digital effects.

b) Design Approach 2

Not wanting the power supply to be close to the analog and digital designs, it was decided the power would have its very own PCB. Isolating the power reduced the original size of the board and also isolated the power so it would not affect the analog and digital board and the LED display unless it was connected to them.

c) Design Approach 3

While still keeping the power isolated, it was now time to decide how to distribute the power between the analog, digital and LED display PCB's.

It was decided the power supply would supply 9V to the pedal box and 5V to the LED display through barrel jack plugs.

IX. POWER IMPLEMENTATION

After the final design was chosen, it was time to decide on how to implement this design.
First, approximately 115 VAC – 120 VAC at 60 Hz would need to be dropped down a lower voltage of 16 Vrms. This was done by using a 7.5 VA transformer.

The output from the transformer was then connected to a full bridge rectifier. Two capacitors, placed in parallel, would clean up the reaming ripple, resulting in 16 VDC.
The 16VDC is dropped down to 9V and 5V using two different regulators. LM22674 was used to drop the voltage down to 5V. LMR14203XMKE was used to drop the 16 V down to 9 V. Both boards will be receiving 500mA. The voltage is dropped down once more on each PCB to obtain the desired input.

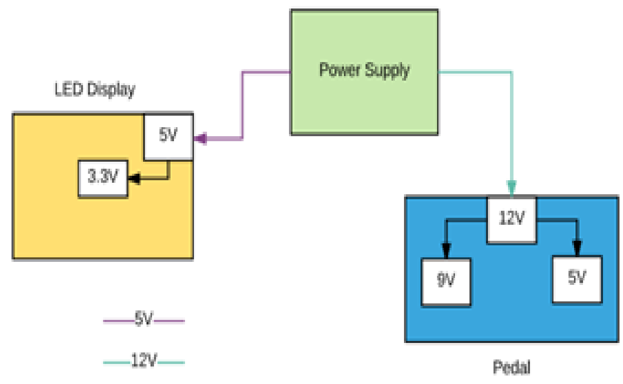

Fig. 2. Block Diagram of broken-down power distribution.

With everything put together on the final power PCB, the guitar pedal and the LED should be fully powered and able to run. While in use, the noise and current fluctuations would need to be monitored closely to avoid undesired outputs that could damage the receiving PCBs.

X. PCB FABRICATION

Once the schematic designs were complete, the boards were sent to PCBWay for fabrication. Due to the simplicity of the PCB designs and the maximum 100mm x 100mm size of the board layout, we were able to get each board fabricated and shipped for approximately $30.

Most of the components on each board were chosen to be surface mount technology, which helped in reducing the size of the boards. Some components, such as transistors and diodes, were chosen as through hole due to availability. Although most all components were surface mount, initial testing and prototyping was done using breadboards and through hole components. Once all components for each board were gathered we used our own soldering station to solder each component to the final boards.

XI. CONCLUSION

The research done for the G12 PedalVision was extensive due to the well-established nature of this topic. In addition to researching the engineering aspect of this project, we had to research the artistic side as well. In order to determine which final design we would use, it was necessary to test multiple options, leading to a consensus as to which method gave the better sounding result. The process for getting a complete product included many simulations, breadboard tests, designing and testing PCBs.

The design of the G12 PedalVision was split into multiple portions. The analog effects, digital effects, LED feedback matrix, and power supply were all individual portions that were combined to create a final product.

The analog effects were researched and tested in a simulation environment before any hardware was used. Once the simulations were completed, a breadboard was set up to test the signal going through the circuits. Once complete, a PCB was designed and ordered. The digital effects hardware was set up using the Hoxton OWL documentation. This allowed for a great baseline for making modifications and designing the board in a way that works for our application. The code was also a necessary portion of the design in order to implement the digital effects and optimize them. The LED matrix was designed to allow for an added dimension to the musician's performance or practice. The frequency and amplitude of the signal will be used to create different color and motion patterns depending on the setting currently selected. Although for now this project is just a prototype, with the lessons learned we plan to continue development and optimization of this product. Also we will take this project out to the public for testing by professionals in order to gauge its real world performance.

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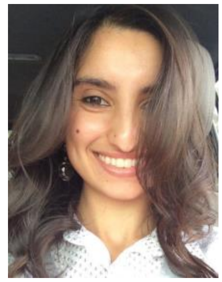
The Engineers

Ayesha Arif:



Ayesha Arif is an Electrical Engineer. For the G12 PedalVision, she was in charge of the power supply.

Brian Boga:

Brian Boga is an Electrical Engineer focused on Embedded Systems. For the G12 PedalVision, he was in charge of the hardware and software for the LED Matrix. He will be joining Northrop Grumman upon graduation as an Electrical Engineer.

Kevin Leone:

Kevin Leone is a Computer Engineer focused on software and artificial intelligence. For the G12 PedalVision, he was in charge of the digital effects hardware and software. He will be working Lockheed Martin as a Software Engineer after graduation.

Jose Ramirez:

Jose Ramirez is an Electrical Engineer focused on analog design and digital signal processing. For the G12 PedalVision, he was in charge of all analog effects and assisted in the hardware design and testing for the digital effects portion of the project. Upon graduation he will be joining Texas Instruments as an applications engineer.