Electric Guitar Amplifier With Digital Effects

By

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Abstract

During the course of this project, the research and development of an electric guitar amplifier was undertaken. This amplifier is divided into three distinct sections. The power distribution circuitry provides the proper voltages to safely operate all necessary components. The DSP and digital effects stage provides processing and allows the user to add digital effects to the original guitar sound. The amplification stage increases the power of the input signal to drive a loudspeaker.

Due to several setbacks throughout the course of this project, the overall amplifier was not completed. All three of the sections were developed to working functionality, but complete integration wasn't achieved. Due to the unsuccessful integration of the complete system, additional developments such as a refined user interface, integrated PCB design, and enclosure were not completed.

I. Introduction

During the course of this senior project, a fully functional electric guitar amplifier (Amp) and guitar effects synthesizer was researched and designed, and to a certain degree implemented. The amplifier's design consists of three major elements. The DSP and digital effects stage will handle the volume and tonal modifications of the electric guitar signal. The amplification stage will deliver enough signal gain to power a 25W speaker. Finally, the power stage will provide the proper power levels to safely operate the individual components of the system.

I conceived the idea of developing this system because of several reasons. Since I started to learn how to play the guitar in late 2009, I've become more interested in audio engineering, especially guitar amplifiers. While looking at retail guitar amplifiers and other guitar accessories, I noticed that there were few amplifiers that could also add effects. And finally, developing this project required a great deal of knowledge acquired over the past three years.

Definition of Intended User and Environment

This amplifier will have a power output of approximately 25 W, placing it within the Practice Amplifier / Studio Amplifier classification, which is the lowest classification of guitar amplifiers. This system will be equipped with a small assortment of effects, which offer slightly more flexibility than a standalone guitar amplifier. Because of its relatively low power output and limited functionality, this amplifier is intended for a beginning to intermediate-level guitar player.

Even though there are countless guitar amplifiers and effects synthesizers on the market, there seems to be very few devices that combine an amplifier with more than one effect, which is usually a form of added distortion. The fact that this project will combine an amplifier with at least three effects doesn't make it unique, but it does make it rare.

There are several well established musical instrument and electronics manufacturers such as Fender, Gibson, Behringer, and Marshall that produce both guitar amplifiers and special effects systems. A standalone amplifier with similar specifications to the one to be designed for this project will cost on average between $50 and $100. An amplifier with the same approximate power output combined with a suite of effects will cost from $80 to $200.

Alternatives

While a system such as this provides the convenience of providing guitar amplification and effects in one unit, it does have its limitations. A system such as this can be programmed with virtually any conceivable effect, but may lack the quick and simple customization that is provided with a separate amplifier and effect pedal system.

II. Background

This project is essentially divided into three major parts, signal processing and amplification.

- **Signal Processing:** In order to provide the effects to the guitar's signal, several components must be employed. In order to translate the analog signal produced by an electric guitar's pickups to a digital signal, an Analog to Digital Converter (ADC) will be used. Once the signal is in digital form, a Digital Signal Processor (DSP) will be used to make the necessary modifications to the signal. The DSP will use two types of memory to perform its function, Read Only Memory (ROM) for program storage, and Random Access Memory (RAM) for temporary signal data storage. Once the signal processing is complete, the digital signal will be translated back into an analog signal using a Digital to Analog Converter (DAC). In order to synchronize the digital components, a constant clock signal will be provided by an Oscillator.
- **Amplification:** The signal produced by an electric guitar is far too weak to power conventional speakers. The signal must be amplified significantly for this project, Once the signal has been converted back into an analog format after being processed, a series of Operational Amplifiers (Op-Amps) are used to provide the signal increase. The power output from the op-amp will be sufficient to power a 25W speaker.
- **Power Distribution:** In order to provide safe power levels for many of the more sensitive components, the voltage levels must be regulated. In order to provide safe and effective voltage levels, a combination of voltage regulators and op-amps will be used.

Terminology
These terms are related to this project, and may be used frequently in its description.

- **Analog to Digital Converter (ADC):** A component which converts a continuous analog signal into a series of digital values. These values are always between a log and high reference voltage.
- **DC Offset:** A constant voltage which a continuous AC voltage signal is added to. DC offset can be desired in some situations, but unwanted in others.
- **Digital to Analog Converter (DAC):** A component that converts discrete numerical values into a continuous analog signal. The DAC is the counterpart of the ADC.
- **Digital Signal Processing (DSP):** The process of capturing either an analog or digital signal, and potentially making modifications to the signal.
- **Gain:** The amount in which a signal's amplitude is increased.
- **Gain Bandwidth Product (GBP):** This is a relationship between an amplifier's signal gain, and its maximum range of frequencies (bandwidth). As the signal gain is increased, the maximum frequency rang the amplifier can handle before signal degeneration occurs is decreased.
- **Operational Amplifier (Op-Amp):** An active electrical component that can be used in various circuits, including filters, signal amplification, and voltage rail adjustment.
- **Pickup:** A passive electrical component used on an electrical guitar to convert the vibrations made by the guitar's strings to an electrical signal.

- **Slew Rate:** A relationship between the amount of change in the output signal of an amplifier with respect to time. The larger the slew rate, the greater the signal can change in a shorter period of time without distortion.
- **Total Harmonic Distortion (THD):** The amount of signal distortion caused by a system, compared to the original signal.
- **Voltage Regulator:** An active electrical component which can convert a widely varying electrical voltage to a relatively constant voltage.

III. Requirements

Design Requirements

Functional Requirements

- **Amplification:** The system will receive a low-current AC signal produced by an electric guitar, and provide adequate amplification to make the signal clearly audible through a built-in speaker.
- **Digital Effects:** The system will provide a suite of at least three digital effects.
    - **Distortion:** Modify the standard sound produced by the guitar to produce an "overdriven" Rock / Heavy Metal sound
    - **Flange:** Modifies the original sound to create a wavering, metallic like tone.
    - **Chorus:** Modifies the original sound to produce a sound that resembles what sounds like several guitarists playing the same note at roughly the same time.
- **Output:** The system will be able to output the sound produced by the guitar to the built-in speaker. There will also be a built-in headphone jack, which can be used in situations where the player wishes to use the amplifier in places where consideration is necessary.
- **Controls:** The player will be able to control the sound produced by the amplifier in the following ways.
    - **Power:** Turns the system on or off.
    - **Volume:** Controls the loudness of the output signal.
    - **Tone:** Controls the relative bass and treble of the output sound.
    - **Effects Selection:** The player will be able to select between any or all of the available effects to add to the output sound.
    - **Effects Gain:** The player will be able to set the amount of each effect.
- **Portability:** The amplifier will be lightweight enough to carry around with relative ease. A carrying handle will be provided.

Performance Specifications

- **Input Power:** ~38 W
- **Input Power Source:** 120 V 60 Hz AC power converted to 48V DC
- **Output Power:** ~25 W_{RMS} at 8 Ω
- **Output Frequency Response:** 80 Hz to 20 KHz
- **Total Harmonic Distortion (THD):** <5%
- **Inputs:** 1/4" musical instrument jack
- **Outputs:** Built-in speaker, 1/8" Headphone jack
- **Controls:** Power, Volume, Tone, Effects Select, Effects Gain, LCD display
- **Maximum Physical Dimensions:** H:11.5" (29.2 cm) x W:10.5" (26.7 cm) x D:5.75" (14.6 cm)
- **Maximum Weight:** 8.5 lb (3.9 Kg)

Critical System Parameter Selections / Settings

- **Amplification Parameters**
  - **DC Offset:** The amplifiers used must have very little DC offset, because this will degrade the audio signal. What DC offset accumulates will need to be removed using DC-blocking capacitors.
  - **Gain Bandwidth Product (GBP):** This factor determines the frequency range an amplifier will accurately amplify a signal to a desired gain. The higher the GBP the better, but other factors may cause high GBP amplifiers to be unreasonable.
  - **Slew Rate:** This factor determines how accurately an amplifier can handle an input signal at higher frequencies. This factor will most likely not be an issues, since the maximum audible frequency is still relatively low.

- **DSP Parameters**
  - **ADC And DAC Bit Width:** In order to ensure accurate audio production, the bit width of the ADC and DAC must be at least 16-bits wide.
  - **Operating Frequency:** The DSP system must be able to operate at a frequency considerably larger than the maximum audio frequency of 20 KHz. Most DSP systems have clock frequencies well above this frequency.

Memory & I/O Address Map

The memory and I/O device address map has yet to be determined at this time.

System Testing & Verification Plan

In order to verify that the amplifier performs within the pre-defined specifications, a series of tests shall be performed to verify its functionality. These tests will ensure the functionality of each subsystem, and ensure the overall functionality of the system. The testing plan is described below.

- **Signal Processing Tests:** These tests will verify the functionality of all components within the signal processing subsystem.
  - **Signal Conversion Testing**
    The Analog to Digital Converter (ADC) and Digital to Analog Converter (DAC) will be tested to ensure proper signal translation from analog to digital, and back to analog. This will be accomplished by producing a known voltage signal using either a function generator or a constant voltage source, while observing the translations performed by the ADC and DAC using an oscilloscope. If necessary, the digital signal may be observed using a Function Analyzer. In order to reduce unwanted distortion, the analog-to-digital-to-analog conversion must have less than a 10% total error.

- **Digital Signal Processing (DSP) Unit Testing:** Due to its complexity, the DSP Unit will require extensive testing to ensure its functionality. Both hardware and software testing will be necessary. More tests may be added as the project progresses.
    - **Hardware Testing**
      In order to test the hardware capability of the DSP, it will be subjected to several tests. These tests include basic power-on and power-off testing, feature testing, and performance stress testing.
    - **Software Testing**
      In order to test the algorithms used to process the guitar's signal, extensive debugging may be necessary. Debugging the basic algorithm will likely be accomplished using software test drivers, but the full program will have to be tested using the DSP itself. In order for the software to be verified as completely functional, all signal processing hardware must be verified as functional first.

- **Amplification Testing:** Signal is a key aspect of this project, and it may also be the most complex. Due to the nature of an analog signal, there is almost no margin for error when performing amplification. For this reason, the amplification subsystem must be extensively tested.
  - **Main Amplification Testing:** The main amplification system will increase the output signal from the Signal Processing Subsystem, enough to be outputted from the main speaker. It will require thorough testing.
    - **Component Testing**
      The overall amplifier will require several smaller Op-Amp integrated circuits to step-up the signal in small increments. Each of these components will be tested with a function generator and an oscilloscope to ensure it performs as expected.
    - **Overall Amplifier Testing**
      Once every sub-component has been tested and the overall amplifier has been assembled, the entire circuit must be tested. Testing will be similar to that performed on each component, with special attention paid to signal accuracy and achieved gain over the specified operating frequency range.
- **Power System Testing:** The systems which make up the overall device all have specific power requirements. In order to ensure proper functionality of the system's components, as well as prevent damage to any component within the system ,every aspect of the Power Subsystem must be tested.
  - **Power Supply Testing**
    Every device will require power from a single power supply. This power supply will be tested to ensure that it can supply the necessary voltage and current to power all necessary devices. Testing will be performed using a multimeter.
  - **Power Regulation Circuitry Testing**
    Since each device has its own power requirement, each device will require its own power regulation. In order to ensure the functionality and safety of each component, all power regulation circuits will be tested. To ensure each device will be provided voltage and current within acceptable limits, a multimeter will be used to test all regulation circuitry.

- **User Interface Testing:** Once all other components have been verified, the User Interface Subsystem must be tested. Even though the complete subsystem can only be verified once all other components have been tested, testing of individual user interface components may be done as the project progresses.
    - **Controls Testing**
      Testing must be performed to ensure that all system controls, including power switch, volume and tone controls, and automatic switching from speaker to headphone output, are in proper working order. These tests will be accomplished using various devices, including a multimeter and an oscilloscope.
    - **Display Testing**
      The display must be tested to ensure proper visual feedback is provided to the user. Since the display is required for effects selection, it is a key piece of the user interface. The display will be visually tested to ensure it provides the proper output when given the proper signals.
- **I/O Component Testing:** To ensure that all input and output components are fully functional, they will be tested. Although it may not be the most desirable method of testing, some or all of these components may be tested using some of the system components themselves. If this is necessary, then all of the components required to test the I/O components must be previously verified.
    - **Input Testing**
      The 1/4" input jack and connecting wiring must be tested to ensure that there are no physical defects or wiring faults. This will be done by comparing an input signal with an output signal from the wiring, using an oscilloscope.
    - **Output Testing**
      The speaker and 18" inch headphone jack, and connecting wiring must be tested to ensure that there are no physical defects or wiring faults. The 1/8" headphone jack can be tested using an oscilloscope, as the input was. The speaker will have to be tested audibly, which is more inaccurate, but the only way of testing its performance.
- **Overall System Testing:** This will be testing performed on the finished project. Once all of the subsystems have been verified, and system assembly is complete, the finished system will be tested to ensure that it meets all predefined specifications.

IV. Design Approach Alternatives

When research was first began for this project, little was known as to which components should be used. The components chosen for this project were chosen for various reasons, including cost, simplicity, and stability. In retrospect, some of alternative components may have been better choices.

Amplification

After doing a small amount of research, and consulting with a professor, it seemed that the best way to accomplish the signal amplification, was to use several op-amps. Each op-amp would have a relatively small gain to avoid drastic decreases in their GBP. It was believed that different op-amps would need to be chosen for different stages, for example, an op-amp with a low DC offset would be used for the input stage, while an op-amp with a large slew rate would be used for the latter stage. After finding the LM1875, it was determined that this op-amp could supply more than enough current, had an excellent GBP and slew rate, and relatively low DC offset. Therefore, it was determined that the signal amplification itself could be handled with a single op-amp. The one drawback with this op-amp, is that it requires either +/-24V rails, or a single +48V rail.

Given the uncommon nature of the LM1875, no accurate SPICE model exists, making circuit simulation virtually impossible. After several instances of trial and error, including two burned resistors and a blown speaker, a stable and relatively safe amplification circuit was found.

A second op-amp, a TL081, is used as a voltage buffer to connect the output of the DSP stage to the signal amplification stage. This was an attempt to achieve proper interfacing between the two stages. It unfortunately didn't have the desired results. An alternative for this problem was not found.

Power Supply

Due to cost, a single rail power supply was chosen. On average, split-rail power supplies were at least $20 more expensive. In order to keep the input signal well within the rails of the op-amp, a level-shifting circuit was added. Considering that this level shifter may be a potential cause of the interfacing problem, the split-rail power supply may have been far more cost-effective in the long run. Each voltage was tested to ensure it was within the proper levels. Although the values weren't exact as desired (+5.43V instead of +5V for example), this was as close to desired levels as could be achieved with the given resistors.

DSP and Digital Effects

In order to create the digital stage of the system, ADCs, DACs, a DSP chip, and the necessary components to interface everything together are needed. To save time, and to reduce the complexity, at least in the short term, a prefabricated evaluation board was chosen. Had this project's design been successful, it was planned to design and fabricate a custom-designed digital stage.

The digital board was tested to ensure that the input signal is reproduced, as well as that the audio effects are functional.

V. Project Design

Interface Specifications
Black Box Diagram

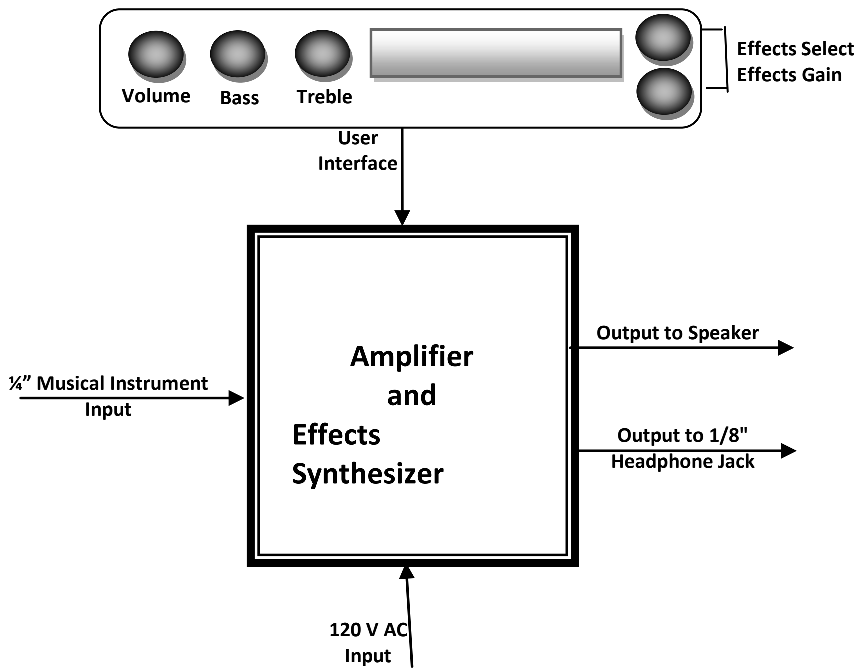


Figure V.1: *Basic User Interface Diagram*

Definition of User Interface

- **Power Switch (not shown):** Turns the amplifier on and off.
- **Volume:** Control the loudness of the output signal, either to the speaker or the headphone jack.
- **Bass:** Increase / decrease the gain of the lower frequencies of the output signal.
- **Treble:** Increase / decrease the gain of the higher frequencies of the output signal.
- **Effect Select:** Scroll through and select from the available audio effects. Effect names and selection status appear on the LCD display.
- **Effect Gain:** Determines the amount of effect added to the output signal.

Detailed System Functional Block Diagram

A top-level block diagram of the original system design is shown below in Figure V.2. Complete system integration hasn't been possible at this point.

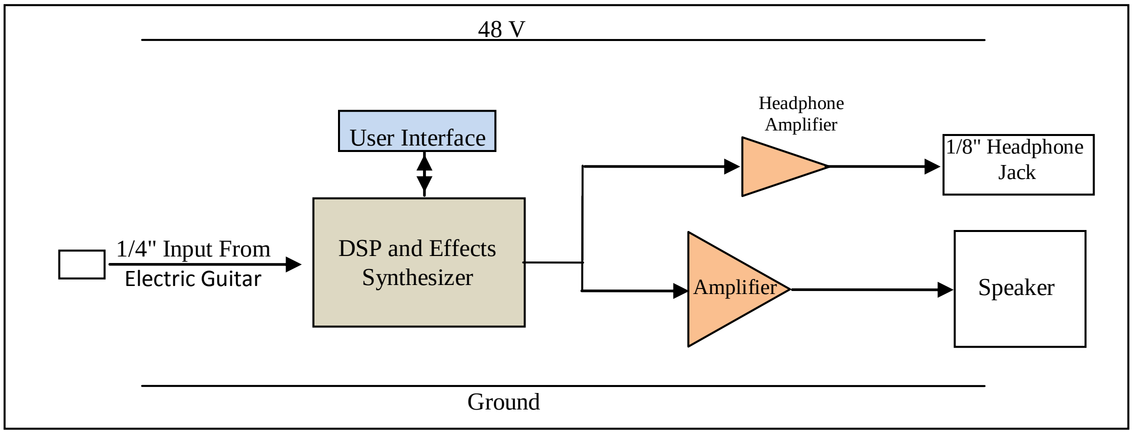


Figure V.2: *Top-Level Subsystem Diagram*

DSP and Effects Synthesizer Subsystem

To handle the digital effects, a prefabricated development board centered around the Analog Devices ADAU1701 audio DSP chip is used. This development board is fitted with all the components necessary to implement this subsystem, including ADCs, DACs, and a headphone output.

Functional diagrams of the ADAU1701 and the development board are shown below in Figures V.3 and V.4 respectively, and a complete list of specifications for the ADAU1701 are shown in Appendix A. The components used for this design are shaded.

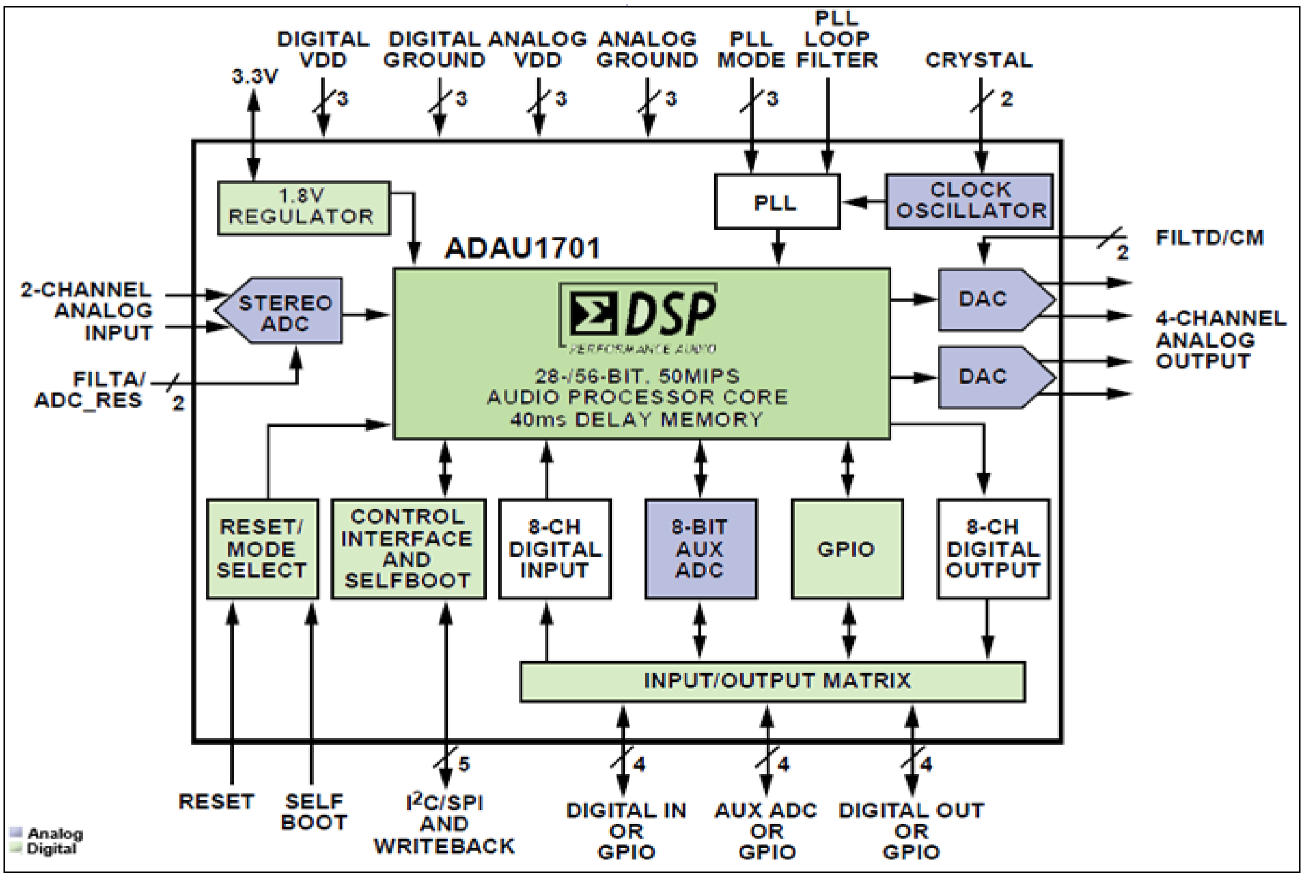


Figure V.3: *Functional Diagram of the ADAU1701 DSP Chip*

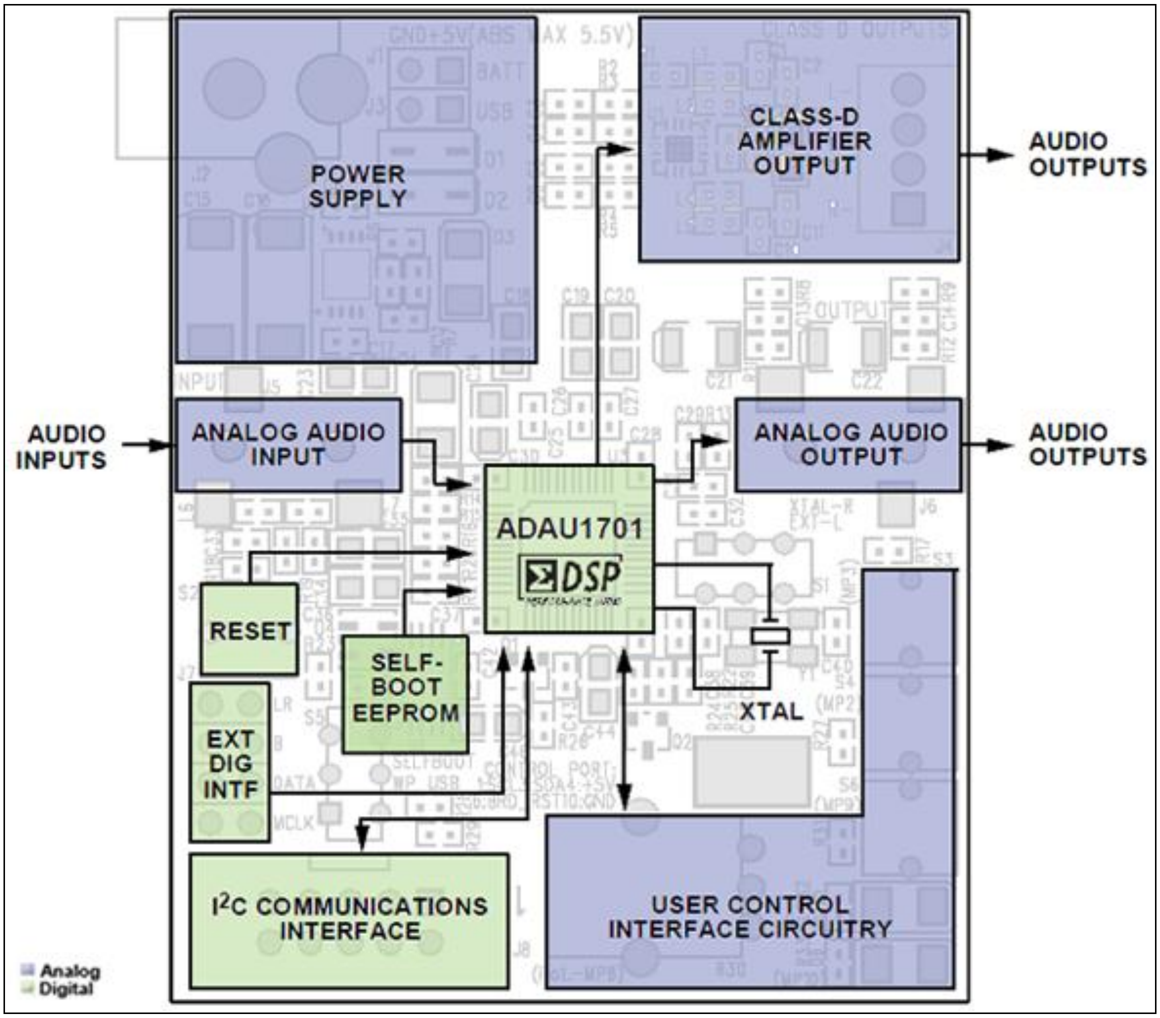
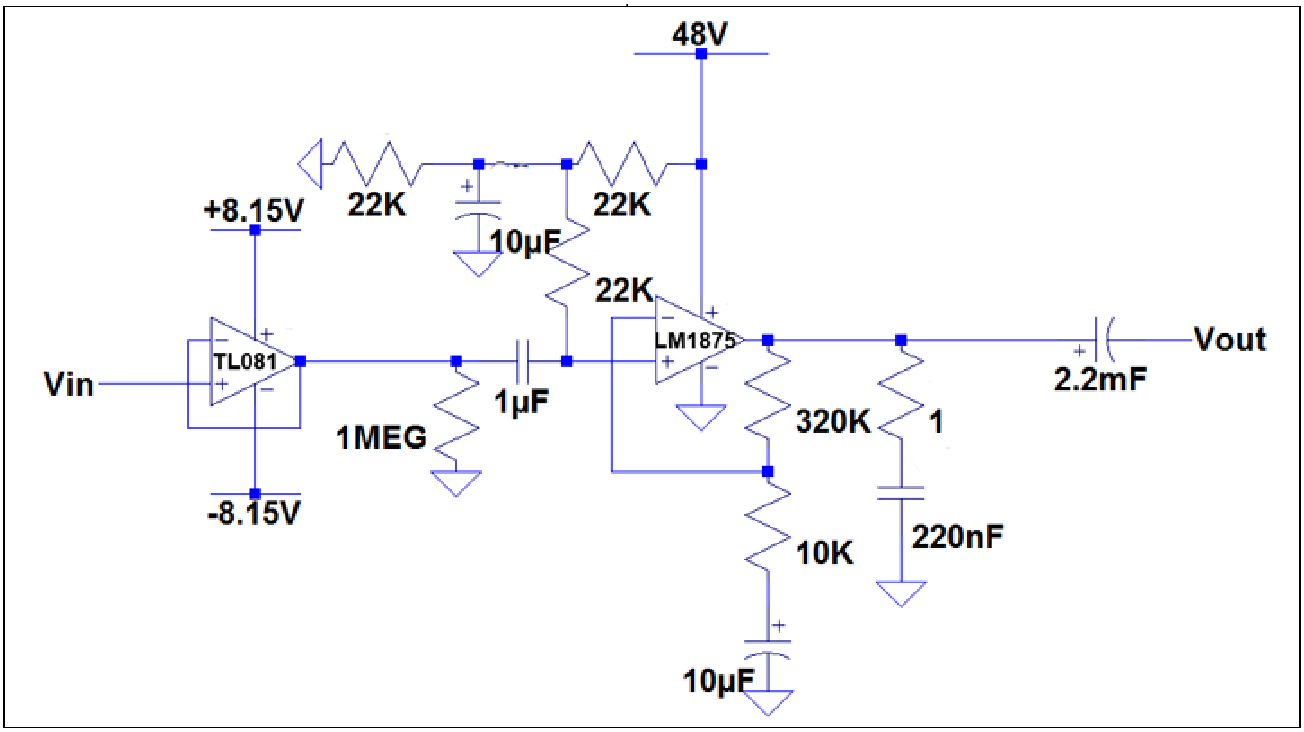


Figure V.4: *Layout of Evaluation Board*

Amplification

In order to amplify the output signal from the DSP stage, the National Semiconductor LM1875 Audio Power Amplifier has been determined to be the best option for this system. This is a single channel op-amp which has been specifically designed for audio applications such as this. The LM1875 is configured in a non-inverting topology suitable for a single-rail design. Capacitors are used to prevent large DC currents from entering the feedback loop and loudspeaker. The schematic for the amplification stage is shown below in Figure V.5.

The TL081 connecting the input signal to the main signal amplifier is an attempt at preventing the signal from the DSP stage from being overdriven by the main signal amplifier. The ADAU1701 chip has the capability of providing enough amplification for headphones, and therefore a headphone isn't necessary. For complete specifications for all op-amps used in this project, see Tables 2, 3, and 4 in Appendix A.



***FigureV.5:** Amplifier Stage Schematic*

Power

The power for the system will be supplied by a 120 V AC to 48 V DC power supply, which takes power from a standard U.S. AC wall outlet. For this application, the V-Infinity VF-S250-48A power supply has been selected. This model supplies the relatively high voltage and current needed for the amplification stage. A power switch will be added to the power circuit to allow easy power-up and power-down of the system.

The DSP components require far less power than the amplifier stage, and would be easily destroyed by the raw power from the supply. In order to lower the voltage from 48 V to ~5 V, a series of voltage regulators are used. A complete set of schematics of the power distribution circuitry are shown in Figures V.6 and V.7.

Notice that the output voltage for the 5V rail is not exact. This is due to inaccuracies in the resistors. The single-to-split voltage rail circuit is used for the voltage buffer connecting the output of the DSP stage to the signal amplifier.

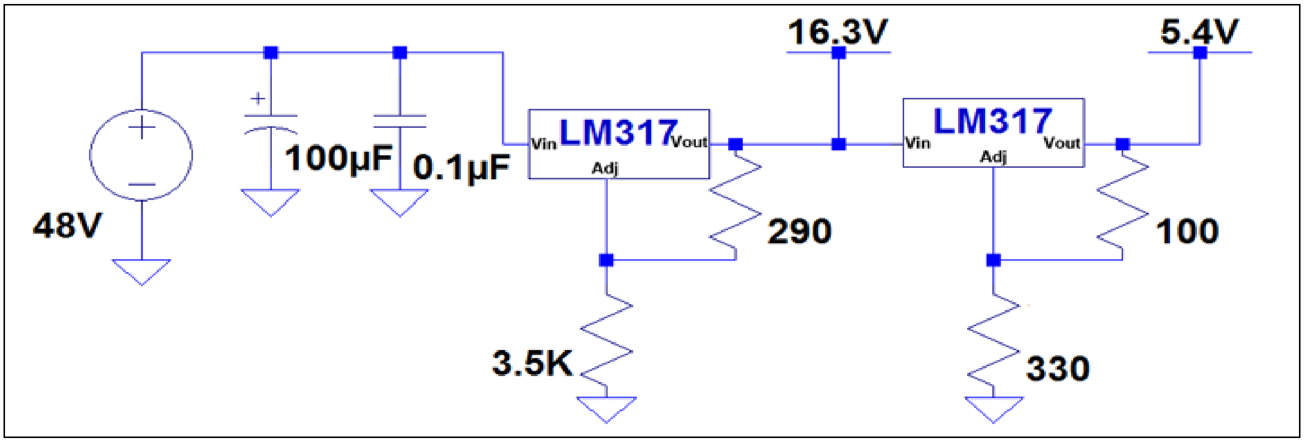


Figure V.6: *Primary Power Distribution Schematic*

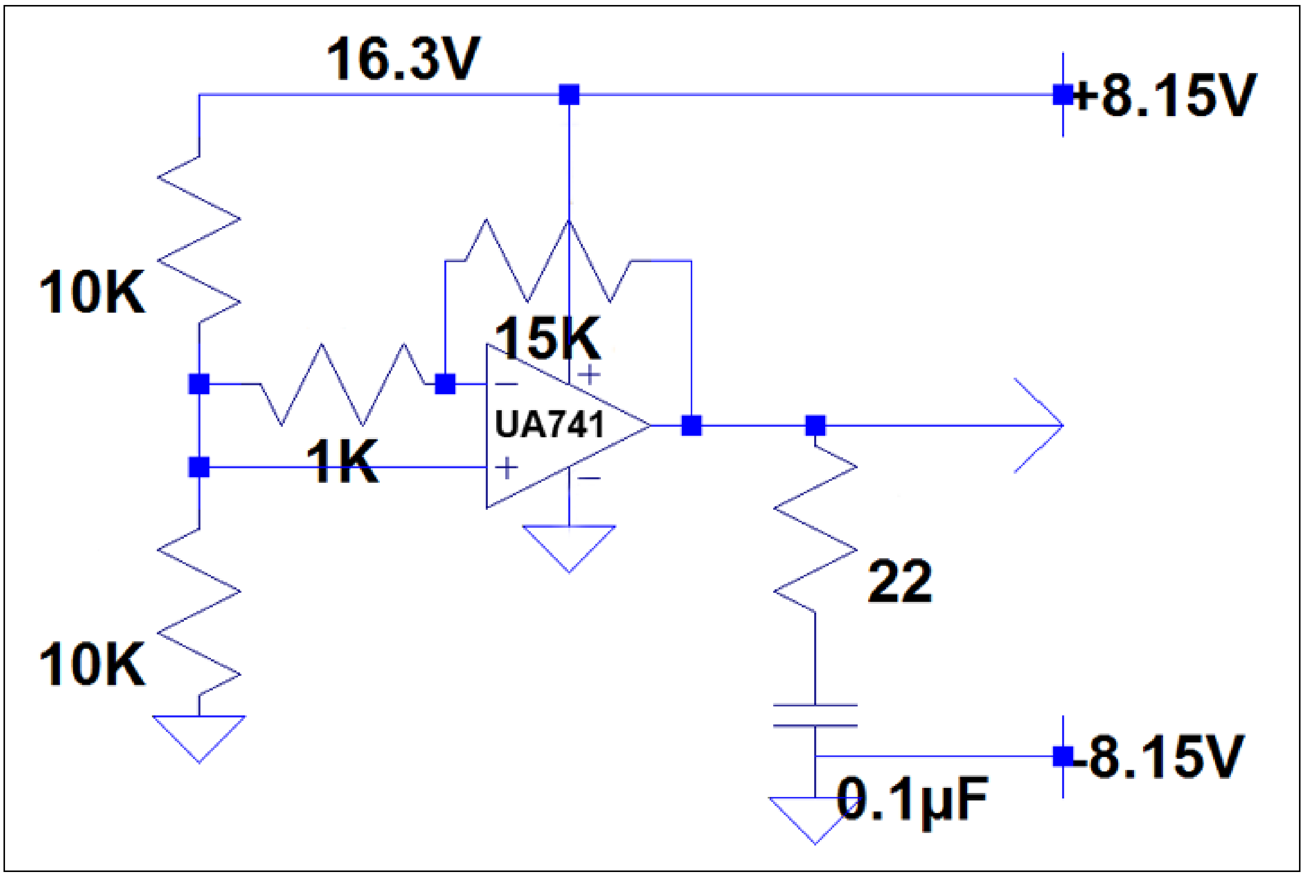


Figure V.7: *Single-to-Split Voltage Supply Conversion Schematic*

User Interface

Originally, the user interface was supposed to be a console consisting of knobs, buttons, a power switch, and an LCD display. Due to design setbacks, and the incomplete functionality of the system, a complete user interface wasn't implemented.

The ADAU1701 development board has a primitive user interface consisting of buttons and a potentiometer connected to an auxiliary ADC. Between these physical controls, and the software interface, the system can be controlled.

Enclosure

In order to protect the system from the environment, as well as enhancing the speaker output, the entire system would've been enclosed in an amplifier cabinet. Due to the incomplete nature of the system, an enclosure was not built.

VI. Physical Construction and Integration

Since the project never made it to complete system integration, the physical constraints didn't have much of an impact on the final result. The entire prototype consists of several components wired together on breadboards. The ADAU1701 development is itself a prefabricated circuit board, but none of the other components was deemed successful enough to make permanent. There was a general plan for the physical design however.

PCB Design

The original physical design called for the amplifier stage and power distribution circuitry to be placed on a prototype board, or preferably a custom fabricated PCB. The LM1875 requires a heat sink to remain within its operating temperature. In order to reduce noise, like components such as resistors and capacitors would be placed in a parallel orientation on the board. Traces would make gradual bends when possible to reduce noise, and the traces would be as short as possible. Any long power wires would be twisted to help prevent noise. The board would have to be able to fit inside the required dimensions.

Enclosure

In order to provide protection from the environment, as well as improve the performance of the speaker, an enclosure was planned. It would need to be no larger than the required dimensions, yet be able to accommodate the speaker, circuit boards, power supply, and control interface. The enclosure may have also needed to provide vibration protection for the electronics, as well as ventilation for the power supply and amplifier.

The optimum layout for such an enclosure would place the speaker in a mostly isolated space at the bottom of the enclosure. The other components would be placed in an upper compartment, away from the speaker. The power supply would be placed in the rear of this compartment, allowing easy access for the power cord. The amplifier circuit board would most likely be placed above the DSP board for better ventilation, assuming there wasn't a single unified circuit board.

VII. *Integrated System Tests and Results*

Since the system wasn't fully integrated, it is impossible to test a fully integrated system with respect to the original specifications declared earlier. The only option therefore, is to test the individual stages to the best of their abilities.

Power Ratings

Using a multimeter, the maximum input current to the system was shown to be 875mA. With a maximum voltage rail, this translates to an input power rating of 42W. This is slightly higher that the desired specification for input, but this only occurs when driving the speaker at its recommended peak level. When the speaker isn't being driven, the maximum input current is 64mA, therefore the input power at rest is only 3.1W.

In order to drive the loudspeaker at ~25W_{RMS}, the amplifier must be able to maintain an output voltage of 20V_P. As shown by the oscilloscope trace below in Figure VII.1, the amplifier easily achieves this. In practical application however, this volume was found to be somewhat painful at close distances, and is not recommended.

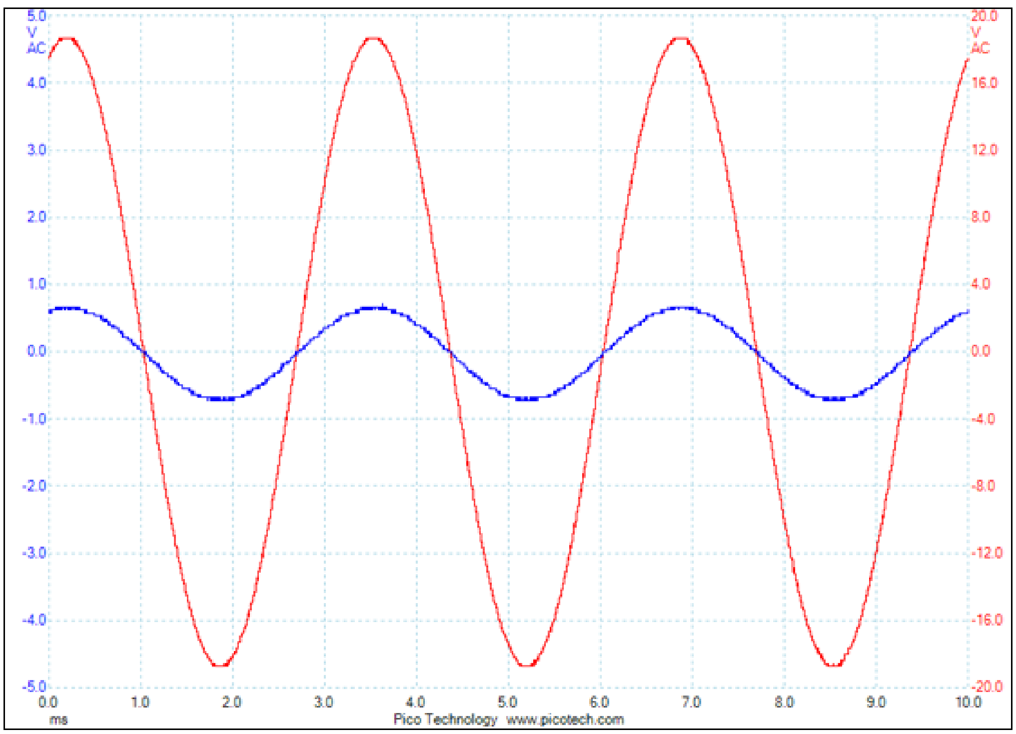


Figure VII.1: *Amplifier output Oscilloscope Trace*

As shown by the figure above, the amplifier produces a very clean signal when the input is a function generator.

DSP and Effects

Testing the digital effects stage is slightly more difficult. Due to the sensitive nature of the input ADC, the input and output signals couldn't be measured simultaneously. A break in the connection between the electric guitar and the ADC would cause a great deal of noise, and makes testing virtually impossible. Therefore, two waveforms were measured at the output of the electric guitar. These inputs will be used as reference waveforms for the other tests. The waveforms are shown below in Figure VII.2.

It should also be noted that an electric guitar is far from being a stable signal generator. The waveform produced will vary with every pluck of the string. Differences in temperature, humidity, and being slightly out of tune will all change the string's vibration, and as a result, the waveform itself. Therefore, there will be differences in each waveform shown.

The next several figures are examples of the digital effects added to the original guitar signal. The first figure is the unaltered output from the DSP stage. The effects on the signal is difficult to determine by visual inspection, and is far clearer when heard. Therefore, a description of each effect is provided below.

- **Distortion:** The distortion effect works by removing, or clipping, any part of the signal that reaches a certain level. This version of the effect leaves most of the signal, and just removes enough to cause a more abrasive Hard Rock sound.
- **Chorus:** Chorus works by using delays and level adjustments on the original signal, and then feeding the signal back in. The result gives the impression of multiple guitars playing slightly out of sync with each other.
- **Flange:** The flange effect is caused by taking the original signal and sweeping its frequency. The modified signal can then be added back to the original signal. The result is a wavering sound.

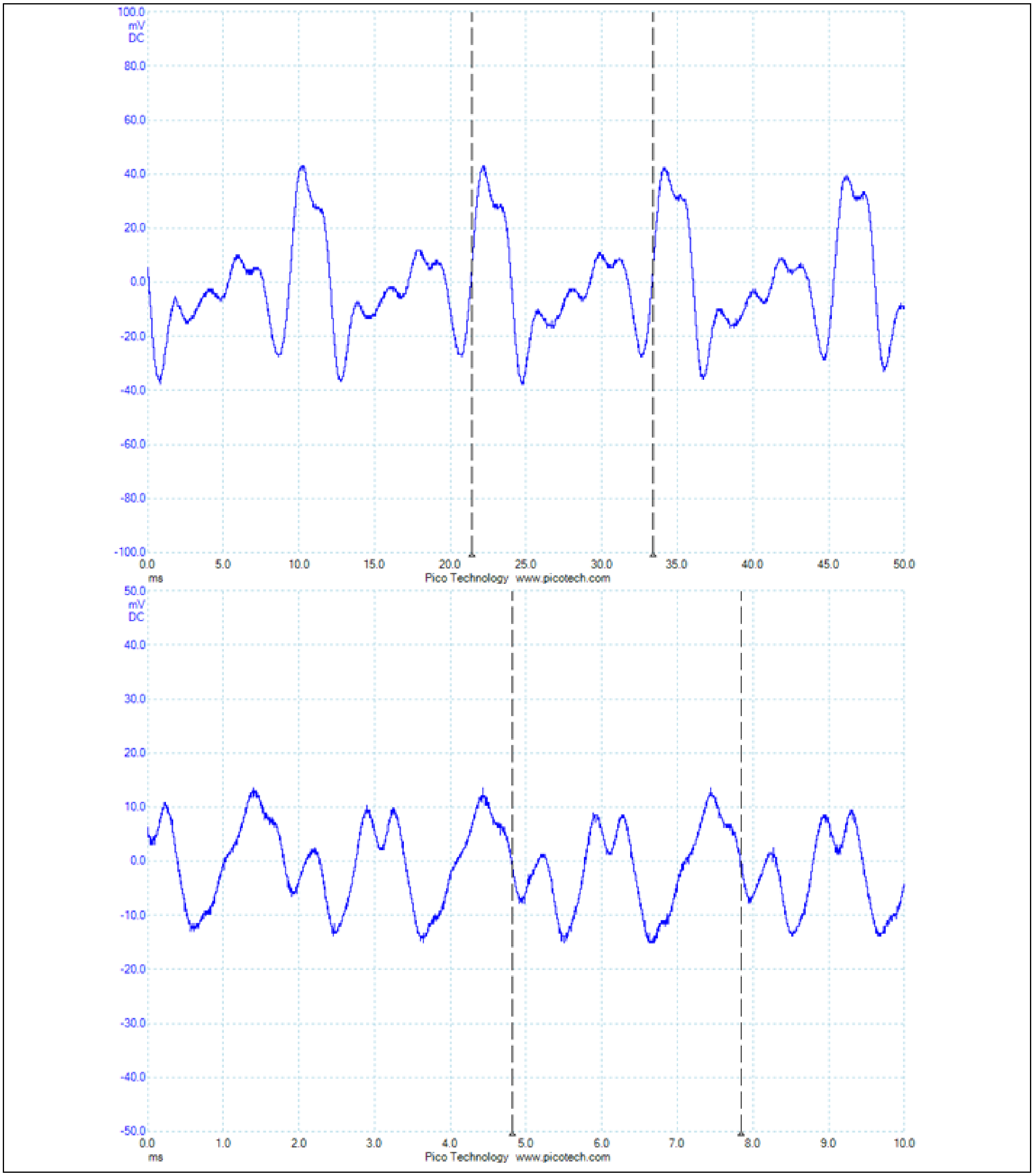


Figure VII.2: *Baseline electric guitar waveforms for Low-E (Top) and High-E (Bottom)*

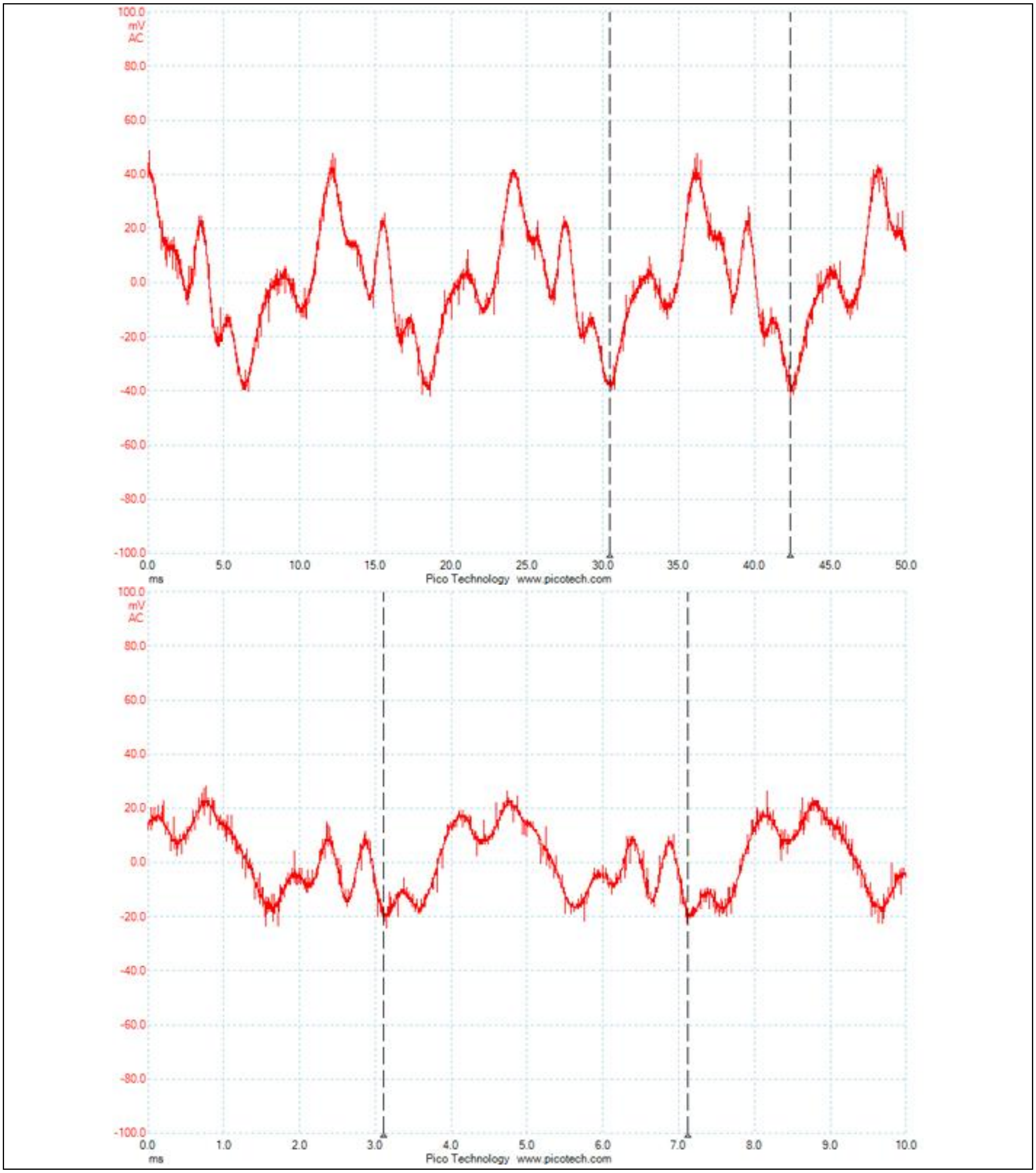


Figure VII.3: *Low-E (Top) and High-E (Bottom) Output from DSP Stage*

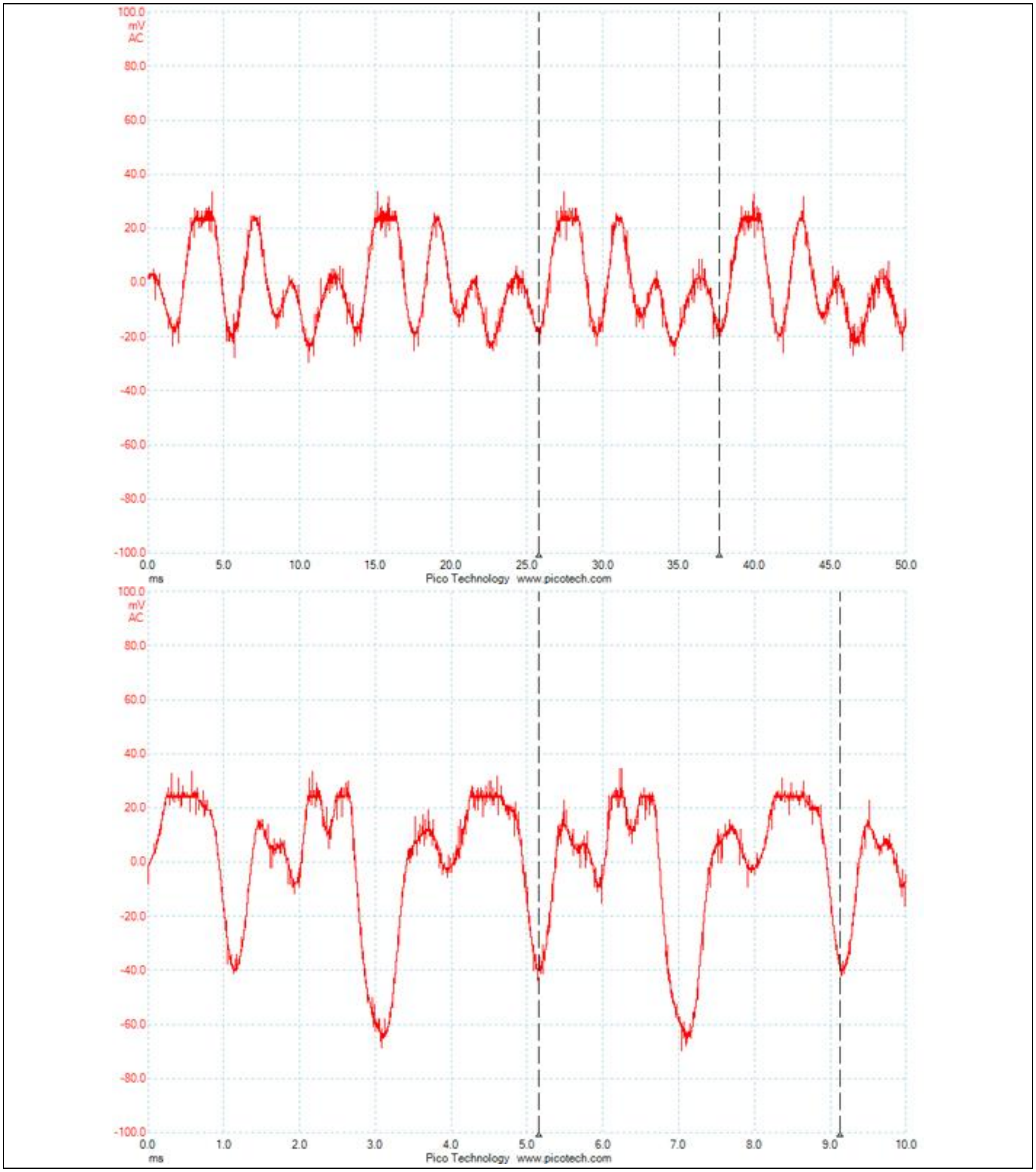


Figure VII.4: *Low-E (Top) and High-E (Bottom) with Distortion Effect*

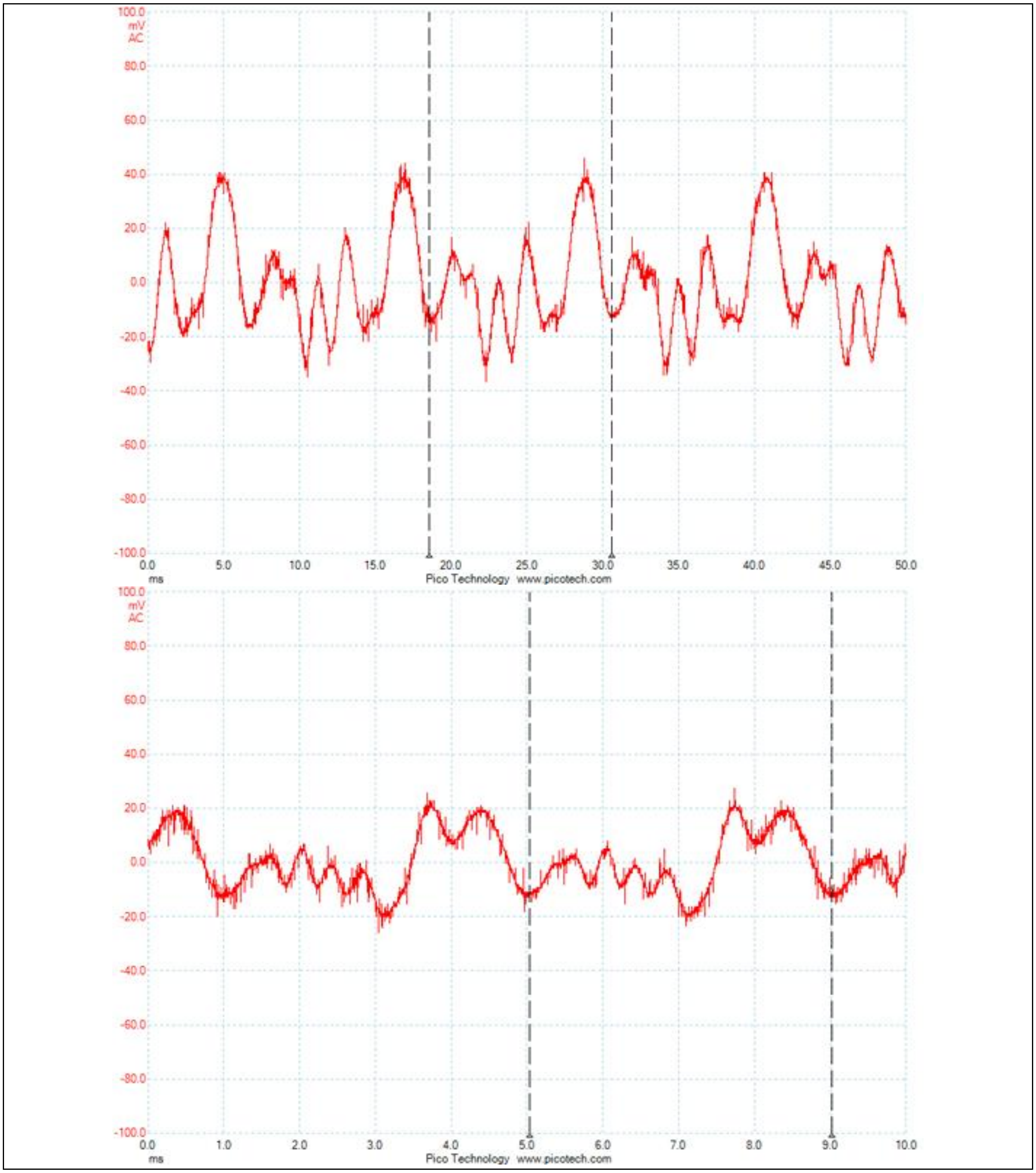


Figure VII. 5: *Low-E (Top) and High-E (Bottom) With Chorus Effect*

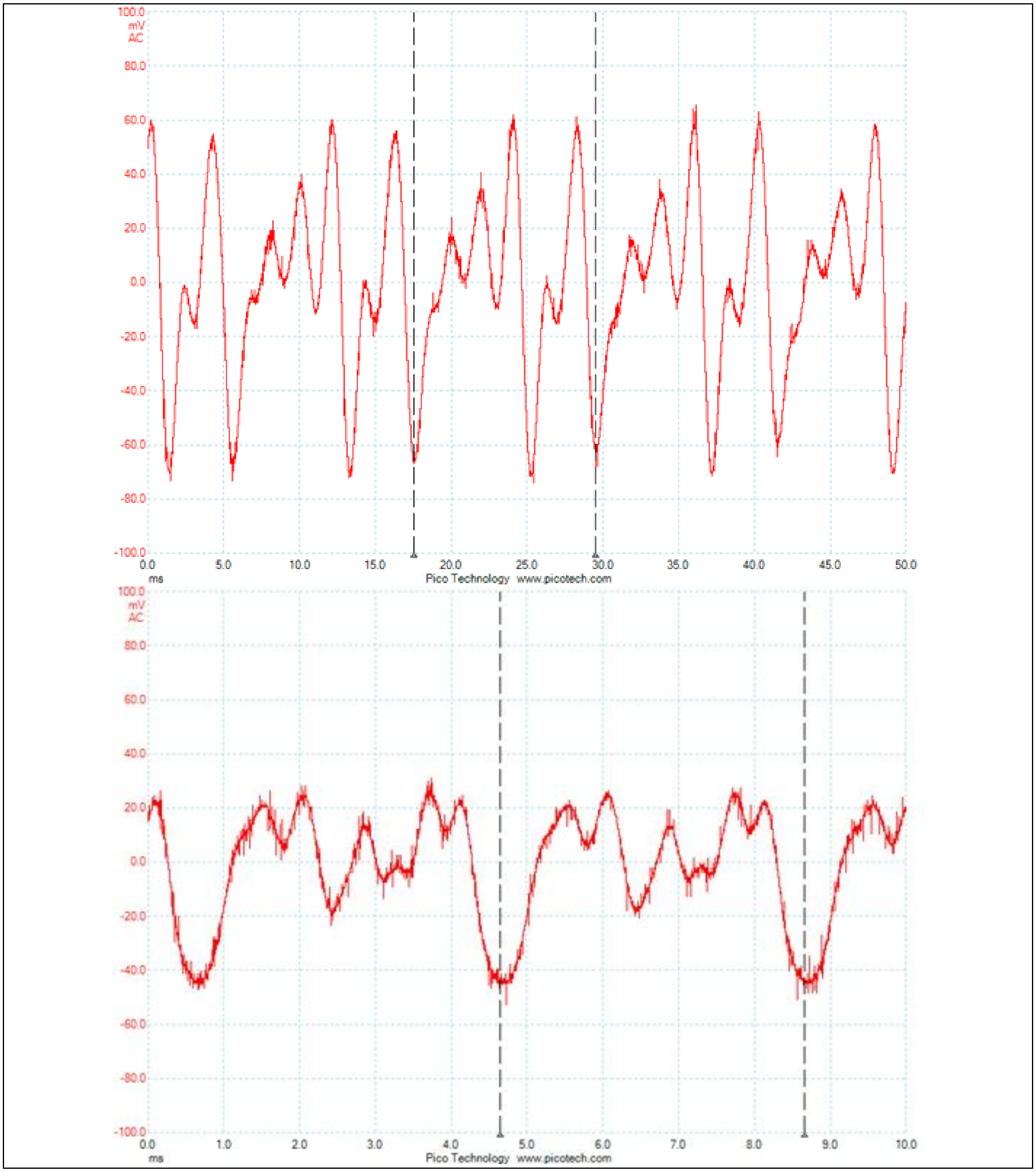


Figure VII.6: *Low-E (Top) and High-E (Bottom) With Flange Effect*

VIII. Conclusion

Unfortunately, a cohesive working design wasn't accomplished within the time constraints of the project. This was due to several reasons. Component damage, a lack of knowledge about various aspects of the design, and time conflicts all had a part to play in the final disarray of the design.

It should be considered that all of the separate stages work on their own. The amplifier could successfully drive the loudspeaker, and accurately reproduce an input signal at a higher gain. The DSP stage could accurately reproduce the guitar's original signal, and add the digital effects when desired. And the power distribution circuitry provided safe and reliable power to the more sensitive components in the design.

Though it seemed more cost effective at the time, a split-rail power supply would've been a far more desirable alternative to a single-rail supply. Without the level-shifting circuit needed to properly use the LM1875, the interfacing issues between the DSP and amplifier stages would've probably been eliminated.

If this project were continue, and it very well may. A different power supply may be employed, or perhaps a way to eliminate the interfacing issues may be found. The issue may be caused by the low the low impedance output of the DSP stage's DAC being overwhelmed by the level-shifting circuit. If a way were found to preserve the output from the DAC, then the system would work. If these problems were solved, the project would continue with further implementation. This would include a permanent circuit board design, a refined user interface, and an enclosure.

Even though the project as a whole wasn't successful, there are some positives to consider. Building test circuits in the lab is far easier than designing an entire system such as this. In the labs, power amplifiers aren't even considered, and a function generator is a far more clean and forgiving signal source compared to a DAC. Valuable lessons were learned for future projects, and perhaps this project still has more to teach.

IX. Bibliography

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- Various other websites were researched for this project.

Appendix A. Component Specifications

National Semiconductors LM1875 Audio Power Amplifier

Absolute Maximum Ratings (Note 1)

Supply Voltage	60V
Input Voltage	$-V_{EE}$ to V_{CC}
Storage Temperature	−65˚C to + 150˚C
Junction Temperature	150˚C
Lead Temperature (Soldering, 10 seconds)	260˚C
θ_{JC}	3˚C
θ_{JA}	73˚C

Electrical Characteristics

V_{CC}=+25V, $-V_{EE}$=−25V, $T_{AMBIENT}$=25˚C, R_L=8Ω, A_V=20 (26 dB), f_o=1 kHz, unless otherwise specified.

Parameter	Conditions	Typical	Tested Limits	Units
Supply Current	P_{OUT}=0W	70	100	mA
Output Power (Note 2)	THD=1%	25		W
THD (Note 2)	P_{OUT}=20W, f_o=1 kHz	0.015		%
	P_{OUT}=20W, f_o=20 kHz	0.05	0.4	%
	P_{OUT}=20W, R_L=4Ω, f_o=1 kHz	0.022		%
	P_{OUT}=20W, R_L=4Ω, f_o=20 kHz	0.07	0.6	%
Offset Voltage		±1	±15	mV
Input Bias Current		±0.2	±2	μA
Input Offset Current		0	±0.5	μA
Gain-Bandwidth Product	f_o=20 kHz	5.5		MHz
Open Loop Gain	DC	90		dB
PSRR	V_{CC}, 1 kHz, 1 Vrms	95	52	dB
	V_{EE}, 1 kHz, 1 Vrms	83	52	dB
Max Slew Rate	20W, 8Ω, 70 kHz BW	8		V/μs
Current Limit	V_{OUT} = V_{SUPPLY} −10V	4	3	A
Equivalent Input Noise Voltage	R_S=600Ω, CCIR	3		μVrms

Note 1: "Absolute Maximum Ratings" indicate limits beyond which damage to the device may occur. Operating Ratings indicate conditions for which the device is functional, but do not guarantee specific performance limits.

Note 2: Assumes the use of a heat sink having a thermal resistance of 1˚C/W and no insulator with an ambient temperature of 25˚C. Because the output limiting circuitry has a negative temperature coefficient, the maximum output power delivered to a 4Ω load may be slightly reduced when the tab temperature exceeds 55˚C.

Analog Devices ADAU1701 Audio DSP Chip

SPECIFICATIONS

AVDD = 3.3 V, DVDD = 1.8 V, PVDD = 3.3 V, IOVDD = 3.3 V, ambient temperature 25° C, master clock input 12.288 MHz, unless otherwise noted.

ANALOG PERFORMANCE

Table 1.

Parameter	Min	Typ	Max	Unit	Test Conditions/Comments
ADC INPUTS					
Number of Channels		2			Stereo input
Resolution		24		Bits	
Full-Scale Input		100 (283)		μA_{rms} ($\mu A_{p\text{-}p}$)	2 V_{rms} input with 20 kΩ (18 kΩ external + 2 kΩ internal) series resistor
Signal-to-Noise Ratio					
A-Weighted		100		dB	
Dynamic Range					−60 dB with respect to full-scale analog input
A-Weighted	95	100		dB	
Total Harmonic Distortion + Noise		−83		dB	−3 dB with respect to full-scale analog input
Interchannel Gain Mismatch		25	250	mdB	
Crosstalk		−82		dB	Analog channel-to-channel crosstalk
DC Bias		1.5		V	
Gain Error	−11		+11	%	
DAC OUTPUTS					
Number of Channels		4			Two stereo output channels
Resolution		24		Bits	
Full-Scale Analog Output		0.9 (2.5)		V_{rms} ($V_{P\text{-}P}$)	
Signal-to-Noise Ratio					
A-Weighted		104		dB	
Dynamic Range					−60 dB with respect to full-scale analog output
A-Weighted	99	104		dB	
Total Harmonic Distortion + Noise		−90		dB	−1 dB with respect to full-scale analog output
Crosstalk		−100		dB	Analog channel-to-channel crosstalk
Interchannel Gain Mismatch		25	250	mdB	
Gain Error	−10		+10	%	
DC Bias		1.5		V	
VOLTAGE REFERENCE					
Absolute Voltage (CM, FILTA, FILTD)		1.5		V	
AUXILIARY ADC					
Full-Scale Analog Input		3.0		V	
INL		0.5			
DNL		1.0			
Offset		15		mV	
Input Impedance		30		kΩ	

ADAU1701

DIGITAL INPUT/OUTPUT

Table 2.

Parameter	Min	Typ	Max	Unit	Comments
Input Voltage, High (V_{IH})	2.0		IOVDD	V	
Input Voltage, Low (V_{IL})			0.8	V	
Input Leakage, High (I_{IH})			1	µA	Excluding MCLKI
Input Leakage, Low (I_{IL})			1	µA	Excluding MCLKI and bidirectional pins
Bidirectional Pin Pull-Up Current, Low		150		µA	
MCLKI Input Leakage, High (I_{IH})			3	µA	
MCLKI Input Leakage, Low (I_{IL})			3	µA	
High Level Output Voltage (V_{OH}), I_{OH} = 2 mA	2.0			V	
Low Level Output Voltage (V_{OL}), I_{OL} = 2 mA			0.8	V	
Input Capacitance			5	pF	
GPIO Output Drive		2		mA	

POWER

Table 3.

Parameter	Min	Typ	Max[1]	Unit
SUPPLY VOLTAGE				
Analog Voltage		3.3		V
Digital Voltage		1.8		V
PLL Voltage		3.3		V
IOVDD Voltage		3.3		V
SUPPLY CURRENT				
Analog Current (AVDD and PVDD)		50	85	mA
Digital Current (DVDD)		40	60	mA
Analog Current, Reset		35	55	mA
Digital Current, Reset		1.5	4.5	mA
DISSIPATION				
Operation (AVDD, DVDD, PVDD)[2]		286.5		mW
Reset, All Supplies		118		mW
POWER SUPPLY REJECTION RATIO (PSRR)				
1 kHz, 200 mV_{P-P} Signal at AVDD		50		dB

[1] Maximum specifications are measured across a temperature range of −40°C to +130°C (case) and across a DVDD range of 1.62 V to 1.98 V and an AVDD range of 2.97 V to 3.63 V.
[2] Power dissipation does not include IOVDD power because the current drawn from this supply is dependent on the loads at the digital output pins.

TEMPERATURE RANGE

Table 4.

Parameter	Min	Typ	Max	Unit
Functionality Guaranteed	0°C		70°C	°C ambient

PLL AND OSCILLATOR

Table 5.

Parameter	Min	Typ	Max	Unit
PLL Operating Range	MCLK_Nom − 20%		MCLK_Nom + 20%	MHz
PLL Lock Time			20	ms
Crystal Oscillator g_m (Transconductance)		78		mmho

REGULATOR

Table 6. Regulator[1]

Parameter	Min	Typ	Max	Unit
DVDD Voltage	1.7	1.8	1.84	V

[1] Regulator specifications are calculated using a Zetex Semiconductors FZT953 transistor in the circuit.

DIGITAL TIMING SPECIFICATIONS

Table 7. Digital Timing[1]

Parameter	Limit		Unit	Description
	T_{MIN}	T_{MAX}		
MASTER CLOCK				
t_{MP}	36	244	ns	MCLK period, 512 f_S mode.
t_{MP}	48	366	ns	MCLK period, 384 f_S mode.
t_{MP}	73	488	ns	MCLK period, 256 f_S mode.
t_{MP}	291	1953	ns	MCLK period, 64 f_S mode.
SERIAL PORT				
t_{BIL}	40		ns	INPUT_BCLK low pulse width.
t_{BIH}	40		ns	INPUT_BCLK high pulse width.
t_{LIS}	10		ns	INPUT_LRCLK setup. Time to INPUT_BCLK rising.
t_{LIH}	10		ns	INPUT_LRCLK hold. Time from INPUT_BCLK rising.
t_{SIS}	10		ns	SDATA_INx setup. Time to BCLK_IN rising.
t_{SIH}	10		ns	SDATA_INx hold. Time from BCLK_IN rising.
t_{LOS}	10		ns	OUTPUT_LRCLK setup in slave mode.
t_{LOH}	10		ns	OUTPUT_LRCLK hold in slave mode.
t_{TS}		5	ns	OUTPUT_BCLK falling to OUTPUT_LRCLK timing skew.
t_{SODS}		40	ns	SDATA_OUTx delay. Time from OUTPUT_BCLK falling in slave mode.
t_{SODM}		40	ns	SDATA_OUTx delay. Time from OUTPUT_BCLK falling in master mode.
SPI PORT				
f_{CCLK}		6.25	MHz	CCLK frequency.
t_{CCPL}	80		ns	CCLK pulse width low.
t_{CCPH}	80		ns	CCLK pulse width high.
t_{CLS}	0		ns	CLATCH setup. Time to CCLK rising.
t_{CLH}	100		ns	CLATCH hold. Time from CCLK rising.
t_{CLPH}	80		ns	CLATCH pulse width high.
t_{CDS}	0		ns	CDATA setup. Time to CCLK rising.
t_{CDH}	80		ns	CDATA hold. Time from CCLK rising.
t_{COD}		101	ns	COUT delay. Time from CCLK falling.
I²C PORT				
f_{SCL}		400	kHz	SCL frequency.
t_{SCLH}	0.6		µs	SCL high.
t_{SCLL}	1.3		µs	SCL low.
t_{SCS}	0.6		µs	Setup time, relevant for repeated start condition.
t_{SCH}	0.6		µs	Hold time. After this period, the first clock is generated.
t_{DS}	100		ns	Data setup time.
t_{SCR}		300	ns	SCL rise time.
t_{SCF}		300	ns	SCL fall time.
t_{SDR}		300	ns	SDA rise time.
t_{SDF}		300	ns	SDA fall time.
t_{BFT}	0.6			Bus-free time. Time between stop and start.
MULTIPURPOSE PINS AND RESET				
t_{GRT}		50	ns	GPIO rise time.
t_{GFT}		50	ns	GPIO fall time.
t_{GIL}		1.5 × 1/f_S	µs	GPIO input latency. Time until high/low value is read by core.
t_{RLPW}	20		ns	$\overline{\text{RESET}}$ low pulse width.

[1] All timing specifications are given for the default (I²S) states of the serial input port and the serial output port (see Table 66).

National Semiconductors LM317 Voltage Regulator

LM317A and LM317 Electrical Characteristics (Note 3)

Specifications with standard type face are for $T_J = 25°C$, and those with **boldface type** apply over **full Operating Temperature Range**. Unless otherwise specified, $V_{IN} - V_{OUT} = 5V$, and $I_{OUT} = 10$ mA.

Parameter	Conditions	LM317A			LM317			Units
		Min	Typ	Max	Min	Typ	Max	
Reference Voltage		1.238	1.250	1.262	-	1.25	-	V
	$3V \le (V_{IN} - V_{OUT}) \le 40V$, $10\ mA \le I_{OUT} \le I_{MAX}$	**1.225**	**1.250**	**1.270**	**1.20**	**1.25**	**1.30**	V
Line Regulation	$3V \le (V_{IN} - V_{OUT}) \le 40V$ (Note 4)		0.005 **0.01**	0.01 **0.02**		0.01 **0.02**	0.04 **0.07**	%/V
Load Regulation	$10\ mA \le I_{OUT} \le I_{MAX}$ (Note 4)		0.1 **0.3**	0.5 **1**		0.1 **0.3**	0.5 **1.5**	%
Thermal Regulation	20 ms Pulse		0.04	0.07		0.04	0.07	%/W
Adjustment Pin Current			**50**	**100**		**50**	**100**	μA
Adjustment Pin Current Change	$10\ mA \le I_{OUT} \le I_{MAX}$ $3V \le (V_{IN} - V_{OUT}) \le 40V$		**0.2**	**5**		**0.2**	**5**	μA
Temperature Stability	$T_{MIN} \le T_J \le T_{MAX}$		**1**			**1**		%
Minimum Load Current	$(V_{IN} - V_{OUT}) = 40V$		**3.5**	**10**		**3.5**	**10**	mA
Current Limit	$(V_{IN} - V_{OUT}) \le 15V$ K, T, S Packages EMP Package H, MDT Packages	 **-** **1.5** **0.5**	 **-** **2.2** **0.8**	 **-** **3.4** **1.8**	 **1.5** **1.5** **0.5**	 **2.2** **2.2** **0.8**	 **3.4** **3.4** **1.8**	A
	$(V_{IN} - V_{OUT}) = 40V$ K, T, S Packages EMP Package H, MDT Packages	 - 0.112 0.075	 - 0.30 0.20		 0.15 0.112 0.075	 0.40 0.30 0.20		A
RMS Output Noise, % of V_{OUT}	$10\ Hz \le f \le 10\ kHz$		0.003			0.003		%
Ripple Rejection Ratio	$V_{OUT} = 10V$, f = 120 Hz, $C_{ADJ} = 0\ \mu F$		**65**			**65**		dB
	$V_{OUT} = 10V$, f = 120 Hz, $C_{ADJ} = 10\ \mu F$	**66**	**80**		**66**	**80**		dB
Long-Term Stability	$T_J = 125°C$, 1000 hrs		0.3	1		0.3	1	%
Thermal Resistance, θ_{JC} Junction-to-Case	K (TO-3) Package T (TO-220) Package S (TO-263) Package EMP (SOT-223) Package H (TO-39) Package MDT (TO-252) Package		- - - 23.5 21 12			2 4 4 23.5 21 12		°C/W
Thermal Resistance, θ_{JA} Junction-to-Ambient (No Heat Sink)	K (TO-3) Package T (TO-220) Package S (TO-263) Package (Note 6) EMP (SOT-223) Package (Note 6) H (TO-39) Package MDT (TO-252) Package (Note 6)		- - - 140 186 103			39 50 50 140 186 103		°C/W

Note 1: Absolute Maximum Ratings indicate limits beyond which damage to the device may occur. Operating Ratings indicate conditions for which the device is intended to be functional, but do not guarantee specific performance limits. For guaranteed specifications and test conditions, see the Electrical Characteristics. The guaranteed specifications apply only for the test conditions listed.

Note 2: Refer to RETS117H drawing for the LM117H, or the RETS117K for the LM117K military specifications.

Note 3: I_{MAX} = 1.5A for the K (TO-3), T (TO-220), and S (TO-263) packages. I_{MAX} = 1.0A for the EMP (SOT-223) package. I_{MAX} = 0.5A for the H (TO-39), MDT (TO-252), and E (LCC) packages. Device power dissipation (P_D) is limited by ambient temperature (T_A), device maximum junction temperature (T_J), and package thermal resistance (θ_{JA}). The maximum allowable power dissipation at any temperature is : $P_{D(MAX)} = ((T_{J(MAX)} - T_A)/\theta_{JA})$. All Min. and Max. limits are guaranteed to National's Average Outgoing Quality Level (AOQL).

Note 4: Regulation is measured at a constant junction temperature, using pulse testing with a low duty cycle. Changes in output voltage due to heating effects are covered under the specifications for thermal regulation.

Note 5: Human body model, 100 pF discharged through a 1.5 kΩ resistor.

Note 6: When surface mount packages are used (TO-263, SOT-223, TO-252), the junction to ambient thermal resistance can be reduced by increasing the PC board copper area that is thermally connected to the package. See the Applications Hints section for heatsink techniques.

Appendix B: Part List and Pricing

LM1875 Power Audio Amplifier	$3.98
EVAL-ADAU1701 Evaluation Board	$230.00
LM317 Adjustable Voltage Regulator	2x$83 = $1.66
UA741 General Op-Amp	$0.59
TL081 High Input Resistance Op-Amp	$5.00
Jensen 25W Speaker	$30.00
V-Infinity VF-S250-48A	$100.00
Resistors, Wires, Switch, Capacitors, Breadboard, Miscellaneous	$20.00

Appendix C: Analysis of Senior Project Design Form

Analysis of Senior Project Design

Please provide the following information regarding your Senior Project and submit to your advisor along with your final report. Attach additional sheets for your responses to the questions below.

Project Title: _Electric Guitar Amplifier With Digital Effects ____

Quarter / Year Submitted: _Fall 2011________

Student: (Print Name) _Shawn Garrett__________ **(Sign)** ____________________________

Advisor: (Print Name) _Dr. Pilkington________ **(Initial)** __________ **Date:** _02/14/2011_

• **Summary of Functional Requirements**

The design accepts an input signal from an electric guitar. You then have the option to adjust the volume, and provide digital effects to the original signal. The signal is then amplified so it may drive a 25W loudspeaker.

• **Primary Constraints**

This system consisted of three main stages, DSP, amplification, and power distribution. Each stage had its own challenges. Raw AC power from a standard 120V outlet needed to be regulated for the sensitive components of the system. The DSP stage needed to accurately reproduce the guitar signal, plus add the desired digital effects. Finally the amplification stage needed to amplify the signal to a sufficient level to drive the 25W speaker, while being powered by a single-rail power supply. Getting all of these stages to work together was the most significant challenge however.

• **Economic**

o Originally, the overall cost of the project was believed to be $455.

o The final overall component cost of the project was $367.

o *A bill of materials is provided in the proceeding appendix.*

o In order to complete the project, two breadboards, wire, a multimeter, an oscilloscope, a DC power supply, and a function generator were all required. All of these components were already owned, or were available for use free of charge.

o The expected development time was approximately eight months, with the addition of the three month Summer months if needed. The deadline was scheduled for mid-December, 2010.

o An additional month of development time was necessary due to component damage. The final deadline was set at November 9, 2011.

• **If manufactured on a commercial basis:**

o Assuming this project was completed, and packaged in a marketable fashion, as well as the consideration of manufacturing time per unit without aid, an estimated 30 units a year could be sold.

o The estimated cost of production per unit is $165

o The estimated retail price would be $190

o The estimated profit per year would be $750, or a profit of 15%.

o Assuming the system were run at peak performance for an entire hour without stopping every day, the cost to the electricity cost to the user would increase by approximately $1.30. This assumes current residential electricity costs.

• **Environmental**

As a result of the changing environmental views and regulations, most if not all of the components in this project were certified Led-free. When manufacturing this design, very little waste would be created, and most of that could be recycled. Excess wood from the construction of the enclosure can be ground up and used as compost for example.

Like most mid to high output consumer audio equipment, efficient components such as Class D Amplifiers aren't useable in this design. Therefore, this system can consume as much as 42W when driving a load. At idle however, the system requires less than 5W.

• **Manufacturability**
The project wasn't developed to the point of manufacturability, so it is impossible to know every outcome of the manufacturing process. In this design, the digital stage consisted of a prefabricated PCB based around the selected DSP chip. If this system were to be marketed, a custom PCB would be developed to save money.

• **Sustainability**

o The device wasn't entirely completed. This was due in part to interfacing issues between the DSP stage and the amplifier. Sustainability issues weren't the cause of the project not being completed. `

o The system requires electricity, which at this point is still a largely non-renewable resource. While the system doesn't have a power saving mode per se, it does require relatively little power when there is no input signal. Once manufactured, the unit generates no additional waste, and the vast majority of the components can be recycled or reused once the system has reached the end of its life.

o An upgrade that would make the system more sustainable, would to implement an automatic power-off circuit, which would power the system after a certain amount of idle time. This could be done using a digital timer, and power transistors.

o If a power saving circuit were implemented, there would need to be a way of detecting when user wishes to use the system again, whether it be an active signal from the guitar or input from the user interface. Therefore, there will still be some power usage in the system.

• **Ethical**
Due to the relatively loud volume capable of being produced by the system, the sound may be annoying or uncomfortable to others in the vicinity. It is the responsibility of the user to consider this possibility.

• **Health and Safety**
The potentially loud volume of the system's output may cause hearing damage to the user or others in the vicinity of the system. It is the responsibility of the user to ensure this doesn't occur. Electric shock is a possibility with the system's internal components, due to the relatively high current produced by the power supply. If this system were marketed, clear warnings would be in place to warn the user.

• **Social and Political**
There are no foreseeable political effects from this design. The only social issues would be caused by noise pollution, which is more the fault of the user and not the device.

• **Development**
I have had experience using all of the testing tools before I began this project. I did learn more about dealing with power amplifiers as a result of implementing the amplification stage of this project. Furthermore, I learned more about implementing a DSP system, which included the limitations that may be placed on the system as a whole. This project was also the first time I have ever researched, selected, purchased, and developed an entire system from the ground up. Granted many of the components were prefabricated, such as the power supply and the DSP board, but there were still several aspects that needed to be researched, as well as issues to be solved.