Design and Implementation of a Saturating Audio Interface

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ABSTRACT

This paper concerns the construction of a four-part microphone amplifier. The goal of the project is to explore the design process of a quality pre-amplifier and to craft a musically-interesting tool. First, the power supply steps down the wall AC voltage to DC, creating sources for all op-amps including phantom power. Second, the pre-amplifier gains dynamic or condenser microphones in two stages. Third, the line-level signal passes through a parametric equalizer. Last, the signal passes through a saturation unit and can be equalized again. The intention is that the tone produced by the equalizer-sandwiched saturator is shapeable. The samples can be fed into an ADC on a Daisy Seed microcontroller. The design constraints, challenges, and modifications are considered. Then, the total parameters of the prototype are evaluated and its musical functions are observed. Lastly, the anticipated future upgrades are discussed.

1. INTRODUCTION

In this project, a pre-amplifier is designed with built in pre- and post-equalizers about a saturation unit. The intent of the project is to evaluate the realistic non-linearities of audio-grade passive components while creating an interesting and expandable musical tool. When processing, audio engineers often trust in distortion to boost loudness precisely while creating harmonic color, creating a stronger signal body. This amplifier employs a rudimentary foldback saturation with tone control.

The pre-amplifier is intended to operate competitively when measured against a Scarlett Focusrite 2i2. The goals are high signal-to-noise ratio, low unintended distortion, and the capacity to output a line-level signal and sample it using a Daisy Seed-mounted analog-to-digital converter. The Seed is an inexpensive and tiny microcontroller that provides the potential to expand the interface for any task.

The equalizers allow the tone of the distortion to be shaped. Being parametric, the gain, resonance, and bandwidth of both bands can be controlled. The output band can be set opposite the input band to restore the balance of the original signal, or be used creatively to enhance an alternate portion of the signal.

The saturation unit does not induce simple clipping. Being a foldback unit, amplitudes exceeding the threshold are reflected back down upon it.

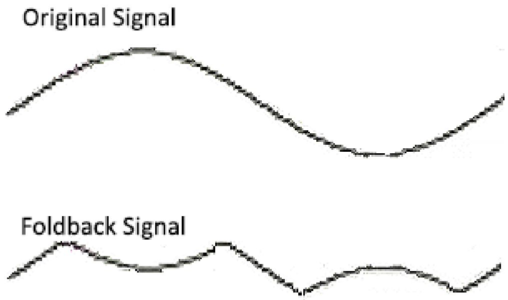

Figure 1. Foldback Distortion

Foldback saturation is preferred because it produces far less abrasive high-frequency harmonics when compared to clipping. With tone-shaping, it can accent a lower frequency, building a fuller body in any signal. With smoothing of the peaks about the threshold, fulfilled with analog amplification deliberately, the resultant achieves loudness effectively while compromising fidelity far less.

2. BACKGROUND

The goals of this project are four: design a pre-amplifier, configure an ADC, design EQs and a saturation unit, and construct housing. The diagrams are drawn using program easyEDA. The objectives of this project are to create a creative, expandable, and easily-shareable pre-amplification interface. All components are mountable on proto-boards, making the interface a reproducible do-it-yourself guide. Through-hole components and simple integrated circuit packages are prioritized.

Initially, the requirements of the power supply are evaluated. Given that audio products both professional and consumer rely on max peak amplitudes beneath 3.5 volts, a polarized 15 volt supply powers all operational amplifiers. A sufficient 48 volt phantom power is employed for condenser microphones.

Next, the pre-amplifier stage is designed. It needs to have high impedance inputs for each channel and sufficient audio specifications. The design should have minimal distortion and a high signal-to-noise ratio. Plenty of open-source designs for pre-amplifiers already exist, and are followed closely in this project. Because the design needs to be tested at each stage, the pre-amplifier circuit includes XLR inputs and potentiometers. A design by Jules Ryckebusch is observed as the main reference because it is comprehensive and flexible [1]. The design is evaluated using a guide from THAT Corp, which details simple amplifier circuits, the hidden "gremlins," and how to assess distortion and noise levels with varying gain [2]. Though the chosen design is compatible with many amplifier ICs, the SSM2019 is selected for this project. Following, an INA134 is included for gain control. They are available, inexpensive, well-packaged, and well-documented.

Then, the parametric bell equalizer is designed for use before and after the saturator. Four potentiometers each control the parameters of the band-shelves. Proper audio-grade filters are a central goal of this endeavor, and in this stage the largest circuit is implemented. The LM358 is chosen because it houses two amplification circuits, allowing the proto-board space to be easily distributed. The design is guided by user Swagatam, who details the implantation of op-amps in RC circuits to control the range of the equalizer's parameters [3]. It is inexpensive and meets satisfactory specifications.

Between the equalizers lies the saturation unit. Given it necessitates four amplifiers, including a buffer, it is accomplished with an LM324, having four amplification circuits but less connections to the provided polarized supply voltages. Though this elementary design may not produce a most pleasing distortion due to low slew rate, it is satisfactory and a guide is supplied by user Hamuro [4]. Two potentiometers are attached to control the input and output gains.

Last, the project is housed and the circuits are isolated. Switches and knobs are purchased from Do-It-Yourself Electronics in Stoneham, Massachusetts. Their prices and product specifications are far superior to any options available online. The housing is built with cardboard and the circuits are shielded using copper with felt padding.

In total, this project comprises a simple amplification component and an ambitious effects section. Pre-amplifiers are musically ubiquitous, and this project produces an inexpensive, effective, and semi-modular system to enhance recordings. Foldback distortion has been long a popular choice for saturation, and implementing it with a functional interface creates a useful instrument. This system will support digital and analog outputs, making it a flexible and creative interface.

3. PROJECT DESCRIPTION

3.1 Design Constraints

Because this project does not include the layout of a PCB, the constrains are few. Given that the design is installed on proto-boards, a small form is unreasonable to achieve. However, high voltages must be converted using solutions that produce the least amount of EMI, those being appropriate ICs. Considering that this interface is an amateur, original, and novel design, noise and distortion will not be optimal. Also, this interface is built for mono input and output. Only TR cables are applicable. Finally, the description of the interface does not detail an instrument-line input and only accommodates microphones.

3.2 Design Modifications

3.2.1 Power Supply

The initial design of the power supply included a 120 VAC to 24 VCT transformer feeding into a dual bridge rectifier with filtration, finally passing ~17 VDC signals into a pair of ±15 VDC regulators. This solution requires heavy, expensive components and also produces great EMI. To reduce interference and cost, a medium-sized Recom AC/DC converters is employed as the polarized 15 volt supply. For the same reasons, the phantom power supply is too supplied by a Recom IC.

3.2.2 Phantom Power

It was expected that most modern microphones do not require a full 48 volt supply. However, considering further research into many professional condenser microphones, as well as the supply voltage measured of the Scarlett, 48 volts is necessary for a comprehensive, professional audio interface. Because the prototype is expected to support 48 volts, the pre-amplifier circuit is updated to safely impede the connection to condenser microphones.

3.2.3 Equalizer

Mainly, the most substantial modification to this project is the decision to avoid 3-band parametric equalizer circuits. For shaping the tone musically, often only one bell band is necessary. Soldering a parametric bell takes a long time.

3.3 Implementation

3.3.1 Power Supply

The power supply is created using two Recom ICs fed from the live and neutral wires of a pre-purchased dissected wall outlet cable. The ±15 VDC provides supply for all operational amplifiers. The 48 VDC is used solely to power condenser microphones. This solution produces little EMI, and is further easily insulable.

Figure 3. Power Supply ICs

3.3.2 Pre-amplifier

The pre-amplifier brings the microphone signal to line-level. Consumer audio systems accept a maximum voltage amplitude of .894 volts. This system comprises phantom power, a 10 V/V amplifier, and an amplifier with 0 to 10 V/V. That is a gain total of 20. The initial stage boosts the microphone level with low noise. The next stage is controlled by potentiometer to move between no output to maximum. From testing, microphone voltages reach a reasonable level while the potentiometer is set to the maximum, above 300 mV. This is acceptable considering further gain through the following stages. This amplification stage requires one potentiometer for gain and one 48 VDC switch for phantom power.

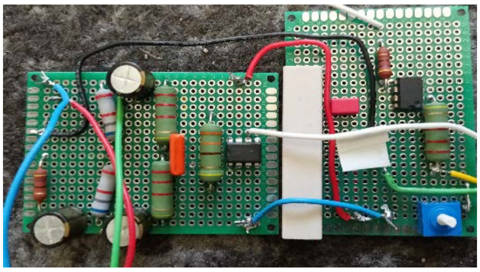

Figure 4. Pre-Amplifier Circuit

3.3.3 Parametric Equalizer Band

The equalizer band is a reproducible circuit that can be laid in series, fulfilling the semi-modular intent of this project. In this implementation, only the bell band is used to accentuate or attenuate the signal. It is constructed using three ICs that house two op-amps. Initially, the signal is buffered. Then, a low- and high-pass create a band-pass filter with controllable frequency. Finally, the resonance and gain of the band can be adjusted.

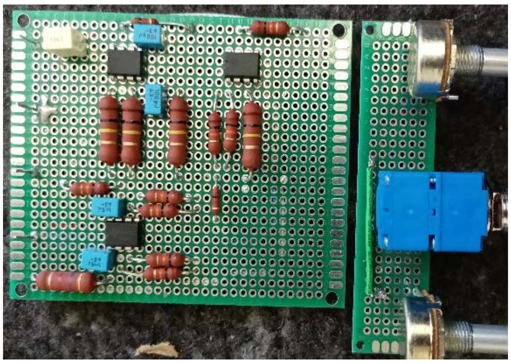

Figure 5. Equalizer Bell Circuit

3.3.4 Saturator

The saturation unit is built to fold the signal back. That is, as amplitudes exceed the threshold, they are reflected about it. To accomplish this, a mathematical equation is followed.

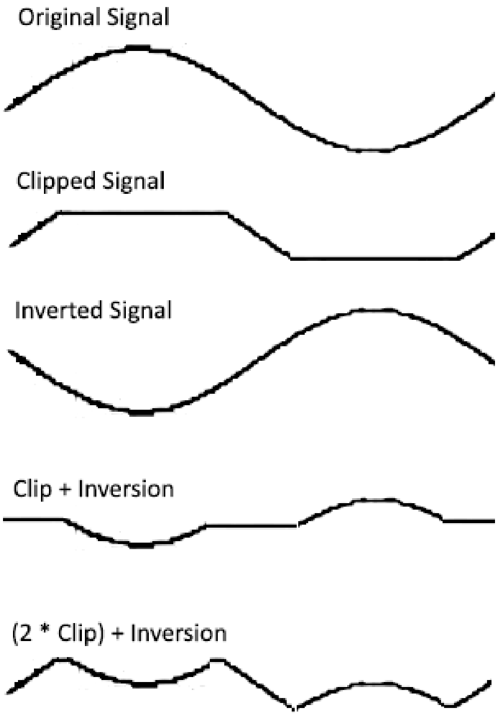

Figure 6. Foldback Formula

To achieve this configuration, a four-output amplification IC is employed. The first buffers the signal. The second provides the drive gain for the saturator. The third clips the signal at 1 volt using a network of diodes in feedback. The fourth acts as a summing amplifier, combining the clipped signal with the inverse of that which makes it clipped.

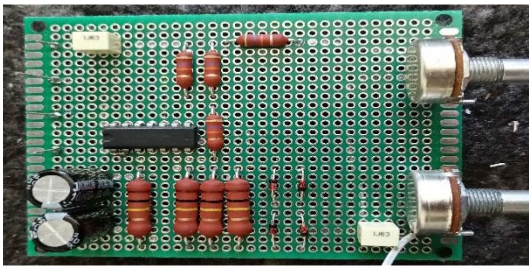

Figure 7. Saturation Circuit

3.3.5 Microcontroller

At the end of the chain, the signal can be sampled to a computer using the Daisy Seed microcontroller. It includes a microprocessor, data memory, and analog-to-digital converter. For this project, it uses an ADC input to read the level of the microphone. It is fit to include many effects and simply considering the semi-modular design of this prototype.

3.3.6 Housing

Avoiding noise interference is a main goal of this project. Made of cardboard, the housing is padded with highly insulatory metal sheets, isolated using felt pads. Further, there is an attention to insulation of exposed wiring.

3.4 Engineering Use

This project details a simple and cheap DIY interface made with accessible components and a well-documented microcontroller. The equalizer circuit can be run in series for more bands, providing greater tone control through the saturator. With slight foldback distortion, an audio signal can be recorded with a greater loudness and more predictable peaks for control with compression. The saturation can be driven radically to destroy sound in an interesting way, increasing loudness without the harshness created by clipping.

Considering the project is based around the Daisy Seed microcontroller, it is greatly expandable. Compressors, filters, and other effects can be implemented simply. However, though this design handles balanced audio signals, it is not suited for the processing of stereo signals. With the Seed, a stereo expansion can be implemented easily for communication with a computer.

3.5 Artistic Use

Musically, this paper describes an interesting and useful product. Though saturation is most often used in a processing chain following pre-amplification, compression, gating, and EQ, it can provide an unmatched perception of excitation when used early. Prior to gain staging and fader riding, this design can accentuate tense vocal moments and lower their peak amplitude while bringing the quieter portions closer in level. Proper placement of the vocalist to the microphone should be considered when recording. Moreover, this design works best in a live setting. It reduces peak amplitudes, gives more body to the instrumentalist or performer, and can be adjusted in the moment for volume, distortion, or equalization control.

4. DISCUSSION

4.1 Passive Components

All of the resistors and capacitors of this design lie in the audio signal path. Considering passive components do exhibit non-linearities, it is important to source audio-quality components. Research from engineers advises that wirewound, high wattage-rated resistors are the standard for audio technology [5]. Premium capacitors are made of film, particularly polypropylene [6]. Mica capacitors are superior, though PPL proves effective. In one case, in the saturation unit, back-to-back electrolytic capacitors are used for a smoothing application. Aluminum electrolytic capacitors also provide low total harmonic distortion.

4.2 Buffering

Signal buffers are used in each equalizer stage and the saturator. A buffer requires one operational amplifier and supplies unity gain, but it prevents the current supply to each circuit from interference from the load created by resistances, other amplifiers, and potentiometer changes. The buffers achieve effective impedance matching between each stage, stabilizing the audio signal path.

4.3 Noise

With the implemented components, the pre-amplifier stage should introduce less than .005% total harmonic distortion, considering datasheets. Though the signal is amplified cleanly, there is measurable noise in the signal. Through the equalizers, the THD may be as high as .003%. In the saturator, up to .002% is expected. However, the distortion stage is intended to distort, thus, higher saturator input voltages should be avoided if the intention is to not distort.

To reduce noise interference between separated, powered components, copper is installed in the housing. The power supplies and effects section are insulated. From measurement, the insulation does nothing to reduce noise. It is assumed that noise is created from poor and inequal wiring connects and environmental interference.

4.4 Error

In constructing this device using solely through hole components, many errors were made. Though there were no solder-bridging nor connection errors, many techniques were learned. It is important to attempt to use component leads to create bridges between related components. It is to be noted, however, that exposed leads can create and intercept greater electromagnetic interference. When designing audio devices, it is important to match wire lengths, particularly given the effect of interference from capacitors at long-range. In this project, wire length was ignored. Further, power wires are not intwined to reduce environmental interreference.

5. CONCLUSIONS AND FUTURE WORK

This paper describes the construction of a microphone pre-amplifier, saturator with pre- and post-equalizers, and an ADC. Through this project, passive component characteristics, effective amplifier ICs, circuit design, and circuit implementation were studied. The final design works sufficiently, though it likely does not compare to a cheap commercial amplifier given its housing and through-hole design.

Through designing, it is important to use the component leads as much as possible to bridge connections and bridge them closely. When building with through-hole components, it is important to maximize space.

This project can be expanded in many ways, considering its modular capacity with analog circuitry and the digital possibilities of the Daisy Seed. Each hardware component is modular, given the use of buffers, and software effects can be implemented simply with few connections. First, more equalizer bell bands can be run in series. Next, the Seed can be used to create any digital effect.

6. ACKNOWLEDGMENTS

I am thankful for the assistance given by the employees at You-Do-It Electronics and Mr. Music. In different ways, both assisted me in understanding EMI protection and knob standardization. Anyone who enjoys the control of music should visit both of these establishments. You-Do-It has a wealth of high-power passive components, audio components, and general tools available off the shelf. In the greater Boston area, there is simply no other place to go.

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