Master Bus Processor

Team 14 - ECE 445 - Spring 2019
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Project Proposal - TA: Zhen Qin

1. Introduction

1.1. Objective:

A common issue facing aspiring musicians is the very high cost barrier to entry. Integrated and digital circuits have allowed home studios to become increasingly popular, however these digital processors force the user miss out on the more traditional acoustic properties of formal analog processing equipment, which historically have been very costly and essentially made to order by manufacturers. This is not ideal for most musicians as they prefer to use analog processing over digital processing to differences in how the sound is perceived by a listener. As of right now there is no affordable option designed for the home user, and certainly nothing that combines the performance of analog processing with the ease of digital control.

The objective of our Master Bus Processor is to create a solution tailored for a home user by offering an affordable product that is simple to use and understand. By combining analog audio processing with digital control, we can create a very flexible design will still keeping costs down and and maintain high quality audio performance.

1.2. Background:

With the advent of digital recording technology, more musicians and producers are adopting home recording than ever before. While modern digital audio software and hardware provide unparalleled flexibility, home recordists often lament that digital audio equipment does not provide the same musical character as the analog equipment found in commercial studios. This is due in part to the harmonic enhancement provided by the inherent, subtle non-linearities of analog audio circuits. While some audio equipment manufacturers have created analog processors which could be used to reintroduce these qualities, this equipment is not affordable or accessible to home recording studio owners.

Our goal is to combine the desirable audio properties of analog processing circuits with the flexibility and control of a digital system in order to produce a hybrid analog-digital signal processor which can be used in home studios. By designing with modern manufacturing techniques in mind, we plan to develop a Master Bus Processor which is both high quality and affordable to home recordists.

1.3. High-level requirements list:

There are three main quantitative characteristics that a successful project will exhibit:

1. *Affordability:* Must be affordable enough to reach the intended prosumer and home studio user. This would mean a sub-$2500 price point.
2. *Performance:* Must be able to provide the non-linear audio response of analog processing. There should be proper implementation of compression, equalization, and saturation control.
3. *Accessibility:* Device controls should be intuitive and simple enough for an amature user. Must provide all the necessary basic functionality and flexibility for this user base.

2. Design

2.1. Block Diagram:

Our solution consists of both digital and analog circuits to allow for the flexibility of digital control with the audio quality of analog processing. We will be using a microcontroller to control relays in the analog circuit that can change the orientation and exposure of different analog component blocks. This will allow the user to use a series of buttons and an LCD display to control circuit parameters such as the order of each analog block and the particular mode that each block performs its desired function. All audio processing will be performed using analog circuits and the user will be able to use knobs to adjust specific analog block parameters.

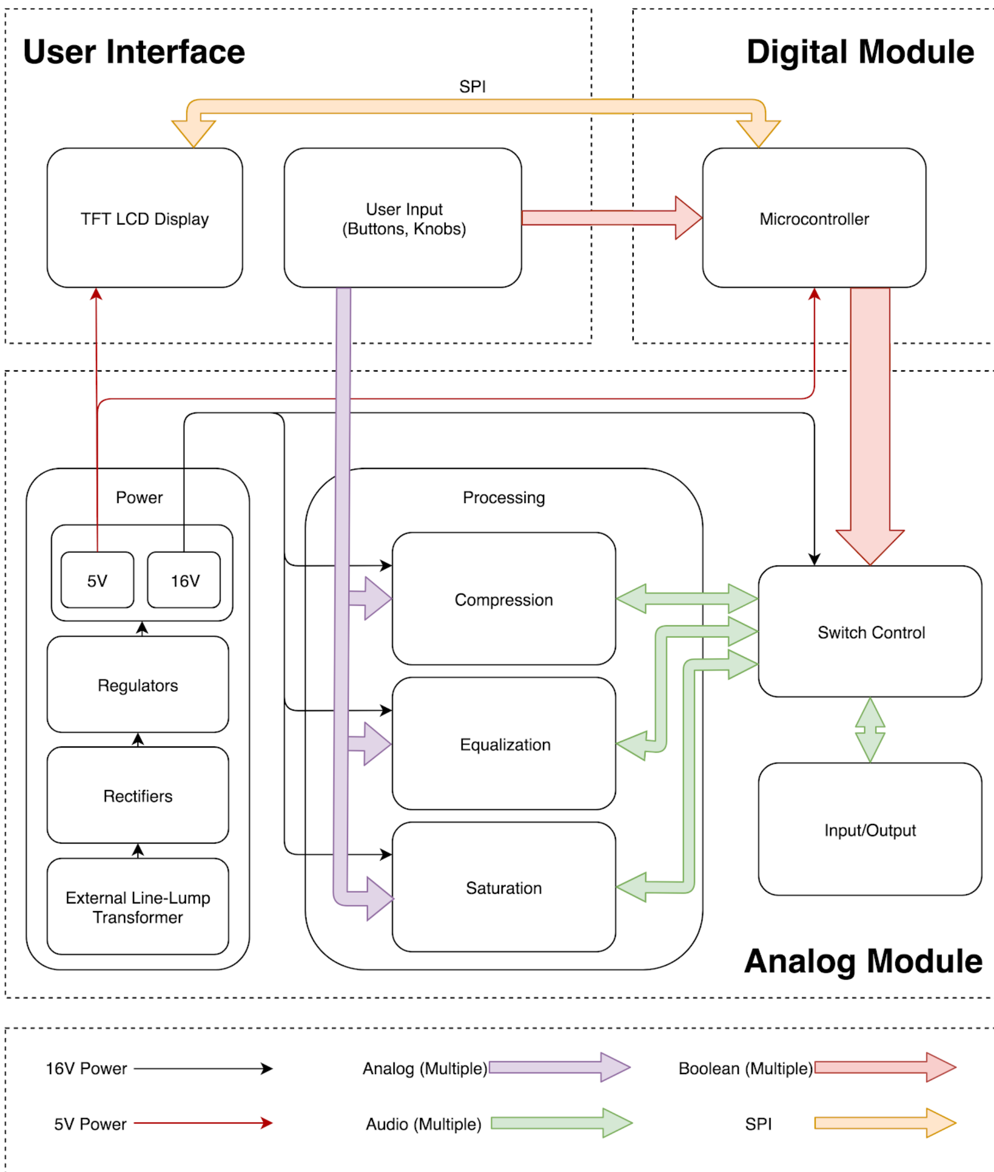

Figure 1. Block Diagram

2.2. Physical Design:

The Master Bus Processor (MBP) will be designed to fit inside an industry-standard 19” rackmount enclosure. The front control panel will occupy two rack-units (2U). As a *reach goal* we will design a custom 19” rackmount enclosure for the MBP.

Figure 2. Standard 2U Rackmount Enclosure - Amazon.com [2]

The User Interface--consisting of the LCD screen, selection buttons, and processor control knobs, will be mounted on the front panel of the enclosure.

Inside the enclosure, the system will be divided into two separate printed-circuit-boards (PCB’s) in order to reduce noise from interference. The power systems, analog processing blocks, I/O connections, and control knobs will be mounted on a large, base PCB. The digital controller will be housed on a separated PCB which will be mounted above the analog PCB. The two PCB’s will exchange power and control signals through a central ribbon cable connection.

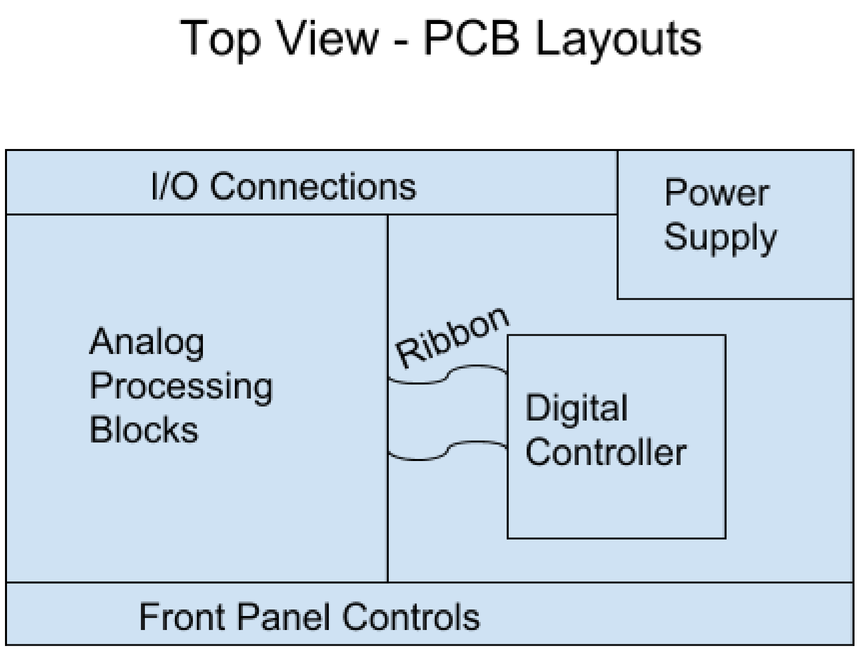

Figure 3. PCB Layouts

2.3. Functional Overview:

2.3.1. Analog

1. *External Line-Lump Transformer*

    The External 'Line-Lump' Transformer will convert US AC wall mains power (120V 60Hz) to levels which are tolerable by the Voltage Regulators down-line.

    Requirement 1: Must convert 120V AC to 20-24V AC

    Requirement 2: Must safely provide 500mA to 2A current

    Requirement 3: Must be non-switching, transformer only line-lump

2. *Rectifiers*

    The AC input voltage provided by the External Transformer will be converted to DC by diode Rectifiers and smoothing capacitors in preparation to be received by non-switching Voltage Regulators down-line.

    Requirement 1: The Rectifiers must convert the voltage AC power input to dual positive and negative DC voltage buses

    Requirement 2: The Rectifiers must provide at least 2000µF total smoothing capacitance

    Requirement 3: The Rectifiers must feature high-power rated impedance to suppress input current surges

3. *Regulators*

    The Voltage Regulators will accept the smoothed positive and negative DC buses from the Rectifiers and output stable voltage buses to power the analog and digital systems.

    Requirement 1: The Voltage Regulators must be non-switching (linear)

    Requirement 2: The digital power Voltage Regulator must provide 5.0-5.1V for the digital control system

    Requirement 3: The analog power Voltage Regulator must provide 15.90-16.10V for the analog control system

    Requirement 4: The Voltage Regulators must be trimmable to achieve precise output voltages

4. *Compression*

The Compressor will be an automatic-gain-controller that is specialized for use with audio signals. When the audio signal at the input of the compressor exceeds a fixed threshold level, the Compressor will reduce its gain by a proportional amount until the signal has fallen below the threshold again. The output of the Compressor is then normalized so that the average amplitude of the audio signal has been increased. By this process, the Compressor is able to improve the perceived loudness of the audio signal. In order to reduce distortion and act as linearly as possible, the Compressor operates on a timescale that spans at least multiple low-frequency wave cycles. The Compressor can be connected in various places in the analog signal chain, where it will interact differently with the other analog processing blocks. As a *reach goal* the Compressor will be able to switch into a 'Limit' mode and act as a peak-limiter.

Requirement 1: The Compressor must be able to provide at least 20dB gain reduction without overloading its input.

Requirement 2: The Compressor must be able to engage full gain reduction within at least 100ms.

Requirement 3: The Compressor must be able fully disengage gain reduction in a range of settings from 100ms to 1s.

5. *Equalization*

The Equalizer will be series of high and low-frequency focused filters that can be applied to the audio signal. The filters in the Equalizer block will allow the user to boost or cut certain bands of the audio spectrum in order to shape the overall tonal response of the Equalizer block. The Equalizer will utilize an operational amplifier (op amp) gain stage with filters connected to the inverting and non-inverting inputs in order to achieve active equalization. The Equalizer can be connected in various places in the analog signal chain, where it will interact differently with the other analog processing blocks. As a *reach goal* the Equalizer will be able to switch between 'Shelving' and 'Bell' modes so that the user can select different filter types in addition to controlling the frequency response.

Requirement 1: The Equalizer must provide at least two controllable frequency bands

Requirement 2: One of the available bands must be in the region of 60-150Hz

Requirement 3: One of the available bands must be in the region of 5-10kHz

Requirement 4: The Equalizer must not introduce significant additional distortion below the voltage 'headroom' clipping level.

6. *Saturation*

    The Saturator block will allow the user to enhance the harmonic content of the audio signal by mixing in small amounts of harmonic distortion. The Saturator will use an op-amp gain stage with diode-incorporated feedback network in order to provide non-linear gain. The output from this gain stage will then be added back to the original audio signal in small amounts by the user through a ‘blend’ control circuit and summing amplifier stage. The Saturator can be connected in various places in the analog signal chain, where it will interact differently with the other analog processing blocks. As a *reach goal* the Saturator will be able to switch between ‘Asymmetric’ and ‘Symmetric’ modes so that the user can choose between the different harmonic orders emphasized by each mode.

    Requirement 1: The Saturator must generate harmonic distortion through ‘soft clipping’

    Requirement 2: The Saturator must be able to contribute 0-1% total harmonic distortion to the audio signal

7. *Input/Output*

    The I/O section will consist of the hardware connectors and electronic signal balancing circuits which allow the Master Bus Processor to be connected to other professional audio equipment. The I/O connections will use differential amplifiers receive and drive signals to and from other equipment and a user-selectable hardware bypass will be incorporated to directly connect the inputs and outputs of the MBP when necessary.

    Requirement 1: The I/O connections must be fully balanced

    Requirement 2: The I/O connections must use industry standard ¼” TRS or 3-pin XLR connectors

    Requirement 3: The I/O connections must not overload at less than +/-15V swing

8. *Switch Control*

    The switch control will take boolean inputs from the microcontroller to orient and expose the different analog blocks. It will consist of a series of power transistors and double contact relays. This will allow the microcontroller to change the layout of the analog circuits based on what is inputted by the user. A fail-safe route will be provided in the case of a digital circuit failure.

    Requirement 1: Must be rated for use with audio signal

    Requirement 2: Must not allow discharges onto the audio path when switching

    Requirement 3: Must input less than 40 mA per control line, 200 mA total, from I/O pins on microcontroller

2.3.2. Digital

1. *Microcontroller*

The microcontroller will send boolean output signals to the switch control to alter the layout of the analog signal blocks. It will also take input from the user using a series of hardware buttons on the front of the 2U case that allow the user to intuitively interface with the microcontroller. Feedback of these inputs will be displayed on the LCD display. As a *reach goal*, the microcontroller will perform elementary dsp functions to give some feedback of analog block performance as well as store contents of an onscreen manual for the user.

Requirement 1: Must have at least 32 output pins for connection to switch control
Requirement 2: Must have at least 5 input pins for connection from buttons
Requirement 3: Must support SPI and have required pins for this standard.
Requirement 4: Must have non-volatile storage for program data (>100 KB)
Requirement 5: Must use open source standards and libraries for cost/reliability

2.3.3. User Interface

1. *LCD Display*

A display will allow the user to see the current settings of the digital controls including the order of the analog blocks, and their current mode. The display will communicate with the microcontroller using a SPI connection.

Requirement 1: Must be a full color TFT display
Requirement 2: Must support SPI to allow for connection to microcontroller
Requirement 3: Must fit in a 1U space (1.75” tall)

2. *User Interface*

A series of buttons will allow the user to interface with the microcontroller. There will be buttons labeled: sequence, compression, equalization, saturation, and Enter. These buttons will allow the user to interact with the microcontroller to set circuit parameters.

Requirement 1: Must have ample travel to allow for easy feedback to user
Requirement 2: Must be debounced for use with microcontroller

A series of knobs will be separated based on their corresponding analog processor block. They will allow the user to make circuit alterations in real time.

Requirement 1: Must be able to give user feedback on current angle
Requirement 2: Must be rated for audio manipulation.

2.4. Risk Analysis:

The switch control is the block that poses the most risk to the successful completion of our product. It is an integral part of our design as it is what allows the digital control to alter the design of the analog circuit. We will need to make sure that the PCB layout of the relays and routes are correct as it cannot be changed once the board is printed. This means we will have to carefully test and verify our relay logic before applying it to the final product in order to minimize risk of failure.

Each set of relays must be able to switch between three different states. Each analog processing block must be able to connect to the other two processing blocks as well as the input and output, depending on the user input. Different bias circuits that determine the analog block mode can also be exposed using relays controlled in the switch control module. These functionalities are a core feature of our design and set our product apart from other products currently on the market.

Because of this risk, we will include a fail-safe that will still maintain full functionality of the analog hardware. This will be done by having the relays default to a certain setting that will put the analog processing blocks into a predetermined setting. This means the user won't be able to make alterations, however the product will still perform audio processing capabilities.

3. Ethics and Safety

Because our device will use wall power, it is important that all safety guidelines are adhered to in respect to the design of the power block of our device. Improper circuit design can damage other circuits in our device as well as other devices connected to the input and output connections. Most importantly, improper design can lead to injury of the user, due to electrocution. In the scope of this class, we hope to mitigate these concerns by following the direction of course supervisors for power supply design. In the case of an ultimate go to market strategy, UL certification will be obtained to allow potential clients to know that our product is safe to purchase and interact with.

A potential breach of ethics in our project stems from the use of open source hardware/software. Open source projects allow use greater flexibility, affordability, and reliability using work that is not our own. Although this work is legal to use in the scope of our project, it is important that we follow proper ethics of the open source community. Specifically, it is important to not claim anyone else's work as your own work as this would go against IEEE Code of Ethics #7: "...to credit properly the contributions of others" [1]. To avoid this breach of ethics we will explicitly credit any contributions to our project from open source hardware/software projects and make sure that protected intellectual property is not copied without consent.

References

[1] ieee.org, "IEEE Code of Ethics", 2019. [Online]. Available: http://www.ieee.org/about/corporate/governance/p7-8.html. [Accessed: 7- Feb- 2019].

[2] Amazon.com, “New 2U Rackmount Enclosure ET2/35B - Electronic Enclosures | Circuit Specialists”
https://www.amazon.com/New-Rackmount-Enclosure-ET2-35B/dp/B01M5999AM