Master Bus Processor
Team 14 - ECE 445 - Spring 2019
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Design Document - 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 due 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:**

The current market for Master Bus Processors is fairly limited, especially for a more casual, home user. The market is also split between digital, which is generally cheaper, and analog, which is very expensive, especially for a home user. What our product will offer is a hybrid of the two. All of the audio processing circuits are strictly analog, and these are controlled by a digital interface. The analog circuitry is so important because many musicians and audiophiles claim that analog audio sounds “much deeper and fuller”, and as a result is much more desirable [1]. Our MBP will make use of digital control of the circuits in order to provide ease of use to the user. Many of the products on the market have many different knobs and controls that are initially confusing to use and take a while to learn and get used to. By offering an LCD screen as well as minimal knobs and buttons to control the device, we can make it much easier to use for a more casual user.

Specifically, what makes our product unique from others that are currently available is the use of digital controls on an analog circuit, use of SMD/SMT components, control of both channels, and the ability to rearrange the order of processing blocks. When comparing to the Rupert Neve Designs Portico II Master Buss Processor, we can find many fundamental differences in the two designs [2]. The Portico II MBP is entirely analog and consequently does not allow the user to change the order of processing blocks. Our design uses digital control to allow the user to select any order of compression, equalization, and saturation through a very simple button and screen
interface. The Portico II MBP also cost $4,000 while ours is projected to be under $2,500, which is significantly cheaper and more attractive to a home user.

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 with at least 20dB gain reduction and 100ms response, equalization with at least 2 EQ bands centered at 55-65 Hz and 9-11kHz boost/cut of at least 10dB at each band, and saturation with the ability to provide 0-1% total harmonic distortion to the mix.
3. *Accessibility:* Device controls should be intuitive and simple enough for an amateur user. Controls should be limited to at most 2 knobs per analog block, 5 total buttons and an LCD display with at most 4 menu screens.
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](image-url)
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 [3]](image)

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.

**Top View - PCB Layouts**

![Figure 3. PCB Layouts](image)
2.3. **Block Design:**

2.3.1. **Analog**

1. **AC/DC Conversion and Regulators**

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. 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.

<table>
<thead>
<tr>
<th>Requirement</th>
<th>Verification</th>
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<tbody>
<tr>
<td>Must convert 120V AC to 20-24V DC</td>
<td>Input 120V AC and measure open-circuit voltage confirming below 24 VDC</td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>Requirement</th>
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</table>
| must provide 4.9-5.1V for the digital control system and safely provide 500mA to 2A current | - Measure open-circuit voltage and ensure that it is below 5.1V  
- Connect a resistive load until voltage reaches 4.9V and ensure that at least 500mA current available using an ammeter |
| must provide 15.90-16.10V for the analog control system and safely provide 500mA to 2A current | - Measure open-circuit voltage and ensure that it is below 15.10V  
- Connect a resistive load until voltage reaches 14.9V and ensure that at least 500mA current available using an ammeter |
2. 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. The Compressor circuit is based on the THAT Corp 4305 Analog Engine standard application circuit [5] with adjustments to accommodate 2-channel operation.

<table>
<thead>
<tr>
<th>Requirement</th>
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<tbody>
<tr>
<td>must be able to provide at least 20dB gain reduction without overloading its input.</td>
<td>Input a 1KHz sine wave. Measure the input and output an oscilloscope and a verify at least 20dB gain reduction at maximum setting. Verify stability of output using oscilloscope.</td>
</tr>
<tr>
<td>must be able to engage full gain reduction within at least 100ms.</td>
<td>Input a 1KHz sine wave. Measure the input and output an oscilloscope and a verify response time in MATLAB.</td>
</tr>
<tr>
<td>must be able fully disengage gain reduction in a range of settings from 100ms to 1s.</td>
<td>Input a 1KHz sine wave. Measure the input and output an oscilloscope. Set to maximum setting and record response time in MATLAB, ensuring at least 1s. Repeat experiment at lowest setting and verify that response time is at most 100ms.</td>
</tr>
</tbody>
</table>

3. 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. The Equalizer circuit is similar to a standard “graphic equalizer” layout, as described by Rod Elliott [6]. Each band of the Equalizer contains a resonant RLC circuit which allows the frequency bands to be boosted or attenuated by connecting the resonant circuits to the feedback network of an op-amp.

<table>
<thead>
<tr>
<th>Requirement</th>
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<tbody>
<tr>
<td>One available band must be in the region of 55-65Hz</td>
<td>Sweep input signal frequency from 20Hz-20kHz, measure relative gain at 60Hz</td>
</tr>
<tr>
<td>One available band must be in the region of 9-11kHz</td>
<td>Sweep input signal frequency from 20Hz-20kHz, measure relative gain at 10kHz</td>
</tr>
<tr>
<td>must not introduce significant additional distortion below the voltage ‘headroom’ clipping level (+/-14V).</td>
<td>Input a 1KHz sine wave. Measure the input and output an oscilloscope. Set to maximum setting and verify total harmonic distortion is within range, set to minimum setting and verify also within range.</td>
</tr>
</tbody>
</table>

4. *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. The Saturator circuit is based on a standard soft-clipping “fuzz” circuit, as described by Rikupetteri Salminen [7]. The Saturator incorporates diodes into the feedback network.
network of an op-amp in order to produce highly non-linear gain. The output from this stage is then summed back with the original, unaffected signal at a very low level in order to give a subtle effect.

<table>
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<tr>
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<tbody>
<tr>
<td>must be able to contribute 0-1% total harmonic distortion to the audio signal</td>
<td>Input a 1KHz sine wave. Measure the input and output an oscilloscope. Set to maximum setting and verify total harmonic distortion is within range, set to minimum setting and verify also within range.</td>
</tr>
</tbody>
</table>

5. **Input/Output Balancing**

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. The Inputs and Outputs utilize the THAT Corp 1200 and 1646 Line Receiver/Drivers and the recommended implementation circuits [8] [9], respectively.

<table>
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<th>Requirement</th>
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<tbody>
<tr>
<td>Must use ¼” TRS or 3-pin XLR connectors</td>
<td>Confirm that connectors correspond to desired standards</td>
</tr>
<tr>
<td>must not overload at less than +/-14V swing</td>
<td>Input a 1KHz sine wave with magnitude +14V. Measure the input and output an oscilloscope and confirm stability using matlab</td>
</tr>
</tbody>
</table>

6. **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 | Verification
--- | ---
Must have less than 0.01% distortion | The test circuit from figure 4 will be built and connected to oscilloscope. Biased in the on position, input and output signals will be compared in MATLAB.

Must not allow discharges onto the audio path when switching | The test circuit from figure 4 will be built and connected to oscilloscope. Biased in the off position, output signals will tested for leakage.

Must input less than 40 mA per control line, 200 mA total, from I/O pins on microcontroller | The test circuit from figure 4 will be built and input control current will be measured at both logic high and low.
2.3.2. Digital

1. Microcontroller

The microcontroller (included in figure 5) 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.

<table>
<thead>
<tr>
<th>Requirement</th>
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<tbody>
<tr>
<td>Must have at least 20 output pins for connection to switch control</td>
<td>Datasheet will be referenced for number of digital output pins</td>
</tr>
<tr>
<td>Must have at least 5 input pins for connection from buttons</td>
<td>Datasheet will be referenced for number of digital input pins</td>
</tr>
<tr>
<td>Must support SPI and have required pins for this standard</td>
<td>Datasheet will be referenced for communication standard supported</td>
</tr>
<tr>
<td>Must have non-volatile storage for program data (&gt;10 KB)</td>
<td>Datasheet will be referenced for total amount of flash memory</td>
</tr>
<tr>
<td>Must have at least 5 analog input pins for reading voltage ranges</td>
<td>Datasheet will be referenced for number of analog pins with DAC support</td>
</tr>
</tbody>
</table>

2.3.3. User Interface

1. LCD Display

A display (included in figure 5) 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.

<table>
<thead>
<tr>
<th>Requirement</th>
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<tbody>
<tr>
<td>Must support SPI to allow for connection to microcontroller</td>
<td>Datasheet will be referenced for communication standard supported</td>
</tr>
</tbody>
</table>
2. **User Interface**

A series of buttons (included in figure 5) 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.

<table>
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<th>Requirement</th>
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<tbody>
<tr>
<td>Must travel at least 2 mm to allow for easy feedback to user</td>
<td>Press button and measure travel using a ruler</td>
</tr>
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A series of knobs (potentiometers) will be separated based on their corresponding analog processor block. They will allow the user to make circuit alterations in real time.

<table>
<thead>
<tr>
<th>Requirement</th>
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<tbody>
<tr>
<td>Must maintain desired position</td>
<td>Turn knob and release. measure voltage differential on leads using a voltmeter for consistency</td>
</tr>
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2.4. Schematics:

Figure 5. Digital Control Board Schematic
Figure 6. Analog Processing Schematic Overview
Figure 7. Analog Power Supply Schematic
Figure 8. Analog Hardware-Bypass Schematic
Figure 9. Analog Digital-Connector Schematic
Figure 10. Analog Controls Schematic
Figure 11. Analog Inputs Schematic
Figure 12. Analog Input Relays Schematic
Figure 13. Analog Compressor Schematic
Figure 14. Analog Compressor Relays Schematic
Figure 15. Analog Equalizer Schematic
Figure 16. Analog Equalizer Relays Schematic
Figure 17. Analog Saturator Schematic
Figure 18. Analog Saturator Relays Schematic
Figure 19. Analog Outputs Schematic
Figure 20. Analog Output Relays Schematic
2.5. **PCB Layouts**

Figure 21. Digital Control Board PCB

Figure 22. Display Adapter Board PCB
Figure 23. Analog Board PCB Overview
Figure 24. Analog Board: Power Supply Layout

Figure 25. Analog Board: Analog Controls Layout

Figure 26. Analog Board: Input/Output Layout
Figure 27. Analog Board: Compressor Layout
Figure 28. Analog Board: Equalizer Layout
2.6. Flowcharts

![Flowchart Image]

Figure 30. Controller Flow Chart
2.7. **Discussion of 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. In our project we will be using an ATmega microcontroller as well as the Arduino bootloader. Open source projects allow us greater flexibility, affordability, and reliability by 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 guidelines 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” [4]. To avoid this breach of ethics we will explicitly credit any contributions to our project from ATmega hardware and Arduino software and make sure that protected intellectual property is not copied without consent.
3. **Citations:**


