Class-D Audio Amplifier with a 5-Band Equalizer

Senior Design Document

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Alan Stewart, Dana Conrad, Troy Bowers, Kyle Maginot, Jing Li, Xilu Wang

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Team

Client/Advisor:

Ayman Fayed **aafayed@iastate.edu**

Team Info

Bowers, Troy Lynn, Communicator
tlbowers@iastate.edu
702-481-5777

Conrad, Dana Jay
halborr@iastate.edu
319-830-3109

Li, Jing, Key Concept Holder
kobelk@iastate.edu
515-708-0218

Maginot, Kyle Paul
kyle0729@iastate.edu
630-532-9026

Stewart, Alan Autry, Leader
alans@iastate.edu
712-943-1525

Wang, Xilu, Webmaster
xiluw@iastate.edu
515-817-3852

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Definition of Terms

Abbreviations

EQ- Equalizer
LPF- Low Pass Filter
HPF- High Pass Filter
BPF- Band Pass Filter
I/O- Input and Outputs
Op Amp- Operational Amplifier

Executive Summary

This project targets the implementation of a class-D audio amplifier system with a 5-band equalizer. An audio signal is to be sent through an equalizer system that can emphasize or de-emphasize certain bands and then amplified via a switching amplifier to 200 watts.

System Design

Requirements:

Functional

Class-D amplifier
A Class-D amplifier is required in the project description with the intent of developing a high-efficiency design at a high power level.

Equalizer
With the hearing spectrum divided into 5 bands, the user will be able to adjust the amplitude of one or more frequency ranges independently of the others.

Our Equalizer system is required to be a 5-band audio equalizer. Beyond that definition, it was left up to us what our specification should be. We chose to break up the 20 Hz - 20 kHz spectrum in the following way:

Filter	High-Pass 3dB Frequency	Low-Pass 3dB Frequency
Low Pass Filter		80 Hz
Low Band-Pass Filter	90 Hz	325 Hz
Middle Band-Pass Filter	300 Hz	900 Hz
High Band-Pass Filter	900 Hz	3000 Hz
High Pass Filter	4000 Hz	

Table 1 - Current Filter Range Definition

Non-functional

SNR
The fidelity of the audio output should be better than 96dB SNR.

Power System
Power system should be able to drive a high power speaker. For example: 200W.

Power System Efficiency
Power System Efficiency should be higher than 80%.

Operability
Users should be able to change the scaling factors on any the five bands, and there should be visual indicator that shows the amplified level.

System Description

Overview

Block Diagram

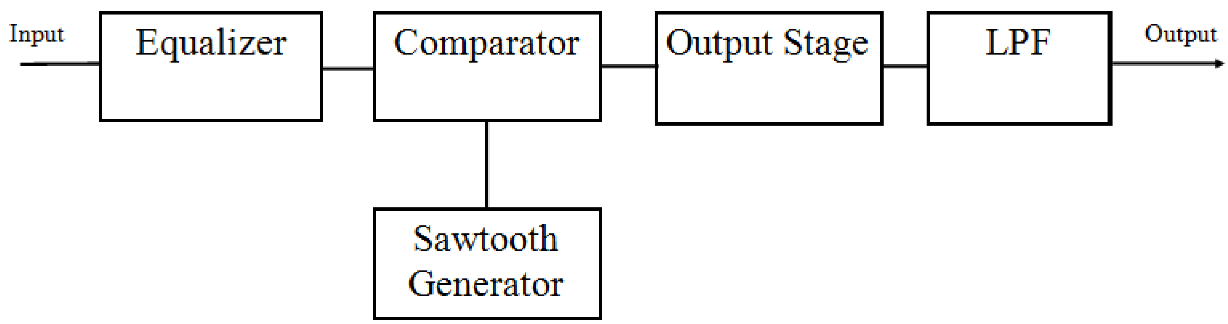


Table 2. Initial System Block Diagram

Equalizer

Overview

The equalizer system is divided into five separate filters in order to be able to attenuate each band individually. We have a high-pass filter, low-pass filter, and 3 band-pass filters in parallel. After filtering, the level of each band is adjusted in a mixer. The output of the summing amplifier is then passed to the amplifier section.

Block Diagram

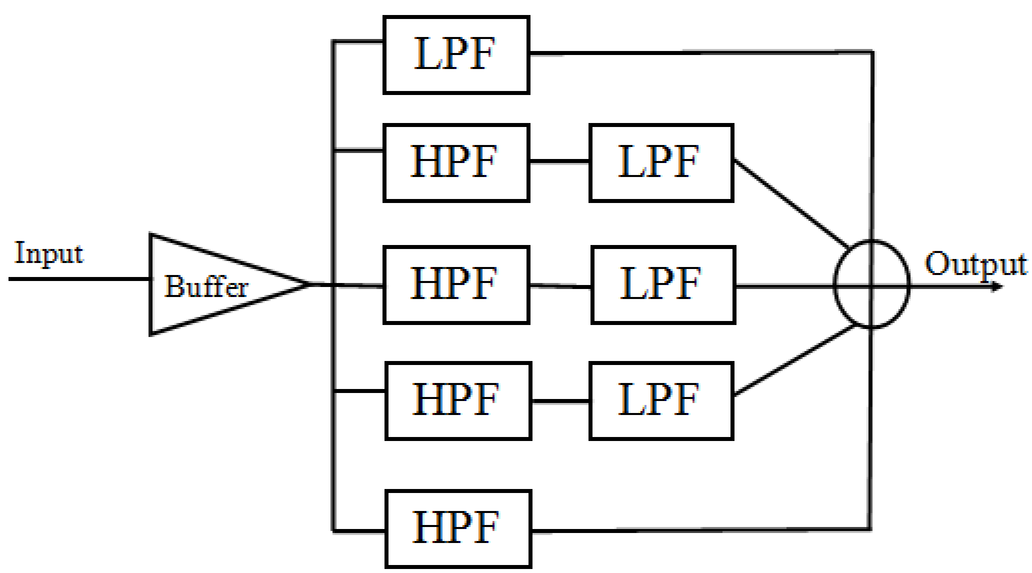


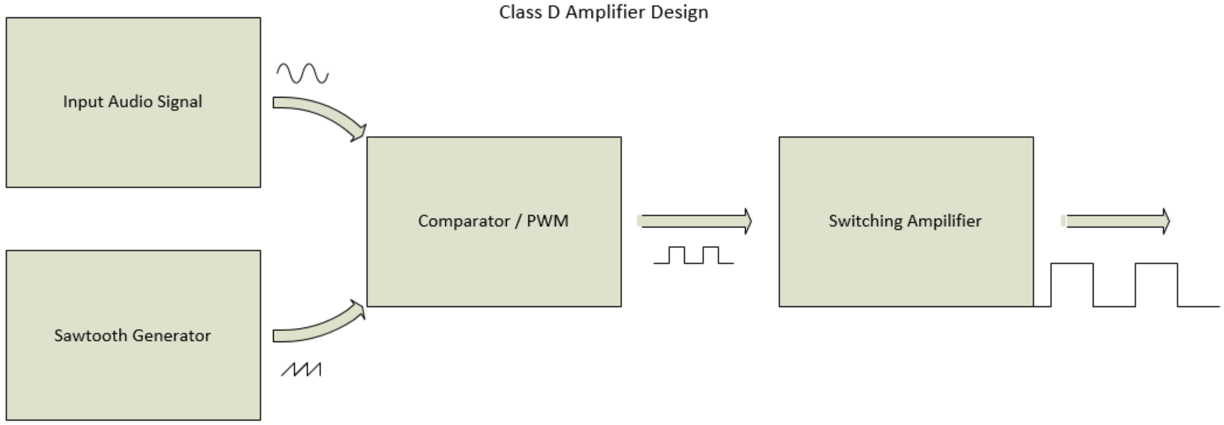
Table 3. Equalizer Block Diagram

Detailed Design

Class-D Amplifier

A Class D amplifier uses transistor switches that oscillate between on and off at very high frequencies to recreate the input signal. This process is accomplished by using Pulse Width Modulation (PWM) which translates an incoming audio signal into a high frequency square wave. In the first stage, the audio signal is compared to a sawtooth or triangle wave with a frequency significantly higher than that of the input. The comparator then creates a square wave signal that digitally represents the original audio signal. This PWM signal is passed to a switching MOSFET that amplifies the input to the value of the voltage rails. Finally the signal is sent through a low pass filter designed to eliminate the switching frequency and smooth out the signal so that it again represents the input signal with an increase in voltage and power.

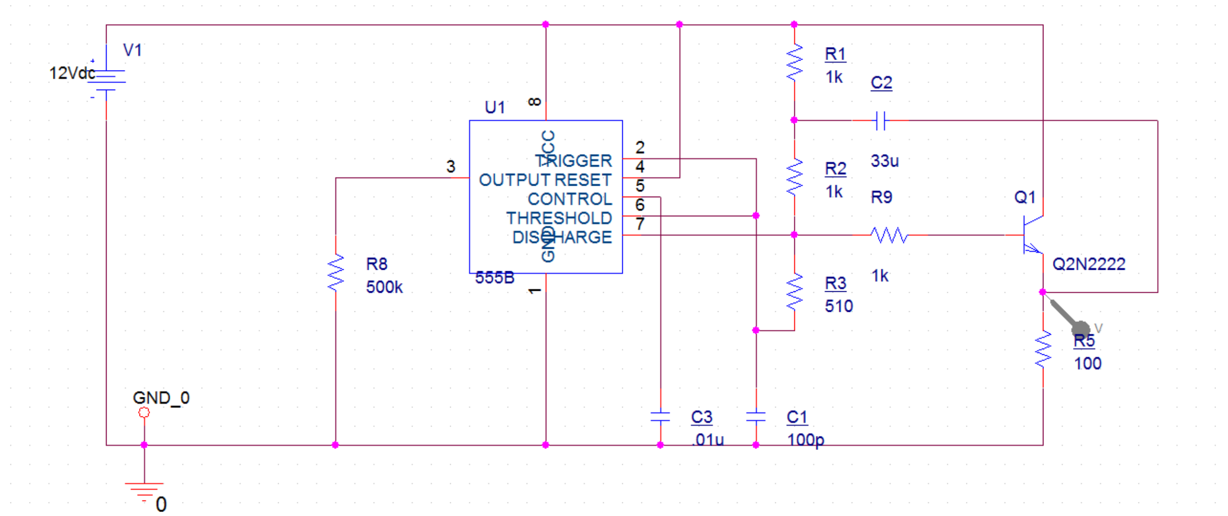
Class D Amplifier Design



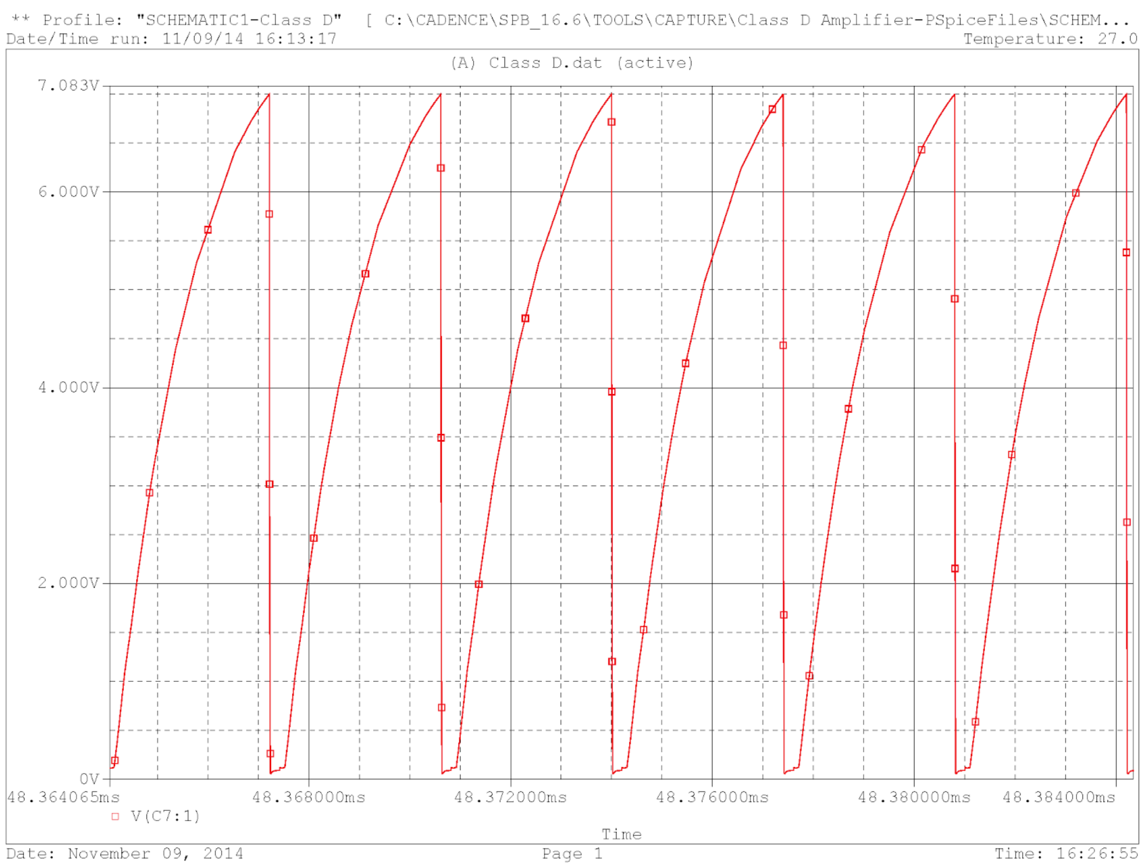
Sawtooth Generator

The purpose of the sawtooth generator is to create a waveform that can be used with the comparator. We have chosen to use a sawtooth wave rather than a triangle wave because of the decrease in duty cycle and simplicity of design. By only including half of the triangle shape, we can save valuable power and increase efficiency. Since the generator will be running at 333K Hertz, the square wave signal will be negligibly affected by the difference between the two waveforms.

Sawtooth Generator

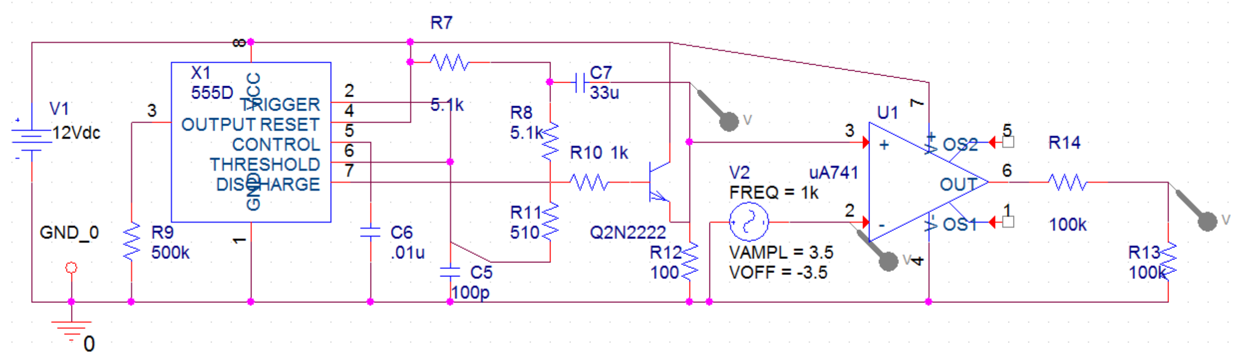


Expected Output for this Sawtooth Design

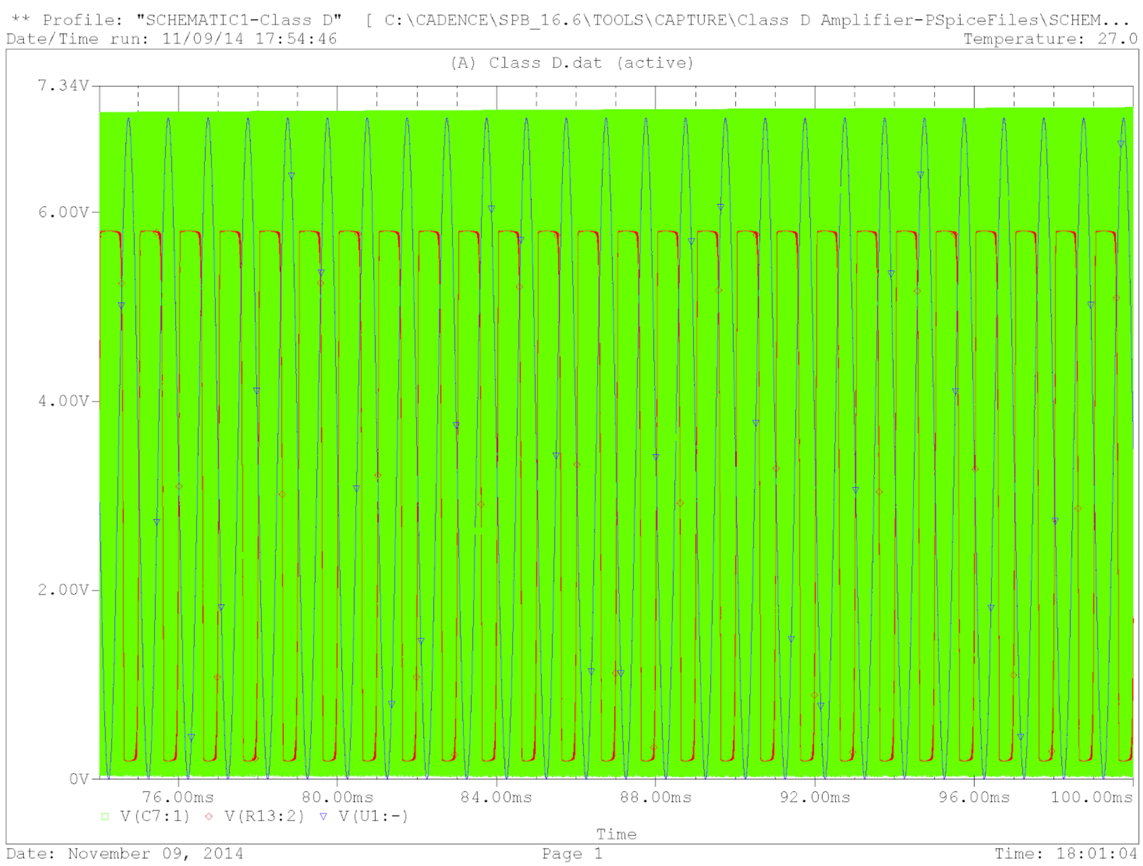


Pulse Width Modulator

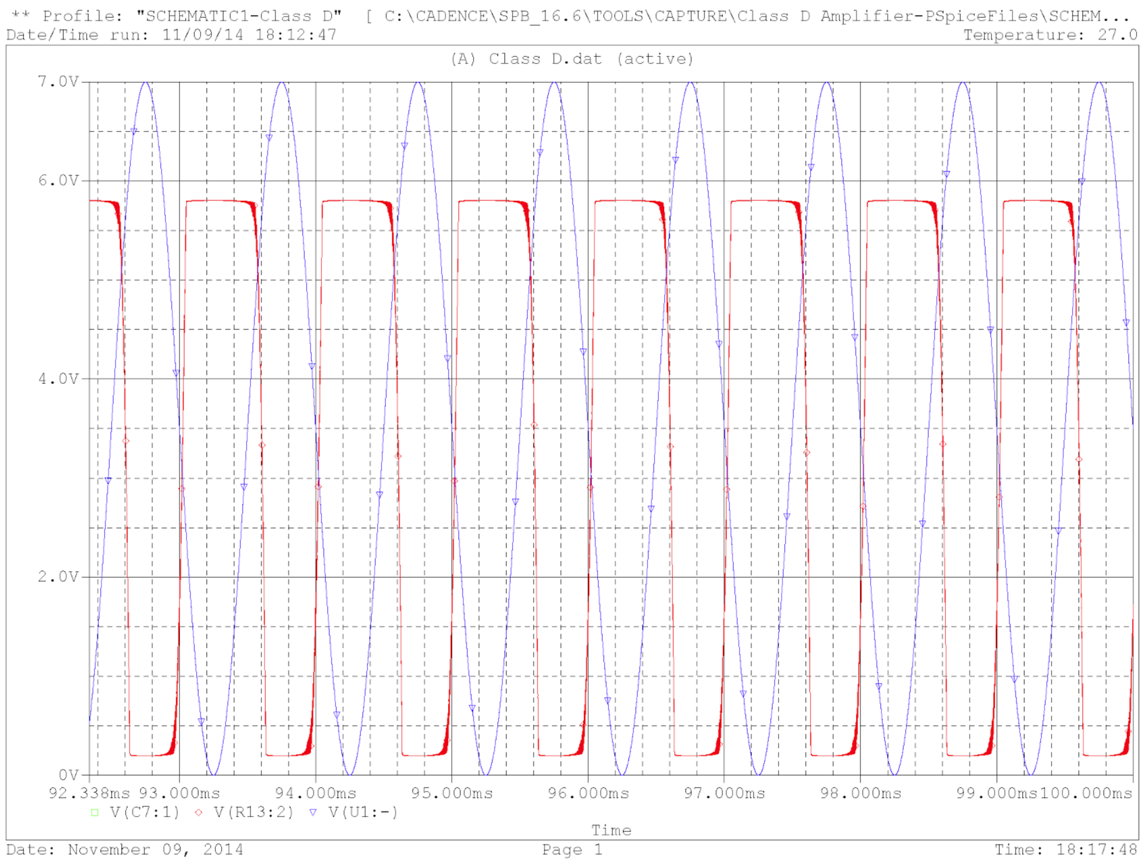
The Pulse Width Modulator (PWM) uses the output of the sawtooth generator and compares it to the output of the Equalizer circuit. In this circuit, the EQ output has been replaced by a Sine Wave generator operating at 1k Hz in order to test the circuit's functionality.



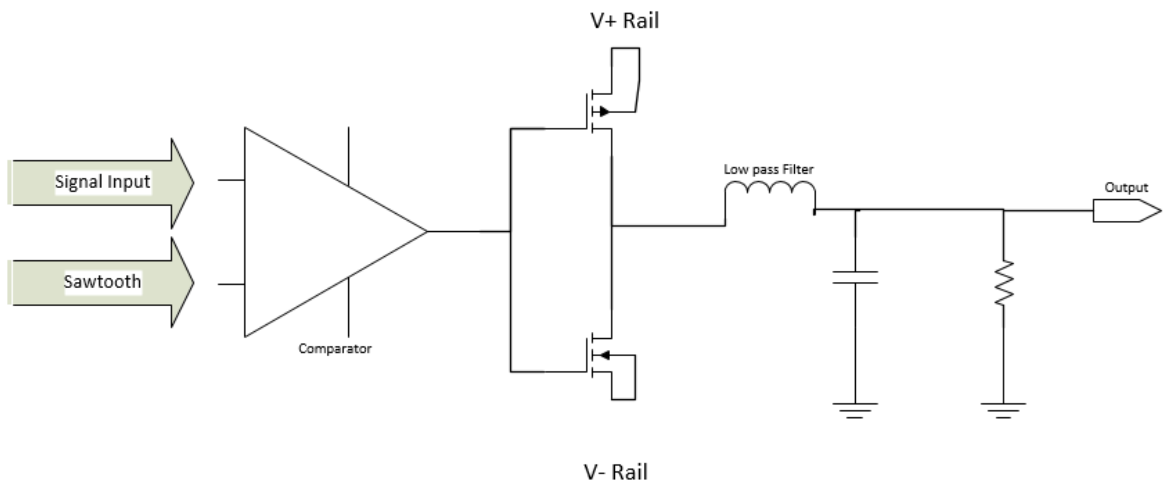
Expected Output for this Circuit at 1k Hz (With Sawtooth Trace Included)



Expected Output for this Circuit at 1k Hz (Sawtooth Trace Removed)



Switching Amplifier



A switching amplifier consists of a comparator, a set of MOSFETS, and a low pass filter. The comparator takes the signal input and the sawtooth waveform and creates a PWM signal. This signal is then passed to the MOSFETS. Theses MOSFETS use the PWM signal to control the gate which is connected to the high power circuit. By switching these MOSFETS we produce an amplified version of the original signal plus high frequency noise from the sawtooth generator. Using a lowpass filter we can remove this noise and can apply the amplified signal to the speaker load.

Equalizer

Filter Design Process

Our filter design process is as shown in the *Analog Filter and Circuit Design Handbook*[1]. In this approach, we choose a normalized filter design with a 3dB frequency at 1 Hz. We then choose two values- a Frequency Scaling Factor (FSF) to convert the normalized filter into a denormalized filter at our desired frequency, and an impedance (Z) to transform the component values into realistic values. In a low-pass filter, this means our resistors are equal to Z, and each capacitor is

$$C_{n,denormalized} = C_n\left(\frac{1}{FSF*Z}\right)$$

Where C_n is a standard capacitance from a table out of the above book.

A low-pass filter can be transformed into a high-pass filter by swapping the locations of the capacitors and resistors. In this case, the capacitances are

$$C = \frac{1}{FSF * Z}$$

And the resistances are

$$R_{n,denormalized} = \frac{1}{C_n} * Z \,.$$

Because our band-pass filters are wider than a factor of (approximately) 2, they are a simple cascading of high- and low-pass filters, designed as described above.

Realized Filters

During the process of simulating the system as a whole, we ran into variation in the output level due to overlap in our bands. Even with fourth order filters, the attenuation roll-off is slow enough that one band may interfere with another two or even three bands over. To reduce these effects, we adjusted our 3dB frequencies as shown in the following tables.

[1] *Analog Filter and Circuit Design Handbook*, Williams, Arthur B., McGraw Hill. New York, 2014.

Initial Bands

Filter	High-Pass 3dB Frequency	Low-Pass 3dB Frequency
Low Pass Filter		180 Hz
Low Band-Pass Filter	150 Hz	325 Hz
Middle Band-Pass Filter	300 Hz	1000 Hz
High Band-Pass Filter	900 Hz	3000 Hz
High Pass Filter	3000 Hz	

Table 4 - Initial Filter Range Definition

Current Bands

Filter	High-Pass 3dB Frequency	Low-Pass 3dB Frequency
Low Pass Filter		80 Hz
Low Band-Pass Filter	90 Hz	325 Hz
Middle Band-Pass Filter	300 Hz	900 Hz
High Band-Pass Filter	900 Hz	3000 Hz
High Pass Filter	4000 Hz	

Table 5 - Current Filter Range Definition

Filter Schematics

We chose fourth order active filters which require two operational amplifiers, ideally four resistors, and four capacitors per high/low-pass filter. The number of components therefore doubles when a band-pass filter is required. The schematics are available in the reference image section.

Simulations

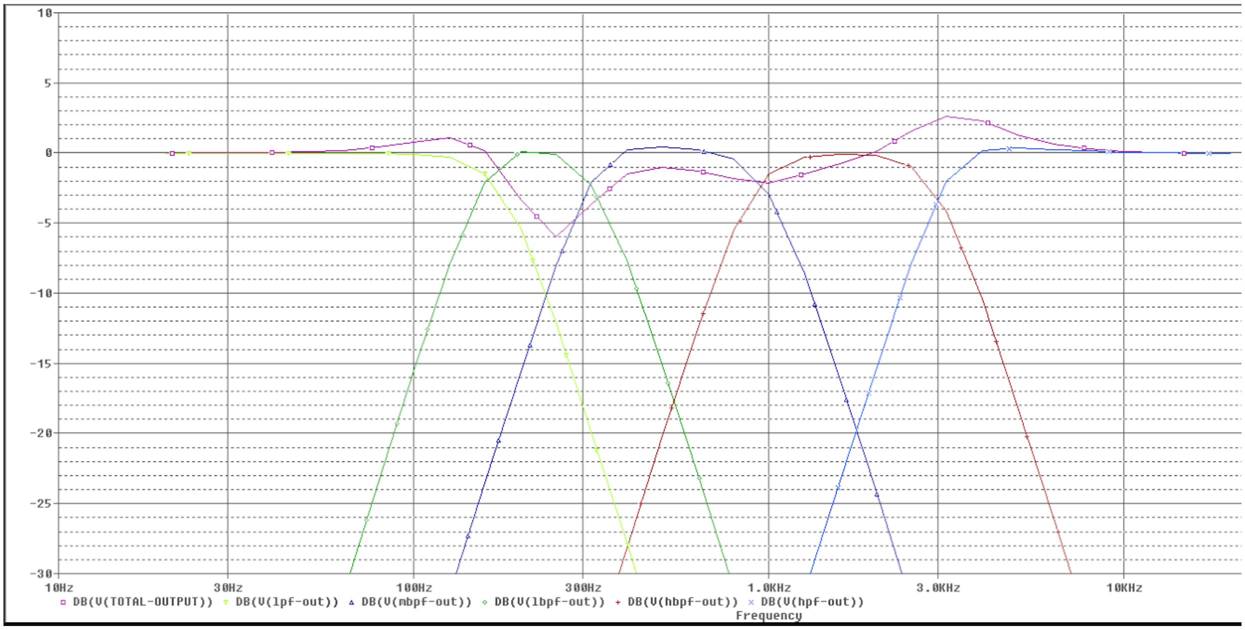


Figure 1. Initial Overall Filter Simulation

This is the pre-realized filter simulation. At this stage we discovered very large variations in gain - approximately 9 dB - over the spectrum. The output is on average lower than 0dB.

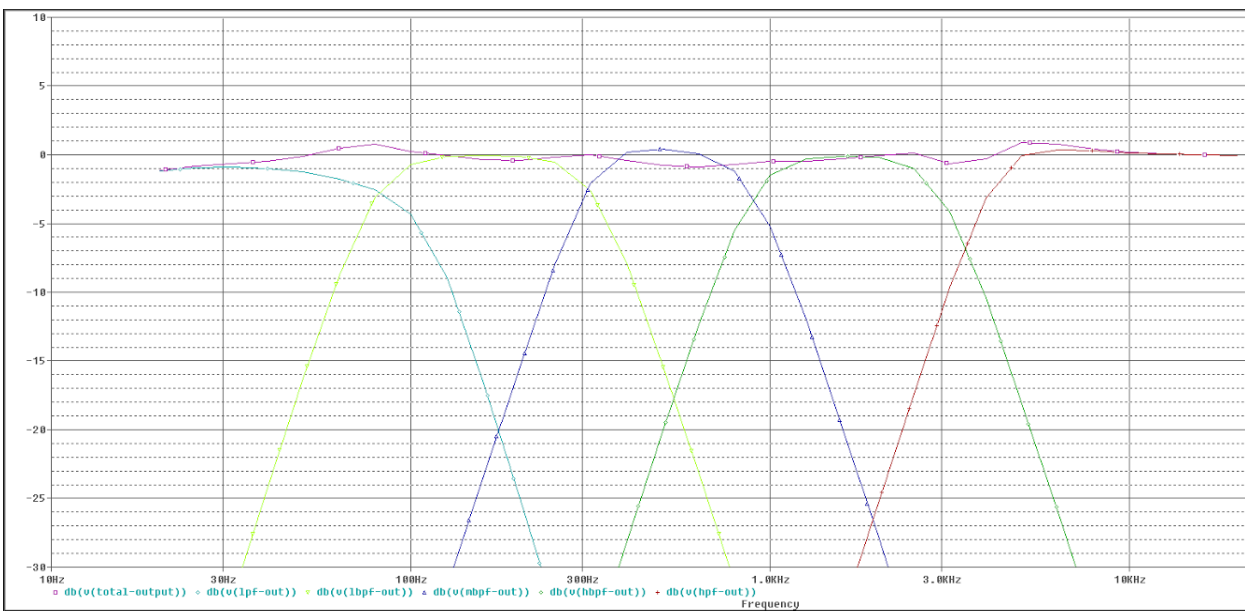


Figure 2. Current Overall Filter Simulation

Here is the new simulation after the adjustments in 3 dB frequencies. We see that the output is much flatter - within 1 dB of 0 across the spectrum. The level also changes less rapidly than in the original design.

Mixer Design

For the mixer, we use a summing amplifier with potentiometers to adjust the gain of individual bands. This is given by, in the case of the HPF,

$$\frac{R40}{R35 + R34}$$

where R34 is adjustable between 0 and 500 kΩ. Thus we are able to vary the gain between 0.06 and unity.

Mixer Schematic

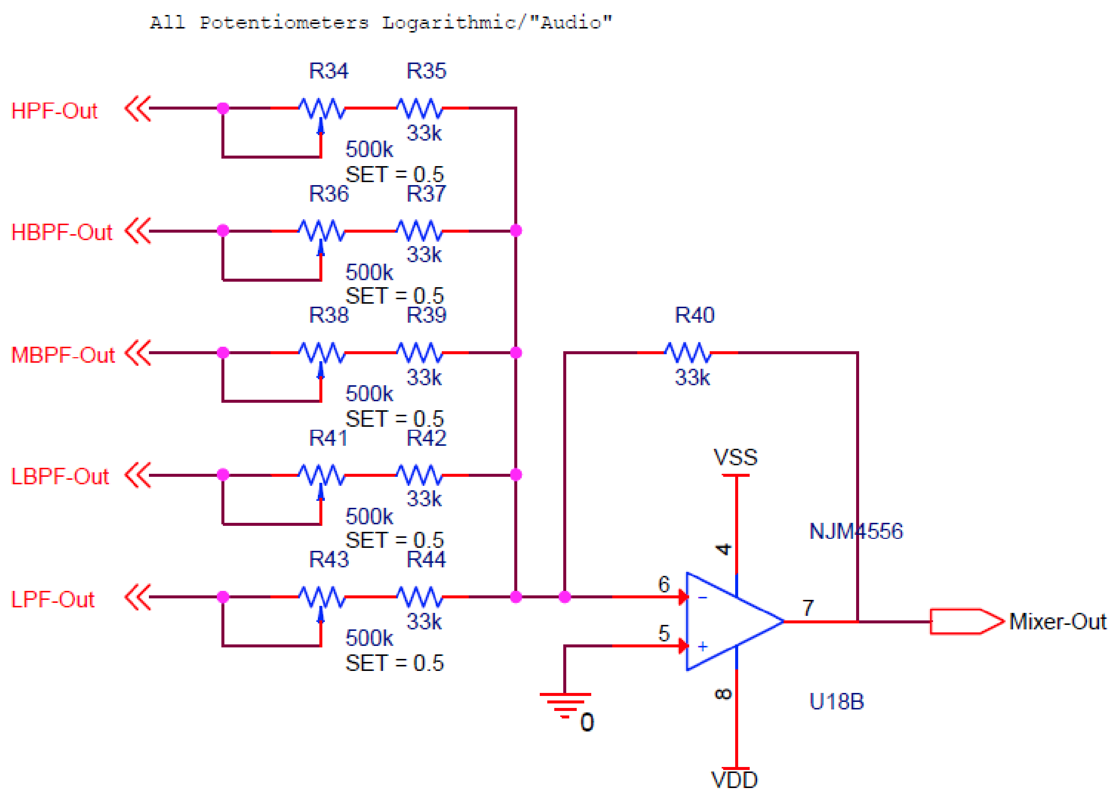


Figure 3. Mixer Schematic

Input Buffer

Our input buffer is a simple non-inverting op-amp stage with unity gain and a high-pass filter. This is to decouple our circuit from the input device for biasing purposes.

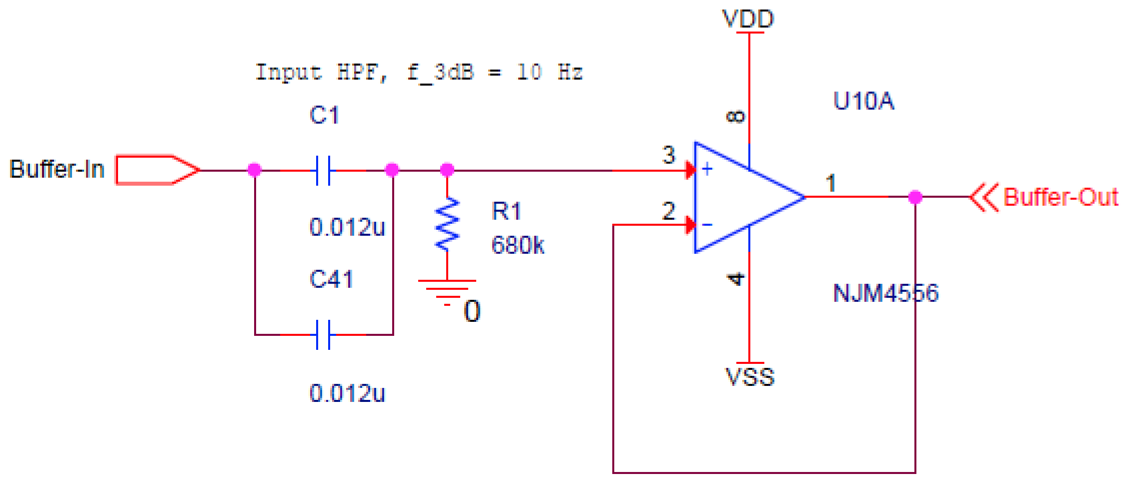


Figure 4. Input Buffer Schematic

User Interface

We are using 10-segment bar graph LEDs to give users a visual indication of each band's current level.

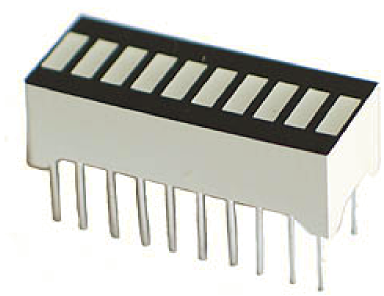


Fig. 10 segment bargraph LED

Test Methods

Bill of Materials

Project Schedules

Fall 2014

- 9/29-Block Diagram Done
- 10/15-Power or Signal initial Design
- 11/1-Models Simulation Finished
- 11/21-Project Circuit Design Final
- 12/1-Bill of Materials
- 12/10-Final Project Design, Parts ordered

Spring 2015

- February-Construction Design/ Process Complete
- March-Signal Equalizer built & test
- Mid-April-Power Amp Construction
- May-Amplifier Built & test