Maximizing Amplifier Audio Quality Based on Loudspeaker Characteristics

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Abstract

Each unique loudspeaker and amplifier has its own set of characteristics throughout the frequency range. Particularly, speakers are almost always characterized as an impedance, while they actually behave in the same way as an RLC network unique to each unit. Differences in linearity can create bothersome distortion in audio amplification systems, and effort must often be made to match amplifiers to loudspeakers. A system which characterizes the nonlinearity of an audio chain and corrects for it by altering the input audio signal in real time can alleviate this issue by correcting errors accrued through the full system.

1 System Design

1.1 Overall Goals

This project aims to develop a very high-fidelity audio amplifier system which will use digital signal processing in a feedback paradigm to achieve maximum audio quality, no matter which amplifier and loudspeaker are used downstream. A classic audio amplifier system (audio chain), consists of a signal source, such as a mixer, computer, or microphone; followed by a preamplifier, usually consisting of operational amplifiers or small-signal discrete transistor amplifiers, to boost the signal to a maximum voltage; which feeds through a buffer containing a filter, to condition the signal, and a unity-gain amplifier, to present a high output impedance. This stage supplies the signal to a power amplifier, composed of power transistors, which boosts the current of the audio signal so that it can be used to drive a loudspeaker. The final stage of this classic system is the loudspeaker, which converts the electrical audio signal to air pressure waves audible as sound.

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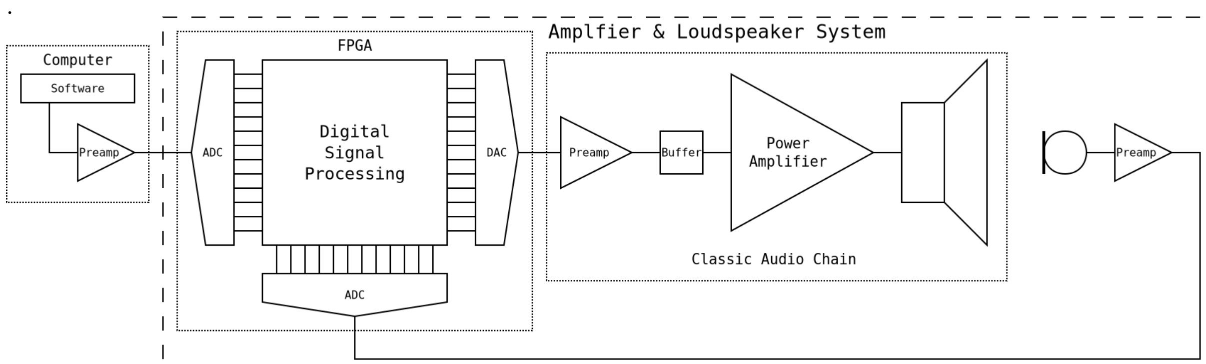

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My design adds four additional stages to the classic audio chain model; these extend the concept of negative feedback, ubiquitous in power amplifier designs, to the entire amplifier system, from the preamp to the loudspeaker. These stages are a microphone positioned close to the loudspeaker output, its preamp, a digital signal processing (DSP) subsystem, and a supporting preamp. The DSP subsystem, which could comprise a configured FPGA alone or linked to external conversion circuitry, will accept as input the original audio signal, passed through modest amplification, and the output from the microphone preamp. The effect of this scheme is to pass the actual audio chain output, as heard by a listener, back to an early stage of the system as negative feedback. The DSP system will use this input to alter the original audio before passing it to the amplifier, in order to make the audio heard by a listener match the original audio signal as closely as possible.

1.2 Preamplification

My design requires preamplifiers at three stages: between the audio input and the DSP system, between the microphone and the DSP system, and between the DSP system and the power amplifier. The purpose of each preamplifier is to increase the voltage of an audio signal to make it more robust and less susceptible to external noise. In my system, the preamplifier circuits will be constructed using operational amplifiers, to take advantage of their simple, low-noise operation and excellent properties for both input and output. My operational amplifier of choice is the ADA4700-1, rated for $+50$V to -50V, 20 V/ms slew rate, and a relatively high 30mA output current. It displays fantastic characteristics and may be used to amplify audio by many times in a single step. The preamplifiers running into the DSP system will only need a single stage, while the preamplifier before the power amplifier will have two or three op-amps. In the first design, all of these will be set up in the non-inverting configuration, except for the very first stage, which will be an inverting mixing amp.

1.3 Microphone

The feedback microphone is a core component of this design, and care must be taken to ensure that it has as linear a response as possible and passes undistorted audio signal into the DSP. In testing, an electret microphone (AOM-6738P-R) has been used, which exhibits good characteristics. Care must be taken, though, with electret microphone input circuitry. They contain a small capacitor which changes its value when air pressure on a film changes, and a JFET which acts to convert this change in capacitance to a variable voltage. Thus, the microphone requires a stable external voltage source, and the voltage input also acts as the audio output. The design must account for this with a resistor between voltage regulator and microphone, as well as a mechanism to eliminate the DC offset before amplification. These requirements are worth meeting, because electret microphone characteristics are so superior to those of dynamic microphones.

1.4 Analog to Digital Conversion

Both of the inputs to the DSP system at the start of the audio chain are analog signals: they vary more or less constantly with time. In order to manipulate them using digital methods, which are necessary for versatility, these signals must be converted to a digital format. Performing conversion requires recording the analog level at many instants through time, as one of many discrete values. There are many ADC chips available for use, and our FPGAs themselves can also perform this function. Which specific hardware to use for this application still requires investigation. An important consideration for my design is that, in order to accurately reconstruct the signal without loss, the sampling rate must exceed the maximum signal frequency by at least double. This means that the ADC system must be capable of taking samples at 40 kHz or more, and subsequently passing the quantized values to the FPGA. Since the ADC is outside of the DSP feedback loop, it may be a source of error, so it must be as accurate as possible. Dithering may be useful to avoid aliasing.

1.5 Signal Processing

Digital signal processing is at the core of this design. It will be implemented in Verilog, aboard a Xilinix field-programmable gate array, for maximum processing speed. The function the DSP must perform is comparing an input audio signal to a feedback audio signal, and synthesizing an altered audio signal as output, functioning similarly to an operational amplifier with feedback, but able to alter amplitude, frequency, and phase as necessary. A possible way to achieve this is to analyze "chunks" of each signal, both in the amplitude domain and the frequency domain. Amplitude analysis is simple, and could be used simply to detect and fix phase shifts and differences in volume. Frequency analysis will require performing a real-time Fast Fourier Transform on both input audio signals to find and correct discrepancies in frequency response across the audio band. The FPGA program will need to accept two input signals, record them as chunks, run an FFT on each one, compare the chunks and FFTs, and output a modified signal, aiming to bring the inputs as close as possible to equivalence.

1.6 Digital to Analog Conversion

The DSP system works entirely in the digital domain, using a processor to make alterations to chunks of signal represented by groups of bits. The rest of my design, however, uses analog signals varying constantly with time, so it is necessary to convert from digital to analog between the DSP system and the loudspeaker preamplifier. Digital to analog converters take as input binary values representing voltage, just like an ADC outputs. The DAC then outputs the intended voltage, run through a low-pass filter to smooth out the "steps" as the voltage changes. Similarly to the ADC, the DAC must have a sampling frequency of at least double the maximum signal frequency for the signal to be reconstructed completely from the changing digital signal. As long as the DAC is fast enough to reconstruct the digital signal, its error can be fully corrected for by the DSP, because it is within the feedback loop. The FPGA may also be able to perform this function internally. As with the ADC, though, the DAC may be better assembled as a separate component of the system.

1.7 Power Amplifier

Final amplification is the most difficult part of an audio chain to get right. It consists of a transistor output stage designed to supply driving current to the loudspeaker while faithfully reproducing the audio signal. The transistors in a power amplifier require careful biasing and thermal management to avoid distortion or damage to the parts while supplying high current. In my design, the DSP system will account for some issues with the power amplifier. To prevent excessive processing or unavoidable distortion, though, the power amplifier should be as high-fidelity as possible. Important characteristics of a power amplifier are quiescent current, input impedance, output impedance, and maximum current drive through eight ohms. Quiescent current should be kept as low as possible, to save power and maintain thermal stability. Input impedance should be high, to treat previous stages as voltage sources. Output impedance should be low, to act as a stiff voltage source for the loudspeaker, preventing distortion. Maximium current, along with operating voltage, determines the maximum power loudspeaker that may be driven.

The class AB amplifier, consisting of a biased push-pull pair of output transistors, has good characteristics in every category. I have built class AB amplifiers before, but for this design, I want a power amplifier with lower quiescent current, smaller output impedance, and much better thermal stability. An amplifier design that should work well here is a class AB with a power MOSFET output stage. Using power MOSFETs reduces the current necessary at the gate, makes crossover distortion easier to deal with, removes the necessity of current ballasting resistors on the output, and greatly increases thermal stability at high currents. All of these qualities will make it worthwhile to design and use a high-fidelity amplifier with a MOSFET output stage. I will continue to learn more about the design considerations and limitations of these amplifiers as I design and test my system, but I will start simple at first.

2 Nonlinearity Correction

2.1 Feedback Analogy

2.2 Characterizing Distortion

2.3 Sources of Error

3 Digital Processing

3.1 Using an FPGA

3.2 Signal Processing Inputs

3.3 Algorithm Design

3.4 Coding in Verilog

4 High Fidelity Amplification

4.1 Audio Amplifier Goals

4.2 Types of Amplifiers

4.3 Using Power MOSFETs

5 Project Plan

5.1 Tentative Timeline

5.2 Necessary Research

5.3 Materials

5.4 Deliverable Systems