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# Methods of High-Fidelity, High-Efficiency Class-D Audio Amplification

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**Abstract**

Gallium nitride-based field effect transistors (FETs) have opened a path for full-frequency-range class-D audio amplifiers with low distortion and noise, thanks to their ability to switch at much higher frequencies than that of the upper range of human hearing. Compared to traditional silicon-based transistors, GaN-based transistors offer superior efficiencies, particularly at power levels below their maxima. Paired with an analog-to-digital converter, digital signal processor, and pulse-code modulation to pulse-width modulation converter, these transistors are used to design and implement a solid-state amplifier capable of generating 100 watts of output through speakers with an impedance of 8 ohms using a 1-volt line-level input. This digital signal processor, together with Analog Devices's SigmaStudio development software, allows for equalization, filtering, and other modification of the signal in this design. Together, these equalization features, the use of GaN transistors, and various digital encoding methods are examined for their benefits in producing high-power, high-fidelity audio in small packages.

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## Introduction

In recent years, the rise of new-age semiconductors have opened the gates for a new wave of class-D audio amplifiers. These semiconductors feature wide bandgaps between their valence and conduction bands, thereby allowing the fabrication of transistors capable of withstanding high voltages and frequencies. This project aims to determine how gallium nitride (GaN) specifically can enable high-efficiency, low-distortion, and low-noise audio amplification in conjunction with an Analog Devices SigmaDSP digital signal processor (DSP), external analog-to-digital converter (ADC), and pulse-code modulation (PCM) to pulse-width modulation (PWM) converter. The ultimate goal of the corresponding capstone project is to build an amplifier using silicon carbide (SiC) or gallium nitride (GaN) transistors with an output power of 100 watts, an impedance of 8 Ω, and a total harmonic distortion + noise value (THD+N) of less than 0.5 percent. Techniques to achieve this output power and low amount of distortion centered predominantly around input analog filtering and a deliberate choice of transistors based on the project's needs. The design process of this project and the benefits of the encoding methods and material choices are outlined in the following sections.

## Digital Signal Processing Overview

Digital signals are expressed as binary (1/0) outputs measured in discrete time domains, as opposed to the continuous time and value domains of analog signals. As a result, any audio signal, which is inherently continuous-time and continuous-value, must be encoded into a digital signal before being processed digitally. Methods of encoding analog signals to digital equivalents without a significant loss of data are well-documented, and ADCs have become commonplace in many applications. These converters operate by comparing analog input values to reference values, approximating these signals with preset register values. Using this analog-to-digital conversion, it is possible to perform all of the standard digital transforms on this signal, including filtering, splitting, merging, and analysis of the sound it carries. Before these DSP operations are performed, a number of considerations should be made regarding the signal path, including the sample rate and the types of digital audio encoding used throughout.

First, the appropriate encoding frequencies should be determined. The Nyquist-Shannon sampling theorem requires that the sampling rate of a signal should be twice the highest frequency of the signal to capture the signal properly. ("Highest" is the key word here, particularly with audio signals, which are rarely a single frequency.) However, for the purposes of encoding high-fidelity audio, it is imperative that a fair amount of oversampling occur, on the order of about eight times this Nyquist frequency [1]. Given that the normal human ear can detect audio signals between about 20 and 20 kHz, this amounts to about 320 kHz. The digital signal processor used in testing, the Analog Devices ADAU1466, supports inter-integrated circuit (I2C) communications, including inter-IC sound (I2S), up to 400 kHz, above this eight-times-Nyquist frequency. This ensures that little to no data is lost along the digital signal path.

**Digital Encoding**

Class-D amplifiers require high-frequency encoding to be useful for a wide range of audio frequencies. Previously, limitations on the maximum allowable switching frequencies of these amplifiers without heavy distortion confined these amplifiers for use in only low-frequency applications – an example of which would be subwoofers in low-power stereo systems.

Several methods of encoding are available for class-D audio amplification. The test board used in this project takes advantage of a select few of them. At the beginning of the digital portion of the signal path, the ADC takes the analog signal and encodes it through a process known as sigma-delta (or delta-sigma) modulation, in which changes (deltas) in voltage at the analog input are integrated (sigma) into a stream of pulses that are then sent to the DSP. This form of modulation is also known as “pulse density modulation” in that the density of pulses determine the digital signal’s strength and frequency due to the integration functions of the ADC. The ADC sampling frequency, given the 12.288 MHz clock input from the DSP’s built-in crystal, is 48 kHz, higher than the required Nyquist frequency of the analog signal [2]. Delta-sigma modulation does not require a very high sampling rate like other forms of modulation, including pulse-width and pulse-code modulation, due to its predictive nature and automatic smoothing of signals through integration [3].

Pulse-code modulation (PCM) is the output encoding style of the DSP. PCM differs from sigma-delta modulation in that it encodes the magnitude of signals, as opposed to changes in them. PCM is commonly found in digital audio applications, and it is the most popular audio encoding method in consumer electronics and archival audio storage [4]. Its ability to encode audio signals at high frequencies in a way that doesn’t require ultra-high frequencies makes it desirable for these applications. This method of encoding operates as a single $2^n$-bit stream, and in the Analog Devices DSP used in this project, a 16-bit stream is used. Values are then transferred to the PCM-PWM converter. The TI TAS5086 PCM-PWM converter uses this 16-bit audio stream to generate a PWM signal to be sent to the GaN transistors’ gate drivers, the final digital portion of the signal path [6].

Pulse-width modulation (PWM) is used primarily near the end of the signal path, between the PCM-PWM converter chip and the GaN transistors’ gate drivers. At standard audio frequencies, this encoding method is capable of carrying audio information digitally, albeit with a significant amount of noise. The use of GaN transistors, however, is meant to reduce this noise significantly, making PWM a viable digital encoding method for the final stages of the design [5]. This encoding technique works by encoding signal strength through a duty cycle – i.e., 90 percent strength would translate to 90 percent on-time, 40 percent to 40 percent, and so forth. This technique, however, is limited by the cycle time, which is always a constant value, and, until the advent of feasible high-frequency transistors, was highly undesirable for the output stage of digital audio transmission. (delta-sigma source) If these challenges are truly overcome, pulse-width modulation would be considered the ideal method of encoding for class-D amplifiers; its single stream facilitates output since it does not need to be merged into a single channel.

A handful of researchers have attempted to determine the most effective modulation method for audio and other high-frequency applications, at times going into the near-gigahertz range. Motoi, et al., attempted to determine the most effective form of modulation at 800 MHz, a common wireless communication band for mobile devices [7]. This article highlights the benefits that class-D amplifiers and the usage of GaN provide in ultra-high-frequency transmission of wireless signals. Even in relatively lossy media and high-power applications with "back-offs" – similar to lowering the power output of an audio amplifier, the researchers are able to create a transmitter using delta-sigma (pulse-density) modulation with efficiencies over 50 percent.

Hiorns, et al., assert that a proper high-fidelity sample rate for pulse-width modulation of audio signals should be about eight times the Nyquist frequency, using the audio output of compact disc (CD) media for their analysis [1]. Since their study is from 1991, it does not consider the effects that newer modulation methods and GaN transistors can have on the processing of audio signals. This analysis instead focuses on the techniques of pulse-width modulation that produce the best outcomes, namely the "trailing PWM" and "asymmetric" techniques, in which pulses are modulated on trailing edges of a triangular reference signal and both edges of a triangular signal, respectively, as opposed to on the leading edges. The author finds that trailing PWM modulation is best for reducing distortion resulting from the non-linearity of PWM signals with respect to their analog inputs.

McKenzie and Ng explore the ability of delta-sigma (pulse-density) modulation to reduce harmonics that result from the usage of various PWM techniques [8]. These include feedback loops, in which the output of the amplifier, complete with errors, is fed back to the modulation stage and used to issue corrections to the signal in real time. It is also found that, using these feedback loops, it is possible to decrease the THD+N by using GaN transistors, as opposed to silicon transistors, and this is attributed to the lower rise and fall times and faster “settling” of changed signals causing fewer harmonics, particularly seventh harmonics. The group finds that, using a proper pulse-density modulation system, they are able to reduce harmonics to levels of under -80 dB.

Dallago, et al., found that sigma-delta modulation often produced lower harmonic content than simple pulse-width modulation, a difference of 6.2 percent versus 3.4 percent [3]. This difference is attributed to the sigma-delta method’s high “ripple rejection,” in which fluctuations among line currents within these modulation schemes are, in effect, canceled out by the sigma-delta modulation’s more predictive algorithm in determining digital encoding values.

Considered together, these studies appear to show that sigma-delta, or pulse-density, modulation is an appropriate way to diminish the negative harmonic effects of using pulse-width modulation with class-D amplifiers and should be used through as much of the signal path as possible to ensure minimal distortion.

## Challenges of Class-D Amplification

Many scholars have shown the obvious drawbacks of using class-D audio amplifiers. These drawbacks include severe harmonic distortion and errors in transcoding audio into digital signals due to the avoidance of "overshoot" characteristics. These studies are explored in the following paragraphs, and methods to overcome these challenges are highlighted, as well.

Kovačević et al. note that class-D power amplifiers have higher relative distortion than linear amplifiers, but this effect diminishes significantly as switching frequencies enter the hundred-kHz to few-MHz range [9]. The team finds that the total harmonic distortion (THD) is affected by this switching frequency in addition to feedback resistances within their design. They also note that transistors with low "on" resistances (typically denoted $V_{DSon}$) have lower power dissipation and therefore are more efficient. The team's analysis shows that there appears to be a trade-off between the maximum power output of an amplifier and its total harmonic distortion.

Peregrine Semiconductor Corporation does an overview of these drawbacks, as well, and their article highlights many of the same trade-offs as the Kovacevic article – distortion reduction versus efficiency and overall power output [11]. This article also highlights a severe issue that can result from a class-D amplifier's PWM technique – an instability in the clock for the signal or its reference value can result in significant amounts of noise. Thus, a stable clock signal is crucial for proper class-D audio amplification. Further, the article shows the benefits of GaN in creating these amplifiers. GaN power devices are generally smaller, leading to lower parasitic values, and they store no charge during "dead times," times in which conduction should be turned off in a PWM signal to prevent overshoot. The authors reference a handful of GaN-based class-D amplifier boards and state that they are each capable of over 100 watts of output with less than 0.05 percent THD+N.

**Benefits of GaN and Other Wide-Bandgap Transistors**

For the purposes of this application, GaN is particularly useful in the fabrication of the output transistors. These transistors need to carry relatively high voltage per unit area at frequencies above the high corner frequencies of many silicon-based transistors. They need to be efficient, as a small board design is incapable of handling high power dissipation into heat. Research in the area shows that GaN transistors possess both of these qualities. Gallium nitride is a wide-bandgap semiconductor that, when used in transistors, is capable of switching in the megahertz range [12]. Traditional class-D amplifiers suffer from severe roll-offs at high frequencies due to the internal capacitances of their transistors, well short of the desired frequencies for high-fidelity audio. In addition, they suffer from significant amounts of harmonic distortion without feedback. These harmonics can be measured through the standard THD+N, calculated with the equation

$$THD + N = \frac{\sqrt{\sum_{n=2}^{\infty} V_n^2} + \xi}{V_1} \quad (1),$$

where $V_1$ is the root-mean-square magnitude of the input voltage $V_2$, $V_3$, …, $V_n$ are the rms voltages of the harmonics of this input frequency, and ξ is the rms noise level of the output, unrelated to harmonics. The test amplifier seeks to lower this value to a value of less than 0.5 percent, on par with high-end audio amplifiers and likely possible given the performance of non-GaN amplifiers [9]. To make for a fair comparison between various types of amplifiers Chierchie and Paolini seek to understand the source of these harmonics, and they pin them on the "dead times" of the amplifiers [10]. Even in a theoretical sense, these dead times can raise the noise floor of an audio amplifier to values above -20 dB, which are unacceptable for high-fidelity audio. The most significant factor in improving audio quality is to find ways to address this distortion, and GaN offers a potential solution.

Omura, et al., find that GaN's benefits lie most heavily in its volume reduction per unit area – that is, smaller volumes help GaN transistors achieve lower levels of heat dissipation and parasitics, which in turn lead to much more efficient devices [12]. This analysis also links the reduction of volume within these transistors to their ability to attain higher switching frequencies, a welcome characteristic for audio processing.

The efficiency of GaN-based transistors is significantly improved over silicon-based transistors and somewhat improved over silicon-carbide transistors, particularly at lower frequencies [13]. An analysis done by researchers at the National University of Ireland, Galway, at 100 watts shows a clear improvement over both other types of transistors [13]. In addition to the increased frequency capabilities of GaN transistors, high efficiency over a wide range of output power levels makes them a natural fit for audio applications that require variable volume. Bakker and Duffy's analysis examines the methods of increasing the efficiency of class-D amplifiers, and they assert that this is *best* done by modifying output stages, which are generally the most inefficient portions of the amplifier. They cite the need for greater efficiency in other stages of class-D amplification, particularly in their modulation techniques, but, more importantly, their analysis finds a sort of "untapped potential" in reducing the power consumption of the output stage. The article examines the various methods of increasing the efficiency of class-D audio amplifiers, including changes in power topologies and the use of SiC and GaN transistors in the output stages. The article finds that there is a stark difference between the efficiencies of power stages based on GaN transistors and those based on SiC and silicon at lower power levels. While this difference levels off close to 100 watts, the difference between 94 and 97 percent efficiency is significant, as this translates to silicon-based transistors consuming twice the amount of power GaN-based transistors consume. A graph of these results is shown below. Note that, at output powers above 20 watts, the differences in efficiency between the types of transistors amounts to about half of the wasted power being saved by switching to GaN-based transistors. At 10 watts and below, the difference between GaN transistors and traditional silicon transistors becomes more than 10 percent. Therefore, while the graph does not show significant differences in losses between the various types of semiconductors, these minor-looking losses make the difference when creating high-density audio devices.

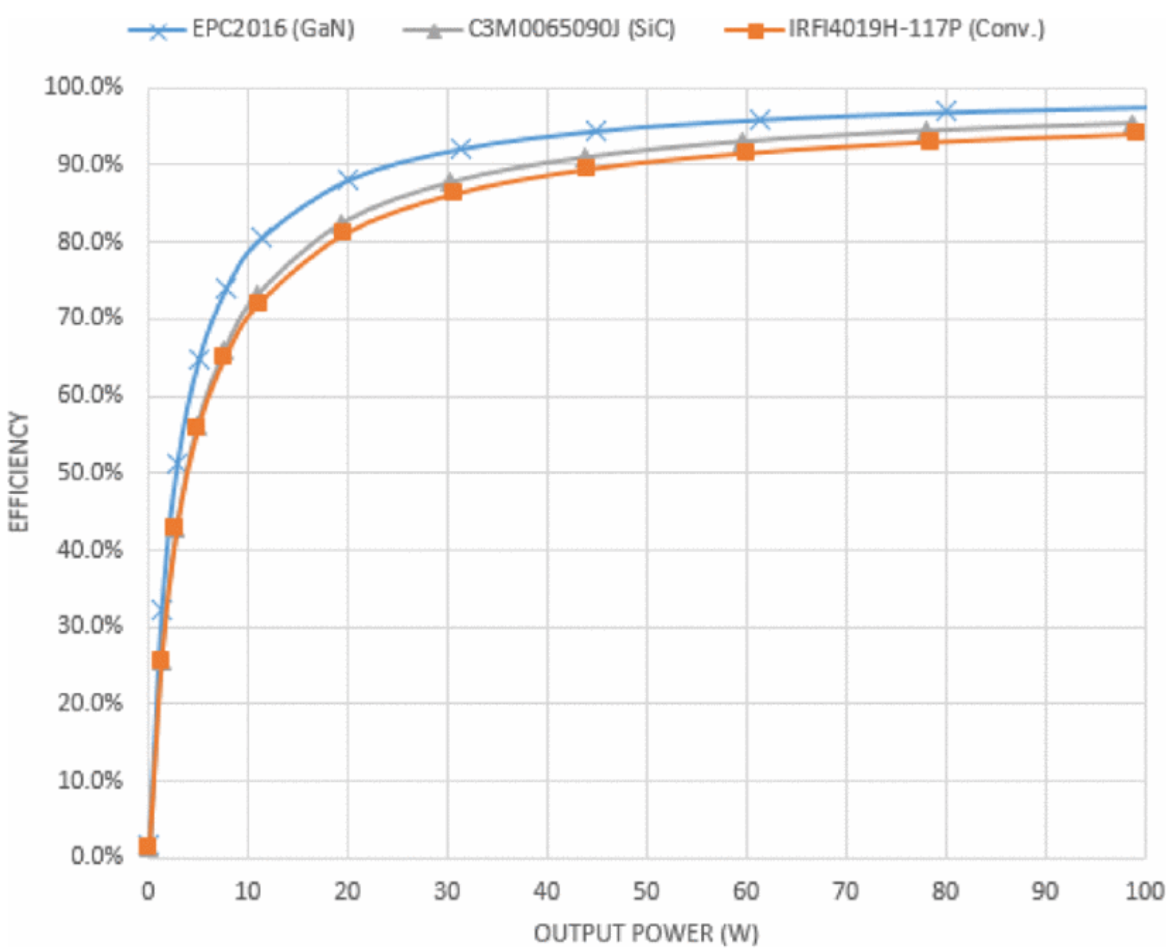


**Figure 1.** Output power efficiency of class-D amplifiers using GaN, SiC, and silicon [13].

**Reference Board Design**

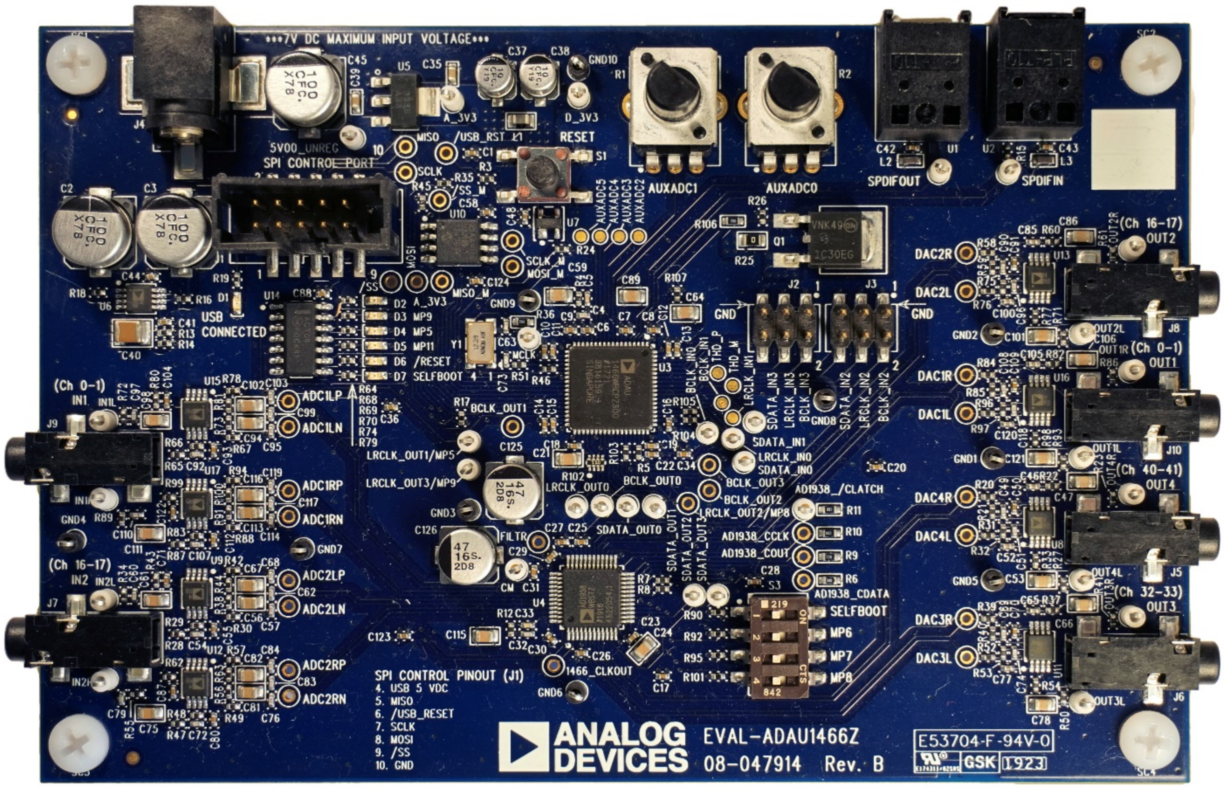


**Figure 2.** Analog Devices EVAL-ADAU1466Z reference board .

The reference board for the ADAU1466 digital signal processor, the EVAL-ADAU1466Z, contains two 3.5-millimeter inputs and four 3.5-millimeter outputs, each with left and right audio channels, along with a Sony/Philips Digital Interface (S/PDIF) input and output. Using this board and a graphical programming interface within Analog Devices's SigmaStudio software, it is possible to explore the filtering techniques of this digital signal processor and how it can be applied to minimize the distortion that results from class-D amplification. Because this board contains multiple inputs and outputs, it is possible to perform tasks beyond the scope of the test board, including mixing and splitting of audio signals.

**Test Board Design**

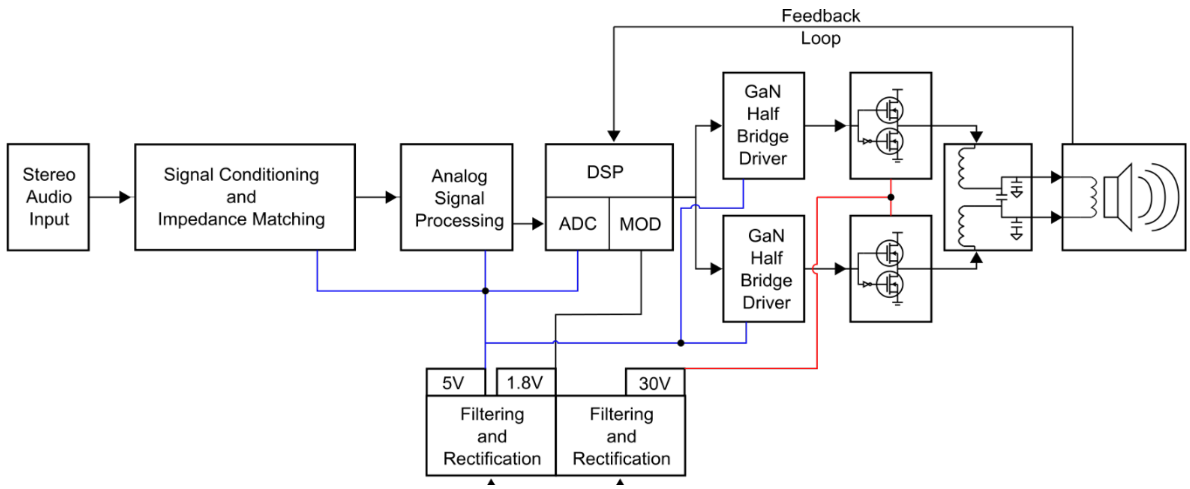


**Figure 3.** Block diagram of test board.

Using the specifications given for the project and the EVAL-ADAU1466Z board's communication protocols and recommended layout techniques, a schematic for a GaN-utilizing test board is created. Key differences result from the usage of GaN, namely the lack of a digital-to-analog converter and a much higher output power necessitating wider traces capable of handling more current. An external ADC is used in the final test board due to its enhanced bitrate capabilities over the DSP's built-in ADC. Going from left to right, an audio input is sent first through a signal conditioning phase, in which impedances are matched and noise is filtered out. This resultant signal is sent to the analog-to-digital converter (ADC), which processes the audio and converts it to an I2S signal to be used by the DSP. The DSP is able to adjust and filter audio according to the program built in Analog Devices's SigmaStudio, including such filters as FIR (finite impulse response), low-, high- and band-pass filters, pink noise filters, and several others. The physical version of this test board is shown below, color-coded by related section within Figure 3.

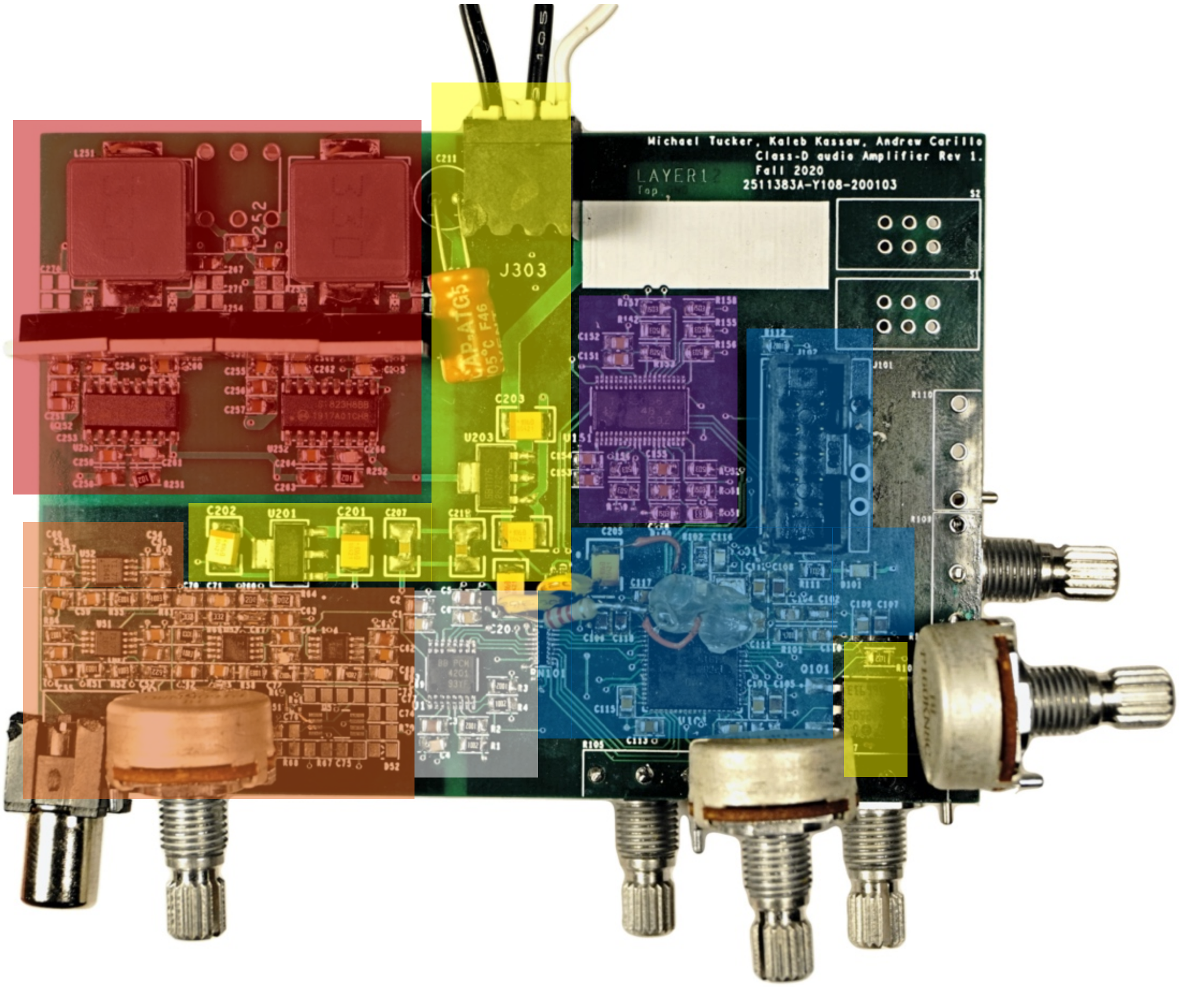


**Figure 4.** PCB of the test board, with component sections highlighted.

On this board are a litany of components that can be separated into a handful of basic functions. Analog signal conditioning and its associated components are marked in orange. Power supplies and regulation are in yellow. The ADC and associated pull-ups are marked in white, the PCM-PWM converter and associated pull-ups are in purple, and the DSP and its respective communications are in blue. Finally, the gate drivers, GaN field-effect transistors (FETs), and shielded inductances are highlighted in red. (Throughout the course of assembly, it was discovered that a handful of computer-to-DSP I2C communication lines were swapped, necessitating the short wires and long-leaded resistors in the blue DSP section.) This board is used to demonstrate a proof-of-concept for the techniques and research topics explored in this paper, and it is currently being tested for its ability to maintain a low noise floor at high power levels.

### DSP Software Demonstration

Because the test board has not yet been fully programmed, the necessary filtering features to counter its individual distortions are as-of-yet unknown. However, an overview of the SigmaStudio software's filtering techniques can be given.

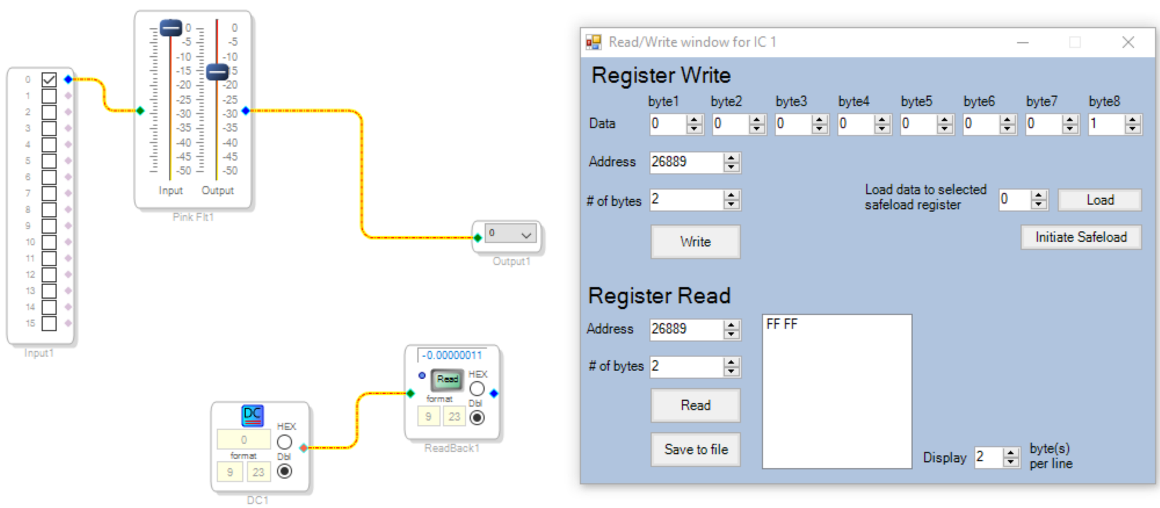


**Figure 5.** SigmaStudio simple volume filtering demonstration.

SigmaStudio's graphical interface allows for the easy manipulation of signals through various filtering techniques. These techniques include pink noise, finite and infinite impulse responses, band-pass filtering, and, together, simple volume controls. On the reference board, addition and subtraction of digital signals may also be performed, thanks to the EVAL-ADAU1466Z's multiple input and output channels. The graphical interface allows for a signal path to be evaluated from the DSP input (typically an analog reference, but potentially an ADC output) all the way to a digital or analog audio output. The software package also includes real-time monitoring of signals and equalization features. Adapted to the test board schematic, the DSP features numerous multipurpose pins and related control registers to communicate with the other chips on the test board, including the ADC and PCM-PWM converter.

The test board design uses one stable clock in conjunction with the digital signal processor, running at 12.288 MHz for a 48-kHz sigma-delta sampling rate, and the DSP functions as an I2C master device for the other devices on the board [14]. To improve sound quality for this application, the DSP is used to compensate for phase shifts and harmonic distortions that have resulted from imperfect analog signal filtering. The related Bode plot is shown below.

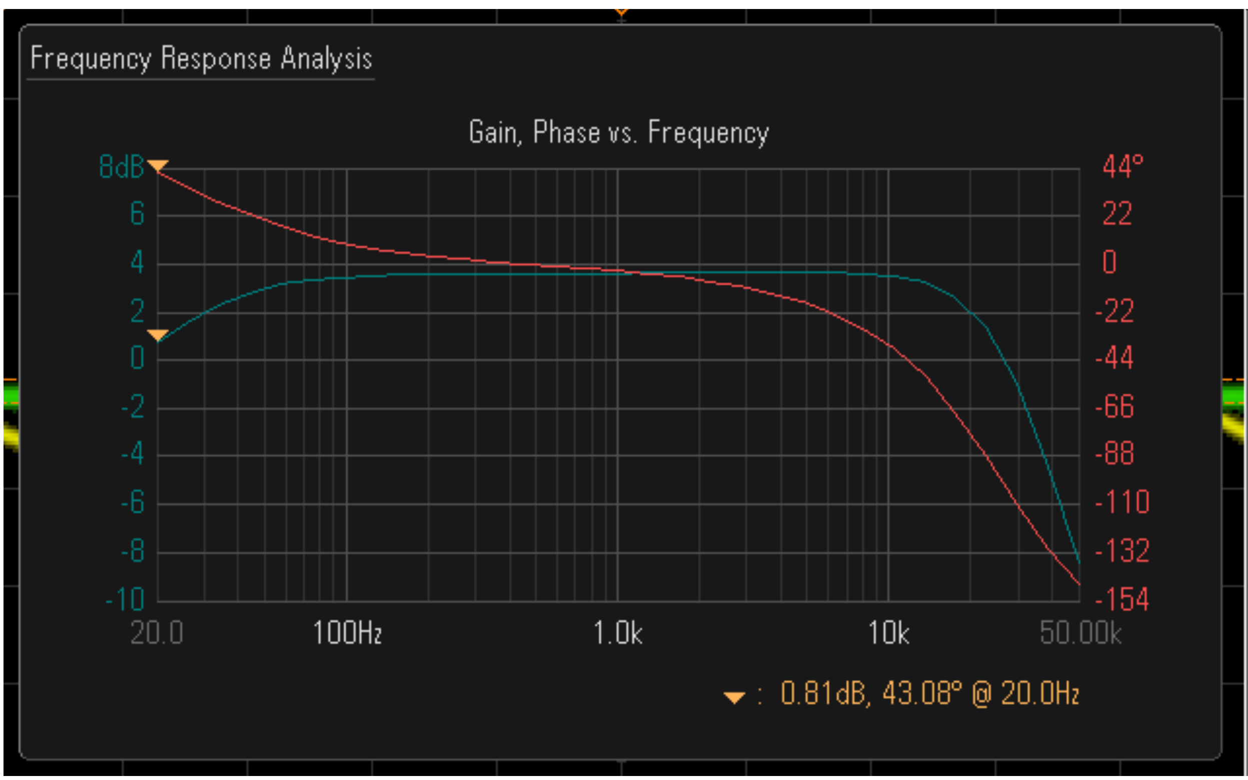


**Figure 6.** Bode plot of test board analog filtering section.

Clearly, a digital filter needs to compensate for this mismatch in phase near the upper levels of the human hearing range, as the phase response becomes significantly skewed at about 10 kHz. This can be done with a phase-shifter circuit in hardware, but a complex filter can be implemented in SigmaStudio to address these concerns with less work.

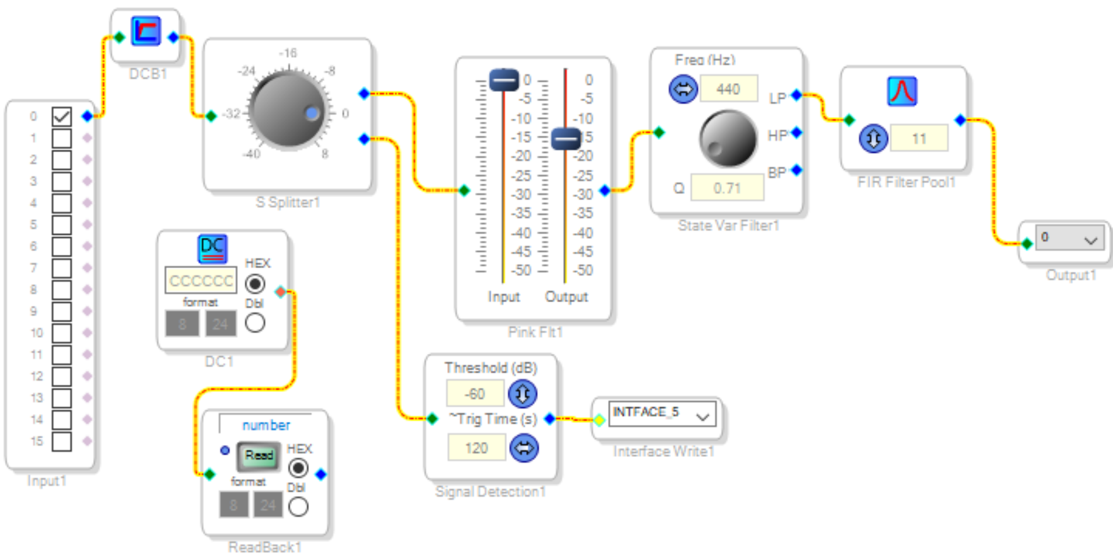


**Figure 7.** Example audio signal flow in SigmaStudio.

## Conclusions

With the introduction of mainstream gallium nitride transistors, the choice between high fidelity and high efficiency in audio systems may finally be put to rest. GaN transistors' relatively small size and low active resistances have made them valuable components in creating high-efficiency, high-fidelity class-D audio amplifiers. Of course, it is yet to be determined whether another superior modulation technique exists to reduce the distortion brought about by pulse-width modulation, and, to a lesser extent, sigma-delta modulation. Taken together, new modulation techniques, digital signal processing algorithms, and GaN have paved the way for a new generation of high-power amplifiers that have bridged the age-old trade-off between sound quality and power efficiency.

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