Back

Digital Amplifiers

The Audio *Path to the Future*

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A few years ago, an engineer/audiophile at Texas Instruments decided to evaluate a new amplifier technology and compare it against his $20,000 system, which included a Mark Levinson amplifier and Sonus Faber speakers. He brought one of TI's amplifier evaluation kits home and wired it into his system in place of his normal amplifier. He then put in one of his favorite CDs, which he used as a reference to judge audio quality.

Although he had listened to this CD literally thousands of times, he was shocked to hear a "new" guitar solo on one particular track that he had never heard before. He quickly skipped backwards and played the track again. After hearing the guitar solo again, he became convinced that this new amplifier technology would revolutionize the audio market.

This event occurred in 2001, and the new amplifier technology the engineer evaluated was a digital amplifier. The first commercially available product that used digital amplifiers debuted in 1998 with the release of the TacT Millennium, developed jointly by TacT Audio and Toccata Technology. Toccata was a small company in Denmark that had developed the digital amplifier technology used in the Millennium. Two years later, Texas Instruments purchased Toccata and implemented this digital amplifier technology in silicon, which now ships in millions of audio products under the PurePath Digital technology brand.

WHAT ARE DIGITAL AMPLIFIERS?

The digital amplifier is classified as a subset of a "class D" amplifier. Just as a digital signal has only two states, the digital amplifier has only two output states. Using power MOSFETs as on-off switches, the digital amplifier applies a fixed DC voltage across a speaker in either a positive or negative direction, causing either a positive or negative current to flow into the speaker.

The digital amplifier reproduces the audio signal by adjusting the duty cycle of this positive/negative signal at a high rate (typically 384kHz). To generate a positive current into the speaker, the duty cycle must be greater than 50%. Conversely, a negative current requires less than 50%. When the duty cycle runs right at 50%, the speaker current is zero.

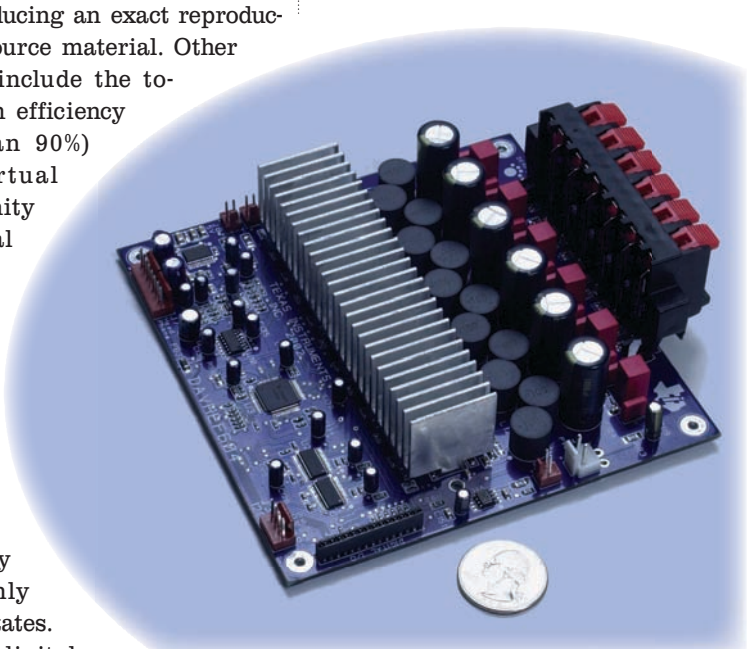
The speaker can be controlled in a very precise manner with a digital amplifier, generating very fine detail—essentially producing an exact reproduction of the source material. Other advantages include the topology's high efficiency (greater than 90%) and the virtual noise immunity of the digital signal path.

Some in the industry also call analog-input class D amplifiers "digital amplifiers" because they also have only two output states. TI has both digital-input and analog-input product lines. The analog-input is generally used in systems where high efficiency is required (i.e., cars, cell phones, laptops, and so on), but where the audio comes from an analog source, such as the tuner in an automotive head unit. But for the purposes of this discussion, "digital amplifier" will refer to digital-input only.

THE AUDIO SIGNAL PATHWAY

As shown in **Fig. 1**, the digital amplifier system keeps the audio signal in the digital domain for nearly the entire signal path, while the analog-based signal converts the signal to analog very early in the process.

The digital amplifier consists of two devices, the pulse width modulation (PWM) processor and the power stage. The PWM processor con-



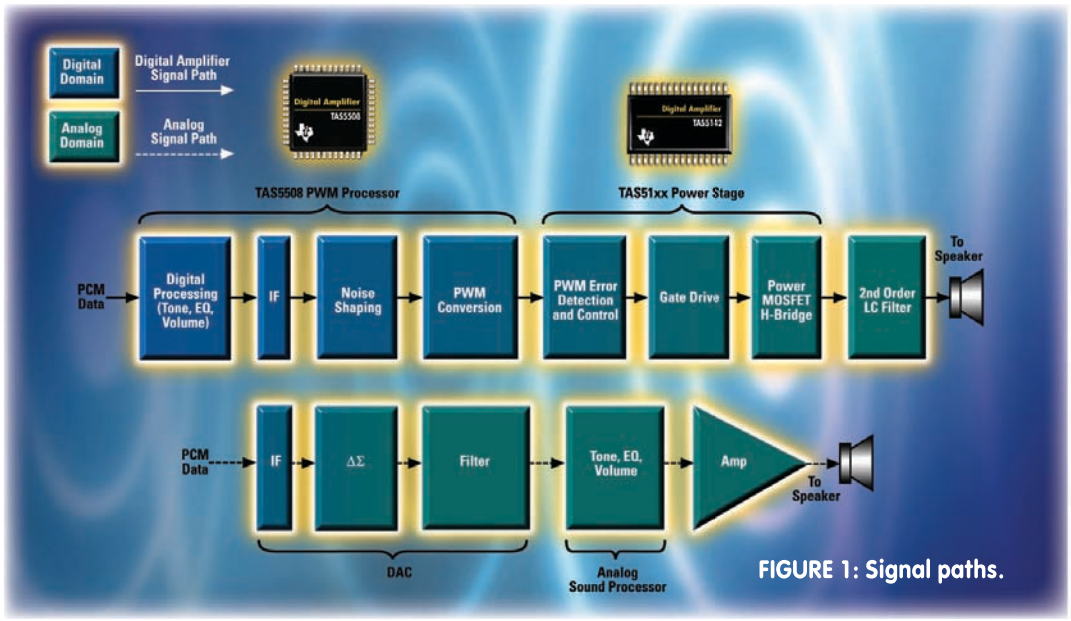


FIGURE 1: Signal paths.

verts the pulse code modulation (PCM) signal to PWM in addition to performing 48-bit digital processing, such as basic treble, bass, volume control, and more advanced functions (dynamic range compression or loudness). In contrast, the same functions are usually done after the DAC in an analog-based system, typically with op-amp-based integrated circuits.

Digital domain processing offers many advantages to the audio design engineer. For example, it is much easier to change configurations or functionality digitally, as this can be done on the production line or in the end-user's home. The 48-bit processing ensures that artifacts from the processing remain far below the noise floor.

Both the digital- and analog-based systems contain an interpolation filter (IF), which oversamples the original signal. Following the IF, the digital amplifier performs fifth-order noise-shaping, which pushes noise from the conversion process out to ultrasonic frequencies. This leaves the audio band virtually free from noise.

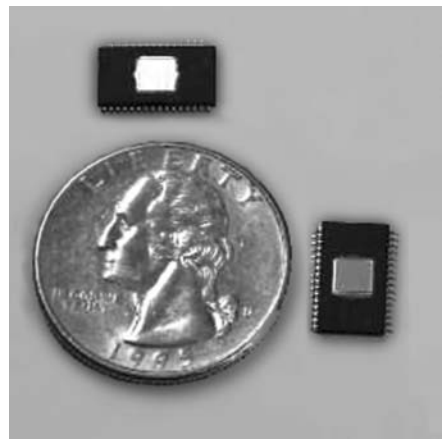
Next, the signal is converted to PWM and, because this conversion is nonlinear, correction must be applied at this point. The PWM signal is a one-bit (two-level) signal that is sampled at an extremely high rate (4,096 times the input sample frequency, or about 200MHz). Timing is critical with PWM to reproduce the input signal, thus, internal clocks with extremely low jitter are

essential. Using the analog-input class D amplifier, the PWM signal would have been generated through advanced analog circuitry.

Once the signal is converted to PWM, it then passes to the power stage, where it is checked for error conditions and timing control is applied. The signal is then passed to gate drivers, which control the MOSFET switches. Other functions in the power stage include detection of and recovery from error conditions, such as over-current, over-temperature, and under-voltage.

After the power stage, the last step of the audio signal pathway is through a low-pass LC filter, which removes high-frequency components from the audio signal to reduce EMI. This is effectively the digital to analog conversion. Without this inductor and capacitor filter, the speaker itself becomes the digital to analog converter (DAC). Most of TI's lower-power analog-input class D products do not require this filter, but it can be included to minimize EMI.

Because the digital amplifier converts the signal to analog with passive components, this conversion introduces no distortion to the signal. The conversion from digital to analog occurs at the final output voltage, avoiding unnecessary noise that will be audible. In contrast, the analog conversion occurs as the first step in an analog amplifier and is at a low-level voltage, which is then processed and amplified. Any



noise on this low-level signal from processing or from coupling will be amplified and affect the final signal quality.

CONCLUSION

As previously mentioned, the digital amplifier provides several advantages over the traditional analog-based circuit, such as fine detail, noise immunity, high efficiency, and configurability. In addition, because audio processing design is now software-centric, changes can be made quicker with less impact to the design schedule.

The digital amplifier is changing the audio and home theater market, and will continue to do so. The benefits of the digital amplifier have been enjoyed by mass consumer markets for the past three years, and home theater systems are now commonplace, in large part due to the digital amplifier. Digital amplifiers will expand across various markets (e.g., automotive head units, portable systems, in-wall speakers, and so on) and enable new product categories, replacing applications that have traditionally been dominated by analog. Texas Instruments has been one of the driving forces behind this shift toward digital amplifiers and is at the forefront of future home audio innovations, changing the way consumers are able to experience entertainment. *aX*