ROHM SEMICONDUCTOR

How a High-Resolution D/A Converter Can Help Capture the Sound Quality of a Symphony Orchestra

The new BD34301EKV, a 32bit D/A converter IC designed for high fidelity audio equipment

White Paper

Introduction

Since the invention of the phonograph, musicians and engineers alike have combined their talents in order to capture the dynamic range of musical instruments and translate them seamlessly into a format to be recorded and replayed. This, however, comes with major challenges (even in the 21st century) where optimizing digital signal quality parameters such as sampling rate, resolution, signal-to-noise ratio (SNR), and total harmonic distortion plus noise (THD + N) requires close attention in all stages of production from circuit design to wafer fabrication and packaging. Since the 1970s, ROHM has been steadily developing audio ICs, optimizing devices with a low distortion and high resolution to adequately capture and reproduce the sound image (effects), timbre (natural sound without coloring), and low-noise characteristics of acoustic instruments in the ambient experience of a concert hall.

This whitepaper dives into the qualities and features of ROHMs latest musical IC — the MUS-IC™ BD34301EKV, a high-resolution audio digital-to-analog converter (DAC). This component, along with an array of others (including a high-fidelity power supply and sound processor), offers the basic building blocks of a Hi-Fi audio device.

BD34301EKV developed with three critical elements for classical music

To improve the listening experience of classical music, ROHM introduced a development concept that ultimately emphasizes three major elements in the BD34301 audio DAC. *__(see Figure 1 on the next page):__*

- Spatial reverberation
- Quietness
- Dynamic range

Spatial reverberation refers to the attenuation of sound pressure waves as they are emitted from the source and reflected off a multitude of surfaces within a facility. While this may seem like a simple natural phenomenon, in fact, it is a complex interaction that is critical for high-quality sound. Large concert halls must mitigate this decay of sound pressure waves in order to create a “fuller” sound and grant the audience a more immersive experience. On the other hand, orators would require a more rapid decay in sound energy for the speech to remain intelligible and not muddled. For instance, in a large gothic cathedral, sound energy can take up to 9 seconds to decay — this is massive compared to the rapid decay found in small offices/conference rooms. Concert halls will almost exclusively leverage hard surfaces to reflect the sound, while soft surfaces (such as the seating and curtains) would absorb these sound waves.

Typically, classical music is meant to be experienced for extended periods of time, with periods quiet (decrescendo) and loud (crescendo). This is further complicated by the fact that each acoustic instrument picked out from any musical family exhibits a timbre (i.e., a distinct musical sound/tone color). In order to capture the unique overtones of each instrument, as well as the full experience of multiple instruments playing loudly, the DAC must have a high dynamic range, high signal-to-noise ratio (SNR), and low distortion. This way, the audio is natural and flat across the sound spectrum, easing the experience for the listener over extended periods of time.

Figure 1: When approaching the design of the BD34301 audio DAC, ROHM developed a concept that focuses on three critical elements to successfully extract the full amount of information from the sound sources.

Parameters that affect sound quality

Sound quality can be directly impacted by very specific parameters found in every stage of development of an audio IC. ROHM has been able to fundamentally change the sound quality offered by ICs by comparing the audio output of their chips specifically designed for sound quality over a standard audio IC. An exhaustive investigation into all aspects from the circuit and layout design to the various fabrication and packaging steps led to the identification of 28 distinct parameters that affect the sound quality in terms of resolution, realism, and power ***(Figure 2)***. This section illuminates only a few of these parameters.

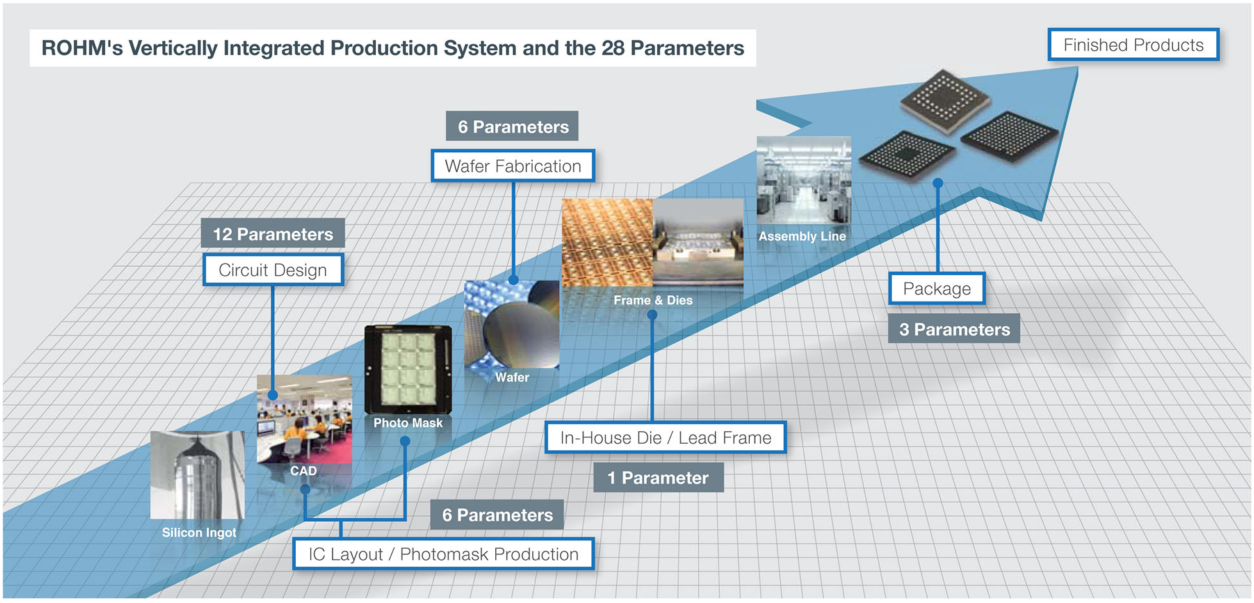

Figure 2: ROHM's engineers were able to establish 28 distinct parameters in the development and production chain that directly impact the sound quality of an audio IC.

Within the circuit design and layout, there are several optimizations that allow for an ideal sound quality ***(Figure 3)***. It is known that common impedance coupling in power supply and signal wiring can generate noise and distortion. When more circuits share a common ground return path, the signal ground can suffer from noise peaks during the charging of decoupling capacitors. Also, when the ground connections exhibit a high impedance from too much inductance (at high frequencies) or too much resistance (at low frequencies). This also occurs when there is a failure to separate signal ground from power ground. Reducing the impedance of the power supply wiring eliminates the potential for common impedance coupling from each current signal to the power supply pin. This makes it possible to match the impedance of the current segments to the power supply pin to minimize total harmonic distortion and noise (THD+N). The result — more optimal bass power and depth for a better sound range and balance.

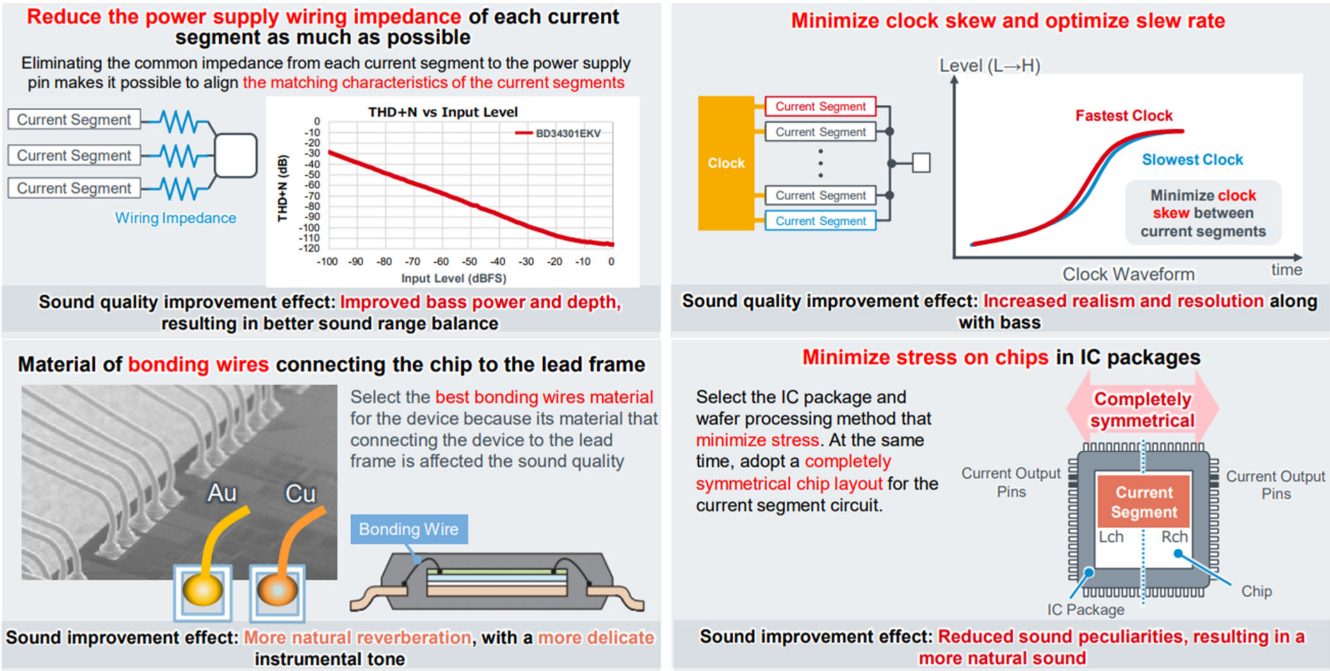

Figure 3: Sample parameters that are optimized in the production chain to generate a quality audio IC.

High fidelity audio equipment will have to take data from a number of sources (e.g. handsets, SD cards, Bluetooth, radio/TV tuners, etc.) and decode various discrete digital samples to a continuous analog signal to be outputted by speakers after sufficient signal processing and DAC ***(Figure 4)***. In an ADC and DAC, the clocks are fundamental to the process of measuring time units in order to process and convert continuous and discrete analog/digital signals. The delta-sigma (ΔΣ) modulator often used in ADCs and DACs leverage a fast-sampling clock to put these numerous digital signals together and generate the analog signal. There are known solutions for minimizing the sampling clock jitter that is critical to the SNR of the system. However, clock skew can occur at the current segments at the output of the ΔΣ modulator and can also be a culprit for the deterioration of sound quality. Minimizing the clock delay differences with a control circuit will optimize the chip's slew rate for increased realism and resolution.

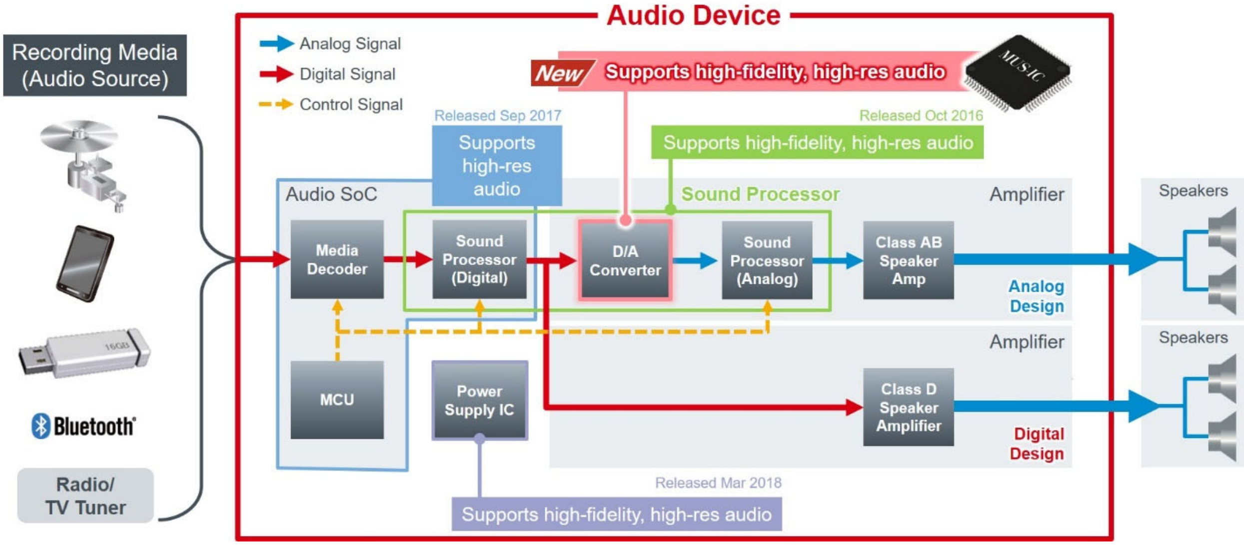

Figure 4: Sample of high-fidelity audio equipment.

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Outside of the design stage, the wafer fabrication, frame and die, and packaging stages all come with their own very specific audio IC optimizations. A noticeable improvement in sound can be perceived by carefully selecting the ideal bonding wire material for the respective device, as it is the physical interface that ultimately connects the device to its lead frame ***(See Figure 3)***. The combination of a completely symmetrical chip layout for the current segments leading to the output pins, and the selection of the ideal wafer processing and package method, can minimize stresses on the chips in the IC package. This effectively cleans the audio output from the IC by mitigating the minor sound defects that are caused by package-induced mechanical strains.

Understanding the MUS-IC™ BD34301EKV

In order to successfully reproduce the concert hall experience (i.e., spatial reverberation, quietness, and dynamic range), the BD34301 audio DAC has three main qualities that diverge from the standard audio IC ***(Figure 5)***:

- The incorporation of a circuit in the signal processing block to check for audio quality
- Best-in-class low noise and distortion (130dB SNR, -115dB THD+N)
- A customizable digital filter

The ability to check for audio quality in the signal processing circuit combined with the excellent signal-to-noise ratio (SNR) and total harmonic distortion and noise (THD+N) characteristic allows for a cleaner recreation of the target sound. Lastly, the customizable digital filter allows the user to process even minute sound signals — making it possible to achieve different sound quality tunings for audio equipment. Additional key features include reduction of the power supply wiring impedance, the minimization of clock skew, completely symmetrical layout and terminals, and a separate power supply for various functions and channels to generate the desired sound quality.

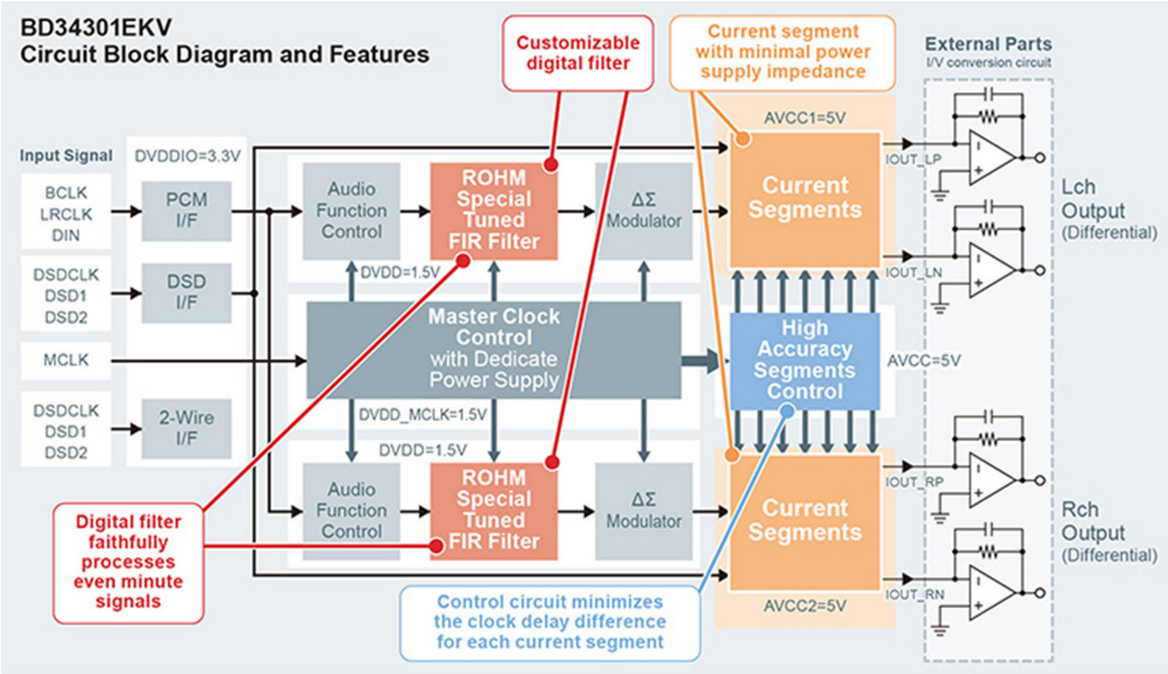

Figure 5: BD34301EKV block diagram and notable features.

The benefits of a customizable digital filter

The customizable digital filter is a key function of the digital signal processing circuit. The finite impulse response (FIR) differs from the traditionally employed infinite impulse response (IIR) filter: its impulse response is finite in duration while the IIR filter has internal feedback and continues to respond infinitely. An IIR filter typically has up to 10 coefficients while FIR filters can have over 100 coefficients — the output of the FIR filter is the sum of a finite number of finite multiples of the input values. This property allows these filters to be inherently more stable than IIR filters, with the tradeoff of consuming considerably more computing power. In a Hi-Fi audio DAC, however, this is worth the tradeoff - it allows for a much closer reproduction of the original signal, no matter how soft the signal may be.

Ultimately, the purpose of the filter in an audio DAC is to simplify the incoming signal. The oversampling technique is often used to reconstruct the phase of the digital signal into an analog one. Digital interpolation, or an intermediary high sampling rate, is used by the filter to add samples between the recorded samples. However, this interpolation can have a significant impact on sound quality, where inefficient interpolations can yield fewer samples and the inability to accurately recreate small signals. This is not acceptable to fully capture both the quiet and loud sounds found in acoustic instruments. The FIR filter found in the BD34301 audio DAC is customizable, as both the filter's calculation coefficients and oversampling rate are programmable.

The oversampling process of the FIR filter improves the smoothness and continuity of the waveform, generating a clearer sound image that lines up much more precisely with reality ***(Figure 6)***. This way, it is possible for the user to construct a unique digital filter to achieve different sound quality tunings for a particular audio equipment design — generating the sound ideally by the manufacturer while also reducing development time.

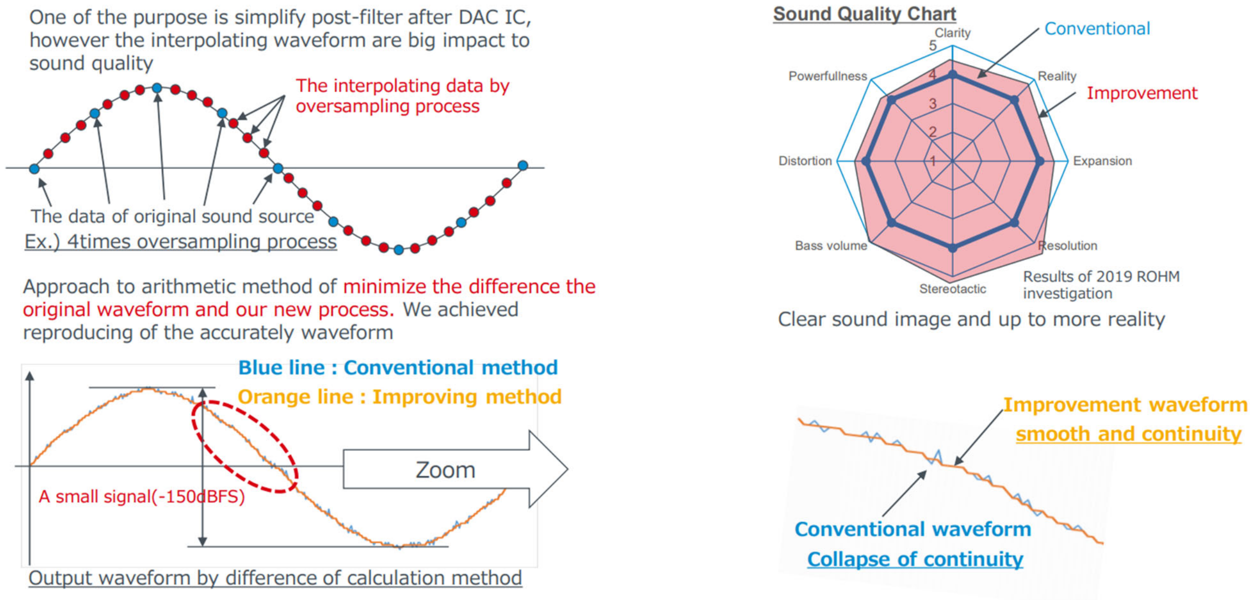

Figure 6: The FIR filter found in the BD34301 features customizable calculation coefficients and oversampling rate so that audio equipment manufacturers can fine-tune the sound quality to better replicate the original sound.

The MUS-IC™ BD34301EKV: Replicating the classical music experience through engineering craftsmanship.

When it comes to replicating the fullness of classical music in a concert hall, it is critical that the outputted digital or analog format very closely represents the original sound spectrum. The DAC is a cornerstone component critical to this process when very low distortion and noise are fundamental to its performance. Whether or not one is an audiophile, the flexibility of the Hi-Fi DAC through its customizable FIR filter allows for a robust level of tuning. Regardless, ROHM's audio engineers carefully assess and verify each audio IC's sound quality with very specific testing parameters in a sound quality chart (Figure 6) and through a vertically integrated production line, carefully optimizing the chip every step of the way. The MUS-IC™ BD34301EKV audio DAC is a direct result of highly specialized design, fabrication, and packaging to offer the "spatial reverberation", "quietness", and "dynamic range" found in classical music.

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