Why Not 16-bit No-OS No-Filter Conversion?

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ABSTRACT

It seems everyone these days is jumping on the bandwagon of non-oversampled and non-filtered 16-bit DACs. Yet Hagerman Technology has chosen to go the opposite direction. Why is that? Simple, we want to reproduce the truest, most pure signal possible. As will be shown, a properly upsampled and filtered 24-bit converter produces fewer artifacts, adding less unwanted and non-existent information.

BACKGROUND

It has long been realized in the hi-fidelity world that less can be more. Better to leave out musical information than to artificially add some. Speakers provide a good example of this. Rather than trying to reproduce the bottom octave of bass in a very poor fashion, most speakers just leave it out. You can still have excellent sound.

This is exactly the problem with non-OS, non-filtered DACs, which create unwanted and irrelevant signals by mistake. These added signals impact the quality of sonic reproduction, changing the musical character. The corrupted signals are caused by amplitude, timing, and ultrasonic errors. Differences are heard in the small stuff. It is in the micro-details where we begin our analysis.

AMPLITUDE

In any digitizing process of a continuous analog signal, there will be truncation errors in both amplitude and time. You cannot sample at infinite frequency with an infinite number of bits. Redbook CD provides us with 16 bits sampled at 44.1kHz. A direct reconstruction of a very small amplitude 1kHz sine wave would typically appear as in Fig. 1.

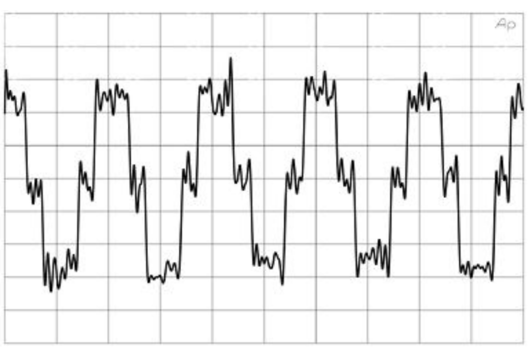

Fig. 1. 1kHz (-90dBFS) 16-bits.

The truncation error is obvious, as there are only three discrete levels available. The signal is no longer a sinewave, but a stair step-like recreation with steep edges and plateaus. This waveform has high harmonic content, plenty of odd-order distortion components. Only the fundamental should exist, and all harmonics are artificially added information. The result is a sound that can be harsh and edgy, or at best overly detailed and exciting.

In comparison, an upsampled digital filter can interpolate, or fill in, information between samples. By applying our "less is more" theory, these filters assume the input is more likely to be a sine wave than a square wave, thereby rounding the edges with the lowest frequency content possible. Fig. 2 shows the result of interpolating to 24 bits. Clearly, the waveform is much better representation of a sine wave.

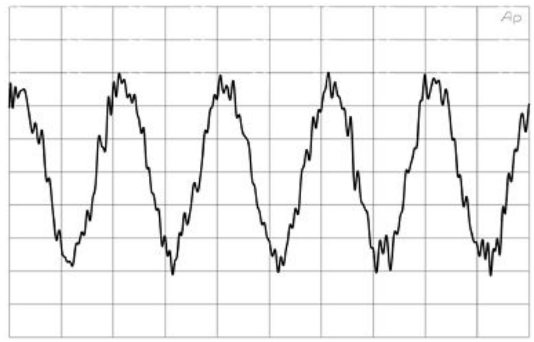

Fig. 2. 1kHz (–90dBFS) 24-bits.

TIMING

Perhaps you have heard comments like "but the square wave performance of my non-OS DAC is perfect!" First of all, we are not trying to recreate test waveforms, but music. Nonetheless, square wave testing is a classically good performance benchmark, as both amplitude and phase anomalies are immediately apparent. So let us examine the errors in reproducing a 1kHz square wave.

The obvious problem here is asynchronous sampling. The 44.1kHz rate does not offer an integer number of samples per cycle. There are 44 samples 9 times in a row, then one with 45 samples. Repeat. The result is a mixture of two fundamental frequencies, 1002.3Hz and 980Hz, but never 1kHz! Now in reality, things are not so bad, as the anti-aliasing low pass filter in the original ADC process will remove the steep edges of the input square wave. Nevertheless, we get some pattern dependent jitter.

In contrast, 24-bit digital interpolation will smooth out the edges through band limiting and interpolation, thus greatly decreasing the problem.

RATE CONVERSION

Some converter architectures use asynchronous upsampling, which will have similar timing issues as described above, with beat frequency and dynamic interpolation errors. A high clock rate can greatly reduce their impact. Nevertheless, there will still be a residual amount, compared to a fully synchronous type.

FILTERING

The final problem with non-OS, non-filtered DACs is the amazing level of ultrasonic output content. Without a brick-wall filter, both switching edge harmonics and aliased images are sent downstream for other system components to deal with. Each system must be hand tuned so that cables, amplifiers, loudspeakers, and ears provide the necessary filtering.

Some amplifiers react benignly to the ultrasonic and RF error components, acting themselves as decent low pass filters. Others, especially those with transistor front ends, may rectify and demodulate, causing bias shifts, instability, and even continued amplification.

The output filter of an upsampled converter is relatively simple, not having to be a nasty brick-wall type. They can maintain a linear phase response *and* remove the aliased images and RF content. This eliminates the need for special system tuning, placing no limitations on your choice of amplification.

CONCLUSION

It has been shown that a non-OS, non-filtered, 16-bit DAC architecture contributes extraneous and unwanted error information to the output signal, corrupting the purity of reproduction. In contrast, a properly designed upsampled, filtered, 24-bit converter can greatly reduce these errors. The result is a more realistic replication of music and tonal integrity.