Asynchronous Upsampling to 110 kHz

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Benchmark DAC1 converters use upsampling techniques to improve the quality of the digital to analog conversion. Benchmark's choice of 110 kHz is a bit unorthodox. It may seem more logical to upsample by 2X or 4X and convert at standard sample rates such as 88.2 kHz, 96 kHz, 176.4 kHz or even 192 kHz. Our intuition may suggest that integer ratios would produce the best results. Our intuition may also suggest that higher is better, and therefore 192 kHz would be the "best" choice for an output sample rate. Unfortunately, intuition often leads us down the wrong path if it is not balanced with reason and scientific analysis. Benchmark's analysis and testing showed that 110 kHz offered advantages over the choices that seemed more reasonable. The following is a short summary of the decisions that went into our choice of 110 kHz:

Asynchronous Upsampling Solved the Jitter Problem

Early in the design process, we identified a distinct performance advantage that could only be achieved using asynchronous upsampling. The digital signals feeding any outboard D/A converter contain too much jitter to directly drive the D/A sampling circuits. The conversion clock can be entirely independent from the data transport clock if upsampling is asynchronous. The performance of an upsampling circuit is limited only by the precision of the digital signal processing (DSP) allocated to the process. Our intuition may suggest that synchronous conversion is superior to asynchronous, but this is not the case. Our tests showed that asynchronous upsampling could achieve the same performance as synchronous upsampling, but with the distinct advantage of jitter attenuation. On the basis of jitter attenuation, we chose an asynchronous upsampling process.

The Asynchronous Upsampling Process does not Require Integer Upsampling Ratios

Once we had chosen an asynchronous up-sampling process, we discovered that we were free from the restriction of integer upsampling ratios. Our upsampling algorithm is identical for all up-sampling ratios. In other words, the process required to upsample by 2.29166 is identical to the process required to upsample by a factor of 2. The digital filtering process does not get simpler or better when integer ratios are chosen. Spectrum analysis tests showed that integer and non-integer upsampling ratios produced the same audio quality. This allowed us the freedom to explore non-integer upsampling ratios.

Frequency-Shifting Solved Digital Filter Performance Problems

As designers of high-performance converters, we were often frustrated by the fact that we had no control over the digital filters that were built into A/D and D/A conversion chips. With outboard DSP processors it is possible to build digital filters that outperform the filters that are built into the best pro-audio conversion chips, but there is usually no means of bypassing the internal filters. We soon discovered that the asynchronous upsampling system gave us a mechanism for frequency shifting the filter that was built into the D/A chip. The near-Nyquist transition band of the filter could be moved above the Nyquist frequency of the incoming audio. This frequency shifting essentially makes this filter invisible to the incoming audio. Given this frequency-shifting tool, we now had the flexibility of replacing the built-in filter with a high-performance external filter. This development was very exciting, and we began to explore the possibilities.

Minimum Upsampling Ratio to Replace Built-in Filters

The filters that are built into D/A conversion ICs can effectively be removed from the signal path by frequency shifting upward by a ratio as small as 1.1 to 1.2. The upsampling ratio must be sufficient to move the transition band of the brick-wall filter to a frequency that is greater than ½ of the input sampling rate. For example, a 44.1 kHz D/A converter usually has a brick-wall filter with a transition band that begins at 20 kHz. An upsampling ratio of 22.05/20= 1.1025 will move this brick-wall filter upward to 22.05 kHz. 44.1 kHz x 1.1025 is 48.51 kHz. In other words 44.1 kHz should be upsampled to at least 48.51 kHz. If an upsampling ratio of 1.1025 is sufficient, 48 kHz should be upsampled to at least 52.8 kHz, and 96 kHz should be upsampled to at least 105.6 kHz. Perhaps Benchmark's choice of 110 kHz is becoming clearer. But wait, our intuition would suggest that 192 kHz should be upsampled to at least 211.2 kHz. Why not upsample everything to 211.2 kHz? Unfortunately intuition alone could take us down the wrong path. For performance reasons, we chose to downsample 186.4 kHz and 192 kHz to 110 kHz.

What Converter Manufacturers Don't Want You Know

An examination of converter IC data sheets will reveal that virtually all audio converter ICs deliver their peak performance near 96 kHz. The 4x (176.4 kHz and 192 kHz) mode delivers poorer performance in many respects. In most cases, noise, distortion, pass-band ripple, stop-band attenuation and other key performance measurements are significantly better in the 2X (88.2 kHz and 96 kHz) mode of operation. Every A/D and D/A conversion IC that we have tested performs better at 96 kHz than at 192 kHz. In most cases THD+N, SNR, passband ripple, and stopband attenuation are all poorer at 192 kHz than at 96 kHz. Based upon these tests, I am not surprised that there is not yet any conclusive evidence that 192 kHz is better than 96 kHz. Given the current state of the art, 192 kHz should sound poorer than 96 kHz. 192 kHz provides additional bandwidth between 48 kHz and 96 kHz but there is no real evidence that this is useful given the

limitations of our microphones, speakers, and hearing. 192 kHz adds useless bandwidth while decreasing performance.

If 105 kHz is High Enough, Why Does Benchmark Use 110 kHz?

We chose 110 kHz because this is near the maximum frequency that the AD1853 can reliably operate in the 2x (88.2 kHz and 96 kHz) mode. We measured the performance of the AD1853 vs. sample frequency before choosing 110 kHz. At 110 kHz, the passband of our frequency-shifted AD1853 is 45.8 kHz. When combined with our outboard filters, the DAC1 is almost entirely free of images (alias effects). Given a different converter IC, we could choose a different conversion frequency (our asynchronous upsampling topology allows this flexibility).