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Dear Reader,

There has been a great deal of misinformation regarding the benefits and possibilities of upsampling and oversampling. I want to give an overview of the differences between upsampling and oversampling. I will then briefly traverse the history of these techniques as employed in CD players and elucidate the reasons for some common misconceptions about them. Finally, I would like to set expectations for audiophiles as to what these techniques can and cannot do for a 44.1/16 playback medium.

I have yet to see an "official" distinction between upsampling and oversampling. From recent usage, I can only conclude that upsampling is any technique that increases the sample rate, for example from 44.1 to 96 kHz. Oversampling, in contrast, is a form of upsampling where the rate is increased by an integer multiple, that is, 4x, 8x, etc. For audiophile purposes, both processes are used for one reason: to reduce the artifacts of the digital to analog converter. No new information is created -- once the original has been sampled at 44.1 kHz, anything above 22 kHz is lost forever. The whole idea of upsampling is just to make the d/a converter behave more ideally and introduce fewer artifacts into the analog audio signal.

About 7 years ago, a new technique was developed to do "asynchronous" sample-rate conversion using inexpensive silicon. The original application was for professional use, where mixing digital recordings made from different master clocks required either conversion to analog and back to digital, or had to be done on an expensive computer workstation. The advent of this new silicon allowed studio consoles to provide "real-time" mixing of these different tracks. Some designers in the audiophile community began using them as jitter-reduction devices, even though they introduced new artifacts as a result. Then some suggested that upsampling was a substitute for higher-resolution formats, implying that "upsampled CD" is comparable, or nearly so, to DSD or 192kHz/24-bit PCM. But alas, there is no free lunch. For if this were true, digital audio could be distributed with a sample rate of, say, 100Hz, and then simply upsampled upon playback.

There is no question that different up/oversampling methods sound different. Consider that with 16x oversampling, 15 new data points are interpolated between successive samples. Every 16th point that comes out of the filter is the original data straight off the disc. The intermediate points require additional precision, hence the common use of a 24-bit filter. In contrast, converting from 44.1 to 96 kHz represents a 2.17687... ratio. This means that every output sample is the result of a very complex calculation, and invariably contains some artifacts not related to the original music signal. Every upsampling product I have heard sounds significantly better at integer multiples -- for example, 176.4 kHz instead of 192 kHz on CD playback.

These methods do not address the CD red book limitation: the sampling rate could have been higher, and the number of bits could have been higher as well. Fortunately, the audiophile community has spent incredible energy working to improve what was originally hailed as "perfect sound forever." In short, 44.1/16 has come a long way -- consider that jitter effects were discovered in the early 1990's, and today even the large manufacturers acknowledge the importance of low-jitter design.

The goal of CD player design is to get the most out of what's on the disc. There are many ways to approach this, but any claim that "hidden" information can be restored by a magical upsampling process should be met with a healthy dose of skepticism.

Regards,

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