Intersample overshoots

Introduction

When a continuous-time waveform gets sampled, the samples will usually not be exactly on the peaks of the waveform. Conversely, when a continuous-time waveform is reconstructed from a sampled signal, some of its peaks will usually be greater than the largest sample.

Many digital recordings have their largest sample normalized to full scale. As a result, the waveform in between samples has to exceed full scale. To prevent clipping or folding, interpolation and reconstruction filters need to have enough headroom for this.

Digital interpolation filters are often designed with no headroom at all. Hence, DACs with digital interpolation filters are often incapable of reproducing peak sample normalized recordings without hard clipping. Similar problems can occur with sample rate converters, either in hardware or in software.

Much more information can be found at the Benchmark Media website, for example https://benchmarkmedia.com/blogs/application_notes/intersample-overs-in-cd-recordings

The effect is generally at its worst at relatively low sample rates. The lower the sample rate, the larger the distance between the peak of the continuous-time waveform and the nearest samples can be.

One can argue, and probably correctly, that normalizing the largest sample to full scale is a poor practice and should simply not be done. This doesn't help much when you want to listen to a CD that has been peak sample normalized, though. When you use computer files rather than CDs or other physical carriers, you can simply process the file and reduce its volume slightly to solve the whole issue.

Test signal to check for headroom

A simple two-tone test can clearly show whether or not a given DAC has headroom for intersample overshoots. I've generated a signal

$$\frac{1}{2}\sin\left(\frac{1}{2}\pi\left(n+\frac{1}{2}\right)\right)+\frac{1}{2}\sin\left(\frac{2}{3}\pi\left(n+\frac{1}{2}\right)\right)$$

where the positive and negative full scale are +1 and -1. The integer *n* is the sample number. At 44.1 kHz sample rate, the signal consists of frequency components of 11025 Hz and 14700 Hz.

For integer *n*, the result of the equation never exceeds $\frac{1}{4}\sqrt{2}+\frac{1}{4}\sqrt{3}\approx 0.7865660925$

After ideal interpolation by (for example) a factor of four, the same equation applies, but *n* can then be any multiple of 1/4. The peak value then becomes 0.9619397663.

Interpolation by two: peaks increase to 0.9330127019

Interpolation by four: peaks increase to 0.9619397663

Interpolation by eight: peaks increase to 0.9733555533

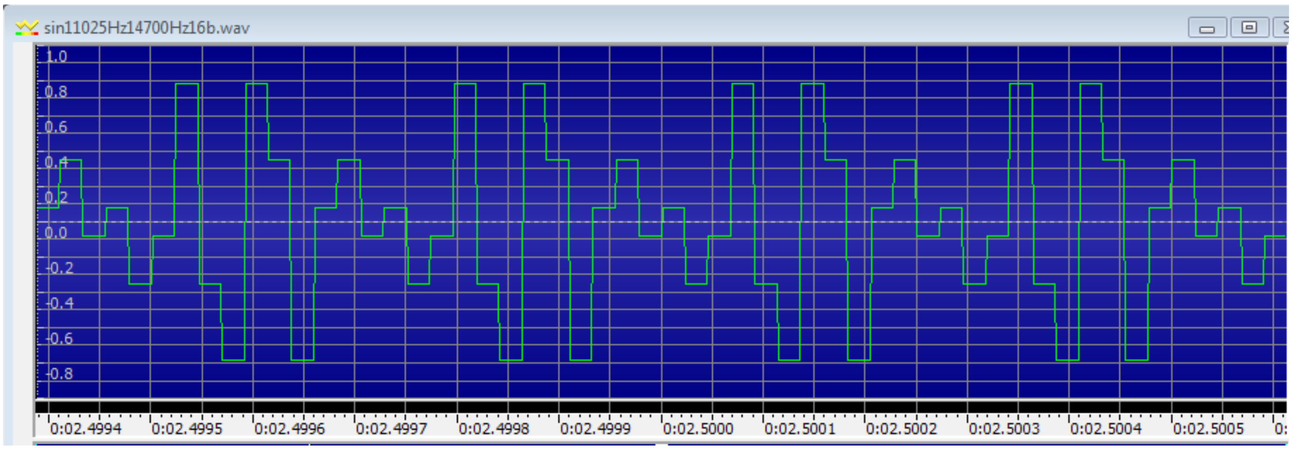
Interpolation by sixteen: peaks increase to 0.9741925986

Interpolation by thirty two: peaks increase to 0.9754082668

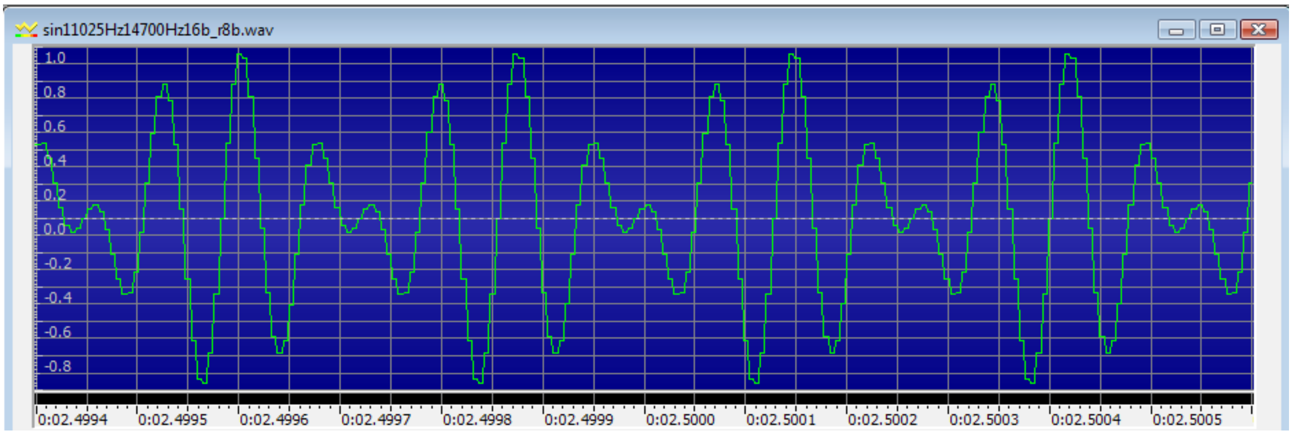
I've also made a peak sample normalized version of the same signal, that is

$$\frac{4}{\sqrt{2}+\sqrt{3}}\left(\frac{1}{2}\sin\left(\frac{1}{2}\pi\left(n+\frac{1}{2}\right)\right)+\frac{1}{2}\sin\left(\frac{2}{3}\pi\left(n+\frac{1}{2}\right)\right)\right)$$

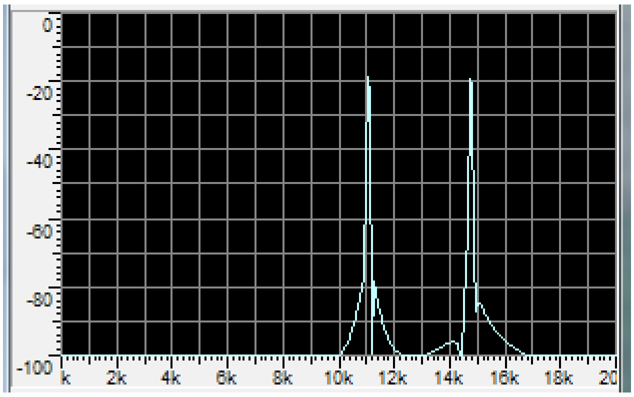
The waveform of the test signal without normalization is:



Interpolating this to 176.4 kHz sample rate using r8brain results in:

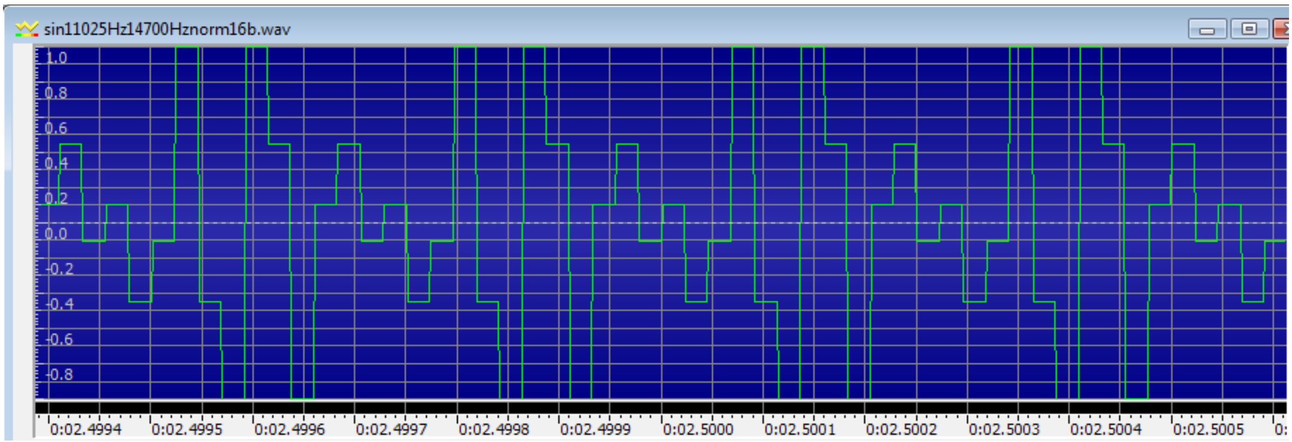


of which the spectrum is:

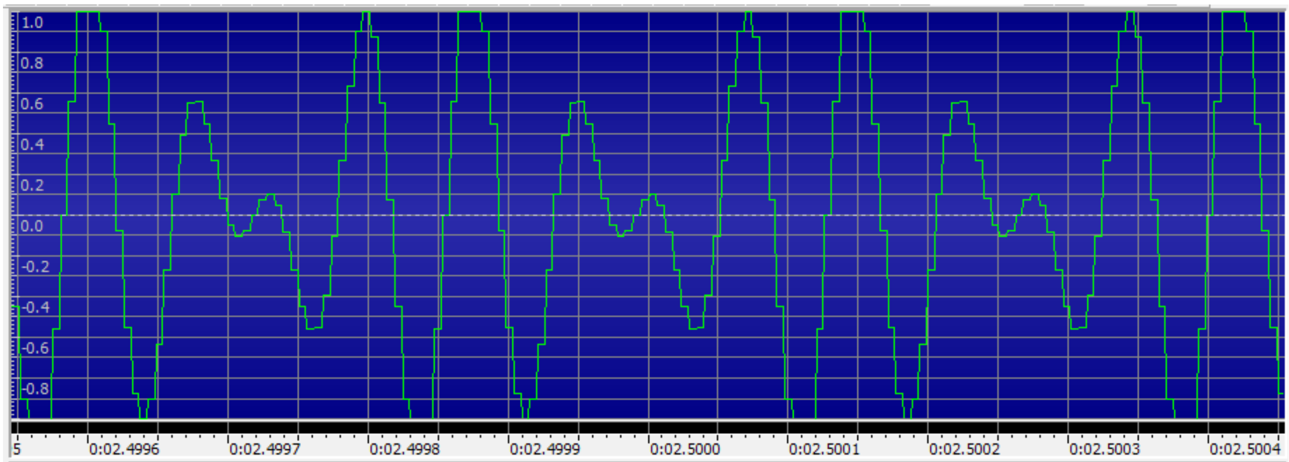


so still peaks at 11025 Hz and 14700 Hz.

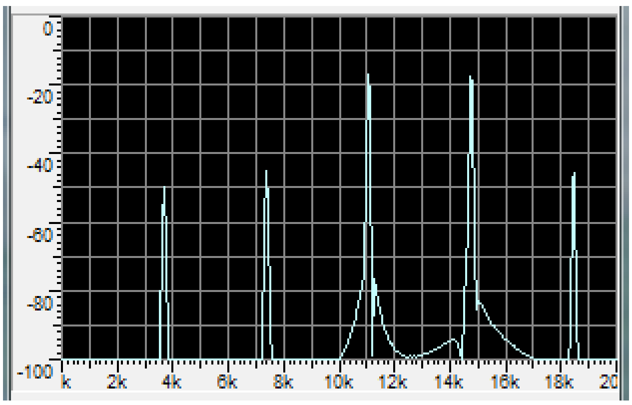
The waveform of the peak sample normalized test signal is:



which r8brain turns into



clearly showing hard clipping. The spectrum is



which has severe intermodulation distortion with peaks at all multiples of 3675 Hz.

In order to check whether a DAC has headroom for intersample overshoots:

1. Listen to the normalized file sin11025Hz14700Hznorm16b.wav (at a moderate volume!)

2. Compare the sound to that of the file without normalization, sin11025Hz14700Hz16b.wav. If this file sound softer but otherwise very similar, apparently there is a headroom of at least 1.48 dB...1.87 dB.

3. Also compare it to sin11025Hz14700Hznorm16b_r8b_r8b_att.wav. This file is full of intermodulation: it has been clipped by r8brain, converted back to the original sample rate and slightly attenuated. If the sound in step 1 is more similar to step 3 than to step 2, there is an intersample overshoot problem.

Audibility on music

I haven't a clue whether there are any music signals around on which this effect is audible. I don't really care either, audio DACs should simply not clip on music. In any case, intersample overshoots are not something rare; according to Benchmark Media, Steely Dan's *Gaslighting Abbie* from *Two Against Nature* has about 3.7 intersample overshoots per second.