The “Pure Nyquist” filters of the MU1

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Our MU1 Music Player is receiving critical acclaim for its sound quality. Many people ask us how a unit with just digital inputs and outputs can improve the sound quality so much, so we decided to write this explainer text.

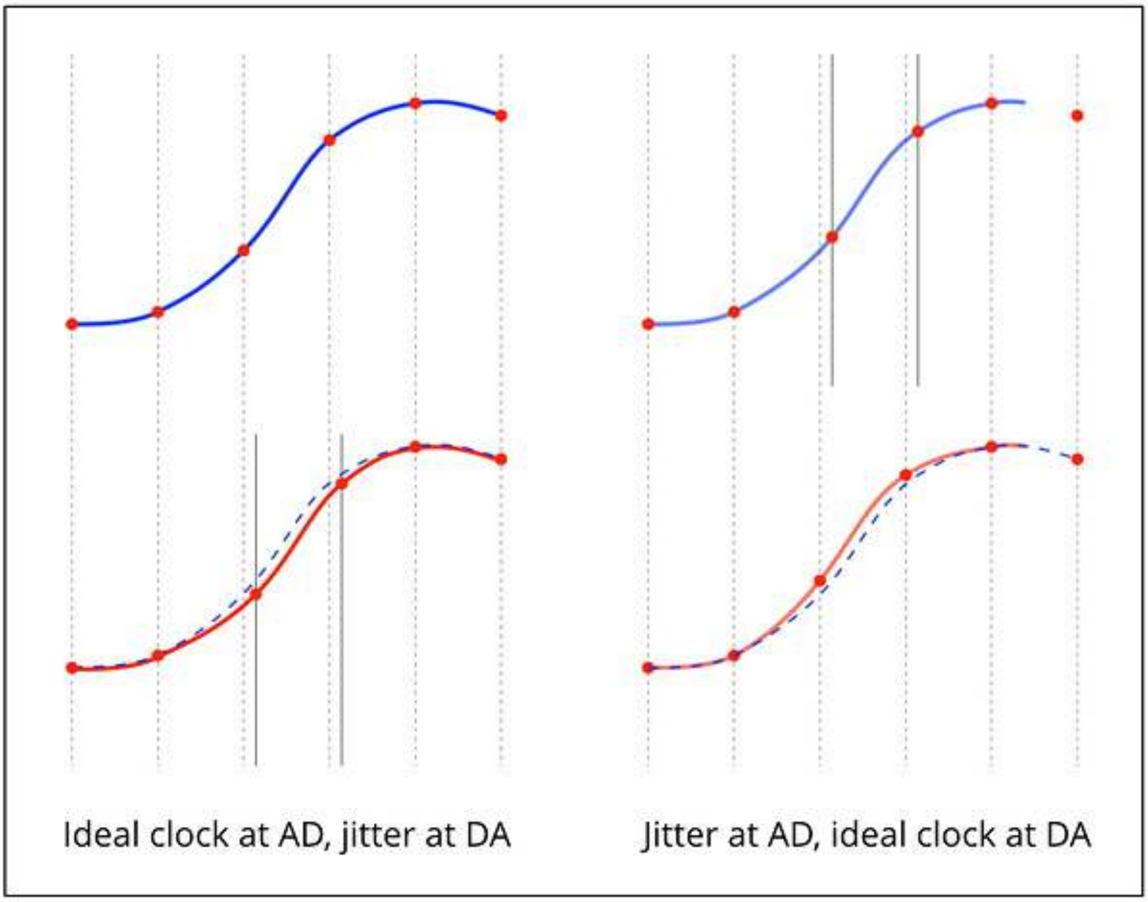
At the core of the MU1 a so called ‘FPGA’ processor is found. This is not some sort of magical element, but just a very powerful calculator that happens to be well suited for audio processing such as filters. Whenever audio is up or down converted from one sample rate to another a filter is needed to remove frequencies that are above half the lowest sample rate of the two, which is called the ’Nyquist frequency’. The up- and downsampling filters of the MU1 have extreme resolution in all dimensions. We discovered that precision like this is needed to recover microdynamic detail from humble CD tracks as well as from pristine DSD files.

Those of us who remember listening to the first digital CD recordings may still clearly recall the thrill from the absence of wow and flutter, noise and distortion. Would audio replay become ‘perfect’ as the measurements suggested? When we all got more used to the improved aspects of the sound quality, it became clear that perfection had not been achieved. Digital seemed to have a sound of its own, often described as harsh, fatiguing and non-involving. This intrigued members of the Grimm Audio team from the very start. What caused this seemingly perfect system to sound imperfect?

What we learned is that the masking effect of analog noise had some influence. It seems that when veils are removed, it shows that what’s beneath is not always pristine gold. However for some reason the small amounts of distortion added by the digital replay chain seemed to harm a pleasurable sound more than most analog aberrations. This called for a lifelong quest to find what causes this.

One clue was that digital converters are analog for a large part. In a DA converter the power supplies and the output circuitry are clearly analog. But also the final stage of conversion within the DA chip, where digital bits are turned into voltages, is more analog than digital. This final stage on the edge of digital and analog appears to be one of the most sensitive ‘analog’ parts of the whole audio system. The noise on its power supply must be very low to guarantee that the amplitude of the output signal is as intended. But even more important, the timing of the conversion moment from digital to analog must be extremely precise. A variation in timing of the ‘clock’ that conducts this process is called jitter. As it turned out, extremely low jitter is required for digital audio system to prevent harshness, lack of detail and stereo image blurring. When you realize that these high demands on the performance exist in an environment where nearby circuitry switches at high speed, it becomes clear that designing mixed digital and analog circuitry is much more demanding than that of purely analog audio circuitry.

A second clue was found in the digital domain. We realized that the required extremely low clock jitter for ‘perfect’ audio playback implied that an extreme calculation precision for digital signal processing would be needed as well. It can be shown that digital signal ‘quantization’ not only sets limits to the noise floor but simultaneously affects the timing precision of signals. This means that whenever a process is performed with insufficient calculation power, an amplitude error is introduced that affects sound quality in a similar way as jitter does. This is illustrated in the picture below. On the top left side an analog signal (in blue) is acquired by an AD with a ‘perfect’ clock. The down left graph shows what happens to the reproduced signal (in red) if the DA converter has a jittery clock where not all sample moments are played out in time. The right side of the picture shows that the same problem occurs when the AD has a jittery clock and the DA a ‘perfect’ clock. As it shows, these timing errors lead to an amplitude error. But then of course the reverse is also true: an amplitude error can be seen as a timing error (the amplitude is wrong due to the error, but it would have been right at a slightly different time). So jitter and amplitude errors have a similar effect on the waveform, although they have a completely different origin.



Timing errors and amplitude errors distort the waveform in a similar way.