Digital Audio with an Analogue Soul

How QRONO d2a Improves Digital Audio Playback

Background

The early days of digital audio promised, and mostly delivered, a better way to store and transfer audio recordings. Compared to the best analogue sources like open reel tape and vinyl, digital audio offered immediate benefits: no generation loss when copying, and a much more stable medium that didn't deteriorate with time and repeated playback. Traditional measures of fidelity, such as noise and distortion, along with mechanical variations like wow and flutter, also showed improvement with digital audio recording. Yet, despite these advancements, there was something about digital audio that was less engaging, less lifelike than pure analogue audio.

Careful listeners commented that the sound seemed flatter and more mechanic, with a subtle harshness that, over time, could cause listener fatigue. What was missing from digital music? Why did analogue recordings, with all their obvious imperfections, sound more alive? These questions inspired creative scientists and engineers to dig deeper into the mysteries of human hearing and how they relate to digital audio capture and playback.

Modern Neuroscience Offers Hearing Insights

When we think of hearing, we first think about the ear and how it translates tiny air pressure variations into electrical impulses which travel to the brain via our nervous system. Our ability to perceive even the quietist sound of danger and pinpoint the direction it came from, was critical for our survival.

Our brain processes these sensations to create a broad impression of sounds that help us live. Sound can generate great pleasure and meaning. Research shows that all regions of the brain can be stimulated by the sounds we call 'music'. Rhythm, melody and harmony all stimulate different brain structures and, collectively, can create levels of emotion that engage higher levels of brain activity.

Our hearing mechanisms are far more sophisticated than our eardrum and pinnae simply capturing and delivering sound to our brain. The brain and nervous system are just as important. When we listen, more information is going from the brain to the ear, than from the ear to the brain. We can actively focus on certain elements of sound and tune out others. Like most human activity, hearing acuity can be improved with learning and practice.

It's About Time

One of the most surprising recent discoveries in neuroscience has been the precision of our hearing when discriminating sounds with fine time detail. We can perceive distinct auditory signals as close together as 7 microseconds [1-3]. This precision was critical for early humanity's survival. As it applies to music perception, it provides the fine texture, clear delineation and spatial impression of musical instruments. Audio processing that fails to preserve these time details will blur sounds together.
Much of what you will read later in this white paper is not new information. But the fact we can perceive sounds as close together as 7 microseconds is new information. For decades, digital audio technologies and systems overlooked the critical importance of this level of accuracy. Armed with this new knowledge, it's imperative to double-down on efforts to improve and fix this aspect of digital audio. This time-domain precision is the key to making digital audio sound as natural as analogue, while maintaining its core advantages.

Finding Time

Commonly used and accepted methods of translating analogue-to-digital and digital-to-analogue require filtering. In Analogue to Digital Convertors (ADCs), we wish to use filtering to prevent downward aliasing when reducing the very high sample rate from the modulator to a useable rate for recording, mixing, mastering and distribution. In Digital to Analogue Convertors (DACs), we wish to use filtering to control the reverse of that problem known as upward imaging. This is required when turning that lower sample rate back to a very high sample rate to supply to the DAC's modulator.

Conventional design of DAC filters prioritises traditional measurements, ensuring that there is no upward imaging and a completely flat pass band often above the highest frequency of any musical information. The trade-off is that these "brick wall" designs create time smearing. This can alter music perception in subtle but significant ways. In mechanical terms, these filters would be considered severely underdamped. Think of your car driving over a single bump in the road and the suspension being so soft that the car continued to bounce up and down for some time. This is like the concept of an impulse response, which characterises a system's response to a sudden and short-lived input. Figure 1 shows a perfect digital impulse or delta function. The signal is full scale at a single sample and zero everywhere else. This can be used as an input signal for measuring the impulse response of a system. The output of the system shows the response of the system to that impulse.

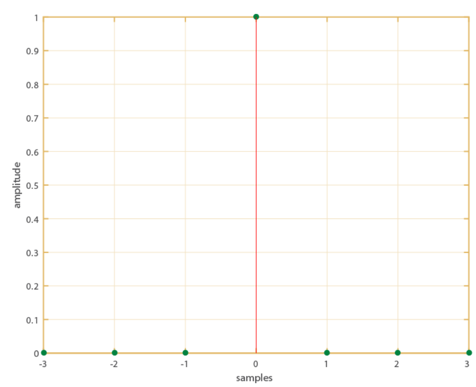

Figure 1: A digital impulse or delta function

When looking at an impulse response of a typical analogue component (shown in Figure 2), we can see what we call 'post-ringing'. In a physical system, this is caused by the resonances of the masses and springs which can't stop resonating fast enough to accurately follow the input signal. You can see that energy of the input impulse is spread slightly in time by the ringing of the filter.

Audiophiles often describe the speed or the 'jump factor' of music. When the drumstick taps the cymbal, or the piano hammer strikes the string, there is an instantaneous burst of energy that brings immediacy and vitality to the sound that is extremely engaging to the listener when reproduced faithfully by the audio system.

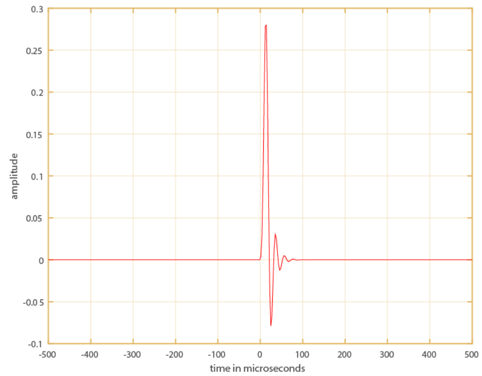

Figure 2: A simulation of the impulse response of a typical analogue system (5th order Butterworth with 50kHz cut off simulated at 384kHz)

With the advent of Digital Signal Processing (DSP), it became possible to implement equivalents of existing analogue filter designs and more complicated designs that weren't practical to create in an analogue system. One of the most popular classes of new digital filters were called linear phase filters. On paper, these filters sound like a great idea with many useful properties. As they are symmetric, they were easy to implement and common in early DAC chips. They are often still the default option in modern DAC chips.

A typical filter of this kind is shown in Figure 3. This filter has both pre and post-ring. The energy of the input impulse is spread both before and after the main peak of the filter. Pre-ring doesn't occur in nature or physical systems. This may be why this kind of filtering is often perceived as unnatural and may be one of the irritants many careful listeners complained about with early digital audio. Imagine if your car started bouncing up and down before you hit a bump in the road!

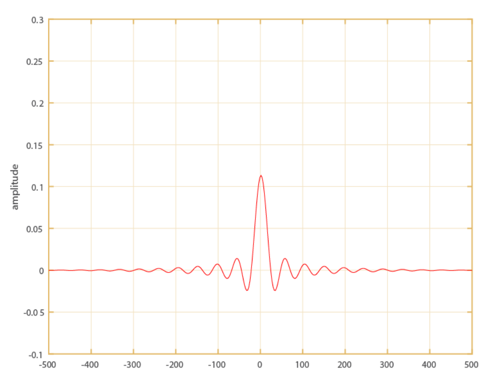

Figure 3: A typical DAC fast roll-off linear phase filter shown with 48kHz input and the same axis as Figure 2 for comparison

The post-ringing is similar to natural reverberation and can be more easily ignored by the brain, but it is still beneficial to minimise it to accurately preserve the time information in a recording. If the energy of one sound event is allowed to overlap the space of another, what was once sharp and well defined will become blurry and indistinct.

Addressing The Problem: QRONO d2a by MQA Labs

Now that we understand the problem, we can start to create solutions. This is what QRONO d2a is all about. Designing DAC filters that do not exhibit pre-ringing and stop ringing fast enough to not blur events that can be discerned by our hearing.

QRONO d2a uses proven engineering practice combined with thoughtful and creative solutions to create a breakthrough in digital audio reproduction.

Almost all DAC chips in popular use today employ upsampling of the input signal to drive their high-speed modulators. Typically, these filters are designed to create the most impressive spec sheet numbers for low distortion and noise. They look good on paper, but do they sound good? Too often the time domain performance suffers in a quest for impressive measurements of other factors like noise and dynamic range, with input signals that don't represent what we want to listen to with our audio equipment.

To design better filters, MQA Labs started by concentrating on the actual spectrum of natural sounds and music which are the signals we want to listen to. Environmental sounds show a tendency for amplitude to decrease as frequency increases. Music has similar properties. [4] What Audiophiles refer to as the bass and midrange frequencies have the greatest amplitude, while the higher frequencies are much lower in level. While they are lower in level, they are still critical to our perception of timbre and ambience cues, so important in musical enjoyment.

MQA Labs designed a set of filters that provide adequate anti-imaging performance with musical input but with a significantly better time response than conventional filters. We can use these filters instead of the conventional ones built into our DAC chip. In some instances, we can also program our own filters into the DAC chip. This allows us to further improve the time domain performance by considering the whole system of our upsampling filters and the upsampling filters in the DAC chip.

Most DAC chips also use the same filters for inputs at different sample rates. This often means they are unnecessarily aggressive with high resolution inputs of 88.2kHz and higher. By contrast, QRONO d2a filters are optimised for the musical spectrum of each input sample rate. In all cases, they preserve musicality but ensure there are no ultrasonic outputs that could damage downstream equipment like amplifiers and loudspeakers.

Finally, we use a technique called noise shaping to mask any low-level imaging in the ultrasonic region with benign noise. This noise also helps improve the performance of the DAC chip. The shaped noise maintains the signal's dynamic range in the audible band but reduces the number of bits required to represent the signal. This means we can use the most linear region of the DAC chip's dynamic range. [5]

Each QRONO d2a implementation is specifically tuned for the DAC chip in that product.

Measuring The Impact

Let's look at some impulse response measurements of product with and without QRONO d2a with both 48k and 192kHz input. Before QRONO d2a was enabled on this product, a minimum phase filter had been chosen which was a more natural sounding starting position than something like the linear phase filter we saw early in Figure 3.

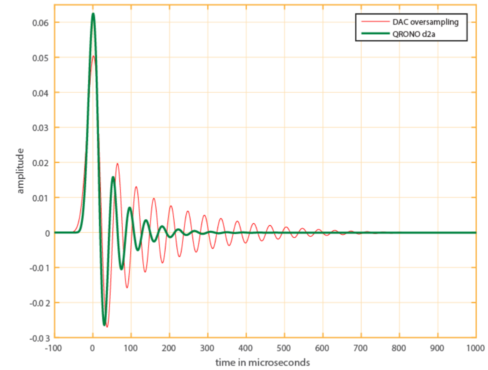

Figure 4: Analogue output measurement with DAC oversampling and QRONO d2a. 48kHz input linear amplitude scale.

Figure 4 shows the 48kHz response of the DAC's built-in filter and the shorter QRONO d2a filter. 44.1kHz and 48kHz are challenging rates to upsample as the top of the audio band is close to the maximum frequency representable at that sample rate.

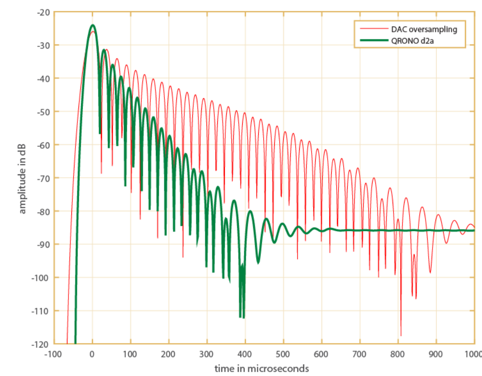

Figure 5: Analogue output measurement with DAC oversampling and QRONO d2a. 48kHz input dB amplitude scale.

Our perception of envelopes is closer to logarithmic, so Figure 5 shows the same measurements as Figure 4 but with a logarithmic dB scale. This allows us to clearly see the length of the ringing of the filters before it falls into the noise floor of the measurement. It's now possible to see that the QRONO d2a filter is twice as compact as the DAC's built-in filter.

Figure 6 shows the device measured with 192kHz input. You can see that the DAC is using the same filter for 192kHz input as it was for 48kHz input but running 4x faster, so it stops ringing four times quicker. We have made the time axis four times shorter so this can be seen more easily. By contrast to the DAC's built-in filter, QRONO d2a is using a totally different filter designed specifically for 192kHz input. It is much shorter with a near perfect time response.

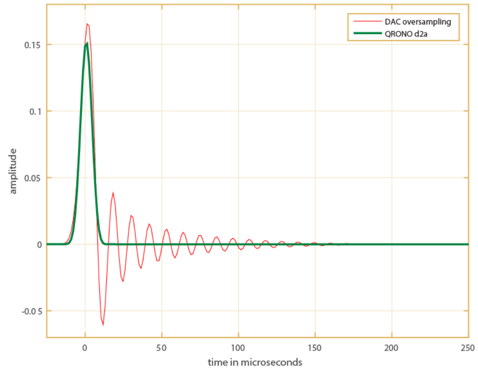

Figure 6: Analogue output measurement with DAC oversampling and QRONO d2a. 192kHz input linear amplitude scale.

It is important to note that this measurement only represents the playback response of the DAC and its analogue output circuitry. This doesn't account for or correct any blur in a recording from the ADCs used. That requires a different technology!

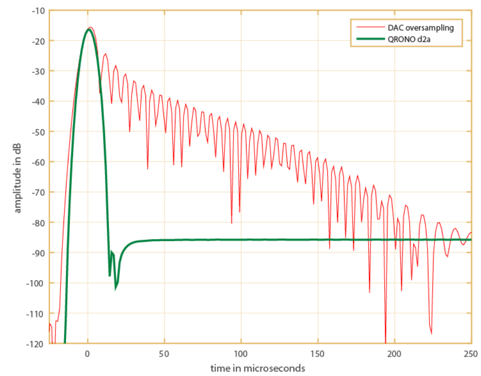

Figure 7: Analogue output measurement with DAC oversampling and QRONO d2a. 192kHz input dB amplitude scale

Figure 7 shows the measurement in Figure 6 with a dB amplitude scale. Here we can see the QRONO d2a time response is around twenty times more compact than the DAC's built-in filtering.

We started with a question, "what was missing from digital music?". Perhaps we should have asked, "what was added by digital capture and playback?". We now know that the ADCs used in recording studios, and our DACs used for playback, added time blur from the ringing in their filters.

QRONO d2a playback of a CD-level audio master has the time response of conventionally played 96kHz Hi-res file. A 192kHz hi-res file played with QRONO d2a can exceed the time performance of the best analogue systems. The audible improvement, while subtle, is immediate. Removing the time smear reveals textures previously obscured. Micro dynamics are improved, and instruments are clearly delineated, enhancing the stereo soundstage and image. Music flows naturally, adding realism and reducing listener fatigue. Listen for yourself.

References

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