From: info@mqa.co.uk
Attention Cameron
13 April 2021

Cameron, thank you for being in touch with us. We appreciate that. If you decide to include our note in your posts, please use the reply in full. We would like our feedback available to as many of your audience members as possible.

We think some of the confusion described in your note originates in what seems to be a misunderstanding of the philosophy and basis of MQA. Your evaluation seems to look at MQA from a very 'conventional' perspective whereas, by its nature, MQA takes a different frame-of-reference - because that is how we can overcome ‘loose thinking’ in the arena of ‘high-resolution’.

MQA is not a simple codec, it is a system that can be used to make the closest sounding connection between an analogue acoustic source and the sound finally delivered to the listener. Instead, MQA is a system based on human auditory science and modern sampling theory. It is intended to efficiently deliver extraordinary clarity and, for 'sensible' content, routinely sounds closer to the original performance than more old-fashioned and data-hungry digital chains.

Being analogue-to-analogue in conception, it goes beyond straightforward lossless and (when used correctly) embraces the conversion and processing at each end of the chain. As we have said many times, having developed the first practical lossless compression in the early 1990s, the MQA team understands lossless processing and compression. MQA moves beyond that and is addressing embedded shortcomings in the sound of a conventional digital chain, including temporal blur and modulation noise introduced before and after the distribution.

None of that is hidden by us. In fact MQA has been thoroughly evaluated by mastering engineers, Grammy P&E Wing, academics and industry bodies such as JAS and RIAA. We have also published peer-reviewed articles in AES, various Q&A in audio magazines and insights on Bob's blog. (We suggest you study the links given at the end).

We don't understand your frustration about evaluating MQA.

There are literally many millions of songs encoded in MQA that can be evaluated on many hundreds of playback devices. MQA is designed to deliver the sound of the original performance.
Of course, these goals are met most easily when MQA tools are used in the studio, when the mastering engineer or Label are involved in adjusting and final approval of the encode.

To help young artists and small labels get their music encoded in MQA and on to Tidal we recently enabled the service you used. However, that service is limited in flexibility and places conditions and obligations on the user. First, the encoder is fully automatic, which means it will use analysis to set parameters for each song as a whole; second, it is intended strictly for music. This encoder is not configured to deal with content where, for example, the statistics change mid-song, or where the audio does not resemble natural sound. The onus is on the submitter to check the content when it arrives in Tidal and confirm the sound. In this way, we can all be sure that the provider is happy with the Master and that, because of the ‘light’, everyone with an MQA decoder is getting the intended sound.

Before we get to your specific questions, we should explain that your experiment, presumably intended to see what MQA is doing, is flawed. There is no single MQA characteristic.

As described in the 'Hierarchical paper' [1], the model conceives an infinite cascade of kernels based on (but not being) B-splines. The role of the MQA encoder is to identify the content information and to encode it with the absolute minimum of temporal blur consistent with inaudibly low or non-existent artefacts. This allows MQA to capture a signal which is 'natural' and accessible. The encoder then uses 'Origami' and lossless processing within that information, for robust delivery.
MQA decoders unfold the Core signal and, if it is available, the MQA Renderer reconstructs to analogue under instruction from the encoder.

We cannot listen to a file; it has to be converted to analogue. The vast majority of DACs use delta-sigma and/or upsampling architectures, with the actual conversion happening with a small number of bits (1-6) at high rates (in the range 2.8 - 48MHz). Part of the role of the MQA decoder is to prevent the playback system reintroducing blur, while the Renderer is specifically tuned for each DAC to match the analogue output as far as possible.

Going back to the front end, there is a spectrum of MQA encoders, ranging from real-time for Live, to encoders supporting the full white-glove process in which the characteristics of the A/D and mastering suite can be accounted for directly. In between, there are encoders used for music release by a Label with controls and options as mentioned earlier. The encoder is flexible because, unlike a basic codec, the end-end chain of MQA is responsive to the content but on a song-by-song or work-by-work basis. For non-real-time, the encoder performs extensive analysis to both identify the information and optimise the hierarchical chain, but also look for problems in the source.

As you may have seen in the messages returned from the service, the analysis phase of the encoder objected outright to 11 of the 14 files you submitted, with the message: “There was an error in this audio file. The MQA encoder was unable to encode it.”
The three that passed through presented these various warnings in combinations:
"Audio invalid - Excessive alias …"
"Audio invalid - Input audio is predominantly 16 bits while file container is 24 bits
"Input audio appears to have been wrapped",
"Input audio contains a band edge …"
"**Encode may not have worked as desired and may require further QC**"

Moving to some specific points:

The impulse response you show is not 'the impulse response of MQA', we have published that on several occasions. Instead, what you are seeing is the reverse (de-convolution) of the encoder's estimate of the temporal blur in the input data treating it as 'one continuous object' - it might give a hint about the music in which the test signals were buried, but more likely it just got confused.

Noise-floor in the 44.1 and 88.2 kHz files. The encapsulation process and addition of signalling require dither for transparency. Your files had no dither (which is a naive mistake). When the input file is 24b, this dither is subtracted in the decoder and has no impact beyond linearising the channel. With no decoder the dither is visible, and it is deliberately shaped for minimum audibility. The shaper used follows best practice and enables noise-floor below 16b at 4kHz (as we have described before [3] [4]). When, as in your case, the audio was sometimes 16b in a 24b file, all bets are off and the file should not be passed by a label for distribution until that error is corrected.

'Leaky filter'. This has been explained many times and yet is still misunderstood. Your use of “leaky” as a derogatory word is inappropriate. If the ultimate reproduced audio wasn’t “leaky”, it would be brick-wall bandlimited. Such audio will have lost any beneficial characteristics a wider bandwidth, version might have had. Shannon's theorem states that a bandlimited signal can be reconstructed perfectly …, but neither Nyquist nor Shannon state that audio should be bandlimited to 20kHz because we can't hear higher frequency sine waves. The difference is often missed but important.

For user convenience, the MQA Core decoder can (optionally) normalise all signals to 2x (88.2/96kHz). When the audio is 1x, a low-blur upsampler is used. The conventional alternative would be a 'brick-wall'. This process was optimised over many hundreds of hours of listening over a decade and matches the Renderer perfectly. Such constrained upsampling is almost always better than the same step going on inside the DAC. Some MQA decoders don't use it, most do because it gives a seamless playback experience.

Aliasing. The ultrasonic twin-tone exceeded the performance envelope of any normal music and, as mentioned above, the encoder warned the content was suspect. Such ultrasonic levels would be the result of something very wrong upstream This is a clear case of 'garbage in, garbage out.

Distortion: Is non-existent. MQA does not add distortion. Your picture seems to show inaudible high-frequency dither noise (that may result from the mixed statistics but will, in any case, be both benign and inaudible); you will note that the ultrasonic noise is not correlated to the sine-wave.

Finally, a testing exercise does not provide reason for conclusions about business practises. Your specific points are answered here:

1. Every MQA file will tell you the sample-rate of the original audio.
2. MQA makes no restrictions on what labels or services can sell or stream. The choice of what to stream is made by the service.
3. MQA does provide test streams to our licensed decoder partners.

In conclusion, we are grateful for your outreach and we remain happy to answer your questions and comments. Your enthusiasm and desire for understanding is appreciated. The following are some links to reading on MQA.

[1] Stuart, J. Robert; Craven, Peter G., 'A Hierarchical Approach for Audio Archive and Distribution', JAES Volume 67 Issue 5 pp. 258-277; May 2019. Open Access http://www.aes.org/e-lib/browse.cfm?elib=20456

[2] Stuart, J. Robert; Craven, Peter G., 'The Gentle Art of Dithering', JAES Volume 67 Issue 5 pp. 278-299; May 2019. Open Access

http://www.aes.org/e-lib/browse.cfm?elib=20457

[3] MQA Q&A Sidebar 1: https://www.stereophile.com/content/mqa-questions-and-answers-sidebar-1-example-nielsen-2l-120-track-1

[4] MQA Q&A Sidebar 2: https://www.stereophile.com/content/mqa-questions-and-answers-sidebar-2-example-portland-state-chamber-choir

[5] MQA Sidebar 3: https://www.stereophile.com/content/mqa-questions-and-answers-sidebar-3-example-portland-state-amazing-grace-audibility-analysis

[6] Bobtalks: MQA CD https://bobtalks.co.uk/blog/mqaplayback/origami-and-the-last-mile/