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TECHNICAL BRIEF: UPSAMPLING IN ANTHEM STATEMENT PROCESSORS

Background: The Shannon-Nyquist Theorem

Harry Nyquist came up with the idea of a minimum required sampling rate in 1928 and Claude Shannon proved it mathematically in 1949. The theorem states that to avoid distortion when an analog signal is converted to digital form it must be sampled at a rate at least two times higher than the highest frequency being converted. For example, an audio signal with a bandwidth of 20 kHz must be sampled at a rate of at least 40 kHz. This is done to prevent aliasing—copies of the input signal, but at the wrong frequency. A loose analogy is the result of a video camera recording a computer monitor's screen—the image on the screen becomes corrupted with a strobing effect because its frame changes more quickly than the video camera can capture.

The same applies when converting the digital signal back to analog. If the digital signal was created at a sampling rate of 40 kHz, byproducts are generated above 20 kHz upon conversion back to analog. This "garbage" may not be audible but it can cause intermodulation distortion which would be audible. In extreme cases it may be enough to damage power amplifiers and tweeters.

This makes low-pass filters before the ADC and after the DAC necessary, to remove everything above half the sampling frequency. The low-pass filter before the ADC is called an anti-alias filter and the one after the DAC is called a reconstruction filter. Unless these filters operate far enough from the audible range, they can cause audible phase and frequency response errors. So how can the artifacts be removed without affecting the sound?

Upsampling and oversampling

The answer lies in changing the sampling rate to a higher one so the filter can be moved away from the audible range. There are two ways of doing this. When done by the ADC or DAC as an even multiple of the original sampling frequency (e.g. 8x), it is referred to as oversampling. When done elsewhere, and not necessarily as an even multiple of the original, it is referred to as sample rate conversion (SRC) or more commonly, upsampling.

Anthem Statement processors do both. Every channel of digital (PCM, Dolby Digital, DTS) or analog-DSP input is converted to 192 kHz before going to the DAC—the Statement D1 was the first, and for a long time the only, pre/pro on the market to do this! The DACs then oversample by 128x, raising the sample rate to 24.576 MHz. Now we only have to get rid of the garbage above half of 24.576 MHz (12.288 MHz)—very far from the audio range. Easy! We do this very gently with 3rd-order Bessel filters since they have the lowest phase error. The result is frequency response that is down only 3 dB at 100 kHz, and in the audible band remains flat, only 0.2 dB down at 20 kHz. At the same time the garbage which is now above 12.288 MHz is down by at least 120 dB.

Upsampling has another advantage since it uses its own clock—it provides a stage of jitter reduction by attenuating bit-to-bit phase errors that may be present in the input data. If a bit arrives too soon or too late due to jitter, the size of the error can be reduced provided that the upsampler's clock is a decent one.

Point of clarification for SACD

For signal processing such as bass management or room equalization, the 1-bit signal must be converted to PCM (the same applies on the production end for mastering although this isn't usually made clear by its practitioners). SACD's native sampling rate in multibit form is 88.2 kHz. The Anthem Statement processor converts this to 192 kHz.

A word about distortion measurements

When THD+N (total harmonic distortion + noise) is measured in products that contain DACs, an AES17 filter, specified by the Audio Engineering Society, is normally used. This filter restricts measurement bandwidth to 20 kHz so the high amount of "garbage" above this range which would obscure the audio range is removed. However, this high-frequency distortion that's ignored can cause audible intermodulation distortion (IMD results from mixing two or more signal frequencies).

To appreciate the superiority of the DAC stage in Anthem Statement processors we must measure to a higher bandwidth, such as 80 kHz. Through the AES17 filter, THD+N for the legendary AVM 20 is 0.004% (digital input) and without the AES17 filter THD+N within an 80 kHz bandwidth is still very good at 0.1% to 0.2%. In contrast, the Statement's THD+N without the AES17 filter is less than 0.01% up to 80 kHz—10 to 20 times better! This is the result of the upsampling/oversampling design.

Is upsampling this good?

Skeptics may point out that upsampling makes the output worse than the input due to generation of spurious tones. When designed poorly this is true but Fast Fourier Transform (FFT) analysis on the output of Anthem Statement processors shows no such design mistake. In the Statement series it really is that good!