ANTHEM™ THE KEY TO HIGH END AUDIO-VIDEO™ STATEMENT

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TECHNICAL BRIEF ON THE STATEMENT D1 UPSAMPLER DESIGN

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 when an analog signal is being converted to digital form, in order to avoid distortion, 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 (frequency) of at least 40 kHz. This is done to prevent aliasing—copies of the input signal, but at the wrong frequency. It doesn't stop there, however. The same applies when converting a digital signal back to analog. If the digital signal was created at a sampling rate of 40 kHz, by-products are generated above 20 kHz upon conversion back to analog. This garbage itself may not be audible, but it can cause intermodulation distortion which would be audible. In extreme cases, it may be enough to cause damage to power amplifiers and tweeters.

It is necessary, therefore, to put low-pass filters before the ADC and after the DAC in order to remove everything above half the sampling frequency. The filter placed before the ADC is called an anti-alias filter, the one after the DAC is known as a reconstruction filter. Unfortunately, unless these filters operate far enough away from the audible range, they can cause audible phase- and frequency-response errors. So how can we filter all the garbage without affecting the audible range?

Upsampling and Oversampling

The answer lies in changing the sampling rate to a higher one so that the filter can be moved away from the audible range. There are two ways of doing this. When done inside 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. The Statement D1 does both.

What this means for the Statement D1

The D1 has multiple AD1896 sample rate converters which upsample every channel of digital or analog-DSP input up to 192 kHz (including Dolby Digital and DTS—the D1 is the first pre/pro on the market to do this!). The DACs then do further 128x oversampling to raise the final sample rate to 24.576 MHz. Now we only have to get rid of the garbage above one half of 24.576 MHz (12.288 MHz). Easy! We do this with very gentle filters of the 3rd-order Bessel type because they ensure the least amount of 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 Analog-DSP

As mentioned in the previous paragraph, the D1 not only upsamples all digital inputs, it also upsamples all Analog-DSP inputs (2-Ch and 6-Ch). While this is an obvious benefit for analog inputs such as phono, tape, etc., it also provides major benefits for DVD-Audio and SACD. Why?

The DSP provides sound management, which includes: bass management, time alignment (listener position), bass/treble controls, THX® post-processing, audio group delay and 2-channel downmix for multichannel material. After the DSP, the D1 upsamples every channel to 192 kHz, this signal then goes to the D1’s DACs, also operating at 192 kHz, thereby ensuring unparalleled transparency.

Points of clarification for DVD-Audio

- DVD-Audio is available in multichannel (6-ch) at 24/96 or stereo (2-ch) at 24/192.
- In both cases, the D1 allows DSP sound management which is followed by 24/192 upsampling to ensure matchless transparency (Analog-DSP vs Analog-Direct).
- For comparison (or preference) the D1 also provides an Analog-Direct option for DVD-Audio.

Points of clarification for SACD

- For DSP sound management, SACD must be converted to PCM. This applies regardless of whether SACD is connected via analog or digital (future Firewire).
- The D1 then upsamples all channels to 192 kHz, once again providing matchless transparency.
- For comparison (or preference) the D1 also provides an Analog-Direct option for SACD.

The D1 upsamples both DVD-Audio and SACD to 24/192 to ensure singular transparency while also providing full DSP sound management. The AVM 20's 6-channel Analog-DSP is widely considered to be among the most transparent designs available. The D1's upsampler is measurably more transparent: 10-20 times better than the AVM 20, ensuring that Analog-DSP is indistinguishable from Analog-Direct.

A word about distortion measurements

When THD+N is measured in products that contain DACs (including CD players, pre/pros, etc.), an AES17 filter is used. This filter, specified by the Audio Engineering Society (AES), restricts the measurement bandwidth to 20 kHz to better reflect what is going on in the audible range by removing the high amount of artifacts which otherwise obscure this range. However, the very distortion artifacts that are ignored by this measurement can actually cause audible intermodulation distortion. To fully appreciate the superiority of the D1 upsampling/oversampling design, we have to eliminate the use of the AES17 filter and then measure to a higher bandwidth, such as 80 kHz. Using the standard AES17 filter, THD+N for the highly respected AVM 20 is 0.004% (digital input); without the AES17 filter THD+N within an 80 kHz bandwidth is still very good at 0.1% to 0.2%. However, the Statement D1's THD+N without the AES17 filter is less than 0.01% up to 80 kHz—10 to 20 times better! This is a direct result of the upsampling/oversampling design incorporated in the D1.

But wait, there's more …

Upsampling has another inherent advantage—it provides a stage of jitter reduction by attenuating bit-to-bit phase errors that may be present in the input data. This occurs because the original and final sampling rates use two different clocks. If a bit arrives too soon, or too late due to jitter, the size of the error can be reduced through the upsampling process—provided that the upsampler's clock is a decent one. It cannot completely eliminate jitter because, ultimately, the input and output are locked to each other—data cannot pile up at the input like a train wreck, nor is it possible for the output to get ahead of the input.

Could Upsampling really be this good?

Some skeptics have expressed misgivings about upsampling, suggesting that when based on poor design execution, it makes the output worse than the input due to generation of spurious tones. Fast Fourier Transform (FFT) analysis on the Statement D1 shows no such design mistakes. In the case of the D1, upsampling really is that good!