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MARANTZ MUSICAL MASTERING

SA-11S3

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Introduction

Since the start of the era of digital recording of music, technology has been faced with the challenge of maximising the accuracy of reproduction. The initial problem was the small number of bits available for the conversion of analogue signals into digital. In the early ‘80s the first attempts were made to create more audible resolution than the bits actually produced. In this way, the first CD player from Philips and Marantz worked with the 14-Bit-D/A-Converter using a process called noise-shaping and achieved real precision of 16 bits and four-fold oversampling, and thus played a leading role.

In recent years, the issue has in fact been reversed, because current master and media can easily deal with 24 or even 32-bit and provide the necessary digital filtering for playing up to 48 bits per sample. Now the issue centers on reducing the surplus resulting resolution and the ensuing huge bandwidths by hundreds of kilohertz as efficiently as possible to usable frequencies and the common resolutions of 16 (CD) to 24 bits. If the excess bits were shortened to simply remove the excess bits, audible errors would be created and any higher resolution gained would be irretrievably lost.

In the SA-11S3 Marantz has implemented new algorithms for digital signal processing developed in-house for the first time and the result is a surprisingly low loss of resolution. For this purpose, Marantz has utilised new signal processing technology previously exclusively reserved for high-end, professionally equipped mastering studios.

Solution

To maximise audible resolution, three methods are combined: Oversampling, Noise Shaping and Dithering.

Through oversampling intermediate samples between 2 original samples are calculated, thus achieving a multiplication of the sampling rate. To calculate the intermediate samples with sufficient accuracy the resolution of the audio data needs to be increased significantly. In our application we found additional 24 bits adequate giving us a total resolution of 48 bits.

The noise shaping offsets digital noise components through clever filtering from the audible frequency range into the frequency range beyond 20 kHz, which is no longer perceptible.

With dithering, the desired signal is modulated with noise to reduce inaccuracies caused through the forcible conversion of resolutions. The type and distribution of the noise determine the quality of the sonic result. Marantz established the optimal variant through extensive listening tests and trial series.

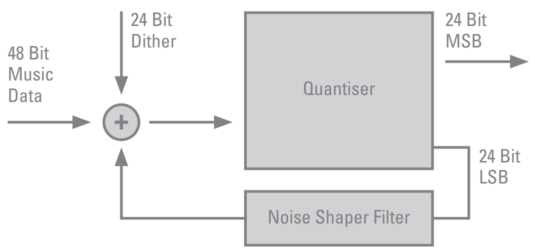
As a result of the correct combination of the three methods and the correct choice of the respective algorithm and its application, more details than the simple conversion of the original signal could be perceived, thus significantly higher-resolution sound reproduction could be achieved.

Oversampling

Unlike many procedures used previously, Marantz deploys no rational numbers for the upsampling of the sampling rate, e.g. 44.1 kHz from a CD to 96kHz (factor 2.1768 ...), as this is often results in a deterioration of the "flow" and the "rhythm" of the playback. In its frequency multiplier Marantz uses exclusively integer factors, and converts the data with a FIR filter (finite impulse response filter) at either to 352.8 or 384 kilohertz. At the same time, this increases the quantisation depth up to 48 bits; depending on the corresponding resolution of the input signal (16 to 24 bits) results in a 2 to 8-fold oversampling.

Noise Shaping

The noise shaping quantifies a mathematical return of rounding errors back to the beginning of the processing of the audio data. The effect of this feedback is similar to the "transfer" in manual, column-by-column addition to or subtraction from two numbers on the paper. This feedback from the rounding error of the quantiser is offset against the next digital word (1 sample with up to 48 bits), added up along with the next random value of the dither and in turn sent to the quantiser. Over time each rounding error of the voltage axis (bits) shifts to a correct value on the time axis (sampling frequency) in this way. The result corresponds more accurately with the original audio signal with the higher resolution and therefore sounds better than simply rounding or truncation the signal. The random element of the dithering signal also prevents the occurrence of noise by the feedback of the transfer to the input, as could occur with the previously mentioned noise shaper procedure without dithering.



The noise shaper filter only acts with the LSB (the least significant bit, the "smallest" bit) and returns it to the input of the quantizer

Dithering

Dithering is, put very simply, the signal mixed with noise. In the reduction from high to low resolution, artifacts which can be perceived as a distortion occur, as the patterns now appear that do not belong to an actual music signal.

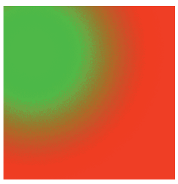
This can be illustrated through graphic sequencing. The colour gradient of the diagram, originally 24-bit, was reduced to 4 bits to illustrate the effect. The problem is transferable between graphics and audio.

Original:
24-Bit graphics

Here is a diagram quantised with 24-bit colour depth. In this analogy it corresponds with an original, high-resolution audio signal. It consists of a similar representation of an analogue image, with smooth, seamless colours.

Linear scaling:
Reduction to 4 bits.

If the graphics to illustrate a lower colour depth are added below, anomalies that do not match the original (24-bit) appear. Even artifacts, new image details will become visible in the form of ring patterns that do not belong to the original image. The same occurs in the downsampling of audio signals. When flattening higher resolutions, interference (distortion) is created that does not belong to the original.



If the reduced image is allocated to a suitable noise, artifacts occurring as a result of the reduction of the original low-resolution image disappear almost completely. With the displaced dither, the image has a significantly greater resemblance to the the original high resolution, despite the lower depth in colour. Something similar takes place In the audio processing. With the right dither displacement, the rounding error is distributed in a statistically favourable manner, the sound is closer to the original.

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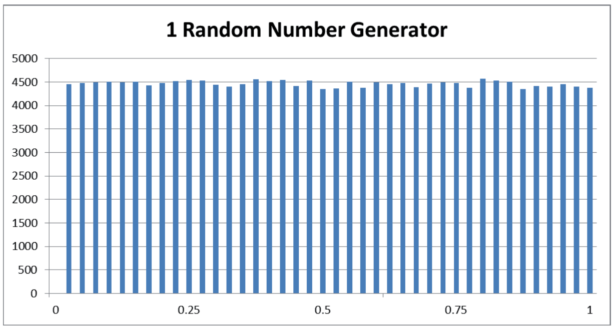
To ensure the generated random signal (noise) remains random from an acoustic point of view, it may not be repeated at audible repetition rates. To achieve a minimum repetition rate of 1 second or higher very long cycle of 180,000 samples are needed. A random number generator was specifically developed for this purpose.

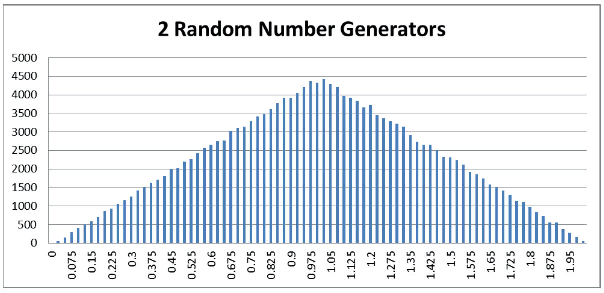
When development was in the initial stage there was a noise signal with a homogeneous uniform distribution of frequency and amplitude, a so-called ‘white noise’. The graph below shows the scaled amplitude statistics of this noise signal.

As can be seen the 180.000 samples show very equal probability between 0 and 1. This means that all amplitudes between 0 and 1 occur equally often.



White noise: a very uniformly distributed random signal.

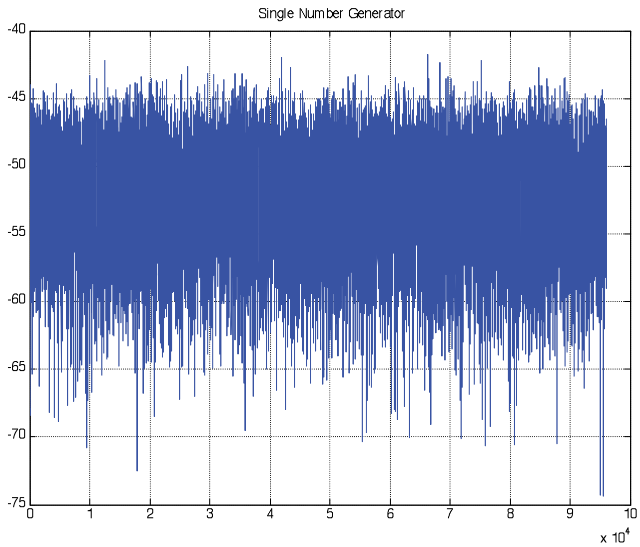
Listening tests and research have shown that this linear probability function can be improved further if 2 such noise generators are used in parallel. Below the scaled amplitude statistics of such a noise generator.



The resulting noise retains that perfect,yet weighted, random distribution

So far we investigated only the amplitude behaviour of the noise generator. The frequency behaviour (e.g. spectrum) of our noise generator looks like this.

It can be seen that all frequencies have about the same amplitude – thus our noise generator really generates ‘white noise’.



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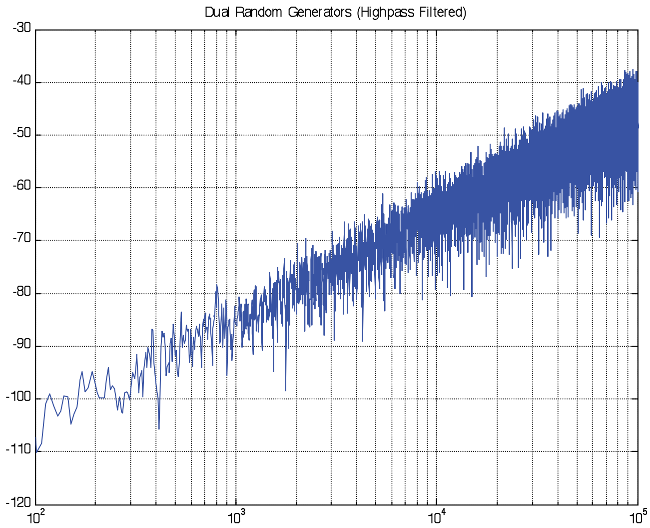
Again several listing tests have been conducted and finally the best sonic performance is achieved with filtered white noise. A high-pass filter led to a significant improvement in sound. Below the plot of the spectrum of the final dither signal.



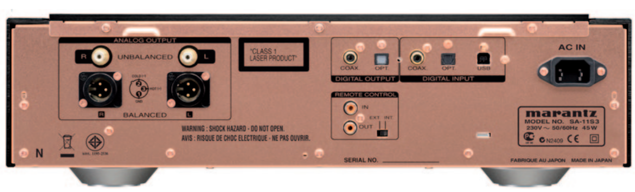
This improves the noise spectrum of the resulting dithered audio signal

Here you can see the distribution of the dither signal, along with its increasing frequency weighted influence on the audio spectrum

The weighted noise of the dither signal developed by Marantz is now stronger at high frequencies than at low frequencies and potential interference components are shifted to the frequency range that is no longer perceptible, beyond the auditory threshold.

Result

This advanced method of signal processing increases and maintains the high resolution of both new and old recordings. The distinct improvement in sound is manifested in detail in individual instruments, space and richer tones.



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