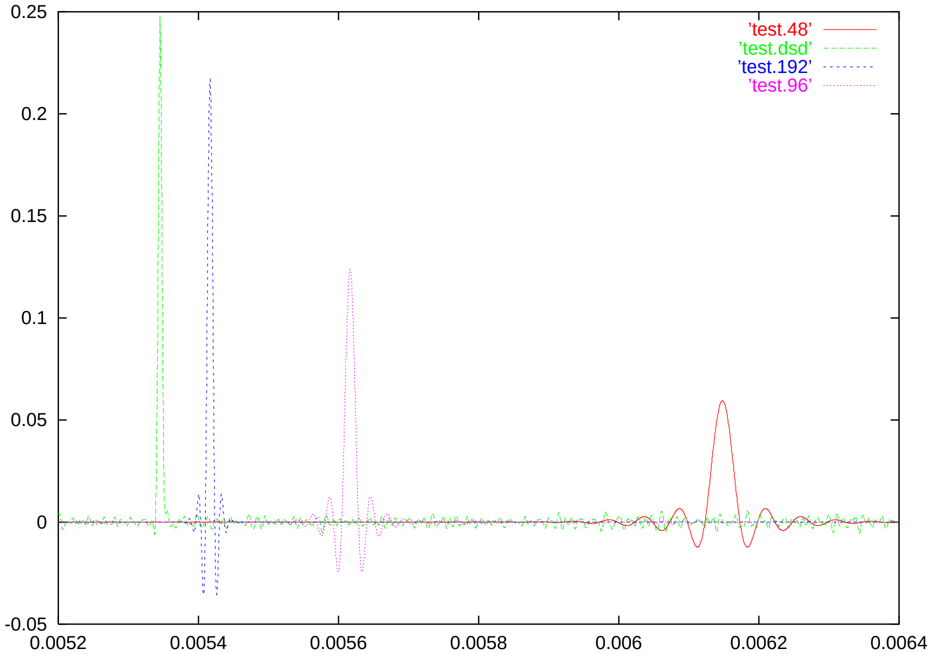

Fig. 9: *Responses (from left to right) of a DSD, a 192 kHz, a 96 kHz and a 48 kHz system on a -6 dB block input ('click') of 3 μs duration.*

accustomed to from CD.

AD AND DA CONVERSION AND OUTLOOK

The original idea for 1-bit coding stems from the observations in the 80-ties and early 90-ties, where the performance of AD convertors improved dramatically with respect to their multi-bit predecessors by using the $\Sigma\Delta$ concept. Indeed, the DA conversion in the feedback path of a SDM is inherently linear, which caused a tremendous increase in overall linearity. Total harmonic distortion ratios of < -90 dB became possible using this technique, concommittant with a dynamic range exceeding 100 dB.

With the increasing demand, however, for better performance, AD/DA manufacturers turned to multi-bit again. In this case not the 14-16 bit flash or folding architectures, but 4-8 bit noise shaping designs, which run at increasingly high speeds; 128 f_S is no exception [19]. The sole reason for returning to multi-bit is in the fact that these designs are less sensitive to clock-jitter; the linearity problems have been effectively solved for these few-bit designs by advanced calibration techniques, or the use of dynamic element matching [4].

From this viewpoint, the best possible scenario is to use the native AD format itself as a pro-audio format: high speed, few bits. After all necessary signal processing, this format is converted to the DSD format, which maintains all necessary psycho-acoustical characteristics such has high bandwidth, filtering with wide transition bands *etc.*, while using the least bits from the disk.

CONCLUSIONS

It is shown that DSD signals can be produced that can be properly dithered. Here, properly dithered means that non-linear artifacts caused by the 2-level quantizer can be removed from the band 0-100 kHz. Moreover, it is argued that the DSD format complies with the minimal requirements set by the human hearing system in order to avoid audible artifacts caused by digitizing the analog signal. For example, the minimal sampling rate needed is about 350 kHz, which is not covered by a 192 kHz PCM recording. Moreover, from 20 kHz and onwards the necessary signal-to-noise ratio becomes increasingly less important, which is in concordance which the natural behaviour of a SDM. Due to the latter feature of DSD, the signal becomes very bit-efficient compared to existing PCM formats closest to the afore mentioned minimal requirements. These statements include lossless coding for both kinds of signals. For example, on SACD a 6-channel DSD recording of 95 minutes can be stored, whereas for 20-bit, 192 kHz PCM only 55 minutes can be stored.

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