DSD Myth

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DSD is hot. True, the concept is so seductive: simply one bit, at a very high sampling rate, producing a signal which is near-analogue. Is DSD the best possible sound quality? Yes and no. There is no simple answer, alas.

The problem is that the format alone does not tell the whole story. Pure, original, 'native' DSD using truly 1-bit dsd converters for A-to-D AND D-to-A can sound stunning. If the converters are done right technically of course (implementation, layout, components, etc), and if the subsequent amplification isn't bothered by the relatively high levels of high-frequency noise specific to DSD.

But most SACD-players and 'DoP' capable DAC's are not equipped with 1-bit DAC chips. And of the recordings on SACD's and DSD internet downloads, the vast majority is not pure, native DSD but has seen many different phases, starting perhaps as 5 bit inside the A-to-D converter, being converted to 24 bit PCM for editing and ending as 1 bit DSD after mastering. Meaning they all have undergone one or more conversions including going back to 1 bit DSD, which is not a lossless process in itself. If you want maximum sound quality once a recording is in a PCM format, you're better off listening to it in PCM than to a flawed DSD transformation of the latter.

This article will explain why.

History

Digital audio is fundamentally Pulse Code Modulation (PCM): each sample being a pulse of a given amplitude, the resolution thereof being determined by the code's word length. We are familiar with 16 bit PCM on CD, 24 bit PCM in computer audio and 1 bit PCM on a SACD disc (the latter is called DSD for marketing reasons). The number of bits determines the noise floor, roughly 6 dB less noise for each extra bit. Proper dithering avoids distortion when truncating (but you need a minimum of 3 bits word length for dither to work properly).

In fig. 1 the general digital audio flow diagram is shown. Assuming the filters fulfil the Nyquist

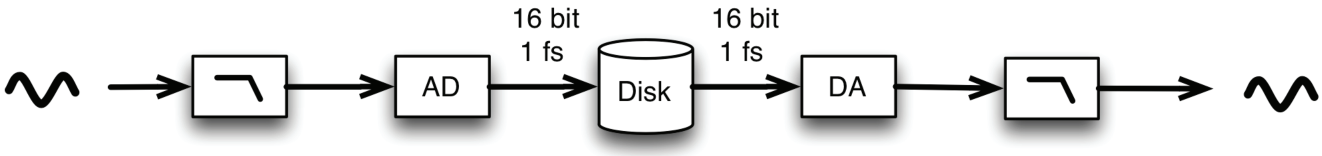

Fig. 1. General digital audio flow diagram

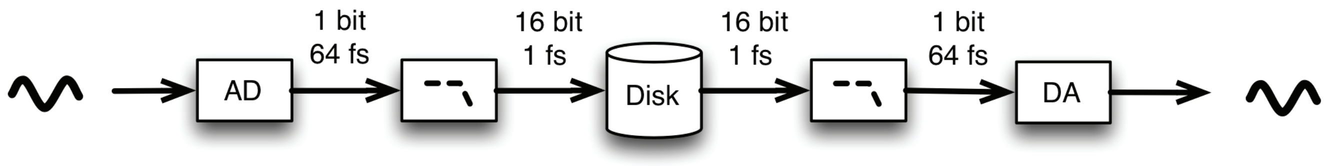

Fig. 2. Digital audio flow diagram with 1 bit converters and digital filtering

criterium (ie. they pass no signal at frequencies of 0.5fs or higher) this signal chain is completely indistinguishable from just a noisy low-pass filter.

The very first 1983 Philips CD player featured an 'oversampling' D-to-A converter, i.e. it had a higher sampling rate than present in the original audio data. The advantage was that this allowed the use of digital 'brickwall' anti-imaging filtering which potentially sounds a lot better than the steep analogue filters Sony originally used. A second advantage was that Philips could use 14 bit converters (16 bit linearity was nearly impossible at the time). The higher noise floor could be shifted to above 20 kHz, rendering it inaudible for human ears. Lateron one realized that this technique could be used to create converters with only a few bits and still higher sampling rates. In the early 1990s it had become standard to use A-to-D and D-to-A converters featuring just 1 bit and a high 2.8 MHz sampling speed. Of course, the CD format itself had remained at 44.1 kHz. In the A-to-D converter a 'decimator' circuitry was used to transform the recorded 1 bit signal into the desired 44.1 kHz 16 bit signal which was recorded to a CD master. Then again, when the CD was played at home the 44.1 kHz signal went through an 'oversampling' or 'modulator' process to 1 bit 2.8 MHz before conversion to analogue. The mentioned digital brickwall filter was an integral part of both the decimator in the A-to-D and the modulator in the D-to-A. By the way, the 2.8 Mhz sample rate is 64 times the CD sample rate of 44.1 kHz, which is written as '64 fs'.

In that same period a discussion had emerged on whether higher sampling rates than 44.1 kHz would sound better, and also on the audible influences of parameter choices in digital filters. Sony Music's archiving department needed a future-proof format when digitizing their analogue tapes. In 1995 they came to the conclusion that a discussion about what sample rate would be enough could be avoided by simply storing the 1 bit signal directly from the A-to-D. Every conceivable consumer format could subsequently be derived, without loss. Implementation was easy as the Crystal A-to-D chip of the time ran at 1 bit 2.8 Mhz internally and happened to have one 'test' pin that carried the 1 bit signal to the outside world. This simple recording/reproduction chain totally avoided the necessity of a decimator in the A-to-D or a modulator in the D-to-A (see fig. 3).

Sony Consumer Electronics soon adopted this new idea as the basis for a successor of the CD. The 1 bit data stream was baptized 'Direct Stream Digital'. Sony joined forces with Philips to develop the Super Audio CD, a new 13 cm disc that used DSD for both stereo as well as surround. A few problems submerged though. By the time the SACD reached the market A-to-D chip technology had already evolved to a higher 128 fs sampling speed and had adopted more than one bit (1.5 to 5 for instance) to attain higher quality. To convert this signal to DSD required extra digital processing, which undermined the original 1 bit signal chain simplicity. Philips and Sony were unable to choose a higher quality format as the new optical disc's capacity didn't allow

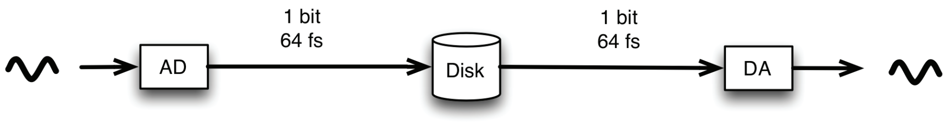

Fig. 3. DSD flow diagram as originally intended

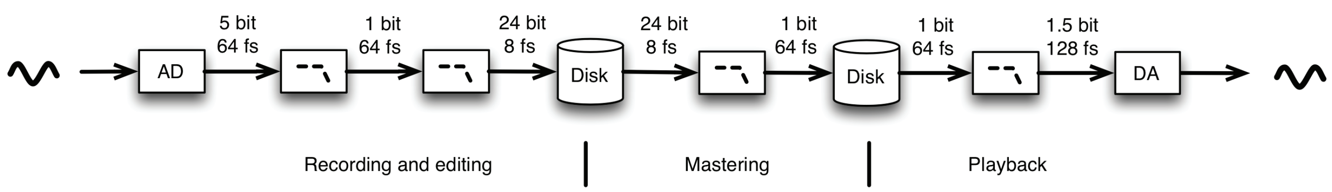

Fig. 4. DSD flow diagram in reality (example)

more than 64 fs data for stereo + surround. But even if it had been possible, then what format to choose as all manufacturers of A-to-D chips had worked out different solutions.

In order to provide the DSD market with a true 1 bit A-to-D converter Grimm Audio introduced its AD1, in 2004. Built with discrete components and featuring a continuous time modulator the AD1 maximizes 64 fs performance to the limit, it still sets a respectable low-distortion benchmark for all other available converters.

Secondly, the recording studios expressed a few comments. The original purpose of DSD, archiving, posed no problem as the digitizing of analogue tapes required no further digital processing. However, at the introduction of SACD it had been standard practice in studios for years to edit and mix digitally, for pop as well as classical music. Any processing of a 1 bit data stream (even a slight adjustment of signal loudness) irrevocably produces multibit data. The only way to produce a (near-)ideal pure DSD master would be to isolate a small portion to be edited for a crossfade from one take to another, convert that to multibit, calculate and apply the crossfade, reconvert it back to DSD and reinsert it in the original data stream. Some editing systems have been developed that offer this function but as far as we know Channel Classics is the only one to have used this approach routinely, in combination with a true 1 bit A-to-D converter. Not going that route would mean that in order to obtain a 'pure' DSD master no editing would be possible; unthinkable, except maybe for remasters of analog tapes. The pragmatic way out has become: make your recording in PCM, edit and mix digitally, then convert to DSD. This is how the vast majority of SACD's has been made. This also allowed to convert the A-to-D output directly to PCM in stead of via DSD, thus avoiding at least one DSD step.

Next question of course was which sample rate would be best for this route. Philips chose 8 fs 24 bit (later baptized 'DXD'), we have good reason to believe that 4 fs and probably even 2 fs can be transparent as well. The choice of the decimation parameters involved is crucial however, and this is what eventually separates the men from the boys. The better the manufacturer the lower the sample rate can be!

A third problem was that already in 1997 D-to-A converters for use in SACD players internally also worked with 1.5 bits or more. So inside the D-to-A chip the 1 bit data stream from the disc had to be converted to an appropriate higher bit format first before it reached the onboard D-to-A converters. Recording the raw internal A-to-D data directly to tape is a noble idea, but as this format seldom matches the reverse process at people's homes this practice is in fact useless and devoid of sense.

To summarize, already at the introduction of SACD had the DSD promise of an 'almost analogue, natural data path' become marketing speak rather than reality. The real data path looked more like the one in fig. 4. Not quite a 'direct stream'.

All this wouldn't be too serious if it had been without consequences for the sound quality. To go from 1.5 or 5 or 24 bits to 1 bit however is no lossless process, alas. 1 bit modulators are plagued by unavoidable 'idle-tones' and vast amounts of noise above the audio band.

Best practice

What then is really sound practice when it comes to high quality audio? The point of departure should be to check how theory works out in practice. The experiments we ran indicated that a 1 bit 64 fs A-to-D-to-A chain can sound totally transparent. So 1 bit 64 fs already enables maximum audio quality. A higher sample rate would therefore not be needed, as it would not add anything. That's good news since it means the consumer SACD format can be transparent.

However, when making a recording, edits and mixing will be necessary. So what to do as a recording professional if you want to keep that transparent? Two good choices are apparent:

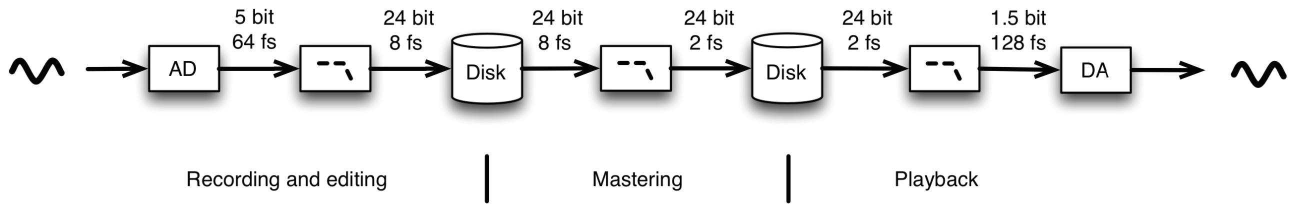

Fig. 5 Preferred digital audio flow diagram (example)

1. Mix everything on an analogue desk, record the result digitally using a native 64 fs DSD converter (like Grimm Audio's AD1). Don't change any levels or other properties of the audio, except some crossfades where necessary, using special software that leaves the rest of the audio untouched (see above). The end result will be 99% pure DSD. Almost no one does that, Channel Classics may well be the only one in the trade. We should admire them for this and encourage them to issue their beautiful recordings in 64 fs for many years to come. And believe it or not, if they were to convert these recordings to 128 fs this would deteriorate the audio quality.

2. Do your recording with any A-to-D converter to your PC in either 2 fs, 4 fs or 8 fs 24 bit format. On the PC do the editing and mixing you need to do. Finally, release the result in 'native' 2, 4 or 8 fs 24 bit. Conversion to 1 bit at 128 or 256 fs is second best. 1 bit 64 fs is third best, still better in bandwidth and noise level than 44.1 kHz 16 bit. And 64 fs will be necessary for SACD, as long as it lasts.

One point remains. Various listening tests have indicated that DSD files sound 'different'. If a 192/24 file is converted to 64 fs DSD and both files are played back through the same converter, some listeners prefer the sound of the DSD file. Two aspects characterize the technical difference between the two files. First, there is some inevitable loss of signal quality after the transfer to DSD. But in view of the listening results the 'damage' probably falls below the hearing threshold or is euphonic. Secondly, the DSD file will have a lot more noise between 20 and 100 kHz (the exact amount depending on the D-to-A used). An example of this can be seen in fig. 6 that shows the spectrum of a -60 dB tone through our AD1. This noise is inaudible to human ears but can have second order effects. For instance, the presence of the noise could influence jitter performance in some converter designs. Some have suggested that this HF noise may change the behaviour of capacitors, cables and electrical contacts. We have not done any research into this ourselves, but it should be relatively simple to set up a listening test featuring a third file: the original 192/24 file but with added HF DSD noise taken from a silent DSD recording. If the ultrasonic noise is responsible, this file should sound 'better' than the original 192/24 without the added noise.

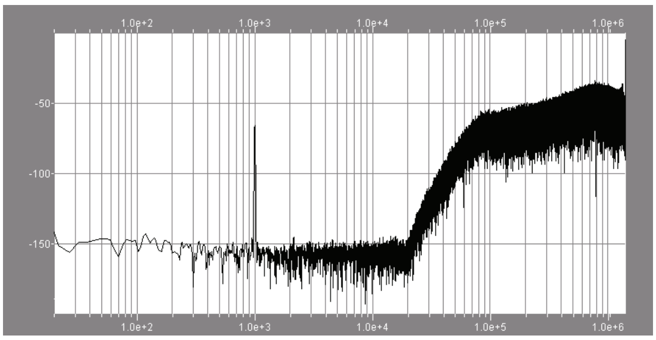

Fig. 6 Output spectrum of the AD1 DSD converter

A final remark. If in our opinion 192/24 files are a better choice than DSD files, then why do we include the 'DoP' DSD format in the USB interface for our LS1? The reason is simple. Until recently there were just two important formats for musical content, CD's with 44.1/16 and SACD's with 64 fs DSD. A lot of wonderful music has been released in these formats and recently the SACD masters are becoming available as online downloads. We want to enable our customers to enjoy that music and therefore we will support every quality format. In our view it is music over format, not the other way around.

Eelco Grimm & Peter van Willenswaard