DSD – the new Addiction

By Andreas Koch

Abstract

A new drug – no, but a new "can't-be-without-it-anymore" audio format bursting into our listening rooms. Direct Stream Digital (DSD) actually has been around for a while already, but it has been so married to a physical media, SACD, that it has yet to receive the attention from audiophiles that it deserves. It is only recently with the growing interest in downloading high resolution audio via the internet that DSD surged to the surface of news coverage. What were compelling reasons to use this encoding scheme for SACD over 10 years ago are now becoming convenient truths for the new era of high resolution internet audio. This paper explains the background and nature of DSD and what might happen to it in the near future.

What is DSD?

The term Direct Stream Digital (DSD) was coined by Sony and Philips when they jointly launched the SACD format. It is nothing else than processed Delta-Sigma modulation first developed by Philips in the 1970's. Its first wide market entry was not until later in the 1980's when it was used as an intermediate format inside A/D and D/A converter chips.

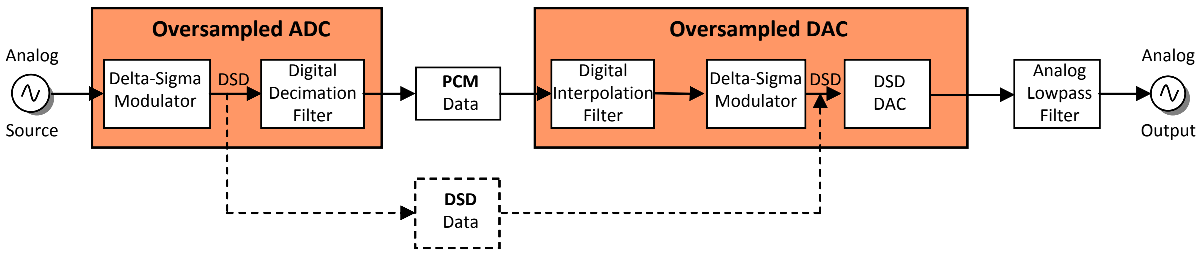

Fig. 1

Figure 1 shows how an analog source is converted to digital PCM through the A/D converter and then back again to analog via the D/A converter. The A/D internally contains 2 distinct processes:

1. Delta-Sigma modulation: the analog signal is converted directly to DSD with a very high sampling rate. Various algorithms are in use depending on the application and required fidelity. They can generate 1-bit DSD or multibit DSD oversampled at 64x or 128x compared to regular CD rate.

2. Decimation filter: the DSD signal from the previous step is downsampled and converted to PCM. Word length is increased (for instance 16 or 24 bits) and sample rate reduced to CD rate or a low multiple of it for high resolution PCM formats.

The D/A process is very similar where:

1. the PCM signal is upconverted to a much higher sample rate.
2. then converted to DSD via the Delta-Sigma modulator (to reduce word length)
3. then converted to analog.

This technology was chosen because of its improved linearity and consistent quality behavior across physical components, as most of the heavy duty signal processing was shifted to the digital domain where it was not susceptible to variability of electronic components. It was quickly adopted in most converter systems and we can say that since about the late 1980's we have been listening to some form of DSD without even knowing it.

As science progressed as well as our experience with digital audio, we started to realize that the algorithms for the DSD-to-PCM and PCM-to-DSD conversions can have a profound impact on the sonic performance when they are developed according to classic formulas. These are relatively complicated algorithms and they introduced a new phenomenon that we describe as "digital sound" or ringing effects. Hence the motivation by the engineering teams of Sony and Philips to remove these steps altogether from the conversions between analog and digital. This simplified DSD path that bypasses the PCM path is shown in Fig. 1 above. As is usually the case most simplifications in the signal path lead to sonic improvements and so it didn't come as a surprise when first listening tests were so astonishing that this format was considered as an archiving format for recording studios. That alone says something about its sonic fidelity. At the time no recording studio was even considering using any PCM format to archive its analog recordings.

The proliferation of the new DVD disc format happened around the same time and its owners immediately raised the question if it was time to use it as replacement for the aging Redbook CD format. The license holders of CD (Sony and Philips) were very concerned about that, of course, and were quick to suggest a competing audio disc that would use the DSD encoding scheme to better address the music industry's requirements. A full fledged format war started raging between SACD and DVD-Audio. Luckily SACD won and through that victory DSD has been used in digital recording studios around the world. Meanwhile a large library of DSD recordings exists today and many of these recordings have only been published as PCM conversions.

While DSD is used at a sample rate of 2.8224MHz (64 x 44.1kHz) mostly for SACD production, recording equipment has also been used at double that rate at 5.6448MHz (128 x 44.1kHz). Often studios use this format to archive their library of analog recordings. Recording equipment for this double rate DSD is available relatively inexpensively at great quality so that consumers can use it to archive their beloved vinyl and tape recordings onto a digital format and then play that back directly via an audiophile grade D/A converter (such as any Playback Designs product) in the comfort of their own listening room.

DSD as a high resolution audio format

The theoretical frequency bandwidth of a DSD signal with a sample rate of 2.8224MHz (64 x 44.1kHz) is 1.4112MHz. Compare this to a 96kHz PCM signal which has a theoretical bandwidth of 48kHz, or 192kHz PCM signal with a bandwidth of 96kHz. However, this wide bandwidth comes at a price: pure Delta-Sigma signals are quantized to 1 bit and, therefore, do not have a great dynamic range by themselves. That is why Delta-Sigma converters need to incorporate a process called “noise shaping” that increases the dynamic range in the usable audio range (0-20kHz) and then slowly decreases it over higher frequencies. It is this noise-shaped delta sigma signal that is then called DSD. Fig.2 below shows the typical dynamic range of a DSD signal sampled at 2.8224MHz and at 5.6448MHz. The slowly rising noise floor at higher frequencies also follows to some degree our hearing threshold for transient signals that have been proven to be audible up to 100kHz.

Of course, DSD at double the rate (5.6448MHz) has an extended audio range of 0-40kHz above where the noise floor then starts to rise gently.

Fig.2 also shows the theoretical dynamic ranges of high resolution PCM signals at various sample rates. Note the steep brickwalls that PCM signals typically have. It is those brickwalls that can generate very audible side-effects such as pre-ringing, if not processed with special algorithms (such as in all Playback Designs products). By design DSD signals do not generate these side effects.

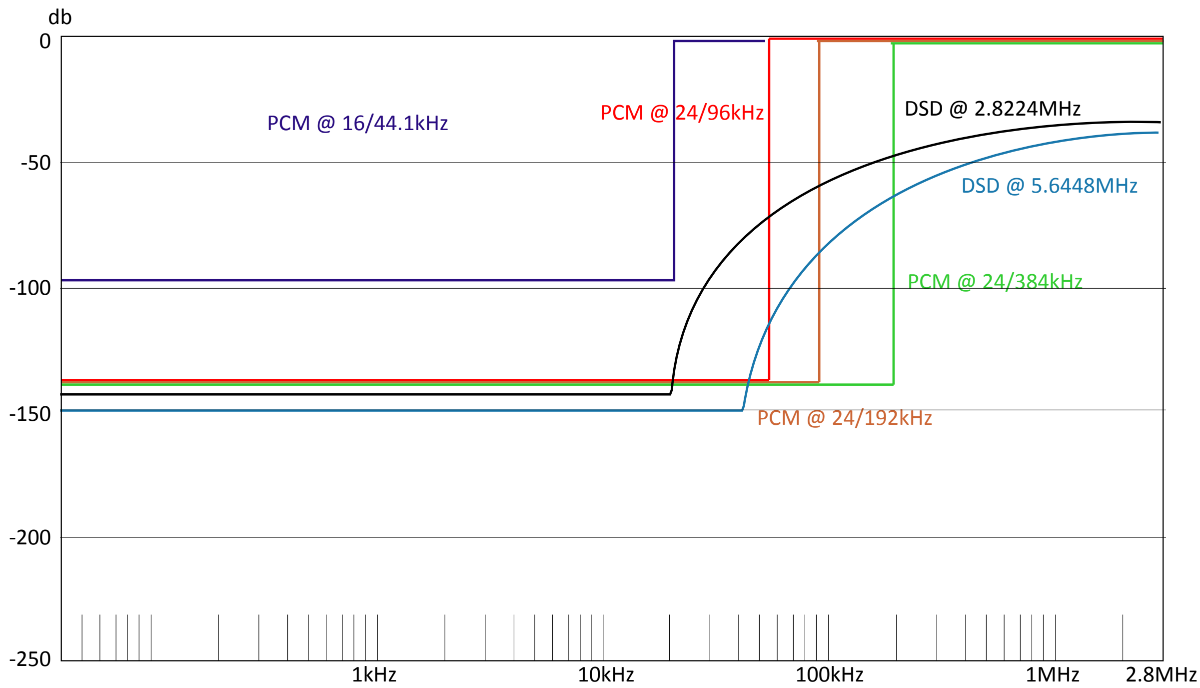

Fig. 2

As we can see from this, DSD is characterized by the following:

- great dynamic range in the audio band (0-20kHz)
- slowly rising noise floor in higher frequencies (no brickwalls)
- extended frequency range into MHz

This makes DSD a serious contender in the choices of high resolution audio formats. Sometimes DSD is criticized for its high frequency content (as shown in Fig.2). But all DACs limit the amount of noise that actually gets through to the analog side. This noise is generally not correlated to the music signal and therefore is easy for our psychoacoustic hearing system to filter out, but most listeners do not even hear it. Double rate DSD addresses this problem by pushing the ramp of the rising noise floor up on the frequency axis by about 20kHz thus reducing the absolute noise floor in the higher frequencies quite dramatically.

Internet Download Bottleneck

Clearly, packaged media is slowly on its way out. It has limited the evolution of audio formats for too long. As the computer platform does not favor one particular format over another, it is becoming an artistic decision for the recording engineer and producer which format to use for which application and market. With fewer and fewer disc pressing plants in the world the era of high resolution downloads to computer has already started. When combined with an external DAC and configured for high quality playback the computer can be a formidable audio platform.

The various high resolution formats that are currently in use can have very different bit rates, which impacts the time it takes to download the files from the internet. Fig. 3 lists different formats along with their files sizes for a 3 minute song and download times assuming a 5Mb/sec internet connection.

Format	File size	Download time
Redbook CD (16/44.1kHz)	32MB	1 min.
PCM 24/88.2kHz	95MB	2.6 min.
PCM 24/96kHz	103MB	2.8 min.
PCM 24/176.4kHz	190MB	5 min.
PCM 24/352.8kHz (DXD)	380MB	10 min.
DSD 2.8224MHz	127MB	3.4 min.

Fig. 3

As the sample rate is increased beyond 96kHz PCM formats create relatively large files that take quite long to download. The file size and download time for DSD on the other side are comparable to 24/96kHz PCM, yet offer greater performance as we saw in Fig.2. Often the DXD format is compared to DSD in terms of sonic quality, but as we see from this table DSD is about 3 times more bit efficient. This comes from the fact that DSD doesn’t have a flat frequency response as seen in Fig.2, but rather adjusts resolution in very high frequencies where our hearing only has very limited resolution as well. For this reason and the fact that a large library of DSD recordings already exists, DSD may become the dominant format for high resolution downloads.

Playing DSD Files

The computer is a very good platform for reading files, shuffling bits from one place to another and it offers various possibilities for user control, local and/or remote. Hardly any other machine can offer the amount of storage, upgradeability, ease of use and controlability via tablet PC or even smart phone, at least not for the price of standard computers today. However, the computer is generally not a good platform to combine with analog audio signals – there are too many fans, disk drives, CPUs, auxiliary processors generating mechanical and electrical noises and running on asynchronous clock signals. This all would have a very negative impact on the analog audio signal, if it was too close to this noisy environment. Fortunately, this can be solved quite easily by moving the critical element, the DAC, outside and away from the computer. This raises the question, of course, of what the best link between computer and external DAC should be. Ideally the requirements for such a link should be:

- no limitation on audio format
- no limitation on sample rate
- single cable to allow the DAC to be clock master (computer does not generate audio clock)
- standard on most common computers
- long cable lengths

The most common audio links that we already know, such as Coax, AES/EBU or Toslink all have some limitations. None of them support DSD and none of them can handle sample frequencies beyond 192kHz or clock master setups without adding a second cable. Fortunately, a few audio companies got together with a couple of large software companies in the late 90's to define a flexible audio link protocol that could be implemented on top of the standard USB interface. This protocol has almost no limitation on format, sample rate or clock setup other than what the transmission rate of the underlying USB protocol allows. It is standard on all computers, even the lowest end versions, and there are numerous USB extenders available that allow cable runs well beyond the basic USB limitation of 15 feet. Seems like a winner and so it is not surprising to see an increasing number of DACs becoming available with the USB interface. If you want the most future proof audio system it seems clear that a standard computer with an external USB DAC should be in your budget.

There are several DSD file standards and they have all their history and reason to be there:

- .dff: introduced by Philips in 2000.
- .wsd: introduced by 1-bit-Audio-Consortium in 2002 composed mostly of Japanese companies.
- .dsf: introduced by Sony in 2005. This format is very similar to .dff, but has more flexibility for including meta data such as graphics for liner art etc. that can be used to display visual information while playing the related song. This format is used on DSD Discs, which are recordable DVD media playable on Sony's Playstation, some PCs and some SACD players. Because of its additional capacity for meta data this format will most likely become predominant.

All 3 formats are currently in use, but most playback software available today for computers support all 3 of them. That is the beauty of a software based platform, it usually takes the software developer only a few trivial tasks to support an additional file format and so we don't have to be too concerned about these multiple file formats. And if it ever should be then Korg's free Audiogate software allows you to convert files from any format to any format. Various manufacturers already offer playback software that supports any PCM sample rate, DSD and double rate DSD.

Once the playback software reads the DSD file it sends the data to the USB driver where it then gets rearranged into containers ready for transmission via USB. Both Windows and Apple OS operating systems have implemented drivers with partial support for the USB Audio specification:

- Windows implements USB Audio very poorly and cannot support more than 24/96kHz PCM. No support for DSD. In other words without a 3rd party driver Windows by itself is useless for high resolution audio. Luckily the professional audio company Steinberg created a high level audio interface driver (ASIO) that not only supports any sample rate for PCM, but also for DSD. It is widely used by many manufacturers that it became a de-facto standard in the professional audio industry. It is also gaining acceptance in the audiophile industry.

- Apple OS has native support for any PCM format. Sadly though it does not support DSD and in its most recent release of OS 10.7 removed a special mode ("integer mode") that before could be used to transmit DSD safely without the danger of any involved software confusing it with PCM. Because Apple creates very closed systems, unlike the PC / Windows platform, the only choice at this moment to play DSD files natively to an external DAC is to stuff the DSD bits into PCM containers so that the operating system thinks it is PCM. It is then up to the software developer and DAC manufacturer to implement enough safe guards that no confusion can ever happen (which could possibly teach your speakers flying lessons if a confusion would ever happen). Several manufacturers have been working as a team to standardize a common method for this so DSD could be played natively via normal PCM paths without any conversion from DSD to PCM and back again. The advantage of this is that no additional driver software is required. The Apple native USB Audio driver would be able to support PCM and DSD at any rate. The Playback Designs products already implement this solution.

The Linux platform is used for music servers also, but unless the user is computer savvy they are not so easy to configure and software and driver support is not as massive as for the Windows and Apple platforms. To my knowledge no playback software and no driver with DSD support is available for this operating system at this time.

Conclusion

While DSD continues being used on every SACD, it may also have an additional new growth life as a separate download format. Its sonic performance makes it competitive with any high resolution PCM format, many listeners would argue it is even superior. Its bit efficiency alone almost guarantees a success in that application. Yesterday any audio format was strongly married to some type of hardware carrier (i.e. vinyl, CD, SACD etc.) and that hindered the evolution of the encoding formats, be that in PCM or DSD. But today we are entering an era where the hardware does not impose the same limitations. It is becoming flexible and upgradeable thanks to software control and computer platforms. This is not only true for storage, processing, simple playback functionality, but also for physical links (i.e. USB) all the way to the place where music is made, in the DAC. When yesterday the encoding format had to adapt to the hardware carrier (packaged media), today the table is being turned: the hardware adapts to the encoding format. In other words today’s computer technology can grow with whatever format we may choose for today or tomorrow. Today it may be a combination of high rate PCM and DSD, tomorrow it may be mostly DSD.