PerfectWave DirectStream DAC

February 27, 2014

Innovation in audio design is not stagnant.

Introducing the PerfectWave DirectStream DAC. Handwritten, discrete, perfection based Delta Sigma Conversion that breaks new ground, distinguishes itself against price-no-object products and provides excellence in every category important to the serious digital Audiophile.

The inspiration
The DirectStream DAC project began in 2002 at one of Sony's early multichannel DSD demos. DirectStream DAC lead designer, Ted Smith[1], was so taken by SACD's sound that day, he began what turned out to be a seven year journey into the science of perfecting DSD.

The History
DSD has been around for years. First rising to consumer consciousness in 1999 with Sony's introduction of the SACD it was short lived. In less than a decade SACD was relegated to the sidelines by the consumer electronic giant. But interest in DSD never died and has been kept alive by a small core of dedicated enthusiasts, recording engineers and labels.

Renewed consumer interest in DSD picked up speed in 2011 and has been growing at a fast clip ever since. It is expected that an increasing number of music labels, including Sony, will be offering DSD versions of their libraries at an accelerated pace through 2014.

A New Approach
Designer Smith has taken a very different approach in the DirectStream DAC: a pure DSD instrument for any digital format including PCM.

The advantages to this approach, relative to the classic PCM multi-bit Sigma Delta architecture, are many including the ability to reveal musical details embedded in digital media that were previously masked by the PCM decoding process.

The DirectStream DAC is built not from dedicated DAC IC's, but from the ground up with hand coded silicon. Smith explains:

"*An obvious problem of using a dedicated DAC chip is they force the digital circuitry to be next to the analog circuitry. This can have serious noise and jitter issues.*

DAC chips have finite resources for DSP processing. With limited power/heat budgets they don't have the physical real-estate for complicated algorithms, forcing compromise in their processing. By building our own device from the ground up we can use *adequate power, real estate, and other resources to do the cleanest possible upsampling, filtering and other DSP duties such as low pass filtering.*

Low pass filtering a noise shaped signal needs some quality capacitors. It's hard to make precision capacitors on a DAC chip and even harder to make accurate high order filters. To use lower order filters with less critical components, DAC chips employ a hybrid approach digitally converting the DSD to a multibit noise shaped signal so there's less analog filtering required, resulting in a loss to DSD's inherent linearity.

Our approach does not use DAC chips, thus allowing us to employ a full width DSD architecture without compromise.

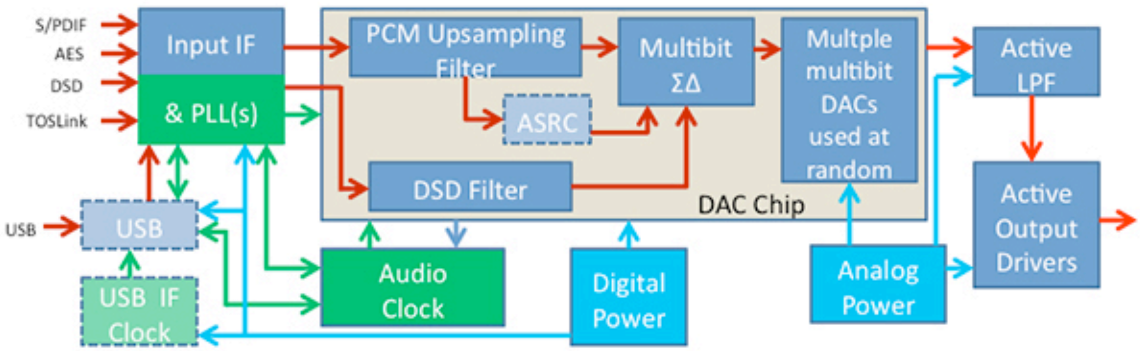
The end result is more information revealed on the digital media than we thought possible, based on classic PCM decoding techniques."

The Next Generation
Building the next generation DAC, capable of a higher level of performance and detail for both PCM and DSD data than that available through a DAC chip, necessitated an approach utilizing an open platform: FPGA (Field Programmable Gate Array). Smith explains:

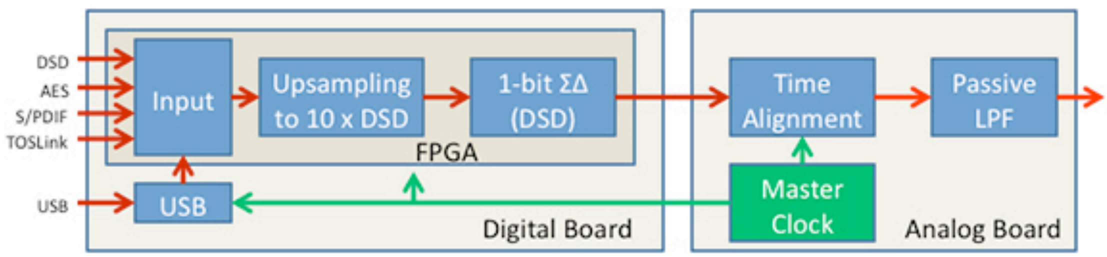
"The choice of a FPGA over a DAC chip was easy. FPGA's allow as much or little full width math to be used as we need. Unlike typical small DAC chips, we're not limited by the heat needed for intensive upsampling and other processing duties. We use more power and physical area to dissipate heat without conflicting with other functions on the board, most of all the analog output.

The Xilinx Spartan 6 FPGA allows complete freedom of processing power and clock choices: our design runs the input processing at 170MHz, the oversampling at 56MHz, and the output at 5.6MHz."

Liberation from a chip and complexity



ESS Sabre Block diagram



PS Audio DirectStream

Why DSD?

The DirectStream DAC is entirely DSD based, even for PCM inputs. DSD was chosen as the core engine for this instrument for a number of compelling reasons:

- **Simplicity**. DSD is simple to convert to analog: just low pass filter it.
- **Linearity**. DSD is inherently linear: it's hard to build a PCM DAC that always takes the same sized step in the output for any possible unit increment of the PCM value because of component matching challenges. DSD doesn't need this level of component matching.
- **Soft clipping**. Like magnetic tape, DSD soft clips when overdriven: signals which exceed the nominal full scale value (by less than, say 3-4dB) only get slightly compressed if at all. With PCM the consequences are flat tops which induce extra energy at the squared off edges, or worse, wrap around which is very audible.
- **Common architecture**. Ironically most reasonably priced PCM players actually use DAC chips that utilize a sigma-delta modulator to get a DSD-like signal anyway. Similarly most A/Ds are sigma-delta based.
- **Immunity to cable and input differences**. Our design handles the PCM conversion from AES/EBU, S/PDIF, TOSLINK, I^2S and USB without recovering a clock, by simply watching for the edges and making decisions

about what they mean in context. The result is that any jitter present on the input clocks is lost entirely in the FPGA. There is no audible difference between TOSLINK and, for example, I^2S because the output clock's rate only depends on the long term average rate of the inputs, not on any clock edge or other local feature.

- **Unamsking of subtle musical details.** Separating the digital and analog stages from each other, operating from a single master clock, 10x DSD processing and a purely passive output stage reveal musical details formerly buried within the digital audio media and masked by classic PCM processing and decoding techniques.

The DSD Engine

The heart of the DirectStream DAC is its DSD engine:

- **10X DSD rate**. Regardless of input format, whether PCM or DSD, all data are upsampled to 30 bits running at 10 times the standard DSD rate and then back down again to double rate DSD for noise shaping.

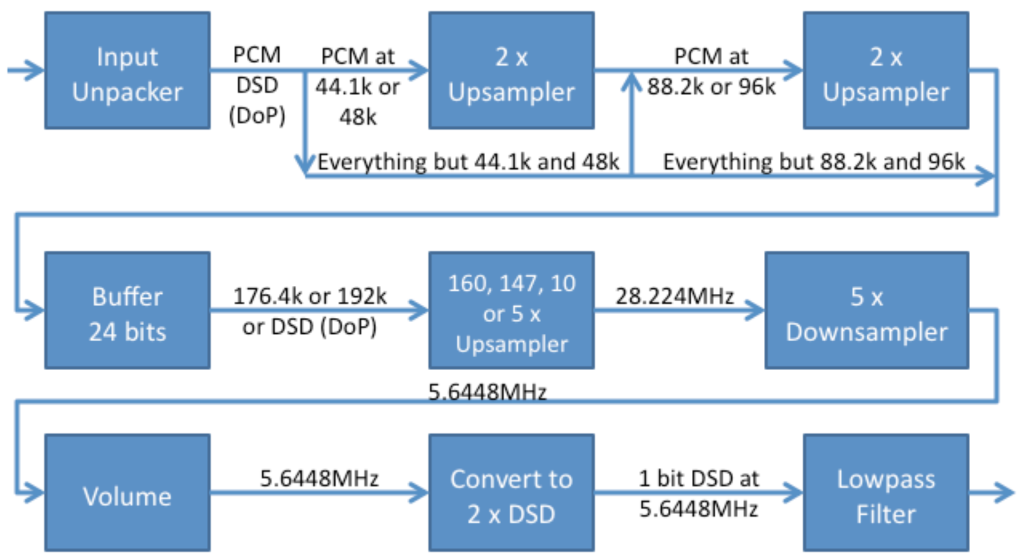
- **No rounding**. The internal volume control keeps complete precision: every bit in the input affects the output of the DAC for any volume level. Except for the sigma-delta modulation process itself there is no rounding, dither or other trimming.

- **30 bit frames**. The design utilizes a minimum width of 24 bits and achives at least 144dB S/N in all of the upsampling filters. Use of full precision in the upsampling filters and significant number of "guard bits" in the IIR filters and the sigma-delta modulator help maintain our goal of perfecting the audio output.

- **Headroom**. Depending on the source material some designs may run out of headroom or approach saturation levels. The PS design opts for at least one extra top bit everywhere in the digital path, coupled with an extra 6 dB of head room in the analog path above and beyond the 6 dB of headroom that SACD needs.

- **Transformer coupled output**. The output of the DSD engine is fed directly into the output stage which is based on high speed video amplifiers, passive filtering, including a carefully crafted high bandwidth audio transformer at the output.

Data flow



System highlights
Overall system design choices that help define the DirectStream DAC's performance:

- **One master clock**. Two clocks would mean more jitter. Even being able to shut one of them completely off while using the other adds stubs and possibly worse, logic or switches between the clocks and the final retiming-flip-flop.

- **10X sample rate**. All input sample rates supported are synchronously upsampled to 10x the standard DSD sample rate and then back down to double rate DSD. There's no need for other clocks to interpret the inputs, regardless of their sample rate, because of the instrument's single clock input sampling architecture.

- **Low noise transfer**. Connections between sections of the design have large impedances to slow down the edges and lessen any noise transfer, in addition to significant power supply isolation.

- **Low intereference**. I^2C, SPI and other control signals run as slowly as they can possibly go, plus control of their transition times limits the amount of induced noise (and hence possible jitter) into the main digital processing area.

- **Balanced architecture**. Balanced signals are used throughout (when practical) reducing radiation and noise in the ground and power rails.

- **Non-saturated logic**. Is used throughout much of the device to improve the predictability of the transition timing. Coming out of saturation is an undesireable statistical process that introduces noise and jitter.

- **Careful terminations**. High rate signals (and/or signals with fast edges) are isolated from control signals and especially each other. If they have to be fast they are terminated appropriately to help keep jitter low.

- **Low temperature coefficient parts**. 0.1% precision thin film low temperature coefficient 1/8W or 1/4W resistors are incorporated everywhere in the audio path, lowering the temperature coefficient of the components below standard practices.

- **Resonance matching**. Low noise techniques are employed, such as the liberal use of low inductance capacitor bypassing, each of which has a self resonance frequency at the main clock rate to keep as much noise from ever getting into the voltage rails.

- **Hand routing**. Every trace on the PC board is hand routed: even for the digital sections. No autorouting is employed, so that each trace, each critical path, is calculated and designed for lowest noise, jitter and isolation.

In Conclusion
The PerfectWave DirectStream DAC represents a significant departure in the design and execution of PS Audio products. We believe this new instrument will help further our industry and the faithful reproduction of music around the world, at prices available to many.

Public Release
The PerfectWave DirectStream DAC will be announced to the public March 1, 2014 and shipping end of April, 2014.

MSRP
$5995 in the United States. PS will provide an upgrade path for all PerfectWave DAC MKI and MKII owners to the new DirectStream standard. Price of the upgrade to be determined. Upgrades will be sold through distribution.

[1] **Ted Smith Biography**
Ted Smith is the lead designer of the PerfectWave DirectStream DAC. While the product is a team effort requiring many disciplines to produce a finished product, this device is principally the work of Designer Smith, who labored over every aspect of the code and hardware for nearly a decade.

- Studied EE/CS at MIT

- Wang Labs: systems programming for the OIS (Wang Word Processor)

- Cadnetix: EE CAD workstations: Wrote file system, kernel, virtual memory, database, and (among other things) the grid generator for the gridless router.

- WaveFrame: Digital Audio Workstations: wrote database, file sysem, DSP code for hard disk based sound editing, led systems software group.
- DA: an independant three person partnership to cost reduce WaveFrames hardware by porting to run in PC, including Windows driver to implement real time portions
- At DA founded two companies:
    - PeakAudio: sound reinforcement systems: first project was the US Senate's sound system.
    - AudioLogic: A DSP based hearing aid company: wrote UI fitting software and DSP code for bringing up various DSP chips.
- Microsoft: various groups from both product development and research.
- Music from grade school on: piano, clarinet, trombone, choirs and later an audiophile.

Experience and training to work with Music performance, being an audiophile, working with hardware, software, embedded systems, DSP, CAD/CAM systems all lead to this project.

For Further Information

Contact PS Audio, 4826 Sterling Drive, Boulder Colorado, 80301.
or your local distributor - Aspera Audio (info@aspera-audio.com)