INDUSTRY VOICE

Q&A; with Thorsten Loesch of AMR/iFi

By Michael Lavorgna • Posted: Apr 7, 2014

Thorsten Loesch

Most readers are familiar with iFi Audio whose affordable products include DACs, portable headphone amps, a USB power supply, Tube Buffer/Pre-Amplifier, and more. We've reviewed a number of their products including the iFi iDAC (see review), the iUSBPower (see review), and the iTube Active Tube Buffer/Preamp (see review). Abbington Music Research (AMR) also produces a full line of "Reference Class" products including amplifiers, preamplifiers, loudspeakers, disc players, DACs, and more. The Chief Designer behind both product lines, Thorsten Loesch, was kind enough to participate in this Q&A; with AudioStream where we dig into some of the nitty gritty behind PCM versus DSD, human hearing, the joy of vacuum tubes, and more. I'd like to thank Thorsten for the time and effort he put into his response and I hope you enjoy it as much as I did.

Can you give us some background related to how you got into designing hi-fi gear?
In a way I have designed and built HiFi gear since I was 11, which is really quite forever. I started meddling with audio gear even earlier in my life. I was always fascinated with music, from an early age.

When I grew up in the former East Germany we owned a lovely old big "Dampfradio" (lit. Steam Radio) – a radio using Tubes) named Stradivari 3 hulking on a commode in a corner. The whole room was awash with great sound. I was drawn to the full, warm and dynamic sound of the "Dampfradio", I was enticed by the exotic place names on the big dial, Brazzaville, London, Moskau, New York, Peking, Timbuktu, Yerewan and so many others. Over time I would get to visit most of the places that were on the dial of this old radio, but that is another story.

East Germany made Stradivari 3 Tube Radio, made by Stern Radio Rochlitz

We also had many LP's of great music, from Satchmo to Aida. But the record player was a small and badly-made solid-state unit with tiny, "tinny" speakers built into the top cover. I did not like the "tinny" sound from the record player. I had just started my 2nd year (i.e. primary school) in school and now had a library card for the Junior Library. We could borrow LP's from a small but well cared for selection of music which broadened my musical horizons. All I knew was that when I did play the records, I just did not like the sound of the record player. This kept bothering me.

East Germany made Ziphona Solid 523 Record Player

I wondered if there was any way to play the music from the record player through the "Dampfradio". There was no simple way, so I asked an uncle of mine, who was an electronic engineer. He helped me modify the radio with a connector that could take a line input from the record player and helped make up a long cable to carry the signal. Now every afternoon I would run home from school, listen to our own records and play new ones borrowed from the library. I was giddy. I discovered The Kinks, Jethro Tull, Pink Floyd and the Beatles as well as Händel and Bach this way.

I was also intrigued with this technology that could make music sound so great or so bad. I started reading up and teaching myself electronics. The bug had bit and it was bad. Less than 5 years later I entered my own home made HiFi system at the school science fair. It included a radio tuner with digital frequency readout, a 100W amplifier with a 6-band equaliser and 3-way speakers with an 8" woofer, 2" midrange dome and 1" tweeter. Little did I know I had made my very first "audio product" at the grand old age of 12.

At the time the domestic HiFi industry in our country (East Germany) produced nothing that could be seen as a match, no 3-Way speakers, 100W amplifiers and equalisers were limited to professional equipment and all radio's had old-fashioned string and pointer tuning systems. What I had built we only knew from western magazines ordered on loan from the central library in Berlin, and from visits to the "Intershop" where for Dollars or Deutschmark (neither of which we had) you could buy the latest gear from Japan.

Of course, the finish was a bit rough, aluminium front panels cut, drilled and brushed by hand, labelled with dry transfer lettering and mostly wood chassis, still, it worked and sounded pretty good. I had spent much time in class doodling circuits and front panel layouts for this system. Sadly no photos survive of this system.

I won first prize which was nice and totally bowled me over. Even today, when our designs win awards, I still get that same elation.

"To be allowed to charge money for my services as a sound engineer, I had to complete a part-time course (totalling nearly 2 years) after which I was able to call myself "Tonmeister" ("Soundmaster")."

Later I studied electronics, worked for a time in a small private pro audio (PA) company (rare in the communist system of the time) that made mixing desks for the state-controlled Radio & TV network. I also got involved with bands as a sound engineer, and built PA systems. To be allowed to charge money for my services as a sound engineer, I had to complete a part-time course (totalling nearly 2 years) after which I was able to call myself "Tonmeister" ("Soundmaster").

Part of that course required me to make classical recordings, even though I was by far, more into hard rock. Such recordings were single-take, minimally-miked, no second chances, no "we will fix it in the mix". One had to be able to read the score, one had to attend the rehearsals and learn how to gain-ride on the old analogue consoles.

The combination of both recording and live sound work and the pro audio designs I worked with, much as it was very "workman like", with gigs on many a night, it helped to lay a real world foundation I still heavily draw upon to this day when designing Ultra-Fidelity components. Eventually I emigrated to the UK, got a 2nd degree in computer sciences and started a career in financial computer systems, but always kept audio and electronics as an obsession.

Through Joe Roberts "Sound Practices" magazine and E-Mail list in the very early days of the Internet, my love of tube sound was revived (I never lost the fascination with LP's). I got quite involved with the "Ultra-Fidelity audio" collective. Being European and German I was also exposed to the French scene, and grew up around the Swiss and (West) German scene. The Japanese journals MJ and Radio Guitsu provided significant influences. Having a solid electronics engineering background and experience sure helped.

Ultra-Fidelity Audio Publications of the 1990's

I wrote occasional reviews and technical articles in the early pioneering days of Audio Webzine's, like TNT-Audio, Enjoythemusic.com and "VALVE the magazine of astounding sound". When others started to commercialise designs I had released as in effect "open source" DIY Kit's into finished products, later I was drawn into the commercial side of High-End audio, where I and my partners started AMR/iFi.

You are responsible for designing gear for AMR and iFi which exist at opposite ends of the price spectrum. Can you talk about the challenges that each approach presents?
At AMR/iFi we have a substantial and strong team that works on all aspects of the design. While my role is substantial, I am just one of several who contribute to the final design.

Challenges both differ and are similar. The overriding challenge is to deliver the best possible sound quality possible for a given product and is the same for iFi as it is AMR. What differ are the constraints.

With AMR products we are largely free to design in technologies or exotic parts and even purely old stock parts that are no longer manufactured with scant regard for cost or time to implementation. New Old Stock DAC Chip's not made for over a decade or two, no sweat. New Old Stock tubes over 50 years old – sure, put them in. Take a year to perfect a small aspect of the circuitry? No, make that two years, well if that is what it takes, then that is what we do.

With iFi we must live with time and cost constraints; use readily available technology and we do not have years to perfect a single-design, but need to be able to release new products much quicker.

> "To many, the manufacturers' applications notes are the golden standards; to us, they are merely the minimum performance standards and only serve as starting points for the real designs."

Yet in both cases we can and do employ unusual solutions. We do not limit ourselves to "cookie cutter" style from manufacturers' applications notes. To many, the manufacturers' applications notes are the golden standards; to us, they are merely the minimum performance standards and only serve as starting points for the real designs. Designing and voicing extreme high-end products gives us a sure and steady hand in selecting and voicing what goes into iFi products.

In the end, iFi products have very much the same "High End" DNA and are designed and assembled with as much care as AMR's "High-End" gear, cost limitations simply means that iFi misses the final degree of refinement that extreme high-end products offer, yet the fundamental performance is as high as we can make it.

AMR CD-77 (with old stock TDA-1541A DAC and old stock Mullard Tubes) left, iFi iDSD nano and iCAN nano (using the latest tech) right

iFi offers the iFi nano iDSD which, "plays ALL high-resolution formats: PCM/DSD/DXD natively." Could you explain what "natively" means in this context since there is some confusion especially surrounding DSD playback and what actually constitutes "native DSD".

PCM and DSD are radically different formats. This becomes clear if we observe the raw digital output of the digitised waveforms for PCM and DSD (Delta Sigma, Bitstream and DSD are in effect different trade names for what is fundamentally the same process).

PCM vs. DSD – Wikipedia.org

Each format has different strengths and weaknesses. For some more background and history, please refer to the separate article (see the Addendum). In short a key issue is that, whenever we convert from one format to another, unless we are living in a perfect world, one can never create a perfect copy in the new format, losses are unavoidable. And worse, in the conversion process we tend to (i) remove whatever makes one format exceptional, and at the same (ii) impress on it the limitations of the other format.

If we convert from 24-Bit at 352.8kHz (DXD-PCM) to 1-bit at 2.822MHz (DSD) – we need to throw away around 99.96% of the amplitude information the PCM format is capable of, while we are only having 12.5% of the time domain information that the DSD system is capable of. If we convert to DSD from DXD, that is 1-bit at 2.822MHz to 24-Bit at 352.8kHz – we need to throw away 87.5% of the time domain information of DSD, though we can theoretically remap all of this into the amplitude domain. So in effect we get the worst of both formats, rather than the best of one.

Modern ADC/DAC parts, such as those from Analogue Devices (AD), Asahi Kasei Microdevices (AK), Cirrus Logic (CS), ESS (ES), Texas Instruments (PCM) and Wolfson Micro (WM) are generally developed for and targeted at a PCM dominated market. In other words, because in the industrial, recording/editing/mastering/release all happen in PCM, so the ADCs generally output PCM and the DACs expect PCM input and they tend to be well-optimised for this operation. In spite of this, inside each DAC, they use a variation of Delta Sigma (i.e. 1-bit with another name) as the underlying conversion mechanism.

Currently manufactured DACs commonly have a complete PCM audio path with digital filtering and a digital volume control integrated in the DAC Chip. Digital filters and digital domain volume controls ONLY work for PCM. DSD is often added as a mere afterthought and in order to provide “buzzword compliance” as required by the marketing department. Any DAC that does not convert DSD to PCM first cannot have a digital volume control and it cannot have a digital filter for DSD.

"Any DAC that does not convert DSD to PCM first cannot have a digital volume control and it cannot have a digital filter for DSD."

In order to allow the use of these features, DSD is first converted to PCM then filtered digitally (adding all the problems of converting PCM to the DSD data stream AND of digital filters) and finally converted into Multi-bit Delta Sigma. So we have double the undesirable conversion at the heart of the Black Box we call the DAC Chip.

Modern DAC chipsets* – focused on PCM, but not on DSD
*Modern ADC/DAC parts, such as those from Analogue Devices (AD), Asahi Kasei Microdevices (AK), Cirrus Logic (CS), ESS (ES), Texas Instruments (PCM) and Wolfson Micro (WM)

Converting DSD first to PCM, processing it as PCM and then playing it back as Multi-bit Delta Sigma (Multi-bit Delta Sigma is still 1 bit technology, just many of them running in parallel. This is very different from running a true multi-bit DAC), is in fact no different from turning DSD immediately into PCM and then releasing it as PCM, yet this is what happens in a lot of so-called "DSD DACs" . One part from Wolfson Micro offers such an option to bypass the DSD > PCM conversion, digital filter and digital volume and to convert DSD directly, however in this case a completely different analogue stage is needed that is optimised for DSD and incorporates the required steep 50kHz low pass filter. Up to today I am unaware of any DSD DAC that implements this DAC chip in "Direct DSD" mode.

> "So usually any sonic differences we hear with modern converters between PCM and DSD releases tell us strictly speaking nothing about the relative merits of each format and everything about the conversion algorithms."

So usually any sonic differences we hear with modern converters between PCM and DSD releases tell us strictly speaking nothing about the relative merits of each format and everything about the conversion algorithms. Losses in sound quality compared to the original untouched DSD or PCM stream are certain.

Ideally we play PCM back as PCM, with a true Multi-bit DAC (no matter what the original ADC source is – we invariably save one stage of manipulation and losses). And we play back DSD as pure Delta Sigma, with no manipulation in the digital domain at all (no matter what the original ADC source is – we invariably save one stage of manipulation and losses). This is what we call "native" playback. DSD remains DSD and is converted directly to analogue. PCM remains PCM and is converted directly to analogue.

In the iDSD nano (and the whole upcoming iDSD range) we go to great length to provide that. Finding a readily available DAC chip that treats both DSD and PCM fairly was a challenge. Manufacturers generally are quite mum about what goes on inside their chipsets, so often you have to actually test the part in detail to figure out what is really going on.

The DAC chip we use in the iDSD nano offers a rather unusual way to handle things. It uses a 6-bit true Multi-bit DAC for the upper 6-bits of PCM Audio and delivers the warmth and slam Burr Brown Multi-bit DAC's are so famous for. Any bits below this are converted with a low order 256 speed Delta Sigma modulator (in effect DSD256), giving PCM playback the smoothness Delta Sigma DAC's and DSD are famed for.

The Burr-Brown True Native DSD/PCM chipset – handling PCM and DSD natively

When playing DSD the same Delta Sigma Modulator is used as directly to convert the DSD bitstream to analog. Of course, there is no digital filtering available for DSD and no digital volume control, so we have to add these features in the analogue domain, where they arguably should belong.

Do you see any benefits to native DSD over PCM?
This question needs to be considered in the context of the whole recording/playback chain. Rather than just the playback chain which is often the case.

I view DSD (and sadly native PCM) as a historical format, much like LP or CD... Either native PCM recording/playback or native DSD recording/playback will deliver the maximum quality of each format. But in the real world, this is seldom the case.

As in our modern times native DSD ADC/DAC's are rare and native PCM ADC/DAC's are even rarer, comparisons become pointless, especially if the ADC/DAC's used are "hybrid" types.

We are not even comparing apples to oranges any longer, but rather apples and oranges processed to freeze-dried juice powder and mixed with water and sugar and the precise processes used at each step. Freshly-squeezed orange juice compared to orange juice from concentrate – there is no comparison.

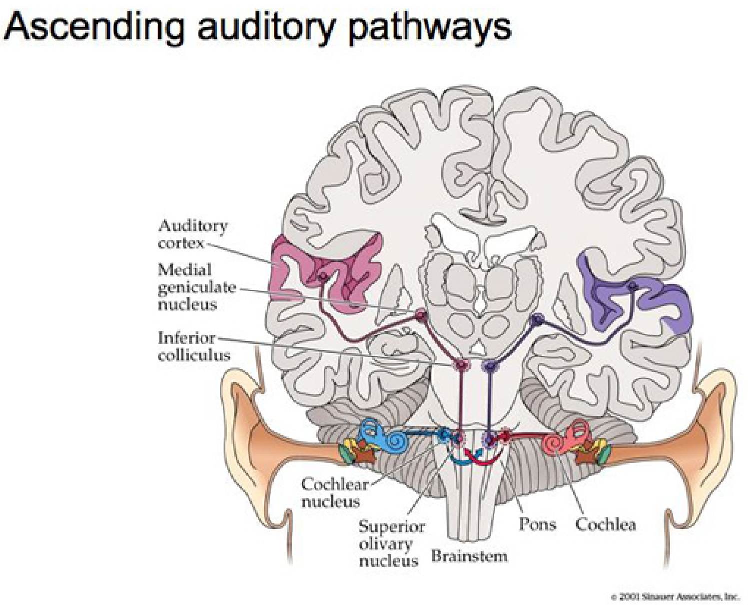
Certainly native DSD has a massive advantage over DSD converted to something else (PCM, Hybrid PCM) and native PCM has a similar advantage over PCM converted to DSD.

There is also some debate over the benefits of higher PCM sampling rates. Some claim, Monty Montgomery being one example (http://xiph.org/~xiphmont/demo/neil-young.html), that 192kHz is actually a

(http://xiph.org/~xiphmont/demo/neil-young.html), that 192kHz is actually a step down in sound quality from lower sample rates. What is your position on higher sample rates including 192kHz and DXD?

Well, Mr. Montgomery has a certain point, insofar that the human hearing has limitations. He may not be quite so accurate as to the actual limits of what we can hear and perceive.

The human hearing mechanism is a marvel. It uses entirely digital "transducers" (hair cells) coupled with an incredibly non-linear acoustic system (the ear canal, diaphragm, attached bones sinews etc. It even amplifies tiny sounds using positive feedback, which if it goes off track is one of the causes of tinnitus. Sometimes the ear oscillates at such SPL's that a person standing next to a sufferer can hear the ringing! And then the digital signal obtained is processed with what amounts to an analogue computer (the brain) with a substantial learned response to sound.



The human auditory system illustrated

If the human hearing was an (electro) mechanical sound recording and analysis system it would be considered as broken by design and completely useless – yet at the same time it endows us humans with the facilities for some exceptional feats of acoustic analysis (we casually call it "hearing"). In fact, we have so far not produced a viable mechanical hearing prosthesis that can be "jacked" into the nervous system, so we really do not understand the human auditory system sufficiently to replicate it mechanically.

Indeed, the human (and to a degree animal) hearing may serve equally well as argument for and against "intelligent design". Usually nature evolves the simplest possible solution to a given problem. Extreme elaboration is extremely rare. So the extreme complexity and anti-simplicity of the human hearing could only been elaborated by an intelligent designer. Yet equally only an utter madman would design such a Rube Goldberg'ish contraption as the human hearing to equip a being with an acoustical sense, so it must have been the blind force of evolution.

Leaving metaphysics aside, we have evidence for example for the perception of ultrasonic content in music in the research of Oohashi et al. Lee/Geddes, J.J. Johnston and many others continually push the boundaries of our knowledge what and how we hear. Much of the cutting-edge research suggest that we both over-estimate and underestimate the human hearings discrimination in all domains and the commonly accepted limits be it in frequency or level are not particularly accurate. So much work still needs to be done before we can have confidence in asserting what can be heard and what cannot be heard.

"So much work still needs to be done before we can have confidence in asserting what can be heard and what cannot be heard."

If we look strictly at the electrical signal, it is easy to see that higher sample rates and greater word-length improves the resemblance of the recorded electrical signal to the acoustic original. Coupled with suitable electronics and loudspeakers or headphones we can certainly claim that we can create a sound field that more closely resembles that present at the original acoustic event with higher sample rates and greater wordlength.

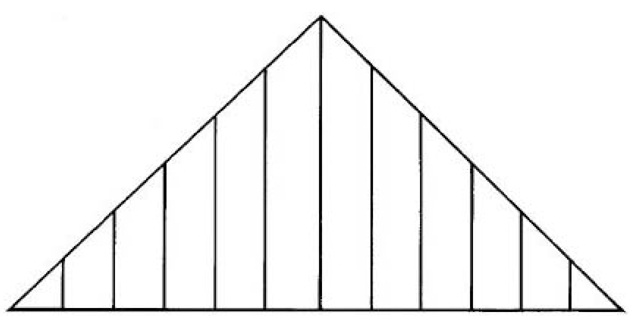
Until we have a reliable working model of the human hearing system (which means that we no longer need amplifiers, speakers, headphones etc., but simply can "jack into the nervous system" instead) the smart money rides on maximising the resemblance to the original acoustic event and thus the sample rate and word length, especially as it is not that difficult to achieve any longer.

The AMR Digital Processor-777 employs "Zero Jitter Mode" to "...virtually eliminate jitter coming from the source". Can you talk about what jitter is and why we need to eliminate it?
Jitter is quite well-explained, but perhaps the concept is still obscure.

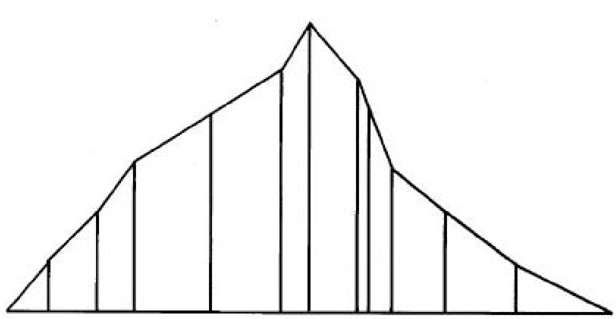
Ignoring DSD for the moment, digital audio records signals as an absolute value at a given precision. As long as we can restore this value as an analogue signal, we will match the original signal as close as allowed within the precision of the digital system.

This is true however ONLY if each sample is obtained during the AD conversion with PERFECT periodicity and then replayed with equally PERFECT periodicity.

If there is any variation in the timing, that is if there is any variation in the WHEN either the AD Conversion or the DA Conversion happens, then distortion will result. An oft found illustration is shown here:



A) Analog signal reconstructed correctly with jitter-free clock.



B) Analog signal reconstructed with jittered clock. (Exaggerated for clarity).

The impact of Jitter illustrated (from Stereophile)

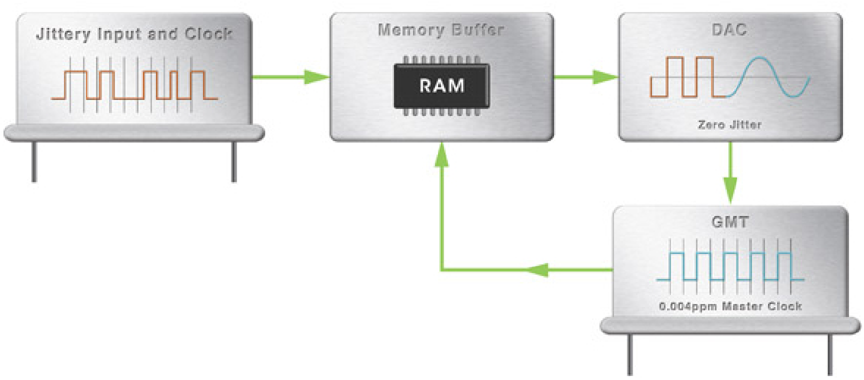
The need to minimise such a kind of distortion is obvious. The methods of how to and the origins of the timing variations are less so.

In the DP-777 we create a completely new, very low jitter clock that only tracks the very long-term average of the incoming clock. It does not use a PLL or similar control loop to control the clock, instead the control is based on non-linear fuzzy logic, which keeps the clock absolutely fixed, unless the control system decides the clock frequency must be changed to keep the memory buffer operating correctly.

> "Femto Clocks by the way, do have quite low phase-noise/jitter. But actually only exhibit Femto-seconds of jitter above 12kHz, meaning they only come into their own towards the top of the audible range."

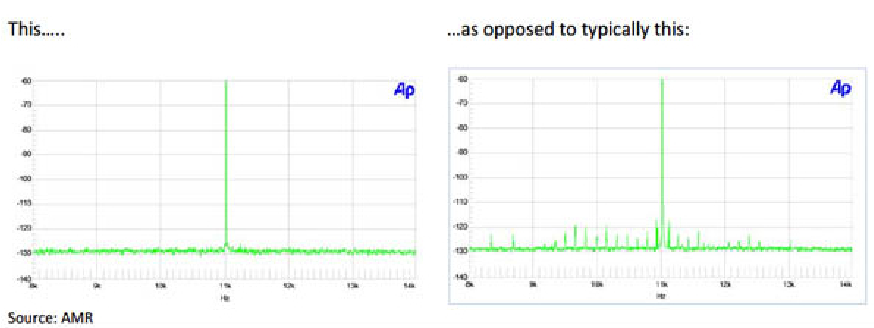
This clock, which we call Global Master Timing, has similar levels of jitter/phase noise as the current fad, so-called Femto Clocks. Femto Clocks by the way, do have quite low phase-noise/jitter. But actually only exhibit Femto-seconds of jitter above 12kHz, meaning they only come into their own towards the top of the audible range. This is because the "Femto Clock" has much lower noise above 10kHz, which is what matters for the intended application (SONET, the Internet backbone, not high-end audio).

GMT exhibits across the board jitter that is so low as to be irrelevant. We simply named it “Zero Jitter.” Typically the new clock changes by less than 0.004ppm over a period of 10 minutes or so. Any clock variations from the source are soaked up by a memory buffer, one short enough to still cause no lip-sync issues in video playback, but long enough to absorb any form of jitter.



AMR DP-777 Zero Jitter principle

AMR’s Zero Jitter circuit = Immaculate Signal



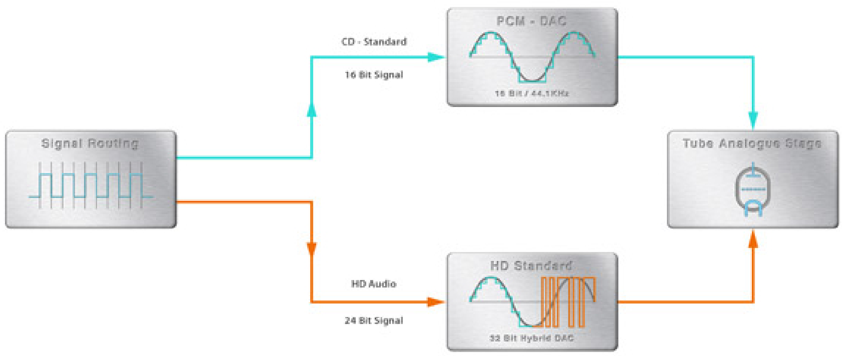
Source: AMR

AMR also employs the "Gemini Digital Engine (GDE)" in the Digital Processor-777 which essentially processes CD-quality data with a different chipset than higher bit/sample rates. What are the benefits in using separate chipsets?
The reasons to use a classic Multi-bit DAC for CD replay should be obvious, after the aforementioned missive on PCM, Bitstream and “native playback.” There is simply nothing to be gained for CD signals by using anything else.

The number of such chips that offer the kind of sound of the classic "Dynamic Element Matching" Philips CD-DAC's is very short, all were discontinued a decade or so and all were made by Philips. Plus none offer more than 16-Bits of operation and nothing else sounds quite like them. So we use it for CD standard signals.

Yet the lack of an ability to treat higher resolution/sample rate at their native resolution would create problems with modern signals. Hence we spent a lot of time finding a modern DAC that offered an excellent sound quality to handle "HD" signals to "future-proof" the DP-777, while offering the classic Philips DAC + Tubes CD sound our CD-Players are rightly famous for and which won many plaudits and awards, when playing CD or CD ripped to files.

In many ways this is the same principle which the iFi iDSD range extends to DSD, play each kind of signal with the optimum solution.



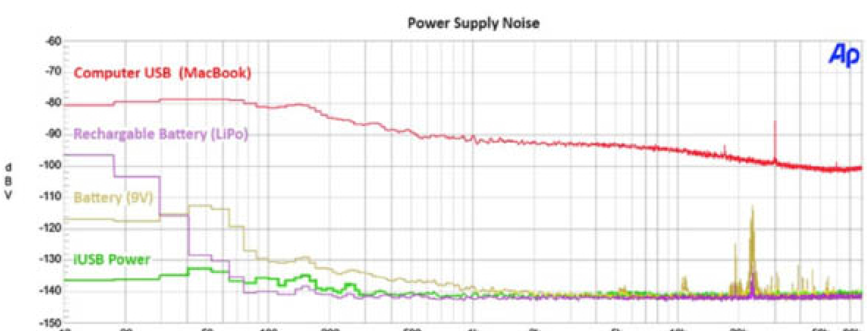
AMR DP-777 Gemini Digital Engine

iFI offers the iUSB Power for USB DACs. Can you talk about the importance of the power supply for USB DACs?
In principle, USB is just a data-pipe. And every DAC benefits from an excellent power supply.

As both USB and Firewire (now Thunderbolt) connections carry power and as external power supplies or mains cables are inconvenient in modern settings (and cost money – arguably), using this power supply is a natural choice when making budget-priced DAC's. It is perhaps less excusable in expensive high-end equipment, yet it is also not uncommon there to power USB interfaces from the computer's USB power.

"Sadly, most of the power delivered via the USB/Firewire Bus is far from what you or I would like to feed a high-quality audio device. At best it is awfully noisy, from 100Hz to several GHz, at worst, well, the less said the better."

Sadly, most of the power delivered via the USB/Firewire Bus is far from what you or I would like to feed a high-quality audio device. At best it is awfully noisy, from 100Hz to several GHz, at worst, well, the less said the better. One can apply a degree of filtering inside the audio device, but there are limits.



Source: iFi (measured on AP2, closely adhering to Stereophile Measurements setup)

The best choice is of course to not use USB Bus power. If this option was omitted “by design” the iUSB Power not only restores this option, but provides a power source that even dry cells and Lithium-ion Batteries cannot match. Think PS Audio Mains generation for 21st century USB audio.

You employ vacuum tubes in a number of your products output stage (OptiValve) including your digital products. Could you talk about the reasons behind their use?
You actually ran a review of our iTube and covered one of the really good reasons:

> “Why tube? iFi references a study by Jürgen Ackermann for the Frankfurt University of Music and Performing Arts where 50 participants took part in a blind listening study listening to a tube and a solid state system.
>
> The results showed the participants felt the tube system improved their overall level of enjoyment by 187% where the solid state system made 30% of participants feel worse after listening. This study was referenced in the excellent piece by Markus Sauer for Stereophile titled, "God is in the Nuances", highly recommended reading. So why tube? It can make you enjoy listening to your music more, the proof of which lies in the listening."


Simple, Tubes subjectively sound better and listening to tubes makes us feel less stressed/more relaxed and brings one to be more emotionally connected to the music – which is something we are all in the pursuit of.

What do you see as representing the next big advance in file-based playback?
2014 will be the year that Hi-Res, be it DXD, DSD or HD-PCM go mobile, portable and streaming.

Smartphones, Pads and related products start to seriously support HD audio (with Japan as usual, leading the way).

Portable headphones and audio devices rapidly gaining sound quality.

Europe leads in streaming music services, giving the music industry a new and solid revenue model to replace the outdated “physical media” model rooted in the early 20th century.

This “physical media” based sales approach was already outdated in the 70’s when record sleeves warned “Home taping kills music” (funny thing, its 2014 and music is still alive and well).

Source: redbubble.com

High-Speed mobile and city-wide WiFi networks give the bandwidth to stream HD Audio seamlessly and start gaining serious market share in Europe and Asia.

Source: vizio.com

To those who cling to old styles of music delivery it may sound like cloud coocoo land. To others it may be more a case of “the future is streaming from the cloud to my DXD DAC and Sennheiser HD800”.

No matter what anyone thinks, it is coming.

The future is NOW. And it is HD/Quality Audio everywhere, in any format and in any package, from Earbuds to Wilson Alexandria’s. These are exciting times for the audio industry and those who are part of it can read the *mene tekel*.

NEXT: Q&A with Thorsten Loesch of AMR/iFi Addendum: PCM vs DSD »

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COMMENTS

great interview

Submitted by deckeda on April 7, 2014 - 9:53am

Intuitively, we all know converting from one thing to another is never perfectly seamless. This interview is one of the best examinations and explanations of the challenge of reproducing music digitally. Thank you Mr. Loesch for your candor.

Fascinating how the music industry, decades ago, has walked away from engineering a dedicated platform that not only gets the job done, but does so in a way that conceptually makes sense. Seems like this is a nut someone has yet to crack because they don't understand it exists. Digital is largely handled as a "workaround" at the precise steps where it shouldn't be.

All of the internal PCM/DSD/PCM conversions that appear harmless can be more simply highlighted in the fact that DSD mixing consoles or other editing/manipulations do not exist. I got no dog in this fight; I am NOT "anti DSD" per se but that's a red flag IMO. (Or maybe red herring, you decide.)

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time domain vs frequency domain

Submitted by ball3901 on April 7, 2014 - 10:59am

Although this is likely a gross oversimplification, high bite rate, low sample rate signals (pcm) have superior amplitude resolution and inferior time resolution. low bit rate, high sample rate signals have superior time resolution, but inferior amplitude resolution. (Not taking into account the effect of filtration on either)

So, take your pick. How about if we came up with a high bit rate and high sample rate format? The best format we have at this time that represents both is DXD. And it just so happens that the native DXD files I own are perhaps the best sounding examples as well? Of course, we have to take into consideration the entire signal chain. But I don't think it is a coincidence that DXD is their delivery format.

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One of the best tech article ever

Submitted by ball3901 on April 7, 2014 - 10:29am

Between this and the q&a with John Swenson, Audiostream has provided the most in depth detailed, and absolutely sensible insight into the workings of digital audio.

Thorsten has always been willing to answer questions in great detail with candor. Sadly, this is lacking in the audio design world, and is very, very refreshing.

Note that some of this (obviously) is simplified for understanding, but conceptually it is an excellent analysis of very complex and debated subject matter.

All "audiophiles" should read this over and over and over again until it sinks in.

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Great explanation of a difficult subject

Submitted by bobflood on April 8, 2014 - 3:04pm

This is the first truly comprehensive explanation of the digital audio capture/playback chain that a lay person can understand.

Thank You

Log in or register to post comments

pure DSD

Submitted by ball3901 on April 7, 2014 - 10:41am

Another thing worth adding, is that pure scarlet book DSD is something that audiophiles wouldn't want to listen to anyway. The official DSD noise-shaper is a 7th order filter with a corner frequency below 20KHz. Which means officially it has no more frequency or impulse response than CD!

Most implementations, including the BB chip described in the article, use an analog FIR at a much higher cutoff. Which in addition to the increase in impulse/frequency response, leads to an increase in ultra increase in impulse/frequency response, leads to an increase in ultra sonic noise released into the signal chain. Which means an increase in IM distortion, which I believe (but don't know for a fact) is a major part of DSD's "smooth, analog" sound.

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Very interesting reading.

Submitted by labjr on April 8, 2014 - 4:37pm

Very interesting reading. Thanks.

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Concepts

Submitted by fmak on April 8, 2014 - 4:08am

"Note that some of this (obviously) is simplified for understanding, but conceptually it is an excellent analysis of very complex and debated subject matter."

The explanations are gross simplifications of the clocking concept adopted by AMR (with fancy names) and in my view does not enable a direct degree of comparison with more conventional approaches (as to which is better). This is fine, but the assertions of zero jitter and 'global master timing' are just buzz words that have no meaning in themselves and so they are not technically based explanations of superiority.

The chip used in the DSD Nano is an old chip used in SACD players.

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Can we keep it simple ?

Submitted by bigrasshopper on April 8, 2014 - 7:32pm

I tend to treat all this complexity as the simplest argument in favor of analog formats, when it comes to sound. In spite of its own set of limitations it sounds pretty good to me, when I can get it made by people who care. Though I can't say for sure that I've heard a digital format that's pure from end to end. I can say there seem to be more of this purity now, available in analog circles. Though that is also not a sure bet either, as non-audiophile vinyl is a hybridized product.
I would really like to see how the "analog" ladder DAC fits into this breakdown, where it's similar and where and it avoids some the those comprises that were outlined. AMR may not be making them, but in a complete history like this the absence of or mention of a ladder Dac seems kind of conspicuous. If Thorston or anyone else would care to comment on them, I am all ears. If I were going to purchase a reference Dac, that's what I would be looking at. And by the way, what about a ladder ADC ?

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Ladder DACs

Submitted by iFi Audio on April 10, 2014 - 6:19am

Hi,

Ladder DAC's are the exact same thing as pure PCM DAC and while the DACs we use are from Philips and do not use resistor Ladders (but something much cleverer), they operate on the same

Ladders (but something much cleverer), they operate on the same fundamental principle.

The Pacific Microsonics Model 2 mentioned has Ladder DAC's and the ADC equivalent a SAR (Successive Approximation) ADC for playback and recording.

At any extent, Ladder" DAC's are just a subset of true PCM DAC's, as opposed to Delta Sigma.

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Beauty and the beast

Submitted by Vincent Kars on April 19, 2014 - 10:38am

:)

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Mistake

Submitted by JIMIXY on April 21, 2014 - 3:37pm

This article reads, "If we convert to DSD from DXD, that is 1-bit at 2.822MHz to 24-Bit at 352.8kHz"

should it not be from DSD to DXD?

Regards

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Mistake?

Submitted by JIMIXY on July 3, 2014 - 6:59am

This article reads, "If we convert to DSD from DXD, that is 1-bit at 2.822MHz to 24-Bit at 352.8kHz"

Shouldn't it be from DSD to DXD ??

Please please correct or explain

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