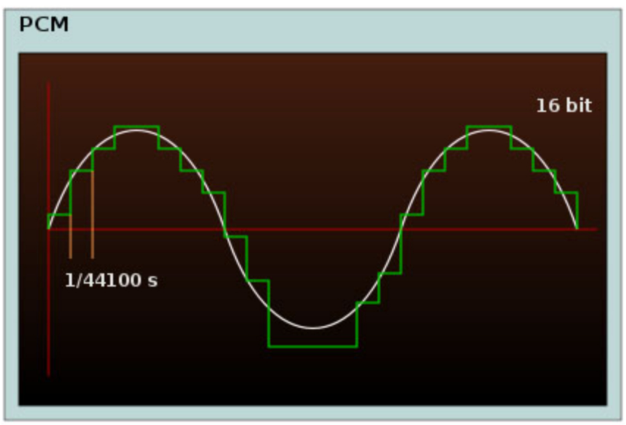
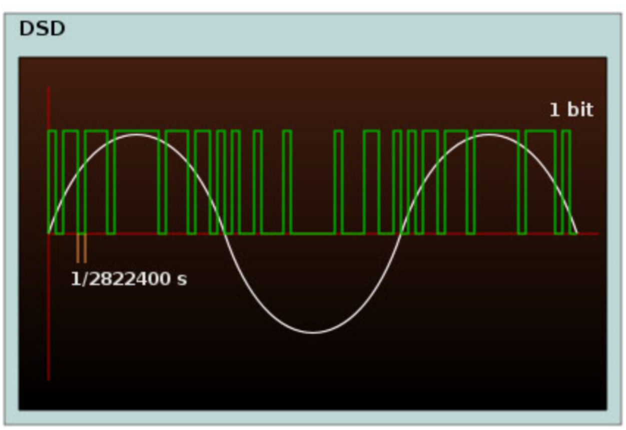
Q&A; with Thorsten Loesch of AMR/iFi Addendum: PCM vs DSD

What actually constitutes "native DSD" and/or native PCM has become an issue of substance. 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:





Each format has different strengths and weaknesses. Whenever we convert from one format to another we incur unavoidable losses. And worse, in the process we tend to remove whatever makes one format exceptional and at the same impress upon it the limitations of the other format. So in effect we get the worst of both, rather than the best of each or even the best of one.

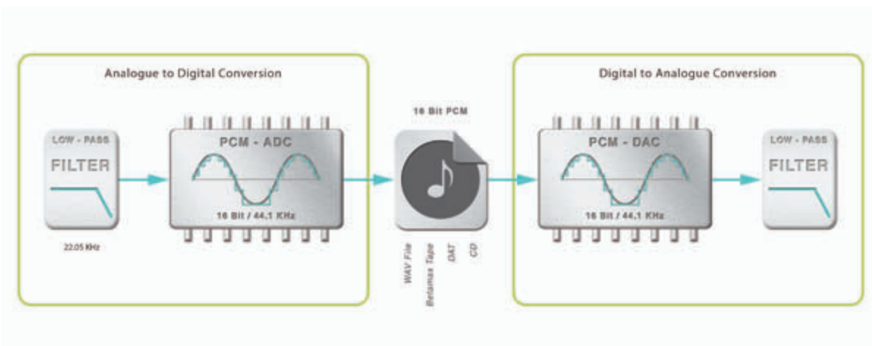
Maybe a bit of history is in order. So please bear with me.

Maybe a bit of history is in order. So please bear with me.

Digital audio started out with both PCM (Japanese EIJA Standard 14 and 16-Bit PCM mainly, but also non-standard systems like that employed by Decca) and a Bitstream system principally similar but inferior to DSD in the form of the dbx Model 700.

The original CD standard PCM system translated the music signal every 22.7 microseconds (44.1kHz sample rate) into a single value out of 65536 values (2^16 – 16 Binary weighted bits) possible. So at each point in time there is a defined absolute value compared to some reference, just like it is the case for analogue systems. The key difference is that there is no continuous waveform, but rather a staircase like approximation of the original waveform, correct analogue low-pass filtering “smooths out” the staircase waveform.

If we wanted to visualise 44.1KHz/16Bit PCM as an image – taking 1 second, we would have an image 44100 pixel wide and 65536 pixels high. PCM always had a high-degree of absolute precision and resolution for amplitude. As a downside, we must low-pass filter the analogue signal upon recoding severely, with attendant phase and time domain errors and time-domain resolution which is quite coarse.



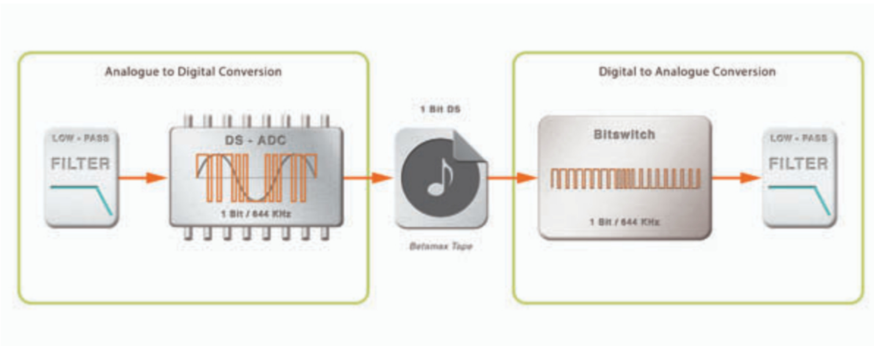
44.1KHz PCM digital audio system (e.g. Sony PCM F1)

Mr. Loesch’s own Sony PCM F1 portable recording system, originally owned by Alan Parsons

By comparison, the dbx bitstream system determined every 1.55 microseconds (644kHz sample rate – around ¼ of that of DSD) if the signal had changed upwards or downwards compared to the last sample. So at each point there is no absolute value and for a 22.7 microsecond long time window (this is the PCM 44.1kHz window), only 14.6 values can be represented, compared to the 65536 values of PCM.

A technique called noise-shaping is used to allow more values to be represented, but the time window is required to be made longer (apply averaging). DSD's higher Sample rates improve this situation to some degree.

If we wanted to visualise the 644kHz dbx bitstream system as an image – taking 1 second, we would have an image 644000 pixels wide and 2 pixels high. Single-bit / Bitstream systems tend to be lacking on amplitude domain precision and resolution but have high time domain precision. On the bonus side, this Bitstream system did not require a steep slope anti-alias filter of the kind needed with PCM, but low-pass filters are still required.

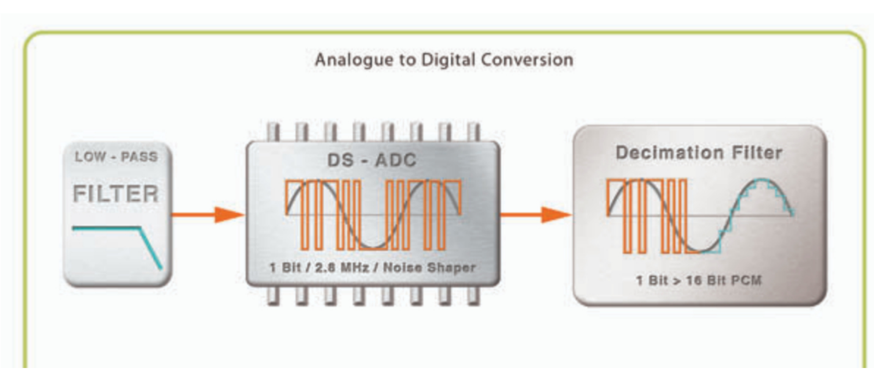


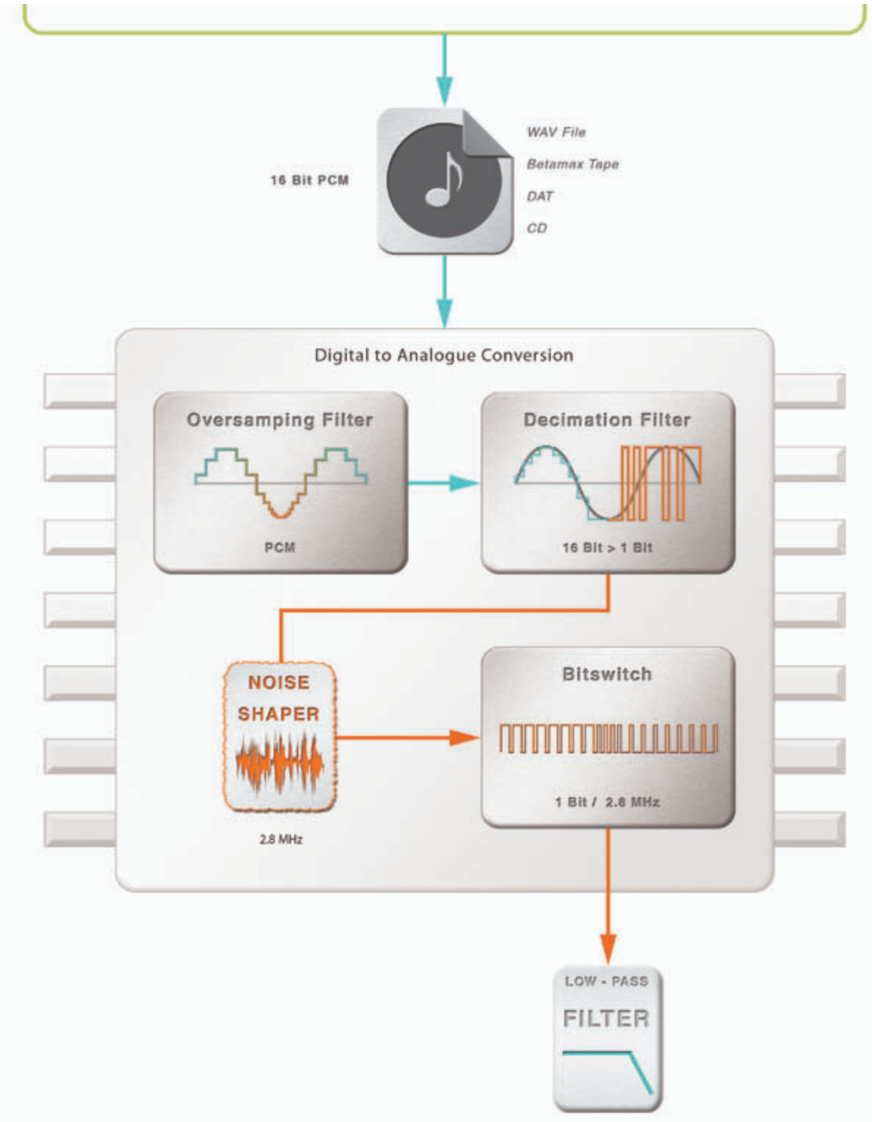
Classic Bitstream digital audio system (dbx Model700)

dbx Model 700 Bitstream AD & DA Processor

Either system requires some form of low-pass filter on the playback side, with a wide variety of trade-offs between suppressing ultrasonic noise and Phase/Time domain errors.

In this "Format War" between PCM and Bitstream, PCM won the first round resoundingly and became the de-facto standard for digital audio, later also becoming the standard for CD and later still, as audio system for DVD. The portable semi-pro Sony PCM-F1 and the studio pro-grade Sony PCM1630 became the de-facto standards for early digital recording while the dbx model 700 is merely a side note to history and forgotten by almost everyone.





appx. late 90's digital audio system – single-bit ADC to CD or DVD to single-bit DAC in 16 Bit / 44KHz or 24Bit / 96KHz

True PCM audio recordings and playback used Multibit Analogue-to-Digital Converters (ADC) during recording and Multibit Digital-to-Analogue Converters (DAC). Such Multibit converters are complex and time-consuming and hence expensive to make, but they dominated the first decade of digital audio.

By comparison ADC's and DAC's of the Single-Bit / Bitstream type are much easier and thus cheaper to manufacture. The hardware (ADC and DAC Chip) market moved towards Single-Bit / Bitstream converters in the early 90's and away from true PCM.

Crystal CS4303 Delta Sigma DAC and Asahi Kasei AK5327 Delta Sigma ADC

Exceptions existed. However by the late 1990's the last holdout making Multibit ADC systems was Pacific Microsonic for their Model 1 (and the subsequent, largely identical Model 2). Multibit DAC's for audio managed to hang for longer and are still used today in the more extreme Ultra-Fidelity replay systems, though the low-end is dominated by cheap Single-bit / Bitstream devices.

We now had a situation where most of the ADC and DAC were Single-Bit / Bitstream devices but the ADC Single-Bit / Bitstream had to be converted to PCM for release on CD and then from the CD PCM signal converted back to Single-Bit / Bitstream to be output by the Single-Bit / Bitstream DAC. Such double conversion of course is really a worst case scenario: we accrue losses of information twice, sound quality as a result suffered twice. With the parallel impact of the so-called loudness war, the quality of commercial music recordings reached a massive low in the decade from the mid 90's to the mid 00's.

Hence, we may presume that many "PCM" recordings released after the mid-1990's are in fact sourced from single-bit ADC's (so something akin to DSD) and then converted to PCM for editing, mastering and release, even if they are marketed as "HD" PCM.

Yamaha 01V Digital Mixer with single-bit ADC's and 16 Bit PCM output to CD to Marantz CD-Player with single-bit DAC

In fact, pretty much only recordings made through Pacific Microsonics Model 1 or 2 can be considered actually true HD PCM and as such converters are very rare, so are such recordings. Hence, few people will have truly heard much in terms of real "PCM" Audio, even less so "HD" PCM Audio; a sad fact but true.

appx. late 90's HD PCM digital audio system – Pacific Microsonic Model 2 Studio 24-Bit / 176.4KHz Multi-bit AD/DA Processor with HDCD

In light of the problems (and for other more commercial reasons) in the late 90's, Sony and Philips tried to (re) launch a commercial Bitstream format under the trade name DSD; first as an archival format and later as a CD replacement format called Super Audio CD (SACD). DSD eliminates the conversion from Single-Bit to PCM and back by taking directly the Single-Bit / Bitstream and recording it. As such DSD/SACD was a major improvement on CD's sourced from Single-Bit / Bitstream ADC's and played back via Single-Bit / Bitstream DAC's.

appx. late 90's DSD digital audio system – SACD

Yet this attempt on marketing a Bitstream format was also not blessed by success. Over this period, even Vinyl sales continued to eclipse that of SACD.

With higher sample rates and greater word length recordings becoming standard for DVD and Movies, the hardware industry was under pressure to deliver ADC and DAC solutions capable of “beyond CD” quality. They found generally the Single-Bit / Bitstream technology poorly-suited and thus a variety of so-called “hybrid” systems became the new standard, where several Bits worth of ”Multibit” conversion are combined with Bitstream conversion.

appx.2013 typical DSD revival digital audio system – “DSD capable DAC”

The best of these ADC’s and DAC’s using this technology have become very good and this technology is now in effect as the new standard for conversion. There is of course a problem here. We now have ADC’s that operate at 6-8 Bit and 256 or even 512 times oversampling, and DAC’s that have similar specifications. This is perhaps not quite the same as a good 24 Bit – 768KHz capable true Multibit ADC or DAC, but in principle the potential is way beyond the traditional Single-Bit / Bitstream devices.

Yet the “transport” between these ADC’s and DAC’s are available only as either DSD at varying speeds or as PCM at varying speeds. There is no “native” format for this new system. This has created again the same issue that DSD attempted to resolve, but at a higher level of quality. To release the recordings we need to convert to either DSD or PCM and either conversion will lose some of the unique qualities of the original recording.

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 off, while we are only having 12.5% of the time domain information that the DSD system is capable of.

If we convert to DSD to 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.

As these ADC/DAC parts generally were developed for and targeted at a PCM dominated market, where recording/editing/mastering/release all happen in PCM these ADC’s generally output PCM and the DAC’s expect PCM input and they tend to be well-optimised for this operation.

DSD is often added as a mere afterthought and in order to provide “buzzword compliance”. *Many such DAC’s for example have a complete PCM Audio path with digital filtering and a digital volume control. Inside, DSD is first converted to PCM then filtered digitally (adding all the problems of converting PCM to the DSD data stream) and finally converted into Multibit Delta Sigma.* So we have **double the undesirable conversion** at the heart of the Black Box we call the **DAC Chip**.

Finally, we come back from the past to the present and to “native” DSD and PCM. If we really want the best from DSD, then converting it first to PCM, processing it as PCM and then playing it back as Multi-Bit Delta Sigma 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 DAC’s.”

Single-bit ADC with DSD output to File played on Mac/Audirvana and “Brand X” “DSD DAC” with hybrid DAC

Any sonic differences we hear with such converters between PCM and DSD releases tell us strictly about conversion algorithms, NOT about the formats. Losses in sound quality compared to the original untouched DSD stream are unavoidable.

So ideally we play PCM back as PCM, with a true Multibit 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). If this is our desire, we find that most current “Flagship” DAC’s fare very badly. They mangle both PCM and DSD.

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 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.

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 belong. In the end, as they say in my adopted fatherland (England) – “the proof of the pudding is in the eating”.

ARTICLE CONTENTS

Log in or register to post comments

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.)

Log in or register to post comments

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.

Log in or register to post comments

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.

Log in or register to post comments

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 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.

Log in or register to post comments

Very interesting reading.

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

Very interesting reading. Thanks.

Log in or register to post comments

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.

Log in or register to post comments

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 ?

Log in or register to post comments

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 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.

Log in or register to post comments

Beauty and the beast

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

:)

Log in or register to post comments

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

Log in or register to post comments

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

Log in or register to post comments

Related Latest Reviews Recommended

Conversations With Ayre (Video)

Sampling: What Nyquist Didn't Say, and What to Do About It, Tim Wescott, Wescott Design Services

Open and Tolerant: MBL's Juergen Reis on Listening, Measurements, and (Un)Certainty

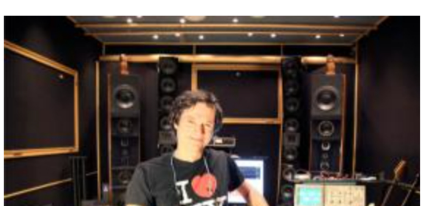
The Subjectivist/Objectivist Synthesis, by Jason Stoddard

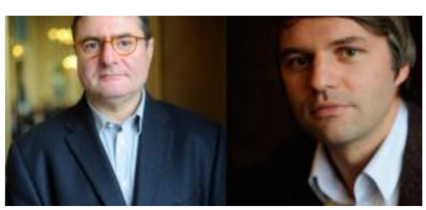
USB Audio Gremlins Exposed: Beyond 1s and 0s, by iFi Audio

Q&A; With Mike Moffat, Schiit Audio

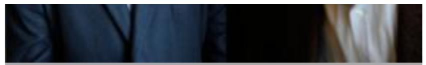
Bruno Putzeys Talks DACs

Q&A; With Xuanqian Wang of AURALiC





Q&A; Michal Jurewicz, Mytek Digital



Q&A; With Yves Riesel, CEO, Qobuz

Emil Torick On Why The MP3 Was Good Enough

Q&A; with John Banks, Chief Brand Officer, Bluesound

AudioQuest White Paper: Evaluation of Digital Devices and Proper Warm-Up for Ideal Listening and Measurements

Q&A; with Thorsten Loesch of AMR/iFi

PS Audio Presents Ted Smith on Jitter

Hello Mr. Soul: My Interview with Neil Young

CES 2012
RMAF 2011