ПК Аудиофил

# Russia :

‘From Russia with Love’ – PC Audiophile Review

## Metrum Acoustics Pavane - a unique R2R matrix DAC


In 2014, as reported by *PC audiophile,* a DAC passed the test without resampling (*The non-oversampling* sound *of NOS)* – Metrum Group Acoustics the Hex. Its sound made a strong impression on us. The sound was so natural that one could talk about the absence of digital artifacts that are so annoying to music lovers.

Now we have an opportunity to assess another converter from Dutch company ***All** Engineering - Metrum Acoustics Pavane,* which was released in 2015 and incorporates the latest technical developments from its makers.

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|---|---|
| **Name** | Metrum Acoustics Pavane |
| **Manufacturer** | *All Engineering,* Netherlands www.metrum-acoustics.com |
| **Exclusive distributor in Russia** | *Distribution company "Plan A"* hi-audio.ru |
| **Price** | 330 000 p. |

The main feature of *Metrum Acoustics* DACs, is the rejection of modern standard converter circuits and the use of technical solutions that eliminate the need for oversampling in the digital signal.

The first model converters from Metrum Acoustics have won a lot of fans due to their sound quality and sold well. The designer behind the company Seis Ruytenberg (Cees Ruijtenberg) did not stop there and decided to go further by creating Metrum’s own design NOS chips based on R2R-matrices.

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*Why R2R matrix*

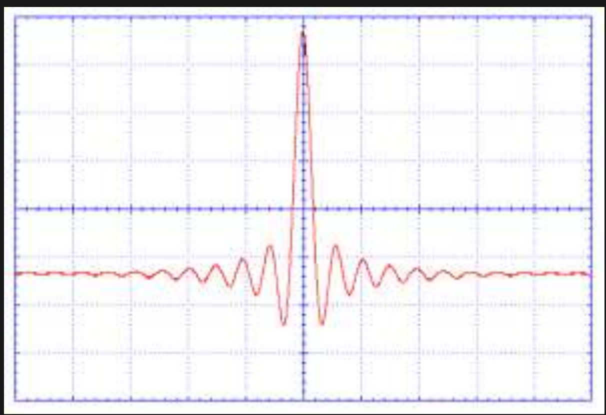
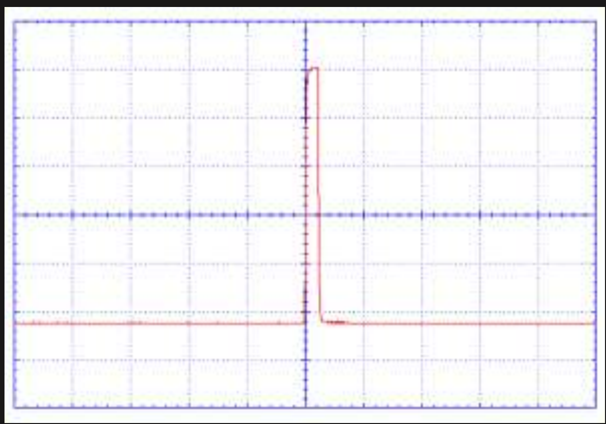
Matrix [R2R-1] - one of the classic designs used in electronics for converting digital code to an analog signal. Originally using this principle, many DACs were based on the use of this type of converter chip. Among them are multibit chips such as the Burr-Brown PCM1704 and PCM63, Philips TDA1541A and TDA1547, Analog Devices AD1865. Today, they are no longer available, although there are examples of modern DACs operating with such converter *chips:* Aqua an Italian company, the English *Audio Note,* Russian *Markan* and *MyST (Mycroft)* and others.

Since the technical features and cost of production of such chips were poor, the development of digital to analog conversion technology has gone the other way and now all the ‘off the shelf’ chips are based on the use of oversampling with digital filtering. Technical specifications have improved, costs reduced, but what about the sound?

Unfortunately, as many experts and lovers of good sound have found, modern DACs with oversampling, despite the various technological tricks, cannot get rid of some harshness or gritty sound - "Digital Audio".

One of the first to note these sound problems from converters with digital filtering and the benefits of avoiding such an approach, was Japanese Rёhey Kusunoki (Ryohei Kusunoki) back in 1996. The cause of these problems in converters with oversampling - distortion in the temporal region, to which the human ear is particularly sensitive.

To illustrate this on the *Metrum Acoustics* website are images of two converter response waveforms from a single pulse.

- 
- 

On the left - a typical response from a converter with oversampling. Immediately evident is that the temporal characteristics of the process contain oscillations which precede the peak response and follow on afterwards. This phenomenon is manifested in the converter, using digital signal filtering and in English literature is referred to as *pre - ringing ( «predzvon")* and after *post - ringing ( «poslezvon").* In the right picture - the output of the converter without oversampling : the oscillations are absent but the pulse shape has not changed.

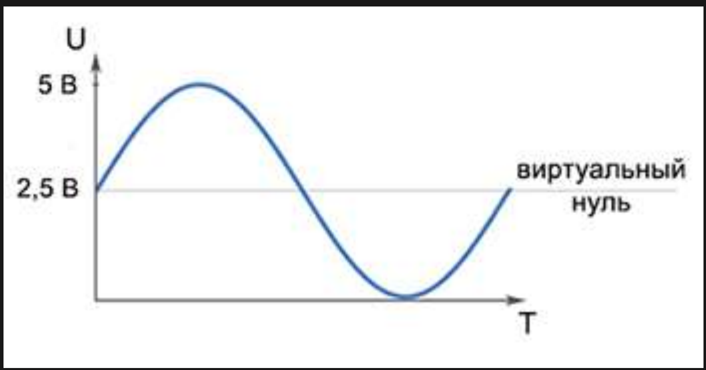
Attempts to eliminate such oscillations have been made by changing the characteristics of the digital filter. To this end, the British company *Meridian Audio* proposed a so-called apodized filter *(apodising* filter), which eliminates the "predzvon" (pre ringing). Unfortunately, this filter not only eliminates the subsequent oscillation but may even increase its duration. Currently other digital filters are used, which are also designed to reduce oscillations. Such filters are built into the BOF (Beginning of Frame data stream codes) in modern chips, included as desired to provide a change of sound. However even with custom (user) adjustable filtering, this still has not completely solved this problem with a resampling circuit.

But because of the presence of such oscillations in digital to analog conversion, the signal suffers in reproducing the natural sound of musical instruments, especially in the high frequency range, as well as characteristics such as precision sound playback, preserving natural-sounding musical attack. Therefore to overcome this problem, the use of R2R resistor converters and this type of circuit are considered the answer, thanks to the absence of distortions in the temporal region - providing the right sound.

It may be added that, despite the widespread use of converter chips with oversampling converters, R2R resistor designs have shown "persistence" and now the technology has successfully developed but to new levels.

### *Innovations in R2R-matrices DAC Pavane*

As a result of hard work, Cees Ruijtenberg has managed to improve the R2R-circuit matrix and eliminated some weaknesses.




Firstly, the classical R2R scheme disadvantage is the problem of so-called "virtual zero" (conversion of a variable sound signal), which appears at the moment where the transformation matrix AC audio signal is an average value of output voltage equal to half the supply voltage. These points in the circuit must be switched all at once and even a slight asynchrony in their shift for example (because of the magnitude of the deviations of resistances), leads to disruption of the conversion process.

For the solution to this problem, Cees chose the method offered a few years ago. He has applied two parallel R2R-matrices, having connected them in a balanced design. At the same time as one matrix begins to work with a positive half wave of a signal, the second works – with negative. This decision has completely fixed the problem of "virtual zero" and with that "has straightened" out characteristics of the converter around the zero position - that is in the field of the weakest signals.

By the way, simultaneously and independently from the Dutch engineer, known Russian developer of audio equipment L. Burtsev also came to the same conclusion (utility model patent №146932 «DAC").

The second problem relates to the fact that a 24-bit resistor matrix contains a large number of "steps" - resistors impeding the passage of all the same weak signals. Because of this, weaker signals (least significant bits in the digital code) are converted under worse conditions than the larger amplitude signals (most significant bits), which leads to distortions in the linearity of the conversion and switching noise.

To solve this problem, the 24-bit digital code was divided into two 12 bit - streams of most significant bits and least significant bits before conversion, each of which is directed to a separate R2R-matrix. Moreover, to improve the conditions for converting the lower bits of the corresponding digital stream, they are increased in amplitude by 67dB (equivalent to 12 bits) and after conversion - decreased by the same amount. As a result, decreasing the switching noise.

According to the principle of such action, this method reduces noise interference and resembles Compander technology (compressing and then expanding dynamic range), for example the well-known Dolby noise reduction system (Dolby NR), which was designed for magnetic tape recording in the late 60-ies of the last century. In this system NR is introduced to a recording, where the high-frequency responses are particularly affected by noise and then a corresponding reduction is made when replaying. The amplitude of the noise also decreases and provides higher quality sound reproduction. Fans of magnetic recordings on Cassette will remember this helpful system.

The technology used in the **Pavane** DAC**,** enables transformation of weak signals under the same optimal conditions as the larger amplitude signals, which positively affects the quality of the digital to analog conversion. Managing this process is performed by the FPGA and the corresponding schematic is called "pre-emptive correction” *module»* (forward correction module).

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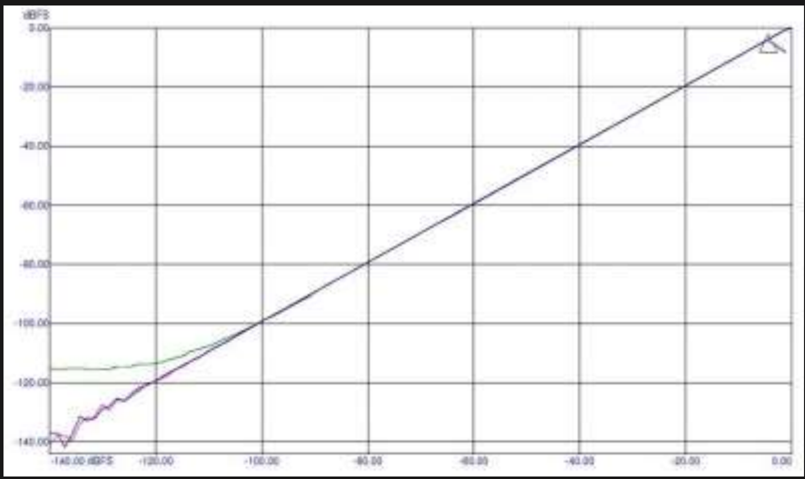
As a result of these two innovations a unique converter module has been developed - *Transient,* having properties previously unobtainable to R2R circuits - high linearity conversion in the weakest of signal information down to -140 dB, low noise and distortion.

The DAC **Pavane** incorporates eight such *Transient* modules - four in each channel. The other new models from *Metrum Acoustics include - Musette, Menuet* and Adagio - also using *Transient* modules.

To illustrate the capabilities of this module, compared to a typical converter chip with oversampling - *Metrum Acoustics* site contains the following diagram.

- 

The graph shows that the linearity of the *Transient* module for weak signals (blue and red lines) greatly exceeds the capabilities of modern chips with oversampling (green line).

It is no coincidence therefore, that the *Transient* converter module*,* is a serious technological breakthrough. The company also offers componentry for sale - for assembly of DACs. By the way, a set of other audio components are also available, including board converters, USB-ins and pre-emptive correction modules etc.

### *Specifications DAC Metrum Acoustics Pavane*

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|---|---|
| *general characteristics* | DAC without oversampling<br>Proactive correction in the FPGA<br>On 4 converter module per channel, connected on the balanced line |
| *Full power* | 45 V · A, 3 toroidal transformer |
| *power consumption* | In standby mode: less than 1.5 W<br>In operation: 18 W |
| *Supply voltage* | 110/115 V or 220/230 V, 60/50 Hz |
| *inputs* | 1 optical<br>2 coaxial (1x BNC and 1x RCA)<br>AES / EBU<br>USB |
| *Outputs* | Unbalanced: 2x RCA, gold plated Neutrik<br>Symmetric: 2x XLR |
| *Output voltage* | RCA: 2 V (rms *.)<br>XLR: 4 (well *.) |
| *frequency range* | 1 Hz - 20 kHz (-1 dB, the sampling frequency of 44.1 kHz)<br>1 Hz - 65 kHz (-1 dB, the sampling frequency of 192 kHz and 384, USB input) |
| *CED* | 0.01% |
| *Signal / noise* | 145 dB at an output voltage of 2 V (rms. *) |
| *Output impedance* | RCA: 100 ohms<br>XLR: 200 Ohm |

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|---|---|
| *Sampling frequency* | Optical input: 44.1 - 96 kHz<br>Coaxial inputs and AES / EBU: 44,1 - 192 kHz<br>Login USB: 44,1 - 384 kHz |
| *Dimensions* | 440 x 320 x 85 mm |
| *Weight* | 10 kg |

** DH. (English RMS.) - root mean square (value).*

The characteristic of a noteworthy excellent signal / noise ratio of the 145 dB, is achieved primarily through the use of *Transient R2R-ins.* Nowadays, DACs with similar characteristics in terms of noise, can be counted on only a few fingers. By the way, the use of dual mono designs also enables a Dac to obtain very good separation between channels - up to 120 dB across the entire audio range.

If we talk about the design of **Pavane** DAC**,** we must also note the following :

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Much attention was paid to the power supply system, for high quality power is required - a prerequisite for accurate digital to analog conversion. At the input a Schurter power line filter is installed. The design uses three separate toroidal transformers for better power ; one for each channel and the third - for the USB module and control circuits. The power supply circuit for each channel - a pair of 22,000 uF capacitors.

As an R2R-balanced design, the entire converter circuits from beginning to end are balanced, allowing you to fully take advantage of such a circuit design. And the *Transient* module design is such, that it further allows you to opt out of any output stages - for complete direct signal output. Obviously, this gives an additional gain in sound quality because the inclusion of any additional circuits for each gain stage, may introduce noise and distortion.

For asymmetric balanced output the signal is summed via a Lundahl transformer, after which a set stage FET. Thus, the balanced and unbalanced outputs are electrically separated from each other and can run simultaneously.

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Digital inputs : AES / EBU, Coaxial BNC and RCA - in order to avoid unwanted interference connections are galvanically isolated.

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The quality USB-module uses a modified production *M2Tech* unit, similar to that installed in the latest model of the famous Italian firm hiFace Evo Two. The module operates in an asynchronous mode, equipped with a pair of high-precision clocks and supports digital signals with a frequency up to 384 kHz. As I mentioned, this module is USB powered from a separate internal power supply in order to prevent ingress of noise and interference through the USB-powered interface.

Designed and assembled by *Metrum Acoustics,* the proactive correction Module is an FPGA Altera Cyclone IV and operates at a clock frequency of 400 MHz.

The converter has an elegant appearance, with more character than the simpler designs from *Metrum Acoustics.* Available in silver and black. In testing I had black.

The DAC housing is made of thick aluminum, of double frame construction. Housing cover is also double: Lower steel plate and the top - toughened glass 4 mm thick.

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The front panel has a power button, five button input selection and related indicators, a window for the remote control signal, as well as the error indicator, which lights up in orange when no digital signal is detected.

The complete box-set includes: remote control with a single button to select the digital input; power cable, good Hi-Speed ACT 1 USB-cable, length .8 m from Dutch company *Intronics;* Adapter BNC-RCA; two discs - one with drivers and operating instructions for the *Metrum Acoustics* DACs and the other - a test CD 'The Absolute Sound Reference' from Dutch company *STS Digital.*

***Listening (standard definition)***

Before listening, as expected, the DAC has been thoroughly warmed up. The instructions indicate that the machine reaches the maximum level of performance, for proper evaluation in three to four weeks.

I started listening to music in standard definition - wanted to hear, is there a difference in **Pavane's** sound compared to a DAC with oversampling. For comparison a second transducer was used - an Audio Research DAC7, which has served me as reference for a number of years. The signal fed from a Transport-player, an Esoteric X-05 via RCA input.

As we know, with an oversampling DAC - with an increase in the sampling rate there can be reduced distortion of the signal in the time domain. When converting signals of higher resolution, it can reduce the duration of oscillation. Therefore, a sound problem in these DACs is often more visible on material with a standard resolution, rather than high.

In this test the *Transient* DAC R2R-matrices in all respects, beat the converter chips in my Burr-Brown oversampling PCM1792's.

Although I was expecting to hear good sound drawing on my experience with the Hex DAC, what I heard surpassed all my expectations. The sound seemed to me, to become even more natural and almost completely got rid of the digital "plaque." Natural sound and ease of perception of music with the Pavane DAC - amazing.

It was like years ago, back when the presentation of digital sound and appropriate equipment was not available and we listened to gramophone first issue recordings. There was more digital signal processing to come, after which newly issued records became even more "digital" than CDs, due at least to double conversion analog-digital-analog processes. In those years, in spite of the limited dynamic range of Vinyl, squeaks and clicks, increase in the harmonic distortion at the center of the disc, it was a natural sound that we could live with for a long time.

 Similar sensations of lightness and naturalness in sound manifested in the Pavane DAC. Particularly impressive listening to music from a good quality recording, for example, a test CD The Ultimate Demonstration Disc (1995) by *Chesky Records.*

On the other hand, when you get a low quality recording, I felt that **Pavane** can extract subtle musical detail, previously inaccessible. It certainly gives new life to CDs and can be especially important for those who have a large music library on CD.

I must admit that in the process of listening to CD, I gradually came to believe that **Pavane** sounds very close to the limit of possibilities incorporated in this digital format.

### *Why 384 kHz*

After listening to music in standard definition, I switched to HiRes files. **Pavane** DAC supports Optical input frequencies up to 96 kHz, for Coaxial and AES / EBU - up to 192 kHz, and USB - to 384 kHz. The question arises: why such a high sampling rate, the more so because as it turned out, this DAC works fine even with material at 44.1 kHz and the music in the format DXD (352.8 kHz / 24-bit) is generally very limited?

Furthermore, it is widely known that even the CD standard covers the entire frequency range of audible hearing and increasing the sampling rate to, say, 96 kHz moves the upper limit frequency response to the digital audio device 40 kHz. Why more ? And for what ? If the increase in the sampling rate cannot be heard, there is a sneaking suspicion that the campaign in support of High-Definition audio, is nothing but an opportunity to sell the same music content in a new "package", plus a huge amount of new equipment to play it on.

Of course, I'm exaggerating but still: why do we need 384 kHz?

To answer this question, we turn to the theory. As is known, the choice of sampling rate according to the Nyquist sampling theorem seems justified. From which it follows that the frequency $f_d$ should be at least two times higher than the highest frequency in the spectrum of the signal f. For example, the frequency of the standard CD sampling ${}_d$ f = 44.1 kHz – this is sufficient to provide all the audible spectrum of signal sound in a range up to 20 kHz.

However, this argument does not take into account that the sampling theorem is formulated for continuous signals with a finite spectrum. But the actual music signal is not.

To get around this situation it is proposed for example, to make the margin requirements set forth in the Nyquist Theorem - a sampling frequency of say twice that estimated i.e. 88.2 or 96 kHz. Or make a margin of sampling rate even more - five times.

Let's look at this issue from the other side - from the viewpoint of the properties of real music signals. An acoustic music signal is constantly changing in temporal and dynamic structure, that amongst other reasons is determined by the speed of musical instrument attack. It is known that the attack process is particularly important as a sound recognition tool.

If we talk about the attack speed then, for example when playing the trumpet - sound may reach a peak for 10ms and cymbals 7-10 microseconds. A temporal sampling interval i.e. the interval between samples used in CD recording is 22.7ms, which is clearly insufficient.

Let us now look at the table, which presents the sampling intervals for different sampling rates.

| *The sampling frequency, kHz* | *Sampling interval, ms* |
|---|---|
| 44.1 | 22.7 |
| 48 | 20.8 |
| 96 | 10.4 |
| 192 | 5.2 |
| 384 | 2.6 |

The table shows that in order to track fast changes in the signal, the sampling interval should be as small as possible i.e. the sampling rate - as much as possible. Moreover, when using frequencies of 96 and even 192 kHz, it is difficult to speak of the highest accuracy digital recording / playback attacks for musical instruments. This argument - not the only possible illustration of why it is so important to use a high sampling rate. But this is enough to understand that HiRes did not arise out of nowhere.

However, we must admit that at the moment there are not many companies recording music in the DXD format, although the number is growing as with selection of music in the HD format. Apparently - this is a matter of time, related in particular to the modernisation of expensive equipment in digital recording studios. But DACs released with support for sample rates up to 384 kHz, including the DXD format are already in full swing. Therefore the **Pavane** converter fits in from a perspective of HD development .

### *Listening (high resolution)*

Returning to listening results, now in HD format.

I was able not to rush into this and I have carefully listened to music from different genres, recorded by different labels. To give a general description, I would compare this to the most enjoyable listening experience to recorded music in recent years.

The device proved to be very good in all respects. Not to go into a detailed description of the individual pieces of music played/formats and ancillaries connected to the digital inputs - not to abuse the attention of readers, I would note only the following sound features :

- The soundstage was complete, stable and transparent. The scale of the studio room or concert hall is transmitted accurately. There is no veil in front of the instruments (I believe that this is mainly due to the *Transient* R2R-matrices and proactive correction module that provides high linearity up to -140 dB).
- High focus of each instrumental sound in the sound stage. Even the most complex orchestral music becomes more understandable, thanks to the precision of each instrument’s location and the lack of any overlap. (As noted above, it is definitely attributed to the impact of the design with separate dual mono power supplies, which gives 120dB crosstalk attenuation).
- The sound is lively, exciting from the first song. Amazing and sounds natural. The lack of digital artifacts. (Design without oversampling and digital filtering).

- The accuracy of timbre and full instrument recognition. Bass - defined and detailed, you can feel the difference between electric and acoustic instruments. Mids - full and rich. Highs - without excessive voicing, probably the most natural sound compared to other DACs that I have heard, including my Audio Research DAC7.
- No stress listening - as if listening to live music. After all, when we listen to live music, you do not think about the quality of sound reproduction. Approximately the same thing - with the **Pavane** DAC**.**

I also note that the DAC works particularly well and is full of musical meaning with HD. I mean music that is recorded and edited in HD formats (e.g. *2L studios,* AIX, Chesky, Linn Records, etc.). Unlike those that are recorded in standard resolution, such as a tape or PCM 44/16 and then transferred to a higher sampling rate format. In these cases, one can hardly speak of a musical recording being HiRes.

### *No support for DSD. And do you need it?*

DAC's working on an R2R-matrix do not imply support for the DSD standard. Not that it reduces the value of the **Pavane** DAC for lovers of good sound! Indeed, in recent years music files in this format are becoming more affordable and most modern DACs running on standard chips can also support DSD.

Indeed, the music in the DSD format, especially DSD128 and above may sound very good. But what are we listening to really with DSD?

Here, I would single out two sides to this issue. The first concerns the properties of the DSD format, primarily the presence of high frequency noise in the signal spectrum. This noise does not only have a negative impact on the audio spectrum - necessitating the need for a low-pass filter but most importantly, it prevents the carrying out of digital editing in this format in the recording studio. In this regard, the majority of music DSD-Files, officially distributed by download – finds the files recorded, edited and mixed in PCM format and then, at the last stage, converted to DSD. For example, very high-quality music files in DSD produced by renowned audiophile recording studio *2L,* are first made in DXD, a PCM format and then transferred to DSD64, DSD128 or DSD256.

There are of course digital music soundtracks really recorded in this format, that have not undergone any digital processing but the number of "live" recordings are low. There are also studios which record sound in analog form on magnetic tape and then convert it into DSD. But here we are not dealing with a high but rather standard recording resolution, in which the HD format is only used as a container for storage and transmission of information in digital form.

The second aspect of the problem is connected to technical aspects. The fact is that the vast majority of DAC's with support for DSD, are made on the basis of using standard chips. And even those converters where the technical data indicates that they support this format in its pure state *(native* format), in reality one bit DSD can be converted to PCM format – in equivalent bit form. To convert special converter circuits are used in a few models - manufacturers including *Light Harmonic, Meitner Audio, Playback Designs* and Lampizator.

We can say that in reality, when we hear DSD under that brand name – it is in one form or another, the result of converting DSD to PCM-flow and / or back. On the other hand, music lovers of this format may themselves employ DSD to PCM conversion, with the help of modern software players, even "on the fly" i.e. during playback of a DSD-file. You can do it yourself.

I can say that the lack of support for DSD is not very important for me. I think high-quality conversion of PCM signals are more important and the **Pavane** DAC ensures this.

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### *conclusion*

You may have noticed that in my report on testing **Metrum Acoustics Pavane** DAC, that a lot of superlatives have been used. This - not the desire to flatter the device but rather to convey impressions of what was really heard.

However, progress does not stand still and it is possible that one day, with the aid of digital technology in a different way perhaps, oversampling technology or otherwise, high-quality natural sound will be received. But today in my opinion, the **Pavane** DAC provides a digital signal conversion result, which is unattainable even from the most expensive models using standard chip circuits.

I think **Pavane** is one of the best DACs available today - standing amongst the most high-quality and advanced devices currently on sale. Moreover, it has one advantage that makes it truly unique - it is value for money. For this characteristic it certainly is more attractive than the other DACs using R2R-matrices, such as Light Harmonic Da Vinci DAC MKII (US$31,000), Totaldac d1-dual DAC (€9,900 Euros) or the MSB Signature DAC V (US$24,990).

After several weeks with **Metrum Acoustics Pavane** DAC, I was so fascinated by the sound, I had no choice but to order one for myself. My previous really good converter, the Audio Research DAC7 has faithfully served me for many years. But his time has passed and he will be replaced by that which to date, has the most natural sound.

Of course, I can honestly recommend it to you.

Y. Kuzmin

June 2016

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*Notes.*

1. Other names: resistor (resistive) matrix r-2r, a matrix of resistors R-2R, ladder-type converter R-2R, converter resistors R-2R, resistive matrix of constant impedance, the R-2R-circuit chain.

2. More on this can be read in an interview with Andreas Koch, one of the SACD format and hardware developers, online edition *Audiostream (USA).*

Information on Metrum Acoustics Pavane online @ www.metrum-acoustics.com (Eng. Lang.). And on the website hi-audio.ru (Russian lang.)

PC audiophile thanks Distribution company OOO “Plan A” for providing the test Pavane DAC from Metrum Acoustics.

Photo: PC audiophile, www.metrum-acoustics.com and hi-audio.ru.

*Photo Gallery*

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