Weiss Medea D/A Converter

Swiss precision

Product: Weiss Medea D/A Converter
Producer: Weiss Engineering LTD. - Switzerland
Approx.cost: approx 11,000 €
Reviewer: Giorgio Pozzoli - TNT Italy
Reviewed: September, 2004

[Italian version]

Introduction

Up to few years ago, Weiss name was famous only among audio professionals, while it remained unknown in the consumer audio environment.

In facts, Weiss Engineering achievements were so widely out and over the consumer area to make any direct involvement or connection difficult to imagine.

Weiss have been pioneering the extreme borders of digital audio since the 80's and have substantially been increasing through the years their mastery in this area. The best confirmation is the diffusion of Weiss products in the most important recording studios all over the world.

I had already read about Medea in USA magazines, where it was described as one of the best units in the world, also from the jitter measurements point of view. I had been testing for weeks a new jitter measurement deck, and I would have liked a lot to be able to review one of these units, also to have a chance of comparing measurements.

Unfortunately, the chances of reviewing such high end products are always very scarce: there are normally big problems due essentially to the fact that importers and local distributors tend to keep the stock to a minimum (which too often just means zero...).

So, when I visited Milano Hi End 2004, I was really excited to discover that Weiss Medea exclusive distributor for Lombardia (Milan region) was Stefano Zaini, the Milan Show organizer.

Asking a unit for a review was a matter of minutes. Some more man-hunting :-)) has proved necessary, there was a lot of interest and the sample was continuously required for demos, but in the end here is the article.

The looks

Medea is indeed a nice looking object. In the silver dressing it appears as a slim silver stripe with a very pure design.

In my view it has exactly the amount of complications and features such an object should have; each detail must have been studied and discussed throughly: such a clean line is hardly casual. Even the Weiss Logo blends perfectly in the fascia.

As said, the front panel is clean, but not empty as in many other DACs. In facts on the left, just under the logo, we find a standby pushbutton, which allows moving all the circuits in a standby mode. In the center panel, a protruding slab applied over the front panel, there are other four pushbuttons that implement the input selection control, each with a beautiful blue led showing the input status: flashing for selected but not jet locked, stable for selected and locked.

On the right, there are only two little holes, which allow setting the level of the variable outputs independently for each channel. Note that a beautiful (ceramic?) screwdriver is supplied for this purpose: even the screwdriver is at the level of the unit, it looks like those fantastic supertools of SF films...

All the labels appear written in a very clear and pleasant font.

The pushbuttons are solid, with a pleasant feeling and never any kind of malfunctioning. It takes a little to get acquainted with the rather long locking time, but the flashing of the led is there to remember you this fact.

The look can appear rather cold in the photos. Indeed is not the warmest looking unit I have reviewed, but in truth the photos are not fair with it, and in any case the physical perception you have as soon as you take it in your hands is so strong that the look is in some way overwhelmed by other embedded characters.

First of all, it is HEAVY. Very heavy for its size. Second, the strength of the panels you touch is such that the feeling is exactly the same of a large solid slab of stone. The weight, the price tag and consequent need for a careful manipulation, make moving the unit a sort of rite, requiring a careful preparation of the destination area before going to fetch it.

The visible details confirm the feelings. When you look at the unit from the top, you see the side slab apparent thickness is around 1cm at least. Well, it is the real thickness: the frame is extremely heavy. Even the top and bottom panels are far thicker than the usual, and below the aluminium slab you can find a further stainless steel box, that utterly increases the unit robustness and weight, let alone EMI protection.

Guys, if this is not a tank, then it is an armoured battleship...

The back panel is devoted to the connections, which are, due to the four inputs, far more than usual in a DAC. From the left we can see the power supply IEC socket with the integrated supply voltage selector, four couples of digital input sockets and finally the two couples of output sockets. Explanation is required: each of the four independent and selectable digital inputs presents two different connectors: an unbalanced one (SPDIF) on RCA pin, and either a balanced (AES) one, on XLR (first three inputs), or an optical one on TOSLINK (the fourth one).

Each connector supports any bit length and sampling rate up to 24/96K, but you can in any case use only one connector of each couple at any time. The first two inputs can also be used together to support flows up to 24bit/192K, as one single line has not a sufficient throughput.

The analog outputs are doubled: you have an unbalanced output on RCA and a balanced one on XLR. Here again you should use only one of the two options. Between the analog output pins, a small switch allows selecting the output level as high or low; the output level can be controlled through the trimmers on the front panel.

Technical aspects

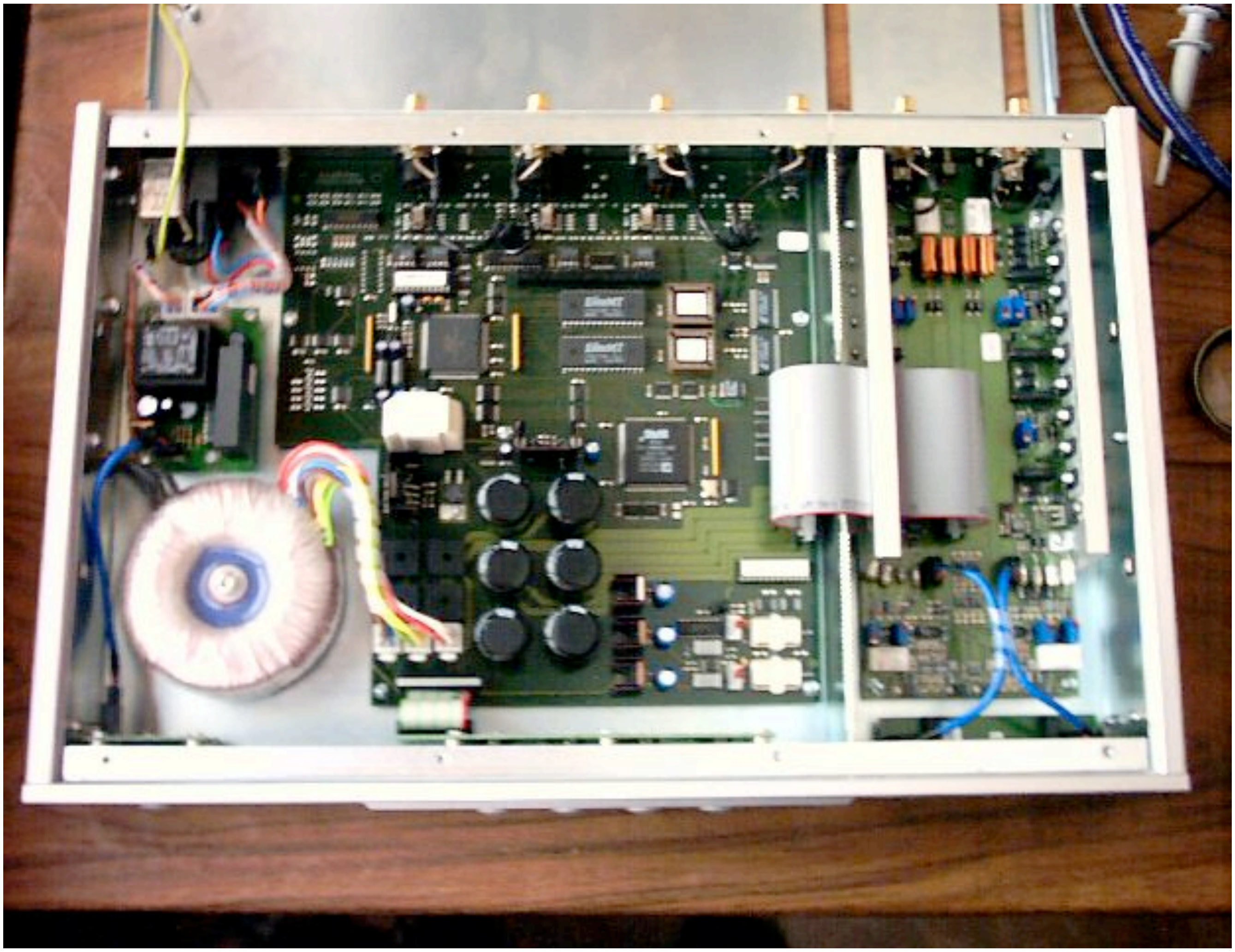
The Weiss Medea is a stereo D/A converter: this means that can be used only with two channel input flows with any frequency up to 24bit/192K. But only stereo. Do not try to connect a multichannel flow, it would not be able to recognize it.

Most components are surface mount, a technology that helps a lot in sparing expensive PCB real estate. The quality is very good, without falling continuously in the audiophile hype: there is exactly what is needed where it is needed, with very little room for phantasy, as in professional systems. The unit contains two main boards plus a front panel board and a further small board with a separate transformer for services. This apparently remains active (as is normal in any servo controlled system) even when the unit is switched off.

The two boards are mounted in separate enclosures inside the box. The interconnection of the two parts is obtained with a large flat cable. We'll come back to this interconnection later on. Wiring is for the rest almost absent, and in any case very tidy.

The main power supply is placed by the power transformer, sized like a small power amp's one. There are also many voltage regulators spread throughout the unit.

The structure of the unit from the hardware point of view seems rather straightforward and classical. The input is managed by a couple of CS8413, the microprocessor controlled version of the well known 8414 digital audio receiver. Two units are obviously needed because of the double flow through input 1 and 2 in case of higher sampling rates.



The clock recovered by these units is passed through a second PLL; from the direct analysis, this seems based on a VCXO, a voltage controlled crystal oscillator. As a matter of fact, if you switch from one input to another, the system takes a significant amount of time (more then one second, I would say) to lock, while we all know very well the very short time required by any standard dac to lock. This is a typical side effect of very stable PLL systems, which can provide very high protection from input clock jitter down to very low frequencies (here fractions of hertz!!!), but are necessarily very slow in getting locked.

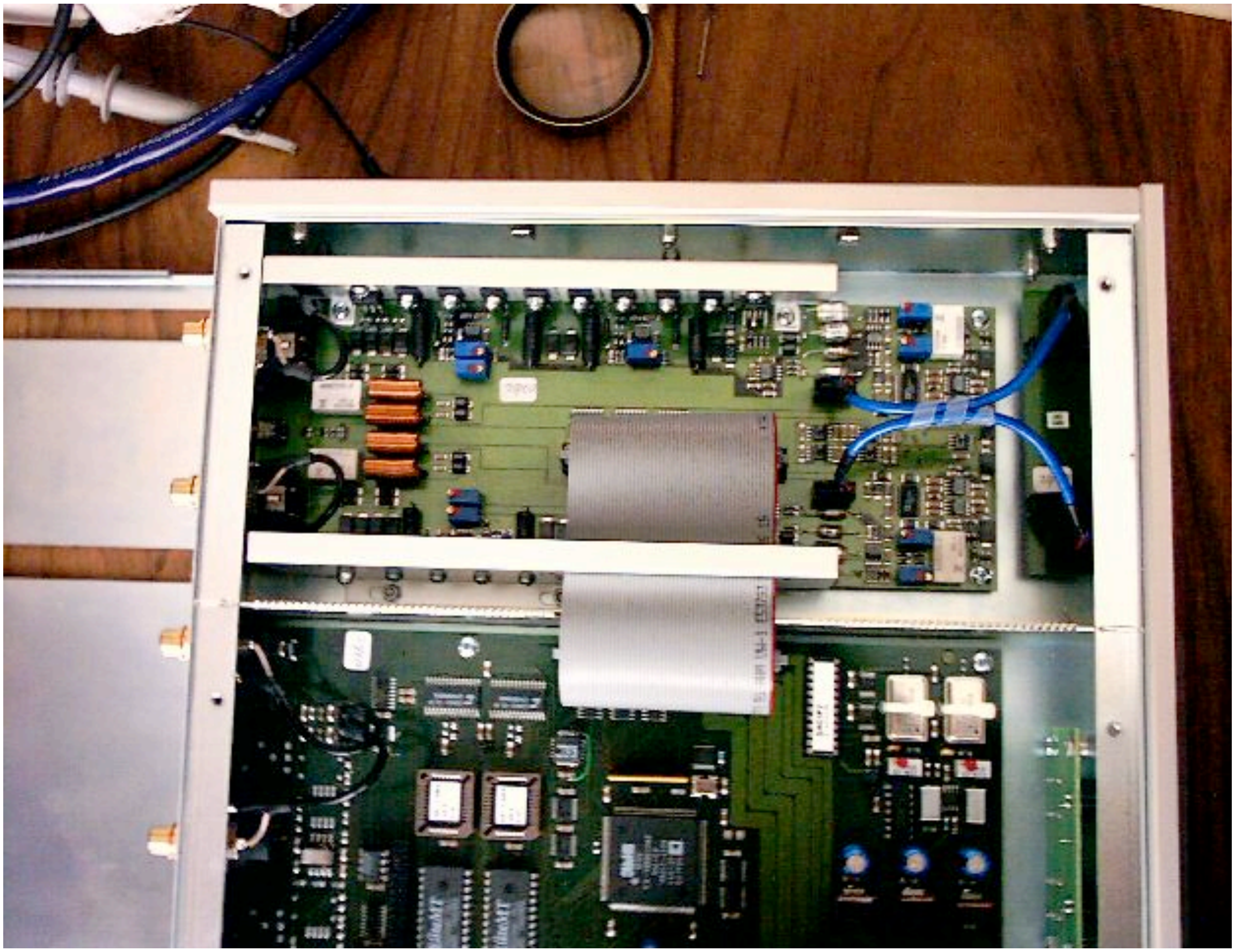
Also, the locking range, as stated in the specifications (+-80ppm), is not huge: there might be problems if you try driving Medea with a transport not having an acceptable quality level, at least in theory. In facts I had no locking problem referable to clock tolerance with any of my digital sources, and in any case for sure whoever can afford Medea can also afford a reasonable transport... Actually, the only limitation is that it is not possible, given the longish locking time, to switch instantaneously from one source to another, but I cannot see in this any real issue for the normal user.

The data is then passed on to a very powerful DSP of the Sharc series by AD, which implements most of the required digital processing, including the most critical part of upsampling, jitter reduction and de-emphasis. The DSP makes by the way use of a large memory bank. This is the real digital heart of the system, the one that probably most makes the unit different. In facts the most critical parts of the upsampling algorithms are performed by this chip: the DAC used can perform similar upsampling functions, but its upsampling factor can be configured and here is reduced to a minimum.

Finally the digitally processed data are passed on to the other board, isolated by metal walls from the digital main board, where the digital to analog conversion and analog processing take places.

The converters are two AD1853, used according Weiss literature in a special configuration to minimize any distortion whatsoever. The analog processing is taken care by op-amps, essentially AD797.

What is really impressive here, however, is the number of transistors used for the analog "power" outputs: even though their power rating is not so high, it is difficult to find such a number in many power amps! The rationale is to try building an output with the lowest possible impedance (virtually 0 ohm, actually 0.2, from literature), and that can tolerate any symmetrical or asymmetrical load impedance down to a short circuit.



Well, yes, a very traditional structure, you'll think. And still, I am pretty sure that, apart very special DSP algorithms, there are many other hidden details, touches of class that make this unit a masterpiece, if not unique.

Just one example: one of the major problems in DACs is how to achieve a good separation of the digital circuits (digital audio receiver and DSP) ground planes and lines from the audio circuits (DAC and op-amps) ones, while allowing them to interoperate correctly. The separation of the two boards, connected through the flat cable, in Medea just follows this kind of considerations.

However, achieving a real ground separation is not simple at all, and it is not even so sure it can give huge benefits with today's technology. In facts, this issue was addressed in the past with optocouplers, but the speed of these units, even in the best cases, is not high enough to convey a precise, unjittered digital signal to the DAC, especially with the higher sampling rates in use today.

On the other end, complete isolation of audio circuits in a separate board would for sure be a positive fact. However, there are a few digital signals that must be moved from one board to the other, and this means that the ground connection between the two boards will have to convey all the return currents of these signals, which would cause a very undesired high frequency voltage to appear between analog and digital ground planes.

To solve this issue, here the digital signals are transferred through a set of RS422 balanced line drivers and receivers, normally used to connect distant units in difficult environments. Balanced operation in facts allows digital currents to loop back through the (dual) line, and not through the ground, minimizing the problem. Original, clean and simple. In one word, smart.

Yes, in this design you can just perceive the constant and continuous search for the best and safest technical solution, which drives so often to very well known and thoroughly tested techniques or circuits. There is nothing diminishing about this: on the contrary, a good designer never spends his time in designing a brand new solution if very good ones are already available. Too often the claims about a new circuit or a new technique are just marketing hype, too often invention is necessary only for ads.

Measures

Well, I do not even try to claim I have a lab capable of serious measures on such a beast. In facts, I was very interested in taking measures of Medea mainly to test my new measure deck, more than vice versa.... So please consider the measures what they are.

I must also point out that I had the unit available for a very short time, and even though I was able to spend a lot of that time with the DAC, I had scarcely any time for any test at frequencies and bit rates different from 44.1K and 16 bits.

However, the measures I got are definitely outstanding, and, what's more important, seem to match the references I was able to gather. Apparently, the unit is so good that is able to make my soundcard work at his best too...

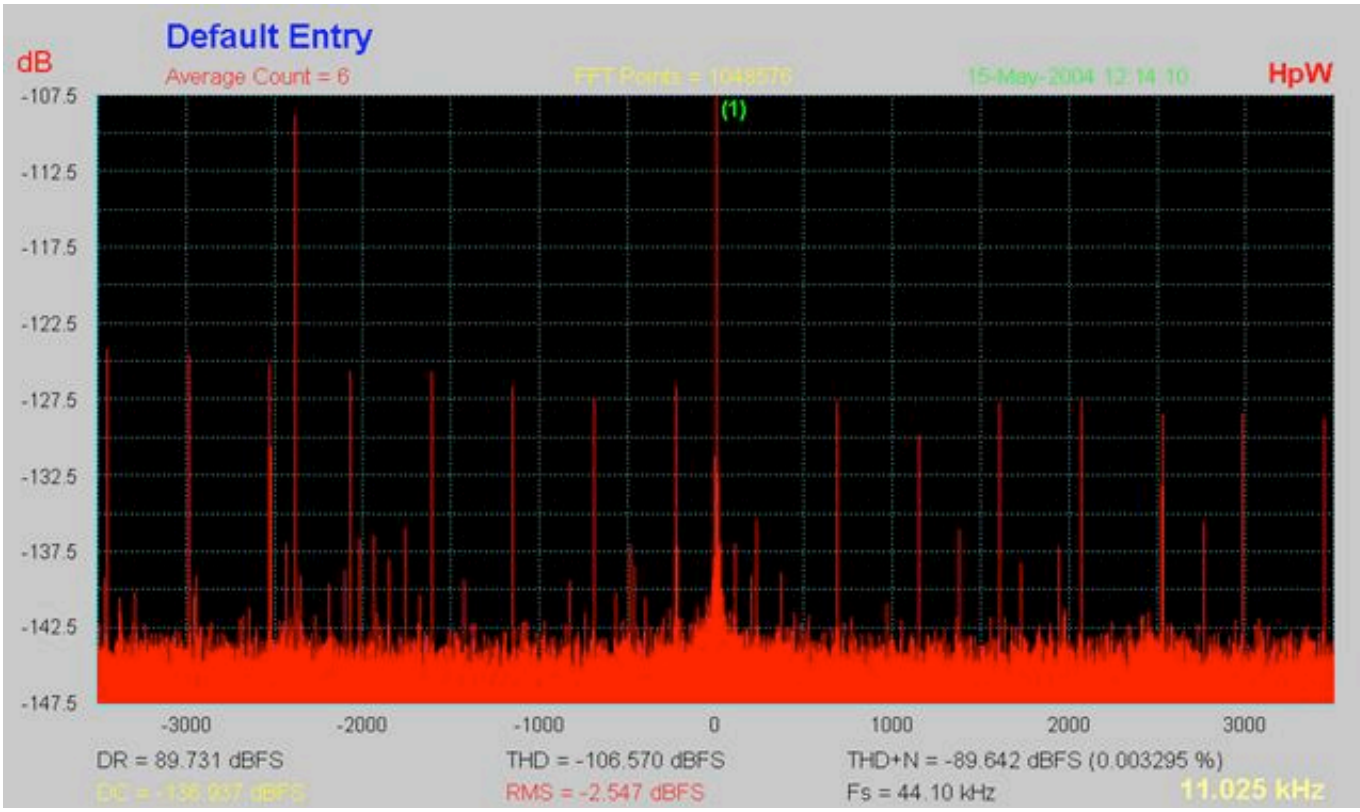
In any case, I was not able to measure any kind of distortion whatsoever down to less than 104dB (0.001%) even at highest output levels at 16bit 44.1k; harmonic distortion seems mainly due to odd (3rd and 5th) harmonics.

Even intermodulation results as low as stated; by the way, apparently the higher the frequencies, the lower intermodulation.

Distortion with dithered tones at 1KHz and 10KHz tones at -90dB is completely hidden in the noise floor at less than -130dB.

I also made several jitter measurements. These are spectra of the analog output centered on 11.025KHz (Fc/4), calculated usually as geometrical average of 16 FFTs with 32K points each, while driving the DAC with a high level 11.025KHz tone with the least significant bit (LSB) toggling on and off with a cycle of 192 samples. This very low level asymmetrical square wave adds a set of very low level lines at 44100/292=229.68...Hz and all odd harmonics which help also in evaluating the low level signal treatment capabilities of the digital units. Occasionally, I take a similar spectrum with single or even multiple acquisitions of 1M points.

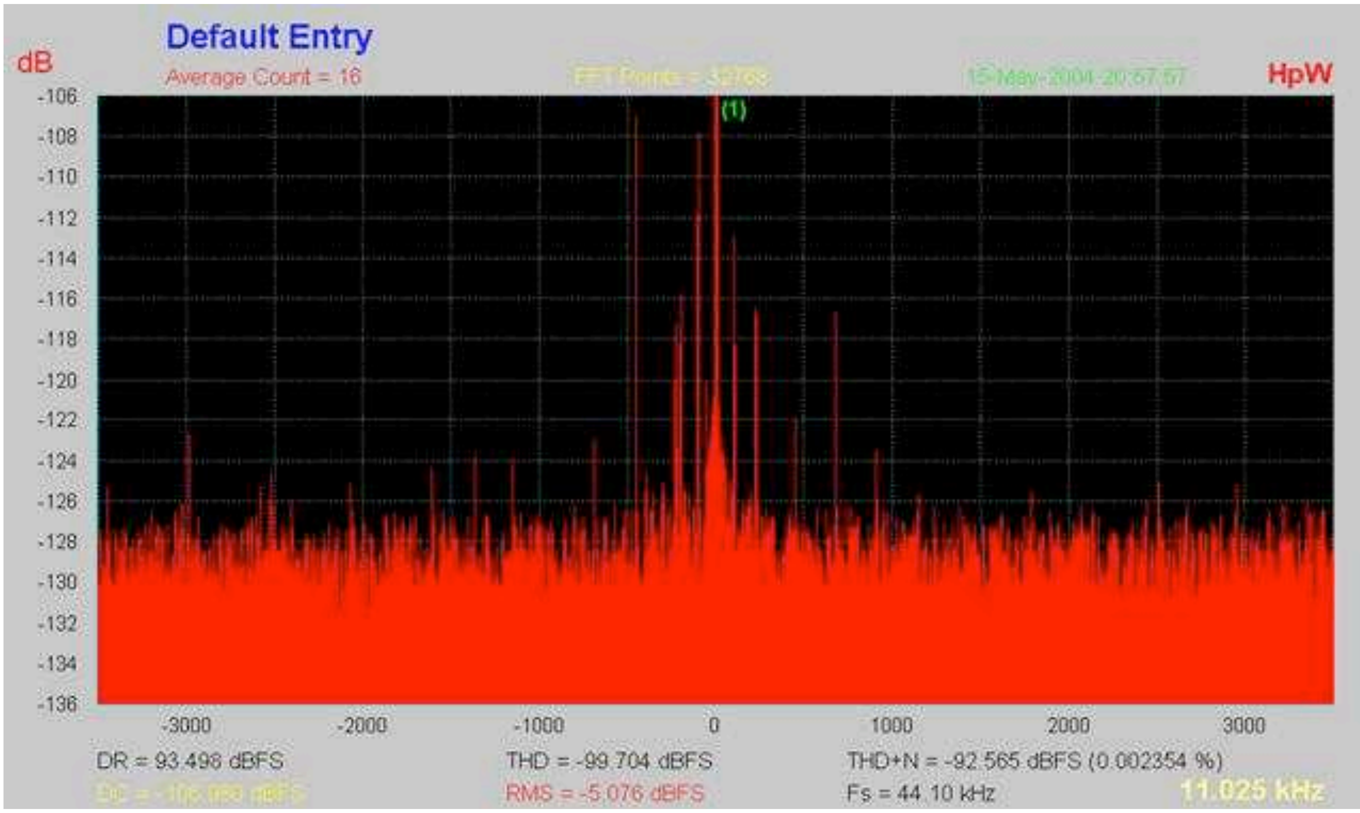
The measured jitter is in the range 100-120ps for all the inputs, and is the lowest I ever measured. I am surprised the card is able to make such measures, indeed, but they match the value published by Stereophile: the spectrum is almost perfect. The line at -2.3KHz is a disturb I was not able to eliminate, but seemed injected at soundcard level, as it was present also in other test measures in which Medea was not involved at all; the problem, as per Murphy's Law, obviously disappeared the next day, as soon as Medea was back to its owner...

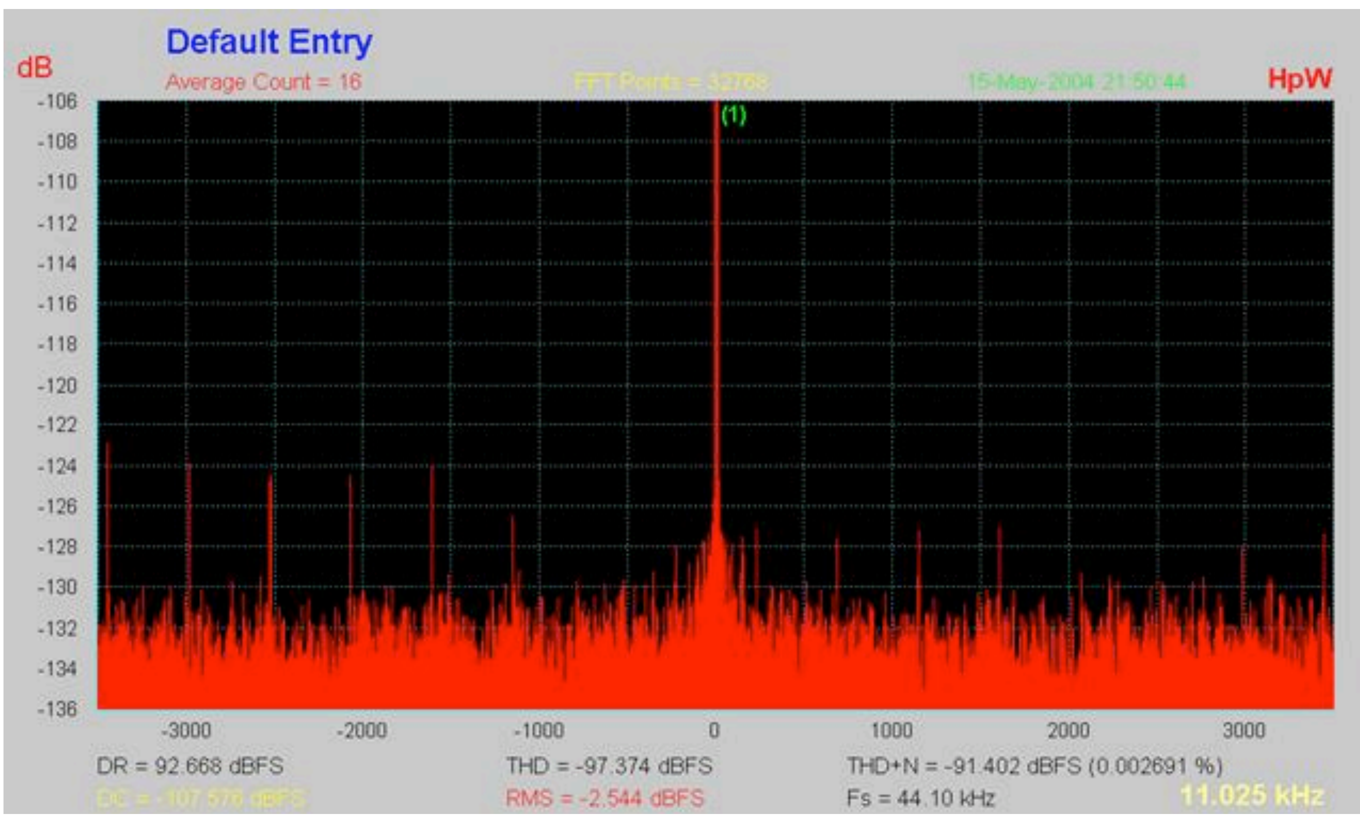


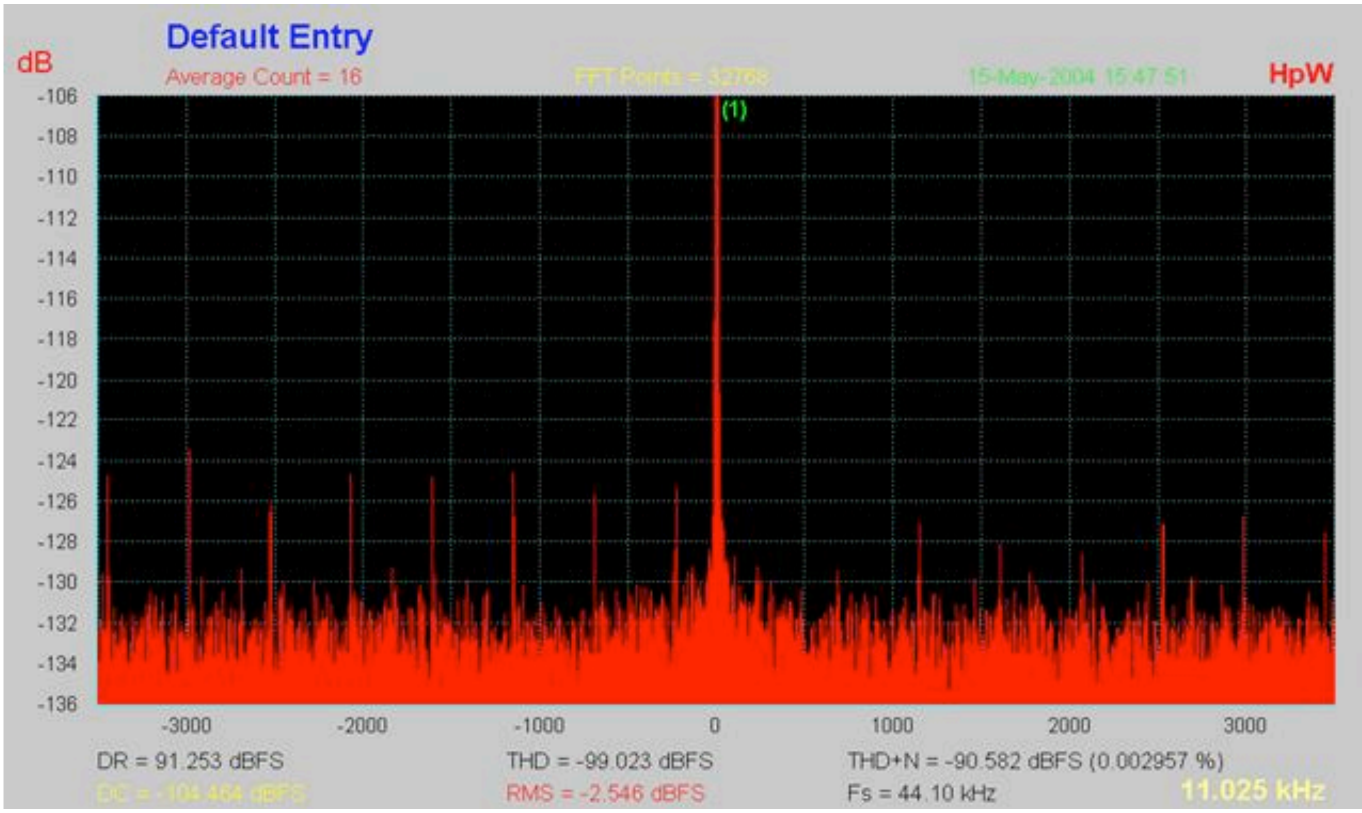
The jitter reduction systems seem to work perfectly. We tested their effectiveness in two ways.

First, we verified the behaviour of the Medea with different digital input flows, from the soundcard and from a low cost modded CD player.

Here you can see from top to bottom the jitter spectra from the CD player (measured jitter 270ps), from Medea driven driven by the CD Player output and from the Medea by the soundcard. As you can see, the jitter spectra from Medea seems almost independent from the source!!!

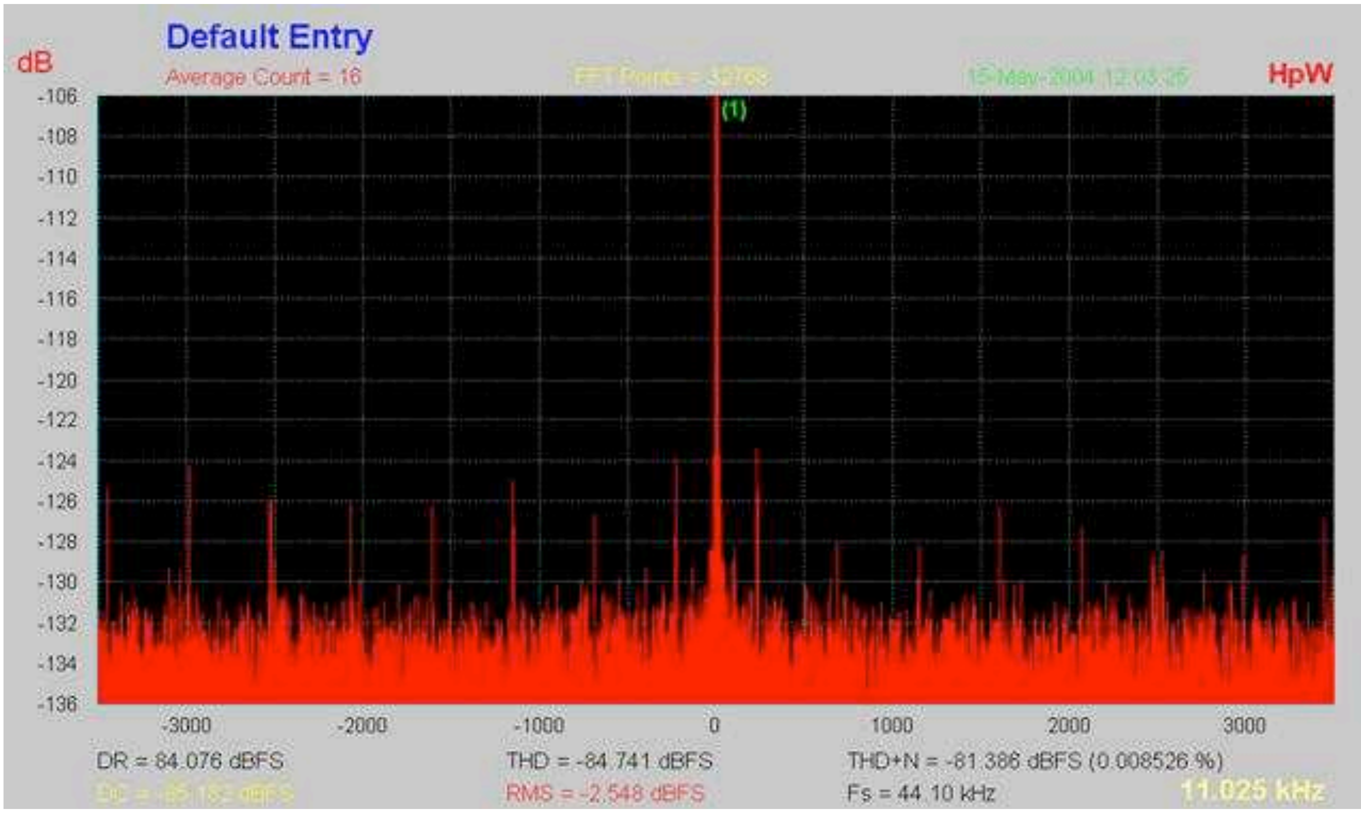

Default Entry
dB
Average Count = 16
HpW
(1)
-106
-108
-110
-112
-114
-116
-118
-120
-122
-124
-126
-128
-130
-132
-134
-136
-3000
-2000
-1000
0
1000
2000
3000
DR = 93.498 dBFS
THD = -99.704 dBFS
THD+N = -92.565 dBFS (0.002354 %)
RMS = -5.076 dBFS
Fs = 44.10 kHz
11.025 kHz

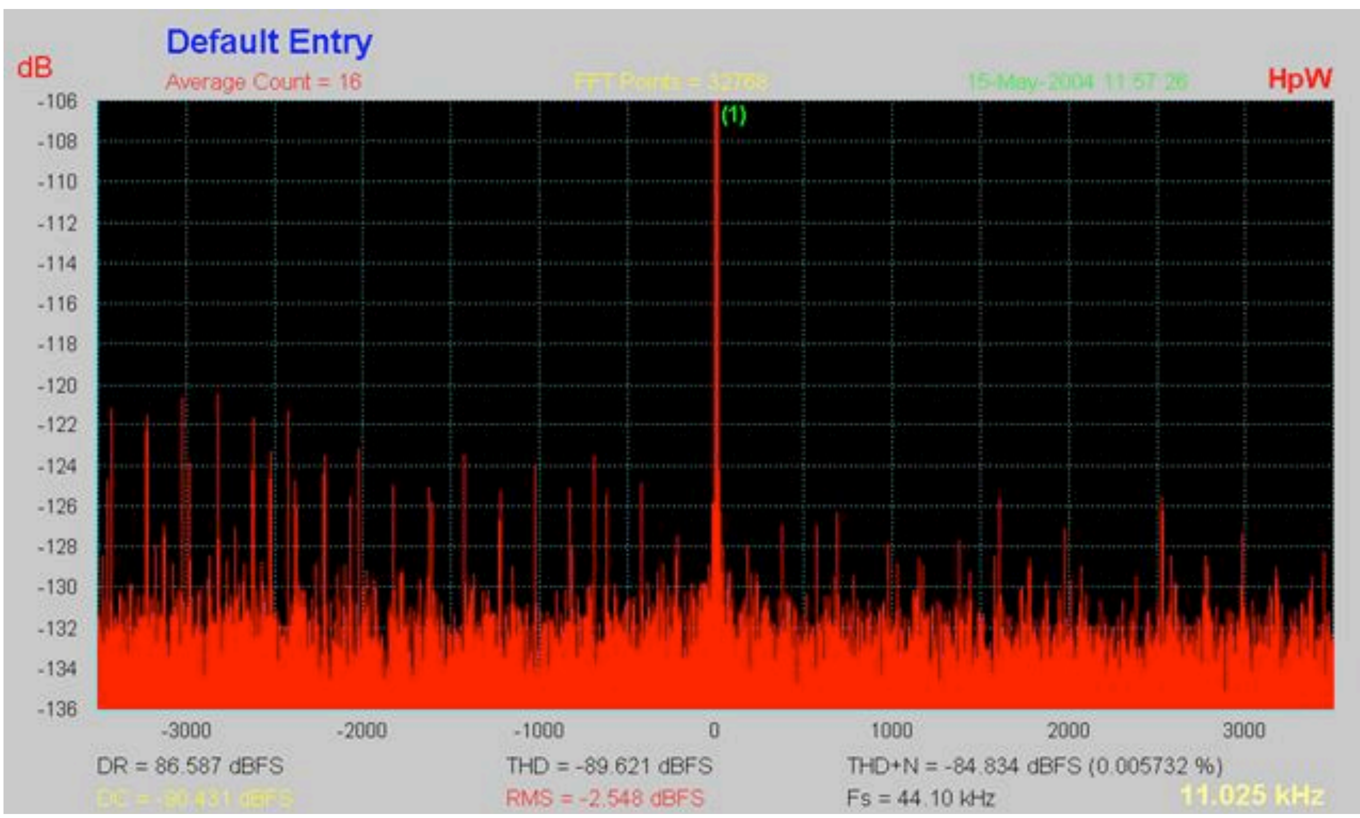

Default Entry
dB
Average Count = 16
HpW
(1)
-106
-108
-110
-112
-114
-116
-118
-120
-122
-124
-126
-128
-130
-132
-134
-136
-3000
-2000
-1000
0
1000
2000
3000
DR = 92.668 dBFS
THD = -97.374 dBFS
THD+N = -91.402 dBFS (0.002691 %)
RMS = -2.544 dBFS
Fs = 44.10 kHz
11.025 kHz



Second, we compared different connections types using the same Medea Input and the same driver, the soundcard.

Here you can see from top to bottom the jitter spectra with TOSLINK and with SPDIF.


Default Entry
dB
Average Count = 16
HpW
(1)
-106
-108
-110
-112
-114
-116
-118
-120
-122
-124
-126
-128
-130
-132
-134
-136
-3000
-2000
-1000
0
1000
2000
3000
DR = 84.076 dBFS
THD = -84.741 dBFS
THD+N = -81.386 dBFS (0.008526 %)
RMS = -2.548 dBFS
Fs = 44.10 kHz
11.025 kHz

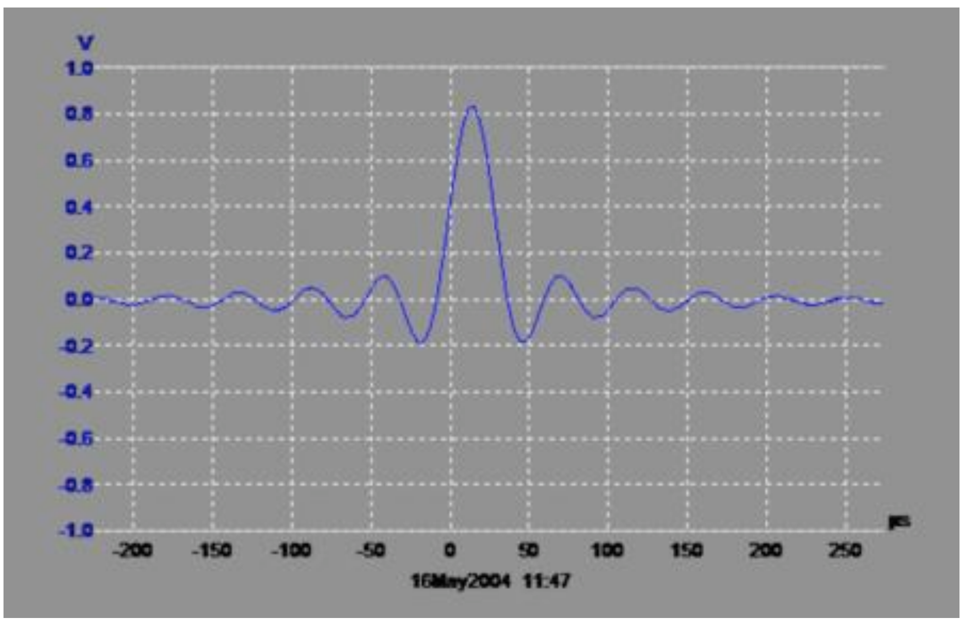

Default Entry
dB
Average Count = 16
HpW
(1)
-106
-108
-110
-112
-114
-116
-118
-120
-122
-124
-126
-128
-130
-132
-134
-136
-3000
-2000
-1000
0
1000
2000
3000
DR = 86.587 dBFS
THD = -89.621 dBFS
THD+N = -84.834 dBFS (0.005732 %)
RMS = -2.548 dBFS
Fs = 44.10 kHz
11.025 kHz

The jitter value measured at input 4 with Toslink connection resulted not so significantly different from the one at the same input with SPDIF connection, but in the first case the spectrum is much cleaner, due probably to problems in the measuring PC grounds. However, it seems evident that here (and in any case where the jitter has been adequately defeated) the use of a Toslink, which is normally considered a major cause of jitter, should have scarcely any drawback, at least for what regards this aspect.

If you want to have a further proof of the quality of this design, just consider that a tight PLL can eliminate external jitter, but at the same time such a clock is normally much more affected by what is called "intrinsic" jitter, in particular low frequency phase noise that appears on the spectrum as a "skirt", a triangular widening at the base of the main freaquency. Widening that is evidently absent in the Medea measures.

Just one note on noise: during jitter measurements, I normally set the display range to -100 - -130dB, because in this way the noise floor just appears at the bottom of the display area. Here to be able to see the noise floor I had to go down an extra 6dB....

To end with, the pulse response seems definitely "traditional".



Sound

As stated, most of the tests were performed at 16 bit and 44.1KHz. In any case, the results were fully confirmed by brief sessions at 24/96.

The first impression you get listening to Medea first time is of being literally overwhelmed by detail. The unit spreads out such an amount of detail that it takes some time to get used to it. So you start thinking it is too much, and where can it come from and how is it possible for a poor CD to contain all that information...

Second impression, bass extension. It goes really low, as low as my 20Hz 'speakers can get (and from the measure I can say even far lower, -3dB to 1-2Hz...). The message is complete, and at the same time very tight, to the extent it seems reduced, but it is only so much articulated that you are simply not used to such experience. First class is diminutive...

In facts, with the proper program you can really feel the strength of the bass: full, but still perfectly detailed.

Everything is precise, detailed, but never harsh, unless the CD really is. And what is surprising, is that Medea is an absolutely true and faithful player of what is on the media, a real purist in this attitude of correctness at any cost, but at the same time it is not so merciless with less fortunate records. In my experience this is not so common, normally the most correct players make disasters out of bad CDs, showing perfectly all their limits: Medea seems instead to expose their naked limits, but also their most hidden secrets, so that the balance is not as negative as in other cases. A very positive attitude, overall.

With such a detail and precision, presence and sound stage are again exceptional. The feeling is to have the players just in front of you, not so far away, I would better say just nearby, a few steps from your armchair, but this upfront presentation is again completely different from anyone else I ever experienced: in facts, while normally the sound stage is wide but everything seems splashed on the wall in front of you, as though the players were only thin silhouettes, stamped on a flat stage, here the players maintain they three-dimensional placement in space, their depth, their correct and realistic size. Something very near perfection.

I note that I did not mention transparence. It's correct. Transparent can be something that remains between you and the music: here you feel complete freedom from any veil, any vapour. It is not transparent, it is not, and that's all. It is like having a direct connection with music, with nothing in the way.

In my view the fundamental character of Medea is the ability of reproducing the truth. Something for sure inherited from its pro ancestors and predecessors, like the presentation, typically "monitor".

Don't ask Medea to hide anything or even to make anything more acceptable: it's not its job; it will render you exactly what has been recorded, nothing more, nothing less. But even in less acceptable recordings you'll find out so many new details that you'll not be able to avoid appreciating them.

As expected, and as stated in the accompanying documentation, the digital interconnect effect on sound is very scarce, but SPDIF electrical connections seem a little in advantage on Toslink.

Issues

This is probably the unit nearer to perfection I have ever reviewed. I like everything of it.

I really find nothing to comment about, apart the price. Yes, it is simply outrageous.... and, what's worse, perfectly in line with the price of its natural competitors, the best units in the world.

If I really have to find something wrong, well, OK, I am ready (you know, I am not hard, simply impossible to please...). But do expect neither big nor sure things.

I just report the following for completeness. I had the faint suspect that the sensitivity of the SPDIF (unbalanced) electrical digital inputs might be lower than the standard one. Using one of my transports, which has a tweaked digital out with very low output level, just over the minimum standard level (0.5Vp-p), I had problems in locking. But the problem never presented again with any other transport or digital card I used. Unfortunately, I was in a hurry to complete the tests and I had no time to indagate. Weiss seems very surprised by this problem, I was too, so I really tend to think there might have been a problem somewhere else in the setup.

The second is more of a suggestion for the builder than a real issue, and a suggestion that is reasonable only in front of the price tag. When you switch over to a new input, there is no other way to be sure the connection is correct than to wait (few seconds) until the unit locks up. It would be nice to know immediately if the unit is trying to lock or there is no signal at all at the input.

Conclusion

What else can I say? If you can spend the money, go and listen to it. If you cannot spend... try to listen to it in any case. As any jewel, some caution is required: in particular, a perfectly balanced and neutral system is absolutely necessary, as any imperfection is amplified by the huge amount of information that is delivered, as far as to transform music reproduction into a caricature.

The biggest risk, for the few fortunate that can afford it, is not to be able to understand, to bear its outstanding introspection, its completely different presentation, its incredible precision and detail. This would really be a pity, because these qualities are so rare, but in my view it is an actual risk, as it really takes some time to get used to them.

The brain seems momentarily overwhelmed by such an amount of information. But after a while you'll suddenly realize that everything is as it should be: detail, music, sound stage, impact are exactly as you have always been dreaming of, and the presentation as a whole is even more involving than you can imagine.

So, do not loose any opportunity to listen to it.

But please, if you cannot afford it, after listening, do not come back and accuse me to have robbed you of the pleasure of listening with your own system. I have already enough problems with mine, now...

I would like to thank again Daniel Weiss and Stefano Zaini for the help and the pictures.