DAC Lynx D60 module.

About 10 years ago, or a little more than that, back accidentally fell into the hands of a DAC chip of the then little-known Japanese company Asahi Kasei. Judging by the description in the reference material, this DAC did not represent anything special and in many ways resembled the products of the notorious Cirrus Logic, which never differed in sound quality. For some coincidence, I urgently needed to make a simple DAC, without any frills - a reliable and easy to manufacture design guaranteed to work in inept hands. It is on the basis of the very chip - AK4394 that the DAC with SPDIF input was quickly manufactured (receiver - DIR1701). Of course, I did not expect miracles from such a mediocre design, and what was my surprise when the sound of this DAC when working with the Sony 559 player turned out to be in many respects comparable to much more serious products that cost quite decent money (PS Audio Ultralink + Sony 559 and Meridian 508), not to mention different versions of DACs based on Cirrus Logic microcircuits. Several experiments comparing DACs based on CS chips and similar AKs showed that the sound of the Asahi Kasei DACs is very different from that of the Cirrus Logic DACs, without scintillation, vivid synthetics and fatigue of the latter, but at the same time maintaining high resolution, clarity of detail and good drawing a scene. In addition, for myself, I noted the calm and even sound nature of the AK4394 and AK4395 chips that I had at that time, and took note of the products of this company as very, very successful ...

10 years have passed, and at the High End 2011 exhibition in the glorious Bavarian city of Munich, among a considerable number of low-, medium- and good-sounding devices - signal sources, my attention was drawn to the universal CD-SACD player Esoteric K-01, which stood out from the total mass exclusively natural full and balanced sound. I went to the Esoteric booth several times, and every time I turned on this unit, even despite the failure of the exhibition installations, I heard its wonderful “voice”. A more detailed acquaintance with the product made it possible to find out that the digital-to-analog conversion in this device is carried out by the "top" Asahi Kasei - AK4399 microcircuits. And then I remembered the events of a decade ago, when the inexpensive DACs of this company showed an excellent result among the then available Delta-Sigma converter circuits with built-in output filters. Upon my return home, we carefully studied the reference materials for the latest Asahi Kasei DACs, and after a little discussion decided to make a universal DAC using such DACs. Initially, we considered two versions of the microcircuit - the AK4397 and AK4399 DACs, but a number of not very significant parameters, at first glance, unambiguously convinced us of choosing the 4399s.

Due to the recent widespread use of computer versions of phonogram repositories, it is obvious that it is desirable to equip the DAC module with a set of standard and specialized inputs that allow it to work with a variety of signal sources - both the widespread SPDIF, AES / EBU and Toslink, as well as with alternative receivers data - directly to IIS, or (via appropriate adapters) - USB, Firewire, etc. The DAC must be able to work as in the master mode with the masterclock generation with low jitter of its own oscilator, or as a slave taking masterclock recovered from SPDIF receiver. Given these points, it was decided to place the SPDIF receiver on the same board as the DAC, by running it on the AK4115 chip. Previously, I had little experience using Asahi Kasei receivers. (specifically AK4112), and the impressions of the work of these receivers were not bad, in any case, clearly better than from CS8414. In turn, the AK4115 has much better parameters than older receivers, in particular, jitter (about 70 ps) very low for receivers with an analog PLL and, importantly, does not require the use of external capacitors for the filter circuit of the PLL controller and is very simple to management in the parallel mode. During experiments with the AK4115, its very good quality was confirmed, the subjective sound of the test DAC when working with this chip was no worse than when working with the DIR9001, but the AK4115 supports data reception with a sampling frequency of up to 216 kHz. In addition, these receivers have a number of interesting functions - this is an automatic de-emphasis implemented by the filter directly in the receiver itself and the ability to select the PLL operation mode either with a constant masterclock frequency (e.g. 128fs) or with a constant masterclock frequency (24.576 MHz or 22.5792 MHz - depending on the grid). The presence of the intrinsic function of de-emphasis allows you to simplify the control of the DAC chip using its de-emphase only when working from IIS inputs, and the choice of the PLL masterclock formation mode allows you to optimize the DAC clock frequency depending on the type of reproduced material. Since in the “parallel mode” AK4115 is able to work with one of 4 inputs, this is exactly how the number of SPDIF-compatible inputs was adopted. The number of IIS inputs was initially also assumed to be four, but for dimensional reasons and reasonable sufficiency, we limited ourselves to three, each of the inputs can be independently configured to switch the sampling frequency base (44.1 / 48 kHz), turn off the masterclock output and its frequency (768fs or 384fs) Also, for IIS inputs, muting and de-emphase signals are independent, data on these modes coming from the selected input are used for operation.

When designing the device, I had to think about which AK4399 DAC control mode to use. With parallel control, the control process is simplified, but the ability to use switching functions such as a digital filter (maximum "squareness" or minimum delay) and determine the "zero" sequence is lost. Based on this, we decided to control the AK4399 modes according to SPI, which required some complication of the FPGA device design and its optimization.

The AK4399 DAC has a voltage output, the primary current-voltage converters and the low-pass filter are built-in. But it looks like they were designed very carefully - the DAC distortions are quite low, and there are practically no traces of interference from the low-pass filter on switched capacitors. Noteworthy is the unusually large amount of current consumed through the analog power circuit - up to 90 (!) MA. This is a very large value, uncharacteristic for other DACs with voltage output. Apparently, the analog cascades at the output of the microcircuit work with large quiescent currents, which determines both their high intrinsic linearity and good frequency properties, and this is important when working with signals with a high level of high-frequency components. During the study of the properties of this DAC, it turned out that its sound is very critical to the organization of analog power. It sharply worsens when using stabilizers with a low intrinsic linearity, in particular, parametric parallel stabilizers on zener diodes, even with a low differential resistance. The best result was the use of sequential compensation stabilizers based on highly linear op-amps, which are separate in both channels: for power supply and for voltage reference.

When designing the analog circuits of the device, special attention was paid to ensuring maximum linearity of the circuit and good load-carrying capacity for output. To ensure the most “comfortable” operation of the critical node - the subtractor of differential signals, signals are sent to it that have already passed low-pass filtering and are independent for each of the differential outputs of the DAC. An additional buffer is used at the output of the subtractor, eliminating the influence of an external load on the operation of the output stage of this unit. To provide analog circuits with “clean” power, highly linear compensating stabilizers with an aperiodic reaction and shunting of power lines with high-quality high-capacity capacitors were used. This construction of power circuits, in contrast to all kinds of surrogate like "Electronic capacitors", guarantees a high intrinsic linearity of power sources, which as a result gives a very clean, detailed and at the same time lively and natural sound.

The circuit diagram of the DAC Lynx D60, developed taking into account all the above points, is shown in Fig. 1. SPDIF receiver DD1 type AK4115 receives the sound stream from one of the 4 inputs selected by its internal multiplexer. The differential input is used to work with the AES / EBU signal and is equipped with an isolation transformer L1. Three single-phase inputs can be used either to directly receive an SPDIF signal, or to receive a signal from the outputs of optical receivers. The operating mode of the receiver is set by the signals on the XS1 connector “AK4115 MODE”. In the mode of receiving SPDIF signals, the clock generators providing the clock signal in the leading mode of the device operation are completely turned off, which is achieved by removing the supply voltage from the generators using a special switching circuit. Such a solution allows the use of ANY quartz oscillators at the appropriate frequencies, since they do not require either an output with three states or complete quenching of the generation in an inactive mode. The combination of generator outputs is provided by external buffer elements and generators power switches. When the device is working with IIS inputs, the SPDIF receiver is put into "sleep mode" and completely stops the generation of clock signals, thereby eliminating interference between clock frequencies.

The IIS signals of the inputs through the buffer input elements DD13 - DD15 are fed to the FPGA, which implements the switching of the IIS bus signals from both external inputs and the SPDIF receiver, generating all the necessary signals for automatic control of the generators and DAC chips and the receiver, measuring the actual sampling frequency the flow supplied to the DAC and the output of data on the current frequency to an external display device. Each of the three IIS inputs has an independent configuration of the frequency base, the resolution of the output of the masterclock frequency, the multiplicity of the masterclock frequency, and the additional multiplicity of the input sampling frequencies. This data is set individually for each IIS input. In the leading mode of the DAC, one of the clock generators is always on. An idle generator is completely de-energized by the power switch and, in principle, does not interfere with the active generator.

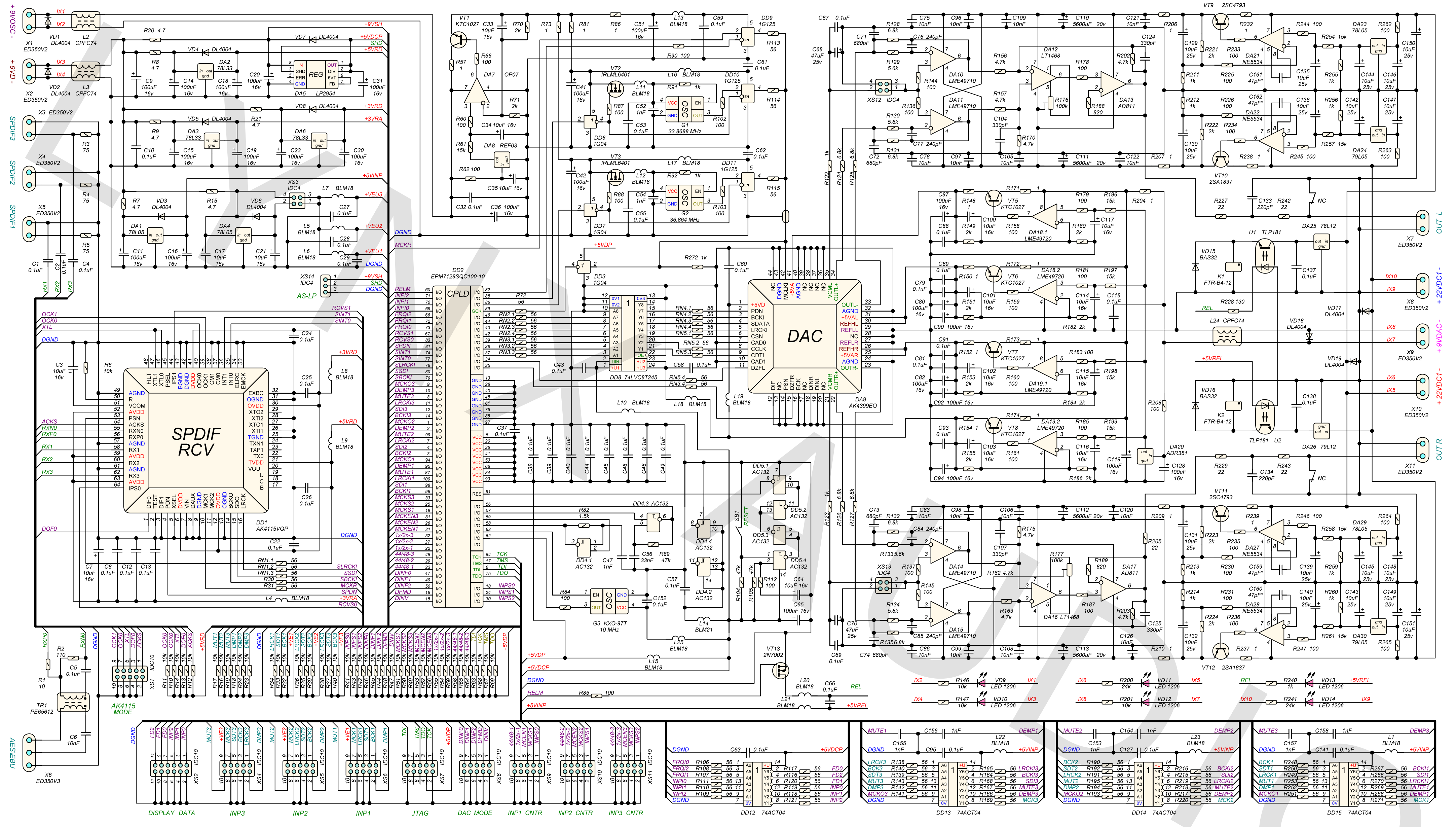

Рис. 1

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UNIVERSAL MULTIFORMAT DAC LYNX D60V1

14.06.2012

The power switch of the generators and the buffer elements at their outputs are controlled by signals from the FPGA outputs.

In the new DAC, as in the three previous models, galvanic isolation of the analog and digital parts of the device is not used due to the large number of problems caused by this solution and the ambiguous results of its application. In order to reduce the penetration of interference from the digital part into the analog one, a 74LVC8T245 buffer microcircuit is used for data transmission, which has independent power supply to sides A and B during data transmission. One side of the chip is powered by + 5V digital power, the second from + 5V DAC power. In addition, to minimize interference, design measures have been taken implemented in the corresponding topology of the printed circuit board.

The AK4399 chip is used in sequential control mode. The necessary sequences for its organization are formed in the FPGA depending on the control signals. The FPGA project and the device circuit are designed in such a way that in the operating mode all auxiliary generators are completely disabled, which are turned on only for the duration of the command to change the device mode. When working from the SPDIF input, it is only possible to switch the type of digital filter (the greatest “squareness” is the shortest delay time) and invert the signal, and when working from IIS inputs, you can additionally select one of 8 input data formats (4 RJ options, 2 LJ options, 2 options I2S). A kind of feedback is used from the DAC chip - according to the data of the “zero” sequence detector, the output blocking relay is switched. The AK4399 is powered by 4 low-noise high-linear stabilizers on the DA18 and DA19 op amps LME49720 type, VT5-VT8 transistors and DA20 reference voltage source are type ADR381.

To exclude abnormal modes, for the DAC chip after any mode switching, a reset signal is generated and only after it a new configuration is recorded. In addition, at the time of switching, recording configurations and transients, the analog output of the device is blocked by relay contacts to ground.

Independent low-pass filters for each of the 4 DAC outputs are implemented according to a multi-loop feedback scheme that provides the minimum value of the common-mode component of the signal at the op-amp inputs and the minimum penetration of the high-frequency components in the input and negative feedback circuits of the op-amp. For filters used op amp DA10, DA11, DA14, DA15 type LME49710. The differential signal subtractors are made on highly linear op amps DA12 and DA16 of type LT1468 with additional output buffering by repeaters on op amps with current feedback, or simply by buffer microcircuits.

The power supply of the analog circuits is carried out from compensating series stabilizers of the type “reference voltage source - noise filter - OpAmp", which have a very low level of intrinsic noise, high intrinsic linearity and an aperiodic transition characteristic.

The appearance of the two options for the assembled Lynx D60 DAC board is shown in Fig. 2 and fig. 3

Fig. 2

Lynx D60 DAC board with op-amp in DIP8

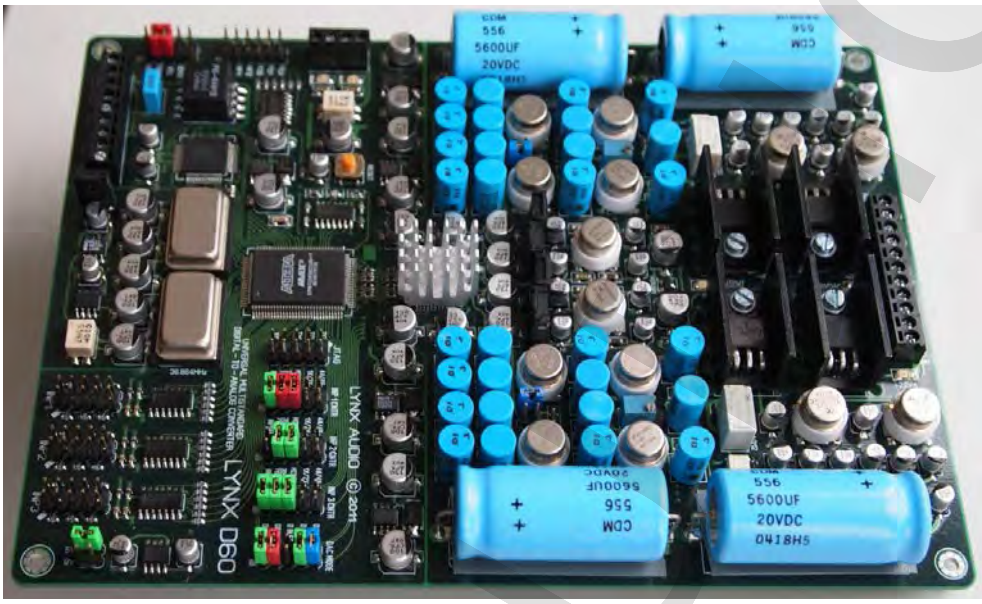

Fig. 3
View of the Lynx D60 DAC board with op-amp in TO-99 enclosures and a radiator for AK4399

The DAC is made on a four-layer printed circuit board 190 x 140 mm 1.5 mm thick. The use of multilayer PCBs made it possible to significantly get rid of the fundamentally unrecoverable disadvantages of single and double layer boards when used in devices with high-speed digital components and a wide dynamic range of analog ones, to create an optimal topology of the ground, signal and power circuits with a minimum level of mutual interference and to realize full shielding of clock circuits, thereby protecting the clock circuits with low phase noise from external noise from one side, and drastically reduce interference RFI from these signals to the analog circuit - on the other.

The board is designed in such a way that allows the use of various types of components. It is possible to install any type of generator in the DIL14, DIL8 and SMD packages 7.5 x 5mm (including with uncontrolled output), polypropylene capacitors in analog filters and power shunts - ERO and EMZ type KP1837 and Wima FKP2, op amp analog circuits in the cases DIP8 or TO-99, power supply transistors of generators and DACs - in TO-126 and TO-92MOD cases, transformer of type PE65812 or PE65612. Analog power electrolytic capacitors can be either axial with housing dimensions up to 40 mm in length and up to 20 mm in diameter (pictured is the Cornell Dubilier 556 series), or SMD with a housing diameter of 16 mm (e.g. Panasonic FK 2200uF x 25V).

In the digital part, passive components of standard size 0603 are mainly used, only some of the sizes 0805 and 1206. In analog circuit resistors MELF 0204 and 0207 are used only. Electrolytic capacitors are Panasonic FK SMD type. Ceramic Power Shunt Capacitors AK4399 - 0.1uF NP0 manufactured by Murata. AK4399 during operation emits a significant amount of heat (up to 1W) and it is desirable to provide its body with a small radiator, which allows to reduce the temperature of the crystal by 15 ... 25 C.

The DAC consumes the following currents from power supplies:

1. +9VOSC – not more than 100 mA
2. +9VD – not more than 600 mA
3. +9VDAC – not more than 150 mA
4. 22VDC1 – not more than 200 mA
5. 22VDC2 – not more than 200 mA

The Lynx D60 DAC power supply is standard (Figure 4). It contains two unstabilized sources with a large capacity of filtering capacitors for supplying analog circuits and three stabilized sources for supplying the digital part of the DAC (+ 9V),

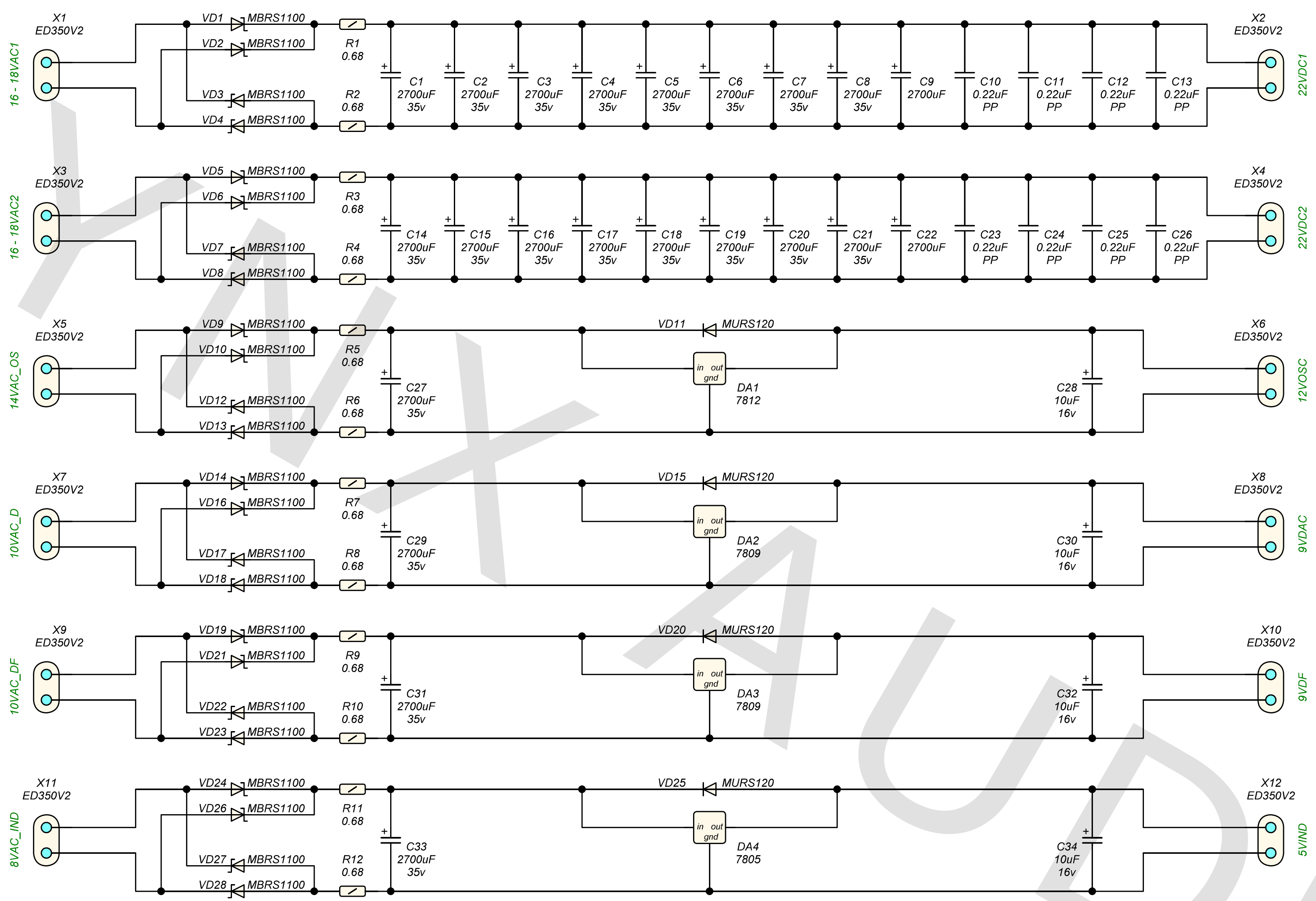

Рис. 4

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LYNX PWR60 POWER SUPPLY

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actually DAC (+ 9V) chips and generators. Another stabilized source (+ 5V) is designed to power the control and display system. To reduce interference, all rectifiers are made on Schottky diodes and are equipped with limiters for the amplitude of the charge current of the filter capacitors. A double-sided board is used for the power source. A view of the assembled source board in which Cornell Dubilier 361 series capacitors are used is shown in Fig. 3. The board allows the installation of electrolytic capacitors with a diameter of 16 and 18 mm, stabilizers in TO-220 cases and rectifier diodes in SMA and SMB cases.

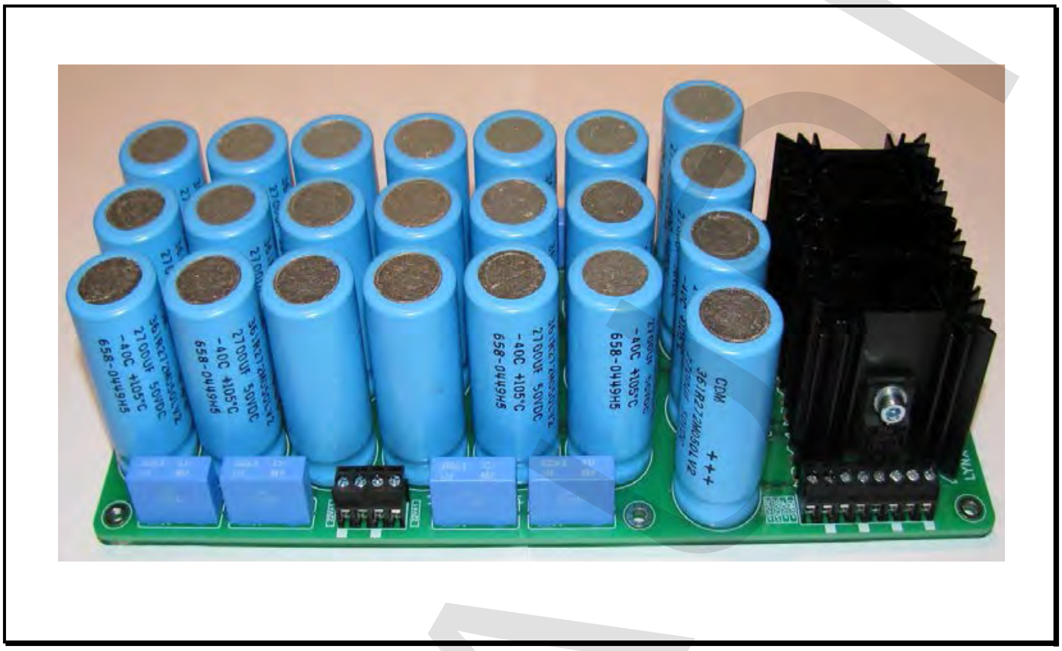

Fig. 5

View of the power supply board for the Lynx D60 DAC

In fig. 6 ... 11 shows spectrograms of the output signal with a frequency of 1 kHz of different levels and a total signal of frequencies of 19 kHz and 20 kHz with a peak level of -0.5 dB from the full scale.

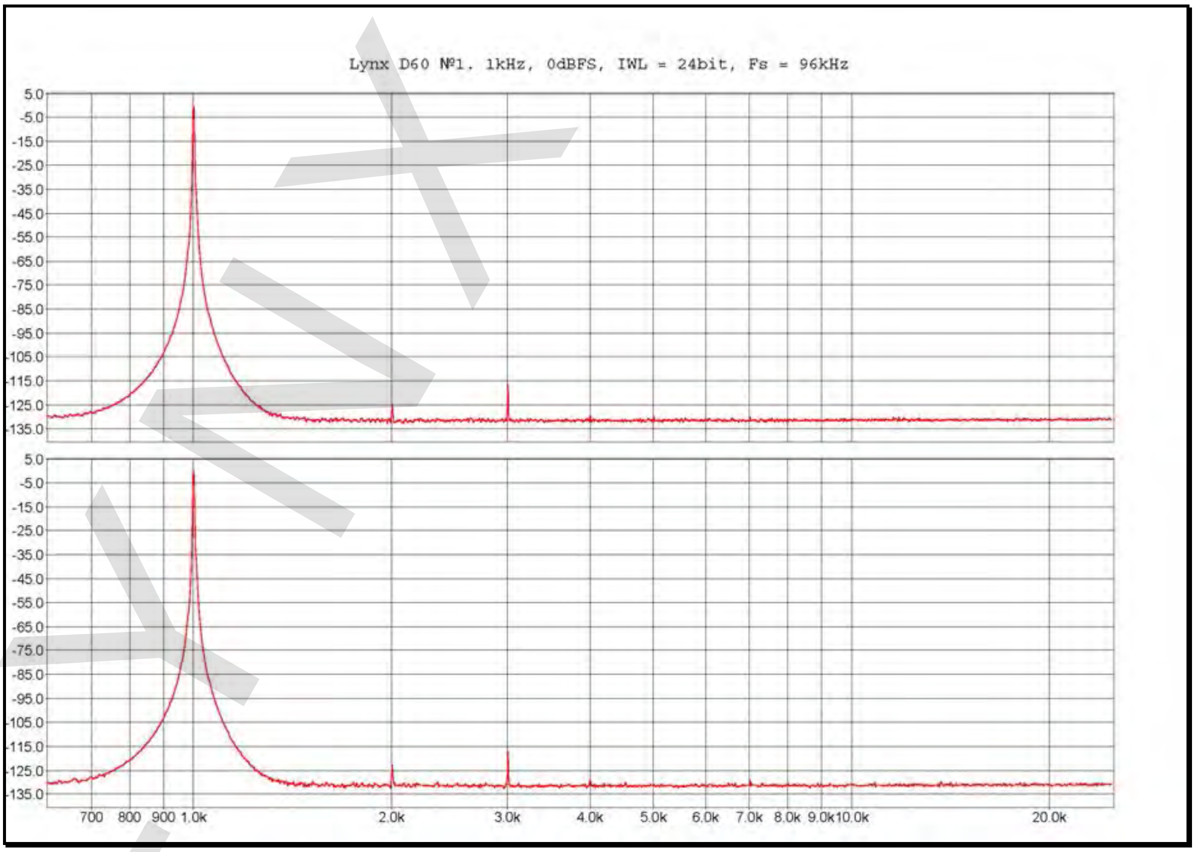

Fig. 6 0dB output signal spectrum

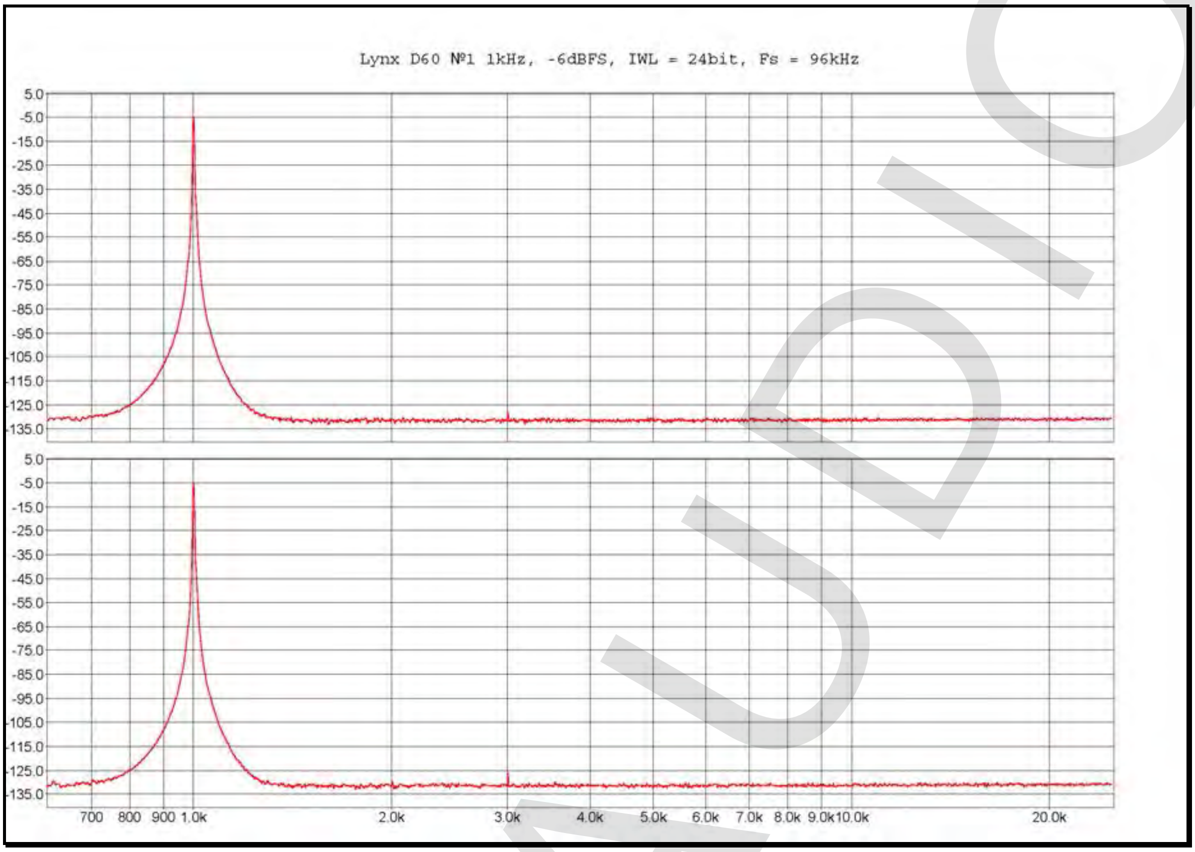

Fig. 7 -6 dB output signal spectrum

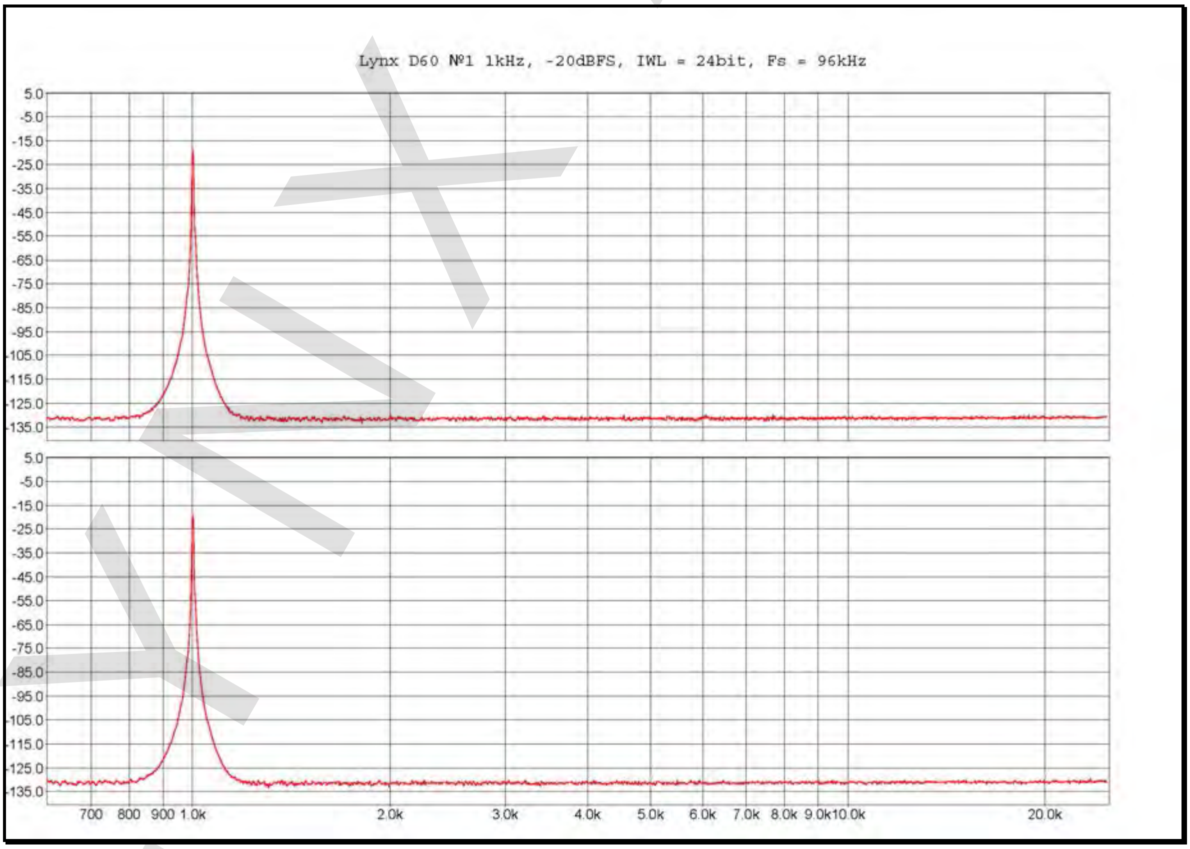

Fig. 8 -20 dB output signal spectrum

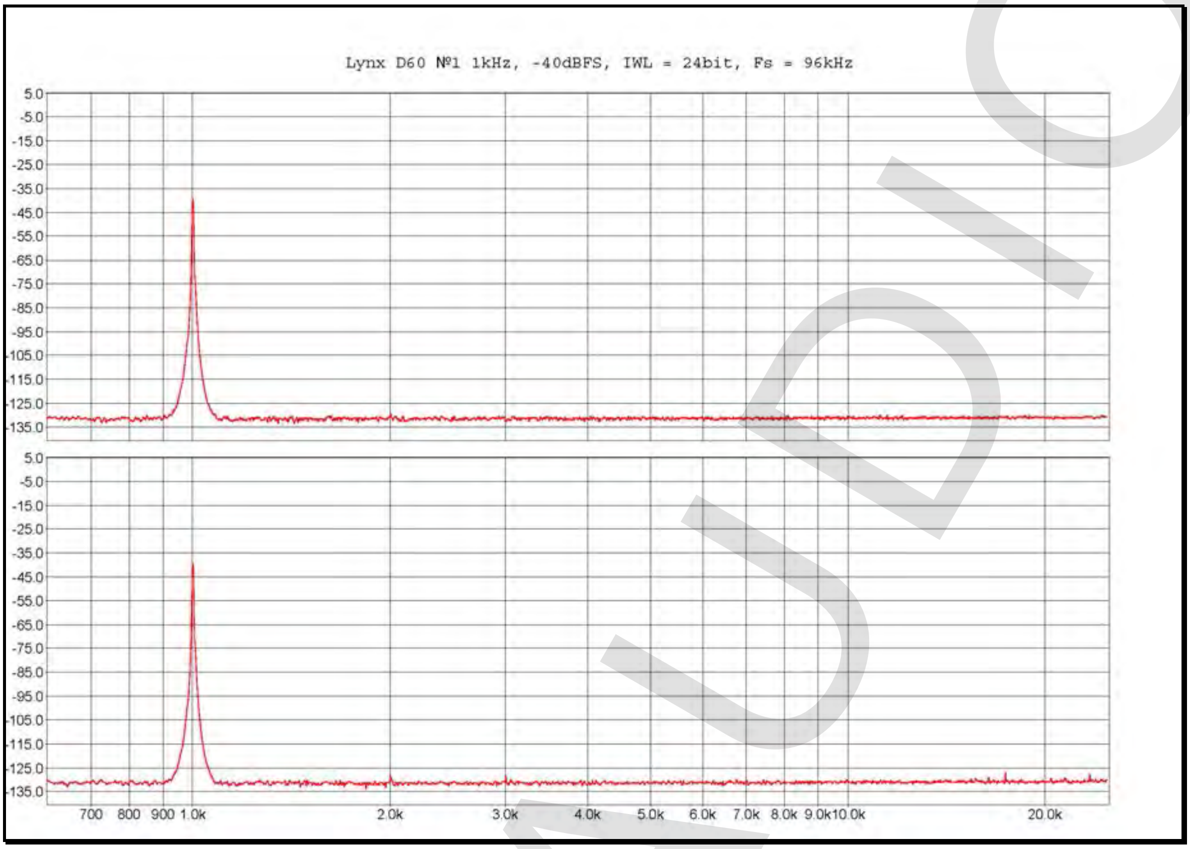

Fig. 9 -40 dB output signal spectrum

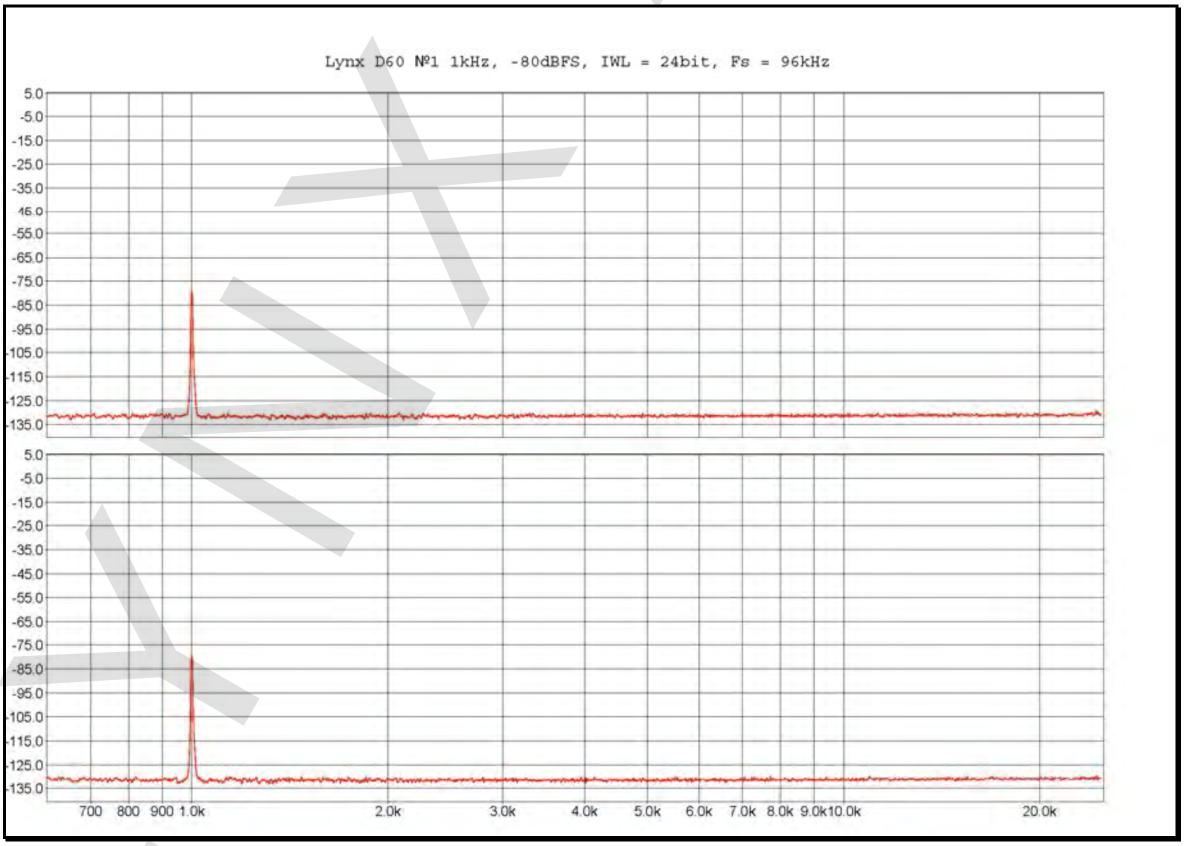

Fig. 10 -80dB output signal spectrum

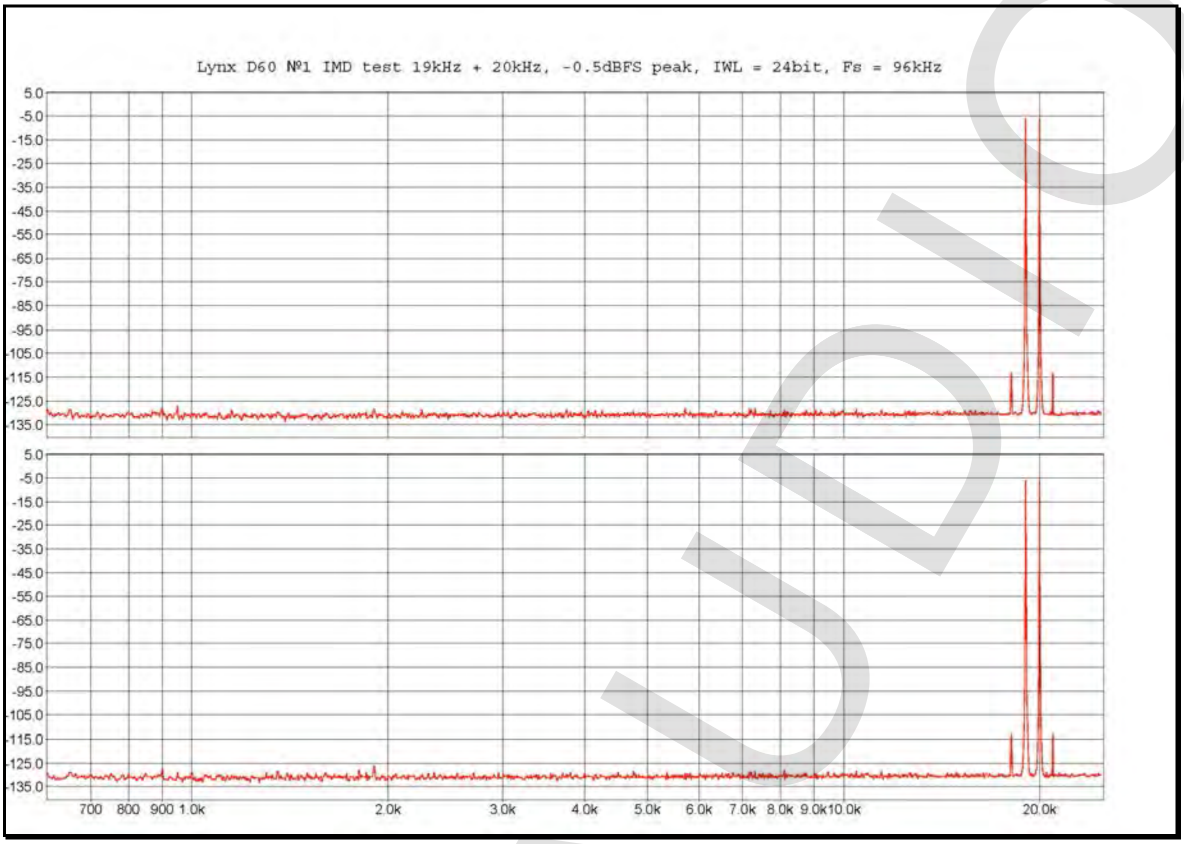

Fig. 11 The spectrum of the output dual-frequency signal 19 kHz + 20 kHz peak level -0.5 dB

A description of the pin assignment of the external connections of the module and control signals is given in Appendix 1.

The first and second versions of the Lynx D60 DAC have the following technical characteristics:

1) rated output voltage corresponding to a full scale conversion, V (RMS)	1.98
2) relative noise level at the exit (at zero input signal), dB	< -130
3) relative harmonic distortion and interference in the 48 kHz band for a 24-bit signal with a frequency of 1000 Hz full scale, dB	< -117*
4) relative harmonic distortion and interference in the 48 kHz frequency band for a 24-bit signal with a frequency of 1000 Hz at a level of -6 dB from full scale, dB	< -128*
5) relative level of intermodulation components, dB	< -115*
6) Noise level in a band of 100 MHz on analog outputs, dB	< -82
7) Dynamic range of conversion, dB	> 128

(*) distortion levels were measured using a selective microvoltmeter

The impressions from listening to the DAC turned out to be very musical. It has a clear, bright and rich sound and at the same time very clearly forms a sound picture. The sound of the device is timbreally correct, very reliably transmitting the sounds and after-tones of natural instruments, such as a violin, saxophone, acoustic guitar, etc. D60 was also listened to by professional musicians and musicologists. According to them, the one on which they played when recording this phonogram is precisely determined against the general background of the instruments. It was also noticed that this device is characterized by timbre and spatial reliability of sound and accurate transmission of the scale of musical instruments, ensuring realism of perception of the phonogram. The DAC fully reveals the potential of not only CDDA, but also high-resolution formats, and all the flaws that occurred during recording cannot be hidden from it.

Lynx D60 easily copes with complex classical pieces performed by large and small compositions, as well as with pop music. Long listening does not cause fatigue, the device can be used for detailed analytical analysis of music recordings, and for easy background playback of any music genres.

In conclusion, we want to express our gratitude to our friends and colleagues, whose goodwill, timely advice and support contributed to the successful process of creating this device, to our families, whose love and patience made possible the creativity of the companies - suppliers of electronic components and sellers of eBay auction, who timely and accurately performed our orders when ordering parts.

Dmitry Andronikov (Lynx Audio)
Sergey Zhukov (Lynx Audio)
Munich - St. Petersburg - Moscow - Malaya Vishera
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