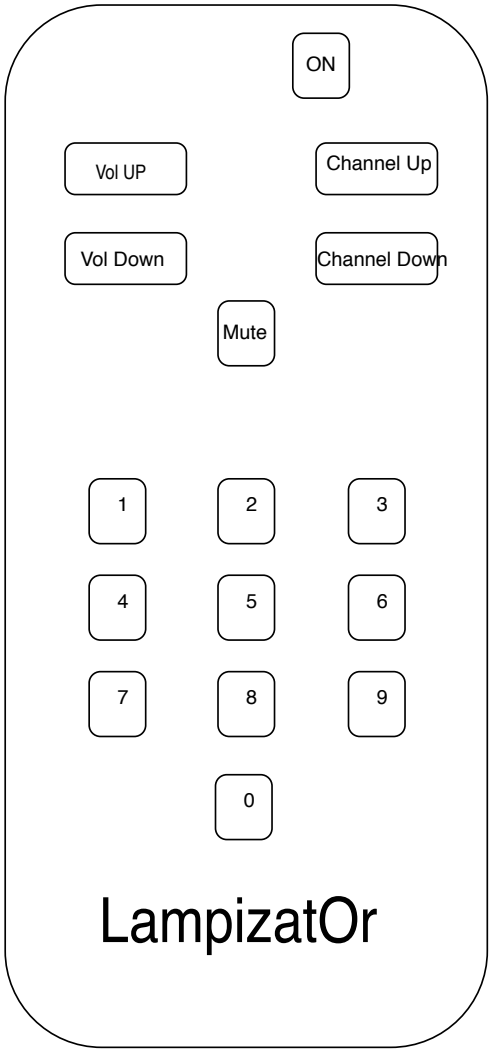
LampizatOr DAC – User Manual

WARNING: as every DAC comes with a 7 days testing period (USA, New Zealand, South Africa, Australia and Poland - ask your dealer first), during this time it is not allowed to open the DAC. The screws are protected with a seal. You have to decide, if you like the sound and you want to keep it. After the 7 days period expire – your DAC is a keeper, and you may open the hood. This does not invalidate the warranty, however – any modifications – no matter how small – invalidate the 5 years warranty. Changes, upgrades and mods must be pre-authorized in writing , even tube change. DACS returned during the test period with the seal broken will not be refunded and will be sent back.

Contents:



The description of REMOTE CONTROL
functions for VC03 systems

(the one with orange display)

ON powers up/down the whole DAC or just the standby function (depending on the DAC level)

Vol UP and Down - changes the volume from complete MUTE (-63 dB) to complete bypass - 0 dB

Channel Up/Down - changes the analog inputs to the tube section. Usually 1 = DAC PCM, 2 = DSD DAC, 3=Line Preamp input 1, 4 = Line Preamp input 2. DACS without DSD and without analog preamp inputs don't have this feature enabled at all. Also - use numeric buttons 1,2,3 and 4.

MUTE does just that - mute. After pressing again - the DAC will go back to the last volume used. Touching the Vol-Up and Vol Down functions will un-mute automatically.

Numeric buttons 1:9 and 0

The 1-4 buttons were described above and they choose analog “client” inputs of the tube section.

Button 5 is reserved.

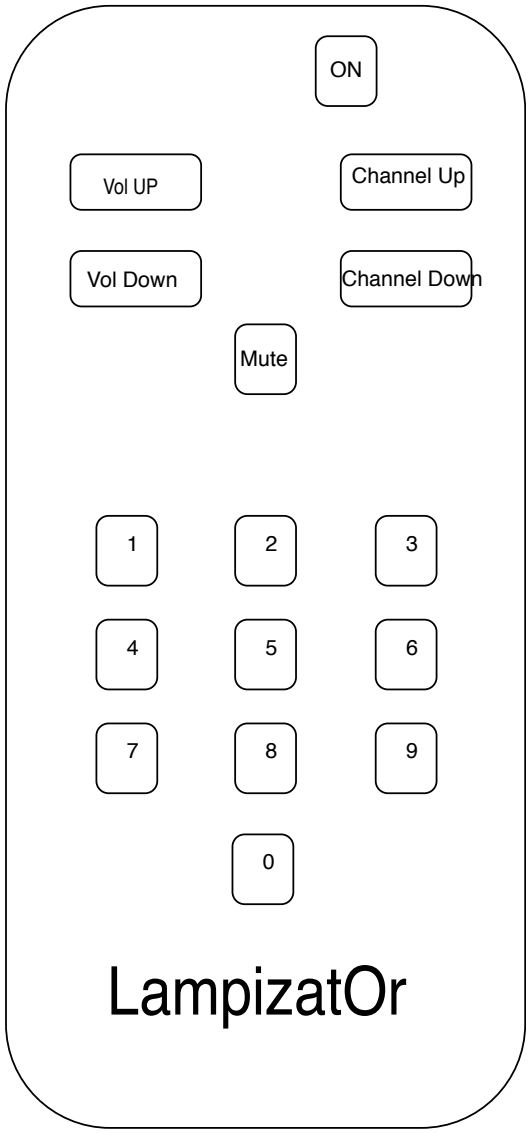
Button 6 as well as 9 can be programmed as a switching relay and can perform any desired relay function. Usually it will be digital input selector like Toslink, AES/EBU or BNC. Note: These button functions - 6 and 9 - do NOT HAVE an alphanumeric representation on the display. You must “trust” the action was performed (you will hear a relay click). Pressing the same button again reverses the action.

Button 7 and 8 - this is a pair which switches the relay inside. It can be freely programmed as any function we can imagine by a relay. In 99% of our DACs this was used to switch between USB(PCM) input and SPDIF group of inputs. By group we mean all of the bi-phase inputs like SPDIF, Toslink, AESEBU, BNC, etc.

You will see on the display - an alphanumeric text describing the input - for

example 7 = S/PDIF and 8 = USB. You can change the descriptors freely under the SERVICE MENU - the ***Dig. Input Name*** section.

To choose DSD - always just press button 2



The description of REMOTE CONTROL
functions for DANZUP systems

(the one with blue display)

ON powers up/down the whole DAC or just the standby function (depending on the DAC level)

Vol UP and Down - changes the volume from (-63 dB) (not complete MUTE however) to complete bypass - 0 dB

Channel Up/Down - changes the analog inputs to the tube section. Usually 1 = DAC PCM, 2 = DSD DAC, 3=Line Preamp input 1, 4 = Line Preamp input 2.

Also - use numeric buttons 1,2,3 and 4.

MUTE does just that - mute. After pressing again - the DAC will go back to the last volume used. Touching the Vol-Up and Vol Down functions will un-mute automatically.

Numeric buttons 1:9 and 0

The 1-4 buttons were described above and they choose analog “client” inputs of the tube section.

Buttons 5,6,7 and 8 are reserved.

Button 0 as well as 9 are fixed and programmed as a switching relay and can perform any desired relay function. Usually it will be digital input selector like USB, Toslink, AES/EBU or BNC. Note: These button functions - 0 and 9 - do HAVE an alphanumeric representation on the display. Pressing the same button again reverses the action.

Button 0 and 9 - in 99% of our DACs this was used to switch between USB (PCM) input and SPDIF group of inputs. By group we mean all of the bi-phase inputs like SPDIF, Toslink, AESEBU, BNC, etc. You will see on the display - an alphanumeric text describing the input - for example 0 = S/PDIF and again 0 = USB. When in S/Pdif the 9 toggles between two of them like Toslink and RCA SPDIF and is described o the screen as Si1 and Si2.

To choose DSD - always just press button 2

A quick guide to a smooth start

THE RING BUTTON
In all DACs up to serial #500 the ring button used to be just the mains power switch. Recently we started using the ring for other purposes, namely - for whatever we think will be most useful (in non-volume DACs).
If your DAC is a non-volume type (without display) the ring button can be:

1. Power button if there is no power toggle at the back right next to the IEC AC power inlet socket.
2. USB to NON-USB switch - in DACs which don't have DSD but do have USB.
3. DSD to non-DSD switch on dacs equipped with DSD separate engine. DSD must be selected by hand.
4. Other, but only if agreed with the user.

VOLTAGE: All DACs are shipped with the voltage of MAINS according to the country of destination. If you bought the DAC second hand and you are in different voltage zone - the DAC can be converted by a qualified technician. There is no switch or anything like that. We need to change transformer windings arrangement of the primaries. And do it on numerous transformers - sometimes 4 or 5 per DAC.

TOGGLE SWITCHES
All toggle switches are safe to use even if by mistake. Dont be afraid to throw a toggle if for example you get no sound while you think you should.
We orient our toggles physically so their position and orientation of the lever corresponds to the placement of the respective input or output.
Example: if the toggle is marked USB-SPDIF - the lever towards USB meand USB is selected, lever towards the RCA means RCA is selected. Same for Toslink, AESEBU, BNC etc.
The USB selector - selects USB against "all other digital inputs". Only then "all others" can be individually selected by the respective switches. USB selection to USB overrides all other inputs. Similarly, DSD is higher on the hierarchy ladder and DSD choice overrides USB, and all others.

Introduction

Thank you for choosing Lampizator DAC. We created it with huge research effort to deliver not only world class musical performance, rivaling the most expensive DACS money can buy, but also to offer very long life of the product. Simply speaking – if you adhere to some basic precautions listed below – the product should last a lifetime and hopefully in this period – will never be outperformed by a competing product.

Whose lifetime one might ask – well – let's not go into details – enough to say it should work flawlessly for the foreseeable future.

The DAC should be future-proof. Shall we ever launch a major upgrade to the digital part – you can get the upgrade at very reasonable cost. Shall you decide you need some added features – you can also get them at reasonable cost anytime in the future. Be it AES/EBU input, Toslink input, BNC input, XLR outputs, volume control, information status LEDs – any option you initially forgot.

We can't be 100% sure, but it is extremely unlikely that the market and the industry in the future will embark any technology of music storage faster than 192 kHz and with more resolution that 24 bits. We already hit the human ear limits, not to mention the real needs of mass consumers (MP3).

Data formats

The DAC is capable of automatic recognition of all sampling rates from 44 to 192 kHz and bit rates from 16 to 24. Since few if any transports offering SPDIF format of the 192 kHz exist in the consumer market, it is hard to guarantee the operation but on the professional ones which we tried – it worked. From our experience the transmitters of SPDIF are incapable of making good square wave over 48 kHz, so if you play a 192 kHz file, be aware that on one hand you "play" more detailed data, but at the same time your signal is waaay more distorted so at the end of the day for this reason alone it may not be worth it to chase the hi-rez rabbit.

If you use USB connection, all our DACs (except Big6) will play up to 384kHz and 32 bits. This theoretical limit does not imply that you need RECORDINGS of that resolution, which don't exist by the way, but that you can use up sampling to play regular files. We however listen to all recordings at the resolution settings they were recorded.

By PCM files we mean all known file formats like: MP3, MP4, Aiff, Flac, WMA, WAV, Ogg, and many more less known types. PCM abbreviation stands for pulse code modulation.

DSD

Direct Stream Digital, also known as DSD format - this format is not new as many people think, it is as old as digital but it wasn't used for consumer audio or home audio - before. It became very popular after 2010 and continues to make its way into our homes. It is VERY different than our well known PCM format as found in our CD files, MP3, FLAC or WAV - AIFF. It encodes the music in the data stream differently, looks differently and sounds differently. It is the format in which the SACD discs were recorded and a format in which the analog master tapes were backed up by record companies. It is currently the format in which the master recordings are made in record industry.

For this reason we decided to use SEPARATE, no compromise electronic board to handle DSD. It is not common with PCM processing.

Our two "digital engines" - PCM and DSD share common enclosure and the user must switch from one to the other by hand.

Audio volume level

Tube technology allows us to set practically unlimited volume level at the output, up to 100 x higher than from a normal CD player. We have decided to adhere to one internally set standard: the test tone of 1 kHz at -20 dB produces an output of sine wave 300 mV AC under the amp load of 47K. Thats equivalent of circa 3 V pp. Shall this be inconvenient for some reason – it is adjustable in the range of 0-1800 mV by just one resistor change. The test tone is available from me via email in the form of WAV or AIFF or FLAC or MP3 file.

Generally - we prefer the sound of the DAC with high output levels, and most amps don't have any problem with that. A simple potentiometer or stepped attenuator takes care of that. Only solid state chip based preamps will saturate and distort thats why we need to know in advance about such solid state chip volume system being driven by the DAC. We will keep then the volume level at the "book" level of 2 V pp. Having said that - chip volume systems and preamps with opamps belong in home theatre (cheap one) and DEFINITELY not in high end.

LampizatOr DAC should not be used with opamp based preamp, no matter how good. Because the op-amp feedback loops will remove the whole joy of music as delivered by the tubed DAC.

AC requirements and overseas operation

The DAC has a multi-voltage power transformer.

In the TOROIDS: Looking from the top – the first wire in the bunch is neutral AC input power and it is always engaged.

For 115 V operation the second wire looking clockwise is the AC phase. For 230-240-220 V operation the third wire replaces the second wire.

If your dac has a serial number lower than 20 – the second wire is formed by a twisted pair. For 230 V this pair is joined together.

It is not necessary, but advisable that the power cable is a quality one, not simply a computer cable. It is also advisable to use some kind of AC filter – in many cases this brings nice results. Generally under-filtering is better than over-filtering.

The EI type transformers have their primary windings on the top. The two external of the three pins are 230 VAC, one external and the center = 115 V primary.

The DACs with volume control, the green board which powers the volume unit has two jumpers set for 230 V and in case of 115 V - these jumpers are open and one jumper is added where it is written 115V.

The heat issue

Many people are concerned about the heat inside the player.

We want you to relax about it - that this is NOT an issue. The DAC operates well below half of its maximum allowed temperature. Tubes are DESIGNED to be hot, this is their very nature. Thats why they have internal heaters and when they are not at optimal operating temperature – they sound bad.

The other components are guaranteed up to 105C and we are expecting no more than 45 degrees Celsius in the air inside the DAC.

Our only advice is do not heat the box additionally by placing it - for example - on top of a hot class A amplifier. Give it some space around to allow free air flow and adequate cooling. Do not cover it with blankets or mats.

Optimal placement

Apart from the heat issue as described above, the DAC has no special placement requirements. Just remember to keep the S/PDIF cable not longer than 1,5 m (5 feet) and RCA chinch cables – not longer than that either. Since tubes are microphonic, they hate vibrations. Therefore it is forbidden to place the dac on top of the speakers or a sub. Choose least vibrating location, preferably one foot behind the plane of the speakers.

Power on-off cycle

The tube lifetime, almost like the life of a car engine in polar climate – is determined largely by the on-off cycle. The heat expansion coefficient of the glass is so much different than that of the metal, that the air-tight seal of the metal pins can leak oxygen inside the tube and eventually kill it. Even if it is just one molecule per day. So in other words it is better to keep the DAC always on, than to switch it on and off more than necessary.

The lampizator DAC (level 3 and up) has a slow start feature which brings the high voltage supply gradually up, at the rate of two- to five volts per second. The PSU reaches 300 V DC after 90 seconds. This helps to extend tube life. The DAC is also equipped with voltage down feature (bleeders) which reduce the power voltage upon switch-off at roughly the same rate.

On top of that – the tubes are operated always around 25% of full nominal power, which greatly increases their life expectancy. Combining all the factors together, the tube lifetime should be anywhere between 10 and 20 years, assuming the player is switched off only once per day, for the night.

Cabling and cable handling

Just to be sure that we know what we are doing:

- AC cable can be freely plugged and unplugged during operation. It is OK for the DAC but NOT OK for the amplifier and speakers. A loud thump may appear after switch off. Please turn your volume fully down before switching off the DAC.
- SPDIF cable should be plugged and unplugged when the transport is powered off. The DAC can be on. However doing it on „hot" when all is working – is not dangerous for the DAC as long as the AC power supply has the GND for all products (DAC, transport, amps).
- Signal cables can be plugged / unplugged with the amplifier volume turned fully down.

Please use a decent AC cable. We suggest spending around 100 Euro for a good AC cable, not more but not much less. The free AC "computer grade" cables are not good enough for serious audio.

Please use a decent digital interconnect. In our DAC it is completely unimportant what is the wave characteristic impedance of the cable (the famous 75 Ohms). Just use the cable that sounds good to you. Analog as well as digital interconnects can be tried. Best results are obtained with silver cables. Let your ears decide, not specs of the cable.

Tube rolling and replacement

W took an expensive and painful decision to sell the DAC with the best tubes we can find in consistent sustainable supply. Therefore we feel you should not be tempted to change them for any reason.

If you feel that you MUST try other tubes – we need to pre-authorize it in writing. Otherwise you loose the warranty, sorry.

This does not concern Big7, in which tubes can be rolled freely.

It is possible to order the special version of DAC for tube rolling fans. It will have specially configured multiple sockets to enable the use of almost all existing tubes - from 6SN7GT to ecc83 or e182cc

Here are some practical tips for tube rolling:

1. Tube compatibility- many people ask “is the tube X compatible with Y?” and the answer is of course - it depends. Tubes can have completely different bases but be compatible by parameters and can be swapped by means of an adaptor. A good example are ECC40, and 6SN7GT - different bases but very close parameters.

2. Other scenario is when the tubes have same base (say - noval) but they have different pinouts. So we CAN NOT inter-change the two tube types but we CAN use an adaptor. Same base type and same pinout DOES NOT MEAN that the tubes are interchangeable - best example is cc81 and cc82 - same base, same exact pinout but completely different parameters. Or octal 6SN7 and VT99 - both octal, same parameters, different pinout.

3. Some tubes can have same base, same pinout and same parameters except the different heaters. Best example is ECC82 and 12BH7 - the former uses half heating of the latter. They can be used with a switch or within limited timing or with extra care, depending on the heater arrangement in our DAC. Another example are completely different tubes that miraculously are perfectly interchangeable - like E182CC with 5687.

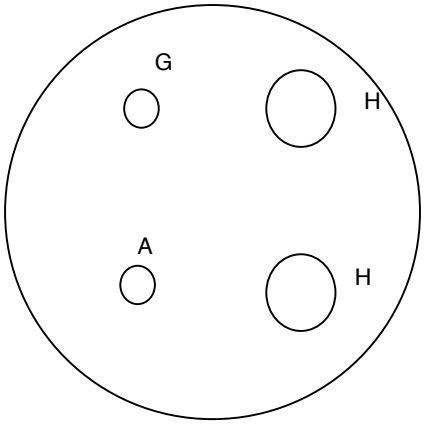
4. DHT triodes used in our Level7 and Big 7 DAC are yet another can of worms. Most DHT tubes have the same base (four pin) and the same pinout (two fat legs are heaters and cathode, two slim legs are Grid and Anode. The problem is that these tubes have completely different heater demands. Our DAC7 is designed to accept ALL KNOWN dht triodes from this group: 101D, 45, 245, 345, 6A3, 2A3, 300B. People keep discovering more and more compatible triode types every month.

How is it possible to run so many triode types in one circuit ? DON’T THEY HAVE DIFFERENT HEATERS ?

Our genius circuit automatically detects the current demand and adjusts voltage accordingly. We added a switch which divides the tubes into two groups: high heaters and low heaters. High heaters are 2A3, 300B, 6A3 and low heaters are 101D, 45, 245, 345.

Our favorite tubes are: (New) Sovtek and Electro Harmonics 2A3 - this is actually the same tube but Sovtek is a cheaper brand.

45 by Emission Labs (New). 245 and 345 NOS from Sylvania, Tung-Sol and RCA. 101D by Psvane WE Replica (New). Nos 6A3 from any US brand. 300B Mesh Black from Psvane.



BOTTOM VIEW

Above: the WE 101D replica triode (designed 100 years ago in 1914.)

Rectifier Rolling

Rectifiers are generally less rolled but many customers report that huge leaps in synergy can be achieved when, after choosing the optimal music triodes, we also choose optimal rectifier.

How can we tell the rectifiers ?
At Lampizator we use generally two sub-groups of rectifier diodes: directly heated and indirectly heated.
Indirectly heated diodes are those which have separate heater coil mounted underneath the cathode. The heater current does not mix with the cathode current. This has several advantages (diodes are smaller, they live longer, they are much more vibration-resistant, and they can be connected in series with many tubes sharing one heater source. They have 5 active pins on an octal base. We use 6X5 (USA) or Soviet 6c5s (6u5C) type. They are very short bottles. We used them on all Amber DACs, on all dual mono Level5 and Level6 and Level6 two-box machines. On all Dual mono Big machines. On 50% of our Level4 DAC as well as on all preamps.
PINOUT: 2-7 is heater 6,3 V AC. Pin 8 is cathode. Pins 3 and 5 are two anodes.

Directly heated diodes are older in design, physically larger, and have 4 pins versus 5 and use 5V heaters versus 6,3.
PINOUT: 2-8 is heater 5,0 V AC. Pin 8 (or2) is also cathode. Pins 4 and 6 are two anodes. Other pins - even if existing - are not connected. To test - just use a meter and check resistance in ohms between the pins. IN A RECTIFIER THE ONLY TWO PINS WHICH SHOW ANY OHM READING AT ALL, ARE HEATER PINS. THE READING SHOULD BE IN SINGLE OHMS.
Some people report back that the directly heated diodes sound better than their indirectly heated counterparts, but this hasn't been verified in any semi scientific way. Generally we expect the directly heated diodes to have up to 400% higher current capability as well as voltage max. It al depends on the DEMAND of our circuit. Some Lampizator tube stages demand only 2 mA in total, some can demand 40mA and more.

We used directly heated diodes in our amps, in one-rectifier Big 5, Big 6, Big 7, and on some Level4 DACs.
Rectifiers compatible: 274B, 5c3s, 5Y3, 5r4, 5U4, GZ34, 5c4s, 5u4c
Rectifiers can be changed safely **during playback**. Just grab them by the base.

To change music tubes you must switch off the amp. DAC can continue to work.

Aging problems

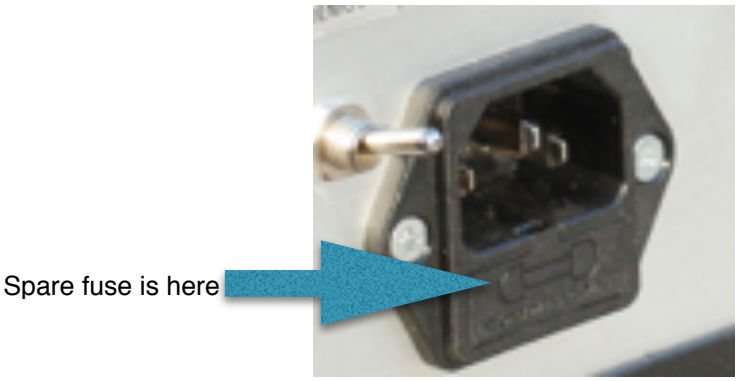
As already explained above, the DAC should age very very slowly.

The digital PCB should last a lifetime. The transformer, the paper in oil caps, the cables, plugs, sockets – should last a lifetime. There are only 4 electrolyte caps which we selected from premium brands and they should last circa 25-30 years. Other than that we suggest to change tubes every 10 years.

So - short of a thunderstrike – we expect no failures or aging problems before 20-30 years.

Fuse Change

The DAC is equipped with a non-repairable 25 mm glass fuse circuit breaker inside the IEC-AC socket at the back. There is also one spare fuse provided in the little drawer removable when changing the fuse. The fuses are 1A (1,6A for USA) and overrated by the factor of 3. Therefore it is impossible for the fuse to blow without a specific reason - a failure inside the player. Consequently, if the fuse burns, it is a signal to send the dac for service and NOT change the fuse. Obviously the second fuse will burn as well.



WE ABSOLUTELY DO NOT ALLOW changing the fuses for any larger size than 1,6A or installing the “audiophile silver bolts” in place of the fuse. Fuses are there mainly to SAVE YOUR LIFE. And we mean that.

Volume control VC03

(This concerns DACs equipped with TAIV volume control modules with display and remote).

The volume control is an extremely nice module, that changes the way we use the DAC. The module consists of 5 elements:

1. Power supply with the DAC power management relay, allowing to switchON/OFF the whole DAC via the remote while keeping the volume module powered. For space reasons some DACs especially Level4 do away with the power supply control feature and provide only volume/remote features.

2. The display - amber backlit 2 x 16 alpha LCD, which is our GUI (a blue display indicated Danzup module - a different brand of the same thing.

3. The microprocessor board with memory, firmware and the chip that controls the relays

4. Relay volume board with resistor ladder. The microprocessor connects the resistors in such way that they form a resistive attenuation L-Pad with 63 steps of logarithmic attenuation. The overall impedance is held at almost constant 30k and the steps are calibrated in 1 db distances.

5. The virtual potentiometer with push action

The module is capable not only of volume adjustment but also of the input selection- both analog inputs as well as digital.

Using the volume module: Press the volume knob down for 3 seconds and the display will show selection of 4 analog inputs . Turn the knob to select the desired input. Press for 3 s again to exit this menu.

The input selection is available from the remote in direct mode - just press digit 1 to 4 to choose input or use left - right cursor buttons.

** in most DACS that we ship, this function of input selection is disabled because the DAC is the only active input. You can bring it back by entering the PROGRAMMING menu.

PROGRAMMING THE MODULE

Press the volume knob for 10 seconds continuously and the display will enter the programming mode. Turn the knob clockwise to go to next menu. Press the knob for 2 s to enter the selected sub-menu. To exit the sub menu and go back to menu list - press the knob for 2 s. To exit programming mode - turn the knob right till you choose EXIT SETUP and press knob for 2 s.

1. Volume step speed: it is default set at 3 , the higher the number the slower is the reaction of volume level to turning the knob. For example, choosing 6 means that it takes 3 full turns 360 degrees to adjust the volume full scale. We like it at 6.
2. Number of inputs: choose a limitation of analog inputs to avoid having to deal with non-existing inputs. The normal DACS have a limitation set to one.
3. Backlight display - choose a percentage of DIMMING - from full light intensity of display that dims AFTER the period of inactivity. Setting 40 % means, that the display will be lit 100% during any operation with knob or remote, but when we listen - the display will dim to 40 % of the full level. We like it on 35 because we like listening in the dark room.
4. IR Learning - you can program any remote control you want to perform the desired actions. For example - if you use a CD transport remote, you can choose two buttons on it that aren't used for CD and program them to be Vol-Up and Vol-Down, so you use just one remote. After entering this sub-menu the display will ask you to press the remote button and you acknowledge the signal code by pressing the knob for 2 s and proceeding to next function. The display will guide you. At the end choose exit.
5. OUTPUTS: the module has 3 freely programmable latching or non latching outputs activated by the remote control. They can be wired to perform a specific function inside the DAC (say reset, change frequency, activate de-emphase or something like that.
6. Logo change: you can change the welcome logo to your own.
7. Alphanumeric descriptors: you can add alpha descriptors to the 4 inputs. the first input by default is DAC. Add Phono, Tuner, TV to the other inputs of analog stereo signals. (The DAC must be equipped with the RCA inputs to use this function).
8. After reaching the end of menu list - the display will show EXIT SETUP. The options we programmed will be written to a non volatile memory and stored forever. There is no limit to how many times we can repeat this operation.

MUTE function: this is useful to use instead of turning the volume all the way down. Available ONLY via remote.

POWER OFF - the DAC will be switched off fully but the remote module will be always alive to enable you to power it ON again.

*** Do not be upset about the C2013 text - it means the date the firmware was written for this module and not the year of manufacturing of the DAC.

0dB operation: at full volume - 0dB the DAC operates as if there was no volume module. This position is recommended for systems with own volume control: preamp or integrated amp. The resistor ladder is completely by-passed.

DANZUP (blue display) remote control

Alternatively to TAIV VC03 control we use DANZUP remote system. The volume adjustment function is THE SAME and the sound quality is the same as VC03.

From the user perspective the functional differences are:

- in DANZUP system the volume dial knob does not have the “push” function. In VC03 the volume dial knob has a push function, and works similarly to the BMW iDrive controller. To enter the hidden functions of DANZUP system, you must press the service button which we locate on the floor plate of the dac. Usually for convenience the secret button is located near the front panel - around the center point , under the ring button. So it is easy to access by index finger.
- Press the button once to change analog input channels (PCM-DSD-Preamp-Line)
- Press button more than once to enter service functions like: enroll new remote control unit, choose dimming backlight function of the display, choose alpha-numeric descriptors of the names of inputs, and we keep adding extra functionality.

COOPERATION WITH THE PREAMP

The DAC with volume control should sound audibly cleaner and more direct without any preamp between the DAC and the amp. The preamp, however good, will veil a lot of the DAC’s natural clarity, speed and directness. If you feel you need the preamp nevertheless, use DAC at the full volume or order your DAC without volume module.

The load presented by the preamp or amp or simply the next analog component that the DAC sees, should be as high as possible. It is measured in kilo-Ohms and 100Kilo Ohms is a perfect ballpark value. More is VERY rarely seen. 47 K is next common value, and it is great too. 20 K is kind of on a low side, but we can handle that. Lower than 20k is bad news. We must configure the DAC with additional cathode follower buffer stage.

The DAC will not be damaged in any way, but at around 10K of load the dynamics of the dac will start to fade away.

Having said that - every properly designed amp or preamp keeps the load value above 40k. And if it doesn’t - we simply don’t choose such amp because it was not designed with audiophiles in mind.

DIGITAL INPUTS

There are three data types that our DAC can read internally: biphase, i2s and USB. The bi-phase can come in many forms, but the most common are:

S/PDIF (sony/Philips data inter face) by means of single ended square wave of amplitude around 0,5 V pp

AES/EBU - the same as S/PDIF but the signal is a mirrored (balanced) pair of square waves around 2,5 V pp (max. 5 V pp)

TTL - just as S/PDIF but 5 V pp

TOSLINK - a fiber optic transmission of S/PDIF producing at the DAC the 5 V TTL electrical signal.

RS422 - it is practically the same as AES/EBU

The i2S is the same as biphase but separated into 4 signals - each carrying only one type of information. Biphase encodes 4 groups of informations in one signal stream. Specifically they are: System Clock, Bit Clock, Left/Right Clock and Data. We can install these four in any type of connector, because there is no standard. Most customers use RJ45 LAN socket or simply four RCA sockets just like in TV RGB.

I2S Pinouts

The JKI2S uses a RJ45 connector to output I2S signals according to the following pin outs:

1	SLCK
2	SLCK GND
3	SDATA
4	SDATA GND
5	L/RCLK
6	L/RCLK GND
7	MCLK
8	MCLK GND

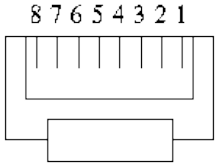


Figure 1:
End view of RJ45 Plug

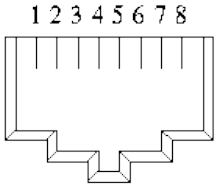


Figure 2:
Looking into an RJ45 Jack

The TOSLINK connection
Is toslink bad or not ? That is the question. Like everything in life - it can be bad or it can be good.

By using own experiments and oscilloscope observations we concluded, that Toslink is not bad and not inferior to RCA SPDIF if implemented properly. Toslink is EXTREMELY demanding about the power supply quality. That's why we build for toslink separate dedicated power supply and with this supply the response is instantaneous and there is no deformation of square wave. Usually Toslink ports are installed in cheap low end gear and the power supply to Toslink is completely neglected. Not in LampizatOr DAC. If you have Toslink in your DAC you can be sure it will sound good and not degrade the sound. Of course providing that the transmitter part of the link is at least semi decent.

NOTE: All Apple products which have headphone output (iMac, MacBook, Power MAC, MAcMini, iPhone, iPad, iPod) - have a secret toslink transmitter hidden inside that port. Just buy the special cable - Toslink Minijack and when placed in the headphone output of an Apple product - will emit light with SPDIF in it. That is a very good way of using MAC computers as transports.

USB playback

USB data requires installation of additional converter module to convert the “packet” data into a steady i2S stream. Our asynchronous converter has internal RAM and two own clocks and own power supply and own power transformer secondary winding.

The USB module requires a driver for Windows to recognize it. MAC OS works without any drivers.

We use three vendors of USB modules:

for Level 6 DACs we use “eXD” - the driver for Windows is available at www.lampizator.eu/USB.zip

For all levels except the 6 and Amber we use **“**Amanero” USB module: the driver is at www.amanero.com/drivers

For Amber DAC and Amber amp we use XMOS driver : http://lampizator.eu/xmos.rar

Do not install windows driver with the DAC USB connected to PC.

USB module can be added as a paid option to existing DACs.

Our USB converter is capable of working with 32 bit files with 384kHz signal frequency.

Only USB2 rated cables will work. The USB standard printer cable will not work.

USB cables with ferrite filters (the “thing” on the cable) will not work.

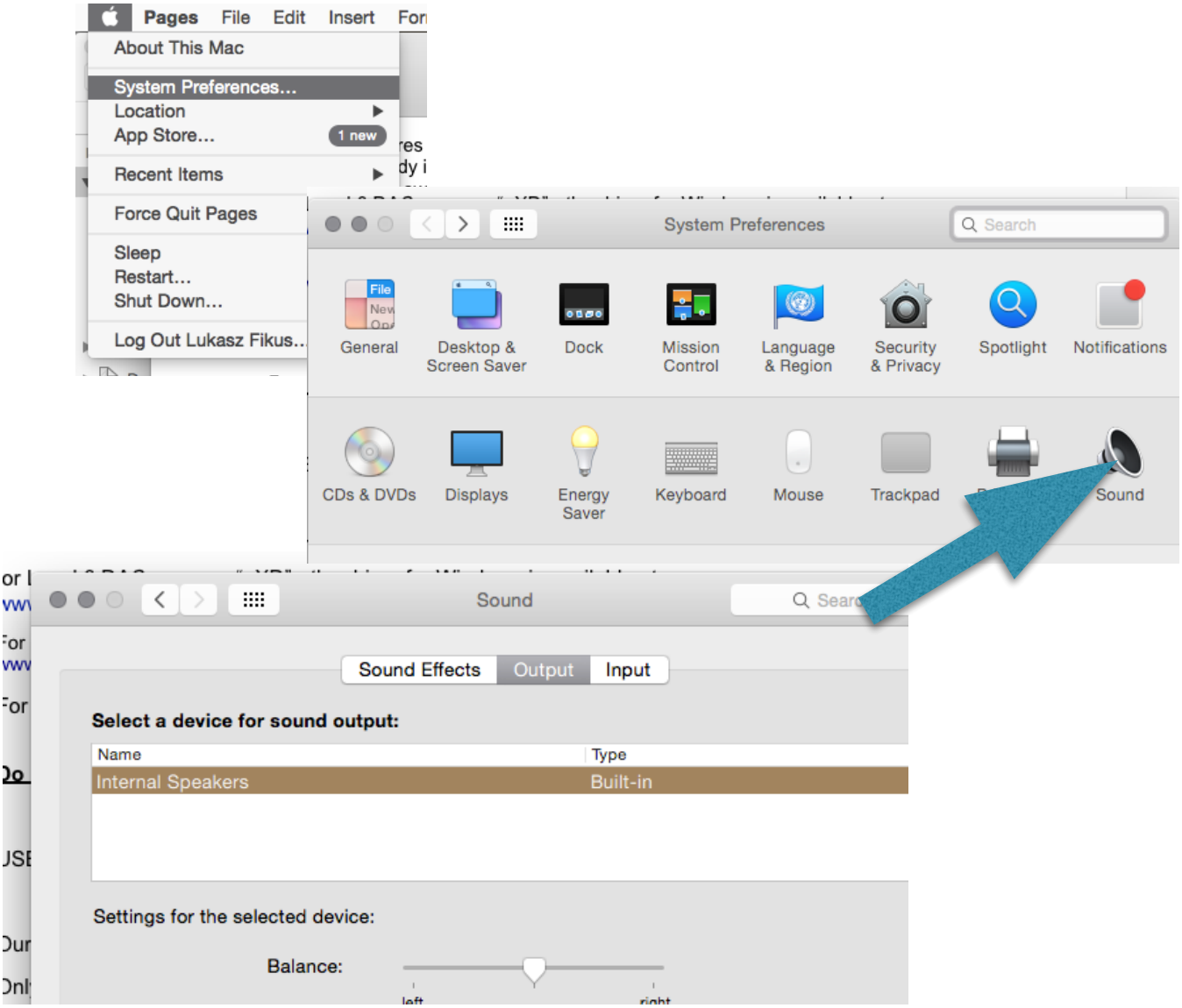
For best results please download the jPlay software to manage the Windows parameters and optimize them for working with music via USB.

MAC OS operation of USB output:

MAC OS dos not require any driver installation. Somehow miraculously the MAC computer knows how to handle all USB devices. After plugging the DAC by the USB cable and turning it on, within 3 seconds the device should show up on the MAC.

Note: the device will NOT be described as Lampizator DAC but as Amanero Combo module.

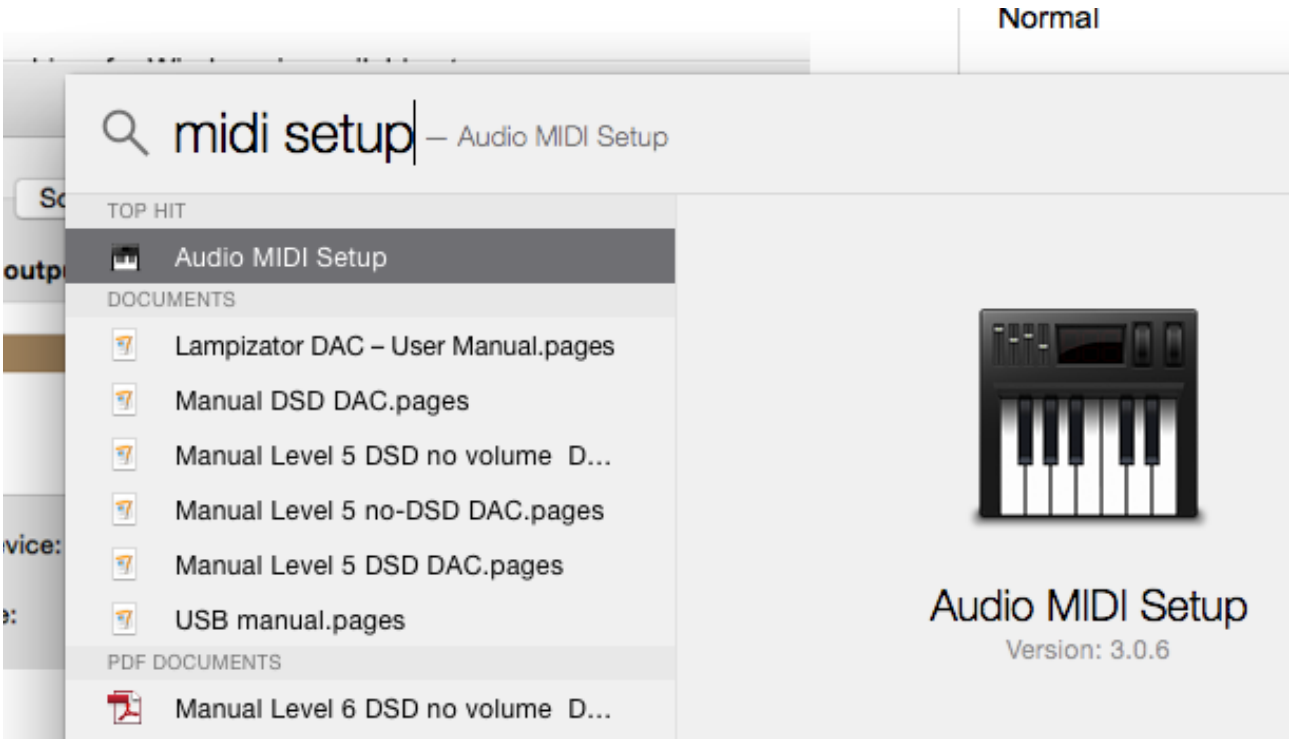
To verify what is going on, please go to the “apple sign” in top left corner of the screen and choose PREFERENCES and then the loudspeaker icon - SOUND.



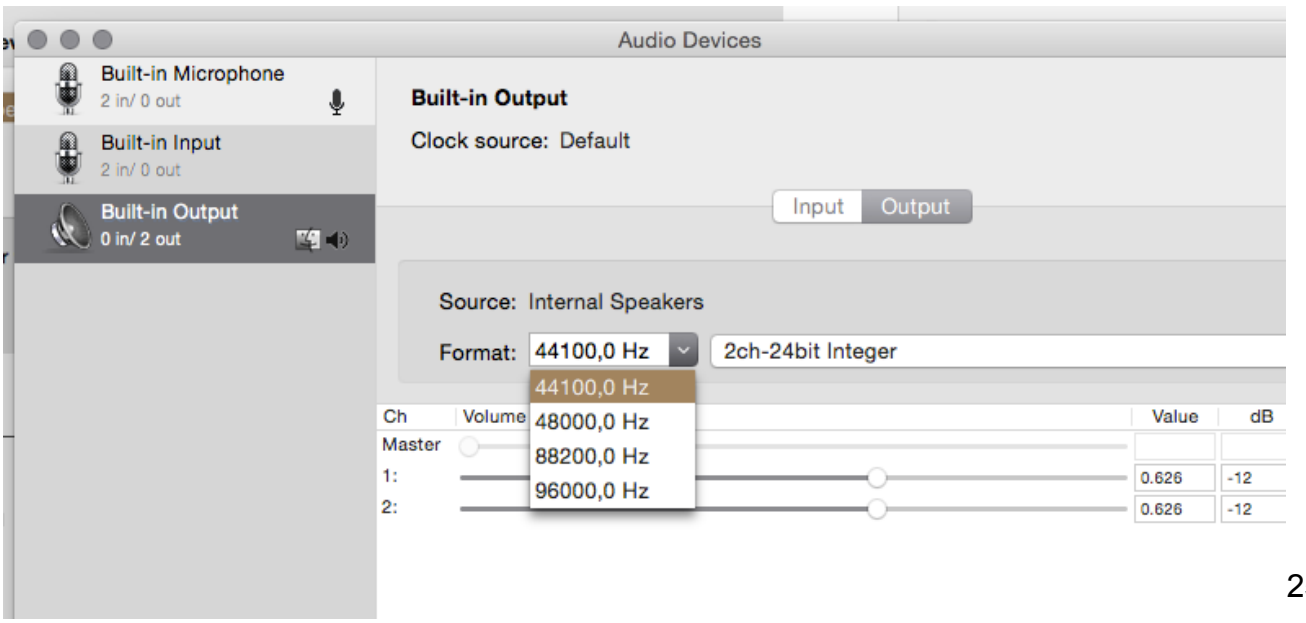
Above: on that list the Amanero should appear.

Next thing to check is MIDI SETTINGS of the MAC computer. We go to the top right corner of the screen and press SPOTLIGHT :

We type MIDI SETUP and enter.



In the MIDI setup we can choose frequencies of sampling we use for the Amanero output. We don't think that the higher the better but your own test should confirm that.



LISTENING TO THE MUSIC

some practical tips

Please use good shelf for the DAC. Do not place it on speakers, subs, or even on transports or amps. Again - tubes hate vibrations.

If you try the special devices for placement, we feel that: granite or marble is bad (ringing). Cones are just plain ridiculous. Ceramic ball bearing feet are great. Good wood is great if thick. Others - please try.

The way stereo sound is created inside the DAC can - under optimal condition - re-create the musical experience as it sounded live. It means that two speakers can cause us listeners to hear sounds everywhere around us, above, below, far in front, almost close to our face, and also behind us. This type of imaging is our goal. The sound must be able to get detached from the speakers (so called disappearing act) and the more our DAC helps doing it - the higher we value it (and price accordingly). We voice our DACs to be as 3-D as possible.

From our experience speakers should be positioned following the basic rules of LampizatOr Nirvana Room:

1. Speakers and listener’s head form unilateral triangle (3 x 60 degrees) with the distance between speakers being exactly equal to distance head-speaker.
2. Head must be in exactly middle of the speaker base and the speaker base must be exactly symmetrical versus side walls. We place speakers and measure the distance from side walls with 1 cm accuracy.
3. The distance of the speakers to the side walls and speakers to rear wall should not be equal. We recommend 1,4 times smaller or 1,4 times larger distance- but not equal. We measure that counting from the magnet of the bass driver.
4. Distance from rear wall of speaker and rear wall of the room should be no less than 0,5 m or 2 feet.
5. Ideally, the tweeter should be at the height of the ear or up to 10 cm higher, but nOT LOWER. Speakers with tweeters lower than 90 cm sound terribly wrong.
6. The chair or sofa should not have the back support higher than the person’s shoulders - in other words - should not be just behind the ears
7. Feet are the second ears of our body. They receive a lot of vibration stimulation and the brain combines this with the hearing. So we advise to have a piece of floor without any carpet directly where our feet are. Listening with feet (preferably bare) on the hard floor greatly enhances our perception of music. It is advisable to have rug or carper between listener and speakers but not under the feet.
8. It is advisable to put something soft directly on the wall behind the speakers

9. The so called toe-in - the degree by which the speakers face the listener and not alongside the walls straight - is very critical. The rule of thumb is to toe in half way between standing straight and aiming at the listeners ear. Or slightly more straight, but not more towards the head.

BURN IN PERIOD

The DAC comes straight from our factory after around 24 Hours of testing so it is not exactly “new” but it is not burned-in enough. Our customers report back, that after 3 days of constant powering (playing or not) the DAC opens up significantly. Further improvements are observed after up to 7 days when things stabilize.

Additional one day burn in is needed after every time the DAC: travels somewhere (vibrations), or is disconnected for over a month or is subject to cold temperature - like in the car trunk, when left overnight.

When the DAC is fully burned in, the sound quality is stable, and we only need to warm it after powering every day.

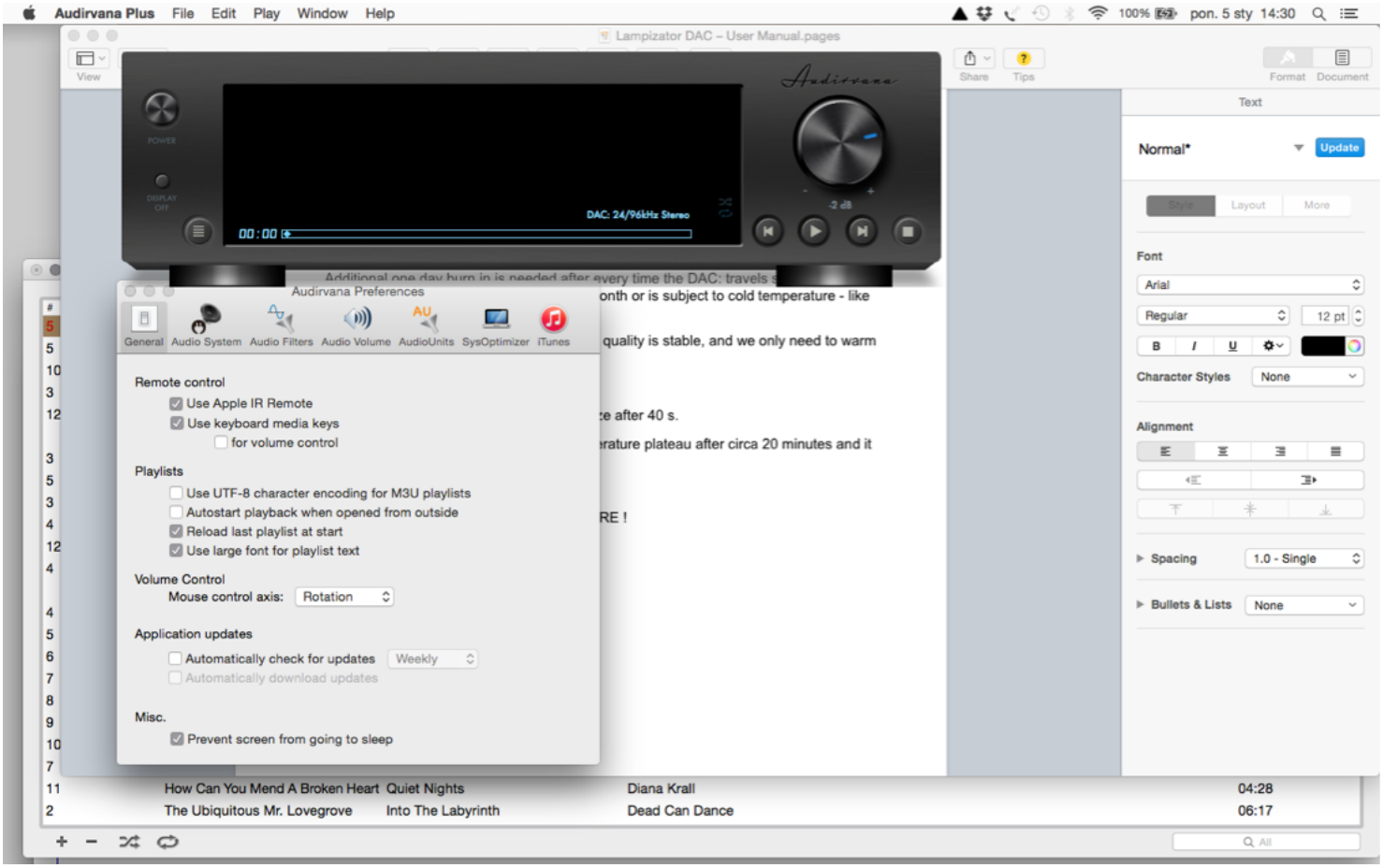
The DAC starts to play after 20 seconds.

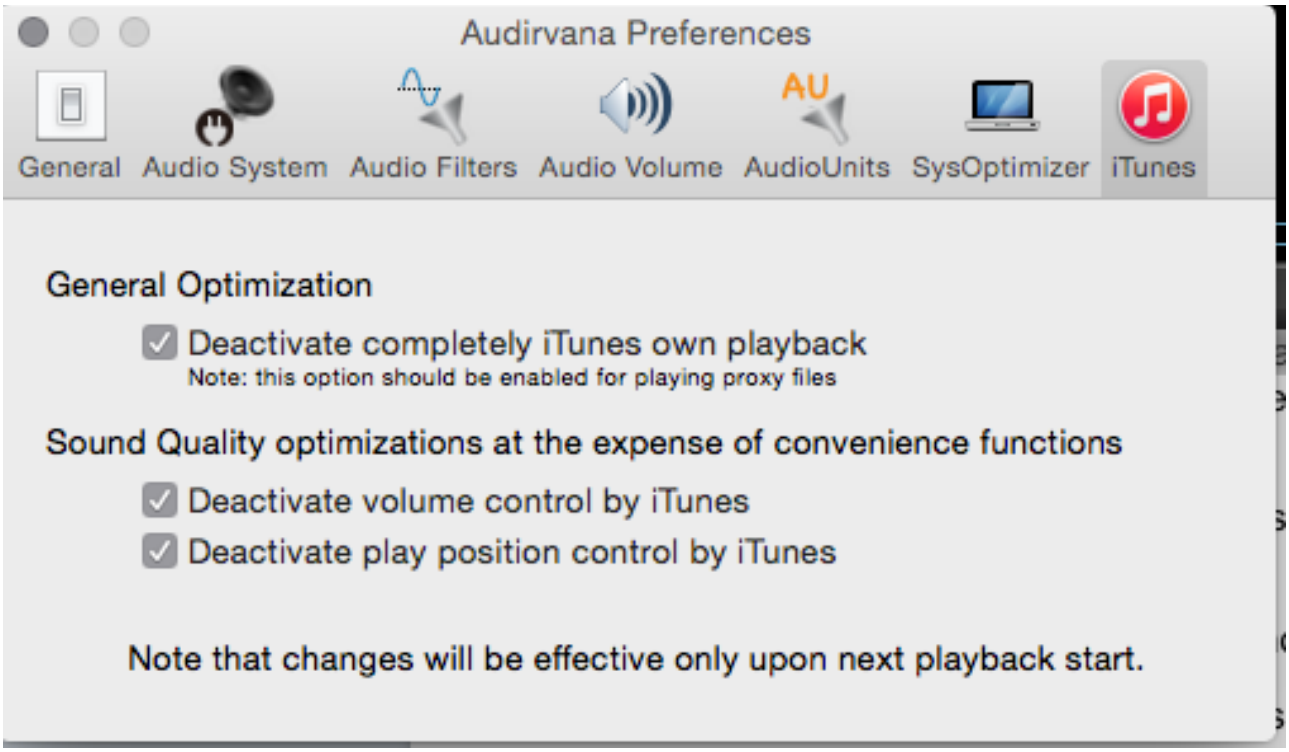
The tubes reach full parameters and stabilize after 40 s.

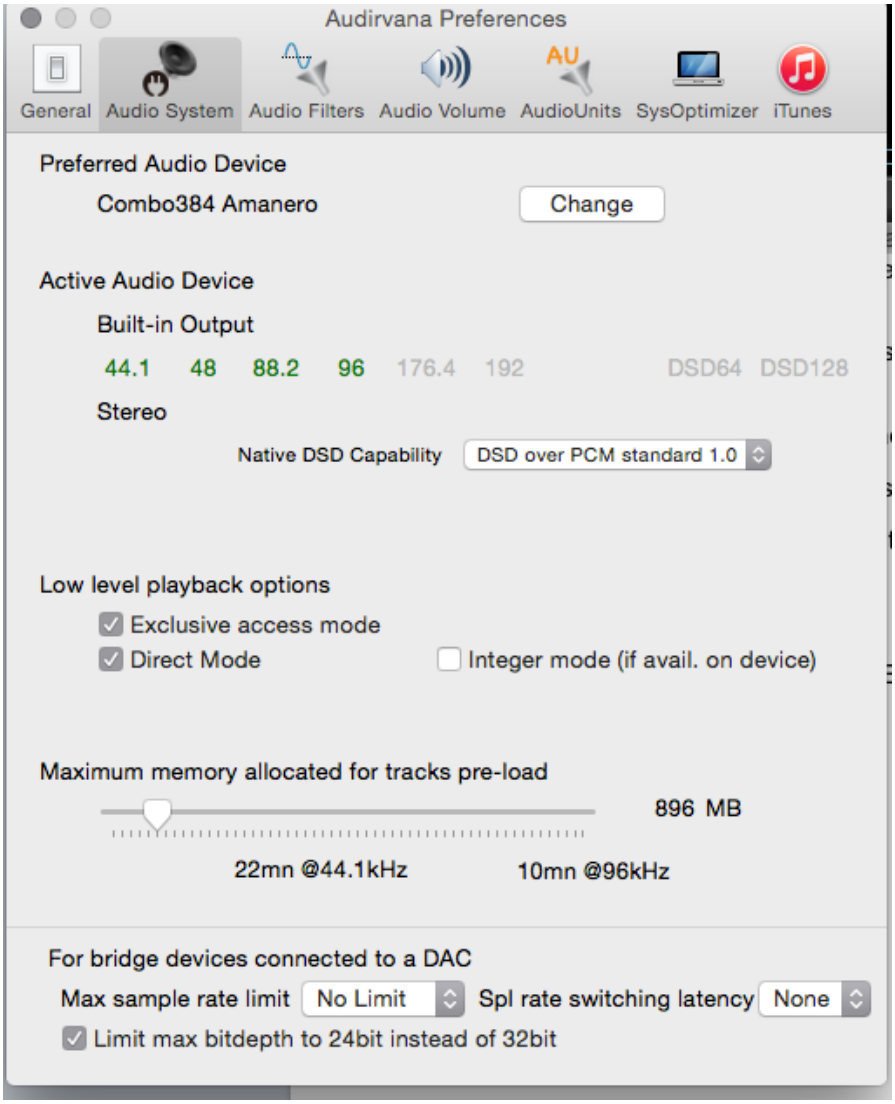
The whole system reaches operating temperature plateau after circa 20 minutes and it is ready for serious listening.

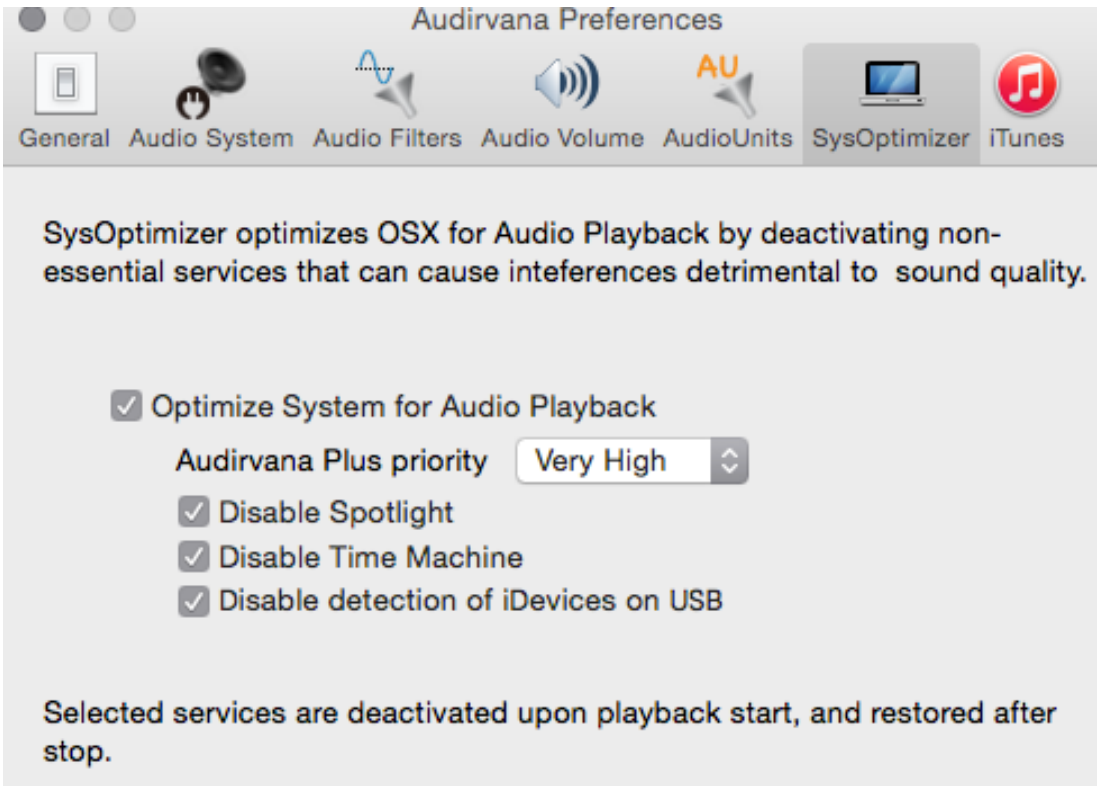
ENJOY YOUR MUSIC LIKE NEVER BEFORE !

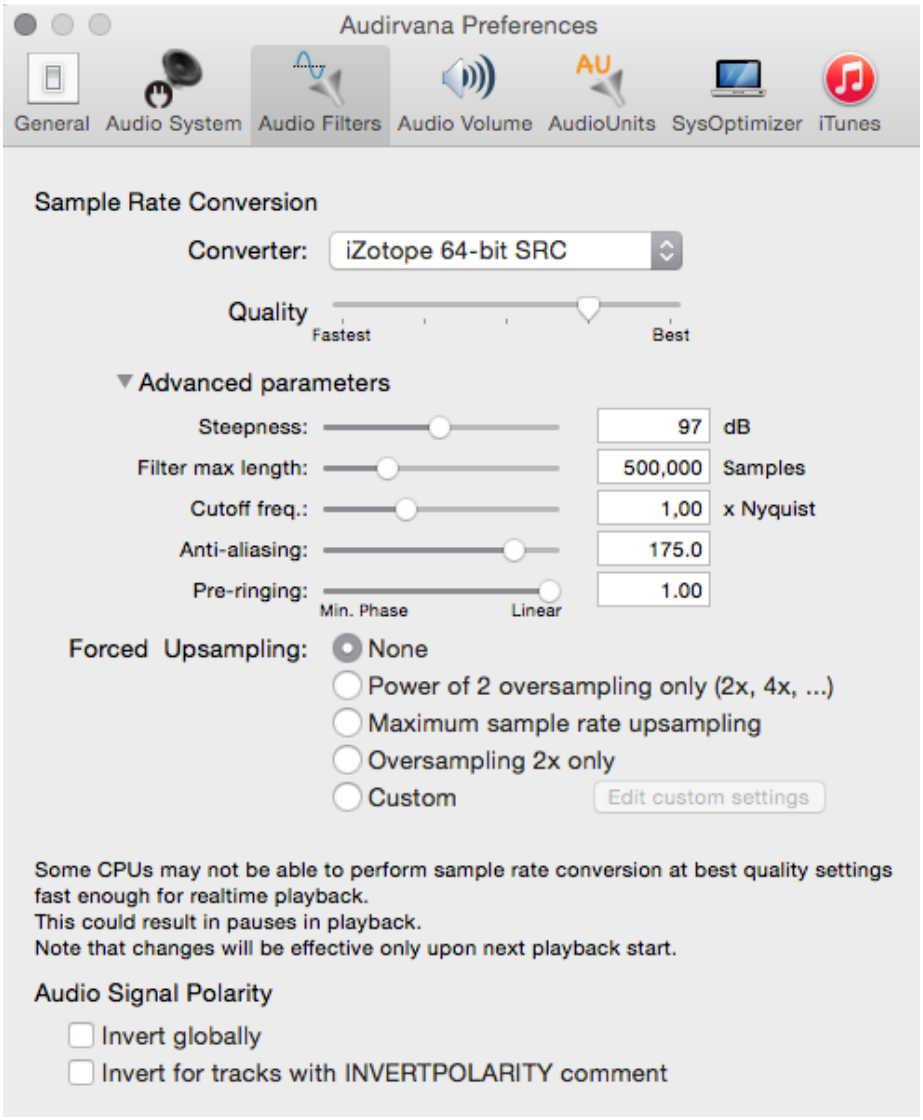
Audirvana Screenshots of working MAC OS configuration:

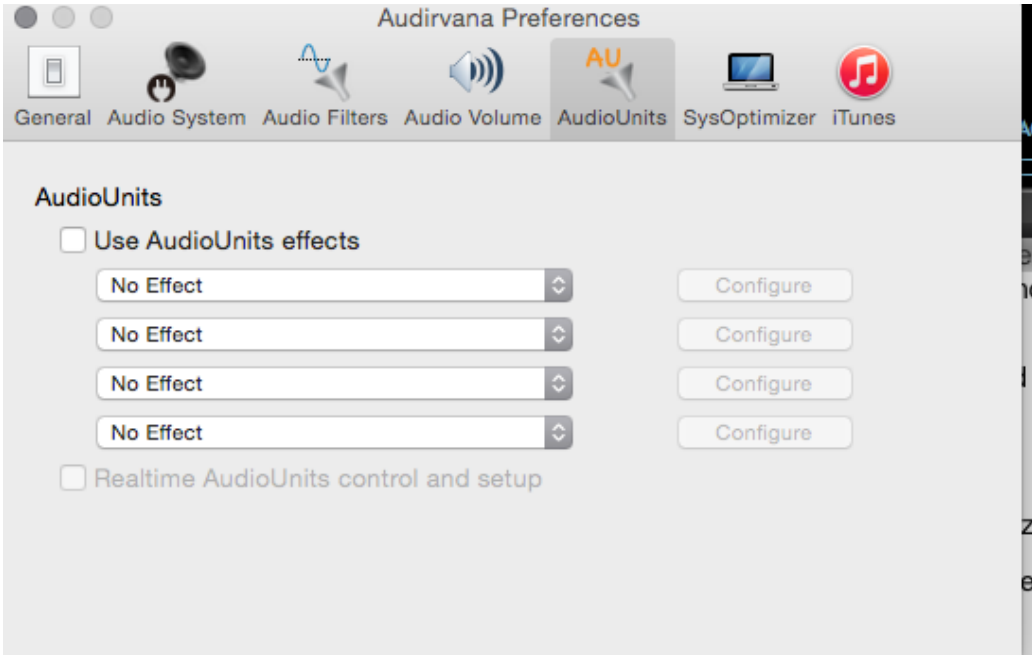


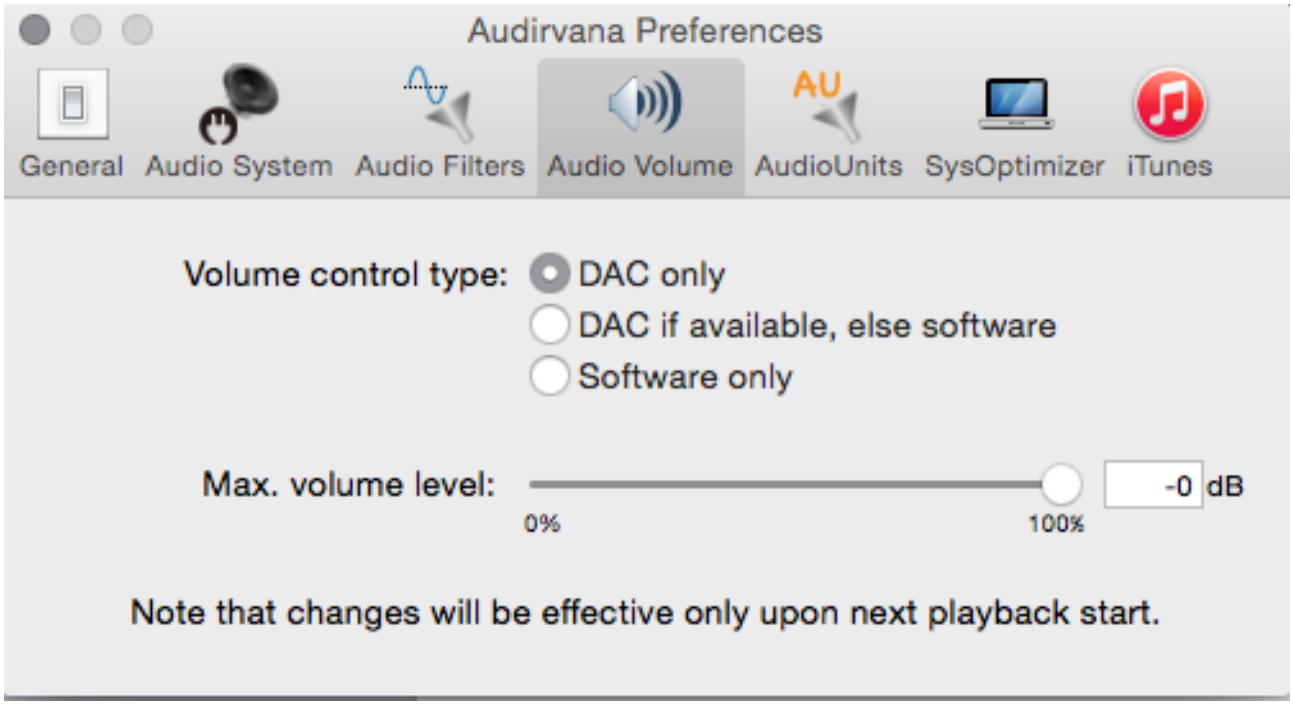



Audirvana Preferences
General
Audio System
Audio Filters
Audio Volume
AudioUnits
SysOptimizer
iTunes
Preferred Audio Device
Combo384 Amanero
Change
Active Audio Device
Built-in Output
44.1 48 88.2 96 176.4 192 DSD64 DSD128
Stereo
Native DSD Capability
DSD over PCM standard 1.0
Low level playback options
Exclusive access mode
Direct Mode
Integer mode (if avail. on device)
Maximum memory allocated for tracks pre-load
896 MB
22mn @44.1kHz
10mn @96kHz
For bridge devices connected to a DAC
Max sample rate limit
No Limit
Spl rate switching latency
None
Limit max bitdepth to 24bit instead of 32bit

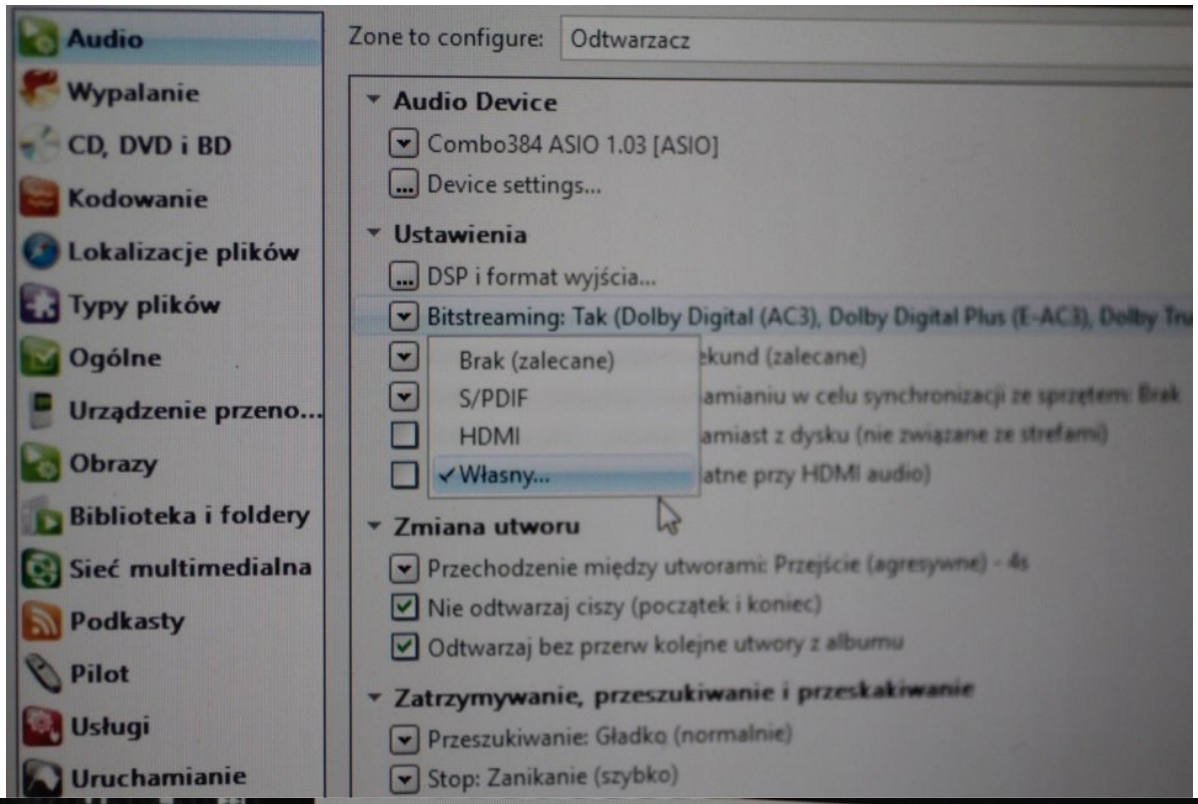

Audirvana Preferences
General
Audio System
Audio Filters
Audio Volume
AudioUnits
SysOptimizer
iTunes
SysOptimizer optimizes OSX for Audio Playback by deactivating non-essential services that can cause inteferences detrimental to sound quality.
Optimize System for Audio Playback
Audirvana Plus priority
Very High
Disable Spotlight
Disable Time Machine
Disable detection of iDevices on USB
Selected services are deactivated upon playback start, and restored after stop.

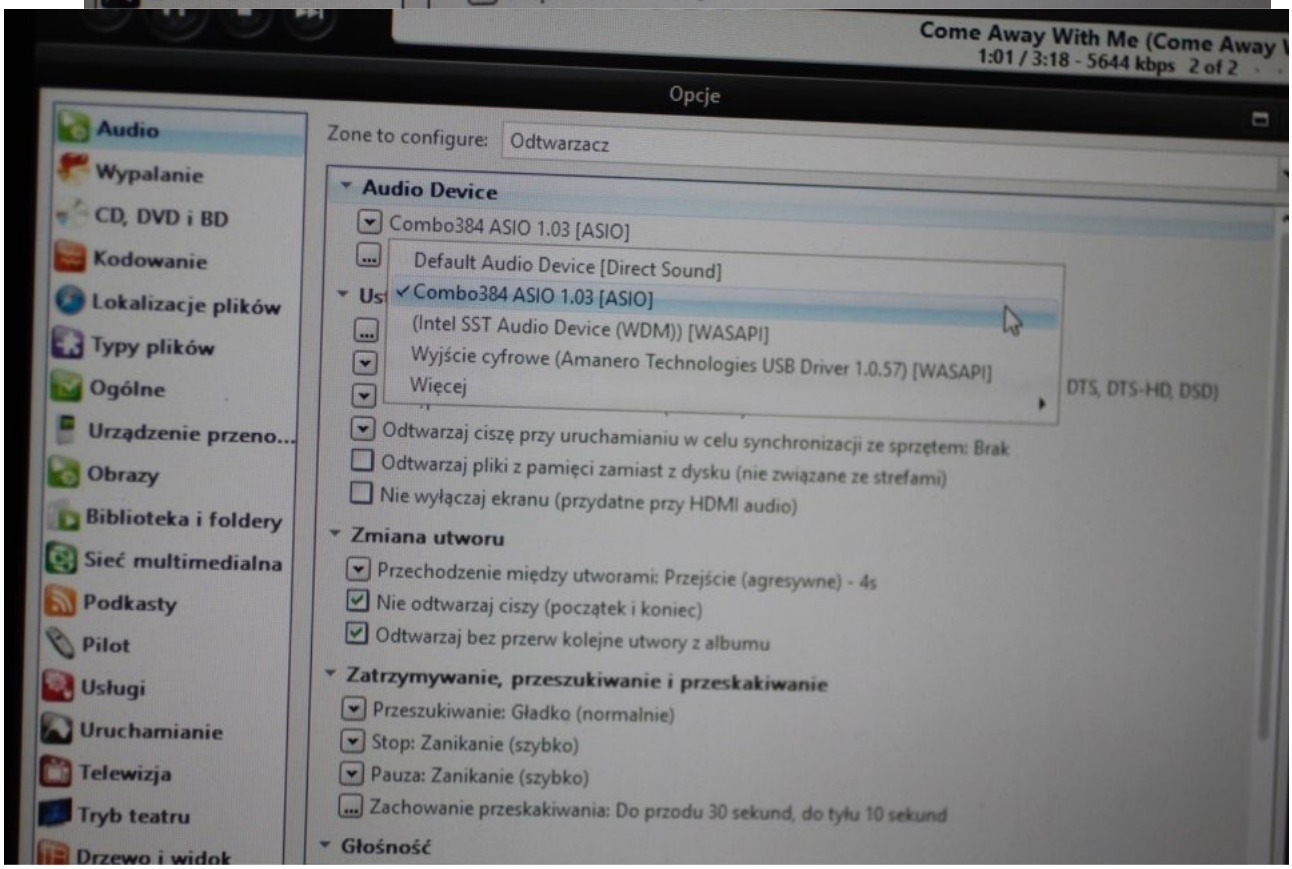

Audirvana Preferences
General
Audio System
Audio Filters
Audio Volume
AudioUnits
SysOptimizer
iTunes
Sample Rate Conversion
Converter:
iZotope 64-bit SRC
Quality
Fastest
Best
Advanced parameters
Steepness:
97
dB
Filter max length:
500,000
Samples
Cutoff freq.:
1,00
x Nyquist
Anti-aliasing:
175.0
Pre-ringing:
1.00
Min. Phase
Linear
Forced Upsampling:
None
Power of 2 oversampling only (2x, 4x, ...)
Maximum sample rate upsampling
Oversampling 2x only
Custom
Edit custom settings
Some CPUs may not be able to perform sample rate conversion at best quality settings
fast enough for realtime playback.
This could result in pauses in playback.
Note that changes will be effective only upon next playback start.
Audio Signal Polarity
Invert globally
Invert for tracks with INVERTPOLARITY comment

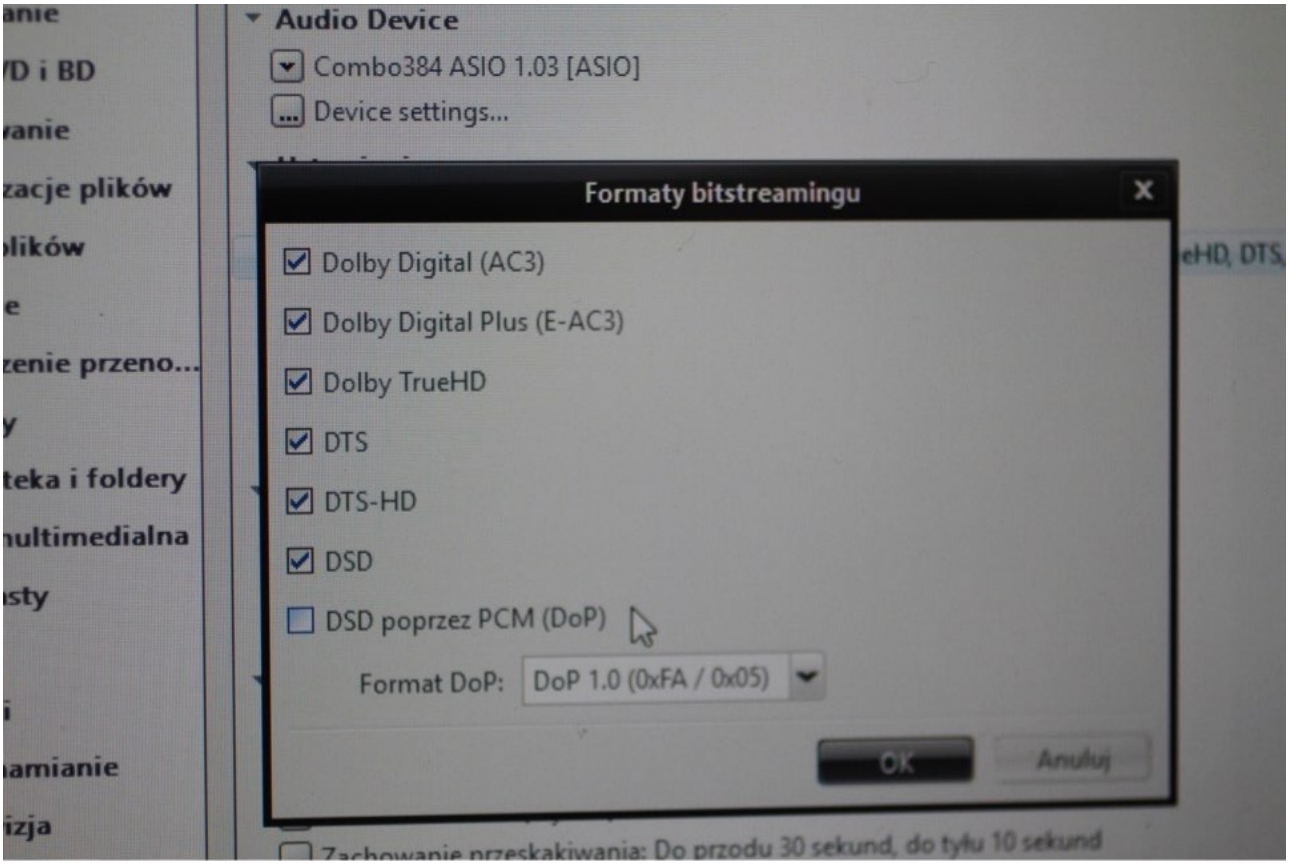

Audirvana Preferences
General
Audio System
Audio Filters
Audio Volume
AudioUnits
SysOptimizer
iTunes
AudioUnits
Use AudioUnits effects
No Effect
Configure
No Effect
Configure
No Effect
Configure
No Effect
Configure
Realtime AudioUnits control and setup


Audirvana Preferences
General
Audio System
Audio Filters
Audio Volume
AudioUnits
SysOptimizer
iTunes
Volume control type:
DAC only
DAC if available, else software
Software only
Max. volume level:
-0 dB
0%
100%
Note that changes will be effective only upon next playback start.

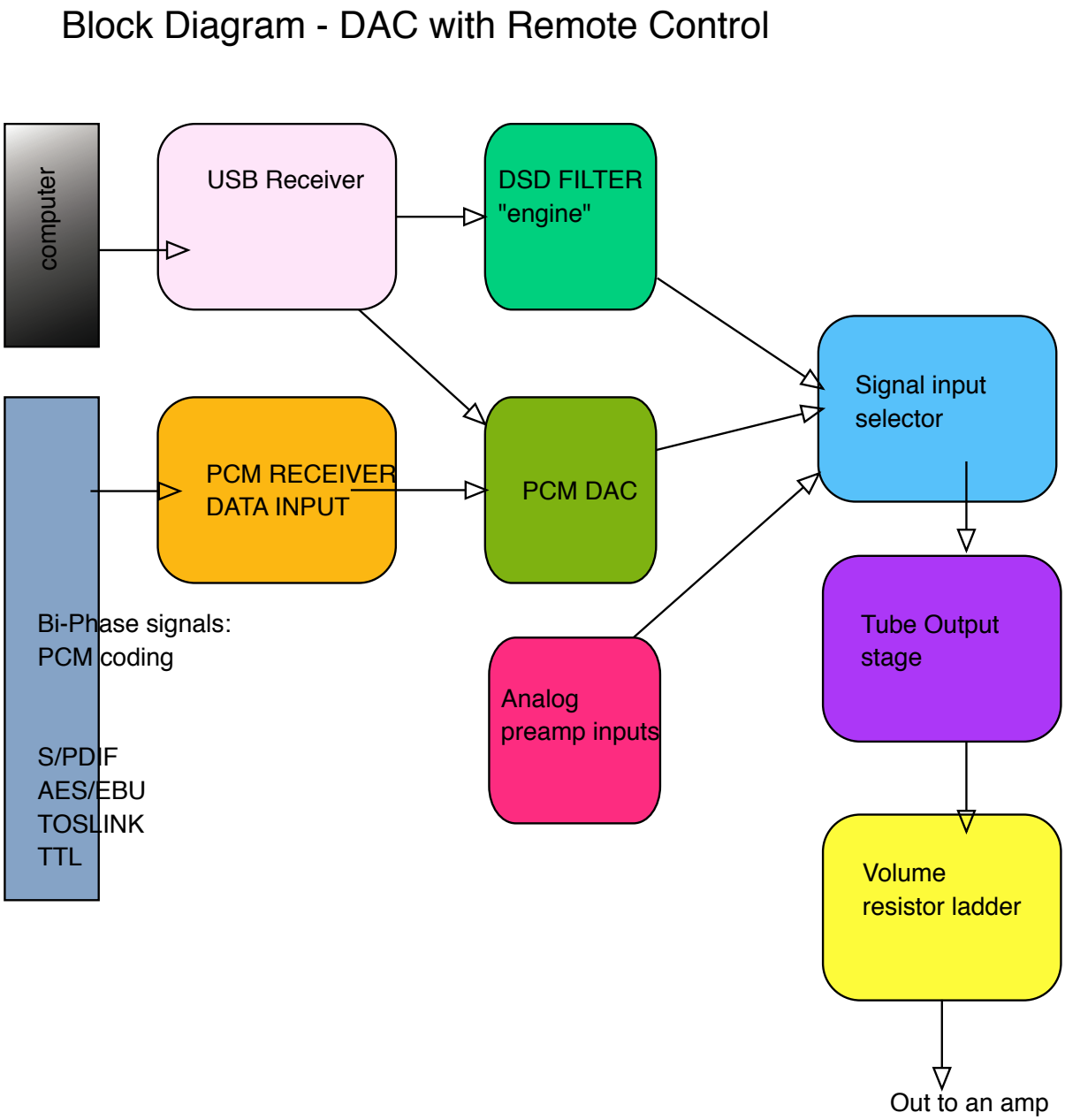
JRiver DSD setting screenshots






Audio Device
Combo384 ASIO 1.03 [ASIO]
Device settings...
Formaty bitstreamingu
Dolby Digital (AC3)
Dolby Digital Plus (E-AC3)
Dolby TrueHD
DTS
DTS-HD
DSD
DSD poprzez PCM (DoP)
Format DoP:
DoP 1.0 (0xFA / 0x05)
OK
Anuluj

Block Diagrams



Block Diagram - DAC without Remote Control