DAC501-502 MK2 - 4ch d/a converter

DAC501-MK2-4ch is Weiss 5 Series 2nd generation D/A converter and network renderer with unique DSP technologies. The different between DAC501 & DAC502 are ***1)*** *the width of the chassis* ***2)*** *DAC502 has 4pin XLR headphone amplifier output*. They have identical sound quality and features.

The DAC501 / DAC502 (or DAC50x in short) are extremely versatile and pleasantly sounding D/A Converters. They are available for close to 6 years now and still going strong. In our ongoing journey to achieve a live-like music reproduction we now introduce the **Mark II versions**, called the **DAC501-4ch** and **DAC502-4ch**. Our residential audio engineer, Joschka Weiss, says in comparison to the former DAC section the -4ch version ***"achieves another level of sonic performance"***.

>> What is the purpose of the MK2-4ch versions?

The **MK2-4ch** versions use one of the latest DAC chips with stellar technical specifications. So the **MK2-4ch** versions make for ***an enhanced 2 channel DAC***, with the same features as the DAC501 / DAC502 units sport. The sonic characteristics of the **MK2-4ch** versions are on a higher level with incredible transparency and openness. Both line output stages and headphone output stages are built with discrete electronics developed from the standard DAC50x.

All standard DAC50x units can be upgraded with the **MK2-4ch** version. It is a new analog section module to be installed in the unit plus a new software needs to be loaded. It is our goal to keep our equipment upgradeable for long-term securing of your investment. The standard DAC501 / DAC502 units we continue to manufacture.

In addition the **MK2-4ch** versions will be capable to play 4 channels simultaneously once we introduce the necessary software. This also with the goal to get a more live – like reproduction at home and to accommodate for the upcoming 3D Audio productions in a high-end HiFi setup where a very limited number of available speaker channels often is a matter of fact. This will be the future of high-end audio reproduction.

>> What 4 channel applications are planned?

Here is a list of 4 – channel features we have in mind. There is no guarantee that we will implement all of them, though. We would be pleased to get information from you about what 4 channel features you would like to see.

The audio sources for the setups mentioned below can be two or more channels. The input channels are automatically rendered such that they fit the format of the four outputs.

- *2 channel front speakers, 2 channel rear speakers for surround playback.*
  This setup adds another pair of speakers at the back for an enveloping sound.

- *2 channel front speakers, 2 channel front speakers at a different angle.*
  A very interesting setup for a more enveloping sound. Based on a design by the Swiss audio company "Pawel Acoustics".

- *2 channel front speakers, 1 channel front center speaker, 1 channel subwoofer.*
  In early stereo times there have been made 3 channel recordings with left, right and center channels. The center channel is of great use for a realistic reproduction. The fourth channel can be configured to feed a subwoofer speaker.

- *2 channel front speakers, 2 channel subwoofers.*
  For a decent full range system, based on satellites and subwoofers, the use of a stereo instead of a mono subwoofer is advantageous.

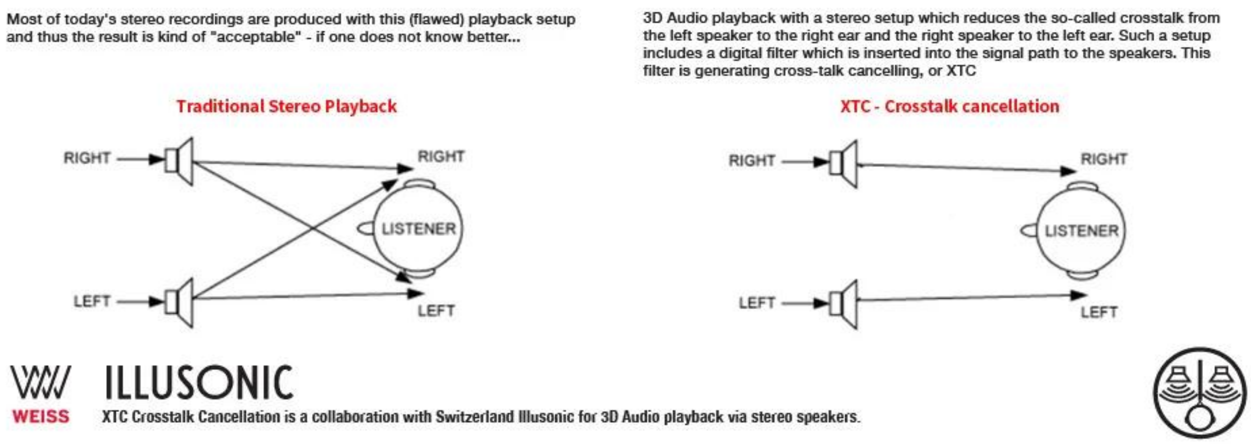
- *2 channel front speakers with crosstalk cancellation, 2 channel rear speakers with crosstalk cancellation.* Crosstalk Cancellation (XTC) means that a pair of speakers becomes kind of a headphone in that the left speaker only reaches the left ear and the right speaker only reaches the right ear – similar to a headphone. This has the effect of generating a huge soundstage in front of the listener with a very life-like reproduction of a life recording. By adding a second pair of XTC speakers at the back an enveloping life-like scenery can be achieved.

- *2 channel headphones type A, 2 channel headphones type B for two persons listening simultaneously with two different headphone models.* Allows two persons to listen to the same music program each with his/her own set of headphones and signal processing.Some more "exotic" ideas which need some refinement in the design process:

- *2 channel front speakers, 2 channel front elevated speakers.*
  Another way to generate a more live – like sound with a sound stage which extends to the space above the listener. Similar to a concert hall speaker setup.

- *2 channel front speakers, 2 channel headphones for simultaneous listening via speakers and headphones.*
  With extremely open headphones it is possible to listen to speakers and headphones simultaneously. This makes for a new listening sensation not experienced before.

We are considering even more setup variants – let us know what you would like to see.



>> What technology is used?

The DAC501-4ch and DAC502-4ch units are based on the following hardware parts:

- **ARM processor** for Network and USB interfacing and general control of the unit
- **SHARC Signal Processor chip** for various signal processing tasks including sampling frequency conversion and high precision volume control
- Extensive power supply with separate power transformers and many high precision voltage regulators
- Local, very low jitter clock generator for the D/A converter
- State of the art D/A conversion
- Discrete analog sections following the D/A conversion
- Very low impedance outputs
- Analog output level selection to accommodate for amplifiers and headphones

The **DAC501-4ch** / **DAC502-4ch** are our new state of the art D/A Converters with an unprecedented level of sophistication and versatily.

The **DAC502-4ch** uses a larger frame but else sports the same features as the **DAC501-4ch**. Except for an an additional 4 pin headphone socket at the back of the unit.

With the **DAC50x-4ch** we are creating a new paradigm for what used to be a black box device. A typical D/A Converter is a “set and forget” device. Not so with the **DAC50x-4ch**. It adds a number of interesting signal processing features and sports a variety of digital inputs. Balanced, unbalanced and headphone outputs are provided.

Weiss Engineering has a 35 year history in D/A Converter design. In that time span we have learned a thing or two about converter design. The **DAC50x-4ch** is the essence of our experiences.

DAC501-4ch and DAC502-4ch are Roon Ready.

Daniel Weiss Awarded 2021 Technical Grammy

MAIN FEATURES

Mechanics
The **DAC50x-4ch** uses a stainless steel chassis with a solid 10 mm aluminium front plate.

Power Supply
A powerful non-switching power supply is used. All sensitive voltages have their own regulators which are separated between left and right channels. The result is an analog output free of “digital noise” and channel crosstalk. The power switch activates a semiconductor relay which only switches on or off at zero crossings of the mains voltage. This assures a glitch free power switching. The two mains transformers are toroidal types. Mains voltage selection is done automatically by measuring the mains voltage before power is applied to the rest of the electronics.

Synchronization
An internal high precision / low jitter clock generator is responsible for clocking the D/A converter section. The sampling frequency of that generator is fixed at about 195kHz. The input signals are converted to the 195kHz sampling frequency for optimal signal quality. This scheme also helps significantly in reducing any jitter related effects. All standard sampling frequencies up to 384 kHz plus DSD x64 and x128 are supported.

Digital Inputs
There are a total of five inputs:

- AES/EBU or S/PDIF via XLR, Toslink and RCA sockets
- UPnP / DLNA via Ethernet
- USB
- Roon Ready®

Accepted formats: PCM 44.1kHz up to 384 kHz, DSD 64x / 128x.

Future formats can be accommodated for via software updates.

Signal Processing

The **DAC50x-MK2-4ch** has a digital signal processing chip built in (**DSP**).

The following DSP algorithms are currently implemented or implemented soon:

- **Room Equalizer** – to suppress room modes for a decent bass reproduction.

- **Creative Equalizer** – a tone control with low boost/cut, high boost/cut and mid boost/cut. Very useful to correct those recordings which do not quite sound right.

- **De-Essing** – the automatic removal of overly bright sibilances from human voices. The sibilance effect can be more or less pronounced depending on your speakers or room acoustics.

- **Dynamic Adaption** – adjusts the audio volume (loudness) to a constant value across all tracks played. Useful for “party mode” when the volume control should stay untouched.

- **Vinyl Emulation** – get that special sonic character of a record player based playback chain. We also employ an emulation of the **DMM-CD** procedure offered by the **Stockfisch label**.

- **Crosstalk Cancelling (XTC)** – for the playback of dummy head recordings or live recordings via speakers for an incredible live sensation. Dummy head recordings usually are listened to via headphones because they only work properly if the left channel goes to the left ear only and the right channel to the right ear only. With speakers this is difficult to achive as the left channel goes to the left and the right ear. But with some clever signal processing of the speaker channels is is possible to suppress the crosstalk, i.e. the audio going from the left speaker to the right ear and vice versa. If that works properly then the recording sounds as if one would be in the space where the recording has taken place. All the reverberation and 3D representation of the sound sources is there.(For speaker based playback only.)

- **Out Of Head Localization algorithm** – tries to get the music “out of your head” when listening via headphones. The goal is to achieve a similar listening sensation as one gets when listening via speakers.
  (For headphone based playback only.)

- **Audeze Headphones EQ** – The advanced Headphone Equalizer currently sports the large range of Audeze headphones. Select your Audeze headphone model and enjoy listening with it even more than before.LL Audeze headphone models.

- **The Loudness Equalizer** – The Loudness Equalizer is a premier implementation of a patented design by Mr. Bernhard Schwede. It does not adhere to the commonly used Fletcher-Munson frequency responses but uses a completely new approach with incredible results.

Synchronization
Via the input signal
Via the internal oscillator
Sampling frequencies supported: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz
USB & UPnP/DLNA supports 352.8 kHz, 384 kHz, DSD64, DSD128

Converters
Two of the latest 32 bit D/A Converter chips are used. Two D/A conversion channels are used per audio channel, resulting in exceptional performance specifications.

Analog Outputs

- Line out unbalanced on RCA connector
- Line out balanced on XLR connector
- Headphone out on 1/4" Jack Analogue
- For the DAC502: Addtional headphone output on a 4 pin XLR connector.

Discrete output stages for both line and headphone outputs are employed.
The output levels can be set in a coarse manner with 4 steps to adapt for the amplifier or headphone connected. The levels can be set independently for line and headphone outputs.
No sound degrading servo mechanisms are used.

Front panel controls

- A rotary encoder knob for changing parameters and for powering the unit on/off.
- A touch screen colour LCD display.
- An 1/4 inch headphone socket.
- An IR receiver

Back panel elements

- Analog outputs on XLR and RCA connectors.
- Digital inputs on XLR, RCA, TOSLINK, USB, Ethernet connectors.
- USB type A connector for various applications.
- Mains connector with fuses.

IR Remote Control
Allows to control several functions of the **DAC50x**, namely:

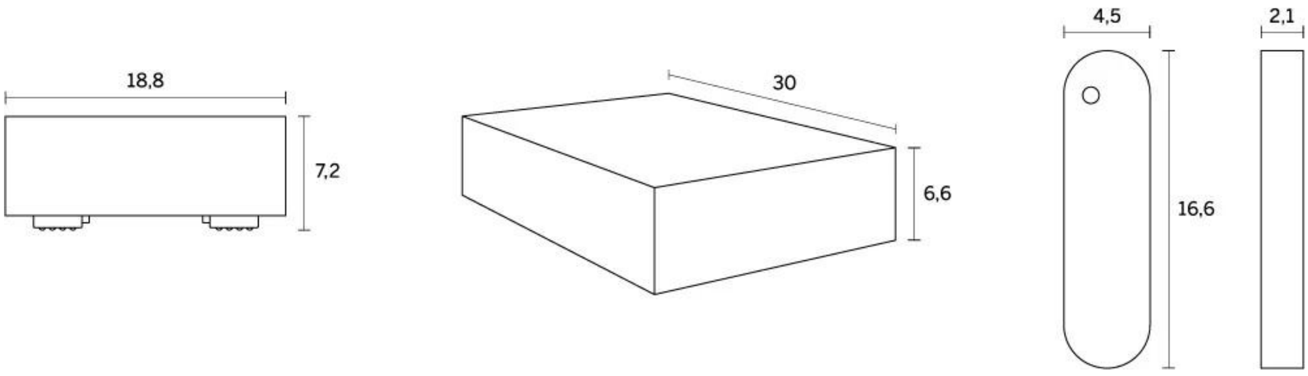
- The input selected for conversion.
- The output type, level, muting, absolute polarity.
- Power on/off.
- DSP presets.

Web Interface

The **DAC50x** has a web interface for configuring it via a web browser. The following functions are accessible via the web interface:

- Volume and Balance controls.
- Input selection.
- Output type.

- DSP algorithms configuration.
- DSP snapshots configuration.



TECHNICAL DATA

Power
Mains voltage: 100…120 V or 200…240 V
Fuse rating: 500 mA slow blow at 100…120 V, 250 mA slow blow at 200…240 V
Power consumption: 15 VA max.
Power consumption in standby: 0.5 VA max.

Size DAC501
Depth: 30 cm
Width: 18,8 cm
Height: 6,6 cm
Height with feet: 7,2 cm

Size Remote
Depth: 2,1 cm
Width: 4,5 cm
Height: 16,6 cm

Available Color
Silver, Black

Digital Inputs
(1) XLR connector
(1) RCA connector
(1) TOSLINK connector (optical)
(1) USB type B connector

(1) RJ45 Ethernet connector
All inputs accept professional or consumer standard , i.e. accept AES/EBU or S/PDIF signals .
Sampling frequencies: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz or 192 kHz on any of the inputs
Maximum input word-length: 24 Bits

Digital Outputs
none (Please see **DSP501**/**DSP502**)

Analog Outputs
(2) XLR connectors (hot on pin 2), short circuit proof output circuitry, output impedance tbd Ω
(2) RCA connectors, short circuit proof output circuitry, output impedance tbd Ω

The output level is selectable via the LCD menu; four settings are provided as shown below:

XLR Output:
8.15 Vrms +20.44 dBu with a 0 dBFS sine wave input
4.15 Vrms +14.57 dBu with a 0 dBFS sine wave input
2.12 Vrms +7.74 dBu with a 0 dBFS sine wave input
1.06 Vrms +2.72 dBu with a 0 dBFS sine wave input

These levels are achieved with a 0.0 dB setting for the level control on the LCD screen.

Suggested subsequent amplifier input impedances:
8.15 Vrms setting: 500 Ω or higher
4.15 Vrms setting: 300 Ω or higher
2.12 Vrms setting: 150 Ω or higher
1.06 Vrms setting: 70 Ω or higher

RCA Output:
4.08 Vrms +14.42 dBu with a 0 dBFS sine wave input
2.08 Vrms +8.55 dBu with a 0 dBFS sine wave input
1.06 Vrms +1.72 dBu with a 0 dBFS sine wave input
0.53 Vrms −3.30 dBu with a 0 dBFS sine wave input

These levels are achieved with a 0.0 dB setting for the level control on the LCD screen.

Suggested subsequent amplifier input impedances:
4.08 Vrms setting: 250 Ω or higher
2.08 Vrms setting: 150 Ω or higher
1.06 Vrms setting: 75 Ω or higher
0.53 Vrms setting: 35 Ω or higher

Headphone Output:
(1) stereo 1/4-inch jack connector, DC coupled, short circuit proof output circuitry. The output level is selectable via the LCD menu; four level settings are provided as shown below:

5.2 Vrms +16.53 dBu
2.7 Vrms +10.84 dBu
0.9 Vrms +1.30 dBu
0.2 Vrms −11.77 dBu

These levels are achieved with a 0.0 dB setting for the level control on the LCD screen.

Suggested headphone impedances:
5.2 Vrms setting: 100 Ω or higher with a 0 dBFS sine wave input
2.7 Vrms setting: 50 Ω or higher with a 0 dBFS sine wave input
0.9 Vrms setting: 16 Ω or higher with a 0 dBFS sine wave input
0.2 Vrms setting: 4 Ω or higher with a 0 dBFS sine wave input

Measurements Main Output

The measurements below have been taken at the following conditions (unless noted otherwise): 1 kHz measurement frequency, maximum selectable output level, 192 kHz sampling frequency (Fs), 22 kHz measurement bandwidth, unweighted, 0 dBr equals the output level at 0 dBFS input.

Frequency Response:
Fs = 44.1 kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 44.1 kHz, Filter B, 0 Hz…20 kHz: within ±1.3 dB
Fs = 88.2 kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 88.2 kHz, Filter A, 0 Hz…40 kHz: within ±0.8 dB
Fs = 88.2 kHz, Filter B, 0 Hz…20 kHz: within ±0.25 dB
Fs = 88.2 kHz, Filter B, 0 Hz…40 kHz: within ±1.5 dB
Fs = 176.4 kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 176.4 kHz, Filter A, 0 Hz…40 kHz: within ±0.8 dB
Fs = 176.4 kHz, Filter A, 0 Hz…80 kHz: within ±2.5 dB
Fs = 176.4 kHz, Filter B, 0 Hz…20 kHz: within ±0.25 dB
Fs = 176.4 kHz, Filter B, 0 Hz…40 kHz: within ±0.8 dB
Fs = 176.4 kHz, Filter B, 0 Hz…80 kHz: within ±3.5 dB

Total Harmonic Distortion plus Noise (THD+N):
116 dBr (0.00016 %) at −3 dBFS input level
125 dBr (0.000056 %) at −40 dBFS input level
125 dBr (0.000056 %) at −70 dBFS input level

Linearity:
At 0 dBFS to −120 dBFS input level: less than ±0.4 dB deviation from ideal

Spurious components (including harmonics):
At 0 dBFS input level, maximum output level, 1 kHz, all components at less than −120 dB
At 0 dBFS input level, maximum output level, 4 kHz, all components at less than −115 dB

Crosstalk:
Better than 120 dB, 20 Hz…20 kHz

Interchannel Phase Response:
±0.05° 20 Hz…20 kHz
±0.30° 20 Hz…80 kHz

Measurements Headphone Output
The measurements below have been taken at the following conditions (unless noted otherwise): 1 kHz measurement frequency, maximum selectable output level, 192 kHz sampling frequency (Fs), 22 kHz measurement bandwidth, unweighted, 0 dBr equals the output level at 0 dBFS input.

Frequency Response:
Fs = 44.1kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 44.1kHz, Filter B, 0 Hz…20 kHz: within ±1.3 dB
Fs = 88.2kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 88.2kHz, Filter A, 0 Hz…40 kHz: within ±0.8 dB
Fs = 88.2kHz, Filter B, 0 Hz…20 kHz: within ±0.25 dB
Fs = 88.2kHz, Filter B, 0 Hz…40 kHz: within ±1.9 dB
Fs = 176.4 kHz, Filter A, 0 Hz…20 kHz: within ±0.25 dB
Fs = 176.4 kHz, Filter A, 0 Hz…40 kHz: within ±0.8 dB
Fs = 176.4 kHz, Filter A, 0 Hz…80 kHz: within ±2.6 dB
Fs = 176.4 kHz, Filter B, 0 Hz…20 kHz: within ±0.25 dB
Fs = 176.4 kHz, Filter B, 0 Hz…40 kHz: within ±0.8 dB
Fs = 176.4 kHz, Filter B, 0 Hz…80 kHz: within ±3.8 dB

Total Harmonic Distortion plus Noise (THD+N):
−115 dBr (0.00016 %) at −3 dBFS input level
−122 dBr (0.0000795 %) at −40 dBFS input level
−122 dBr (0.0000795 %) at −70 dBFS input level

Linearity:
At 0 dBFS to −120 dBFS input level: less than ±0.4 dB deviation from ideal

Spurious Components (including Harmonics):
At 0 dBFS input level, max. output level, 100 kΩ load, 1 kHz, all components at less than −120 dB
At 0 dBFS input level, max. output level, 600 Ω load, 1 kHz, all components at less than −120 dB
At 0 dBFS input level, max. output level, 300 Ω load, 1 kHz, all components at less than −120 dB
At 0 dBFS input level, max. output level, 100 kΩ load, 4 kHz, all components at less than −120 dB
At 0 dBFS input level, max. output level, 600 Ω load, 4 kHz, all components at less than −120 dB
At 0 dBFS input level, max. output level, 300 Ω load, 4 kHz, all components at less than −115 dB

Crosstalk:
Better than 110 dB, 20 Hz…20 kHz

Interchannel Phase Response:
±0.15° 20 Hz…20 kHz
±0.50° 20 Hz…80 kHz

WEISS ENGINEERING USB AUDIO DEVICE DRIVER (VERSION V4.11.0)

WEISS ENGINEERING USB AUDIO CLASS DRIVER FOR **WINDOWS**, POWERED BY THESYCON. COMPLIANT TO USB AUDIO 2.0 OR USB AUDIO 1.0 DEVICE CLASS SPECIFICATION. PROVIDES ASIO 2.2 COMPLIANT SOFTWARE INTERFACE AND INTEGRATES WITH WINDOWS AS A STANDARD WDM/DIRECTX COMPATIBLE SOUND DEVICE.
SUPPORTED OS: WIN 10, WIN 8.1, WIN 8, WIN 7 (32 & 64-BIT)
Mac OS and Linux OS requires no driver. V1.3 supports USB and UPnP up to 384kHz and DSD64/128