DAC202

Owner's Manual

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Introduction

Dear customer,

Congratulations on your purchase of the DAC202 D/A Converter and welcome to the family of Weiss equipment owners! The DAC202 is the result of an intensive research and development process. Research was conducted both in analog and digital circuit design, as well as in signal processing algorithm specification.

On the following pages I will introduce you to our views on high quality audio processing. These include fundamental digital and analog audio concepts and the DAC202 converter.

Yours sincerely,
Daniel Weiss

President, Weiss Engineering LTD.

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A Short History of Weiss Engineering Ltd.

Having studied electrical engineering, Daniel Weiss joined the Willi Studer (*Studer-Revox*) company in Switzerland. His work included the design of a sampling frequency converter and of digital signal processing electronics for digital audio recorders. In 1985, Mr. Weiss founded the company *Weiss Engineering* LTD. From the outset the company concentrated on the design and manufacture of digital audio equipment for mastering studios. Its first product was the modular 102 Series system. After more than 25 years, this system is still up to date (24 bit/96 kHz).

In the early nineties the Gambit Series was launched, taking ergonomics and sonic quality to new heights. The Gambit Series consists of stand-alone units like equalizer, denoiser/declicker, dynamics processor, A/D converter, D/A converter, sampling frequency converter, dithering etc. 40 bit floating point processors and sampling rates up to 192 kHz are employed. In 2001 we have decided to enter the High-End Hi-Fi market, which offers a comparable clientele to that of the mastering studios. Both consist of critical and discerning listeners, who are in constant search for the best audio reproduction equipment or the best audio tools respectively.

Our list of clients includes big names, like *SONY*, *BMG*, *EMI*, *Warner*, *Hit Factory*, *Abbey Road*, *Teldec*, *Telarc*, *Gateway Mastering* (Bob Ludwig), *Bernie Grundman Mastering*, *Masterdisk*, *Sterling Sound*, *Whitfield Street*, *Metropolis* and hundreds more. For a more comprehensive list you are invited to visit our pro audio website at www.weiss.ch.

Today Weiss Engineering LTD. employs nine people, five of them in the engineering department.

Our Mission and Product Philosophy

The wealth of experience we have gained in over 25 years of designing products for top mastering engineers, we now also apply to the design of outstanding High-End Hi-Fi products. Our mission is to create equipment which becomes classic right from the outset — outstanding in sonics and design. These are some of the milestones at Weiss Engineering LTD.:

1985 Introduction of the 102 Series, a 24 bit modular digital audio processor for mastering studios

1986 Introduction of one of the first sample rate converters for digital audio

1987 Introduction of one of the first digital equalizers

1989 Introduction of one of the first digital dynamics processors

1991 Introduction of the Ibis digital mixing console, built for the mix-down of classical music

1993 Introduction of the Gambit Series of digital audio processors, which employ 40 bit floating point processing and sport an extremely ergonomic user interface

1995 First 96 kHz sampling rate capable products delivered

2001 Introduction of the MEDEA High-End Hi-Fi D/A Converter, the first product in our High-End series

2004 Introduction of the JASON CD Transport

2007 Introduction of the CASTOR High-End Hi-Fi Power Amplifier

2008 Introduction of the MINERVA Firewire D/A Converter and the VESTA Firewire-AES/EBU interface

2010 Introduction of the DAC202 Firewire D/A Converter, the INT202 Firewire Interface and the ATT202 Passive Attenuator

2011 Introduction of the MAN301 Music Archive Network Player and the INT203 Firewire I/O Interface

2012 Introduction of the MEDEA+ D/A Converter, the MEDEA with an enhanced analog section

2013 Introduction of the INT204 USB/DSD Interface and the MEDUS State Of The Art D/A Converter. Continuing development of the MAN301 Network Player with new software features.

Contents

Chapter 1

Advanced Digital and Analog Audio Concepts Explained

1.1 Jitter Suppression and Clocking

What is jitter and how does it affect audio quality? In the audio field the term jitter designates a timing uncertainty of digital clock signals. In an analog to digital converter (A/D) the analog signal is sampled (measured) at regular time intervals; in the case of a CD, 44 100 times a second or every 22.675737 μs (microseconds). If these time intervals are not strictly constant then one talks of a jittery conversion clock. In practice it is of course not possible to generate *exactly* the same time interval between each and every sample. After all, even digital signals are analog in their properties and thus are influenced by noise, crosstalk, power supply fluctuations, temperature etc.

Hence a jittery clock introduces errors to the measurements taken by the A/D, resulting from measurements being taken at the wrong time. One can easily observe that the level of the error introduced is higher during high audio frequencies, because high frequency signals have a steeper signal form. A good designer takes care that the jitter amount in his/her design is minimized as well as possible.

What type of equipment can be compromised by jitter? There are three types: The A/D converter as described above, then there is the D/A converter where the same mechanism as in the A/D converter applies and the third is the asynchronous sample rate converter (ASRC). The ASRC is not something usually found in Hi-Fi systems. It is used by sound engineers to change the sample rate from e.g. 96 kHz to 44.1 kHz, or e.g. for putting a 96 kHz recording onto a 44.1 kHz CD.

You may now argue that in High-End Hi-Fi there are such things as „over-samplers“ or „upsamplers“. Yes, those are in essence sampling rate converters, however in a well-designed system these converters employ a synchronous design, where jitter does not play any role. Of course a conversion between 96 kHz and 44.1 kHz as in the example above can be done in a synchronous manner as well. An ASRC in fact is only required either where one or both of the sampling frequencies involved are changing over time („varispeed“ mode of digital audio recorders) or where it is unpractical to synchronize the two sampling frequencies.

So basically in Hi-Fi jitter matters where there are A/D or D/A converters involved. CD and DVD players are by far the most numerous type of equipment employing D/A converters. And of course stand-alone D/A converters. Jitter, being an analog quantity, can creep in at various places. The D/A converter built into CD or DVD players can „infected“ by jitter through various crosstalk mechanisms, like power supply contamination by power hungry motors (spindle/servo), microphony of the crystal generating the sampling clock or capacitive/inductive crosstalk between clock signals etc.

In the standalone D/A converter jitter can be introduced by inferior cables between the source (e.g. CD transport) and the D/A converter unit or by the same mechanisms as described above — except for the motors of course. In the case of a stand-alone D/A converter (as the DAC202), one has to take two different jitter contamination paths into account. One is the internal path where internal signals can affect the jitter amount of the sampling clock generator. Here, all the good old analog design principles have to be applied. Such as shielding from electric or magnetic fields, good grounding, good power supply decoupling, good signal transmission between the clock generator and the actual D/A chip.

The other path is the external signal coming from the source to which the sampling clock has to be locked. In other words the D/A converter has to run synchronous to the incoming digital audio signal and thus the frequency of the internal sampling clock generator has to be controlled so that it runs at the same sampling speed as the source (e.g. CD transport). This controlling is done by a phase locked loop (PLL), which is a control system with error feedback. Of course the PLL has to be able to follow the long term fluctuations of the source, e.g. the sampling rate of the source will alter slightly over time or over temperature, it will not be a constant 44.1 kHz in the case of a CD. But the PLL should not follow the short-term fluctuations (jitter). Think of the PLL as being like a very slow-reacting flywheel. In the DAC202 we employ a two-stage PLL circuitry, which very effectively suppresses jitter. A common problem with most PLLs used in audio circuitry is that they suppress jitter only for higher frequencies. Jitter frequencies that are low (e.g. below 1 kHz or so) are often only marginally suppressed. It has been shown that low frequency jitter can have a large influence on the audio quality though. The DAC202 suppresses even very low frequency jitter components.

This means that the DAC202 is virtually immune to the quality of the audio source regarding jitter. For a CD transport as a source this means that as long as the data is read off the CD in a correct manner (i.e. no interpolations or mutes) you should hardly hear any difference between different makes of CD transports or between different pressings of the same CD. Also „accessories“ like disk dampening devices or extremely expensive digital cables will not make any difference in sonic quality. Of course it is always a good idea to have a good quality cable for digital (or analog) audio transmission — but within reason.

1.2 Upsampling, Oversampling and Sampling Rate Conversion in General

In consumer audio circles the two terms oversampling and upsampling are in common use. Both terms essentially mean the same, a change in the sampling frequency to higher values. Upsampling usually means the change in sampling rate using a dedicated algorithm (e.g. implemented on a digital signal processor chip (DSP)) ahead of the final D/A conversion (the D/A chip), while oversampling means the change in sampling rate employed in today's modern D/A converter chips themselves.

But let's start at the beginning. What is the sampling frequency? For any digital storage or transmission it is necessary to have time discrete samples of the signal which has to be processed. I.e. the analog signal has to be sampled at discrete time intervals and later converted to digital numbers (also see section *Jitter Suppression and Clocking*, p. 3). This sampling and conversion process happens in the so-called analog to digital converter (A/D). The inverse in the digital to analog converter (D/A).

A physical law states that in order to represent any given analog signal in the digital domain, one has to sample that signal with at least twice the frequency of the highest frequency contained in the analog signal. If this law is violated so called aliasing components are generated which are perceived as a very nasty kind of distortion. So if one defines the audio band of interest to lie between 20 Hz and 20 kHz, then the minimum sampling frequency for such signals must be 40 kHz.

For practical reasons explained below, the sampling frequency of 44.1 kHz was chosen for the CD. A sampling frequency of 44.1 kHz allows to represent signals up to 22.05 kHz. The designer of the system has to take care that any frequencies above 22.05 kHz are sufficiently suppressed before sampling at 44.1 kHz. This suppression is done with the help of a low pass filter, which cuts off the frequencies above 22.05 kHz. In practice such a filter has a limited steepness, i.e. if it suppresses frequencies above 22.05 kHz it also suppresses frequencies between 20 kHz and 22.05 kHz to some extent. So in order to have a filter which sufficiently suppresses frequencies above 22.05 kHz, one has to allow it to have a so-called transition band between 20 kHz and 22.05 kHz where it gradually builds up its suppression.

Note that so far we have talked about the so-called anti-aliasing filter, which filters the audio signal ahead of the A/D conversion process. For the D/A conversion, which is of more interest to the High-End Hi-Fi enthusiast, essentially the same filter is required. This is because after the D/A conversion we have a time discrete analog signal, i.e. a signal that looks like steps, having the rate of the sampling frequency.

Such a signal contains not only the original audio signal between 20 Hz and 20 kHz but also replicas of the same signal symmetrical around multiples of the sampling frequency. This may sound complicated, but the essence is that there are now signals above 22.05 kHz. These signals come from the sampling process. There are now frequencies above 22.05 kHz which have to be suppressed, so that they do not cause any intermodulation distortion in the amplifiers and speakers, do not burn tweeters or do not make the dog go mad.

Again, a low pass filter, which is called a „reconstruction filter“, is here to suppress those frequencies. The same applies to the reconstruction filter as to the anti-aliasing filter: pass-band up to 20 kHz, transition-band between 20 kHz and 22.05 kHz, stop-band above 22.05 kHz. You may think that such a filter is rather ,,steep“, e.g. frequencies up to 20 kHz go through unaffected and frequencies above 22.05 kHz are suppressed to maybe $\frac{1}{100\,000}^{\text{th}}$ of their initial value. You are right, such a filter is very steep and as such has some nasty side effects. For instance it does strange things to the phase near the cutoff frequency (20 kHz) or it shows ringing due to the high steepness. In the early days of digital audio these side effects have been recognized as being one of the main culprits for digital audio to sound bad.

So engineers looked for ways to enhance those filters. They can't be eliminated because we are talking laws of physics here. But what if we run the whole thing at higher sampling rates? Like 96 kHz or so? With 96 kHz we can allow frequencies up to 48 kHz, so the reconstruction filter can have a transition band between 20 kHz and 48 kHz, a very much relaxed frequency response indeed. So let's run the whole at 96 kHz or even higher! Well — the CD stays at 44.1 kHz. So in order to have that analog lowpass filter (the reconstruction filter) to run at a relaxed frequency response we have to change the sampling frequency before the D/A process. Here is where the upsampler comes in. It takes the 44.1 kHz from the CD and upsamples it to 88.2 kHz or 176.4 kHz or even higher. The output of the upsampler is then fed to the D/A converters, which in turn feeds the reconstruction filter. All modern audio D/A converter chips have such an upsampler (or oversampler) already built into the chip. One particular chip, for instance, upsamples the signal by a factor of eight, i.e. 44.1 kHz ends up at 352.8 kHz. Such a high sampling frequency relaxes the job of the reconstruction filter very much; it can be built with a simple 3$^{\text{rd}}$ order filter.

So, how come that upsamplers are such a big thing in High-End Hi-Fi circles? The problem with the upsamplers is that they are filters again, digital ones, but still filters. So in essence the problem of the analog reconstruction filter has been transferred to the digital domain into the upsampler filters. The big advantage when doing it in the digital domain is that it can be done with a linear phase response, which means that there are no strange phase shifts near 20 kHz and the ringing can also be controlled to some extent. Digital filters in turn have other problems and of course have quite a few degrees of freedom for the designer to specify. This means that the quality of digital filters can vary at least as much as the quality of analog filters can. So for a High-End Hi-Fi designer it is a question whether the oversampling filter built into the D/A chips lives up to his/her expectations. If not, he/she can choose to design his/her own upsampler and bypass part of or the whole oversampler in the D/A chip. This gives the High-End Hi-Fi designer yet another degree of freedom to optimize the sonic quality of the product.

1.3 Reconstruction Filters

Reconstruction filters have been mentioned in section ,,Upsampling, Oversampling and Sampling Rate Conversion in General“ (p. 5). If you have read that paragraph you know what the purpose of the reconstruction filter is. The main point about this analog filter is that its frequency response should be as smooth and flat as possible in order to have a virtually linear phase response. The DAC202 employs a 3rd order filter for that purpose.

1.4 Analog Output Stages

The DAC202 employs separate output stages for the main output and the headphone output. Both stages use state of the art operational amplifiers with high slew rate. A topology with a very low output impedance has been chosen. This assures that the performance of the DAC202 and the subsequent amplifier combination is not compromised by the cables between the two or by the input impedance characteristics of the amplifier.

1.5 Dithering

You have probably not heard the term dithering in conjunction with audio. Actually it is a term widely used in the professional audio realm but not so much in the High-End Hi-Fi market.

What is dithering? Suppose a digital recording has been made with a 24 bit A/D converter and a 24 bit recorder. Now this recording should be transferred to a CD, which has just 16 bits per sample, as you know. What to do with those 8 bits, which are too many? The simplest way is to cut them off, truncate them. This, unfortunately, generates harmonic distortions at low levels, but which nonetheless cause the audio to sound harsh and unpleasant. The harmonic distortion is generated because the eight bits, which are cut off from the 24 bits, are correlated with the audio signal, hence the resulting error is also correlated and thus there are distortions and not just noise (noise would be uncorrelated). The dithering technique now is used to de-correlate the error from the signal. This can be achieved by adding a very low level noise to the original 24 bit signal before truncation. After truncation the signal does not show any distortion components but a slightly increased noise floor. This works like magic — the distortion is replaced by a small noise which is much more pleasant. I have given the example of a 24 bit recording, which has to be truncated to 16 bits. Where is the application in High-End Hi-Fi audio? More and more signal processing is implemented in the digital domain. Think of digital equalizers, digital volume controls, upsamplers, digital pre-amplifiers, decoders for encoded signals on DVD etc. All those applications perform some mathematical operations on the digital audio signal. This in turn causes the wordlength of the signal to be increased; an input signal to an upsampler may have a wordlength of 16 bits (off a CD), but the output signal of the upsampler may have 24 bits or even more. This comes from the fact that the mathematical operations employed in such devices increase the word length. A multiplication of two 2-digit numbers results in a four-digit number. So after the upsampler the word length may be higher than the subsequent processor may be able to accept. In this example, after the upsampler there may be a D/A converter with a 24 bit input word length capability. So if the upsampler generates a word length of more than 24 bits it should be dithered to 24 bits for maximum signal fidelity.

Another application where dither is important is described in the next paragraph.

1.6 Digital Level Control

In High-End Hi-Fi circles a level control done in the digital domain is often viewed as being inferior to one operating in the analog domain. Let's look on how a digital level control works and why it can be an excellent solution if it is properly implemented.

A level control is a multiplication of the audio signal with a constant, the ,,gain factor“. The gain factor usually is in the range of zero (signal fully off) to one (signal untouched). A factor of 0.5 then means that the audio signal is attenuated to half of its amplitude. What exactly happens when we multiply two numbers? If we e.g. multiply a 2 digit and a 3 digit number, the resulting number can be up to 5 digits long (the sum 2 plus 3). As an example: 30 times 500 equals 15 000; 2 digits times 3 digits yields a 5 digit result.

In digital audio, the numbers are represented in the binary system, not the decimal system. A decimal number consists of digits 0 through 9, a binary number of digits 0 and 1. So a binary number may look like this: 1011 0011 0101 1101. This is a 16 digit or 16 bit binary number, the grouping into 4 bit chunks is for better readability. The audio samples on a CD are represented with such a binary number system with each sample value represented with 16 bits. Now let's assume we have a 8 bit gain factor for a level control. If we apply that to a signal coming off a CD we multiply a 8 bit gain factor with a 16 bit sample value. The result is up to 24 bits long (the sum of the wordlengths of the two factors). An example:

$$0100\ 1001 \times 1001\ 0110\ 0111\ 1011 = 0010\ 1010\ 1110\ 1001\ 0001\ 0011$$

The question now is what do we do with the 24 bit long result? The digital to analog converter which converts the samples after the level control may only be capable to handle 16 bit wide samples. Thus what should we do with the excessive 8 bits? The simplest solution is to truncate the 24 bit sample to 16 bits, i.e. to cut off the 8 least significant bits. The truncated 24 bit result above then would look like this: 0010 1010 1110 1001 i.e. the first 16 bits of the 24 bit result above.

The remaining bits (0001 0011) are discarded. If these bits are discarded an error is introduced. This error is called a quantization error, because the 24 bit result is requantized to 16 bits. Unfortunately the quantization error is part of the audio signal — and if we take that part away from the signal, the signal undergoes some distortion, the so called quantization distortion. The sound example at the link below shows how such a distortion sounds. In this music example a 16 bit signal is truncated to 8 bits; 8 bits in order to clearly show the effect. Notice how the noise (distortion) is modulated by the music signal.

www.weiss-highend.ch/computerplayback/nodither.mp3

This is how a badly implemented digital level control works... Fortunately there is a better way to handle the re-quantizing. One solution would be to use a D/A converter with a higher wordlength, e.g. a 24 bit converter, to accommodate for the 24 bit samples coming out of the level control. This of course would already help a lot, but there is another technique: dithering.

The idea about dithering is to de-correlate the quantization error from the audio signal. As we have seen in the example above, the quantization error depends on the audio signal (i.e. it is correlated with the audio signal). On the other hand, if dither noise is added to the 24 bit sample after the level control and before the re-quantization to 16 bits, the quantization error can be fully decorrelated from the signal. This means instead of distortion there is noise. The music is undistorted. The audio example at the link below is again a 16 bit signal quantized to 8 bits, but with dither noise added. A much more pleasant experience. Notice how the noise stays untouched by the music, i.e. there is no noise modulation.

www.weiss-highend.ch/computerplayback/flatdither.mp3

Dithering does not stop here. More elaborate dithering schemes shape the noise such that it is mainly present at higher frequencies where the human ear is less sensitive. This means that the audible noise is much lower. The link below is again the 16 bit source quantized to 8 bits with noise-shaped dithering. Probably hard to believe that this is only an 8 bit system! Note that the music is not distorted at all, despite the 8 bit resolution. Remember, a 16 bit system has 65 536 quantization steps while a 8 bit system has only 256 quantization steps — a huge difference. And still, the properly dithered 8 bit system sounds great.

www.weiss-highend.ch/computerplayback/shapeddither.mp3

This is what a properly dithered level control is capable to do. You have heard the 8 bit version, imagine that with today's 24 bit converters — no question that a level control with a 24 bit wordlength easily rivals the best analog level controls. By the way, 24 bits means 16 777 216 quantization steps.

The last example below toggles the noiseshaping dither on and off to give a good contrast between dither/no dither versions.

www.weiss-highend.ch/computerplayback/togglingdither.mp3

Dithering is used in many disciplines. Also see

en.wikipedia.org/wiki/Dithering

I hope these excursions into the theory and practice of audio engineering has been useful for you. If you would like to dive further into those issues I recommend visiting the website of Mr. Bob Katz, a renowned mastering engineer and a Weiss Engineering LTD. customer. He publishes articles on dithering and jitter and many other topics at

www.digido.com

Chapter 2

The DAC202 D/A Converter

2.1 Features in Alphabetical Order

Absolute Phase Switch

The absolute phase of the outputs can be inverted for optimizing the sonic impression.

Audio Inputs

One XLR, one RCA and one Toslink connector for AES/EBU or S/PDIF signals. Two Firewire connectors for computer connection.

Audio Outputs

Two XLR and two RCA connectors for analog audio output. One XLR and one RCA connector for AES/EBU and S/PDIF audio output. One $\frac{1}{4}$-inch jack socket for headphones.

Backpanel Elements from Left to Right

- Analog outputs on XLR and RCA connectors
- Digital outputs on XLR and RCA connectors
- Digital inputs on XLR and RCA connectors
- Wordsync input and output on BNC connectors
- Digital input on Toslink connector
- Firewire connectors
- Mains connector with fuse

Figure 2.1: DAC202 backpanel

Converters

Two converters per channel are employed in order to lower the converter imperfections. Separate converters are used for the main outputs and the headphone outputs.

Dual/Single Wire Modes

The AES/EBU inputs/outputs of the DAC202 normally work in the so called *Single Wire* mode, i.e. both audio channels are transferred via a single cable. The DAC202 also supports the *Dual Wire* mode where the two audio channels are transferred via two cables, i.e. left channel is on the XLR connector and the right channel on the RCA connector. This applies to both input and output connectors. The dual wire mode, when activated, is active only at sampling rates of 176.4 kHz or 192 kHz. In dual wire mode the frequency of the wordclock synchronization on the BNC connectors can be chosen to be the sampling rate (i.e. 176.4 kHz or 192 kHz) or half the sampling rate (i.e. 88.2 kHz or 96 kHz).

Frontpanel Elements

- Standby LED
- IR receiver
- Headphone socket
- LCD display
- Rotary encoder with switch

Insert Mode

If the insert mode is activated an external digital audio device (e.g. a digital equalizer) can be looped into the signal path via the XLR input/output con-

Figure 2.2: DAC202 frontpanel

nectors. The resulting signal path thus looks as follows: Firewire (or RCA or Toslink) input → XLR output → external device → XLR input → DAC chip.

LCD Brightness

The brightness of the LCD can be set with two different choices: One brightness level is active when operating the rotary encoder knob or the remote control. The other brightness level is active when the DAC202 or the remote control are not touched. This allows dimming the LCD when the information on the LCD screen is not required.

Level Control Main Output

The output level of the main output can be adjusted in the analog domain in four coarse steps in order to accommodate for the input sensitivity of the subsequent amplifier. A higher resolution level control is implemented in the digital domain and operated from the frontpanel knob or the remote control. The high-resolution level control can be defeated for the main output in case there is another level control available in the audio chain.

Level Control Headphone Output

The output level of the headphone output can be adjusted in the analog domain in four coarse steps in order to accommodate for the headphone sensitivity. A higher resolution level control is implemented in the digital domain and operated from the frontpanel knob or the remote control.

Power Supply

A powerful non-switching power supply is used. All sensitive voltages have their own regulators.

Figure 2.3: DAC202 remote control

Remote Control

The IR remote control allows to control the following parameters:

- Power on/off
- Volume up/down
- Input source (Firewire, XLR, RCA, Toslink)
- Output mute
- Absolute phase normal/inverted
- Upsampling filter type

Signal Routing

Due to the various possible settings for input source, insert mode, dual/single wire modes and sync source there are quite a few routing paths possible. Refer to the operation instructions below for a detailed list of the signal routing.

Synchronization

Wordclock input and output on BNC connectors. Supported sampling rates on all inputs: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz. For all input modes the synchronization source can be freely chosen. With Firewire as input the *Internal* synchronization is typically chosen, which means that the DAC202 is the master clock for the computer.

Transparency Check

This feature can verify the bit transparency of any player software running on a computer. For that purpose audio files are supplied with the DAC202. Playing back these files via the player software to be checked allows the DAC202 software to recognize the bit pattern of the files. If the bits of the files are changed during playback e.g. because of a volume control or EQ or upsampling algorithm etc., the bit transparency check fails. The files supplied cover all the DAC202 supported sampling rates (44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz) as well as 16 bit and 24 bit wordlengths.

Upsampling Filter Selection

The upsampling filter can be selected between *A* and *B*. Filter A has a steeper frequency response than B. Future DAC202 software will offer more filter choices. All DAC202 units can be software updated via Firewire.

2.2 Installation/Operation

Unpacking and Setup of the DAC202

Carefully unpack the DAC202. The following items should be enclosed:

- The DAC202 D/A converter unit
- The IR remote control unit
- A CD with the necessary Firewire drivers for Windows and Mac OS X and with the audio files for the bit transparency check
- This owner's manual
- A certificate of guarantee

Firewire Connection

> *Before connecting the Firewire cable between computer and DAC202 unplug both the computer and the DAC202 from the mains power.*

Mains Connection

Before connecting the mains cable make sure the label on the back of the unit (near the mains inlet) shows the appropriate mains voltage. If this is not the case then the proper mains voltage may have to be selected with a jumper cable inside of the DAC202 unit. Contact your dealer in that case.

Figure 2.4: DAC202 main menu display

First Time Operation

After connecting the necessary cables (the DAC202 can also be operated without computer, e.g. by connecting a CD transport to one of its inputs) switch on the unit by pressing on the rotary encoder knob.

> *When power is applied to the DAC202 the blue standby LED is lit. When the DAC202 is switched on, the blue LED is turned off and after a short while the LCD screen comes on.*

If a computer is connected to the DAC202 via Firewire, the sync source and sync frequency (if applicable) is preferably selected from within the *Weiss Firewire IO* control panel on the computer. This ensures that sync settings on both control panel and device coincide. After a short while the LCD screen lights up and shows the basic start-up screen.

2.3 Main Menu

Figure 2.4 shows the main menu display. In the upper left corner the volume is displayed in dB (decibel), a value of 0.0 is maximum volume. Below the dB figure there is a bar which also represents the volume. In the upper right corner the absolute phase is shown with φ.[1] „$\varphi+$“ means the signal is not inverted, „$\varphi-$“ means the signal is inverted (both channels). In the lower right corner the selected upsampling filter type is shown. It can be *A* or *B* and for later software versions it may be even *C*, *D* etc.

Below the volume bar the current input source is shown. It can be *Firewire*, *AES (XLR)*, *SPDIF (RCA)* or *SPDIF (TOS)*. These are the four input sockets to the DAC202, i.e. Firewire, XLR, RCA and Toslink (optical). Below the input source the sampling rate is shown, if there is a valid signal present at the selected input, otherwise *unlocked* indicates that there isn't a valid input signal.

Rotating the knob causes the volume control to change the value. Pressing the knob when the display shows the main menu activates the selection of the

[1] The Greek character φ (*phi*) is used for the phase angle in electrical engineering.

Figure 2.5: *Options Menu* highlighted

Figure 2.6: *Input mode* selection

Options Menu as shown in figure 2.5. Pressing the knob again enters the *Options Menu* (p. 18).

Input Mode

Rotating the knob instead of pressing it navigates to the input source select as in figure 2.6. Pressing the knob again allows selecting the input source.

If any parameter is shown with the two dots in the upper left and lower left corners, then rotating the knob can change that parameter. To confirm a setting the knob has to be pressed again. The two dots vanish and the parameter is set to the value indicated. Figure 2.6 shows the two dots in the input selection menu. Selectable input sources are:

- Firewire
- AES (XLR)
- SPDIF (RCA)
- SPDIF (TOSLINK)
  Note: up to 96 kHz sampling rate only!

Figure 2.7 shows the AES/EBU input on XLR selected. There isn't any valid signal at the AES/EBU input thus it shows *unlocked*.

Figure 2.7: *AES (XLR)* input unlocked

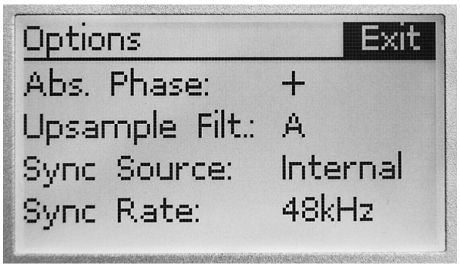

Figure 2.8: The *Options Menu*

2.4 Options Menu

Upon entering the options menus the display as shown in figure 2.8 appears. The highlighted item is the one which can be changed/executed by pressing the knob. E.g. if the knob is pressed with the display as shown, the options menu is exited. Here is a rundown of all entries in the options menu:

Absolute Phase „+“ or „−“. A „+“ means the signal is not inverted when passing through the DAC202. A „−“ means the signal is inverted.

Upsample Filter Upsample filter type *A* or *B*. Later software versions may allow to select *C*, *D* etc. *A* uses a steeper filter than *B*. Also see the Technical Data section (p. 33ff.).

Synchronization Source For all possible input sources (*Firewire*, *AES (XLR)*, *SPDIF (RCA)* and *SPDIF (TOS)*) the following sync sources can be selected:[2]

XLR This selects the XLR input as the synchronization source.

RCA This selects the RCA input as the synchronization source.

Toslink This selects the Toslink input as the synchronization source.

[2]These instructions assume that neither dual wire nor insert modes are selected, for dual wire and/or insert modes check the instructions further down.

WC BNC This selects the BNC connector at the rear of the DAC202 as the synchronization source. If the DAC202 is used in dual wire mode read the instructions for the dual wire mode regarding external synchronization.

1394 bus This slaves the DAC202 clock to the Firewire bus. This setting is only required if more than one DAC202 unit is connected to the same computer for multichannel playback. In that case one of the DAC202 is the master clock and the other DAC202 units have to be slaved to that master DAC202. This is done by setting the slave DAC202 to *1394 bus*.

Internal The DAC202 generates the sampling rate clock internally. Note that in this mode the source has to be synchronized to the internally generated sync. With *Firewire* as input source this is done automatically via Firewire. With the other inputs the source, e.g. a CD transport, has to be synchronized via e.g. the sync out BNC connector at the back of the DAC202.

If a computer is connected to the DAC202 via Firewire, the Sync Source parameter has to be selected from the Weiss Firewire IO *control panel on the computer.*

Synchronization Rate Depending on the sync source selected there is either the sampling rate shown or the word *autolock*. If the sampling rate is indicated (i.e. *Internal* is selected as sync source) then it is possible to change the rate to the appropriate value. Usually the sampling rate is set by the player program running on the computer in that case.

All sync settings (source and rate) are saved separately for each input mode such as to ensure consistency when switching from one input to another.

LCD Brightness Sets the active mode's LCD brightness.

LCD Dim Level sets the LCD brightness when in dimmed mode. The dimmed mode is entered after some time of inactivity from frontpanel knob or remote control.

Dual Wire *Disabled* means that the signals at the XLR or RCA or Toslink inputs are treated as single wire AES/EBU signals with sampling rates up to 192 kHz. Also the XLR and RCA outputs operate in single wire mode up to 192 kHz. If *enabled*, the XLR and RCA inputs (or outputs) are a dual wire pair, i.e. the XLR connectors carry the left channel and the RCA connectors carry the right channel. This is the case only for sampling rates of 176.4 or 192 kHz though! All other sampling rates are disabled for external sync sources and are operated in single wire mode for internal sync.

Figure 2.9: *Options Menu*: LCD and DW settings

DW WCLK (Dual Wire Wordclock) May be set to *audiorate* or *halfrate*. *Audiorate* means that the wordclock signal at the BNC connectors (input or output) is at the actual audio sampling rate when the unit operates in dual wire mode. I.e. the wordclock rate at the BNC connectors is:

Audio sampling rate:	BNC connectors rate:
44.1 kHz	44.1 kHz
48 kHz	48 kHz
88.2 kHz	88.2 kHz
96 kHz	96 kHz
176.4 kHz	176.4 kHz
192 kHz	192 kHz

If *halfrate* is selected the wordclock rate at the BNC connectors is:

Audio sampling rate:	BNC connectors rate:
44.1 kHz	44.1 kHz
48 kHz	48 kHz
88.2 kHz	88.2 kHz
96 kHz	96 kHz
176.4 kHz	**88.2 kHz**
192 kHz	**96 kHz**

Insert Mode When enabled, an external digital audio device (e.g. a digital equalizer) can be inserted into the signal path between e.g. the source via Firewire and the D/A converter. The insert mode possibilities are explained further down.

Main Output Attenuation If engaged, the volume knob and the remote control volume work on both the volume of the main output and the headphone output. This mode is selected if the DAC202 is used as a preamplifier. If bypassed, the volume knob and the remote control volume work only on the headphone output. The main outputs are set to full volume (0.0 dB). This mode is used if there is another volume control used in the chain.

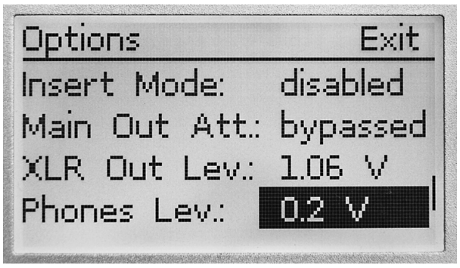

Figure 2.10: *Options Menu*: insert and level settings

XLR Output Level Main output level in V_{rms}. There are four settings to chose from: 8.15 V_{rms}, 4.15 V_{rms}, 2.12 V_{rms} and 1.06 V_{rms}. Best is to start off with the lowest value (1.06 V_{rms}) and have the volume knob at 0.0 dB. If the audio volume is at a comfortable level, i.e. does not need to be louder, leave the setting at 1.06 V_{rms}. If it needs to be louder select the next setting (2.12 V_{rms}). I.e. select the setting, which gives you a comfortably loud level with the volume knob, set to 0.0 dB, i.e. the maximum level.

Phones Level The same as the main output level, but for the headphone output. Be careful when selecting that level! The settings are: 0.2 V_{rms}, 0.9 V_{rms}, 2.7 V_{rms}, 5.2 V_{rms}. Start off with the lowest level (0.2 V_{rms}). This level is fine for many low impedance headphones. If the volume is too low even for a 0.0 dB setting of the volume knob then get to the next higher setting. The highest setting (5.2 V_{rms}) is used for very insensitive headphones like e.g. the AKG K1000.

Transparency This allows checking the player program on your computer for bit transparency. To do this you need to play the audio files supplied on the CD coming with the DAC202. Copy these files onto your drive holding your audio files. There are two files for each sampling rate, one at a 16 bit wordlength (i.e. the system is checked for 16 bit transparency) and one at a 24 bit wordlength for 24 bit transparency checking. The files are in `.wav` format, which is an uncompressed format, supported by most players. When playing a particular file make sure the DAC202 shows the same sampling rate as the file played has. If the two rates do not match then there is a sampling rate conversion going on and bit transparency cannot be achieved. When this is all fine, play the file and activate the transparency check by pressing the button when the *run* word is highlighted. If the player software is bit transparent then the wordlength of the file played is shown, i.e. 16 bit or 24 bit. If the player software is not bit transparent the word *fail* is shown. *Fail* means that the bits of the original audio file get changed somewhere on the path between the hard disk and the DAC202.

If the player does not seem to be bit transparent then this can have several causes, like:

- A volume control not at 0 dB gain

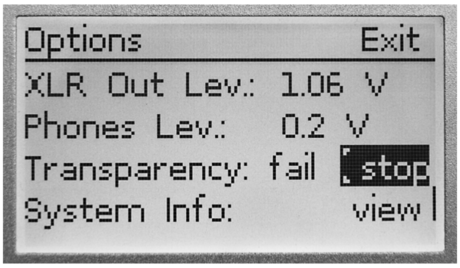

Figure 2.11: *Options Menu*: transparency check

- An equalizer
- A sampling rate conversion
- A ,,sound enhancer" feature and more

Make sure all those processing elements are bypassed. Particularly the sampling rate conversion can creep in ,,unnoticed". The sampling rate in the *Weiss Firewire IO* window has to match the sampling rate of the file played; else a conversion is going on in the operating system. For iTunes there is another thing to know: Whenever the sampling rate is changed in the AudioMidi setup or the *Weiss Firewire IO* window, the iTunes program has to be restarted to gain bit transparency again. For iTunes running on a Mac OS X computer a program like Sonic Studio's *Amarra* is highly recommended. With Amarra it is possible to switch the sampling rate in AudioMidi (i.e. in the DAC202) automatically depending on the sampling rate of the file played. Amarra works in conjunction with iTunes.

On a Windows based system the use of ASIO or WASAPI is highly recommended. These systems make it simple to achieve bit transparent playback. In addition the sampling rate of the DAC202 is switched automatically depending on the sampling rate of the file played. Note that the test audio files do not generate any audible audio signal. This makes sure that your speakers are protected when doing the test.

Notice that transparency check is applicable on either input mode, i.e. you may also check for bit transparency when playing the supplied files into either of the AES inputs.

System Info Includes factory reset, information on the firmware version, device identifier, etc. Factory reset allows to reset the DAC202 device configuration to our default factory settings.

2.5 Signal Routing in Various Operation Modes

This paragraph describes some advanced features of the DAC202: The Insert mode, the Dual Wire mode and external synchronization via Wordclock on the

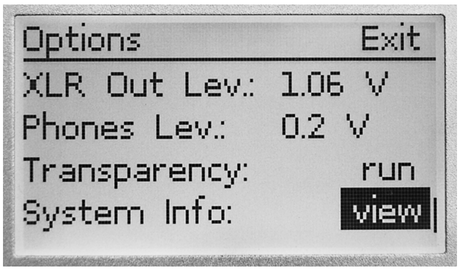

Figure 2.12: *Options Menu*: system info

BNC connector or via the other digital audio inputs. There are four basic modes, which specify the input source for the DAC202:

- Firewire
- AES (XLR)
- SPDIF (RCA)
- SPDIF (TOSLINK)
  Note: up to 96 kHz sampling rate only

The respective input source is routed to all possible output destinations (Firewire, XLR, RCA, DAC chip) as well as to the *APB* (ARM processing buffer) for the bit transparency check.

When the Insert mode is engaged, the input source is not routed to the DAC chip; the DAC chip receives its signal from a secondary input instead. See Insert routing below. This allows the user to insert an external processing stage into the audio stream, e.g. a digital equalizer. The secondary input (so called insert return) is selected within the options menu (p. 20). Each basic mode can be operated in either:

Single Wire L and R channels on a single AES/EBU or S/PDIF cable (the normal mode).

or

Dual Wire L and R channels on two separate AES/EBU or S/PDIF cables. If the dual wire mode is activated it is active only with sampling rates of 176.4 kHz or 192 kHz. With all other sampling rates the DAC202 automatically switches to dingle wire mode. Dual wire mode is useful for digital audio equipment connection to the DAC202 where the units only support the high sampling rates (176.4 kHz or 192 kHz) in dual wire mode.

Besides the inputs as sync sources, wordclock („sync“) input/output on BNC connectors are available. The applicable wordclock rate for dual wire mode is configurable to be:

halfrate (88.2 kHz and 96 kHz)

or

audiorate (174.6 kHz and 192 kHz)

In the following we will discuss the various operating modes.

Mode: Firewire input/single wire/no insert

- The Firewire input is routed to: DAC chip, XLR out, RCA out, APB.
- The Firewire output (going to the computer for recording) is fed from the source specified as the sync source. XLR selected as sync source: XLR signal goes to Firewire. RCA selected as sync source: RCA signal goes to Firewire. Toslink selected as sync source: Toslink signal goes to Firewire. Internal or WC BNC or 1394 bus selected as sync source: The Firewire output is muted.

Mode: Firewire input/single wire/insert active

- The Firewire input is routed to: XLR out, RCA out, APB.
- The Firewire output (going to the computer for recording) is fed from the source specified as the sync source. XLR selected as sync source: XLR signal goes to Firewire. RCA selected as sync source: RCA signal goes to Firewire. Toslink selected as sync source: Toslink signal goes to Firewire. Internal or WC BNC or 1394 bus selected as sync source: The Firewire output is muted.
- The DAC chip is fed from the source selected in the Insert mode menu, i.e. from either the XLR or RCA or Toslink input.

Mode: Firewire input/dual wire/no insert

- The Firewire input is routed to: DAC chip, XLR out (left channel), RCA out (right channel), APB.
- The Firewire output (going to the computer for recording) is fed from the source specified as the sync source. XLR/RCA selected as sync source: XLR is the left channel signal going to Firewire and RCA is the right channel signal going to Firewire. Toslink selected as sync source: Not supported. Internal or WC BNC or 1394 bus selected as sync source: The Firewire output is muted.

Mode: Firewire input/dual wire/insert active

- The Firewire input is routed to: XLR out (left channel), RCA out (right channel), APB.
- The Firewire output (going to the computer for recording) is fed from the source specified as the sync source. XLR/RCA selected as sync source:

XLR is the left channel signal going to Firewire and RCA is the right channel signal going to Firewire. Toslink selected as sync source: Not supported. Internal or WC BNC or 1394 bus selected as sync source: The Firewire output is muted.

- The DAC chip is fed from: XLR input (left channel) and RCA input (right channel).

Mode: AES(XLR) input/single wire/no insert

- The XLR input is routed to: DAC chip, XLR out, RCA out, Firewire out, APB.
- The Firewire input, RCA input and Toslink input are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: AES(XLR) input/single wire/insert active

- The XLR input is routed to: RCA out, APB.
- The DAC chip, Firewire output and XLR output are fed from the RCA input or Toslink input as selected in the insert menu.
- The Firewire input is not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: AES(XLR) input/dual wire/no insert

- The XLR input (left channel) and the RCA input (right channel) are routed to: DAC chip, XLR out (left channel) RCA out (right channel), Firewire out, APB.
- The Firewire input and Toslink inputs are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: AES(XLR) input/dual wire/insert active

This mode is not supported.

Mode: S/PDIF(RCA) input/single wire/no insert

- The RCA input is routed to: DAC chip, XLR out, RCA out, Firewire out, APB.
- The Firewire input, XLR input and Toslink input are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(RCA) input/single wire/insert active

- The RCA input is routed to: XLR out, APB.
- The DAC chip, Firewire output and RCA output are fed from the XLR input or Toslink input as selected in the insert menu.
- The Firewire input is not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(RCA)/dual wire/no insert

- The XLR input (left channel) and the RCA input (right channel) are routed to: DAC chip, XLR out (left channel) RCA out (right channel), Firewire out, APB.
- The Firewire input and Toslink inputs are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(RCA) input/dual wire/insert active

This mode is not supported.

Mode: S/PDIF(TOS) input/single wire/no insert

- The TOS input is routed to: DAC chip, XLR out, RCA out, Firewire out, APB.
- The Firewire input, RCA input and XLR input are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(TOS) input/single wire/insert active

- The TOS input is routed to: APB plus either RCA or XLR out depending on the insert routing selected, i.e. the insert can go via the RCA or the XLR connectors.
- The DAC chip, Firewire output and RCA (or XLR) output are fed from the XLR (or RCA) input as selected in the insert menu.
- The Firewire input is not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(TOS)/dual wire/no insert

- The XLR input (left channel) and the TOS input (right channel) are routed to: DAC chip, XLR out (left channel) RCA out (right channel), Firewire out, APB.
- The Firewire input and RCA inputs are not routed anywhere.
- The sync source can be specified as the XLR input, the RCA input, the Toslink input the WC BNC input, the 1394 bus or the internal sync generator.

Mode: S/PDIF(RCA) input/dual wire/insert active

This mode is not supported.

2.6 Software Installation

Perform the following installation procedure before connecting the DAC202 to the computer. The necessary files are supplied on the enclosed drivers CD.

Windows

1. Do not connect the device.
2. Double click *WeissFirewireInstaller.exe*.
3. Click *Next*.
4. Select the directory where you'd like to install the tools. Usually you can use the default values and click *Next*.
5. Select if you'd like to create a desktop icon. *Next*.
6. Click *Install*.

7. You will be asked if you'd like to continue the installation because the driver/software didn't pass the Windows-Logo-Test. Select *Continue*.

8. Select *Yes, restart the computer now* and click *Finish*.

Mac OS X

1. Mount the *WeissFirewire.dmg* disk image by double clicking it.

2. From the mounted drive double click *WeissFirewire-3.5.3.8786.pkg* (the version numbers can be different of course).

3. Click *Continue*.

4. Select the drive (usually you leave it at the defaults).

5. Click *Continue*.

6. Click *Upgrade* or *Install*.

7. You'll be asked to login as administrator.

8. Confirm *Continue Installation* when warned that the computer requires a reboot after install.

9. Click Restart.

After installing the drivers, connect the DAC202 to the computer and connect the power cord to the DAC202. Switch the DAC202 on. The DAC202 should now be recognized automatically. In Windows tell the installation window that you do not want to check the Microsoft website for drivers and then let the drivers be installed automatically. Ignore warnings concerning *Windows Logo Test* and continue the installation until completed. You will be asked to install drivers for *Weiss Engineering Ltd. –Firewire Unit–*.

2.7 Software Setup

The connected DAC202 device can be controlled through the *Weiss Firewire IO* control panel.

Windows

The control panel can be accessed by clicking on the *Weiss Firewire IO* icon on the desktop. Available tabs:

Global Settings/Bus

Master Is the device which is sync master on the virtual bus in case multiple devices (DAC202s) are connected.

Sampling Rate The sampling rate of the device when internally clocked. When clocked/syncing to one of the other interfaces (*XLR*, *RCA*, *TOS*) the sampling rate indicated reflects the one fed from the external device.

Sync Source The clock to which the DAC202 should sync to. Usually this is the DAC202's internal clock generator.

Buffer Size Larger buffer sizes increase robustness against dropouts; lower buffer sizes provide low latency.

Operation Mode Determines the stability of the system. Try other modes if there are clicks in the music.

Global Settings/WDM Enables the WDM driver.

Global Settings/DPC Determines your computer's performance and recommends an operation mode.

Global Settings/System Some utilities to determine the chipset in your computer and to get information on the supported chipset. Required for debugging if problems with the Firewire connection are encountered.

Global Settings/Info Information about the driver version.

Device Settings/General The device settings should be pretty self-explanatory.

Device Settings/Firmware Loader Allows uploading new firmware to the DAC202. Not used for normal operation.

Mac OS X

Configure the DAC202 via the *Audio MIDI Setup* (*Applications* → *Utilities*) and the *WeissFirewire Control Panel* (*Applications*). Note that most settings controllable from the control panel are available only in Firewire mode.

Global Settings/Bus

Master Is the device which is sync master on the virtual bus in case multiple devices (DAC202s) are connected.

Sync Source The clock to which the DAC202 should sync to. Usually this is the DAC202's internal clock generator.

Sampling Rate The sampling rate of the device when internally clocked. When clocked/syncing to one of the other interfaces (*XLR*, *RCA*, *TOS*) the sampling rate indicated reflects the one fed from the external device.

Operation Mode Determines the stability of the system. Try other modes if there are clicks in the music. Note that when using the *XLR*, *RCA* or *TOS* inputs there is no need to hook up a computer to the DAC202.

Global Settings/Info Information about the driver version.

Device Settings/General The device settings should be pretty self-explanatory.

Device Settings/Firmware Loader Allows uploading new firmware to the DAC202. Not used for normal operation.

Chapter 3

DAC202 Technical Data

3.1 Digital Inputs

(1) XLR connector, (1) RCA connector, (1) Toslink connector (optical), (2) Firewire connectors. All inputs accept professional or consumer standard, i.e. accept AES/EBU or S/PDIF signals.

Supported sampling frequencies are 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz or 192 kHz on any of the inputs, except Toslink which handles 96 kHz maximum. The dual wire mode can be activated for sampling rates of 176.4 kHz or 192 kHz exclusively. Maximum input wordlength is 24 bit.

3.2 Digital Outputs

(1) XLR connector, (1) RCA connector, (2) Firewire connectors. Professional channel status data on the XLR and RCA outputs. The dual wire mode can be activated for sampling rates of 176.4 kHz or 192 kHz exclusively.

3.3 Main Analog Outputs

(2) XLR connectors (hot on pin 2), DC coupled, short circuit proof output circuitry, output impedance 44 Ω. (2) RCA connectors, DC coupled, short circuit proof output circuitry, output impedance 22 Ω. The output level is selectable via the LCD menu; 4 settings are provided as shown below.

XLR Output

8.15 V_{rms}	+20.44 dBu	with a 0 dBFS sinewave input
4.15 V_{rms}	+14.57 dBu	with a 0 dBFS sinewave input
2.12 V_{rms}	+7.74 dBu	with a 0 dBFS sinewave input
1.06 V_{rms}	+2.72 dBu	with a 0 dBFS sinewave input

These levels are achieved with a 0.0 dB setting for the level control on the LCD screen. Suggested subsequent amplifier input impedances:

$8.15\,V_{rms}$ setting:	$500\,\Omega$ or higher
$4.15\,V_{rms}$ setting:	$300\,\Omega$ or higher
$2.12\,V_{rms}$ setting:	$150\,\Omega$ or higher
$1.06\,V_{rms}$ setting:	$70\,\Omega$ or higher

RCA Output

$4.08\,V_{rms}$	$+14.42$ dBu	with a 0 dBFS sinewave input
$2.08\,V_{rms}$	$+8.55$ dBu	with a 0 dBFS sinewave input
$1.06\,V_{rms}$	$+1.72$ dBu	with a 0 dBFS sinewave input
$0.53\,V_{rms}$	-3.30 dBu	with a 0 dBFS sinewave input

These levels are achieved with a 0.0 dB setting for the level control on the LCD screen. Suggested subsequent amplifier input impedances:

$4.08\,V_{rms}$ setting:	$250\,\Omega$ or higher
$2.08\,V_{rms}$ setting:	$150\,\Omega$ or higher
$1.06\,V_{rms}$ setting:	$75\,\Omega$ or higher
$0.53\,V_{rms}$ setting:	$35\,\Omega$ or higher

3.4 Headphone Output

(1) stereo $\frac{1}{4}$-inch jack connector, DC coupled, short circuit proof output circuitry. The output level is selectable via the LCD menu; 4 settings are provided as shown below.

$5.2\,V_{rms}$	$+16.53$ dBu	with a 0 dBFS sinewave input
$2.7\,V_{rms}$	$+10.84$ dBu	with a 0 dBFS sinewave input
$0.9\,V_{rms}$	$+1.30$ dBu	with a 0 dBFS sinewave input
$0.2\,V_{rms}$	-11.77 dBu	with a 0 dBFS sinewave input

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

$5.2\,V_{rms}$ setting:	$100\,\Omega$ or higher
$2.7\,V_{rms}$ setting:	$50\,\Omega$ or higher
$0.9\,V_{rms}$ setting:	$16\,\Omega$ or higher
$0.2\,V_{rms}$ setting:	$4\,\Omega$ or higher

3.5 Synchronization

Synchronized via the input signal, the internal oscillator or via a wordclock signal (TTL level/75 Ω) on the BNC input. Sampling frequencies: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz. Wordclock output (TTL level/75 Ω) on BNC for synchronization of other equipment.

3.6 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: 1 VA max.

3.7 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 (F_s), 22 kHz measurement bandwidth, unweighted, 0 dBr equals the output level at 0 dBFS input.

Frequency Response

F_s= 44.1 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 44.1 kHz,	Filter B,	0 Hz–20 kHz:	within ±1.3 dB
F_s= 88.2 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 88.2 kHz,	Filter A,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 88.2 kHz,	Filter B,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 88.2 kHz,	Filter B,	0 Hz–40 kHz:	within ±1.5 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–80 kHz:	within ±2.5 dB
F_s= 176.4 kHz,	Filter B,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 176.4 kHz,	Filter B,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 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

3.8 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 (F_s), 22 kHz measurement bandwidth, unweighted, 0 dBr equals the output level at 0 dBFS input.

Frequency Response

F_s= 44.1 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 44.1 kHz,	Filter B,	0 Hz–20 kHz:	within ±1.3 dB
F_s= 88.2 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 88.2 kHz,	Filter A,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 88.2 kHz,	Filter B,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 88.2 kHz,	Filter B,	0 Hz–40 kHz:	within ±1.9 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 176.4 kHz,	Filter A,	0 Hz–80 kHz:	within ±2.6 dB
F_s= 176.4 kHz,	Filter B,	0 Hz–20 kHz:	within ±0.25 dB
F_s= 176.4 kHz,	Filter B,	0 Hz–40 kHz:	within ±0.8 dB
F_s= 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, maximum output level, 100 kΩ load, 1 kHz, all components at less than −120 dB
- At 0 dBFS input level, maximum output level, 600 Ω load, 1 kHz, all components at less than −120 dB

- At 0 dBFS input level, maximum output level, 300 Ω load, 1 kHz, all components at less than −120 dB
- At 0 dBFS input level, maximum output level, 100 kΩ load, 4 kHz, all components at less than −120 dB
- At 0 dBFS input level, maximum output level, 600 Ω load, 4 kHz, all components at less than −120 dB
- At 0 dBFS input level, maximum 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

Appendix A

Contact

For any questions, suggestions etc. feel free to contact us at:

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