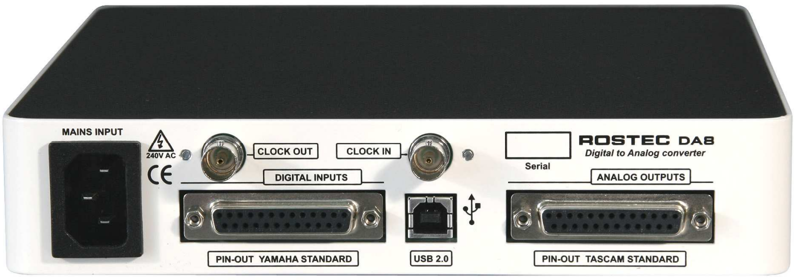

ROSTEC DA8
8 channel Digital to Analog Converter

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DA8 Digital to Analog Converter

Features

- **8 channels of Digital to Analog conversion from the DAW via USB.**
- **8 channels of Digital to Analog conversion from the AES inputs.**
- **Hardware Sample Rate Conversion of AES inputs (selectable).**
- **AES inputs is compatible with SPDIF format. SRC up-sampling provides an enhanced listening experience in HI-FI applications**
- **192 kHz 24 bit high-end latest generation D/A converters.**
- **Dynamic range 118 dB. Higher internally, but limited by op-amp noise.**
- **USB 2.0 class compliant bit perfect interface.**
- **Transformer balanced 110 ohm AES inputs.**
- **Industry leading Low Latency performance.**
- **Easy interfacing via standard D-sub 25 and USB B connectors.**
- **Internal low jitter Grade 2 word clock generator.**
- **Exceptional high immunity against jitter and digital noise from computer work stations.**
- **Word clock Input has an automatic "sweet spot" detector, providing auto-slicing and clean-up.**
- **Word clock input accept from 0.2 V to 10 V.**
- **Electronically floating balanced analog industrial output buffers.**
- **Analog JFET and BJT cascode technology with the same characteristics as vacuum tubes for exquisite sonic performance.**
- **Analog level signal switching by hermetically sealed gold contact relays.**
- **Analog and digital true balanced architecture throughout the unit.**
- **Linear low noise analog power supply.**
- **Internal magnetic and electrical screening.**
- **Digital Inputs and outputs are ESD protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5.**
- **Sturdy steel metal casing, half size 19” 1U height.**
- **Affordable price. High value for money.**

General description

With its true class A and carefully designed analog filter circuitry, Rostec DA8 represents a sound quality that is far above what you will find in the standard digital interfaces flooding the market today. DA8 is constructed with a throughout balanced architecture, meaning that audio signals, analog as well as digital, are routed internally as balanced signals, not using ground to transfer audio signals at all.
This architecture keeps the audio path free from non-linear distortion from currents in the ground mesh and free from “non musical” signals from external electrical fields, power supply noise and crosstalk from other channels.

All analog level switching is performed by hermetically sealed relays with gold contacts. There are no digital analog switching, no digital level adjustments, and there are no digital processing, except from the DA conversion and SRC processes.
The analog circuit features balanced floating industrial output buffers, which support balanced as well as unbalanced configuration. The analog circuit has, throughout the unit, a 10 dB higher headroom than the digital converter circuit, which means that analog clipping cannot occur.

Also, the power supply is pure analog for a quiet and noise-free environment.
We want none of that severe digital noise pollution from switch mode power supplies, thank you!

This highly sophisticated design gives DA8 a large dynamic range, huge headroom and ultra low noise and distortion, and the result is a beautiful relaxed, open, warm and natural sound, with an impressive amount of details and an astonishing depth and perspective.
DA8 is a highly efficient tool for professional Sound Engineers, who need to accurately evaluate recordings and mix-downs. Sound Engineers are forced to listen for hours and hours every day, and the ugly, cold, flat and lifeless sound you get from a standard digital interface, create substantial ear fatigue, which severely impairs judgment.
In contrast, the natural clear and open sound from DA8 (and its companion AD8) creates a healthy working environment for musicians, producers and engineers. This allows for long hours of work without tiring the ears or compromising abilities.

Digital Inputs

The DA8 has four AES/SPDIF digital inputs (eight audio channels) on the standard 25 pin D-sub connector at the back panel. The connector follows the YAMAHA DB-25 industrial pin-out standard.

The digital inputs are 110 ohm transformer balanced and compatible with the AES3, SPDIF, IEC60958 and EIAJ CP201 interface standards.

Normally a SPDIF connection requires a 75 ohms termination impedance, but this is quite irrelevant for shorter cable lengths of 1 meter or less. When using the SPDIF format with longer cable lengths, impedance matching should be applied, by placing a 240 ohm resistor across the input terminals.

The digital receiver circuit latches the incoming data into a FIFO buffer memory (First In First Out) in order to properly handle any malformed or jittery data from noisy computer workstations or semi-professional equipment.

Latching the data out of the FIFO again with the internal master clock, makes the device virtually immune to incoming jitter. The penalty is a short latency of approx 0.2 msec, but it has a huge positive impact on the sonic performance.

Sample Rate Conversion

The digital inputs feature 4 individually high performance hardware Sample Rate Converters. The SRC works by up-sampling the input audio signal to a very high frequency, and dividing down to the chosen output sampling frequency.

The SRC conversion is automatically turned on and off whenever it is necessary for the smooth continuation of the audio signal.
The SRC is turned off when these three frequencies are the same <u>and</u> EXT LOCK is shown in the display.

1. The selected frequency on the front display.
2. The clock sync input frequency.
3. The sample rate frequency of the incoming audio.

In all other situations, the Sample Rate Conversion is turned on. The user can freely play around with these parameters without penalty.
The Sample Rate Conversion glides smoothly back and forth without any interruption of the audio signal, and the accuracy of the SRC is impressive, with artifacts and distortion products below -140 dB.

USB input

The USB connector is a type B , the most physical robust of all USB connectors. The USB connection features a bit-perfect transmission protocol, which is USB 2.0 class compliant and directly compatible with USB 2.0, 3.0, 3.1, 3.2 and USB-C.
DA8 supports Core-Audio on MAC OS X systems, which means that when plugged in, the 8 channels show up on the MIDI-setup panel and are immediately available as system resources for any DAW.
DA8 is also compatible with Windows 10. When plugged in, the Windows system detects the unit as a 7.1 audio device. Older Windows versions are not supported.

Running the DA8 in USB mode has some restrictions that must be observed.

1. The DAW controls the sampling frequency. Unlike in AES mode, the sampling frequency cannot be changed manually with the arrow buttons.
2. There is no Sample Rate Conversion available. The USB format will not allow that. Any Sample Rate Conversion must be performed within the Work Station or the DAW.
3. If more than one USB unit is connected to the Work Station, all units must be synchronized to the same Word Clock. Else there will be clicking in the sound.

When running in USB mode, the actual sampling rate frequency is shown in the display at the upper right corner.
When USB mode is selected, but no connection to the workstation is made, the sampling frequency will be indicated as NONE, and the unit will be in pause mode, i.e. there will be no clock output and no audio output.

Analog outputs

The DA8 has eight analog audio outputs on the standard 25 pin D-sub connector at the back panel. The connector follows the TASCAM DB-25 industrial pin-out standard.

The analog outputs are intended for use with active control room monitors, featuring high performance balanced floating industrial buffers, able to handle any impedance from 600 ohm to infinite.
The buffers are true floating, which means that their outputs are not referenced to ground, but is configured as the difference between the negative and positive terminals.
The outputs are normally used in balanced mode, but can be configured to run in unbalanced mode. When running in unbalanced mode, the negative terminal MUST be connected to ground.

The analog output level can be selected at the front. The nominal output level is +18 dBu at 0 dBFS, which is the official EBU standard.
The +10 dBu setting is optimal for use with standard (vintage) equipment. Older equipment has a 0 VU level which corresponds to +4 dBu electric level. Setting the maximum digital level to +10 dBu leaves 6 dB of extra headroom for peaks above 0 VU. This setting provides an operating level that brings out the best of most vintage equipment.

Clock sync input and output

The DA8 has input and output for clock synchronization on standard BNC connectors at the back.
The input is 1.5 kohm (i.e. not terminated) and the output clock is 75 ohm 5 Volts, TTL compatible.

The clock input features a high-speed comparator with hysteresis and a "sweet spot" detector, which performs an accurate auto-slicing of the input.
This means that the circuit automatically chooses the most useful part of the input signal, thus being able to clean-up and reconstruct a ringing and noisy input clock into a perfect output clock used to internally synchronize the unit.

The unit is built with sync safety in mind. It always uses the internal low jitter oscillator as the master clock for all internal digital signals, and it always uses the master clock oscillator to generate the output clock.
When it locks to an incoming clock, it continues to use the internal oscillator as the master clock, but it gently adjust the oscillator frequency to match the frequency of the incoming signal, creating a phase lock between clock input and clock output.
When the incoming clock signal is lost or discontinued, the master clock oscillator gently glides back to its internal reference, and continues to generate sync signals without disruption. Thus, there are never gabs or disturbances of any kind when the clock sync is lost and re-established.

The unit works as an excellent clock reference generator for a studio setup. In AES mode, the output clock frequency is manually controlled at the front panel with the arrow buttons. In USB mode, the output clock frequency is controlled by the Work Station or the DAW.
The output clock accuracy is typically +/-2 ppm, and the pulse shape of the long-haul output buffer is close to theoretically perfect, with impressive jitter specifications, steep rise time and no ringing, even with long cables.

Power Supply

The power supply is pure analog, with an oversized toroidal transformer and low noise linear regulators.
This configuration creates an electromagnetic quiet environment, free from the usual radiation pollution from a switch-mode power supply.
A switch mode power supply generates strong repetitive electromagnetic pulses that travel through air and sharp current pulses that travel through the ground system.
When this pollution hit the analog circuitry, it disrupt the smooth operation of the circuitry by pressing the amplifiers into slew-rate-mode momentarily

But what is slew-rate-mode?

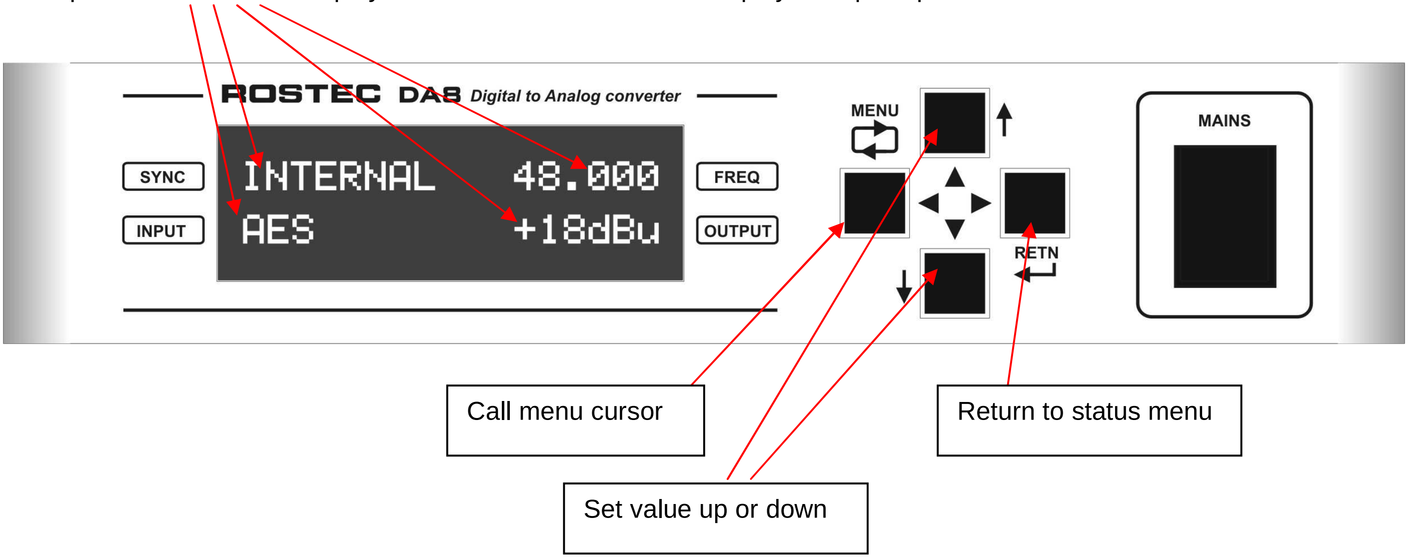
It is when an amplifier is presented with a signal, that moves faster that the amplifiers maximum speed capability. It then tries to “slew” as fast as it can, to cope with the signal. When the amplifier is in this mode, it cannot process any further information, it is in fact blocked from reproducing incoming audio signals.
This happens in short durations, when the pulses from a switch mode power supply hit the circuit.
The result is that 60.000 a second the analog circuit looses, in small intervals, the ability to reproduce audio, and this is in fact the main reason why audio products with switch mode power supplies sound harsh, flat and lifeless, with a degraded ability to process details and depth in the audio.

DA8 maintains a clean electromagnetic environment in the box, and the reward is a natural, pleasant, open and detailed sound.

Front panel quick guide.

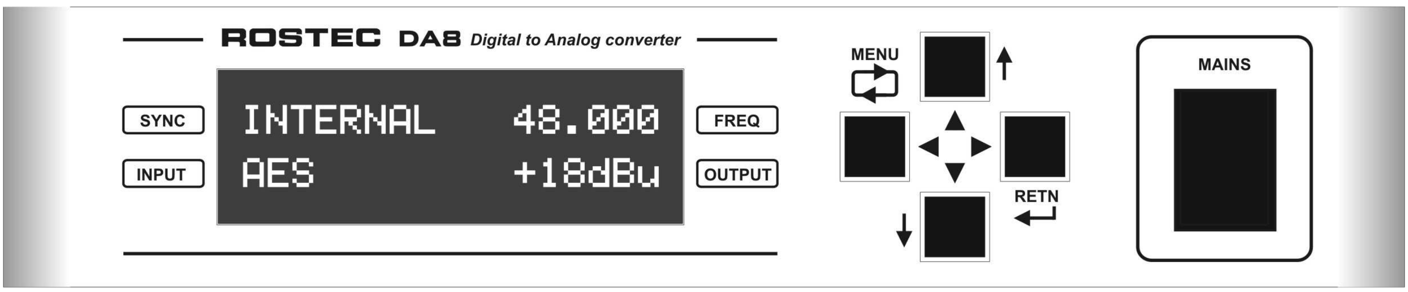
Status menu (default start-up):

The operational **status** is displayed at the four corners of the display. The principle is **“What You See Is What You Get”**



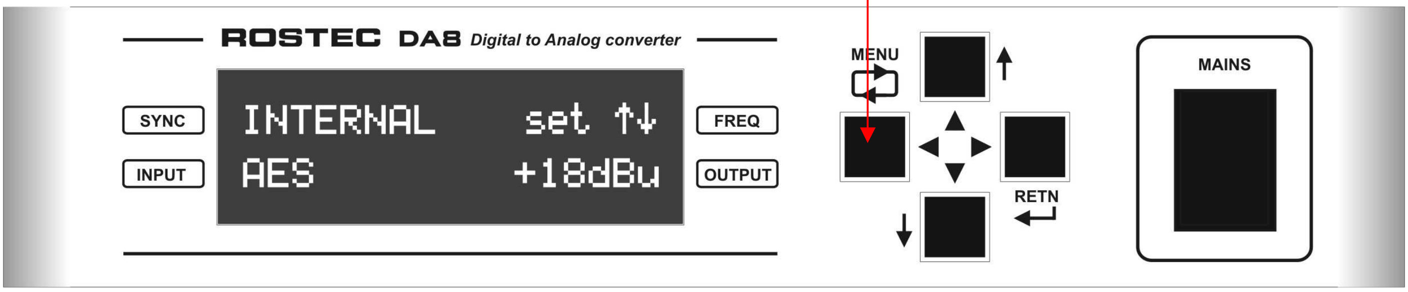
How to change parameters (example):

Step 1: For example, you start with this status menu:



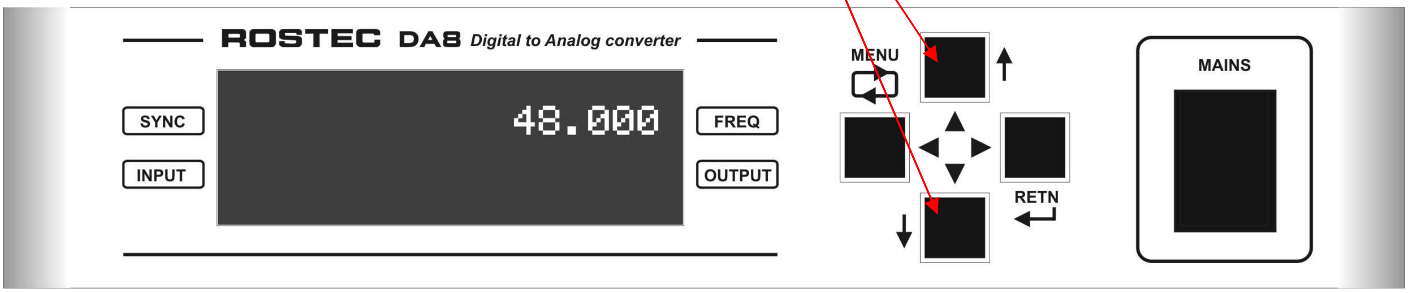
Step 2: Press the MENU button

Observe that the setting cursor appears. The arrows indicate that you now can change the value <u>at the cursor position</u> with the arrow buttons. Press MENU again to change the position of the cursor. If you don't wish to change anything, just press RETN to return to the status menu.



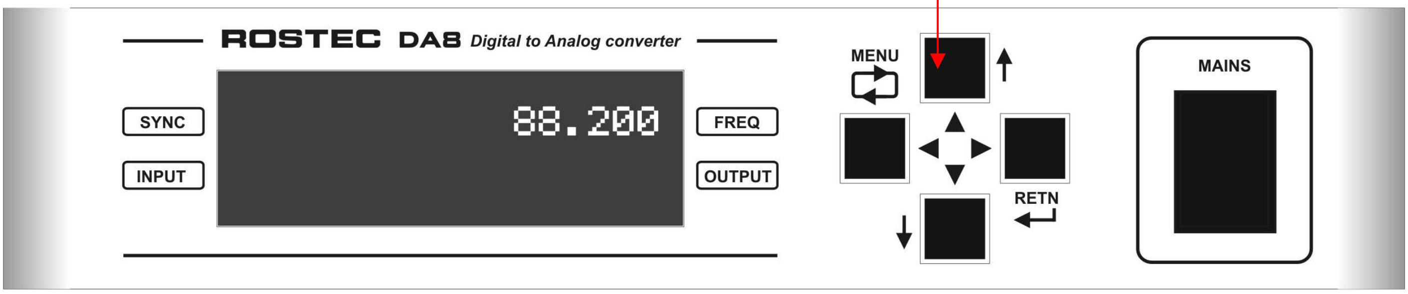
Step 3: Press one of the ARROW buttons. It doesn't matter which one.

Observe that the current parameter <u>at the cursor position</u> is shown. Everything else is blanked out.



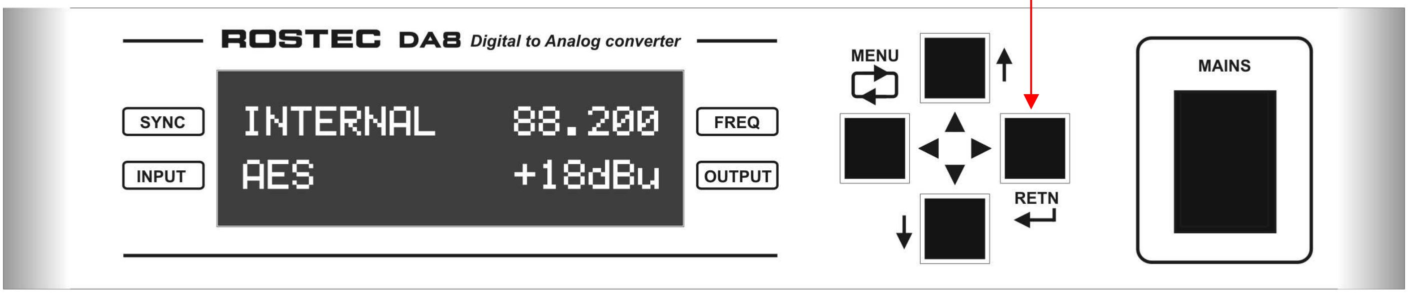
Step 4: Press the UP ARROW button

Observe that the parameter changes from 48.000 to 88.200



Step 5: Press the RETN button

All done. Remember that pressing RETN will always bring you back to the status menu and consolidate and save the setting.



SYNC:
INTERNAL: The unit runs on its internal high grade clock oscillator.
EXT ----: Synchronization of the internal clock oscillator is enabled. The internal clock continues to run, and when a valid clock is detected at the clock input (BNC connector, back panel), the ---- symbol changes to LOCK.

FREQ:
When input **OFF** or input **AES** are selected, the sampling frequency is controlled manually by the arrow buttons.
When **USB** is selected, the sampling frequency is controlled by the connected Workstation (DAW)
The available frequencies are: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz.
*OBS: When input **USB** is selected, but **no USB connection is made,** the FREQ will show NONE. The unit will be paused, and no clock output will be available.*

OUTPUT:
OFF: The analog outputs are disabled. Digital signals are not affected.
+18 dBu: EBU standard output level. +18 dBu analog output equals Digital Full Scale (0 dBFS)
+10 dBu: This output level is optimized for use with vintage equipment, especially vintage multi channel consoles.

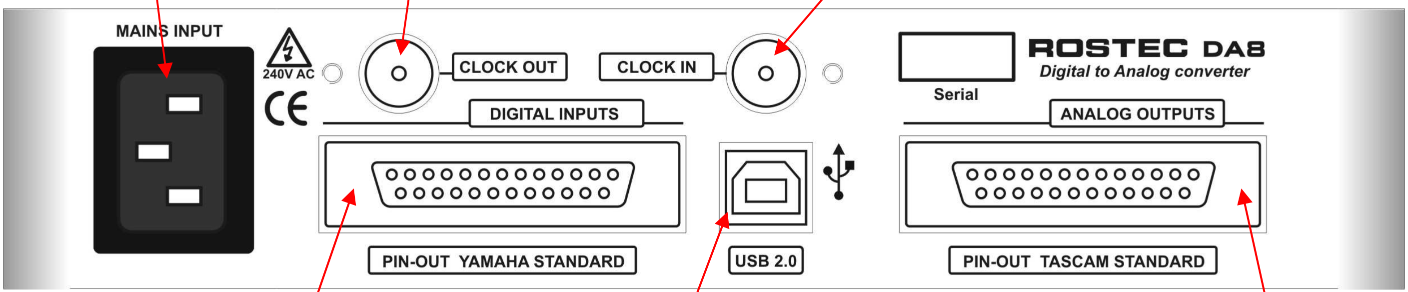
INPUT:
OFF: Digital inputs are disabled. Clock output is still running at the selected sampling frequency.
AES: The AES signals on the D-SUB 25 pole Digital Input Connector are active. The USB signal is disabled.
USB: The USB signal on the type B USB connector is active. The AES signals are disabled..

Back panel quick guide.

Mains input
190 – 250 VAC
IEC Connector

Word Clock Output, Zout = 75 ohm.
5.0 Vpp no load, 2.5 Vpp into 75 ohm.
BNC Connector.

Word Clock Input, Zin = 1.5 kohm.
Min 0.2 Vpp, Max 10 Vpp.
BNC Connector.



Digital inputs, 4 x AES3 (8 channels).
Zin 110 ohm, transformer balanced.
Max in 7.0 Vpp Min in 0.2Vpp (SPDIF)
D-SUB 25 pole, Female.

USB type B connector.
Class 2.0 compliant.

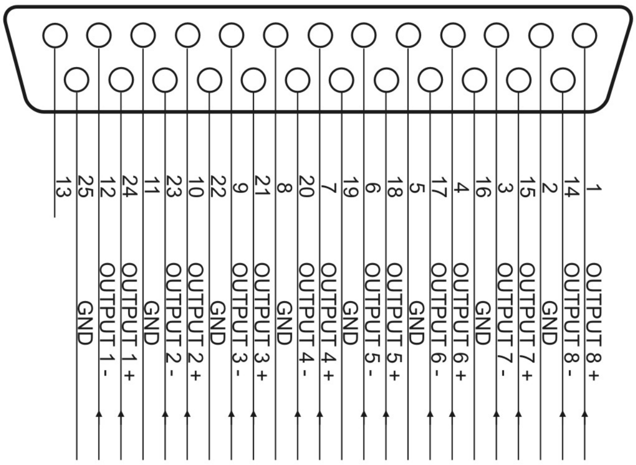
Analog outputs, 8 channels.
Electronically balanced, Zout 75 ohm.
Max output +28 dBu.
D-SUB 25 pole, Female.

Digital Inputs and outputs are ESD protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5. Analog outputs are ESD protected to 2000 V, human body model.

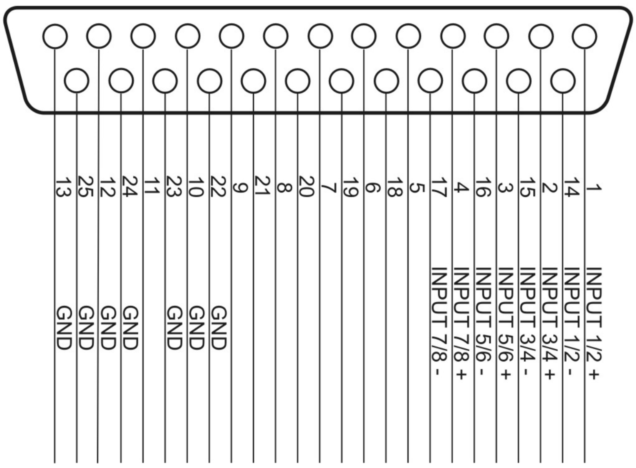
DA8 D/A Converter pin-out

25 pole D-SUB connectors seen from outside

Analog outputs, TASCAM standard



Digtal inputs, YAMAHA standard



Mechanical and electrical specifications

Digital to Analog interface:

8 balanced analog line outputs at the back panel, D-SUB 25 female, TASCAM STANDARD.
4 transformer balanced AES3/SPDIF inputs (8 channels) at the back panel, D-SUB 25, YAMAHA STANDARD.
1 USB 2.0 Digital interface, TYPE B connector.
The USB connection is compatible with Mac OS X 10.6.8 or later. Windows 10 is supported.
DA Resolution: 24 bits.
Sampling frequencies: 44.1k, 48k, 88.2k, 96k, 176.4k, 192k.
AES latency: down to 0.5 mSec, depending on sampling frequency
USB latency: down to 1.5 mSec. This is highly dependent on the host computer speed and sampling frequency!
Interchannel Gain Mismatch: 0.1 dB.
Linearity inside pass band (pass band ripple): 0.05 dB.
Sample Rate Conversion of AES/SPDIF: Distortion/residual artifacts -140 dB. Ratio.16:1 to 1:16, smooth glide and auto detect.
Frequency response: 2nd order analog anti-aliasing filters at 1 Hz - 60 kHz -3 dB. Brick-wall digital filters at 0.5 x Fs
THD+N: 0.00039 % (-108 dB) at 1 kHz.
THD: 0.00015 % at 1 kHz (-116 dB). See detailed FFT distortion analysis).
Crosstalk between adjacent channels: -105 dB, 20 Hz - 20 kHz.
Noise: 118 dB RMS, A weighted, 10 Hz - 22 kHz measurement bandwidth.
Analog output level: +18 dBu +/- 0.2 dB (EBU standard), switchable to +10 dBu +/- 0.2 dB (for vintage mixing consoles, equipment etc).
Analog output buffers internal headroom: + 28 dBu.
Hum residue from analog power supply, 1st, 2nd, 3rd and 4th harmonic: less than -140 dB referenced to 0dBFS.

Clock system:

Word clock input:: BNC, unbalanced, unterminated (1.5 kohm), min. 0.2 Vpp, max. 10 Vpp.
Word clock output: BNC, unbalanced, impedance 75 ohm, 5 Vpp no load, 2.5 Vpp into 75 ohm.
Internal AES GRADE 2 crystal oscillator.
Temperature stability: +/-2 ppm from 0 degC to +70 degC.
Ageing: 2 ppm per year.
Word clock frequencies: 44.1k, 48k, 88.2k, 98k, 176.4k, 192k.
Internal crystal oscillator jitter: 2 ps rms.
Word clock output jitter: 50 ps rms.
AES/USB internal data jitter: 18 ps rms.
Clock input capture range: +/- 80 ppm.
Lock time approx: 0.2 - 0.4 sec.

Power supply:

Linear analog, passively cooled, high power, very low noise.
Thermally and overload protected.
Mains voltage: 180 - 264 VAC 50-60 Hz (115 VAC version available by request).
Power consumption: 15 Watts.

General:

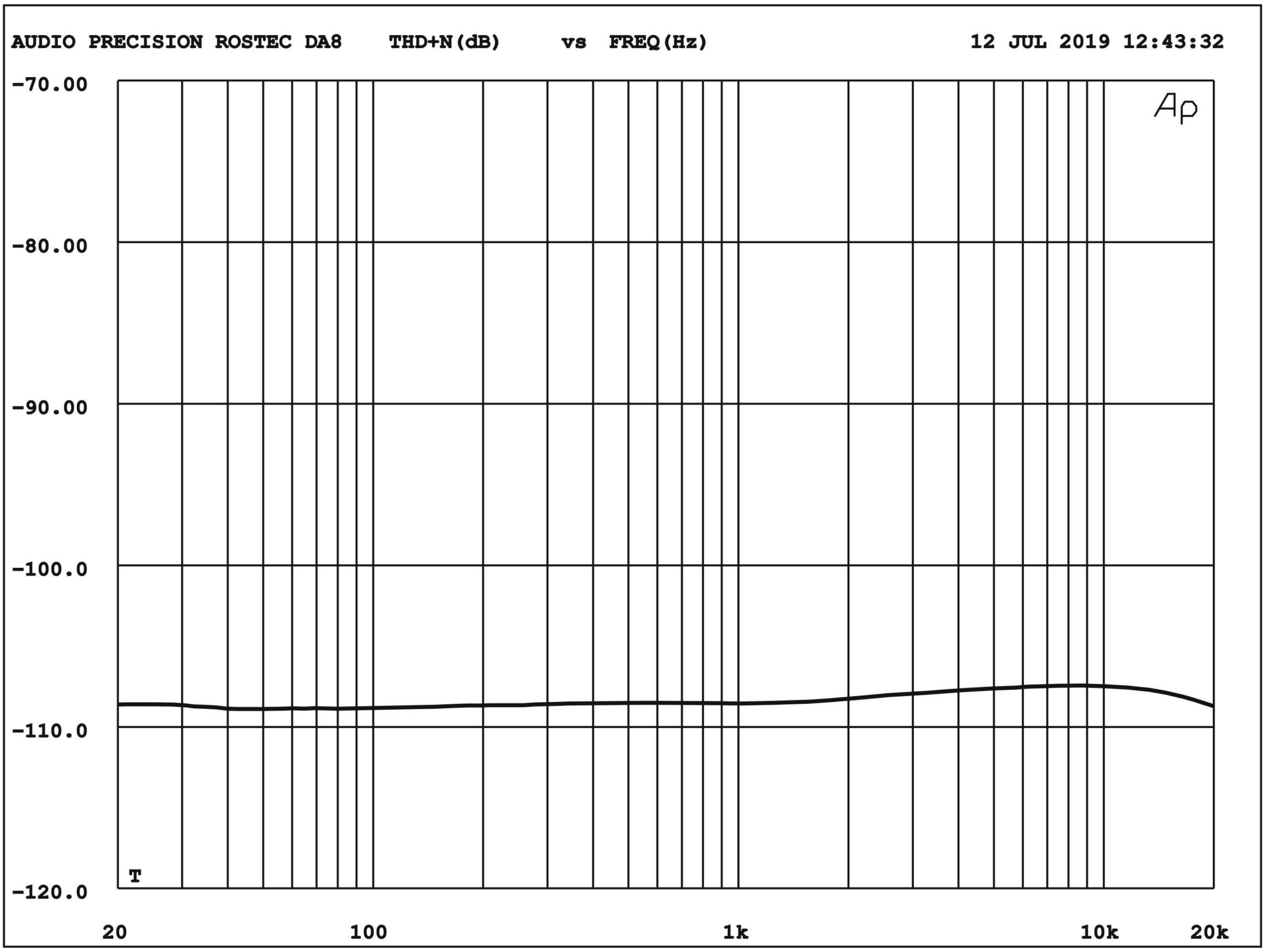
Sturdy steel casing, magnetically and electrically screened. Size 210 mm x 210 mm x 42 mm.
Analog and digital circuits are magnetically and electrically isolated by internal metal screening and separate ground.
All digital inputs and outputs are ESD protected to 23 kV 15 A surge (IEC61000-4-2 and IEC61000-4-5).
All analog outputs are ESD protected to 2000V, human body model.

Technical section

Distortion characteristics

There has always been focus on distortion, when it comes to judging audio equipment. It can be argued, that any distortion product (2nd, 3rd harmonic etc) below -60 dB is inaudible, simply because the distortion tones are a consequence of the incoming tone, and thus are always masked by it. In some sense this is true, but distortion is more than just about sound, it is also an indicator that shows whether there is something wrong with the circuit or not.

Classical distortion analysis.

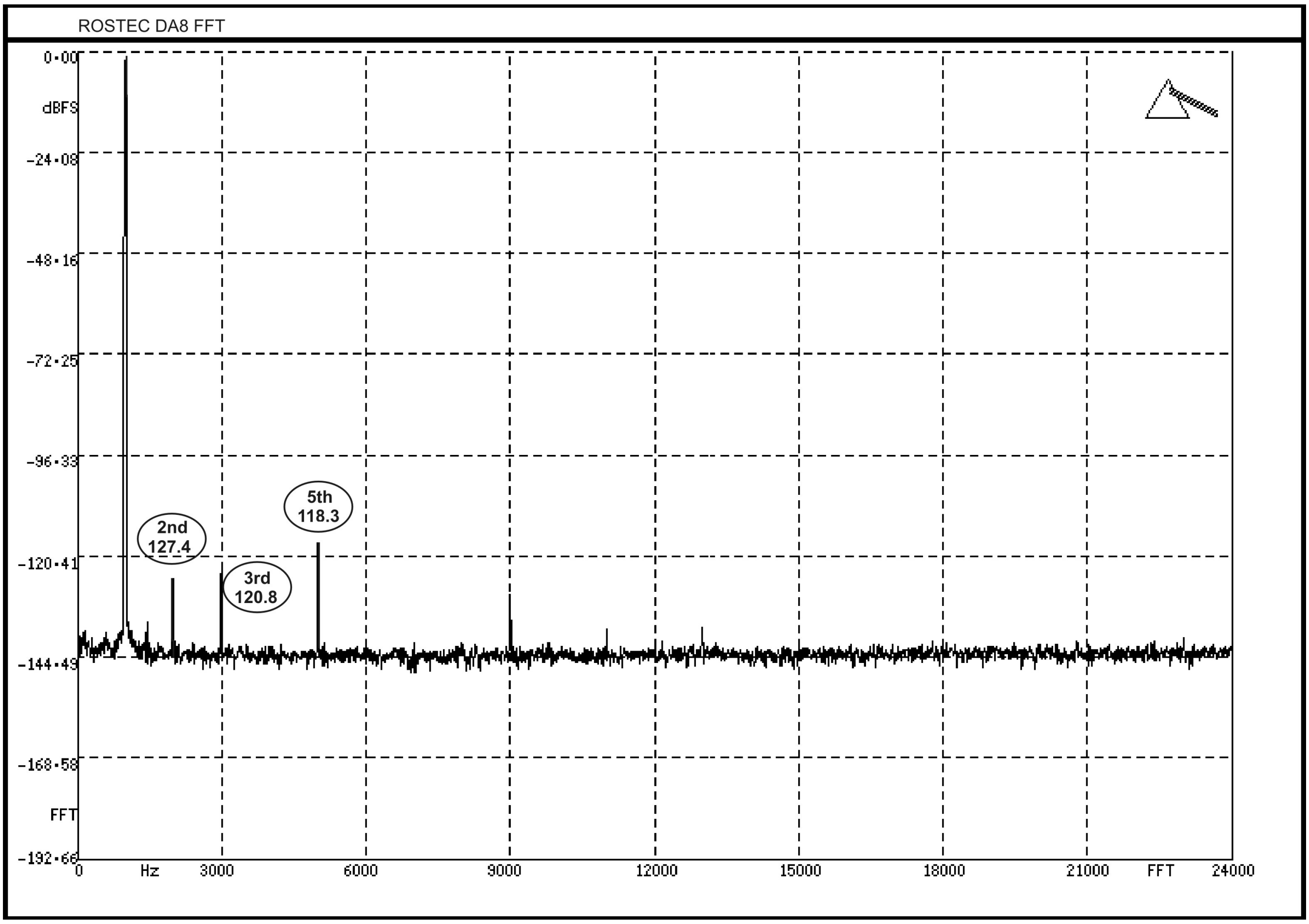


THD+Noise versus frequency. 20 Hz - 20 kHz @ -0.2 dBFS. Only the measurement @ 48 kHz sampling frequency is shown. The Characteristics are the same with very little variations at all samplings frequencies.

Distortion in depth analysis. FFT (Fast Fourier Transform)

A more detailed picture of the distortion characteristics can be seen using FFT analysis, which allows the magnitude of the individual distortion components to be quantified. In this FFT analysis only the distortion analysis at 48 kHz sample rate is included. Distortion characteristics at the other sample rates are identical within 0.5 dB.

Harmonic distortion measurement, 16 passes of FFT superimposed to suppress noise components.



Input is 1 kHz @ -0.2 dBFS. Sampling rate 48 kHz. The resulting distortion sums up to -116.4 dB RMS. Only the 2nd, 3rd and 5th harmonic contribute to distortion result. The harmonics above the 5th are considered negligible as they are buried in the noise.

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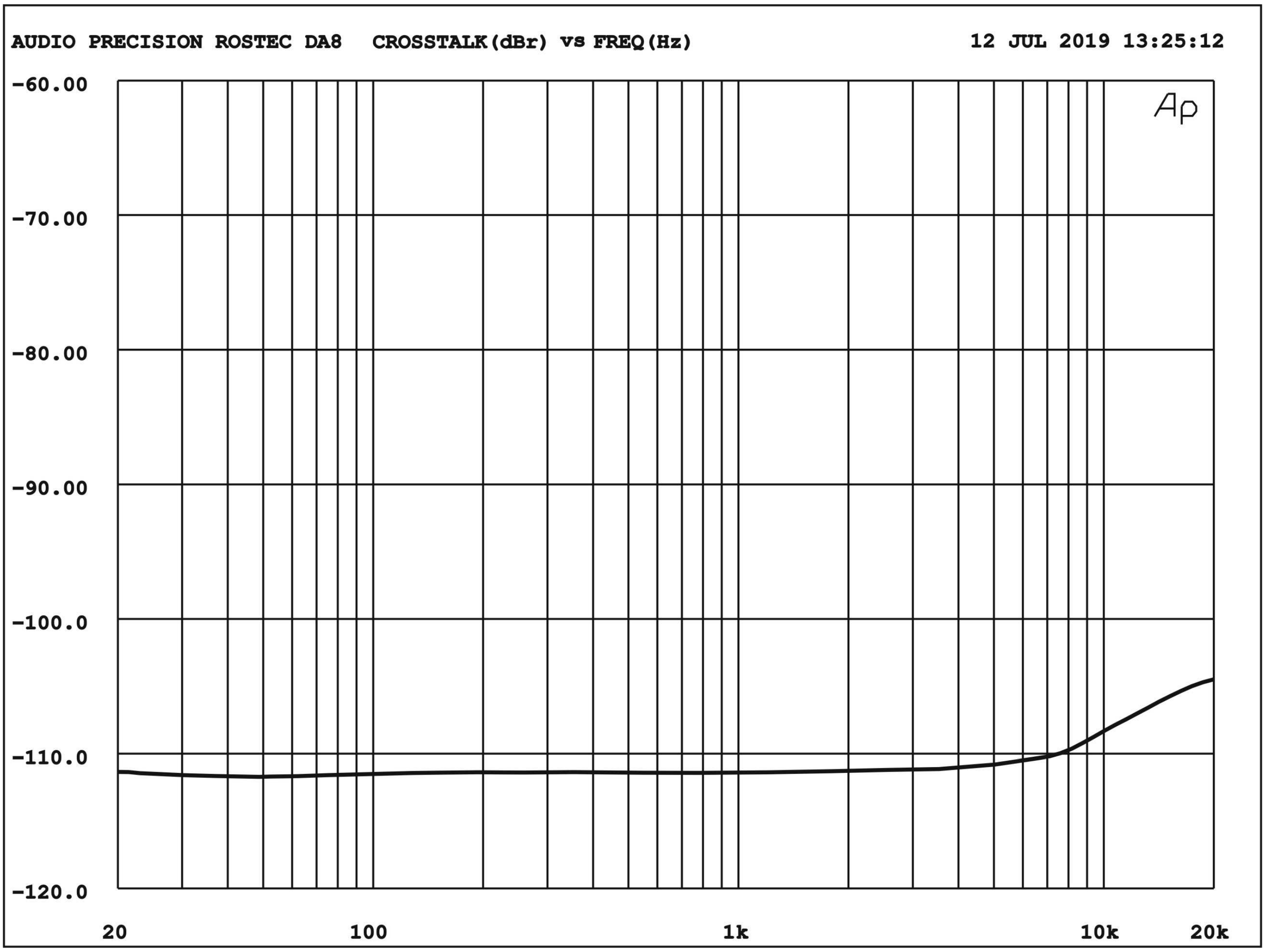
Crosstalk

Below is the crosstalk measured between adjacent channels (worst case). A sine sweep from 20 - 20 kHz at full amplitude was sent to one channel, and the output level was measured at the adjacent channel. The channel was terminated by 75 ohm at the input in order to eliminate cable transferred crosstalk.

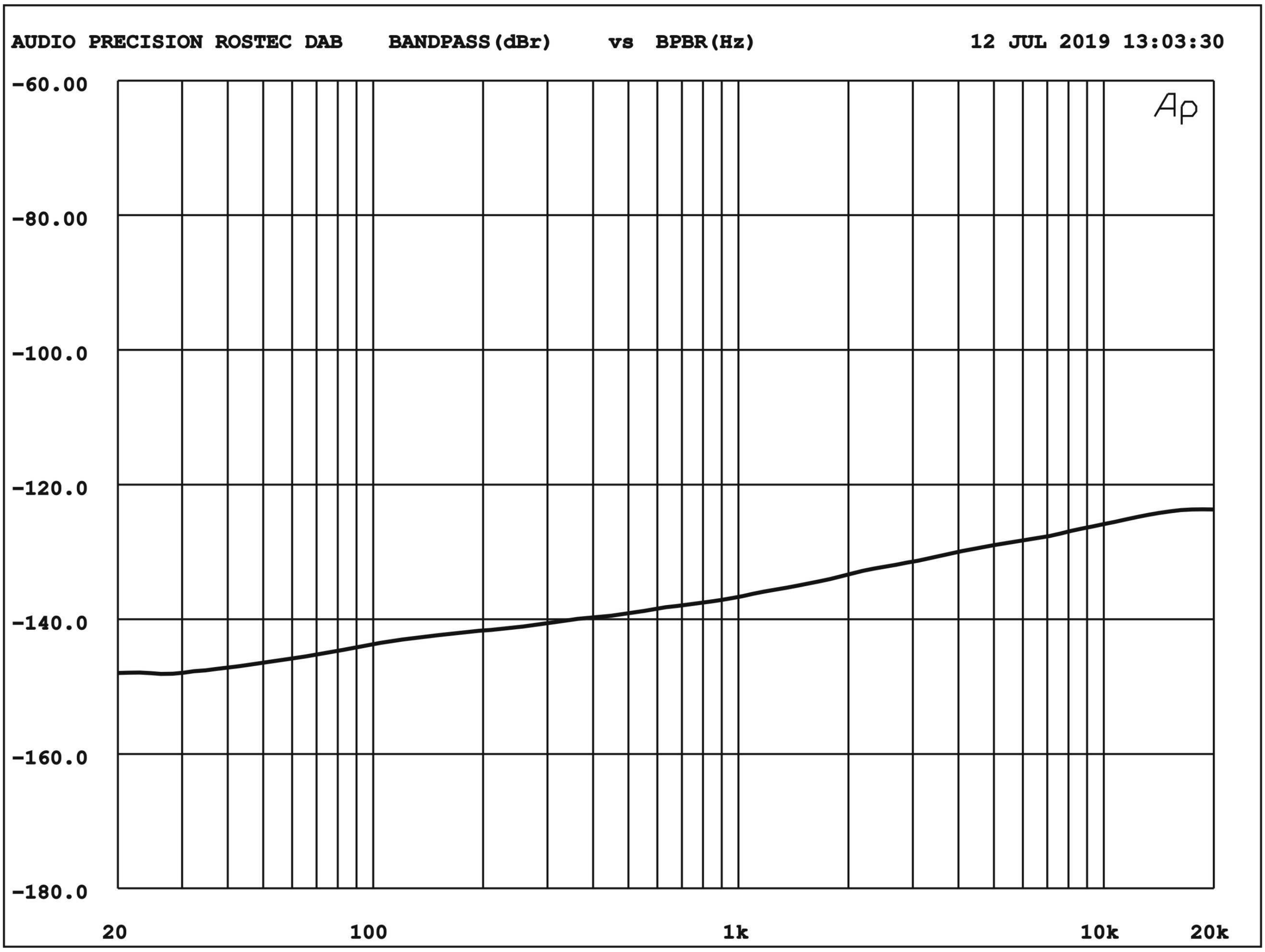


The true balanced nature of the DA8 really pays off. The crosstalk is below -110 dB in the main part of the spectrum. Only above 10 kHz is a slight increase seen, reaching approx. -105 dB at 20 kHz

Note: This is a linear and unweighted measurement. Using A-weighting would give a much better picture, but that would be cheating!

Hum and noise

Using an analog power supply with an oversized toroidal transformer and linear regulators could be cause for concern. However, not to worry. The carefully designed power circuits creates an electromagnetic quiet environment, free from the usual radiation pollution from a switch-mode power supply, and the true balance nature of the analog circuits really pays off. The measurement speaks for itself. There is no measurable power supply hum transferred to the analog outputs.

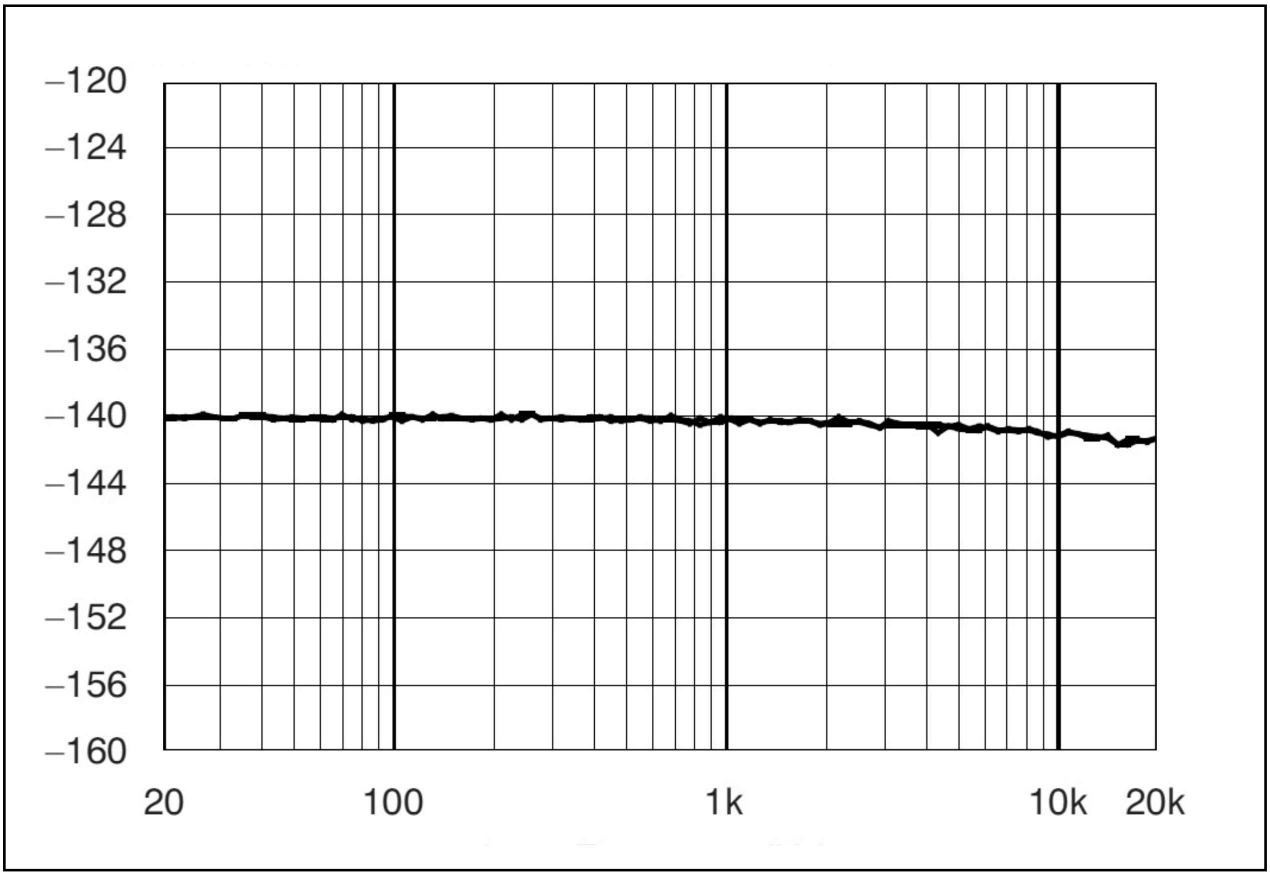


The graph shows the spectral density of the output noise by measuring with a sweeping 1/3 octave band pass filter across the audio range.
The output was measured directly at the output buffers. Digital input was off, and all sampling frequencies were tested.
The noise spectrum is linear and smooth from 1 kHz and up and exhibits the typical characteristics of white noise.
Below 1 kHz, the 1/f corner noise is dominant. No noise component from the power supply can be seen in the critical range from 50 Hz to 400 Hz. The measured result in this range is comfortably below -140 dB, referenced to 0 dBFS.
The total RMS sum of the noise from 20 Hz - 20 kHz is -118 dB A weighted.

Sample Rate Converters

The DA8 has four hardware Sample Rate Converters, one on each AES input. Their quality is truly exceptional and the sonic performance is outstanding, Absolutely the best that today's industry can offer.

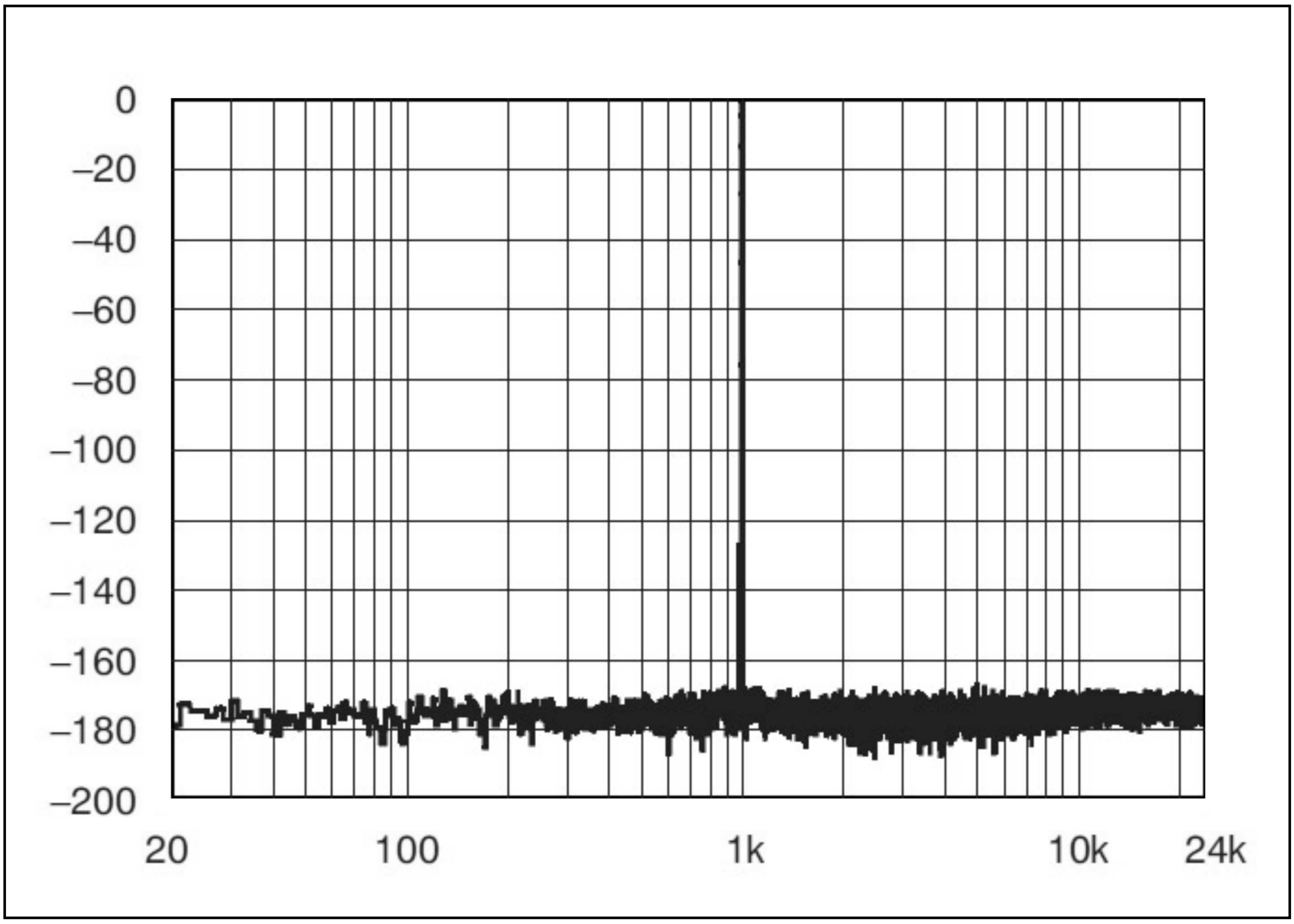
Classical distortion analysis of the Sample Rate Converters



Input is 1 kHz @ -0.2 dBFS. Sampling rate conversion is from 44.1 kHz to 48 kHz. The distortion is measured in the audio range from 20 Hz – 20 kHz, and the result is -140 dB.
All various combinations of sample rate conversions, up and down, show the same excellent result.

Distortion in depth analysis of Sample Rate Converters. FFT (Fast Fourier Transform)

A more detailed picture of the distortion characteristics can be seen using FFT analysis, which allows the magnitude of the individual distortion components to be quantified. In this FFT analysis only the result of the **s**ampling rate conversion from 44.1 kHz to 48 kHz is included. Distortion characteristics at the other sample rates are identical within the measurement uncertainty.



Input is 1 kHz @ 0 dBFS. Sampling rate conversion is from 44.1 kHz to 48 kHz. The distortion is measured in the audio range from 20 Hz – 24 kHz (half the output sampling frequency).
The surprising result is, that there are no artifacts or distortion components in the measurement. This indicates, that the -140 dB THD+N result of the classical measurement is purely noise, and with no distortion products.