...the measure of excellence!™

Benchmark *DAC2 Series*
Instruction Manual

Reference Stereo Preamplifier
PCM and DSD D/A Converter
Asynchronous USB

Safety Information

Fuses

CAUTION: FOR CONTINUED FIRE HAZARD PROTECTION ALWAYS REPLACE THE FUSES WITH THE CORRECT SIZE AND TYPE (0.5A 250 V SLO-BLO® 5 X 20 MM – LITTELFUSE® HXP218.500 OR EQUIVALENT). THE FUSE DRAWER INCLUDES TWO FUSES. ALWAYS REPLACE BOTH FUSES AT THE SAME TIME.

Voltage Selection

THE DAC2 IS EQUIPPED WITH A UNIVERSAL POWER SUPPLY. THERE IS NO VOLTAGE SELECTION SWITCH. AC VOLTAGE RANGE IS 88-264 VAC, 50-60 HZ. THE PRODUCT MAY ALSO BE OPERATED FROM DC POWER OVER A VOLTAGE RANGE OF 125-373 VDC.

Power Cord

CAUTION: ALWAYS USE A GROUNDED POWER CORD. THE PRODUCT IS EQUIPPED WITH A STANDARD IEC POWER ENTRY MODULE. USE AN IEC POWER CORD THAT IS EQUIPPED WITH THE APPROPRIATE CONNECTOR FOR YOUR LOCATION. CORDS ARE AVAILABLE FROM YOUR DEALER.

Modifications

CAUTION: DO NOT SUBSTITUTE PARTS OR MAKE ANY MODIFICATIONS WITHOUT THE WRITTEN APPROVAL OF BENCHMARK MEDIA SYSTEMS, INC. MODIFICATION MAY CREATE SAFETY HAZARDS AND VOID THE WARRANTY.

NOTICE: CHANGES OR MODIFICATIONS NOT EXPRESSLY APPROVED BY BENCHMARK MEDIA SYSTEMS COULD VOID THE USER'S AUTHORITY TO OPERATE THE EQUIPMENT UNDER FCC REGULATIONS.

Repairs

CAUTION: DO NOT SERVICE OR REPAIR THIS PRODUCT UNLESS PROPERLY QUALIFIED. ONLY A QUALIFIED TECHNICIAN SHOULD PERFORM REPAIRS.

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Features

The ***DAC2*** *series* is a reference-grade digital to analog converter, stereo system pre-amplifier, and headphone amplifier with infrared remote control. It supports D/A conversion of PCM sample rates up to 192 kHz, as well as direct DSD conversion.

- ***HGC™*** (Hybrid Gain Control) – combines motor-driven active analog potentiometer, 32-bit digital attenuation, and passive analog attenuators, to achieve state-of-the-art performance
- ***SABRE*** - 32-bit PCM D/A conversion system, four 32-bit D/A converters per channel
- ***SABRE*** – Native DSD D/A conversion system, four 1-bit DSD D/A converters per channel
- ***HPA2™*** reference-grade ***headphone amplifier*** with dual outputs - “0-Ohm”, high-current (*DAC2 HGC and DAC2 D model only*)
- ***Multi-Mode Asynchronous USB Audio 2.0*** – 24 bit/192 kHz, DSD (DoP 1.1)
- Driverless ***Asynchronous USB Audio 1.1*** – 24-bit/96 kHz
- ***Benchmark UltraLock2™ Jitter Attenuation System*** – eliminates jitter-induced distortion
- ***Sample Rate Display*** – 44.1, 48, 88.2, 96, 176.4, 192 kHz and DSD
- ***Word Length Display*** – 16-bit, 24-bit
- ***2 Coaxial Digital Inputs*** – 24-bit/192 kHz PCM, DSD (DoP 1.1)
- ***2 Optical Digital Inputs*** – 24-bit/96 kHz PCM
- ***1 Coaxial Digital Output*** – digital pass through from USB, Coax, and optical inputs when jumpered
- Aluminum ***IR Remote*** provides control of all functions
- ***2 Stereo Analog Inputs*** – 2 pairs, unbalanced (RCA) *(DAC2 HGC and DAC2 L model only)*
- ***3 Stereo Analog Outputs*** – 1 pair, balanced (XLR), plus 2 pairs unbalanced (RCA)
- ***Low-Impedance Passive Output Pads*** – 0, 10, and 20 dB – optimize output level to power amplifiers to maximize SNR
- ***2 HPA2™ Headphone Outputs*** – one output has option to automatically mute main outputs (*DAC2 HGC and DAC2 D model only*)
- ***HPA2™*** gain jumpers for customizing headphone output gain for headphone sensitivities (*DAC2 HGC and DAC2 D model only*)
- ***12V Trigger I/O*** – bi-directional 12V trigger can act as input, output, or both *(DAC2 HGC and DAC2 L model only)*
- ***Home Theatre Bypass*** – places analog input(s) in a fixed-gain pass-through mode (*DAC2 HGC and DAC2 L model only*)
- ***High Throughput Mode*** - places the digital input(s) in a fixed-gain
- ***Polarity Switch*** – inverts the polarity of selected digital inputs
- ***Mute and Dim Functions*** – accessible from remote or front panel
- **Automatic De-Emphasis** - in response to consumer pre-emphasis bit (44.1, 48 kHz)
- ***Power Switch*** - very low standby power - <0.5 W at 120 VAC
- ***High-Efficiency Low-Noise Power Supplies*** – only 12-15 W, 88-264 VAC, 50-60 Hz
- Meets FCC Class B and CE emissions requirements
- Tested for immunity to radiated and conducted RF interference

Overview

The ***DAC2*** builds upon Benchmark's highly successful ***DAC1*** product family. Every ***DAC1*** subsystem has been redesigned and upgraded to achieve higher performance. The ***DAC2*** includes an updated version of Benchmark's highly-effective UltraLock™ jitter-attenuation system.

- New features have been added to extend the versatility of the product, and improve the listening experience. These features include: native DSD conversion, asynchronous USB 2.0, asynchronous USB 1.1, home theater bypass, digital pass-through, polarity control, word-length display, sample-rate display, a bi-directional 12V trigger *(DAC2 HGC and DAC2 L model only)*, and additional I/O.

The ***DAC2*** includes Benchmark's high-performance ***HPA2™***headphone amplifier (*DAC2 HGC and DAC2 D model only*), and Benchmark's aluminum remote control.

Performance Improvements

Lower Noise than the DAC1

Four balanced 32-bit digital-to-analog audio converters are summed together to form each balanced output channel. The 4:1 summing reduces noise by about 6 dB. Overall, the ***DAC2*** is about 10 dB quieter than the ***DAC1***. Low-level musical details are faithfully reproduced over a breathtakingly quiet noise floor.

Lower Distortion than the DAC1

Benchmark's ***DAC1*** converters are known for their very low distortion (THD and IMD). The ***DAC2*** sets new benchmarks for clean and transparent musical reproduction.

Low Power Consumption

The ***DAC2*** uses high-efficiency low-noise power supplies. Each critical subsystem has at least one dedicated low-noise regulator. The unit runs cool while providing substantial power to the headphone and output drivers. A power switch is included.

UltraLock2™ Clock System

UltraLock2™ provides the outstanding jitter attenuation of the older ***UltraLock™*** system while providing a higher SNR.

High-Headroom Digital Processing

All digital processing includes at least 3.5 dB headroom above an input level of 0 dBFS. This prevents all clipping in the digital processing, and provides clean and transparent audio reproduction.

New Features

Native DSD Conversion

The ***DAC2*** supports native DSD conversion. DSD signals can be delivered to the USB or Coaxial inputs in DoP 1.1 format. The DSD signal is then routed directly to a bank of 1-bit DSD D/A converters. Four balanced 1-bit converters are summed together for each balanced output.

Asynchronous USB 2.0

The USB interface supports DSD and 192 kHz, 24-bit PCM. The ***DAC2*** generates the conversion clocks and totally eliminates the USB interface as a source of jitter. No drivers are required for Apple operating systems. Drivers are provided for Windows operating systems at http://www.benchmarkmedia.com/dac/dac2-drivers.

Native Asynchronous USB 1.1

The ***DAC2*** has a driverless USB 1.1 mode that supports 96 kHz, 24-bit PCM with all operating systems. This mode provides quick and easy connection to a wide variety of computers and tablets.

New Hybrid Gain Control

HGC™ is Benchmark's unique Hybrid Gain Control that combines active analog, 32-bit digital, and passive analog attenuation systems. ***HGC™*** puts an end to the debate about analog versus digital gain controls, and passive versus active analog attenuation. The dual-domain ***HGC™*** system combines the high dynamic range of Benchmark's **HDR**™ analog control with the low distortion, and accuracy of digital control. ***HGC™*** outperforms traditional analog or digital volume controls, including the two-stage ***DAC1 HDR™*** system. Musical details are preserved over a very wide range of output levels. Analog inputs are controlled in the analog domain. Digital inputs are controlled in both domains.

The volume control is a servo-driven analog potentiometer. This control rotates in response to commands from the remote control while providing the convenience of manual adjustments.

Low-impedance passive output attenuators optimize the gain range of the active analog and digital gain sections.

HT - Home Theater Bypass (DAC2 HGC and DAC2 L model only)

The home theater bypass is useful when you have a home theater system and want to bypass the signal through the **DAC2**. Set the **DAC2** between your AVR and your power amp through the left and right channels of your analog inputs. The ***DAC2*** can drive the left and right power amplifiers, while the home theater system drives all other power amplifiers. When the home theater mode is in use, left and right audio passes through the ***DAC2*** at fixed-gain, and the home theatre system controls the audio level.

HT – High Throughput Mode

Any digital input can be placed at fixed-gain in **HT** mode. In HT mode, the digital audio output is set to a fixed audio level. This mode is useful if the output is going to a stereo preamp. (Note: DAC1 users will find this function to be similar to the DAC1's calibrated mode.)

Digital Pass-Through

The second coaxial input (**D4**) can be reconfigured as a digital output. When operating as an output, any selected digital input is passed through to (**D4**) without any processing. Optical, coaxial, and USB inputs can be passed through to the (**D4**) connector. This even includes special signals such as DoP, DTS, Dolby Digital, etc.

Polarity Control

Each digital input can be inverted to correct polarity problems. Some listeners report that polarity is incorrect on some recordings, and that they enjoy an improved listening experience when this is corrected.

Bi-directional 12V Trigger

(***DAC2 HGC*** and ***DAC2 L*** model only)

The 12V trigger can be connected to other audio components so that an entire audio system can turn on and off in a sequenced fashion. The ***DAC2***trigger I/O could be connected to a preamplifier, power amplifier, or both. The ***DAC2*** will pull the trigger I/O to 12 volts DC when the ***DAC2*** turns on. If an external device pulls the trigger I/O to 12 volts, the ***DAC2*** will turn on.

Expanded I/O

The ***DAC2 Series*** has:

- 2 stereo unbalanced analog inputs *(**DAC2 HGC** and **DAC2 L** model only)*
- 2 optical inputs
- 2 coaxial inputs
- 1 USB input
- 2 stereo unbalanced outputs
- 1 balanced stereo output
- 1 bi-directional 12 volt trigger *(**DAC2 HGC** and **DAC2 L** model only)*

Applications

The ***DAC2*** is designed for maximum transparency and purity. The sonic integrity of the ***DAC2*** makes it well suited for critical playback in recording studio control rooms and mastering rooms. The versatility of the ***DAC2*** makes it an asset to any high-end audiophile application, including: HDTV, DVD, digital cable, music server, digital radio, analog radio, phono playback, portable music player, etc.

Benchmark's **Multi-Mode Asynchronous *USB***™ interface makes the ***DAC2*** an ideal output device for computer-based media playback, including: home media servers, digital audio workstations, desktop audio editing application, and computer-based radio broadcast systems.

DAC1 Heritage

The pristine audio performance of the award-winning ***DAC1*** made it the 'Benchmark' of stand-alone D/A converters. The ***DAC1 USB*** and ***DAC1 PRE*** , and ***DAC1 HDR*** added features and minor performance improvements. The ***DAC2*** series continues the tradition of perfectionism by adding many new technologies, features, and major performance improvements.

With the introduction of the ***DAC1 USB*** we added an advanced USB input with native 96-kHz / 24-kHz capability, an auto-mute function for headphone use, customizable headphone gain range, an auto-standby feature, and a high-current LM4562/LME49860 output stage designed to drive difficult loads.

The ***DAC1 PRE*** added the versatility of a stereo analog input and three S/PDIF digital inputs. The LM4562/LME49860 opamps were used throughout the analog section, and all RCA connectors were upgraded to premium gold plated, Teflon insulated bulkhead mounted RCA connectors for maximum durability and superior grounding.

The ***DAC1 HDR*** added a remote-control and the ***HDR-VC***™ volume control. The ***HDR-VC***™ volume control is built with a custom-made, motor-driven Alps potentiometer. The intelligent volume control allows the user to easily control the 'Normal' and 'Dim' / 'Soft-Mute' settings independently for ultimate flexibility.

The ***DAC2*** is a complete redesign with 32-bit D/A conversion, native 24-bit/192kHz PCM, native DSD conversion, multi-mode Asynchronous USB, sample rate and word length displays, ***UltraLock2***™ jitter attenuation, polarity switch, home theater bypass *(DAC2 HGC and DAC2 L model only)*, bi-directional 12V trigger *(DAC2 HGC and DAC2 L model only)*. It also adds 1 additional stereo analog input *(DAC2 HGC and DAC2 L model only)*, an additional analog output, 1 additional optical input, a digital pass-through, and high-efficiency low-noise power supplies.

HPA2™ Headphone Amplifier

(***DAC2 HGC*** and ***DAC2 D*** model only)

The ***HPA2™*** is one of the most transparent headphone amplifiers available. It also is able to deliver high current and/or high signal levels, and is well suited for a wide variety of headphones. The near 0-Ohm output impedance provides outstanding damping of headphone drivers. This damping reduces distortion, and maintains control of the frequency response.

Quick Start Guide

Audio Inputs

The ***DAC2*** features two stereo RCA analog inputs (***DAC2 HGC*** *model and* ***DAC2 L*** *only*) and five stereo digital inputs (2 coaxial, 2 optical, and 1 USB). The coaxial and optical inputs accept professional (AES) and consumer (S/PDIF) data formats at word lengths up to 24-bits. The optical inputs are limited to 96 kHz sample rates. Use the coaxial inputs and the USB input for 192 kHz applications.

Remote Control

The remote-control has the following functions (and their respective icons):

OFF	Press the button twice to turn off the unit. Press the button once to put the unit in standby mode. Once engaged in standby mode, all the lights will remain on for 5 minutes, and the unit will turn off. Press and hold **OFF** button, it will turn off all devices hooked connected to Benchmark's 12V connector. (*DAC2 HGC and DAC2 L model only*)
ON	Turns unit on. Hold down the **ON** button to engage or disengage the **Polarity Switch**
VOLUME	Turns volume up and down
DIM	Engage or disengage dim capabilities
MUTE	Mutes or un-mutes the **DAC2**. Hold down **MUTE** to toggle the **Home Theater Bypass** on the selected input.
INPUT	Cycles through the different inputs
D1	Select **D1** digital optical input
D2	Select **D2** digital optical input
D3	Select **D3** digital coaxial input
D4	Select **D4** digital coaxial input
USB	Selects USB input. Hold down **USB** button to either engage the unit in USB 2.0 mode or USB 1.1 mode. (*Note USB must be plugged into the computer in order to switch modes*)
Analog	Cycle through analog input **A1** and **A2** *(DAC2 HGC and DAC2 L model only)*

MUTE and DIM Functions

The 'Mute' and 'Dim' functions are used to gracefully silence the ***DAC2***. The 'Mute' function will fade the volume down before completely muting, and will ramp the volume up after un-muting. The 'Dim' function will also fade the volume down, but will not completely mute the audio. Dim is convenient for reducing volume to low levels during television or radio commercials or while conducting a conversation.

The level of the 'Dim' volume setting can easily be set by the user with the remote control. The ***DAC2*** will remember the user's preferred 'Dim' setting upon returning to 'Normal' mode, and will recall it when 'Dim' mode is engaged again.

To engage 'Dim' mode, press the 'Dim' button. To set the level of the 'Dim' mode, simply press the 'Volume up' or 'Volume down' button on the remote control until you achieve the desired 'Dim' level. To exit 'Dim' mode and return to 'Normal' mode, simply press the 'Dim' button again.

The 'Dim' level cannot be set higher than the 'Normal' level. A minimum offset will be reached when adjusting the 'Dim' level upward. This minimum offset occurs just below the 'Normal' level setting. If the user continues to raise the volume above the minimum offset, the ***DAC2*** will enter 'Normal' volume mode.

The 'Mute' button quickly fades the volume to a full mute, while moving the rotary volume control to the 'Dim' setting. When exiting 'Mute' mode, the volume will ramp up to the 'Normal' volume setting.

While in 'Normal' or 'Dim' mode, pressing the 'OFF' button will immediately mute the ***DAC2*** and place the system in standby. After 5 minutes of inactivity, the displays will shut down, but all circuits will remain active. Press the power button twice to shut the system down and save power.

Direct Interfacing to Power Amplifiers

The ***DAC2*** is designed to be able to interface directly to power amps and powered studio monitors. This configuration provides the cleanest and shortest path from the digital source to the monitor output and often results in a substantial improvement in sound quality.

The ***DAC2's*** XLR output is equipped with 0, 10 and 20 dB output attenuators for optimal interfacing. The pads optimize the output signal level of the ***DAC2*** to the input sensitivity of virtually any load (amplifier, preamp, etc). Most power amplifiers and powered monitors require the 10 dB or 20 pad setting. The ***DAC2*** is factory-set with the 10 dB pad enabled.

Headphone Mute Switch

(***DAC2 HGC*** and ***DAC2 D*** model only)

The left-hand headphone jack includes a switch that mutes the main analog outputs (XLR and RCA) when a headphone plug is inserted. This feature allows the listener to switch from loudspeaker to headphone playback seamlessly. This mute feature can be disabled with internal jumpers.

Front Panel


DAC2 HGC
Benchmark
D1
D2
U
D3
A2
D4
A1
HT
DSD
4X
2X
48
44
24
16
POWER
DIM/MUTE
POLARITY
INPUT


DAC2 D
Benchmark
D2
D3
D1
D4
U
HT
DSD
4X
2X
48
44
24
16
POWER
DIM/MUTE
POLARITY
INPUT


DAC2 L
Benchmark
D1
D2
U
D3
A2
D4
A1
HT
DSD
4X
2X
48
44
24
16
POWER
DIM/MUTE
POLARITY
INPUT

Input Status Display

The ***DAC2 HGC*** and ***DAC2 L*** have sixteen status LED indicators on the front panel.

The ***DAC2 D*** has fourteen status LED indicators on the front panel.

Indicator Functions

DIM/MUTE – A solid red LED indicates that the unit is in DIM mode. A flashing red LED indicates that the unit is in MUTE mode.

POLARITY – Yellow LED indicates that the polarity is switched

A1 – A solid blue LED indicates analog input 1 is selected. *(DAC2 HGC and DAC2 L model only)*

A2 – A solid blue LED indicates analog input 2 is selected. *(DAC2 HGC and DAC2 L model only)*

U – The blue blinking LED indicates no signal is transmitted to USB input. A solid blue LED indicates a signal is being transmitted to USB input.

D1 – The blue blinking LED indicates no signal is transmitted to optical digital input 1 or data transmitted is incompatible. A solid blue LED indicates a signal is being transmitted to optical digital input 1.

D2 – The blue blinking LED indicates no signal is transmitted to optical digital input 2 or data transmitted is incompatible. A solid blue LED indicates a signal is being transmitted to optical digital input 2.

D3 – The blue blinking LED indicates no signal is transmitted to coaxial RCA digital input 3 or data transmitted is incompatible. A solid blue LED indicates a signal is being transmitted to coaxial RCA digital input 3.

D4 – The blue blinking LED indicates no signal is transmitted to coaxial RCA digital input 4 or data transmitted is incompatible. A solid blue LED indicates a signal is being transmitted to coaxial RCA digital input 4. *Note: if **D4** is not selectable by not lighting up, it indicates that the **Digital Pass Through** is engaged.*

HT – When you choose an input to engage Home Theater Bypass or High Throughput mode using the analog inputs, the **HT** yellow LED will blink and then turn solid when it is engaged. When the **HT** yellow LED is lit, it is accompanied by a blue lit LED on either **A1** *(DAC2 HGC and DAC2 L model only)*, **A2** *(DAC2 HGC and DAC2 L model only)*, **U**, **D1**, **D2**, **D3**, or **D4**.

24 – The solid blue LED lit indicates that the word-length is 24 bits. If the **24** LED and the **16** LED are on at the same time, this indicates that the word-length is between 17-23 bits.

16 – The solid blue LED lit indicates that the word-length is 16 bits. If the **24** LED and the **16** LED are on at the same time, this indicates that the word-length is between 17-23 bits.

44 – The solid blue lit LED indicates that the sample rate is 44 kHz.

48 – The solid blue lit LED indicates that the sample rate is 48 kHz.

2X – The solid blue lit LED indicates that the multiplier is used in conjunction with either **44** to indicate 88.2 kHz or **48** to indicate 96 kHz. If both the **2X** and the **4X** LEDs are lit, this indicates that the signal coming into the DAC is DSD.

4X – The solid blue lit LED indicates that the multiplier is used in conjunction with either **44** to indicate 176.4 kHz or **48** to indicate 192 kHz. If both the **2X** and the **4X** LEDs are lit, this indicates that the signal coming into the DAC is DSD.

Input Status Display

Under normal operation, the **Input Status Display** shows which of the inputs is selected. A single steady light indicates that a proper signal is present. A flashing light indicates that an error is occurring on the selected input.

Input Error Codes:

- **Very slow flashes –** No signal – audio muted
- **Slow flashes –** Data transmission errors or Non-PCM – audio muted
- **Rapid flashes** – Non-audio – audio muted
- **Very rapid or intermittent flashes** – Invalid sample(s) (v-bit) – no mute

Common Causes of Input Errors:

- Disconnected cable
- Data drop-outs due to a bad cable
- Incompatible data type (AC3, ADAT, etc.)
- Non-Audio data

There are no error indications on the analog inputs.

Word-Length Display

The word-length display is indicated by the two LEDs labeled **16** (16-bit) and **24** (24-bit). When a 16-bit track is played, the **16** LED will light up and vice versa when a 24-bit track is played. If a DSD track is played, both the **16** and **24** LEDs will turn off. Compressed MP3 files will display as 24-bits when originating from a player with a 24-bit MP3 decoder. When both the **16** and **24** LEDs are lit, it indicates that the word-length is between 17 to 23-bits. If the **24** LED is lit while playing a 16-bit file, the music player is performing some processing. Please review your music player's settings.

Sample Rate Display

The sample rate display is indicated by the four LEDs labeled **44, 48, 2X, and 4X.**

Sample Rate Reference
44.1 kHz = **44** LED
48 kHz = **48** LED
88.2 kHz = **44** and **2X** LEDs
96 kHz = **48** and **2X** LEDs
176.4 kHz = **44** and **4X** LEDs
192 kHz = **48** and **4X** LEDs
DSD = **2X** and **4X** LEDs

Button Functions

Power – Turns the unit on and off. Press once to turn on. Press once to enter standby mode, press twice to turn off.

Dim/Mute – Pressing the button will engage or disengage the dim and mute functions. If the button is held down for more than 2 seconds, it will engage the ***Home Theater Bypass/High Throughput*** on the selected channel. The **HT** yellow light will turn on in conjunction with the input that is designated with the Home Theater Bypass/High Throughput.

Polarity – Toggles the polarity of the selected input (digital inputs only). LED is on when polarity is inverted.

Input – Press the input select buttons to change the input. To switch between USB 1.1 and 2.0, plug in the USB to your computer. Select the USB input, then hold down both the top and bottom input buttons until either the **4X** (USB 2.0) or **2X** (USB 1.0) flashes. More information can be found on the next section.

USB Mode Selection

To change the USB mode, plug in your ***DAC2*** into the computer through the USB. Select the USB input, and then press and hold both input select buttons (on the faceplate). After holding the buttons for 2 seconds, either the **4X** LED or the **2X** LED will flash once indicating the new USB mode. The **4X** LED flash indicates that the unit is engaged in USB 2.0 for up to 24-bit/192 kHz file playback. The **2X** LED flash indicates that the unit is engaged in USB Audio 1.1 for up to 24-bit/192 kHz playback. The **4X** or **2X** LED will flash once every time the USB input is selected. This flash provides an indication of the USB mode. **It is very important your computer playback is stopped before changing the USB mode**. **Doing so might freeze your computer.** Pressing and holding the USB button on the remote for 2 seconds will also change the USB mode.

HPA2™ Headphone Jacks

(***DAC2 HGC*** and ***DAC2 D*** model only)

The ***DAC2*** features two headphone jacks. The left-hand jack is equipped with a switch that automatically mutes the XLR and RCA analog outputs when a headphone plug is inserted. The right-hand jack has no switch. This feature enables seamless muting of the main outputs when headphones are being used. This auto-mute feature can be enabled or disabled via an internal jumper. Instructions for setting the auto-mute jumper are detailed in the '**Internal Settings**' section of this manual.

TIP: Use the left-hand jack to mute your loudspeaker system. Use the right-hand jack to keep all outputs active.

The dual jacks also allow two listeners to monitor and compare notes on what is heard. When comparing, we recommend using identical headphones because headphone sensitivities differ significantly. The **Volume Control** adjusts the level for both jacks.

The gain of the ***HPA2***™ is high enough for the most difficult headphones, but it may be too high for some other headphones. The ***HPA2***™ in the ***DAC2*** features three gain ranges to customize the output level for a particular set of headphones. These gain ranges are set using internal jumpers. The jumpers reduce the input to the ***HPA2***™ by 0, 10 or 20 dB. These jumpers are factory-installed at 10 dB below full gain. Instructions for setting the headphone gain range are detailed in the 'Internal Settings' section of this manual.

TIP: For optimal performance, the headphone gain jumpers should be set so that comfortable listening levels occur when the 'Volume Control' is set above the '11 o'clock' position.

HGC™ Volume Control

Hybrid Gain Control™

"HGC" is Benchmark's unique **Hybrid Gain Control™** system. The **DAC2** combines active analog gain control, passive low-impedance attenuators, a 32-bit digital gain control, and a servo-driven volume control. All inputs are controlled by the rotary volume control. This volume control moves in response to commands from the remote control. Analog inputs are never converted to digital, and digital inputs never pass through an analog potentiometer. Digital inputs are precisely controlled in the 32-bit DSP system. The DSP system preserves precise L/R balance, and precise stereo imaging, while avoiding any source of noise and distortion. Benchmark's unique passive output attenuators provide distortion-free gain reduction without reducing the dynamic range of the converter. The attenuators optimize the gain staging between the **DAC2** and the power amplifier. This optimization is absolutely essential for maximizing the dynamic range of the entire playback system. Much of the success of the **DAC1** converters can be attributed to the passive output attenuators. Musical details can be obscured by system noise whenever a preamplifier and power amplifier are improperly matched. The **HGC** system will make full use of your power amplifier's dynamic range. Experience the new details in your favorite recordings.

The front-panel volume control is a servo-driven gain circuit control built around a custom-made Alps potentiometer. The custom Alps pot is equipped with remote-controllable motor drive.

This potentiometer is equipped with a clutch which prevents damage from overriding the motor drive. If the pot is driven beyond the end of its range, it will not damage the motor. Also, if the pot is manually overridden, it will not damage the motor.

The XLR outputs have jumper-enabled passive attenuators that can be used to optimize the gain structure of the playback system.

TIP: For optimal performance and minimal noise, the XLR gain jumpers should be set so that comfortable listening levels occur when the 'Volume Control' is set above 11 o'clock.

Instructions for setting the XLR pads are detailed in the 'Internal Settings' section of this manual. The ***DAC2*** is shipped with the XLR attenuation set to -10 dB.

Rear Panel

DAC2 HGC and ***DAC2 L***

DAC2 D

Inputs

There are seven stereo inputs on the ***DAC2***: 2 x Analog *(DAC2 HGC and DAC2 L model only)*, 1 x USB, 2 x Optical, and 2x Coaxial. These inputs are selected using the front-panel **Input** control, or the remote.

The optical and coaxial can decode AES/EBU and S/PDIF input signals in either professional or consumer formats.

TIP: The *DAC2* will not decode AC3 or ADAT signals. The 'Status Display' will flash when AC3, ADAT, or other non-PCM input signals are connected to the selected digital input.

The Benchmark ***UltraLock2***™ system removes interface jitter from all digital inputs (including the USB input). The result is that all digital inputs have identical jitter performance.

Analog Inputs – RCA Unbalanced

The ***DAC2 HGC and DAC2 L*** have two sets of unbalanced stereo analog inputs. The DAC2 D does not have analog inputs.

The analog inputs can be used for devices such as:

- Phono preamplifiers
- FM Tuners
- Tape Transports
- Analog VCR outputs
- iPod and MP3 devices
- Outputs from analog mixing consoles

Computer Input – USB

The USB input accepts a 'B-type' male USB 1.1 or USB 2.0 connector. An 'A-B type' USB cable is provided with the ***DAC2***. The USB cable connects the ***DAC2*** directly to a computer's USB output. The USB interface utilizes USB 1.1 and USB 2.0 protocol, and is compatible with both USB 1.1 and USB 2.0 ports.

The USB input supports 44.1, 48, 88.2, 96, 176.4, and 192 kHz sample rates at word lengths up to 24-bits. The USB also accepts DSD. The USB interface acts as a 'native' USB audio device and does not require the installation of any custom drivers if listening to music up to 96 kHz on either Mac OS X or Windows. Drivers are required for listening to music up to 192 kHz and DSD only for Windows. Drivers are not required for Mac OS X for 192 kHz and DSD playback.

On USB 1.1, the Benchmark USB interface is truly a plug-and-play solution. The ***DAC2*** can begin streaming high resolution 24-bit/96 kHz audio bit-transparently within seconds after being plugged into a computer for the first time. No software or hardware configuration is required.

The ***DAC2's*** USB 1.1 is designed, tested and proven compatible with Windows XP/Vista/7/8, Mac OS X, and iPads using the 30-pin to USB Camera Kit with no driver installation or system configuration required.

The ***DAC2's*** USB 2.0 is designed, tested and proven compatible with Windows XP/Vista/7/8 with driver installation. It was also test on Mac OS X versions 10.6, 10.7, and 10.8 with no driver installation.

For the up-to-date information about more recent operating systems and suggestions for optimization, go to: www.benchmarkmedia.com/wiki.

Optical Digital Inputs - D1 and D2

The optical input connector is commonly known as a TOSLINK connector. The TOSLINK optical connector used on the ***DAC2*** is designed to work well at sample rates up to 96 kHz. Maximum word length is 24-bits. All sample rates between 28 and 96 kHz are supported. The optical input will accept professional AES/EBU data formats or consumer S/PDIF data formats.

Coaxial Digital - D3 and D4

The coaxial inputs use female RCA connectors that are securely mounted directly to the rear panel. The input impedance is 75 Ohms. Maximum word length is 24-bits. All sample rates between 28 and 195 kHz are supported.

The Coax inputs are DC isolated, current limited, and diode protected. The RCA body is bonded directly to the chassis to prevent currents in the internal ground system. This direct bonding also maximizes RF shielding.

TIP: Shielded 75-Ohm coaxial cable is required for stable performance. Do not use 50-Ohm cables or twisted pair cables, or any non-coaxial cables.

The Coaxial inputs accept AES/EBU or S/PDIF digital audio formats. AES3-id and SMPTE 276M standards specify 75-Ohm 1 Vpp professional format digital audio signals and these are commonly used in video production facilities. IEC 609588-3 specifies 75-Ohm 0.5 Vpp consumer-format digital audio signals (commonly known as S/PDIF). The coaxial inputs on the ***DAC2*** are designed to accept either type of signal.

Outputs

Analog Outputs

The ***DAC2*** has one pair of balanced XLR outputs and two pairs of unbalanced RCA outputs.

The ***DAC2*** features high-current output drivers that are capable of driving 300-Ohm loads without an increase in distortion. They are also well suited for driving long cables or high-capacitance loads.

Balanced XLR Analog Line Outputs

The Left and Right balanced outputs use Neutrik™ gold-pin male XLR jacks. The XLR shell and pin 1 (ground) are both directly bonded to the chassis to prevent currents in the internal ground system. This direct bonding also maximizes RF shielding.

The XLR outputs have passive attenuators that allow direct connections to a wide variety of audio devices without a loss of dynamic range. The 10 or 20 dB pads are usually required for direct interfacing to power amplifiers and powered speakers. The ***DAC2*** ships with the 10 dB pads enabled. A full description of the output attenuators and instructions for configuration is located in the **Internal Settings** section of this manual.

Industry-standard XLR wiring:
XLR pin 2 = + Audio Out
XLR pin 3 = - Audio Out
XLR pin 1 = Cable Shield

CAUTION: If the balanced XLR outputs are wired to an unbalanced input (using a special adapter cable), pin 3 must be left floating. Shorting pin 3 to ground will increase the temperature of the output drivers, will increase power consumption, and may cause distortion.

Unbalanced RCA Analog Outputs

The Left and Right unbalanced outputs use standard RCA style jacks. The ground connections are bonded to chassis ground at the location where analog ground is bonded to the chassis. This minimizes the effects of ground loops caused by AC currents in the cable shield.

TIP: Mono summing with an RCA 'Y' cable is not recommended as this will cause high amounts of distortion. Mono summing with a 'Y' cable can be accomplished with the use of a modified cable by implementing a 1k Ohm series resistor in each leg of the 'Y'.

Note: The XLR pads do not have any effect on the level of the RCA outputs.

The RCA output impedance is very low (30 Ohms). This makes these outputs well suited for driving high-capacitance loads and/or high-capacitance cables.

AC Power-Entry and Fuse Module

Fuses

CAUTION: FOR CONTINUED FIRE HAZARD PROTECTION ALWAYS REPLACE THE FUSES WITH THE CORRECT SIZE AND TYPE (0.5A 250 V SLO-BLO® 5 X 20 MM – LITTELFUSE® HXP218.500 OR EQUIVALENT). THE FUSE DRAWER INCLUDES TWO FUSES. ALWAYS REPLACE BOTH FUSES AT THE SAME TIME.

Voltage Selection

THE DAC2 IS EQUIPPED WITH A UNIVERSAL POWER SUPPLY. THERE IS NO VOLTAGE SELECTION SWITCH. AC VOLTAGE RANGE IS 88-264 VAC, 50-60 HZ. THE PRODUCT MAY ALSO BE OPERATED FROM DC POWER OVER A VOLTAGE RANGE OF 125-373 VDC.

Power Cord

The AC power input uses a standard IEC type connector. One USA-compatible power cord is included with ***DAC2*** converters. IEC style power cords in country-specific configurations are available in your locality.

CAUTION: ALWAYS USE A GROUNDED POWER CORD. THE PRODUCT IS EQUIPPED WITH A STANDARD IEC POWER ENTRY MODULE. USE AN IEC POWER CORD THAT IS EQUIPPED WITH THE APPROPRIATE CONNECTOR FOR YOUR LOCATION. CORDS ARE AVAILABLE FROM YOUR DEALER.

Internal Settings

Removing Top Cover

The ***DAC2*** cover must be removed to gain access to the jumpers. Do not attempt to remove the faceplate or rear panel.

CAUTION: The *DAC2* contains static sensitive components and should only be opened by qualified technicians. Static discharge may cause component failures, may affect the long-term reliability, or may degrade the audio performance. Use a static control wrist strap when changing jumper settings.

CAUTION:

- **Disconnect AC power by unplugging the power cord at the back of the *DAC2*.**
- **Remove only the 8 screws holding the cover (4 on each side).**
- **Do not remove any screws on front or rear panels.**
- **Never remove the power entry safety cover in the rear corner of the *DAC2*.**
- **Always connect a static-control wrist strap to the chassis before touching any internal component.**

Low-Impedance Passive Pads

The XLR outputs are equipped with low-impedance passive pads that may be used to reduce the output levels while preserving the full dynamic range of the ***DAC2***. The ***DAC2*** ships with the 10 dB pads enabled.

TIP: When directly driving power amplifiers and powered speakers, start with the factory default 10 dB pad setting. If necessary, change the pads so that normal listening levels are achieved with the 'Volume Control' above the 11 o'clock position.

When the output pads are enabled, the output impedance changes slightly, and the maximum allowable cable length should be reduced as shown in Table 1 (assuming 32 pF/foot and a maximum allowable loss of 0.1 dB at 20 kHz).

Table 1 - Cable Drive Capability

Balanced Output Drive Capability:

Attenuator Setting (dB)	Output Impedance	Maximum Cable (ft)	Loss in dB at 20 kHz
0	60	680	0.1
10	425	96	0.1
20	135	302	0.1

Unbalanced Output Drive Capability:

Output Impedance	Maximum Cable (ft)	Loss in dB at 20 kHz
30	1360	0.1

TIP: To set the XLR outputs to typical professional studio levels, set the pads to 0 dB. For most home installations, set the pads to 10 dB or 20 dB.

Jumpers

The following functions are jumper configured:

- Headphone Gain Range Adjustment (*DAC2 HGC and DAC2 D model only*)
- Headphone Switch Disable (*DAC2 HGC and DAC2 D model only*)
- XLR Output Pads
- Digital Pass Through Enable

XLR Output Pad Selection (P8, P9, P10, and 11):

Four 6-pin headers (P8, P9, P10, and P11) allow selection of the output level at the XLR jacks.

One pair of 6-pin headers controls the output level at each XLR jack as follows:

- 0 dB - (Attenuator disabled) – (Jumper plug between pins 1 and 2 of each header)
- -10 dB – *** (Jumper plug between pins 3 and 4 of each header) – *Factory Default*
- -20 dB – (Jumper plug between pins 5 and 6 of each header)

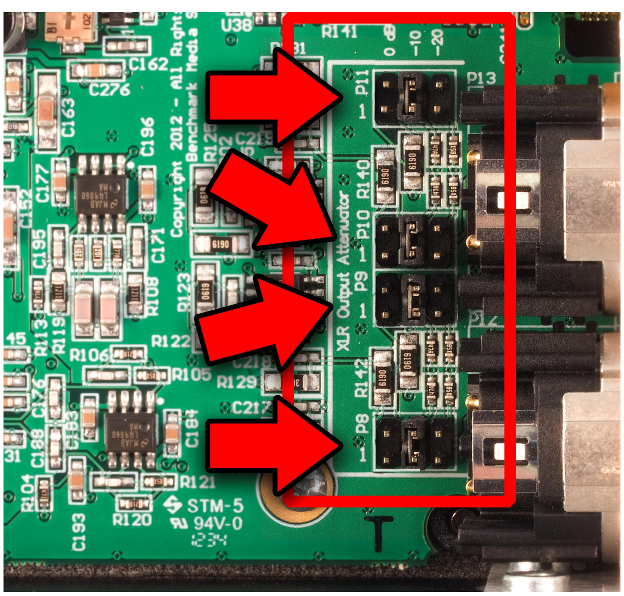


Figure 1 - 10 dB (Factory Default)

Headphone Switch Disable (JP1 and JP2):

(***DAC2 HGC*** and ***DAC2 D*** model only)

The ***DAC2*** is configured so that the analog outputs will mute when a headphone plug is inserted into the left-hand jack. This is convenient when the user wishes to switch between headphones and speakers. This feature can be defeated by adding jumpers at JP1 and JP2.

JP1 and JP2 should be configured as follows:

- Headphone Switch enabled*** (Jumpers Removed)
- Headphone Switch disabled (Jumpers Inserted)

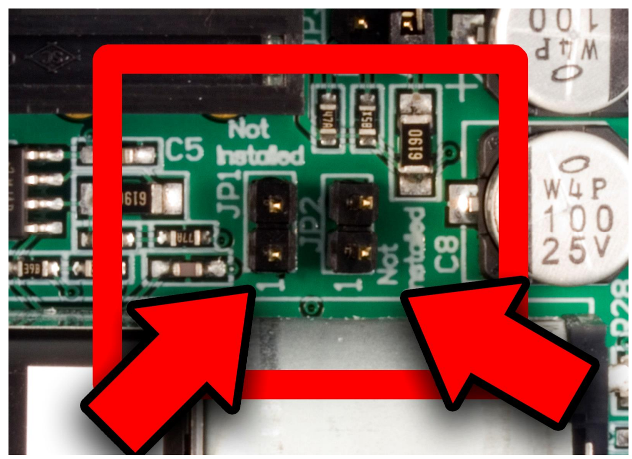


Figure 2 * - Factory Default**

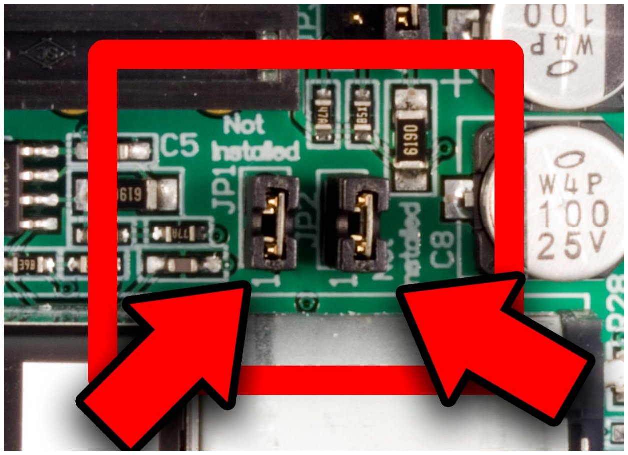


Figure 3 - Disable Headphone Switch

Headphone Gain Reduction (JP3 and JP4):

(***DAC2 HGC*** and ***DAC2 D*** models only)

The gain range of the ***HPA2***™ can be set using jumpers JP3 and JP4. When jumpers are installed at position "A" the headphone amplifier gain is decreased by 20 dB. When jumpers are installed at position "B" the headphone amplifier gain is decreased by 10 dB.

The ideal gain setting permits the user to set the front-panel **Volume Control** above 40% (10 o'clock) without the headphone volume being too loud.

JP3 and JP4 are factory installed for a headphone attenuation of 10 dB. This setting is best for most applications as shown in **Figure 4 - 10 dB (Factory default)**. Move the jumpers according to example in **Figure 5 - 0 dB** for more gain or move the jumpers according to example in **Figure 6 - 20 dB** for less gain.

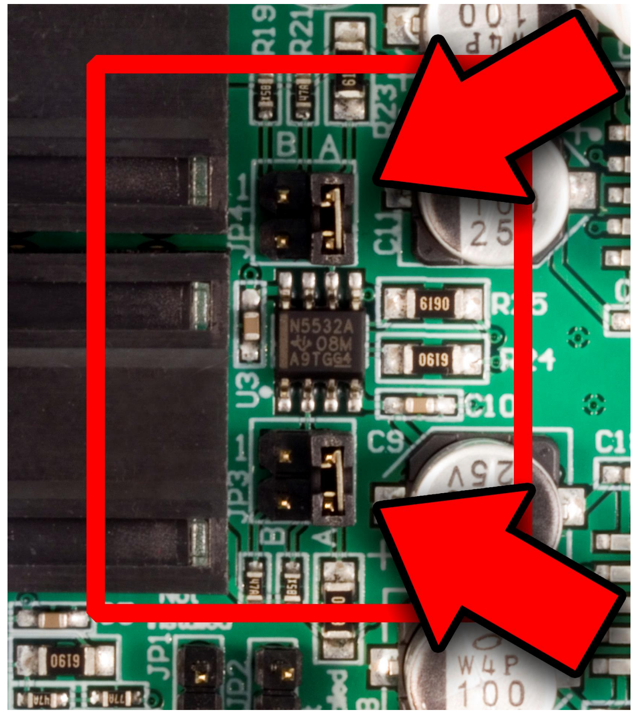


Figure 5 - 0 dB

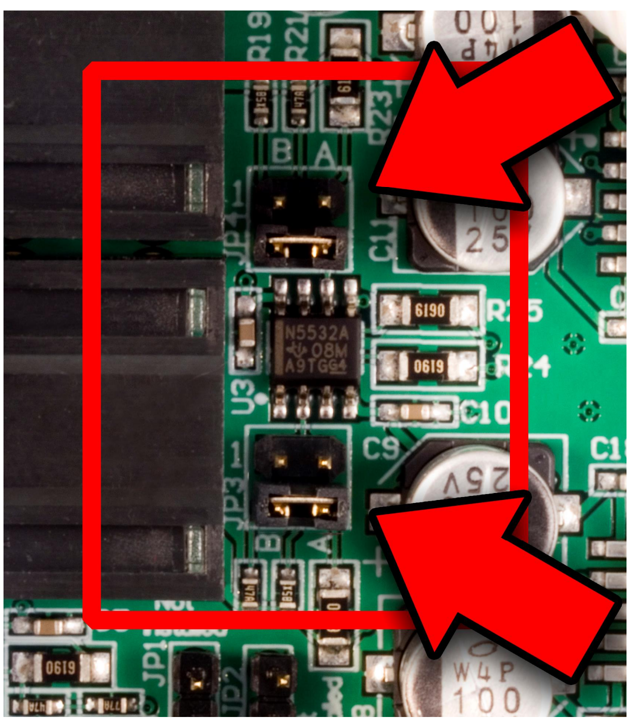


Figure 4 - 10 dB (Factory default)

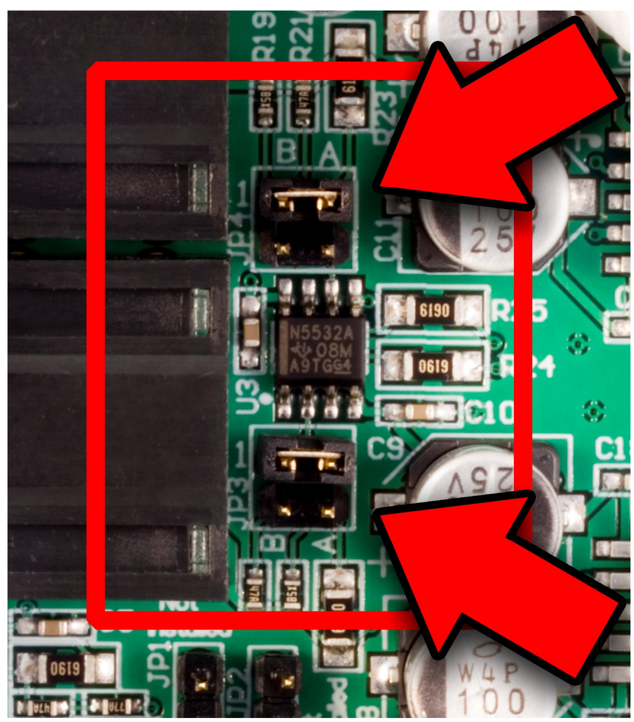


Figure 6 - 20 dB

Digital Pass Through

The Digital Pass Through can be enabled by moving both P14 jumpers towards the faceplate shown in **Figure 7**. Once the jumpers are moved into the position shown in **Figure 7**, input **D4** is active to function as a digital pass through.

By default, **D4** functions as a digital input so the jumpers are set according to **Figure 8**.

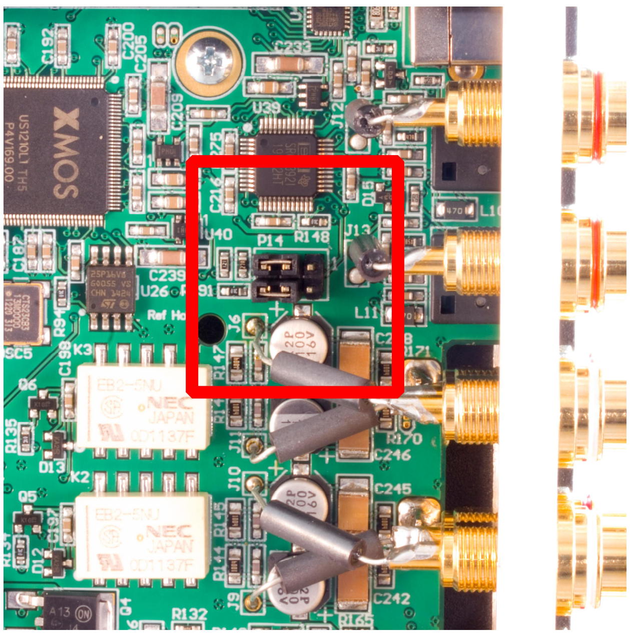


Figure 7 – Digital Pass Through enabled

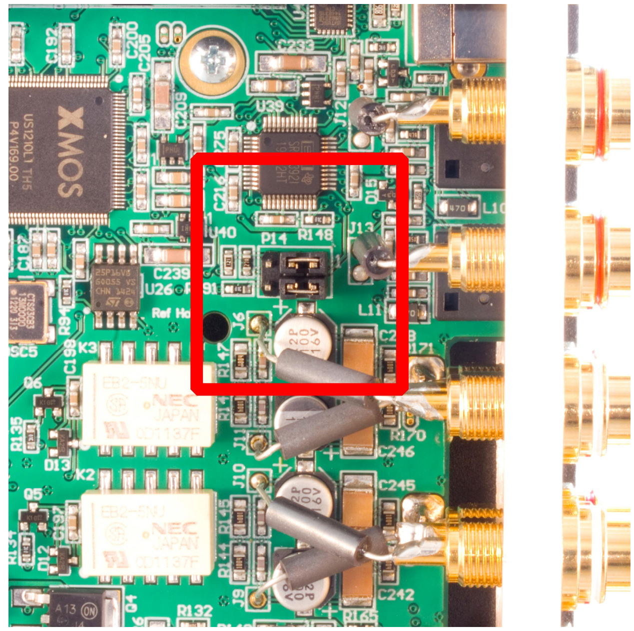


Figure 8 - (Factory default)

Rack Mounting

An optional rack mount adapter allows the mounting of any two Benchmark ***System1™*** products in a single rack space. A **Blank Rack Panel** can be added when only one unit is installed in the rack mount adapter.

The ***System1™ Universal Rack Adapter*** and ***Blank Rack Panel*** are available from Benchmark.

Call us, visit our website (http://www.BenchmarkMedia.com), or contact your dealer to purchase these accessories.

System1™ Universal Rack Adapter

The ***Universal Rack Mount Adapter*** is a tray that mounts up to two ½ RU Benchmark products in a single race space. The tray accepts any combination of ½ RU Benchmark products (with or without rack-mount type faceplates).

Blank Rack Panel

The ***Blank Rack Panel*** is a ½-wide 1-RU black-anodized aluminum panel for covering an unused slot in a ***System1™ Universal Rack Adapter***.

Benchmark Technologies

Hybrid Gain Control™

"HGC" is Benchmark's unique **Hybrid Gain Control™** system. The **DAC2** combines active analog gain control, passive low-impedance attenuators, a 32-bit digital gain control, and a servo-driven volume control. All inputs are controlled by the rotary volume control. This volume control moves in response to commands from the remote control. Analog inputs are never converted to digital, and digital inputs never pass through an analog potentiometer. Digital inputs are precisely controlled in the 32-bit DSP system. The DSP system preserves precise L/R balance, and precise stereo imaging, while avoiding any source of noise and distortion. Benchmark's unique passive output attenuators provide distortion-free gain reduction without reducing the dynamic range of the converter. The attenuators optimize the gain staging between the ***DAC2*** and the power amplifier. This optimization is absolutely essential for maximizing the dynamic range of the entire playback system. Much of the success of the ***DAC1*** converters can be attributed to the passive output attenuators. Musical details can be obscured by system noise whenever a preamplifier and power amplifier are improperly matched. The "HGC" system will make full use of your power amplifier's dynamic range. Experience the new details in your favorite recordings.

Native DSD Conversion

The digital coaxial inputs and the USB 2.0 input on the **DAC2** support native DSD conversion. DoP 1.1 is automatically detected on all digital inputs, and the system seamlessly switches to native DSD conversion. DoP 1.1 is supported by many media players, and DSD downloads are now available from several sources.

Visit our website for an up-to-date list of DSD and PCM high resolution download sites.

Multi-Mode Asynchronous USB Audio

Benchmark's USB system supports USB Audio 2.0, DSD, and USB Audio 1.1. It is frequency agile, and will follow sample rate changes initiated by the computer and/or the media playback software. In all modes the USB communications are asynchronous in order to eliminate unnecessary sources of jitter.

The **DAC2** has a low-jitter master clock which controls the transfer of audio data from the computer to the USB sub-system. The computer asynchronously transfers audio data to a buffer in the ***DAC2***. The contents of the buffer are then asynchronously transferred to the D/A conversion sub-system. This second asynchronous transfer eliminates any traces of jitter that accumulate as the data is transferred between the USB and conversion subsystems. No traces of jitter-induced distortion are measurable to our measurement limits (better than -160 dBFS). This truly represents the state-of-the art. Enjoy the convenience of computer playback without compromise.

The Asynchronous USB system supports USB Audio 2.0 for high-resolution 192kHz, and DSD playback. No drivers are required for MAC operating systems. An easy-to-install driver adds 192 kHz and DSD capabilities to Windows operating systems. You can download the latest drivers from:

http://www.benchmarkmedia.com/dac/dac2-drivers

An asynchronous driverless USB Audio 1.1 mode supports sample rates up to 96 kHz. This USB mode can be selected from the front panel or from the remote control. The driverless USB Audio 1.1 mode allows quick plug-and-play connections to Windows, MAC, iOS, and Linux operating systems without installing drivers. Just plug in the USB, and the ***DAC2*** becomes an available audio device.

The USB subsystem remains active when the **DAC2** is powered down. This prevents interruptions to the computer playback operations and eliminates the need to reconfigure the computer every time the converter is turned on.

Jitter-Immune *UltraLock2™*

UltraLock2™ is an improved version of the ***UltraLock™*** system used in the DAC1 product family. DSP processing is 32-bits, DSP headroom is 3.5 dB, sample rate is 211 kHz, and jitter-induced distortion and noise is at least 160 dB below the level of the music - well below the threshold of hearing. Benchmark's ***UltraLock2™*** system eliminates all audible jitter artifacts.

High Head-Room DSP

All of the digital processing in the ***DAC2*** is designed to handle signals as high as +3.5 dBFS. Most digital systems clip signals that exceed 0 dBFS. The 0 dBFS limitation seems reasonable, as 0 dBFS is the highest sinusoidal signal level that can be represented in a digital system. However, a detailed investigation of the mathematics of PCM digital systems will reveal that inter-sample peaks may reach levels slightly higher than +3 dBFS while individual samples never exceed 0 dBFS. These inter-sample peaks are common in commercial releases, and are of no consequence in a PCM system until they reach an interpolation process. But, for a variety of reasons, virtually all audio D/A converters use an interpolation process. The interpolation process is absolutely necessary to achieve 24-bit state-of-the art conversion performance. Unfortunately, inter-sample overs cause clipping in most interpolators. This clipping produces distortion products that are non-harmonic and non-musical. We believe these broadband distortion products often add a harshness or false high-frequency sparkle to digital reproduction. The ***DAC2*** avoids these problems by maintaining at least 3.5 dB of headroom in the entire conversion system. We believe this added headroom is a groundbreaking improvement.

32-bit SABRE Conversion System

Four balanced 32-bit D/A converters deliver audio to Benchmark's low-impedance current to voltage converters. The 4:1 redundancy reduces noise and distortion to levels that set new benchmarks. The conversion system at the core of the ***DAC2*** is as good as it gets. The analog circuits that follow the D/A converter are carefully designed. Benchmark has leveraged its long history of building reference analog audio equipment to create an outstanding output stage.

Diagnostic Display

Ever wonder why that 192 kHz 24-bit download on your computer just doesn't sound right? Your media player may be down-sampling to 44.1 kHz and/or truncating to 16-bits. Many media players apply poor-quality sample rate conversion and truncation. Fortunately these problems can be eliminated with the selection of a good frequency-agile media player. The sample-rate and word-length displays on the ***DAC2*** confirm the proper operation of your media player.

Bi-Directional 12 Volt Trigger

(***DAC2 HGC*** and ***DAC2 L*** models only)

Benchmark re-invents the 12 volt trigger. The trigger connection on the ***DAC2*** can be used as an input or output or both, and is compatible with any common 12 volt trigger input or output. The trigger can be used to turn a power amplifier on or off automatically. The ***DAC2*** will also respond to a 12 volt trigger and follow the actions of another audio component.

Distributed Power Regulation

To achieve the lowest possible noise, the ***DAC2*** uses distributed power supply regulation. Each critical subsystem has at least one dedicated low-noise voltage regulator.

HPA2™ Headphone Amplifier

(***DAC2 HGC*** and ***DAC2 D*** models only)

The ***DAC2*** headphone output is driven by Benchmark's signature ***HPA2***™ headphone amplifier. This high-current, high-output amplifier has an output impedance of near 0-Ohms. It is designed to drive loads as low as 30 Ohms without any increase in distortion. It also has sufficient amplitude to drive low-sensitivity 600-Ohm headphones.

The ***HPA2***™ includes current-limiting circuits that fully protect against damage from short circuits. This is important because the right channel of a headphone amplifier will experience a short whenever a mono phone plug is inserted into the stereo headphone jack. Shorts may also occur when a plug is partially inserted.

0-Ohm Output Impedance

Most headphone amplifiers use series resistors to maintain stability and protect against short-circuit conditions. These resistors are usually at least 30 Ohms, and have a negative impact on performance. A headphone amplifier with series resistors may measure very well when driving resistive loads. However, the same amplifier will measure very poorly when driving a headphone load. Unfortunately, most manufacturers do not specify headphone amplifier performance with anything other than ideal resistive loads. Our measurements show that headphones do not behave like resistive loads.

Headphone Performance

In our tests we have measured substantial distortion across resistors that are wired in series with headphones. We conducted measurements with a variety of headphones. In general, distortion increases as headphone impedance decreases. This distortion can be eliminated with a properly designed 0-Ohm headphone amplifier.

The performance of the ***HPA2***™ does not change when headphones are driven. THD+N measurements for no-load, 30-Ohm resistive loads, 30-Ohm headphone loads, and 600-Ohm headphone loads are virtually identical. The ***HPA2***™ will substantially improve the sound of 30 and 60-Ohm headphones. It will make very noticeable improvements with 600-Ohm headphones.

Differential Amplifiers

Differential amplifiers remove common-mode distortion components from the D/A converter outputs. This feature is critical for achieving low-distortion in down-stream devices. Benchmark addresses common-mode distortion so that it will not cause distortion in power amplifiers and other connected devices. Common-mode distortion can cause audible distortion while escaping the scrutiny of an audio analyzer. The balanced and unbalanced outputs on the ***DAC2*** deliver very similar performance.

Many D/A converters omit the differential amplifiers after the converters. Specifications usually ignore common-mode distortion. A balanced signal with high common-mode distortion can measure just fine when feeding a precisely balanced input on a high-quality audio analyzer. However, any imbalance in a downstream device will expose the common-mode distortion.

UltraLock2™ Clock System

Accurate 24-bit audio conversion requires a very low-jitter conversion clock. Jitter can very easily turn a 24-bit converter into a 16-bit converter (or worse). There is no point in buying a 24-bit converter if clock jitter has not been adequately addressed.

Jitter is present on every digital audio interface. This type of jitter is known as '**interface jitter**' and it is present even in the most carefully designed audio systems. Interface jitter accumulates as digital signals travel down a cable and from one digital device to the next. If we measure interface jitter in a typical system we will find that it is

10 to 10,000 times higher than the maximum allowable level for accurate 24-bit conversion. Fortunately, interface jitter has absolutely no effect on the audio unless it influences the conversion clock in an analog-to-digital converter (A/D) or in a digital-to-analog converter (D/A).

Many converters use a single-stage Phase Lock Loop (PLL) circuit to derive their conversion clocks from AES/EBU, Wordclock, or Superclock reference signals. Single-stage PLL circuits provide some jitter attenuation above 5 kHz but none below 5 kHz. Unfortunately, digital audio signals often have their strongest jitter components at 2 kHz. Consequently, these converters can achieve their rated performance only when driven from very low jitter sources and through very short cables. It is highly unlikely that any converter with a single-stage PLL can achieve better than 16 bits of performance in a typical installation. Specified performance may be severely degraded in most installations.

Better converters usually use a two-stage PLL circuit to filter out more of the interface jitter. In theory, a two-stage PLL can remove enough of the jitter to achieve accurate 24-bit conversion (and some do). However, not all two-stage PLL circuits are created equal. Many two-stage PLLs do not remove enough of the low-frequency jitter. In addition, two-stage PLL circuits often require several seconds to lock to an incoming signal. Finally, a two-stage PLL may fail to lock when jitter is too high, or when the reference sample frequency has drifted.

***UltraLock*™** converters exceed the jitter performance of two-stage PLL converters, and are free from the slow-lock and no-lock problems that can plague two-stage PLL designs. ***UltraLock*™** converters have extremely high immunity to interface jitter under all operating conditions. No jitter-induced artifacts can be detected using an Audio Precision System 2 Cascade test set. Measurement limits include detection of artifacts as low as –160 dBFS, while applying jitter amplitudes as high as 12.75 UI, over a frequency range of 2 Hz to 200 kHz. Any AES/EBU signal that can be decoded by the AES/EBU receiver will be reproduced without the addition of any measurable jitter artifacts.

The ***DAC2*** employs Benchmark's ***UltraLock2*™** technology to eliminate jitter-induced performance problems. ***UltraLock2*™** technology isolates the conversion clock from the digital audio interface clock. Jitter on a D/A digital audio input, or an A/D reference input can never have any measurable effect on the conversion clock of an ***UltraLock2*™** converter. In an ***UltraLock2*™** converter, the conversion clock is never phase-locked to a reference clock. Instead the converter oversampling-ratio is varied with extremely high precision to achieve the proper phase relationship to the reference clock. The clock isolation of the ***UltraLock2*™** system ensures that interface jitter can never degrade the quality of the audio conversion. Specified performance is consistent and repeatable in any installation with cables of any quality level!

How does conversion clock jitter degrade converter performance?

Problem #1: Jitter phase-modulates the audio signal. This modulation creates sidebands (unwanted tones) above and below every tone in the audio signal. Worse yet, these sidebands are often widely separated from the tones in the original signal.

Jitter-induced sidebands are not musical in nature because they are not harmonically related to the original audio. Furthermore, these sidebands are poorly masked (easy to hear) because they can be widely separated above and below the frequencies of the original audio tones. In many ways, jitter induced distortion resembles intermodulation distortion (IMD). Like IMD, jitter induced distortion is much more audible than harmonic distortion, and more audible than THD measurements would suggest.

Jitter creates 'new audio' that is not harmonically related to the original audio signal. This 'new audio' is unexpected and unwanted. It can cause a loss of imaging, and can add a low and mid frequency ‘muddiness’ that was not in the original audio.

Jitter induced sidebands can be measured using an FFT analyzer.

Problem #2: Jitter can severely degrade the anti-alias filters in an oversampling converter. This is a little known but easily measurable effect. Most audio converters operate at high oversampling ratios. This allows the use of high-performance digital anti-alias filters in place of the relatively poor performing analog anti-alias filters. In theory, digital anti-alias filters can have extremely sharp cutoff characteristics, and very few negative effects on the in-band audio signal. Digital anti-alias filters are usually designed to achieve at least 100 dB of stop-band attenuation. But, digital filters are designed using the mathematical assumption that the time interval between samples is a constant. Unfortunately, sample clock jitter in an A/D or D/A varies the effective time interval between samples. This variation alters the performance of these carefully designed filters. Small amounts of jitter can severely degrade stop-band performance, and can render these filters useless for preventing aliasing.

The obvious function of a digital anti-alias filter is the removal of audio tones that are too high in frequency to be represented at the selected sample rate. The not-so-obvious function is the removal of high-frequency signals that originate inside the converter box, or even originate inside the converter IC. These high-frequency signals are a result of crosstalk between digital and analog signals, and may have high amplitudes in a poorly designed system. Under ideal (low jitter) conditions, a digital anti-alias filter may remove most of this unwanted noise before it can alias down into lower (audio) frequencies. These crosstalk problems may not become obvious until jitter is present.

Stop-band attenuation can be measured very easily by sweeping a test tone between 24 kHz and at least 200 kHz while monitoring the output of the converter.

Put *UltraLock*™ converters to the test:

We encourage our customers to perform the above tests on ***UltraLock2™*** converters (or let your ears be the judge). There will be absolutely no change in performance as jitter is added to any digital input on an ***UltraLock2™*** converter. Try the same tests on any converter using conventional single or two-stage PLL circuits. Tests should be performed with varying levels of jitter and with varying jitter frequencies. The results will be very enlightening. Jitter related problems have audible (and measurable) effects on A/D and D/A devices. Practitioners of Digital Audio need to understand these effects.

Is it possible to eliminate all of the effects of jitter in an entire digital audio system?

Interface jitter will accumulate throughout even the most carefully designed digital audio system. Fortunately, **interface jitter** can only degrade digital audio if it affects the sampling circuit in an analog-to-digital or digital-to-analog converter. Any attempt to cure jitter outside of an A/D or D/A will prove expensive and, at best, will only partially reduce jitter-induced artifacts. Dedicated clock signals (word clock, and super clock, etc.) are often distributed to A/D converters and D/A converters in an attempt to reduce jitter. Again, these are only partial solutions because jitter even accumulates in these clock distribution systems. Furthermore, a poor quality master clock generator can degrade the performance of the entire system (if converter performance is dependent upon reference clock quality). Jitter free A/D and D/A converters are the only true ensurance against the ill effects of jitter. ***UltraLock2™*** converters are jitter-immune under all operating conditions (they will never add audible jitter induced artifacts to an audio signal).

What *UltraLock*™ converters cannot do:

UltraLock2™ converters cannot undo damage that has already been done. If an A/D with a jitter problem was used to create a digital audio signal, then there is nothing that can be done to remove the damage. Jitter-induced sidebands are extremely complex and cannot be removed with any existing audio device. Therefore, it is very important to attack jitter at both ends of the audio chain. The ***DAC2*** is a great start, as it will allow accurate assessment of various A/D converters. It is impossible to audibly evaluate A/D performance without a good D/A. The consistent performance delivered by the ***DAC2*** eliminates one major variable: jitter.

Multi-Mode Asynchronous USB Audio

Plug it in and Start Listening… Immediately

Benchmark's ***Advanced USB Audio*** technology is truly 'Plug and Play'. When connecting to a USB port on a computer running Windows or Mac OSX, the computer will automatically and instantaneously recognize the presence of the Benchmark USB device in USB 1.1, playing tracks up to 96 kHz 24-bit. Any audio played from the computer will then be routed to the Benchmark USB device immediately. There is no software to install or configure.

To play tracks up to 192 kHz or DSD, the **DAC2** needs to be engaged in USB 2.0 mode. To engage it in USB 2.0 mode, hold down the USB button on the remote control until you see the **4X** LED blink. If the **2X** LED blinks instead of the **4X** LED, hold down the USB button again on the remote control until you see the **4X** LED blink. In Mac OS X, no driver is required to play tracks up to 192 kHz or DSD. On Windows XP, Vista, or 7, a driver is required for 192 kHz or DSD playback.

To download the driver and get the instructions for Windows XP, Vista, 7, and 8, please visit: http://www.benchmarkmedia.com/dac/dac2-drivers

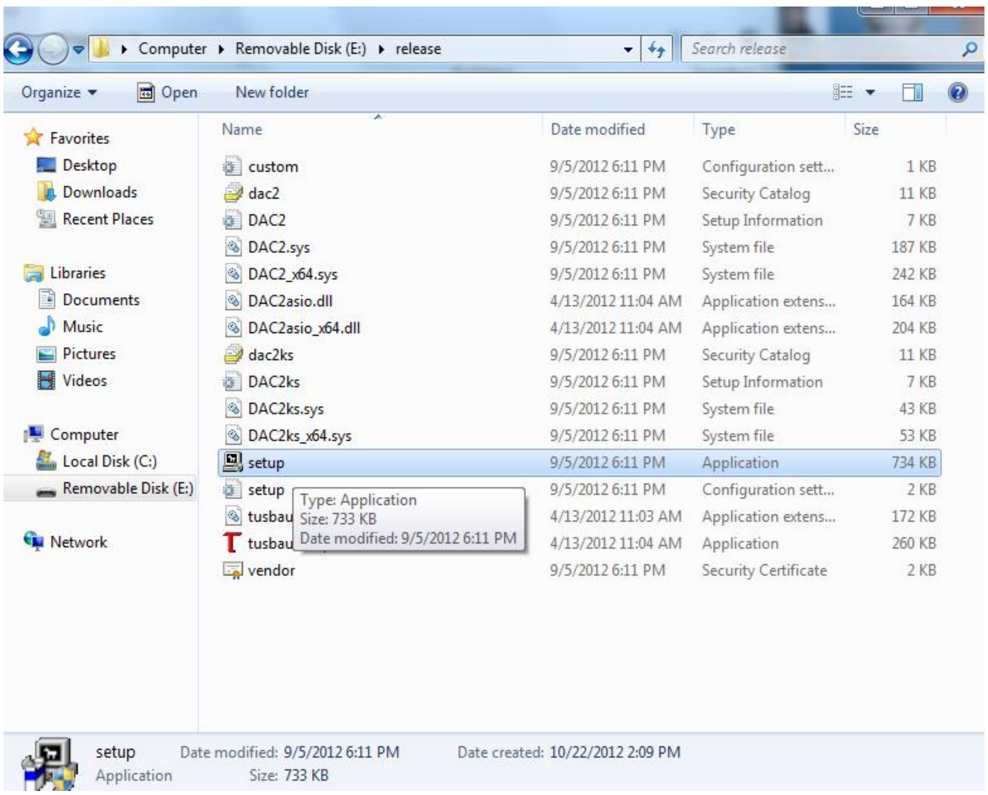
USB Driver Installation

Windows XP, Vista, 7

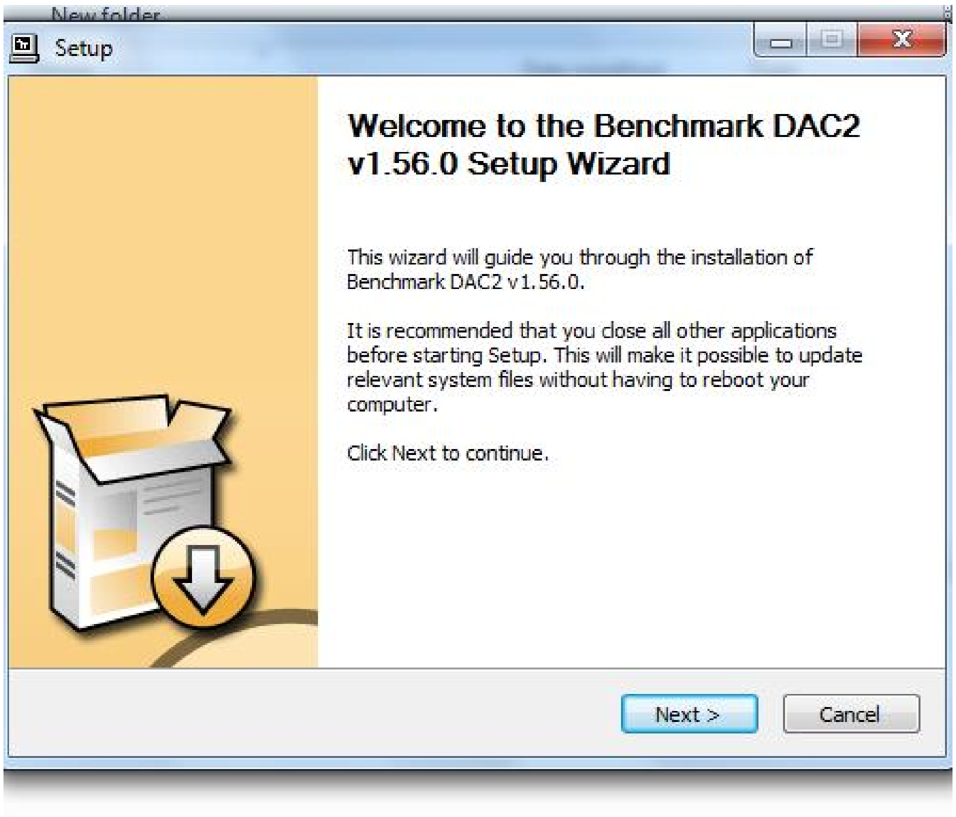
Note: The ***DAC2*** driver is available for download at: *http://www.benchmarkmedia.com/dac/dac2-hgc/driver*

Before you install the driver, make sure the USB is unplugged before installation of the driver.

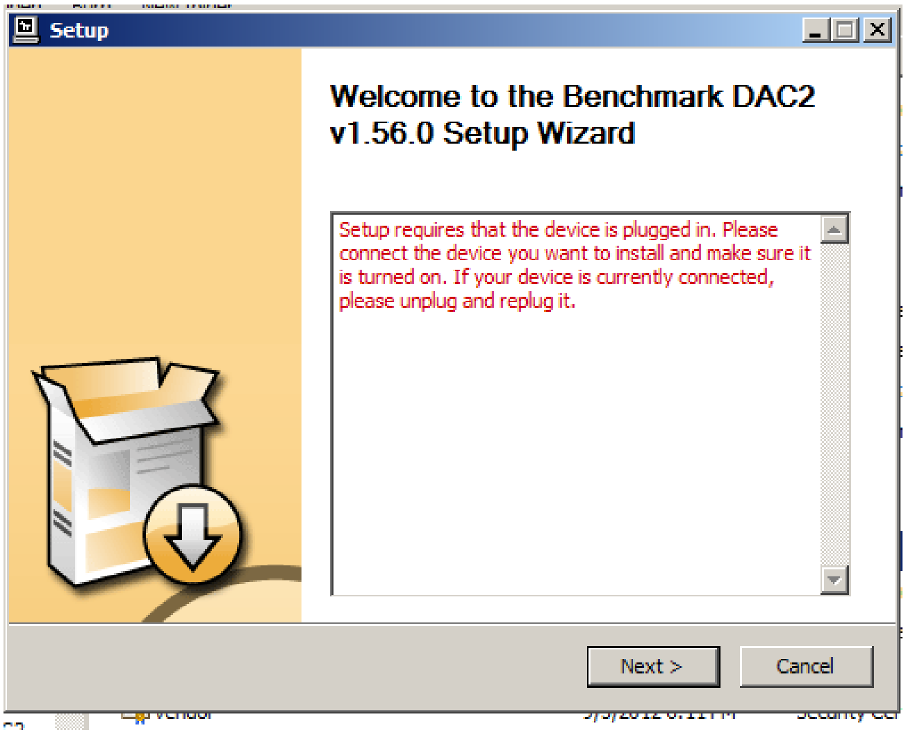
1. **In the DAC2 Driver folder, double click "*setup.exe.*"**



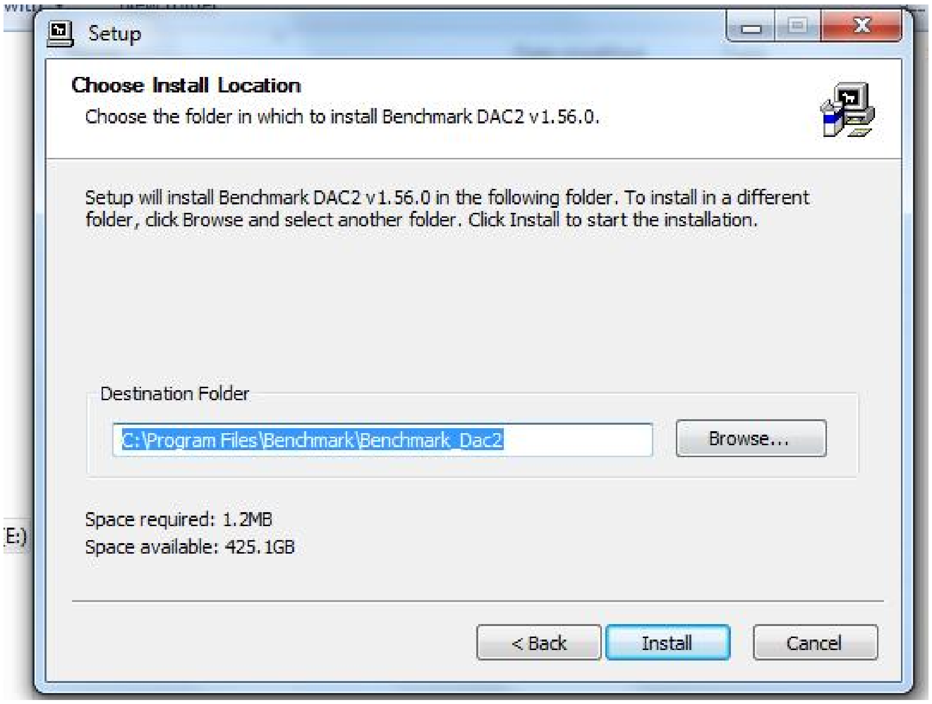
2. **A welcome screen will pop-up. Click "*Next.*"**



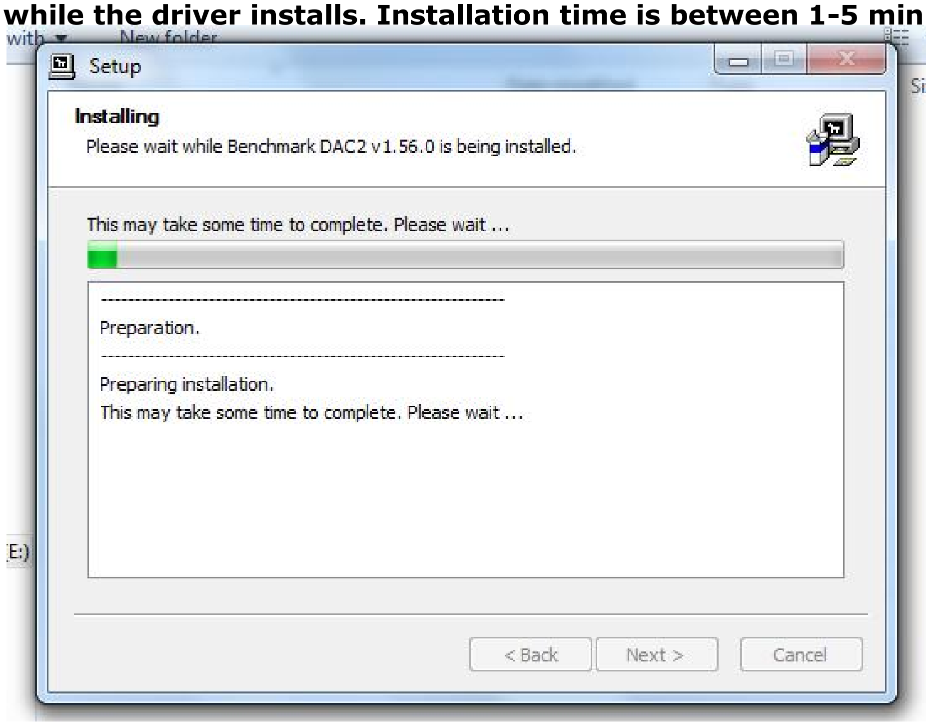
3. **When you see the following screen, turn on the DAC2, and select USB as your input. By default, the DAC2 is shipped in USB 1.0 mode. You can enable your DAC2 in USB 2.0 mode in two ways. 1) Using your remote control, hold down the USB button on your remote control for 3-4 seconds until you see the 4X LED light up for 3-4 seconds. 2) From the front faceplate, hold down both input buttons until you see the 4X LED light up for 3-4 seconds.**



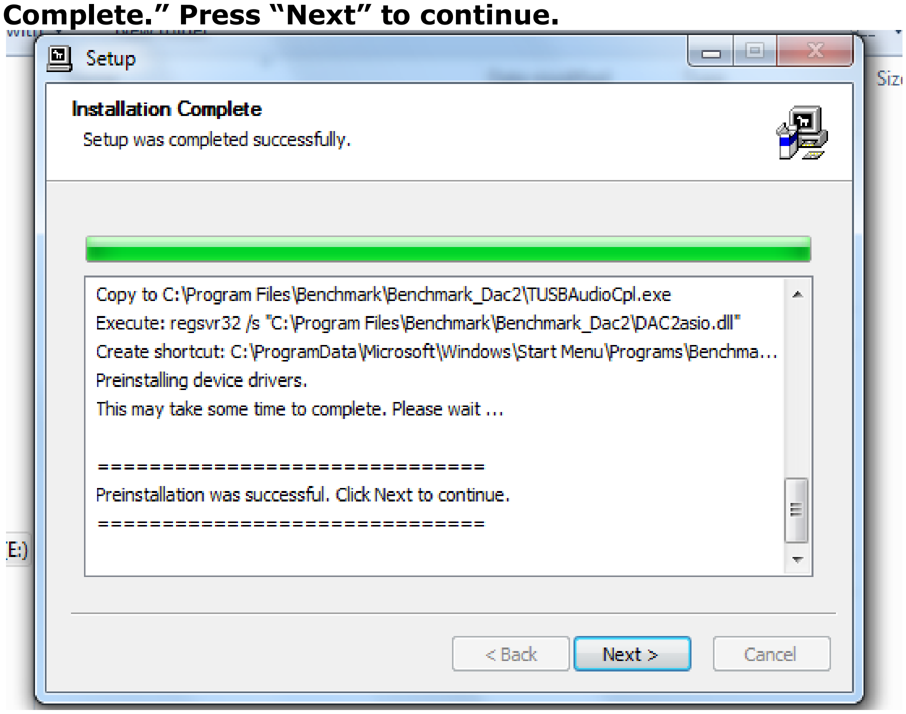
4. **You will be prompted to select a location to install the driver in. It will default to your Program Files folder. If you wish to install it another location, you can change the location. We suggest keeping it in the default destination folder.
Press “Install”**



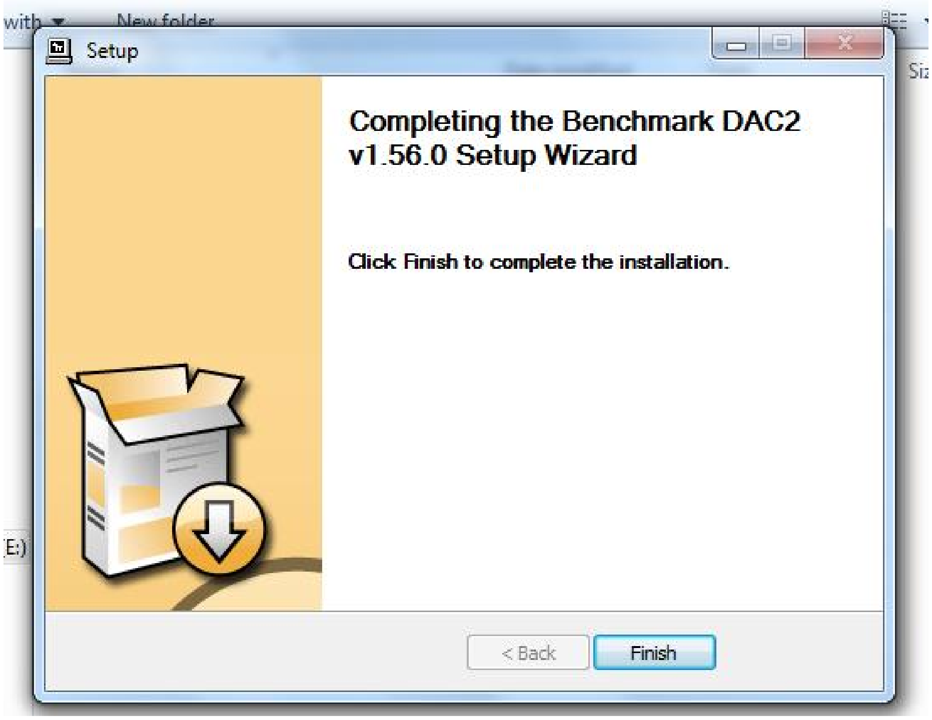
5. **When the installation begins you will see the following screen. Please be patient while the driver installs. Installation time is between 1-5 minutes.**



6. **Once the installation finishes a message at the top will say "Installation Complete." Press "Next" to continue.**

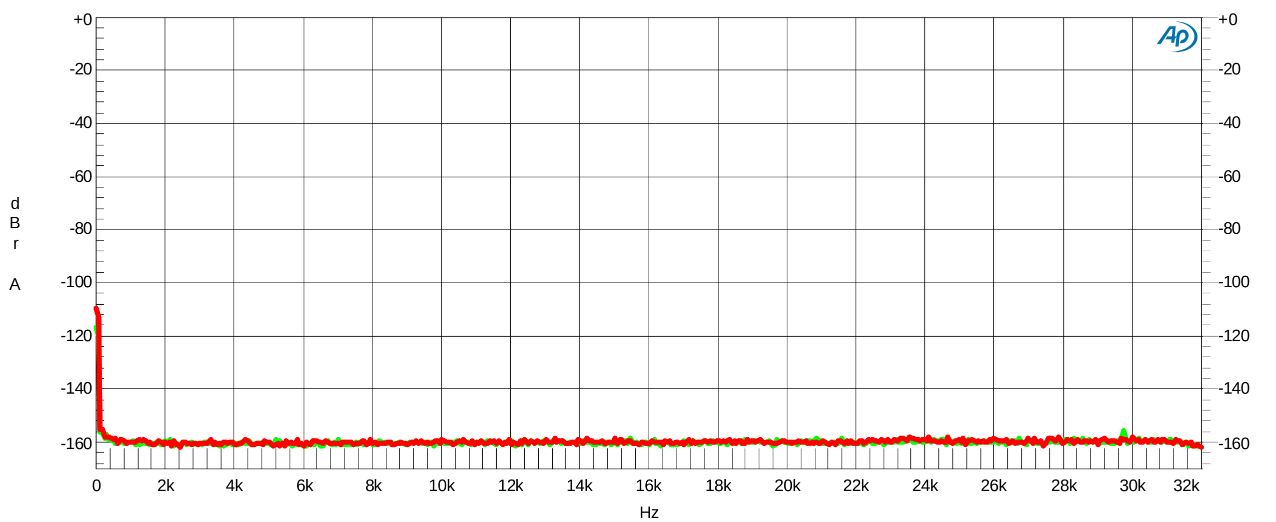


7. **Click "Finish." The Setup will close automatically and this completes the installation process. You can now enjoy music up to 192 kHz and DSD.**



Performance Graphs

Audio Precision FFT Idle Channel Noise, 0 dBr = 0 dBFS = 23 dBu



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right	

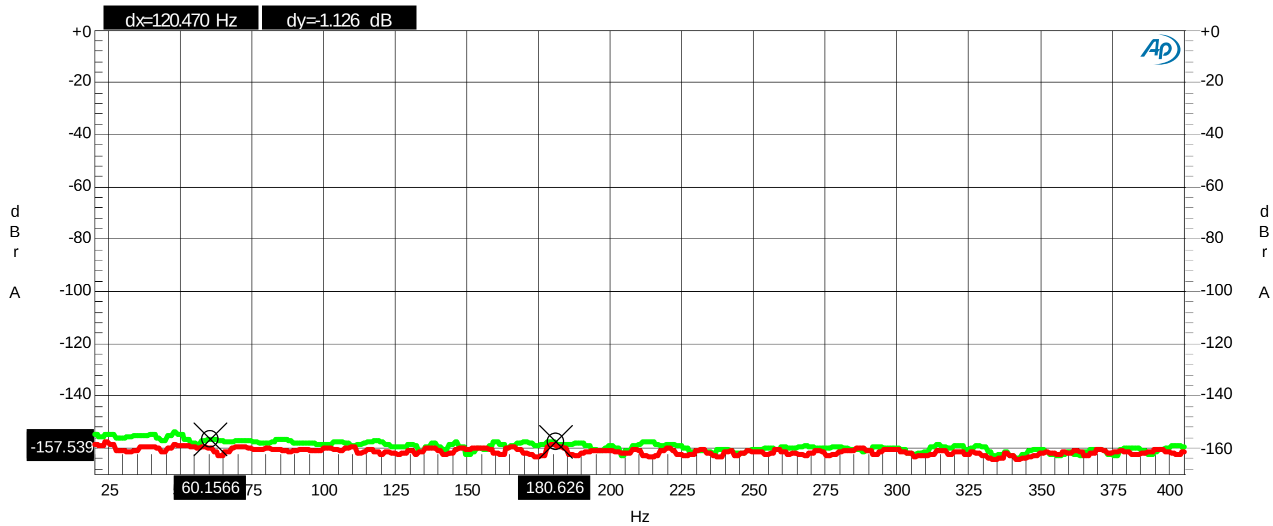
DAC2 - FFT Idle Channel Noise.at27

Graph Plot 1 – FFT Idle Channel Noise

The extraordinary performance of the DAC2 is demonstrated by the FFT plot shown above. There is no sign of any AC hum, there are no idle tones, and there are no spurious tones detected at a measurement limit of -160 dBFS.

The rise at 0 Hz is normal in an FFT analysis and is not an indication of noise. This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging, and spans a frequency range of DC to 32 kHz.

Audio Precision FFT Idle Channel Noise - Low Frequency, 0 dBr = 0 dBFS = 23 dBu



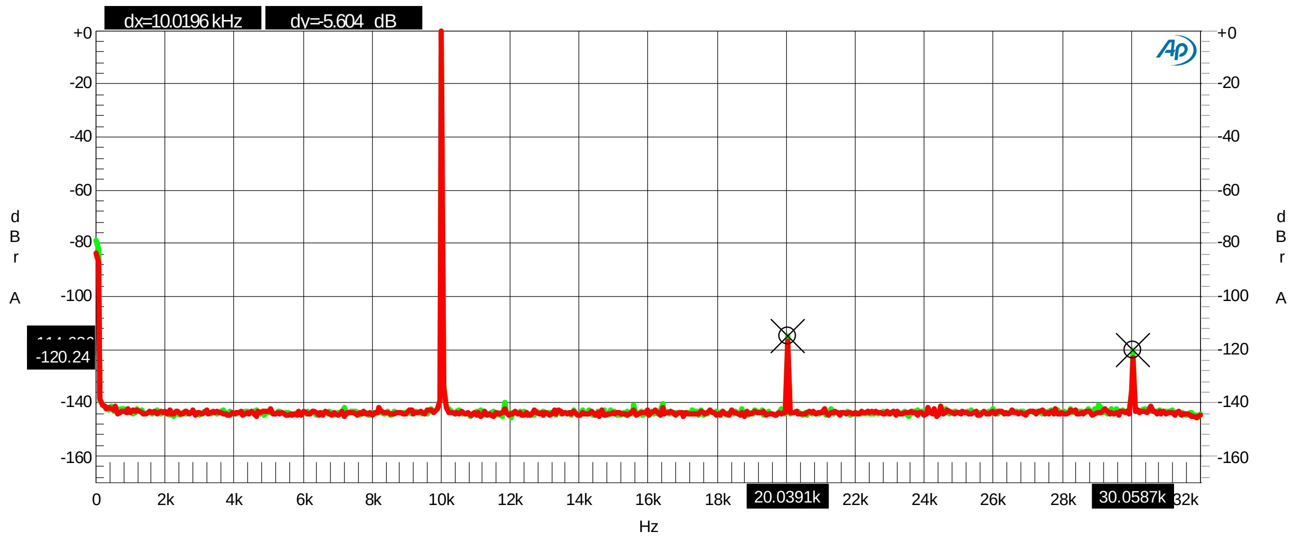
Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left		*-156.413 dBr	*-157.539 dBr
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right		-159.813 dBr A	-158.700 dBr A

DAC2 - FFT Idle Channel Noise - Low Frequency.at27

Graph Plot 2 - LOW FREQUENCY FFT - AC LINE-RELATED HUM

The DAC2 shows no evidence of AC line-related hum to a measurement limit of -160 dBFS. The cursors are placed at 60 Hz and 180 Hz (frequencies where we would expect to see interference from the 60 Hz AC input. There is absolutely no sign of any AC hum.

Audio Precision DAC2 - FFT 10 kHz, 0 dBr = 0 dBFS = 23 dBu



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left		*-114.636 dBr	*-120.240 dBr
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right		-116.765 dBr A	-122.816 dBr A

DAC2 - FFT 10 kHz.at27

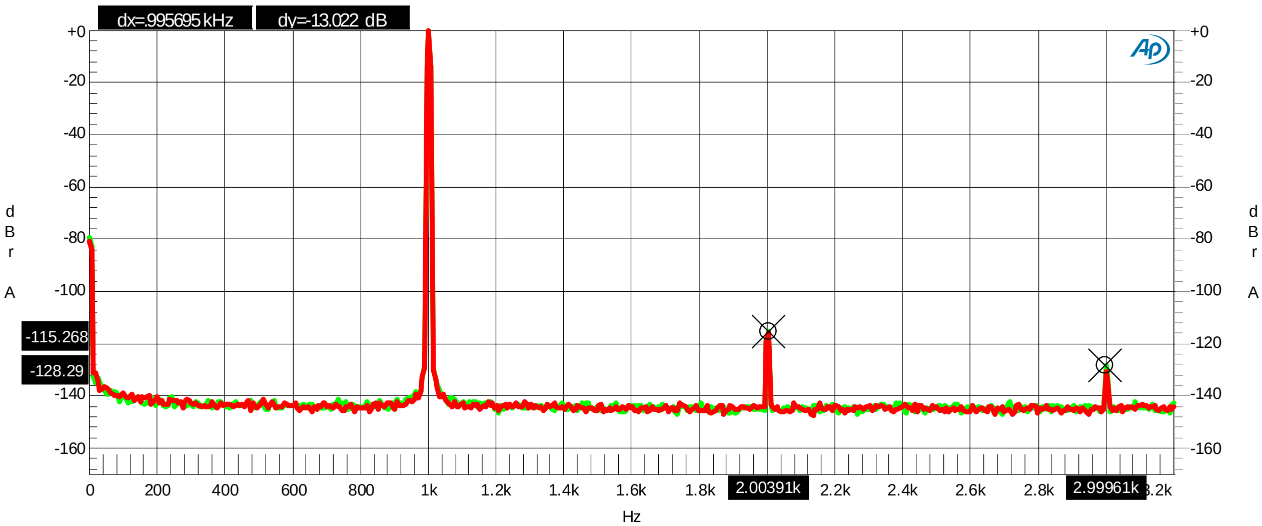
Graph Plot 3 - FFT 10 kHz

The 10 kHz FFT analysis is an excellent test for detecting sample clock jitter. Jitter will create sidebands (unwanted tones) above and below the 10 kHz test tone. A jitter frequency of 1 kHz would create two sideband tones; a lower sideband at 9 kHz, and an upper sideband at 11 kHz. Similarly, a jitter frequency of 2 kHz would produce sideband tones at 8 kHz and 12 kHz. The above plot shows no evidence of jitter-induced sidebands to a measurement limit of about -140 dBFS.

The -140 dB measurement limit is due to the SNR limitations of the Audio Precision 2700 test set, and not the DAC2. Note the very low harmonic distortion; -114 to -116 dB 2nd harmonic (20 kHz), and -120 to -122 dB 3rd harmonic (30 kHz). Please note that these are almost exactly the same harmonic distortion levels that occur with a 1 kHz tone (see ***Graph Plot 4 - FFT 1 kHz***). This demonstrates that the DAC2 analog stages have the high slew rates required to pass high-amplitude high-frequency signals without an increase in harmonic distortion.

The rise at 0 Hz is normal in an FFT analysis and is not an indication of noise. This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging, and spans a frequency range of DC to 32 kHz.

Audio Precision DAC2 - FFT 1 kHz, 0 dBr = 0 dBFS = 23 dBu



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left		*-115.268 dBr	*-128.290 dBr
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right		-115.954 dBr A	-130.429 dBr A

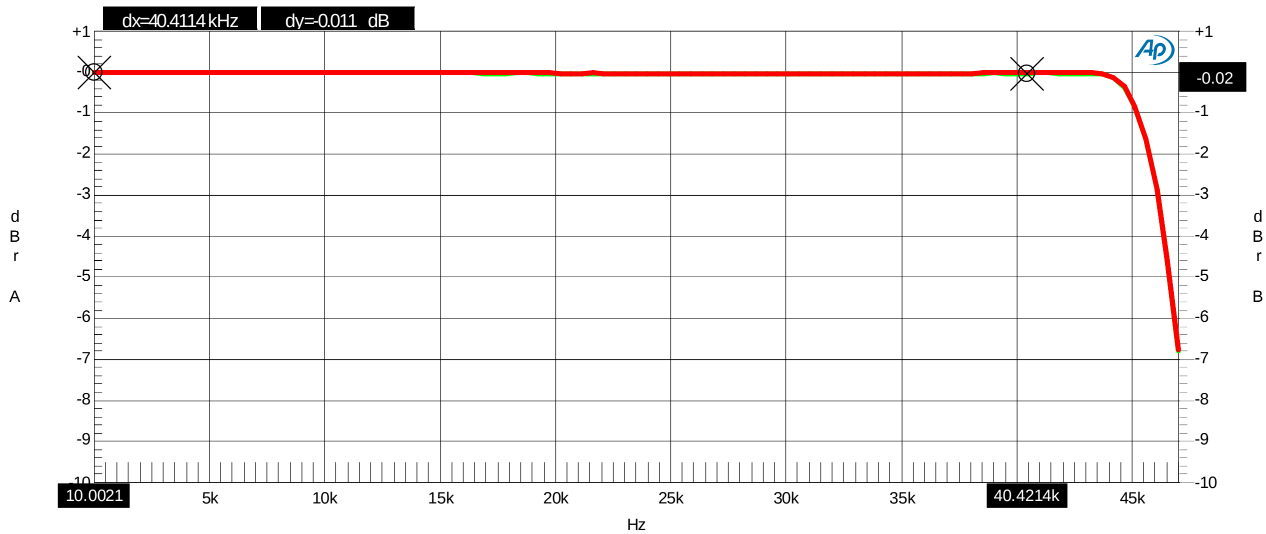
DAC2 - FFT 1 kHz.at27

Graph Plot 4 - FFT 1 kHz

The 1 kHz FFT analysis demonstrates the low harmonic distortion of the DAC2. Second harmonic distortion (2 kHz) measures about -115 dB, while 3rd harmonic distortion measures -128 to -130 dB.

The rise at 0 Hz is normal in an FFT analysis and is not an indication of noise. This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging, and spans a frequency range of DC to 32 kHz.

Audio Precision DAC2 - FREQUENCY RESPONSE 10/24/12 16:02:03
Fs = 96 kHz

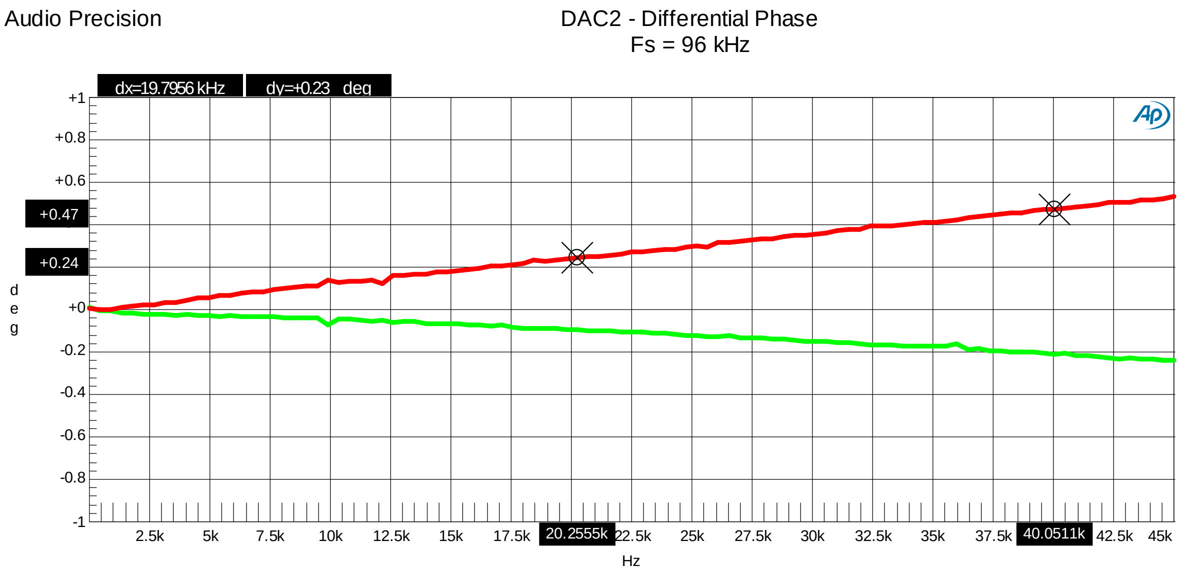


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Anlr.Level A	Left		-0.011 dBr A	-0.026 dBr A
1	2	Red	Solid	4	Anlr.Level B	Right		*-0.009 dBr	*-0.020 dBr

DAC2 - Frequency Response.at27

Graph Plot 5 - FREQUENCY RESPONSE

This plot demonstrates the ruler-flat frequency response of the DAC2. Note that the frequency response measures - 0.01 dB at 10 Hz and -0.02 dB at 40 kHz.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Anlr.Phase	Left		-0.09 deg	-0.21 deg
2	1	Red	Solid	4	Anlr.Phase	Left		*+0.24 deg	*+0.47 deg

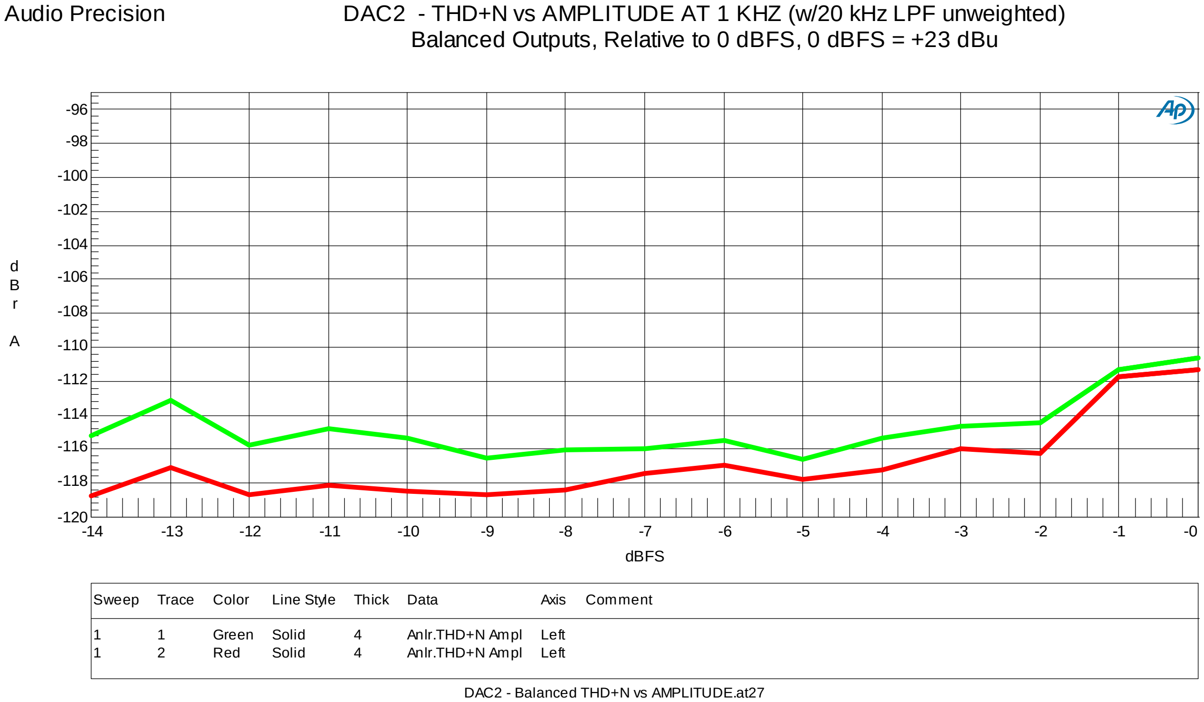
DAC2 - Differential Phase.at27

Graph Plot 6 - Differential Phase

This plot demonstrates the inter-channel phase accuracy of the DAC2. From this plot, the inter-channel phase accuracy is calculated to be +/- 0.17 degrees at 20 kHz, and +/- 0.34 degrees at 40 kHz.

The phase accuracy of the DAC2 is almost the same as the phase accuracy of the Audio Precision 2700 test set. For this reason, the phase error in the AP must be subtracted from the measurement. The green trace assigns the left channel to channel 1 of the AP, and the right channel to channel 2 of the AP. The red trace reverses the inputs. The two traces must be averaged to remove the phase errors of the AP test set.

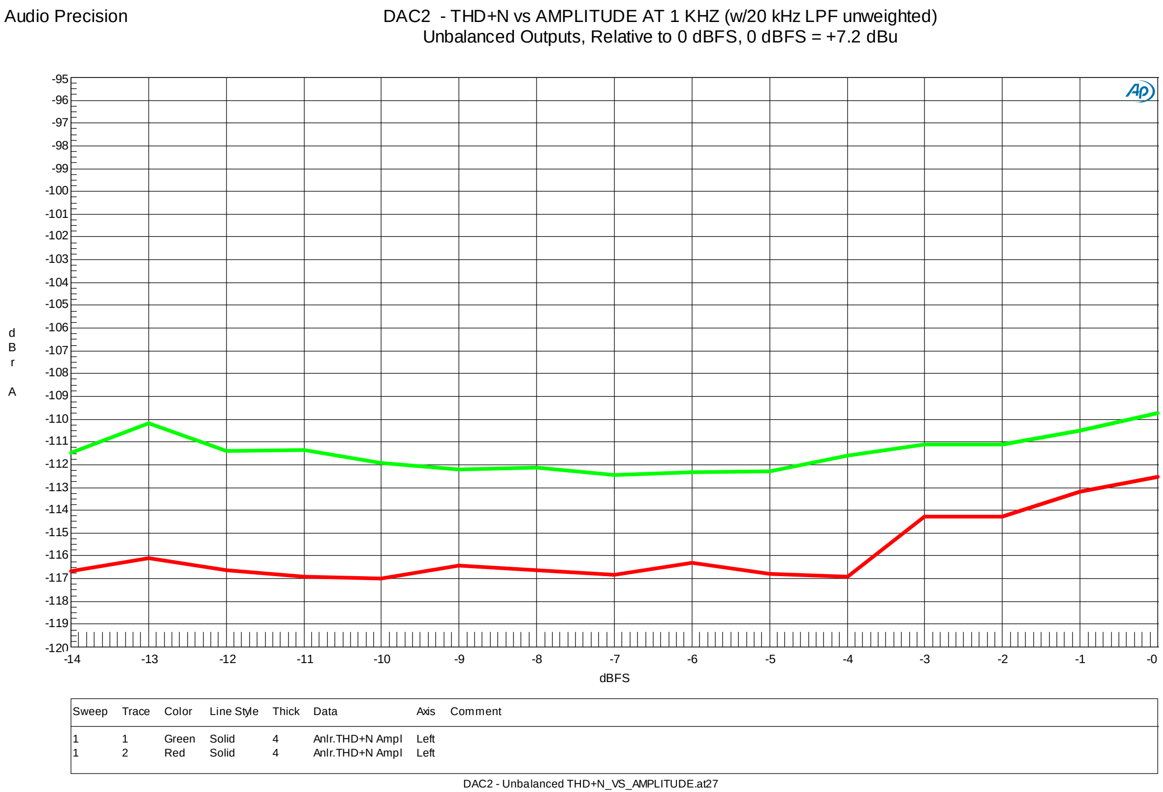
DAC2 converters are phase-accurate between boxes. Three DAC2 converters can be used as a group to form a phase-accurate 5.1 surround system. Four DAC2 converters can be combined to form a 7.1 system. The phase accuracy between any two channels will match the phase accuracy shown above.



Graph Plot 7 - Balanced THD+N vs. Amplitude

This plot demonstrates the very low harmonic distortion produced by the DAC2 at signal levels ranging from -14 dBFS to 0 dBFS. All variations below -4 dBFS are due to the measurement limits of the AP 2700 system. The peaks at -13 dBFS are due to an AP 2700 gain range change. In almost all listening environments, THD will be below the threshold of hearing. The DAC2 is virtually uncolored by any trace of harmonic distortion.

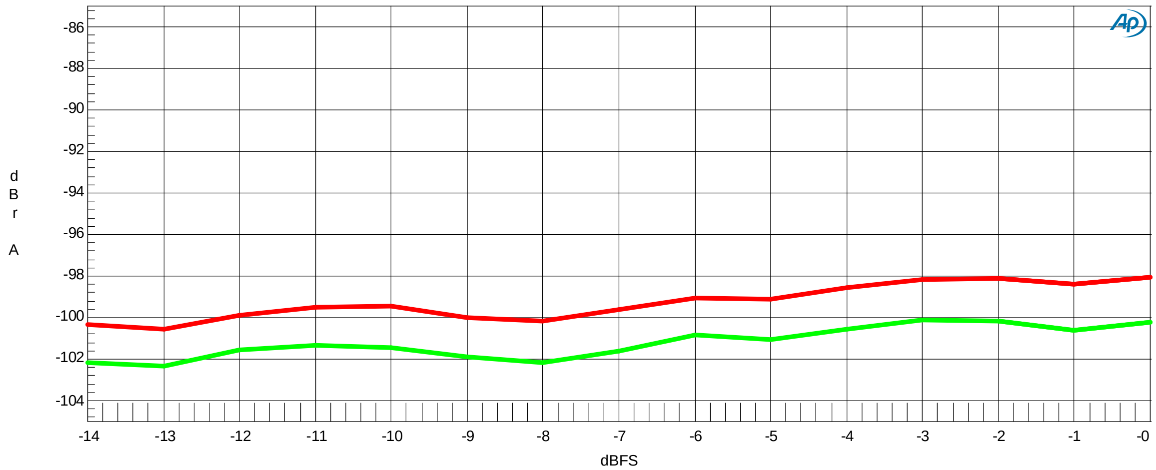
The DAC2 includes differential amplifiers that remove common-mode distortion components. This is important, because common-mode distortion is not detected when measuring with the precisely-balanced inputs on the AP 2700 test set. The THD+N measurements on the unbalanced outputs confirm the effectiveness of the differential amplifiers. Similar results are obtained by measuring either side of the balanced outputs relative to ground.



Graph Plot 8 - Unbalanced THD+N versus Amplitude

This plot shows the THD+N performance of the unbalanced outputs. Note that the THD+N performance of the unbalanced outputs approaches that of the balanced outputs. The DAC2 includes differential amplifiers that remove common-mode THD from the balanced outputs of the SABRE converters. These differential amplifiers give the unbalanced outputs the ability to approach the performance of the balanced outputs. Please note that the differential amplifiers also eliminate common-mode distortion on the balanced outputs.

Audio Precision DAC2 - THD+N vs AMPLITUDE AT 1 KHZ (w/20 kHz LPF unweighted)
Headphone Outputs, Relative to 0 dBFS, 0 dBFS = +7.2 dBu

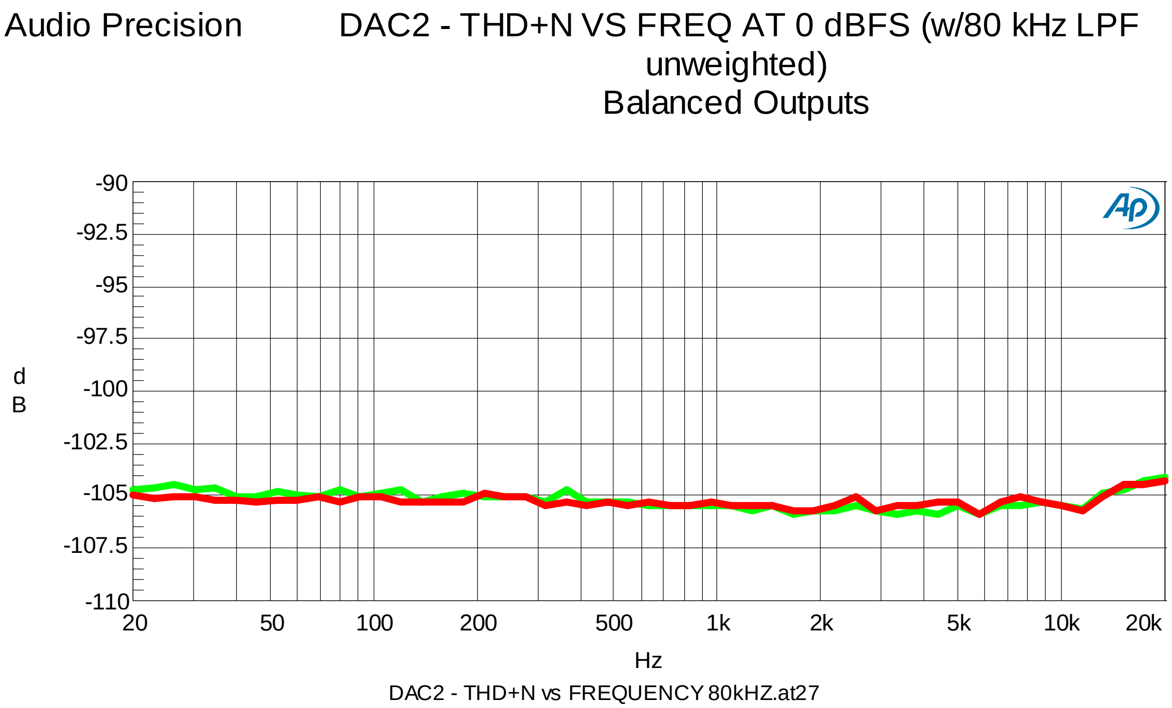


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Anlr.THD+N Ampl	Left	
1	2	Red	Solid	4	Anlr.THD+N Ampl	Left	

DAC2 - HEADPHONE - THD+N_VS_AMPLITUDE.at27

Graph Plot 9 - Headphone Amplifier – THD+N versus Amplitude (*DAC2 HGC and DAC2 D model only*)

This plot shows the THD+N performance of the headphone outputs under load. Note that the THD+N performance of the headphone outputs approaches that of the balanced outputs. The DAC2 includes Benchmark's HPA2(TM) headphone amplifier. The HPA2(TM) has a near 0-Ohm output impedance which provides outstanding control and damping of the headphone drivers. The HPA2(TM) has the voltage and current drive necessary to drive a wide variety of headphones.

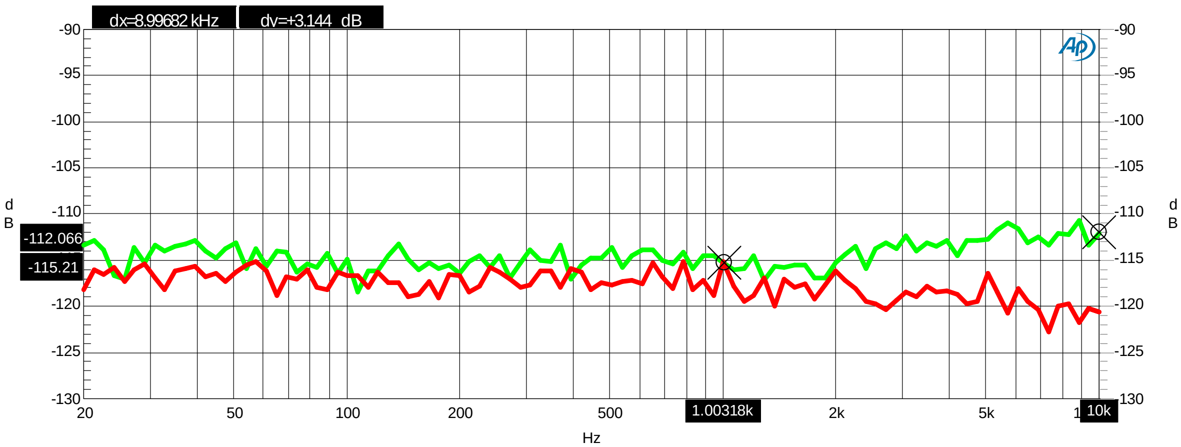


Graph Plot 10 - THD+N vs. FREQUENCY 80 kHz

The analog output stages on the DAC2 have high slew rates and are capable of maintaining low THD levels at high frequencies even when driven to 0 dBFS. Note that there is almost no rise in THD+N with frequency, even when operating at maximum output levels.

Audio Precision

DAC2 - THD VS FREQ AT 0 dBFS
Balanced Outputs



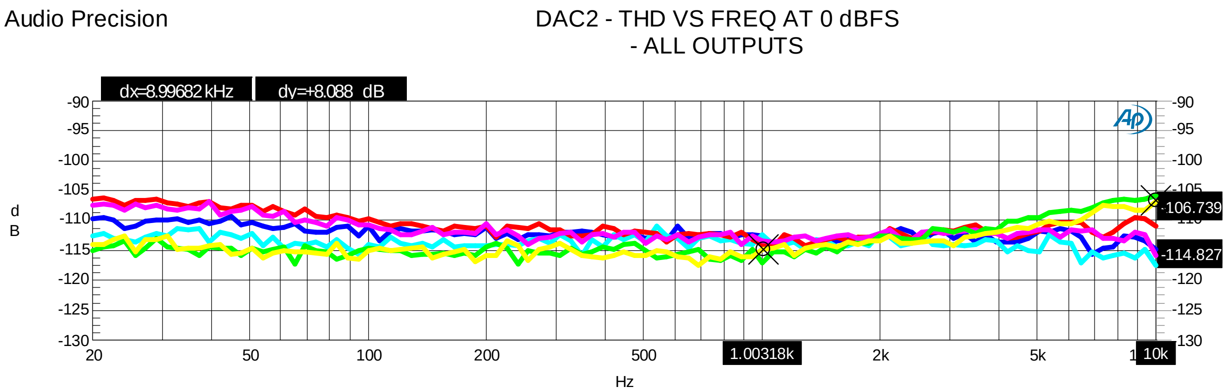
Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Distortion.Ch.1 Harm Sum1 Ratio	Left		*-115.210 dB	*-112.066 dB
1	2	Red	Solid	4	Distortion.Ch.1 Harm Sum2 Ratio	Right		-115.165 dB	-120.544 dB

GREEN = EVEN HARMONIC DISTORTION (2,4,6,8,10,12,14)
RED = ODD HARMONIC DISTORTION (3,5,7,9,11,13,15)

DAC2 - EVEN AND ODD THD vs FREQUENCY.at27

Graph Plot 11 - EVEN AND ODD THD vs. FREQUENCY

This plot demonstrates that the harmonic distortion of the DAC2 is lower than the THD+N numbers would suggest. This plot shows THD not THD+N. Even and odd harmonic distortions are plotted separately. Note that odd harmonics are lower than the more "musical" even harmonics. Both sets of harmonics are very low in amplitude (-112 dB to -120 dB), and should be entirely inaudible.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Source 2	Cursor1
1	2	Blue	Solid	4	Distortion.Ch.1 Harm Sum1 Ratio	Left		: 1.00000 =Swr.Ch. A+B Input	-112.557 dB
1	3	Cyan	Solid	4	Distortion.Ch.2 Harm Sum1 Ratio	Right		: 1.00000 =Swr.Ch. A+B Input	-112.453 dB
2	2	Red	Solid	4	Distortion.Ch.1 Harm Sum1 Ratio	Left		: 3.00000 =Swr.Ch. A+B Input	-113.919 dB
2	3	Magenta	Solid	4	Distortion.Ch.2 Harm Sum1 Ratio	Right		: 3.00000 =Swr.Ch. A+B Input	-113.349 dB
3	2	Green	Solid	4	Distortion.Ch.1 Harm Sum1 Ratio	Left		: 5.00000 =Swr.Ch. A+B Input	-117.017 dB
3	3	Yellow	Solid	4	Distortion.Ch.2 Harm Sum1 Ratio	Right		: 5.00000 =Swr.Ch. A+B Input	*-114.827 dB

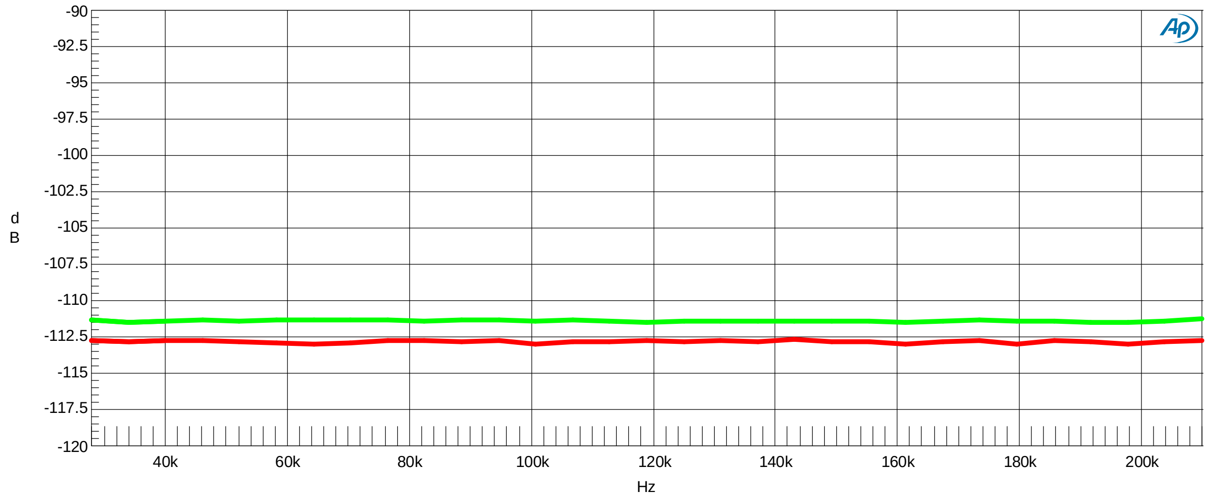
1=Balanced L
2=Balanced R
3=Unbalanced L
4=Unbalanced R
5=Headphone L
6=Headphone R

DAC2 - THD vs FREQUENCY.at27

Graph Plot 12 - THD vs. Frequency - All Outputs

This plot demonstrates that all of the analog outputs on the DAC2 have very similar performance over the entire audio band. The unbalanced outputs, and the headphone outputs, closely match the performance of the balanced outputs. Like ***Graph Plot 11 - EVEN AND ODD THD vs. FREQUENCY***, this plot shows THD (not THD+N).

Benchmark Media Systems, Inc. DAC2 - THD+N vs Sample Rate, 1 kHz at 0 dBFS (w/20 kHz LPF unweighted)
Balanced Outputs

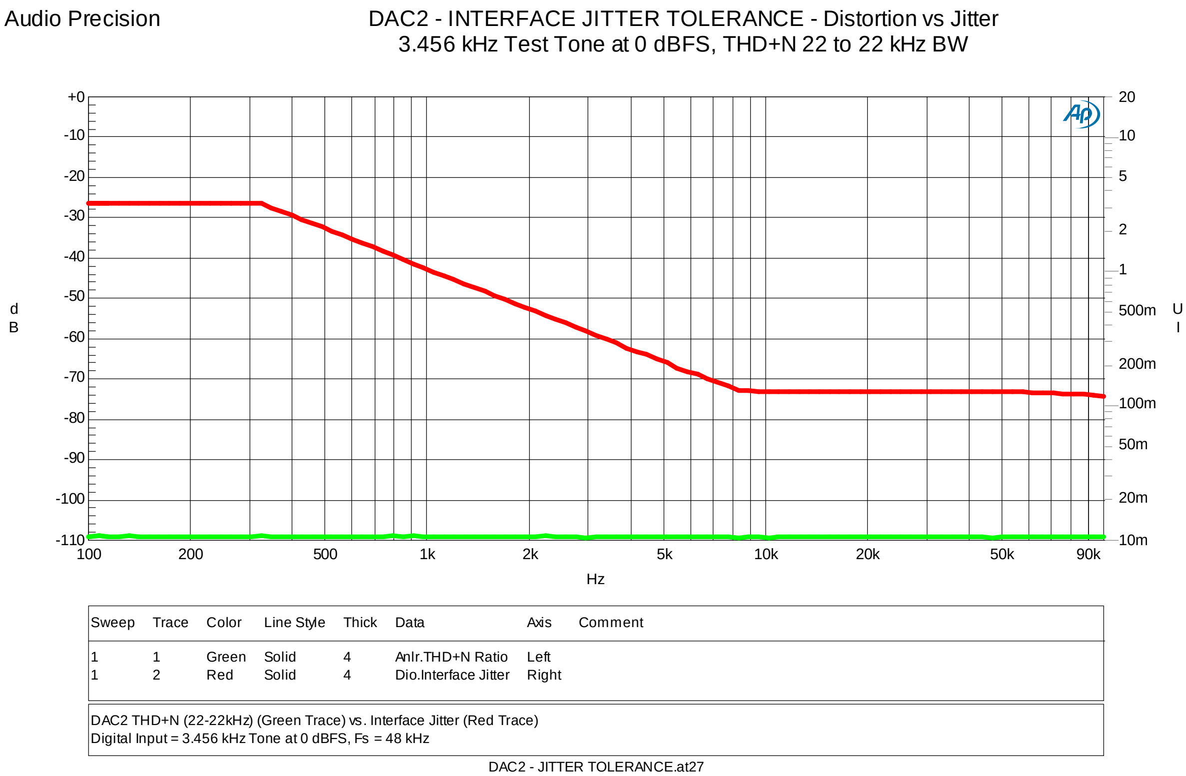


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Anlr.THD+N Ratio	Left	
1	2	Red	Solid	4	Anlr.THD+N Ratio	Left	

DAC2 - THD+N vs Sample Rate.at27

Graph Plot 13 - THD+N versus Sample Rate

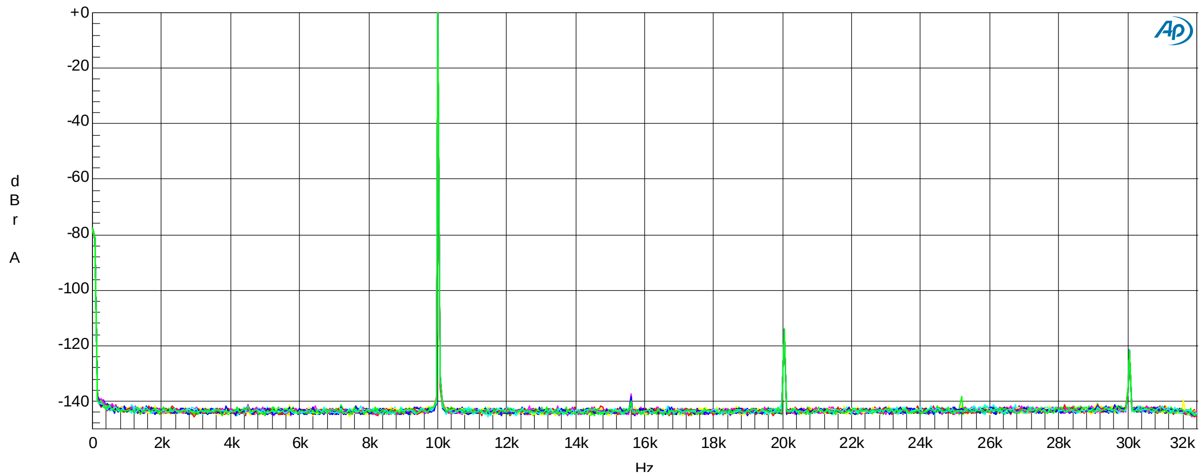
The THD+N performance of the DAC2 is identical at all Sample Rates.



Graph Plot 14 - JITTER TOLERANCE

The Audio Engineering Society (AES) has created a jitter tolerance template for testing digital audio devices. The red curve shows the AES jitter tolerance template. Induced jitter approaches 5 UI at 200 Hz, and is reduced to 0.125 UI above 8 kHz. The green trace shows the THD+N of the DAC2 while being driven with the jitter shown on the red curve. Over the entire range of the AES jitter tolerance test, the THD+N performance of the DAC2 is unchanged. The DAC2 easily passes the AES jitter tolerance test without any THD+N performance degradation.

Audio Precision DAC2 - INTERFACE JITTER TOLERANCE FFT
10 kHz Test Tone at 0 dBFS, AES Jitter Tolerance Sweep

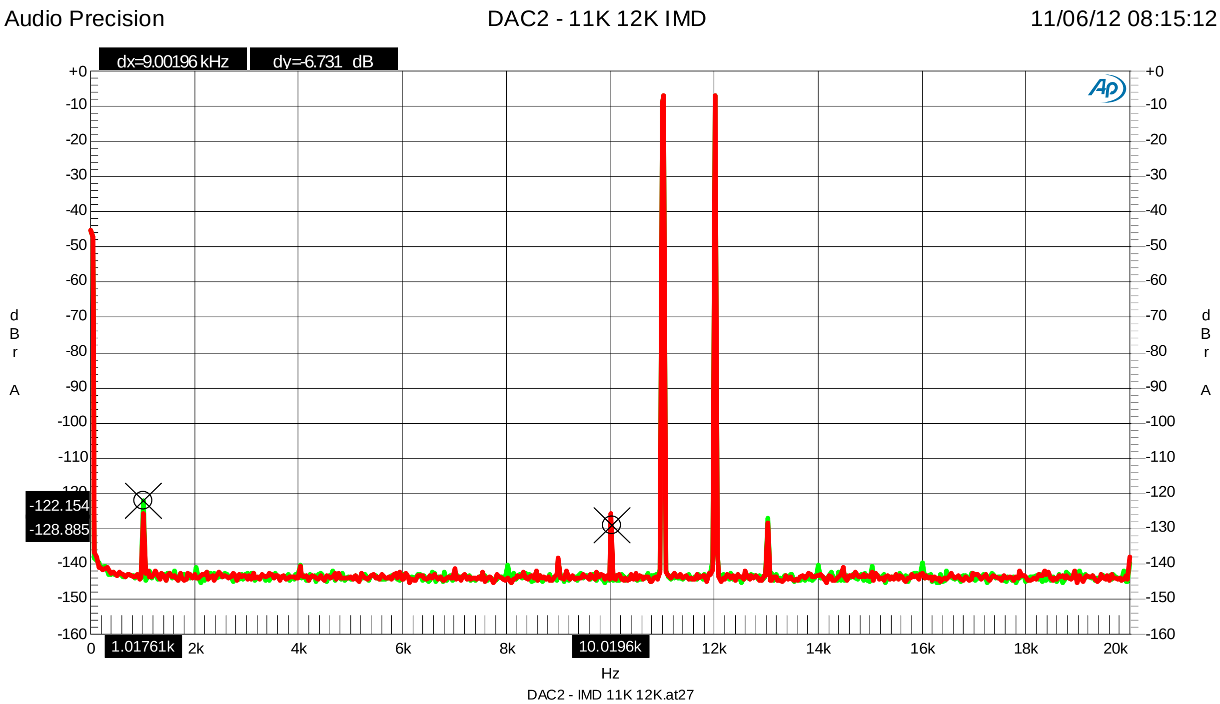


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Source 2
1	2	Green	Solid	4	Fft.Ch.1 Ampl	Left		: 100.000 Hz=Dio.Jitter Freq
2	2	Red	Solid	4	Fft.Ch.1 Ampl	Left		: 125.893 Hz=Dio.Jitter Freq
3	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 158.489 Hz=Dio.Jitter Freq
4	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 199.526 Hz=Dio.Jitter Freq
5	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 251.189 Hz=Dio.Jitter Freq
6	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 316.228 Hz=Dio.Jitter Freq
7	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 398.107 Hz=Dio.Jitter Freq
8	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 501.187 Hz=Dio.Jitter Freq
9	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 630.957 Hz=Dio.Jitter Freq
10	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 794.328 Hz=Dio.Jitter Freq
11	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 1.00000 kHz=Dio.Jitter Freq
12	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 1.25893 kHz=Dio.Jitter Freq
13	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 1.58489 kHz=Dio.Jitter Freq
14	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 1.99526 kHz=Dio.Jitter Freq
15	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 2.51189 kHz=Dio.Jitter Freq
16	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 3.16228 kHz=Dio.Jitter Freq
17	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 3.98107 kHz=Dio.Jitter Freq
18	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 5.01187 kHz=Dio.Jitter Freq
19	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 6.30957 kHz=Dio.Jitter Freq
20	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 7.94328 kHz=Dio.Jitter Freq
21	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 10.0000 kHz=Dio.Jitter Freq
22	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 12.5893 kHz=Dio.Jitter Freq
23	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 15.8489 kHz=Dio.Jitter Freq
24	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 19.9526 kHz=Dio.Jitter Freq
25	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 25.1189 kHz=Dio.Jitter Freq
26	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 31.6228 kHz=Dio.Jitter Freq
27	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 39.8107 kHz=Dio.Jitter Freq
28	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 50.1187 kHz=Dio.Jitter Freq
29	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 63.0957 kHz=Dio.Jitter Freq
30	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 79.4328 kHz=Dio.Jitter Freq
31	2	Green	Solid	4	Fft.Ch.1 Ampl	Left		: 100.000 kHz=Dio.Jitter Freq

DAC2 - JITTER TOLERANCE FFT.at27

Graph Plot 15 - JITTER TOLERANCE FFT

This figure shows an series of FFTs that were acquired while running the AES jitter tolerance test. Note that none of the 31 FFTs show any signs of jitter-induced sidebands. Note that the plots are identical to the plots shown in FIGURE 8. The DAC2 shows no change in performance when the AES jitter tolerance test is applied to the digital inputs. No jitter-induced sidebands are visible to a measurement limit that exceeds -140 dBFS.



Graph Plot 16 - IMD 11k 12K

This plot demonstrates that the DAC2 has very low IMD distortion. The 1 kHz difference frequency measures -122 dB, and the 10 kHz and 13 kHz products measure about -128 dB. IMD distortion should be well below audible levels.

Specifications

Audio Performance	
Fs = 44.1 to 96 kHz, 20 to 20 kHz BW, 1 kHz test tone, 0 dBFS = +24 dBu (unless noted)	
SNR – A-Weighted, 0 dBFS = +20 to +29 dBu	126 dB
SNR – Unweighted, 0 dBFS = +20 to +29 dBu	123 dB
THD+N, 1 kHz at 0 dBFS	-109 dBFS, -109 dB, 0.00035%
THD+N, 1 kHz at -1 dBFS	-110 dBFS, -109 dB, 0.00035%
THD+N, 1 kHz at –3 dBFS	-113 dBFS, -109 dB, 0.00035%
THD+N, 20 to 20 kHz test tone at –3 dBFS	-112 dBFS, -108 dB, 0.00040%
Frequency Response at Fs=96 kHz	+0 dB, -0.04 dB (20 to 20 kHz) -0.04 dB at 10 Hz -0.04 dB at 20 kHz -0.04 dB at 40 kHz -0.7 dB at 45 kHz
Frequency Response at Fs=48 kHz	+0 dB, -0.04 dB (20 to 20 kHz) -0.04 dB at 10 Hz -0.04 dB at 20 kHz
Crosstalk	-116 dB at 20 kHz -130 dB at 1 kHz -137 dB at 20 Hz
Maximum Amplitude of Jitter Induced Sidebands (10 kHz 0 dBFS test tone, 12.75 UI sinusoidal jitter at 1 kHz)	< -144 dB
Maximum Amplitude of Spurious Tones with 0 dBFS test signal	< -138 dB
Maximum Amplitude of Idle Tones	< -147 dB
Maximum Amplitude of AC line related Hum & Noise	< -140 dB
Inter-channel Differential Phase (Stereo Pair – any sample rate)	+/- 0.25 degrees at 20 kHz
Inter-channel Differential Phase (Between ***DAC2*** Units Fs<110 kHz) Any sample rate.	+/- 0.25 degrees at 20 kHz
Maximum Lock Time after Fs change	400 ms
Soft Mute Ramp Up/Down Time	50 ms
Mute on Receive Error	Yes
Mute on Lock Error	Yes
Mute on Idle Channel	No
50/15 us De-Emphasis Enable	Automatic in Consumer Mode
De-Emphasis Method	Digital IIR
De-Emphasis Supported at	Fs = 32, 44.1, 48 kHz

Group Delay (Latency)	
Delay – Digital Input to Analog Output (function of sample rate)	1.36 ms at 44.1 kHz 1.27 ms at 48 kHz 0.90 ms at 88.2 kHz 0.82 ms at 96 kHz 0.51 ms at 176.4 kHz 0.47 ms at 192 kHz

Digital Audio Inputs	
Number of Digital Inputs (switch selected)	5 (1 USB, 2 Optical, 2 Coaxial)
Number of Channels	2
Input Sample Frequency Range	28 to 210 kHz (Coaxial Inputs) 28 to 96 kHz (Optical Inputs) 44.1, 48, 88.2, 96, 176.4, 192 kHz (USB Input)
Maximum Input Word Length	24 bits
Digital Input Impedance	75 Ohms (Coaxial Inputs)
DC Blocking Capacitors on Digital Inputs	Yes (Coaxial Inputs)
Transient and Over-Voltage Protection on Digital Inputs	Yes (Coaxial Inputs)
Minimum Digital Input Level	250 mVpp (Coaxial Inputs)
Jitter Tolerance (With no Measurable Change in Performance)	>12.75 UI sine, 100 Hz to 3 kHz >1.5 UI sine at 20 kHz >1.5 UI sine at 40 kHz >1.5 UI sine at 80 kHz >1.5 UI sine at 90 kHz >0.25 UI sine above 160 kHz
Jitter Attenuation Method	Benchmark ***UltraLock2™*** - all inputs

Balanced Analog Outputs	
Number of Balanced Analog Outputs	2
Output Connector	Gold-Pin Neutrik™ male XLR
Output Impedance	60 Ohms (Attenuator off) 425 Ohms (Attenuator = 10 dB) 135 Ohms (Attenuator = 20 dB)
Analog Output Clip Point	+30 dBu
Factory Set Bypass Level (at 0 dBFS)	+13 dBu (Attenuator = 10 dB)
Output Level Range (at 0 dBFS) In 'Variable' Mode	Off to +23 dBu (Attenuator off) Off to +13 dBu (Attenuator = 10 dB) Off to +3 dBu (Attenuator = 20 dB)
Output Level Variation with Sample Rate (44.1 kHz vs. 96 kHz)	< +/- 0.006 dB

Unbalanced Analog Outputs	
Number of Unbalanced Analog Outputs	4
Output Connector	RCA
Output Impedance	30 Ohms
Analog Output Clip Point	+13.5 dBu
Factory Set Home Theater Bypass Output Level (at 0 dBFS)	+8.2 dBu (2 Vrms)
Output Level Range (at 0 dBFS)	Off to +8.2 dBu
Output Level Variation with Sample Rate (44.1 kHz vs. 96 kHz)	< +/- 0.006 dB

HPA2™ Headphone Outputs	
Number of Headphone Outputs	2
Output Connectors	¼" TRS with switch on left-hand jack
Output Impedance	< 0.11 Ohms
Output Level Control	Stereo Control on Front Panel
Output Level Range (at 0 dBFS) into 60-Ohm Load	Off to +17 dBu
Maximum Output Current	250 mA
Overload Protection (independent per channel)	Current limited at 300 mA, Thermal
Bandwidth	> 500 kHz
THD+N	-106 dB, 0.0005% into 30 Ohms at +18 dBu (1.26W)

Status Display	
Indicators - Type and Location	16 LED’s on Front Panel
Selection/Status Indication	1 – Dim/Mute 1 - Polarity 6 – Input 2 – Bypass/Calibrated Output 2 – Word length 4 – Sample Rate

AC Power Requirements	
Nominal Input Operating Voltage Range (VAC RMS)	100 – 240V
Frequency	50-60 Hz
Power	< 0.5 Watts Idle 12 Watts Typical Program 15 Watts Maximum
Fuses	5 x 20 mm (2 required) 0.5 A 250 V Slo-Blo® Type
Min/Max Operating range (VAC RMS)	90 – 260 47 – 63Hz

Dimensions	
Form Factor	½ Rack Wide, 1 RU High
Depth behind front panel	8.5” (216 mm)
Overall depth including connectors but without power cord	9.33” (237 mm)
Width	9.5” (249 mm)
Height	1.725” (44.5 mm)

Weight	
DAC2 only	3 lb.
DAC2 with remote control, power cord, extra fuses, and manual	4 lb.
Shipping weight	7 lb.

Regulatory Compliance

FCC and RoHS Compliance Statements

FCC Notice (U.S. Only)

NOTICE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received including interference that may cause undesired operation.

Instructions to Users: This equipment complies with the requirements of FCC (Federal Communication Commission) equipment provided that following conditions are met:

- RCA Digital Connections: Shielded 75-Ohm coaxial cable must be used.

NOTICE: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

RoHS Compliant Information

This statement clarifies Benchmark Media Systems, Inc. product compliance with the *EU*'s (European Union) directive 2002/95/EC, or, *RoHS* (Restrictions of Hazardous Substances).

As of July 01, 2006, All Benchmark Media Systems, Inc. products placed on the European Union market are *compliant* (containing quantity limit weight less than or equal to 0.1% (1000 ppm) of any homogeneous Lead (Pb), Mercury (Hg), Hexavalent Chromium (Cr VI), and flame retardant Polybrominated Biphenyls (PBB) or Polybrominated Diphenyl Ethers (PBDE)).

CE Certificates of Conformity

Certificate of Conformity

Diversified T.E.S.T. Technologies, Inc. has tested the product to the current appropriate standards and finds that the product is in compliance with those requirements.

EMC Directive:	**2004/108/EC**	
Generic Emissions Standard:	EN 61000-6-3: 2007/A1:2011	
Product Specific Emissions:	EN 55011 Class A	
Generic Immunity Standard:	EN 61000-6-1: 2007	
Immunity:	EN 61000-4-2	Electrostatic Discharge
	EN 61000-4-3	Radiated Susceptibility
	EN 61000-4-6	Conducted Susceptibility

Manufacturer's Name:	**Benchmark Media Systems**
Manufacturer's Address:	203 East Hampton Suite 2
	Syracuse, NY 13206
Product:	DAC2HGC
Model Number:	500-14800-XXX *

* Where XXX indicates a color code.

This Certificate of Compliance issued September 21, 2012 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature: ____

Annelle Frierson
Vice President
Diversified T.E.S.T. Technologies, Inc.
4675 Burr Drive
Liverpool, NY 13088
Phone: 315-457-0245
Fax: 315-457-0428

DIVERSIFIED T.E.S.T. TECHNOLOGIES, INC.
CORPORATE
SEAL
1991
NEW YORK

Warranty Information

Benchmark 1-Year Warranty

The Benchmark 1-Year Warranty

Benchmark Media Systems, Inc. warrants its products to be free from defects in material and workmanship under normal use and service for a period **of one year from the date of delivery.**

This warranty extends only to the original purchaser. This warranty does not apply to fuses, lamps, batteries, or any products or parts that have been subjected to misuse, neglect, accident, modification, or abnormal operating conditions.

In the event of failure of a product under this warranty, Benchmark Media Systems, Inc. will repair, at no charge, the product returned to its factory. Benchmark Media Systems, Inc. may, at its option, replace the product in lieu of repair. If the failure has been caused by misuse, neglect, accident, or, abnormal operating conditions, repairs will be billed at the normal shop rate. In such cases, an estimate will be submitted before work is started, if requested by the customer.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for export.

The foregoing warranty is in lieu of all other warranties, expressed or implied, including but not limited to any implied warranty of merchantability, fitness or adequacy for any particular purpose or use. Benchmark Media Systems, Inc. shall not be liable for any special, incidental, or consequential damages, and reserves the right to change this information without notice. This limited warranty gives the consumer-owner specific legal rights, and there may also be other rights that vary from state to state.

Benchmark Extended Warranty Options

The Benchmark Extended 5-Year Warranty *

Benchmark Media Systems, Inc. optionally extends the standard 1-year warranty to a period of **five years from the date of delivery.**

*For the extended warranty to become effective, the original purchaser must register the product at the time of purchase either by way of the enclosed registration card or through the product registration section of the Benchmark Media Systems, Inc. website. This optional warranty applies only to products purchased within the US and Canada and is extended only to the original purchaser.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for export. The terms of the extended warranty are subject to change without notice. For products purchased outside the US and Canada, please refer to the Extended Two 2-Year International Warranty.

The Benchmark Extended 2-Year International Warranty **

Benchmark Media Systems, Inc. optionally extends the standard 1-year warranty to a period of **two years from the date of delivery.**

**For the extended warranty to become effective, the original purchaser must register the product at the time of purchase either by way of the enclosed registration card or through the product registration section of the Benchmark Media Systems, Inc. website. This optional warranty applies only to products purchased outside the US and Canada and is extended only to the original purchaser.

Attempts to deliberately deface, mutilate, or remove the product's label will render this warranty void. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for export. The terms of the extended warranty are subject to change without notice. For products purchased in within the US and Canada, please refer to the Extended Five 5-Year Warranty.

Notes on Warranty Repairs

An RMA (return merchandise authorization) number, issued by our Customer Service Department, is required when sending products for repair.

They must be shipped to Benchmark Media Systems prepaid and preferably in their original shipping carton with the RMA number clearly visible on the exterior of the packaging. A letter should be included giving full details of the difficulty.

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