...the measure of excellence!™

Benchmark *DAC2 DX*
Instruction Manual

Reference Stereo D/A Converter
Native PCM and DSD D/A Conversion
Headphone Amplifier
Asynchronous USB
Dual Output Buses
ESS9018 Conversion System

(Version 2.X Firmware)

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.

AC Input Voltage Range

NOTE: 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.

CAUTION: 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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- Tested for immunity to radiated and conducted RF interference

Introduction

Applications

The ***DAC2 DX*** is a professional reference-grade audio digital to analog converter with Benchmark's ***HPA2™*** headphone amplifier. The ***DAC2 DX*** supports 24-bit D/A conversion of PCM at sample rates up to 192 kHz. It also supports direct conversion of 1-bit DSD at a 2.8224 MHz sample rate. It is designed to be very transparent and this makes it well-suited for critical monitoring in studio control rooms and mastering rooms.

The ***DAC2 DX*** is also well-suited for high-end hi-fi environments. It includes a generous collection of inputs and outputs and can serve as the central component in any stereo hi-fi system where all inputs are digital. The ***DAC2 DX*** provides D/A conversion, source selection, volume control, and headphone amplification. A remote control, 12V trigger, and volume control bypass function provide the features needed in a home environment.

The ***DAC2 DX*** is designed to directly drive a wide variety of power amplifiers and powered monitors. The balanced outputs include low-impedance passive pads that can be adjusted to optimize the gain staging between the ***DAC2 DX*** and the power amplifier. This gain optimization can provide very substantial improvements in the system-level SNR and THD+N performance.

DAC2 vs. DAC1

The ***DAC2*** adds these features that are not found on the ***DAC1***:

- Asynchronous 192kHz USB Audio 2.0
- 32-bit D/A conversion system
- Word Length Display
- Sample Rate Display
- Polarity Control
- Direct DSD D/A Conversion
- -20 dB DIM
- Bi-Directional 12V Trigger
- Power Switch with Auto-On Function
- Volume Control Bypass
- Digital Pass-Through
- High-Headroom DSP
- Dual-Domain Hybrid Gain Control
- Additional I/O

DAC2 Technologies

Parallel Conversion Structure

The conversion system in the ***DAC2 DX*** achieves a 4.8 dB signal to noise improvement through the use of 3:1 summing on the main outputs. The ES9018 D/A is an 8-channel 32-bit converter. In the ***DAC2 DX***, three channels are summed in the analog domain to form the main outputs. The remaining two channels provide the auxiliary outputs.

The 3:1 summing also improves the THD. The non-linearities in individual conversion channels are averaged across the four summed channels and incoherent non-linearities are attenuated by almost 4.8 dB.

High-Headroom Digital and Analog Processing

The ***DAC2 DX*** has generous amounts of analog and digital headroom. The analog clip point is above 29 dBu. The digital clip point is 28 dBu. When operating at a typical -20 dB at +4 dBu studio calibration, the ***DAC2 DX*** has 4 dB of digital headroom above 0 dBFS. This digital headroom prevents the clipping of intersample overs.

No Clipping of Intersample Overs

The ***DAC2*** is one of very few D/A converters that can accurately reproduce intersample overs without clipping. Intersample peaks can reach +3.01 dBFS and commonly occur many times per second in most 44.1 kHz and 48 kHz recordings. When recordings are ripped using lossy compression systems (such as MP3), additional intersample overs are often created. Most converters (including the ***DAC1***) produce bursts of distortion at every occurrence of an intersample over. In contrast, the ***DAC2*** and ***DAC3*** converters cleanly reproduce all intersample overs.

Low-Noise Power Supplies

The ***DAC2 DX*** uses high-efficiency low-noise power supplies. Each critical subsystem also has at least one dedicated low-noise regulator. The high-efficiency supplies deliver the substantial power required by the low-impedance circuits, the headphone amplifier, and the output line drivers. A power switch is included. The standby power consumption is less than 0.5 W when the unit is off.

Low Magnetic Emissions

The magnetic components in the ***DAC2 DX*** power supplies operate at over 800 kHz. This allows the use of very small magnetic components that emit correspondingly small magnetic fields. This virtually eliminates all traces of line-frequency components in the output spectrum of the ***DAC2 DX***. This also means that the ***DAC2 DX*** can be placed in close proximity to any audio component without causing interference with the other component.

UltraLock2™ Clock System

UltraLock2™ provides the outstanding jitter attenuation of Benchmark's ***UltraLock™*** system while providing an improved SNR.

Dual-Mode USB Input

The ***DAC2 DX*** has a USB input that can be operated in two modes; driverless **USB Audio 1.1**, and a high sample rate **USB Audio 2.0**. Both use asynchronous clocking to eliminate the USB interface as a source of clock jitter.

Note: To provide full backward and forward compatibility, the ***DAC2 DX*** uses the ***DAC2*** USB drivers. This prevents the need to install two different sets of drivers. Please note that the ***DAC2 DX*** USB input will be identified as "Benchmark DAC2" in your computer control panels. This is intentional.

Asynchronous USB Audio 2.0

The USB Audio 2.0 interface supports DSD and 192 kHz, 24-bit PCM. No drivers are required for Apple operating systems. Drivers are provided for older Windows operating systems at: BenchmarkMedia.com/drivers

Native Asynchronous USB 1.1

The ***DAC2 DX*** has a driverless USB Audio 1.1 mode that supports 96 kHz, 24-bit PCM with all operating systems. This mode provides a quick and easy connection to a wide variety of computers and tablets without installing a driver.

32-bit Digital Gain Control

The ***DAC2 DX*** uses the digital section of Benchmark's dual-domain ***HGC™*** system (used in the ***DAC2 HGC***).

Benchmark’s unique motor-driven volume control sets the gain of a 32-bit dithered digital gain control. The 32-bit digital output feeds the 32-bit D/A conversion system.

The 32-bit digital gain control delivers low distortion, accuracy, and precise left-right gain matching. The noise-free 32-bit dithered system preserves musical details over a very wide range of output levels.

The XLR outputs leverage the low-impedance passive analog attenuation system. When properly configured, the entire dynamic range of the ***DAC2 DX*** can be lined up with the dynamic range of the power amplifier. This matching can provide a dramatic improvement in the system-level signal to noise ratio.

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 with a physical knob. The potentiometer produces a DC voltage that controls the gain of a dithered 32-bit multiplier. The outputs of the multiplier drive the 32-bit D/A converters.

Low-Impedance Passive Attenuators

Like the ***DAC1*** and ***DAC2*** includes low-impedance passive attenuators on the XLR outputs (**Page 29**).

These attenuators can be adjusted in 10 dB steps to optimize the interface with the power amplifier or powered monitors. This optimization places the volume control in its best operating range. This exclusive Benchmark feature can provide substantial improvements in the overall performance of the playback signal chain.

Native DSD Conversion

The ***DAC2 DX*** supports native DSD conversion. This feature was not available on the ***DAC1***. 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. Three balanced 1-bit converters are summed together for each of the **MAIN** outputs.

Digital Pass-Through

The second coaxial input (**D5**) can be reconfigured as a digital output (**Page 32**).

When **D5** is configured as an output, any selected digital input is passed through to **D5** without any processing. Optical, XLR, coaxial, and USB inputs can be passed through to the **D5** connector. PCM and DoP formatted DSD can both be passed through the **D5** connector while also being sent to the D/A converter. The pass-through function even works with special signals such as DTS, Dolby Digital, even though these signals cannot be decoded by the ***DAC2 DX***.

Volume Control Bypass

The **CALIBRATED** mode can be activated for either or both output buses (**Page 19**).

The factory default calibration at 0 dBFS is +24 dBu on the XLR outputs (pads at 0 dB) and 2 Vrms on the RCA outputs. If your studio calibration is different, the calibration can be adjusted in 1 dB increments from +20 dBu to +28 dBu using the removable jumpers on connector **P6** (**Page 33**).

The **M** and/or **A** lights will be on when the **MAIN** and/or **AUXILIARY** outputs are in the **CALIBRATED** mode. A slow flashing light indicates that a calibrated output is muted. The **DIM** function is disabled on any output group that is operating in **CALIBRATED** mode.

When the **CALIBRATED** mode is off, the **M** and/or **A** lights will flash rapidly when the volume of the **MAIN** and/or **AUXILIARY** outputs are being adjusted.

The **CALIBRATED** mode is similar to the **CALIBRATED** switch setting on the ***DAC1*** except that the ***DAC2 DX*** system is much more flexible. The ***DAC2 DX*** has two independent output buses that can be programmed differently. In addition the settings for these buses are individually programmable for each digital input on the ***DAC2 DX***. This flexibility has many applications in studio and home environments.

Relay-Muted Analog Outputs

The XLR and RCA analog outputs are equipped with mute relays that keep the outputs muted while powering on or off. These relays eliminate pops and clicks at the unit power up or down.

Bi-directional 12V Trigger

The 12 Volt trigger can be connected to other audio components so that an entire audio system can turn on and off in a sequenced fashion (**Page 17**).

The ***DAC2 DX*** trigger I/O can be connected to a preamplifier, power amplifier, or both.

The ***DAC2 DX*** will pull the trigger I/O to 12 volts DC while the ***DAC2 DX*** is on. If the ***DAC2 DX*** is off and an external device pulls the trigger I/O to 12 volts, the ***DAC2 DX*** will turn on.

Auto-On Function

The ***DAC2 DX*** can be programmed to automatically turn on when AC power is applied (**Page 16**).

Preamp COMPATIBILITY Mode

This feature was added in Version 2.0. If you have an older version, an update is available.

The **COMPATIBILITY** mode disables the volume control when the ***DAC2*** feeds a preamplifier that will be used to control the system playback volume. When this mode is used with a Benchmark preamplifier or line amplifier, such as the ***HPA4***, a single Benchmark IR remote can be used to control both devices (**Page 20**).

Front Panel

Rear Panel

Quick Start Guide

Audio Inputs

The ***DAC2 DX*** features six stereo digital inputs (1 AES XLR, 2 coaxial, 2 optical, and 1 USB). The XLR, coaxial and optical inputs accept professional (AES), consumer (S/PDIF) and DoP DSD data formats.

Tip: We recommend using the coaxial or USB inputs for DSD and for PCM sample rates above 96 kHz. Optical interfaces are rated for 96 kHz data rates and may not be reliable for DSD or sample rates above 96 kHz.

Remote Control

The remote control is designed to have a long operating range. In most applications it is not necessary to point the remote directly at the ***DAC2 DX***.

The chart at the right summarizes the functions of the IR remote control.

OFF	Turns the unit off. Any devices slaved to the **12V TRIGGER** will also turn off in a controlled sequence. Press and hold the **OFF** button for 3 seconds to force the **12V TRIGGER** off (only necessary when another device is acting as a **TRIGGER BUS MASTER**).
ON	Turns the unit on. Any devices slaved to the **12V TRIGGER** will also turn on in a controlled sequence.
VOLUME	Turns the volume up or down.
MUTE	Toggles the **MUTE** function. Press and hold the **MUTE** button for 10 seconds to toggle the **CALIBRATED** mode on the **MAIN** outputs.
DIM	Toggles the -20 dB **DIM** function. Press and hold the **DIM** button for 10 seconds to toggle the **CALIBRATED** mode on the **AUX** outputs.
INPUT	Selects the inputs.
D1	Selects optical digital input **D1**.
D2	Selects optical digital input **D2**.
D3	Selects XLR digital input **D3**.
D4	Selects coaxial digital input **D4** then toggles between **D4** and **D5**.
USB	Selects **USB** input. Press and hold the **USB** button for 3 seconds to toggle between the **USB 1.1** and **USB 2.0** modes.
Analog	Selects analog input **A1** and then toggles between **A1** and **A2.**

Front Panel Controls

The front panel controls duplicate all of the functions that are available from the remote control.

Two additional functions, **AUTO-ON**, and **COMPATIBILITY** mode are only controllable from the front panel.

- The **AUTO-ON** function keeps the ***DAC2*** on whenever AC line voltage is supplied.
- The **COMPATIBILITY** mode disables the volume control when the ***DAC2*** feeds a preamplifier that will be used to control the system playback volume. mote can be used to control both devices.

Tip: When **AUTO-ON** is enabled, a switched AC outlet can be used to turn your system on and off. The **12V TRIGGER I/O** can be used as a trigger output to control the power state of additional components.

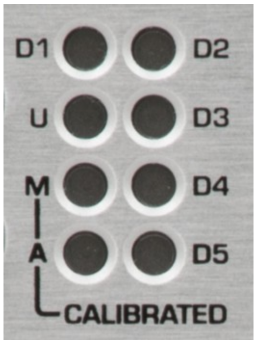
The chart at the right summarizes the functions of the front-panel controls.

POWER	Turns the unit on or off. Any devices slaved to the **12V TRIGGER** will also turn on or off in a controlled sequence. Starting with the unit off, press and hold the **POWER** button for 10 seconds to set the **AUTO-ON** function. Starting with the unit on, press and hold the **POWER** button for 10 seconds to clear the **AUTO-ON** function. If **AUTO-ON** is set, the **POWER** button will toggle **MUTE** on and off (the unit will remain on).
MUTE	Press once to toggle the -20 dB **DIM** function. Press and hold the **MUTE** button for 10 seconds to toggle the **CALIBRATED** mode on the **MAIN** outputs. Simultaneously hold the **MUTE** and **INPUT-UP** keys for 10 seconds to activate or deactivate the **COMPATIBILITY** mode.
DIM	Press once to toggle the -20 dB **DIM** function. Press and hold the **DIM** button for 10 seconds to toggle the **CALIBRATED** mode on the **AUX** outputs.
INPUT	Selects the inputs. Select USB and press lower button for 10 seconds to toggle between the **USB 1.1** and **USB 2.0** modes.
VOLUME (knob)	Sets the volume of all outputs that are not in **CALIBRATED** mode.

Front Panel Displays

There are sixteen status indicator lights on the front panel. At least one light will be illuminated whenever power is on.

Calibration and Input Indicators



The **M** and **A** indicators show that the **CALIBRATED** mode is active for the **MAIN** and/or **AUX** outputs.

The input indicators (**U** and **D1**-**D5**) show which input is selected.

A	This light will flash rapidly when the volume of the **AUX** bus is being adjusted. A solid light indicates that the **AUX** bus is in **CALIBRATED** mode. A slow-blinking light indicates that the **AUX** bus is in **CALIBRATED** mode but the output is muted or dimmed.
M	This light will flash rapidly when the volume of the **MAIN** bus is being adjusted. A solid light indicates that the **MAIN** bus is in **CALIBRATED** mode or in **COMPATIBILITY** mode. A slow-blinking light indicates that the **MAIN** bus is in **CALIBRATED** mode but the output is muted.

U	A solid light indicates that the **USB** input is selected and operating normally. A blinking light indicates that the input is selected but a connection to a computer has not been established.
D1	A solid light indicates that optical input **D1** is selected and operating normally. A blinking light indicates that the input is selected but audio data is not being received.
D2	A solid light indicates that optical input **D2** is selected and operating normally. A blinking light indicates that the input is selected but audio data is not being received.
D3	A solid light indicates that XLR input **D3** is selected and operating normally. A blinking light indicates that the input is selected but audio data is not being received.
D4	A solid light indicates that coaxial input **D4** is selected and operating normally. A blinking light indicates that the input is selected but audio data is not being received.
D5	A solid light indicates that coaxial input **D5** is selected and operating normally. A blinking light indicates that the input is selected but audio data is not being received.

Note: **D5** cannot be selected if the **Digital Pass Through** function is enabled. Instructions for configuring this jumper-selected function can be found in the **Internal Settings** section of this manual (**Page 29**).

Input Error Codes

An input indicator (**U** or **D1**-**D5**) flash when an error is present on the selected digital input. Use the following table to diagnose the problem:

Slow Flash (2Hz)	No digital signal (output muted)
Med. Flash (7Hz)	Data transmission errors or Non-PCM (output muted)
Rapid flashes (14Hz)	Non-audio data is being received (output muted)
Intermittent flashes	Some data corruption is occurring, converter may be interpolating to replace invalid samples, check the cable.

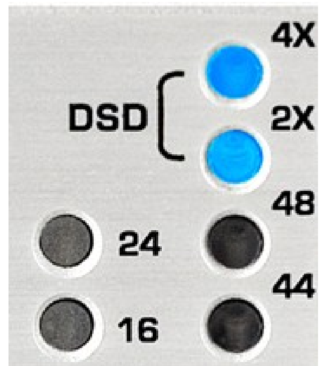
Tip: Common causes of input errors:

- Disconnected or faulty cable
- Use of excessively long digital cables
- Use of analog cables for digital signals
- Use of optical cables for sample rates exceeding 96 kHz
- Incompatible data type (AC3, ADAT, etc.)
- Non-audio data is being received

MUTE and DIM Indicators

MUTE	Indicates that all outputs are muted.
DIM	Indicates that all non-calibrated outputs are dimmed by 20 dB.

Digital Format Indicators



Two lights indicate the measured word length of the selected digital input.

Four lights indicate the measured sample rate and format of the selected digital input.

Tip: Computers, disk players and streaming devices often subject the digital signal to sample rate conversion, changes in word length, PCM to DSD conversions, and other forms of digital processing that may degrade the quality of the audio. This display makes it easy to detect these processes.

Word Length Indicators

The **16** and **24** lights indicate the measured word length of the selected digital input. The ***DAC2 DX*** detects active data bits and displays the results as follows:

16 Only	Measured input word length is 16 bits.
16 and 24	Measured input word length is 17 to 23 bits.
24 Only	Measured input word length is 24 bits.
Both Off	Measured input word length is less than 16 bits.

Tip: The **DIM** button will not change the level of outputs that are in the **CALIBRATED** mode.

Format indicators

The **44**, **48**, **2X**, **4X** and **DSD** lights indicate the sample rate and format of the selected digital input as follows:

44 Only	The input format is PCM at a sample rate of 44.1 kHz (CD sample rate).
48 Only	The input format is PCM at a sample rate of 48 kHz (often used with video).
44 and 2X	The input format is PCM at a sample rate of 88.2 kHz (high-resolution audio format).
48 and 2X	The input format is PCM at a sample rate of 96 kHz (high-resolution audio format).
44 and 4X	The input format is PCM at a sample rate of 176.4 kHz (high-resolution audio format).
48 and 4X	The input format is PCM at a sample rate of 192 kHz (high-resolution audio format).
DSD **(4x and 2X)**	The input format is 1-bit DSD at a sample rate of 2.8224 MHz (high-resolution audio format). Note: DSD must be streamed in DoP format.
All Off	Digital signal is not present or is not in a supported format.

Headphone Jacks

By default, the left-hand jack mutes the **MAIN** outputs.	By default, the right-hand jack keeps all outputs active.

Headphone Mute Switches

Both headphone jacks includes include switches that can be programmed to mute the **MAIN** outputs. When enabled, the **MAIN** analog outputs (XLR and RCA) are muted when a headphone plug is inserted. This feature allows the listener to switch from loudspeaker to headphone playback seamlessly. This **Auto-Mute** feature can be disabled or enabled on one or both jacks using internal jumpers.

Note: Instructions for setting the **Auto-Mute** jumpers can be found in the **Internal Settings** section of this manual (**Page 29**).

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

Driving Two Sets of Headphones

The ***HPA2***™ is specifically designed with enough power to drive two sets of headphones.

Tip: The **Volume Control** simultaneously adjusts the level for both jacks. If two listeners will be simultaneously using the headphone outputs, we recommend using headphones with identical or similar voltage sensitivities.

HPA2™ Headphone Power Amplifier

The headphone jacks are driven by Benchmark's ***HPA2***™ headphone power amplifier. This very clean power amplifier can deliver the current and voltage required by some of the most demanding headphones. The output impedance of the ***HPA2***™ is very close to 0 Ohms. This low output impedance delivers a high damping factor so that the amplifier can maintain precise control over the headphone transducers.

Headphone voltage sensitivities vary substantially, so we have equipped the ***HPA2***™ with **Gain-Range** jumpers that can be used to customize the headphone amplifier to your favorite headphones. If you find that you have too much output (volume control set below 11 o'clock), there are internal jumpers that can be adjusted to decrease the output level by 10 dB or 20 dB relative to the factory default setting.

Note: Instructions for setting the headphone **Gain-Range** jumpers can be found in the **Internal Settings** section of this manual (**Page 29**).

Tip: For optimal performance, the headphone **Gain-Range** jumpers should be set so that comfortable listening levels occur when the volume control is set above the 11 o'clock position.

Operational Details

DIM and MUTE Functions

Definition: If the **DIM** and **MUTE** and **COMPATIBILITY** modes are all off, the ***DAC2 DX*** is in **NORMAL** mode.

DIM Function

The **DIM** function reduces the output level by 20 dB. To toggle between **NORMAL** and **DIM** volume, press the **DIM** button on the remote or the **DIM** button on the front panel.

The **DIM** light will turn on whenever **DIM** is active.

The **DIM** function makes it convenient to fade back and forth between normal and background playback volume levels.

If the volume is adjusted while **DIM** is active, the **NORMAL** volume setting will change by the same amount.

If an output bus is set to **CALIBRATED** mode, **DIM** is disabled on that bus.

TIP: In the studio, the **DIM** function allows a temporary reduction in level without losing the volume setting that was being used for monitoring.

Tip: In home applications the **DIM** function allows temporary reductions in volume during TV commercial breaks, phone calls, or other interruptions, without losing the volume setting that was being used for normal listening.

MUTE Function

The **MUTE** function immediately mutes all outputs. To toggle this function, press the **MUTE** button on the remote or press the **MUTE** button on the front panel.

The **MUTE** light will turn on whenever **MUTE** is active.

Tip: If the unit is in **MUTE**, press the **DIM** key to immediately enter the **DIM** mode.

Tip: If the unit is in **MUTE** and/or **DIM** press the **ON** key (on the remote) to enter the **NORMAL** mode.

Tip: if the unit is in **DIM**, press the **MUTE** key to toggle between **DIM** and **MUTE**.

AUTO-ON Function

The ***DAC2 DX*** can be programmed to automatically turn on whenever AC power is applied. This function allows automation using switched AC outlets. When **AUTO-ON** is enabled, the ***DAC2 DX*** cannot be turned off without removing AC power.

The **AUTO-ON** function is programmed by pressing and holding the **POWER** button on the front panel for 10 seconds. It is the only function that cannot be programmed from the remote control. This limitation prevents accidental access to this special feature.

Enabling AUTO-ON

Starting with the ***DAC2 DX*** off, press and hold the **POWER** button for 10 seconds.

If **AUTO-ON** has been successfully enabled, the **POWER** button will now function as a **MUTE** button. The **OFF** button on the remote will also function as a **MUTE** button when **AUTO-ON** is enabled.

Disabling AUTO-ON

Starting with the ***DAC2 DX*** on, press and hold the **POWER** button for 10 seconds. At the end of 10 seconds the ***DAC2 DX*** will power down if the **AUTO-ON** mode has been successfully disabled.

Bi-directional 12V Trigger

Benchmark has reinvented the 12 volt trigger by adding bi-directional signaling. The trigger connection on the ***DAC2 DX*** can be used as an input, an output, or both. It is compatible with any common 12 volt trigger input or output. The **12V TRIGGER** I/O can be used to turn other audio components on when the ***DAC2 DX*** turns on. The ***DAC2 DX*** can also turn on and off in response to other connected components. The Benchmark bi-directional **12V Trigger** is compatible with virtually all trigger systems.

The **12V TRIGGER** I/O can be connected to the trigger input or output ports on a preamplifier, power amplifier, or both.

The ***DAC2 DX*** can send a 12 Volt DC trigger signal to start other components in the system, or it can wake up in response to an externally generated trigger signal. The ***DAC2 DX*** automatically configures its trigger I/O port as an input (slave) or output (master).

Trigger Output (DAC2 DX is Master)

When the ***DAC2 DX*** is turned on using the **POWER** button (on the front panel), or the **ON** button (on the remote), the ***DAC2 DX*** configures itself as a trigger master and will drive the **12V TRIGGER** I/O to 12 volts DC and hold it there while the ***DAC2 DX*** is on. The trigger output signal generated by the ***DAC2 DX*** is delayed so that the ***DAC2 DX*** can stabilize before downstream devices (such as power amplifiers) turn on. When powering down, the ***DAC2 DX*** will mute before allowing the trigger line to drop low. The ***DAC2 DX*** keeps the internal power supplies running for 10 seconds after dropping the trigger. This delay gives other triggered components ample time to mute and shut down.

If the **AUTO-ON** function is enabled, the ***DAC2 DX*** will automatically turn on when AC power is applied, configure itself as a trigger master, and ignore any external signaling on the **12V TRIGGER** I/O line. In **AUTO-ON** mode, the ***DAC2 DX*** will always drive the **12V TRIGGER** I/O line to 12 V (after a short start-up delay).

Trigger Input - (DAC2 DX is Slave)

If the ***DAC2 DX*** is off and an external device pulls the trigger I/O to 12 volts, the ***DAC2 DX*** will configure itself as a trigger slave and will follow the actions of the trigger input. The ***DAC2 DX*** will then turn off when the external device stops sending the 12 V trigger.

Typical Trigger Applications

In most systems, the **12V TRIGGER** will be used to connect the ***DAC2 DX*** to one other device. The ***DAC2 DX*** can be connected to the first trigger input at the beginning of a trigger chain, or it can be connected to the last trigger output at the end of the chain (less common).

Typical trigger applications:

- ***DAC2 DX*** → Amplifier
- ***DAC2 DX*** → Amplifier → Amplifier
- ***DAC2 DX*** → Preamplifier → Amplifier

Trigger Bus Applications

The Benchmark bi-directional trigger system also supports multiple trigger ports wired together on a bus.

A group of Benchmark trigger ports can be connected to a group of non-Benchmark trigger input ports to form a single trigger bus. A bus should never be connected to more than one non-Benchmark trigger output port. If an output port is connected to the bus, this device should be used to start the audio system.

A 3.5 mm (1/8") TRS "Y" cable can be used to split the trigger output of the ***DAC2 DX*** to feed more than one trigger input.

Benchmark ***AHB2*** power amplifiers have two trigger I/O ports that are wired in parallel. This makes it easy to connect more than one power amplifier to a trigger bus (without the use of a "Y" cord). Connect a trigger cable between the ***DAC2 DX*** and the first amplifier. Use another trigger cable to connect this amplifier to the next amplifier. Any number of Benchmark amplifiers can be added to the trigger bus. The ***DAC2 DX*** will turn on first, and after a delay, all of the amplifiers will turn on together.

Bi-Directional Trigger Applications

Benchmark products support bi-directional communications over a trigger bus. Any Benchmark product connected to the bus can turn the entire system on or off. Because of the bi-directional design, any power button on a Benchmark ***DAC2 DX*** or ***AHB2*** can be used to start or stop the system.

The Benchmark device that starts the system will become the trigger master. If the trigger master is turned off, all slave devices will follow. If a slave device is turned off, all other devices will stay on.

If the ***DAC2 DX*** is used to turn the system on, any connected ***AHB2*** amplifiers will become slave devices and they can be turned off without shutting down the ***DAC2 DX***. This feature makes it easy to turn the ***AHB2*** amplifier(s) off when listening to headphones.

Slave devices can force the entire trigger bus to shut down if the **POWER** button or **OFF** button is pressed and held for 3 seconds.

Tip: Press and hold the **POWER** button on any Benchmark device for 3 seconds to force a shutdown of the entire trigger-connected system.

Trigger Specifications

The Benchmark **12V TRIGGER** I/O has a wide operating range to allow interfacing with most other DC trigger systems. It should only be used with trigger inputs that are designed to tolerate 12 VDC.

- 12 VDC 200 mA current-limited output
- Input responds to 3.3 V logic and higher
- Maximum input voltage = 30 VDC
- Maximum reverse input voltage = -0.3 VDC
- Input Impedance = 20 k Ohms
- 1/8" (3.5 mm) TRS jack
- Tip = 12 Volt Trigger I/O
- Ring = no connection
- Sleeve = chassis ground

CALIBRATED Mode

The **CALIBRATED** mode sets the **MAIN** and/or **AUX** output buses to calibrated levels. The two output buses are individually programmable for each digital input.

The factory preset calibration is +24 dBu at 0 dBFS on the XLR outputs, and 2 Vrms at 0 dBFS on the RCA outputs. The calibration is adjustable up or down by up to 4 dB in 1 dB steps using internal jumpers. On the XLR outputs the calibration range is +20 dBu to +28 dBu. On the RCA outputs the calibration range is 1.26 to 3.17 Vrms.

The **CALIBRATED** mode is similar to the **CALIBRATED** switch setting on the ***DAC1*** except that the **CALIBRATED** mode can be programmable separately for each input.

The **CALIBRATED** mode has two distinct applications:

- **Volume Control Bypass** - useful when the system has an upstream digital volume control or a downstream analog volume control
- **Calibrated Output** - useful in studio applications where calibrated levels are needed

The **M** light indicates that the **MAIN** bus is in **CALIBRATED** mode.

The **A** light indicates that the **AUX** bus is in **CALIBRATED** mode.

Enabling the CALIBRATED Mode

1. Select the input channel that you wish to program
2. Press and hold the **MUTE** button for 10 seconds to toggle **CALIBRATED** mode on the **MAIN** outputs
3. Press and hold the **DIM** button for 10 seconds to toggle **CALIBRATED** mode on the **AUX** outputs.

CALIBRATED Mode - Volume Control Bypass

The **CALIBRATED** mode is useful whenever the system volume will be controlled before or after the ***DAC2 DX***. It is usually best to avoid having two cascaded volume controls in a playback system. Dual controls will usually degrade the noise performance of the system and they can lead to confusion.

If the ***DAC2 DX*** feeds a preamplifier, the preamplifier will provide a downstream analog volume control for the system and the ***DAC2 DX*** should be placed in **CALIBRATED** mode for all inputs.

If the ***DAC2 DX*** is directly feeding an amplifier, but one or more sources have volume controls, the sources with volume controls can be set to **CALIBRATED** mode.

Example 1: The **USB** input is fed from a computer that has an internal digital volume control. If you wish to use the volume control in the computer exclusively, you will want to enable the **CALIBRATED** mode on the **USB** input. If you do not wish to use the computer volume control, leave the **CALIBRATED** mode off and disable the computer volume control (or set it to maximum).

Example 2: Digital input **D3** is fed from a digital audio workstation and the workstation will be controlling the playback level in the control room. The volume control on the ***DAC2 DX*** will be controlling the playback level in another room. The **MAIN** outputs will be driving the control room monitors and the **AUX** outputs will be driving the monitors in the other room. Select **D3** and enable the **CALIBRATED** mode on the **MAIN** outputs only.

Example 3: Input **D4** is being driven by a music server. A power amplifier in the same room is driven from the **MAIN** outputs. Another room needs a fixed line-level feed to an integrated amplifier (with volume control). Select **D4** and enable the **CALIBRATED** mode on the **AUX** output only.

Preamp COMPATIBILITY Mode

The **COMPATIBILITY** mode sets the MAIN output bus to its calibrated output level and disables the motorized volume control. The main outputs can then feed a preamplifier that will be used to control the system playback volume. When this mode is used with a Benchmark preamplifier or line amplifier, such as the ***HPA4***, a single Benchmark IR remote can be used to control both devices.

In **COMPATIBILITY** mode:

- **M_CAL** mode is on for all inputs.
- **MUTE** is disabled.
- **DIM** is disabled.
- Remote **INPUT** select arrows are disabled.
- Remote **VOLUME** commands are disabled.

Enabling the COMPATIBILITY mode:

- Simultaneously hold the **MUTE** and **INPUT-UP** keys for 10 seconds to activate or deactivate this feature.
- The **M** light will be illuminated on all inputs when **COMPATIBILITY** mode is enabled.
- The **DIM** and **MUTE** keys will be disabled when **COMPATIBILITY** mode is enabled.
- The ***DAC2*** will not respond to the **DIM**, **MUTE**, **VOLUME-UP**, **VOLUME-DOWN**, **INPUT-UP**, or **INPUT-DOWN** keys on the IR remote when **COMPATIBILITY** mode is enabled.

USB MODE Selection

The ***DAC2 DX*** supports two **USB MODES**:

- **USB Audio 1.1** mode - up to 24 bits at 96 kHz
- **USB Audio 2.0** mode - up to 24 bits at 192 kHz plus DSD in DoP 1.1 format

Caution: Close all USB audio playback applications before changing the **USB MODE**. If an audio application is playing while the **USB MODE** is changed, the audio application may freeze.

To change the **USB MODE**, select the **USB** (**U**) input on ***DAC2 DX*** and then press and hold the **USB** button on the remote control for 10 seconds. If a remote control is not available, select the **USB** input, and then press and hold the bottom **INPUT** button on the front panel for 10 seconds.

After holding the button for 10 seconds, either the **4X** lamp or the **2X** lamp will flash once indicating the new **USB MODE**. A flash of the **4X** lamp indicates that the unit is now in **USB Audio 2.0** mode. A flash of the **2X** lamp indicates that the unit is now in **USB Audio 1.1** mode.

Tip: The **4X** or **2X** lamp will flash once every time the **USB** input is selected. This flash provides a convenient indication of the current **USB MODE**.

Tip: Avoid any unnecessary switching between **USB MODES**. Rapid switching between modes can confuse some operating systems.

Tip: USB Audio 1.1 and USB Audio 2.0 are industry standard protocols for the transmission of digital audio over USB interfaces. Mac operating systems and Windows operating systems starting with Windows 10 support both modes. Windows operating systems prior to Windows 10 only support USB Audio 1.1. For this reason, we provide a Windows driver for USB Audio 2.0 that can be used with older Windows systems. Do not attempt to install this driver on Windows 10 systems.

Driving Power Amplifiers

The ***DAC2 DX*** is designed to directly drive virtually any audio power amplifier or powered monitor. This direct connection provides the cleanest and shortest path from the digital source to the monitor output.

Tip: In most cases, Benchmark does not recommend placing an audio device between the ***DAC2*** and the power amplifier. One notable exception is the ***HPA4*** line amplifier. The ***HPA4*** provides stepped relay gain control with exceedingly low THD and noise. The ***HPA4*** provides the ultimate analog volume control and it can operate transparently between the ***DAC2*** and a power amplifier.

The **RCA** and **XLR** outputs on the ***DAC2 DX*** are equipped with low-impedance high-current drivers. These robust outputs are well equipped to drive a wide variety of input impedances. The ***DAC2 DX*** outputs remain clean when driving amplifiers that present difficult loads (high input capacitance and/or low input impedance).

The **XLR** outputs on the ***DAC2 DX*** are equipped with jumper-configured passive low-impedance output pads. These pads can be set to an attenuation of **0 dB** (pad off), **10 dB**, or **20 dB**. The pads should be used to match the output level of the ***DAC2 DX*** to the input sensitivity of the power amplifier. Most power amplifiers and powered monitors will require the use of the **10 dB** or **20 dB** pads. Use the high-output **0 dB** setting when driving a Benchmark ***AHB2*** power amplifier.

Tip: The Benchmark ***AHB2*** power amplifier has a unique low-gain topology that allows it to accept full studio-level input signals. This high-level interconnection provides a very low-noise connection between the ***DAC2 DX*** and the ***AHB2***. Set the input **SENSITIVITY** switch on the ***AHB2*** to **22 dBu** (all the way down). This places the ***AHB2*** full-power output point at an input level of 22 dBu. This level is exactly 2 dB lower than the **CALIBRATED** output level of the ***DAC2 DX*** (when the pads are set to **0 dB**). This configuration optimizes the gain-staging between the ***DAC2 DX*** and the ***AHB2*** while placing the ***DAC2 DX*** volume control in the proper range.

Tip: If you are using a ***DAC2 DX*** with non-Benchmark power amplifiers, the **XLR** pads should be set so that comfortable listening levels occur when the **VOLUME** control is set above 11 o'clock. This will optimize the gain-staging between the ***DAC2 DX*** and your power amplifier.

Tip: Increase the pad setting if a comfortable listening level is reached at a **VOLUME** control setting below the 11 o'clock position.

Tip: Decrease the pad setting if a comfortable listening level cannot be reached when the **VOLUME** control is fully clockwise.

Instructions for setting the **XLR** pad jumpers are detailed in the **Internal Settings** section of this manual (**Page 29**).The ***DAC2 DX*** is shipped with the **XLR** pads disabled (set to **0 dB**). No adjustments will be necessary if you will be using a Benchmark ***AHB2*** power amplifier.

HPA2™ Headphone Amplifier

The ***DAC2 DX*** is equipped with two 1/4" stereo headphone jacks. The audio output on both jacks is wired in parallel and is driven by Benchmark's ***HPA2™*** headphone power amplifier. The ***HPA2™*** is an ultra-clean power amplifier that is capable of delivering 1.25 W into 30 Ohms. It easily drives a pair of headphones.

Both headphone jacks are equipped with mute switches that can be programmed to mute the analog outputs on the back of the ***DAC2 DX***. By default, the switch on the left-hand headphone jack is enabled and the switch on the right-hand headphone jack is disabled. Instructions for setting the headphone switches are detailed in the **Internal Settings** section of this manual (**Page 29**).

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

The ***HPA2™*** has a set of 3-position gain-range jumpers that can be used to increase or decrease the gain by 10 dB relative to the factory calibrated setting.

The jumpers change the gain of the ***HPA2***™ headphone amplifier without changing the output impedance. This keeps the output impedance of the ***HPA2***™ constant and very near 0 Ohms. External attenuators should never be inserted after a headphone amplifier as this would change the output impedance and alter the frequency response of the headphones.

Proper gain settings are important for maximizing the SNR of the headphone monitoring system. With proper settings, the full performance of the ***DAC2 DX*** can be delivered to the headphones for critical monitoring tasks and for maximum musical enjoyment.

Tip: When the headphone gain jumpers are set properly, a normal listening level will be achieved at a **VOLUME** control setting above the 11 o'clock position.

Tip: If a normal listening level is achieved below an 11 o'clock **VOLUME** setting, the headphone gain is too high, and the gain should be decreased.

Tip: If the level is too low at the maximum **VOLUME** setting, the headphone gain is too low, and the gain should be increased.

Digital Pass-Through

The second coaxial input (**D5**) can be reconfigured as a digital output. When operating as an output, any selected digital input is passed through to **D5** without any processing.

Optical, coaxial, and USB inputs (**U**, **D1**, **D2**, **D3** and **D4**) can be passed through to the **D5** connector. The signals are buffered but are not processed in any way. For this reason, any data format can be passed through to the **D5** connector, even when these formats cannot be decoded by the ***DAC2 DX***. Surround formats, such as DTS, Dolby Digital, cannot be decoded by the ***DAC2 DX***, but they can be passed to a surround system using the digital pass-through function.

The digital pass-through can also be used to provide the following digital signal conversions:

- Optical to Coaxial
- USB to Coaxial
- AES XLR Digital to Coaxial
- Coaxial to Coaxial (buffering)

DoP encapsulated DSD can also be passed through **D5**. For example, DSD files on a computer can be sent in DoP to the **USB** input on the ***DAC2 DX***. The **USB** input can be routed to coaxial output **D5**. This output can be recorded by any 24-bit, 176.4 kHz digital recorder with a coaxial input. The PCM digital recorder can then be used to play the DSD recordings in DoP format.

Firmware Version Identification

The firmware version is displayed during the lamp test while the ***DAC2*** is turning on. At least one lamp in the **INPUT INDICATOR** will flash rapidly while the remaining lamps will be on. The flashing lamps identify the firmware version. The values of each lamp are shown in this chart below.

Add the values of all flashing lamps to determine the version number. If no lamp flashes in the second column, the second digit is a 0.

Digit 1			**Digit 2**	
8	**D1**		**D2**	.8
4	**U**		**D3**	.4
2	**M**		**D4**	.2
1	**A**		**D5**	.1

Example 1: The **A** lamp is the only lamp that flashes. The firmware version is 1.0.

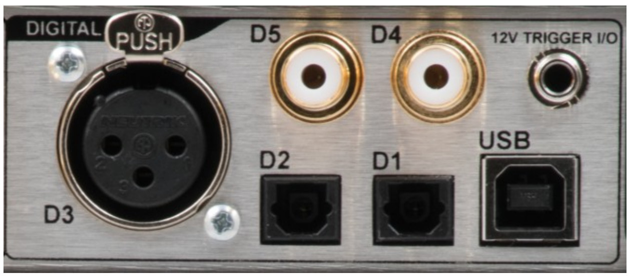
Example 2: The **A** and **D5** lamps flash. The firmware version is 1.1.

Example 3: The **M** lamp is the only lamp that flashes. The firmware version is 2.0.

Example 4: The **M**, **A**, **D5**, **D4** and **D3** lamps flash. The firmware version is 3.7.

Rear Panel

Inputs



There are six stereo digital inputs on the ***DAC2 DX***:

- **USB** - USB Audio 1.1 or 2.0 Input
- **D1** - Optical Digital Input
- **D2** - Optical Digital Input
- **D3** - XLR Digital Input
- **D4** - Coaxial Digital Input
- **D5** - Coaxial Digital Input or Output*

These inputs are selected using the **INPUT** buttons on the front-panel or on the remote-control.

Tip: There is no **D5** button on the remote control. Use the **INPUT** up-down buttons or press **D4** twice to select input **D5**. Pressing **D4** again to return to input **D4**.

* **D5** can be jumper-configured as a digital **PASS-THROUGH** output. When enabled, the selected digital input will be routed to the internal D/A converter and to output **D5**. The selected input will be buffered and sent to output **D5** even if the format cannot be decoded by the ***DAC2 DX***.

The digital inputs support PCM stereo AES/EBU and SPDIF digital formats. Maximum word length is 24-bits. Maximum sample rate is 192kHz. The digital inputs also support DSD stereo at a sample rate of 2.8224 MHz using DoP 1.1 encapsulation.

The **USB** input has two operating modes:

- **USB Audio 1.1** - PCM up to 24-bits at 96 kHz
- **USB Audio 2.0** - PCM up to 24-bits at 192 kHz and DSD (DoP 1.1 format)

Caution: The optical inputs (**D1** and **D2**) are not recommended for DSD or for sample rates above 96 kHz. Optical connections may be unreliable at sample rates above 96 kHz.

Tip: The ***DAC2 DX*** will not decode multichannel digital formats such as AC3, and Dolby Digital. The audio will mute and the **INPUT INDICATORS** will flash whenever an incompatible format is connected to the selected digital input. If the **PASS-THROUGH** mode is enabled, these multichannel formats can be sent to a surround processor using connector **D5** as a digital output.

Caution: The **12V TRIGGER** I/O is not an audio connection! This is a 12V DC connection for synchronizing the on and off sequencing of an entire audio system.

Digital Inputs - Details

All of the digital inputs on the ***DAC2 DX*** use Benchmark's ***UltraLock2***™ system to remove virtually all of the interface jitter. The result is that all digital inputs deliver identical audio performance. The USB, optical, XLR and coaxial digital inputs will all sound identical if they receive identical data.

Computer Input – USB

The **USB** input accepts a **Type-B** male USB connector. A **Type-A** to **Type-B** USB cable is provided with the ***DAC2 DX***. The USB cable connects the ***DAC2 DX*** directly to a computer's USB output.

The **USB** input supports 44.1, 48, 88.2, 96, 176.4, and 192 kHz PCM sample rates at word lengths up to 24-bits. The **USB** input also accepts DSD in DoP 1.1 format.

The ***DAC2 DX*** can be configured as a **USB Audio 1.1** or **USB Audio 2.0** device. Press and hold the **USB** button on the **REMOTE** for 10 seconds to toggle the **USB MODE**. If a remote is not available, select the **USB** input and then press and hold the bottom **INPUT** button on the front panel for 10 seconds.

The **USB AUDIO 1.1** mode never requires the installation of a driver. It allows a quick driverless connection to Windows machines when playing sample rates of 96 kHz or less. In this mode, Windows machines can begin streaming audio within seconds after the ***DAC2 DX*** is connected for the first time. No software or hardware configuration is usually required.

USB Audio 2.0 is required for DSD and for all PCM sample rates exceeding 96 kHz. Windows computers require a driver to support the **USB Audio 2.0** mode.

The **USB Audio 1.1** mode was tested for compatibility with Windows XP, Vista, 7, 8 and 10, Mac OS X, and iPads using the 30-pin to USB Camera Kit. No driver installation is required for any of these systems when operating in **USB Audio 1.1** mode.

The **USB Audio 2.0** mode was tested for compatibility with Windows XP, Vista, 7, and 8 (driver installation is required for these Windows versions). Beginning with Windows 10, no driver is required. Do not attempt to install the driver on Windows 10.

The **USB Audio 2.0** mode was also tested for compatibility with Mac OS X starting with version 10.6 (operation is driverless for all OS X versions).

Optical Digital Inputs - D1 and D2

The optical input connectors (**D1** and **D2**) are commonly known as **TOSLINK** connectors. The **TOSLINK** optical connectors used on the ***DAC2 DX*** are 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 inputs may be unreliable at sample rates above 96 kHz. The optical inputs will accept professional AES/EBU data formats or consumer S/PDIF data formats.

Tip: The optical inputs include dust caps. Keep these in place if the input is not being used.

XLR Digital Input - D3

The XLR digital inputs (**D3**) uses a female XLR connector. The input impedance is 110 Ohms. Maximum word length is 24-bits. All sample rates between 28 and 195 kHz are supported. The XLR digital input will accept professional AES/EBU data formats or consumer S/PDIF data formats. The XLR input also accepts DSD in DoP 1.1 format.

The XLR digital input is transformer coupled. The transformer provides galvanic isolation of the signal pins (2 and 3). A capacitor protects the transformer from DC currents. Diodes at the transformer outputs protect the digital audio receiver. Pin 1 (ground) and the XLR shell are tied directly to the chassis to prevent currents in the internal ground system. This direct bonding also maximizes RF shielding.

Caution: Use 110-Ohm digital audio cables for digital XLR audio connections. Do not use analog XLR cables. Digital interfaces require the use of matched impedances. The digital input may not function, or may be unreliable if the incorrect cable is used.

Coaxial Digital Inputs - D4 and D5

The coaxial digital inputs (**D4** and **D5**) use female RCA connectors. The input impedance is 75 Ohms. Maximum word length is 24-bits. All sample rates between 28 and 195 kHz are supported. The coaxial digital inputs will accept professional AES/EBU data formats or consumer S/PDIF data formats. The coaxial inputs also accept DSD in DoP 1.1 format.

The coaxial digital 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.

Caution: Use 75-Ohm coaxial cables for digital audio connections **D4** and **D5**. Digital interfaces require the use of matched impedances. Do not use 50-Ohm coaxial cables, twisted pair cables, or any non-coaxial cables for digital audio. The digital inputs may not function, or may be unreliable if the incorrect cable is used.

Note: The Coaxial inputs (**D4** and **D5**) accept professional or consumer digital audio formats. The AES3-id and SMPTE 276M standards specify a 75-Ohm, 1 Vpp, professional format which is also known as AES/EBU and is commonly used in video production facilities. The IEC 609588-3 standard specifies a 75-Ohm, 0.5 Vpp, consumer format which is also known as S/PDIF, and is commonly used in hi-fi equipment. The coaxial inputs are designed to accept either type of signal.

12V TRIGGER I/O

The Benchmark bi-directional **12V TRIGGER** is compatible with virtually all trigger systems. The **12V TRIGGER** I/O connection on the ***DAC2 DX*** can be used as an input, an output, or both. It is compatible with most 12 volt trigger inputs and outputs. The **12V TRIGGER** can be used to turn other audio components on when the ***DAC2 DX*** turns on. The ***DAC2 DX*** can also turn on and off in response to trigger signals sent from other components.

The **12V TRIGGER** I/O can be connected to the trigger input or output ports on a preamplifier, power amplifier, or both.

The ***DAC2 DX*** can send a 12 Volt trigger signal to start other components in the system, or it can wake up in response to an externally generated trigger signal. The ***DAC2 DX*** automatically configures the **12V TRIGGER** I/O port as an input (slave) or output (master). See the **Bi-directional 12V Trigger** section on **Page 17** for more information.

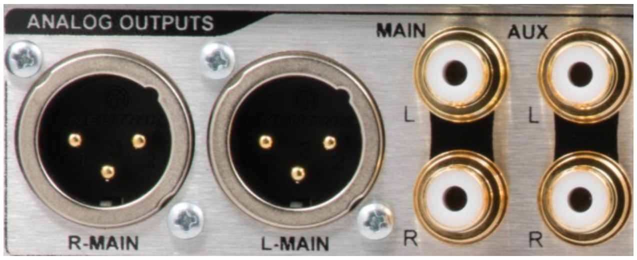
The Benchmark **12V TRIGGER** I/O has a wide operating range to allow interfacing with most other DC trigger systems. It should only be used with trigger inputs that are designed to tolerate 12 VDC.

- 12 VDC 200 mA current-limited output
- Input responds to 3.3 V logic and higher
- Maximum input voltage = 30 VDC
- Maximum reverse input voltage = -0.3 VDC
- Input Impedance = 20 k Ohms
- 1/8" (3.5 mm) TRS jack
- Tip = 12 Volt Trigger I/O
- Ring = no connection
- Sleeve = chassis ground

Caution: The **12V TRIGGER** I/O is not an audio connection! This is a 12V DC connection for synchronizing the on and off sequencing of an entire audio system.

Outputs

Analog Outputs



The ***DAC2 DX*** has two analog output buses (**MAIN** and **AUX**). By default, both buses are controlled by the volume control. Either or both buses can be programmed to bypass the volume control. The **CALIBRATED** mode bypasses the volume control and sets the output(s) to a preset level.

MAIN Bus

The **MAIN** bus drives the XLR outputs and one pair of RCA outputs. The **MAIN** bus delivers the highest performance because it uses three conversion channels wired in parallel for each XLR connector. The main bus uses 6 of the 8 channels in the ES9018 D/A conversion chip. The remaining two channels in the ES9018 drive the **AUX** bus.

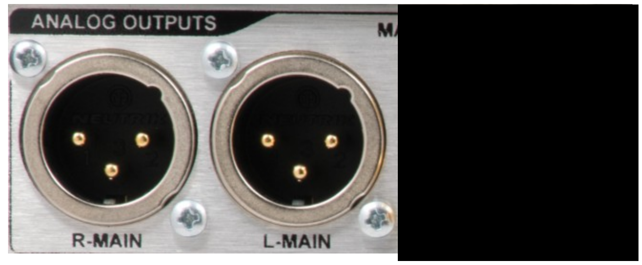
AUX Bus

The **AUX** bus drives the second pair of RCA outputs.

Output Drivers

The ***DAC2 DX*** 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.

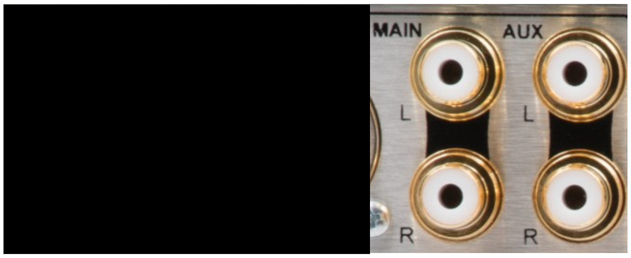
The XLR outputs have passive attenuators that allow direct connections to a wide variety of audio devices without a loss of dynamic range. The 10 or 20 dB pads are usually required for direct interfacing to power amplifiers and powered speakers. The ***DAC2 DX*** ships with the pads disabled (0 dB setting). Use the 0 dB setting with the Benchmark ***AHB2*** power amplifier. A full description of the output attenuators and instructions for configuration is located in the **Internal Settings** section of this manual (**Page 29**).

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, may cause distortion, and may shorten the life of the output drivers.

Unbalanced RCA Analog Outputs



The Left and Right unbalanced outputs use female RCA 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. 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.

Caution: Mono summing with an RCA 'Y' cable is not recommended as this can 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.

AC Power-Entry and Fuse Module

Input Voltage Range

Note: The ***DAC2 DX*** is equipped with a universal power supply. There is no voltage selection switch. AC voltage range is 88-264 VAC, 50-60 Hz.

Power Cord

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

Caution: Always use a grounded power cord. The *DAC2 DX* 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.

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.

Internal Settings

Jumper-Configured Options

The following functions are jumper configured:

- XLR Output Pads
- Headphone Mute Switches
- Headphone Gain
- Digital Pass-Through
- Calibration Level

Removing Top Cover

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

Caution: The ***DAC2 DX*** contains static sensitive components. 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.

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

XLR Output 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 DX***. The ***DAC2 DX*** ships with the pads disabled (**0 dB** setting).

Tip: The XLR outputs are factory-preset to deliver professional studio levels. Most power amplifiers and powered monitors will require the use of the **10 dB** or **20 dB** pads.

Tip: Use the factory-default **0 dB** setting with Benchmark's ***AHB2*** power amplifier. When directly driving most other power amplifiers (or powered speakers), start with the **10 dB** pad setting. If necessary, change the pads so that normal listening levels are achieved when the **VOLUME** control is between the 11 o'clock and 3 o'clock positions.

When the output pads are enabled, the output impedance changes slightly, and the maximum recommended XLR cable length is reduced as shown in Table 1. The table assumes a cable capacitance of 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

XLR Output Pad Jumpers

Four jumpers on four 6-pin headers (**P8**, **P9**, **P10**, and **P11**) allow selection of the output level at the XLR jacks. The jumpers are properly configured if a normal playback level is achieved when the **VOLUME** control is set above the 11 o'clock position.

One pair of 6-pin headers control the attenuation at each XLR jack as follows:

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

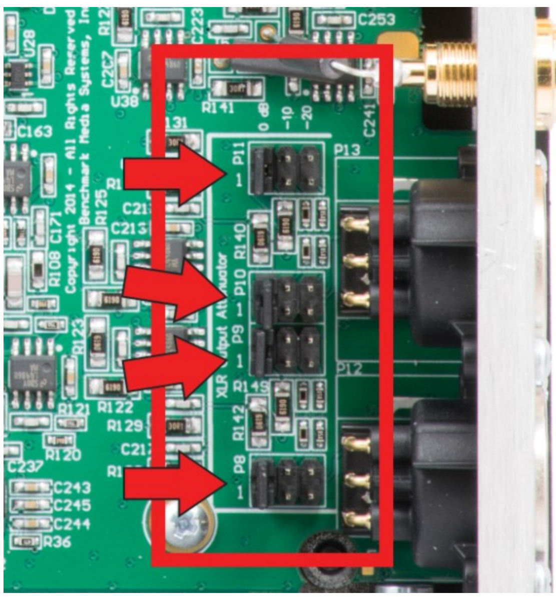


Figure 1 - Attenuators set to -0 dB

Headphone Switch Configuration

Both headphone jacks are equipped with switches that can mute the **MAIN** outputs. The **AUX** outputs are not muted by the headphone switches. The switches can be enabled by placing jumpers on header P1 (on the back side of the faceplate). By default, the mute switch is enabled on the left-hand jack and disabled on the right-hand jack. In this default configuration the left-hand jack automatically mutes the **MAIN** outputs. In most cases it is convenient to have one jack that mutes the outputs and one that does not mute the outputs. If your requirements are different the **HEADPHONE SWITCHES** can be enabled or defeated.

Headphone Switch Configuration

The **HEADPHONE SWITCH** on the left-hand headphone jack is enabled by placing a jumper across the bottom two pins of **P1** (on the back of the faceplate circuit board), as shown in **Figure 2**.

The **HEADPHONE SWITCH** on the right-hand headphone jack can be enabled by adding a jumper across the top two pins of **P1**. A spare jumper is supplied with the ***DAC2 DX*** for use at this location.

- No Jumpers - Both switches disabled
- Top Jumper Only - Right-hand switch enabled
- Bottom Jumper Only - Left-hand switch enabled (factory default)
- Top and Bottom Jumpers - Both switches enabled
- Jumper(s) rotated vertically - Storage of unused jumper(s)

Unused jumper(s) may be stored vertically on either pair of pins.

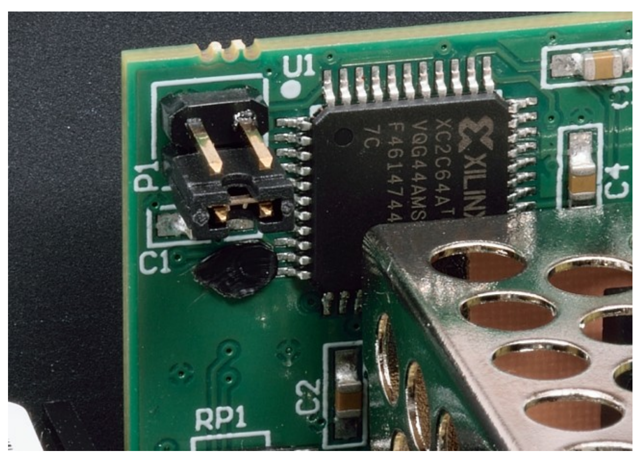


Figure 2 - Left-Hand Headphone Switch Enabled (Factory Default)

Headphone Amplifier Gain

The gain range of the ***HPA2***™ can be set using jumpers **JP3** and **JP4**:

- **Gain = 0 dB** - see **Figure 3**
- **Gain = -10 dB** - see **Figure 4** (Factory Default)
- **Gain = -20 dB** - see **Figure 5**

The jumpers change the gain of the ***HPA2***™ headphone amplifier without changing the output impedance. This keeps the output impedance of the ***HPA2***™ constant and very near 0 Ohms. External attenuators should never be inserted after a headphone amplifier as this would change the output impedance and alter the frequency response of the headphones.

Proper gain settings are important for maximizing the SNR of the headphone monitoring system. With proper settings, the full performance of the ***DAC2 DX*** can be delivered to the headphones for critical monitoring tasks and for maximum musical enjoyment.

Tip: When the headphone gain jumpers are set properly, a normal listening level will be achieved at a **VOLUME** control setting above the 11 o'clock position.

Tip: If a normal listening level is achieved below an 11 o’clock **VOLUME** setting, the headphone gain is too high, and the gain should be decreased.

Tip: If the level is too low at the maximum **VOLUME** setting, the headphone gain is too low, and the gain should be increased.

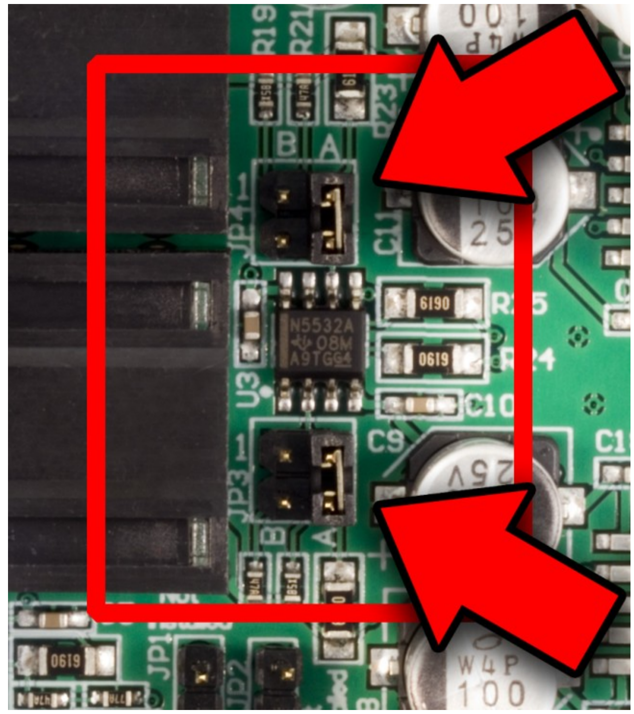


Figure 3 - *HPA2*™ Gain is 0 dB

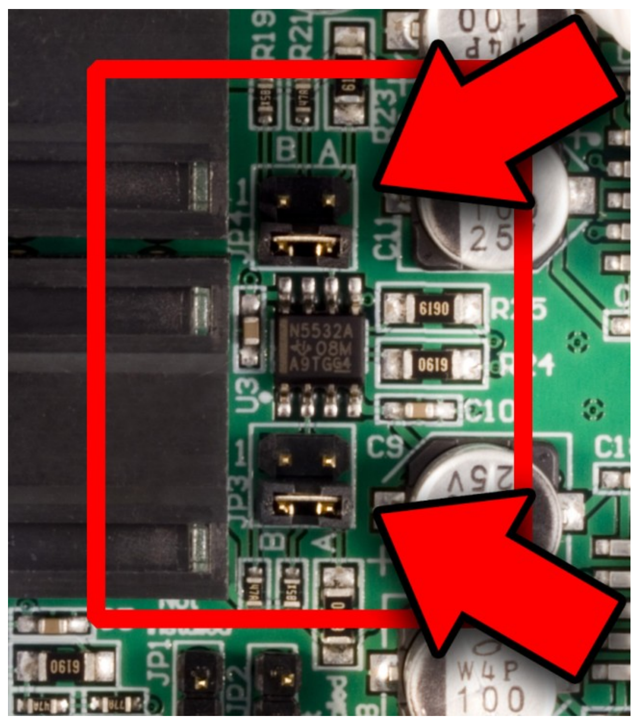


Figure 4 - *HPA2*™ Gain is -10 dB (Default)

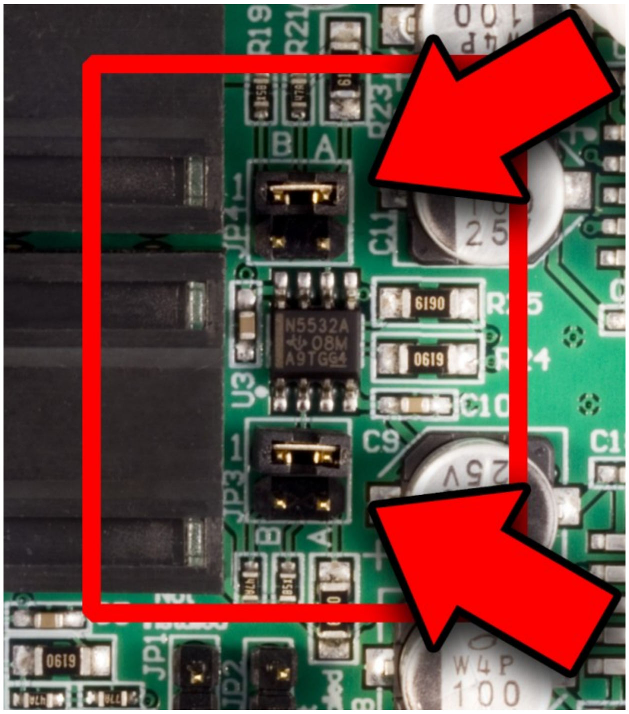


Figure 5 - *HPA2*™ Gain is -20 dB

Digital PASS-THROUGH Function

The digital **PASS-THROUGH** function can be enabled by moving both **P14** jumpers toward the faceplate shown in **Figure 6**. Once the jumpers are moved into the position shown in **Figure 6**, **D5** is configured as a digital audio output. When the **PASS-THROUGH** function is enabled, **D5** cannot be selected as an input. Any other selected digital input will be routed to both the internal D/A converter and to output **D5**. The digital output at **D5** is buffered, but is not processed. Many digital audio formats can be passed through to **D5** (even when these formats cannot be decoded by the ***DAC2 DX***).

By default, **D5** functions as a digital input and the jumpers are set according to **Figure 7**.

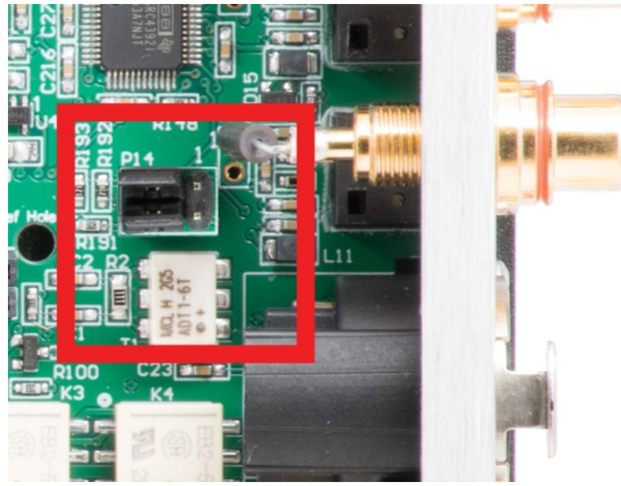


Figure 6 – Digital PASS-THROUGH Enabled

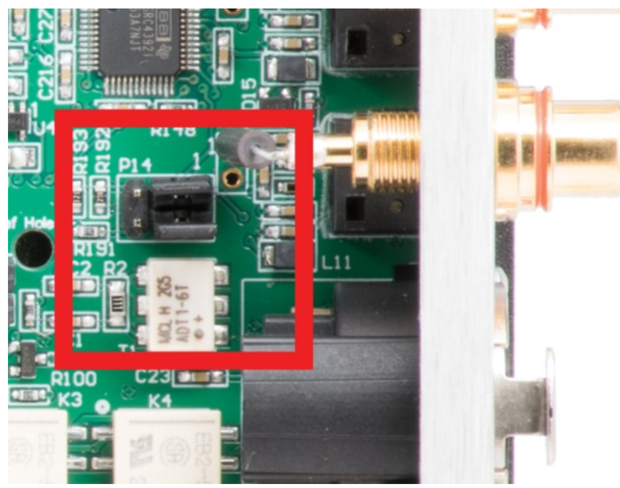


Figure 7 - Digital PASS-THROUGH Disabled (Factory Default)

Calibration Settings

The factory default calibrated output levels are +24 dBu at 0 dBFS on the XLR outputs and 2 V RMS at 0 dBFS on the unbalanced outputs. This calibration is activated when the **MAIN** or **AUX** busses are placed in **CALIBRATED** mode. It is also activated on the **MAIN** bus when the **COMPATIBILITY** mode is turned on.

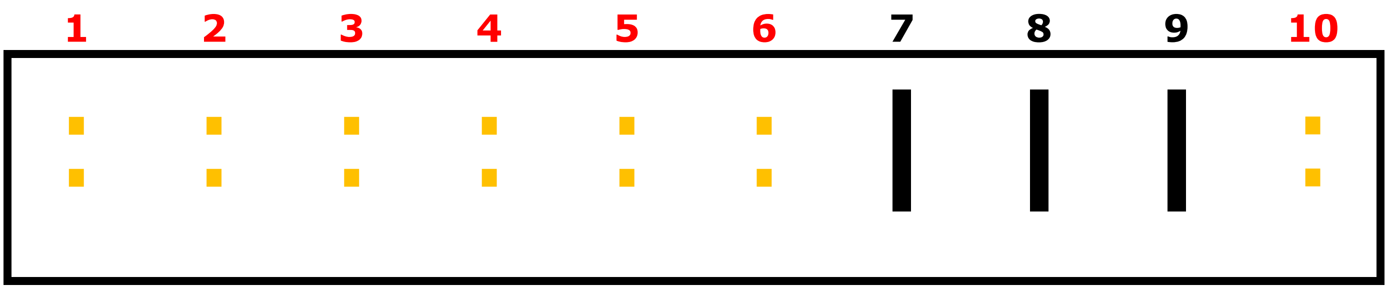
If you wish to change the calibration, you will need to remove or install jumpers on header P6. There are 10 pairs of pins on P6 that provide 10 jumper positions. Only use positions 7, 8 and 9!

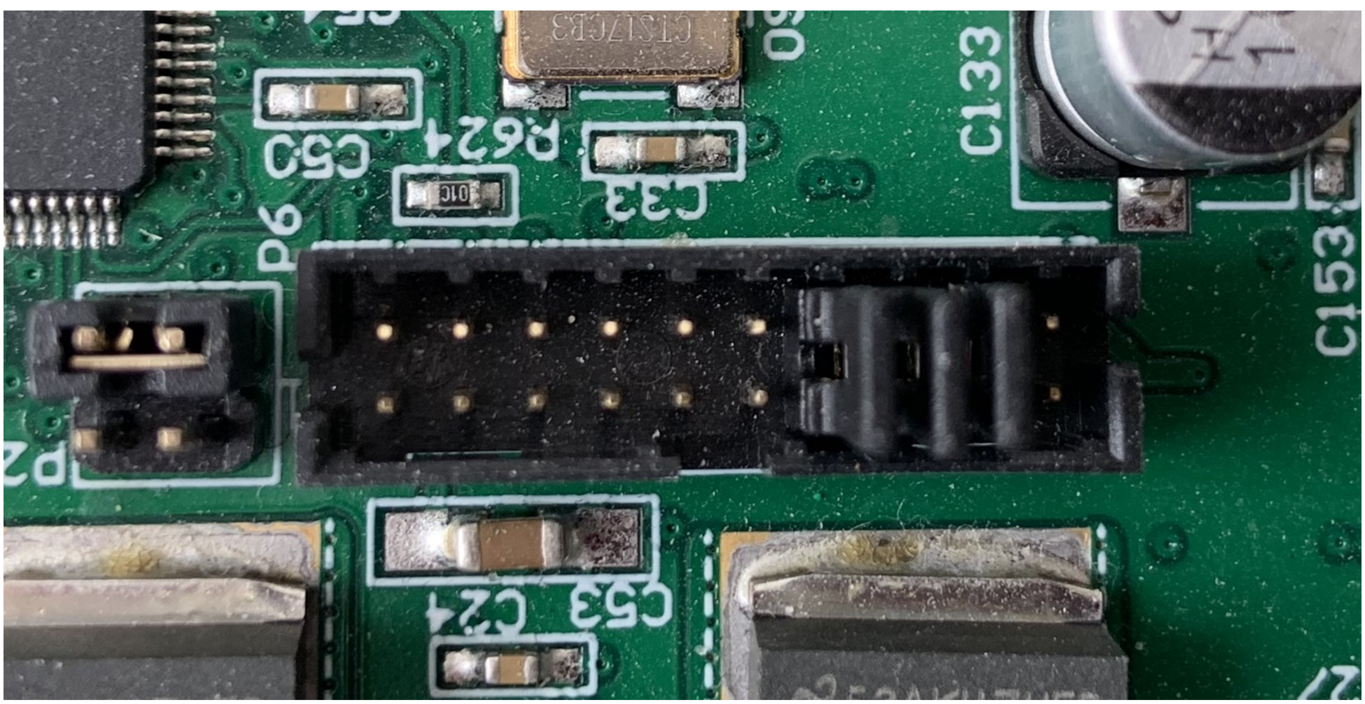
The factory default calibration is selected when all three jumpers are installed, or all three jumpers removed.

Caution: Do not install jumpers in positions 1, 2, 3, 4, 5, 6 or 10!

The chart and photo below show **P6** with jumpers installed in the factory-default positions (7, 8, and 9). The **P6** legend is located near jumper position 1 (see photo).

P6





Calibration Settings (continued)

When header P6 is viewed front the front of the unit, position 1 is on the far left and position 10 is on the right. Count the pins starting from the left side of the header. Position 7 is the 7th pair of pins. Position 8 is the 8th pair of pins. Position 9 is the 9th pair of pins. Do not install jumpers on any other pins on this header!

Use the following chart to set the P6 jumpers to the desired calibration level:

P6

1	2	3	4	5	6	7	8	9	10	XLR Level (dBu)	RCA Level (Vrms)
:	:	:	:	:	:	:	:	:	:	**+24**	**2.00**
:	:	:	:	:	:	:	:	|	:	**+20**	**1.26**
:	:	:	:	:	:	:	|	:	:	**+21**	**1.42**
:	:	:	:	:	:	:	|	|	:	**+22**	**1.59**
:	:	:	:	:	:	|	:	:	:	**+23**	**1.78**
:	:	:	:	:	:	|	:	|	:	**+26**	**2.52**
:	:	:	:	:	:	|	|	:	:	**+28**	**3.17**
:	:	:	:	:	:	|	|	|	:	**+24**	**2.00**

+28 dBu at 0 dBFS is not recommended due to a loss of digital headroom in the ESS DAC chip. Intersample peaks may clip.

Tip: The factory default setting can be obtained by installing all three jumpers or by removing all three jumpers.

Tip: The factory default setting is recommended when using the LA4 or HPA4 as a volume control.

Rack Mounting Options

The ***DAC2 DX*** is available with or without rackmount ears. Either version can be rack mounted with the appropriate accessories. There are several accessories available for rack mounting a ***DAC2 DX***:

- ***Connector Block*** - Joins two ½-wide Benchmark products when both have rackmount-type faceplates. It can also join a rackmount ***DAC2 DX*** directly to a blank plate (no tray required).
- ***Premium ½-Wide Blank Panel*** - Fills one side of the rack tray, or mounts directly to a rackmount ***DAC2 DX***. Panel is 1/4" brushed aluminum with a durable clear-anodized finish. This premium panel is styled to match the silver rackmount ***DAC2 DX***, and includes a large engraved Benchmark logo.
- ***Black ½-Wide Blank Panel*** - Fills one side of the rack tray, or mounts directly to a rackmount ***DAC2 DX***. Panel is 1/8" brushed aluminum with a durable black-anodized finish.
- ***Universal Rack Adapter Tray*** - Mounts two ½-wide Benchmark products with any combination of rackmount and non-rackmount faceplates.

Call us, contact your dealer*, or v*isit http://BenchmarkMedia.com to purchase accessories for your ***DAC2 DX***.

Rack-Mount Version of the *DAC2 DX*

Connector Block

The connector block joins two ½-wide Benchmark products with rack-mount type faceplates. The joined units fill a standard 19 inch wide 1-RU rack space. The connector block can also join a single ½-wide Benchmark product to a Benchmark ½-wide blank panel.

Premium ½-Wide Blank Rack Panel

This premium ¼" thick blank panel matches the finish on the silver rackmount version of the ***DAC2 DX*** and features an engraved Benchmark logo. It can also be used to cover an unused slot in the ***Universal Rack Adapter Tray***.

Black ½-Wide Rack Panel

This ⅛" thick blank panel matches the finish on the black non-rackmount version of the ***DAC2 DX.*** This panel can be used to cover an unused slot in the ***Universal Rack Adapter Tray***. It can also be mounted directly to a rackmount version of the ***DAC2 DX***.

Universal Rack Adapter Tray

The ***Universal Rack Adapter Tray*** is a tray that mounts up to two ½-wide Benchmark products in a standard 19 inch wide 1-RU rack space. The tray accepts any combination of ½-wide Benchmark products (with or without rack-mount type faceplates). The tray is not necessary if both units are equipped with rack ears.

Rack Mounting Example

Blank Plate and ***Connector Block***:

The ***Connector Block*** creates a rigid connection:

Ready to mount in a standard 19" rack:

***DAC1* and *DAC2* Family History**

The pristine audio performance of the award-winning ***DAC1*** made it the 'Benchmark' of stand-alone D/A converters. The ***DAC1 USB***, ***DAC1 PRE*** , and ***DAC1 HDR*** added many features to the basic ***DAC1*** platform. Benchmark converters are in use in many of the world's top studios.

The following is a brief overview of the various Benchmark ***DAC1*** and ***DAC2*** models:

***DAC1* Series**

DAC1

Benchmark's original ***DAC1*** converter.

The ***DAC1*** features included:

- Three digital inputs
- XLR outputs with passive pads
- RCA outputs
- Analog volume control
- ***HPA2™*** headphone amplifier
- ***UltraLock™*** jitter-attenuation system

DAC1 USB

The ***DAC1 USB*** introduced these improvements:

- ***AdvancedUSB™*** computer input
- Mute switch on the left headphone jack
- Two headphone gain ranges
- High-current LM4562/LME49860 output stages - designed to drive difficult loads
- Lower output impedances

Benchmark's ***AdvancedUSB™*** computer input was the first USB audio interface to support 96 kHz audio without the need to install special drivers.

DAC1 PRE

The ***DAC1 PRE*** added these improvements:

- Three coaxial digital inputs
- Three Headphone Gain Ranges
- LM4562/LME49860 opamps throughout
- Premium bulkhead-mounted RCA connectors

In order to provide room for the analog inputs, we removed the XLR digital input and replaced it with two additional coaxial digital inputs.

DAC1 HDR

The ***DAC1 HDR*** added:

- IR Remote Control
- ***HDR-VC™*** (high dynamic range volume control).

The ***HDR-VC™*** features a custom-made, motor-driven Alps potentiometer. The motor-driven control provides the audio performance of a manual control while adding the convenience of remote control.

***DAC2* Series**

DAC2 HGC

The ***DAC2 HGC*** maintains the familiar ½-wide ***DAC1*** form factor, but the entire product was redesigned from the ground up.

The ***DAC2 HGC*** features:

- Four 32-bit converters per channel
- Native 24-bit/192kHz PCM conversion
- Native 64X DSD conversion
- High-headroom digital processing
- ***UltraLock2™*** jitter attenuation
- Multi-mode asynchronous USB audio input
- Sample rate and word length displays
- Polarity control
- Home theater bypass
- Digital pass-through
- Bi-directional 12V trigger
- Two stereo analog inputs
- Three stereo analog outputs
- Two optical inputs
- High-efficiency low-noise power supplies

DAC2 L

The ***DAC2 L*** is identical to the ***DAC2 HGC*** except that the ***DAC2 L*** has no headphone amplifier.

DAC2 D

The ***DAC2 D*** is identical to the ***DAC2 HGC*** except that the ***DAC2 D*** has no analog inputs and no 12V trigger.

DAC2 DX

The ***DAC2 DX*** replaced the ***DAC2 D.*** The new model added an XLR digital input and the 12V trigger. It also added a second output bus so that one set of outputs could be placed in calibrated mode while the other was controlled by the volume knob.

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 (included on ***DAC2 HGC*** and ***DAC2 L*** models) are never converted to digital. On all ***DAC2*** models, 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*** and ***DAC2*** 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 in your ***DAC2*** will make full use of your power amplifier's dynamic range. Experience newly revealed details in your favorite recordings.

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

Native DSD Conversion

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

High-Headroom DSP

All of the digital processing in the ***DAC2*** is designed with a headroom of 3.5 dB above 0 dBFS. A sinusoid that just reaches the maximum positive and negative digital codes has a level of 0 dBFS. If the peaks of the sinusoid occur between samples, higher signals can be captured without clipping. For a pure tone, the maximum intersample peak that can be represented by a PCM system is +3.01 dBFS (see **Figure 8**).

Benchmark's high-headroom DSP can handle intersample peaks without overloading or clipping. Intersample peaks are cleanly rendered by the ***DAC2*** and are delivered to the analog outputs without clipping or distortion (see **Figure 10**). **Very few D/A converters can make this claim!**

In most D/A conversion systems, intersample peaks cause overloading of the upsampling interpolators and digital filters that are found in all sigma-delta converters (see **Figure 9**). When overloads occur, bursts of non-harmonic distortion are produced. These bursts of high-frequency distortion may occur many times per second and may add a false brightness and harshness to the sound. This defect impacts PCM formats but does not impact 1-bit DSD formats. The absence of intersample clipping may explain some people's preference for DSD. The ***DAC2*** delivers clean PCM conversion that meets or exceeds the clarity of DSD.

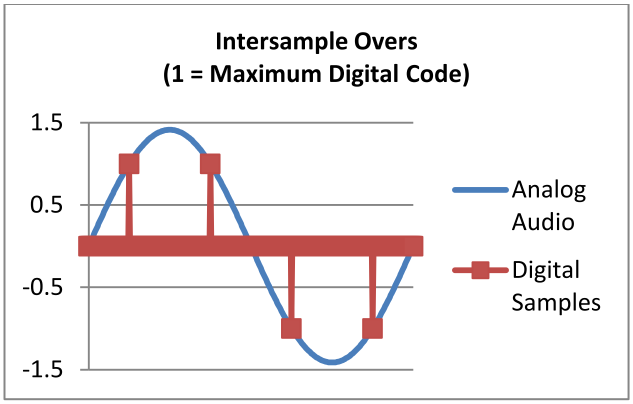


Figure 8 - Intersample Over at +3.01 dB

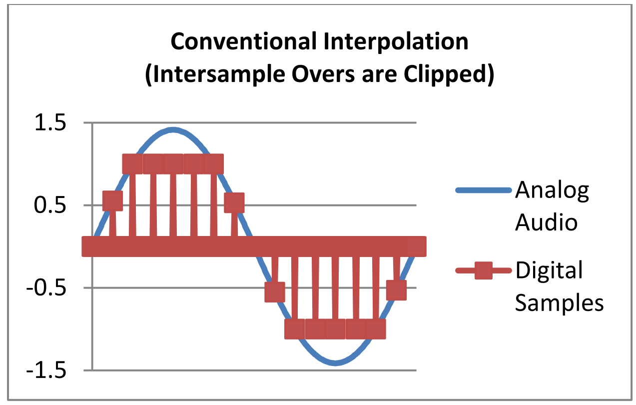


Figure 9 - Clipped Intersample Overs

Intersample overs are common in low sample rate (44.1 kHz and 48 kHz) commercial releases. Due to the mathematics and the bandwidth of typical input signals, intersample overs are less of a problem in high sample rate recordings. The reason for this is that the worst-case (+3.01 dB) intersample overs occurs for pure tones that are exactly 1/4 of the sample rate (see **Figure 8**). At the 44.1 kHz CD sample rate, the worst case occurs at 11.025 kHz. It turns out that many recordings have substantial peaks near this frequency. In contrast, at a sample rate of 88.2 kHz, the worst-case intersample overs occur at a frequency of 22.1 kHz where most musical sources have insufficient energy to produce significant intersample overs. The 88.2 kHz sample rate is still susceptible to intersample overs, but the magnitude of the worst-case overs tends to be much lower. For example, at 1/8 of the sample rate (11.025 kHz), the maximum intersample peak is about +0.66 dB instead of the 3.01 dB worst case at a sample rate of 44.1 kHz.

The biggest advantage of higher sample rates may be the immunity to intersample overs. If higher sample rates sound better, this difference may be entirely due to the absence of DSP overloads caused by intersample overs. Benchmark's high-headroom DSP (**Figure 10**) renders low sample rates with the clarity and detail normally associated with high sample rates.

PCM systems can accurately capture peaks that exceed 0 dBFS, but these peaks will overload the oversampling interpolators in most delta-sigma D/A converters. The solution is not to eliminate the interpolation process; the solution is to build interpolators with more headroom.

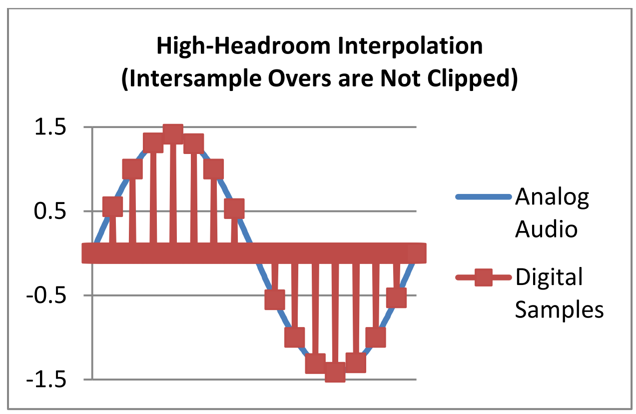


Figure 10 - High-Headroom Interpolation

The interpolation process is absolutely necessary to achieve 24-bit state-of-the art conversion performance. Unfortunately, most interpolators clip! 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 brightness to digital reproduction. Benchmark ***DAC2*** converters have high-headroom non-clipping interpolators. We believe these interpolators provide groundbreaking improvements to digital to analog conversion.

32-bit ESS SABRE D/A System

Eight balanced 32-bit D/A converters deliver audio to Benchmark's low-impedance current to voltage converters. Two converters drive the two **AUX** output channels. The remaining six converters drive the two **MAIN** output channels. The 4:1 redundancy on the **MAIN** outputs reduces noise by about 4.8 dB. This redundancy also reduces the THD. 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 Displays

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

Many disk players also downsample all sources to 44.1/16. This processing can do significant damage to the audio quality.

The sample-rate and word-length displays on the ***DAC2*** confirm the proper operation of your disk player, media player, and computer.

Bi-Directional 12 Volt Trigger

Benchmark has re-invented 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.

Benchmark components can communicate bi-directionally on the trigger I/O ports. This bidirectional communication provides greater flexibility. In a given system, the power button on any Benchmark device can be used to start or stop the entire audio system in a sequenced manner.

Distributed Power Regulation

To achieve the lowest possible noise, the ***DAC2*** uses distributed power supply regulation. There are 20 voltage regulators in the ***DAC2***. Each critical subsystem has at least one dedicated low-noise voltage regulator.

We have created two discrete ultra low-noise regulators for the ES9018 D/A converter. We use separate regulators for the left and right channels. This Benchmark exclusive feature improves the noise and THD performance of the already-outstanding ES9018 converter chip.

HPA2™ Headphone Amplifier

The ***DAC2*** headphone output is driven by Benchmark's signature ***HPA2***™ headphone power amplifier. This high-current, high-output amplifier has an output impedance that is nearly 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 on a test bench when driving resistive loads. However, the same amplifier will measure very poorly when driving a headphone load. Unfortunately, most manufacturers do not measure or specify headphone amplifier performance when loaded with real headphones. The measurements use 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 have 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.

Jitter-Immune *UltraLock2™*

Benchmark's ***UltraLock2™*** clock system is featured in the ***DAC2***. The 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 while achieving instantaneous locking.

The Importance of Eliminating Jitter

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 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), the conversion clock in a digital-to-analog converter (D/A), or the rate estimator in an asynchronous sample rate converter (ASRC).

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 often 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. Two-stage PLL circuits may fail to lock when jitter is too high, or when the reference sample frequency has drifted.

UltraLock™ converters exceed the jitter attenuation performance of two-stage PLL converters while achieving near instantaneous lock time. They 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.

The ***UltraLock™*** system is so effective that no jitter-induced artifacts could be detected using an Audio Precision System 2 Cascade test set while the inputs to the ***DAC2*** were exposed to high levels of interface jitter. The measurement limits included the ability to detect artifacts as low as –144 dBFS, but none could be detected, even 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 in the ***DAC2*** will be reproduced without the addition of any measurable jitter artifacts.

Benchmark's ***UltraLock™*** technology eliminates jitter-induced performance problems. ***UltraLock™*** 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 <u>any</u> measurable effect on the conversion clock of an ***UltraLock™*** converter. In an ***UltraLock™*** 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 ***UltraLock™*** 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 while the converter plays a pure high-amplitude tone. We typically use a full-scale 10 kHz test tone to test for the presence of jitter-induced side bands (**Graph 15 - JITTER TOLERANCE FFT**). This FFT shows that the ***DAC2*** is free from any jitter-induced sidebands to a measurement limit of about -144 dB relative to the level of the test tone. The graph plots the output spectrum of the ***DAC2*** when exposed to 31 different jitter frequencies ranging from 100 Hz to 100 kHz. All 31 output spectra are identical and are free from any signs of jitter-induced distortion.

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, the 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 ***UltraLock™*** 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 ***UltraLock™*** 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 insurance against the ill effects of jitter.

UltraLock™ 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:

UltraLock™ 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 jitter-induced distortion that happened inside the A/D converter. 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*** system supports the industry standard **USB Audio 1.1** and **USB Audio 2.0** protocols. These audio protocols should not be confused with USB port speeds. The **USB Audio 1.1** protocol requires at least a USB 1.0 (Full Speed) port. **USB Audio 2.0** requires at least a USB 2.0 (High Speed) port.

Benchmark's **USB Audio 1.1** and **USB Audio 2.0** modes are frequency agile. This means that the sample rate is controlled by the computer. The ***DAC2*** will follow sample rate changes initiated by the computer and/or the media playback software.

Tip: The **USB Audio 1.1** protocol will run on any USB port, but if it is run on a USB 1.0 (Full Speed) port, it will require all of the available bandwidth to support 96 kHz sample rates. If a USB 1.0 port is the fastest port available, make sure that there are no other devices sharing the USB hub that services the port. When possible, connect the ***DAC2*** to a port that supports USB 2.0 or higher.

Tip: The **USB Audio 2.0** protocol is required for sample rates above 96 kHz. This protocol will not run on USB 1.0 ports. When possible, connect the ***DAC2*** to a port that supports USB 2.0 or higher.

Asynchronous USB

In all modes the USB communications are asynchronous. An ultra low jitter conversion clock is generated inside the ***DAC2***. The asynchronous USB interface pulls data from the computer without using computer-generated clocks. The D/A conversion in the ***DAC2*** is completely isolated by the asynchronous USB interface and by the ***UltraLock2™*** jitter-attenuation system.

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 detectable at our measurement limits (about -144 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.

The 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. In many cases, audio will automatically be routed to the newly connected device. If not, it can be selected as the current or default playback device.

The industry-standard **USB Audio Mode 2.0** mode is natively supported by Windows operating systems starting with Windows 10. A driver is required for older Windows operating systems. The driver supports Windows XP, Vista, or 7, and 8. This driver is required for DSD and sample rates above 96 kHz when using these older Windows systems.

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

The Windows **USB Audio 2.0** driver is available at:
BenchmarkMedia.com/drivers

USB Driver Installation

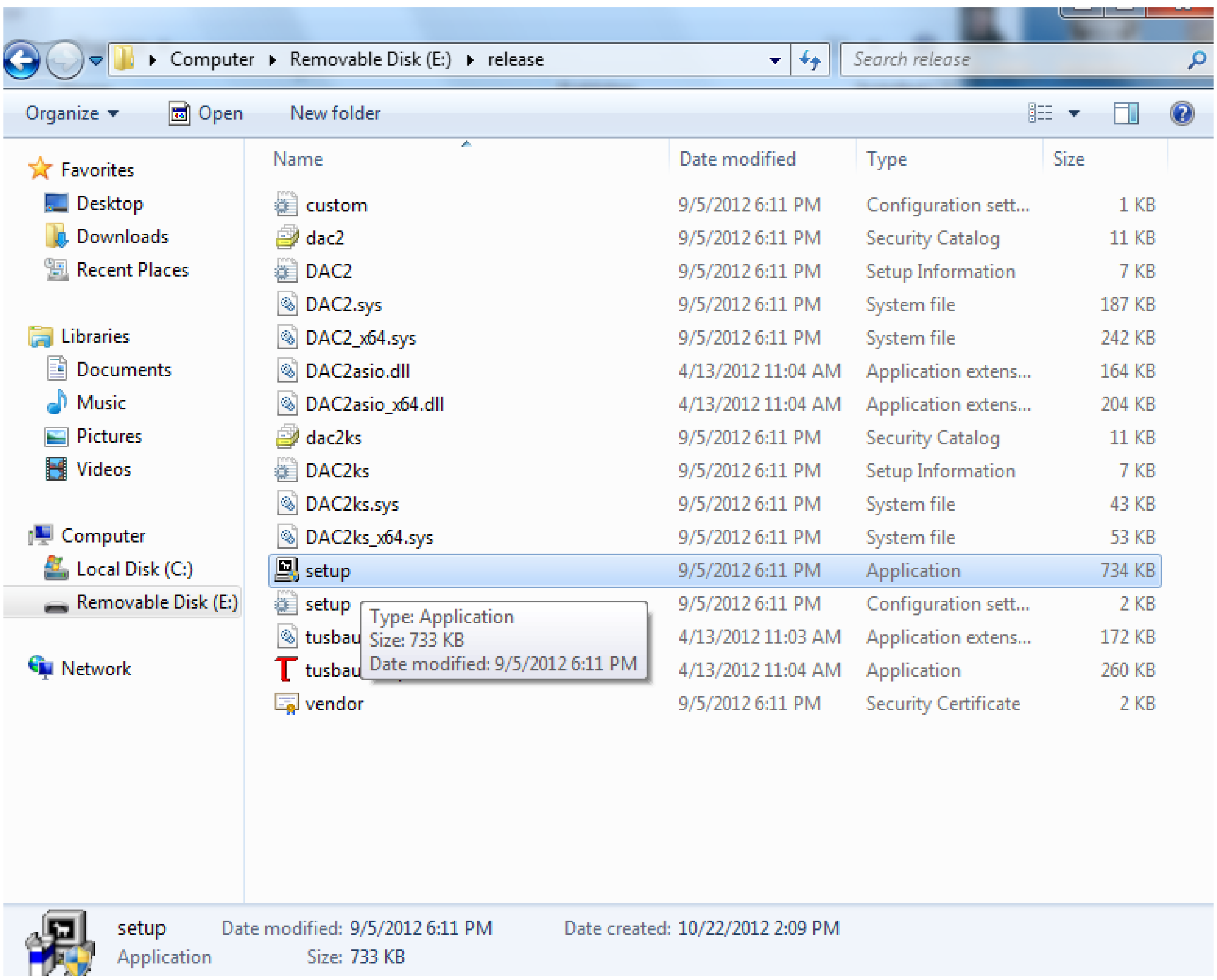
Required for Windows operating systems prior to Windows 10 only!

Note: The ***DAC2*** driver is available for download at: BenchmarkMedia.com/drivers

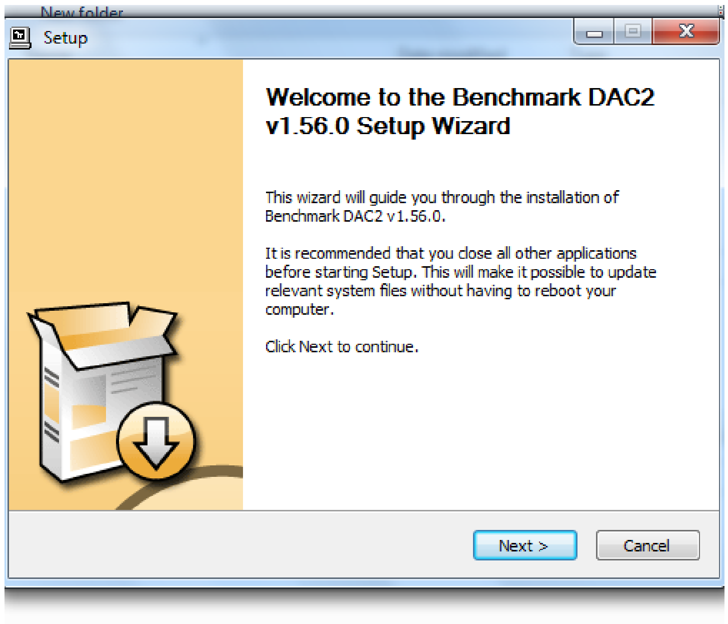
Note: To provide full backward and forward compatibility, the ***DAC3*** uses the ***DAC2*** USB drivers. This prevents the need to install two different sets of drivers. Please note that the ***DAC3*** USB input will be identified as "Benchmark DAC2" in your computer control panels. This is intentional.

Before you install the driver, make sure the USB cable is unplugged.

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

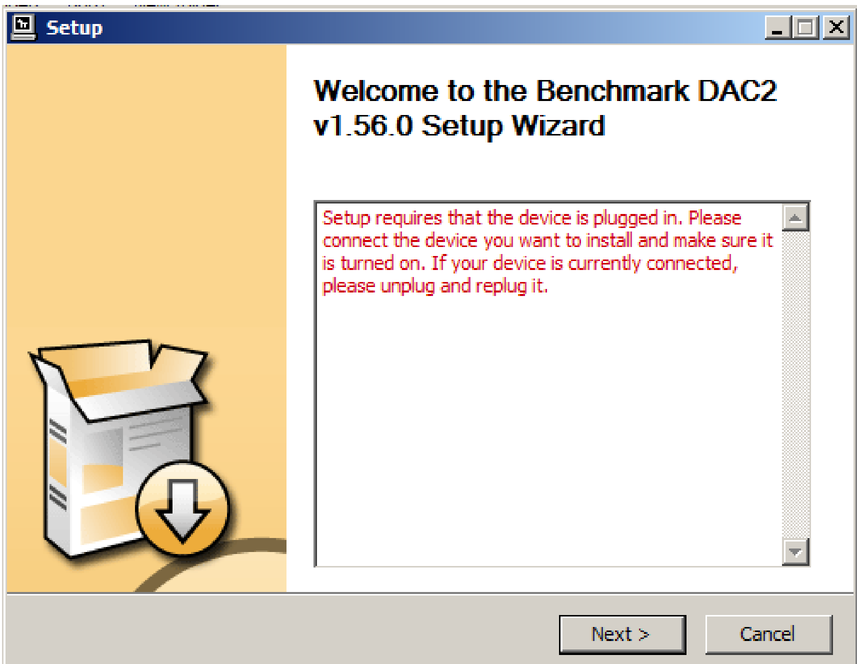


2. A welcome screen will pop-up. Click "*Next.*"

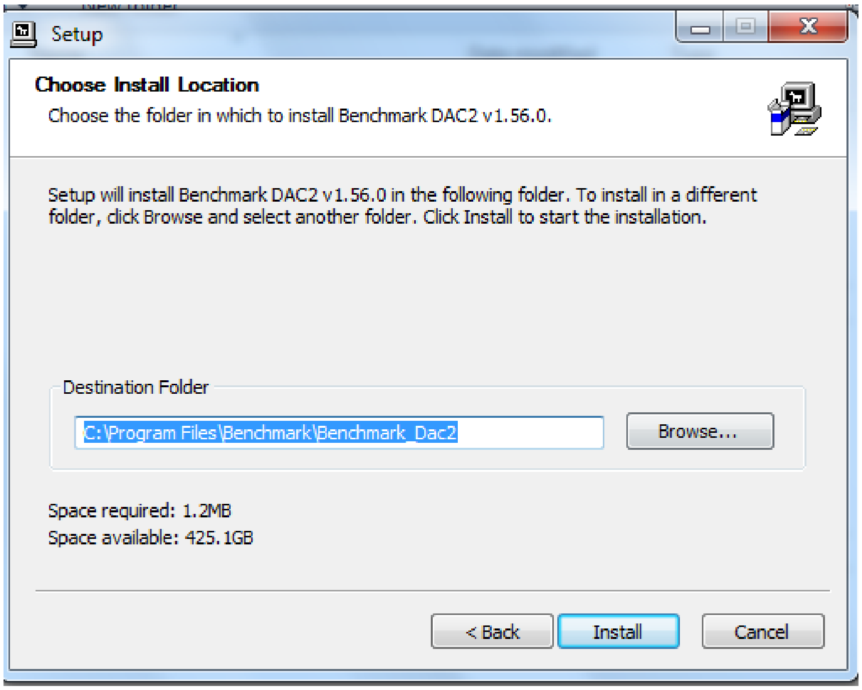


3. When you see the following screen, turn on the ***DAC2***, plug in the USB cable, and select the **USB** input on the ***DAC2***. By default, the ***DAC2*** is shipped in **USB Audio 1.0** mode. Do not proceed to the next step until the **USB Audio 2.0** mode has been enabled. You can enable the **USB Audio 2.0** mode using one of the following two methods:

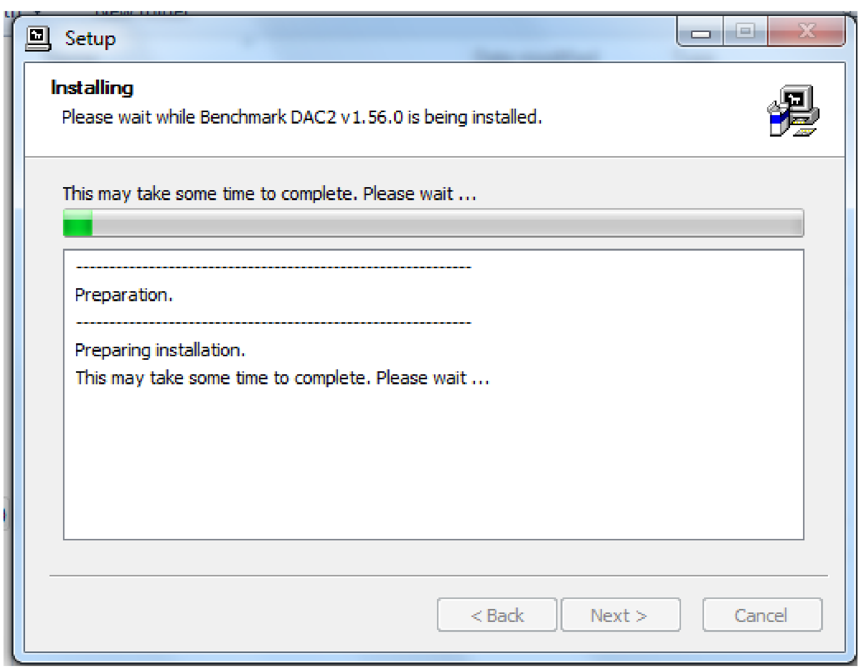
    - **METHOD 1 - REMOTE CONTROL:** Using your remote control, press and hold USB button for 10 seconds (the **4X** light should flash once). If the **2X** light flashes instead of the **4X** light, repeat this step.
    - **METHOD 2 - FRONT PANEL:** From the front panel, select the **USB** input and then hold the lower **INPUT** button for 10 seconds (the **4X** light should flash once). If the **2X** light flashes instead of the **4X** light, repeat this step.



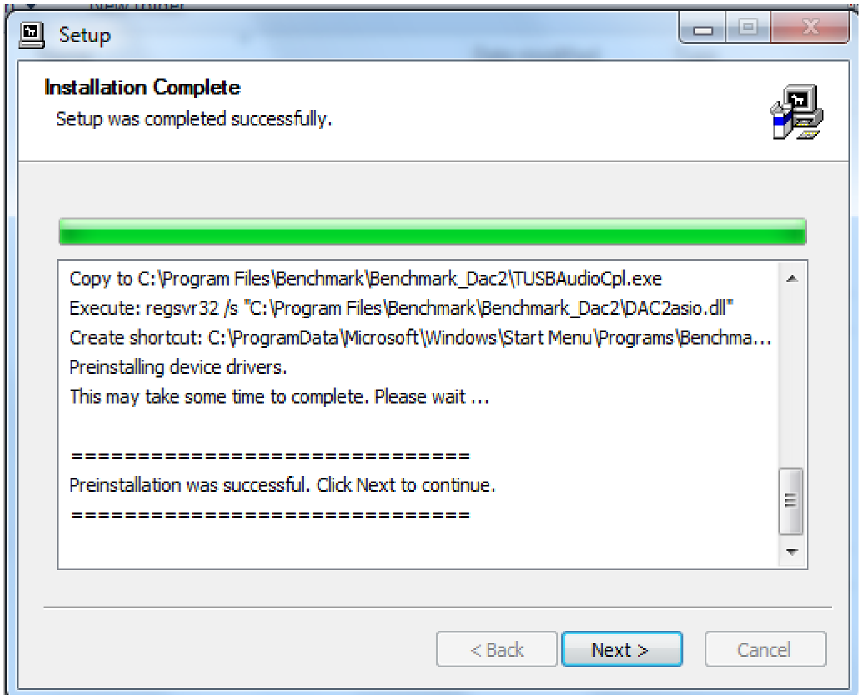
4. You will now be prompted to select a location to install the driver. It will default to your Program Files folder. If you wish to install it another location, you can change the location. We suggest keeping it in the default location. Click “Install”.



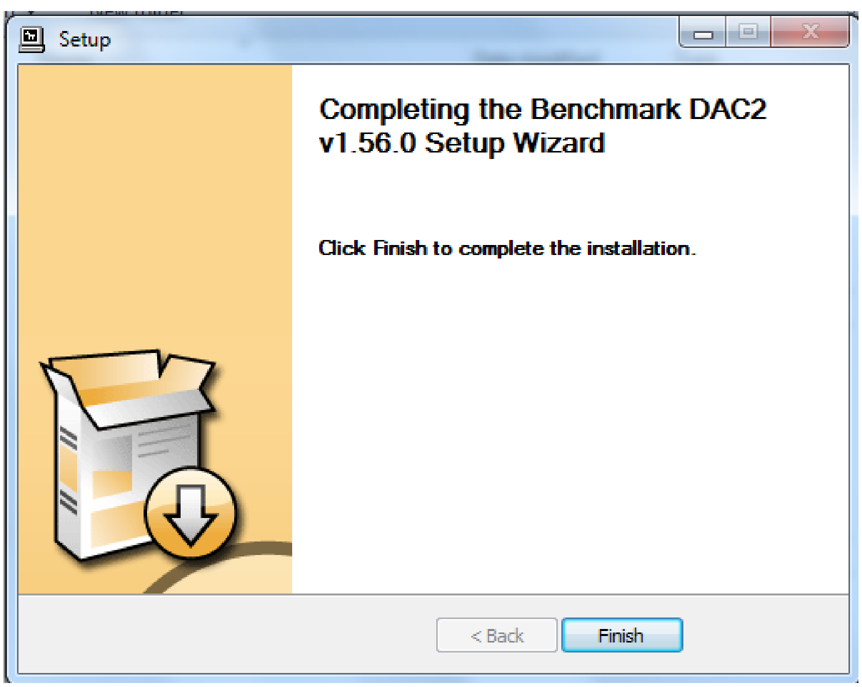
5. When the installation begins you will see the following screen. Please be patient while the driver installs. Installation may require several minutes.



6. When the installation finishes a message at the top will say “Installation Complete.” Click “Next” to continue.

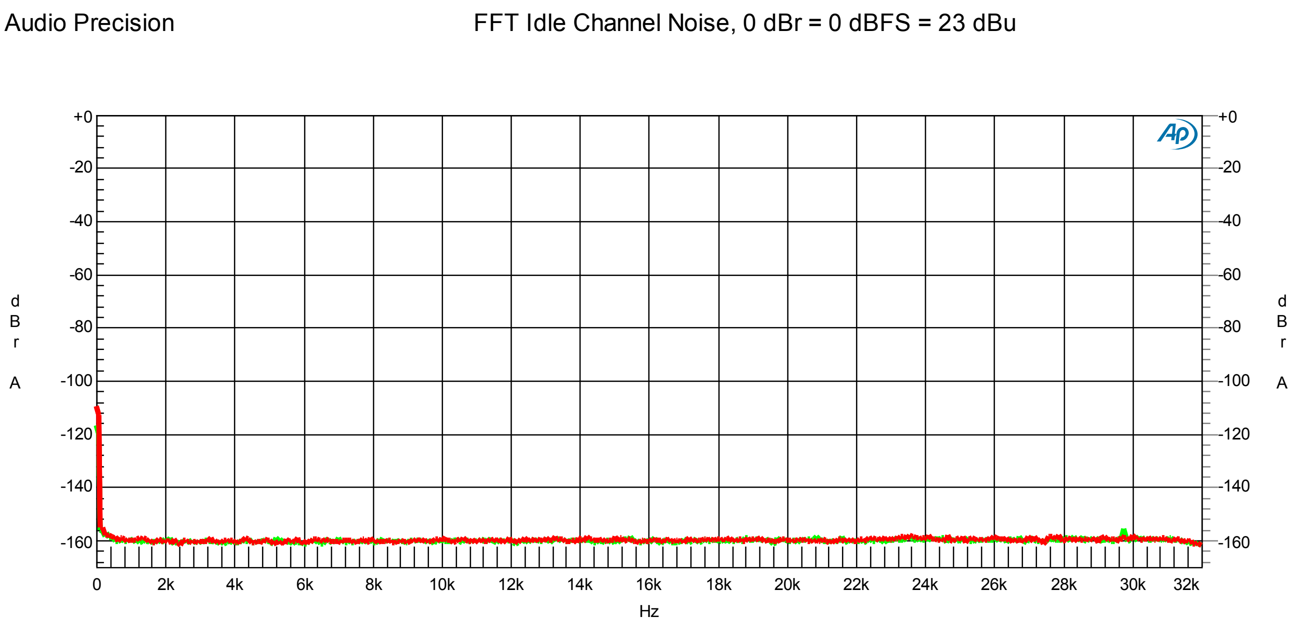


7. Click “Finish.” The Setup will close automatically. This completes the installation process. You can now enjoy music over the **USB Audio 2.0** connection at sample rates up to 192 kHz. DSD can also be played in DoP 1.1 format.



Performance Graphs

The following graphs were produced using a ***DAC2 HGC***. The ***DAC2 DX*** provides similar performance.



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	

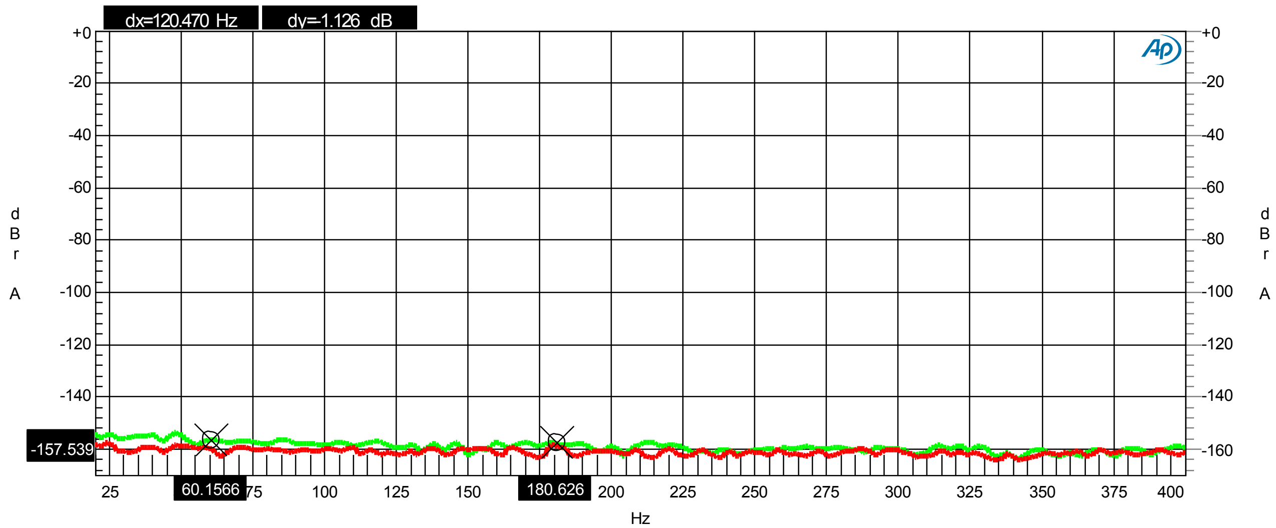
DAC2 - FFT Idle Channel Noise.at27

Graph 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



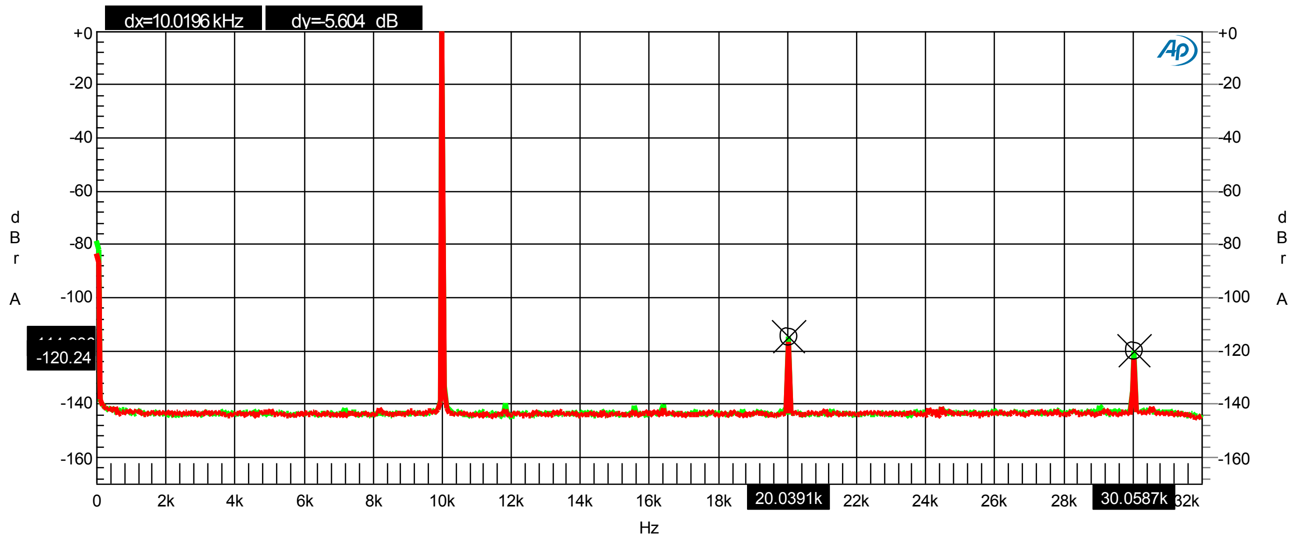
Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left		*-156.413 dBr	*-157.539 dBr
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right		-159.813 dBr A	-158.700 dBr A

DAC2 - FFT Idle Channel Noise - Low Frequency.at27

Graph 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

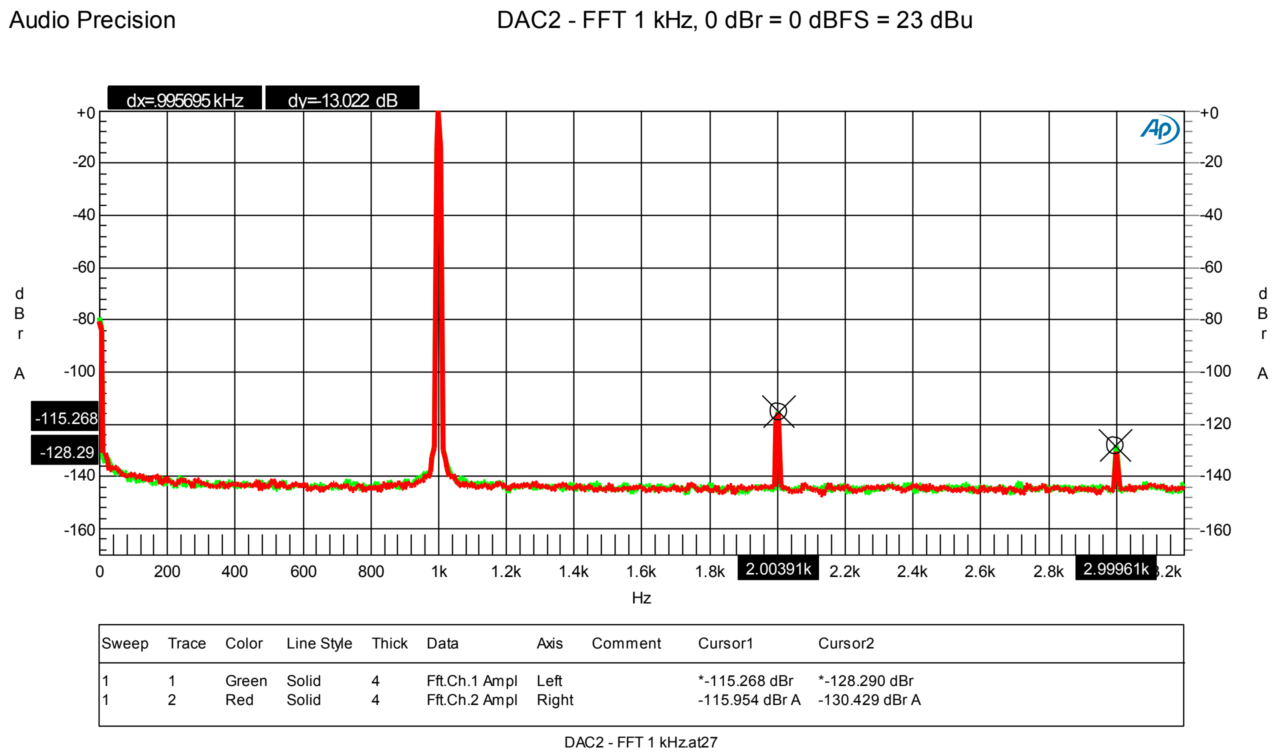
DAC2 - FFT 10 kHz.at27

Graph 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 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.

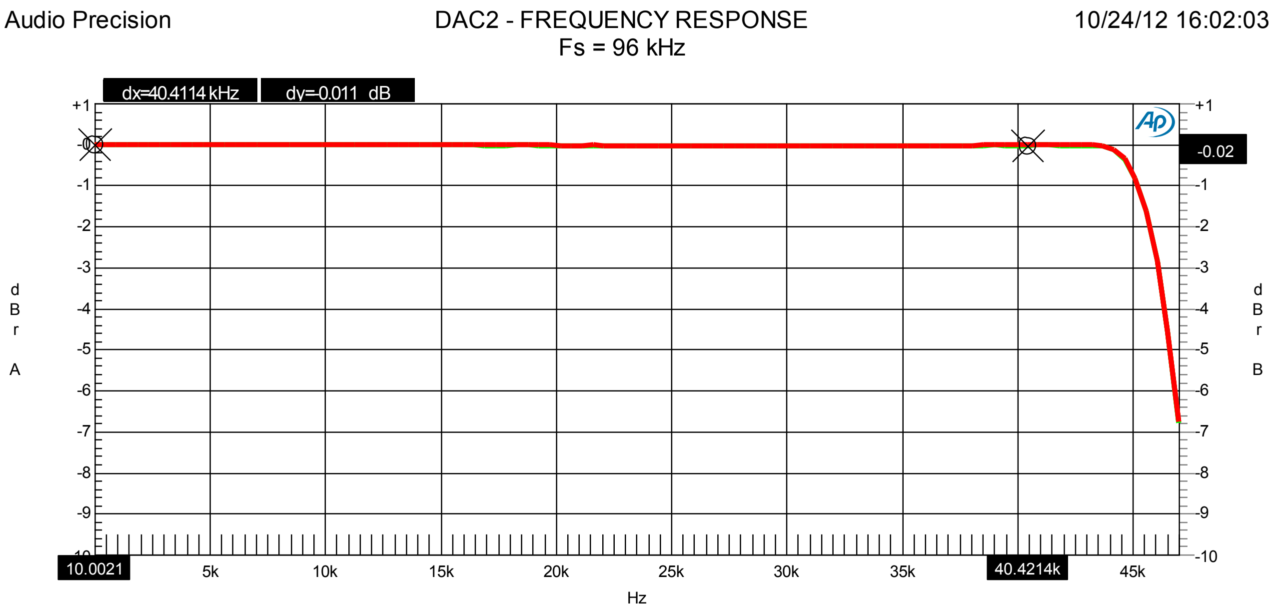
The rise at 0 Hz is normal in an FFT analysis and is not an indication of noise. This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging, and spans a frequency range of DC to 32 kHz.



Graph 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.

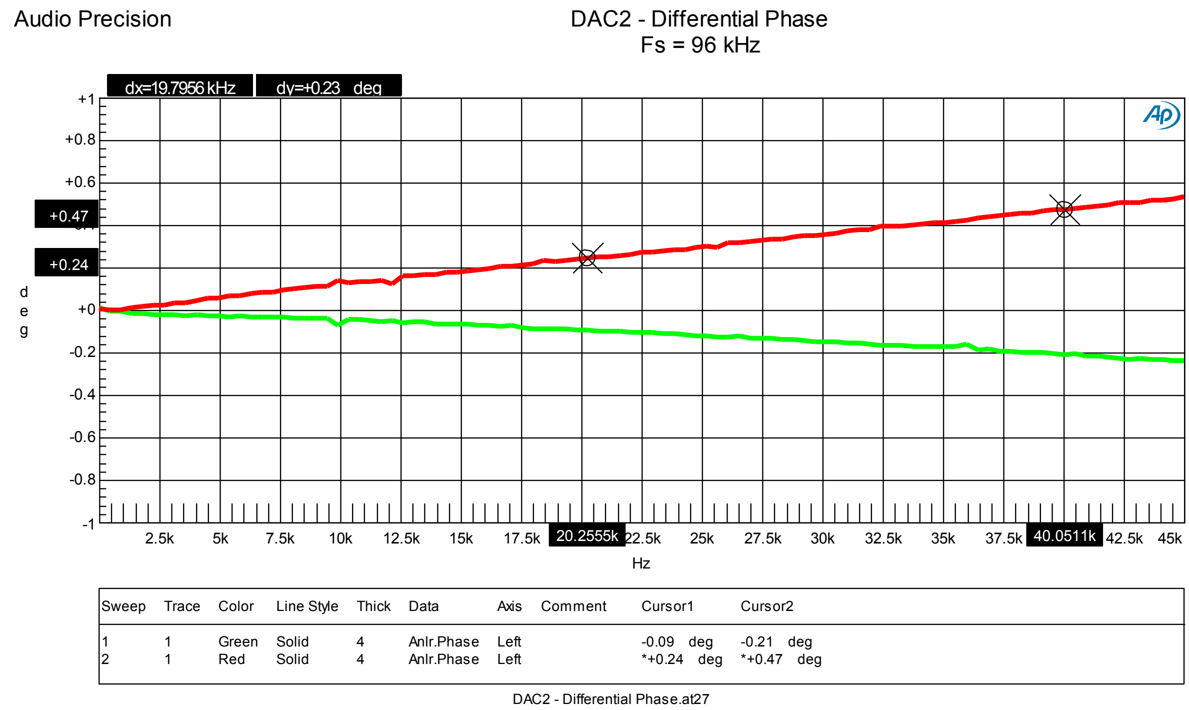


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Anlr.Level A	Left		-0.011 dBr A	-0.026 dBr A
1	2	Red	Solid	4	Anlr.Level B	Right		*-0.009 dBr	*-0.020 dBr

DAC2 - Frequency Response.at27

Graph 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.

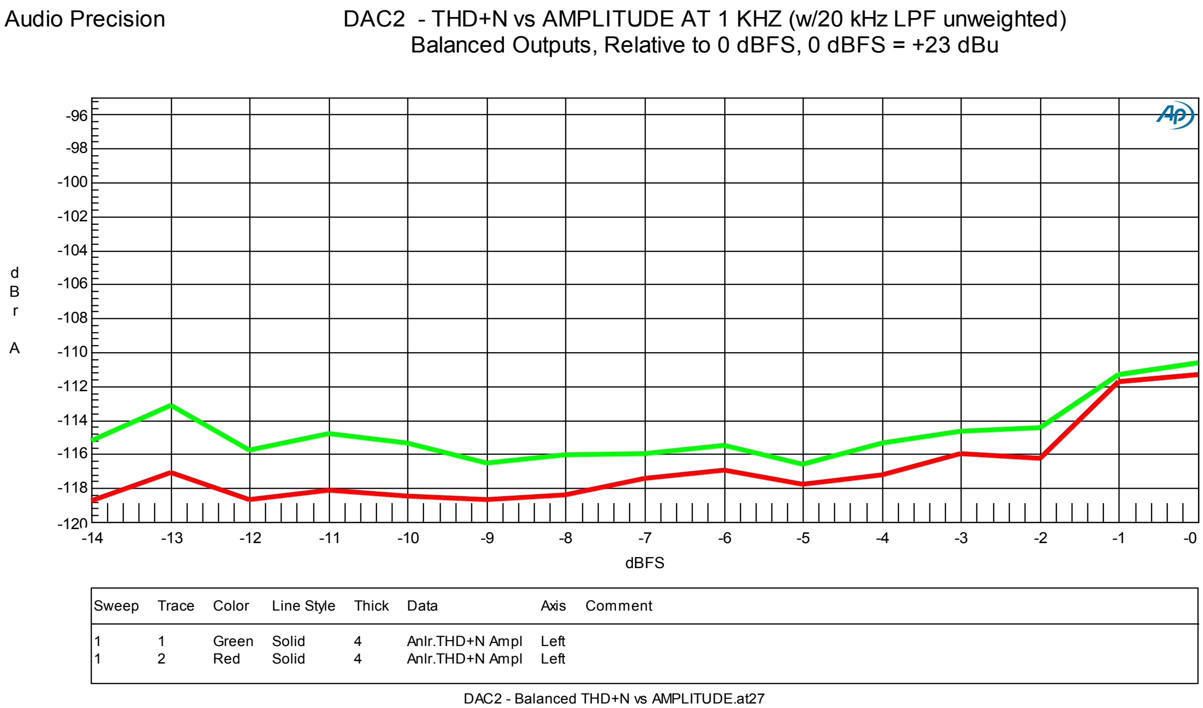


Graph 6 - Differential Phase

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.

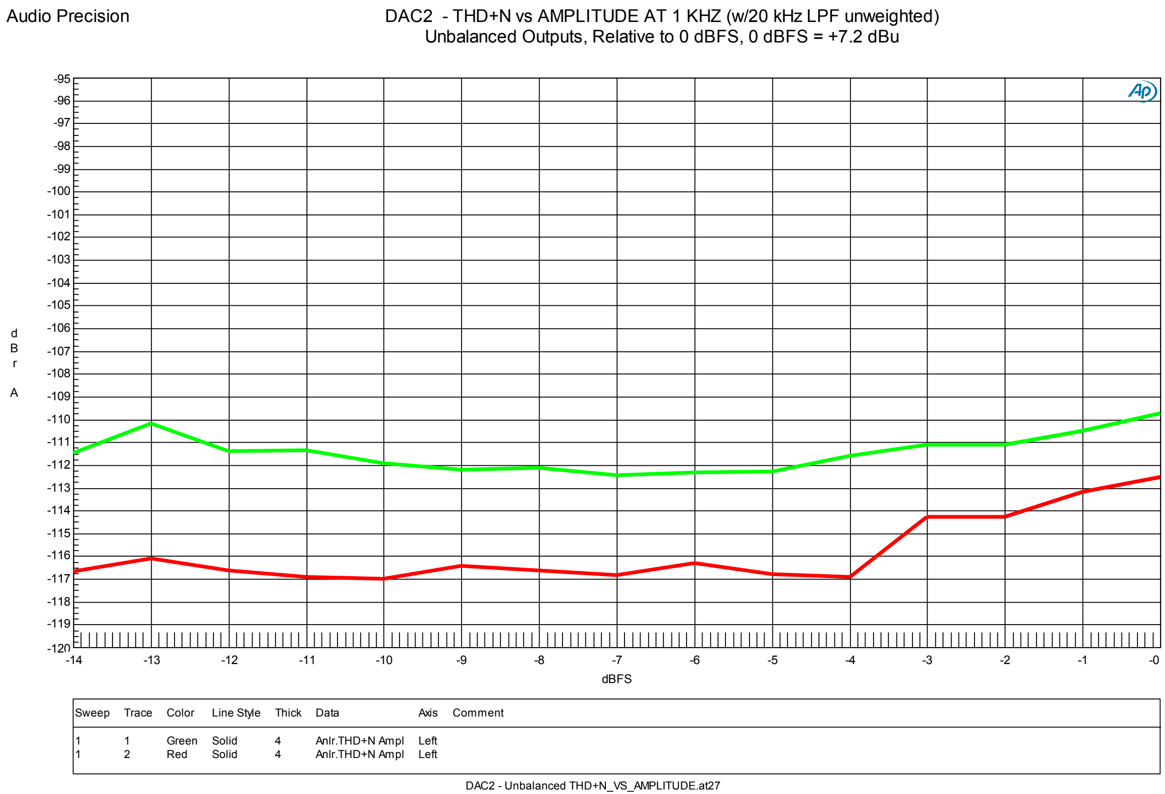
DAC2 converters are phase-accurate between boxes. Three ***DAC2*** converters can be used as a group to form a phase-accurate 5.1 surround system. Four ***DAC2*** converters can be combined to form a 7.1 system. The phase accuracy between any two channels will match the phase accuracy shown above.



Graph 7 - Balanced THD+N vs. Amplitude

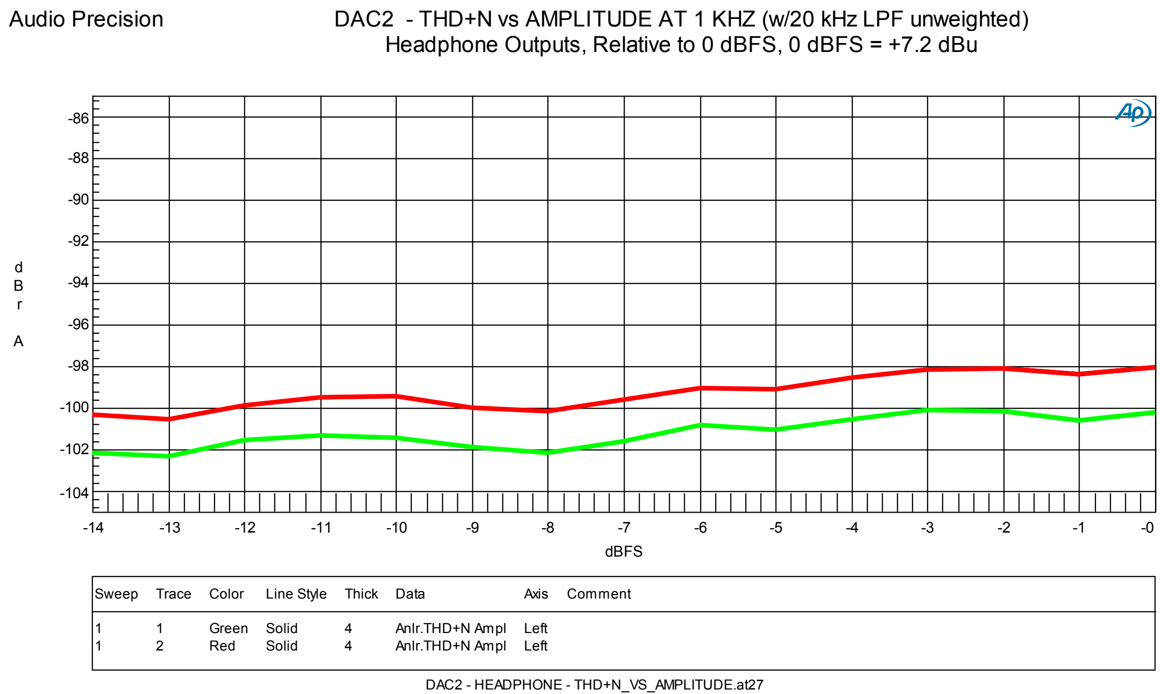
This plot demonstrates the very low harmonic distortion produced by the ***DAC2*** at signal levels ranging from -14 dBFS to 0 dBFS. All variations below -4 dBFS are due to the measurement limits of the AP 2700 system. The peaks at -13 dBFS are due to an AP 2700 gain range change. In almost all listening 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.



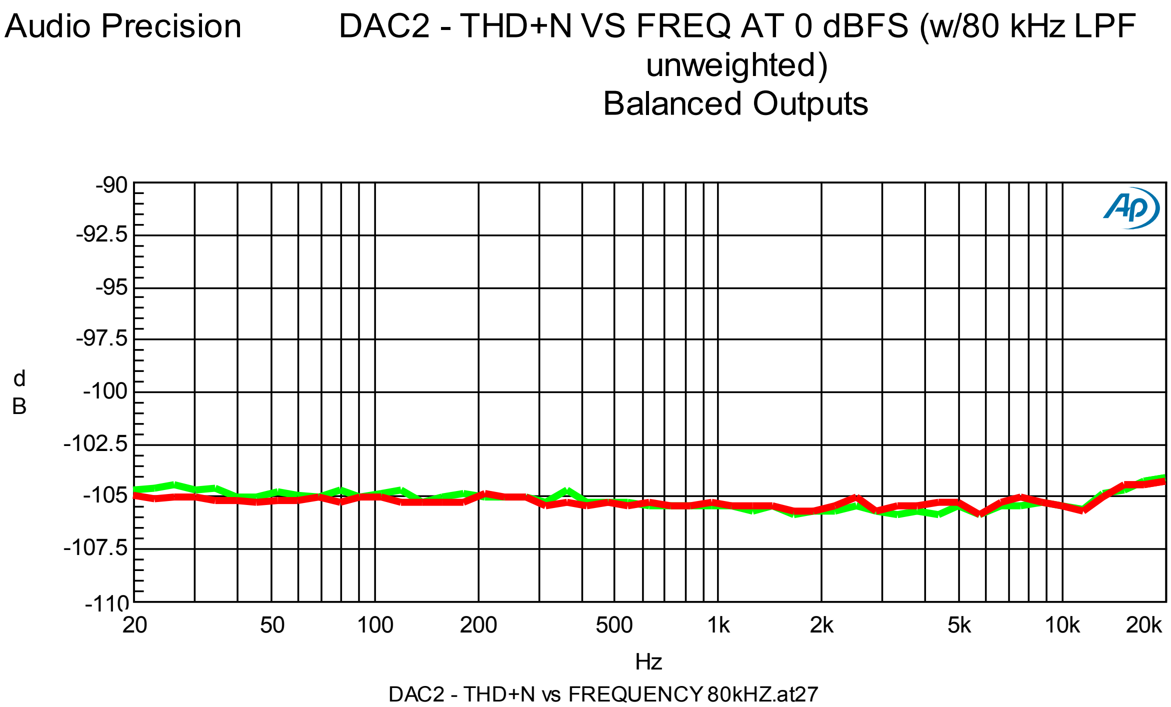
Graph 8 - Unbalanced THD+N versus Amplitude

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.



Graph 9 - Headphone Amplifier – THD+N versus Amplitude

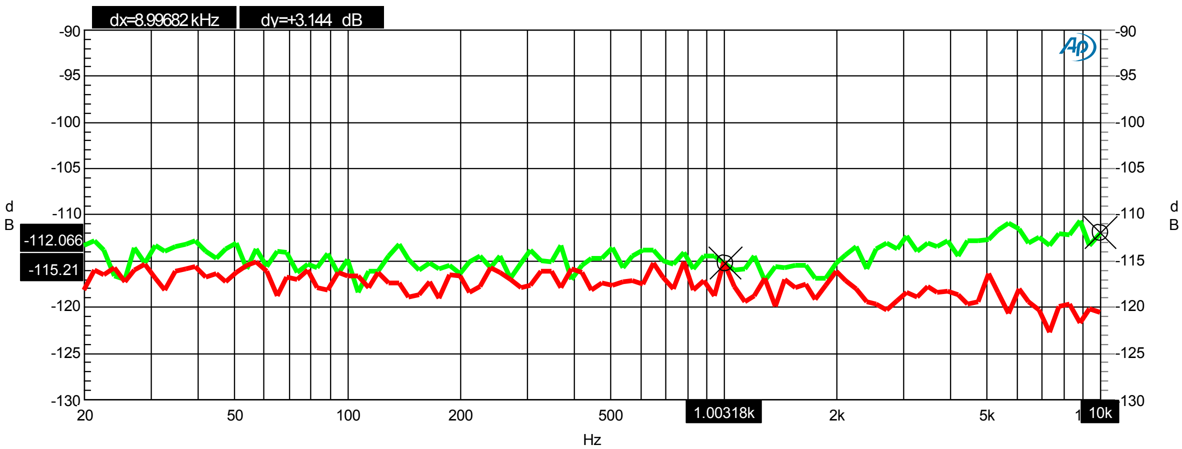
This plot shows the THD+N performance of the headphone outputs under load. Note that the THD+N performance of the headphone outputs approaches that of the balanced outputs. The ***DAC2*** includes Benchmark's ***HPA2™*** headphone amplifier. The ***HPA2™*** has a near 0-Ohm output impedance which provides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



Graph 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

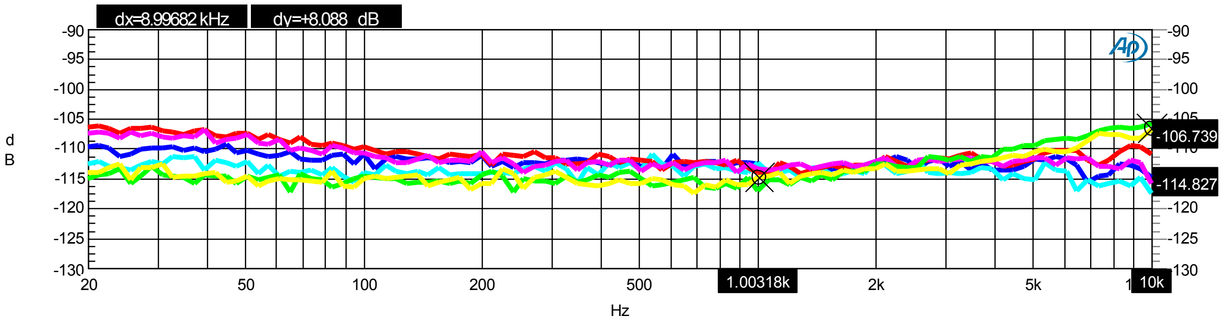
GREEN = EVEN HARMONIC DISTORTION (2,4,6,8,10,12,14)
RED = ODD HARMONIC DISTORTION (3,5,7,9,11,13,15)

DAC2 - EVEN AND ODD THD vs FREQUENCY.at27

Graph 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.

Audio Precision DAC2 - THD VS FREQ AT 0 dBFS - ALL OUTPUTS



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

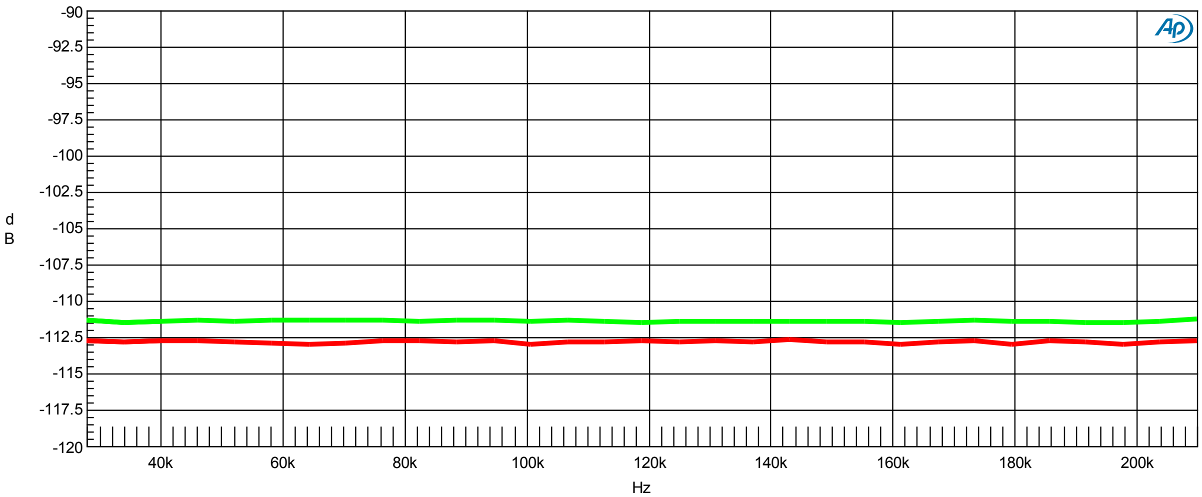
1=Balanced L
2=Balanced R
3=Unbalanced L
4=Unbalanced R
5=Headphone L
6=Headphone R

DAC2 - THD vs FREQUENCY.at27

Graph 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 11**, 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



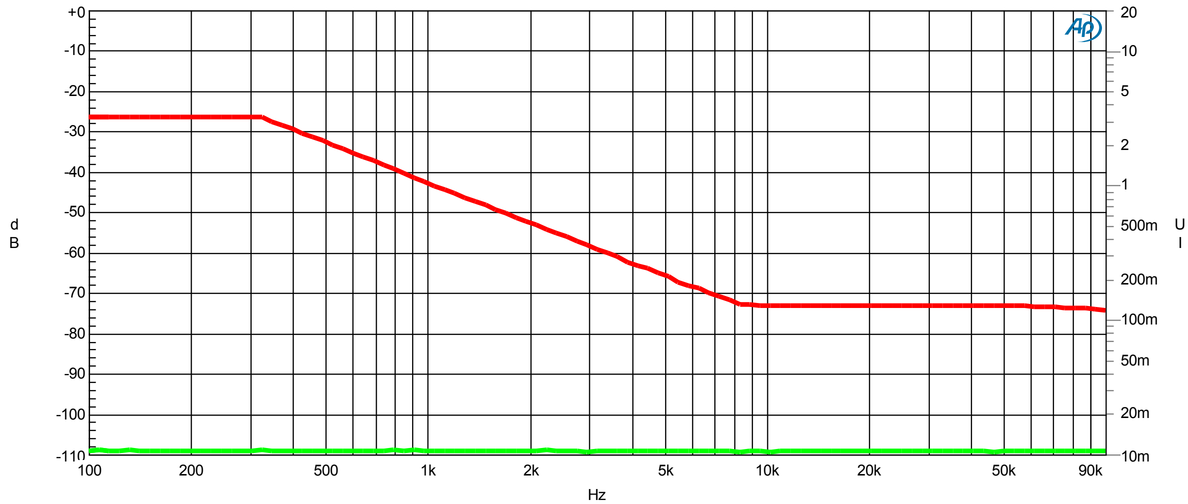
Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Anlr.THD+N Ratio	Left	
1	2	Red	Solid	4	Anlr.THD+N Ratio	Left	

DAC2 - THD+N vs Sample Rate.at27

Graph 13 - THD+N versus Sample Rate

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

Audio Precision DAC2 - INTERFACE JITTER TOLERANCE - Distortion vs Jitter
3.456 kHz Test Tone at 0 dBFS, THD+N 22 to 22 kHz BW



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

DAC2 THD+N (22-22kHz) (Green Trace) vs. Interface Jitter (Red Trace)
Digital Input = 3.456 kHz Tone at 0 dBFS, Fs = 48 kHz

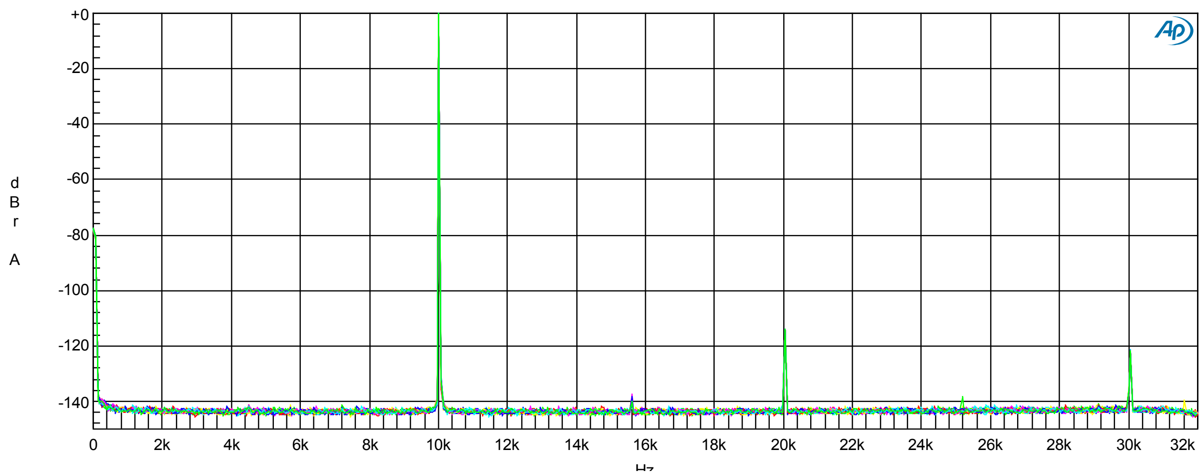
DAC2 - JITTER TOLERANCE.at27

Graph 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

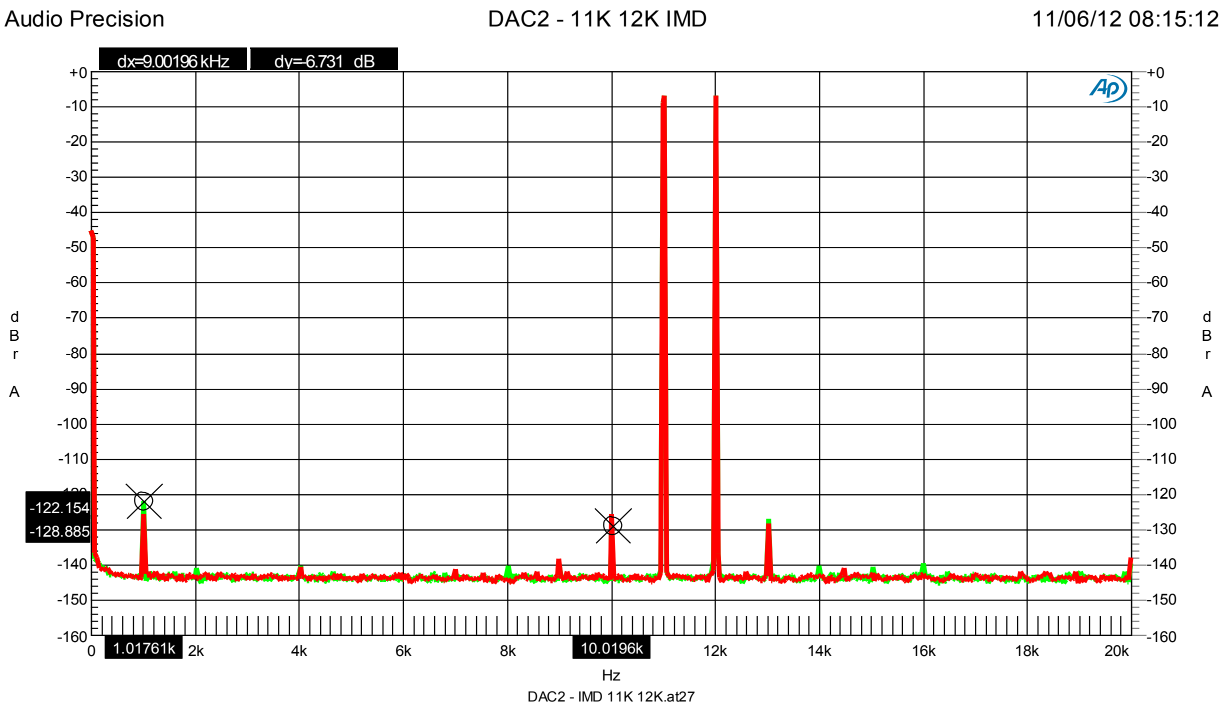


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Source 2
1	2	Green	Solid	4	Fft.Ch.1 Ampl	Left		: 100.000 Hz=Dio.Jitter Freq
2	2	Red	Solid	4	Fft.Ch.1 Ampl	Left		: 125.893 Hz=Dio.Jitter Freq
3	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 158.489 Hz=Dio.Jitter Freq
4	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 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 15 - JITTER TOLERANCE FFT

This plot 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 **Graph 3 - FFT 10 kHz**. 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 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 HGC*** Units Fs<110 kHz) Any sample rate.	+/- 0.25 degrees at 20 kHz

Audio Performance (continued)	
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)	6 (1 USB, 2 Optical, 1 AES, 2 Coaxial)
Number of Channels	2
Input Sample Frequency Range	28 to 210 kHz (Coaxial and XLR) 28 to 96 kHz (Optical) 44.1, 48, 88.2, 96, 176.4, 192 kHz (USB)
Maximum Input Word Length	24 bits
Digital Input Impedance	75 Ohms Coaxial, 110 Ohms XLR
DC Blocking Capacitors on Digital Inputs	Yes (Coaxial and XLR Inputs)
Transient and Over-Voltage Protection on Digital Inputs	Yes (Coaxial and XLR Inputs)
Transformer Coupling on Digital Inputs	Yes (XLR Input)
Minimum Digital Input Level	250 mVpp (Coaxial and XLR 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)	+24 dBu (Attenuator = 0 dB)
Output Level Range (at 0 dBFS) In 'Variable' Mode	Off to +24 dBu (Attenuator off) Off to +14 dBu (Attenuator = 10 dB) Off to +4 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	2 – Dim/Mute 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 Compliance 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 Certificate 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

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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...the measure of excellence!™