...the measure of excellence!™

Benchmark *DAC3 B*
Instruction Manual

Reference Stereo
PCM and DSD D/A Converter
with
ESS9028PRO Conversion System

(Version 1.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 *DAC3* IS EQUIPPED WITH A UNIVERSAL POWER SUPPLY. THERE IS NO VOLTAGE SELECTION SWITCH. AC VOLTAGE RANGE IS 88-264 VAC, 47-63 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.

Contents

Features

- ***Fixed Gain Calibrated Outputs*** – XLR outputs are calibrated to +24 dBu at 0 dBFS (pads at 0 dB), RCA outputs are calibrated to 2 Vrms at 0 dBFS (calibration is equivalent to "HT MODE" or "CALIBRATED MODE" on other ***DAC3*** models)
- ***CONTROL LOCK*** - Disables keypad and remote control to prevent accidental changes
- ***Low-Impedance Passive Output Pads*** – 0, 10, and 20 dB – optimize output level to power amplifiers and other downstream devices to maximize system SNR (Page 25)
- ***SABRE PRO*** - 32-bit PCM D/A conversion system, four 32-bit D/A converters per channel
- ***SABRE PRO*** – Native DSD D/A conversion system, four 1-bit DSD D/A converters per channel
- ***Benchmark UltraLock3™ Jitter Attenuation System*** – eliminates jitter-induced distortion
- ***High Headroom DSP*** - provides 3.5 dB of analog and digital headroom above 0 dBFS to completely eliminate the clipping of intersample peaks
- ***Multi-Mode Asynchronous USB Audio 2.0*** – 24 bit/192 kHz, DSD (DoP 1.1)
- Driverless ***Asynchronous USB Audio 1.1*** – 24-bit/96 kHz
- ***Sample Rate Display*** – displays the measured sample rate, and format (PCM or DSD)
- ***Word Length Display*** – displays the measured word length
- ***2 Coaxial Digital Inputs*** – 24-bit/192 kHz PCM, DSD (DoP 1.1)
- ***2 Optical Digital Inputs*** – 24-bit/96 kHz PCM
- ***1 Coaxial Digital Output*** – digital pass through from USB, Coax, and optical inputs when function is enabled (Page 26)
- ***2 Stereo Analog Outputs*** – 1 pair balanced (XLR), plus 1 pair unbalanced (RCA)
- ***IR Remote*** with metal housing provides control of all functions (optional)
- **Automatic De-Emphasis** – automatically responds to consumer pre-emphasis bit (44.1, 48 kHz)
- ***12V Trigger I/O*** – bi-directional 12V trigger can act as input, output, or both (Page 14)
- ***AUTO-ON Function*** - can be programmed to turn on when AC is applied (Page 13)
- ***Power Switch*** – very low standby power , <0.5 W at 120 VAC
- ***High-Efficiency Low-Noise Power Supplies*** – only 12-15 W, 88-264 VAC, 47-63 Hz
- Meets FCC Class B and CE emissions requirements
- Tested for immunity to radiated and conducted RF interference

Introduction

Applications

The ***DAC3 B*** is the ideal converter to use in front of Benchmark's ***HPA4*** headphone amplifier or ***LA4*** line amplifier. It is also ideal for many professional studio applications. It delivers the full performance of the flagship ***DAC3 HGC***, but eliminates the volume control, the analog inputs, the mute and polarity controls, and the headphone amplifier.

The ***DAC3 B*** is a professional reference-grade audio digital to analog converter with a fixed-gain output. The ***DAC3 B*** 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 ***DAC3 B*** provides D/A conversion and digital source selection. The ***DAC3 B*** is ideal for use with the Benchmark ***HPA4*** headphone/line amplifier or the ***LA4*** line amplifier. These devices provide a very high-performance analog volume control that is a perfect complement to the fixed-gain output of the ***DAC3 B***.

The fixed-gain ***DAC3 B*** has many applications in a professional environment where a calibrated output is required. The **CONTROL LOCK** function can be enabled to prevent accidental changes to the settings.

DAC3 vs. DAC2

The ***DAC3*** builds upon Benchmark's highly successful ***DAC2*** product family. The ***DAC3*** maintains the familiar ***DAC2*** form factor, but adds the higher performance available from the new ES9028PRO D/A converter. The ***DAC3*** offers the following improvements over the ***DAC2***:

- Active 2nd Harmonic Compensation
- Active 3rd Harmonic Compensation
- Lower THD
- Lower passband ripple
- Improved frequency response
- Increased Dynamic Range
- Faster PLL lock times
- Faster switching between inputs

DAC3 vs. DAC1

The ***DAC3 and DAC2*** add 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
- Digital Pass-Through
- High-Headroom DSP
- Additional I/O

DAC3 Technologies

4:1 Parallel Conversion Structure

The conversion system in the ***DAC3*** achieves a 6 dB signal to noise improvement through the use of 4:1 summing. The ES9028PRO D/A is an 8-channel 32-bit converter. In the ***DAC3***, four channels are summed in the analog domain to form each of the two output channels.

The 4:1 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 6 dB.

Harmonic Compensation

The ES9028PRO has two distortion compensation systems that independently remove most of the 2nd and 3rd harmonic distortion in the D/A converter. Benchmark's ultra-clean analog output stages allow these systems to be fully leveraged in the ***DAC3***.

High-Headroom Digital and Analog Processing

The ***DAC3 B*** has generous amounts of analog and digital headroom. The analog clip point is above 29 dBu. The digital clip point is +3.5 dBFS and is aligned to 27.5 dBu. The factory calibration is +24 dBu at 0 dBFS. This means that the ***DAC3 B*** has 3.5 dB of digital headroom above 0 dBFS, and another 1.5 dB before reaching analog clip. The digital headroom prevents the clipping of intersample overs. The analog headroom allows accurate reproduction of intersample peaks without clipping.

No Clipping of Intersample Overs

The ***DAC3*** 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 without clipping.

Low-Noise Power Supplies

The ***DAC3*** 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 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 ***DAC3*** 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 ***DAC3***. This also means that the ***DAC3*** can be placed in close proximity to any audio component without causing interference with the other component.

UltraLock3™ Clock System

UltraLock3™ provides the outstanding jitter attenuation of Benchmark's ***UltraLock2™*** system while providing virtually instantaneous (6 ms) lock times.

Dual-Mode USB Input

The ***DAC3*** 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 ***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.

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 or Linux operating systems. Drivers are provided for Windows operating systems at: BenchmarkMedia.com/drivers

Native Asynchronous USB 1.1

The ***DAC3*** 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.

Low-Impedance Passive Attenuators

Like the ***DAC1*** and ***DAC2***, the ***DAC3*** includes low-impedance passive attenuators on the XLR outputs. These attenuators can be adjusted to optimize the interface with preamplifiers, power amplifiers or powered monitors. This optimization places the system volume control in its best operating range. This exclusive Benchmark feature can provide substantial improvements in the performance of the playback signal chain.

Native DSD Conversion

The ***DAC3*** 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. Four balanced 1-bit converters are summed together for each balanced output.

Digital Pass-Through

The second coaxial input (**D4**) can be reconfigured as a digital output. When operating as an output, any selected digital input is passed through to **D4** without any processing. Optical, coaxial, and USB inputs can be passed through to the **D4** connector. This even includes special signals such as DoP, DTS, Dolby Digital, even when these signals cannot be decoded by the ***DAC3***.

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. The ***DAC3*** trigger I/O can be connected to a preamplifier, power amplifier, or both. The ***DAC3*** will pull the trigger I/O to 12 volts DC while the ***DAC3*** is on. If the ***DAC3*** is off and an external device pulls the trigger I/O to 12 volts, the ***DAC3*** will turn on.

Auto-On Function

The ***DAC3*** can be programmed to automatically turn on when AC power is applied. (Page 13)

Front Panel

- IR Sensor
- Power Button
- Control-Lock Button and Indicator
- Input Select Buttons
- Input Indicators
- Word-Length Indicators
- Sample-Rate Indicators

Rear Panel

- Balanced Outputs
- Unbalanced Outputs
- Digital Inputs
- 12V Trigger I/O
- IEC Power Entry with Fuses

Quick Start Guide

Audio Inputs

The ***DAC3 B*** features five stereo digital inputs (2 coaxial, 2 optical, and 1 USB). The coaxial and optical inputs accept professional (AES) and consumer (S/PDIF) data formats at word lengths up to 24-bits.

Tip: We recommend using the coaxial or USB inputs for sample rates above 96 kHz. Optical interfaces are not always reliable at sample rates above 96 kHz.

Remote Control (optional)

Tip: The remote control is designed to have a long operating range. In most applications it is not even necessary to point the remote directly at the ***DAC3***. The IR sensor is located above the **POWER** button.

The chart at the right summarizes the functions of the optional 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	Not used.
DIM	Not used.
MUTE	Not used.
INPUT	Not used.
D1	Selects optical digital input **D1**.
D2	Selects optical digital input **D2**.
D3	Selects coaxial digital input **D3**.
D4	Selects coaxial digital input **D4**.
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	Not used.

Tip: To provide system compatibility with the Benchmark ***HPA4*** and ***LA4***, the **DIM**, **MUTE**, **VOLUME UP/DOWN**, **INPUT UP/DOWN** and **ANALOG** keys are not used by the ***DAC3 B***.

Tip: The **CONTROL-LOCK** button disables the remote control and all other buttons on the front panel.

Front Panel Controls

Tip: The **CONTROL-LOCK** button disables the remote control and all buttons on the front panel. Use this function in dedicated applications to prevent loss of audio. If you are not using the **12V TRIGGER I/O**, you will also want to activate the **AUTO-ON** feature because the **POWER** button is disabled.

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.

Tip: The IR remote control sensor is located directly above the **POWER** button.

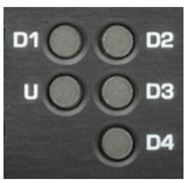
The chart at the right summarizes the functions of the front-panel controls.

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 3 seconds to set the **AUTO-ON** function. Starting with the unit on, press and hold the **POWER** button for 3 seconds to clear the **AUTO-ON** function. If **AUTO-ON** is set, the **POWER** button is disabled. If **CONTROL-LOCK** is on, press the **POWER** button for 3 seconds to turn the unit off.
CONTROL LOCK	Press and hold for 3 seconds to toggle **CONTROL-LOCK** on or off. When the **CONTROL-LOCK** light is on, the front panel keys and remote control are disabled.
INPUT	Press to select inputs. If **CONTROL-LOCK** is on, the **INPUT** buttons are disabled. Press and hold both input buttons for 3 seconds to toggle between **USB 1.1** and **USB 2.0**.

Front Panel Display

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

Input Indicators



The input indicators show which input is selected.

A flashing light indicates an error on a digital input.

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

Note: D4 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 26).

Input Error Codes

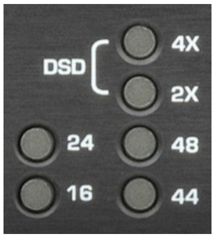
The input indicators flash when errors are present on the selected digital input. There are no error indications for analog inputs. Use the following table to diagnose the problem:

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

Digital Format Indicators



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

Four lights indicate the 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 ***DAC3*** 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.

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.

Operational Details

AUTO-ON Function

The ***DAC3*** 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 ***DAC3*** 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 three seconds. This function cannot be programmed from the remote control. This limitation prevents accidental access to this special feature.

Tip: The **AUTO-ON** function cannot be changed if the **CONTROL-LOCK** is on. Turn the **CONTROL-LOCK** off before attempting to change the **AUTO-ON** function.

Enabling AUTO-ON

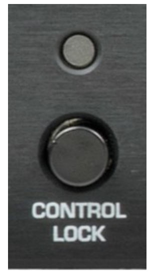
Starting with the ***DAC3*** off, press and hold the **POWER** button for 3 seconds.

If **AUTO-ON** has been successfully enabled, the unit cannot be turned off using the **POWER** button or the **OFF** button on the remote.

Disabling AUTO-ON

Starting with the ***DAC3*** on, press and hold the **POWER** button for 3 seconds. At the end of 3 seconds the ***DAC3*** will power down if the **AUTO-ON** mode has been successfully disabled.

CONTROL-LOCK Function



The **CONTROL-LOCK** function disables the remote control and all buttons on the front panel.

Use the **CONTROL-LOCK** function when the ***DAC3*** will be used in a dedicated application using just one digital input. This feature prevents accidental loss of audio in these dedicated applications.

Controls are locked when the **CONTROL-LOCK** light is on.

Press the **CONTROL-LOCK** button for 3 seconds to toggle this function on or off.

Tip: If **CONTROL-LOCK** is on, but **AUTO-ON** is off, the unit can be powered down by pressing and holding the **POWER** button for 3 seconds. This turns the unit off, but does not turn off the **CONTROL-LOCK** function.

Bi-directional 12V Trigger

Benchmark has reinvented the 12 volt trigger by adding bi-directional signaling. The trigger connection on the ***DAC3*** 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 ***DAC3*** turns on. The ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** automatically configures its trigger I/O port as an input (slave) or output (master).

Trigger Output (*DAC3* is Master)

When the ***DAC3*** is turned on using the **POWER** button (on the front panel), or the **ON** button (on the remote), the ***DAC3*** configures itself as a trigger master and will drive the **12V TRIGGER** I/O to 12 volts DC and hold it there while the ***DAC3*** is on. The trigger output signal generated by the ***DAC3*** is delayed so that the ***DAC3*** can stabilize before downstream devices (such as power amplifiers) turn on. When powering down, the ***DAC3*** will mute before allowing the trigger line to drop low. The ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** will always drive the **12V TRIGGER** I/O line to 12 V (after a short start-up delay).

Trigger Input - (*DAC3* is Slave)

If the ***DAC3*** is off and an external device pulls the trigger I/O to 12 volts, the ***DAC3*** will configure itself as a trigger slave and will follow the actions of the trigger input. The ***DAC3*** 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 ***DAC3*** to one other device. The ***DAC3*** 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:

- ***DAC3 B*** → Preamplifier → Amplifier
  - ***DAC3 B → LA4 → AHB2***
- ***DAC3 B*** → Headphone Amplifier
  - ***DAC3 B → HPA4***
- ***DAC3 B*** → Amplifier
  - ***DAC3 B → AHB2***

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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3***.

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

HPA4, LA4 Compatibility

The ***DAC3 B*** is designed to work seamlessly with Benchmark line amplifiers (such as the ***HPA4*** and ***LA4***).

The line amplifier will control the system volume and analog input switching. The ***DAC3 B*** will be used to select the digital inputs. Both are controlled from a single Benchmark remote control.

USB MODE Selection

The ***DAC3*** 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. Avoid any unnecessary switching between **USB MODES**. Rapid switching between modes can confuse some operating systems.

Note: The computer and ***DAC3*** must be connected and both must be on before the **USB MODE** can be changed.

To change the **USB MODE**, press and hold the **USB** button on the remote control for 3 seconds. If a remote control is not available, simultaneously press and hold both **INPUT** buttons on the front panel for 3 seconds.

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**. A flash of the **4X** lamp indicates that the unit is in **USB Audio 2.0** mode. A flash of the **2X** lamp indicates that the unit is in **USB Audio 1.1** mode.

High-Current Outputs

The **RCA** and **XLR** outputs on the ***DAC3 B*** are equipped with low-impedance high-current drivers. These robust outputs are well equipped to drive a wide variety of input impedances. The ***DAC3 B*** outputs remain clean when driving inputs that present difficult loads (high input capacitance and/or low input impedance).

The **XLR** outputs on the ***DAC3 B*** 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 ***DAC3 B*** to the input sensitivity of the downstream device.

Tip: In most applications we recommend placing a Benchmark ***HPA4*** or ***LA4*** line amplifier between the ***DAC3 B*** and the power amplifier. These Benchmark line amplifiers will optimize the analog interface between the ***DAC3*** and the power amplifier while providing a stepped relay gain control with exceedingly low THD and noise. The ***HPA4*** or ***LA4*** provide a transparent path between the ***DAC3 B*** and a power amplifier.

Driving Power Amplifiers

If the system volume is being controlled digitally by an upstream device, the ***DAC3 B*** can be connected directly to an audio power amplifier or a set of powered monitors. This direct connection provides a clean path between the ***DAC3 B*** and the amplifier, but it is important to match the signal levels between the ***DAC3 B*** and the power amplifier in order to prevent excessive use of the upstream digital volume control.

The XLR outputs are equipped with adjustable pads that can be used to optimize the system gain structure in order to maximize the system noise performance.

Most power amplifiers and powered monitors will require the use of the **10 dB** or **20 dB** pads. Use the **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 ***DAC3*** 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 ***DAC3 B*** (when the pads are set to **0 dB**). This configuration optimizes the gain-staging between the ***DAC3*** and the ***AHB2***.

Tip: If you are using a direct connection between a ***DAC3 B*** and a non-Benchmark power amplifier, the **XLR** pads should be set so that a normal listening level is reached when the system volume control is near its maximum setting. This will optimize the gain-staging between the ***DAC3 B*** and your power amplifier.

Tip: Increase the pad setting if a comfortable listening level is reached while the system volume control is well below maximum.

Tip: Decrease the pad setting if a comfortable listening level cannot be reached when the system volume control is at maximum.

Instructions for setting the **XLR** pad jumpers are detailed in the **Internal Settings** section of this manual. The ***DAC3 B*** 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 or Benchmark line amplifier.

Digital Pass-Through

The second coaxial input (**D4**) can be reconfigured as a digital output. When operating as an output, any selected digital input is passed through to **D4** without any processing.

Optical, coaxial, and USB inputs (**U**, **D1**, **D2** and **D3**) can be passed through to the **D4** connector. The signals are buffered but are not processed in any way. For this reason, any data format can be passed through to the **D4** connector, even when these formats cannot be decoded by the ***DAC3***. Surround formats, such as DTS, Dolby Digital, cannot be decoded by the ***DAC3***, 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
- Coaxial to Coaxial (buffering)

DoP encapsulated DSD can also be passed through **D4**. DSD files on a computer can be sent in DoP to the **USB** input on the ***DAC3***. The **USB** input can be routed to coaxial output **D4**. 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.

Firmware Version Identification

The firmware version is displayed during the lamp test while the ***DAC3 B*** 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. 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**	
2	**D1**		**D2**	.4
1	**U**		**D3**	.2
			D4	.1

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

Example 2: The **U** and **D4** lamps flash. The firmware version is 1.1.

Example 3: The **D1** lamp is the only lamp that flashes. The firmware version is 2.0.

Rear Panel

Inputs

There are five stereo digital inputs on the ***DAC3 B***:

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

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

*** D4** 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 **D4**. The selected input will be buffered and sent to output **D4** even if the format cannot be decoded by the ***DAC3 B***.

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 ***DAC3 B*** 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 **D4** 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 - Overview

All of the digital inputs on the ***DAC3 B*** use Benchmark's ***UltraLock3***™ system to remove virtually all of the interface jitter. The result is that all digital inputs deliver identical audio performance. The USB, optical, 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 ***DAC3 B***. The USB cable connects the ***DAC3 B*** 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 ***DAC3 B*** 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 three seconds to toggle the **USB MODE**. If a remote is not available, simultaneously press and hold both input buttons on the front panel for three seconds.

The **USB AUDIO 1.1** mode does not require 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 ***DAC3 B*** 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, 8 and 10 (driver installation is required for all Windows versions). It 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 ***DAC3 B*** 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.

Coaxial Digital Inputs - D3 and D4

The coaxial digital inputs (**D3** and **D4**) 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 **D3** and **D4**. 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.

Tip: The RCA analog outputs have no restrictions on cable type. But, to avoid confusion, we recommend using 75-Ohm coaxial cables for all RCA connections. 75-Ohm coaxial cables are compatible with digital audio, analog audio, and video.

Note: The Coaxial inputs (**D3** and **D4**) 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 ***DAC3*** 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 ***DAC3 B*** turns on. The ***DAC3 B*** 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 ***DAC3*** 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 ***DAC3*** automatically configures the **12V TRIGGER** I/O port as an input (slave) or output (master). See page 14 for more information.

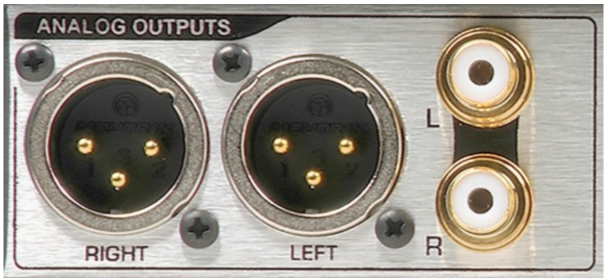
The Benchmark **12V TRIGGER** I/O has a wide operating range to allow interfacing with most other DC trigger systems. It should only be used with trigger inputs that are designed to tolerate 12 VDC.

- 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 ***DAC3 B*** has one pair of balanced XLR outputs and one pair of unbalanced RCA outputs.

The ***DAC3 B*** 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 ***DAC3 B*** ships with the pads disabled (0 dB setting).

A full description of the output attenuators and instructions for configuration is located in the **Internal Settings** section of this manual.

TIP: Use the 0 dB pad setting when driving Benchmark products and other professional products with XLR inputs that support +24 dBu signal levels. Use the 10 dB pad when driving consumer-grade XLR inputs.

Industry-Standard XLR Wiring

- XLR pin 2 = + Audio Out
- XLR pin 3 = - Audio Out
- XLR pin 1 = Cable Shield

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

Unbalanced RCA Analog Outputs

The Left and Right unbalanced outputs use female RCA jacks. The ground connections are bonded to chassis ground. This prevents ground loop currents in the internal analog ground.

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 produce 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 ***DAC3 B*** is equipped with a universal power supply. There is no voltage selection switch. AC voltage range is 88-264 VAC, 47-63 Hz.

Power Cord

Note: The AC power input uses a standard IEC type connector. One USA-compatible power cord is included with ***DAC3 B*** converters. IEC style power cords in country-specific configurations are available in your locality.

Caution: Always use a grounded power cord. The *DAC3 B* 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
- Digital Pass-Through

Removing Top Cover

The ***DAC3*** cover must be removed to gain access to the jumpers. Do not attempt to remove the faceplate or rear panel.

Caution: The ***DAC3*** 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 *DAC3*.
- 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 *DAC3*.
- Connect a static-control wrist strap to the chassis before touching any internal component.
- If a static-control wrist strap is not available, keep one hand on the chassis while moving the jumpers. Avoid touching other components.

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 ***DAC3***. The ***DAC3*** ships with the pads disabled (**0 dB** setting).

Tip: To set the XLR outputs are factory-preset to deliver professional studio levels. Most home installations will require the use of the **10 dB** or **20 dB** pads.

Tip: Use the factory-default **0 dB** setting when driving Benchmark products such as the ***HPA4***, ***LA4*** or ***AHB2***. 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 system volume control is near maximum.

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)

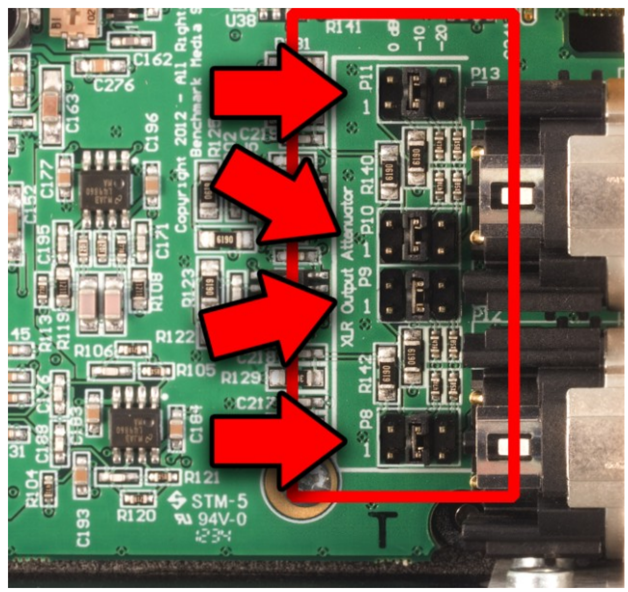


Figure 1 - Attenuators set to -10 dB

Digital PASS-THROUGH Function

The digital **PASS-THROUGH** function can be enabled by moving two jumpers. When the **PASS-THROUGH** function is enabled, **D4** cannot be selected as an input. Any other selected digital input will be routed to both the internal D/A converter and to output **D4**. The digital output at **D4** is buffered, but is not processed. Many digital audio formats can be passed through to **D4** (even when these formats cannot be decoded by the ***DAC3***).

Digital PASS-THROUGH Jumpers

To enable the **PASS-THROUGH** function, move both **P14** jumpers toward the faceplate as shown in **Figure 2**. Once the jumpers are moved into the positions shown in **Figure 2**, **D4** is configured as a digital audio output.

By default, **D4** functions as a digital input and the ***DAC3*** is shipped with the jumpers set according to **Figure 3** (both jumpers toward the rear panel).

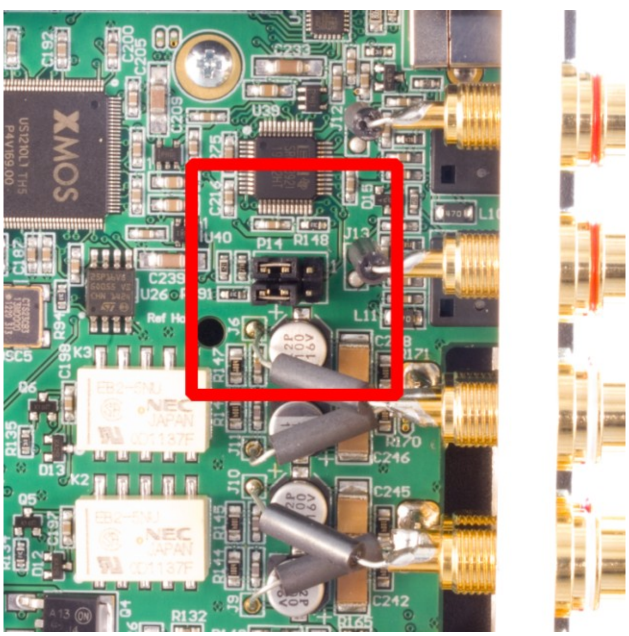


Figure 2 – Digital PASS-THROUGH Enabled

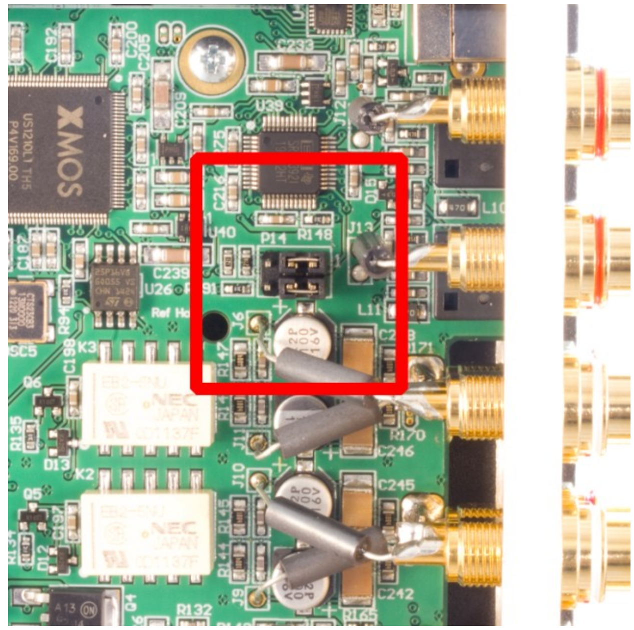


Figure 3 - Digital PASS-THROUGH Disabled (Factory Default)

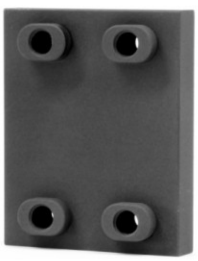
Rack Mounting

The ***DAC3 B*** is available with or without rack mount ears. The chassis is one half the width of a standard rack and occupies a one rack-unit height.

Units without rack-mount ears will require a mounting tray. Units with rack-mount ears will require a ***Rack-Mount Coupler*** which joins the ***DAC3*** to another ½-wide Benchmark product or to a ***Blank Plate***.

The ***Rack-Mount Coupler***, ***Rack-Mount Tray*** and ***½-Wide Blank Plate*** are available from Benchmark.

Benchmark Rack-Mount Coupler



The ***Rack-Mount Coupler*** joins two ½-wide 1-RU products. Both products must be equipped with rack ears. Order with an optional ***Blank Plate*** if only one unit will be mounted in the rack.

No tray is required when using the ***Rack-Mount Coupler***.

Benchmark Rack Mount Tray

The ***Benchmark Rack Mount Tray*** mounts up to two ½-wide Benchmark products in a single race space. The tray accepts any combination of ½-wide Benchmark products (with or without rack-mount type faceplates). A blank plate can be added when only one unit is installed in the rack mount tray.

Benchmark ½-Wide Blank Plate

The ***Benchmark ½-Wide Blank Plate*** is a ½-wide 1-RU anodized aluminum panel for filling an unused ½-wide slot. It is available in black or silver. The silver panel includes an engraved Benchmark logo.

Visit BenchmarkMedia.com for a complete selection of cable, accessories, and replacement parts.

DAC1, DAC2 and *DAC3* 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***, ***DAC2*** and ***DAC3*** 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.

DAC3 Series

On the surface, the ***DAC3*** series converters look exactly like the ***DAC2*** converters. They have the same controls, the same connectors, and even the similar-looking circuit boards. The difference is that many critical components and systems have been upgraded. The new ***DAC3*** delivers lower THD, improved digital filtering, and faster PLL lock times.

The firmware, digital signal processing, and ***UltraLock™*** clock system have all been upgraded in the new ***DAC3***. Most notably, the ***DAC3*** series includes the new ES9028PRO D/A converter. This groundbreaking D/A converter IC offers several significant improvements over the ES9018 converter used in the ***DAC2*** series. Until the ESS PRO series was introduced, the ES9018 was the highest performance D/A converter IC available. The ESS PRO series converters are now setting this benchmark. The Benchmark ***DAC3*** is one of the first products to feature this new 32-bit D/A converter.

Benchmark's ultra-clean analog stages, low-jitter ***UltraLock3™*** clock system, and high-headroom DSP leverage the full capabilities of the new ESS PRO series converters.

DAC3 vs. *DAC2*

The ***DAC3*** series adds the following improvements over the ***DAC2*** series:

- ESS SABRE-PRO D/A Conversion
- ***UltraLock3™*** clock system
    - Instantaneous lock
    - Instantaneous input switching
- THD reduction system
    - 2nd-harmonic compensation
    - 3rd-harmonic compensation
- Improved digital filters
    - Lower passband ripple
    - Flatter frequency response
- Higher maximum output level
- Increased Dynamic Range

Benchmark Technologies

Native DSD Conversion

The digital coaxial inputs and the **USB Audio 2.0 input** on the ***DAC3*** 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.

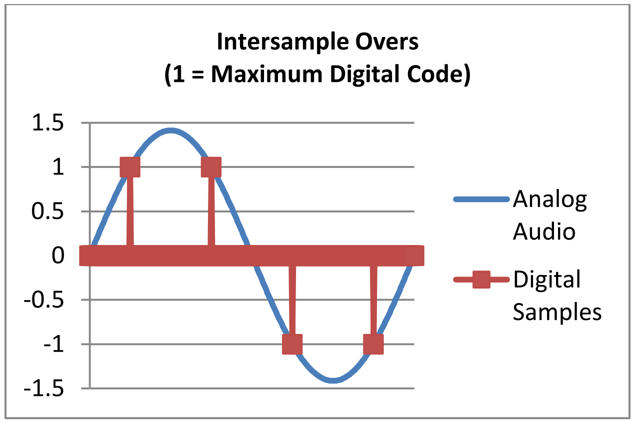
Visit our website for an up-to-date list of DSD and PCM high resolution download sites.

High Headroom DSP

All of the digital processing in the ***DAC3*** 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.

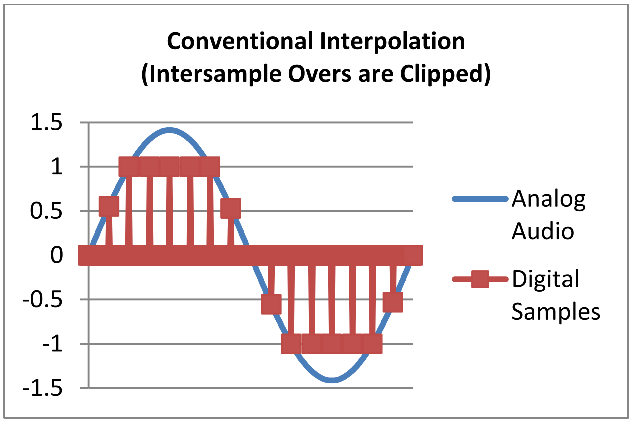
Benchmark's high-headroom DSP can handle intersample peaks without overloading or clipping. Intersample peaks are cleanly rendered by the ***DAC3*** and are delivered to the analog outputs without clipping or distortion. **Very few D/A converters can make this claim!**

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. 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 ***DAC3*** delivers clean PCM conversion that meets or exceeds the clarity of DSD.

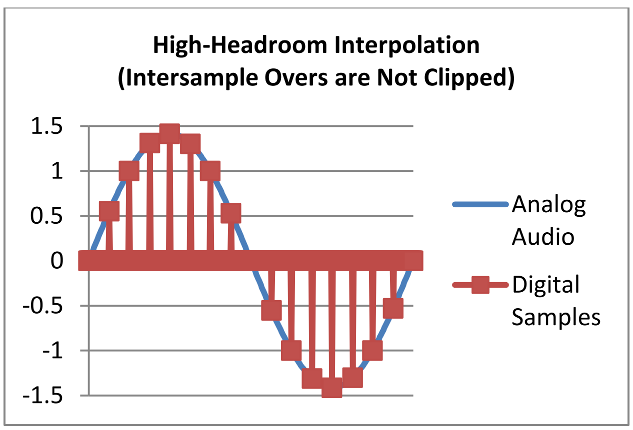


Intersample overs are common in low sample rate (44.1 kHz and 48 kHz) commercial releases. Due to the mathematics and the bandwidth of typical input signals, intersample overs are less of a problem in high sample rate recordings. The reason for this is that the worst-case (+3.01 dB) intersample overs occurs for pure tones that are exactly 1/4 of the sample rate (as shown in the figure above). 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 1/8th 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 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.



The interpolation process is absolutely necessary to achieve 24-bit state-of-the art conversion performance. Unfortunately, intersample overs cause clipping in most interpolators. This clipping produces distortion products that are non-harmonic and non-musical. We believe these broadband distortion products often add a harshness or false high-frequency brightness to digital reproduction. The ***DAC3*** avoids these problems by maintaining at least 3.5 dB of headroom in the entire conversion system. We believe this added headroom is a groundbreaking improvement.

32-bit SABRE-PRO D/A System

Four balanced 32-bit D/A converters deliver audio to Benchmark's low-impedance current to voltage converters. The 4:1 redundancy reduces noise by 6 dB. The redundancy also reduces the THD. The conversion system at the core of the ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** uses distributed power supply regulation. Each critical subsystem has at least one dedicated low-noise voltage regulator.

We have created a discrete ultra low-noise regulator for the ES9028PRO D/A converter. This Benchmark exclusive feature improves the noise performance of the already-outstanding ES9028PRO.

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 ***DAC3*** 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 *UltraLock3™*

UltraLock3™ is an improved version of the ***UltraLock2™*** clock system used in the ***DAC2***. The new ***UltraLock3™*** system provides faster lock times than the older ***UltraLock2™*** and ***UltraLock™*** systems. 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 ***UltraLock3™*** system eliminates all audible jitter artifacts.

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 <u>unless</u> 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 ***DAC3*** 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 ***DAC3*** 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 any 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 (see Graph 14). This FFT shows that the ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3*** 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 ***DAC3***. The asynchronous USB interface pulls data from the computer without using computer-generated clocks. The D/A conversion in the ***DAC3*** is completely isolated by the asynchronous USB interface and by the ***UltraLock3™*** jitter-attenuation system.

The ***DAC3*** 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 ***DAC3***. 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 ***DAC3*** 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 not yet natively supported by the current Windows operating systems. For this reason, a driver is required for Windows operating systems. The driver supports Windows XP, Vista, or 7, 8 and 10. This driver is required for DSD and sample rates above 96 kHz when using Windows.

The USB subsystem is computer powered (through the USB cord) and it remains active when the ***DAC3*** 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

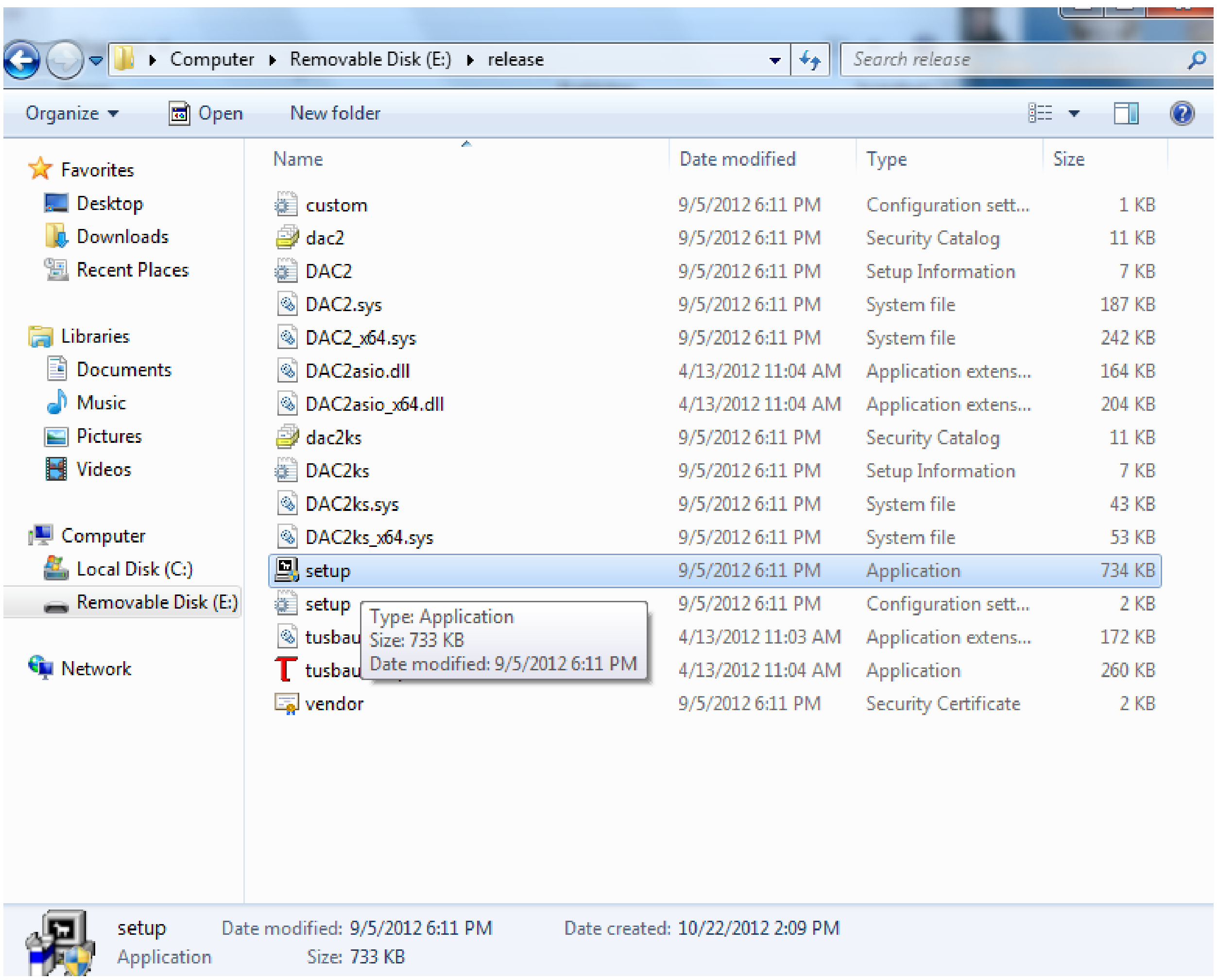
Windows Operating Systems

Note: The ***DAC2/DAC3*** driver is available for download at: BenchmarkMedia.com/drivers

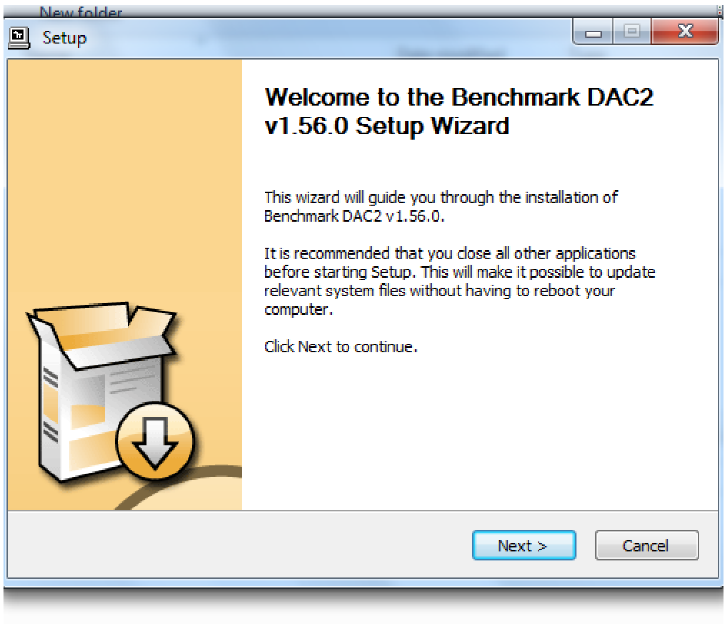
Note: To provide full backward and forward compatibility, the ***DAC3*** uses the ***DAC2*** USB drivers. This prevents the need to install two different sets of drivers. Please note that the ***DAC3*** USB input will be identified as "Benchmark DAC2" in your computer control panels. This is intentional.

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

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

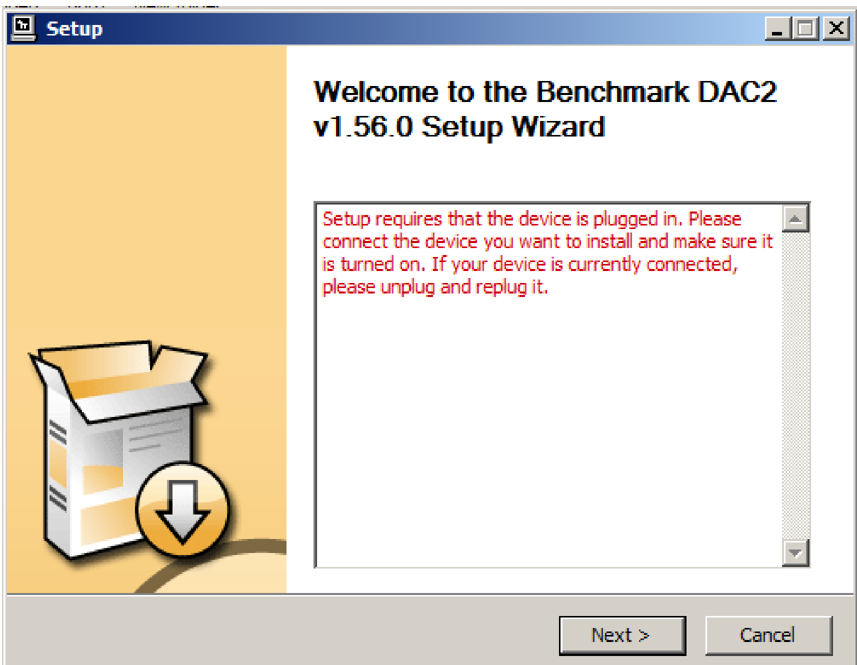


2. A welcome screen will pop-up. Click "*Next.*"

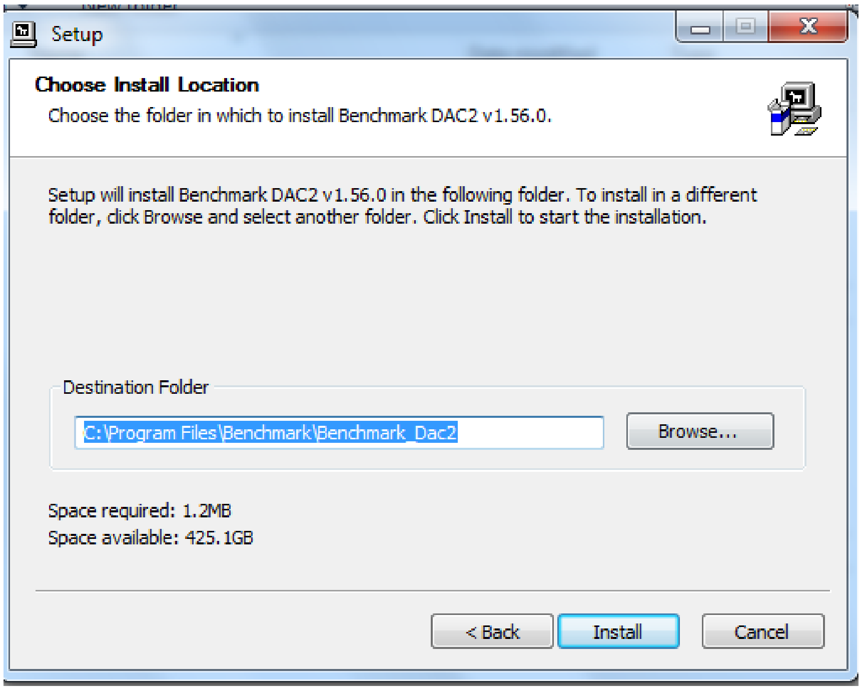


3. When you see the following screen, turn on the ***DAC3***, plug in the USB cable, and select the **USB** input on the ***DAC3***. By default, the ***DAC3*** is shipped in **USB Audio 1.0** mode. Do not proceed to the next step until the **USB Audio 2.0** mode has been enabled. You can enable the **USB Audio 2.0** mode using one of the following two methods:

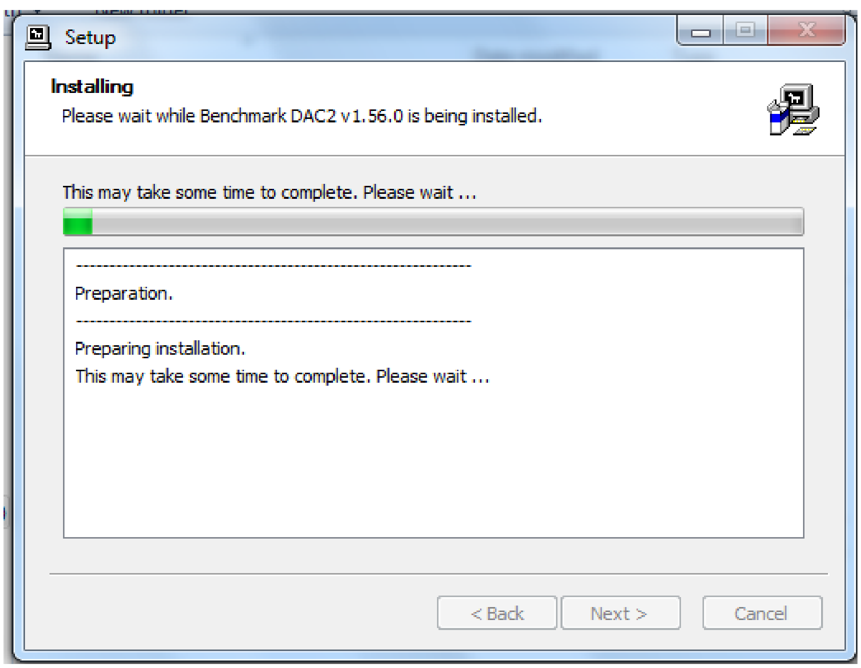
    - **METHOD 1 - REMOTE CONTROL:** Using your remote control, press and hold USB button for 3 seconds (the **4X** light should flash once). If the **2X** light flashes instead of the **4X** light, repeat this step.
    - **METHOD 2 - FRONT PANEL:** From the front panel, simultaneously press and hold both **INPUT** buttons for three seconds (the **4X** light should flash once when selecting the **USB** input). If the **2X** light flashes instead of the **4X** light, repeat this step.



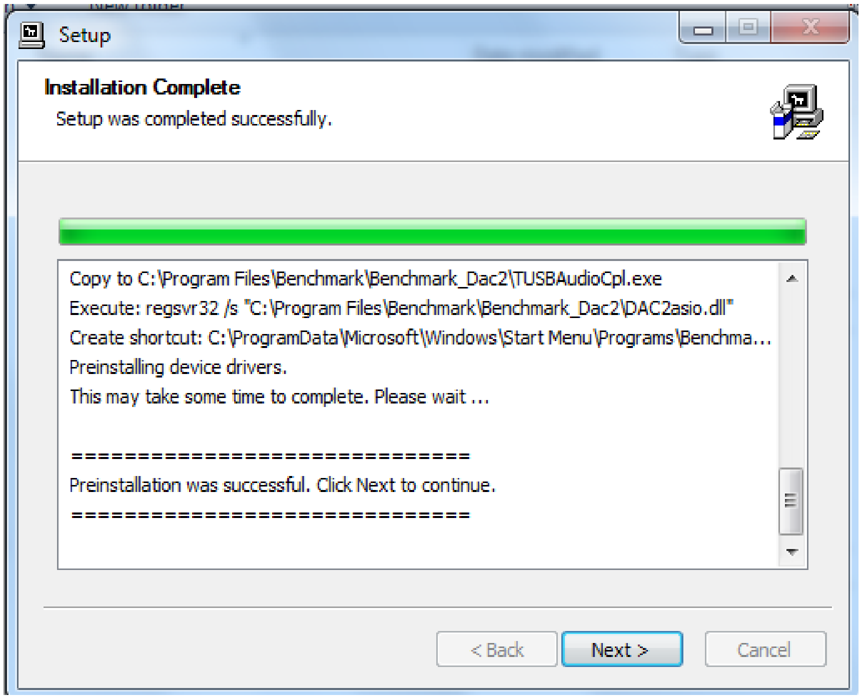
4. You will now be prompted to select a location to install the driver. It will default to your Program Files folder. If you wish to install it another location, you can change the location. We suggest keeping it in the default location. Click “Install”.



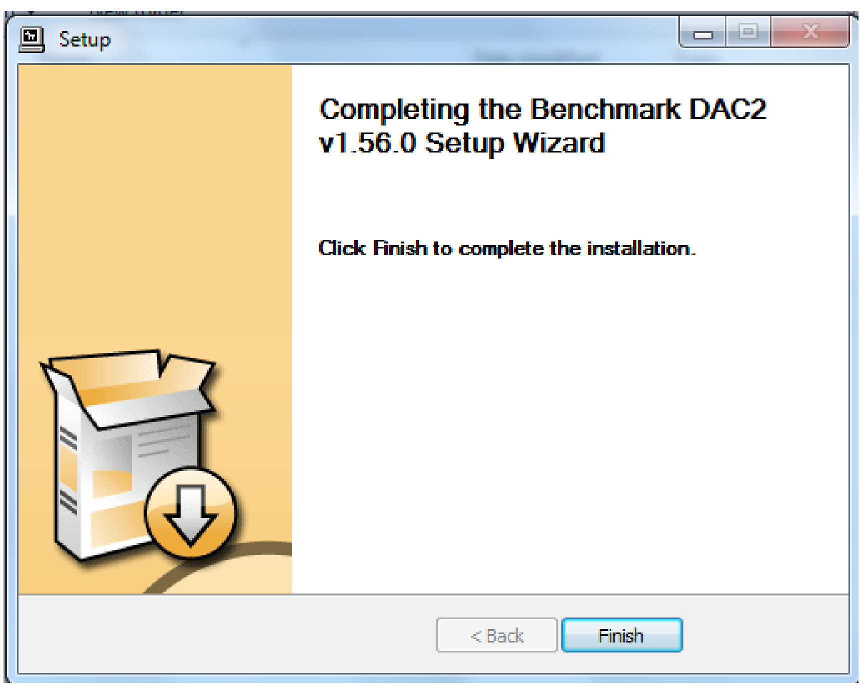
5. When the installation begins you will see the following screen. Please be patient while the driver installs. Installation may require several minutes.



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

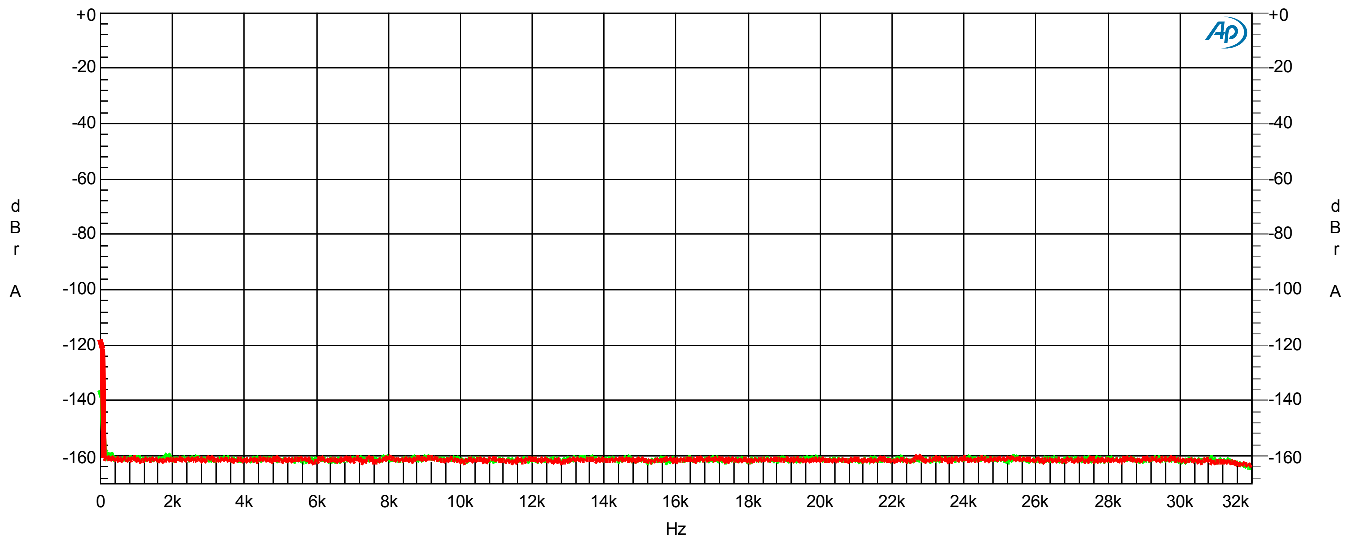


7. Click “Finish.” The Setup will close automatically. This completes the installation process. You can now enjoy music over the **USB Audio 2.0** connection at sample rates up to 192 kHz. DSD can also be played in DoP 1.1 format.



Performance Graphs

Audio Precision DAC3 - FFT Idle Channel Noise at Max Gain, 0 dBr = 0 dBFS = 27.5 dBu



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

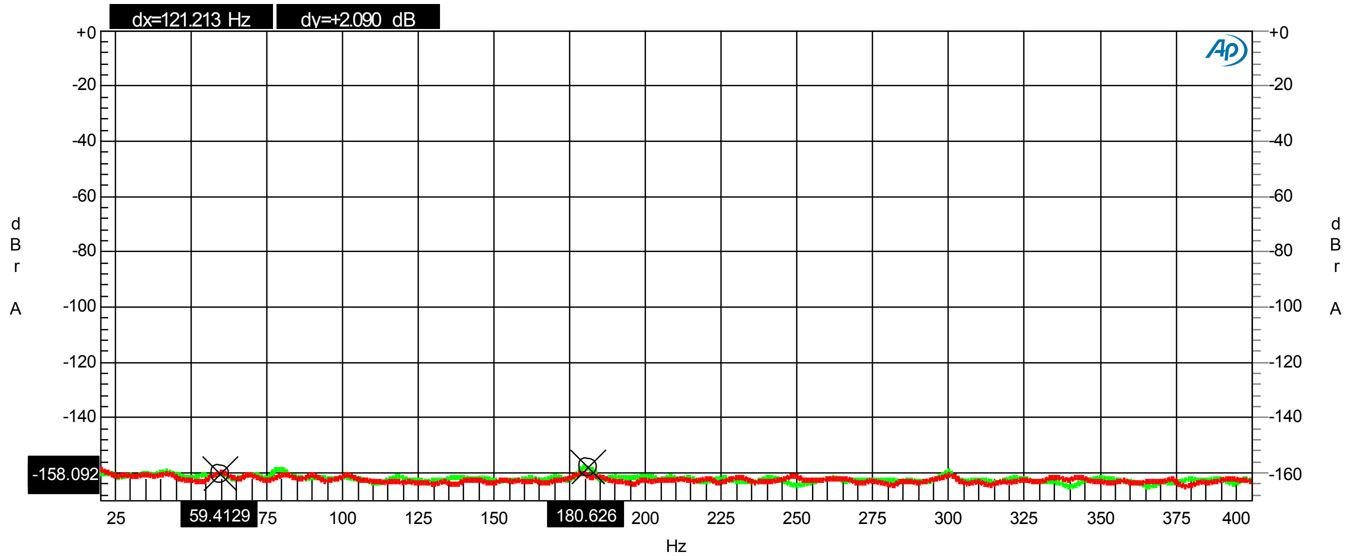
DAC3 - FFT Idle Channel Noise.at27

Graph 1 – FFT Idle Channel Noise

The extraordinary performance of the ***DAC3*** 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 DAC3 - FFT Idle Channel Noise - Low Frequency, 0 dBr = 0 dBFS = 27.5 dBu



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	Left Channel	*-160.182 dBr	*-158.092 dBr
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right	Right Channel	-159.850 dBr A	-160.611 dBr A

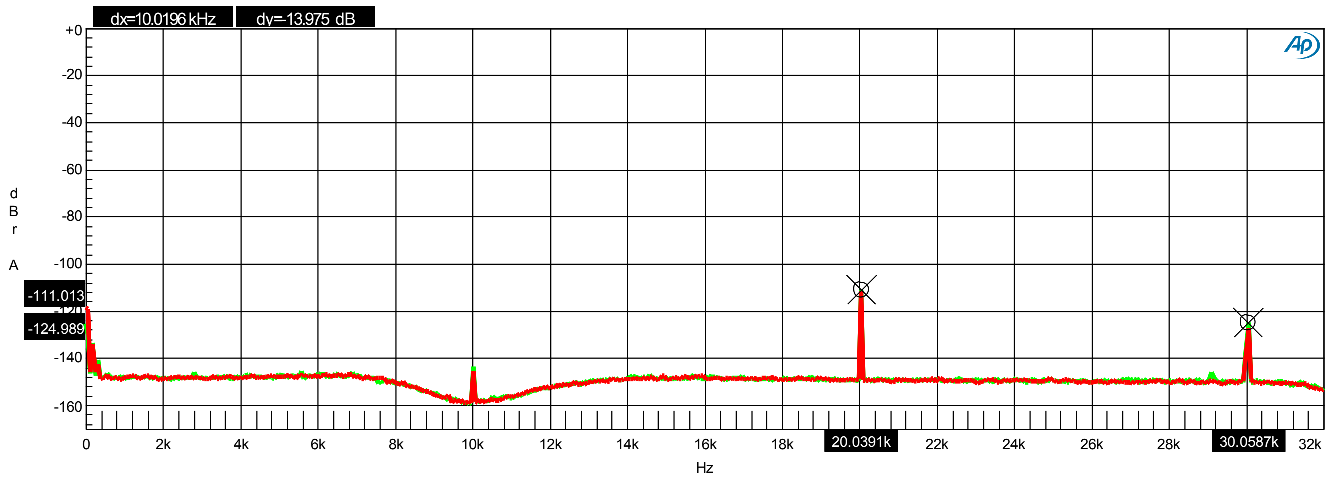
DAC3 - FFT Idle Channel Noise - Low Frequency.at27

Graph 2 - Low Frequency FFT - AC Line-Related Hum

The ***DAC3*** shows no evidence of AC line-related hum to a measurement limit of about -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. In the idle channel noise spectrum there is absolutely no sign of any AC hum! At full output, these line-related frequencies still measure better than -133 dB (see Graph 4).

This graph demonstrates one of the advantages of switching power supplies. The switching power supplies in the ***DAC3*** operate at frequencies above the audio band and this eliminates the strong line-frequency magnetic fields that would have been created by line-frequency power transformers.

Audio Precision DAC3 - FFT 10 kHz, 0 dBr = 0 dBFS = 24 dBu



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Source 2	Cursor1	Cursor2
1	2	Green	Solid	4	Fft.Ch.1 Ampl	Left		: 1.00000 =Swr.Ch. A Input	*-111.013 dBr	*-124.989 dBr
2	2	Red	Solid	4	Fft.Ch.1 Ampl	Left		: 2.00000 =Swr.Ch. A Input	-111.241 dBr A	-127.362 dBr A

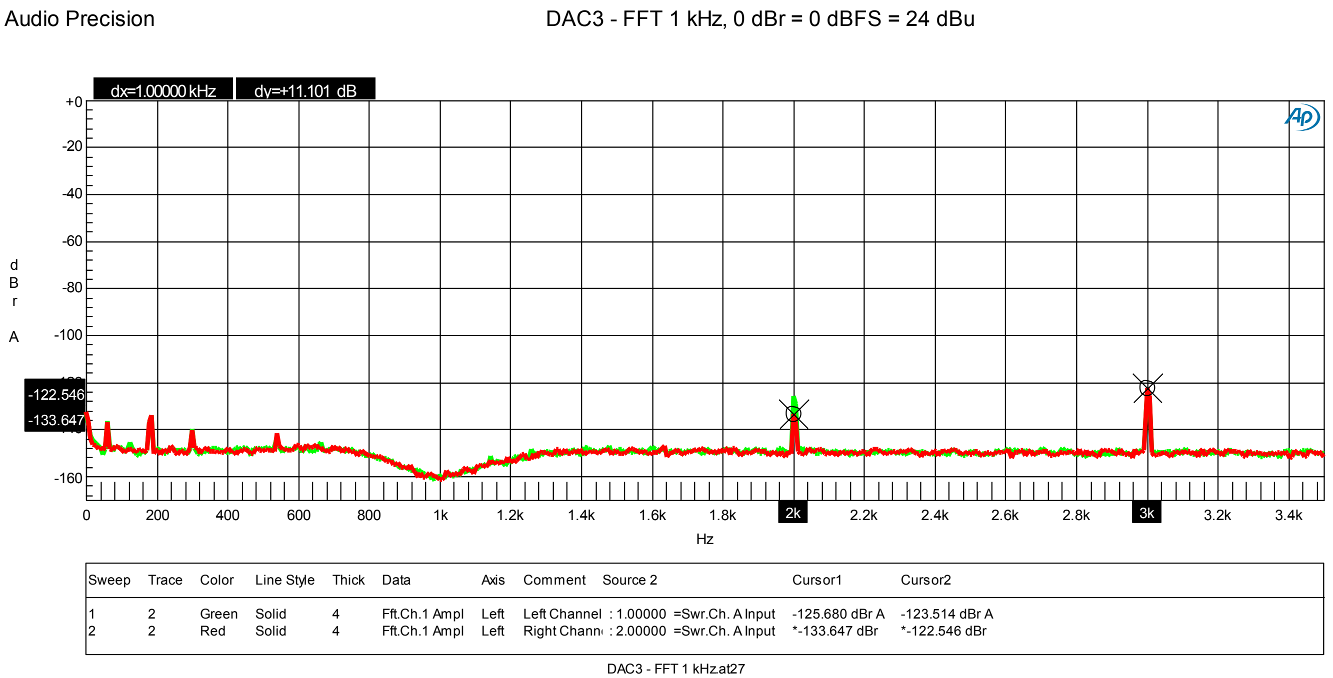
DAC3 - FFT 10 kHz.at27

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. For example, 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 -149 dB relative to the amplitude of the 10 kHz test tone.

Note the very low harmonic distortion; the 2nd harmonic (20 kHz) measured -111 dB, and the 3rd harmonic (30 kHz) measured -125 to -127 dB.

This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging. The 10 kHz fundamental has been removed by a notch filter in order to increase the resolution of the A/D converter in the AP2722 test set.

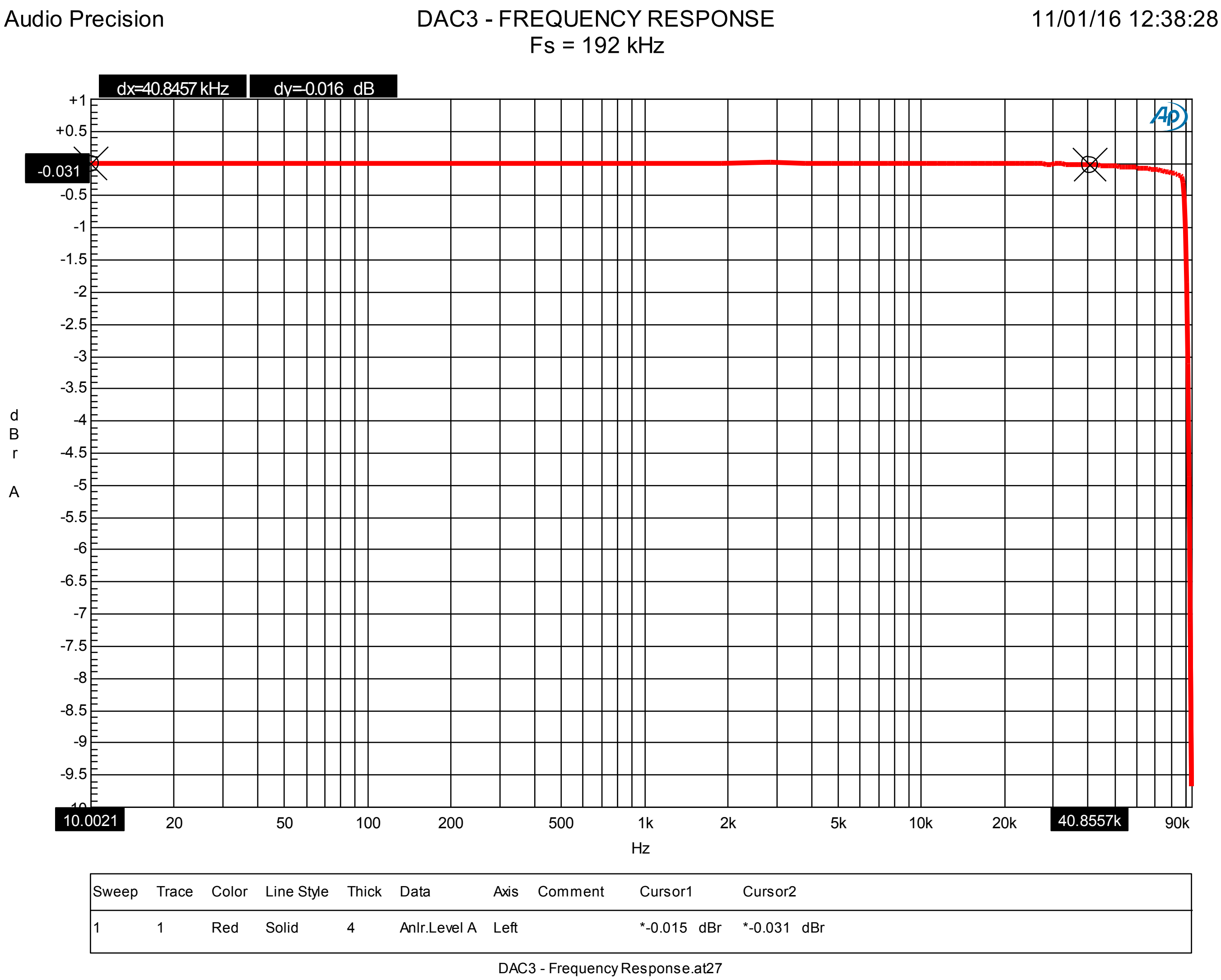


Graph 4 - FFT 1 kHz

The 1 kHz FFT analysis demonstrates the very low harmonic distortion of the ***DAC3***. The second harmonic distortion (2 kHz) measured better than -126 dB, while the 3rd harmonic distortion measured better than -122 dB relative to the amplitude of the 1 kHz test tone.

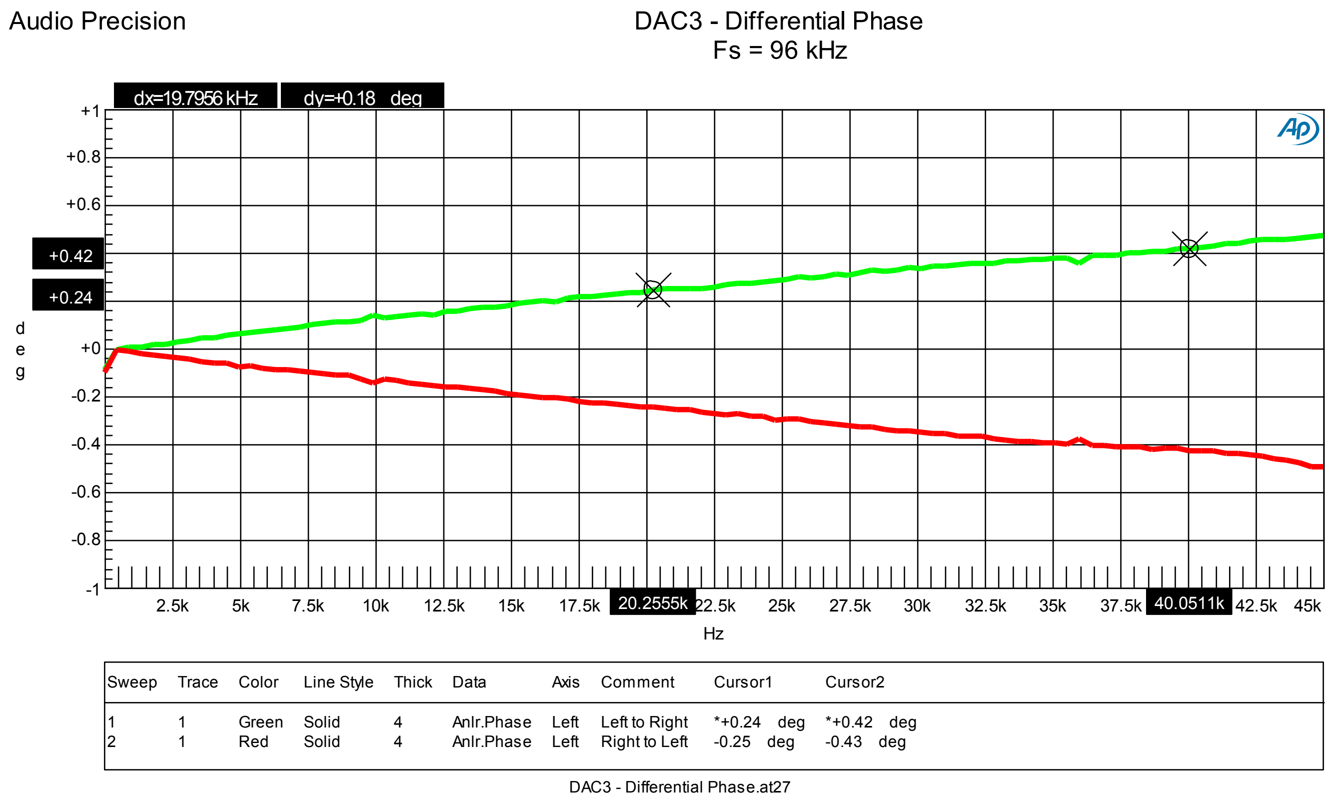
This 32k point FFT analysis uses a Blackman-Harris window with 16x power averaging. The 1 kHz fundamental has been removed by a notch filter in order to increase the resolution of the A/D converter in the AP2722 test set.

Note that under this full-output condition, the AC line-related hum frequencies (60 Hz, 180 Hz, and 240 Hz) measure better than - 133 dB.



Graph 5 - Frequency Response

This plot demonstrates the ruler-flat frequency response of the ***DAC3***. Note that the frequency response measures - 0.015 dB at 10 Hz and -0.031 dB at 40 kHz. The extreme low-frequency extension of the ***DAC3*** virtually eliminates the phase shifts that often occur at low frequencies (near 20 Hz). Bass is rendered in the proper timing relative to high-frequency content.

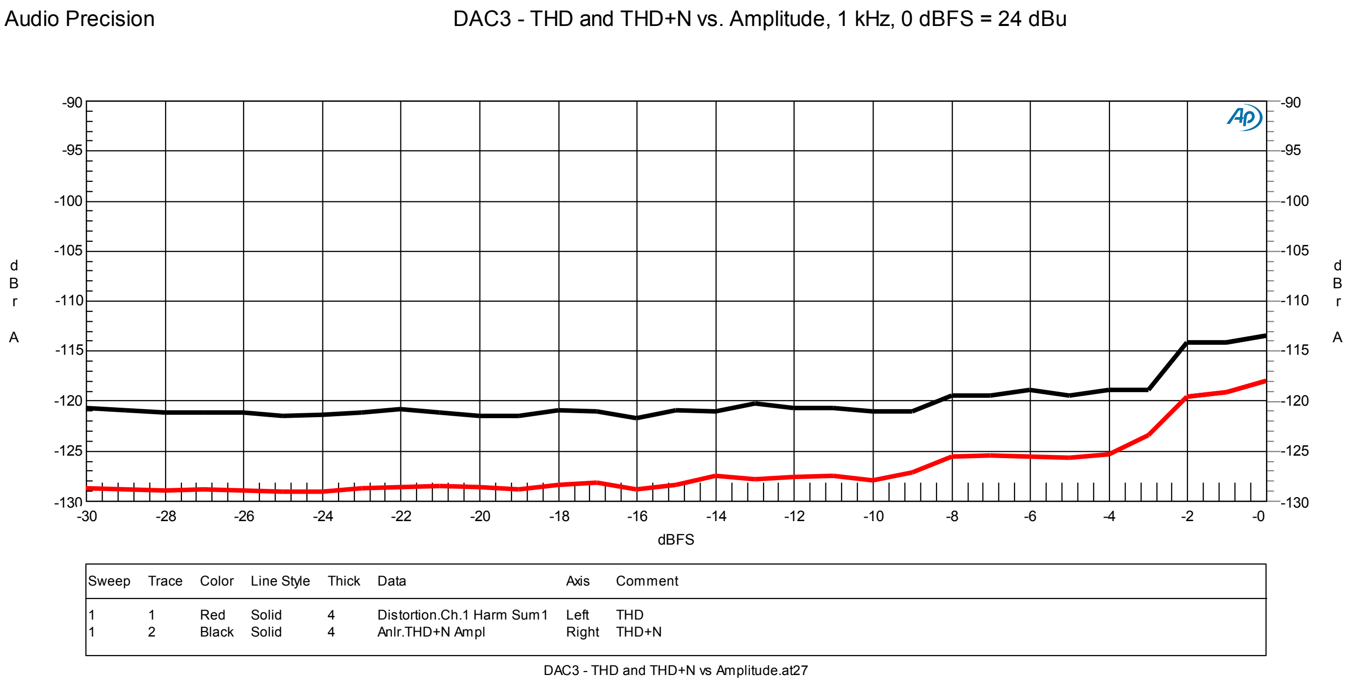


Graph 6 - Differential Phase

This plot demonstrates the inter-channel phase accuracy of the ***DAC3***. This is a highly expanded scale, spanning only +/- 1 degree at 45 kHz. From this plot, the inter-channel phase accuracy is calculated to be +/- 0.25 degrees at 20 kHz, and +/- 0.43 degrees at 40 kHz.

The phase accuracy of the ***DAC3*** approaches the phase accuracy of the Audio Precision AP2722 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.

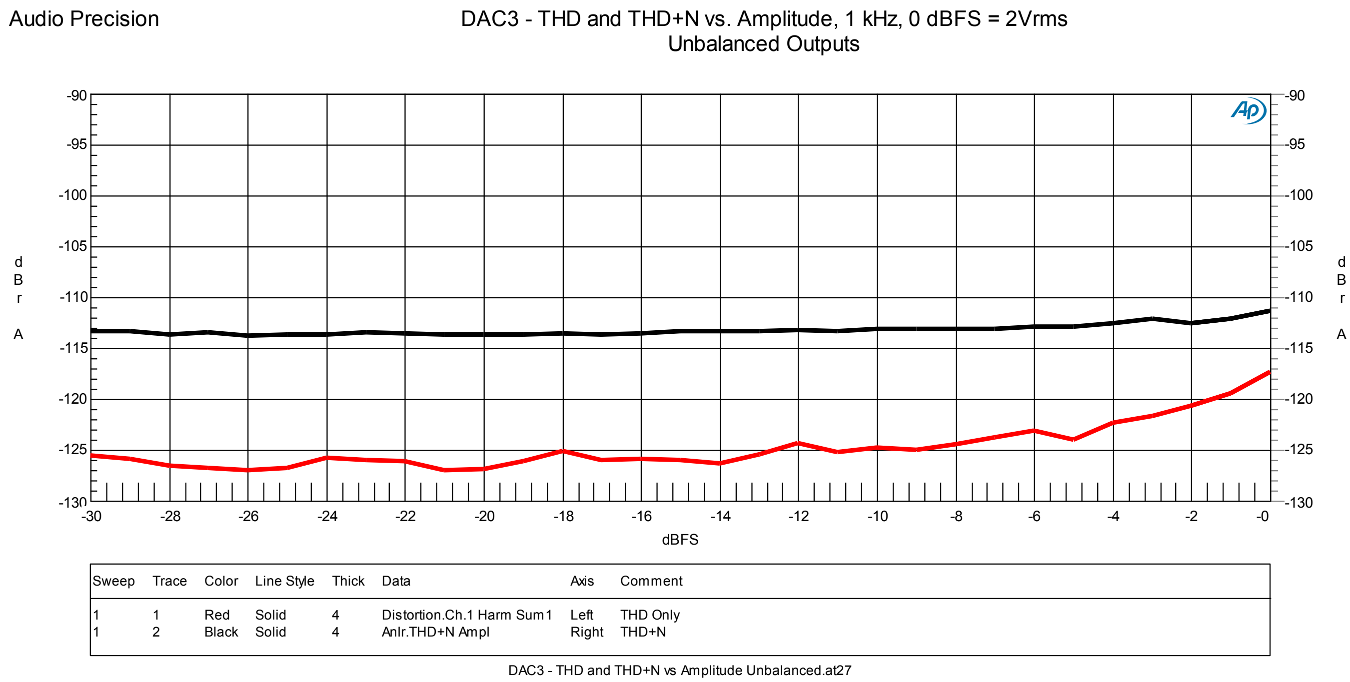
DAC3 converters are phase-accurate between boxes. Three ***DAC3*** converters can be used as a group to form a phase-accurate 5.1 surround system. Four ***DAC3*** converters can be combined to form a 7.1 system. The phase accuracy between any two channels will match the typical phase accuracy shown above.



Graph 7 - THD and THD+N vs. Amplitude (Balanced Outputs)

These plots demonstrate the very low harmonic distortion produced by the ***DAC3*** with digital input signal levels ranging from -30 dBFS to 0 dBFS. These plots show that THD should never reach the threshold of hearing in most listening environments. This is proof that the ***DAC3*** is virtually uncolored by any traces of harmonic distortion.

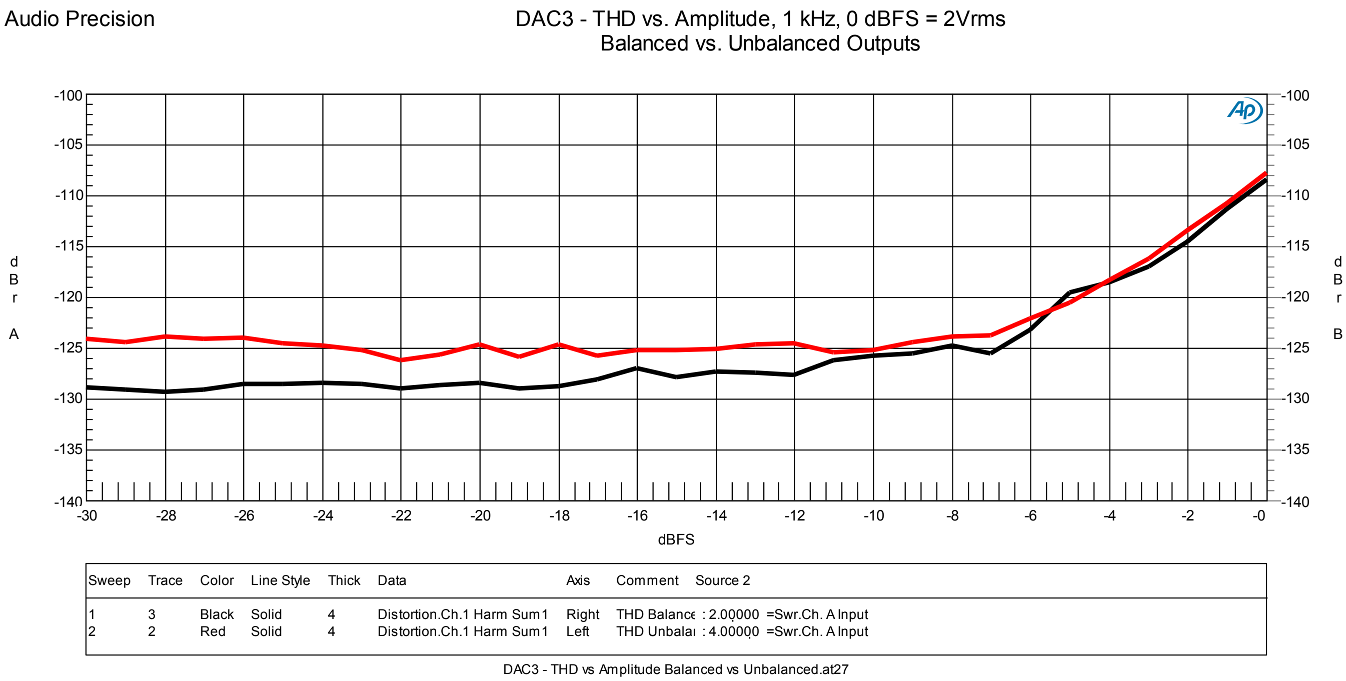
The black curve is a plot of THD+N, band limited to 22 kHz, and was acquired using the analog analyzer in the AP2722 test set. The red curve is a plot of THD (harmonic distortion only) and was acquired using the analog notch filter, A/D converter, and digital harmonic distortion analyzer in the AP2722 test set. This THD curve includes all harmonics falling below 32 kHz. The steps near -3 dBFS and -9 dBFS are due to the auto ranging of the AP2722 test set and are an indication that the THD performance of the ***DAC3*** is very close to the measurement limits of the AP2722.



Graph 8 - THD and THD+N vs. Amplitude (Unbalanced Outputs)

This plot shows the THD and THD+N performance of the unbalanced outputs.

Note that the THD performance of the unbalanced outputs approaches that of the balanced outputs. The THD+N levels are slightly higher on the unbalanced outputs. This is due to the relatively low (2 Vrms) signal levels used on unbalanced interfaces. This noise difference highlights one advantage of professional-level balanced interconnects.



Graph 9 - THD vs. Amplitude - Balanced vs. Unbalanced Outputs

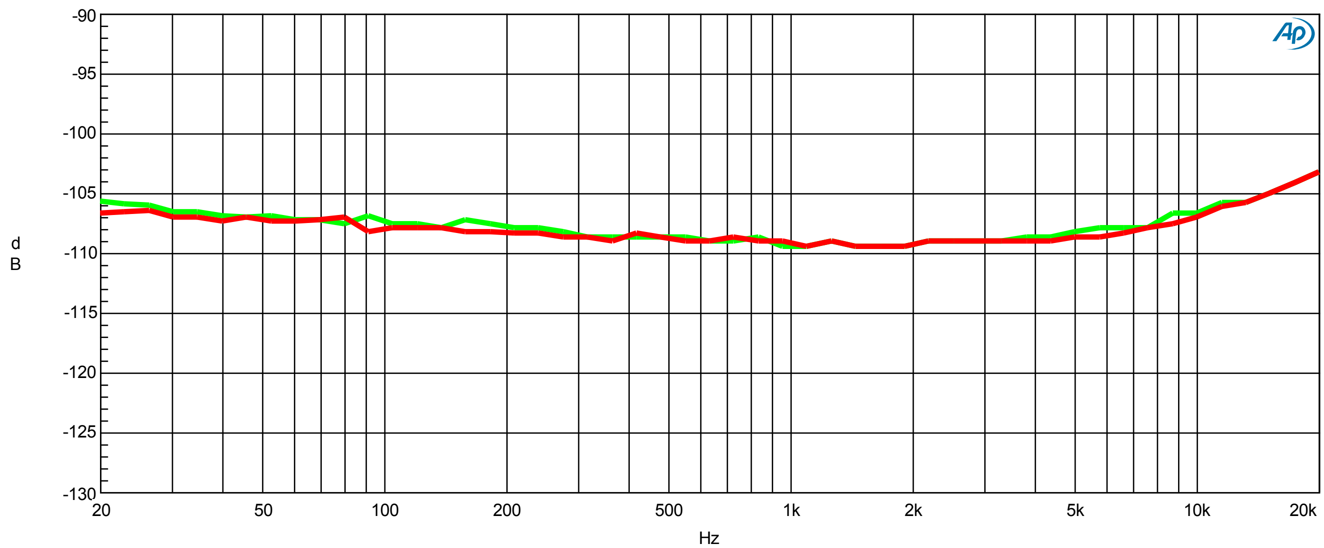
This plot demonstrates that the balanced and unbalanced analog outputs on the ***DAC3*** have very similar THD performance. The unbalanced outputs (red curve) closely match the performance of the balanced outputs (black curve) at high signal levels. The separation between the curves at signal levels below -10 dBFS is due to the improved SNR provided by the balanced interfaces. At 0 dBFS, the balanced interfaces were calibrated to +24 dBu (12.28 Vrms) while the unbalanced interfaces were calibrated to +8.24 dBu (2 Vrms). The higher signal levels used with the balanced interfaces make it easier to achieve high signal to noise ratios. This is just one reason why Benchmark recommends balanced interconnects.

The ***DAC3*** includes differential amplifiers that remove common-mode THD from the balanced outputs of the SABRE-PRO converters. These differential amplifiers give the unbalanced outputs the ability to approach the THD performance of the balanced outputs. Please note that the differential amplifiers also eliminate common-mode distortion on the balanced outputs.

Top-quality D/A conversion chips (such as the ES9028PRO) are equipped with balanced outputs. These balanced outputs allow a significant reduction of THD <u>if</u> they are followed by precision differential amplifiers. Conversion chips tend to produce significant common-mode distortion products that should be removed by a differential amplifier. The ***DAC3*** includes precision differential amplifiers following the outputs of the ES9028PRO. Many competing products omit these differential amplifiers. The omission of the differential amplifiers would make the THD much higher on the unbalanced outputs. The differential amplifiers also improve the system performance when the balanced outputs are driving balanced inputs that are not precisely trimmed.

The THD measurements shown above confirm the effectiveness of the differential amplifiers in the ***DAC3***. Additional confirmation can be obtained by measuring the THD of either side of the balanced outputs relative to ground.

Audio Precision DAC3 - THD+N VS FREQ AT 0 dBFS, 24 dBu (w/80 kHz LPF unweighted)
Balanced Outputs



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

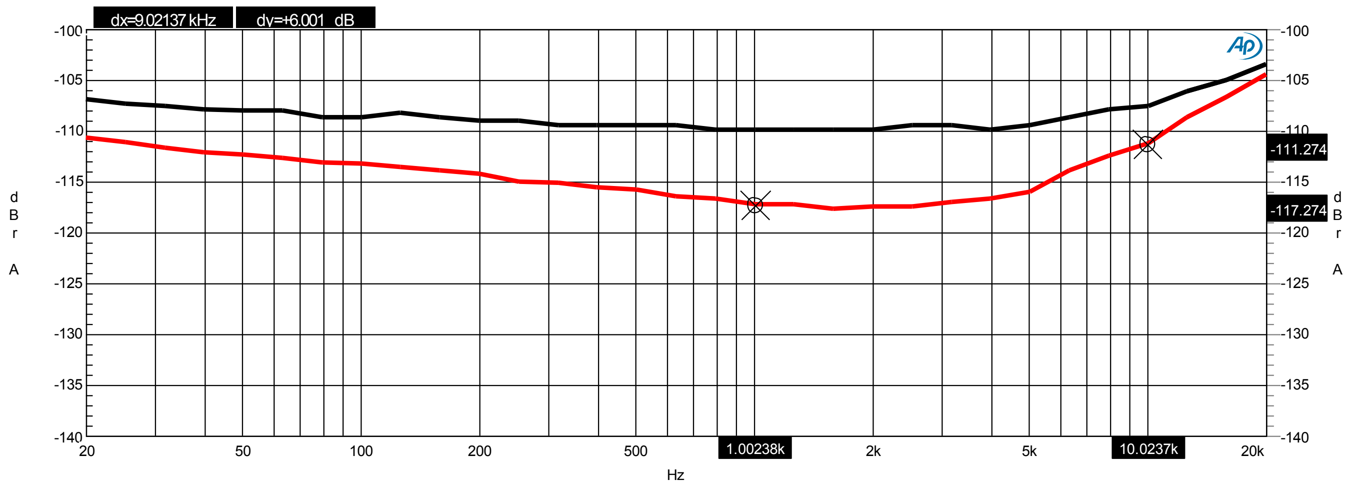
DAC3 - THD+N vs FREQUENCY 80kHZ.at27

Graph 10 - THD+N vs. Frequency 80 kHz

The analog output stages on the ***DAC3*** have high slew rates and are capable of maintaining low THD levels at high frequencies even when driven to 0 dBFS. Note that the THD+N remains very low at 20 kHz, even when operating at maximum output levels.

Audio Precision

DAC3 - THD and THD+N vs. Frequency, 1 kHz, 0 dBFS, 24 dBu
Digital In to Balanced Analog Out

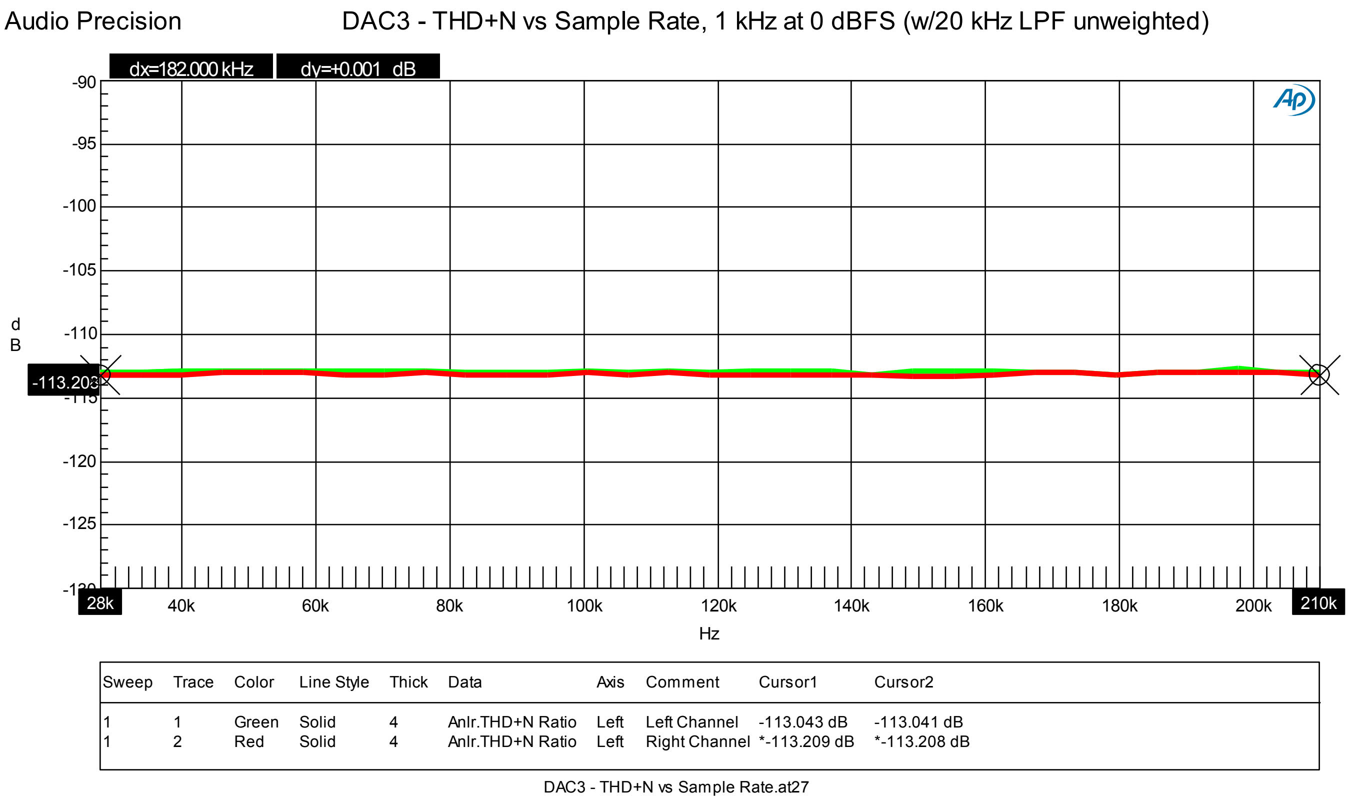


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Black	Solid	4	Anlr.THD+N Ampl	Left	THD+N, 80 kHz	-109.838 dBr A	-107.584 dBr A
1	2	Red	Solid	4	Distortion.Ch.1 Harm Sum1	Right	THD, 48 kHz LP	*-117.274 dBr	*-111.274 dBr

DAC3 - THD and THD+N vs Frequency Digital to Analog.at27

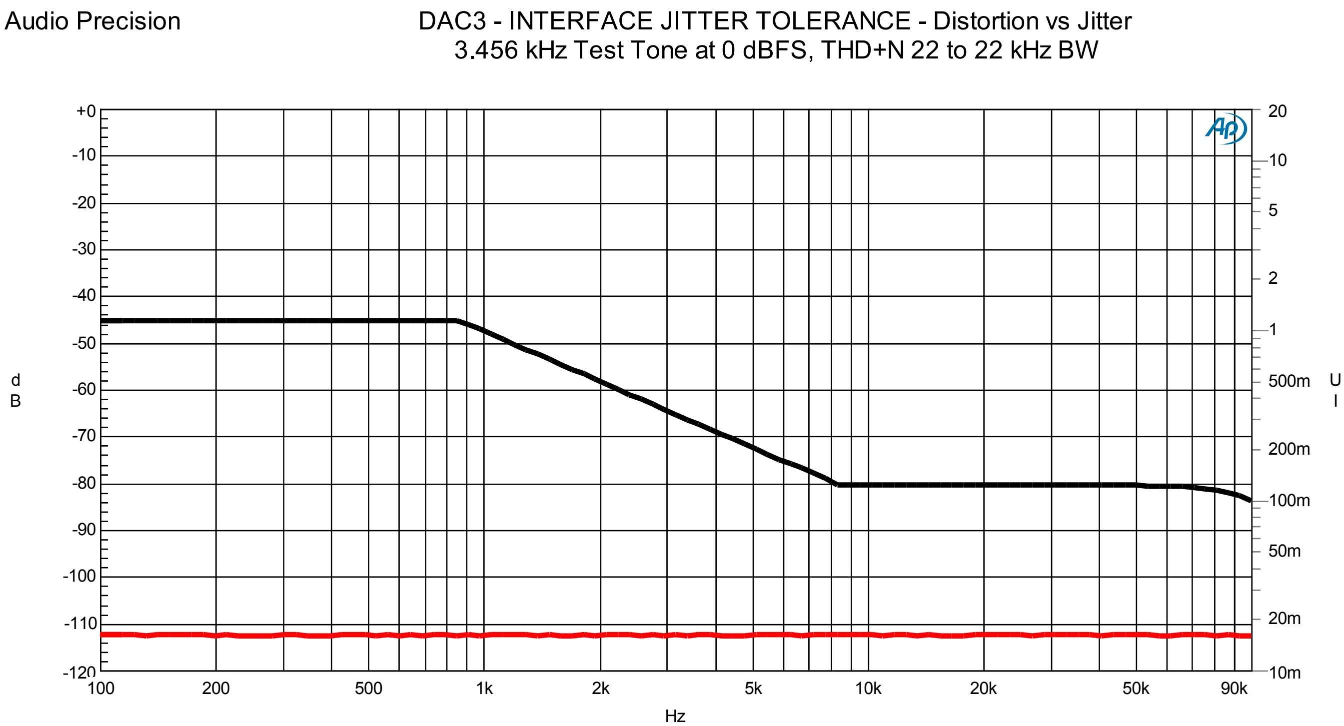
Graph 11 - THD and THD+N vs. Frequency

These plots demonstrate that the harmonic distortion of the ***DAC3*** is lower than the THD+N numbers would suggest. This black curve is a plot of THD+N. The red curve is a plot of THD only. Below about 10 kHz, the THD is so low that the THD+N measurement is dominated by noise.



Graph 12 - THD+N versus Sample Rate

The THD+N performance of the ***DAC3*** is identical at all Sample Rates.



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

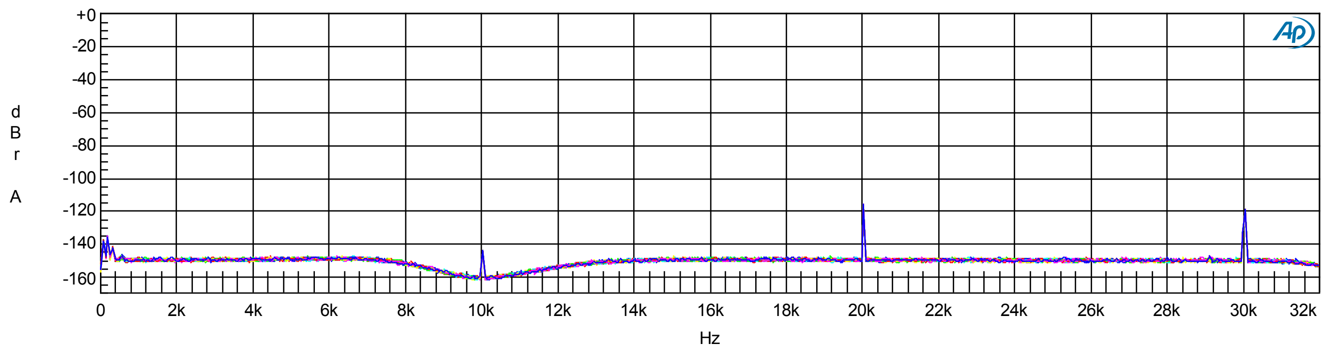
DAC3 - JITTER TOLERANCE.at27

Graph 13 - Jitter Tolerance

The Audio Engineering Society (AES) has created a jitter tolerance template for testing digital audio devices. The black curve shows the AES jitter tolerance template (see right-hand vertical axis). Induced jitter approaches 5 UI at 200 Hz, and is reduced to 0.125 UI above 8 kHz. The red trace shows the THD+N of the ***DAC3*** (left-hand vertical axis) while being driven with the jitter shown on the black curve. Over the entire range of the AES jitter tolerance test, the THD+N performance of the ***DAC3*** is unchanged. The ***DAC3*** easily passes the AES jitter tolerance test, and it does so without any performance degradation.

Audio Precision

DAC3 - 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	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 100.000 Hz=Dio.Jitter Freq
2	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 125.893 Hz=Dio.Jitter Freq
3	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 158.489 Hz=Dio.Jitter Freq
4	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 199.526 Hz=Dio.Jitter Freq
5	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 251.189 Hz=Dio.Jitter Freq
6	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 316.228 Hz=Dio.Jitter Freq
7	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 398.107 Hz=Dio.Jitter Freq
8	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 501.187 Hz=Dio.Jitter Freq
9	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 630.957 Hz=Dio.Jitter Freq
10	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 794.328 Hz=Dio.Jitter Freq
11	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 1.00000 kHz=Dio.Jitter Freq
12	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 1.25893 kHz=Dio.Jitter Freq
13	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 1.58489 kHz=Dio.Jitter Freq
14	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 1.99526 kHz=Dio.Jitter Freq
15	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 2.51189 kHz=Dio.Jitter Freq
16	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 3.16228 kHz=Dio.Jitter Freq
17	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 3.98107 kHz=Dio.Jitter Freq
18	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 5.01187 kHz=Dio.Jitter Freq
19	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 6.30957 kHz=Dio.Jitter Freq
20	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 7.94328 kHz=Dio.Jitter Freq
21	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 10.0000 kHz=Dio.Jitter Freq
22	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 12.5893 kHz=Dio.Jitter Freq
23	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 15.8489 kHz=Dio.Jitter Freq
24	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 19.9526 kHz=Dio.Jitter Freq
25	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 25.1189 kHz=Dio.Jitter Freq
26	2	Cyan	Solid	1	Fft.Ch.1 Ampl	Left		: 31.6228 kHz=Dio.Jitter Freq
27	2	Green	Solid	1	Fft.Ch.1 Ampl	Left		: 39.8107 kHz=Dio.Jitter Freq
28	2	Yellow	Solid	1	Fft.Ch.1 Ampl	Left		: 50.1187 kHz=Dio.Jitter Freq
29	2	Red	Solid	1	Fft.Ch.1 Ampl	Left		: 63.0957 kHz=Dio.Jitter Freq
30	2	Magenta	Solid	1	Fft.Ch.1 Ampl	Left		: 79.4328 kHz=Dio.Jitter Freq
31	2	Blue	Solid	1	Fft.Ch.1 Ampl	Left		: 100.000 kHz=Dio.Jitter Freq

DAC3 - JITTER TOLERANCE FFT.at27

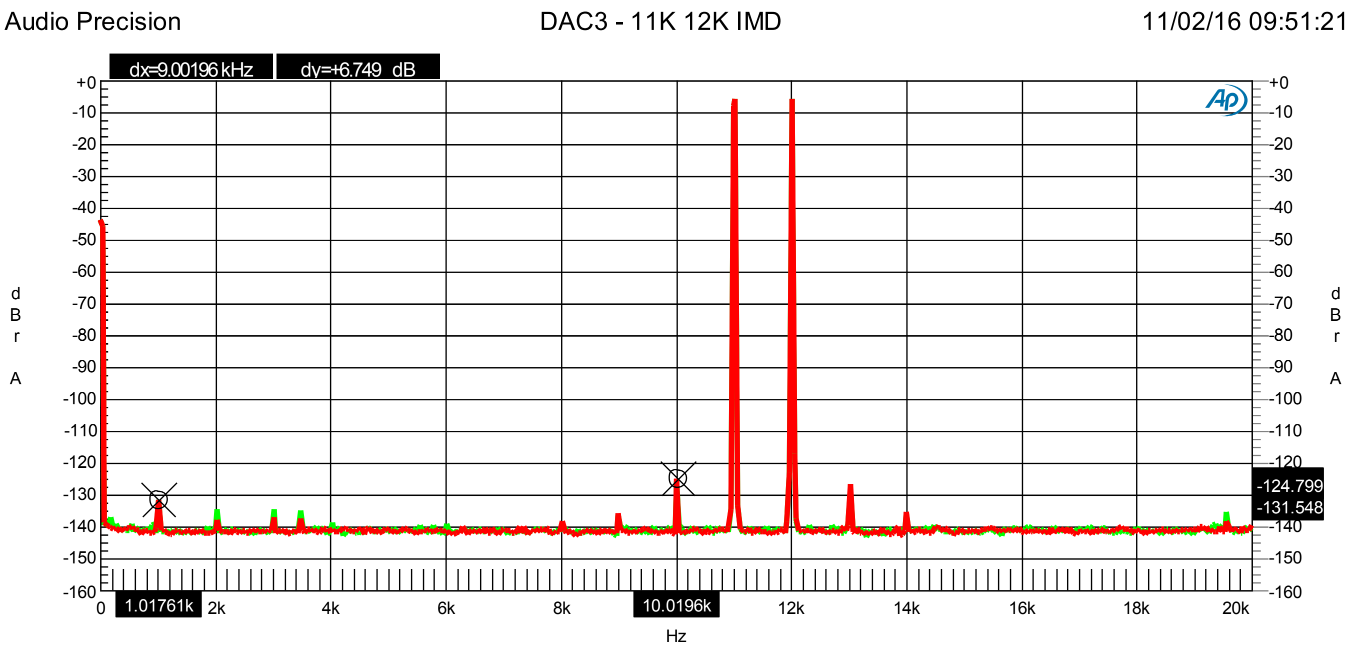
Graph 14 - Jitter Tolerance FFT

This figure shows a series of FFTs that were acquired while running the AES jitter tolerance test. This is the ultimate jitter-immunity 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**.**

The ***DAC3*** shows no change in performance when the AES jitter tolerance test is applied to the digital inputs. No jitter-induced sidebands are visible anywhere in this measurement.

Please note that the 10 kHz fundamental needed to be notched out in order to make this very high-resolution measurement. If the fundamental was not notched out, the spurious tones, produced by the A/D converter in the AP2722 test set, would have obscured the virtually-perfect results.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment	Cursor1	Cursor2
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	Left Channel	-131.652 dBr A	-125.861 dBr A
1	2	Red	Solid	4	Fft.Ch.2 Ampl	Right	Right Channel	*-131.548 dBr	*-124.799 dBr

DAC3 - IMD 11K 12K.at27

Graph 15 - IMD 11k 12K

This plot demonstrates that the ***DAC3*** has very low IMD distortion. The 1 kHz difference frequency measures -131 dB, and the 10 kHz and 13 kHz products measure about -125 dB. The actual IMD may be better. The results shown are partially limited by the performance of the AP2722 test set.

All IMD distortion products 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 = +24 dBu	124.5 dB, 128 dB relative to dig. clip
SNR – Unweighted, 0 dBFS = +24 dBu	122.5 dB, 126 dB relative to dig. clip
Dynamic Range - A-Weighted, 0 dBFS = +24 dBu	124.5 dB, 128 dB relative to dig. clip
Dynamic Range - Unweighted, 0 dBFS = +24 dBu	122.5 dB, 126 dB relative to dig. clip
THD+N, 1 kHz at 0 dBFS	-113 dBFS, -113 dB, 0.00022%
THD+N, 1 kHz at -1 dBFS	-114 dBFS, -113 dB, 0.00022%
THD+N, 1 kHz at –3 dBFS	-119 dBFS, -116 dB, 0.00016%
THD+N, 20 to 20 kHz sweep at –3 dBFS	-113 dBFS, -110 dB, 0.00032%
Frequency Response at Fs=192 kHz	+0 dB, -0.015 dB (20 to 20 kHz) -0.015 dB at 10 Hz -0.005 dB at 20 kHz -0.031 dB at 40 kHz -0.15 dB at 80 kHz
Frequency Response at Fs=48 kHz	+0 dB, -0.015 dB (20 to 20 kHz) -0.015 dB at 10 Hz -0.005 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	< -133 dB
Inter-channel Differential Phase (Stereo Pair – any sample rate)	+/- 0.25 degrees at 20 kHz
Inter-channel Differential Phase (Between ***DAC3*** Units Fs<110 kHz) Any sample rate.	+/- 0.25 degrees at 20 kHz
Maximum Lock Time after Fs change	6 ms
Soft Mute Ramp Up/Down Time	9.6 ms
Mute on Receive Error	Yes
Mute on Lock Error	Yes
Mute on Idle Channel	No
50/15 us De-Emphasis Enable	Automatic in Consumer Mode
De-Emphasis Method	Digital IIR
De-Emphasis Supported at	Fs = 32, 44.1, 48 kHz

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

Digital Audio Inputs	
Number of Digital Inputs (switch selected)	5 (1 USB, 2 Optical, 2 Coaxial)
Number of Channels	2
Input Sample Frequency Range	28 to 210 kHz (Coaxial Inputs) 28 to 96 kHz (Optical Inputs) 44.1, 48, 88.2, 96, 176.4, 192 kHz (USB Input)
Maximum Input Word Length	24 bits
Digital Input Impedance	75 Ohms (Coaxial Inputs)
DC Blocking Capacitors on Digital Inputs	Yes (Coaxial Inputs)
Transient and Over-Voltage Protection on Digital Inputs	Yes (Coaxial Inputs)
Minimum Digital Input Level	250 mVpp (Coaxial Inputs)

Jitter Tolerance	
(With no Measurable Change in Performance)	>12.75 UI sine, 100 Hz to 3 kHz >1.5 UI sine at 20 kHz >1.5 UI sine at 40 kHz >1.5 UI sine at 80 kHz >1.5 UI sine at 90 kHz >0.25 UI sine above 160 kHz
Jitter Attenuation Method	Benchmark ***UltraLock3***™ - 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	+29 dBu
Factory Calibration (with 0 dBFS digital input)	+24 dBu (Attenuator = 0 dB)
Maximum Amplitude of Intersample Peaks	+27.5 dBu (Attenuator = 0 dB) +17.5 dBu (Attenuator = 10 dB) +7.5 dBu (Attenuator = 20 dB)
Output Level Variation with Sample Rate	< +/- 0.006 dB

Unbalanced Analog Outputs	
Number of Unbalanced Analog Outputs	2
Output Connector	RCA
Output Impedance	30 Ohms
Analog Output Clip Point	+13.5 dBu (3.7 Vrms)
Factory Calibration (with 0 dBFS digital input)	+8.2 dBu (2 Vrms)
Maximum Amplitude of Intersample Peaks	+11.7 dBu (3 Vrms)
Output Level Variation with Sample Rate	< +/- 0.006 dB

Status Display	
Indicators - Type and Location	12 LEDs on Front Panel
Selection/Status Indication	1 – Control Lock 5 – Input 2 – Word length 4 – Sample Rate

AC Power Requirements	
Nominal Operating Range	100 – 240 VAC, 50 - 60 Hz
Min/Max Operating range	88 – 264 VAC, 47 - 63 Hz
Power	< 0.5 Watts Standby 12 Watts Typical Program 15 Watts Maximum
Fuses (2 required)	5x20 mm, 0.5 A 250 V Slo-Blo® Type

Dimensions	
Form Factor	½ Rack Wide, 1 RU High
Depth behind front panel	8.5″ (216 mm)
Overall depth including connectors	9.33″ (237 mm)
Width	9.5″ (249 mm)
Height	1.725″ (44.5 mm)

Weight	
DAC3 B only	3 lb.
DAC3 B with accessories and manual	4 lb.
Shipping weight	7 lb.

Regulatory Compliance

FCC and RoHS Compliance Statements

FCC Notice (U.S. Only)

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

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

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

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

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

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

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

RoHS Compliant Information

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

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

CE 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
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.

Revision A - 10/22/2018

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