Benchmark *DAC1 PRE*
Instruction Manual

2-Channel 24-bit 192-kHz
Digital-to-Analog Audio Converter
with Preamp Functions and USB Input

...the measure of excellence!™

Safety Information

Voltage Selection

CAUTION: THE FUSE DRAWER INCLUDES A VOLTAGE SELECTION SWITCH WITH TWO SETTINGS: '110' AND '220'. CHECK TO SEE THAT IT IS PROPERLY CONFIGURED FOR YOUR LOCATION BEFORE CONNECTING AC POWER.

Incorrect configuration may blow fuses or cause erratic operation.

Repairs

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

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).

Modifications

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

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

Contents

Features

- Reference-quality 2-channel 192-kHz 24-bit digital-to-analog audio converter
- "System Preamp" (control amplifier) functions include input selection and volume control
- Designed to connect directly to power amplifiers and/or powered monitors and/or preamps
- 6 stereo inputs – 1 analog (RCA), 1 computer (USB), 1 optical (TOSLINK), and 3 coaxial (RCA)
- 4 stereo analog outputs – balanced (XLR), unbalanced (RCA), and 2 headphone outputs
- Rotary input selector switch with standby on/off function
- Two reference-grade ***HPA2™*** "0-Ohm", high-current, ¼" headphone outputs
- Front-panel volume control for headphone outputs
- Front-panel volume control of all analog outputs (in **Variable** mode)
- Rear-panel **Variable/Calibrated** switch selects the volume/mute mode of analog outputs
- In **Calibrated** mode, output levels are set by 10-turn trimmers (20 dB Range, 2 dB/turn)
- Benchmark's ***AdvancedUSB Audio™*** input supports high-resolution 96-kHz 24-bit digital audio
- ***AdvancedUSB™*** is compatible with Windows and Mac computers without driver installation
- Coaxial digital inputs support professional (AES) and consumer (S/PDIF) data formats at up to 192-kHz, 24-bits
- Optical digital inputs support professional (AES) and consumer (S/PDIF) data formats at up to 96-kHz, 24-bits
- Benchmark's ***UltraLock™*** technology eliminates jitter on all digital inputs (including USB)
- Jumper-selected low-impedance 10, 20, or 30 dB pads on balanced outputs
- ***HPA2™*** gain jumpers for customizing headphone output gain for headphone sensitivities
- Left-most headphone jack auto-mutes XLR and RCA outputs (feature may be disabled)
- Status LED's - display input selection and error conditions
- Automatic **Standby Mode** – activated after 15 seconds of loss of digital input signal
- Instant wake-up from **Standby Mode** - no loss of audio
- Automatic de-emphasis in response to consumer pre-emphasis bit (44.1, 48, 88.2, and 96 kHz)
- 115 V, 230 V, 50-60 Hz international power supply with very wide operating range
- Low radiation toroidal power transformer significantly reduces hum and line related interference
- Low power consumption (8 Watts typical program, 16 Watts peak)
- Meets FCC Class B and CE emissions requirements

Overview

The ***DAC1 PRE*** is a reference-quality, 2-channel 192-kHz 24-bit digital-to-analog audio converter, stereo preamplifier, and headphone amplifier. It features Benchmark's ***AdvancedUSB Audio™*** technology, ***UltraLock™*** clock system, and ***HPA2™*** headphone amplifier.

DAC1 and *DAC1 USB* Heritage

The pristine audio path of the award-winning ***DAC1*** has made it the 'Benchmark' of stand-alone D/A converters. The ***DAC1 USB*** and ***DAC1 PRE*** preserve the exact topology of the DAC1 audio path while adding some of the most frequently requested features.

With the introduction of the ***DAC1 USB*** we added a very unique USB input with native 96/24 capability, an auto-mute function for headphone use, customizable headphone gain range, an automatic standby feature, and a high-current LM4562/LME49860 output stage designed to drive long cables and/or difficult loads, such as high-end power-amplifiers.

The ***DAC1 PRE*** adds the versatility of a stereo analog input and three S/PDIF digital inputs. National LM4562/LME49860 opamps are used throughout, and all RCA connectors are premium bulkhead mounted RCA connectors for maximum durability and superior grounding.

The ***DAC1 PRE*** looks, sounds, and measures the same as the ***DAC1*** and ***DAC1 USB***. We have added convenience and flexibility without altering the performance or changing the signal path.

Applications

The ***DAC1 PRE*** is designed for maximum transparency and is well suited for critical playback in studio control rooms, mastering rooms, and high-end audiophile applications.

Benchmark's ***AdvancedUSB Audio™*** interface makes the ***DAC1 PRE*** an ideal primary output device for digital audio workstations, desktop audio editing applications, computer-based media playback, home media servers, and computer-based radio broadcast systems.

A rugged rack-mount adapter makes the ***DAC1 PRE*** an excellent choice for location recording, broadcast facilities, and mobile trucks.

AdvancedUSB Audio™ Technology

The USB input is compatible with Windows Vista/XP/2000 and Mac OS X with no driver installation or system configuration required (see www.benchmarkmedia.com/wiki for up-to-date compatibility information).

Benchmark's ***AdvancedUSB Audio™*** technology supports sample rates up to 96 kHz and word lengths up to 24 bits.

The ***DAC1 PRE*** is a true plug-and-play solution, and it will be ready for playback immediately after the unit is connected to a USB port for the first time.

Jitter-Immune *UltraLock™*

The Benchmark ***UltraLock™*** system is nearly 100% jitter-immune. The D/A conversion clock is isolated from the input digital audio clock in a topology that outperforms two-stage PLL designs. In fact, no jitter-induced artifacts can be detected using an Audio Precision System 2 Cascade test set. Measurement limits include detection of artifacts as low as -140 dBFS, application of jitter amplitudes as high as 12.75 unit intervals (UI) and application of jitter over a frequency range of 2 Hz to 200 kHz.

Any signal that can be decoded by the USB or AES/EBU receivers will be reproduced without the addition of any measurable jitter artifacts. The AES/EBU receiver IC has been selected for its ability to accurately recover data in the presence of very high jitter levels.

HPA2™ Headphone Amplifier

Two ¼" headphone jacks are driven by the ***HPA2***™ - Benchmark's signature high-current, 0-Ohm headphone amplifier. The ***HPA2™*** is capable of delivering the full performance of the ***DAC1 PRE*** into the difficult loading presented by headphones. The ***HPA2***™ maintains less than 0.0003% THD+N under full load.

High-Current Output Drivers

The ***DAC1 PRE*** features new 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.

'Audio-Always' Design Philosophy

The ***DAC1 PRE*** is designed to perform gracefully in the presence of errors and interruptions at the digital audio inputs. A soft mute circuit eliminates pops when a digital signal is applied. Power management circuitry controls the muting and resetting of all digital circuits upon removal and application of power. Audio is present at the outputs only 60 ms after applying, selecting, or restoring a digital input signal and only 500 ms after applying power to the unit.

The ***DAC1 PRE*** is designed to avoid all unnecessary mute scenarios. Muting is only enabled upon loss of power, or when digital transmission errors occur. The ***DAC1 PRE*** does not mute when the AES or S/PDIF input data is all zeros. Consequently, no audio is lost when an audio transient follows full silence. Furthermore, the ***DAC1 PRE*** signal-to-noise specifications represent the true system performance, not just the performance of an output mute circuit.

The ***DAC1 PRE*** will operate even when sample rate status bits are set incorrectly. Sample rate is determined by measuring the incoming signal. Lack of sample rate status bits or incorrectly set status bits will not cause loss of audio.

The ***DAC1 PRE*** includes non-volatile memory that saves the state of control settings when AC power is removed for a period of up to several hours. The unit will resume normal operation after interruptions in AC power.

Low-Noise Internal Power Supply

The internal power supply supports all international voltages with generous margins for over and under voltage conditions. It has excellent immunity to noise on the AC line and no external AC filtering is required.

Phase-Accurate Multi-Track and 5.1

The ***DAC1*** is phase-accurate between channels at all sample rates, and is phase accurate between any combination of ***DAC1***, ***DAC1 USB***, and ***DAC1 PRE*** converters at sample rates up to 96 kHz. Phase-accurate multi-track and 5.1 surround systems are easily constructed using any combination of ***DAC1*** series converters.

Automatic Digital De-Emphasis

Pre-emphasis was used on many early CD recordings. It is rarely used on newer recordings and consequently some D/A converters omit de-emphasis. The ***DAC1 PRE*** will correctly apply precise digital de-emphasis when and if it is needed. The de-emphasis circuit supports 44.1, 48, 88.2 and 96-kHz sample rates and is automatically enabled in response to the pre-emphasis status bits in consumer format digital signals.

Quick Start Guide

Audio Inputs

The DAC1 PRE features one stereo analog input (RCA) and five stereo digital inputs (3 x coaxial, 1 x optical, and 1 x USB). The coaxial and optical digital inputs accept professional (AES) and consumer (S/PDIF) data formats at word lengths up to 24-bits. The coaxial inputs support sample rates up to 192 kHz. The optical inputs support sample rates up to 96 kHz.

Volume Control

The front-panel **Volume Control** sets the output level of the headphone jacks, and can also be used to control the output level of the main outputs (balanced XLR and unbalanced RCA analog) when in **Variable** output mode.

A rear-panel switch selects **Variable** or **Calibrated** output mode. In **Variable** output mode, all analog outputs are controlled by the **Volume Control**. In **Calibrated** output mode, the volume is fixed at the level set by the calibration trim-pots.

Direct Interfacing to Power Amplifiers

The ***DAC1 PRE*** is designed to be able to interface directly to power amps and powered studio monitors. This configuration provides the cleanest and shortest path from the digital source to the monitor output, and often results in a substantial improvement in sound quality.

The DAC1 PRE is equipped with 10, 20, and 30 dB output attenuators for optimal interfacing. The pads optimize the output signal level of the ***DAC1 PRE*** for the input sensitivity of virtually any load (amplifier, preamp, etc). Most power amplifiers and powered monitors require the 20 dB pad setting. The ***DAC1 PRE*** is factory-set with the 20 dB pad enabled.

Source Selector and On/OFF Switch

A rotary Source Selector control selects any of the 6 inputs to the ***DAC1 PRE***. Pressing the **Source Selector** toggles the ***DAC1 PRE*** on and off. The on/off function features a very fast soft mute/un-mute and doubles as a mute control.

Input Status Display

Under normal operation, the **Input Status Display** shows which of the 6 inputs is currently selected. A steady light indicates that a normal signal is present. Flashing lights indicate error conditions. If the error condition continues, the automatic-standby mode will begin.

Automatic Standby/Resume

The ***DAC1 PRE*** features an automatic standby mode that eliminates the need to turn the converter on and off. **Standby Mode** starts 15 seconds after a digital source device is turned off, disconnected, or contains errors that prevent D/A conversion. All lights are off while in **Standby Mode**.

While in **Standby Mode**, the ***DAC1 PRE*** continues to monitor the selected digital input and will immediately resume normal operation when an error-free signal is restored.

Mute on Headphone Insertion

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

Front Panel

Input Status Display

The ***DAC1 PRE*** has a six-LED input-selection indicator on the front panel. These LED's flash when an error condition occurs on a selected digital input. All LED's turn off when the ***DAC1 PRE*** is in **Standby Mode** or is turned off.

The numbers next to the LED's match the numbers adjacent to the digital connectors on the rear panel. Digital input "1" is TOSLINK Optical. Inputs 2, 3, and 4 are RCA Coaxial.

Error Indication

The **Input Status Display** will flash when an error occurs on the selected digital input. The type of error is indicated by the number of flashes before standby engages.

Error Codes:

- No signal – 16 slow flashes – audio muted
- Data transmission errors - 16 flashes – audio muted
- Non-PCM – 16 flashes – audio muted
- Non-audio – 32 rapid flashes – audio muted
- Invalid sample (v-bit) – 64 very rapid flashes – no mute

Common causes of errors are:

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

If the error is not resolved within +/- 15 seconds, the ***DAC1 PRE*** will enter **Standby Mode**. The ***DAC1 PRE*** will resume normal operation when it detects a valid input signal at the last chosen input. There is no error indication on the analog input.

"Source" and "ON/OFF" Switch

The rotary **Source Selector** control is located directly to the right of the **Input Status Display**. Rotate the knob to select an input.

The rotary **Source Selector** switch is equipped with an on/off switch. Press the control knob to turn the ***DAC1 PRE*** on or off.

The on/off function features a soft mute and soft un-mute function that responds very quickly. Because of this fast response, the on/off function also serves as a mute control. Press the **Source Selector** switch to mute all audio outputs. Press again to restore all audio outputs.

No analog or digital audio signals are routed through the Source Selector switch. Source selection is transparent and free from crosstalk.

If the ***DAC1 PRE*** is off or in **Standby Mode** it will resume normal operation when the **Source Selector** is rotated or pressed.

Standby Mode

The ***DAC1 PRE*** features an automatic standby mode that eliminates the need to turn the converter on and off. **Standby Mode** starts 15 seconds after the selected digital source device is turned off, disconnected, or contains errors that prevent D/A conversion. All status LED's are off while in **Standby Mode**.

While in **Standby Mode**, the ***DAC1 PRE*** continues to monitor the selected digital input and will immediately resume normal operation when an error-free signal is restored.

HPA2™ Headphone Jacks

The DAC1 PRE features two headphone jacks. The left-hand jack is equipped with a switch that automatically mutes the XLR and RCA analog outputs when a headphone plug is inserted. The right-hand jack has no switch. This feature enables seamless muting of the main outputs when headphones are being used. This auto-mute feature can be enabled or disabled via an internal jumper. Instructions for setting the auto-mute jumper are detailed in the '**Internal Settings**' section of this manual.

TIP: Use the left-hand jack when you want to listen to headphones and mute your playback system. Use the right-hand jack when you need to keep all outputs active.

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

The original gain-range of the ***HPA2***™ (such as in the original ***DAC1***) is often too high for the headphones of many users. The ***DAC1 PRE*** features three gain ranges for the ***HPA2***™ to suit the sensitivity of any particular headphones. These gain ranges are set using internal jumpers. The jumpers reduce the input to the ***HPA2***™ by 0, 10, or 20 dB. These jumpers are factory-installed at 10 dB below full gain. Instructions for setting the headphone gain range are detailed in the 'Internal Settings' section of this manual.

TIP: For optimal performance, the headphone gain jumpers should be set so that comfortable listening levels occur when the 'Volume Control' is set above the 10th detent.

Volume Control

The front-panel **Volume Control** is a 41-detent potentiometer (see 'Volume Control Curve' in the 'Performance Graphs' section of this manual).

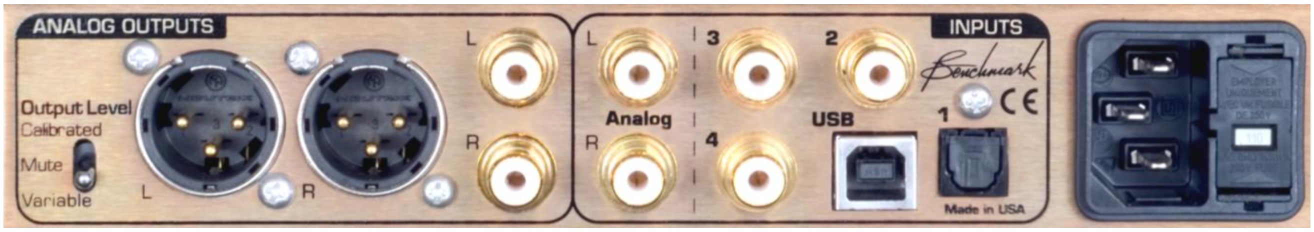
The **Volume Control** always controls the output level of the headphone jacks. It can also be used to control the output levels of the balanced XLR and unbalanced RCA analog outputs when the rear-panel **Output Level Switch** is set to **Variable.**

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

TIP: For optimal performance and minimal noise, the XLR gain jumpers should be set so that comfortable listening levels occur when the 'Volume Control' is set above the 10th detent.

Instructions for setting the XLR pads are detailed in the 'Internal Settings' section of this manual. The *DAC1 PRE* is shipped with the XLR attenuation set to -20 dB.

Rear Panel



Inputs

There are six stereo inputs on the ***DAC1 PRE***: 1 x Analog, 1 x USB, 1 x Optical, and 3 x Coaxial. These inputs are selected using the front-panel **Source Selector** control.

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

TIP: The *DAC1 PRE* will not decode AC3 or ADAT signals. The 'Status Display' will flash when AC3, ADAT, or other non-PCM input signals are connected to the selected digital input.

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

Analog Input – RCA Unbalanced

The DAC1 PRE has an unbalanced stereo analog input via a pair of RCA connectors.

The analog inputs can be used for devices such as:

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

Computer Input – USB

The USB input accepts a 'B-type' male USB 1.1 or USB 2.0 connector. An 'A-B type' USB cable is provided with the ***DAC1 PRE***. The USB cable connects the ***DAC1 PRE*** directly to a computer's USB output. The USB interface utilizes USB 1.1 protocol, and is compatible with both USB 1.1 and USB 2.0 ports.

The USB input supports 44.1, 48, 88.2 and 96 kHz sample rates at word lengths up to 24-bits. The USB interface acts as a 'native' USB audio device and does not require the installation of any custom drivers.

Benchmark's ***AdvancedUSB Audio™*** technology achieves bit-transparent operation without special drivers and without changing system settings.

The Benchmark USB interface is truly a plug-and-play solution. The ***DAC1 PRE*** can begin streaming high resolution audio bit-transparently within seconds after being plugged into a computer for the first time. No software or hardware configuration is required.

The ***DAC1 USB*** is designed, tested and proven compatible with Windows Vista/XP/2000 and Mac OS X with no driver installation or system configuration required. For the up-to-date information about more recent operating systems and suggestions for optimization, go to: www.benchmarkmedia.com/wiki.

TIP – Visit our computer audio application pages for the latest information on media players, media servers, operating systems, and audio-related computer accessories: www.benchmarkmedia.com/wiki

These pages include instructions for maximizing the performance of media servers.

Digital Input 1 – Optical

The optical input connector is manufactured by Toshiba and is commonly known as a TOSLINK connector. The TOSLINK optical connector used on the ***DAC1 PRE*** is designed to work well at sample rates up to 96 kHz. Maximum word length is 24-bits. All sample rates between 28 and 96 kHz are supported. The optical input will accept professional AES/EBU data formats or consumer S/PDIF data formats.

Digital Inputs 2, 3, and 4 - Coaxial

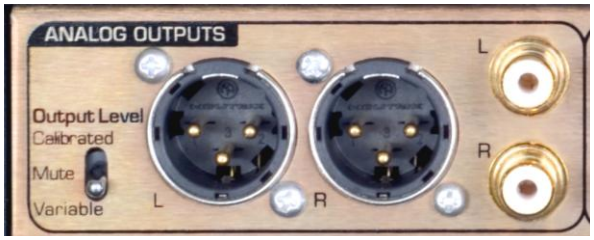
The coaxial inputs use female RCA connectors that are securely mounted directly to the rear panel. The input impedance is 75 Ohms. Maximum word length is 24-bits. All sample rates between 28 and 195 kHz are supported.

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

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

The Coaxial inputs accept AES/EBU or S/PDIF digital audio formats. AES3-id and SMPTE 276M standards specify 75-Ohm 1 Vpp professional format digital audio signals and these are commonly used in video production facilities. IEC 609588-3 specifies 75-Ohm 0.5 Vpp consumer-format digital audio signals (commonly known as S/PDIF). The coaxial inputs on the ***DAC1 PRE*** are designed to accept either type of signal.

Analog Outputs



The ***DAC1 PRE*** has two unbalanced RCA outputs and two balanced XLR outputs.

The ***DAC1 PRE*** features new 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.

Note: The XLR and RCA output levels are controlled by the **Volume Control** level when the **Output Level Switch** is set to variable. The levels are set by the 10-turn calibration trimmers located internally on the circuit board, when the **Output Level Switch** is set to **Calibrated**. The XLR and RCA outputs are muted, when the switch is in the center position.

The XLR outputs are equipped with low-impedance passive output attenuators that can be set at 0 dB, -10 dB, -20 dB, or -30 dB to allow interfacing to a wide variety of audio devices without any loss of dynamic range.

Output Level Switch

The **Output Level Switch** is a three-position toggle switch located on the rear panel. The ***DAC1 PRE*** ships with this switch set in the **Variable** position.

CAUTION: Do not set the 'Output Level Switch' to 'Calibrated' if you are directly driving a power amplifier or powered speakers. The 'Calibrated' setting disables the front panel volume control and will produce an output that may be too loud for your speakers.

Calibrated (UP) – Analog output levels are controlled by 10-turn internal trim controls (see page 14 for information on calibration trimming).

Off (CENTER) – Analog XLR and RCA outputs are muted; headphone outputs remain active.

Variable (DOWN) – Analog output levels are controlled by the **Volume Control**.

The **Output Level Switch** does not affect the operation of the headphone jacks (the headphone outputs are never disabled and the headphone level is always controlled from the **Volume Control**).

TIP: If the *DAC1 PRE* is being used in a critical signal chain (such as a broadcast facility or theater) the headphone mute switch should be defeated using the internal jumpers. See 'Internal Settings' section for instructions.

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 systems. This direct bonding also maximizes RF shielding.

The XLR output levels may be controlled from the front panel, or may be set to fixed levels using the internal **Calibration Trimmers**.

The XLR outputs have passive attenuators that allow direct connections to a wide variety of audio devices without a loss of dynamic range. The 20 dB pad is usually required for direct interfacing to power amplifiers and powered speakers. The ***DAC1 PRE*** ships with the 20 dB pad enabled.

Industry-standard XLR wiring:

XLR pin 2 = + Audio Out

XLR pin 3 = - Audio Out

XLR pin 1 = Cable Shield

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

Unbalanced RCA Analog Outputs

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

The RCA output levels may be controlled from the front panel, or may be set to fixed levels using the internal **Calibration Trimmers**. In **Calibrated** mode the RCA outputs are factory preset to –10 dBV at -20 dBFS. This is typical for most consumer-grade equipment.

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

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

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

TIP: The RCA outputs are capable of driving cables as long as 1360 feet (see Table 1). But, long un-balanced cables will generally suffer from hum problems due to ground loops. We highly recommend using balanced interconnects for long runs.

Low-Impedance Passive Pads

The XLR outputs are equipped with low-impedance passive pads that may be used to reduce the output levels while preserving the full dynamic range of the ***DAC1 PRE***. The ***DAC1 PRE*** ships with the 20 dB pads enabled.

TIP: When directly driving power amplifiers and powered speakers, use 'Variable' mode and start with the factory default 20 dB pad setting. If necessary, change the pads to achieve a normal listening level when the 'Volume Control' is near the 12 o'clock position.

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

Table 1 - Cable Drive Capability

Balanced Output Drive Capability:

Attenuator Setting (dB)	Output Impedance	Maximum Cable (ft)	Loss in dB at 20 kHz
0	60	680	0.1
10	425	96	0.1
20	135	302	0.1
30	43	949	0.1

Unbalanced Output Drive Capability:

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

TIP: To set the XLR outputs to typical professional studio levels, set the pads to 0 dB, and set the 'Output Level Switch' to 'Calibrated'. If the factory settings of the 'Calibration Trimmers' have not been changed, the XLR outputs will be calibrated to +4 dBu at -20 dBFS, and the RCA outputs will be calibrated to -10 dBV at -16 dBFS.

Calibration Trimmers

The **Calibration Trimmers** are located internally on the circuit board behind the **Output Level Switch**. They are 10-turn trimmers and are adjustable using a small screwdriver.

These trimmers provide a 2 dB per rotation adjustment with a total control range of + 9 to +29 dBu at 0 dBFS (full-scale digital input). There are no stops at either end of the 10-turn rotation.

CAUTION: Do not change the calibration trimmers unless you have the ability to accurately measure audio levels.

Factory calibration has been set so that the output level at the balanced XLR connectors is +4 dBu at -0 dBFS. This is exactly 20 dB lower than a typical alignment of +4 dBu at -20 dBFS. The lower level is appropriate for most powered monitors.

TIP: To set the XLR outputs to typical professional studio levels, set the pads to 0 dB, and set the 'Output Level Switch' to 'Calibrated'. If the factory settings of the 'Calibration Trimmers' have not been changed, the XLR outputs will be calibrated to +4 dBu at -20 dBFS, and the RCA outputs will be calibrated to -10 dBV at -16 dBFS.

The factory-preset levels may be increased by 5 dB or decreased by 15 dB in order to conform to other studio reference levels. This range of levels is also well suited for direct connection to the balanced line-level inputs on most power amplifiers. Most professional equipment will work well at these levels.

Note: The **Calibration Trimmers** have no effect on the output levels when the **Output Level Switch** is set to **Variable**.

AC Power-Entry and Fuse Module

The AC power input uses a standard IEC type connector. One USA-compatible power cord is included with ***DAC1 PRE*** converters shipped to North America. IEC style power cords in country-specific configurations are available in your locality.

Fuse Holder

The fuse holder is built into a drawer next to the IEC power connector. The drawer requires two 5 x 20 mm 250 V Slo-Blo® Type fuses. The drawer includes a voltage selection switch with two settings: **110** and **220**. The fuse rating for all voltage settings is 0.50 Amps.

The AC input has a very wide input voltage range and can operate over a frequency range of 50 to 60 Hz. At **110**, the ***DAC1 PRE*** will operate normally over a range of 90 to 140 VAC. At **220**, the ***DAC1 PRE*** will operate normally over a range of 175 to 285 VAC.

Caution: Always install the correct fuses. Always insure that the voltage setting is correct for your locality.

Internal Settings

Removing Top Cover

The ***DAC1 PRE*** cover must be removed to gain access to the jumpers. Do not attempt to remove the faceplate or rear panel.

CAUTION: The *DAC1 PRE* contains static sensitive components and should only be opened by qualified technicians. Static discharge may cause component failures, may affect the long-term reliability, or may degrade the audio performance. Use a static control wrist strap when changing jumper settings.

CAUTION:

- **Disconnect AC power by unplugging the power cord at the back of the *DAC1 PRE*.**
- **Remove only the 8 screws holding the cover (4 on each side).**
- **Do not remove any screws on front or rear panels.**
- **Never remove the power entry safety cover in the rear corner of the *DAC1 PRE*.**
- **Always connect a static-control wrist strap to the chassis before touching any internal component.**

Jumpers

The following functions are jumper configured:

- Headphone Gain Range Adjustment
- Headphone Switch Disable
- XLR Output Pads

XLR Output Pad Selection (P5, P6, P7, and P8):

Four 8-pin headers (P5, P6, P7, and P8) allow selection of the output level at the XLR jacks.

One pair of 8-pin headers controls the output level at each XLR jack as follows:

- 0 dB - (Attenuator disabled) – (Jumper plug between pins 1 and 2 of each header)
- -10 dB – (Jumper plug between pins 3 and 4 of each header)
- -20 dB – ***(Jumper plug between pins 5 and 6 of each header)
- -30 dB – (Jumper plug between pins 7 and 8 of each header)

***** = Factory Default**

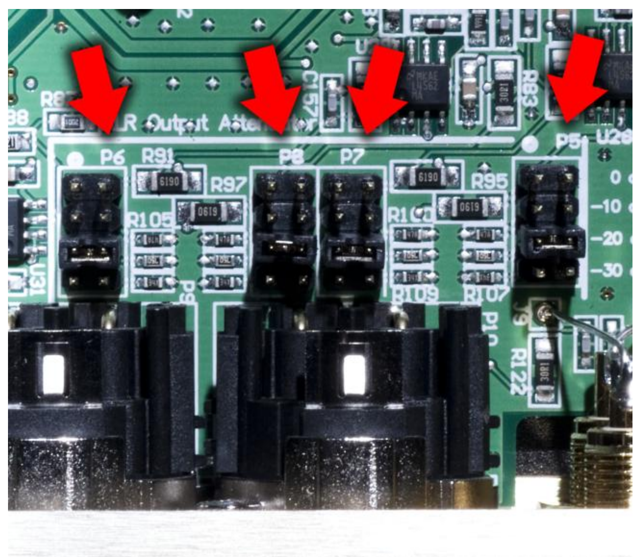


Photo 1 - XLR Output Pad Selection (P5, P6, P7, and P8)

Headphone Switch Disable (JP2 and JP4):

The ***DAC1 PRE*** is configured so that the analog outputs will mute when a headphone plug is inserted into the left-hand jack. This is convenient when the user wishes to switch between headphones and speakers. This feature can be defeated by adding jumpers at JP1 and JP2.

JP1 and JP2 should be configured as follows:

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

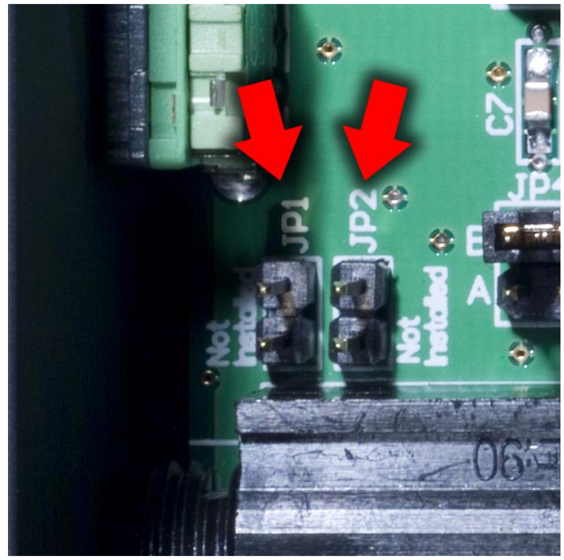


Photo 2 - Headphone Switch Disable (JP1 and JP2)

Headphone Gain Reduction (JP3 and JP4):

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

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

JP3 and JP4 are factory installed at position "B" to reduce the headphone output by 10 dB. This setting is best for most applications. Remove the jumpers if you need more gain, or move them to position "A" if you need less gain.

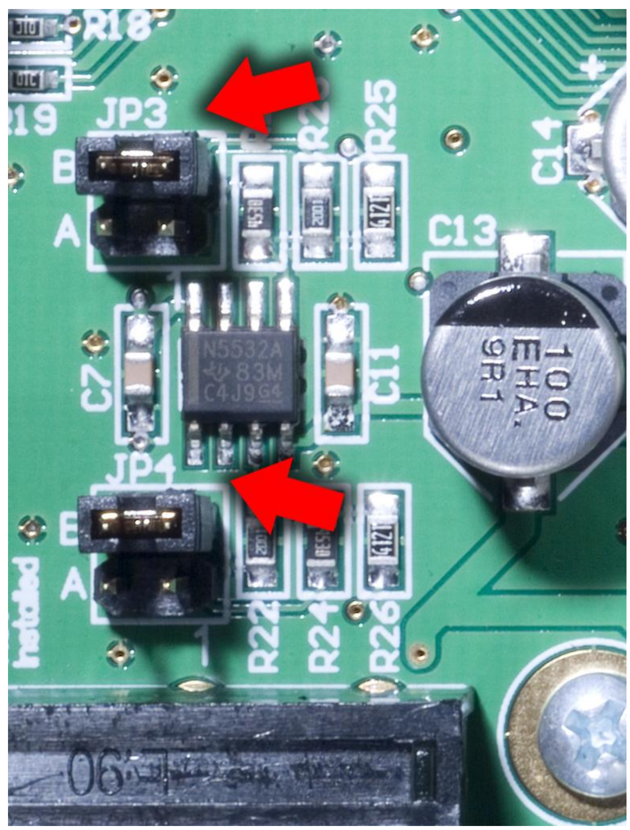


Photo 3 – Headphone Gain Reduction (JP3 and JP4)

Rack Mounting

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

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

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

System1™ Universal Rack Adapter

The ***Universal Rack Mount Adapter*** is a tray that mounts up to two ***System1™*** products in a single race space. The tray accepts any combination of ***System1™*** products (with or without rack-mount type faceplates).

Blank Rack Panel

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

Benchmark Technologies

HPA2™ Headphone Amplifier

The ***DAC1 PRE*** headphone output is driven by Benchmark's signature ***HPA2***™ headphone amplifier. This high-current, high-output amplifier has an output impedance of near 0-Ohms. It is designed to drive loads as low as 30 Ohms without any increase in distortion. It also has sufficient amplitude to drive low-sensitivity 600-Ohm headphones.

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

0-Ohm Output Impedance

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

Headphone Performance

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

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

UltraLock™ Clock System

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

Jitter is present on every digital audio interface. This type of jitter is known as '**interface jitter**' and it is present even in the most carefully designed audio systems. Interface jitter accumulates as digital signals travel down a cable and from one digital device to the next. If we measure interface jitter in a typical system we will find that it is 10 to 10,000 times higher than the maximum allowable level for accurate 24-bit conversion. Fortunately, interface jitter has absolutely no effect on the audio <u>unless</u> it influences the conversion clock in an analog-to-digital converter (A/D) or in a digital-to-analog converter (D/A).

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

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

UltraLock™ converters exceed the jitter performance of two-stage PLL converters, and are free from the slow-lock and no-lock problems that can plague two-stage PLL designs. ***UltraLock™*** converters have extremely high immunity to interface jitter under all operating conditions. No jitter-induced artifacts can be detected using an Audio Precision System 2 Cascade test set. Measurement limits include detection of artifacts as low as –140 dBFS, application of jitter amplitudes as high as 12.75 UI, and application of jitter over a frequency range of 2 Hz to 200 kHz. Any AES/EBU signal that can be decoded by the AES/EBU receiver will be reproduced without the addition of any measurable jitter artifacts.

The ***DAC1 PRE***, ***DAC1***, ***DAC-104, ADC1*** and the ***ADC-104*** employ Benchmark's ***UltraLock™*** technology to eliminate jitter-induced performance problems. ***UltraLock™*** technology isolates the conversion clock from the digital audio interface clock. Jitter on a D/A digital audio input, or an A/D reference input can never have <u>any</u> measurable effect on the conversion clock of an ***UltraLock™*** converter. In an ***UltraLock™*** converter, the conversion clock is never phase-locked to a reference clock. Instead the converter oversampling-ratio is varied with extremely high precision to achieve the proper phase relationship to the reference clock. The clock isolation of the ***UltraLock™*** system insures that interface jitter can never degrade the quality of the audio conversion. Specified performance is consistent and repeatable in any installation with cables of any quality level!

How does conversion clock jitter degrade converter performance?

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

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

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

Jitter induced sidebands can be measured using an FFT analyzer.

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

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

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

Put *UltraLock*™ converters to the test:

We encourage our customers to perform the above tests on ***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 damage. Jitter-induced sidebands are extremely complex and cannot be removed with any existing audio device. Therefore, it is very important to attack jitter at both ends of the audio chain. The ***DAC1 PRE 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 ***DAC1 PRE*** eliminates one major variable: jitter.

AdvancedUSB Audio™ Technology

Benchmark's ***AdvancedUSB Audio™*** technology provides a simple, yet comprehensive, high resolution audio solution for computer audio users. With bit-transparent audio streaming at 96 kHz, 24-bit, the Benchmark USB solution is a dream-come-true for lovers of high quality audio playback. Plus, with no drivers to install, you can enjoy your music as soon as you plug into a computer's USB port. Benchmark's USB technology is compatible with virtually all audio applications and has been extensively tested on four operating systems (Microsoft Vista, XP, 2000, and Mac OSX).

Setting New USB Audio 'Benchmarks'

Benchmark Media Systems has the distinction of presenting the first native, 96-kHz, 24-bit USB audio solution. By intelligently using the capabilities built into the Windows and Mac operating systems, this technology enables bit-transparent audio streams at resolutions up to 96 kHz, 24-bit, when all other native solutions are limited to 44.1-48-kHz, 16-bit. Thus, the fidelity that was originally captured in the recording can be fully appreciated. There is no need to configure and re-configure software to ensure proper bit-rate settings. With this advanced technology, high-resolution audio is automatically passed from the source program to the USB without data modification.

Bit-Transparent Digital Audio Path

A digital audio path can be tested to determine if the digital data is being modified or distorted in any way. This is done by sending a random sequence of bits through the path, and comparing the resulting sequence with the original sequence. If the resulting sequence is always identical to the original, the path is 'bit- transparent'. Benchmark's USB technology is the first native USB solution capable of streaming 96 kHz, 24-bit audio with full 'bit-transparency'.

Beware of 'Custom' Drivers!

Until now, high-resolution USB audio devices required ‘custom’ drivers. These drivers may compromise the stability of the operating system, and may cause conflicts with other installed devices. In addition, custom drivers usually consume more system resources (memory and CPU) than native solutions.

It is also interesting that many of the ASIO high-resolution USB devices we tested failed to deliver bit-transparent audio. In contrast, many of the native USB audio devices delivered bit-transparent audio. Our tests show that custom drivers do not guarantee bit-transparent data transfers, and that bit-transparency can be achieved without custom drivers.

The problem with native USB audio devices has been their inability to stream audio at sample-rates over 48 kHz and at word-lengths over 16-bits. Benchmark's ***AdvancedUSB Audio™*** technology extends bit-transparent native USB audio to resolutions up to and including 96 kHz, 24-bits.

Intelligent Handling of Sample-Rates and Bit-Depths

Benchmark's USB technology will follow the sample-rate and bit-depth of the audio being sent to it without requiring the user to reconfigure any software or hardware. In contrast, devices with custom drivers require the user to make manual changes to the driver setting in order to correctly stream at the sample-rate and bit-depth of the audio they are playing. Incorrect settings usually result in severe distortion. Benchmark's ***AdvancedUSB Audio™*** technology eliminates this problem. This technology allows bit- transparent playback of play lists containing a mixture of sample-rates and word-lengths.

Meticulous Engineering Eliminates Pops and Clicks

A common problem with streaming audio via USB is the presence of pops and clicks. Audio requires constant un-interrupted data flow. Any gaps in the audio data will cause clicks and pops if buffers are not working properly. The Benchmark ***AdvancedUSB Audio™*** solution was engineered to establish and maintain a properly buffered un-interrupted flow of high resolution audio data.

Plug it in and Start Listening… Immediately

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

One USB Audio Solution for All Your Computer Audio Needs

Most devices with custom drivers only connect to one application at a time. This is especially true with devices using ASIO drivers with Windows operating systems. The device will 'lock' to a specific audio application, leaving all other applications unable to access the device. Benchmark's USB technology allows as many applications to access the device as needed. This convenience allows the user to switch between a music player to a video player or web-streaming player without needing to reconfigure any software or hardware.

Advantages of 24-bit Playback of 16-bit Sources

Why do I need a 24-bit USB audio device to play 16-bit 44.1 kHz music files?

The reason is that digital volume controls and digital mixers increase the word-length of the audio. The longer word-length is a result of multiplication and addition. These arithmetic operations produce long word-lengths that must be squeezed back into a shorter word length. Word-length reduction adds noise and/or distortion to the audio. The amount that is added is determined by the output word length.

The noise and/or distortion added by word-length reduction decreases by 6 dB for every additional bit that can be retained. Reduction to 16-bits adds 48 dB more noise than reduction to 24-bits. In general, 16-bit word-length reduction is very audible; while 24-bit word-length reduction produces noise levels that are well below audibility.

Our tests show that 24-bit output devices deliver a dramatic improvement in sound quality when playing 16-bit material. Native USB output devices have had a reputation for poor sound quality. This is primarily due to the 16-bit word-length reduction.

Benchmark's ***AdvancedUSB Audio™*** technology breaks the 16-bit barrier and delivers pristine digital audio to the D/A converter. Benchmark's ***UltraLock™*** system insures that the D/A converters deliver this audio to the analog outputs without adding jitter-induced distortion artifacts.

Recommended Settings for Computer Playback

Benchmark has done extensive testing on various operating systems and media players. These tests determine the optimal settings for high-quality playback.

Although the results of the tests are highly definitive, they are quickly out-dated due to the nature of software updates. For the latest information available, please visit our complimentary information resource center online:

http://www.BenchmarkMedia.com/wiki

Microsoft Windows® Test Results

Windows® 2000 and XP operating systems have a digital mixer known as 'Kmixer'. By default, all audio streams go through the Windows® Kmixer to reach native USB audio devices. The performance of Kmixer is critical to any native USB audio solution, so it was tested extensively by the engineers at Benchmark.

The results indicate that Kmixer can perform with full transparency under the correct conditions. However, under the wrong conditions, Kmixer can do a great deal of damage. Kmixer's sample-rate-conversion is of very poor quality (under XP and 2000) and must be avoided. Benchmark's ***AdvancedUSB Audio™*** solution allows Kmixer to default to a transparent mode of operation that avoids sample rate conversion.

In contrast, sample-rate-conversion is outstanding in Vista. By default, Vista up-samples to the highest sample rate supported by the connected audio device. This up-sampling is so well designed that it should not be capable of generating audible artifacts.

In any Windows® operating system, true bit-transparency is only achieved when the all volume controls are set to maximum. Nevertheless, we discovered that the Windows® volume controls are very well executed and are distortion-free when streaming to a 24-bit output device.

Whenever audio is originated from a single application, Benchmark's ***AdvancedUSB Audio™*** solution prevents Kmixer's sample-rate-conversion so that bit-transparency can be maintained. It also forces Kmixer into a 24-bit output mode so that the Windows® volume control does not degrade the audio quality if it is used.

Benchmark's ***AdvancedUSB Audio™*** solution offers users the convenience of simultaneous high-quality playback from more than one Windows® application. Kmixer's sample-rate-conversion is disabled as long as all applications are playing files at identical sample rates. If the sample rates do not match, sample-rate–conversion is only applied to the lower sample rates, and the high sample rate signals remain at high-quality.

Mac OS X Test Results

The system sample rate must be set to appropriately to optimize playback quality. This setting is different depending on software version and media player. The user is strongly encouraged to check the latest information on this by going to:

http://www.BenchmarkMedia.com/wiki

Like the Windows® XP and 2000 operating systems, most versions of OS X have very poor-quality sample-rate-conversion. The system sample rate is manually set and must be set to match the sample rate of the audio being played.

OS X is capable of bit-transparent audio playback when the system sample rate is set to match the audio, and all volume controls are set to 100%.

TIP – To access the most current information regarding settings for high-quality computer playback, visit our computer audio application pages: www.benchmarkmedia.com/wiki

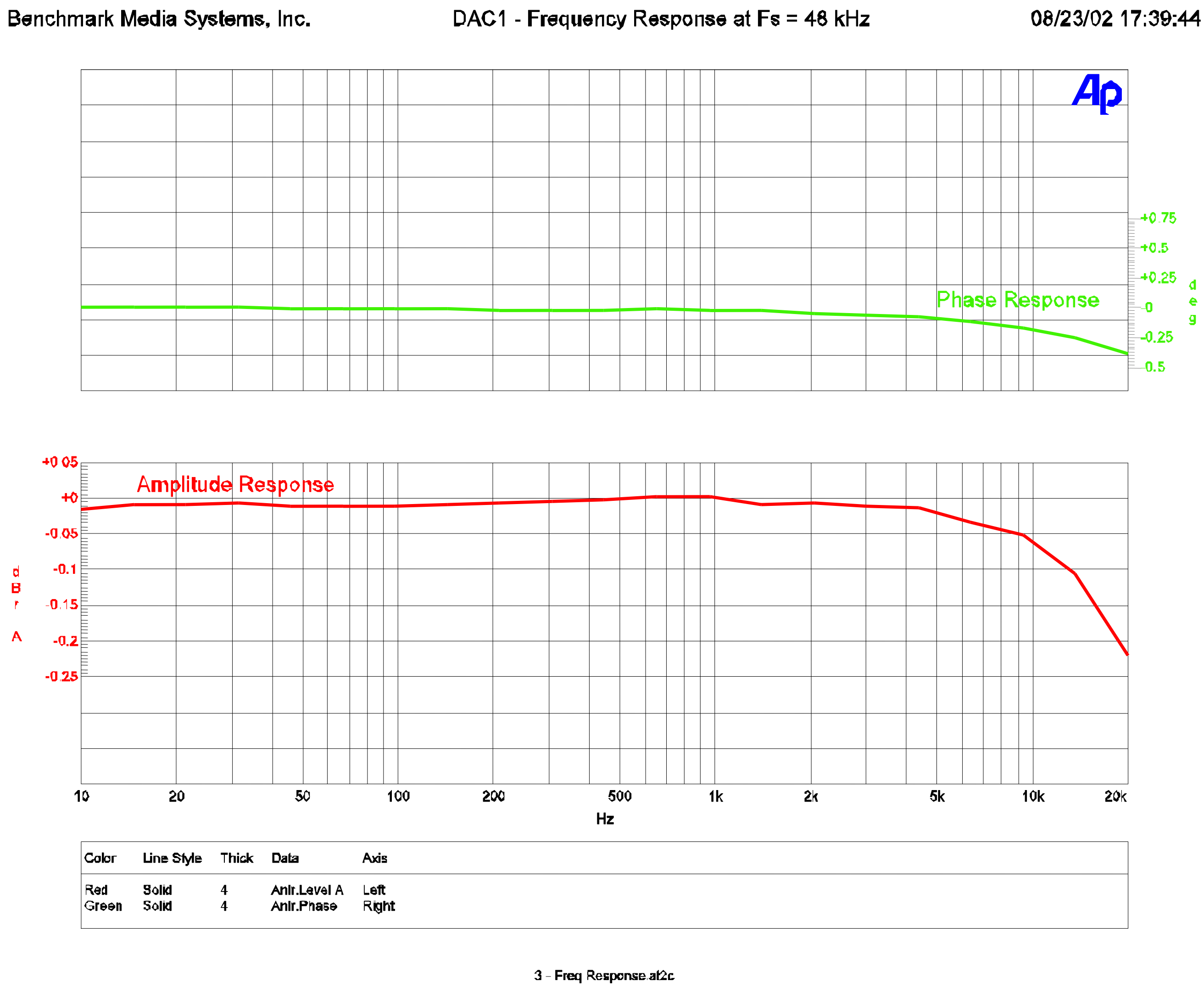
This site provides the latest information on media players, media servers, operating systems, and audio-related computer accessories. It also includes general instructions for maximizing the performance of media servers.

Performance Graphs

The following graphs apply to both ***DAC1*** and ***DAC1 PRE*** converters:

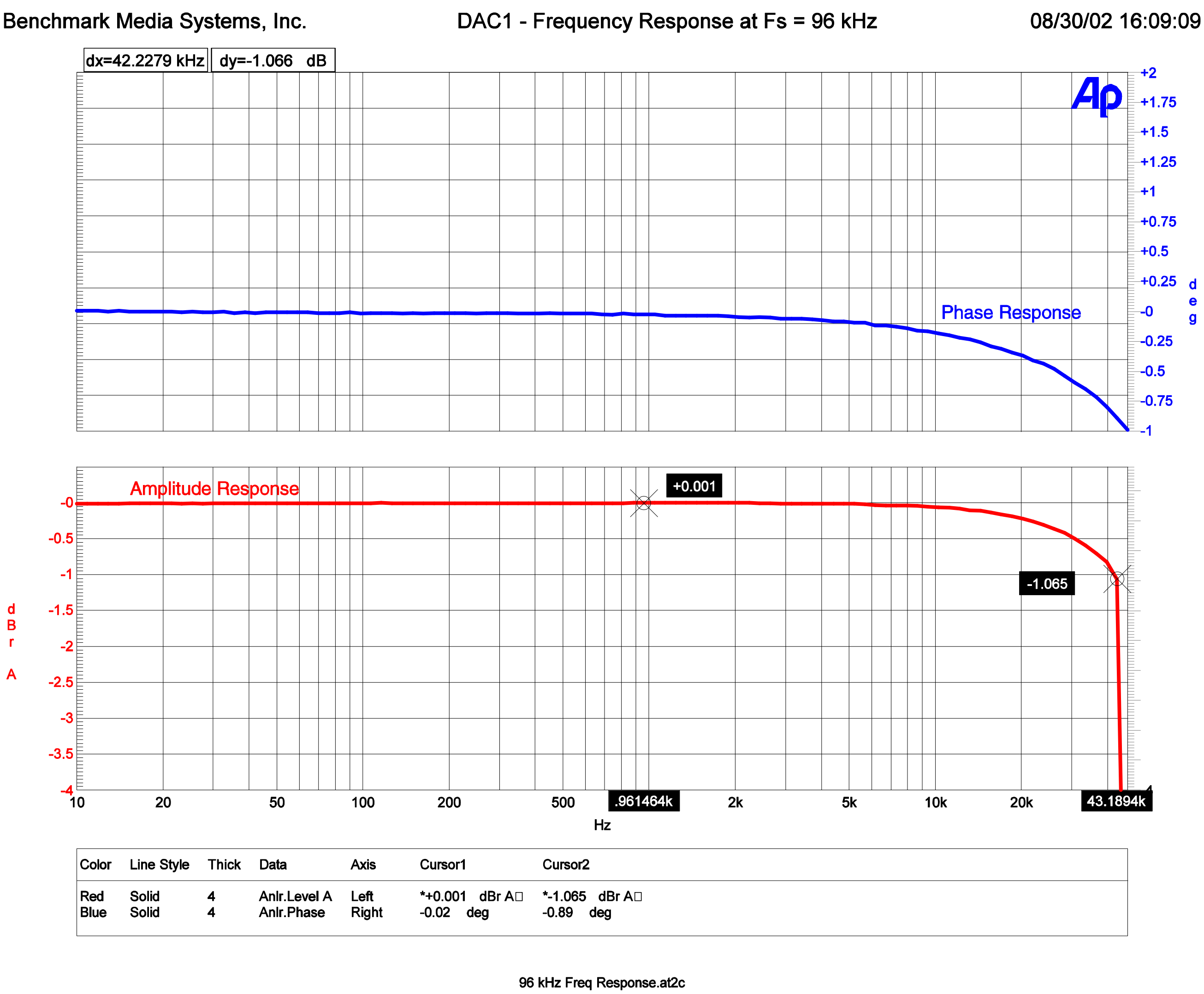
Frequency Response Tests

Frequency Response at Fs = 48 kHz



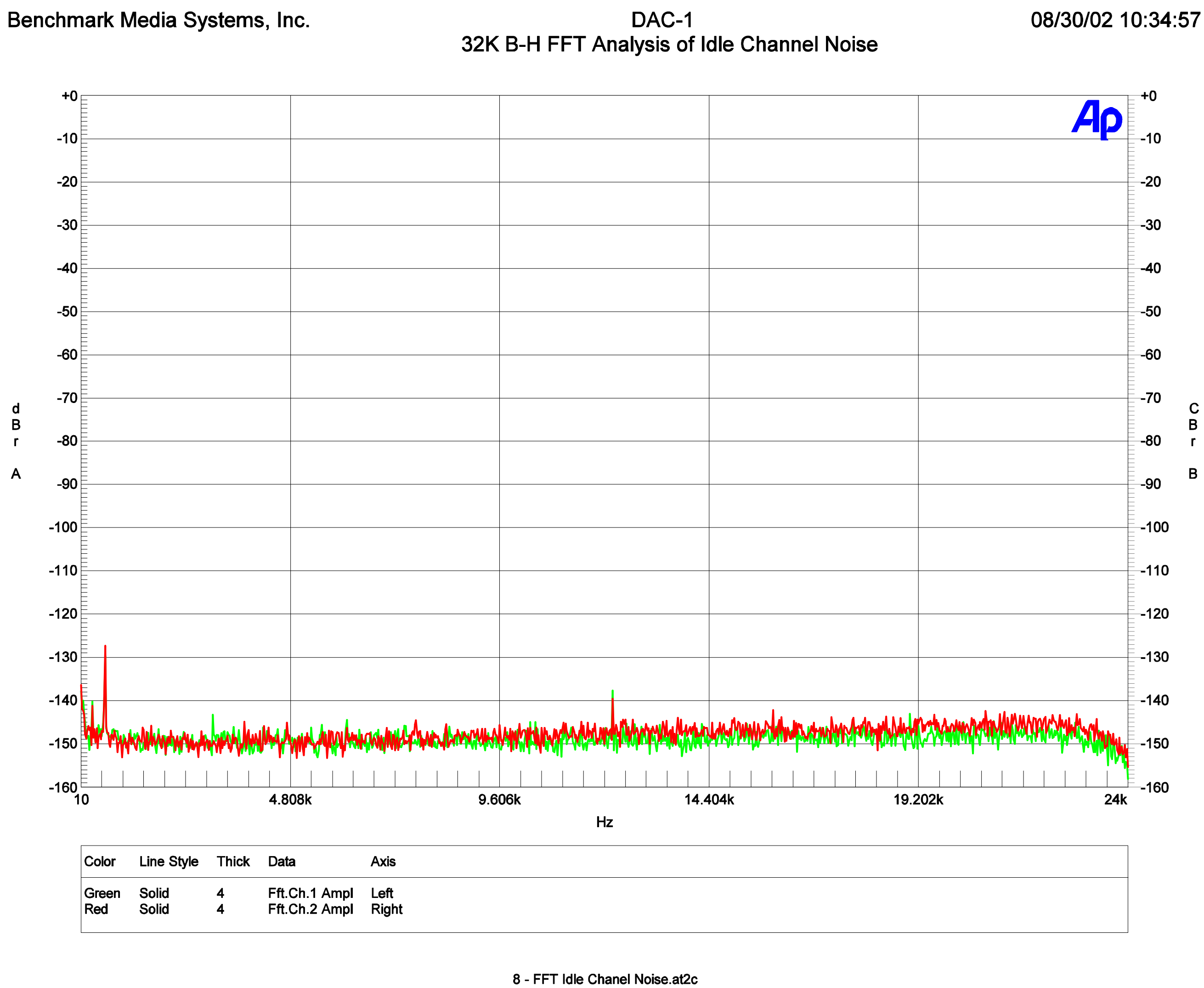
The above graphs show the frequency response of the ***DAC1*** when it is operating at a 48-kHz sample rate. The top graph shows that the differential phase is better than ± 0.5º at 20 kHz. The bottom graph shows the amplitude response on a highly expanded 0.05 dB/division scale. The amplitude response is down by only 0.22 dB at 20 kHz. The bass response extends well below the 10-Hz limitation of the measurement equipment.

Frequency Response at Fs = 96 kHz



The above graphs show the frequency response of the ***DAC1*** when it is operating at a 96-kHz sample rate. The top graph shows that the differential phase is better than ± 0.5º at 20 kHz and better than ± 1º at 43 kHz. The bottom graph shows the amplitude response on a highly expanded 0.05 dB/division scale. The amplitude response is down by only 0.22 dB at 20 kHz and only −1 dB at 43 kHz. The bass response extends well below the 10-Hz limitation of the measurement equipment.

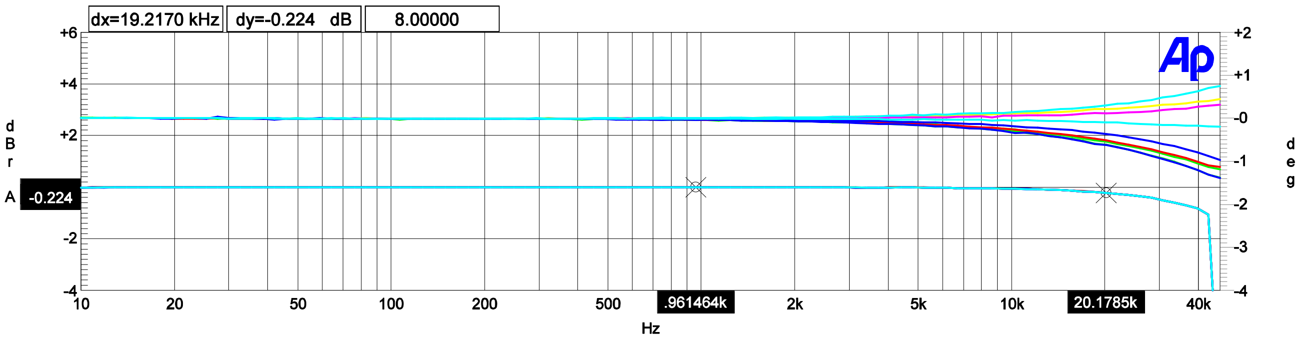
FFT Analysis of Idle Channel Noise



The above graph demonstrates that the ***DAC1*** is free from idle tones and clock crosstalk. The highest spurious tone measures –128 dBFS and is AC line related hum. The highest non-line related tone measures –138 dBFS.

Multi-Unit Phase Response

Benchmark Media Systems, Inc. DAC1 - Phase and Frequency Response Matching at Fs = 96 kHz 08/30/02 16:48:42
10 Channels of Conversion using 5 DAC-1 Converters Selected at Random



Color	Line Style	Thick	Data	Axis	Source 2	Cursor1	Cursor2
Red	Solid	4	Anlr.Level A	Left	: 2.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.223 dBr A
Blue	Solid	4	Anlr.Phase	Right	: 2.00000 =Swr.Ch. B Input	-0.02 deg	-0.36 deg
Magenta	Solid	4	Anlr.Level A	Left	: 3.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.224 dBr A
Cyan	Solid	4	Anlr.Phase	Right	: 3.00000 =Swr.Ch. B Input	-0.01 deg	-0.09 deg
Blue	Solid	4	Anlr.Level A	Left	: 4.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.226 dBr A
Green	Solid	4	Anlr.Phase	Right	: 4.00000 =Swr.Ch. B Input	-0.03 deg	-0.54 deg
Cyan	Solid	4	Anlr.Level A	Left	: 5.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.224 dBr A
Yellow	Solid	4	Anlr.Phase	Right	: 5.00000 =Swr.Ch. B Input	-0.01 deg	+0.21 deg
Green	Solid	4	Anlr.Level A	Left	: 6.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.224 dBr A
Red	Solid	4	Anlr.Phase	Right	: 6.00000 =Swr.Ch. B Input	-0.03 deg	-0.51 deg
Yellow	Solid	4	Anlr.Level A	Left	: 7.00000 =Swr.Ch. B Input	-0.001 dBr A	-0.224 dBr A
Magenta	Solid	4	Anlr.Phase	Right	: 7.00000 =Swr.Ch. B Input	-0.01 deg	+0.12 deg
Red	Solid	4	Anlr.Level A	Left	: 8.00000 =Swr.Ch. B Input	*-0.001 dBr A	*-0.224 dBr A
Blue	Solid	4	Anlr.Phase	Right	: 8.00000 =Swr.Ch. B Input	-0.04 deg	-0.62 deg
Magenta	Solid	4	Anlr.Level A	Left	: 9.00000 =Swr.Ch. B Input	+0.001 dBr A	-0.224 dBr A
Cyan	Solid	4	Anlr.Phase	Right	: 9.00000 =Swr.Ch. B Input	+0.01 deg	+0.30 deg
Blue	Solid	4	Anlr.Level A	Left	: 10.0000 =Swr.Ch. B Input	-0.001 dBr A	-0.226 dBr A
Green	Solid	4	Anlr.Phase	Right	: 10.0000 =Swr.Ch. B Input	-0.04 deg	-0.62 deg

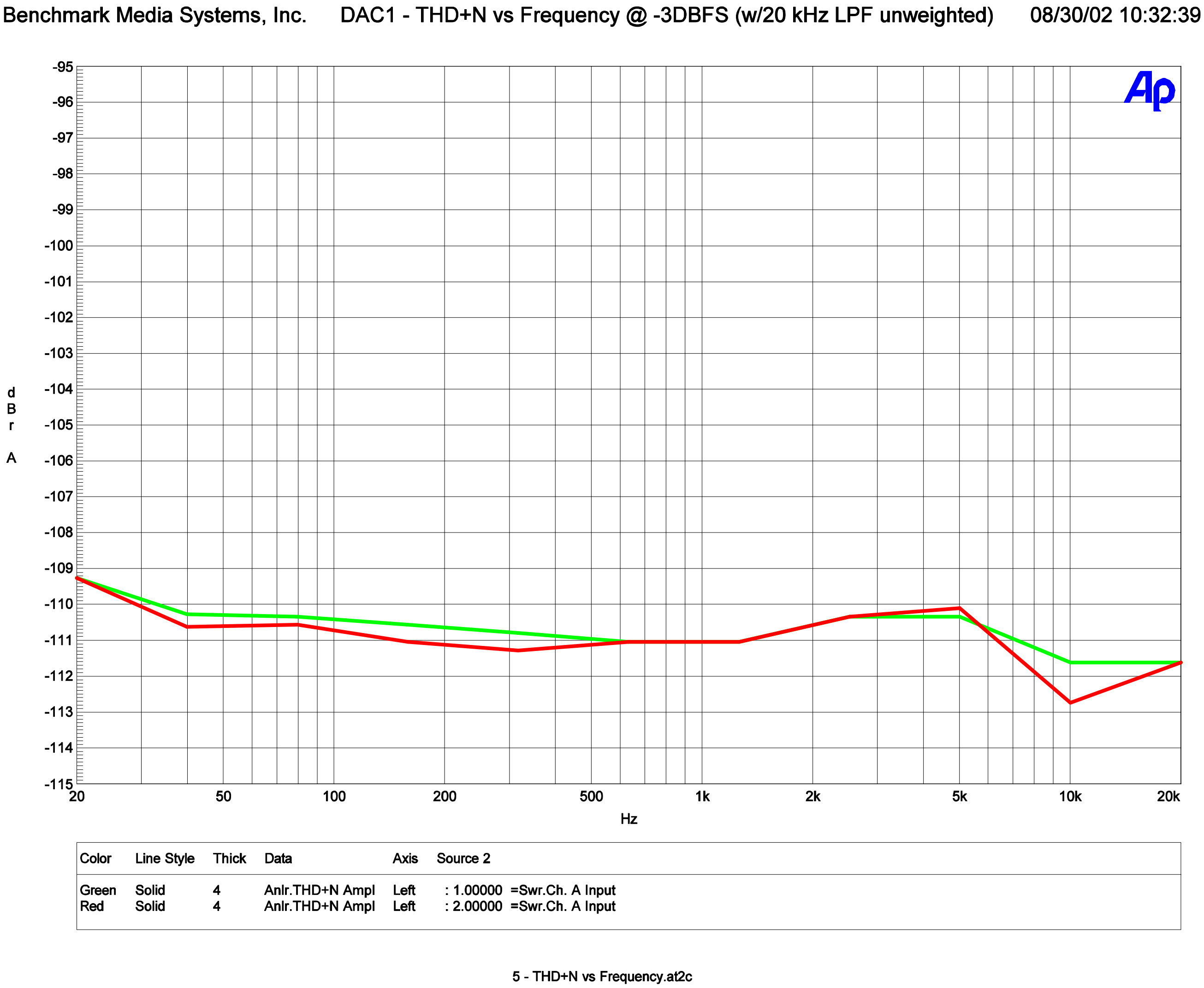
Channel 1 = Left Channel of Unit #1 - Refernce Channel for all Differential Phase Measurements
Channel 2 = Right Channel of Unit #1
Channel 3 = Left Channel of Unit #2
Channel 4 = Right Channel of Unit #2
Channel 5 = Left Channel of Unit #3
Channel 6 = Right Channel of Unit #3
Channel 7 = Left Channel of Unit #4
Channel 8 = Right Channel of Unit #4
Channel 9 = Left Channel of Unit #5
Channel 10 = Right Channel of Unit #5

Phase and Amplitude Matching Between DAC-1 Units.at2c

Any combination of ***DAC1***, ***DAC1 USB***, and ***DAC1 PRE*** converters may be used to create a multi-channel playback system that maintains phase accuracy across all channels at sample rates up to 110 kHz. The above graph shows the differential phase between 10 audio channels using 5 ***DAC1*** converters operating at 96 kHz. The ***DAC1*** converters were chosen from stock at random, and measurements were made using a random combination of Coaxial, XLR, and Optical inputs. The type of digital interface used has no measurable effect on the phase. Please note that no reference or synchronization cables are required to create a phase accurate multi-channel playback system using ***DAC1*** converters at sample rates up to 110 kHz.

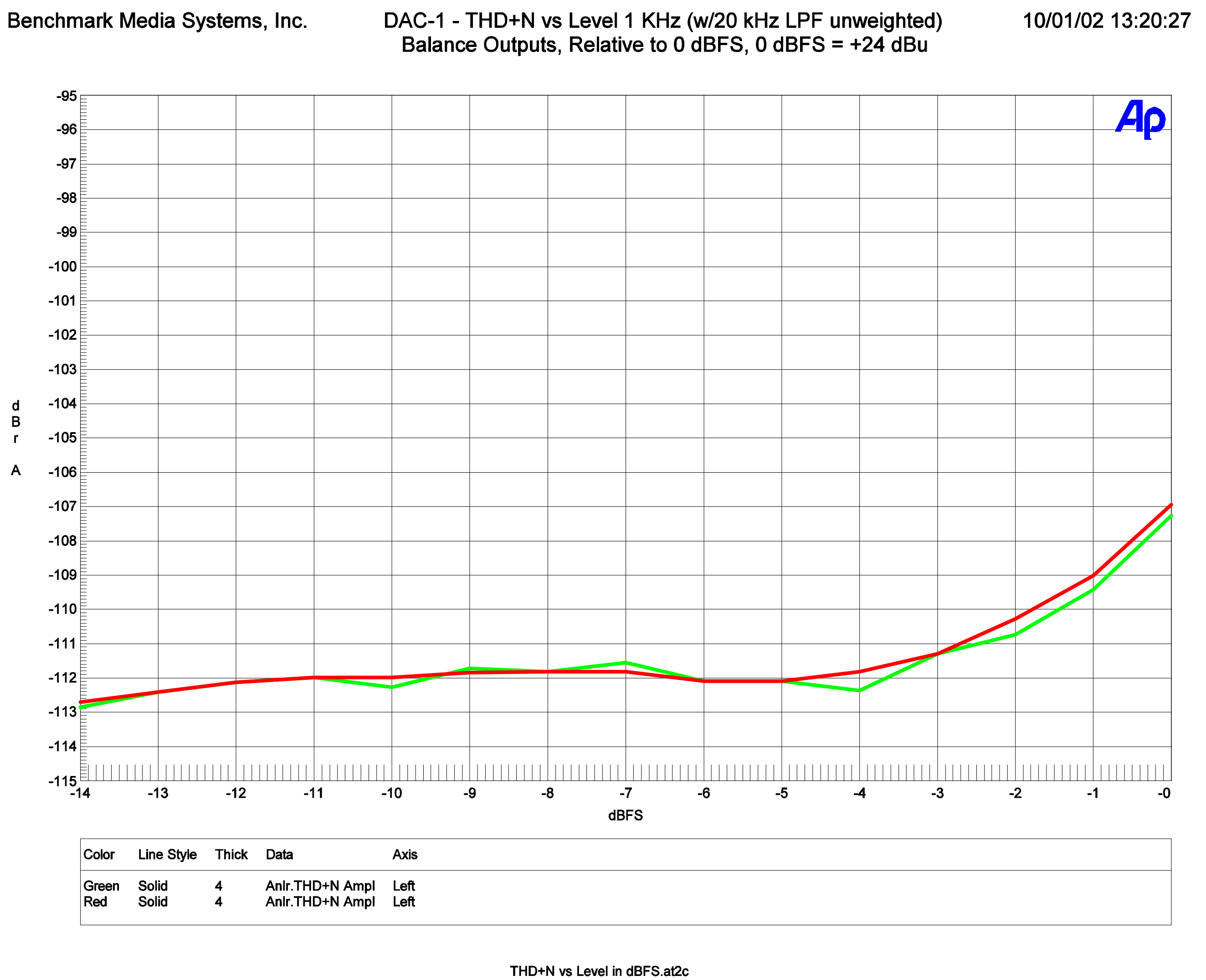
THD+N Tests

THD+N vs. Frequency at –3 dBFS



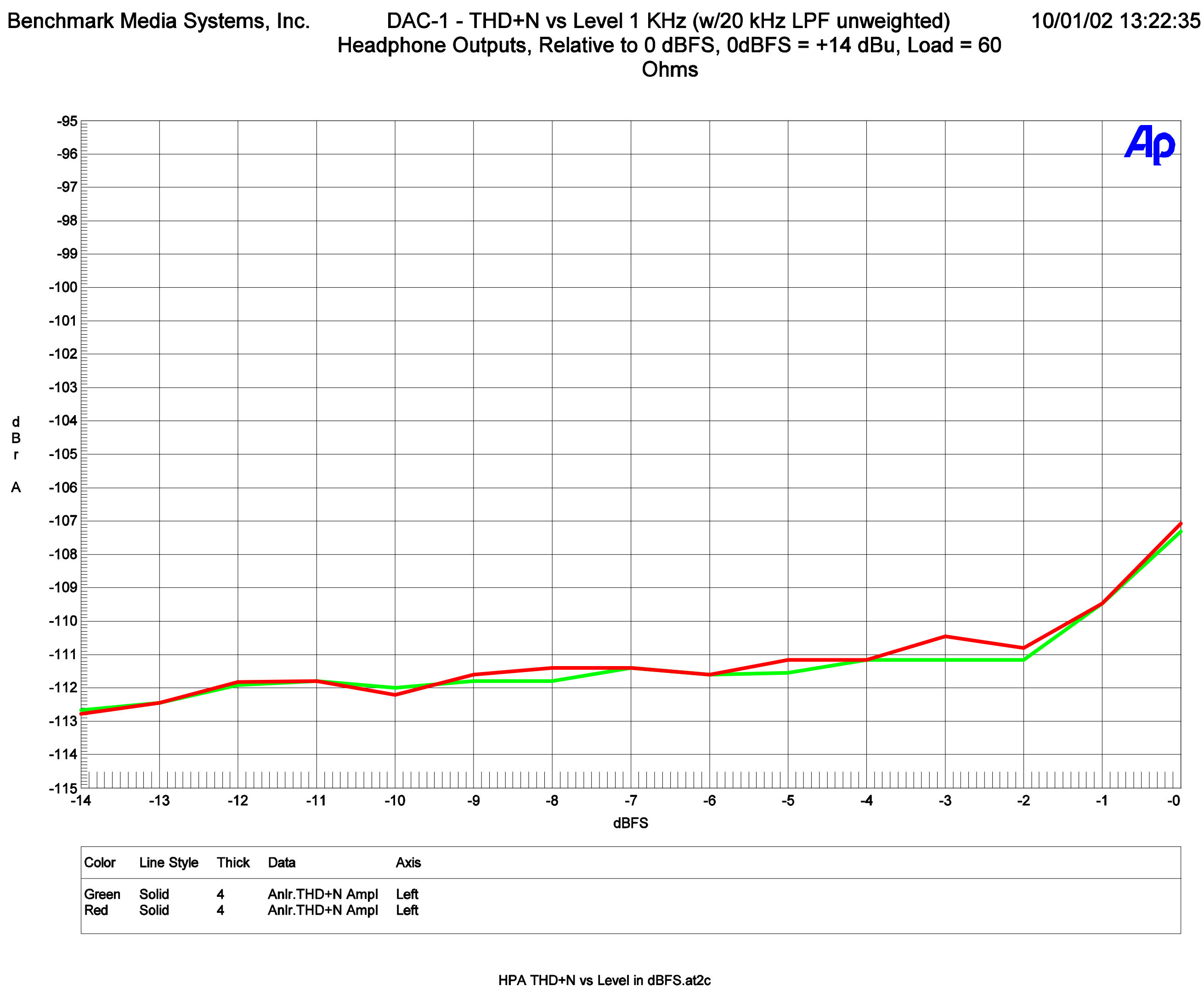
The above graphs demonstrate that the THD+N specifications for the ***DAC1*** are not frequency dependent (the variation from 20 Hz to 20 kHz is very slight). Note that at worst case, the distortion is 109 dB less than the – 3 dBFS test tone (and 112 dB less than the full scale output of the ***DAC1***). This implies that the distortion created by the ***DAC1*** should be below the threshold of hearing unless playback levels exceed 112 dB peak SPL. Distortion should still be well masked at higher playback levels.

THD+N vs. Level at 1 kHz – Balanced Outputs



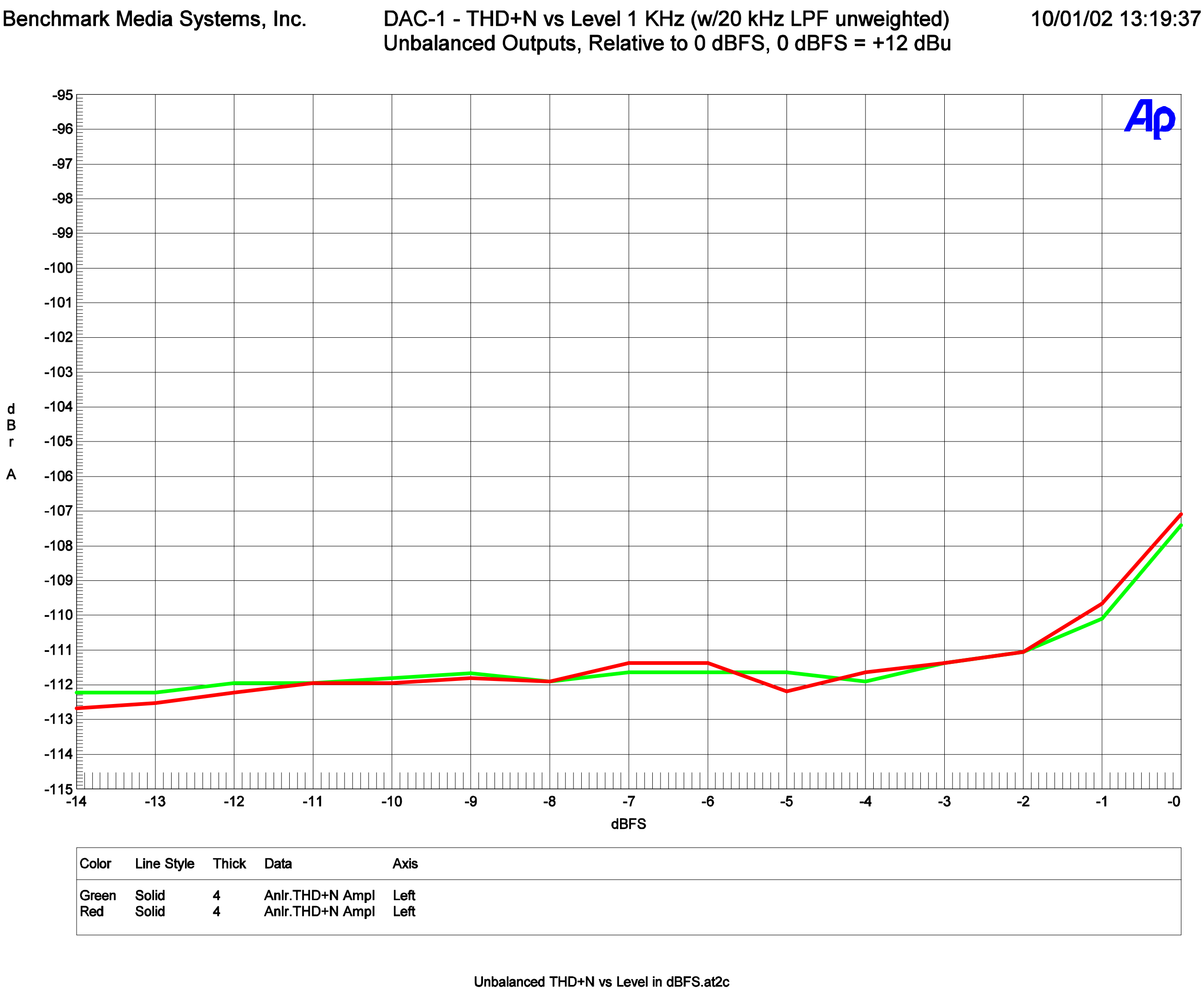
Below –4 dBFS, distortion is lower than the noise floor of the converter. Above –3 dBFS, distortion reaches a maximum value of only –107 dBFS.

THD+N vs. Level at 1 kHz – Headphone Outputs



This graph shows the output of the ***HPA2***™ headphone amp driving a 60-Ohm load at a very high level (+14 dBu). Even under these conditions, the ***HPA2***™ delivers the full rated performance of the ***DAC1***. Compare this to the performance of the balanced outputs (see previous graph).

THD+N vs. Level at 1 kHz - Unbalanced Outputs



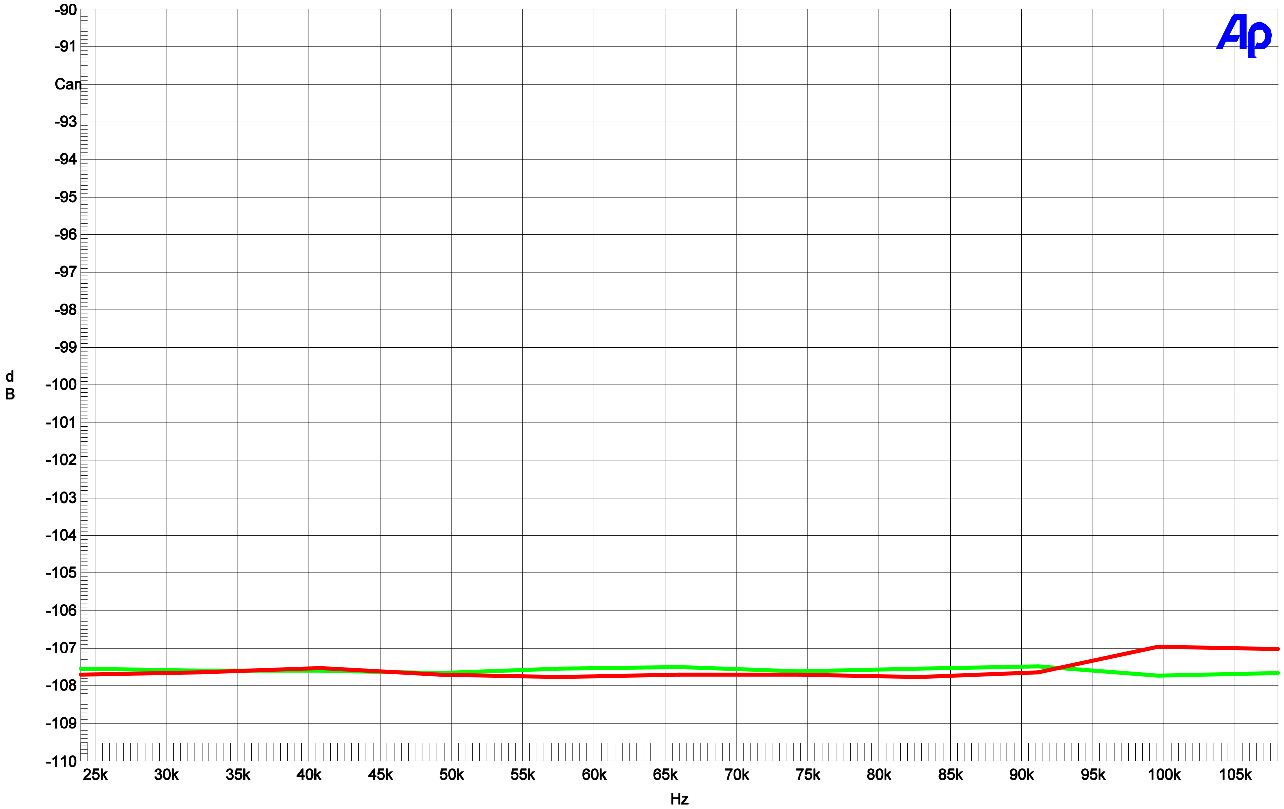
This graph demonstrates the performance of the unbalanced outputs. Note that the performance is nearly identical to that of the balanced outputs.

THD+N vs. Sample Frequency

Benchmark Media Systems, Inc.

DAC-1 THD+N vs Sample Frequency
1 kHz at -3 dBFS, 20-20kHz BW

08/30/02 10:36:09



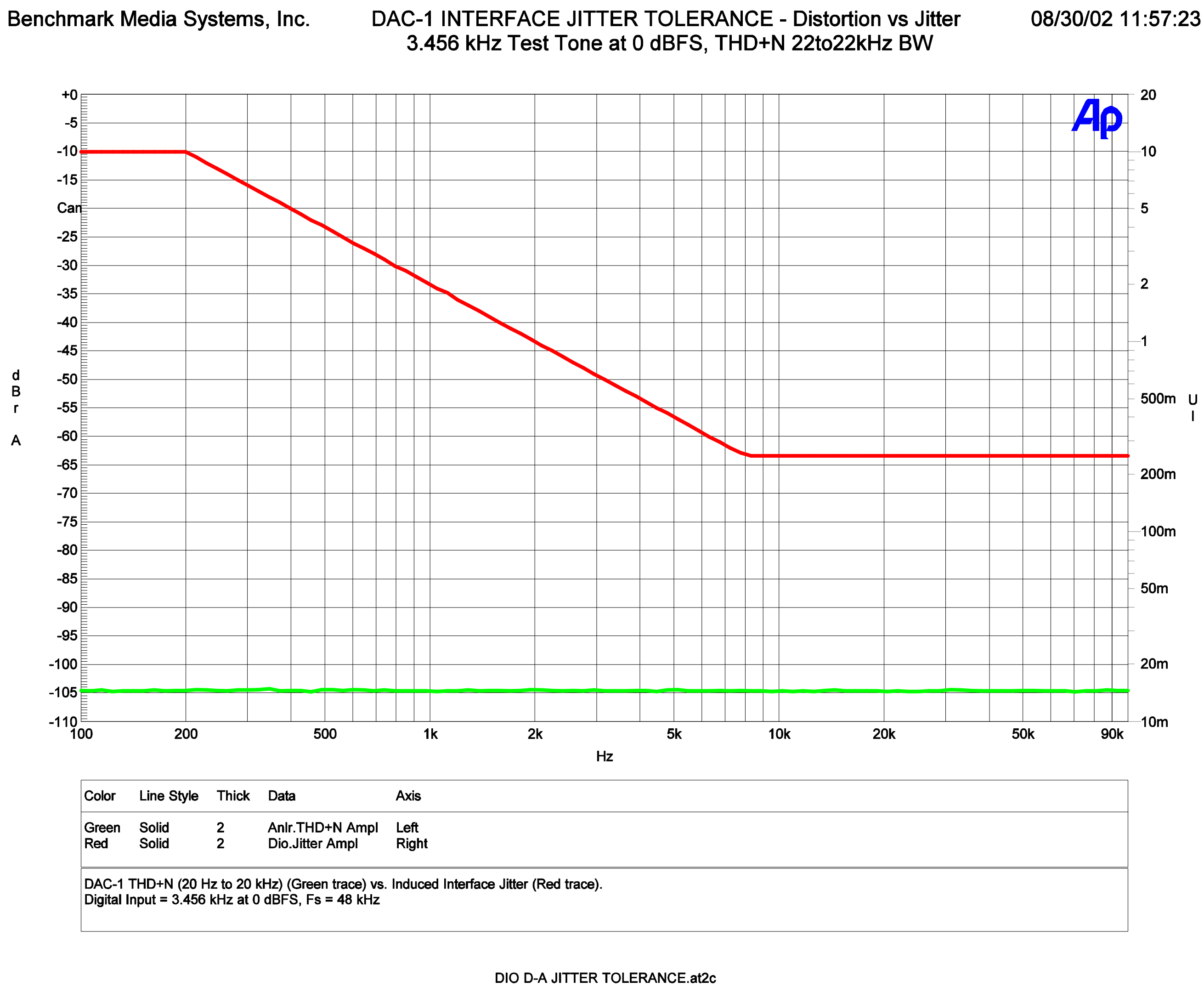
Color	Line Style	Thick	Data	Axis	Source 2
Green	Solid	2	Anlr.THD+N Ratio	Left	: 1.00000 =Swr.Ch. A Input
Red	Solid	2	Anlr.THD+N Ratio	Left	: 2.00000 =Swr.Ch. A Input

9 - Clock Freq Sweep.at2c

The above graph shows that the ***DAC1*** provides consistent performance at all sample rates. Distortion is not a function of sample rate. The minor variations in the above plots are due to measurement limitations.

Jitter Tests

AES Jitter Tolerance Test



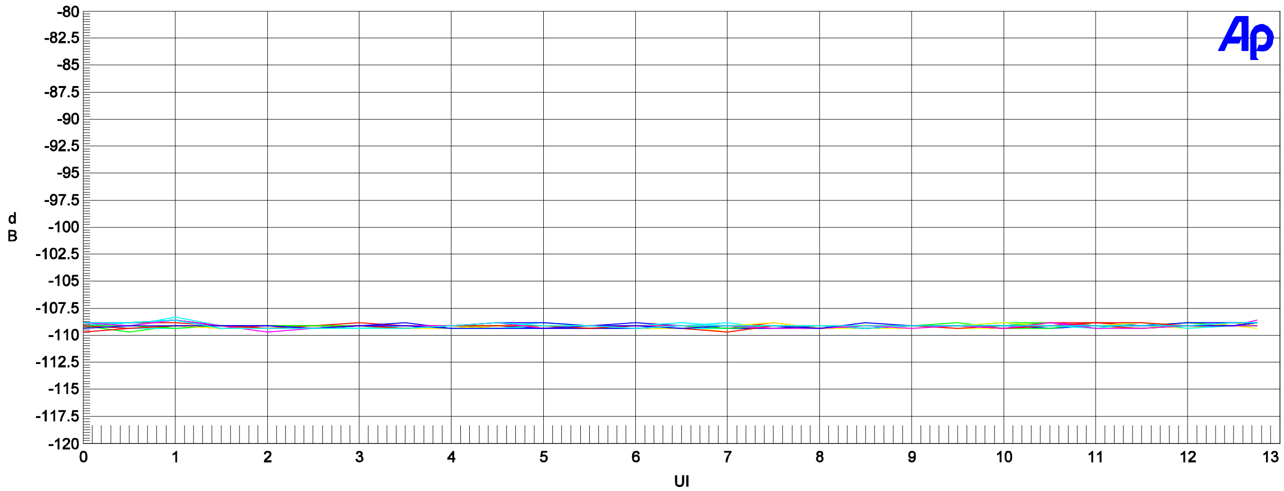
The graph above shows the results of a standard AES jitter tolerance test.

The top (red) curve shows the amplitude of the jitter applied to the inputs of the ***DAC1***. The scale for the top curve is on the right hand side of the graph and is calibrated in UI of jitter.

The bottom (green) curve shows the THD+N of the ***DAC1*** as the jitter amplitude and frequency is varied at the inputs of the ***DAC1***. There is absolutely no change in the ***DAC1*** THD+N measurement over the entire range of jitter test frequencies and amplitudes. In fact, the ***DAC1*** can tolerate much higher levels of jitter without any measurable change in performance (see the next graph).

THD+N vs. Jitter Amplitude and Jitter Frequency

Benchmark Media Systems, Inc. DAC-1 THD+N vs Jitter Amplitude (0 to 12.75 UI) and Jitter Frequency (2 Hz to 9 kHz) 08/30/02 11:10:38
Input = 10 kHz at -3 dBFS, BW = 20 kHz Unweighted



Color	Line Style	Thick	Data	Axis	Source 2
Cyan	Solid	2	Anlr.THD+N Ratio	Left	: 2.00000 Hz=Dio.Jitter Freq
Green	Solid	2	Anlr.THD+N Ratio	Left	: 500.000 Hz=Dio.Jitter Freq
Yellow	Solid	2	Anlr.THD+N Ratio	Left	: 1.00000 kHz=Dio.Jitter Freq
Red	Solid	2	Anlr.THD+N Ratio	Left	: 1.50000 kHz=Dio.Jitter Freq
Magenta	Solid	2	Anlr.THD+N Ratio	Left	: 2.00000 kHz=Dio.Jitter Freq
Blue	Solid	2	Anlr.THD+N Ratio	Left	: 2.50000 kHz=Dio.Jitter Freq
Cyan	Solid	2	Anlr.THD+N Ratio	Left	: 3.00000 kHz=Dio.Jitter Freq
Green	Solid	2	Anlr.THD+N Ratio	Left	: 3.50000 kHz=Dio.Jitter Freq
Yellow	Solid	2	Anlr.THD+N Ratio	Left	: 4.00000 kHz=Dio.Jitter Freq
Red	Solid	2	Anlr.THD+N Ratio	Left	: 4.50000 kHz=Dio.Jitter Freq
Magenta	Solid	2	Anlr.THD+N Ratio	Left	: 5.00000 kHz=Dio.Jitter Freq
Blue	Solid	2	Anlr.THD+N Ratio	Left	: 5.50000 kHz=Dio.Jitter Freq
Cyan	Solid	2	Anlr.THD+N Ratio	Left	: 6.00000 kHz=Dio.Jitter Freq
Green	Solid	2	Anlr.THD+N Ratio	Left	: 6.50000 kHz=Dio.Jitter Freq
Yellow	Solid	2	Anlr.THD+N Ratio	Left	: 7.00000 kHz=Dio.Jitter Freq
Red	Solid	2	Anlr.THD+N Ratio	Left	: 7.50000 kHz=Dio.Jitter Freq
Magenta	Solid	2	Anlr.THD+N Ratio	Left	: 8.00000 kHz=Dio.Jitter Freq
Blue	Solid	2	Anlr.THD+N Ratio	Left	: 8.50000 kHz=Dio.Jitter Freq
Cyan	Solid	2	Anlr.THD+N Ratio	Left	: 9.00000 kHz=Dio.Jitter Freq

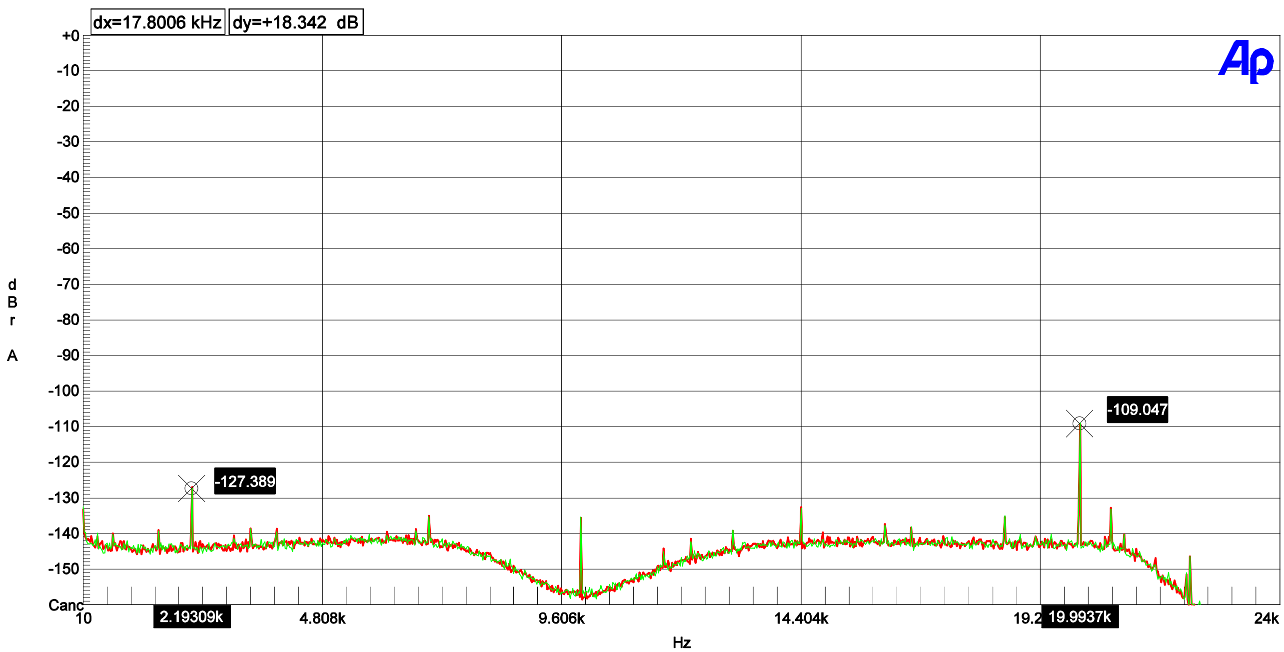
THD+N vs Jitter.at2c

The above graph shows the results from the most severe jitter test that we could create with an Audio Precision System 2 Cascade test set. We selected a 10-kHz audio test tone in order to maximize the sensitivity of the test. We set the interface jitter amplitude to its maximum value of 12.75 UI (2075 ns) of jitter. We then swept the jitter frequency from 2 Hz to 9 kHz and plotted the THD+N from the ***DAC1***. Absolutely no change in THD+N was observed at any test frequency, and the ***DAC1*** performance did not change when the jitter was turned off. The same test was conducted using FFT analysis to look for jitter-induced artifacts. No change was observed on a FFT analysis (see the next graph).

Immunity to Cable-Induced Jitter

Benchmark Media Systems, Inc. DAC-1 - FFT Demonstration of Immunity to Jitter Caused by Long Cables 08/30/02 14:26:29

Performance is Unchanged after 1000 Feet of Cable



Color	Line Style	Thick	Data	Axis	Cursor1	Cursor2
Green	Solid	4	Fft.Ch.2 Ampl	Left	*-127.389 dBr A□	*-109.047 dBr A□
Red	Solid	4	Fft.Ch.2 Ampl	Left	-126.974 dBr A	-109.301 dBr A

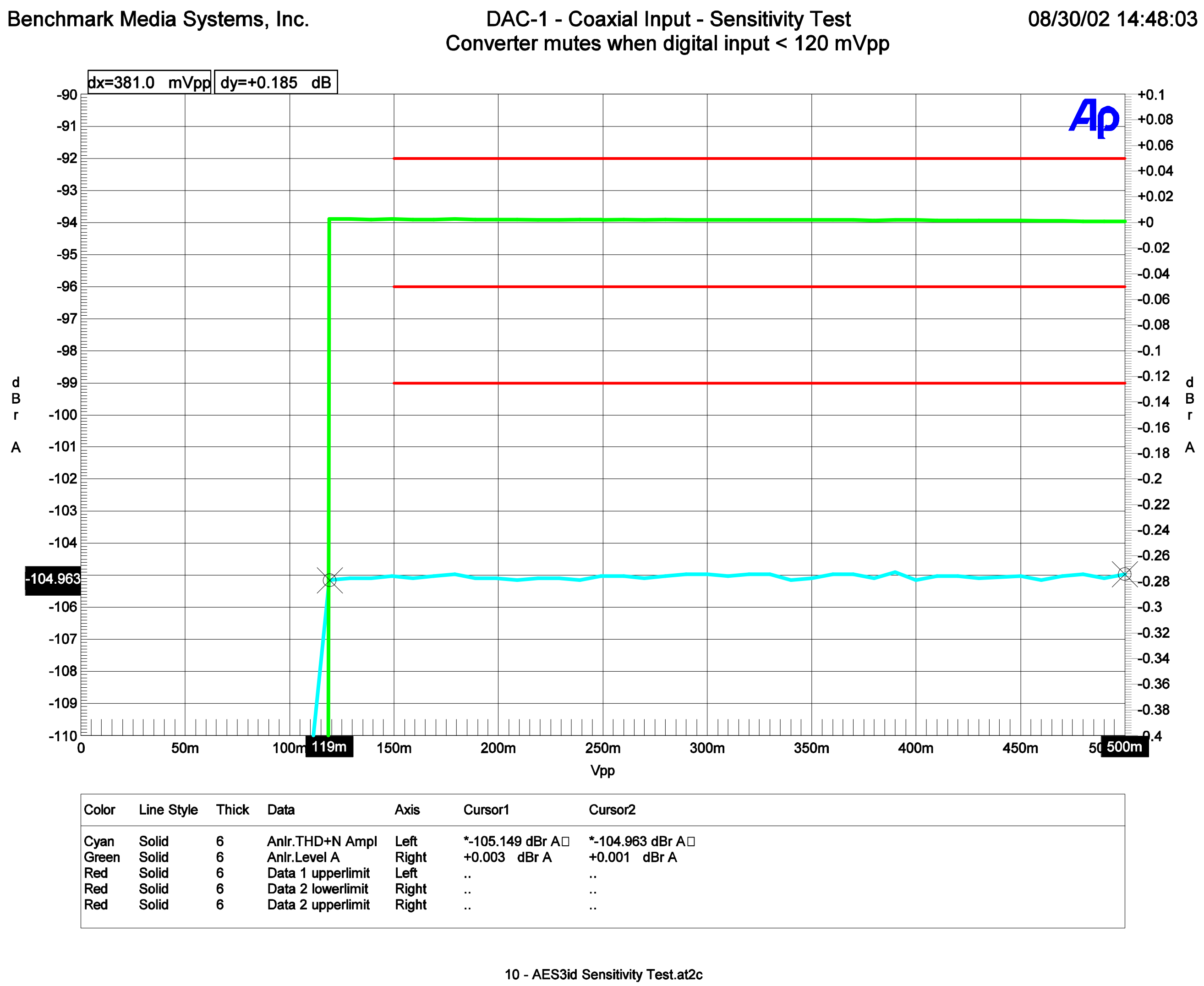
FFT demonstration of jitter immunity.
Digital input is 110 Ohm balanced and is fed directly to the DAC1 (trace 1 - Green) and then through 1000 feet of Category 5e unshielded twisted pair cable (trace 2 - Red).
In both cases, the test signal is a digitally generated TPDF dithered 24-bit 10 kHz test tone at a level of 0 dBFS.
Jitter at the XLR input to the DAC1 measures > 4 ns RMS and > 8 ns peak to peak when the 1000 foot length of cable is added..
Plot shows that spurious tones are at extreemly low levels and are unchanged when compaired to the performance with a jitter free input signal.
Note that there is no evidence of any increase in jitter-induced sidebands when the cable length is increased to 1000 feet.
Tone at 20 kHz is second harmonic distortion and measures -109 dB relative to 0 dBFS.
Highest level spurrious tone measures -128 dBFS with or without 1000 feet of cable.

FFT 10kHz with and without 1000 feet of Cat 5.at2c

The above FFT plots demonstrate that the performance of the ***DAC1*** is not degraded in any way when long cables are used to transmit digital audio to the ***DAC1***.

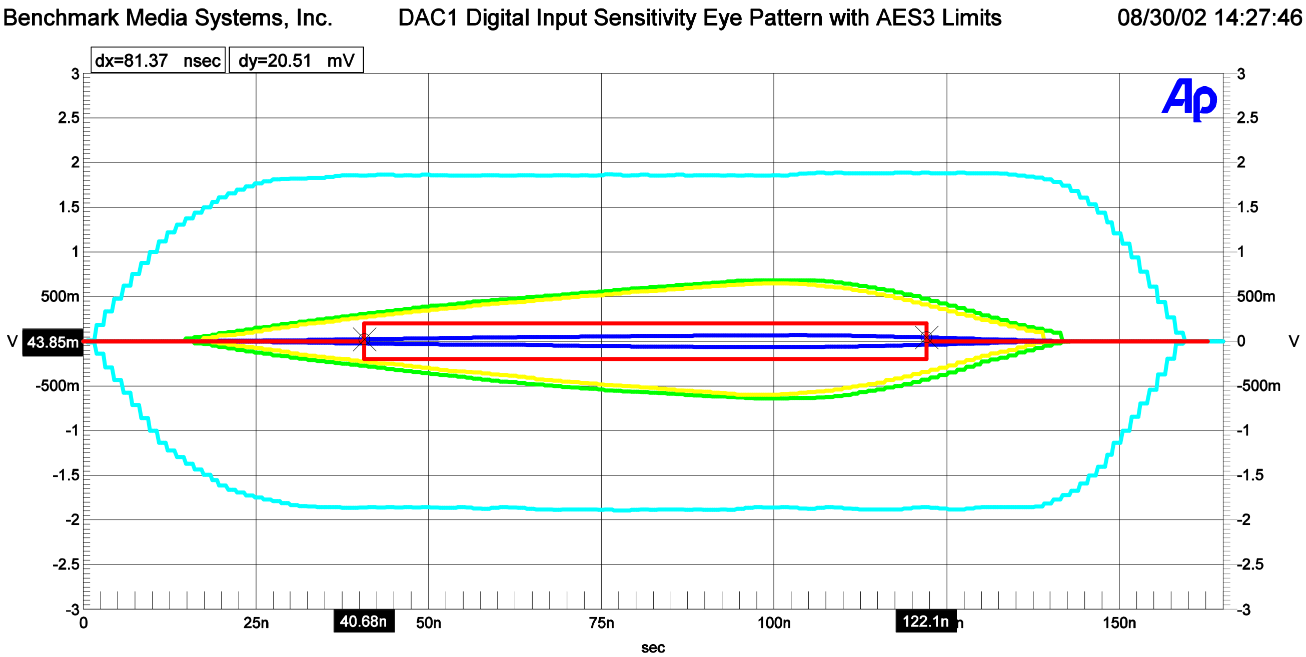
Input Sensitivity Tests

Coaxial Digital Input Sensitivity



The above graph shows that the performance of the ***DAC1*** is not a function of the signal level at the coaxial digital input. When the signal is too low to decode (< 120 mVpp), the converter mutes gracefully.

Minimum Eye Pattern



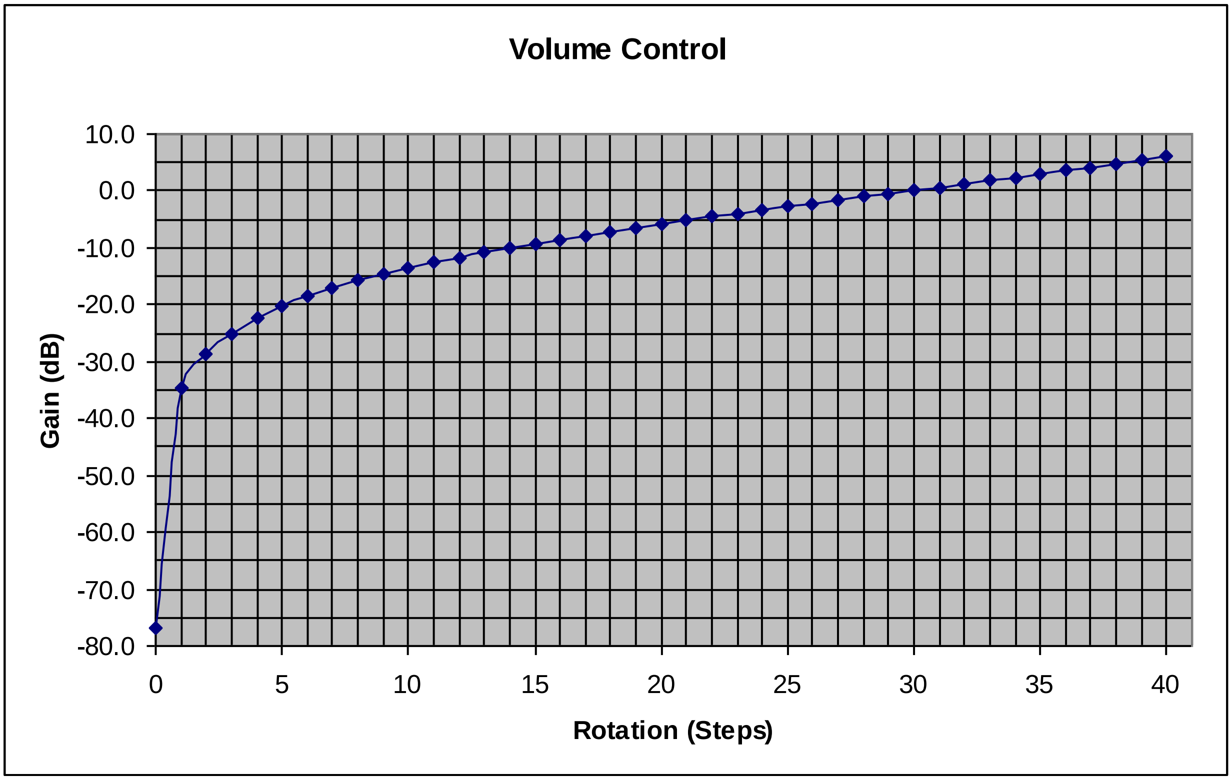
Color	Line Style	Thick	Data	Axis	Cursor1	Cursor2
Green	Solid	3	Intervu.Upper Eye Opening	Left	304.9 mV	479.8 mV
Green	Solid	3	Intervu.Lower Eye Opening	Right	-273.0 mV	-429.1 mV
Yellow	Solid	3	Intervu.Upper Eye Opening	Left	268.0 mV	413.5 mV
Yellow	Solid	3	Intervu.Lower Eye Opening	Right	-225.4 mV	-342.8 mV
Blue	Solid	3	Intervu.Upper Eye Opening	Left	*23.33 mV	*43.85 mV
Blue	Solid	3	Intervu.Lower Eye Opening	Right	-24.13 mV	-40.12 mV
Cyan	Solid	3	Intervu.Upper Eye Opening	Left	1.855 V	1.888 V
Cyan	Solid	3	Intervu.Lower Eye Opening	Right	-1.861 V	-1.857 V
Red	Solid	3	Data 1 upperlimit	Left	..	..
Red	Solid	3	Data 2 lowerlimit	Right	..	..

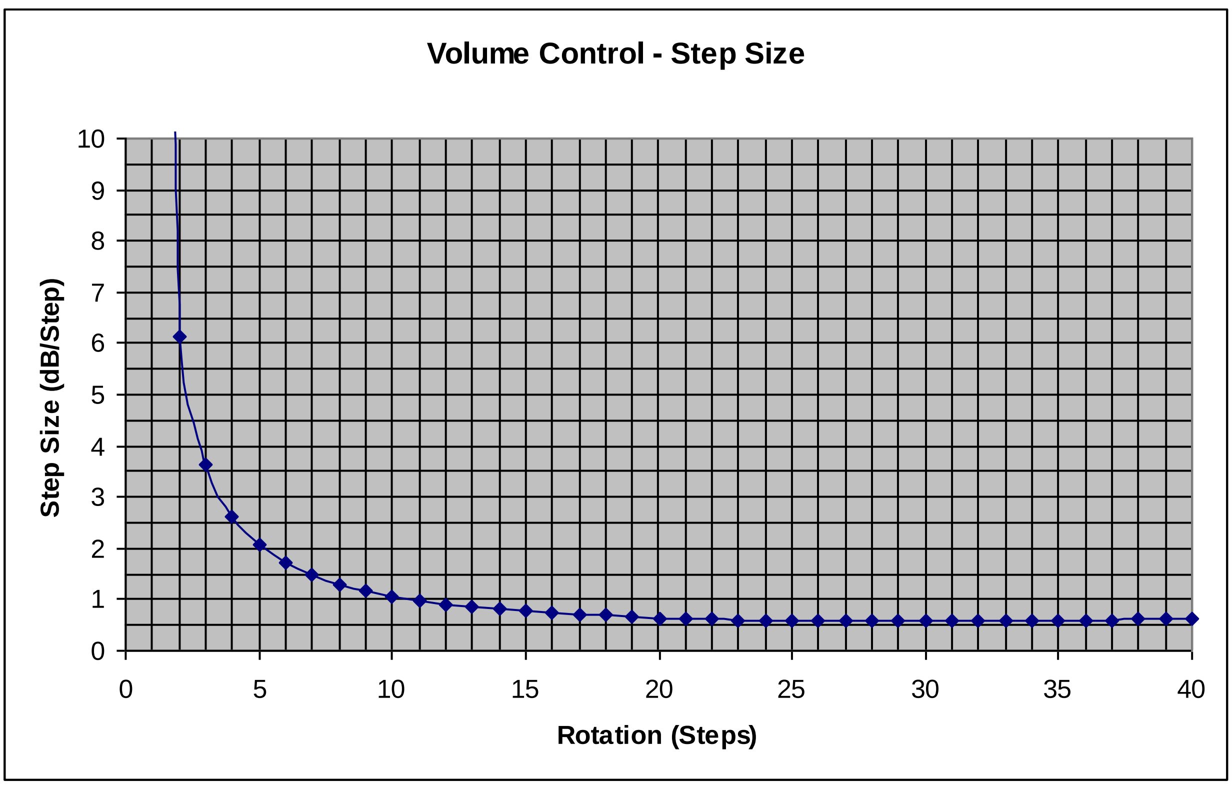
Cyan = Eye pattern from typical AES/EBU 110 Ohm balanced output.
Green = Eye pattern after passing a balanced 110 Ohm AES/EBU signal through 1000 feet of Belden Mediatwist (Category 5e cable).
Yellow = Eye pattern after passing a balanced 110 Ohm AES/EBU signal through 1000 feet of General Cable Category 5e cable.
Blue = Minimum Eye pattern for DAC1 operation.
Red = Minimum Eye pattern for an AES receceiver per AES specifications for 48 kHz.

1000 ft Cat 5e EYE-PATTERN.at2c

The above graph demonstrates that the ***DAC1*** can operate with an eye pattern considerably smaller than specified by the AES. In addition, the above plots show that while the AES minimum eye pattern specifications are barely met at the end of 1000 feet of Category 5 UTP cable, the ***DAC1*** receivers have enough sensitivity to allow reliable operation. The jitter produced by this connection is removed entirely by the Benchmark ***UltraLock™*** clock circuits and the ***DAC1*** operates at full-specified performance.

Volume Control Curve





Specifications

Audio Performance	
Fs = 44.1 to 96 kHz, 20 to 20 kHz BW, 1 kHz test tone, 0 dBFS = +24 dBu (unless noted)	
SNR – A-Weighted, 0 dBFS = +20 to +29 dBu	116 dB
SNR – Unweighted, 0 dBFS = +20 to +29 dBu	114 dB
SNR – A-Weighted at low gain, 0 dBFS = +9 to +18 dBu	114 dB
THD+N, 1 kHz at 0 dBFS	-105 dBFS, -105 dB, 0.00056%
THD+N, 1 kHz at -1 dBFS	-107 dBFS, -106 dB, 0.00050%
THD+N, 1 kHz at –3 dBFS	-110 dBFS, -107 dB, 0.00045%
THD+N, 20 to 20 kHz test tone at –3 dBFS	-110 dBFS, -107 dB, 0.00045%
Frequency Response at Fs=96 kHz	+/- 0.1 dB (20 to 20 kHz) -0.02 dB at 10 Hz -0.20 dB at 20 kHz -0.85 dB at 40 kHz -2.5 dB at 45 kHz
Frequency Response at Fs=48 kHz	+/- 0.1 dB (20 to 20 kHz) -0.02 dB at 10 Hz -0.20 dB at 20 kHz
Crosstalk	-100 dB at 20 kHz -125 dB at 1 kHz -130 dB at 20 Hz
Maximum Amplitude of Jitter Induced Sidebands (10 kHz 0 dBFS test tone, 12.75 UI sinusoidal jitter at 1 kHz)	< -141 dB
Maximum Amplitude of Spurious Tones with 0 dBFS test signal	< -126 dB
Maximum Amplitude of Idle Tones	< -128 dB
Maximum Amplitude of AC line related Hum & Noise	< -126 dB
Inter-channel Differential Phase (Stereo Pair – any sample rate)	+/- 0.5 degrees at 20 kHz
Inter-channel Differential Phase (Between ***DAC1 PRE*** Units Fs<110 kHz)	+/- 0.5 degrees at 20 kHz
Inter-channel Differential Phase (Between ***DAC1 PRE*** Units Fs>110 kHz)	+/- 4.1 degrees at 20 kHz

Audio Performance (continued)

Maximum Lock Time after Fs change	100 ms
Soft Mute Ramp Up/Down Time	10 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, and 96 kHz

Group Delay (Latency)

Delay – Digital Input to Analog Output (function of sample rate) The delay can be calculated using the following formula: **Delay = 1.01 ms + (48/Fs)** Where Fs = the sample rate in Hz.	2.72 ms at 28 kHz 2.51 ms at 32 kHz 2.10 ms at 44.1 kHz 2.01 ms at 48 kHz 1.55 ms at 88.2 kHz 1.51 ms at 96 kHz 1.28 ms at 176.4 kHz 1.26 ms at 192 kHz

Analog Audio Inputs	
Number of Analog Inputs (switch selected)	1 (RCA stereo pair - unbalanced)
Number of Channels	2
Input Impedance	20 k Ohms
Maximum Input Level	+15 dBu
Maximum Input @ Factory-set Calibration Levels	+13 dBu
DC Blocking Capacitors on Analog Inputs	Yes
Transient and Over-Voltage Protection on Analog Inputs	Series R and diode protection
Input Capacitance	10 pF
Analog-Input Gain Range	Off to +3.5 dB (RCA in to RCA out) Off to +19 dB (RCA in to XLR out) Off to +13 dB (RCA in to Headphone)
Factory-Set Analog-Input Gain In 'Calibrated' Mode	+0.5 dB (RCA in to RCA out) -4.0 dB (RCA in to XLR out)

Digital Audio Inputs	
Number of Digital Inputs (switch selected)	5 (1 USB, 1 Optical, 3 Coaxial)
Number of Channels	2
Input Sample Frequency Range	28 to 195 kHz (Coaxial) 28 to 96 kHz (Optical) 44.1, 48, 88.2, 96 kHz (USB)
Maximum Input Word Length	24 bits
Digital Input Impedance on XLR input	110 Ohms
Digital Input Impedance on Coaxial input (jumper selected)	75 Ohms or Hi-Z (Bridging)
Transformer Coupled Digital Inputs	Yes (Coaxial)
DC Blocking Capacitors on Digital Inputs	Yes (Coaxial)
Transient and Over-Voltage Protection on Digital Inputs	Yes
Minimum Digital Input Level	150 mVpp on Coaxial

Jitter Tolerance (With no Measurable Change in Performance):	>12.75 UI sine, 100 Hz to 10 kHz >3.5 UI sine at 20 kHz >1.2 UI sine at 40 kHz >0.4 UI sine at 80 kHz >0.29 UI sine at 90 kHz >0.25 UI sine above 160 kHz
Jitter Attenuation Method	Benchmark ***UltraLock***™ - 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) 43 Ohms (Attenuator = 30 dB)
Analog Output Clip Point	+29 dBu
Output Level Calibration Controls	10-turn trimmers (1 per output)
Calibration Adjustability	2 dB / turn
Output Level Range (at 0 dBFS) In 'Calibrated' Mode	+9 to +29 dBu (Attenuator off) -1 to +19 dBu (Attenuator = 10 dB) -11 to +9 dBu (Attenuator = 20 dB) -21 to -1 dBu (Attenuator = 30 dB)
Factory Set 'Calibrated' Output Level (at 0 dBFS)	+4 dBu (Attenuator = 20 dB)
Output Level Range (at 0 dBFS) In 'Variable' Mode	Off to +27 dBu (Attenuator off) Off to +17 dBu (Attenuator = 10 dB) Off to +7 dBu (Attenuator = 20 dB) Off to -3 dBu (Attenuator = 30 dB)
Output Level Variation with Sample Rate (44.1 kHz vs. 96 kHz)	< +/- 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
Output Level Calibration Controls	Shared with Balanced Outputs
Output Level Range (at 0 dBFS) In 'Calibrated' Mode	-6 dBu to +13.5 dBu
Factory Set 'Calibrated' Output Level (at 0 dBFS)	+8.5 dBu (2 Vrms)
Output Level Range (at 0 dBFS) In 'Variable' Mode	Off to +11 dBu
Calibration Adjustability	2 dB / turn
Output Level Variation with Sample Rate (44.1 kHz vs. 96 kHz)	< +/- 0.006 dB

HPA2™ **Headphone Outputs**	
Number of Headphone Outputs	2
Output Connectors	¼” TRS with switch on left-hand jack
Output Impedance	< 0.11 Ohms
Output Level Control	Stereo Control on Front Panel
Output Level Range (at 0 dBFS) into 60-Ohm Load	Off to +21 dBu
Maximum Output Current	250 mA
Overload Protection (independent per channel)	Current limited at 300 mA, Thermal
Bandwidth	> 500 kHz
THD+N	-106 dB, 0.0005% into 30 Ohms at +18 dBu (1.26W)

Status Display	
Indicators - Type and Location	3 Blue LED’s on Front Panel
Selection/Status Indication	Solid: Digital Input Selection Flashing: Signal Error None: Standby Mode

AC Power Requirements	
Input Operating Voltage Range (VAC RMS)	110 V setting: 90 V min, 140 V max 220 V setting: 175 V min, 285 V max
Frequency	50-60 Hz
Power	8 Watts Idle 8 Watts Typical Program 16 Watts Maximum
Fuses	5 x 20 mm (2 required) 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 but without power cord	9.33" (237 mm)
Width	9.5" (249 mm)
Height	1.725" (44.5 mm)

Weight	
DAC1 PRE only	3.5 lb.
DAC1 PRE with power cord, extra fuses, and manual	4.5 lb.
Shipping weight	7 lb.

Regulatory Compliance

FCC and RoHS Compliance Statements

FCC Notice (U.S. Only)

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

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

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

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

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

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

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

RoHS Compliant Information

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

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

CE Certificates of Conformity

Certificate Of Conformity

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

EMC Directive:	**89/336/EEC**	
Generic Emissions Standard:	EN 61000-6-3: 2001	
Product Specific Emissions:	EN 55022 Class B	
Generic Immunity Standard:	EN 61000-6-1: 2001	
Immunity:	EN 61000-4-2	Electrostatic Discharge
	EN 61000-4-3	Radiated Susceptibility
	EN 61000-4-6	Conducted Susceptibility

Manufacturer's Name:	**Benchmark Media**
Manufacturer's Address:	5925 Court Street Road Syracuse, N.Y. 13026
Product:	Dac1 Digital / Analog Converter
Model Number:	Dac1

This Certificate of Compliance issued January 23, 2007 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature: Annelle Frierson

Annelle Frierson
Vice President
Diversified T.E.S.T. Technologies, Inc.
PO Box 8, 556 Route 222
Groton, NY 13073
Phone: 607-898-4218
Fax: 607-898-4830

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

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: 2001
Product Specific Emissions:	EN55022

Manufacturer's Name:	**Benchmark Media Systems Inc.**
Manufacturer's Address:	5925 Court Street Road Syracuse, NY 13206
Product:	DAC1 PRE
Model Number:	450-14075-011

This Certificate of Compliance issued February 8, 2008 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature: Annelle Frierson __

Annelle Frierson
Vice President
Diversified T.E.S.T. Technologies, Inc.
PO Box 8, 556 Route 222
Groton, NY 13073
Phone: 607-898-4218
Fax: 607-898-4830

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 (1) 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 submitting 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. Any DAC1 PRE returned from the European Union for warranty repair must have the required RoHS logo on the product label; otherwise, repairs will be billed at the normal shop rate. Benchmark will not honor warranties for any products disingenuously purchased on the US or Canadian markets for sale outside the US or Canada.

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 charge this information without notice. This limited warranty gives the consumer-owner specific legal rights, and there may also be other rights that vary form state to state.

Benchmark Extended Warranty

The Benchmark Extended 5* Year Warranty

Benchmark Media Systems, Inc. optionally extends the standard one (1) year warranty to a period of **five (5)* 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 prepaid 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's Extended 2** Year International Warranty

Benchmark Media Systems, Inc. optionally extends the standard one (1) year warranty to a period of **two (2)** 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 prepaid 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.

Benchmark Media Systems, Inc.
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...the measure of excellence!™

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