

Benchmark *DAC1 USB* Instruction Manual

**2-Channel 24-bit 192-kHz
Audio Digital-to-Analog Converter**



Benchmark
...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.

Faceplate Options

The **DAC1 USB** is available with 3 different faceplates. The internal electronics are identical in all versions.



Black



Silver



Black Rack-Mount

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Features

- USB, XLR balanced, BNC coaxial, and TOSLINK optical digital inputs
- Compatible with Windows and Mac operating systems without driver installation
- Jumper-selected low-impedance 10, 20, or 30 dB pads on balanced outputs
- -10 dBV unbalanced RCA analog outputs, +13.5 dBu maximum output level
- High-current output drivers for driving the most difficult loads without sonic degradation
- Two **HPA2™** high-current, “0-Ohm”, high-output ¼” headphone outputs
- **HPA2™** gain jumpers to match gain to headphone sensitivity
- Front-panel volume control for headphone outputs
- Front-panel volume control of all analog outputs (in **Variable** mode)
- Rear-panel **Variable/Calibrated** mode switch enables volume control of analog outputs
- Rear-panel **Variable/Calibrated** mode switch includes a center mute position
- In **Calibrated** mode, levels are set by 10-turn trimmers (20 dB Range, 2 dB/turn)
- XLR outputs are preset to +4dBu at 0 dBFS in **Calibrated** mode (20-dB Pad enabled)
- RCA outputs are preset to +2 Vrms (8.2 dBu) at 0 dBFS in **Calibrated** mode
- Headphone jack 1 mutes XLR and RCA outputs (feature may be disabled)
- Benchmark’s phase-accurate **UltraLock™** technology for total jitter immunity
- 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
- THD+N = -107 dB, 0.00045% @ -3 dBFS input, -105 dB, 0.00056% @ 0 dBFS input
- Automatic de-emphasis in response to consumer pre-emphasis bit (44.1, 48, 88.2, and 96 kHz)
- Reliable and consistent performance under adverse operating conditions
- 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 (12 Watts peak at 120 VAC)
- Meets FCC Class B and CE emissions requirements

Overview



The **DAC1 USB** is a reference-quality, 2-channel 192-kHz 24-bit digital-to-analog audio converter featuring Benchmark's **Advanced USB Audio** technology, **UltraLock™** clock system, and **HPA2™** headphone amplifier.

DAC1 Heritage

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

The **DAC1 USB** includes a very unique USB input with bit-transparent native 96/24 capability, a programmable mute function, programmable headphone gain range, an automatic standby/resume feature, and a high-current output stage designed to drive long cables or low-impedance loads, such as high-end power-amplifiers.

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

Applications

The **DAC1 USB** 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 **Advanced USB Audio** interface makes the **DAC1 USB** 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.

The rugged and compact rack-mount option makes the **DAC1 USB** an excellent choice for location recording, broadcast facilities, and mobile trucks.

Digital Inputs

There are four digital input connectors (coaxial, balanced XLR, optical, and USB). The coaxial, XLR, and optical digital inputs all accept professional (AES) and consumer (SPDIF) data formats.

Advanced USB 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).

Unlike all previous driverless USB audio interfaces, Benchmark's **Advanced USB Audio** technology supports sample rates up to 96 kHz and word lengths up to 24-bits.

The **DAC1 USB** is a true plug-and-play solution, and is designed to begin 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 USB* into the difficult loading presented by headphones. The *HPA2*[™] maintains less than 0.0003% THD+N under full load.

Mute Function

The left-hand headphone jack includes a switch that mutes the XLR and RCA analog outputs when a headphone plug is inserted. This mute feature can be disabled with internal jumpers.

Volume Control

The front-panel **Volume Control** sets the output level of the headphone jacks. It can also be used to control the output level of the balanced XLR and unbalanced RCA analog outputs. A rear-panel switch selects

Variable or **Calibrated** output levels. In **Variable** mode the analog outputs are controlled by the **Volume Control**.

Direct Interfacing to Power Amplifiers

The *DAC1 USB* is designed to interface directly to power amps and powered studio monitors in order to provide the cleanest and shortest path from the digital source to the monitor output. This often results in a substantial improvement in sound quality.

10, 20, and 30 dB pads are provided for interfacing directly to monitors and amplifiers that often have too much input sensitivity to handle high-level (+29 dBu) signal levels.

High-Current Output Drivers

The *DAC1 USB* 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 USB* 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 a digital input signal and only 500 ms after applying power to the unit.

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

performance, not just the performance of an output mute circuit.

The **DAC1 USB** 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.

Error Display

When the selected digital input signal is disconnected, incompatible, or drops out, the **Input Status Display** will flash indicating a signal error. This flashing will stop when the error is corrected. If the error persists for more than 15 seconds, the **DAC1 USB** will enter **Standby Mode**.

Automatic Standby/Resume

The **DAC1 USB** 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 LED's are off while in standby mode.

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

Low-Noise Internal Power Supply

The internal power supply supports all international voltages and has 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** and **DAC1 USB** converters at sample rates up to 96 kHz. Phase-accurate multi-track and 5.1 surround systems are easily constructed using multiple **DAC1** and/or **DAC1 USB** 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 USB** 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.

Front Panel



Input Status Display

The **DAC1 USB** has three LED's on the front panel that indicate which digital input is selected. These LED's flash when an error condition occurs. All LED's turn off when the **DAC1 USB** is in **Standby Mode**.

Input Selector Switch

The input selector toggle switch is located directly to the right of the **Input Status Display**.

The switch is a momentary 3-position switch that scrolls up and down through the 4 digital inputs in a round-robin format.

If the **DAC1 USB** is in **Standby Mode** it will resume normal operation when the **Input Selector Switch** is toggled.

Input Indication

The numbers to the left of the LED's correspond to the following inputs:

1. Coaxial (bottom LED)
2. XLR (middle LED)
3. Optical (top LED)
4. USB (top and bottom LED's)

When the top and bottom LED's are lit simultaneously, the USB (Input 4) is selected.

Error Indication

The **Input Status Display** will flash when an error occurs on the selected digital input. The number of times the display flashes before entering standby indicates the type of error. If the error is not resolved within about 15 seconds, the **DAC1 USB** will enter **Standby Mode**. The **DAC1 USB** will resume normal operation when it detects a valid input signal at the last chosen input.

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 audio data type (AC3, ADAT, etc.)
- Non-Audio data

Standby Mode

The **DAC1 USB** 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 status LED's are off while in **Standby Mode**.

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



HPA2™ Stereo Headphone Jacks

There are two stereo headphone jacks. The left-hand jack is equipped with a switch that mutes the XLR and RCA analog outputs when a headphone plug is inserted. The right-hand jack has no switch. This gives the user the option of muting the analog outputs when using headphones. Internal jumpers are available to disable the headphone switch.

TIP: Use the left-hand jack when you want to mute your playback system and just listen to headphones. 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. We recommend using identical headphones because headphone sensitivities can differ significantly. The **Volume Control** adjusts the level at both jacks.

Because of the variations in headphone sensitivity, the **DAC1 USB** features two gain ranges for the headphone outputs. Our

experience with the classic **DAC1** showed us that most users cannot use the full output of the **HPA2™**. These users benefit from a reduced maximum gain. Internal jumpers reduce the input to the **HPA2™** by 10 dB and place the **Volume Control** near 12 O'clock at a comfortable listening level with most headphones. These jumpers are factory installed, but can be removed if a user needs more output. Removing the jumpers increases the gain to a level that is equal to that of the classic **DAC1**.

TIP: For optimal L/R balance, the headphone 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 headphone gain range and headphone mute switch are detailed in the 'Internal Settings' section of this manual.

Volume Control

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

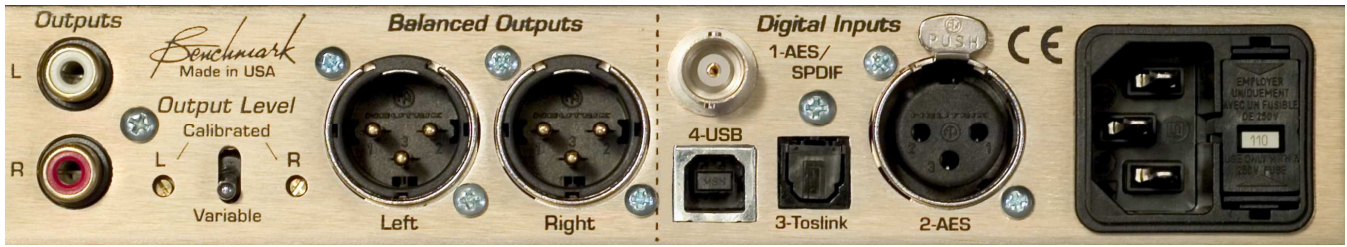
The **Volume Control** always controls the output level of the headphone jacks. It can also be used to control the output levels of the balanced XLR and unbalanced RCA analog outputs when the rear-panel **Output Level Switch** is set to **Variable**.

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

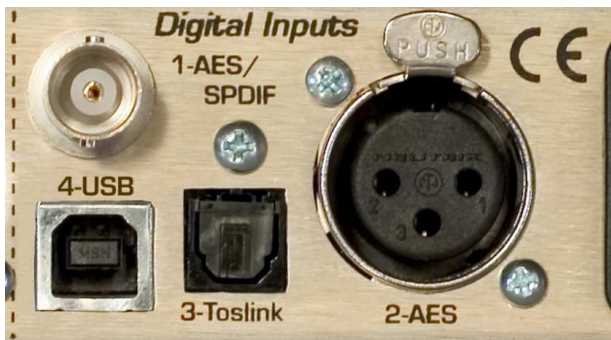
TIP: For optimal L/R balance, and minimal noise, the XLR gain jumpers should be set so that comfortable listening levels occur when the 'Volume Control' is set above the 10th detent.

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

Rear Panel



Digital Inputs



There are four digital inputs on the **DAC1 USB**. These inputs are selected from the front-panel toggle switch.

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

TIP: The DAC1 USB 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 inputs. The result is that all digital inputs have identical jitter performance.

TIP: Internal jumpers can be used to determine which of the four inputs is active when the DAC1 USB is powered up. Factory default is input 4 (USB).

Input 1 - SPDIF/AES BNC

The coaxial input uses a female BNC connector that is 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 BNC input is DC isolated, transformer coupled, current limited, and diode protected. The BNC 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 highly recommended for stable performance. Do not use 50-Ohm cables.

The BNC input accepts AES/EBU or S/PDIF digital audio formats. The BNC connector is specified by the AES3-id and SMPTE 276M standards for 75-Ohm 1 Vpp digital audio signals and is commonly used in video production facilities. IEC 609588-3 specifies the use of an RCA type connector for 75-Ohm 0.5 Vpp consumer-format digital audio signals (commonly known as S/PDIF). The coaxial input on the **DAC1 USB** is designed to accept either type of signal.

We have chosen to use a BNC connector because of its superior reliability and RF performance. The **DAC1 USB** comes with a BNC-to-RCA adapter to allow easy interfacing with consumer S/PDIF equipment.

TIP: The BNC input has a 75-Ohm internal termination that may be

disabled by removing a jumper (JP7). This termination is required for normal operation, but may be removed if the user wishes to loop a single coaxial feed through several other pieces of equipment.

A 75-Ohm termination must be applied at the last device on the loop, and there should be a combined total of less than 6 feet of cable between the first and last receive device.

Input 2 – Digital XLR

This input uses a gold-pin Neutrik™ female XLR connector. The input is balanced and has an input impedance of 110 Ohms. Maximum word length is 24-bits. All sample rates between 28 and 195 kHz are supported.

This input is DC isolated, transformer coupled, current limited, and diode protected. 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 input is designed to accept standard 4 Vpp AES/EBU or S/PDIF signals, and will work reliably with levels as low as 300 millivolts. Receive errors will occur (and the **DAC1 USB** will mute) when the peak-to-peak input voltage is less than about 160 millivolts at the XLR input.

TIP: Shielded 110-Ohm AES/EBU digital audio cable is highly recommended for stable performance. For improved shielding, we also recommend selecting digital cables that have the connector shells bonded to the cable shield.

Input 3 – 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 USB** 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.

Input 4 – 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 USB**. The USB cable connects the **DAC1 USB** 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 **Advanced USB 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 USB** 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.

Analog Outputs



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

The **DAC1 USB** 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. Otherwise the levels are set by the 10-turn calibration trimmers located on the rear panel.

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 USB** 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 produces an output that is near full-volume and may damage your speakers.

Calibrated (UP) – Analog output levels are controlled by 10-turn rear-panel trim controls.

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 USB 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' 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 rear-panel **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 USB** 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 rear-panel **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 USB**. The **DAC1 USB** 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 mid-rotation.

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 to the left and right of the **Output Level Switch**. They are 10-turn trimmers and are accessible through the rear panel 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 Module

The AC power input uses a standard IEC type connector. One USA-compatible power cord is included with **DAC1 USB** 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 USB** will operate normally over a range of 90 to 140 VAC. At **220**, the **DAC1 USB** 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.

Rack Mounting

(**DAC1 USB** Black RM only)

The rack-mount version of the **DAC1 USB** is part of Benchmark's ½-wide **System1™** product family. Each is one rack unit high and is exactly ½ the width of a standard 19" rack panel. The front panels on **System1™** products have rack-mount holes that are machined to conform to standard rack-mount dimensions. Two ½-wide **System1™** units may be joined together to form a single rigid 19" panel that can be installed in any standard 19" rack.

Either ear of a ½-wide **System1™** device can be mounted directly to a standard 19" rack. A **Rack Mount Coupler** connects the other ear to a ½-wide **Blank Rack Panel** or another ½-width **System1™** product

TIP: Use the rack-mount screws supplied with the DAC1 USB (or screws with plastic washers) to avoid scratching the surface of the faceplate.

The **Rack Mount Coupler** 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.

Rack Mount Coupler



The **Rack Mount Coupler** is a machined aluminum junction block that joins any two ½-wide **System1™** devices for rack mounting. It is also used to join a **Blank Rack Panel** to a single ½-wide **System1™** device.

Blank Rack Panel



The **Blank Rack Panel** is a ½-wide 1-RU aluminum panel for mounting a single ½-wide **System1™** device in a standard 19" rack. Installation requires one **Rack Mount Coupler**.

Benchmark Technologies

HPA2™ Headphone Amplifier

The **DAC1 USB** 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.

Headphone Gain Range

Because of the variations in headphone sensitivity, the **DAC1 USB** features two gain ranges for the headphone outputs. Our experience with the classic **DAC1** showed us that most users cannot use the full output of the **HPA2™**. These users benefit from a reduced maximum gain. Internal jumpers reduce the input to the **HPA2™** by 10 dB and place the **Volume Control** near 12 O'clock at a comfortable listening level with most headphones. These jumpers are factory installed, but can be removed if a user needs more output. Removing the jumpers increases the gain to a level that is equal to that of the classic **DAC1**.

TIP: For optimal L/R balance, the headphone 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 headphone gain range and headphone mute switch are detailed in the 'Internal Settings' section of this manual.

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 unless it influences the conversion clock in an analog-to-digital converter (A/D) or in a digital-to-analog converter (D/A).

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

Better converters usually use a two-stage PLL circuit to filter out more of the interface jitter. In theory, a two-stage PLL can remove enough of the jitter to achieve accurate 24-bit conversion (and some do). However, not all two-stage PLL circuits are created equal. Many two-stage 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 USB, 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 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 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 USB** 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 USB** eliminates one major variable: jitter.

Advanced USB Audio Technology

Benchmark's **Advanced USB 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 all major operating systems. 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 up-to-date information about more recent operating systems and suggestions for optimization, go to: www.benchmarkmedia.com/wiki).

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 **Advanced USB 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 **Advanced USB 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 **Advanced USB 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.

Microsoft Windows® Test Results

Windows® 2000 and XP operating systems have a digital mixer known as 'Kmixer'. All audio streams must go through the Windows® Kmixer to reach native USB audio devices. The performance of Kmixer is critical to any native USB audio solution, so we tested it extensively.

We found that Kmixer can perform with full or near full bit-transparency under the right conditions. But, 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 **Advance USB 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 and Win 7. By default, Vista and Win 7 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. Nevertheless, if true bit-transparent operation is desired, Vista's and Win 7's sample rate conversion can be tuned of by manually setting the system sample rate.

In any Windows® operating system, true bit-transparency is only achieved when the 'Wave' volume control is 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 originating from a single application, Benchmark's **Advanced USB 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 **Advanced USB 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 match the sample rate of the files being played, and the system volume control must be set to 100%.

Like the Windows® XP and 2000 operating systems, OS X has 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. This is not a problem if the system will always be used at a single sample rate such as CD's at 44.1 kHz.

OS X also has a poor quality master volume control. It will degrade the quality of the audio if it is not set to 100%. Fortunately it is bit-transparent when set to 100%.

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

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 **Advanced USB 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.

Performance Graphs

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

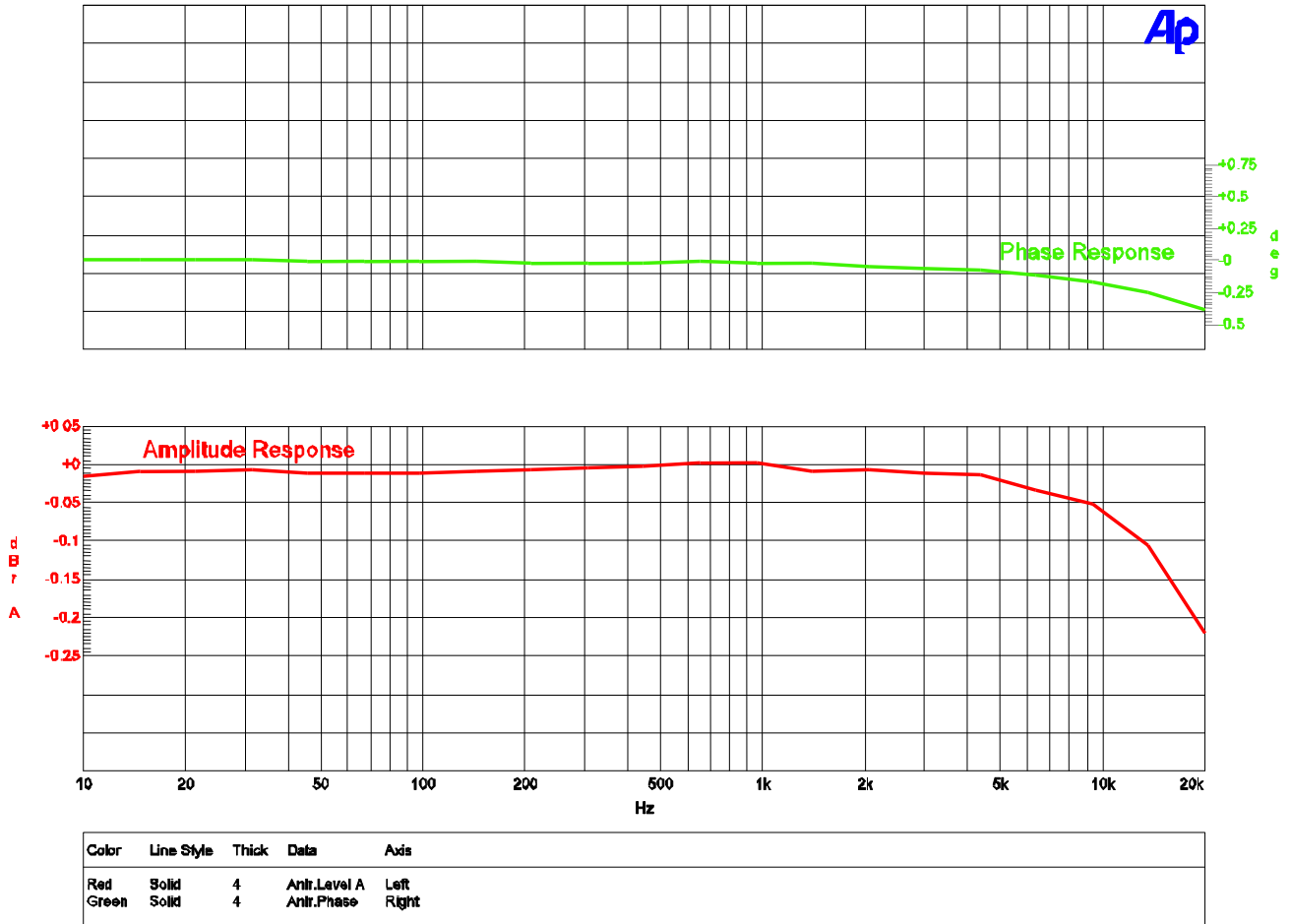
Frequency Response Tests

Frequency Response at $F_s = 48$ kHz

Benchmark Media Systems, Inc.

DAC1 - Frequency Response at $F_s = 48$ kHz

08/23/02 17:39:44



3 - Freq Response at 2c

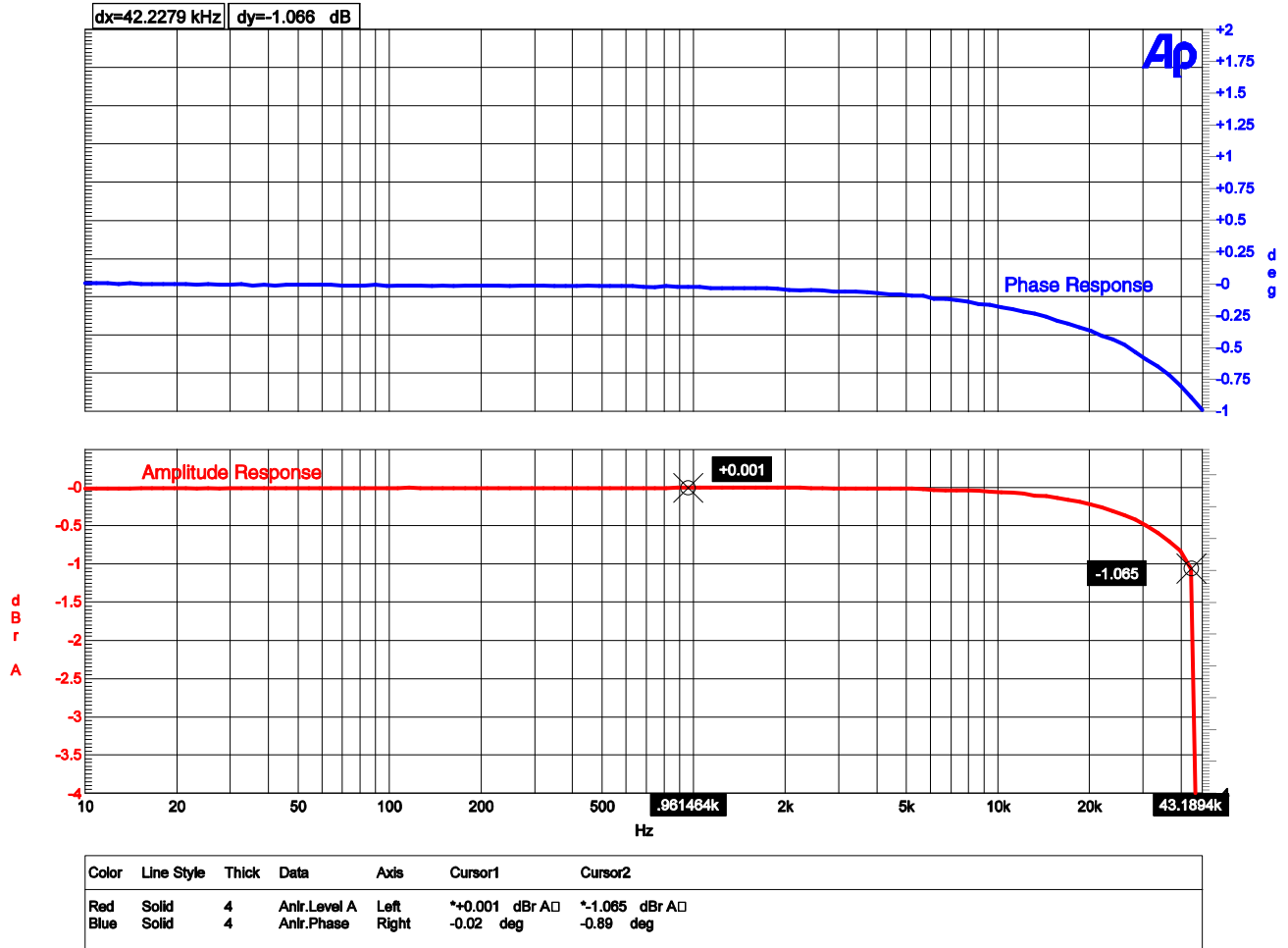
The above graphs show the frequency response of the **DAC1** when it is operating at a 48-kHz sample rate. The top graph shows that the differential phase is better than $\pm 0.5^\circ$ at 20 kHz. The bottom graph shows the amplitude response on a highly expanded 0.05 dB/division scale. The amplitude 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

Benchmark Media Systems, Inc.

DAC1 - Frequency Response at Fs = 96 kHz

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96 kHz Freq Response.at2c

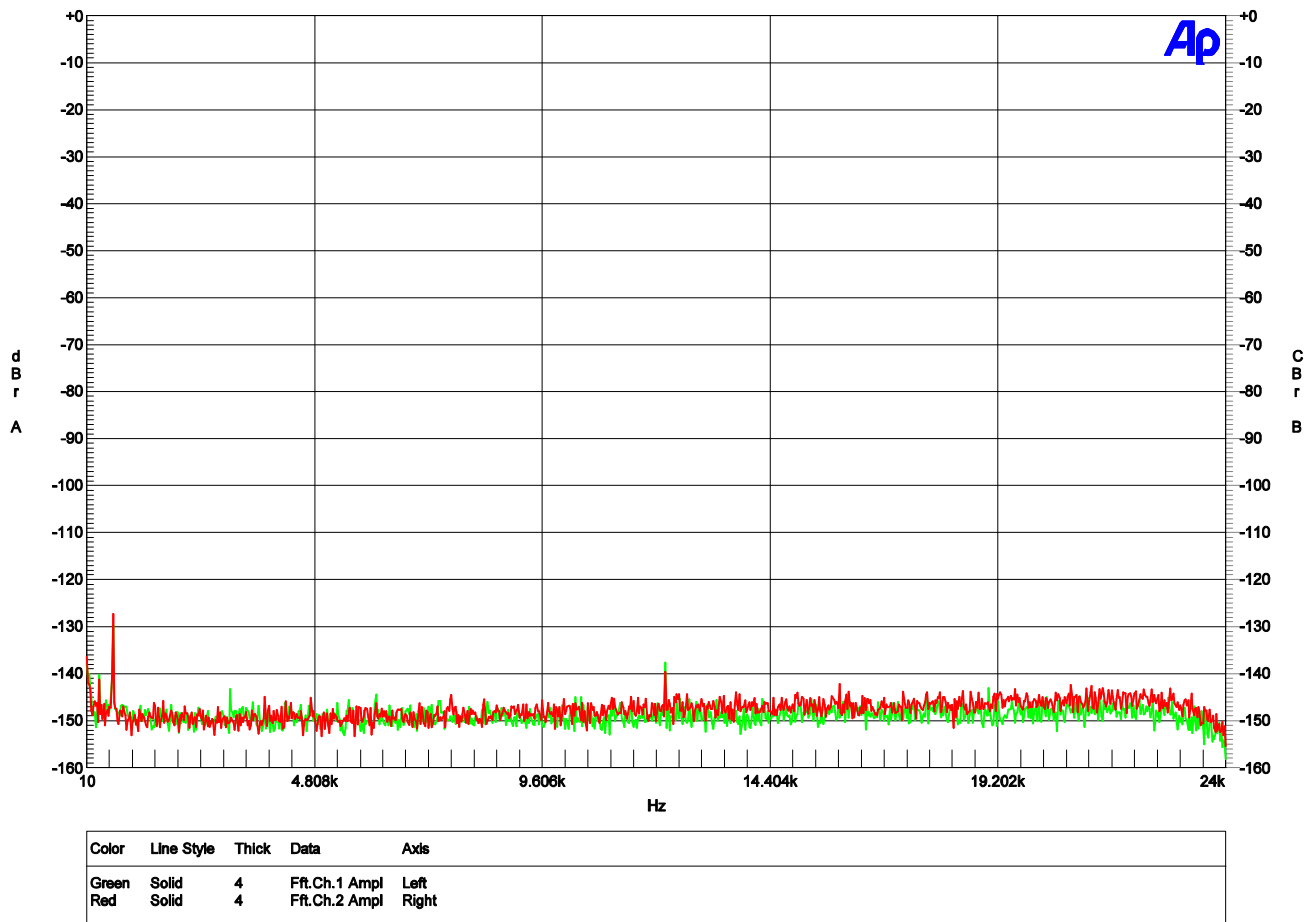
The above graphs show the frequency response of the **DAC1** when it is operating at a 96-kHz sample rate. The top graph shows that the differential phase is better than $\pm 0.5^\circ$ at 20 kHz and better than $\pm 1^\circ$ at 43 kHz. The bottom graph shows the amplitude response on a highly expanded 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

Benchmark Media Systems, Inc.

DAC-1
32K B-H FFT Analysis of Idle Channel Noise

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8 - FFT Idle Chanel Noise.at2c

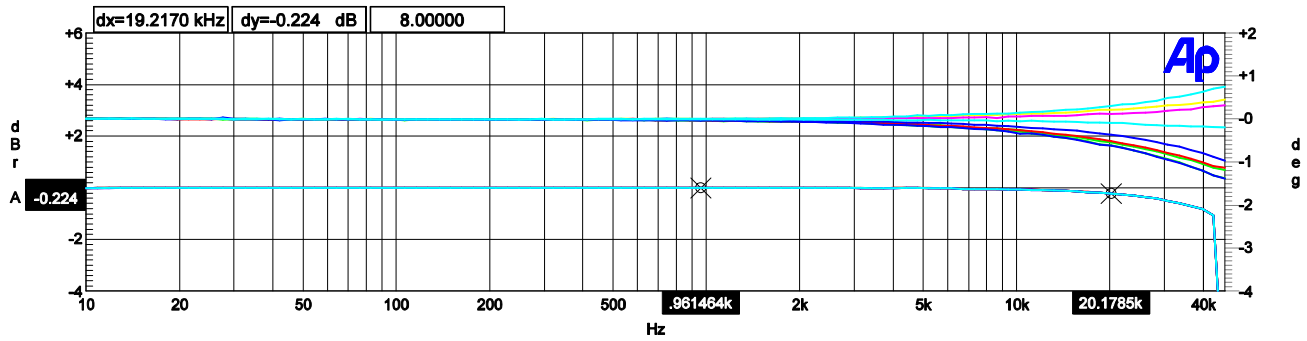
The above graph demonstrates that the **DAC1** is free from idle tones and clock crosstalk. The highest spurious tone measures -128 dBFS and is AC line related hum. The highest non-line related tone measures -138 dBFS.

Multi-Unit Phase Response

Benchmark Media Systems, Inc.

DAC1 - Phase and Frequency Response Matching at Fs = 96 kHz
 10 Channels of Conversion using 5 DAC-1 Converters Selected at Random

08/30/02 16:48:42



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 - Reference 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

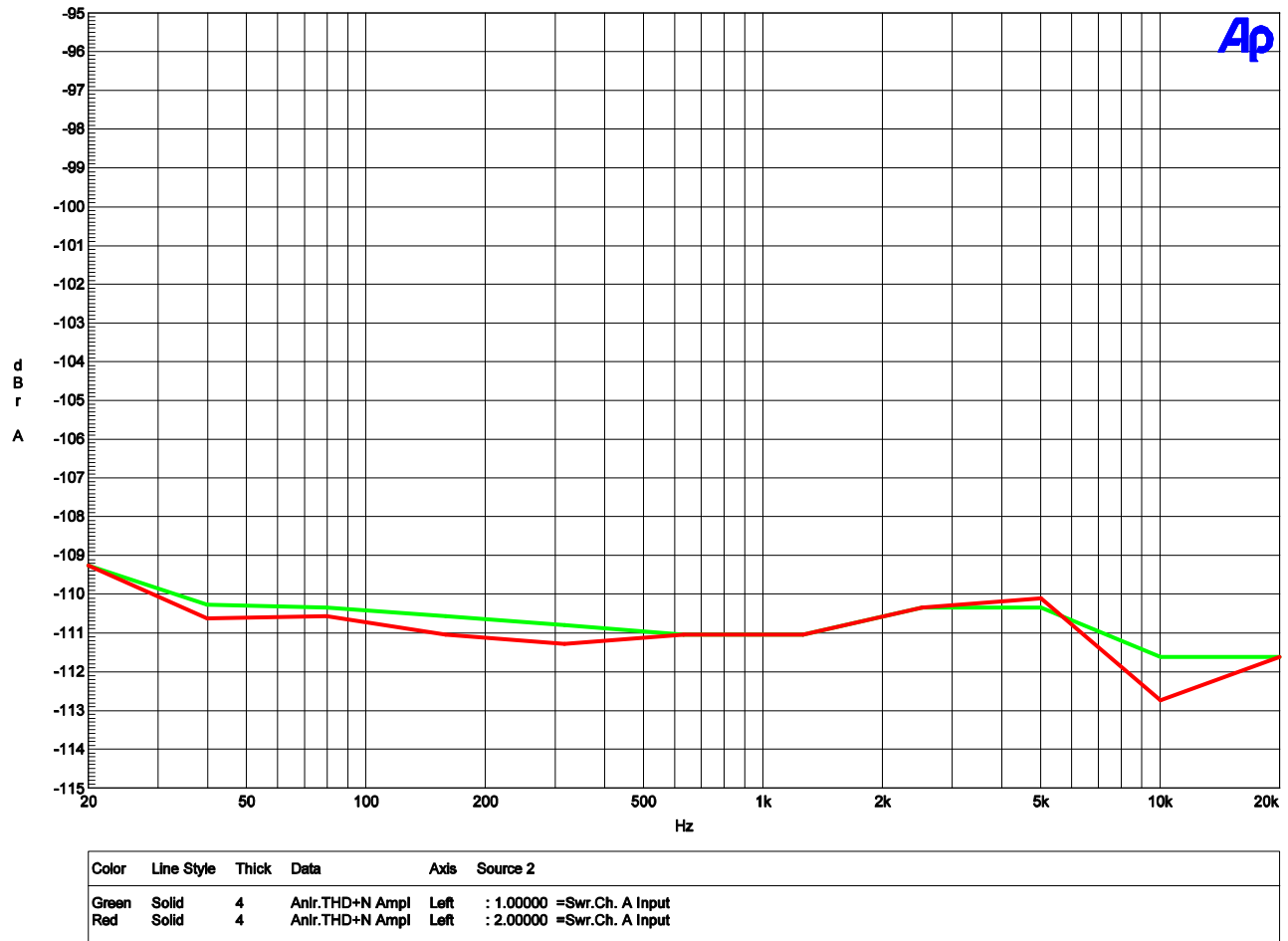
Phase and Amplitude Matching Between DAC-1 Units.at2c

Any combination of **DAC1** and **DAC1 USB** 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

Benchmark Media Systems, Inc. DAC1 - THD+N vs Frequency @ -3DBFS (w/20 kHz LPF unweighted) 08/30/02 10:32:39



5 - THD+N vs Frequency.at2c

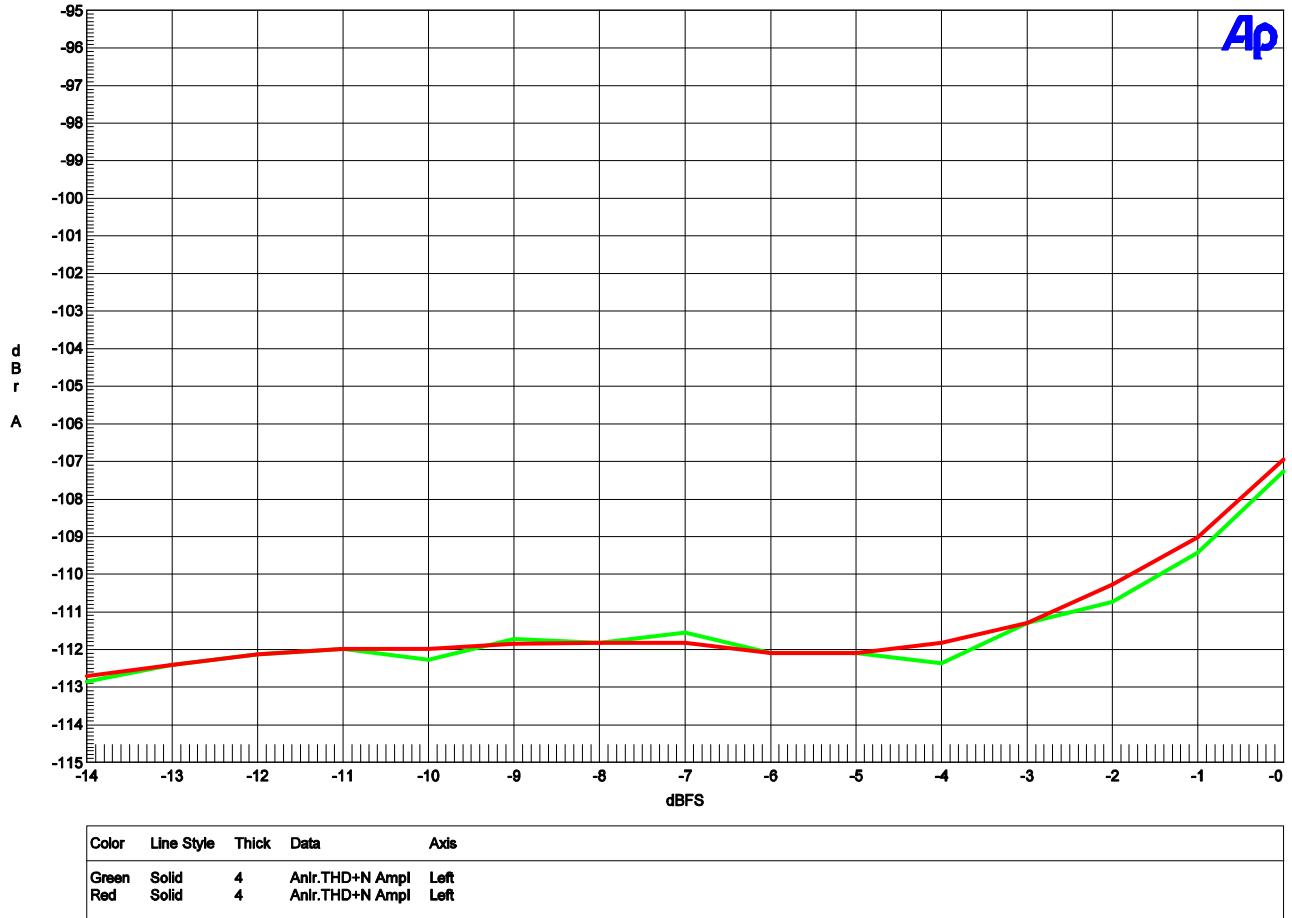
The above graphs demonstrate that the THD+N specifications for the **DAC1** are not frequency dependent (the variation from 20 Hz to 20 kHz is very slight). Note that at worst case, the distortion is 109 dB less than the - 3 dBFS test tone (and 112 dB less than the full scale output of the **DAC1**). 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

Benchmark Media Systems, Inc.

DAC-1 - THD+N vs Level 1 KHz (w/20 kHz LPF unweighted)
Balance Outputs, Relative to 0 dBFS, 0 dBFS = +24 dBu

10/01/02 13:20:27



THD+N vs Level in dBFS.at2c

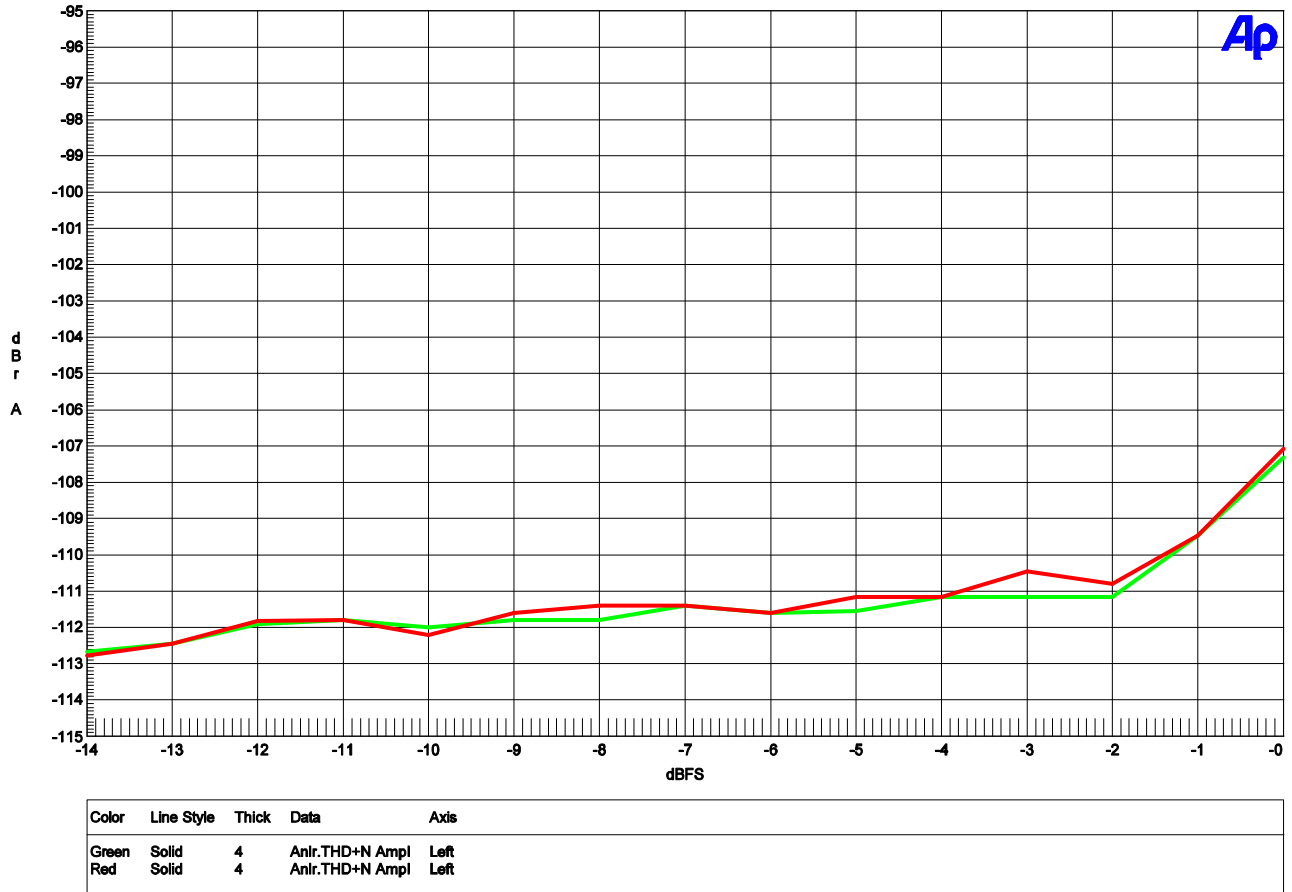
Below -4 dBFS, distortion is lower than the noise floor of the converter. Above -3 dBFS, distortion reaches a maximum value of only -107 dBFS.

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

Benchmark Media Systems, Inc.

DAC-1 - THD+N vs Level 1 KHz (w/20 kHz LPF unweighted)
Headphone Outputs, Relative to 0 dBFS, 0dBFS = +14 dBu, Load = 60 Ohms

10/01/02 13:22:35



HPA THD+N vs Level in dBFS.at2c

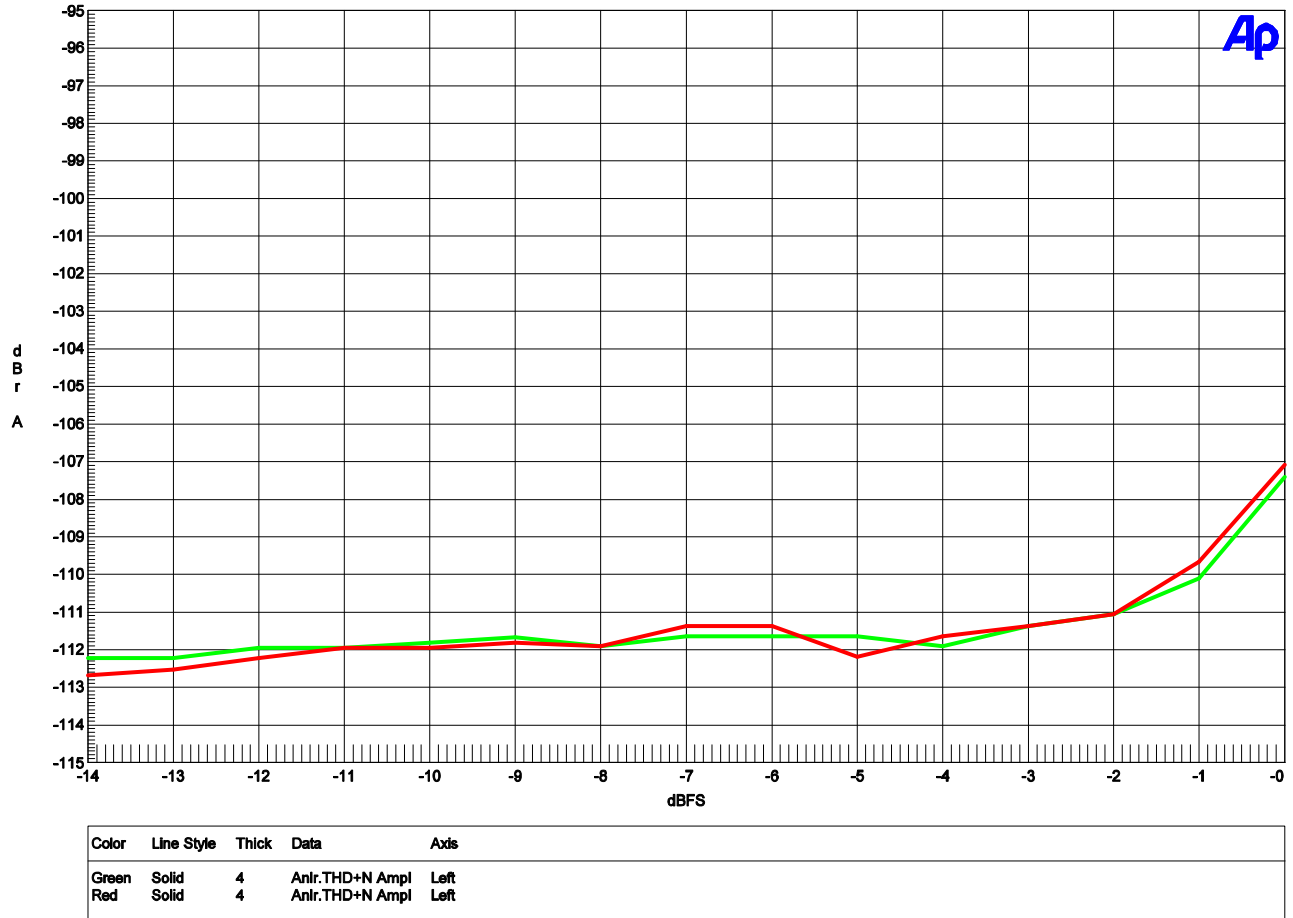
This graph shows the output of the **HPA2™** headphone amp driving a 60-Ohm load at a very high level (+14 dBu). Even under these conditions, the **HPA2™** delivers the full rated performance of the **DAC1**. Compare this to the performance of the balanced outputs (see previous graph).

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

Benchmark Media Systems, Inc.

DAC-1 - THD+N vs Level 1 KHz (w/20 kHz LPF unweighted)
 Unbalanced Outputs, Relative to 0 dBFS, 0 dBFS = +12 dBu

10/01/02 13:19:37



Unbalanced THD+N vs Level in dBFS.at2c

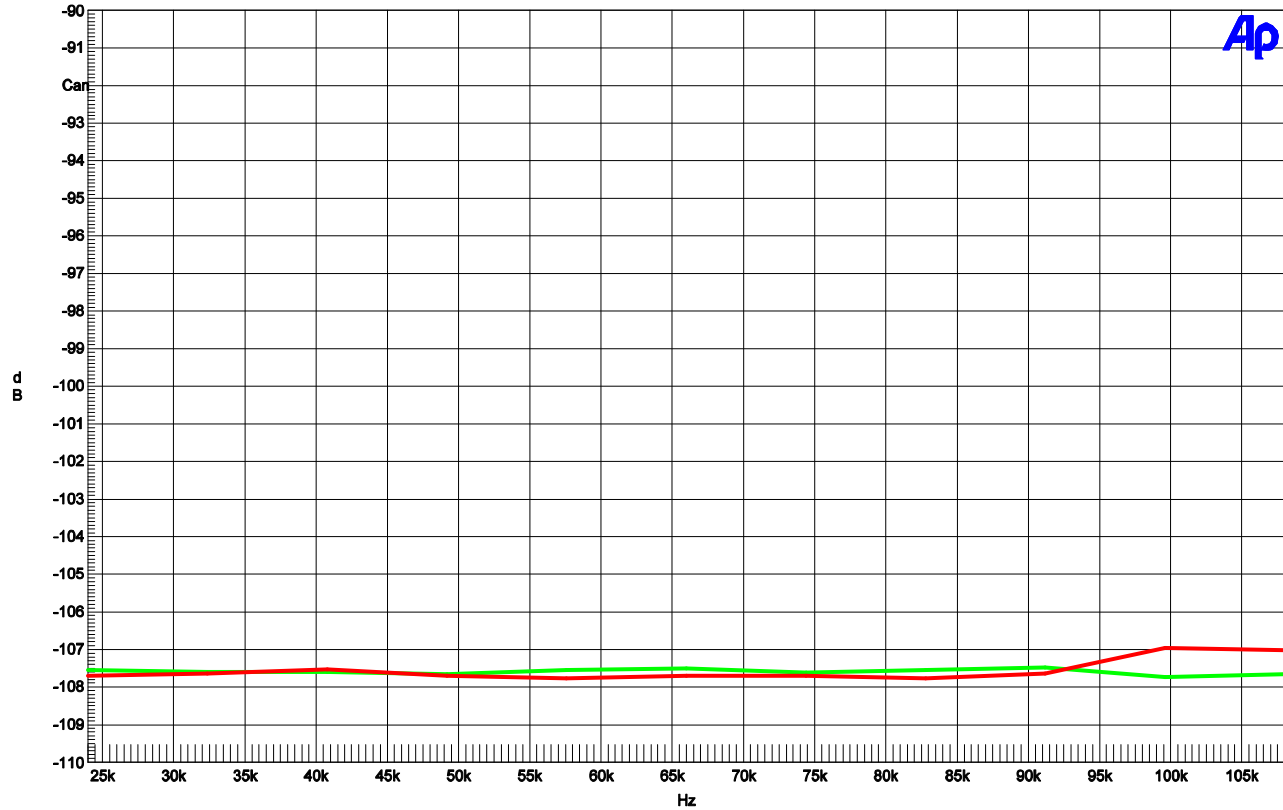
This graph demonstrates the performance of the unbalanced outputs. Note that the performance is nearly identical to that of the balanced outputs.

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.

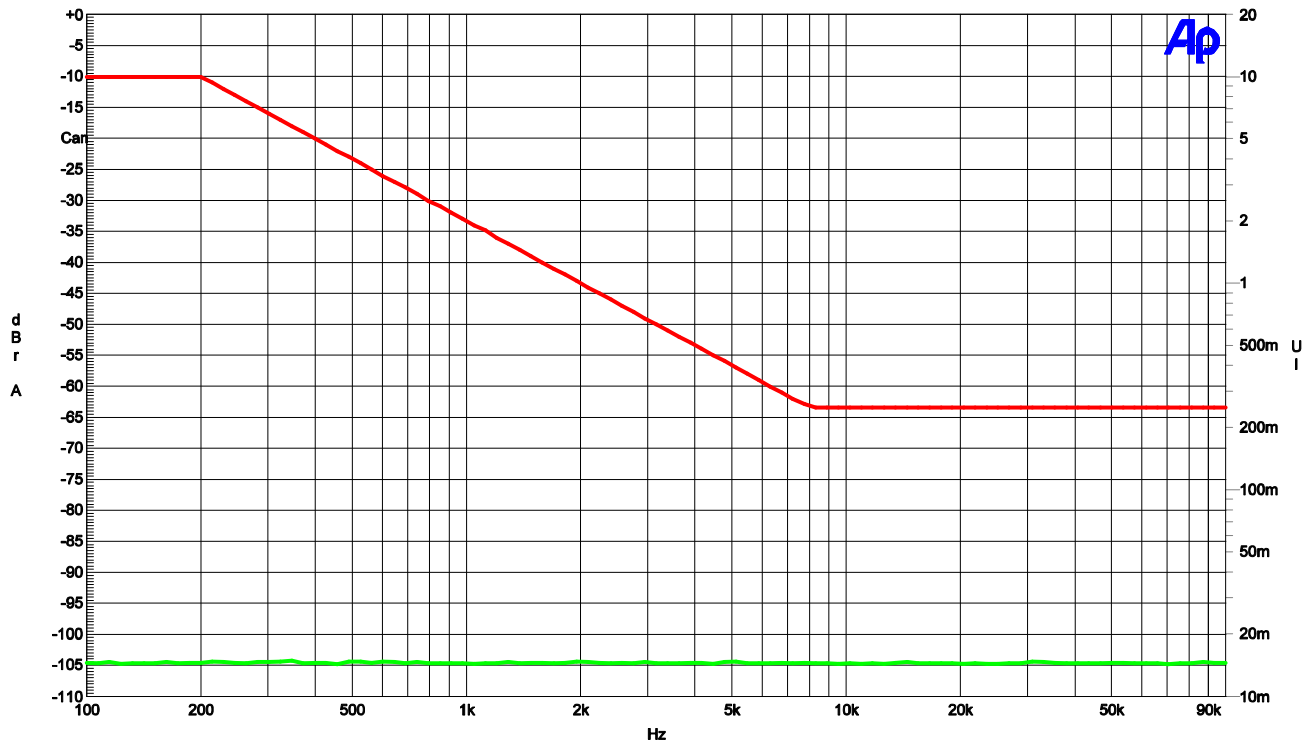
Jitter Tests

AES Jitter Tolerance Test

Benchmark Media Systems, Inc.

DAC-1 INTERFACE JITTER TOLERANCE - Distortion vs Jitter
3.456 kHz Test Tone at 0 dBFS, THD+N 22to22kHz BW

08/30/02 11:57:23



Color	Line Style	Thick	Data	Axis
Green	Solid	2	Anlr.THd+N Ampl	Left
Red	Solid	2	Dio.Jitter Ampl	Right

DAC-1 THD+N (20 Hz to 20 kHz) (Green trace) vs. Induced Interface Jitter (Red trace).
Digital Input = 3.456 kHz at 0 dBFS, Fs = 48 kHz

DIO D-A JITTER TOLERANCE.at2c

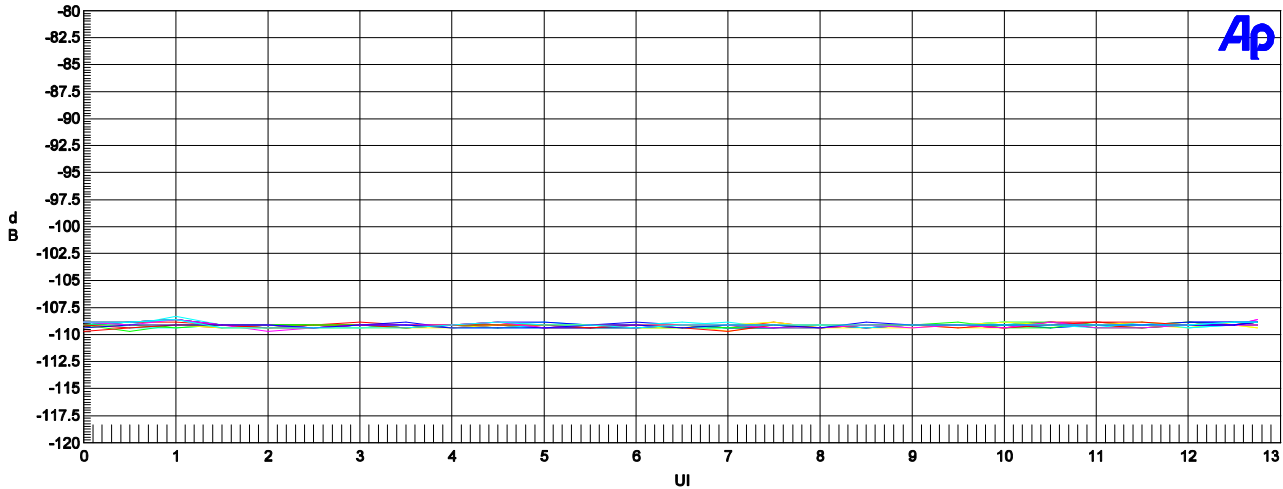
The graph above shows the results of a standard AES jitter tolerance test.

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 08/30/02 11:10:38
 (2 Hz to 9 kHz)
 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).

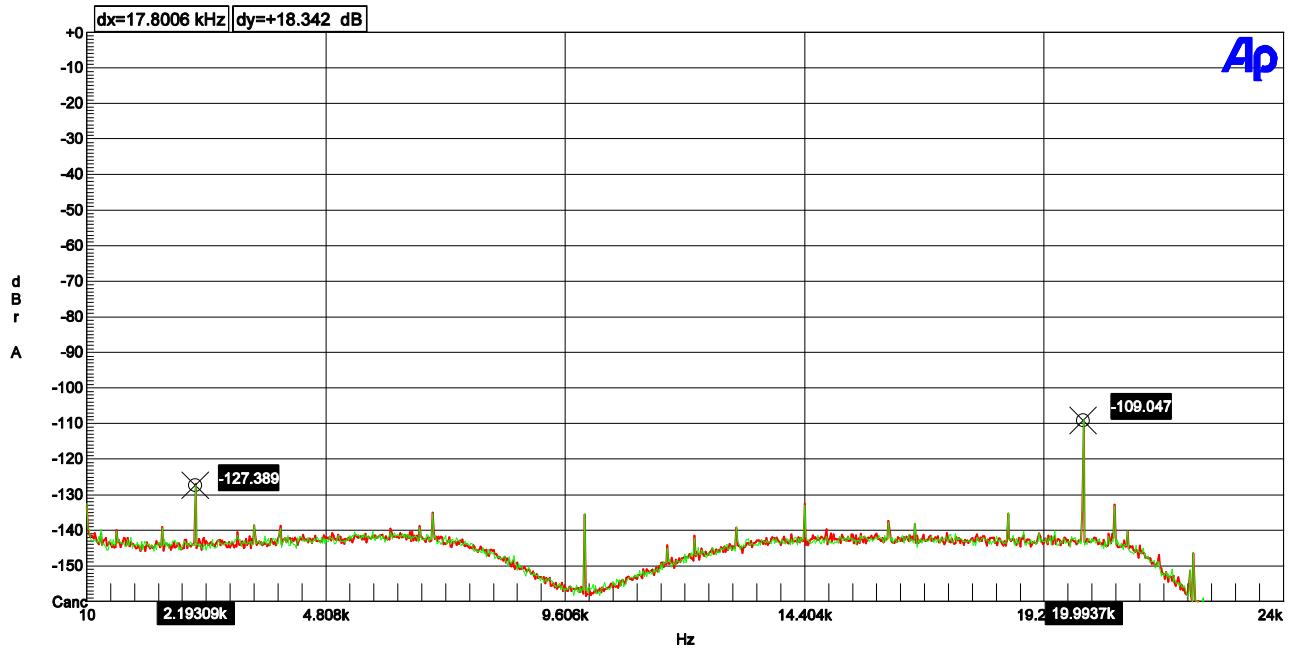
Immunity to Cable-Induced Jitter

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 extremely low levels and are unchanged when compared 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 spurious 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**.

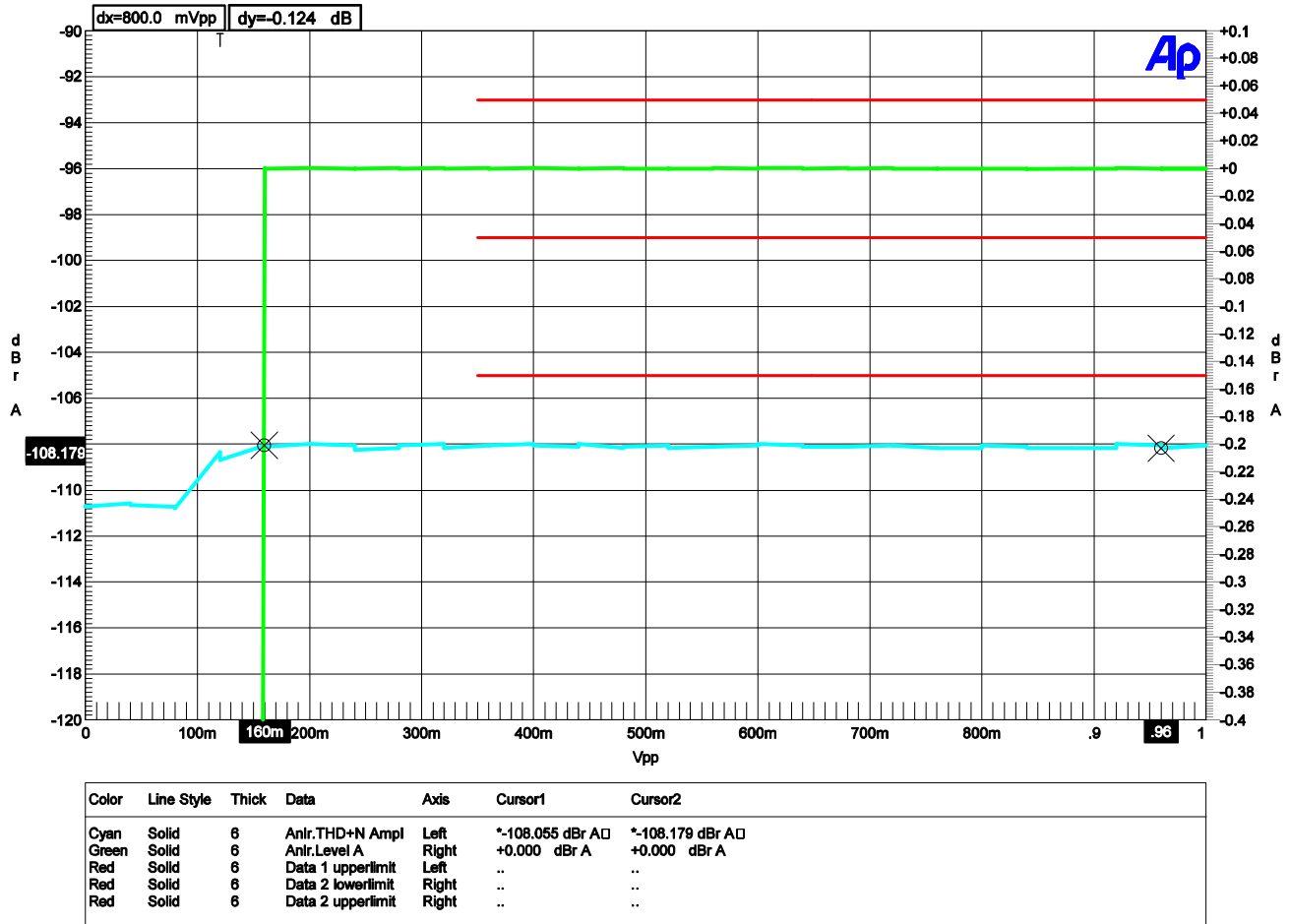
Input Sensitivity Tests

XLR Digital Input Sensitivity

Benchmark Media Systems, Inc.

DAC-1 - XLR Digital Input - Sensitivity Test
 Converter mutes when digital input < 160 mVpp

08/30/02 15:07:27



11 - AES3 Sensitivity Test.at2c

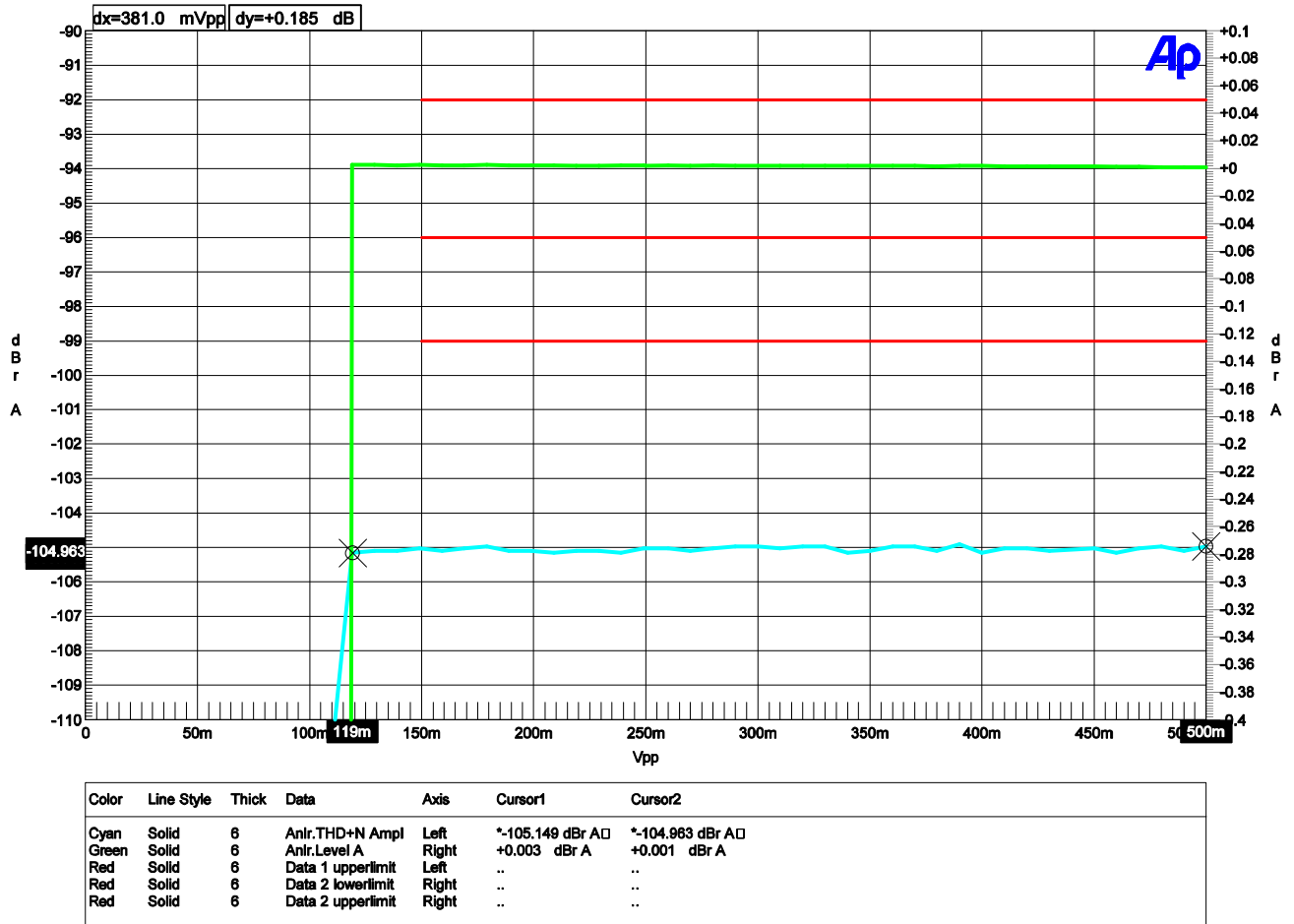
The above graph shows that the performance of the **DAC1** is not a function of the signal level at the XLR digital input. When the signal is too low to decode (< 160 mVpp), the converter mutes gracefully.

Coaxial Digital Input Sensitivity

Benchmark Media Systems, Inc.

DAC-1 - Coaxial Input - Sensitivity Test
 Converter mutes when digital input < 120 mVpp

08/30/02 14:48:03



10 - AES3id Sensivity Test.at2c

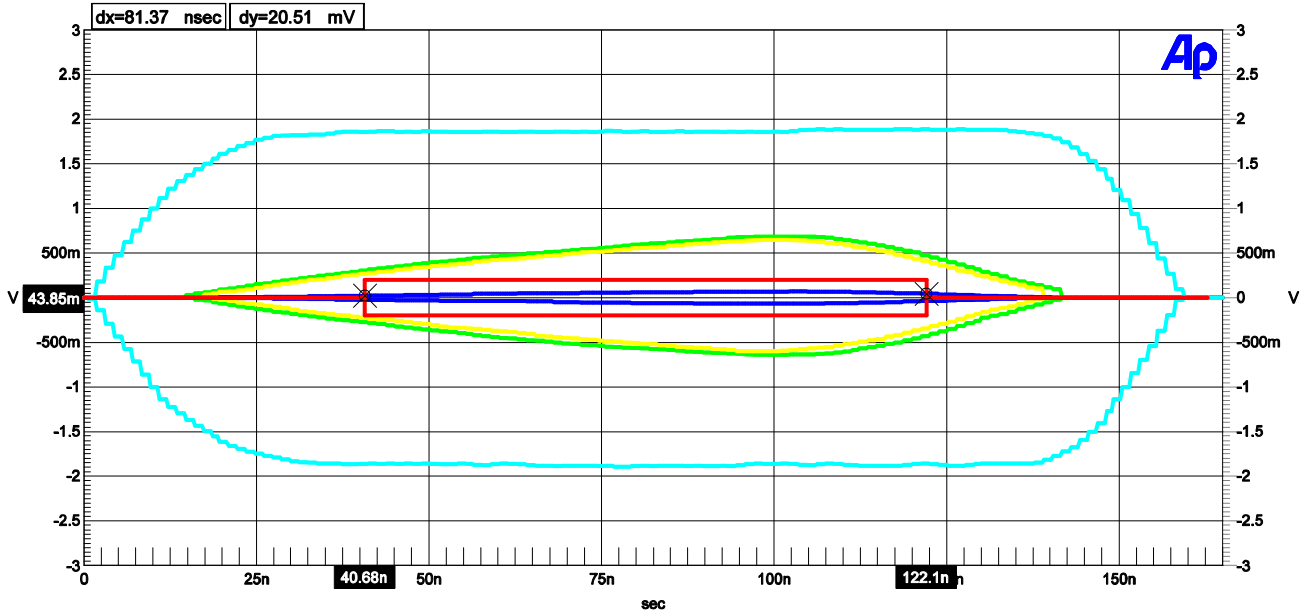
The above graph shows that the performance of the **DAC1** is not a function of the signal level at the coaxial digital input. When the signal is too low to decode (< 120 mVpp), the converter mutes gracefully.

Minimum Eye Pattern

Benchmark Media Systems, Inc.

DAC1 Digital Input Sensitivity Eye Pattern with AES3 Limits

08/30/02 14:27:46



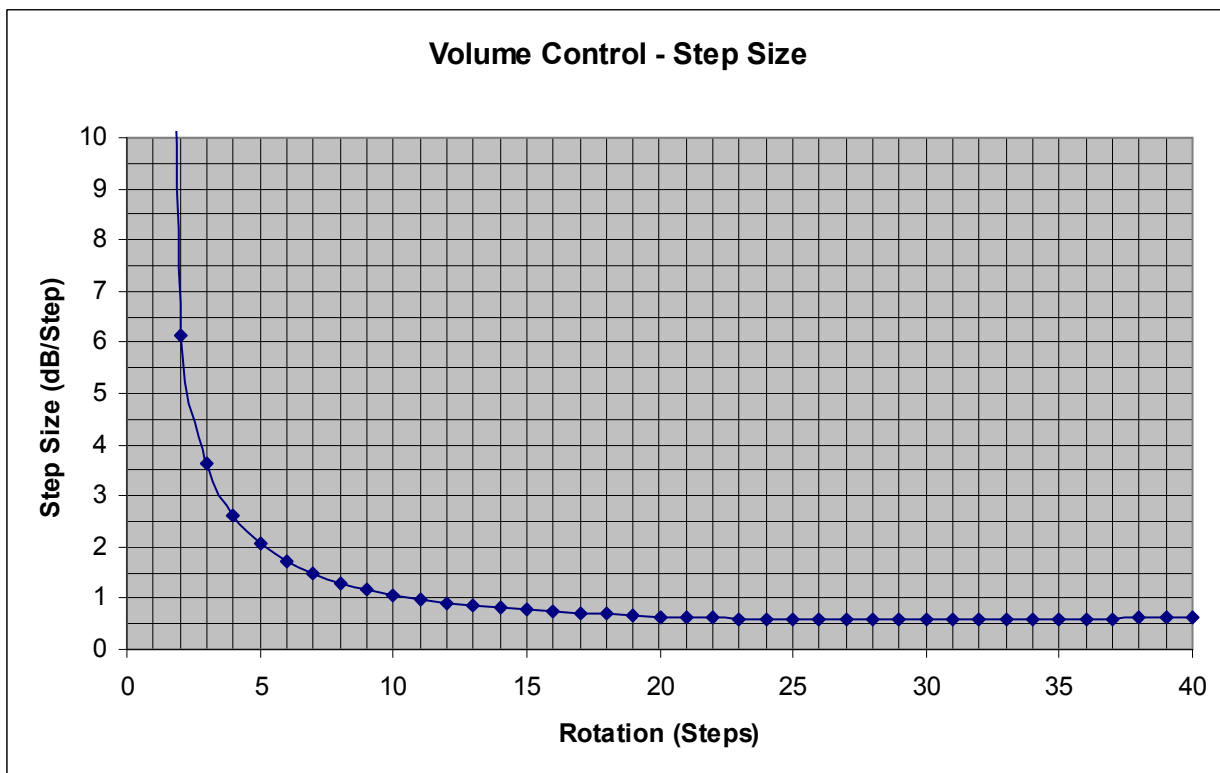
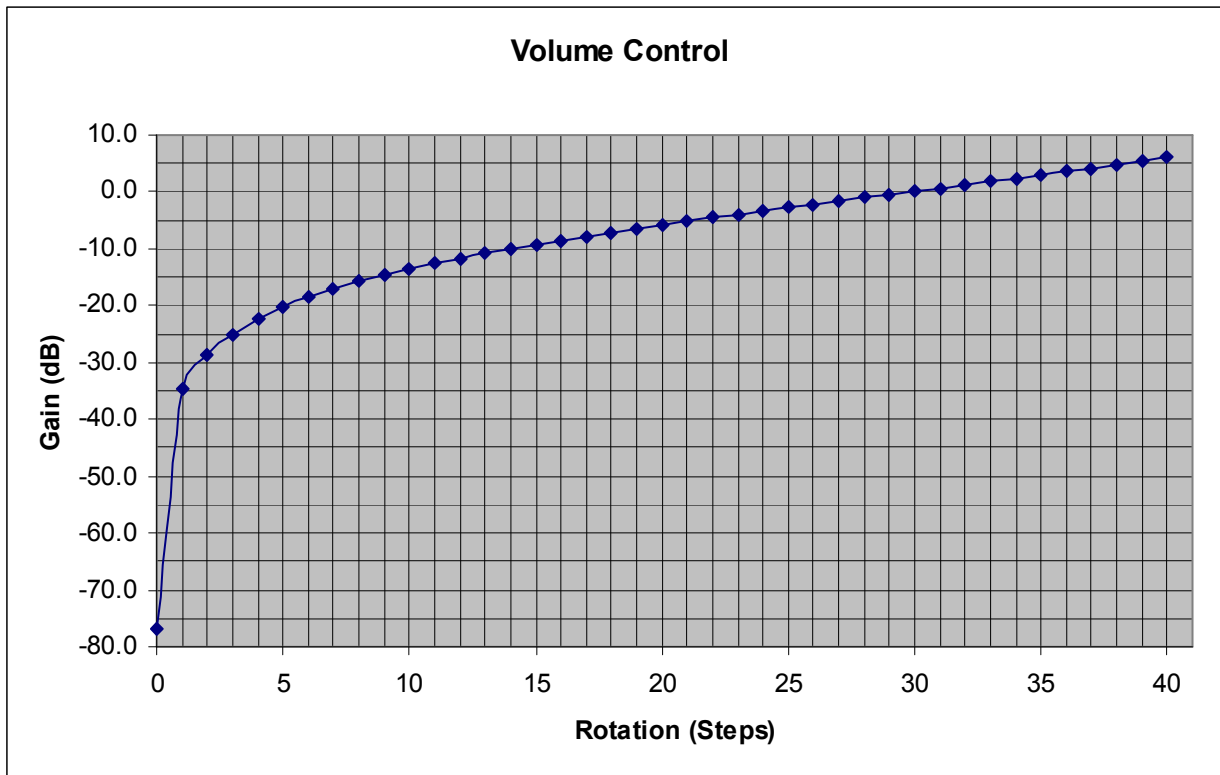
Color	Line Style	Thick	Data	Axis	Cursor1	Cursor2
Green	Solid	3	Intervu.Upper Eye Opening	Left	304.9 mV	479.8 mV
Green	Solid	3	Intervu.Lower Eye Opening	Right	-273.0 mV	-429.1 mV
Yellow	Solid	3	Intervu.Upper Eye Opening	Left	266.0 mV	413.5 mV
Yellow	Solid	3	Intervu.Lower Eye Opening	Right	-225.4 mV	-342.8 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 receiver 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	
<i>Fs = 44.1 to 96 kHz, 20 to 20 kHz BW, 1 kHz test tone, 0 dBFS = +24 dBu (unless noted)</i>	
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 USB Units Fs<110 kHz)	+/- 0.5 degrees at 20 kHz
Inter-channel Differential Phase (Between DAC1 USB Units Fs>110 kHz)	+/- 4.1 degrees at 20 kHz
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:

$$\text{Delay} = 1.01 \text{ ms} + (48/\text{Fs})$$

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

Digital Audio Inputs

Number of Digital Inputs (switch selected)	4 (Coaxial, XLR, Optical, USB)
Number of Channels	2
Input Sample Frequency Range	28 to 195 kHz (Coaxial, XLR) 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, XLR)
DC Blocking Capacitors on Digital Inputs	Yes (Coaxial, XLR)
Transient and Over-Voltage Protection on Digital Inputs	Yes
Minimum Digital Input Level	300 mVpp on XLR, 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
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)
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
Output Level Calibration Controls	Shared with Balanced Outputs
Output Level Range (at 0 dBFS) In 'Calibrated' Mode	-6 dBu to +13.5 dBu
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
Factory Preset	2vRMS (8.2 dBu)

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	12 Watts Idle at 120 VAC 16 Watts Maximum at 120 VAC
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 or BNC-to-RCA adapter	9.33" (237 mm)
Width	9.5" (249 mm)
Height	1.725" (44.5 mm)

Weight

DAC1 USB only	3.5 lb.
DAC1 USB with power cord, 3 BNC-to-RCA adapters, extra fuses, and manual	4.5 lb.
Rack mount kit (blank panel, junction block, and rack-mount screws)	0.32 lb.
Shipping weight	7 lb.

Regulatory Compliance

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:

1. XLR Digital Input: Shielded 110-Ohm AES/EBU digital audio cable with connector shell bonded to shield must be used.
2. BNC 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:	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



Warranty Information

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 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. Any DAC1 USB 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 from 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 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 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.

Internal Settings

Removing Top Cover

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

CAUTION: The DAC1 USB 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 USB.**
- **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 USB.**
- **Always connect a static-control wrist strap to the chassis before touching any internal component.**

Jumpers

The following functions are jumper configured:

- 10 dB Headphone Gain Reduction
- Headphone Switch Disable
- XLR Output Pads
- Default Digital Input
- Coaxial Input Termination

A 2-pin jumper plug at header P2 can be moved in order to select a default input which will be selected upon each power-up. A jumper at JP7 can be removed to disable the 75-Ohm termination on the coaxial digital input. Four 8-pin headers (P4, P5, P6, and

P7) allow selection of the output level at the XLR jacks.

CAUTION: Do not remove jumpers at JP3, JP4, JP5, and JP6. The DAC1 USB will not function without these jumpers. Jumpers should never be installed on header P9.

Default Digital Input (P2):

A 2-pin jumper on header P2 (see Photo 1) can be used to choose an input source which will be active upon each power-up.

The 2-pin jumper sets the power-up default input to:

- USB Input *** (Jumper plug between pins 1 and 2 of the header)
- Coaxial Input (Jumper plug between pins 3 and 4 of the header)
- XLR Input (Jumper plug between pins 5 and 6 of the header)
- Optical Input (Jumper plug between pins 7 and 8 of the header)

*** = Factory Default

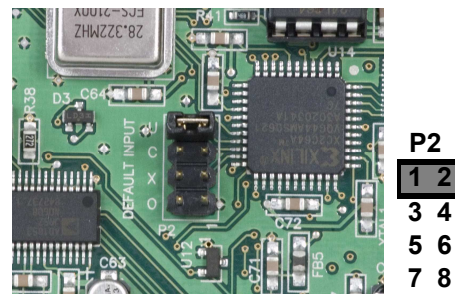


Photo 1 – Default Digital Input (P2)

BNC Input Termination (JP7):

The BNC (Coaxial) input has a 75-Ohm internal termination that may be disabled by removing the 2-pin jumper at JP7 (see Photo 2). This termination is required for normal operation, but may be removed if the user wishes to loop a single coaxial feed through several other pieces of equipment (using a BNC 'T' adapter on the **DAC1 USB**). A 75-Ohm termination must be applied at the last device on the loop, and there should be a combined total of less than 6 feet of cable between the first and last receive device.

JP7 should be configured as follows:

- 75-Ohm Input Impedance*** (Jumper Installed)
- High-Z Input Impedance (Jumper Removed)

*** = Factory Default

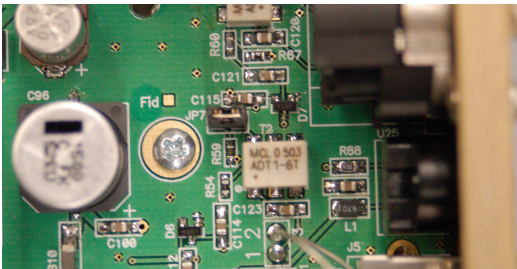


Photo 2 - BNC Input Termination (JP7)

XLR Output Pad Selection (P4, P5, P6, and P7):

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

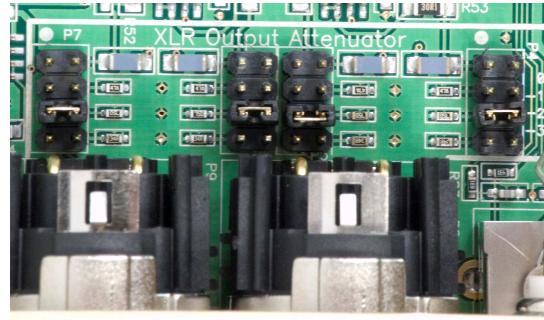


Photo 3 - XLR Output Pad Selection (P4, P5, P6, and P7)

Headphone Switch Disable (JP1 and JP2):

The left headphone jack can be configured so that when a headphone plug is inserted into the jack, the analog outputs are muted. This is convenient when the user wishes to switch between headphones and speakers.

JP1 and JP2 should be configured as follows:

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

*** = Factory Default



Photo 4 – Headphone Switch Disable (JP1 and JP2)

Headphone Gain Reduction (JP8 and JP9):

The gain range of the **HPA2™** can be set using jumpers JP8 and JP9. When jumpers are installed at JP8 and JP9, 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.

JP8 and JP9 are factory installed to reduce the headphone output by 10 dB. This setting is best for most applications. Remove the jumpers if you need more gain.

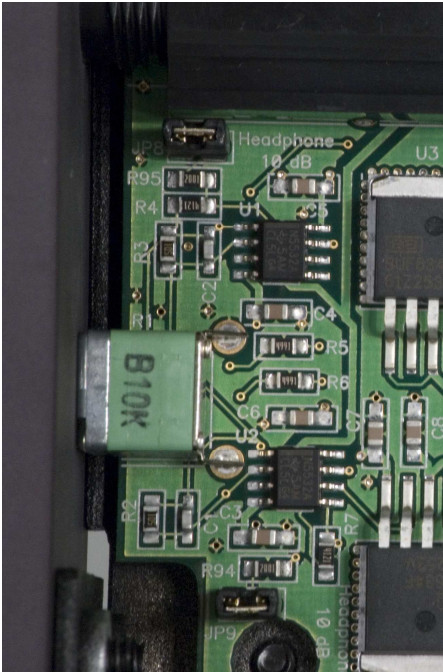


Photo 4 – Headphone Gain Reduction (JP8 and JP9)

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