Benchmark ADC1 USB
Instruction Manual

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

...the measure of excellence!™

Contents

Overview

The ADC1 USB is a reference-quality, 2-channel, 192-kHz, 24-bit, analog-to-digital converter featuring Benchmark's Advanced USB Audio™ technology, phase-accurate UltraLock™ clock system with multi-function clock input and word clock output, and Benchmark’s 9-segment dual-range digital LED metering. The ADC1 USB is designed for maximum transparency and is well suited for the most demanding applications in studios and mastering facilities. A rugged and compact half-wide 1 RU enclosure also makes the ADC1 USB an excellent choice for location recording, broadcast facilities, and mobile rigs. The internal power supply supports all international voltages and has generous margins for over and under voltage conditions.

The ADC1 USB achieves outstanding performance over a wide range of input levels. Each channel has a 41-detent variable gain control, a 10-turn gain calibration trimmer, and a 3-position first-stage gain switch (0, 10, and 20 dB). The gain calibration controls may be used to calibrate the ADC1 USB to precise studio reference levels. The variable gain controls may be used to optimize the gain-staging between a microphone preamplifier and the ADC1 USB. Both the pot and the trimmer have a 24 dB adjustment range. In combination with the first-stage gain switch, these controls provide exceptional SNR and THD+N performance over a 44 dB adjustment range. Each channel has a two-position toggle switch that selects between the variable and calibrated gain control.

The ADC1 USB has a total of 5 stereo digital outputs (1 XLR, 2 coaxial, 1 optical and 1 USB). The digital outputs can operate simultaneously at up to three independent sample rates. The auxiliary coaxial output can be configured for 16-bit TPDF dithered output. This unique flexibility enables simultaneous recording to a CDR, a high-resolution digital recorder, and a high resolution DAW. For example the CDR may operate at 44.1/16, while the DAW operates at 88.2/24, while the digital recorder operates at 192/24. The optical output supports AES or ADAT formats at resolutions up to 192/24. In ADAT mode, high sample rates are supported using SMUX2 and SMUX4. Backup and/or demo recordings can be created with ease while high-resolution outputs are fed to primary recording devices.

The ADC1 USB has a BNC Word Clock output that follows the sample rate of the Main Outputs. Word Clock output is active in all modes of operation.

A multi-format clock-input automatically recognizes AES/EBU, SPDIF, Word Clock, or Super Clock signals. This clock input is used to synchronize the Main Outputs. If desired, the Main Outputs may be driven from internal sources. The ADC1 USB will automatically revert to an internal clock source when the external clock is lost.

The ADC1 USB has two clock modes: External (Ext) and Internal. Both modes support 44.1, 48, 88.2, 96, 176.4 and 192 kHz.

The Ext mode allows the ADC1 USB to lock to an external clock reference. In Ext mode, the ADC1 USB will follow changes in sample rate, and/or changes in the type of reference signal (AES, SPDIF, word clock, or super clock).

When a clock reference is not available, the internal mode must be used, and a sample-rate must be selected (44.1, 48, 88.2, 96, 176.4, or 192 kHz). When the Internal mode is active, the ADC1 USB is acting as clock master, operating at the selected sample rate, and any signal at the clock reference input will be ignored. If internal mode is used, all devices connected to the ADC1 USB digital outputs will need to be configured to lock to the ADC1 USB. The clock output on the back of the ADC1 USB can be used if the connected device(s) require word clock.

The Benchmark UltraLock™ system is 100% jitter immune. The A/D conversion clock is totally isolated from the AES/EBU, SPDIF, ADAT, WC, and super clock interfaces. This topology 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. A poor-quality reference clock will not degrade the jitter performance of the ADC1 USB. In addition, the AES/EBU receiver IC used for AES clock reference has been selected for its ability to decode signals in the presence of very high levels of jitter. The Benchmark UltraLock system delivers consistent performance under all operating conditions.

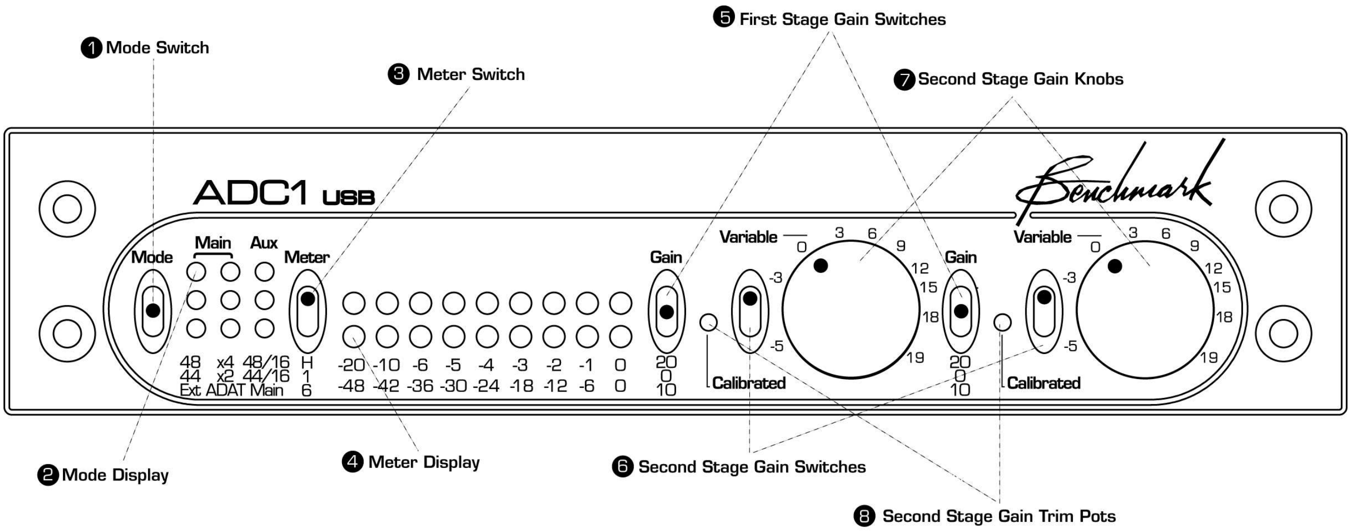
The ADC1 USB is designed to perform gracefully in the presence of errors and interruptions at the clock reference input. The ADC1 USB will even lock to an AES/EBU signal that has its sample-rate bit status set incorrectly since the sample rate is determined by measuring the incoming signal. Lack of sample-rate status bits or incorrectly set status bits will not cause loss of audio.

The ADC1 USB is phase-accurate between channels and between other ADC1 boxes when locked to AES/EBU or word clock reference signals. The word clock output from one ADC1 USB may be connected to the clock input on another ADC1 USB to expand the number of phase-accurate conversion channels.

Features

- Two channels of analog-to-digital conversion
- Two XLR balanced analog inputs providing high-performance over a 44 dB range
- -15 dBu to +29 dBu input sensitivity range (at 0 dBFS)
- First-stage gain switches (1 per channel): 0 dB, 10 dB and 20 dB
- 41-detent variable gain controls (1 per channel) with a 24 dB gain range
- 10-turn gain calibration controls (1 per channel) with a 24 dB gain range
- 9-segment dual-range digital LED meters
- Sample rate LED indicators for ‘Main’ and ‘Aux’ digital outputs
- Sample rate for USB output is determined by computer (D.A.W./software)
- Conversion at 44.1, 48, 88.2, 96, 176.4, and 192 kHz
- USB output resolution 24 bits at 44.1, 48, 88.2, or 96 kHz
- Versatile external and internal clock modes
- Multifunction clock input with auto-recognition of AES, SPDIF, Word Clock, or Super Clock
- Word Clock output
- Total jitter immunity with Benchmark’s, phase-accurate *UltraLock*™ technology
- Simultaneous output of up to three different sample rates
- Simultaneous dithered 16 and 24-bit outputs
- Five digital outputs (1 XLR, 2 Coax, 1 Optical, 1 USB)
- AES/EBU, ADAT, and ADAT S/MUX2, and ADAT S/MUX4 output formats
- THD+N = -104 dB, 0.00063% @ -3 dBFS input, SNR 121 dB A-weighted
- Reliable and consistent performance under all operating conditions
- Internal 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 interference
- Meets FCC Class B and CE emissions requirements

Front Panel Descriptions



1) Mode Switch

The Mode Switch is a non-locking toggle switch which controls the conversion modes of the ADC1 USB.

Mode Switch Operation

The ADC1 USB can be programmed to function in a variety of conversion modes, including simultaneous multiple sample rates, bit depths, and output formats using internal and/or external clock sources.

The USB output sample rate is always controlled by the computer with which it is operating. This is typically set in the recording software (DAW), but it is sometimes set within the audio-utility control panel of the operating systems (e.g. 'AudioMIDI Setup' in Mac OS X).

The conversion modes of the other four outputs (XLR, optical, main coax, and auxiliary coax) are set with the Mode Switch by using press-and-release or press-and-hold actions. These functions are described in the "Conversion Mode Programming" section.

Conversion Mode Programming

- Press-and-release the Mode Switch up to cycle through the clock source and sample rate options for the Main Outputs.
- Press-and-release the Mode Switch down to cycle through the sample rate and bit depth options for the Aux Output.
- Press-and-hold the Mode Switch down for approximately 3 seconds to switch the optical output mode between AES/EBU and ADAT.
- Press-and-hold the Mode Switch up for approximately 3 seconds to reset the ADC1 USB to Factory Default settings.

Factory Default Settings

The ADC1 USB can be easily reset to factory default settings by pressing and holding the Mode Switch up for approximately 3 seconds. The factory default setting syncs the main and aux outputs to the external clock and configures the optical output in AES/EBU format.

2) Mode Display

The Mode Display indicates the clock source, sample rate, Aux output mode, and optical mode (ADAT, etc).

The Mode Display indicates the operating mode of the digital XLR, optical, and coax outputs. It does not display the sample rate of the USB output. The USB sample rate is set within the software of the host computer.

Output Programming

The Main Outputs can be set to operate at a fixed sample rate using the internal clock source or they can be set to follow and lock to an external clock source.

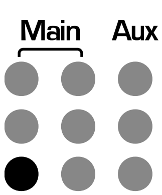
The Aux Output can be set to typical DAT or CD-writer resolutions (TPDF-dithered; 16-bit; 44.1 or 48 kHz), or it can be set to mirror the Main Output (bit-for-bit).

The Optical Output can be set to AES/EBU mode or ADAT mode.

The USB Output is synchronized to the computer host. The sample-rate is determined by the recording software.

External Clock Source Locking

The ADC1 USB can sync to a variety of external clock sources, including Word Clock, Super Clock, AES, and SPDIF. Once the ADC1 USB is locked to the external clock, it will synchronize the Main Outputs (and Aux if selected) to the sample rate of the Ext Ref Input (external reference clock).



Off = Internal Sync

On = Locked to External Sync

Flashing = External Sync is Selected, Not Locked

The bottom left LED in the Mode Display is the Ext Indicator. If the Ext LED is off, then the ADC1 USB is set to operate at a fixed sample rate using the internal clock source.

If this LED is illuminated then the ADC1 USB is locked to an external clock source. When locked, the Mode Display will indicate the sample rate.

If the reference sample rate is changed, the ADC1 USB will automatically switch sample rates to match the change in the reference sample rate. If the Ext LED is flashing, the ADC1 USB is set to sync to an external clock source and is locking to that source. The ADC1 USB should lock in less than 5 seconds. If a lock doesn't occur within a 5 second window, there may be a problem with the external reference clock. If a sync error occurs, verify a secure connection is present between the ADC1 USB and the external clock source. The ADC1 USB is very tolerant of low-level low-quality reference signals.

External Clock Source Synchronization

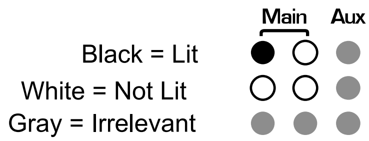
- Press up repeatedly on the Mode Switch and cycle through the Main Output modes until the lower left Ext LED is either on or flashing.

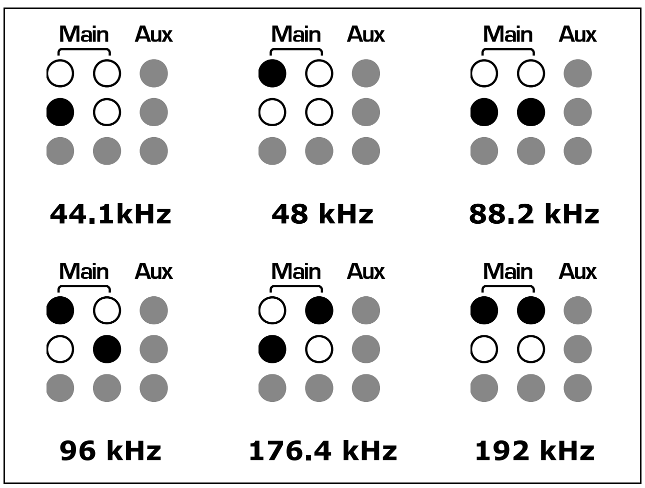
Fixed Sample-Rate Conversion

The ADC1 USB can be programmed to convert at a fixed sample rate using an internal clock source. The following sample rate frequencies are available: 44.1, 48, 88.2, 96, 176.4, and 192 kHz. The Ext Ref Input is ignored when the internal clock source is selected.

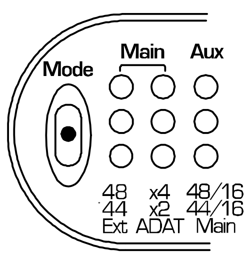
Setting the Fixed Sample Rate on the Main Outputs

- Press-and-release the Mode Switch up repeatedly to cycle through the available sample rates. The four LED's in the upper left of the Mode Display indicate the sample rate, as illustrated in the diagrams below.





Sample Rate Display



Main Output

Column one of the Mode Display has a '48' LED and a '44' LED. These indicate sample rates of 48 kHz and 44.1 kHz respectively. Column two has an 'x4' LED and an 'x2' LED. These indicate 4x or 2x multipliers. Multiply the sample rate shown in column one by the multiplier shown in column two. For example, if the '44' and 'x2' LED's are on, the sample rate is 88.2 kHz (44.1 x 2 = 88.2).

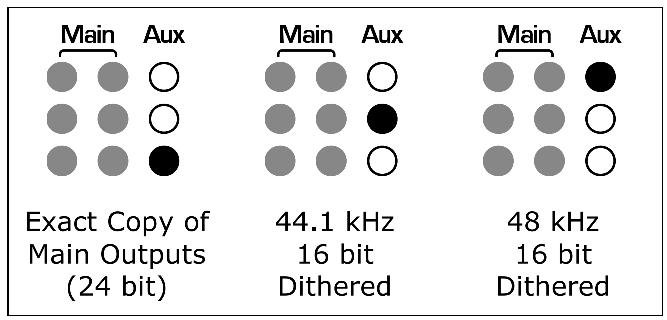
Aux Output

The Aux Output can be programmed to mirror the Main Outputs (bit for bit), or it can provide a lower-resolution copy of the converted signal at an independent sample rate. Column three of the Mode Display displays the Aux Output mode setting.

NOTE: The Aux Output setting does not affect the Main Outputs in any way.

Aux Output Programming

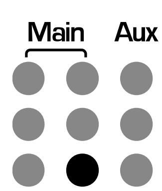
Press-and-release the Mode Switch repeatedly to cycle through the Aux Output mode settings. The right-hand column of LED's in the Mode Display indicates the Aux Output mode as illustrated in the following diagram:



ADAT or AES/EBU Selection for the Optical Output

The Optical Output (one of the three Main Outputs) can provide either AES/EBU or ADAT format. The AES/EBU mode works with most S/PDIF optical inputs.

Press and hold the Mode Switch down to change the Optical Output mode. The bottom-center LED indicates the optical mode as illustrated in the following diagram:



Off = AES/EBU on Optical Output

On = ADAT on Optical Output

When ADAT is active, S/MUX is automatically enabled at all 2X and 4X sample rates (88.2 kHz, 96 kHz, 176.4 kHz, and 192 kHz).

3) Meter Switch

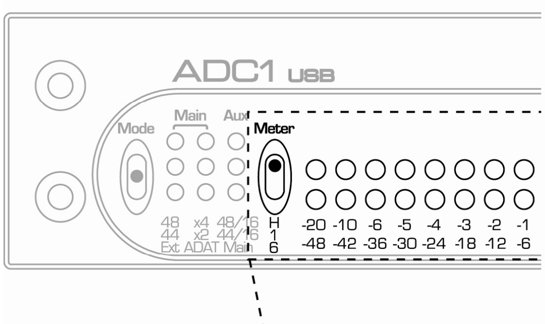
The Meter Switch selects the scale and mode of the 9-segment LED Meter Display.

Meter Scale and Peak Hold Function Selection

- Set the Meter Switch to "H" (top) to enable the Peak Hold function and set the scale to a 1 dB step.
- Set the Meter Switch to "1" (center) to disable the Peak Hold function and set the scale to 1 dB step.
- Set the Meter Switch to "6" (down) to disable the Peak Hold function and set the scale to a 6 dB step.

4) Meter Display

The ADC1 USB is equipped with a multi-function 9-segment LED meter. The Meter Switch selects either a 6 dB step or 1 dB step scale and controls the peak-hold function. Metering is fully digital and post-conversion for absolute accuracy. The units are dBFS (dB below the level of a full-scale sine wave, or more simply, dB below digital clip).



Meter Switch and Meters

Time constants are built into the meters so all transient peaks can be observed easily. If a transient peak has a duration as short as one digital sample, an LED will be illuminated and remain illuminated long enough to be observed by the human eye.

A peak indication mimics the action of the needle on a peak-reading analog meter, while the remaining LED's will follow the instantaneous level of the audio.

The red 0 LED indicates that a full-scale digital code has been reached and that digital clipping has occurred. Full-scale events as short as one digital sample, will light the 0 LED. Short single-sample digital clipping events are often audible, and all 0 dBFS events should be avoided.

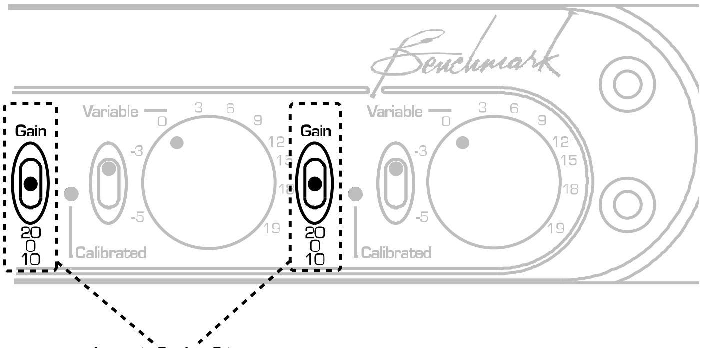
The ADC1 USB has a very large dynamic range (especially when operating at 24-bit output word lengths). It is wise to use some of this dynamic range to provide more headroom as insurance against clipping. Leave some extra headroom between your highest anticipated peak and the red 0 dBFS LED.

Input Gain Adjustment

Each channel on the ADC1 USB is equipped with a 3-position first-stage gain switch. The first gain-stage provides exceptional noise performance at gains of 0 dB, +10 dB, or +20 dB. This stage is followed by a second-stage having a continuously variable gain range of -5 dB to +19 dB. This gain structure provides ultra-high performance at any gain setting between -5 dB and +39 dB. The higher gain settings will allow direct connections from many electronic instruments and instruments with pickups (no DI box required).

At a gain setting of 0 dB, the converter will reach full-scale (0 dBFS) with a +24 dBu input level. The calibrated gain trim-pots are factory calibrated for 0 dB second-stage gain.

5) First-Stage Gain Switch



The First-Stage Gain Switches can apply analog gain (0, 10, or 20 dB) to the left and/or right channel. Both channels have their own dedicated switches. The First-Stage Gain can be used in 'Calibrated' or 'Variable' mode.

To select the first-stage gain

- Set the switch to "0" (center) to select 0 dB (unity) gain for the first-stage.
- Set the switch to "10" (down) to select 10 dB gain for the first-stage.
- Set the switch to "20" (up) to select 20 dB gain for the first-stage.

6) Second-Stage Gain Switch

The Second-Stage Gain Switch determines the type of second-stage gain control. The switches (one for each channel) can be set to either 'Variable' or 'Calibrated'. When the switch is set to 'Variable', the second-stage gain for that respective channel is determined by the Second-Stage Gain Knob. Conversely, when the switch is set to 'Calibrated', the Second-Stage Calibration Trimmer adjusts the gain for finely-tuned, calibrated operation.

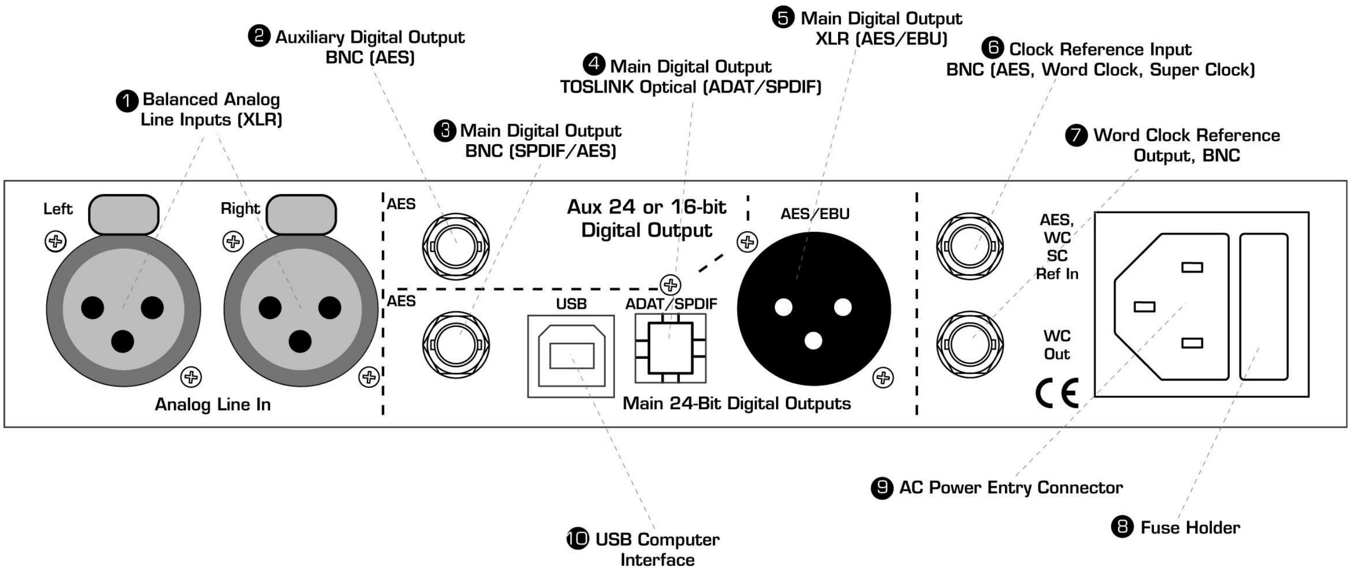
7) Second-Stage Gain Dial

The Second-Stage Gain Dial is a 41-detent gain control with a usable range of approximately -5 dB to +19 dB. In order to use this gain control for a channel, that channel's respective Second-Stage Gain Switch must be set to Variable (up).

8) Second-Stage Calibration Trimmer

The Second-Stage Calibration Trimmers are 10-turn gain trim pots with a useable range of approximately -5 dB to +19 dB. In order to use the trim pots the Second-Stage Gain Switch must be set to Calibrated (down).

Back Panel Descriptions



Digital Outputs

- (#5 above) XLR, 110-Ω, balanced, AES/EBU professional format, 24-bits
- (#3) Main BNC, 75-Ω, un-balanced, AES/EBU professional format, 24-bit compatible with most SPDIF inputs
- (#4) Optical TOSLINK, multi-format (AES professional, ADAT, ADAT S/MUX II & IV), 24-bits
- (#10) USB Computer Interface
- (#2) Aux BNC, 75-Ω, un-balanced, AES/EBU professional format, 24-bit or 16-bit, compatible with most SPDIF inputs

The Main and Aux outputs are controlled by the front-panel Mode Switch. The status of these outputs is shown in the Mode Display adjacent to the Mode Switch.

The Main Outputs always operate at 24-bits. The coaxial, optical, and XLR Main Outputs may be synchronized to an external clock reference or may be controlled by the internal clock. The USB Main Output is always synchronized to the host (computer). The Optical Output has two modes of operation; AES/EBU and ADAT. The ADAT mode supports ADAT (44.1 and 48 kHz), ADAT S/MUX^2 (88.2 and 96 kHz), and ADAT S/MUX^4 (176.4 and 192 kHz).

The Aux Output can operate asynchronously at 44.1 or 48 kHz with a TPDF-dithered 16-bit word length. The Aux Output is provided as a convenience for making safety backups or demo recordings to low-resolution 16-bit recorders (i.e. CDR or DAT). If this low-resolution function is not needed, the Aux Output can be set to mirror the high-resolution Main Outputs.

1) Balanced Analog Line Inputs

'Left' and 'Right' balanced inputs use locking Neutrik™ gold-pin female XLR jacks. These inputs have a wide operating range. The input sensitivity (at 0 dBFS) ranges from -15 dBu (at maximum gain) to +29 dBu (at minimum gain). The input impedance is 200k Ohms balanced, and 100k Ohms unbalanced. The high impedance and sensitivity allow direct connections from many musical instruments (adapter cable required). Direct connection of piezo pickups is not recommended as these pickups require higher input impedances (to prevent low-frequency roll-off problems).

- XLR pin 2 = + Audio In
- XLR pin 3 = - Audio In
- XLR pin 1 = Cable Shield (grounded directly to the chassis to prevent internal ground loops)

Unbalanced Source Adaptation

1. Connect "+" or hot (tip on ¼ phone plug, center pin on RCA plug) to XLR pin 2.
2. Connect ground (sleeve on ¼" phone plug, case on RCA plug) to XLR pins 3 and 1.

NOTE: It's best to used balanced wiring ("+", "-", "shield") and to tie the "-"and "shield" at the unbalanced connector (source).

2) Aux BNC Output

The Aux Output (BNC) has a unique feature that allows it to output at a sample-rate and bit-depth that is independent from the Main Output. Regardless of the sample-rate of the Main Output, the Aux output can convert at 44.1 or 48-kHz, 16-bit (TPDF dithered), for simultaneous use with backup recording devices (CDR, DAT, etc). The Aux Output can also be configured to operate as an additional Main Output with identical data as other Main digital outputs.

- Data Format = AES/EBU professional
- Word Length = 16 bits TPDF dithered, or 24 bits
- Sample Rate = 44.1 or 48 at 16 bits; 44.1, 48, 88.2, 96, 176.4, or 192 kHz at 24-bits
- Clock Source = Internal at 16 bits, internal or external at 24 bits

3) Main BNC Output

The digital data output from the Main BNC Output is identical to that of the Main XLR Digital Output.

- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

BNC Main and Aux Outputs

The two BNC coaxial digital outputs use female BNC connectors. These connectors are securely mounted directly to the rear panel. These are 1 Vpp unbalanced outputs with 75-Ω source impedances. Outputs are DC-isolated, transformer-coupled, current-limited, and diode-protected. 75-Ω coaxial cables should be used when connecting these outputs to other devices. The use of 50-Ω coax is not recommended and may cause data transmission errors.

Consumer-style, SPDIF RCA digital interfaces are common on recording devices. The ADC1 USB ships with BNC-to-RCA adapters to allow easy interfacing with consumer-style digital interfaces. BNC to RCA coaxial cables are also available from Benchmark.

BNC connectors are specified by the AES3-id and SMPTE 276M standards for 75-Ω 1 Vpp digital audio signals and are commonly used in video production facilities and other professional audio applications. RCA connectors are specified by IEC 609588-3 for 75-Ω 0.5 Vpp consumer-format digital audio signals (commonly known as SPDIF). We have chosen to comply with the professional standards because the BNC connectors lock and are generally more reliable than RCA connectors. Compliance with the 1 Vpp digital audio standards increases the reliability of digital connections, and often allows increased transmission distances.

4) TOSLINK Optical Output

The Optical Output has four modes of operation; AES/EBU, ADAT, ADAT S/MUX^2, and ADAT S/MUX^4. The AES/EBU mode is compatible with most SPDIF optical inputs. The ADAT LED on the front panel is illuminated whenever any of the ADAT Modes are active. S/MUX^2 and S/MUX^4 are automatically enabled if required to support the selected sample rate. S/MUX^2 is active at 88.2 or 96 kHz, S/MUX^4 is active at 176.4 or 192 kHz.

The Optical Output uses what is often called a TOSLINK, Type FO5, or 5 mm optical connector. The ADC1 USB uses a special high-bandwidth version that supports AES/EBU digital audio at sample rates up to 192 kHz. Please note that many optical inputs cannot support AES/EBU or SPDIF digital audio at sample rates above 48 kHz, others are limited to 96 kHz. A few products (such as the Benchmark DAC1) support 192 kHz optical inputs. Please note that high-bandwidth optical transmitters and receivers are not required for ADAT, ADAT S/MUX^2, or even ADAT S/MUX^4.

AES/EBU Optical Output Mode

- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

ADAT Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 44.1 or 48 kHz
- Clock Source = Internal or external
- ADAT channel assignments: 1 = Left, 2 = Right, 3-8 = muted

ADAT S/MUX^2 Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 88.2 or 96 kHz
- Clock Source = Internal or external
- ADAT channel assignments *: 1 = Left a, 2 = Left b, 3 = Right a, 4 = Right b, 5-8 = muted
- * a, and b are successive samples

ADAT S/MUX^4 Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 176.4 or 192 kHz
- Clock Source = Internal or external
- ADAT channel assignments **: 1 = Left a, 2 = Left b, 3 = Left c, 4 = Left d, 5 = Right a, 6 = Right b, 7 = Right c, 8 = Right d
- ** a, b, c, and d are successive samples

5) AES/EBU XLR Output

The balanced AES/EBU XLR output uses a gold-pin Neutrik™ male XLR connector. It has an output impedance of 110-Ω as well as DC-isolation, transformer-coupling, current-limiting, and diode-protection. It's designed to drive standard 4 Vpp AES signals into a 110-Ω load. 110-Ω digital cables are recommended when connecting this output to other devices because analog audio cables may cause data transmission errors.

- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rate = 44.1, 48, 88.2, 96, 176.4, or 192 kHz
- Clock Source = Internal or external

6) Clock Reference Input

The Clock Reference Input auto-detects AES/EBU, SPDIF, Word Clock, or Super Clock signals, and automatically follows changes in sample-rate. When Ext mode is active the ADC1 USB will lock to the external clock source. Benchmark's UltraLock™ circuitry isolates the conversion clock from any jitter present on the clock reference. Ext Mode will not degrade the conversion quality of the ADC1 USB even when very high levels of jitter are present on the clock reference.

7) Word Clock Reference Output

This output provides a Word Clock signal which can be passed to downstream components.

8) 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 ADC1 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.

9) AC Power Entry Connector

The AC power input uses a standard IEC type connector. Within the USA and Canada, the ADC1 USB ships with a power cord. In other locations, a location-specific IEC style power cord may be purchased from a local source (including a local Benchmark dealer).

10) USB Computer Interface Output

The USB output supports 44.1, 48, 88.2 and 96 kHz sample rates at 24-bits.

The sample rate of the USB output is independent from the sample rate of the other outputs on the ADC1 USB. For example, the USB output can operate at 96 kHz while the main output is operating and 192 kHz and the auxiliary output is operating at 44.1 kHz. All three outputs are fully independent, and all are equipped with Benchmark's UltraLock™ jitter-attenuation system.

The USB output always acts as a slave. The sample rate of the USB output is always controlled by the computer. The USB cannot be synchronized to the Ext Ref In. The USB receives all of its synchronization signals through the USB port. Benchmark's UltraLock™ system insures that the audio data is free from any jitter-induced artifacts. Performance through the USB port matches the performance through the other digital outputs.

The USB jack on the ADC1 USB accepts a 'B-type' male USB 1.1 or USB 2.0 connector. An 'A-B type' USB cable is provided with the ADC1 USB. The USB cable connects the ADC1 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 interface communicates to the host computer as a 'native' USB audio device and, therefore, does not require the installation of any custom drivers. Benchmark's Advanced USB Audio technology achieves bit-transparent operation without non-native drivers.

The Benchmark USB interface is truly a plug-and-play solution. The ADC1 USB can begin streaming high resolution audio bit-transparently within seconds after being plugged into a computer for the first time.

The ADC1 USB is designed, tested and proven compatible with the following four operating systems:

- Windows Vista
- Windows XP
- Windows 2000
- Mac OS X

Rack Mounting

The ADC1 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

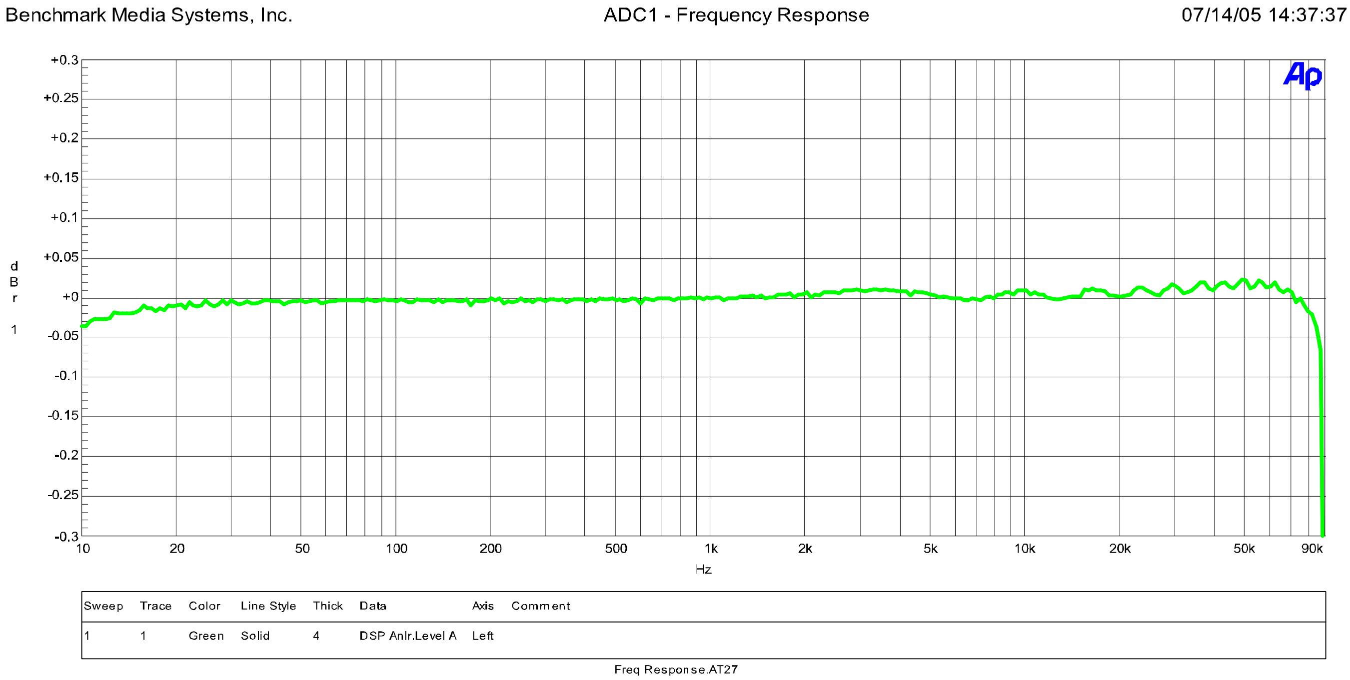
The Rack Mount Coupler is a machined aluminum junction block that joins any two ½-wide System1™ devices for rack mounting. It is also used to join ***a Blank Rack Panel*** to a single ½-wide System1™ device.

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.

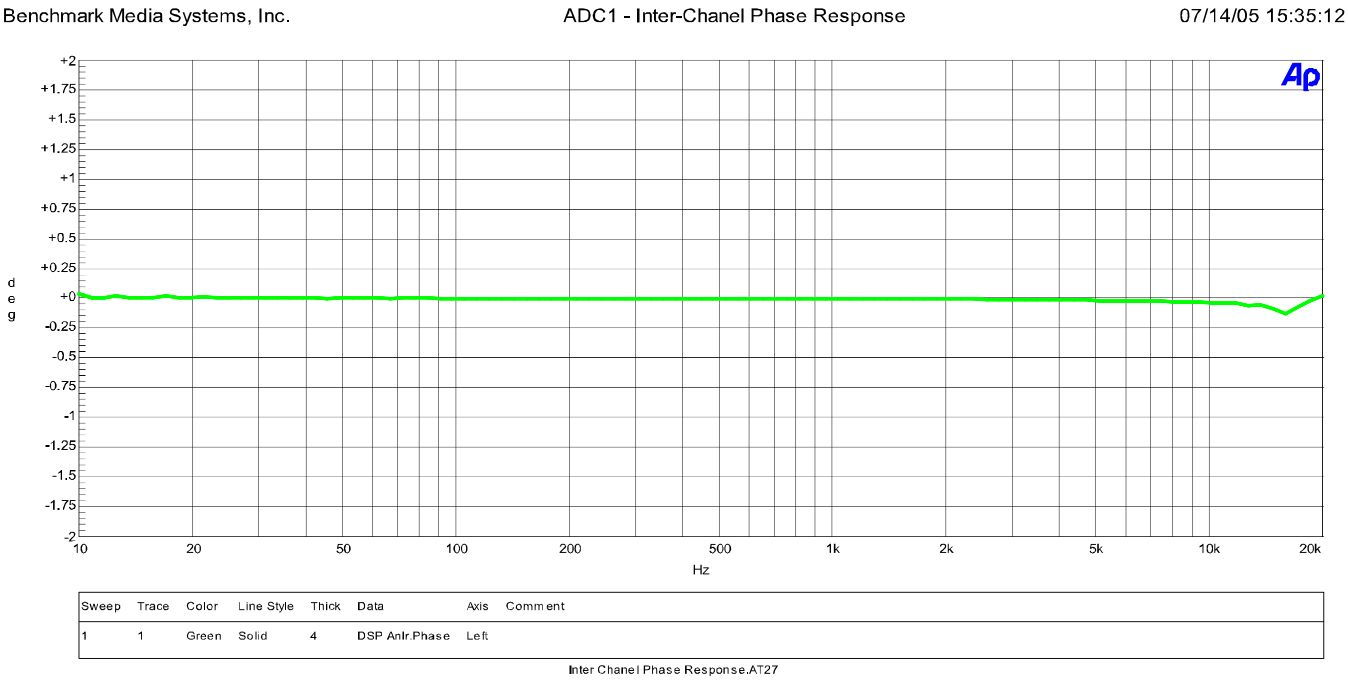
Performance

Frequency Response



The above graphs show the frequency response of the ADC1 USB when it is operating at a 192-kHz sample rate. Note that the amplitude response is down by less than 0.05 dB at 10 Hz and 80 kHz. The bass response extends well below the 10-Hz limitation of the measurement equipment, and the high-frequency analog response extends well above the 96 kHz bandwidth of 192 kHz digital audio.

Inter-Channel Phase Response



This graph shows that the differential phase is significantly better than ± 0.25º from 10 Hz to 20 kHz.

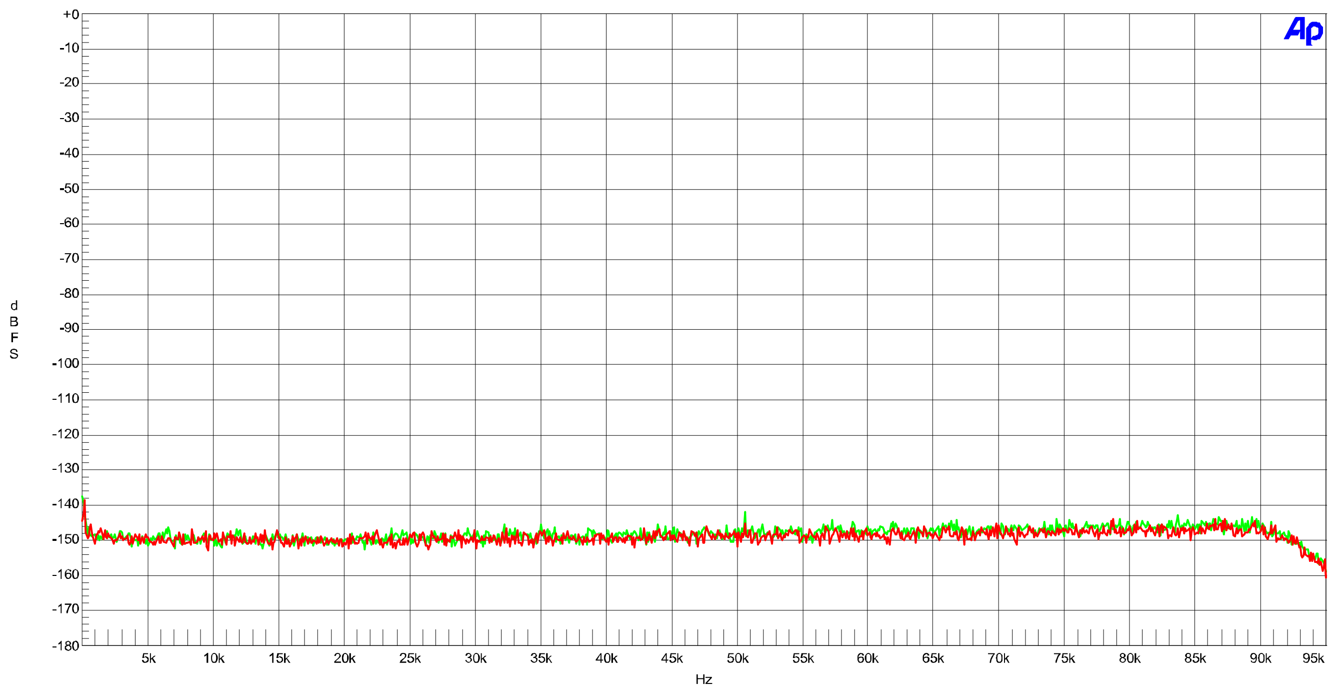
THD+N vs. Level, 1 KHz

w/20 kHz LPF unweighted

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.

32K B-H FFT, Idle Channel Noise

Benchmark Media Systems, Inc. 32K B-H FFT Analysis of Idle Channel Noise 07/14/05 14:11:42



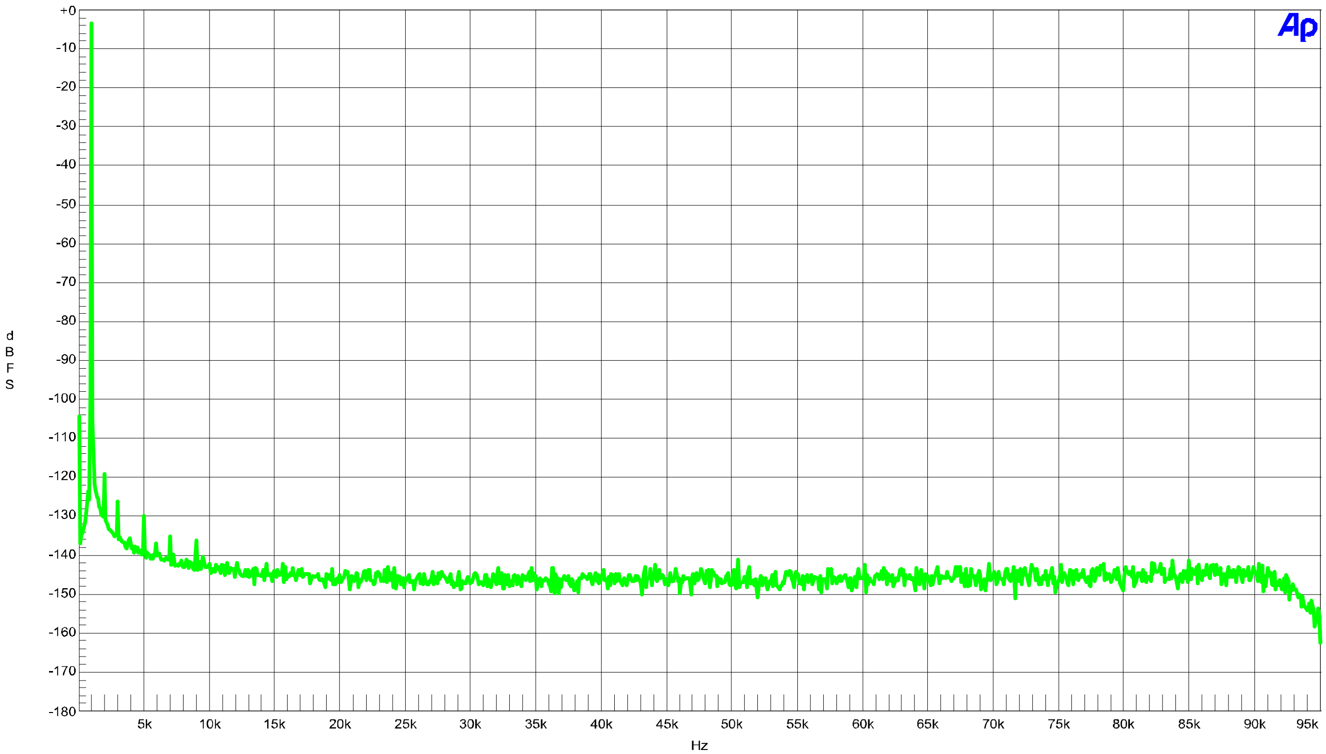
Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	
1	2	Red	Solid	1	Fft.Ch.2 Ampl	Left	

B - FFT Idle Chanel Noise.at27

The above graph demonstrates that the ADC1 USB 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 –135 dBFS.

32K B-H FFT, -3 dBFS, 1 KHz

Benchmark Media Systems, Inc. 32K B-H FFT Analysis of -3 dBFS 1 KHz Test Tone 07/14/05 14:24:24

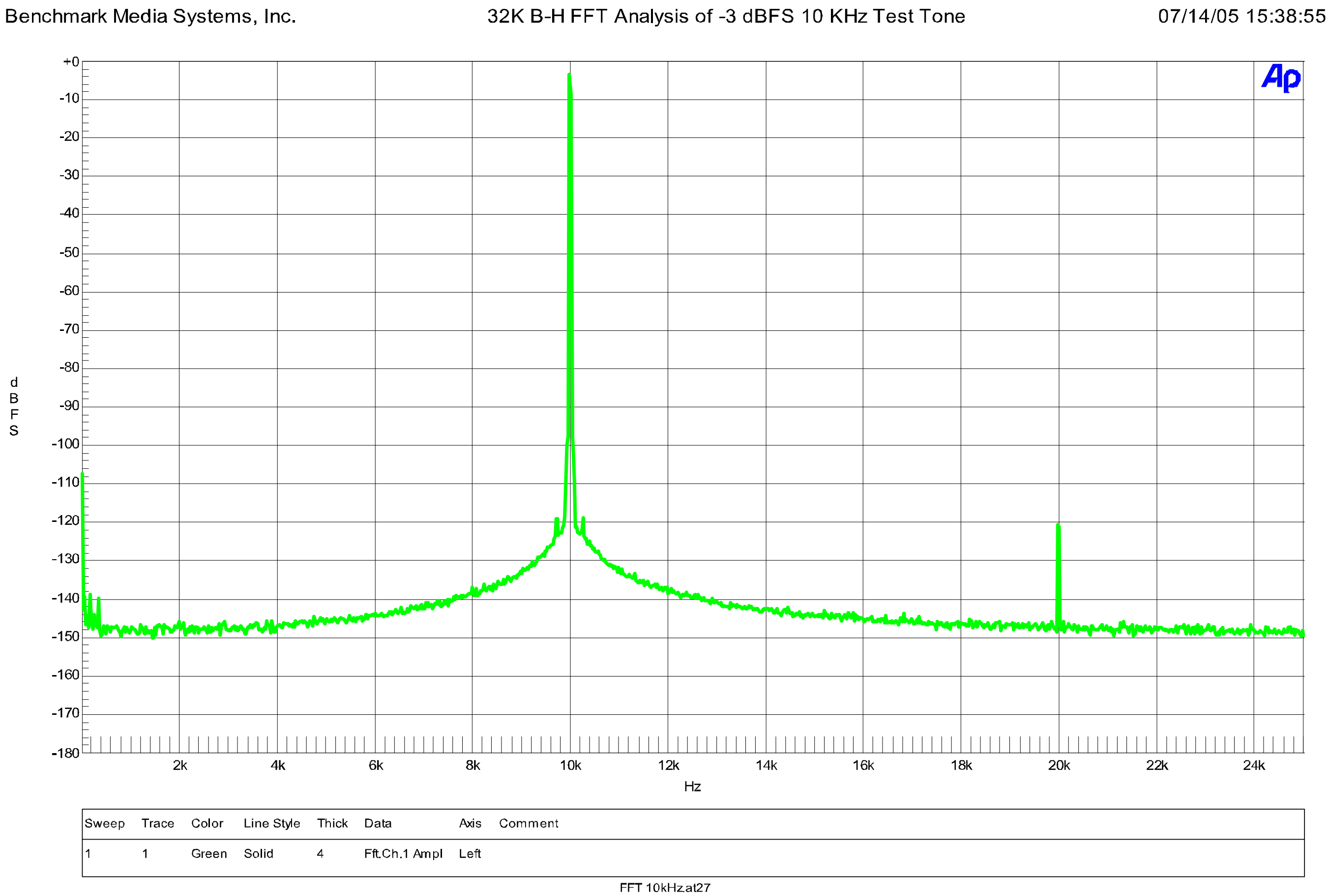


Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	4	Fft.Ch.1 Ampl	Left	

FFT 1kHz.at27

The above FFT plot shows that the ADC1 USB has very little harmonic distortion. Distortion is exceptionally low and is dominated by 2nd harmonic distortion. Note the near absence of spurious tones.

32K B-H FFT, -3 dBFS, 10 KHz



The above FFT plot shows that the ADC1 USB is free from jitter-induced sidebands. Any jitter present at the conversion sampling circuit would produce sidebands equally spaced above and below the 10 kHz test tone. The tone at 20 kHz is due to second harmonic distortion, and measures almost 120 dB below full scale. Note the near absence of spurious tones.

Specifications

Analog Audio Inputs

Number of Inputs (balanced)	2
Connector	Gold-Pin Neutrik™ female XLR
Impedance	200 kΩ
Sensitivity	-15dBu to +29 dBu (at 0 dBFS)

Clock Reference Input

Format	Auto-detect AES/EBU, Word Clock, and Super Clock (256x)
Impedance	75 Ω
Sensitivity	150 mV AES 200 mV Word Clock 750 mV Super Clock
Transformer Coupled	Yes
DC Blocking Capacitors	Yes
Transient and Over-Voltage Protection	Yes
Jitter Attenuation Method	Benchmark UltraLock™

Word Clock Reference Output

Impedance	75 Ω
Level	5 Vpp 2.5 Vpp into 75 Ω
Transformer Coupled	No
DC Blocking Capacitors	No
Transient and Over-Voltage Protection	Yes

Digital Audio Outputs

Number of Digital Outputs	1 XLR Main 1 TOSLINK Main 1 BNC Main 1 BNC Aux 1 USB Computer Interface
Connectors	Gold-Pin Neutrik™ male XLR
Number of Audio Channels	2
Main Output Word Length	24 bits
Main Output Sample Frequencies	44.1, 48, 88.2, 96, 176.4, or 192 kHz
Aux Output Word Length	16 or 24 bits
Aux Output Sample Frequencies	44.1, 48, 88.2, 96, 176.4, or 192 kHz (at 24 bits) 44.1 or 48 (at 16 bits)
Impedance	110 Ω XLR 75 Ω BNC
Level	4 Vpp into 110 Ω XLR 1 Vpp into 75 Ω BNC
Transformer Coupled	Yes
DC Blocking Capacitors	Yes
Transient and Over-Voltage Protection	Yes

Audio Performance

Fs = 44.1 to 192 kHz, 20 to 20 kHz BW, 1 kHz test tone, 0 dBFS = +24 dBu (unless noted)

SNR – A-Weighted, 0 dBFS = +8 to +29 dBu	121 dB
SNR – Unweighted, 0 dBFS = +8 to +29 dBu	119 dB
SNR – A-Weighted at max gain, 0 dBFS = -14 dBu	108 dB
THD+N, 1 kHz at –1 dBFS	-102 dBFS, -101 dB, 0.00089%
THD+N, 1 kHz at –3 dBFS	-107 dBFS, -104 dB, 0.00063%
THD+N, 20 to 20 kHz test tone at –3 dBFS	-106 dBFS, -103 dB, 0.00071%
Frequency Response at Fs=192 kHz	-3 dB, +0 dB, 2 Hz to 92 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.18 dB at 88 kHz -3 dB at 92 kHz -100 dB at 108 kHz
Frequency Response at Fs=96 kHz	-3 dB, +0 dB,1 Hz to 46 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.10 dB at 44 kHz -3 dB at 46 kHz -108 dB at 54 kHz
Frequency Response at Fs=48 kHz	3 dB, +0 dB, 1 Hz to 23 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz -0.00 dB at 20 kHz -0.10 dB at 22 kHz -3 dB at 23 kHz, -110 dB at 27 kHz
Passband Ripple	+/- 0.008 dB
Crosstalk	-105 dB at 20 kHz -130 dB at 1 kHz -200 dB at 20 Hz

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
Maximum Amplitude of Jitter Induced Sidebands	< -134 dB (measurement limit) (10 kHz 0 dBFS test tone, 12.75 UI sinusoidal jitter at 1 kHz)
Maximum Amplitude of Spurious Tones with 0 dBFS test signal	-130 dBFS
Maximum Amplitude of Idle Tones	-145 dBFS
Maximum Amplitude of AC line related Hum & Noise	-130 dBFS
Interchannel Differential Phase (Stereo Pair)	+/- 0.5 degrees at 20 kHz
Interchannel Differential Phase (Between ADC1 USB Units)	+/- 0.5 degrees at 20 kHz
Maximum Lock Time after Fs change	< 1 s for frequency lock < 5 s for phase lock
Mute on Sample Rate Change	Yes
Mute on Loss of External Clock	No
Mute on Lock Error	No
Mute on Receive Error	No
Soft Mute Ramp Up/Down Time	10 ms

Group Delay (Latency)

Delay (Analog Input to Digital Output)	1.20 ms at 44.1 kHz 1.09 ms at 48 kHz 0.75 ms at 88.2 kHz 0.67 ms at 96 kHz 0.63 ms at 176.4 kHz 0.59 ms at 192 kHz

LED Status Indicators

LED Location	Front Panel
Mode Indicators	9 green
Meter	14 green, 2 yellow, 2 red

AC Power Requirements

Input Operating Voltage Range (VAC RMS)	110 V setting – 95 V min, 140 V max 220 V setting – 190 V min, 285 V max
Frequency	50-60 Hz
Power	16 Watts Idle 16 Watts Typical Program 20 Watts Maximum
Fuses	5 x 20 mm (2 required) 110 V setting – 0.5 A 250 V Slo-Blo® Type 220 V setting – 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

ADC1 USB only	3.6 lb.
ADC1 USB with power cord, 3 BNC-to-RCA adapters, extra fuses, and manual	4.9 lb.
Rack mount kit (blank panel, junction block, and rack-mount screws)	0.32 lb.
Shipping weight	7 lb.

ADAT S/MUX

Proper S/MUX Identification

S/MUX^2 allows the recording of 4 channels at 88.2 or 96 kHz using a standard 8-channel 44.1 or 48 kHz ADAT recorder. S/MUX^4 allows the recording of 2 channels at 176.4 or 192 kHz using a standard 8-channel 44.1 or 48 kHz ADAT recorder. In either case it is important to identify S/MUX recordings so that they can be properly decoded upon playback. Failure to properly decode an S/MUX recording will add unwanted artifacts to the audio. The severity of these artifacts is a function of the high-frequency content of the original digital audio signal, and may range from inaudible to very objectionable. This variation in severity can make it difficult to accurately spot a problem just by listening to a portion of the recording.

An ADAT S/MUX^2 recording will have pairs of nearly identical tracks (1≈2, 3≈4, 5≈6, and 7≈8). Unfortunately this can be mistaken for 4 stereo pairs at half of the original sample rate. There is no substitute for proper labeling. This labeling should include the sample rate of the recording.

An ADAT S/MUX^4 recording is somewhat easier to identify because it will have groups of 4 channels that are nearly identical (1≈2≈3≈4, and 5≈6≈7≈8). In error, S/MUX^4 could be played at ¼ of its original sample rate, and sound almost normal. S/MUX^4 could also be mistaken for S/MUX^2 and could be played at ½ of its original sample rate. Please note that these changes in sample rate will not alter the pitch of the audio but will introduce errors, and these errors may not be discovered until it’s too late to correct them.

S/MUX must be decoded before Digital Processing

No DSP process should be applied to an S/MUX signal before it is decoded. S/MUX must be decoded before it reaches the internal processing in a DAW or a digital console. Many such devices include S/MUX decoders at their digital interfaces and these decoders must be properly enabled for S/MUX and disabled for standard ADAT inputs.

Sample Rate Controls S/MUX

Most devices (including the ADC1 USB) automatically enable and disable S/MUX in response to changes in sample rate. Therefore it is essential that all S/MUX equipped A/D converters, D/A converters, digital consoles, digital audio workstations, and digital processing devices be set to identical sample rates. There is one exception to this rule: A non-S/MUX ADAT recorder can be connected to an S/MUX interface, but the recorder must be set at ½ (S/MUX^2) or ¼ (S/MUX^4) of the actual sample rate.

S/MUX is not used for Sample Rate Conversion

If two devices are connected with an ADAT S/MUX interface and the devices are set to different sample rates, a crude form of sample rate conversion will occur. For example, if an A/D converter is set to 96 kHz, and it feeds a digital console that is set to 48 kHz, the system will appear to down convert from 96 kHz to 48 kHz. This would be a useful feature if the digital filtering was correct. The problem is that this ad-hoc sample rate converter is lacking the low-pass filter that prevents aliasing.

UltraLock™ ... What Is It?

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 level required for accurate 24-bit conversion. Fortunately, this interface jitter has absolutely no effect on the audio *unless* it influences the conversion clock in an analog-to-digital converter (ADC) or in an analog-to-digital converter (DAC).

Many converters use a single-stage Phase Lock Loop (PLL) circuit to derive their conversion clocks from AES/EBU, Word Clock, or Super Clock 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 are 100% immune 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 ADC1 USB, DAC-104 and the ADC-104 employ Benchmark's new UltraLock™ technology to eliminate <u>all</u> jitter-induced performance problems. UltraLock™ isolates the conversion clock from the digital audio interface clock. Jitter on a DAC digital audio input, or an ADC 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 over sampling-ratio is varied with extremely high precision to achieve the proper phase relationship to the reference clock. Interface jitter cannot degrade the quality of the audio conversion. Specified performance is consistent and repeatable in any installation!

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 over sampling converter. This is a little known but easily measurable effect. Most audio converters operate at high over sampling 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 ADC or DAC 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 ADC and DAC devices. Practitioners of Digital Audio need to understand these effects.

Jitter Elimination Possible?

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 analog-to-digital converter. Any attempt to cure jitter outside of an ADC or DAC 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 ADC's and DAC's 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).

UltraLock™ Capabilities

UltraLock™ converters cannot undo damage that has already been done. If an ADC 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. It is therefore important to attack jitter at both ends of the audio chain. The ADC1 USB is a great start, as it will allow accurate assessment of various A/D converters. It is impossible to evaluate ADC performance without a good DAC. The consistent performance delivered by the ADC1 USB eliminates one major variable: jitter.

Regulatory Compliance

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 61000-4-4	Electrical Fast Transient/Burst
	EN 61000-4-5	Surge
	EN 61000-4-6	Conducted Susceptibility
	EN 61000-3-2	Harmonic Current
	EN 61000-3-3	Voltage Fluctuations & Flicker
Low Voltage Directive:	**98/68/EEC**	
Standard:	EN 60950	(ITE) Information Technology Equipment

Manufacturer's Name:	Benchmark Media
Manufacturer's Address:	5925 Court Street Syracuse, NY 13026
Product:	Audio A-D Converter
Model Number:	ADC1

This Certificate of Compliance issued July 13, 2005 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature:

Thomas P. Sims
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

RoHS Compliance

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)).

Warranty Information

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 ADC1 USB with a serial number greater than 00261 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.

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.

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.

Warranty Repair Procedure

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

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

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