Benchmark ADC16

Instruction Manual

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

...the measure of excellence!™

Contents

Overview

The ADC16 is a reference-quality, 16-channel, 192-kHz, 24-bit, analog-to-digital audio converter.

The ADC16 utilizes Benchmark's UltraLockDDS™ clock system with multi-function clock input and word clock output. UltraLockDDS™ is a proprietary clock-sync system that is immune to jitter. UltraLockDDS™ is phase-accurate, even across multiple ADC16 units.

The ADC16 has a wide variety of interfaces, including AES/EBU, coaxial, ADAT, Toslink, and more via an optional interface card. The optical output supports AES or ADAT formats at resolutions up to 192/24. In ADAT mode, high sample rates are supported using S/MUX^2 and S/MUX^4. The optional interface card will enable direct connectivity to a computer and other devices.

The signal-level meters on the ADC16 are dual-range, 9-segment LEDs. The meters can easily be switched to display a range of 48 dB or 20 dB. The meters also have a peak-hold function. The meters are single-sample accurate for precise digital signal metering.

The ADC16 is designed for maximum transparency. The ADC16 is well suited for the most demanding applications in recording and film studios. The internal power supply supports all international voltages and has generous margins for over and under voltage conditions.

The ADC16 achieves outstanding performance over a wide range of input levels. Each channel has a 10-turn gain calibration trimmer with a gain range of 23 dB. The gain calibration controls may be used to calibrate the ADC16 to precise studio reference levels.

The ADC16 has a BNC Word Clock output that follows the sample rate indicated on the front-panel display. 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 digital outputs. If desired, the digital outputs may be clocked to the ADC16's high-precision internal clock. The ADC16 will automatically revert to the internal clock source when the external clock is lost.

The ADC16 has four clock modes: 'DAW' (optional interface card), 'AES' (DB-25 AES/EBU input), 'WC' (BNC), and 'INT' (internal). All modes support sample rates between 28 kHz and 200 kHz.

The 'DAW', 'AES', and 'WC' modes allow the ADC16 to lock to an external clock reference. In these modes, the ADC16 will follow changes in sample rate. In 'WC' mode, the ADC16 will also follow changes in the type of reference signal (AES, SPDIF, word clock, or super clock).

When the 'INT' mode is chosen, the ADC16 is acting as clock master, operating at the selected sample rate. In the 'INT' mode, any signal at the clock-reference input will be ignored. If 'INT' mode is used, all devices connected to the ADC16 digital outputs will need to be configured to lock to the ADC16. Also, if the 'INT' mode is used, all other digital sources (A/D converters, etc) connected to the recording system must be synced to the ADC16's clock. The 'WC' output and any of the digital audio outputs on the back of the ADC16 may be used as clock references to feed connected devices.

The Benchmark UltraLockDDS™ clock-system is highly immune to jitter. When synced to the DAW, AES/EBU, WC, and super clock interfaces, the A/D conversion-clock is optimally conditioned using a technology called Direct Digital Synthesis (DDS). The UltraLockDDS™ system utilizes a temperature-compensated crystal oscillator to achieve an accurate center frequency (important when the ADC16 is operating as a master clock). The DDS system operates at

500 MHz and includes digital filters to remove jitter from the reference clock input. The Benchmark UltraLockDDS™ clock-system outperforms two-stage PLL designs while providing the frequency agility to handle the special sample rates that are often required for video transfers to and from film.

The Benchmark UltraLockDDS™ system delivers consistent performance under all operating conditions. Users should not hesitate to lock the ADC16 to other clock sources. The UltraLockDDS™ will remove jitter from the external clock reference, and supply a clean clock to the internal A/D converters.

The predecessor of UltraLockDDS™ is UltraLock™, Benchmark's pioneering jitter-elimination technology. UltraLockDDS™ meets or exceeds the performance of Benchmark's UltraLock™ system, but does not use asynchronous sample rate conversion (ASRC). The elimination of the ASRC processing reduces system latency and provides the most direct path from the A/D to the digital interface.

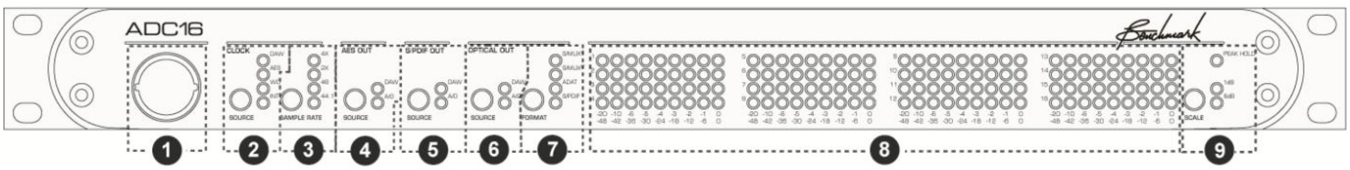
The ADC16 is designed to perform gracefully in the presence of errors and interruptions at the clock reference input. The ADC16 will even lock to an AES/EBU signal that has its sample-rate bit status set incorrectly since the sample rate is determined by measuring the incoming signal. Lack of sample-rate status bits or incorrectly set status bits will not cause loss of audio.

The ADC16 is phase-accurate between channels and between other ADC16 boxes when locked to AES/EBU or word clock reference signals. The word clock output from one ADC16 may be connected to the clock input on another ADC16 to expand the number of phase-accurate conversion channels.

Features

- Sixteen (16) channels of 24-bit analog-to-digital audio conversion
- 9-segment dual-range digital LED meters with 'peak-hold' functionality
- Multifunction clock input with auto-recognition of AES, SPDIF, Word Clock, or Super Clock
- Capable of synchronizing to a master clock, other digital sources, and/or computer
- Benchmark's phase-accurate jitter-immune *UltraLockDDS*™ technology
- High-precision, low-jitter master clock via a temperature-compensated crystal oscillator
- Word Clock output
- Balanced analog inputs via two DB-25 connectors (eight channels per DB-25) - Tascam pinout
- +6 dBu to +29 dBu input sensitivity range (at 0 dBFS)
- 10-turn gain-calibration controls (1 per channel) with a 23 dB gain range
- 'Press-and-hold' buttons to prevent catastrophic incidents
- Sample-rate selector and indicator for digital outputs
- Conversion at sample rates between 28 and 200 kHz
- Five types of digital output (AES/EBU, Coaxial, Toslink, ADAT, and optional interface card)
- All digital outputs can be sourced from internal A/D or from optional interface card (DAW)
- Optical output supports AES/EBU, ADAT, ADAT S/MUX^2, and ADAT S/MUX^4 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 90 - 264 VAC international power supply
- Meets FCC Class B and CE emissions requirements
- Tested to EN 6100-6-2 for RF immunity

Front Panel



1 AC Power Switch - page 7

2 Clock Source - page 8

Settings: Internal Clock, Lock to WC / SC, Lock to AES, Lock to DAW Card

3 Clock Sample Rate - page 8

Settings: 44.1, 48, 88.2, 96, 176.4 and 192 kHz

2 3 Clock Status Indication - page 9

Lock Status, Reference Sample Rate, Error Reporting

4 5 6 Source for Digital Outputs - page 10

Settings: Source = Internal A/D Converters, Source = DAW Card

7 Optical Output Format - page 11

Settings: AES Format, ADAT Format, ADAT S/MUX^2, ADAT S/MUX^4

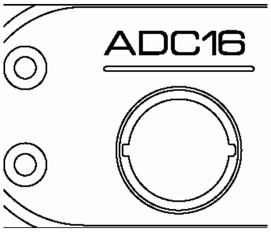
8 Meter Display - page 12

16 Dual-Range 9-Segment LED Meters, Driven from A/D Digital Outputs

9 Meter Scale and Peak Hold - page 12

Settings: Scale Factor = 1 dB/step or 6 dB/step, Peak-Hold On/Off

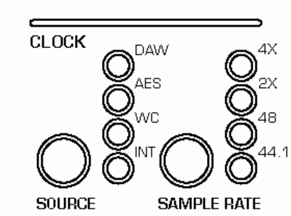
❶ AC Power Switch



DPDT (double-pole, double-throw) switch disconnects both sides of the AC line. When this switch is 'OFF', the ADC16 draws no power. When this switch is turned 'ON', the ADC16 will boot up and calibrate within 5 seconds.

User settings are automatically stored in memory. These settings are automatically recalled when power is restored.

2 3 Clock Settings



The 'CLOCK' control interface consists of a 'CLOCK SOURCE' button, a 'Sample Rate' button. Each button has a corresponding LED display.

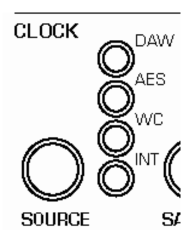
The ADC16 can operate from its internal clock, or it can lock to a variety of external clock sources. Valid clock signals include Word Clock, Super Clock, AES, SPDIF, and the optional 'DAW' interface card.

2 Clock Source

'Clock Source' Button

The 'CLOCK SOURCE' button is a 'press-and-hold' switch. This means that the user must hold in the button for one second before the clock source changes. *The 'press-and-hold' action prevents accidental changes.*

'CLOCK SOURCE' LED Display



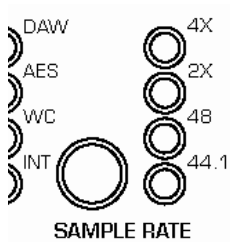
The 'CLOCK SOURCE' display indicates which input is acting as a clock reference for the ADC16. The indicators are as follows:

- **DAW** – On = clock is locked to the optional interface card
- **AES** – On = clock is locked to the AES/EBU input via DB-25
- **WC** – On = clock is locked to the WC input via BNC
- **INT** – On = clock reference is the ADC16's internal clock

3 Clock Sample Rate

The 'SAMPLE RATE' control interface consists of a 'SAMPLE RATE' button and LED display.

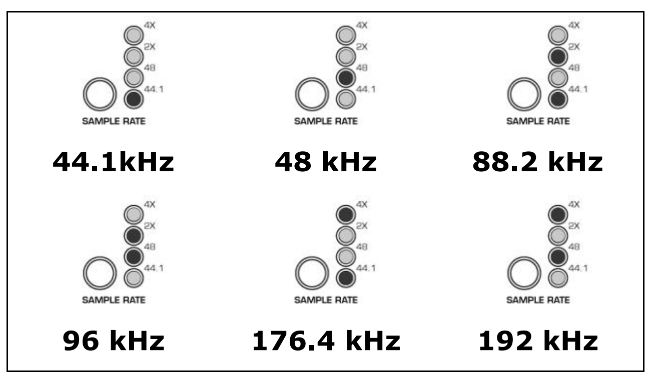
The 'SAMPLE RATE' Button



The 'SAMPLE RATE' button sets the sample rate when the ADC16 is in the 'INT' clock-source mode. This button is only active when the 'INT' LED is on.

To change the internal clock sample rate, press and hold the 'SAMPLE RATE' button. The 'SAMPLE RATE' button must be held for 1 second to change to the next rate. *The 'press-and-hold' action prevents accidental changes.*

'SAMPLE RATE' LED Display



The 'SAMPLE RATE' LED display always indicates the sample rate at which the ADC16 is operating.

In any external clock mode, the ADC16 will automatically follow the sample rate of the external reference signal. The measured sample rate of the external source will be displayed on the LEDs.

If the reference signal is good, the ADC16 will lock within 8 seconds. The ADC16 UltraLockDDS™ system is very tolerant of poor quality clock reference signals.

❷ ❸ Clock Status Indication

Clock Error - Not Locked

'Clock Source' LED flashing rapidly

If selected clock source is present, the corresponding clock source LED will flash rapidly while the system is acquiring lock. Allow up to eight seconds for the system to lock. Failure to lock within 8 seconds is an indication that the clock signal is corrupt or invalid. After eight seconds, the clock system will revert to internal sync, and the 'INT' LED will turn on. The LED corresponding to the selected sync source will continue to flash once per second (indicating the "no-reference" error condition).

Action: Check cables, connect a valid clock signal, switch to a different clock source, or switch the ADC16 to internal clock. Sample rate of external clock must fall within one of the following ranges: 28 – 50 kHz, 75-100 kHz, or 150 kHz-200 kHz

Clock Error - No Reference

'Clock Source' LED flashing once per second, and "INT" LED ON

If no signal is present on the selected clock input, the corresponding 'Clock Source' LED will flash once per second, and the "INT' LED will be on, indicating that the system has reverted to internal sync.

Action: Check cables, connect a valid clock signal, switch to a different clock source, or switch the ADC16 to internal clock.

Clock Error - DAW Frequency

'Sample Rate' LEDs flashing once per second

If the 'SAMPLE RATE' LED display is flashing, this is an indication that the ADC16 and optional DAW card are operating at different frequencies. The DAW card is only functional when the DAW sample rate matches that of the ADC16.

Action: Change the sample rate of either the DAW, the ADC16, or the external clock reference. The ADC16 and DAW sample rates must match.

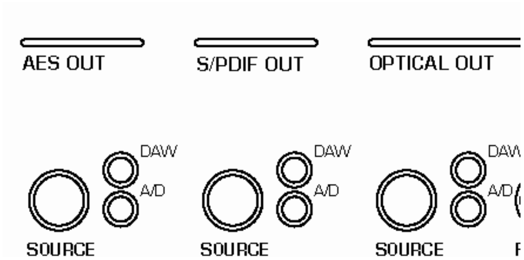
Clock Error - DAW Clock Mode

'DAW' 'Clock Source' LED is flashing once per second

If the DAW clock mode conflicts with the ADC16 clock mode, the 'DAW' 'Clock Source' LED will flash once per second. This can only occur when the ADC16 is not using the DAW as a clock source, or when the DAW card is configured to lock to a source other than the ADC16.

Action: Either change the ADC16 clock source to 'DAW', **or** configure the DAW card to use the ADC16 as a reference.

4 5 6 Source for Digital Outputs



The digital outputs can stream digital audio data from either the ADC16's internal A/D converters or from the source connected to the optional 'DAW' interface card.

To select the source for a set of digital outputs, press and hold the 'SOURCE' button for that set of digital outputs ('AES OUT', 'SPDIF OUT', and 'OPTICAL OUT'). The 'SOURCE' button must be held for 1 second to change. *The 'press-and-hold' action prevents accidental changes.*

Routing A/D Conversion Directly to Outputs

When 'A/D' is selected, the ADC16's internal A/D converters are routed directly to the digital outputs.

In any mode of operation, the A/D converter outputs are also routed to the optional 'DAW' card.

Using the ADC16 as a Digital DAW Interface

The optional DAW card provides a bi-directional 16-channel interface to an external device such as a digital audio workstation (DAW). The 16-channel return path from the DAW can be routed to external D/A converters using any of the digital audio outputs on the ADC16. In such a system, it is not necessary to purchase a DAW interface for the D/A converters. The ADC16 provides the interface between the DAW and the external D/A converters.

Routing A/D outputs through a DAW and Back to Digital Outputs

The optional 'DAW' card provides a bi-directional 16-chanel interface between the ADC16 and a DAW. Analog audio can be converted in the ADC16, sent to the DAW, mixed in the DAW and sent back out through the ADC16 to external D/A converters. Multiple mixes can be created for monitor feeds or other uses.

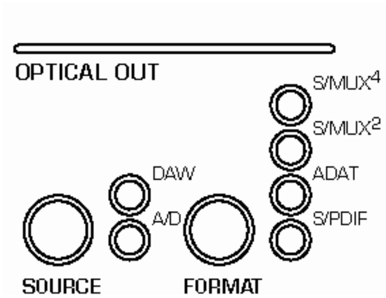
Using the ADC16 with an Analog Summing Bus

The bi-directional 'DAW' interface card can be used to send up to 16 channels of audio to external D/A converters feeding an analog summing bus. The outputs of the summing bus can be connected to the analog inputs on the ADC16 (and sent back to the DAW).

Using the ADC16 to Interface a DAW with Analog Effects

The bi-directional 'DAW' interface card can be used to send up to 16 channels of audio to external D/A converters feeding analog effects. The analog outputs of the effects can be connected to the analog inputs on the ADC16 (and sent back to the DAW).

7 Optical Output Format

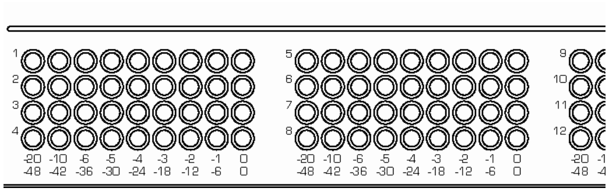


The optical output can provide either AES/EBU or ADAT format. The AES/EBU mode works with most S/PDIF optical inputs.

Press and hold the 'FORMAT' button to change the optical output format. *The 'press-and-hold' action prevents accidental changes.*

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).

8 Meter Display



The ADC16 is equipped with a multi-function 9-segment LED meter. The 'SCALE' button 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 relative to the maximum possible digital signal).

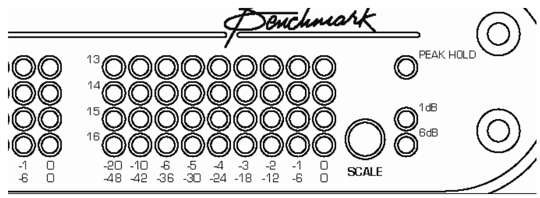
Time constants are built into the meters so all transient peaks can be observed easily. If a transient peak has a duration as short as one digital sample, an LED will be illuminated and remain illuminated long enough to be observed by the human eye.

A peak indication mimics the action of the needle on a peak-reading analog meter, while the remaining LEDs 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 ADC16 has a very large dynamic range. 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.

9 Meter Scale and Peak Hold

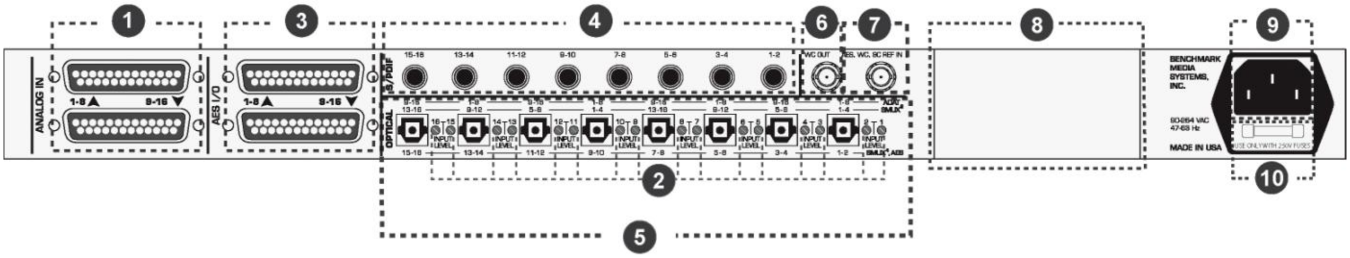


The 'SCALE' button selects the scale and peak-hold mode of the 9-segment LED Meter Display.

Meter Scale and Peak Hold Function Selection

Press the 'SCALE' button momentarily to enable or disable the Peak Hold function. Press and hold the 'SCALE' button to switch the scale between 1 dB steps and 6 dB steps.

Rear Panel



1 Analog Line-Level Inputs - page 15

Tascam Analog DB-25 Standard Pinout - 2 Connectors - 16 Channels In

2 Input Level Trim-Pots - page 15

One 10-Turn Trimmer per Channel, 23 dB Adjustment Range, Preset for +4 dBu @ -20 dBFS

3 AES Digital Audio I/O - page 16

Tascam AES/EBU DB-25 Standard Pinout - 2 Connectors - 16 Channels Out - 2 Channels In

4 Coaxial Digital Audio Outputs page 17

AES Professional Format 1Vpp Coaxial Outputs - 16 Channels Out

5 Optical Digital Audio Outputs - page 18

8 Optical Output Connectors - Supporting AES, ADAT, S/MUX^2, and S/Mux^4

6 Clock Reference Input - page 19

75-Ω BNC Input - Accepts Word Clock, Super Clock, and AES Input - Auto Recognizing

7 Word Clock Reference Output - page 19

75-Ω BNC Output - 2.5 Vpp into 75-Ω Load, DC Coupled

❽ Slot for Optional Interface Cards - page 19

See http://www.BenchmarkMedia.com/ADC16/optional-interface-card

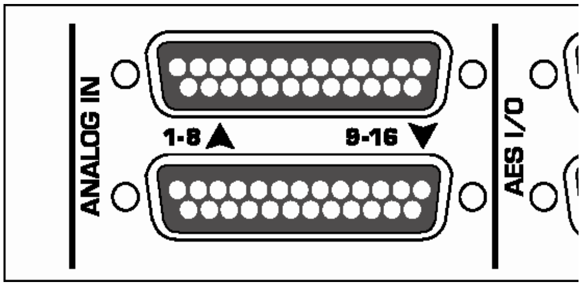
❾ AC Power Entry Connector - page 19

IEC Power Connector

❿ Fuse Holder - page 19

Requires two 0.5 Amp 5 x 20 mm 250 V Slo-Blo® Type fuses

❶ Analog Line-Level Inputs



The balanced "ANALOG IN" line-level inputs follow the Tascam Analog DB-25 standard. Cables are available from many sources.

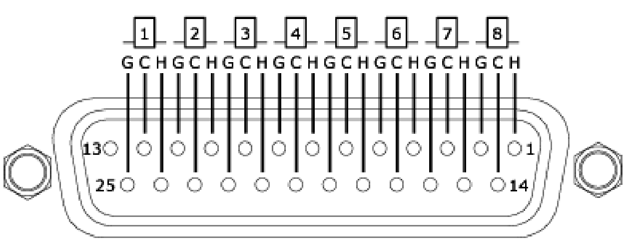
The analog inputs are factory calibrated to an input sensitivity of +4 dBu @ -20 dBFS. Rear-panel trim pots (❺) allow adjustment over a 23-db range. At minimum gain, the sensitivity is +9 dBu at -20 dBFS. At maximum gain, the sensitivity is -14 dBu at -20 dBFS. Unbalanced -10 dBv inputs are easily accommodated within this range.

The input impedance is 200k Ohms balanced, and 100k Ohms unbalanced.

Each analog input is equipped with an RF filter network. The ADC16 exhibits no measurable change in performance when subjected to EN 61000-6-2 RF immunity tests.

'ANALOG IN' DB-25 Pinout

The 'ANALOG IN' DB-25 pinout follows the Tascam Analog DB-25 standard:



H = "+" (or hot), C = "-" (or cold), G = GND

Unbalanced Source Adaptation

1. Connect "+" or hot (tip on ¼ phone plug, center pin on RCA plug).
2. Connect both the ground conductor AND the"-" conductor to the connector's ground (sleeve on ¼" phone plug, or case on RCA plug).

NOTE: When connecting an unbalanced source to the balanced analog input, it's best to use balanced wiring ("+", "-", "shield") and to tie the "-"and "shield" at the unbalanced connector (source).

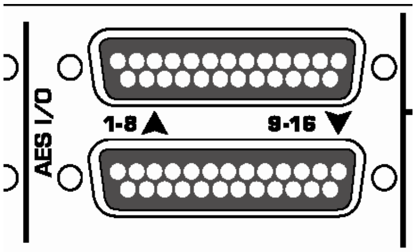
❷ Input Level Trim-Pots

Each channel on the ADC16 is equipped with 10-turn "INPUT LEVEL" trim-pots. These trim pots are located on the back panel near the optical outputs.

The "INPUT LEVEL" trim-pots have a range of approximately 23 dB. At their minimum setting, the ADC16 will clip upon receiving a +29 dBu analog input signal. At their maximum setting, the ADC16 will clip upon receiving a +6 dBu analog input signal.

The "INPUT LEVEL" trim-pots are factory-set to clip at +24 dBu.

❸ AES Digital Audio I/O



The balanced 'AES I/O' digital I/O connectors follow the Tascam AES/EBU DB-25 standard. Cables are available from many sources.

These 110-Ω 'AES I/O' connections include DC-isolation, transformer-coupling, current-limiting, and diode-protection. Outputs are designed to drive standard 4 Vpp AES signals into a 110-Ω load. 110-Ω digital cables are required when connecting these signals to other devices. Analog audio cables may cause data transmission errors, and should not be used to transmit digital audio.

- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rate = 28 to 50 kHz, 75 to 100 kHz, and 150 kHz to 200 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

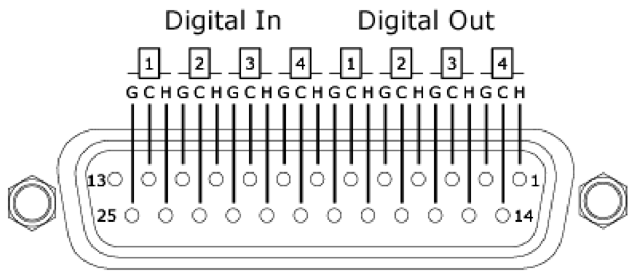
The digital audio data at the ADC16's digital outputs can come from one of two sources: the ADC16's A/D converters ('A/D') or the optional interface card ('DAW').

The user can select digital output sources via the 'SOURCE' buttons on the front panel. Each set of outputs ('AES I/O', 'S/PDIF', and 'Optical') has a 'SOURCE' selection button. These may be used in any combination.

When DAW is selected as the source, the user can assign any tracks or sub-mixes of tracks to the AES outputs.

'AES I/O' DB-25 Pinout

The 'AES I/O' DB-25 pinout follows the Tascam AES/EBU DB-25 standard:



H = "+" (or hot), C = "-" (or cold), G = GND

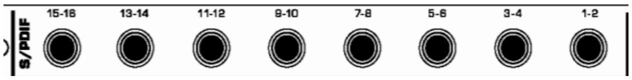
The Tascam AES/EBU DB-25 standard provides four 2-channel inputs and four 2-channel outputs for a total of 8 channels in and 8 channels out (on each DB-25 connector). For example, 'Digital Out' '1' carries channels 1 and 2, and 'Digital Out' '2' carries channels 3 and 4.

The ADC16 uses all 4 of the digital outputs on each DB-25 connector (for a total of 16 audio channels on 8 digital outputs).

The ADC16 only uses one of the 8 available digital inputs. Digital input 1 on the top DB-25 connector (audio channels 1 and 2) can be used as a clock reference input. The ADC16 can lock to an AES digital audio signal on this input when 'Clock Source' is set to 'AES'.

All digital inputs (including the unused inputs), are terminated with 110 Ω. 'Digital In' '2', '3', and '4' are unused on the top DB-25 connector. 'Digital In' '1', '2', '3', and '4' are unused on the bottom DB15 connector.

4 Coaxial Digital Audio Outputs



The ADC16 features eight coaxial digital outputs. There is one RCA coaxial connector for each pair of digital-audio output channels. The coaxial outputs are designed to operate within AES3-id and SMPTE 276M standards, and drive 1Vpp into 75-Ω. 75-Ω coaxial cables are required when connecting these outputs to other devices. Data transmission errors may occur if the wrong cables are used.

The coaxial outputs on the ADC16 are DC-isolated, transformer-coupled, current-limited, and diode-protected. They incorporate filtering to minimize RF interference and susceptibility.

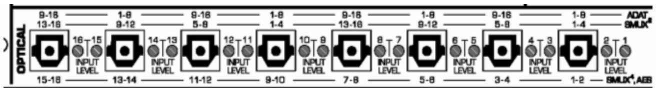
- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rate = 28 to 50 kHz, 75 to 100 kHz, and 150 kHz to 200 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

The digital audio data at the ADC16's digital outputs can come from one of two sources: the ADC16's A/D converters ('A/D') or the optional interface card ('DAW')

The user can select digital output sources via the 'SOURCE' buttons on the front panel. Each set of outputs ('AES I/O', 'S/PDIF', and 'Optical') has a 'SOURCE' selection button. These may be used in any combination.

When 'DAW' is selected as the source, the user can assign any tracks or sub-mixes of tracks to the coaxial outputs.

❺ Optical Digital Audio Outputs



The ADC16 features eight optical digital outputs. These outputs can be operated in four different modes: 'S/PDIF', 'ADAT', 'ADAT S/MUX^2', and 'ADAT S/MUX^4'. The 'FORMAT' button on the front panel toggles between 'S/PDIF' and 'ADAT' formats. If any ADAT mode is selected, S/MUX^2, or S/MUX^4 will be enabled automatically depending upon the system sample rate.

In 'S/PDIF' mode, each optical connector carries two audio channels. Data format is AES professional. The optical transmitters on the ADC16 support a maximum sample rate of 200 kHz in 'S/PDIF' mode . However, most optical inputs do not support S/PDIF or AES formats at high sample rates. Check the specifications on connected equipment before relying on high sample-rate optical connections.

In 'ADAT' mode, each optical carries eight channels of digital audio. In this mode, two optical connectors are required to carry all 16 channels produced by the ADC16. The remaining 6 optical connectors provide two additional sets of outputs. 'ADAT' mode is only available at sample rates between 28 and 50 kHz. If a higher sample rate is selected, 'S/MUX^2' or 'S/MUX^4' is automatically enabled.

ADAT 'S/MUX^2' is available at sample rates between 75 and 100 kHz. In this mode, each optical output carries four channels of audio. Four connectors are required for 16 channels of audio. The 8 optical connectors on the ADC16 carry two identical sets of data when operating in ADAT 'S/MUX^2'.

ADAT 'S/MUX^4' is available at sample rates between 150 and 200 kHz. In this mode, each optical output carries two channels of digital audio. All 8 optical connectors are required, to carry 16-channels of audio.

The optical outputs use what is often called a TOSLINK, Type FO5, or 5 mm optical connector. The ADC16 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. Some 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 'S/PDIF' Optical Mode

- Data Format = AES/EBU professional
- Word Length = 24 bits
- Sample Rates = 28 - 50 kHz, 75 - 100 kHz, and 150 - 200 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

ADAT Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rates = 28 to 50 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

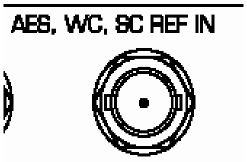
ADAT S/MUX^2 Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 75 to 100 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

ADAT S/MUX^4 Optical Output Mode

- Data Format = ADAT
- Word Length = 24 bits
- Sample Rate = 150 to 200 kHz
- Clock Source = Internal or External
- Data Source = A/D or DAW

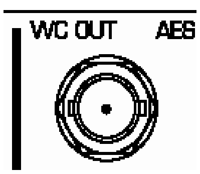
6 Clock Reference Input



The Clock Reference Input uses a BNC connector and is terminated with 75 Ohms. This input is DC isolated and includes overvoltage protection. This input auto-detects AES/EBU, SPDIF, Word Clock, or 256X Super Clock signals. When 'WC' mode is active the ADC16 will lock to this external clock input, and automatically follow changes in sample-rate.

Benchmark's UltraLockDDS™ circuitry isolates the conversion clock from any jitter present on the clock reference. 'WC' mode will not degrade the conversion quality of the ADC16, even when very high levels of jitter are present on the clock reference.

7 Word Clock Reference Output



The Word Clock output provides a clock signal which can be used to synchronize downstream components.

When the ADC16 is running in 'INT' mode (internal clock mode), the clock frequency at this output has an initial accuracy of +/- 5 PPM with a +/- 1PPM/yr aging characteristic. The INT mode is suitable as a studio master clock.

8 Slot for Optional Interface Cards

An interface card can be installed in the ADC16. The card is installed in the slot to the left of the AC power input. This slot provides a bi-directional 16-channel interface to external devices such as a digital audio workstation (DAW).

For more information, go to http://www.BenchmarkMedia.com/ADC16/optional-interface-card

9 AC Power Entry Connector

The AC power input uses a standard IEC type connector. Within the USA and Canada, the ADC16 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 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 fuse rating (for all voltage settings is 0.50 Amps).

Caution: Always install the correct fuses.

The mains power input has a very wide operating range. The ADC16 power supply can operate over an AC frequency range of 47 to 440 Hz, within a voltage range of 90 to 264 VAC. It can also operate from DC voltages between 127 and 300 VDC. There are no settings to change.

Specifications

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 = +16 to +29 dBu	121 dB
SNR – Unweighted, 0 dBFS = +16 to +29 dBu	119 dB
SNR – A-Weighted at max gain, 0 dBFS = -10 dBv	117 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	-105 dBFS, -102 dB, 0.00079%
Frequency Response at Fs=192 kHz	-3 dB, +0 dB, 2 Hz to 94 kHz +/- 0.02 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz +0.02 dB at 20 kHz -0.35 dB at 88 kHz -3 dB at 94.6 kHz -117 dB at 108 kHz
Frequency Response at Fs=96 kHz	-3 dB, +0 dB,1 Hz to 47 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz +0.02 dB at 20 kHz -0.10 dB at 44 kHz -3 dB at 47.3 kHz -115 dB at 54 kHz
Frequency Response at Fs=48 kHz	-3 dB, +0 dB, 1 Hz to 23.6 kHz +/- 0.01 dB, 20 Hz to 20 kHz -0.06 dB at 10 Hz -0.01 dB at 20 Hz +0.02 dB at 20 kHz -0.10 dB at 22 kHz -3 dB at 23.6 kHz, -110 dB at 27 kHz
Passband Ripple	+/- 0.001 dB at 44.1 and 48 kHz +/- 0.003 dB at 88.2 and 96 kHz +/- 0.007 dB at 176.4 and 192 kHz
Crosstalk	-96 dB at 20 kHz -120 dB at 1 kHz < -135 dB at 20 Hz
Jitter Tolerance (With no Measurable Change in THD+N 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	< -116 dB (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	-130 dBFS

Maximum Amplitude of AC line related Hum & Noise	-140 dBFS
Interchannel Differential Phase (Stereo Pair)	+/- 0.25 degrees at 20 kHz
Interchannel Differential Phase (Between ADC16 Units)	+/- 0.25 degrees at 20 kHz
Maximum Lock Time after Fs change	< 8 seconds
Mute on Sample Rate Change	Yes - less than 8 seconds duration
Mute on Loss of External Clock	Yes - after 5 seconds of no clock
Mute on Lock Error	Yes - after 5 seconds of no lock

Group Delay (Latency)

Delay (Analog Input to Digital Output)	1.49 ms at 44.1 kHz
	1.37 ms at 48 kHz
	0.75 ms at 88.2 kHz
	0.68 ms at 96 kHz
	0.37 ms at 176.4 kHz
	0.34 ms at 192 kHz

Balanced Analog Audio Inputs

Number of Inputs	16
Connectors	Two 25-pin DB-25 Connectors
Connector Pinout	Tascam Analog DB-25 Standard
Impedance	200 kΩ Balanced, 100 kΩ Unbalanced
Sensitivity at Maximum Gain	-14 dBu @ -20 dBFS
Sensitivity at Factory Preset Gain	+4 dBu @ -20 dBFS
Sensitivity at Minimum Gain	+9 dBu @ -20 dBFS
Maximum Input Level at Maximum Gain	+6 dBu @ 0 dBFS
Maximum Input Level at Factory Preset Gain	+24 dBu @ 0 dBFS
Maximum Input Level at Minimum Gain	+29 dBu @ 0 dBFS
CMRR	> 90 dB @ 1 kHz
Common Mode Trim	Dual Frequency
	RF Filter
	Transient Protection
	Over-Voltage Protection
	AC Coupled

WC/SC/AES Clock Reference Input

Connector	BNC
Input Impedance	75 Ω
Input Formats (Auto Detected)	Word Clock, 256X Super Clock, AES
Sensitivity with Word Clock Input	320 mVpp
Sensitivity with Super Clock Input	500 mVpp
Sensitivity with AES Input	150 mVpp
	RF Filter
	Transient Protection
	Over-Voltage Protection
	AC Coupled
	UltraLockDDS™ Jitter Attenuation

AES Clock Reference Input

Connector	Uses 'Digital In 1' on Top AES DB-25
Connector Pinout	Tascam AES DB-25
Input Impedance	110-Ω Balanced
Input Formats	AES, S/PDIF - Pro. or Consumer
Sensitivity	150 mVpp
	RF Filter
	Transient Protection
	Over-Voltage Protection
	Transformer Coupled
	DC Blocking Capacitor
	UltraLockDDS™ Jitter Attenuation

Balanced AES/EBU Digital Outputs

Audio Channels	16
Output Format	AES3 Professional Data Format
Output Sample Rate	Up to 200 kHz
Output Word Length	24-bits
Connectors	Two DB-25 Connectors
Connector Pinout	Tascam AES DB-25 Pinout
Output Impedance	110-Ω
Output Level	4 Vpp into 110 Ω
	Transformer Coupled
	DC Blocking Capacitors
	Transient Protection
	Over-Voltage Protection
	RF Filters

Coaxial Digital Outputs

Audio Channels	16
Output Format	AES3 Professional Data Format
Output Sample Rate	Up to 200 kHz
Output Word Length	24-bits
Connectors	Eight RCA Connectors
Output Impedance	75-Ω
Output Level	1 Vpp into 75 Ω
	Transformer Coupled
	DC Blocking Capacitors
	Transient Protection
	Over-Voltage Protection
	Maximum Sample Rate 200 kHz
	RF Filters

Multi-Format Optical Digital Outputs

Audio Channels	16
Output Formats	AES3, ADAT, ADAT S/MUX^2, and ADAT S/MUX^4
Output Sample Rate	Up to 200 kHz
Output Word Length	24-bits
Connectors	Eight Toslink Optical Transmitters

DAW Card Slot

Audio Channels	16 In and 16 Out
Sample Rates	Up to 200 kHz (Card Dependent)
Output Word Length	24-bits
External Interfaces	Card Dependent

Controls

Power Switch	Rocker
Clock Source	Button With Delayed Response *
Sample Rate	Button With Delayed Response *
AES Output Source	Button With Delayed Response *
Coaxial Output Source	Button With Delayed Response *
Optical Output Source	Button With Delayed Response *
Optical Format	Button With Delayed Response *
Meter Range (Uses Meter 'SCALE' Button)	Button With Delayed Response **
Meter Peak Hold (Uses Meter 'SCALE' Button)	Button With Instant Response

* Delayed response (push and hold) prevents accidental changes to these session-critical settings.
** Delayed response (push and hold) allows shared use of the meter 'SCALE' button.

Status Indicators

Clock Source	INT, WC, AES, DAW
Sample Rate	44.1, 48, 2X, 4X
AES Output Source	A/D, DAW
Coaxial Output Source	A/D, DAW
Optical Output Source	A/D, DAW
Optical Format	AES, ADAT, SMUX^2, SMUX^4
Meter Range	1 dB, 6 dB
Meter Peak Hold	Peak Hold On

Meters

Number of Channels	16
Number of LED Segments per Channel	9
Total Number of LED Segments	144
Ranges	2 (1dB/step, and 6dB/step)
Peak Response Time	1 sample
Ballistics	Fast Attack, Slow Decay
Peak Hold Function	Yes, Both Ranges

Power Requirements

Mains Voltage	90 - 264 VAC, or 127 - 300 VDC
Frequency	47 - 440 Hz, or DC
Inrush Current	< 60 A Peak @ 230 VAC
Power	45 W Maximum, 40 W Typical
Power Factor	0.7

Fuses

Type	5 x 20 mm 0.5 A 250 V Slo-Blo®
Quantity	2 Required

Dimensions

Form Factor	1 Rack Wide, 1 RU High 12.125"
Depth Behind Front Panel	16"
Recommended Minimum Rack Depth to Clear Cables	12.5"
Depth	19"
Width	1.725"
Height	

Weight

ADC16 only	7.0 lbs.
Shipping Weight	10 to 12 lbs.

ADAT S/MUX Tutorial

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.

ADAT S/MUX^2 Flag

The ADAT specification was amended in February of 2001 to include an S/MUX^2 flag. The ADAT Interface carries 4 'user bits' per frame, and user bit U2 is now designated as an S/MUX^2 flag. U2 should be set high when the interface is carrying S/MUX^2. The ADC16 follows this standard. Unfortunately, many devices ignore the user bits, and therefore, these devices will not respond to the S/MUX^2 flag. Again, the user must use caution when using an S/MUX connection.

No S/MUX^4 Flag

The February 2001 addendum to the ADAT specification made no provisions for an S/MUX^4 flag. User bit U2 should not be set in S/MUX^4 applications. Unfortunately, this means that the S/MUX^4 user bit configuration is identical to normal ADAT usage. Consequently, at the current time, user bits cannot be used to indicate S/MUX^4. This could change in the future. Again, the user must use caution when using an S/MUX connection.

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.

S/MUX Sample Rates Must Match

Most devices (including the ADC16) 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 to ½ the audio sample rate if S/MUX^2 is in use, or ¼ the audio sample rate if S/MUX^4 is in use.

S/MUX Must Not be Used for SRC

If two devices are connected with an ADAT S/MUX interface and the devices are set to different sample rates, a crude form of SRC (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.

ADAT S/MUX^2 Channel Assignments*

ADAT Channel 1 = Audio Channel 1.a
ADAT Channel 2 = Audio Channel 1.b
ADAT Channel 3 = Audio Channel 2.a
ADAT Channel 4 = Audio Channel 2.b
ADAT Channel 5 = Audio Channel 3.a
ADAT Channel 6 = Audio Channel 3.b
ADAT Channel 7 = Audio Channel 4.a
ADAT Channel 8 = Audio Channel 4.b

* 'X.a' and 'X.b' are successive samples of audio channel X

ADAT S/MUX^4 Channel Assignments**

ADAT Channel 1 = Audio Channel 1.a
ADAT Channel 2 = Audio Channel 1.b
ADAT Channel 3 = Audio Channel 1.c
ADAT Channel 4 = Audio Channel 1.d
ADAT Channel 5 = Audio Channel 2.a
ADAT Channel 6 = Audio Channel 2.b
ADAT Channel 7 = Audio Channel 2.c
ADAT Channel 8 = Audio Channel 2.d

** 'X.a', 'X.b' 'X.c' and 'X.d' are successive samples of audio channel X

UltraLockDDS™ … What Is It?

Accurate audio conversion requires a very low-jitter conversion clock. Jitter can easily cause severe inaccuracies if not adequately addressed, even when the device employs high-performance converter chips.

UltraLockDDS™ is Benchmark's latest jitter-immune clock technology. The ADC16 is the first device to employ Benchmark's new UltraLockDDS™ technology to eliminate all jitter-induced performance problems.

Benchmark's new UltraLockDDS™ clock system utilizes the latest low-jitter clock technology developed for high-frequency RF communications systems. The master oscillator is a low phase-noise, temperature-compensated, fixed-frequency crystal oscillator with a +/- 2 PPM frequency accuracy. This oscillator drives a 500 MHz Direct Digital Synthesis (DDS) system that generates a 3072 x WC system clock. This high-frequency clock is divided by 6 and distributed directly to the A/D converters using a high-speed PECL clock distribution chip. Each of the 8 converters are driven directly from a dedicated, matched-impedance transmission line.

Jitter attenuation is achieved with digital filters in a custom FPGA that controls the DDS system. All jitter-induced distortion artifacts are well below audibility under all operating conditions. Jitter-induced distortion is always at least 135 dB below the level of the music. The jitter-performance of UltralLockDDS™ meets or exceeds the performance of Benchmark's UltraLock™ system, but does not use asynchronous sample rate conversion (ASRC). The elimination of the ASRC processing significantly reduces system latency and provides the most direct path from the A/D to the digital interface.

Does my system have a jitter problem?

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. However, 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 a digital-to-analog 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. Actual performance may be severely degraded below specified performance 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 PLLs do not remove enough of the low-frequency jitter. In addition, two-stage PLL circuits often require many seconds to lock to an incoming signal, and may have a very limited frequency range. Two-stage PLLs may fail to lock if the reference signal is slightly off frequency. Two-stage PLLs may also fail to lock when jitter is too high.

UltraLockDDS™ 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. UltraLockDDS™ converters are highly immune to interface jitter under all operating conditions.

UltraLockDDS™ isolates the conversion clock from the digital audio interface clock. Jitter on the reference input can never have *any* significant effect on the conversion clock of an UltraLockDDS™ converter. Interface jitter cannot degrade the quality of the audio conversion in an UltraLockDDS™ converter. 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 UltraLockDDS™ converters to the test

We encourage our customers to perform the above tests on UltraLockDDS™ 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 UltraLockDDS**™** 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.

Is 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 digital-to-analog converter. Any attempt to cure jitter outside of an ADC or DAC (for example, with a low-jitter master clock) 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 ADCs and DACs are the only true remedy to ill effects of jitter. UltraLockDDS™ converters are jitter immune under all operating conditions (they will never add audible jitter induced artifacts to an audio signal).

UltraLockDDS™ Capabilities

UltraLockDDS™ converters cannot undo damage that has already been done. If a converter with a jitter problem was used to create the 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 ADC16 is a great start.

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:	**2004/108/EC**	
Generic Emissions Standard:	EN 61000-6-3: 2007	
Product Specific Emissions:	EN 55022	
Generic Immunity Standard:	EN 61000-6-1: 2007	
Immunity:	EN 61000-4-2	Electrostatic Discharge
	EN 61000-4-3	Radiated Susceptibility
	EN 61000-4-6	Conducted Susceptibility

Manufacturer's Name:	**Benchmark Media Systems**
Manufacturer's Address:	203 East Hampton Place, Suite 2 Syracuse, NY 13206
Product:	ADC16 -16-Channel Audio Analog to Digital Converter
Model Number:	500-14500-100

This Certificate of Compliance issued July 7, 2011 is valid for the test sample of the product specified above and that it conforms to the Directive(s) and Standard(s).

Signature: Annelle Frierson

Annelle Frierson
Vice President
Diversified T.E.S.T. Technologies, Inc.
4675 Burr Drive
Liverpool, NY 13088
Phone: 315-457-0245
Fax: 315-457-0428

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

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 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 ADC16 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 change this information without notice. This limited warranty gives the consumer-owner specific legal rights, and there may also be other rights that vary from state to state.

Extended 5 Year Warranty - US and Canada

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 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.

Shipments not displaying an RMA number may be refused.

Contact Information

Benchmark Media Systems, Inc.
203 East Hampton Pl, Suite 2
Syracuse, NY 13206
USA
+1-315 437-6300, FAX +1-315-437-8119
http://www.benchmarkmedia.com