Digital Audio Toolbox

User Manual

Version 1.05

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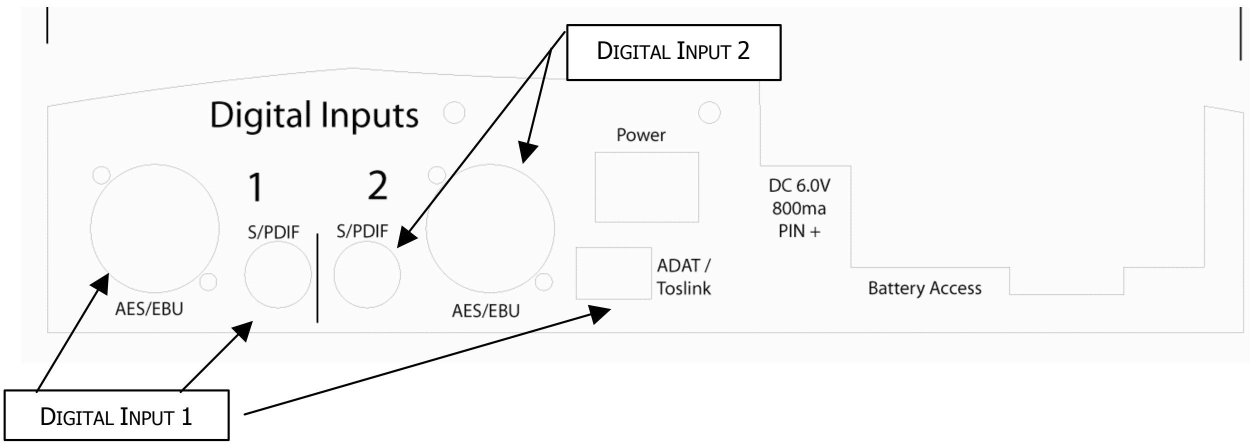
Introduction

Inputs



Digital Input 1

Digital Input 1 is routed to the DSP for decoding and processing. You may select from the AES/EBU input, the S/PDIF input, and the lightpipe input connector, which supports both ADAT and S/PDIF (or Toslink) format. Note that the S/PDIF and Toslink inputs share circuitry, so if one is being used, the other must not have anything plugged into it.

The Digital Toolbox supports input sample rates from 32 to 96kHz.

The Word Clock input (BNC connector) is used to lock the Digital Toolbox to an external clock, and also as an input to the jitter/lock tester board, as a 256xFs sample clock.

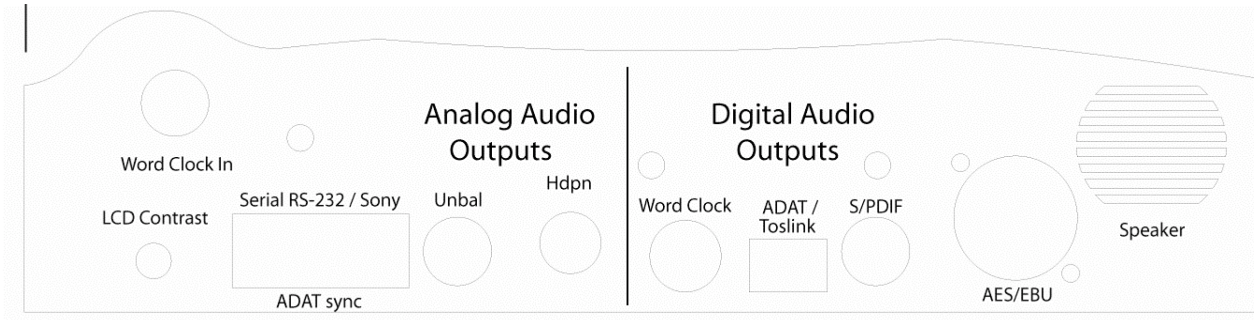
Digital Input 2

These inputs are routed to the jitter detector board. They include one AES/EBU input and one S/PDIF input. They are used for the lock tests, jitter tests, and the digital cable tester. The actual digital audio data is not decoded from these inputs. The input signal is only used for timing, jitter, and lock measurements.

Power Input

The power input powers the Toolbox and charges the battery when the Toolbox is turned off. Using an incorrect adapter can damage the battery, battery charger, and/or the Toolbox.

Outputs



Digital Outputs

These include AES/EBU, S/PDIF, lightpipe, which can be set to Toslink or ADAT format, and word clock out. The ADAT output sends out a copy of the stereo digital audio data on all 4 stereo pairs. The word clock output is derived from the output digital audio transmitter, and so will follow the lock source selected for generator.

Analog Outputs

Analog outputs include an unbalanced RCA phono connector, a 1/4" stereo headphone output. This output may also be used for an unbalanced line-level stereo output.

Word Clock / Video Input

An external word clock may be connected to this BNC jack. The Toolbox digital output can be locked to this source, if desired. Also, this connector can be selected by the sample/clock counter function, to measure the frequency of the signal.

This connector also functions as a video input, for the purpose of testing lock between a video sync signal (black burst or other video signal) and a digital audio AES, S/PDIF, Toslink, or ADAT signal.

DE-9 Connector

This connector is used for multiple purposes. It is used as a serial computer interface port, Sony 9-pin machine control port, and as an ADAT sync port input. The ADAT sync signal is a word clock that is on pin 8 of this connector, and so will not interfere with an RS-232 cable, which for this device only uses pins 2,3, and 5. The ADAT Sync input and the Word Clock Input share circuitry, so only one should be used at a time.

LCD Contrast

The contrast of the blue super-twist display may be set for optimum viewing by using this control.

Speaker

The internal speaker may be used for monitoring digital signals when headphones are not available. The speaker is driven separately from the headphones, and may be turned on or off independantly.

Clocks & Routing

Unlike analog audio, in many cases when feeding a signal into a device the digital audio must be locked to another clock. And any digital audio system, such as a DAW, has to have a reference clock. This can be an internal clock (a crystal), or an external clock, which can come from various sources: a word clock, derived from an AES or SP/DIF signal, or even from an ADAT signal, or sync signal, or even a superclock. The DATB makes use of these clocks in different ways.

The DATB has two main internal clocks: the Master Clock, and the Utility Clock.

Master Clock

The Master Clock is an internal clock that runs at 256 or 512 times the sample rate, or word clock. It is always used to clock the digital output transmitter. Also, the internal analog-to-digital converter (ADC) is clocked by the master clock. The ADC is used for several purposes, including reading the digital voltage level, reading the jitter board output, and measuring the internal battery voltage. The possible sources for the Master Clock are:

- Internal 32k crystal
- Internal 44.1/88.2k crystal
- Internal 48k/96k crystal
- Recovered master clock from the digital input receiver (using its Phase Locked Loop)
- External Word Clock, through the internal Phase Locked Loop (PLL) running at 256x.
- ADAT recovered clock from the ADAT receiver chip.
- ADAT sync (a word clock), through the internal PLL.

The Master Clock source is selected from the Bottom Toolbar. The frequency of this clock is always being counted, and if it is running at a frequency that is too slow to be considered valid, the source selection field will blink on the screen.

Since the Master Clock runs the analog-to-digital converter, none of the features of the ADC are available when the Master Clock is not valid, or simply not present. These include the battery level and AC indicator, and the functions that use the jitter board (Lock Detector, Jitter Meter, Jitter FFT, and the Digital Cable Tester). A valid Master Clock can always be obtained by selecting any of the internal crystals.

Utility Clock

The secondary clock system in the DATB is the Utility Clock. This clock has several functions: it is routed to the jitter board, and so can be used as one of the two inputs to the Lock Detector, it is also routed directly to the DSP to be counted accurately in the frequency counter function, and it can also be routed to the digital receiver chip, so that the digital input can be sample-rate converted to another clock, using the sample rate converter in the receiver. The possible sources for the Utility Clock are:

- Internal 32k crystal
- Internal 44.1/88.2 crystal
- Internal 48k/96k crystal
- Recovered master clock from the digital audio receiver
- External word clock, after running through the PLL at 256x
- ADAT Clock
- ADAT Sync, after running through the PLL at 256x

Output Clocking

The DATB output can be in two modes: generator output, or pass-through mode. In generator mode, the DSP is creating a test signal and sending to the digital output. In Pass-Through mode, it is sending the input signal to the output, possibly modified. In either case, you will need to select a valid clock source for the digital output, using the Bottom Toolbar.

Sample Rate Converter

There are times when may you need to re-clock a digital signal to match a different sample rate. The Sample Rate Converter (SRC) in the Toolbox can convert any sample rate, from

32k to 96k, to any other supported sample rate (32k-96kHz). The SRC in the digital input receiver is used for this purpose.

Analog Output Section

The analog output section can be used in two basic ways: to monitor the digital input, or to provide an analog copy of the digital output, for example to use as an analog generator. In fact, the digital-to-analog converter (DAC) is actually connected to either the digital receiver chip, or to the digital transmitter chip, depending on the requirements. In either case it simply outputs an analog copy of the digital data.

There are two analog outputs, an unbalanced mono RCA output, and a 1/4" stereo headphone output There is also a speaker. The speaker always outputs a mono sum of both left and right channels, and can be turned on and off independently of the analog line output.

The level of the analog output can be controlled independently of the level of the digital output, using the built-in analog attenuator in the DAC. It can also be muted. The speaker can be turned off independently from the other analog outputs, but if the main analog output is muted, the speaker will also be turned off.

All of the analog output controls are located on the left side of the bottom toolbar. See the section below for information on how to control the analog output section.

Power & Battery System

Power Input Jack

The Digital Audio Toolbox requires 6-8vDC unregulated at a minimum of 700ma. The input connector is a 2.1mm coaxial power connector, positive on the center pin.

Power Switch

The power switch is mounted on the input panel of the Toolbox, and will turn the Toolbox on or off whether powered by the internal battery or the AC adapter.

Internal Rechargeable Battery System

The Digital Audio Toolbox includes an internal rechargeable sealed lead acid (SLA) battery system. The battery module can be accessed by unscrewing the thumbscrew and opening the battery compartment door on the front of the unit, and sliding the battery module out of the Toolbox.

The battery charger is mounted directly on top of the battery. It has two connectors mounted on it, one that accepts the AC module power plug, and one that mates with the Toolbox internally. These connectors are different sizes so that the power plug cannot be plugged into the wrong connector. Also, the battery cannot be inserted the wrong way.

Charging

Anytime that the AC power module is plugged into the battery module, the battery will be charging. You cannot overcharge the battery. In fact, we recommend leaving the AC power module plugged into the battery module whenever possible. When the battery is mounted in the Toolbox, and the Toolbox is turned on, it will charge very slowly, since most of the power from the AC power module is being used to run the Toolbox.

You can charge the battery either in the Toolbox, or when it is removed. If you have a spare battery module, you can leave it plugged into the AC power module while you operate the Toolbox on the other battery.

Auto-switch

The battery module uses a latching relay to switch over from AC to battery power, and to prevent the SLA battery from being completely discharged, which can damage it. The switch-over happens automatically as the AC power module is plugged into the battery module. In some cases, it is possible to latch the relay into a state where the battery, even though charged, will not operate the Toolbox. These conditions include plugging the AC power module into the battery module with no or very low AC power present, and extreme physical shock. To correct this condition, just remove the AC power module plug from the battery module, make sure the AC module is getting good power, and plug it back into the battery module.

Battery Life & Charging Times

The display backlight is always on, since the blue LCD is not visible when it is off.

> ***Warning:*** *Storing a Toolbox with the battery in a state of discharge for long periods of time can shorten the life of the battery, or possibly destroy it. We recommend that the battery be kept charged at all times. The battery cannot be overcharged, so it is fine to leave it plugged into the charger when not in use.*

	Battery Life
Backlight On	1.1 hours

	Charge Time
Toolbox Off (or battery removed)	3.0 hours
Toolbox On	7.0 hours

User Interface Features

The DATB has many functions and features, and lots of routing, clocking, and output options. To help make it easier to understand how to use the functions, we have tried to provide a consistent user interface, with commonly used items easily accessible and located in the same place from screen to screen. Not all functions will use all items, but once you learn how to operate one function, you will have an easier time learning the next one.

The following sections discuss common user interface features.

Encoder Operation

The encoder is used like the mouse on a computer; it provides a means of navigating the screens, selecting values, and changing the contents of data fields.

To move around the menus, or the fields in a function, just turn the encoder clockwise or counterclockwise. You will see the highlight move around the screen. When it lands on a field, clicking the encoder will do one of three things:

1) Select the choice (as on a menu)
2) Change the option (if there are only a few choices, like on and off)
3) "Lock" the cursor to allow you to turn the encoder and change the value of a data field. Turning clockwise will increase the value, counterclockwise will decrease the value.

It is actually easier to use the encoder than it is to describe its use, so go ahead and try it out.

Menus

When you initially power up the DATB, you will see 4 menu choices. These are the main, or top-level, menus. Selecting one of these, by clicking on it, reveals another a list of functions. Selecting one of these will start the function running.

To exit a function, or to move from a function list to the main menus, just click on the '<' character in the upper left corner.

Saving Preferred Settings

Many DATB functions have fields that you might want to remember from one day to the next. Some of these settings include the digital input source selection, generator settings, including lock sources, word length, frequency, and level, and sub-menu choices on all of the functions. To store the current settings so that the DATB will start up next time set this way, after setting up the DATB the way that you want it, go to the Setup & Calibration function on the Utilities menu, and select "Save Settings". Next time when you power up the DATB, your settings will be restored.

Top Toolbar

The top line on the screen of every function has items that are used for control of the function and inputs.

Function Control

These items are on the left side of the top line on the screen.

- Exit Arrow The exit arrow in the upper left corner of the screen exits the function and returns you to the menus when you click on it.
- Function Name The name of the function currently running is shown next, in a box. The cursor cannot select this field.

- Sub-menu Some functions have more than one sub-menu, or screen available. If the cursor can select this field, you can click here to select the other screens.

Input Control and Information

These items are on the right side of the top line on the screen. They are shown for all functions that use digital input 1 for the source.

- **Lock Icon** Most functions have a lock icon. This icon will be open if no input lock is detected, or closed (locked) if input lock is detected for the currently selected input. Clicking on this lock will take you to the Bitstream Analyzer screen to get more information about the input signal.
- **Input Selection** Most functions allow you to select an input source. If the function uses the jitter detector board, the sources will be jitter sources. Otherwise, the sources will be from digital input 1: AES, SPDIF/Toslink, or any of the 4 ADAT pairs. If you select SPDIF/Toslink, make sure only one of those inputs is used at a time to prevent conflicts.
- **Sample Rate** The measured sample rate of the input is shown, with a resolution of .1kHz. Note that this is an actual running sample rate, not just a reporting of the status bytes. This sample rate is derived from information from the digital receiver chip, so if the sample rate converter is running, you will see the letters SRC rather than a frequency. This is because the incoming frequency is being sample-rate converted within the receiver chip, and so is not known to the DSP.
- **Level Meter** The far upper right corner of the screen shows the incoming digital level using an 8-segment bar. The bar consists of two lines, for the left and right channel level. Each dot represents approximately 12dB.

Bottom Toolbar

The DATB uses an icon-based toolbar on the bottom of the screen to make it easy to change a lot of common items. It appears at the bottom of most screens (some functions won't need access to the Toolbar and therefore don't display it). Toolbar icons control the analog output section and the digital generator.

Analog Monitor Control

The icons that control the analog output section are on the left side of the screen.

- **Analog Output On/Off** The 1/0 icon turns the analog output section on and off. This turns off both the headphone and the RCA unbalanced output.
- **Level** The bar level gauge icon controls the analog output level. Every click of the encoder changes the output level by approximately 1 dB.
- **Speaker On/Off.** This icon turns the speaker on and off. The speaker has its own amplifier, and so can be operated independently from the headphone output. To hear the speaker output, both the main On/Off and the Speaker On/Off must be turned on.
- **Source** The G/M icon determines the source for the analog output. M selects the digital input for monitoring, the G selects the internal digital generator (or whatever digital output signal is present). Switching from Monitor to Generator changes the source data for the DAC from the input digital receiver chip to the output digital transmitter chip. Therefore, in Monitor mode you will hear exactly what is being received, as selected by

the input select field on the top toolbar, and in Generator mode, an exact copy of the digital output signal will be routed to the DAC. In either case, a valid clock source is required. For example, if G is selected, and the output clock select is flashing, there will be no analog output. Likewise, if M is selected, a valid input signal is required. In this, case, the lock will appear closed on the top toolbar.

> ***Headphone Warning***: *Be careful when wearing headphones and changing sample rates, analog monitor source, and input connections. Although we try to mute the analog output in situations that may cause high analog levels, it is not always possible. We recommend that you remove headphones before switching any clocks, or before changing functions, until lock and a stable clock has been established, and the Toolbox has had a chance to get in control of the analog outputs.*

Digital Output and Generator Control

The icons that control the digital output and generator are located on the bottom toolbar, on the right side of the screen. Some fields control only the digital generator, while others control features of the digital output section, whether the source is the generator or another source, such as the digital input.

- **Jump** The sine icon jumps to the generator control screen for more options, such as frequency control and waveform selection.
- **On/Off** The 1/0 icon turns the generator on and off. This field also applies to functions that use special output signals, such as the transparency test.
- **Level** The bar level gauge controls the level of the digital generator. Every encoder click is exactly 1 dB of level change.
- **Word Length** The number field controls the word length (bit depth) of the digital output. You can select any word length from 8 to 24 bits. Word lengths less than 24 bits may be dithered according to several algorithms. The dither field is set on the Signal Generator screen.
- **Lock Source** The next icon shows the lock source (master clock) used for the digital output. You can select any of the 5 internal crystal speeds (32k to 96k), the word clock input, the incoming AES or S/PDIF input, or the ADAT lightpipe input. The ADAT sync connector (pin 8 of the DE-9 connector) and the word clock BNC input share the same input line. To use the ADAT sync connector (DB-9 connector pin 8), select the word clock input and remove any connectors from it. Likewise, when using the word clock input, do not connect an ADAT sync connector. Note that if you select a lock source that is not running, there will be no digital output. In this case the clock input select field will blink. Also, since the ADC is driven by the output clock, the battery level, jitter functions, and input voltage meter will not work.
- **A/T** The A/T icon selects the output format for the lightpipe output connector. **A** selects ADAT format, **T** selects Toslink. Note that in ADAT format, the digital stereo output signal is copied to all 4 sets of ADAT stereo pairs.

Other Icons

Some functions support saving results to non-volatile memory. These functions will display a memory chip icon on the Toolbar.

Digital Tests Menu

This menu contains functions that are used to test digital audio equipment, and to isolate potential problems.

Digital Lock Test

Description

This function allows you test whether a digital audio input (AES/EBU or S/PDIF) is locked with another digital audio input, the word clock, a superclock, ADAT input or ADAT sync. It uses very accurate crystal-based frequency counters to determine signal lock. Differences in frequency less than 0.01Hz are detected, which provides lock verification to better than 1ppm.

How to Use

- **Setup the test** Connect a digital audio signal to Digital Input Two. Select either the AES or SP/DIF input from the left selector. Connect the other signal to check for lock to either Digital Input One, the word clock input, a super clock (256x sample rate) to the BNC input, an ADAT signal to the input lightpipe connector, or an ADAT sync signal to the DE-9 connector. Select this input on the top toolbar, on the right selector.
- **Read the results** The following information will be shown: the frequency of each signal, whether lock is detected for each digital input signal, and the difference in frequency between the signals. Note that if the phase difference is large (defined here as greater than 4 hertz) the signal is considered out of lock. If the difference is less than 4 Hz, but greater than 0 Hz, the signals will be considered "drifting". If the frequency difference is 0 Hz, the two signals are considered locked.

Applications

Check lock between two digital audio signals, or see if a digital audio signal is locked to a word clock.

Video Lock Test

Description

This function allows you test whether a digital audio signal in AES/EBU, S/PDIF, or ADAT format is locked with a video sync signal (black burst). It uses very accurate crystal-based frequency counters to determine signal lock. Differences in frequency less than 0.01Hz are detected, which provides lock verification to better than 1ppm. Results are shown in terms of video frames per hour of drift. Note that since the BNC input connector is used for the video input, it is not possible to compare video sync directly with word clock or superclock.

Theory

Since the signals that we are comparing are several orders of magnitude apart, true synchronicity testing (like a phase comparison) is not possible. Rather, the ratio of frequencies is computed and compared to the ideal ratio to determine the amount of drift.

First, the digital audio frequency is compared to a list of known, typical sample rates, and the nearest sample rate is chosen. We call this the Nominal Sample Rate. The same is done for the video rate, and then the drift is computed using this formula, where VRa = Actual Video Rate, VRn = Nominal Video Rate, SRa = Actual Sample Rate, and SRn = Nominal

Sample Rate. The results are in units of Video Frames per hour.

```
Vfv/hr = ABS(1 – (VRa/VRn * SRn/SRa)) * 3600 secs/hr * VRn
```

There are several assumptions in effect here. First, it is assumed that the audio track was recorded at a "normal" sample rate. These include the standard 32k, 44.1k, 48k, 8812k, and 96k, as well as these rates pulled up or down by 0.1%. Also, it is important to know the intended sample rate of the audio. The Video Lock Test can show lock, if the audio is playing at a correct ratio, but a different sample rate than the rate at which it was recorded. To help determine this the actual and nominal frequencies are displayed.

How to Use

- **Setup the test** Connect a video sync signal to the BNC input. Connect the other signal to check for lock to either Digital Input One, an ADAT signal to the input lightpipe connector, or an ADAT sync signal to the DE-9 connector. Select this input on the top toolbar.
- **Read the results** The following information will be shown: the actual frequency of each signal, the derived nominal frequencies, whether lock is detected between the digital input signal, and the video signal, and the drift in terms of video frames per hour. Note that if the phase difference is large (defined here as greater than 4 video frames per hour (vf/hr) the signal is considered out of lock. If the difference is less than 2 vf/hr, but greater than 0 vf/hr, the signals will be considered "drifting". If the frequency difference is less than 2 vf/hr, the two signals are considered locked. These values are arbitrary, and in cases of true lock the drift will be less than 0.1 video frames per hour. Remember to verify that both the video frame rate and sample rate are correct.

Applications

Check lock between a digital audio signal and a video signal to determine if an audio track is locked to video.

Also test whether or not a piece of equipment is able to lock to video and generate digital audio that is locked to video.

Transparency Test

Description

Use this function to see if a piece of digital equipment can pass a signal without modifying it. The Transparency Test can generate several bit test patterns, and then monitor the input for those patterns. Any errors are counted and logged.

How to Use

The Transparency Test can run synchronously (analyzing the input as it generates the test signal), or asynchronously (analyzing a previously recorded signal, or a signal transmitted from a remote location).

Setup the connections. Connect any digital output from the DATB to the DUT (Device Under Test). Set the desired output lock source and word length from the Bottom Toolbar. Connect the output of the DUT to the DATB input. Select this input using the Top Toolbar. Or, record the test signal and play it back from the device under test.

Select a waveform. You can select <u>Walking 1s</u>, in which only 1 bit in the word is on at a time, and the bit position is shifted left one bit each sample, <u>Walking 0s</u>, which leaves all bits except one set to 1, moving the single 0 bit one bit left each sample, Monotonicity, which increases the value of the word by 1 each sample (0, 1, 2, 3, …to max value for the selected word length), Digital DC, in which you may select any particular value, in hex, or the J-Test, which is the Julian Dunn test waveform consisting of two superimposed square waves.

Start the test. If running synchronously, just turn on the test using the signal generator on/off icon on the bottom toolbar. If running asynchronously, turn on the output and record the test signal for as long as you wish the test to run.

Read the results. Once lock occurs, you should see a 0 in both left and right results columns. Any non-zero results indicate that the DUT is modifying the digital signal in some way. If you wish to clear the error field, just click on the word "Errors:". Note that in cases where a lot of errors are occurring, such as loss of lock, the error field will "wrap" back around to 0 after it reaches its maximum value of 65,536.

Troubleshooting errors. If you are getting errors from the transparency test, you should first verify that no EQ, compression, or any other signal modifiers are present. Next, turn off an sample rate converters, even if the incoming and outgoing sample rates are nominally identical. If you still have errors, try using the Digital DC waveform. Set a simple value, such as 000100 hex, and see what you are getting back. In some cases, you will be able to tell something from the return value.

Values exactly one less. If you send out 000100 and get back 0000FF (one less in hex) you should suspect that a multiplier is in the circuit, and is not being removed in the case of unity gain.

Value exactly double. This may indicate that the digital stream is being doubled, for example by monitoring a signal and also feeding a copy of it back in.

Applications

The Transparency test was initially designed to test digital recording equipment. It was important to know that whatever was recorded digitally would play back unaltered. This can be even more important today, when often encoded signals such as Dolby E are recorded using PCM recorders. This test can be used for DAT machines, digital multi-track recorders, and even for CD recorders.

Also, increasingly people have been interested to know whether or not a DAW will pass a transparency test at unity gain (theoretically, it should). Of course if ANY gain, attenuation, EQ, compression, reverb, or other effects are present, this test will fail.

This test can also be used to test an ISDN line, or any long-distance digital audio transmission that is supposed to be lossless. You can send the test signal out and back, or record it and send it from the other end.

Latency Test

Description

Use this function to measure the amount of delay that a piece of digital equipment is applying to a signal. Results are shown in both samples and milliseconds.

How to Use

The Latency Test can run either from a digital or analog output. The input is always digital.

Setup the connections. Connect any digital output from the DATB to the DUT (Device Under Test). Set the desired output lock source and word length using the icons on the Bottom Toolbar. Connect the output of the DUT back to the DATB input, and select this input on the Top Toolbar.

Select the output type. You can select analog or digital output. Use analog when you are testing the delay of analog-to-digital converters. Use digital for testing delay through digital systems.

Start the test. Turn on the test use the start icon. If testing a converter, you may need to adjust the converter input level or DATB analog output level to avoid clipping.

Read the results. The delay is shown in samples and milliseconds (ms).

Applications

The Latency test can be used to test the delay through an analog to digital converter (ADC). This is useful if you are using several converters while recording to determine that the difference in delays does not cause phasing errors on the recorded signal.

Also, test the delay through a DAW or digital recording console. Differences in delay through different paths in a console can contribute to phase errors.

Test latency through digital systems, such as live or broadcast systems to eliminate phase errors.

Jitter RMS Meter

Description

This function measures the RMS jitter of a digital input (AES/EBU or S/PDIF: interface jitter) or a clock. It can measure fs interface jitter of digital input in the range of 150ps to 100ns. Clock jitter (superclock, 256 x Fs, or sample clocks) is measured from 35ps to 100ns. Jitter is measured directly, rather than extrapolated from sideband information derived from FFT of analog domain signals.

How to Use

The Jitter Test uses digital input 2 and the BNC input connector.

Setup the connections. Connect a digital input or clock to the DATB. If you are measuring jitter on an AES/EBU or SP/DIF line, plug into Digital Input 2, and select this input on the top input line. Lightpipe cannot be routed to the jitter function. If you are measuring a clock, such as word clock or superclock, connect to the BNC input. Note that this connector is located under the bulge on the output panel of the Toolbox.

Read the results. The jitter is shown in units of nanoseconds (ns). A logarithmic meter shows the relative jitter graphically.

Applications

Measure jitter on digital outputs from any digital equipment. Also, direct measurement of high-speed digital clocks is possible, to find the conversion jitter of ADC OR DACS.

> *Note: The internal analog-to-digital converter (ADC), which is used to measure the jitter from the output of the jitter board, is clocked by the system master clock. For this reason, a valid master clock is required for this function to work. If you get the message "VALID MASTER CLOCK REQUIRED" then the selected master clock (which may be set to the external word clock, for example) is not valid. Select a different master clock on the bottom right toolbar. You can always select one of the internal crystals.*

Analysis Functions Menu

This menu contains functions that are useful in analyzing digital signals.

Bitstream Analyzer

Description

Use this function to analyze a digital signal. This screen can identify digital waveform types and give you a lot of information about the signal.

How to Use

The Bitstream Analyzer works on digital input 1.

> **Setup the connections**. Connect a digital signal to the DATB input. Select this input on the Top Toolbar.
>
> **Start the test.** Make sure the run/stop icon is set to run. (Stop is used for saving results).
>
> **Read the results: status.** There is a lot of information on this screen. The second screen line shows the information translated from the channel status buffer. Below to the left is a box of "confidence indicators", that should all be check marks. They include information received from the digital receiver chip, including biphase status, CRC, eye pattern pass/fail, voltage check, sample rate check, and word length check. There is a voltage meter that includes tick marks at the recommended max and min levels for the signal, a sample rate counter that counts the actual running sample rate, and a bit display with word length computation. The bit positions that are blinking between a '+' and a '-' character are actually changing, and any stuck bits are shown as there actual value (1 or 0).
>
> **More results:** Detail. The Detail sub-menu is specifically for looking at the channel status bytes. It displays all 24 channel status bytes for both channels (A and B), as well as a decoded display of channel A. All byte values are shown in hex format.

Applications

Any time you need to know what's going on in a digital wire, plug it into the Bitstream Analyzer. Use it to check actual word length (compared to reported or claimed word length), actual running sample rate, low voltage, or header information.

Bitscope

Description

Use this function to display a digital signal on an oscilloscope-like screen.

How to Use

The Bitscope works on digital input 1.

> **Setup the connections**. Connect a digital signal to the DATB input and select the input on the Top Toolbar.
>
> **Start the test.** Make sure the run/stop icon is set to run. (Stop is used for freezing results).

Select the scope mode. You can choose to view one input (left or right), or both inputs. To look at the phase relationship between left and right channels, select the X-Y sub-menu.

Set the gain. Using the field below the lower left corner of the graph, you can zoom into the digital signal, essentially applying gain to the input. This field shows the number of bits that can be displayed on the screen without clipping. When set to 24, no gain is present, and a full height waveform indicates a full-scale signal. As you reduce this number, the signal is magnified. For example, setting this field to 23 applies 6dB of gain (1 bit), and a full scale signal is actually at –6dB FS. You can reduce this number to 4, applying 120dB of gain. You can easily see dither and converter noise with this function.

Set the time scale. You can set the time scale as fine as 1 sample per pixel, or as coarse as 100 samples per pixel. Choose whatever time scale provides the best picture.

View & hear the results. The waveform will be shown on the screen. The digital gain is also applied to the analog monitor, so you can listen to amplified signal, either through the small speaker, headphones, or via the line output. Any signal exceeding the current screen upper and lower limits will cause the "OVER" indicator to appear.

X-Y mode. You can select X-Y mode to view the incoming stereo signal on a phase scope. Signals that are in phase will show up as a diagonal line pointing to the 0 mark on the screen. As the phase approaches 90 degrees, the line will approach the vertical. Further phase shift will tilt the line to the left, pointing to the 180 mark on the screen. As the phase angle increases, the line will appear to move back to right. Actually it has moved to the bottom of the screen, but it looks like it is on the top. Therefore, you cannot distinguish between 90 degrees and 270 degrees, but nonetheless the absolute phase relative to 0 appears clearly. The gain field described above can be used to magnify the signal on the screen.

Applications

Look for distortion

Listen to dither, or a signal below the dither threshold.

Listen for converter noise

Align equipment: phase delay. See the program phase content in X-Y mode

Clock/Sample Counter

Description

Use this function to accurately measure incoming sample rates of digital signals. You can also identify word clock, superclock, and bit clock frequencies.

How to Use

The Clock/Sample Counter works on digital input 1 or the BNC word clock input.

Setup the connections. Connect a digital signal to the DATB input 1. Or, if desired, connect a word clock, super clock, or bit clock to the BNC word clock connector. Select the input(s) that you are using on the Top Toolbar. If desired, a composite video signal can be connected to the word clock input connector.

Select the clock source. Select the clock that you wish to measure.

Read the results. The measured sample rate and clock rates are shown on the screen. For superclocks and bit clocks, the corresponding sample rate is also computed.

Applications

Use this function to identify word clocks or superclocks. Both of these signals generally use BNC connectors. You can see the actual clock rate in Hz.

Accurately measure sample rates of digital signals. By displaying accurate sample timing, you can identify equipment that is not running at the correct sample rate, or requires a sample rate converter.

Distortion Meter

Description

This function measures total harmonic distortion plus noise (THD+N) for a digital signal.

Theory

A narrow notch filter is used to eliminate the fundamental frequency from one channel of a digital input. The resulting RMS signal is compared to the unfiltered RMS signal level to determine the percent THD+N.

How to Use

The Distortion Meter works on digital input one.

> **Setup the connections**. Connect a digital signal to the DATB input one. Select the input that you are using on the Top Toolbar.
>
> **Select the channel.** Select either the right or left input channel.
>
> **Select the frequency.** Several frequencies are available for testing, starting at 63Hz and increasing on octaves. See the note below about available frequencies.
>
> **Read the results.** The measured THD+N is shown two ways: as a percent of the input signal magnitude, and as an absolute dB level referenced to 0 dB FS.

Note that the notch filters frequencies are referenced to 48kHz, so if other sample rates are detected at the input, the notch filter and generator frequencies will be adjusted to that sample rate.

Applications

Measure the distortion from a analog-to-digital converter (ADC). The distortion inherent in the analog sine wave generated from the DATB is near –90dB, so any distortion measured above that level is coming from the ADC.

Measure the distortion from a sample rate converter. Sample rate converters are usually implemented using digital filters. These filters can cause distortion. To measure it, route the DATB generator output to the DUT input, and send the DUT output to the DATB input. The distortion in the DATB digital generator is near the theoretical minimum (-145dB), so any distortion above that level is coming from the DUT. This method can also be applied to any digital device which modifies the signal without changing the frequency or waveform content. For example, a ADC/DAC pair can be tested, or the busses in a DAW.

Sweeps

Description

The Amplitude sweep generates a swept sine wave, outputs it to both the analog and digital outputs, and measures the incoming digital audio signal level. Results are plotted on the screen. The Sine sweep generates a swept sine wave, and repeats the sweep until turned off.

How to Use the Amplitude Sweep

The Sweeps measures the signal amplitude on digital input 1. The output can appear on both the analog and digital outputs.

Select the Ampl sub-menu.

Setup the connections. Connect either the digital or analog output to the device under test (DUT). Connect the digital audio output from the DUT to the DATB digital input 1. Select the input that you are using on the Top Toolbar.

Setup the test. If you are using the analog output, set the level using the bottom toolbar. You may need to adjust the Toolbox analog output level, and/or the input level to the DUT to avoid overloads and to optimize the level.

Run the test. Click the Play/Pause icon to begin the test.

Read the results. The measured amplitude response curve is displayed on the screen. You can use the cursor to read any data point on the screen.

Applications

Using the digital output, you can measure digital frequency response, graphically display EQ curves, and check for emphasis/deemphasis.

Using the analog output, you can check the frequency response of A/D converters, as well any equipment that has an analog input and digital output, like a consumer camcorder.

System Tools Menu

General purpose digital audio tools are found on this menu.

Level Meter

Description

Use this function to monitor the level of a digital source signal. Both numeric and VU meters are available.

How to Use

The Level Meter works on digital input 1.

Setup the connections. Connect a digital signal to the DATB input and select the input on the Top Toolbar.

Start the test. Make sure the run/stop icon is set to run. (Stop is used for freezing results).

Select the Level Meter mode. You can choose one input (left or right), both inputs, the sum (L+R), or the difference (L-R). Also, to look at phase between left and right channels, select the X-Y sub-menu.

View the results. The dB level or percent level will be shown on the screen.

Applications

Measure dB levels and calibrate equipment.

Digital Generator

Description

Use this function to create digital test signals.

How to Use

The Digital Generator creates test waveforms that are sent out the digital output. It also displays the unloaded analog output level.

This function can be entered either directly, from the menu, or by clicking on the sine icon on the bottom toolbar. If entered from the toolbar, exiting the generator returns you to the original function.

Note that changing level can be done from the bottom toolbar, in 1dB steps, or from the level field, in 10, 1, or .1 dB steps.

> **Setup the connections**. Connect the digital and/or analog outputs as desired.
>
> **Setup the generator.** You can select a sine wave, square wave, white noise, or pink noise as the signal type. For sine and square waves, you can set the frequency from 1 Hz to nearly 1/2 the sample rate. The output level can be set from 0db fs to –140dB.
>
> **Set the output level.** The output level can be modified in 0.1dB increments.

Applications

Create accurate digital reference levels.

Pass Mode

Description

Use this function to modify a digital signal. Change header information, do sample-rate conversion, change format, add gain or attenuate the signal.

How to Use

Pass Mode routes a signal from digital input one to the digital output.

> **Setup the connections.** Connect a digital signal to one of the DATB input one connectors and select the input on the Top Toolbar. Connect the DATB output back into your system.
>
> **Select the lock source.** The digital input data can be transferred to the digital output at the same sample rate at which it comes in, or it can be sample-rate converted to lock to another signal or clock. To send the signal out without changing the sample rate, select the master clock to match the signal source. For example, if you are using the AES input, set the master clock to AES/SPDIF. In this case the output will be locked to the input. To use the sample rate converter (SRC), just select one of the internal crystals, the word clock or superclock.

Note: *There are some "impossible" cases, such as selecting the AES or SP/DIF input, and selecting ADAT as the lock source. Of course, a valid lock source is required. Selecting Superclock as a lock source when a Word Clock (or no input) is connected to the BCN input will not work. However, it is legal to select ADAT as the input, and the AES input as the lock source, if a valid AES signal is present.*

> **Apply gain.** You can apply digital gain to the signal. Up to 60dB of gain may be added.

Modify the Status Bytes. If desired, change any of the status items, including sample rate, word length, copy protection, machine id, or other items.

Use the output. The incoming digital signal will be modified and sent to the digital output.

Applications

Evaluate A/D converters using a low-level analog input signal and a lot of digital gain.

Modify a signal to be compatible with another system, such as changing a CD input at 44.1k to match a DAW running at 48k or 96k.

Remove copy (SCMS) bit from a digital signal.

Modify channel status information so that a piece of equipment can lock to a digital signal.

Translate a pair of ADAT channels to AES or S/PDIF format, or send the AES or S/PDIF input to the ADAT output.

Dither a 24-bit signal to 16 bits.

Up-sample a 44.1 or 48kHz signal to 88.2 or 96kHz. This can have an advantage, when moving to analog, by pushing the conversion filters up above the audio spectrum.

Lock a free-running digital signal to an AES stream, word clock, or ADAT signal.

Digital Cable Tester

Description

Test a digital AES/EBU or S/PDIF cable for functional quality.

Theory

The TerraSonde digital cable tester uses the Julian Dunn J-test waveform to excite jitter in a digital audio cable, and then uses the jitter detector circuit to measure the resultant jitter.

The Digital Cable tester only works with AES and SP/DIF "wire" cables, not optical. Optical cables do not induce jitter, since they do not have capacitance. To test an optical cable, use the Transparency test, running at the maximum sample rate (96kHz).

How to Use

- Connect the cable to be tested from the digital output to digital input two.
- Read the cable quality on the meter.

Note: The analog-to-digital converter (ADC), which is used to measure the output of the jitter board, is clocked by the system master clock. For this reason, a valid master clock is required for this function to work. If you get the message "VALID MASTER CLOCK REQUIRED" then the selected clock (which may be set to the external word clock, for example) is not valid. Select a different master clock on the bottom toolbar. You can always select one of the internal crystals.

Capture Sample Data

Description

This function can be used to capture a buffer of data from the SP/DIF or AES input. It will capture data whether or not it is PCM (audio) data. You can then scroll through the buffer to examine the data that is store, and search for particular values.

How To Use

Connect any AES data stream to Digital Input 1. This can be normal digital audio (PCM), any encoded bitstream, such as AC-3 or DTS, or any other data stream that complies with the AES specification.

To begin the data capture, click the Play/Pause icon. The data will be stored until the buffer is full.

To examine the data, you can enter a starting address in the Start field, or simply scroll by clicking on the first address and rotating the encoder. Also, you can search for a value by entering the value in the Search field.

Applications

This function was written to debug an AC-3 encoding algorithm, to compare the data being transmitted to the data stored on a DVD, in the audio tracks. No other device was found that could accomplish this with non-PCM audio data.

Digital Watchdog

Description

Use this function to find errors or level overloads on a digital signal. It will run up to 24 hours, auto-save up to 40 sets of results, and will find as little as one sample error in 24 hours. There are 120 data points, so time resolution depends on the test length. Although all errors are logged, errors are combined within each data point (screen bar). For example, if you set the test length to 2 hours (120 minutes), each minute a new bar will be drawn on the screen. If any errors have occurred in this minute, a bar will be drawn. If there are no errors, the screen will stay blank for that data point. Thus, in this case, you can come back later and pinpoint any error to the closest minute.

By using the auto-saves, you can save a full day’s data with better than one-minute resolution. (But note that during the brief auto-save period, it is possible to miss an error).

How to Use

The Digital Watchdog works on digital input 1.

> **Setup the connections**. Connect a digital signal to the DATB input and select the input on the Top Toolbar.
>
> **Setup the test**. You can set a level that, if exceeded, will trigger an overload error. Or, leave it set to 0.0 to look for digital overs (defined here as 4 or more consecutive samples of full signal value). Set the test length using the time field in the lower right corner. If auto-saves are desired, set the “Saves” field to a non-zero value.
>
> **Start the test.** Change the run/stop icon to run.
>
> **View the results.** Any errors will show as bars on the screen. The more errors in a bar, the taller the bar. You can use the cursor to display the error type by dragging it over the graph.

Error list

The error codes are as follows:

Lk – Lock error. Lock has been lost. Usually many other errors will be reported if lock is lost. If this error is occurring intermittently, it may indicate connection problems or non-locked signals.

Ov – Signal over. This error will be triggered if the stated level is exceeded, for even one sample. If the trigger level is 0.0, the error will be triggered if more than 4 consecutive samples at full signal level are detected.

Ey – Eye error. The digital receiver chip has determined that the incoming signal does not meet minimum eye pattern criteria. The eye pattern is used to check that the digital audio waveform is high enough (has enough voltage difference between high and low states) and is wide enough (rise and fall times are not too slow). This pattern looks like an eye when viewed on a high-speed digital oscilloscope. Although we cannot show this pattern, the digital receiver used in the DATB is able to detect waveforms that are on the verge of eye pattern problems. Eye pattern errors may indicate a cable problem, such as poor cable quality (high capacitance will lower rise and fall times) or excessive cable length, which can reduce signal voltage level.

Cr – CRC error. The digital receiver has found a CRC (data integrity) error. This usually indicates a serious problem with the signal or transmitter. Check cabling.

Bp – Biphase error. Digital signals are biphase encoded. This way, the overall signal has no offset – it averages to 0 volts. If the biphase encoding is not done correctly, this error flag will be set. In normal operation, you should never see this error.

Applications

Use this function to find intermittent faults in equipment and cabling. Also, you can use it to monitor digital levels for conformity to level restrictions.

Utilities Menu

Sony 9-pin Tester

Description

Test a Sony 9-pin type serial machine control interface.

How to use

You will need an RS-232 to RS-422 adapter cable. This table describes the wiring for a cable that connects the DATB to a video tape recorder. The same cable should work for any controlled device. To test a controller, swap pins 2 & 3, and use two female DE-9 connectors.

Computer RS-232 DE-9 female	Direction	VTR RS-422 DE-9 male	Signal Name
1		n/c	DCD
2	<-	2	Receive
3	->	8	Transmit
4		n/c	DTR
5	<=>	1, 3, 4, 6, 9, s	Ground
6		n/c	DSR
7		n/c	RTS
8		n/c	CTS
9		n/c	RI

- Connect the cable from the DATB to a device to be tested.
- To try to control a device, select the machine control commands on the DATB screen and click. The command will be transmitted.
- Any response received from the device under test will be displayed on the screen.

Applications

Test a device to see if it is responding properly to Sony serial protocol commands.

Laeq

Description

Computes the average RMS level of the mono (combined) digital audio signal from digital input 1. This information can be used to set the Dolby Digital dialog level constant in a 5.1 stream.

How to use

- **Setup the test** Connect the input to a digital input, select that input on the upper toolbar.
- **Start the test** Click the run/pause icon to start the function.
- **Run the test** Allow the function to run for the desired amount of time.
- **Read the results** Elapsed time will be shown on the screen.

Applications

Determine the equivalent Dolby Digital 5.1 dialog level.

Setup & Calibration

Description

Save user settings in memory.

Calibrate the analog out.

Calibrate the frequency counter.

Computer Interface / About

Description

Displays the firmware version loaded into the DATB, along with tech support contact information.

Also used for communication with a computer, for example to transfer stored information from the DATB to a computer.

Glossary

This section defines commonly used terms in this manual.

ADAT – A digital audio transmission format invented and patented by Alesis. Usually running at 48kHz sample rate, each frame contains 8 channels of 24-bit digital audio. The physical connection uses fiber optics, using the same cable is TOSLINK.

ADC (Analog to Digital Converter) – To start with, you have to get into the digital domain. The ADC will do this. It runs at the desired sample rate – or faster, if over-sampling is used. At each sample, the level of the analog signal is measured and turned into a number. (See word length). It may be hard to imagine that the beautiful, complex music that you hear can be reduced to a single number. But if the conversion happen quick enough (say 48,000 times a second) then it really can sound pretty good.

AES/EBU – defined by IEC-985 and AES-3. It is a balanced line signal that normally uses XLR to interconnect equipment. This format takes its name from the two principal advocates and is the professional audio data i/f. This standard interleaves useful non-audio data with the audio to provide infomation on the sampling rate, etc

Bit Depth – See Word Length.

Clock rate – See sample rate.

DAC (Digital to Analog Converter) – This is how digital audio gets back to analog. The samples are non-continuous (by definition), since there are a finite number of levels (see Word Length) and a finite number of samples in a second (see Sample Rate), so at least the numbers need to be smoothed (filtered) to get back to analog.

Dither – As the signal that is being converted from analog to digital gets lower and lower, and finally approaches the least significant bit, more and more information starts to be lost. Oddly, by adding low-level noise to the incoming signal, the information starts coming through better, at the cost of more noise. The theory behind this is a little messy, but it really works. Imagine looking through your fingers, you would see some things but not all. Now wiggle your fingers, and you can see through them, at the cost of some more blurring.

DATB – Digital Audio Toolbox

DUT – Device Under Test, the unit being tested.

Hex – Short for hexadecimal, or base 16 arithmetic. A digit is 0-9 or A-F (A=10, B=11, ...F=15). Although computers work in binary arithmetic (1's and 0's only), its considerably easier for programmers and others who need to work with computer and digital audio numbers in hex. Here's some examples:

Decimal	Hex
0	0000

1	0001
15	000F
16	0010
-1	FFFF

Lock – In thjs context, lock refers to a property of two or more digital audio devices or data streams. If lock is present, then the devices or data streams are aligned, at least on the word level. Signals that are not locked are referred to as "out of lock".

Nyquist Frequency – 1/2 of the sample rate. No audio data can be stored or represented above this frequency. Just think of plotting samples, as the frequency goes higher and higher, there are less samples per wavelength. When you finally hit exactly 2 samples per sine wave (think of the top and bottom points of the sine wave), that is equal to the Nyquist frequency. Any higher, and you can't even have 2 points on the sine wave. If you keep going, the data actually shows up at a lower frequency – 1/2 of where it is supposed to be. So, very steep filters (so-called brick wall filters) are used to remove frequencies above the Nyquist frequency.

SP/DIF – (Sony/Philips Digital Interface) - defined by IEC-958 and AES-3. This unbalanced signal is commonly connected via coaxial BNC or RCA/phono leads. This format is actually just another mode defined by AES/EBU, a direct descendant of the IEC-985. It is the 'consumer' mode that forgoes some of the less relevant status information but does add the SCMS copy protection system.

Sample Rate – The rate at which samples of digital audio (words) are stored, or captured. In general, faster is better. No audio can be stored at above 1/2 of the sample rate. So, 44.1lHz cannot contain any audio above 22,050Hz, and in most cases the rolloff occurs lower, to prevent any data from sneaking in and aliasing (showing up at 1/2 of its frequency). Also see Nyquist frequency.

Status Bytes – The AES standard has the concept of status bytes. The idea here is that embedded within the data is information about the format of the data. Information such as the sample rate, word length, emphasis, conpyright, consumer, pro, and more is stored here. A few bits are stored in each frame. It takes 192 frames (left-right sample pairs) to store all of the data. The digital audio receiver will extract this data as the frames occur.

Superclock – A square wave that runs at 256 times the sample rate (256x Fs). It is used as an alternative to word clock to lock digital audio devices together.

TosLink -- Defined by Toshiba as a specification that combined S/PDIF with a cheap optical interconnect (those square plugs) designed specificaly for consumer and prosumer equipment. The term 'TOSlink' is now often used generally to describe that particular type of optical cable and connector, regardless of the digital data stream.

Word Clock – (Fs) A square wave that runs at the sample rate. Word clock is often used to lock (or synchronize) digital audio devices. The idea is that there is one master word clock, which is fed to all digital audio devices, so that all digital audio that may be interconnected is locked together.

Word Length – Number of bits in a word of digital audio. Can be an indicator of resolution, at the same sample rate. CDs are 16-bits, which allows for +/-32767 different loudness levels. 20 bit allows +/-524287 different levels, and 8-bit only 127. So at a given sample rate, more bits are better. HOWEVER, even if an analog to digital converter is "24 bit", it doesn't mean that you get all that resolution. In reality, the converter noise may be 17-21 bits, so you wouldn't get the full advantage. And, there are high-resolution 1-bit systems. By running very high sample rates, the overall resolution increases.

Specifications

Digital Specifications

Digital Formats

AES-3

S/PDIF

Toslink

ADAT

Latency Test

+/- 2 samples

Transparency Test

Maximum number of errors per channel: 65,536.

Maximum length of test: unlimited.

Frequency Counter

0.1ppm, or 0.01Hz in a word clock or sample clock, or 4 Hz in a master clock signal (256xFS).

Signal Generator

Sine wave distortion: Better than –130dB

Sine Wave range: 1Hz to 20kHz

Jitter RMS Meter

Input: AES/EBU or sp/dif, jitter range 150ps to 100ns

Input: Clock/Superclock, jitter range 35ps to 100ns

Lock Detect

Lock detect, <1ppm.

Digital Watchdog

Maximum time of test: 24 hours

Minimum length of detected error, all test times: 1 sample

Analog Specifications

Sine wave THD: Better than –85dB

Level range: +7dBu to –71dBu, unbalanced