AD-1000

Portable Reference Analog to Digital Conversion System

Operating Manual and
UV22® License Agreement

Revision 2.1: November 1996

Manual revised for this new edition by:
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Registered User Customer Support:
For customer support, please call (310) 915-1000
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Features and specifications subject to change without notice.

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Part Number AD-1000MAN Rev 2.0v1 August 1996

CAUTION
Any changes or modifications not expressly approved by APOGEE ELECTRONICS CORPORATION could void your authority to operate this equipment under the FCC rules.

The Apogee AD-1000
20 bit Resolution
A/D Converter

"Your first class ticket from analog to digital!"

A Note of Thanks

At Apogee we believe creating new products which really mean something requires putting a lot of one's self into that product. Apogee's AD-1000 was conceived and designed with a passion. The team who put it together put themselves into the product and the results show. We wanted the very best performance and most flexible capabilities – at a reasonable price. We took the pre-production units to the toughest critics, our friends in the mastering world, fussy mixing engineers, and picky end users. When we reached the point where they couldn't reliably pick between the analog original and the AD-1000 digital output, we knew we had arrived at our goal: a transparent, unveiled digital reflection of the analog source. Up to that point we tweaked and honed the design for almost two years, keeping you and our other clients waiting while the perfectionist in us wouldn't let go of the earlier designs. We know you will find the wait was worth it. Thanks for trusting Apogee – we are sure your new AD-1000 will surpass your expectations and provide years of quality service.

Apogee Electronics Corporation

Notes on CE Operation

In order to comply with the new (January 1996) requirements for electronic equipment operated in the European Union, Apogee converters must be equipped and operated with special adapters. These adapters are inserted into all of the I/O connectors on the converters and the normal cabling is then inserted into the adapters. Also, the equipment must be marked to reflect that it meets the CE standards ("CE mark"). The breakdown, by product, is as follows:

AD-1000 – Special adapter kit "CE-AD-1000" must be used for the I/O connectors. The connectors are labeled "Left Input," Right Input," "Digital Output," and "Sync Input." The converter itself must be marked "CE."

DA-1000E-20 – Special adapter kit "CE-DA-1000" must be used for the I/O connectors. The connectors are labeled "Left Output," Right Output," "AES Input," and "S/PDIF Input." The converter itself must be marked "CE."

PS-1000E – The power supply (PS-1000E) is internally modified and marked with "CE." The cable for the power supply has also been internally modified and marked "CE." This cable must be used to to meet the CE requirements.

Conformity to EMC Directive: AD-1000

To Whom It May Concern:

This is to confirm that the Apogee Electronics Corporation product, the AD-1000 2-channel 20-bit analog/digital converter, has been certified as being in conformity with the provisions of the EMC Directive noted below.

The above-referenced unit was tested for compliance with EN 55013:1990 and EN 55020:1988 and was certified as compliant under Letter of Certification number C21337-E issued by Garwood Laboratories, Inc, 7829 Industry Avenue, Pico Rivera, CA 90660 on March 14, 1996.

I, the undersigned, as an officer of Apogee Electronics Corporation, do hereby certify that the above information is true and correct. A copy of the aforementioned Letter of Certification is on file at the offices of Apogee Electronics Corporation.

*Signed:*________________ *Name:* RICHARD G. ELEN

Position: VICE-PRESIDENT, MARKETING *Date:* AUGUST 7, 1996

Licensing and Legal Information

Carefully read the following legal agreement prior to using the UV22 process provided in the AD-1000. *Use of UV22 constitutes your acceptance of these terms. If you do not agree to the terms of the agreement, promptly return the AD-1000 and the accompanying items, including written materials and containers to the location where you obtained them for a full refund.*

1. License Grant APOGEE ELECTRONICS CORPORATION ("Apogee") hereby grants to you, the Purchaser (either as an individual or entity), a personal, non-transferable, and non-exclusive right to use the UV22 Process provided with this license. You agree you will not copy the materials accompanying the AD-1000. The material contained in this manual consists of information that is the property of Apogee and is intended solely for use by the purchasers of the equipment described in this manual. Apogee expressly prohibits the duplication of any portion of this manual or the use thereof for any purpose other than the operation or maintenance of the equipment described in this manual without the express written permission of Apogee.

2. Copyright You acknowledge that no title to the intellectual property in the AD-1000 is transferred to you. You further acknowledge that title and full ownership rights to the AD-1000 will remain the exclusive property of Apogee, and you will not acquire any rights the UV22 process except as expressly set forth above.

3. Reverse Engineering You agree that you will not attempt (and, if you are a corporation, you agree to use your best efforts to prevent your employees and contractors from attempting) to reverse compile, modify, translate or disassemble the UV22 Process Software in whole or in part.

4. Customer Remedies Apogee's entire liability and your sole and exclusive remedy shall be, at Apogee's option, either to (a) correct the error, (b) help you work around or avoid the error or (c) authorize a refund or replacement (at Apogee's option), so long as the AD-1000, documentation and all accompanying items are returned to Apogee according to the instructions on the Warranty Information page opposite, with a copy of your receipts.

OWNER'S RECORD

The serial number is located on the top right hand corner of the rear panel of the unit. We suggest you record the serial number in the space provided below. Refer to it whenever you call an authorized APOGEE repair facility or the manufacturer. Make sure that you return your completed warranty card immediately!

Model No. AD-1000 Serial No. _______________ Purchase Date _________________

Dealer____

Warranty Information

Be sure to return the enclosed warranty card. If you do so, Apogee can contact you with any update information. As enhancements and upgrades are developed, you will be contacted at the warranty registration address. Firmware updates are free for the first year of ownership Please address any inquiries to your dealer or directly to Apogee at:

APOGEE ELECTRONICS CORPORATION, 3145 Donald Douglas Loop South, Santa Monica, CA 90405, USA.
TEL: (310) 915-1000, FAX: (310) 391-6262
email: support@apogeedigital.com. Web: http://www.apogeedigital.com/

APOGEE ELECTRONICS CORPORATION warrants this product to be free of defects in material and manufacture under normal use for a period of 12 months. The term of this warranty begins on the date of sale to the purchaser. Units returned for warranty repair to Apogee or an authorized Apogee warranty repair facility will be repaired or replaced at the manufacturer's option, free of charge. **All units returned to Apogee or an authorized APOGEE repair facility must be prepaid, insured and properly packaged**. Apogee reserves the right to change or improve design at any time without prior notice. Design changes are not implemented retrospectively, and the incorporation of design changes into future units does not imply the availability of an upgrade to existing units.

This warranty is void if Apogee determines, in its sole business judgment, the defect to be the result of abuse, neglect, alteration or attempted repair by unauthorized personnel.

The warranties set forth above are in lieu of all other warranties, expressed or implied, and Apogee specifically disclaims any and all implied warranty of merchantability or of fitness for a particular purpose. The buyer acknowledges and agrees that in no event shall the company be held liable for any special, indirect, incidental or consequential damages, or for injury, loss or damage sustained by any person or property, that may result from this product failing to operate correctly at any time.

USA: Some states do not allow for the exclusion or limitation of implied warranties or liability for incidental or consequential damage, so the above exclusion may not apply to you. This warranty gives you specific legal rights, and you may have other rights which vary from state to state.

Service Information

If the AD-1000 is kept in a clean environment free of **excess dust, moisture and heat**, it will give years of trouble-free service. The only components with a limited life are the electrolytic capacitors used. These are of high quality and will give many thousands of hours service.

The AD-1000 contains no user-serviceable components: refer to qualified service personnel for repair or upgrade. Your warranty will be voided if you tamper with the internal components. If you have any question with regard to the above, please contact Apogee by phone at (310) 915-1000, by fax at (310) 391-6262, or via email to support@apogeedigital.com.

In the event your AD-1000 needs to be upgraded or repaired, it is necessary to contact Apogee prior to shipping, and a Return Materials Authorization (RMA) number will be assigned. This number will serve as a reference for you and helps facilitate and expedite the return process. *Apogee requires that shipments be pre-paid*, and requests that all USA-originated returns be sent via UPS, and all international returns shipped via Federal Express — unless otherwise authorized in advance. **IMPORTANT: Any shipment that is not pre-paid or is sent without an RMA number will not be accepted.**

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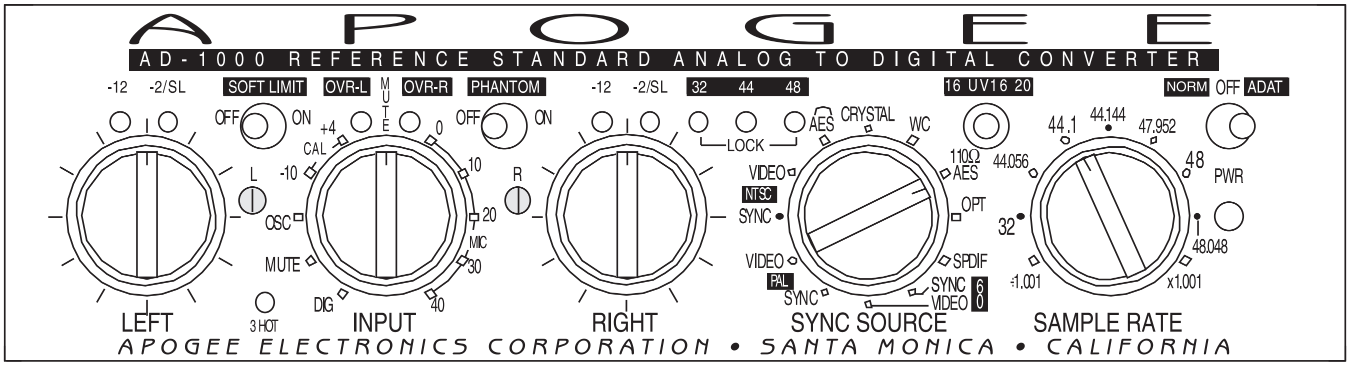
Introduction



The AD-1000 is a multi-purpose stereo analog to digital converter combining a high quality dual-stage, triple servo, analog 'front end' with 20 bit resolution. This combination provides unmatched sonic performance of 20 bit resolution, optimized for 20- and 16-bit applications such as digital tape recorders, disk-based audio workstations, and CD premastering. The AD-1000 includes Apogee's UV22® Encoding system, as used by mastering facilities worldwide to capture 20 bit detail in 16-bit formats such as CD and DAT.

Analog Input Circuitry

The AD-1000 features a proprietary balanced discrete front end with a combined noise and distortion measurement in the 0.001% range. The unit features XLR input connectors that will accept input levels from microphones and line levels up to +28 dBu. The common mode rejection is typically 100 dB – surpassing even the finest transformers – and provides very stable performance over a broad temperature range. Also incorporated, in front of the active circuitry, is a passive RF filter with hand-selected resistors and capacitors – matched to better than one part per thousand – to roll off any outside interference, without the typical intermodulation distortion byproducts of active circuitry.

Input Selector and Gain Controls

The input selector is an 11-position switch providing maximum input flexibility. When the switch is in the center MUTE position, the digital output still delivers a 'digital black' condition (all 0's) used for sync in many mastering and video applications. Rotating the selector clockwise, the next five positions select microphone input gain in 10 dB increments. Used in conjunction with the LEFT and RIGHT input level controls, up to 54 dB of gain is available (40 dB from the input selector and an additional 14 dB if the LEFT or RIGHT controls are fully clockwise). This combination allows the setting of the coarse gain with the input selector and fine calibration using the LEFT and RIGHT controls. *(Note: The LEFT and RIGHT controls can be bypassed and the gain can be routed to the multi-turn CAL pots if desired. See* ***page 31*** *for details of this procedure.)*

Line Level Inputs +4 dBu and –10 dBV

Two CAL positions are available by rotating the INPUT SELECTOR counter-clockwise from twelve o'clock. These positions correspond to +4 (Pro) and –10 (Consumer) line level inputs. When either of these positions are selected, the multi-turn CAL trim pots are enabled. *(Note: The CAL pots can be bypassed and the gain can be routed to LEFT and RIGHT controls if desired. See* ***page 31*** *for details regarding this procedure.)*

Phantom Power

48 volt phantom power is supplied to both microphone inputs when this selector is in the ON position and the input selector is in any MIC or the 12 o'clock mute position. The phantom voltage ramps up and down slowly for quiet activation and uses highly accurate resistors for optimum performance.

Soft Limit™
Included with the AD-1000 is Apogee's highly acclaimed Soft Limit circuitry. (This feature has functioned as many an engineer's "secret weapon" to get extra level in their masters.) Soft Limit functions as a selective peak limiter. Typical peak limiters are very abrupt and can spread unwanted harmonics. With the AD-1000, once an input signal passes the threshold, Soft Limit gently removes the peaks by rounding them off, making the limiting action very difficult to hear. The result is 'hotter' sounding program material.

Filters
Apogee has long been known for filter technology – the result of years of proven, proprietary designs. The AD-1000 continues this tradition with the ultimate in filter performance. A quadruple stage, progressive passive filter is inserted between the analog inputs and the analog to digital converter.

20 Bit Conversion and UV22®
The AD-1000 resolves analog detail to the 20-bit level to capture accurately every subtle nuance of the analog input signal. The output can be either 20-bit resolution or 16-bit. Apogee's UV22 Super CD Encoding system may be utilized when working with 16-bit recording formats. Used by virtually all major mastering facilities, UV22 captures the resolution and detail of the 20-bit conversion in a 16-bit word length; information that would normally be lost.

Sync Sources
It is unusual to use an analog to digital converter in a stand-alone environment. We have included every popular sync source capability via the ground-isolated BNC connectors (with additional loop through). Sync is selected via the front panel SYNC SOURCE SELECTOR SWITCH. Sync source capabilities include: CRYSTAL (selects the internal crystal reference from the sample rate switch); WORD CLOCK (locks to any external clock from 32 to 54KHz, including full vari-speed operation); AES/EBU, S/PDIF; VIDEO and SYNC (NTSC [525line/59.94Hz], Monochrome [525line/60Hz], PAL [625line/50Hz] are provided).

Sample Rate
An accurate 10 PPM internal crystal provides sample rates of 32, 44.1 and 48 KHz and the 0.1% derivatives 44.056, 44.144, 47.952 and 48.048 KHz (useful in film and video transfer work). The AD-1000 also has ability to multiply or divide incoming AES, S/PDIF, Optical and Word Clock sync by 1.001 for added flexibility. (1.001 is the ratio between monochrome video – originally used in mastering CD's – and NTSC video.)

Metering
A simple but very effective LED metering system has been incorporated into the AD-1000 that indicates levels below converter clipping and is designed to be used with the hosts' metering. The –12 LED for each channel can be customized. The –12 labeling is nominal; in fact the threshold of this LED is tied to the digital oscillator level and can be varied from –20 to –12 below 0dBfs. This is done by adjusting various switches on the top of the AD-1000. The –12/Threshold LEDs also serve an additional function. When a signal is within 0.05dB of the nominal gain of the AD-1000 oscillator setting, these LEDs flash rapidly. This is used to calibrate the analog to digital conversion levels. *For additional information, see Section 6, **page 33**.*

When Soft Limit is engaged, the –2 LED acts as a Soft Limit Threshold indicator. When this threshold is achieved, the –2/SL will light. When Soft Limit is *not* engaged, the LED will signal when a peak of –2dBfs is achieved.

Two separate Red LEDs indicate "overs" (digital clipping) when three or more samples in a row reach "all ones".

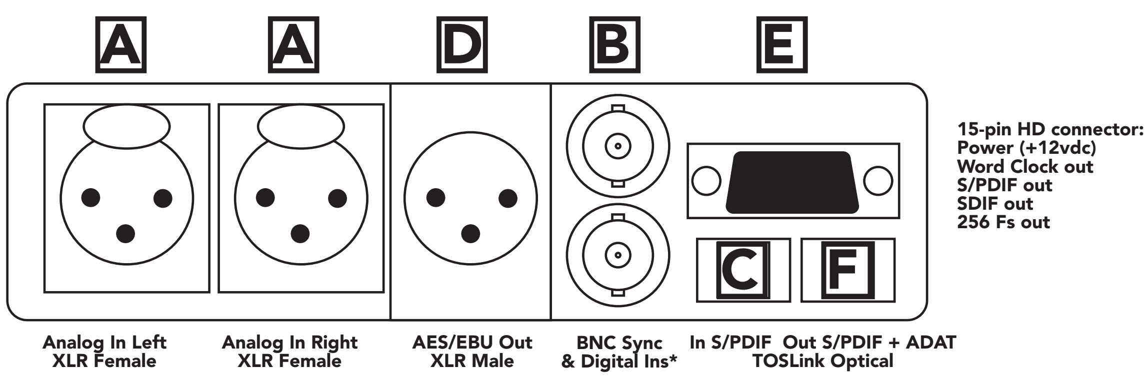
Digital Outputs
A separate transformer-isolated AES/EBU output is provided on the rear of the unit. Also provided is a fiber optic output connector which can supply either TOSLINK or ADAT format. S/PDIF and WORD CLOCK outputs are available on a high density 15-pin HD connector. (SDIF outputs are also optionally available on this connector). Additionally, the WORD CLOCK output is driven by the internal Low Jitter Clock via a high speed opto-isolated driver.

Apogee Patented Low Jitter Clock

The C768 Ultra Low Jitter Slaving Clock references to an external clock source and provides a 'flywheel' effect by smoothing out short-term timing irregularities or jitter. This low jitter slaving clock locks in a predictable and very accurate phase relationship with the incoming reference. The result is superior sonic imaging with excellent soundstage detail. **Whenever possible, the AD-1000 should be the digital timing source for the digital studio**. *Additional information on the sync capabilities of the AD-1000 is included on **page 25** of this manual.*

Specifications

QUANTIZATION	20 Bits/Sample
FREQUENCY RESPONSE	20Hz–10KHz ±0.025dB 10KHz–20KHz +0.025 / –0.1dB
TOTAL HARMONIC DISTORTION PLUS NOISE (THD+N)	
Full Amplitude	Typically –100dB @ 48KHZ sampling; 0.5dB below full scale
–20dB scaled to Fs	Typically –103dB@ 48KHZ sampling; 0.5dB below full scale
SIGNAL TO NOISE RATIO	Typically –104 Unweighted, –109dB A weighted
CROSSTALK	Typically better than –110dB @ 1kHz
SOFT LIMIT THRESHOLD	–1.5dB below digital output full scale (adjustable) selectable on/off
METERING THRESHOLDS	Amber indicates 2dB below full scale (includes peak hold function and SoftLimit Threshold) Green indicates 12dB below full scale (adjustable) (includes peak hold function) Red indicates three overs in a row (Includes peak hold function)
COMMON MODE REJECTION	Typically better than 110dB@100Hz Typically better than 75dB @10KHz
EXTERNAL SAMPLING RATE RANGE	Any frequency from 32KHz to 54KHz via External Sync Input
INTERNAL SAMPLING RATE	32kHz, 44.056kHz, 44.1kHz, 44.144kHz, 47.952 kHz, 48kHz or 48.048kHz
SAMPLING RATE ACCURACY	± 10 PPM
SAMPLING RATE INDICATOR	Amber LEDs indicate output sampling rate range for 32KHz, 44KHz, and 48KHz
INTERNAL CLOCK JITTER	Typically 30 picoseconds RMS
NOMINAL DC POWER INPUT	12vdc @ 1150-1300mA dependent upon function
INPUT VOLTAGE RANGE	+11.5 to +15 Volts DC (well regulated)
INPUT DROPOUT VOLTAGE	10.4 Volts DC
LOW VOLTAGE INDICATOR	Power LED Flashes at <11.5vdc Power Input
WEIGHT	1.3Kg (2 lbs 14 oz)
DIMENSIONS	L=273.0 x W=141.2 x H=39.6 mm (L=10.75 x W=5.56 x H=1.56 in)
OPERATING TEMPERATURE	0°C to 40°C (32°F to 104°F)



*Word Clock, video; AES sync, S/PDIF sync; AES Audio, S/PDIF Audio. Both connectors tied together (loop through) Unused connector should be terminated.

*Note: The AD-1000 will work with any well regulated 12V DC Power source with an output current of 1300 mA or greater. We recommend using linear type power supplies Direct connection to external 12VDC lead acid or NiCad batteries will provide convenient portable operation. See **page 43** for additional details on battery power.*

INPUTS

A. ANALOG INPUTS (two 3 pin female XLR connectors on rear panel)

Line Level:

- +4 dBu (Professional) nominal input level @ 10 kΩ balanced (>5 kΩ unbalanced).
  CAL +4 Maximum input peak level (front panel gain controls at minimum, fully counterclock-wise) +28dBu balanced, +24dBu unbalanced.
- –10 dBu (consumer) nominal input level.

Microphone Level:

- Minimum input peak level (Input Selector at Mic 40 and front gain controls at max): –54dBu Balanced, –50 dB Unbalanced (up to 54dB of gain is available).
- Phantom Power:
  Available in Mic Positions when Phantom Power Selector is enabled.
  (≈ 7mA available per channel to power condenser microphones.)

B. SYNC INPUTS (Two male BNC connectors on rear panel)
BNCs are paralleled for convenient looping to other units or termination (selectable termination).
BNCs will accept either balanced or unbalanced inputs.

WORD SYNC (WC):
32kHz to 54kHz Input Sampling Rates.
TTL or RS422 levels.
(input bridged by > 5 kΩ . Needs to be externally terminated @ 75 Ω).

DIGITAL AUDIO SYNC:
32kHz to 54kHz Input Sampling Rates.
AES/EBU Format (termination selectable at SYNC SOURCE switch to110 Ω or bridged by > 5 kΩ for looping to other devices).
S/PDIF Format (input bridged by > 5 kΩ or 75 Ω terminated - switch selectable).

ANALOG VIDEO:
Black burst/Composite Sync/Composite Video.
NTSC (525line/59.94Hz); Monochrome (525line/60Hz); PAL (625line/50Hz).
(input bridged by > 5 kΩ or 75 Ω terminated – dip switch selectable).

C. OPTICAL SYNC INPUT: (Optical input connector on rear panel)
32kHz to 54kHz Input Sampling Rates.
S/P DIF or AES/EBU Optical Format.

In addition to Analog to Digital Conversion, the AD-1000 offers many Digital format conversion options. Depending on the options installed, the following digital input options are available when the DIG position is selected on the INPUT SELECTOR.

DIGITAL AUDIO INPUT STANDARD (UV22 processing for digital inputs available as an option)
32kHz–54kHz input sampling rates.
AES/EBU format (termination selectable at SYNC SOURCE switch to 110 Ω or bridged by > 5 kΩ).
S/PDIF format (input bridged by > 5 kΩ or 75 Ω terminated – dip switch selectable).

S/PDIF DIGITAL OPTICAL INPUT (Optical input connector on rear panel)
32kHz to 54kHz Input Sampling Rates.
S/PDIF.

ADAT DIGITAL OPTICAL INPUT – available only with special option AD1K-PRT

OUTPUTS

D. AES OUTPUT (Male XLR connector on rear panel)
32kHz to 54kHz Output Sampling Rate.
AES/EBU Format when front panel NORM position selected.
Audio Black when front panel ADAT position selected.

E. S/PDIF OUTPUT (15 Pin HD connector on rear panel)
32kHz to 54kHz Output Sampling Rate.
S/PDIF Format when front panel NORM position selected.
Audio Black when front panel ADAT position is selected.

F. OPTICAL OUTPUT (Toslink Optical output connector on rear panel)
32kHz to 54kHz Output Sampling Rate.
S/PDIF or ADAT Format selectable on front panel power switch.

E. WORD SYNC OUTPUT (15 Pin HD connector on rear panel)
32kHz to 54kHz Output Sampling Rate.
Two types available:
• Balanced – RS422 Compatible.
• Unbalanced – TTL/Sony Compatible.

F. 256 fs OUTPUT (15 Pin HD sub connector on rear panel)

OPTIONAL DIGITAL OUTPUTS

E. SDIF-II OUTPUT (15 Pin HD connector on rear panel)
32kHz to 54kHz Output Sampling Rate.
Balanced - RS422 Compatible or Unbalanced - TTL Compatible.

F. HI-RESOLUTION OPTION - AD1K-PRT (20 bit encoding and decoding across track pairs)
Through a combination of enhancements to the AD-1000, a hi-resolution stereo output can be mapped to stereo pairs of ADAT (or TDIF with the FC-8 format converter) format tracks. This process is compatible with Rane's Paqrat™ format. *Additional information can be found in the Options and Enhancements section* ***(page 35)***.

Optional Accessories

PS-1000E
Model PS-1000E is a rack mountable dual worldwide power source for the AD-1000 with selectable 100, 120, 220, and 240 Volts AC and 50/60Hz providing dual 12.0 Volts DC regulated outputs at 1500mA each. Unit is ⅓ rack size and matches style and finish of the AD-1000. Power is distributed by two 15 Pin HD connectors on the rear panel. This unit can power up to two Apogee converters (A/D or D/A).

PS-1000E / AD-1000 Cable
Interconnect from PS-1000 to AD-1000 for S/PDIF and Word Clock output operation (cable with a breakout of two female BNC connectors – one for S/PDIF [red connector], the other for Word Clock output [white connector]. A male BNC to female adaptor is also provided.

AD-1000 SDIF Option
Adds SDIF and SDIFII output capability Left Data, Right Data and Word Clock via 15 Pin HD connector.

RM-1000
Model RM-1000 is a rack mountable carrier frame in Apogee purple, capable of holding up to three AD-1000, DA-1000E or PS-1000E in a 1U EIA space.

RM-2000
The RM-2000 is an alternative fan-cooled racking frame which holds three units in a 2U space where ventilation is a problem.

"PLATINUM OPTIONS"

AD1K-UVD (firmware option)
UV22 Digital Through option – allows a digital input signal to be UV22 processed (the standard unit only supports UV22 on signals generated by the A/D section from the analog inputs).

AD1K-PRT (firmware option)
20-bit bidirectional ADAT interface option – The standard AD-1000 provides an ADAT output. This upgrade adds ADAT input capability, plus the ability to record and play back two 20-bit signals by using two pairs of tracks on an ADAT machine. The 20-bit capability is compatible with Rane's PaqRat™ system.

FC-8 (Stand-alone unit)
Bidirectional ADAT to TDIF converter – This unit allows ADAT format 8-channel signals to be converted into the TDIF format used by Tascam DA-88 machines and compatibles. It is a bidirectional, stand-alone unit with its own power supply, and may be used independently of the AD-1000. (FC-8 available 3rd quarter 1996.)

COMBINATION OPTIONS

The **AD1K-ADT** option provides UV22 Digital Through and 20-bit ADAT capability as described above. The **AD1K-20FC** option adds an FC-8 to provide all three "platinum options" described above. This option makes a standard AD-1000 equivalent to the "AD-1000 Platinum Edition".

AD-1000 "Platinum Edition"
This is an AD-1000 system ordered with all three "platinum options" listed above.

Unpacking and Installation

Unpacking

Your AD-1000 is packed in a foam lined shipping container. Be sure to save the container for any future shipments of the unit.

Accessories

The following accessories are shipped with the AD-1000.

1 x Operation Manual
1 x Warranty Card
1 x 3/32 Hex Wrench
2 x 10-32 x 7/16 Mounting screw for rack mount
4 x Spare rubber feet

NOTE: Power supplies and associated cables are shipped separately

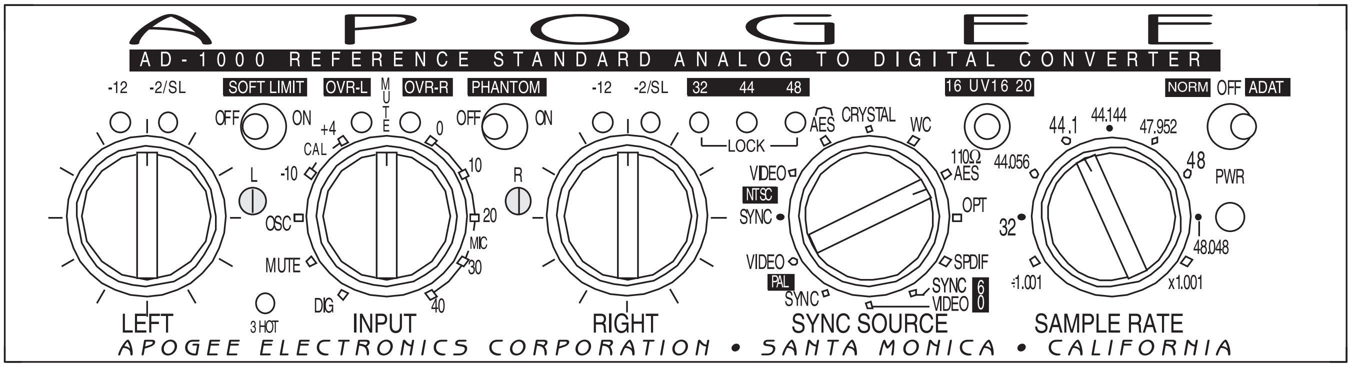
Installation

Your AD-1000 is designed for free standing or rack mount operation. The AD-1000 features a high performance analog input stage consisting of many discrete bipolar and FET transistors. This front end circuitry runs in a special discrete mode and is relatively power hungry. The AD-1000 therefore generates around 14 watts of heat. It is important to allow for adequate ventilation – otherwise it may overheat. It is, however, normal for the AD-1000 to run warm.

When using the unit in a free standing mode, make sure it is sitting on its rubber feet (spares are provided in the accessory kit) and that the cooling slots on the bottom are not obstructed. Operate on a flat surface free of materials such as paper or carpeting that may restrict natural cooling.

For rack mounted operation, check that the rack mount adapter is not tightly sandwiched between other items in the rack so as to restrict ventilation. It is best to allow at least one rack space above and below the units. **Fan cooling may be necessary** if the converters are above other heat-generating equipment. Additionally, the top of the rack should be ventilated to allow hot air to escape. *Remember – heat kills electronic equipment!*

Getting Started Quickly



Your AD-1000 is shipped ready to go. It has spent at least two days 'burning in'. This burn-in procedure involves powered operation at elevated temperatures to isolate units that could possibly fail due to infant mortality. Even if you are familiar with Apogee's AD-500E, we recommend you read the entire manual before using your AD-1000. There are several new features that make the AD-1000 even more useful. If you are anxious to get started and you don't like to read manuals, we suggest you take the following steps:

1. Connect your left and right analog inputs to the rear panel XLR connectors. The inputs are left and right as you look at the rear panel from behind *(refer to the rear panel diagram on* ***page 14 or 30****)*. **The input polarity is pin 2 hot** as per international standards. Balanced or unbalanced signals are accommodated. *If your input is wired for pin 3 hot you should see the information on setting Switch 5 on* ***page 31***.

2. Connect the digital output from either the AES/EBU male XLR connector on the rear panel, or the optical output connector on the right (when viewing the rear panel). To use the coaxial S/PDIF output you need the S/PDIF adapter cable (available from Apogee separately: PS-1000/AD-1000 CABLE) attached to the rear 15 pin HD connector. Alternatively, you can use an optical cable to connect to either an S/PDIF optical input or ADAT.

3. Connect the power cable between your power source and the AD-1000 using the 15 pin HD connector on the rear panel of the converter.

4. Set the front panel power switch to NORM or ADAT depending on the chosen output format. When NORM is selected, the XLR output transmits the AES/EBU format. S/PDIF optical is output on the right-hand TOSLINK connector; S/PDIF coax is available via a BNC connector sourced from the 15 pin HD connector (if the S/PDIF breakout cable is utilized). Selecting ADAT transmits the ADAT output on the optical digital output. Digital black will be present at the other outputs. (i.e. a normal AES or S/PDIF signal with muted audio for sync purposes.)

5. Set the SYNC SOURCE to CRYSTAL and set the SAMPLE RATE selector (typically to 44.1 or 48kHz)

6. Set the INPUT SELECTOR to either a CAL or MIC position.

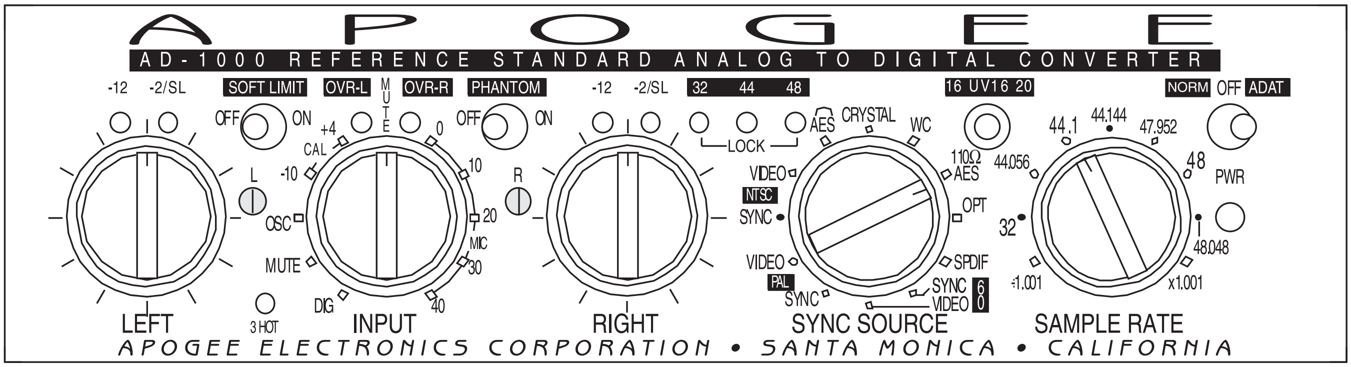
7. If you are utilizing a 16-bit recorder or system, set the RESOLUTION switch to UV16

8. If in a Mic Position, adjust the left and right gain knobs to flash the green '–12' LED's while occasionally flashing the amber '–2' LED's. If in a CAL position, adjust the multi-turn CAL pots located on the front panel, for a similar reading. *Further gain adjustments are discussed later in this manual on* ***page 33***.

9. Your AD-1000 should now be supplying digital audio.

Operation

Locations and Functions: Front Panel Controls



The AD-1000 offers an input section that can receive a wide variety of Analog and Digital inputs ranging from low microphone levels to the hottest line levels. The controls affecting the audio input selection and level are located toward the left of the unit and are detailed above. AES, S/PDIF electrical and optical signals can also be input and reformatted to other digital output types.

Input Selector

The input selector is an eleven position switch providing a wide selection of options, as follows:

Mute

When the analog input selector is switched to MUTE, the outputs of the AD-1000 transmit what is known as ‘audio black’ in their respective digital formats. This is a signal which corresponds to zero level input. Word Clock is still available at the 15 pin HD connector on rear panel. The analog to digital converter is turned off and will not pass audio; the analog and digital sections of the converter are powered down. The second MUTE position located at the 8 o’clock position is the same in function to MUTE at the twelve o’clock position except that phantom power is turned off in the 8 o’clock position.

MIC Input 0, 10, 20 ,30 ,40

Rotating the selector clockwise, these five positions select microphone input gain in 10 dB increments. Used in conjunction with the LEFT and RIGHT input level controls, up to 54 dB of gain is available. (40 dB from the input selector and an additional 14 dB if the LEFT or RIGHT controls are fully clockwise.) This combination allows the setting of the coarse gain with the input selector and use the LEFT and RIGHT controls for precise control.

The LEFT and RIGHT rotary controls can be bypassed and the gain controlled by the multi-turn CAL pots if desired. This is accomplished by changing the DIP switch settings accessed via a slot on top of the AD-1000.

Line Level Input +4, –10

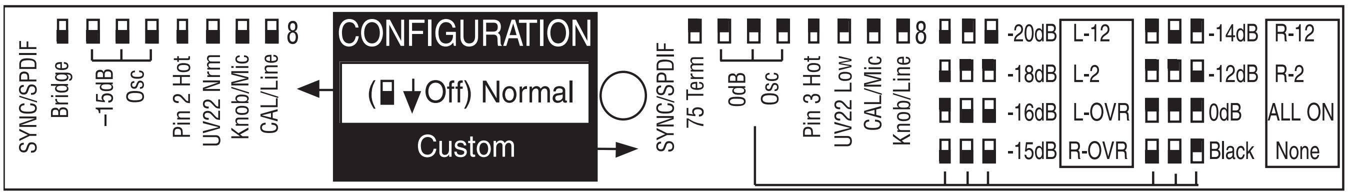
Two CAL positions are available by rotating the INPUT SELECTOR counter-clockwise from twelve o’clock. These CAL positions correspond to a nominal +4 (Pro) and nominal –10 (Consumer) line level inputs. When either of these positions are selected, the multi-turn CAL trim pots located to each side of the INPUT selector are enabled. These multi-turn pots are adjustable by a small straight blade screw driver.

The CAL pots can be bypassed and the gain controlled by the LEFT and RIGHT rotary controls if desired. This is accomplished by changing the DIP switch settings accessed via a slot on top of the AD-1000. *Refer to **page 31** for further information.*

OSC Position

OSC selects a built in 1kHz digital oscillator calibrated to ± 0.01 dB with internal switch selectable output levels for accurate headroom settings. When selecting OSC, the oscillator is selected and output on the corresponding digital outputs. This is used primarily for headroom alignment and testing. The output level can be selected by setting DIP switches located on the top of the AD-1000. The digital oscillator is set at 15dB below full scale as a default. These switches can be changed by using a small screwdriver. The table below lists the available test tone levels and the corresponding switch settings. When in OSC, a front panel LED will light indicating the level of the OSC output as a reminder corresponding to the level that is set by the DIP switches.

Below is a diagram of the switch settings that adjust these and other secondary functions. This is duplicated on top of the AD-1000 for convenience.



The black part of the DIP switches as indicated above represents the actuator of the switch. It may not be this color on the unit.

Level Below dB Full Scale	Switch 2	Switch 3	Switch 4	LED Display
20	Off	On	Off	Left –12 (green)
18	Off	On	On	Left –2 (amber)
16	On	Off	Off	Left Over (red)
15	Off	Off	Off	Right Over (red)
14	On	Off	On	Right –12 (amber)
12	On	On	Off	Right –2 (green)
0	On	On	On	All On
Audio Black*	Off	Off	On	All Off

**Audio Black is defined as a signal that contains clock data but no audio data (0 bits).*

Metering

The front panel metering LED's indicate when the AD-1000 input signals are peaking in a comfortable range. They are designed to be used in conjunction with the meters of the host system such as a DAT recorder, workstation, etc.

The **GREEN "–12" LED** indication for each channel may be customized. The –12 labeling is nominal because the threshold of this LED is tied to the digital oscillator level and can be varied from –20 to –12 below 0dBfs. This is done by adjusting various switches on the top of the AD-1000.

The **AMBER "–2 /SL" LED** indicates when the input signal is 2 dB below full scale digital out. When Soft Limit is engaged, it will also flash when the Soft Limit Threshold has been exceeded.

The **RED "OVER" LEDs** illuminate when three consecutive digital "Over" samples are output. The AD-1000 also includes a unique feature for precise input calibration delivering accuracy unavailable with normal digital bar graph meters of DAT recorders and workstations. Only with digital meters costing as much as the AD-1000 itself can you approach the accuracy of this system. In conjunction with the Digital Oscillator settings, the –12 LED will flash quickly when a tone is connected to the analog inputs and is within 0.05 dB of the oscillator level set on the programming DIP switches, located on the top panels of the AD-1000. *(Section 5, **page 31** has more information regarding these switches and the procedure to complete the calibration of your system.)*

Selecting the DIG Position

The last position counter-clockwise is the DIG position. When selected, an AES or S/PDIF signal (input via the top BNC connector on the back of the AD-1000) or S/PDIF optical signal (via the Toslink input connector) can be re-clocked and formatted into various digital outputs. This allows AES or S/PDIF conversion to ADAT and other useful format conversions. In this mode, the SAMPLE RATE cannot be selected and must remain in the 6 o'clock position. The sample rate of the digital output will be identical to the input, within a range of 32kHz to 54kHz. The format of the input is selected by the SYNC SOURCE selector. AES, Terminated and Bridged, as well as S/PDIF coaxial, terminated and bridged and Optical are supported. ADAT can also be reformatted and selected when the Hi-Resolution Option (AD1K-PRT) is installed. *See the Options & Enhancements section for further information on these options.*

To Input Digital Audio:

1. Connect your digital audio source to the AD-1000 to the top BNC connector on the back of the AD-1000 for AES/EBU* or S/PDIF operation. Optical input is available via the TOSLINK connector on the back of the AD-1000.
** NOTE: A special cable (Apogee part # WE-BX-0.5FT) is required to interface an AES/EBU XLR cable to the BNC connector.*

2. Set the INPUT SELECTOR to the DIG position.

3. Set the SYNC SOURCE to 110 Ω AES, SPDIF, or OPT (depending on type of digital audio source).

NOTES:

- *S/PDIF must be terminated with 75 Ω, either by setting DIP switch 1 (located on top of the AD-1000) to ON or a 75Ω BNC termination (supplied with the AD-1000) can be connected to the unused BNC connector (making sure not to double terminate). Termination on the AD-1000 is not necessary if looping to another device, as the last device in the chain is then terminated.*

- *Two AES positions are available. The position labeled 110 Ω AES terminates the input with a 110 Ω impedance. The AES position with the ^ above it signifies a bridged input of 5 kΩ for the AES. This position would be used if the AD-1000 was* not *the last device in the chain.*

4. Set SAMPLE RATE to the 6 o'clock position (the unmarked position - straight down).
Note: The 1.001 "gearbox" functions are not available with digital inputs.

5. Set the THREE POSITION SWITCH (located between the Sync Source and Sample Rate selectors) to either 16, UV16 or 20 as desired.

When inputting a 20 bit signal:

- the 20 position outputs 20 bits with no processing.
- the UV16 position outputs 16 bits*
- the 16 position outputs 16 bits with triangular dither.

When inputting a 16 bit signal:

- the 20 position outputs 16 bits unprocessed (whatever signal is input).
- the UV16 position outputs 16 bits*
- the 16 position output 16 bits with triangular dither.

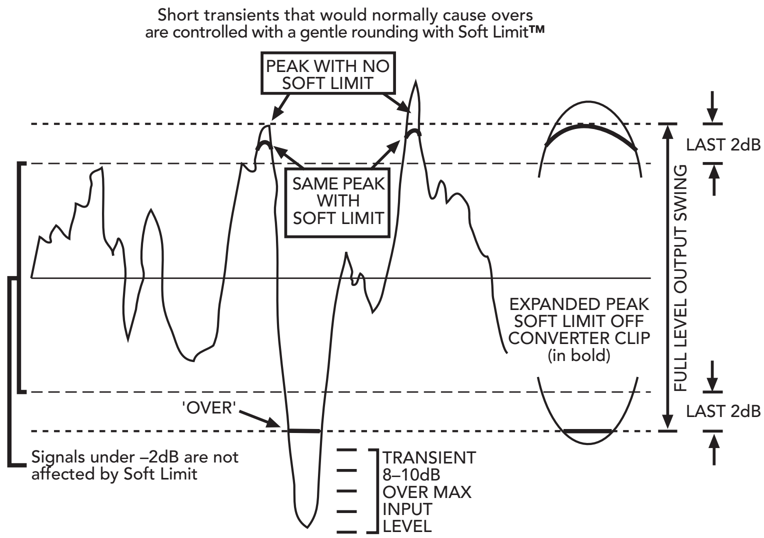
Note: UV22 is not available on a digital input unless a special option AD1K-UV22 is installed. See* ***page 35* *for operation of this option.*

6. NORM / ADAT Switch: Selecting ADAT transmits the ADAT output on the Optical digital output.
When NORM is selected, the AD-1000 simultaneously outputs AES/EBU (Male XLR) and S/PDIF in optical and coaxial formats.

Soft Limit™ Function

Evolved from the legendary AD500E, this unique Apogee feature makes your mixes jump out in the highly competitive market for CD airplay. Short transients can clip analog to digital converters and can produce unwanted harmonics. Typical peak limiters used to remove short transients are very abrupt and actually spread unwanted harmonics. If these short transients are removed without hard clipping, the average recording level can be raised by several dB with little or no sonic penalty (ie your ear doesn't miss the short transients if you remove them cleanly and don't overdo it).

Soft Limit does not make a hard, sharp-edge clip as you would get if the converter was over–driven. Instead, once you pass the threshold, it rounds off any peaks in a manner that makes their removal difficult to hear. These sharp peaks do not usually affect the sound quality if they are cleanly removed. Also, you get none of the typical limiter/compressor byproducts such as pumping and breathing. Depending on the material being recorded, the threshold of audibility will vary. This is best found by experimentation. Just a few dB increase in average level can result in more powerful, hotter sounding CD's without fear of leaving a trail of 'overs'.



THE EFFECTS OF SOFT LIMIT

The Soft Limit feature of the AD-1000, unlike conventional peak limiters, does not have a precise kick-in point. It is, in fact a logarithmic curve whose steepness increases as you approach the "threshold" (therefore "threshold" is a misnomer.) Determining the amount of limiting or increased effective headroom you wish to attain is really best done by ear. However, you may find the following procedure a good reference point, or a way to check left/right symmetry once you've found a pleasing threshold by ear.

Two different methods of calibration are described. One is for calibration with a recording console and the other is for calibration with test bench gear.

Soft Limit Calibration Procedure using a recording console

1. Calibrate the AD-1000 inputs to your optimum headroom setting using an oscillator via your mix desk's stereo output, or any metered source.

2. Patch the oscillator through a mult (parallel) into the analog inputs of the AD-1000.

3. Patch from the same mult into a line input of your mixing desk.

4. With Soft Limit turned off, set the oscillator to 1 kHz, increase the oscillator level to where the "over" leds light up, then reduce the level until they just go out.

5. Making sure there is no signal processing (dynamics, EQ, effects, etc.) dialed into the mixing desk's signal path, push up the line fader until the desk's stereo output meter reads the threshold you wish to attain using the Soft Limit feature. The factory setting is –4, so to check that it had been set correctly you would make your stereo meter read –4. If you wish to decrease the threshold to –6, (2 dB more limiting, effectively 2 dB more headroom) make the meter read –6. Alternatively, if you wish to increase the threshold to –2, (2 dB less limiting) your meter should read –2.

6. Turn on Soft Limit.

7. Increase the oscillator output until the desk's stereo meter reads 0 VU. (Don't touch the console fader at this point.)

8. Flip the AD-1000 over and find the two little multiturn adjustments through one of the ventilation slots (see diagram). The one nearer the front of the unit sets the RIGHT channel; the one nearer the rear sets the LEFT, and turning the pots *clockwise lowers* the threshold.

9. Now simply adjust the pots until each corresponding "over" LED lights, then back off until they just go out.

Soft Limit calibration Procedure using test bench gear:

You will need:
- An oscillator or tone generator capable of delivering 1 kHz up to about +20 dBu.
- An analog-reading meter capable of reading +20 dBu (any good quality high impedance volt meter will do, but you will have to do the conversion from voltage to dB. We recomend the Fluke 8060A (found in most studios) as it is extremely accurate and will read out in decibels, which makes all of this much easier.)
- A 3-way mult (parallel) – one in and three out.

1. Calibrate the AD-1000 to your specific requirements. *See **page 33** of this manual if you need instructions.* Turn Soft Limit OFF.

2. Mult the output of the oscillator to both inputs of the AD-1000 and to the meter. Adjust the oscillator output level until until the red OVR LEDs light, then back it off until the LEDs just go out. Make a note of this level on the meter. If you have an 8060A, just hit the REL (relative) button.

3. Increase the oscillator level by +4dB on meter (see procedure using a recording console step 5 for information about changing the threshold) and turn Soft Limit ON. Adjust Soft Limit Adjustment pots (on the bottom of the AD-1000) until OVR LEDs just go out. Verify that you are not getting any "overs" on a DAT machine or a digital-reading meter.
4. Return to your "normal" operating set-up.

Phantom Power

48 Volt Phantom Power is available when the INPUT SELECTOR is in any of the MIC positions or the 12 o'clock Mute setting and the PHANTOM POWER selector is switched to ON. At all other positions phantom power is disabled, regardless of the position of the PHANTOM POWER selector. The phantom power comes up slowly to minimize clicks and pops when powering microphones. 7mA per channel is available.

Note: Do not use phantom power with unbalanced microphone inputs.

20 Bit Resolution, UV22 and 16 Bit Selector

The AD-1000 outputs digital audio with 20 bit resolution. Most current digital audio recording and distribution systems (CD's, DAT, etc.) are capable of 16 bit resolution. To maintain as much of the AD-1000's 20 bit detail as possible, we have included two methods of capturing the extra resolution into 16 bit formats. The first is similar to standard triangular dither, and the second is Apogee's highly acclaimed UV22 encoding process. The mode of operation is selected by the three-position switch located between the SYNC SOURCE and SAMPLE RATE selectors.

In **20** position: the digital outputs carry the full resolution of the converter in a 20 bit word.

In **16** position: the signal is converted at 20 bits but is reduced to a 16 bit output by adding dither to remove the distortion of truncation.

In **UV16** position: the signal is converted at 20 bits and processed using Apogee's UV22 process to capture the resolution and detail of the 20 bit source in a 16 bit word.

Apogee's UV22 is used by virtually all major mastering facilities for 20 bit to 16 bit reduction for CD mastering. With the AD-1000, this process is available for all analog inputs. Two types of UV22 processing are available, Normal and Low. The Normal position is used almost exclusively. When the recorded signal is to be processed through UV22 more than two or three times, it might be beneficial to use the LOW setting. This setting captures almost as much detail as the NORMAL setting but has 6dB less energy. *Further information on UV22 is contained in Appendix I.*

*Unless your AD-1000 has the UV22 Digital Through Option, when in digital mode, the resolution of the output is equivalent to the digital input's resolution. Additional information on the AD-1000 UV22 Digital Through Mode and other Enhancement Options can be found on **page 35.***

Channel Status

The AES/EBU and S/PDIF formats are very closely related. The main differences are output level, impedance and some differences contained in a repeating train of data packed in with the digital audio. The data are called channel status. This train of information contains codes to tell digital audio products information such as whether it is AES/EBU or S/PDIF format, sample rate, pre-emphasis applied or not applied, if copying is permitted, etc. Most of the information transmitted in the channel status is transparent to users. The two main bits of information that may matter are: a) The data defining AES/EBU or S/PDIF format; and b) The data defining whether a copy can be made or not. Consumer-format copy protection systems are outside the scope of this document, and the copy-protection bit settings of the AD-1000 are not user-adjustable.

Timing and the AD-1000

Selecting Sync Source and Sampling Rate

When recording digitally, it is necessary to select an appropriate sample rate either independent of other equipment or locked to an external reference. The AD-1000 is very flexible, with very accurate internally-generated sampling rates and its ability to lock to a wide range of external sync sources. The AD-1000 can also act as a stand-alone sync generator.

Internal Crystal Sync

A 'crystal' is a thin piece of quartz cut very precisely to vibrate at a very accurate frequency in an electrical circuit. The crystal inside the AD-1000 vibrates within plus or minus ten parts in a million (PPM) of the specified frequency. This translates to less than 0.5 Hz error at 44.1kHz which is equivalent to less than one-fiftieth of a cent at A440.

When the sync source is set to crystal (**o**), the sample rate switch is active in any of the sample rate positions marked with a solid dot (•). Illumination of the appropriate sample rate LED indicates lock.

Locking to External Word Sync Inputs

The AD-1000 locks to external word sync inputs (also known as Word Clock or WC). In this sync source mode the AD-1000 normally outputs the same frequency as the Word Clock connected to the Sync Input. Termination can be applied using the 75Ω terminator supplied with the AD-1000 and applied to the unused BNC connector. The AD-1000 SAMPLE RATE selector should be in the unmarked position (6 o'clock straight down).

Pull-Up and Pull-Down Sync

Two other positions of the SAMPLE RATE Selector provide the unique ability to increase or decrease the input sampling rate by a 1.001 ratio. Utilizing the above settings, any WC (Word Clock/Sync) input will be multiplied or divided by 1.001. For example a 44.056 kHz WC input will deliver 44.1 kHz locked to the input. Illumination of the 44 LED indicates lock. Users tell us this feature has saved the day in situations where the wrong sample rate was used.

The x1.001 and ÷1.001 positions are marked with a square (■) which corresponds to the square on the WC, AES, OPT AND S/P DIF sync source positions.

1.001 is the ratio between the frequency of a NTSC color signal and the original monochrome standard. When engineers in the USA first developed the NTSC color television system, they found interference problems using the same frequencies as the monochrome system they were replacing. The solution was to slightly shift the color video frequency by 0.1% – thus the 1.001 ratio.

When Compact Discs (CDs) were developed, a reliable method to store the digital audio data in preparation for making the CD was necessary. The solution was to use a video cassette recorder to store the wide bandwidth information. A sampling rate of 44.1 kHz was chosen to pack the digital audio data conveniently on a U-matic video cassette in the form of a video signal. The video signal chosen was the NTSC monochrome standard of 525 lines/60 frames per second (525/60). This CD 44.1 kHz sampling rate is not compatible with a NTSC color video signal (525/59.94 Hz). In fact, 44.1kHz is 1.001 times higher than the 44.056 sampling rate that would be compatible with NTSC video. Therefore to lock a 44.1kHz sampling rate to NTSC video requires multiplying the NTSC reference by 1.001.

Digital video machines (D-1, D-2) use 48 kHz sampling locked to NTSC video or PAL video. F-1 type systems use 44.056 Hz sampling in NTSC versions and 44.1 kHz sampling in PAL versions. As an example where you would need to use ÷1.001: in a music video application, a picture running at 59.94 FPS has to sync with a CD sound track running at 60 FPS (44.1 kHz sampling rate).

AES Sync Inputs

Selecting the AES sync source positions enables the AD-1000 to lock to any AES/EBU digital audio source. **The AES/EBU Sync Source is input via the BNC connector**. No audio information is transferred unless a digital input is selected. The sync information is extracted from the AES/EBU data stream. Two AES positions are available. The position labeled 110Ω AES terminates the input with a 110Ω impedance. You would use the Terminated position if the AD-1000 was the last or the only device in the AES chain. The position with the (ˆ) above it signifies a bridged input of 5kΩ for the AES. This position would be used if the AD-1000 was not the last device in the chain.

Depending on the input sampling rate, the appropriate LED will illuminate to indicate lock. The SAMPLE RATE Selector should be in the 6 o'clock position. In this sync source mode, the AD-1000 normally outputs the same frequency as the AES Signal connected to the Sync Input. Two other positions of the SAMPLE RATE Selector provide the unique ability to increase the input sampling rate by a 1.001 ratio. *See the earlier section on Pull-Up and Pull-Down for details.*

S/PDIF Sync Input

Selecting the S/PDIF sync source position enables the AD-1000 to lock to S/PDIF digital audio sources. No audio information is transferred unless the input selector is set to Digital. The sync information is extracted from the S/PDIF data stream. Apply the S/PDIF input signal to the BNC. If you are not looping signal through to another unit, terminate the signal by selecting the correct switch setting on the 8 position DIP switch located on the top of the AD-1000. *Additional details are on **page 31***. If looping the signal to other devices, the last unit in the chain should be terminated with 75Ω.

The SAMPLE RATE Selector should be in the 6 o'clock – straight down – position. In this sync source mode, the AD-1000 normally outputs the same frequency as the S/PDIF signal connected to the Sync Input. Two other positions of the SAMPLE RATE Selector provide the unique ability to increase the input sampling rate by a 1.001 ratio.

Optical Sync Input

The optical sync source comes in via the optical cable input connector located on the rear panel. If the power switch is in the NORM position, then this sync input acts in the same way as the S/PDIF or AES inputs do. If the AD-1000 is selected to ADAT then the incoming signal is expected to be from an ADAT optical output. *ADAT input is only possible if the special AD1K-PRT option is installed.* The SAMPLE RATE Selector should be in the 6 o'clock position. In this sync source mode, the AD-1000 normally outputs the same frequency as the ADAT signal connected to the Optical Input. Two other positions of the SAMPLE RATE Selector provide the unique ability to increase the input sampling rate by a 1.001 ratio.

Selecting an External Video Sync Source

A video signal is often used as a reference for locking a number of different pieces of audio and/or video equipment together. The video signal is usually a very accurate sync reference and conveniently ties sound to picture. The AD-1000's video sync source input is very flexible and can generate all the various sync output requirements including many of the 0.1% (1.0x) derivatives.

In video applications it is important to match sound and picture. If sound and picture are not synchronized they will drift depending on the differences between the two timing references. Timing accuracy is measured in parts per million (PPM) difference from the ideal. Accuracy for various sources can be gauged by the time it takes for two devices to drift *one frame* out of sync. The approximate times below assume worst-case conditions (one reference fast, the other slow) and a frame period of 33 milliseconds.

Timing Accuracy vs. Drift	1 ppm	=	4.6 hours (approximate)
	10 ppm	=	25.25 minutes (approximate)
	100 ppm	=	2.5 minutes (approximate)

Common Timing Accuracy Standards:	NTSC sync generators hold to 3 ppm
	PAL sync generators hold to 1 ppm
	AES/EBU digital audio holds to 10 ppm
	Consumer digital audio typically holds to 50 ppm, but can vary as much as 1200 ppm

Video Levels

A video signal contains picture information in a sequence of thin lines and additional timing information to synchronize (SYNC) the lines into a complete picture. The total video signal is represented as 1 volt peak-to-peak (pk-pk) when driving into a 75Ω load. The picture part uses 0.7V pk-pk and the sync uses the remaining 0.3V pk-pk. If the video signal is to be used just for sync purposes, a black picture can be used which consists of only the 0.3V pk-pk part. Video sync information can also be distributed as a 4 volts pk-pk signal into 75Ω for NTSC and 2 volts pk-pk into 75Ω for PAL.

In the AD-1000, the three different video standards supported are NTSC (525line/59.94 Hz), PAL (625line/50 Hz), and monochrome NTSC (525line/60 Hz). In each of these standards the input can be video, black video or the higher level sync.

Locking to an NTSC Video Sync Source

Connect the video reference to one of the BNC connectors on the rear panel. Termination is not necessary if looping to another device and the last device in the chain is terminated with 75Ω. *NOTE: The video must be correctly terminated for proper AD-1000 sync operation.* 75Ω termination is available by switching a DIP switch on the top of the AD-1000. Select switch position 1 to ON for 75Ω termination. *(Additional information on this switch is available on* ***page 31****.)* Alternatively, a 75Ω BNC termination can be connected to the unused BNC connector. Select NTSC video on the sync source selector and choose the sample rate with the sample rate selector. Illumination of the appropriate sample rate LED indicates sample rate lock to the external video source.

Locking to a PAL Video Sync Source

Connect the video reference to one of the BNC connectors on the rear panel. Termination is not necessary if looping to another device and the last device in the chain is terminated with 75Ω. NOTE: The video must be correctly terminated for proper AD-1000 sync operation. 75Ω termination is available by switching a DIP switch on the top of the AD-1000. Select switch position 1 to ON for 75Ω termination. *(Additional information on this switch is available on* ***page 31****.)*

Select PAL on the sync source selector and choose the sample rate with the sample rate selector. Illumination of the appropriate sample rate LED indicates sample rate lock to the external video source.

Locking to a 60 Hz Video Source (used in CD premastering)

The AD-1000 has two additional sync source selector positions. These two positions generate the seven output sampling rates from a monochrome video sync source. Monochrome refers to 525 line/60 Hz video. These sync sources can also be used with NTSC video when the need to “bump-up” or “bump-down” is required. The effect will be the same as using the 1.001 positions with a monochrome sync signal.

Opposite is a matrix that summarizes all of the possible sync combinations. Those combinations that are “illegal”, such as 48.048 locked to NTSC will blink the three sample rate indicators to indicate an error in the setup.

Digital Audio Output Format and Selection

The AD-1000 features many different digital audio output formats. The front panel power switch is a three position switch located on the front upper right hand corner of the AD-1000. The center position selects power OFF. The other two positions select either NORM or ADAT output format. The selected output format appears at the AES (male XLR), S/PDIF 15 pin HD connector and OPTICAL (optical plug nearest the edge of unit) outputs simultaneously with the appropriate format, level and impedance. For example, with the front panel switch selected to NORM, the XLR will output AES and the S/PDIF and OPTICAL outputs will output S/PDIF format. The AES output will be +4 volts pk-pk/balanced, unterminated, and the S/PDIF output will be 0.5 volts pk-pk /75Ω/balanced (S/PDIF will drive balanced and unbalanced loads). When selected to ADAT, output is ADAT format on optical, with “Audio Black” clock signal (no data bits) output on the AES and S/PDIF outputs.

AES/EBU – Professional digital audio transmission format.

Transformer-isolated balanced line output to a three-wire transmission line.
Output Impedance: 110Ω
Signal amplitude: 4.4 Volts peak to peak into 110Ω
Sampling Rate Range: 32–54kHz
Connector: male XLR mounted on rear panel, pin 1 ground; pins 2 & 3 signal

Note: Polarity of pins 2 and 3 is not important. Reversing pins 2 and 3 will not affect the digital audio transmission because the format is insensitive to polarity.

Sync Source	Pwr Switch	Valid Sample Rate Selector Positions								
		32	44.056	44.1	44.144	47.952	48	48.048	1.001 Postions	Unmarked Postions
Internal Crystal	Either	●	●	●	●	●	●	●		
Word Clock	Either								●	●
AES Sync In	Either								●	●
SPDIF Sync In	Either								●	●
SPDIF Optical In	NOR								●	●
ADAT Optical In	ADA								●	●
NTSC Video In	Either	●	●	●		●	●			
PAL Video In	Either	●	●	●	●	●	●	●		
60 Hz Video In	Either	●	●	●	●	●	●	●		

S/PDIF (Sony/Philips Digital Interface) – Consumer digital interface

Standardized as the CP-340 digital audio interface by the EIAJ (Electronic Industries Association of Japan) and IEC 958, (International Electrotechnical Commission)

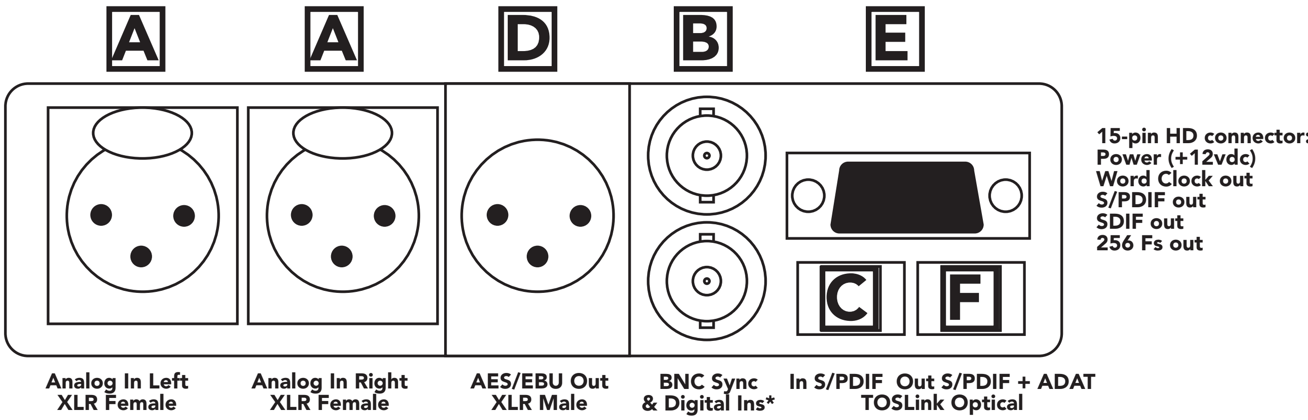
Transformer isolated balanced or unbalanced output to a two wire transmission line.
Output impedance: 75Ω
Signal amplitude: 0.5 Volts peak-to-peak into 75Ω
Sampling Rate Range: 32–54 kHz
Connector: 15 pin HD connector mounted on rear panel, Pin 1: Signal; Pin 6: Shell
(Note: requires adapter cable and/or adaptor to standard female 'RCA' connector or BNC)

OPTICAL – Digital optical interface standard for consumer applications (TosLink) and ADAT (multitrack)

Transmitter characteristics:
Peak emission wavelength 660nm ±30nm
Peak emission power between –15dBm and –21dBm (when measured at the edge of the reference optical fiber which is connected to the output terminal of the fiber optic transmitter)
Connector: standard to 'TOSLINK' style optical interconnect.

Rear Panel Functions and Locations

Analog Audio Inputs



***Word Clock, video; AES sync, S/PDIF sync; AES Audio, S/PDIF Audio. Both connectors tied together (loop through) Unused connector should be terminated.**

Viewing the rear panel of the AD-1000, the female XLR connector on the left is the left (or Channel A) analog audio input. The XLR on the right is the right (or Channel B) analog audio input. (These are labeled "A" on the diagram above.) *The AD-1000 is set up for the worldwide standard pin 2 hot as shipped.* A positive-going voltage into pin 2 will produce a digital audio output of the correct polarity.

Note that some equipment, especially in the USA, remains pin 3 hot. The AD-1000 can be configured for pin 3 hot. This is done by changing the position of switch 5 to of the eight pin DIP switch package located on the top of the AD-1000. ON corresponds to Pin 3 hot and OFF to Pin 2 hot. The Red Pin 3 LED on the front panel will illuminate to warn of the non-standard configuration.

Digital Outputs

The AD-1000 takes in analog audio and outputs digital audio in three formats as standard. The output format is selected on the front panel power switch (additional formats are optional)

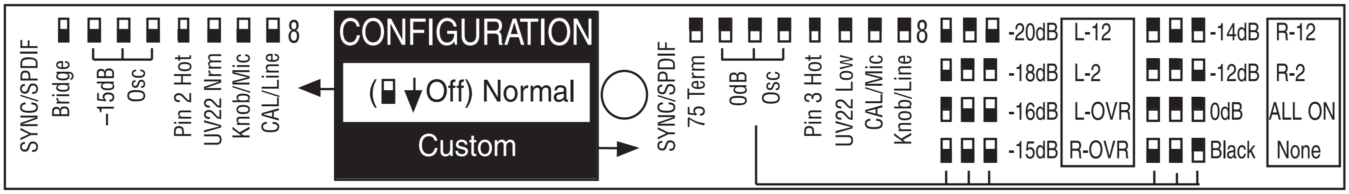
a) AES/EBU for professional applications (NORM Position)
b) S/PDIF for consumer applications (Both coaxial and optical) (NORM position)
c) Alesis ADAT for use with ADAT Optical Digital Formats (ADAT Position)

The AD-1000 simultaneously outputs a professional format digital audio output on the AES/EBU Male XLR connector (D) and S/PDIF format in optical (F) and coaxial (B) formats. The coaxial consumer output (S/PDIF) is available on the 15 pin HD connector (E) on the rear panel via an adapter cable. *Diagrams providing additional information on these cables are located on **pages 44 and 45**.*

Balanced and unbalanced word clock outputs are also available on the rear panel 15 pin HD connector. These outputs can be used to synchronize a wide variety of other digital audio products *(see **pages 44–45**)*.

Advanced Operations

The DIP Switches – Location and Function



The black part of the DIP switches as indicated above represents the actuator of the switch. It may not be this color on the unit.

There are eight small switches located on the top of the unit that are accessible with a small screwdriver or pin The default position is with all switches set to the off position. The switches serve the following functions.

Switch 1 Selects the termination of the SYNC-S/PDIF BNC connector located on the back of the AD-1000
Off Sync/SPDIF input is bridged with > 5kΩ
On Sync/SPDIF input is terminated with 75Ω

Switch 2, 3 and 4 Select the oscillator level output and –12 LED Threshold.
(The related Front Panel LED will light when in Oscillator mode.)

Level Below dB Full Scale	***Switch 2***	***Switch 3***	***Switch 4***	***LED Display***
20	Off	On	Off	Left –12
18	Off	On	On	Left –2
16	On	Off	Off	Left Over
15	Off	Off	Off	Right Over
14	On	Off	On	Right –12
12	On	On	Off	Right –2
0	On	On	On	All On
Audio Black	Off	Off	On	All Off

Switch 5 Selects the polarity of the Analog XLR Inputs
Off Pin 2 Hot (Pin 3 LED off*)
On Pin 3 Hot (Pin 3 LED on*)

Switch 6 Selects the UV22 Process Level (Appendix I, page 46 has more information regarding this selection)
Off Normal*
On Low*

Switch 7 Selects which gain potentiometer controls the microphone input level.
Off Front Panel Knob
On Multi-turn Calibration Pot

Switch 8 Selects which gain potentiometer controls the line input level.
Off Multi-turn Calibration Pot
On Front Panel Knob

**To differentiate between Pin 2 and Pin 3 Hot while in Low UV22 Mode, the following will occur: When Low UV22 and Pin 3 Hot are set, the LED will turn OFF quickly every second or so. With Low UV22 and Pin 2 Hot, the LED will turn ON quickly every second or so.*

Technical Input Information

Input Impedance
The AD-1000 is set up to easily match and interface to most any analog audio input. The AD-1000 input assumes the source impedance (from the unit driving the AD-1000) will be less than 600Ω and probably more like output impedances of 50 to 100Ω. *In the very unlikely event of using higher source impedances such as from some consumer and musical instrument sources, the frequency response at the very highest frequencies will increasingly be rolled off gradually with increasing source impedance.* Professional audio equipment output impedance is almost always less than 100Ω.

Each side of a balanced input encounters a RF filter and then what appears to be a purely resistive input to ground of >5kΩ. This makes the input impedance greater than 10kΩ for balanced inputs and greater than >5kΩ for unbalanced inputs. This is considered to be a bridging input because it sits like a bridge over the output driving it without any significant loading effect. This permits any low impedance output to drive many such bridging inputs.

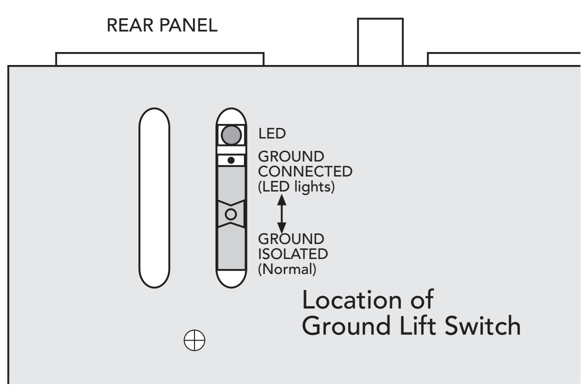
Balanced or Unbalanced Inputs
The AD-1000 input will accept both balanced and unbalanced inputs. No internal jumper selectors are required. The maximum input level is +28 dBu balanced and +24 dBu unbalanced. This is the level which will produce a full scale digital output with the multi-turn CAL pots in their minimum position.

DC Removal
The input to the AD-1000 is direct coupled in the Line +4 and –10 CAL positions: no capacitors are in the signal path. This means the DC inputs will be passed through the direct coupled stages all the way to the conversion stage. The AD-1000 has very effective DC servos to remove both common mode and differential DC inputs well beyond the specified 50mV DC common mode and 50mV differential. The servos remove any residual DC from the digital output to beyond the 16th LSB level to deliver a totally centered digital output. This eliminates the need with most converters to trim for DC removal or run through an external high pass filter which, more often than not, compromises the sonic integrity of the analog input. The Apogee approach goes back to basics for a simple method of extracting DC without the attendant number-crunching of a digital high pass filter.

Common Mode Rejection
Common Mode Rejection is a measure of how well an input ignores interference picked up on pin 2 and 3 together. The AD-1000 discrete input stage features excellent common mode rejection. With a well-balanced input, the common mode rejection at 100 Hz is typically better than 110 dB and at 10 KHz is typically better than 80 dB. (Reference Settings: +15 dBu input signal and –0.5dB f.s. digital audio output.) This common mode performance assures quiet performance in the presence of external noise and interference.

Total Harmonic Distortion and Noise
Technically, analog circuitry is often judged by how little distortion and noise it adds to an input. In reality, this is a good indication, but not the entire story. At Apogee, we trust listening tests over measurement. The AD-1000 gives the best of both worlds: it sounds great and measures great! The entire analog section of the AD-1000 typically has total harmonic distortion plus noise performance better than 0.001% with Soft Limit switched out. (Soft Limit only affects the distortion over the last 2dB.)

Chassis Grounding
A switch located on the top rear of the AD-1000 allows the chassis and audio grounds to be connected together. The default position is pointing toward the front of the unit, which isolates the chassis and audio grounds. If you want the grounds connected to remove a grounding difficulty, moving the switch to the back position will accomplish this. An LED will light to indicate that you are operating in a non-standard mode.



Location of
Ground Lift Switch

Precise Calibration using One LED

When you need to make a mix from your analog mixing console, it is necessary to calibrate your AD-1000 to the output of the mixing console. The AD-1000 incorporates a unique feature to enable you to simply match the digital output to within a hair width of your console's meters. The –12 LED is much more flexible than the labeling would make it appear. When you input an analog signal into the AD-1000, the –12 LED's threshold becomes variable and coincides with the headroom setting of the digital oscillator. This permits precise matching of an analog input to the internal digital oscillator.

If the internal digital oscillator is set to the default –15dBfs position (all three switches in default off) the –12 LED will remain off until the input level reaches –15dB (Peak) below full scale digital output. Passing the threshold turns the LED On. In addition, the LED will blink rapidly to tell you when the signal is within the 0.05dB of the –15dB threshold.

If you are inputing a calibrated analog sine wave oscillator tone from your console (typically +4dBu for an analog meter zero) you can now adjust the selected gain pots on the AD1000 to make the "–12" LED blink. You have now calibrated the AD-1000 to within ±0.05dB of the "perfect" digital oscillator. Switching between the analog input (such as +4) and the digital oscillator will show perfect calibration.

The calibrated headroom can be varied to best suit your application by changing the digital oscillator headroom. The blinking point for the "–12" LED moves right along with it. The mastering world usually chooses –12 or –14dB, tracking tends to be done with –15 to –8 dB and the film world tends to play it safe with up to 20dB of headroom.

Headroom

(See diagrams overleaf.) A hotter-sounding compact disc can be the difference between having a hit or being forgotten. A hotter-sounding CD means not wasting headroom. In analog recording we define a nominal operating level and allow enough *headroom* above to avoid clipping the analog circuitry. This nominal level is usually referred to as 'zero' for the 0dB calibration on analog meters. The analog zero usually represents a nominal +4 dBu output level, i.e. when the meter indicates 0 it is really putting out a level of +4 dBu.

With digital audio, the precise distortion or clipping point is known. This is the point where we run out of numbers to represent the analog input. This maximum positive (or negative) level is often called an 'over' due to the popular labeling of digital meters. The 'over' indicators illuminate when a digital maximum is reached, usually for a total of more than 3 samples in a row.

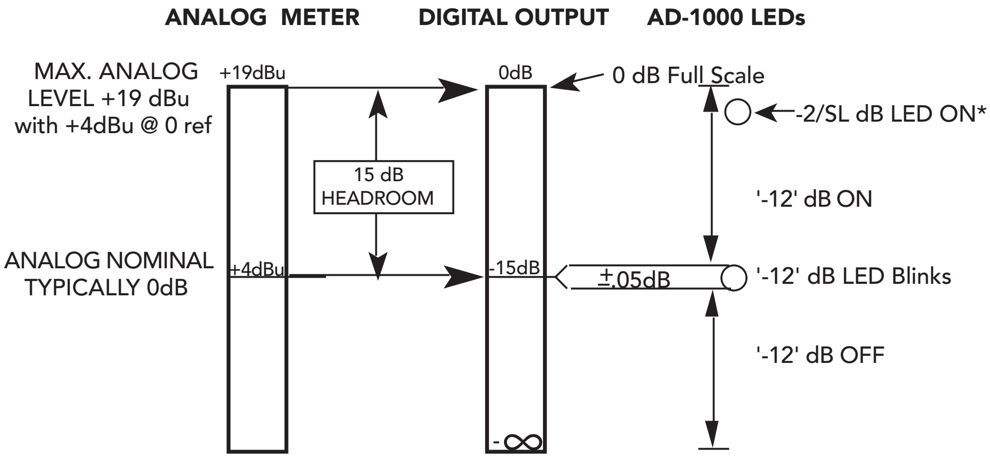
NOTE: Some DAT recorders such as the Panasonic 3700 and 3900, indicate 'overs' with analog inputs, but do not indicate them with digital inputs such as from the AD-1000.

In digital audio we must decide on how much headroom we want above our nominal level (the zero from analog world) before we hit an 'over' or digital clip. The aim is to use as much of the dynamic range as possible. Any wasted headroom means we are closer to the noise floor than necessary.

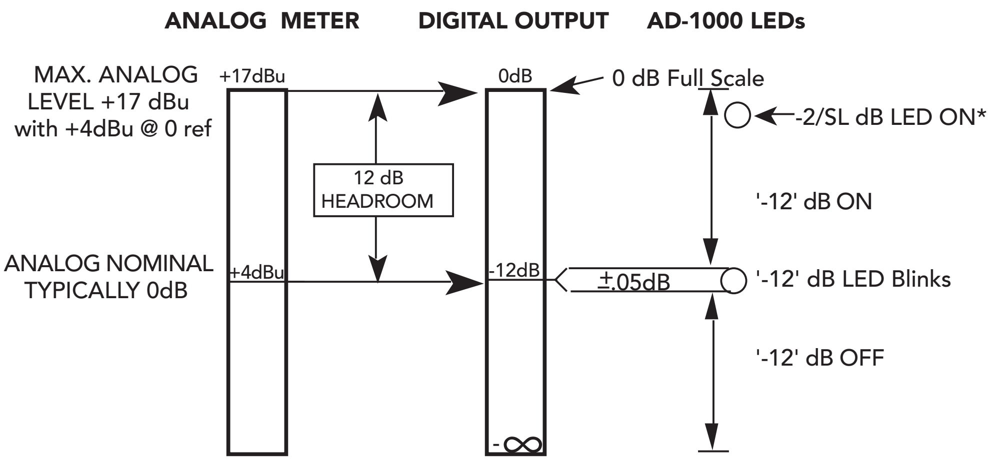
We require more or less headroom, depending on the material being recorded. Mastering engineers typically choose 12 or 14 dB headroom over their nominal input level because they usually have their dynamics tightly controlled. In tracking situations, 15 and 16 dB are the most popular, with some users going as high as 18 and 20 dB. The headroom of analog inputs or some DAT recorders are often fixed, such as at 18 dB for the Panasonic *3700* and *3900.* When using the AD-1000, the headroom is easily adjusted with the front panel controls (purple knobs – essential for high-quality digital audio) or front panel multi–turn pots, and the built in oscillator for generating the a digital test tone at a known digital level and using the precise threshold to input tones and adjust the gain pot or CAL pot. .

Digital audio levels are often referred to the maximum level (or full scale, f.s.). "Zero dB full scale" is a maximum level when referring to digital levels. With 16dB headroom, the nominal level would be then sitting down at –16dB referenced to full scale (f.s.)

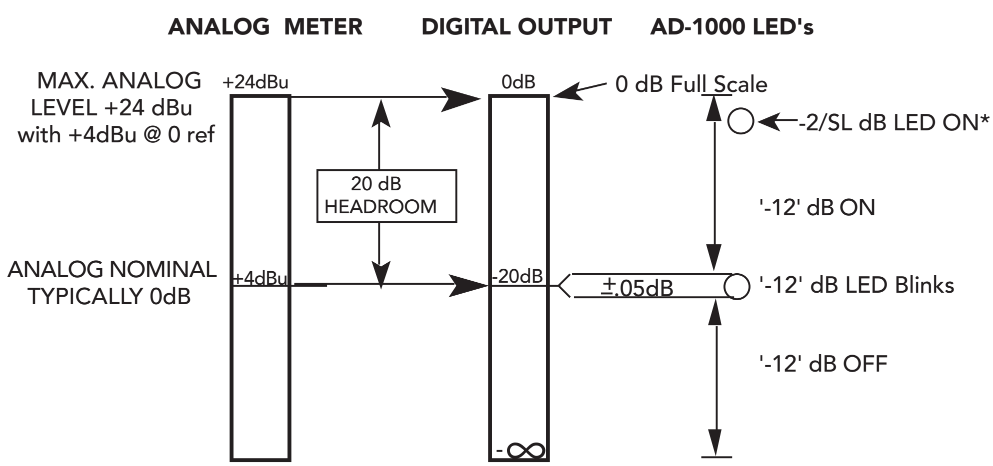
Digital Headroom Explained



COMPARISON OF ANALOG INPUT TO DIGITAL OUTPUT
WITH 15dB HEADROOM AND +4 dBu REFERENCE



COMPARISON OF ANALOG INPUT TO DIGITAL OUTPUT
WITH 12dB HEADROOM AND +4 dBu REFERENCE



COMPARISON OF ANALOG INPUT TO DIGITAL OUTPUT
WITH 20dB HEADROOM AND +4 dBu REFERENCE

*Dependent upon SoftLimit Threshold and activity.

Options and Enhancements

AD1000 UV22 Digital Through Option

The UV22 digital through option gives the AD-1000 the ability to process digital input signals with Apogee's award-winning UV22 encoding technique, which preserves the audio quality of 20-bit signals in a 16-bit environment. (The standard AD-1000 includes UV22 encoding, but only on analog input signals. Digital signals may be format-converted but not encoded in the standard model.) This option allows a studio to work entirely at 20-bit resolution and then run the completed project through the AD-1000 to apply UV22 encoding down to 16-bits.

Operation:
1. Connect your digital audio source to the AD-1000 to the top BNC port on the back of the AD-1000 for AES/EBU* or S/PDIF operation. Optical input is available via the TosLink connector on the back of the AD-1000.
** NOTE: A special cable (Apogee part # WE-BX-0.5FT) is required to interface an AES/EBU XLR cable to the BNC port.*

2. Set the INPUT SELECTOR to the DIG position.

3. Set the SYNC SOURCE to 110Ω AES, S/PDIF, or OPT. (depending on type of digital audio source.)

NOTES:
- *S/PDIF must be terminated with 75 Ω by selecting the DIP switch 1 to ON (located on top of the AD-1000). Alternatively, a 75 Ω BNC termination (supplied with the AD-1000) can be connected to the unused BNC connector. Termination is not necessary if looping to another device, as the last device in the chain is then terminated.*

- *Two AES positions are available. The position labeled 110 Ω AES terminates the input with a 110 Ω impedance. The AES position with the ^ above it signifies a bridged input of 5 kΩ for the AES. This position would be used if the AD-1000 was not the last device in the chain.*

4. Set SAMPLE RATE to the 6 o'clock (unmarked) position.

5. Set the THREE POSITION SWITCH (located between the Sync Source and Sample Rate selectors) to either 16, UV16 or 20 as desired.

When inputting a 20 bit signal:
- the 20 position outputs 20 bits with no processing.
- the UV16 position outputs 16 bits with UV22 encoding.
- the 16 position outputs 16 bits with triangular dither.

When inputting a 16 bit signal:
- the 20 position outputs 16 bits unprocessed (whatever signal is input).
- the UV16 position outputs 16 bits with UV22 encoding.
- the 16 position output 16 bits with triangular dither.

6. NORM / ADAT Switch:
Selecting ADAT transmits the ADAT output on the Optical digital output.
When NORM is selected, the AD-1000 simultaneously outputs AES/EBU (Male XLR) and S/PDIF in optical and coaxial formats.

General notes on ADAT Recording

The AD-1000 has the ability to output the ADAT digital format (on Toslink Optical) allowing you to bypass the A/D converters in the ADAT recorder and get the legendary Apogee sound on ADAT. The following pages contain information on how to accomplish this in a number of different ways. There are, however, some basic concepts that need to be understood first.

The most common questions revolve around the fact that the AD-1000 is a two channel device and the ADAT is an eight track recorder. You will notice that the AD-1000 has only one optical output (and one optical input) and the ADAT has the same number and type of optical connectors. When in ADAT mode (power switch to "ADAT"), the ADAT transmitter in the AD-1000 transmits eight channels of information which is just the original two channels (left and right) duplicated four times. If you were to put all eight tracks of an ADAT into "input," you will see four pairs of information with the left channel of the AD-1000 going to tracks 1, 3, 5 and 7 of the ADAT and the right channel of the AD-1000 going to tracks 2, 4, 6 and 8 of the ADAT. Thus, you can record onto tracks 3 and 8, or 5 and 6, or 1 and 4, or 7 and 8, and so on. This of course, begs the question of "How do you record four different channels worth of music simultaneously on to four tracks of the ADAT" (assuming that you have two AD-1000s)? Unfortunately, you can't. At least not on one ADAT. Remember, the AD-1000 is only a two channel device and there is only one connector, so there is no way to get those other channels into the ADAT. You can connect as many AD-1000s as you wish to that many ADATs (*see diagram on "Using Multiple AD-1000s"*) and record lots of tracks simultaneously, but only two channels per ADAT. Obviously, you can go back and overdub on the remaining tracks.

This brings up the question of using multiple ADATs controlled by a BRC (Big Remote Controller). Recording on to a single ADAT (with no BRC) is easy *(see note on "Recording Without a BRC"– later in this section).* Any time a BRC enters the picture, even with a single ADAT, sync must be provided to (or from) the BRC. For digital equipment to communicate properly, all the gear must be running synchronously, that is they must all have their clocks lined up so that they do exactly the same things at exactly the same time. The BRC provides sync information to the ADAT; therefore if the AD-1000 is providing a signal to the ADAT, the BRC must have the same timing (clocking) information as the AD-1000. Imagine having two different drummers in two different rooms playing the same song. They are both providing a beat (clock) to the song, but in order for the song to work they have to play together, therefore it is necessary to provide both of them with the same information (sync) and they need to be able to hear each other. Digital audio gear must all be "playing together" to work properly. A separate timing signal (usually Word Clock or video sync) is used to accomplish this. Usually one piece of gear is designated the Master and everything else "slaves" to this master clock. The AD-1000 is remarkable in that it can either be a "master" or a "slave" depending on the requirements. *Please see notes on "Recording With a BRC."*

The last (hopefully) question that needs to be addressed is "How does the AD-1000 receive ADAT?" That's easy – it doesn't. Remember that ADAT is an eight channel format and the AD-1000 is a two channel device. The basic AD-1000 cannot decode this information and make anything sensible of it, nor does it have a way to "pass it through" so the AD-1000 cannot be hooked up in the ADAT optical loop. *(The special version AD1K-ADT can decode this information and convert it to AES/EBU format two channels at a time. See later in this section.)* It is recommended that the AD-1000 be connected (via optical) only to the particular ADAT being recorded onto. The optical input on a standard AD-1000 is not an ADAT input.

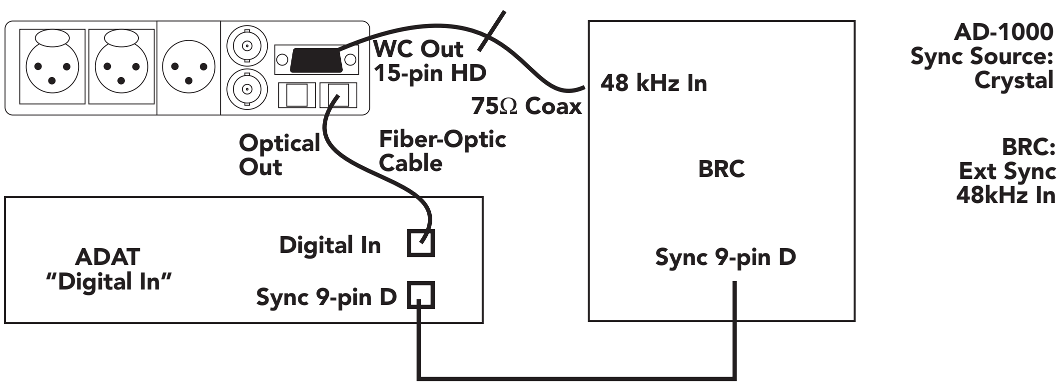
The AD-1000 will output AES "black" (Audio Black – clock signal with no data) from the AES output when in ADAT mode. The ADAT format is a 16-bit format. That is, it records with 16-bit resolution per track. The AD-1000 is a 20-bit converter and can output 20-bit data even in ADAT mode. The AD-1000 gives you three choices for output resolution – 16, UV16, and 20 (16-bit triangular dithered output, 16-bit UV22 processed output, and unprocessed 20-bit output, respectively). With a 20-bit output, the ADAT will truncate (cut off) the "extra" 4 bits, which obviously is not the way to go, so you are left with a choice of "UV or not UV?" *We generally recommend that the UV process not be used twice on a signal that remains in the digital domain* (does not become analog again) and we recommend against doing any digital signal processing after the signal has been UV encoded *(please see UV22 Caveats **page 49**).*

However, many people have come to depend on the superior audio quality of using UV22 to record their 16-bit tracks, and report no significant audio damage by using UV22 again when the recording is mastered. The bottom line here is that you need to be the judge of your own recordings. Listen to the music as it goes through its various stages of tracking, mixing, editing, sequencing and mastering. If you like what you hear, then its OK. If you don't, then its not. Music is art, not science.

Recording with the AD-1000 to ADAT (without the BRC)

1. Connect the output from your console or microphone to the AD-1000 XLR input connectors. (When viewing the AD-1000 from the rear, the Left XLR is on the left side)
2. Select any analog input (+4, –10 or MIC) on input select switch.
3. Set the front power switch of the AD-1000 to the ADAT position.
4. Select the SYNC SOURCE to CRYSTAL.
5. Select the SAMPLE RATE selector to desired setting (44.1 or 48kHz).
6. Connect the optical cable from your AD-1000 optical output to the ADAT optical input of the destination ADAT (When viewing the AD-1000 from the rear, the optical output is on is on the right side).
7. Set the ADAT unit to DIGITAL IN.
8. Arm the desired tracks of the ADAT (left channels are odd numbers 1,3,5,7 and right channels are even numbers 2,4,6,8).

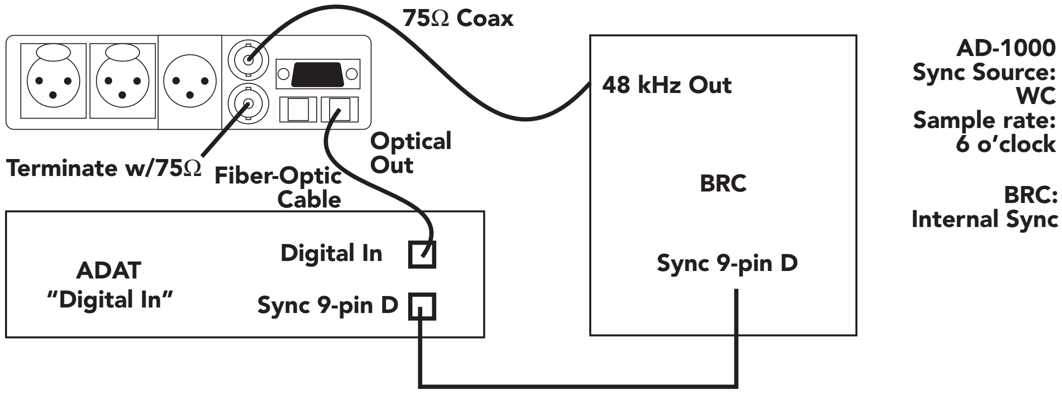
Recording with the AD-1000 as the master (BRC as the Slave)



ADAT w/BRC – AD-1000 as Master

1. Connect the output from your console or microphone to the AD-1000 XLR input connectors. (When viewing the AD-1000 from the rear, the Left XLR is on the left side)
2. Select any analog input (+4, –10 or MIC) on the input select switch.
3. Set the front power switch of the AD-1000 to the ADAT position.
4. Set the SYNC SOURCE to CRYSTAL.
5. Set the SAMPLE RATE selector to the desired setting (44.1 or 48kHz).
6. Connect the word clock cable (75Ω coax) from the WC output of the AD-1000 (on the 15 pin HD connector) to the 48 kHz input of the BRC. *A special cable "PS-1000/AD-1000 Cable" is required for this. It may be purchased from your Apogee dealer or see the diagram to build it yourself.*
7. Set the BRC to external sync. (EDIT, EXT SYNC to 48K Input)
8. Connect the optical cable from your AD-1000 optical output to the optical ADAT input of the destination ADAT . (When viewing the AD-1000 from the rear, the optical output is on is on the right side).
9. Set the ADAT unit to DIGITAL IN.
10. Arm the desired tracks of the ADAT (left channels are odd numbers 1,3,5,7 and right channels are even numbers 2,4,6,8).

Recording with the BRC as the master (AD-1000 as the slave)



ADAT w/BRC – BRC as Master

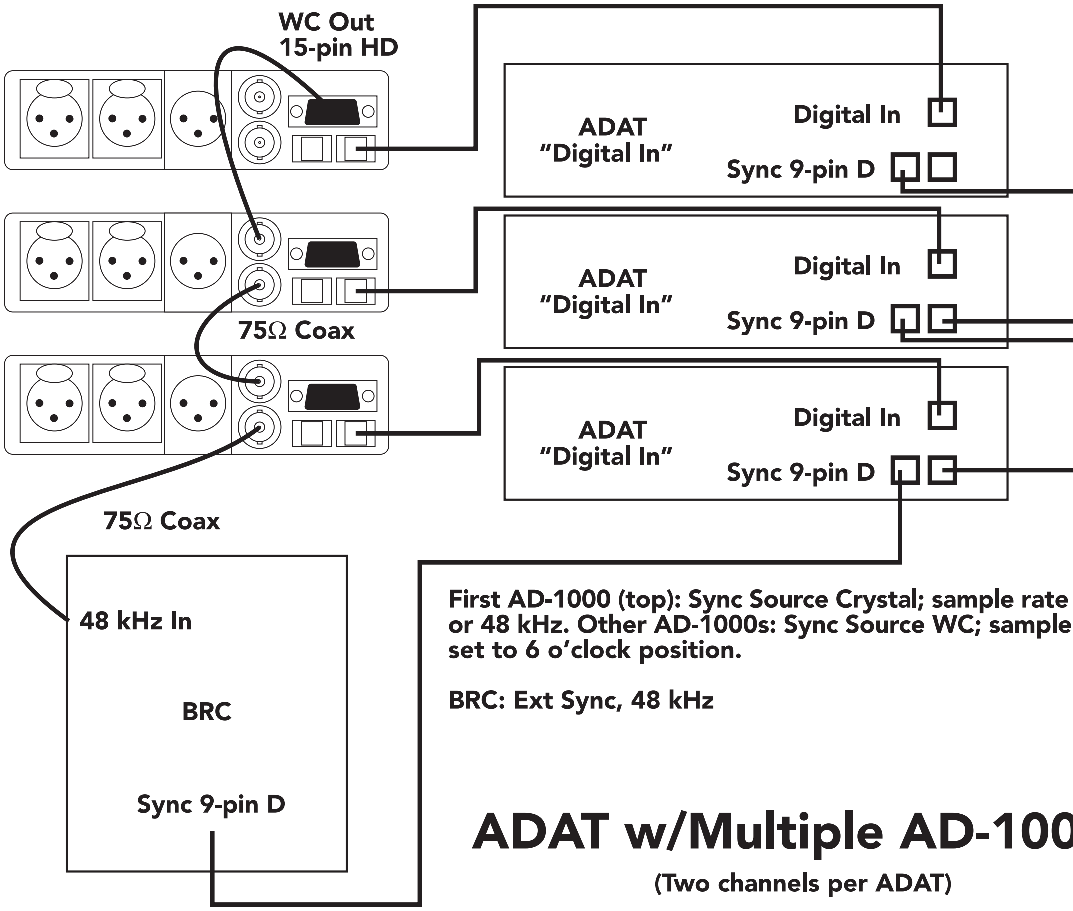
1. Connect the output from your console or microphone to the AD-1000 XLR input connectors. (When viewing the AD-1000 from the rear, the Left XLR is on the left side)
2. Select any analog input (+4, –10 or MIC) on input select switch.
3. Set the front power switch of the AD-1000 to the ADAT postition.
4. Set the SYNC SOURCE to WC.
5. Set the SAMPLE RATE selector to the unmarked position (6 o'clock - straight down).
6. Connect the word clock cable (75 Ω coax) from the 48 kHz output of the BRC to the AD-1000 top BNC connector. Terminate the unused BNC with a 75Ω terminator (supplied in the ship kit).
7. Set the BRC to internal sync.
8. Connect the optical cable from your AD-1000 optical output to the optical ADAT input of the destination ADAT . (When viewing the AD-1000 from the rear, the optical output is on is on the right side).
9. Set the ADAT unit to DIGITAL IN.
10. Arm the desired tracks of the ADAT (left channels are odd numbers and right channels are even numbers).

Resolving ADAT to Video with the AD-1000

Using good quality (Apogee) 75Ω cable, connect a good quality video sync (blackburst) generator to one of the sync inputs (BNC connectors on the rear panel) of the AD-1000. Using the other BNC on the AD-1000, which is connected as a loop through, connect a cable to the VIDEO IN connector (BNC) on the BRC. The BRC is necessary to reference the ADAT to video sync.
Set the BRC to EXTERNAL VIDEO (EDIT, EXT SYNC, VIDEO).
On the AD-1000, select NTSC SYNC on the SYNC SOURCE switch.
The power switch should be set to ADAT.
Set the SAMPLE RATE switch to either 44.1kHz or 48kHz.
The ADAT should be set to DIGITAL IN.

20-bit Recording Using the ADAT

With the addition of the 20-bit bit-splitting option *(AD1K-PRT)*, the Apogee AD-1000 has the ability to record a true 20-bit signal on to tape. Since current modular digital multitracks (MDMs) can record a maximum of 16-bits per track, the 20-bit signal is split onto two tracks, with the 16 MSBs (most significant bits) being recorded onto track one and the 4 remaining LSBs (least significant bits), plus a low level 1 kHz square wave, being recorded on to track two – the tone indicating that the signal on this track is not designed to be used alone.This information makes up the 20-bit signal of the "A" channel (left). The process is repeated for the "B" channel (right) information, with the MSBs recorded onto track three and the LSBs (and low level 1 kHz tone) recorded onto track four. Alternately (or additionally) this can be done on tracks five through eight. Thus we have a two channel 20-bit recording on four tracks. The information given here applies to ADAT format recorders: DA-88-compatible machines can be used with the addition of the Apogee FC-8 Format Converter.

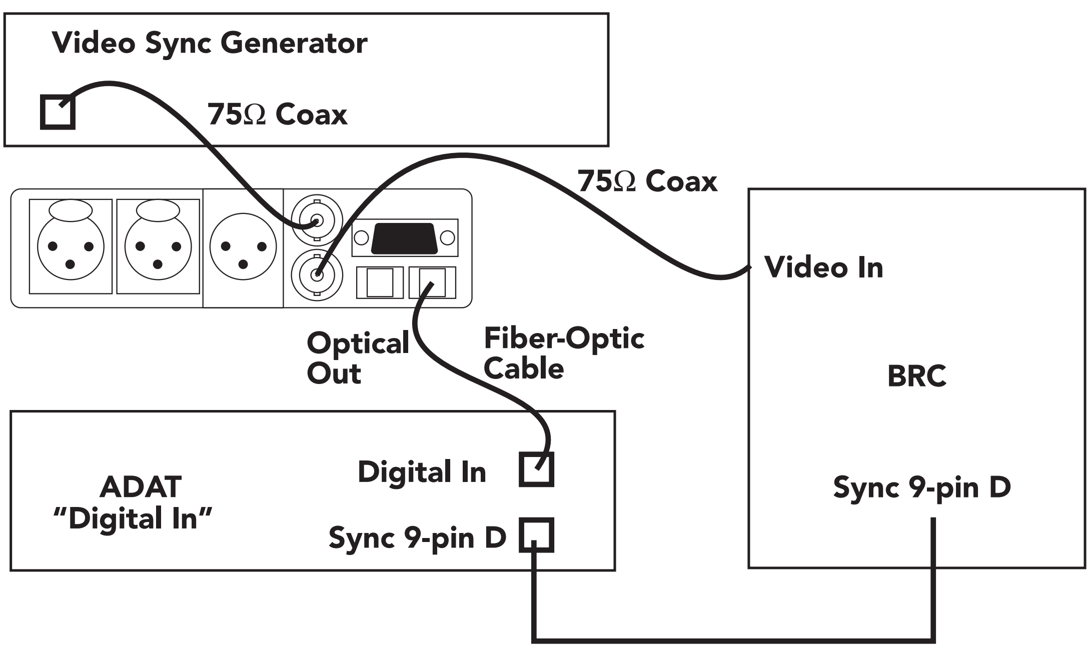


First AD-1000 (top): Sync Source Crystal; sample rate 44.1 or 48 kHz. Other AD-1000s: Sync Source WC; sample rate set to 6 o'clock position.

BRC: Ext Sync, 48 kHz

ADAT w/Multiple AD-1000s

(Two channels per ADAT)



AD-1000
Sync Source:
NTSC
Sample rate:
44.1, 48 kHz
Termination:
Bridge

BRC:
Ext Sync
Video

ADAT w/VIDEO SYNC

Recording

To record a true 20 bit signal onto an ADAT recorder (which is a 16-bit recorder) the following steps must be taken:

1. Connect optical cable from AD-1000 optical output to the optical input of the destination ADAT. Set ADAT to "digital input." Note: the optical cables cannot be "looped thru" ie, multiple input to output connections made as in "digital track- bouncing" mode. Only connect between AD-1000 and the machine to be recorded on to.
2. Select any analog input (+4, –10 or mic pre) or the oscillator.
3. Select the appropriate sync source (crystal, WC, video, etc.).
4. The 16 / UV16 / 20 switch should be set to 20 (20-bit output).
5. Select the sample rate of choice. Please note that ADAT will only accept either 44.1 kHz or 48 kHz, so if you select any variation of 44.1 kHz (44.056 or 44.144) you will still only get 44.1. Likewise with 48 kHz – selecting any of 47.952, 48, or 48.048 will get you only 48 kHz! Also note that when using an external sync source (WC, video, AES, etc.) the gear box function (x1.001 or ÷1.001) does not operate. Further note that when referenced to NTSC video, both sample rates will be pulled down to either 44.056kHz or 47.952kHz.
6. Lastly, the power switch must be switched to the ADAT position – to the right, opposite of NORM.
7. Eight channels of information will be transmitted (two stereo pairs): you will need to select four adjacent tracks, either 1 thru 4 or 5 thru 8, to be able to properly decode this information for playback.

Playback

To play back this special signal format, it must first be recombined into a "normal" 20-bit signal (AES or S/PDIF). It can then be UV22'd to a CD recorder or converted to analog by one of our DA-1000E-20 digital to analog converters.

1. Connect an optical cable from the ADAT output to the AD-1000 optical input. Please see above note about "looping."
2. Select "DIG" (digital input) on the Input Selector Switch.
3. Select "OPT" (optical) on the Sync Source Switch.
4. The 16 / UV16 / 20 switch will select the level of resolution at either the AES or S/PDIF (optical) outputs. Note that in this mode the optical output is in the S/PDIF format, not ADAT.
5. The power switch should be selected to ADAT.
6. The Sample Rate switch now determines which of the eight tracks are received and in what form. The following chart shows the various possible combinations.

Sample Rate Switch Position	***Track***	***Format***	***Channel***
1 o'clock (47.952)	1	16 bit	A
	2	4 bit	B
2 o'clock (48)	3	16 bit	A
	4	4 bit	B
3 o'clock (48.048)	5	16 bit	A
	6	4 bit	B
4 o'clock (x1.001)	7	16 bit	A
	8	4 bit	B
5 o'clock (unmarked)	1&2	20 bit	A
	3&4	20 bit	B
6 o'clock (unmarked)	5&6	20 bit	A
	7&8	20 bit	B

In positions 1 o'clock to 4 o'clock, the output of the AD-1000 (AES or S/PDIF) will be a 16-bit signal on channel A (left) and the 4-bit plus tone signal on channel B (right) from each of the selected track pairs. In positions 5 and six o'clock, the output (AES or S/PDIF) will be a "normal" (recombined) 20-bit two-channel signal from the selected group of four tracks.

Special notes and restrictions:
1. *ADAT format cannot be transmitted while the AD-1000 is set to digital input "DIG."*
2. *ADAT format cannot be transmitted and received simultaneously. As a result, the AD-1000 cannot be connected in an optical "loop."*
3. *While transmitting ADAT, the AD1000 outputs AES "black" - a sync signal with no data.*
4. *While synced to video, the AD-1000 will only transmit 44.056 kHz or 47.952 kHz sample rates in ADAT mode.*
5. *While synced to either AES or S/PDIF, the AD-1000 will only output 44.100 kHz or 48 kHz sample rates in ADAT mode.*
6. *While synced to an external Word Clock, any sample rate received can be transmitted.*

Operating the AD-1000 with the Tascam DA-88

The Tascam DA-88, an eight channel modular digital multitrack (MDM) recorder, has only the proprietary TDIF (Tascam Digital Interface) for its digital inputs. TDIF is specified as a 25-pin "D" connector which would take up almost half of the AD-1000's rear panel. Since ADAT is also a eight-channel digital format which uses only the small optical Toslink connector, Apogee decided it would be easiest to output the ADAT format and then convert that signal to the TDIF format external to the AD-1000.

Thus was born the FC-8, Apogee's ADAT to TDIF (and vice-versa) format converter. The FC-8 is the simplest and best way to get into a DA-88 digitally with the AD-1000. There are other devices that convert AES/EBU into two channels of TDIF, but the FC-8 can be viewed as an extension of the AD-1000 itself and was designed to be just that.

It would be useful to read the previous discussions regarding using the AD-1000 with ADAT, as the same basic principles apply (i.e. the AD-1000 being a two channel device and the DA-88 being an eight- track recorder) and in fact, you will primarily be using the ADAT output of the AD-1000 (and converting it to TDIF). Using the DA-88 is somewhat more simple in that there is no BRC or master controller to contend with, but care must be taken to ensure that all devices in the digital recording chain are properly referenced.

When recording onto a DA-88 digitally, a separate word clock must be provided to the DA-88, since the TDIF receiver does not support a clock signal. The FC-8 will sync to the incoming ADAT signal and provide a Word Clock output to be connected to the DA-88 Word Sync input. If using the IF-88AE (Tascam's AES to TDIF Converter), it will sync to the incoming AES and provide a Word Clock output. In both cases the DA-88 Clock should be set to "WORD".

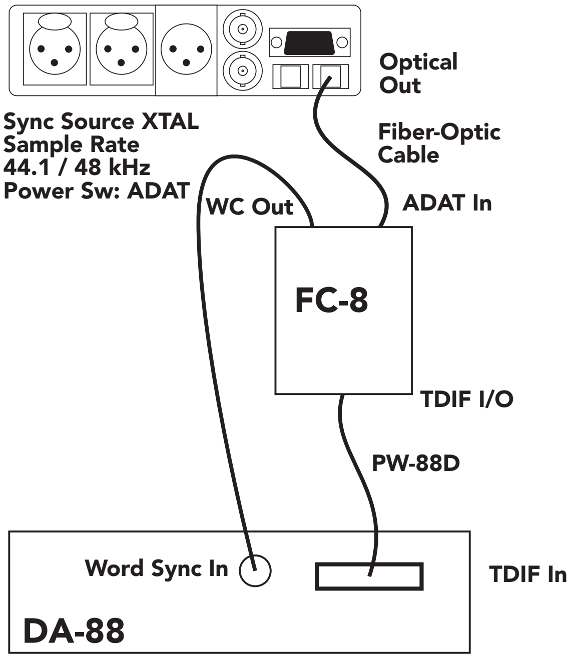
Recording on to the DA-88 from an AD-1000

1. Connect the output of your console or microphones to the AD-1000 XLR input connectors.
2. Select the appropriate analog input (+4 CAL, –10 CAL, or MIC) on the INPUT SELECT switch.
3. Set the front panel power switch of the AD-1000 to ADAT when using the Apogee FC-8 Format Converter. If using the Tascam IF-88AE, select NORM for an AES output.
4. Set the SYNC SOURCE switch to CRYSTAL.
5. Connect an optical cable from the OPTICAL OUTPUT of the AD-1000 to the ADAT IN of the FC-8. When viewing the AD-1000 from the rear, the optical output is on the right side. Alternatively, if using the IF-88AE, connect an AES cable from the AES output of the AD-1000 to one of the AES inputs of the IF-88AE.
6. Connect a TDIF I/O cable (Tascam part number PW-88D) between the TDIF I/O connectors of the FC-8 (or the IF-88AE) and the DA-88.
7. Connect a 75Ω BNC cable from the Word Clock output of the FC-8 (or the IF-88AE) to the Word Sync input of the DA-88.
8. Set the clock of the DA-88 to "WORD".
9. Select DIGITAL IN on the DA-88 and arm the desired tracks. Remember that the left channel of the AD-1000 will go to tracks 1, 3, 5 and 7, while the right channel goes to tracks 2, 4, 6 and 8 of the DA-88.

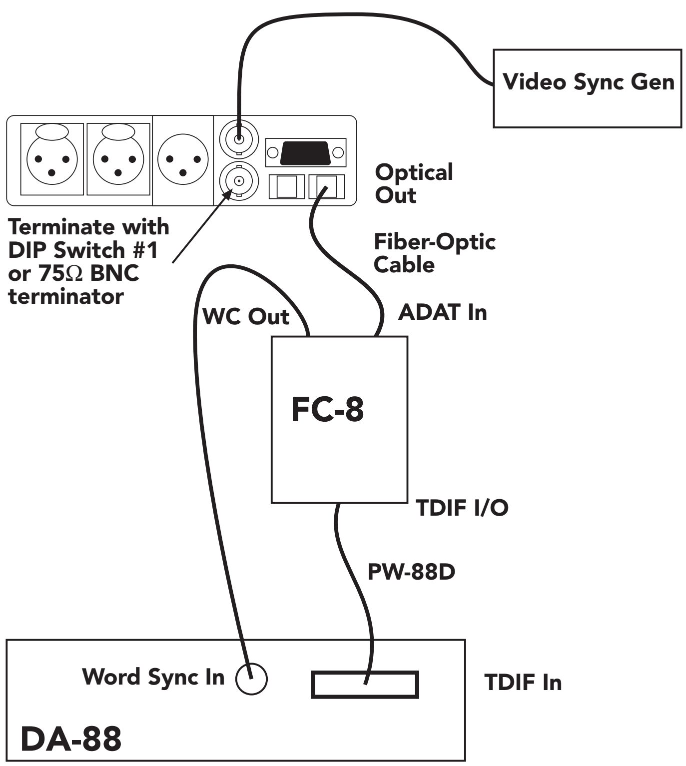
Note: When using the High Resolution option (AD1K-PRT) with the DA-88, follow the steps outlined in "20 Bit Recording Using the ADAT" for coding and decoding the 20 bit signal to and from the DA-88. With the FC-8, the AD-1000 will only see the ADAT I/O, so you can fool it into thinking that it is dealing with an ADAT machine.

Recording On To the DA-88 From an AD-1000 Using Video Sync

1. Connect the output of your console or microphones to the AD-1000 XLR input connectors.
2. Select the appropriate analog input (+4 CAL, –10 CAL, or MIC) on the INPUT SELECT switch.
3. Set the front panel power switch of the AD-1000 to ADAT when using the Apogee FC-8 Format Converter. If using the Tascam IF-88AE, select NORM for an AES output.
4. Set the SYNC SOURCE switch to the appropriate video type – NTSC, PAL or 60 Hz.
5. Connect an optical cable from the OPTICAL OUTPUT of the AD-1000 to the ADAT IN of the FC-8. When viewing the AD-1000 from the rear, the optical output is on the right side or connect an AES cable from the AES output of the AD-1000 to one of the AES inputs of the IF-88AE.
6. Connect a TDIF I/O cable, (Tascam part number PW-88D) between the TDIF I/O connectors of the FC-8 and the DA-88.
7. Connect a 75Ω coaxial cable from your video sync generator to the SYNC INPUT BNC on the rear of the AD-1000, making sure that DIP switch number 1 is on (input is terminated), if this is the last device in the chain.
8. Connect a 75Ω BNC cable from the WC output of the FC-8 (or the IF-88AE) to the Word Sync input of the DA-88.
9. Set the CLOCK of the DA-88 to WORD (not VIDEO).
10. Select DIGITAL IN on the DA-88 and arm the desired tracks. Remember that the left channel of the AD-1000 will go to tracks 1, 3, 5 and 7, while the right channel goes to tracks 2, 4, 6 and 8 of the DA-88.



AD-1000 – FC-8 – DA-88



AD-1000 – FC-8 – DA-88 Video Sync

S/PDIF and Word Clock (WC) Output Operation

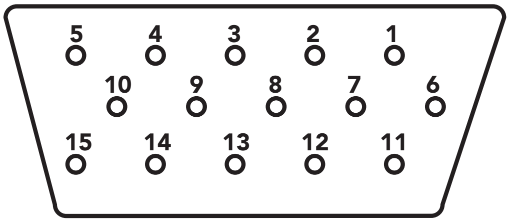
The S/PDIF and Word Clock (WC) outputs are available on the 15-pin HD connector on the rear of the AD-1000 (the same connector that is the power supply input). This cable can be ordered from your Apogee dealer (PS-1000/AD-1000 cable) or use the diagram overleaf if you want to make it yourself. *Please note the WC output is on different pins than on the AD-500, so the equivalent AD-500 cable (PS-1000/AD) will not work on the AD-1000.*

Battery Operation

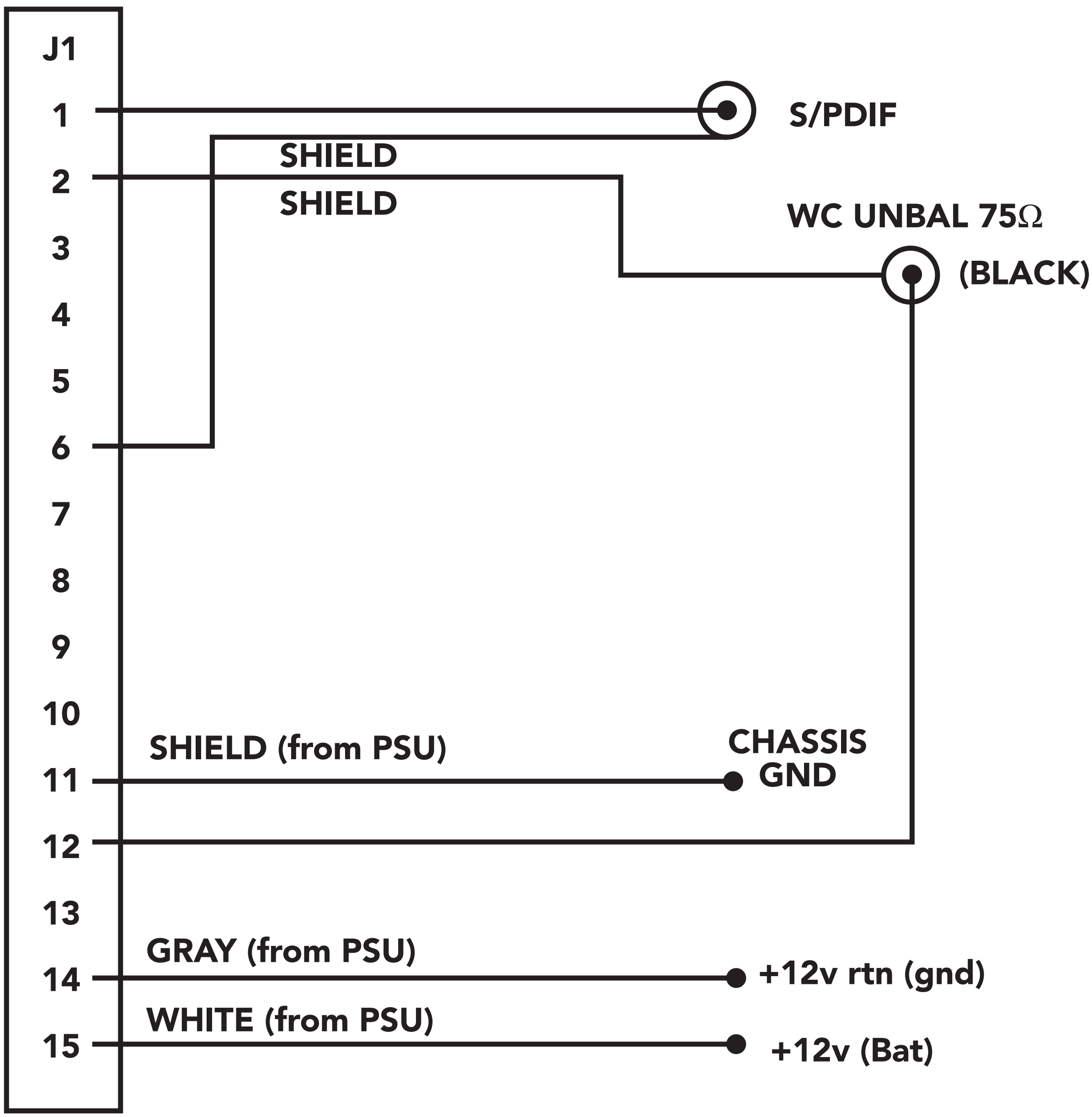
A 12 volt 7.2 Amp/hours lead-acid battery (NiCad batteries are not recommended) can be used for portable or field operation. You will need to interface the battery leads with the 15 pin HD connector – see drawing for pinouts. For battery information contact **Eco Charge** in Boulder, CO 1-800-361-5666. The AD-1000 can also be powered from a car battery.

Other 12 Volt sources

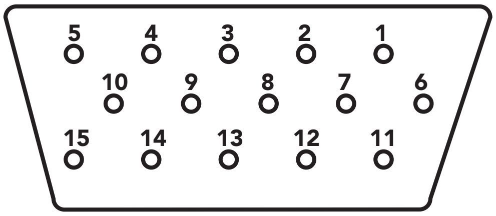
Any good regulated 12 V DC, 1.5 amp linear power supply will sufficiently power the AD-1000. *We do not recommend using a switch-mode type power supply. In addition, we do not recommend the use of the TT-1200 "table-top" PSU designed for the AD-500. The AD-1000 takes more current than the TT-1200 can provide.*



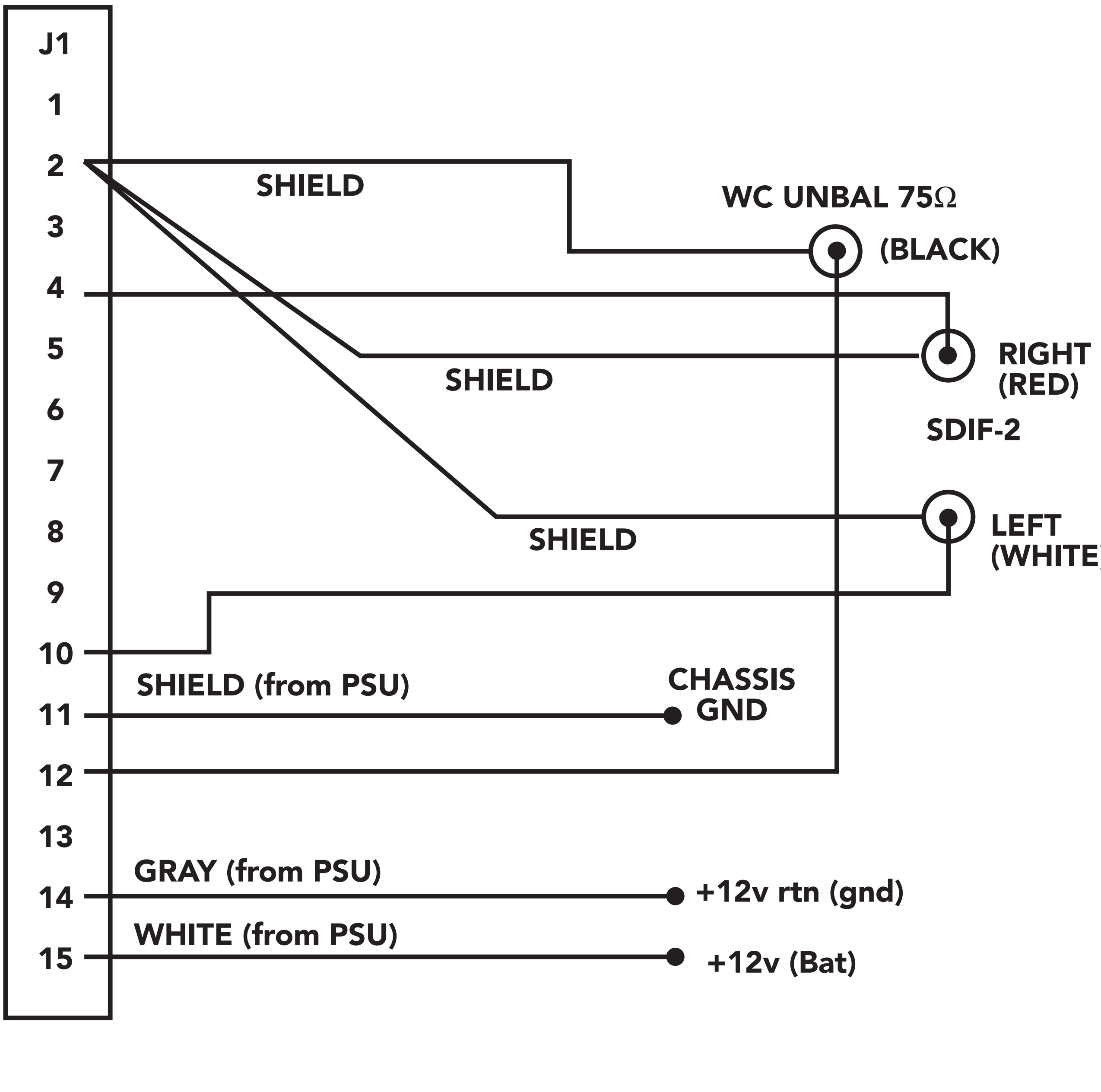
AD-1000 HD15 connector (J1) pin numbering viewed looking into the connector on the unit from the outside (numbers are as molded on the connector itself)



AD-1000 HD15 connections – S/PDIF



AD-1000 HD15 connector (J1) pin numbering viewed looking into the connector on the unit from the outside (numbers are as molded on the connector itself)



AD-1000 HD15 connections – SDIF-2

Appendix I – UV22 Super CD Encoding

Squeezing more performance from a standard CD is not a new idea. It began with adding white noise, called *dither,* to the digital audio. Plain dither was followed by different flavors of dither noise, then a process called 'noise shaping' and more recently various forms of so-called 'bit mapping'. Independent listening tests confirm that these systems either color the recordings we are trying to preserve, or compromise the audible noise floor.

Five years in the making, **Apogee UV22 Super CD Encoding** is an entirely different process. UV22 does its job *without* sonic compromise, and *without* adding a sound of its own, preserving the sound stage and tonal balance of the original 20-bit source. The effects are even audible on original 16-bit recordings.

UV22 Encoding adds an inaudible, high frequency 'bias' to the digital bitstream, placing an algorithmically-generated 'clump' of energy around 22 kHz. Much as the bias on an analog tape recorder smooths out magnetic tape recording non-linearities, UV22 silently captures resolution up to – and even beyond – 20 bits on a standard, 16-bit CD. In addition, this inaudible carrier smooths the rough edges of even the most inexpensive CD player or external converter. UV22 makes your recordings sound better on *all* listening systems. Running already-mastered 16-bit sources through a UV22 processor delivers sonic improvements that any user can realize on equipment they already own.

UV22 is a very special information carrier: it is *not* a new flavor of dither noise. The truly unique statistical properties of UV22 guarantee a constant white noise floor, very similar in character to analog tape noise, no matter what the input source. If you listen to the noise on a UV22 encoded recording, you can hear a stable, accurate sound stage and faithful tonal balance more than 24dB into the noise – just as you do on analog tape. Yet the UV22's low audible noise floor sits at the theoretical limit for a 16-bit system. Nothing is lost – but a great deal is gained.

In listening test after listening test, mastering engineers unanimously choose UV22 over all other systems. Many thousands of CD titles have already been mastered using Apogee UV1000 Super CD Encoders and the Apogees AD-1000.

Beware of "Music Shaping"

Noise-shaping and bit-mapping systems modify the noise floor by changing it from a familiar white noise to one that has been radically modified. Their proponents' theory says that the –96 dB CD noise floor is not low enough to avoid interfering with our listening pleasure, and that our ears would prefer a big dip (about 12 dB) in the noise floor in the 2–3 kHz area, with an accompanying HF boost of as much as 30 dB. What they forget is that few CD releases actually approach the –96 dB noise floor: the noise of almost all sources is significantly higher than this and swamps any of the claimed benefits. In addition, in the process of shaping the 'noise', these systems are also shaping audio information hiding in what they call noise, which results in noticeable shifts in image and colored tonality of the music.

At Apogee, we believe the dynamic range of CD to be fine for many current applications. As a result, we don't try to modify the noise floor. Instead, we make it transparent, allowing clear, clean audio information to be heard up to 30 dB into the noise – just like analog. This information is captured and encoded on to CD – and can be appreciated on any CD playback system.

What Are We Comparing?

With all the grandiose claims flying about, it would be easy to forget that well-executed 16-bit digital audio for CD can already sound amazingly good, and enhancements must therefore be subtle at best. Sometimes a manufacturer will demonstrate a CD enhancement process using two CDs: one recorded using the enhancement process and the other without. What is not usually very clear is that the CD without the process was mastered several years ago using older A to D converter technology, and the CD with the process had the added benefit of the latest A to D conversion. In these comparisons, the converters have a much bigger bearing on the perceived sound quality than the difference between the enhancement processes. If you do your own listening evaluation, be sure to keep all the variables in mind and follow standard good engineering practices when making comparisons – such as accurate level matching.

Something For Nothing?

If we take a well-recorded 16-bit digital audio source and we decide to add some digital EQ, compression, gain change, de-click, de-crackle or even the latest surround-sound process, we are digitally manipulating our 16-bit numbers. We don't get more audio information out than we put in, but we do get numbers with resolution greater than the 16 bit input signal. These extra 'detail' bits we pick up contain some of the results of whatever process we performed to our original 16-bit audio source – and ideally we should hang on to those bits. They show up as improved smoothness, detail, image and depth. The aim of the various encoding schemes is to hold on to the extra resolution after digital processing or A to D conversion when transferring to a 16-bit CD quality output.

Holding on to more bits in a 16-bit CD world

An ideal 'Super CD' system would take as much as 24-bit resolution digital audio and capture the same detail and quality on to our 16-bit CD. We don't live in an ideal world, but it is possible to capture much of the added detail in 20-bit (and greater) systems and bring it into the world of 16-bit DATs and CDs.

Dancing Bits On The Noise Floor

All the encoding systems make the last digital bits dance so they capture extended resolution in the 16-bit CD format. A useful way to separate the different processes (dance steps?) is to look at how each handles the noise floor:

- Common dither methods compromise the 16-bit noise floor – they *add* noise
- Noise shaping and 'bit mapping' trade a reduced noise floor for a large boost at high frequencies
- UV22, Apogee's proprietary process, keeps the audible noise floor solid at the theoretical minimum for 16-bit systems

Although noise shaping and bit-mapping systems (questionably) focus on the noise floor, users often hear this 'improved' noise floor as changing with the music, making it watery and 'fluid-like'. Traditional dither adds noise and raises the noise floor. UV22, on the other hand, presents a constant, smooth and stable noise floor, unobtrusively at the theoretical minimum level, but through which can be heard full 20-bit detail.

How Does It Compare To Analog?

We all know one of the main reasons for going digital: low noise. So why do some engineers still master to ½ inch analog? Technically, analog recordings may appear to be limited by their noise 'floor'. On closer listening, however, the noise floor turns out not to be as solid as the name suggests. A better analogy would be to compare the 'floor' to the surface of a crystal-clear lake, where you can see right into the depths. Analog noise is like that: smooth and constant – but you can hear through it.

This is where digital has differed in the past. The (albeit low) noise floor truly was a limit – more like a stirred, muddy lake. Dither has been used (intentionally and unintentionally) for years to clarify the grunge usually lying on the bottom. The problem, however, was that dither is quite inefficient at capturing full fidelity musical detail, because it is very slow at its job – and it invariably increases the noise.

UV22 is the most efficient method of all in capturing extended resolution into the 16-bit format. This powerful information carrier sits inaudibly out of hearing, yet presents a smooth, white, unvarying noise floor through which can be heard undistorted detail up to 30dB lower in level – extending full- fidelity information beyond 20-bit resolution to your 16-bit CD.

Impressive comments from critical listeners

"The reverb detail and stereo spread are amazing."
—Michael Bishop, Telarc International, Cleveland

Engineers Michael Bishop, Scott Burgess and Elaine Martone tested a number of systems: Apogee UV22 Super CD Encoding; Sony Super Bit Mapping; Gambit; Harmonia Mundi; and Sonic Solutions Turbo Bit Mapping. They used a recording of the Atlanta Symphony Orchestra and Chorus, conducted by Yoel Levi, performing Ravel's Daphnis and Chloe (November 1993 release, Telarc catalogue number CD-80352). The source was 20-bit, recorded to a 20-bit Mitsubishi X-86 2-track.

"Listening tests have shown the Apogee UV22's 16-bit output is the closest to what we hear on our 20-bit source. It's really like getting something for nothing.

"We chose the Ravel recording to test dithering schemes because of its wide dynamic range, distinct imaging and deep sound stage. The piece opens with very low level tympani, high woodwinds and light strings and slowly builds to a 250-voice and orchestra crescendo. Any change from the 20-bit source, especially in those opening bars, is immediately apparent.

"All other systems changed the sound stage and the tonal balance. The Apogee UV22 holds the detail, holds the soundstage and holds the tonal balance across the spectrum. The UV22 was very open and very clean".

Michael further used the UV22 on a recent Brazilian project, Paraiso, featuring Gerry Mulligan and Jane Dubo (October 1993 release, Telarc catalog number CD-83361).

"The UV22 makes all the difference in the world in fades to digital black.

"The reverb detail and stereo spread are amazing; it makes an overall improvement in the final product"

"UV22 kept the 24 bit signal perfectly clean…
all the way down to –120dB."
—Ted Jensen, Sterling Sound, New York

"In our system the UV22 kept the 24-bit signal perfectly clean with no step-type artifacts, all the way down to –120 dB on our digital fader. Also, unlike some…noise shaping/dithering systems we've listened to, the Apogee doesn't color the sound.

"Compared to other systems, or truncation, there is a solidity to the sound. Harmonics are in proper perspective with less 'sizzle', and the image is better preserved.

"You can hear close to 20 bit resolution from a 16-bit CD
without any special decoding hardware… it's amazing."
—Roger Nichols, Engineer and Producer, Nashville

"Warner Bros. Records called me and said that they were going to put out a 20-bit version of Donald Fagen's album, *Kamakiriad,* and would I please supervise the transfers. Our source was a 20-bit Mitsubishi X-86 tape of Donald's album.

"We played back the X-86 tape through the Sony Super Bit Map encoder and cut a CD and a CD master tape for Warner Bros.

"Afterwards, we compared the Super Bit Map CD with the original 16-bit CD. The noise level actually seemed to get louder on some cuts. The quality of the noise floor seemed to change with the signal content. The quality of the lead vocal seemed to get grainier. With our source material, the SBM version of the CD was not any better than the straight 16-bit CD. Maybe worse.

"We performed the same test with the Apogee UV22; we made a CD master tape and a CD of the 20-bit X-86 tape through the UV22. We listened to all three discs, the 16-bit, the SBM, and the UV22. The UV22 version was by far the best. The voice was crystal clear, the noise floor was lower than that on the 16-bit CD, and there was no noise modulation by the program material. It was significantly better than the 16-bit CD, and the difference between the UV22 and the SBM version was like night and day.

"We sent the UV22 tapes to Warner Bros. for the 20-bit version of Donald's album. If there is going to be an expensive gold-plated 20-bit version, it should sound better than the 16 bit CD, right?

"The results are that [with UV22] you can hear close to 20-bit resolution from a 16-bit CD without any special decoding hardware. If you get a chance, listen to both of them and check out the difference. It is amazing."

"UV22 is the closest thing to the 20-bit source that I have heard."
—Bob Ludwig, Gateway Mastering, Portland

Bob had a chance to put our prototype UV22 through its paces with various program material he was working on. He had used several types of material in the 1⁄2-inch 2-track analog format, and had spent some time with our UV22 on the song Nobody's Hero from the forthcoming Rush album Counterparts on Anthem Records. This is what he had to say about the UV22:

"The Apogee UV22 is very impressive… it's the last word in redithering."

"The Apogee UV22 is the closest thing to to the 20 bit source that I have heard. It even makes inexpensive D to A's sound twice as good."

"UV22 rounds out the rough edges of digital"
—Stephen Marcussen, Precision Mastering, Hollywood

"[The Apogee UV22] rounds out the rough edges of digital. I put material through the UV22 at below –60 dB and you could clearly hear the low-level information. It was much smoother and much more intact".

"The low-level stuff was really nice and smooth."

"The signal with the UV22 was a lot clearer than without."

"The Apogee UV22 is an excellent way to utilize 20-bit A/D conversion and 20-bit signal processing. It allows you to capture the improvements of 20-bit even on 16-bit formats. Most importantly, it is very musical sounding; it doesn't change the tonal balance. With the UV22 the 16-bit output sounds very close to the 20-bit source.

"It's simple to use and sounds great. Bravo!"

—Scott Hull, Masterdisk Corporation, New York

"When a 20-bit signal is processed by the UV22, the result is essentially the same as the 20-bit original, and cleaner than Turbo Bit Mapping".

—John Newton, Sound Mirror, Jamaica Plain, MA

UV22 Process Caveats

UV22 Encoding is expected to be the final step in the signal chain before the CD mastering device such as the Sony 1630, etc. No additional process of any kind should be performed on the UV22 processed data or the benefits may be compromised. Other points in the signal chain are possible, but some care must be taken when applying the process.

Because of the addition of the UV22 signal, it is not recommended to use the UV22 process more than once or twice on a signal. Multiple passes through the UV22 process could degrade the noise floor of the system in the upper frequencies.

Experimentation with the Normal and Low settings is recommended for multiple passes. In the Normal mode, the process has been optimized to capture the greatest amount of detail from the high resolution digital input. The Normal mode has the added benefit of smoothing out the non-linearity in almost any DAC. Even with 16-bit sources, the UV22 process gives notably better results on inexpensive DACs. By using the Low setting, less of the detail is captured. The Low setting maintains respectable performance with a reduction in the UV22 "Energy Clump" of 6 dB. The reduction in energy could be desirable when multiple passes on a signal are performed.

Recordings to be used in a Sample or Sound Effects Disc can be encoded if *only pitch shifting upward* is to be used. Pitch shifting *downward* on processed signals could make the UV22 Energy Clump audible.

At this time we do not recommend UV22 processing on signals that are destined for compression systems such as the Sony MiniDisc format. The UV22 Process encodes so much detail that it is possible that the compression algorithms could have adverse effects on UV22 processed material. For this reason, the intricate information captured by the UV22 process would complicate the lives of MD compression systems. We have not done extensive listening tests on this yet, but do advise caution. This caveat is not only true for the UV22 process but for "noise- shaping" systems such as Sony's Super Bit Mapping.

Appendix II – Digital Audio Interconnects

Digital Audio Without Making Your Eyes Glaze Over

You've probably read, or at least started to read many articles on digital audio. Like many people, you may be guilty of skipping the technical diagrams and jumping to the last page for the conclusions. Understanding how digital audio works is akin to getting into the details of how MIDI controls musical instruments… it's handy information, but not necessary for making music. Most digital audio users' eyes glaze over when discussing the technical aspects of the subject. On the other hand, discussing why one digital audio box won't talk to another can make the same eyes bug out and face turn red! Many of us have experienced the frustration of trying to make one piece of digital audio gear connect to another without success. Digital audio is not so new anymore, so it's reasonable to assume that interconnects should be "no brainers". Because they aren't, requires some understanding of what makes them tick so we can get the most out of them. This section will give you insight to deal with the peculiarities of digital interconnects without the usual technical smoke screen.

The Difference Between Good Old Analog And Digital Audio

Sound is transmitted through air as movement of individual air molecules. A microphone turns this movement of air into a changing voltage which represents the air movement. This changing voltage is called an *analog* of the air movement. Sound analogs can also be mechanical, such as a phonograph groove, electrical current, magnetic field, optical energy, or any continuously varying representation.

Digital audio uses numbers to represent sound. These numbers have to be big enough to capture the smallest and biggest details in sounds – accurately. The same numbers also need to be changed fast enough so our ear is not aware of them stepping by. You are probably aware that cartoons consist of a sequence of individual drawings changing fast enough to give the illusion of motion. If we slow the sequence of drawings down, the image starts to flicker like the old movies and motion becomes jerky.

To fool our eyes into seeing fluid motion, the images need to change from one to the next at least 25 times per second. There are some motion picture systems – such as the one from Showscan in Culver City, CA – that increase the rate to 60 per second, resulting in an amazingly grain-less and fluid motion.

The frozen visual images of individual movie frames are analogous to the individual numbers of digital audio. Our ear doesn't get fooled into thinking that these numbers sound real until they change at around 32,000 times a second. The individual numbers are called *samples* and represent audio in narrow slivers of time. The rate these frozen slices of audio change per second is called the *sample rate*.

You will often see sample rates represented as kHz or kiloHertz (k = one thousand; Hz = cycles/times per second). A sample rate of 32 kHz (32 thousand samples per second) is used in digital broadcasting applications. Compact Discs use a 44.1 kHz sample rate (44,100 samples per second). These individual samples are different to the musical instrument or vocal samples used in assembling music tracks. Sound samples are made up from strings of the individual "slices of time" samples much as a video clip is a sequence of individual video frames.

You can see it takes a lot of numbers in the digital world to represent an analog version of the same sound. An analog signal path may need a frequency response of 100 kHz to faithfully reproduce 20 kHz audio. A digital signal path for the same 20 kHz audio requires a frequency response of *several million Hertz* (Megahertz or MHz). Bandwidth is a measure of the lowest to the highest frequency a path can handle. The wide bandwidth required for digital audio is due to the way the individual numbers are transmitted across an interconnect. There are a number of different methods of making digital audio connections inside equipment and externally to other devices.

Digital Audio Interconnects

In the early days of digital audio there was no accepted standard for interconnecting different devices, so the manufacturers invented their own schemes. A interconnect needs to pass the individual numbers of each sample along with timing information and any useful control information such as if pre-emphasis was applied or not.

The numbers of digital audio are transmitted in *binary* form. Instead of using our familiar ten-finger-oriented decimal numbers, we substitute one-finger binary numbers. Any decimal number can be represented as a binary number and vice versa. The big advantage of using binary coding to represent digital audio samples is that each individual digit of a complete binary number takes only one of two values instead of the ten when we use our familiar decimal method of counting. *Binary digits* are called *bits* and, because they have only two values or states, can be easily represented by electronic circuits as either on or off, high or low voltage etc. The most common digital audio numbers in use today are 16 bits long, with a small (but growing) percentage of recorders and workstations capable of handling or storing more.

All At Once or A Bit At A Time

When manufacturers had to come up with schemes to interconnect their products, before they could agree on a standard (pre AES/EBU), the main requirement was to minimize the number of interconnections. When making interconnections *within* a digital device, it is usually most efficient to move the numbers around as complete chunks of the individual bits. Sixteen bit systems can use 16 separate lines to transfer entire samples in single steps. This is known as *parallel* operation. A parallel interconnect between different audio devices is cumbersome, requiring over 32 connections for a stereo 16-bit system – plus additional lines for grounds, timing and control information. A more efficient method is to send the 16-bit numbers across one wire, one bit at a time. This is called a *serial* interconnect and can be visualized as sending individual bits down a hose and reassembling them into complete numbers at the other end. It's important to know when the 16 bit numbers start and finish to correctly unravel them at the other end, so timing information is also included – as either a separate connection or included with the 16-bit audio and identified with an additional unique pattern of bits. You can think of the timing as the pulse of a digital audio system; every time it beats, it signals a sequence of events such as the beginning of a transfer of a sample, one bit at a time. The main pulse is at the sample rate, beating at 44,100 times a second for a CD player. In addition to the sample rate beat, there are additional higher frequency pulses used to co-ordinate all the activity going on between the slower sample rate timing. You could visualize this relationship in musical terms as a one-measure loop with the main pulse on one and the other as 1⁄32 note pulses. The high frequency pulses are often called the bit clock, which is passed across interconnects in one form or another.

It's All In The Timing

A drummer's timing can make the difference between good music and a memorable hit. Digital audio, likewise, needs good timing to make it from one place to another with uncompromised sound quality. The timing in the interconnect is used to unscramble all the bits for accurate recovery of the exact samples transmitted. The timing also needs to be very regular.

Timing jitter is any irregularity in the timing passed across an interconnect. If the samples become messed up in the interconnect, the effects are usually very audible, varying from occasional clicks to a loud, harsh fuzz. Timing jitter can cause more subtle effects. In digital to analog converters for example, the location of instruments across the audio sound stage can become less focused. *Note: A "sound stage" is the mental picture you form when you listen to a piece of music and localize the various instruments and vocals as if they were on stage in front of you (closing your eyes can help form the image). A well defined sound stage has width, depth, and focused locations all defined by subtle reflections, reverb tails and tonal quality in a stereo mix.*

These Interconnects Sound Different!

You may have heard critical digital audio listeners complain "if digital audio is so perfect, then how come it sounds different when I use different interconnects?" Some experts will tell them it must be their imagination because if the numbers are sent correctly on each interconnect they both must sound the same. That makes sense, but it's only part of the story…

When a digital to analog converter receives the samples from an interconnect, it must also extract the timing information and regenerate its own timing "clock". A good analogy is a drummer playing to a click track. If the drummer is good, he can nail the basic tempo of the click and add in faster patterns of his own, such as a sixteenth-note high hat. When digital devices receive the clock from an interconnect, they lock up to the sample rate tempo and add faster multiples many times higher than the drummers sixteenth-note example. Now imagine what would happen to the drummer's playing if we put slight, random variations in his click track reference. The drummer would try to follow the changing tempo but because the changes were unpredictable, he would overshoot the click tempo as it moved up and down. The random click track variations around a perfectly steady tempo could be called tempo *jitter.* The poor drummer ends up with worse jitter in his timing unless he can ignore the small changes and play to the average.

The problem of interconnects affecting the sound can be traced to jitter in the timing of the digital to analog playback. Each time digital audio timing is passed through additional circuits, it picks up slight variations around the original perfect timing. The amount of timing jitter added through successive stages depends on the type of circuits. Inside products, different computer logic families used for digital calculations add varying amounts of jitter. Noise on power supplies and grounds, nearby clocks with similar harmonics, AC power and external interference can all add jitter to perfect timing. Some of it is random and some has specific frequency content. When the internal timing is passed to another device over an interconnect, different types of connections add more or less jitter. A short AES/EBU connection over high quality digital audio cable – such as Apogee's *Wyde Eye* 110Ω AES/EBU cable – will pick up less jitter than the same signal run through a length of microphone cable, XLR connectors and patch bays. A S/PDIF coaxial wire connection (especially one made with *Wyde Eye* 75Ω cable) will be cleaner than the consumer "TOSLINK" optical version, at least partially because of the slower response time of the optical transmitter and receiver.

When the circuits in digital to analog converters (D/A's) recover the timing, they are often negatively influenced by the jitter picked up along the way, much like our miserable drummer trying to follow the varying click track. When the recovered timing starts to wobble around as it tries to track the jittery input, it modulates the analog sound coming out of D/As, causing all sorts of subtle negative effects such as changes in the stereo image and tonal quality. An interesting source of jitter in AES/EBU digital interconnects is due to the changing samples and subcode information. A 1kHz digital audio tone causes 1kHz jitter.

Different interconnects *do not* sound different if the timing circuits of the reference D/A are designed to ignore any jitter and the samples are correctly transmitted. Manufacturers can claim low jitter circuitry – although it's only a relative claim, as at the moment there are no accepted standards for jitter measurement for digital audio. Jitter also has a big influence on the quality of analog to digital converters with very similar side effects, which unfortunately are there forever after.

Interconnect Formats Before AES/EBU

Sony SDIF Interface

The Sony SDIF interconnect is a good example of a basic digital connection between two digital audio units. SDIF format interconnects are found on the Sony 1610 and 1630 processors (used to generate CD masters on U-Matic video cassettes) and many Sony professional digital audio products. The original SDIF and newer SDIF II both use two connections for mono transmission and three for stereo. Each channel is transmitted on its own separate connection along with one word sync connection for any number of audio channels. The main difference between SDIF and SDIF II is unbalanced operation with SDIF and balanced for SDIF II. The 3324 and 3348 digital multitrack recorders have balanced connections which require electronic translation to unbalanced for connection with the SDIF Input/Output on 1610 and 1630 processors. Although the Word Sync transmits electrical pulses at the sample rate for synchronizing the sample transfers, the same information is also contained within each audio channel, so any timing delays between the word sync and channels (due to different wiring lengths) will not corrupt the audio. The SDIF format can handle up to 20-bit samples and includes emphasis identification. The Sony word sync is a symmetrical square wave. The audio samples are transmitted with the most significant bit (MSB) first, the same sequence we read out our familiar decimal numbers.

Mitsubishi (Melco) PD Interface

The Mitsubishi 2-track and multitrack interconnect formats are similar to the Sony SDIF format – but not enough for direct compatibility. Like the Sony format, the Mitsubishi 2 track "DUB" connections are unbalanced, although different in their ability to directly drive high speed optical isolators on the receiving end (for improved ground isolation). The digital audio travels on separate lines along with a word clock line (at the sample rate) and a bit clock line running at 48 times the sample rate. The bit clock runs in bursts, with silence in between, resulting in the 32-bit time slots of a sample being compressed into a shorter duration than the full word clock cycle. The Mitsubishi interconnect uses four lines for a stereo signal instead of Sony's three. Its important that all four run together as any timing skew with the bit clock, caused by different length runs, can result in errors in the audio when it is decoded.

The multitrack version of the Mitsubishi interconnect is a little different. There are still four lines for a stereo connection, except the lines are balanced and the bit clock operates continuously at a 32 times multiple of the word clock. One full cycle of a sample transfer occupies 32 time slots spread evenly over one word clock cycle. No emphasis identification is transmitted in the Mitsubishi Interface standard although Otari made modifications to handle emphasis in some of their PD format multi-track recorders. The interconnect can handle up to 20 bit sample, transmitted MS first. The word clock is a short pulse instead of the evenly spaced rise and fall of the Sony clock.

Yamaha Interconnect

The Yamaha 2-track interconnect and its variations are like the missing link between the above formats and the AES/EBU format. The Yamaha format uses balanced connections and two lines to transmit stereo audio. One line carries the left and right samples transmitted with the least significant bit (LSB) first. The other line carries a symmetrical word clock whose transitions coincide with the LSBs of the left and right samples. The stereo data is transmitted over a single pair by taking the first half of the word clock cycle to send up to 24 bits of a left sample, sitting in 32 times slots, followed by the companion right sample over the next 32 time slots. A total 64 time slots are transmitted for every word clock cycle. This two-line interconnect is also used to combine a number of digital audio products together by cascading from one device to the next in an input to output sequence. Each subsequent device then adds its output to the previous devices output.

ADAT, TDIF Other Interconnects

Several other multi-line digital interconnects exist. Some are proprietary, used to connect products from the same manufacturer, while others are internal interconnects, sometimes adapted for connection to the outside world by another specialty manufacturer. Two of the most common interconnects that are growing in importance are the Alesis ADAT optical and Tascam TDIF 8-track interconnects. These formats were developed specifically for their respective modular digital multitrack (MDM) systems. The ADAT uses an optical format in which the physical transmitters and receivers are the same as for TosLink S/PDIF (which is why the AD-1000 can handle both with the same connector); however the format carries all eight channels rather than just two. As a result, the AD-1000 repeats its two channels across the eight channels in the ADAT (the first AD-1000 channel appears on the od-numbered tracks, while the second appears on the even tracks). Tascam's TDIF interface is a bi-directional multi-line interface using a DB25 connector. It also carries eight-track information.

AES/EBU Interface

AES/EBU, AES3-1985, ANSI S4.40-1985, AES3-1992, EBU Tech.3250- E.CCIR Rec.647 (1986), CCIR Rec.647 (1990) Confused? Well, don't be. These are different standards are lumped together and called AES/EBU, the connection designed to standardize plugging one digital box to another. AES is the Audio Engineering Society and EBU is the European Broadcasting Union. These organizations and others have worked very hard to bring us a standard method of sending professional digital audio across a single interconnect with maximum compatibility. Generally the approach work well as long as the potential weaknesses are kept in mind when stringing things together. A better understanding of how two channels of digital audio flow across a single connection helps highlight the pitfalls.

Electrically, the AES/EBU signal is tailored to use microphone-type cable, although in fact the bandwidth is a good deal wider than regular mic cable can handle successfully. Microphone cable normally carries analog audio on a twisted pair of wires enclosed in an outer metal shield. The shield is usually a continuous, flexible braided wire jacket or in applications where flex is unnecessary, a metal foil wrap is often used (inside patch bays and consoles for example). The shield provides a ground connection and reduces the influence of outside electrical interference on the two wires carrying the audio. Two wires are used instead of one to further reduce the effects of outside interference. Because the two wires are twisted together, they follow almost exactly the same path. Any interference managing to make it through the tubular shaped shield tends to affect both wires equally. An example would be running the microphone cable alongside a power transformer. The magnetic energy radiated from the transformer causes the two wires to develop the same AC mains related hum voltage. If the two wires were driven into a transformer, this hum voltage would not come out the other side of the transformer because both wires have the same voltage at any moment due to the hum. For the transformer to give any output, there must be a voltage difference between the two wires. The transformer input is called *differential* because the analog audio is carried as the voltage difference between the two wires. The noise signals picked up along the way are called *common mode inputs* and the ability of the transformer to ignore them is rated as *common mode rejection*. In professional audio we call differential inputs and outputs *balanced* and because transformers are bulky and expensive, they are outnumbered in modern equipment by their more economical electronic equivalent: electronically balanced inputs and outputs.

As compared to other digital formats which rely on multiple interconnects for clock, left and right data, AES/EBU simplifies the cable connections and uses readily available wire interconnects that are already in use at most professional and semi-professional facilities.

A single line connection of stereo digital audio must transfer a string of data packages containing left and right audio samples repeated at the sample rate. One package is referred to as a frame. The single line AES/EBU interconnect divides each package into 64 little pieces of binary bits with 32 for the left sample and 32 for the right. Each chunk of 32 bits is called a subframe. To make it easy to recover the data on the receiving end, each bit is further divided in two. Patterns of full bits and half bits are coded to indicate whether the bits represent one binary state or another, often referred to as zeros and ones. In some older multi-line interfaces, the location of the beginning of samples is marked with a separate word clock line. To find the beginning of the left and right samples in the AES/EBU format, each 32-bit subframe includes a unique pattern of half bits and at least one delay equal to one full and one half bit joined together. Receiver circuits can recognize the longer one and a half sync bit and use it to extract the left/right synchronizing information for sample decoding and word clock separation.

The audio samples can be up to 24 bits long and the sync pattern uses four more bits. With 32 bits available, there are four extra bits left to send more information. Digital audio samples must change very quickly whereas other information can be updated at a slower rate. For example, emphasis is usually selected at the beginning of a session and remains on or off, so updating the emphasis status 44,100 times a second would be redundant. The AES/EBU interconnect takes two bits of each subframe and calls them user data and channel status bits. To pack more information into the one channel status bit location, 192 bits are sent sequentially, one bit at a time. These 192 bits can represent vast amounts of data at a slower rate than the one bit alone. The beginning of one of these sequences is marked with a special sync pattern in place of the normal sync pattern for a left sample. At the receiving end, the status bit is picked off at every frame and assembled one at a time into a string 192 bits long. The collection of 192 bits repeats every 230 time a second for a 44.1 kHz sampling rate.

The status bits can represent controls for a variety of important data. Sample Rate, Emphasis and Copy protection are represented. Even control of redundancy checking is implemented. Bits for ‘indexing’ are supported. Identification of professional or consumer format is also indicated.

Revision History

Manual Revision
Rev 0.1 1/27/95

Rev 0.2 6/8/95
Installed New Graphic Elements

Rev 0.3 6/19/95
Added changes and revisions to entire manual.

Rev 2.0 8/6/96
Complete revision and reworking including additional section covering new options

Rev 2.1 11/5/96
Minor corrections to the above