APOGEE DA-1000E-20 REFERENCE STANDARD DIGITAL TO ANALOG CONVERTER

ADDENDUM TO STANDARD MANUAL

REVISION A April 15, 1993

This addendum will guide you through the differences that exist between the DA-1000E and the DA-1000E-20 units. Many sections of the original manual are still valid. Until we can update the manual we hope that this will answer many of your questions.

The following sections in the DA-1000E manual are replaced by new sections written in this addendum:

<u>ORIGINAL DA-1000E MANUAL</u>	<u>NEW DA-1000E-20 ADDENDUM</u>
H. Location and functions of front panel controls	A. Location and functions of front panel controls
L3. Additional Inputs	C. Configuration Modes of DA-1000E-20

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WARNING

This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A digital device pursuant to Part 15 of the FCC Rules, which are designed to provide reasonable protection against harmful interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference.

Important- To insure that the complete system (including this peripheral) is capable of complying with the FCC requirements, it is recommended that the user make sure that the individual equipment of the complete system has a label with one of the following statements.

> "This equipment has been tested with a Class A Computing Device and has been found to comply with Part 15 of FCC Rules."
> -or-
> "This equipment complies with the requirements in Part 15 of FCC Rules for a Class A Computing Device."
> -or equivalent.

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

CONFIDENTIAL

OWNER'S RECORD

The serial number is located on the top left 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. DA-1000E   Serial Number________________Purchase Date_________

A. LOCATION AND FUNCTIONS OF FRONT PANEL CONTROLS

1) Digital audio input selector and activity indicators

The Input selector is located on the left side of the DA-1000E-20. A three position switch selects the Digital Audio Input. Two LED's indicate the signal activity for inputs 1 and 3. (See figure a)

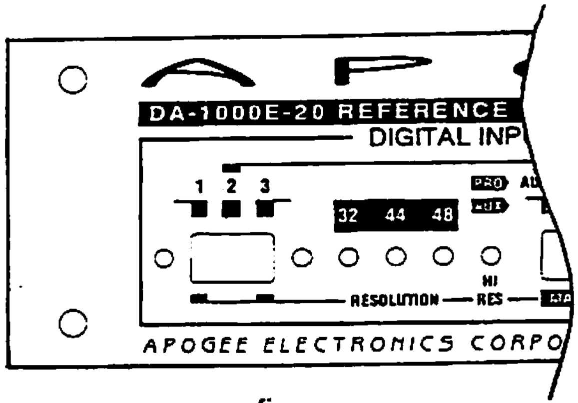

figure a

POSITION 1-
With the Input selector switch in the left most position, the female XLR connector on the rear panel is selected for AES/EBU format. An AES/EBU digital audio input is applied to the rear panel AES/EBU female XLR connector. (See Figure 7 of DA1000E manual) The green indicator directly to the left of the Input selector switch will illuminate whenever there is a signal present. The red **MUTE** indicator will illuminate whenever position 1 is selected and the AES/EBU signal is not present.

POSITION 3-
With the Input selector switch in the right most position, the female BNC connector on the rear panel is selected for S/P DIF format. An S/P DIF digital audio input is applied to the rear panel S/P DIF Optical input port or the S/P DIF electrical input connector(a female BNC or on older models this is a female RCA jack); located on the rear panel. (See Figure 7 of DA1000E manual) The green indicator directly to the right of the Input selector switch will illuminate whenever there is a signal present on either the electrical or optical S/P DIF input. The red **MUTE** indicator will illuminate whenever the signal is not present and position 3 is selected. Both the S/P DIF Optical and electrical connections can be attached at the same time, but the electrical input will override the Optical input.(ie- both inputs are connected and signals are being applied to both. The electrical input will be selected. When the unit generating the signal being applied to the S/P DIF electrical input is turned off or unplugged, then the Optical input then becomes operational)

POSITION 2-
With the Input selector switch in the center position, the auxiliary selector switch located directly to the right of the **HI RES** indicator becomes active. See section A4 of this addendum on the Auxiliary selector switch.

2) Digital audio sample rate indicators

Three amber LED's are located between the Input selector switch and the Auxiliary selector switch.(See figure a) One of the three amber LED indicators will illuminate to indicate the digital audio input sampling frequency range of the digital audio signal that is selected, when it is present. When the DA-1000E-20 is first turned on or when the Input selector switch is changed from one input to another, all three sample rate indicators will illuminate (along with the **MUTE** indicator) until the digital audio input signal has been acquired and locked to.

a) <u>Simple troubleshooting using the front panel LED indicator's</u>

On Power Up -
There is approximately a 5 second delay for the DC servos to settle. During this time both the **EMPH** and **HI RES** indicators will blink, while the **MUTE** indicator is illuminated.

After approximately 5 seconds -
1. If there is no digital audio input on the input selected by the front panel, the **HI RES** indicator will be constantly illuminated and the **EMPH** indicator will continue blinking. (The **MUTE** indicator is also illuminated)
2. If there is a valid digital audio input on the input selected by the front panel, the three amber sample rate indicators will illuminate, indicating the DA-1000E-20 is in the process of locking to the incoming digital audio.

When the DA-1000E-20 locks to the selected input -
The three amber sample rate indicators go off and a single amber LED indicating the incoming sample rate will illuminate.

3) <u>HI RES Indicator</u>

The red **HI RES** indicator located between the three sample rate indicators and the Auxiliary selector switch will illuminate whenever AES/EBU or S/P DIF digital audio input signal has the status flag indicating data up to 22 bits activated in the digital audio data. The **HI RES** indicator also illuminates when **SDIF** digital audio input is selected or the **TEST** tone generator is selected. The SDIF format supports 20 bit inputs. There is no information other than the digital audio to indicate whether the input is 16 or more bits, so the **HI RES** indicator remains illuminated even with 16 bit SDIF inputs.

4) <u>Auxiliary input selector</u>

The switch located between **HI RES** indicator and the **POLARITY** selector has two modes of operation dependent on the position of the **INPUT** selector. (See figure b)

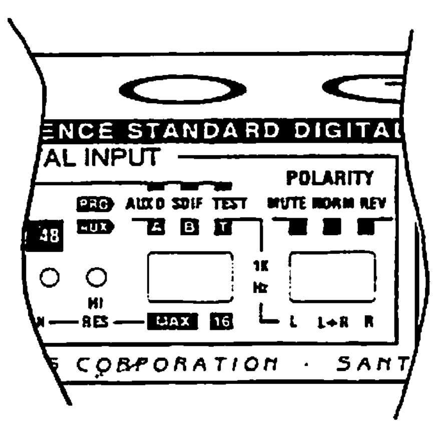

figure b

When the Input selector is switched to position **1** or **3**, this switch controls the resolution of the converters. The legend on the bottom of the Auxiliary selector indicates the function.

MAX -
The leftmost or center position selects the maximum resolution that can be achieved with the digital audio input. (In other words, if 20 bit digital audio data is being applied to the inputs, then the converters can utilize the full digital word. If only 16 bit digital audio data is present at the digital input, then only 16 bit data is applied to the converters.)

16 -
The rightmost position selects 16 bit <u>truncated</u> audio data regardless of the resolution of the digital audio input.

When the Input selector is switched to position **2**, the Auxiliary/Pro switch controls the auxiliary inputs.

The legend above the selector identifies the two possible scenarios and is dependent of the configuration of the DA-1000E-20. The legend that lines up with **AUX** identifies the configuration of all DA-1000E-20 units as they are shipped from the factory. The Auxiliary **A** and **B** inputs are activated allowing multiple S/P DIF inputs. These inputs are located on the 15 pin HD sub-connector on the rear panel. See section C1 for wiring diagram of 15 pin HD sub connector.

AUX A -
When the AUXILIARY selector is in the leftmost position, the **AUX A** input is selected.

AUX B -
When the switch is in the middle position, **AUX B** input is selected.

The legend that lines up with **PRO** identifies another configuration that is available for the DA-1000E-20. The **SDIF** and **AUX D** inputs are activated allowing for one SDIF input (Word Clock, Left Data and Right Data) and one S/P DIF input. Two internal jumper changes are needed for DA-1000E-20 to be activated for this mode of operation. See section C2 for detail on how to implement this configuration and for the wring diagram of the 15 pin HD connector.

AUX D -
When the AUXILIARY selector is in the leftmost position, the **AUX D** input is selected.

SDIF -
When the AUXILIARY selector is in the center position, **SDIF** input is selected.

TEST or T -
When the Auxiliary selector is in the rightmost position and the input selector switch in the middle position, the high resolution test generator is activated. The **HI RES** LED indicator and the 44 sample rate LED will illuminate whenever the test tone generator is activated. This is a 24 bit digitally generated test tone at 1Khz (1.036 KHz @ 44.1KHz sample rate) that is directly applied to the digital to analog converter chips. As shipped from the factory, the signal level is -15dB from the full level output. (The DA-1000E-20 is set up at the factory for unbalanced output with full level at approximately +20dBu therefore with that setup, the test tone generator will output at +5dBu. The DA-1000E-20 is capable of several different output levels of the test tone generator. (0, -12, -13, -14, -15, -16, -18 and -20dB) If another level is desired, consult the configuration change section B of this addendum. The test oscillator is accurate to within 0.01dB relative to the full scale output.

5) <u>POLARITY selector switch</u>

This switch is located between the Auxiliary selector and the **EMPH** indicator and has two separate functions, depending on the setting of the Auxiliary selector switch. (See figure c)

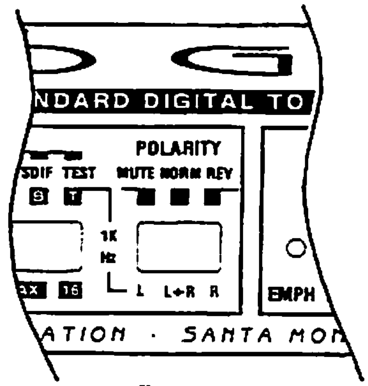

figure c

When the **TEST** position is selected on the Auxiliary selector <u>and</u> the Input selector is in position **2** and the **POLARITY** selector is in the leftmost position, **L**, only the left channel analog output is active. When the **POLARITY** selector is in the center position, **L+R**, both analog output channels are active. When the **POLARITY** selector is in the rightmost position, **R**, only the right channel analog output is active.

When the **A** or **B** position (or the **Aux D** or **SDIF** position, depending on the configuration mode) is selected on the Auxiliary selector; the leftmost position of the **POLARITY** switch selects **MUTE**. The **MUTE** LED indicator (located between the **POWER** selector and the **POLARITY** selector) will illuminate. (See figure d) This selects the analog outputs to off by closing the output mute relays and loading the digital to analog convertors with digital 'audio black' (silence).

The center position, **NORM**, allows the analog outputs to be correct polarity - ie. the digital audio input will produce an analog output voltage of the correct polarity. *note- a correct output polarity requires the correct configuration for pin 2 or pin 3 hot. See section J2 of the standard DA-1000E manual.* The rightmost position, **REV**, outputs both left and right with their polarity reversed from the standard polarity.

6) <u>EMPH Indicator</u>

The **EMPH** indicator located between the **POLARITY** selector and the **MUTE** indicator, will illuminate any time the emphasis flag is present in the digital audio data. The de-emphasis circuits are automatically engaged. (see figure d)

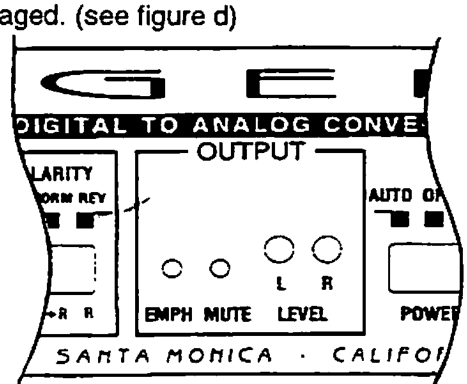

figure d

7) <u>MUTE Indicator</u>

The **MUTE** indicator ,located between the **EMPH** indicator and the analog output **LEVEL** calibration adjustments, will illuminate under two conditions. (See figure d) First, whenever the appropriate Input selector is activated and the digital audio input is not present on that input. Second, whenever power is first applied to the unit or the Input selector is switched from one digital audio input to another one. In other words, whenever the DA-1000E-20 is not in lock.

8) <u>Analog output LEVEL calibration adjustment</u>

Located between the **MUTE** LED indicator and the **POWER** selector switch are two multiturn screwdriver adjustable pots. (See figure d) These pots individually control the output level of both Left and Right analog audio signals which are available from the two male XLR connectors located on the back panel. (See Figure 7 of the DA-1000 manual) Output level range depends on the position of the internal output selection configuration described in further detail in section J2 of the DA-1000 manual.

9) <u>Auto/manual POWER on selector</u>

Located on the right side of the front panel of the DA-1000E is the POWER selector switch. (See figure e)

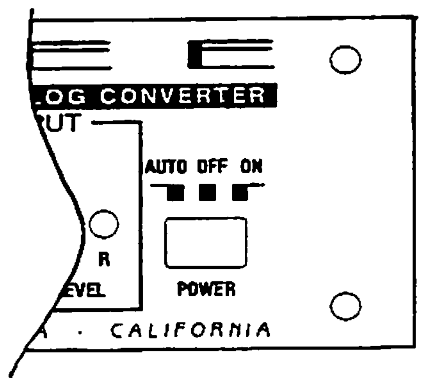

figure e

AUTO position-
With the **POWER** selector switch in the left most position, the DA-1000E is in **AUTO** mode. In this mode, with no digital inputs, the DA-1000E is off, using a tiny trickle of current for low battery operating drain. The **AUTO** position permits you to operate the DA-1000E in conjunction with other equipment (ie, DAT recorder) for automatic power on selection. The DA-1000E will wait for a digital audio input signal to be applied to either the AES/EBU or electrical S/P DIF inputs before it goes into normal operation. This function does work for an optical or auxiliary inputs.

OFF position-
With the **POWER** selector switch in the center position, the DA-1000E is turned off.

ON position-
With the **POWER** selector switch in the right most position, the DA1000E is powered on.

B. TEST TONE GENERATOR OUTPUT LEVEL

When the Input selector is switched to position **2**, and the Auxiliary selector is switched to **TEST**, the high resolution test tone is applied to the analog outputs. The DA-1000E-20 is shipped from the factory to output -15db from analog full output level. This can be changed to one of several levels. The jumpers that need to be changed are JP702, JP703 and JP704. To access the jumpers refer to the DA-1000E manual, section M2 on access to the jumpers.

The analog output is also dependent on the output pin configuration that the unit is set at. (See DA-1000E manual, section J2 for a complete description) Below is a table to easily identify the settings that need to be made to achieve the output level needed when the test tone generator is engaged.

				ACTUAL LEVEL		
J702	J703	J704	SEE NOTE 1 OUTPUT	BALANCED PROFESSIONAL	UNBALANCED PROFESSIONAL	UNBALANCED CONSUMER
on	on	on	0 dB	+26 dBu	+20 dBu	+15.5 dBu
off	on	on	-12 dB	+14 dBu	+8 dBu	+3.5 dBu
on	off	on	-13 dB	+13 dBu	+7 dBu	+2.5 dBu
off	off	on	-14 dB	+12 dBu	+6 dBu	+1.5 dBu
on	on	off	-15 dB	+11 dBu	+5 dBu	+0.5 dBu
off	on	off	-16 dB	+10 dBu	+4 dBu	-0.5 dBu
on	off	off	-18 dB	+8 dBu	+2 dBu	-2.5 dBu
off	off	off	-20 dB	+6 dBu	0 dBu	-4.5 dBu

NOTE 1: RELATIVE TO FULL SCALE DIGITAL INPUT

table 1

C. CONFIGURATION MODES OF OPERATION WITH THE DA-1000E-20

As shipped from the factory, AES/EBU and S/P DIF (electrical and optical) digital audio inputs are functional. The DA-1000E-20 has other inputs, located on the 15 pin HD sub-connector, which enable it to accept other types of inputs and other types of formats. The next two sections will inform you as to how to make the DA-1000E-20 accept these formats and inputs.

a) Standard Mode

The DA-1000E-20 is shipped from the factory in "Standard" mode. Connecting an AES/EBU or S/P DIF format signal to the appropriate connector on the back panel is all that is needed. Standard Mode also includes the ability to receive two additional S/P DIF inputs. These two extra inputs are referred to as **AUX A** and **AUX B** and are available through the 15 pin HD sub connector when using the proper cable. The cables can be purchased from APOGEE or you can follow figure g to assemble your own cable. APOGEE supplies the AUX A/AUX B cable accessory as a TT-1200/DA (Table Top power supply with cabling) or the PS-1000/DA (Cabling for PS-1000 Rack mount Power Supply). Contact your distributor or APOGEE for more information

When AUX A or AUX A and AUX B input operation is required, a special cable assembly is needed. This uses BNC connectors known as AUX A and AUX B. These two BNC connectors on the cable assembly go to the back panel 15 pin D-SUB connector. (See Figure g) You will also need to change the internal Jumper configurations if your unit is already in standard configuration (as shipped from factory unless otherwise specified at time of sale). If you do not know what configuration your unit is in, then it is suggested that you verify the below settings to confirm AUX A/AUX B mode operation. Before making any changes to the jumpers, read section M in the DA-1000E manual on Disassembly.

The AUX A and AUX B connectors do not use transformers as do the standard digital inputs, AES/EBU and S/P DIF (electrical or optical). This option thus makes it useful for interfacing to equipment that does not work optimally with the transformer inputs.
To select between the two AUX inputs change the Input selector to position 2 and the Auxiliary selector to position A or B. With the Auxiliary selector switch to the left, **AUX A** is activated. With the switch to the center, **AUX B** is selected.

The JP200 and JP100 configurations for standard mode are shown below.

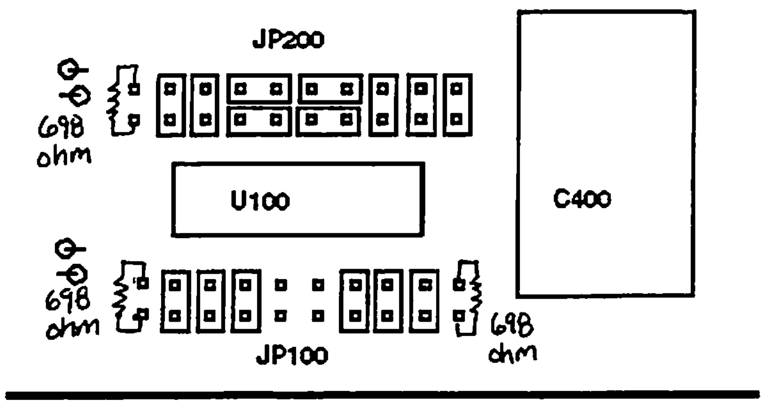

figure f

For AES/EBU inputs, the terminating resistor should be 110 ohms and S/P DIF inputs should be 75 ohms. *NOTE: the diagram in figure g shows 75 ohm resistors thereby being an AUX A/AUX B cable assembly for use in S/P DIF applications.* If you want to drive another piece of digital audio equipment with the same AES/EBU or S/P DIF line, the termination resistor can be omitted and the signal looped on to the next digital input. *(Note: This may not work in all situations due to reflections in the connectors and cables.)*

The diagram below shows the wiring needed to operate the DA-1000E-20 in AUX A/AUX B mode.

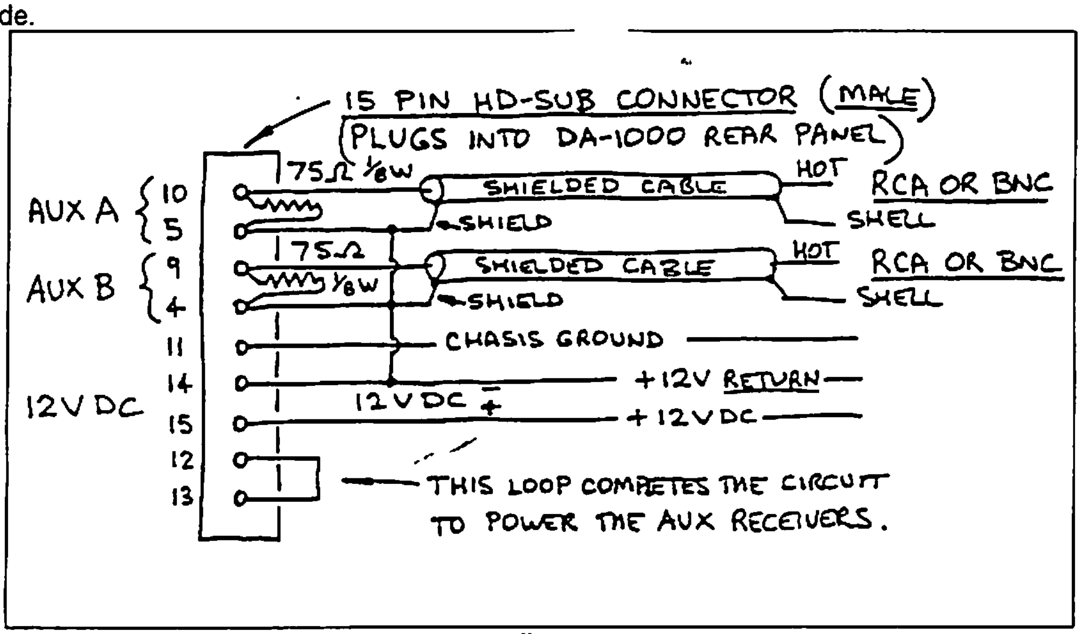

figure g

b) <u>AUX D/SDIF (Sony Digital Interface) Operation Mode</u>

With only two internal jumper changes and the proper cable, this format can be received by the DA-1000E-20. The cables can be purchased from APOGEE or you can follow the diagram below to put together a cable. APOGEE supplies the SDIF cable accessory as a TT1200/DA/SDIF/D (Table Top power supply with cabling) or the PS-1000/DA/SDIF/D (Cabling for PS-1000 Rack mount Power Supply). Contact your distributor or APOGEE for more information.

When AUX D/SDIF input operation is required, a special cable assembly is needed. This uses four BNC connectors known as AUX A(word sync), AUX B(Channel 1-Left Data), AUX C(Channel 2-Right Data) and AUX D(S/P DIF data).

Word Sync (also known as Word Clock) is a symmetrical square wave at the sampling frequency. For example, if the sampling rate is 44.1KHz, then the word sync is a square wave at 44.1KHz. Word Sync (Word Clock,WC) is often used to distribute timing to more than one digital processor to synchronize them at the correct sampling rate and phase.

Data Left is a serial representation of the left (or A) digital audio. The data rides in 32 bit cells for each individual digital audio sample. This repeating train of 32 cells contain a code representing the digital audio in addition to a flag used to indicate whether emphasis was used when converting from analog to digital. If the sampling rate was at 44.1KHz, then these trains of 32 cells pass by at 44,100 times per second.

Data Right (or B) is a serial representative of the right digital audio in the same format as described in the Data Left paragraph above.

On the special cable assembly there are four BNC connectors which go to the back panel 15 pin D-SUB connector. (See figure J) You will also need to change two of the internal Jumper configurations(JP100) to fully activate the SDIF/AUX D mode, if your unit is already in standard configuration(as shipped from factory unless otherwise specified at time of sale). If you do not know what configuration your unit is in, then it is suggested that you verify the below settings to confirm SDIF mode operation.

To select between the AUX D/SDIF inputs change the Input selector to position 2 and the Auxiliary selector to position **AUX D** or **SDIF**. With the Auxiliary selector switch to the left, **AUX D** is activated. With the switch to the center, **SDIF** is selected.

The JP200 and JP100 configurations for AUX D/SDIF mode are as shown below.

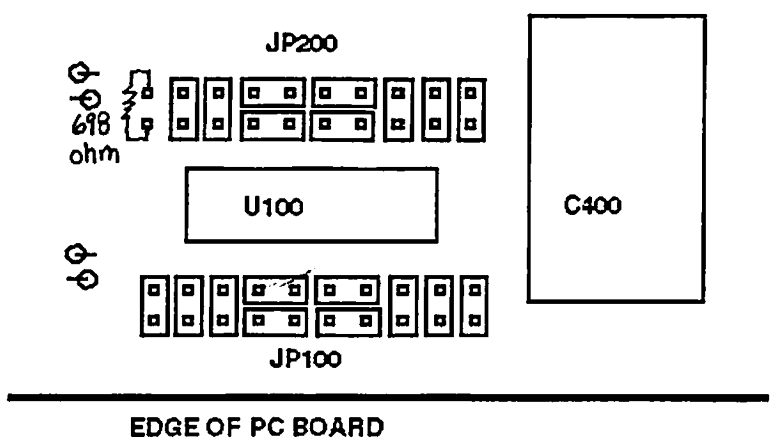

figure h

The diagram below shows the necessary wiring that is needed to operate in AUX D/SDIF mode with the DA-1000E-20.

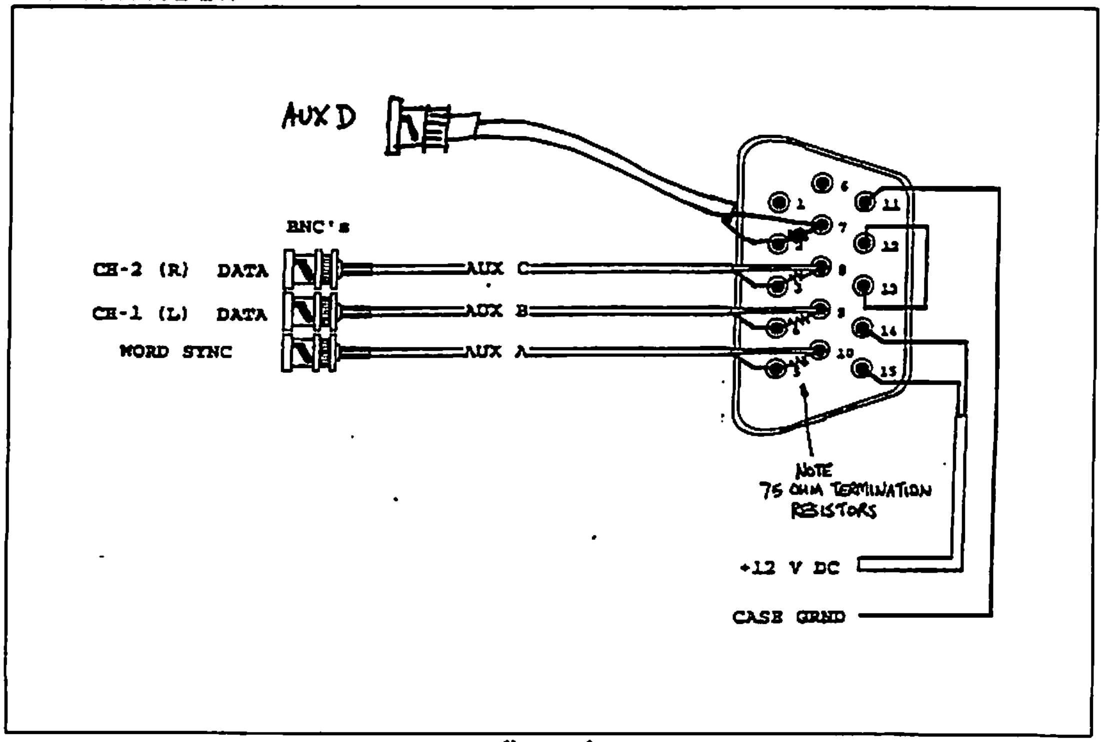

figure j

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CONFIDENTIAL

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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 ELECTRONICS CORPORATION.

OWNER'S RECORD

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Model No. DA-1000E Serial Number_______________Purchase Date_________

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List Of Figures and Illustrations

List Of Tables

A. APOGEE DA-1000E PORTABLE REFERENCE DIGITAL TO ANALOG CONVERTER

SPECIFICATIONS

QUANTIZATION - CONVERTER	20 Bits per sample, 8 times oversampling
QUANTIZATION - INPUT	16 Bits per sample
FREQUENCY RESPONSE	20Hz - 10KHz +/- 0.025dB 10KHz - 20KHz +0.05/-0.2dB
TOTAL HARMONIC DISTORTION PLUS NOISE	-94dB Typically @ 1KHz (@ any sampling rate, 0.1dB below converter full scale output level)
SIGNAL TO NOISE RATIO	Typically -106dB flat Typically -108.5dB (A weighted)
CROSSTALK	Typically 110dB, 20-20KHz
DE-EMPHASIS	50uSec/15uSec
DE-EMPHASIS CONTROL	Automatic, Override On, Override Off
DE-EMPHASIS INDICATION	Amber LED indicates De-emphasis
INPUT SAMPLING RATE RANGE	Any frequency from 32KHz to 54KHz via all digital audio inputs
SAMPLE RATE INDICATOR	Amber LED's indicate sample rate range for 32KHz, 44KHz and 48KHz
INTERNAL RECOVERED CLOCK JITTER	Typically 30 picoseconds RMS
NOMINAL DC POWER INPUT	12 Volts DC @ 1100mA
INPUT VOLTAGE RANGE	+11.0 to +15 Volts DC
INPUT DROPOUT VOLTAGE	+10.5 Volts DC
WEIGHT	1.3Kg (2 lbs 14 oz)
DIMENSIONS	L=27.30 x W=14.12 x H=3.96 cm (L=10.75 x W=5.56 x H=1.56 in)
AMBIENT OPERATING TEMPERATURE	0°C to 40°C (32°F to 104°F)

Note: The DA-1000E will work with any well-regulated 12V DC Power source with an output current of 1200mA or greater. We recommend using linear type power supplies. Direct connection to external 12VDC lead acid or NiCad batteries will provide convenient portable operation.

OUTPUTS

ANALOG OUTPUTS two male gold XLR connectors on rear panel
OUTPUT LEVEL AND POLARITY internally selectable via six internal header positions - selects consumer, professional unbalanced and professional balanced; all with pin 2 or 3 'hot'. Adjustable from front panel multi-turn calibration pots (both left and right)
CONSUMER OUTPUT LEVEL RANGE +8dBu to +15.5dBu for full scale digital audio input
PROFESSIONAL UNBALANCED LEVEL RANGE +13.5dBu to +20.5dBu for full scale digital audio input
PROFESSIONAL BALANCED LEVEL RANGE +19dBu to +26dBu for full scale digital audio input

SEE NEXT PAGE

INPUTS

AES INPUT(female XLR connector on rear panel)
32KHz to 54KHz Input Sampling Rate
Transformer isolated, 110 or 220 ohm via internal jumper, See section J3
AES/EBU Format selectable on front panel Input switch
S/P DIF INPUT(female BNC connector on rear panel)
32KHz to 54KHz Input Sampling Rate
Transformer isolated, 75 ohms
S/P DIF Format selectable on front panel Input switch
OPTICAL INPUT("TOSLINK' optical connector on rear panel)
32KHz to 54KHz Input Sampling Rate
S/P DIF or AES/EBU Format selectable on front panel Input switch
MULTI FORMAT DIGITAL INPUTS(15 pin HD sub connector on rear panel)
32KHz to 54KHz Input Sampling Rate
Maximum of 4 RS-422 digital inputs controlled by internal jumpers and EPROM programmable for the following digital formats and options..........

- **SDIF**(balanced or unbalanced)
- **SDIF-2**(three RS--422 inputs utilized for Word Clock, Left Data and Right Data)
- **AUX A/AUX B**(two RS--422 inputs utilized to expand input capability of S/P DIF and/or AES/EBU formats)
- **YAMAHA**(two RS--422 inputs utilized for operation)
- Contact APOGEE for other supported formats; including JVC, NED and Mitsubishi (both two track and multitrack)

SUPPLIED ACCESSORIES

Operation Manual, warranty card, Male BNC to Female RCA adapter, 3/32 Hex wrench, 10-32 x 7/16 screw for rack mounting, EPROM extraction tool, two spare rubber feet, two 698 ohm miniature programmings and ten miniature programming jumpers.

OPTIONAL ACCESSORIES

APOGEE Model TT-1200/DA is a compact table top power supply for the DA-1000E accepting 110 Volts AC @ 60Hz and providing 12V DC regulated output @ 1200mA. Includes 15 pin HD sub connector with two coaxial cables with female BNC connectors for AUX A and AUX B digital audio inputs.

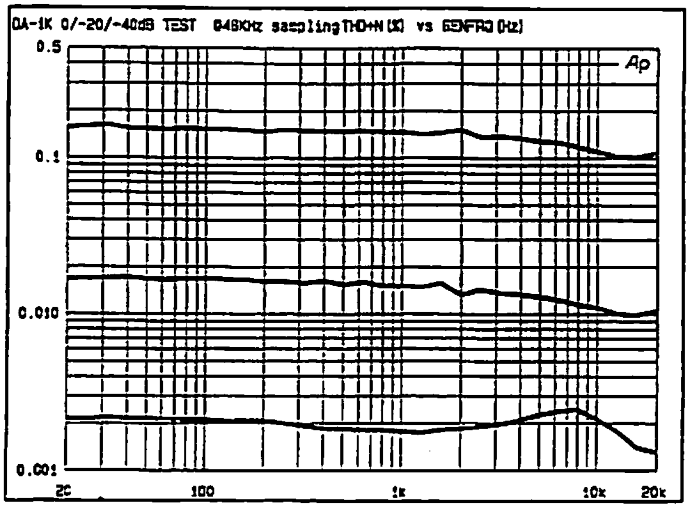
APOGEE Model TT-1200/DA/SDIF is a compact table top power supply for the DA-1000E accepting 110 Volts AC @ 60Hz and providing 12V DC regulated output @ 1200mA. Includes 15 pin HD sub connector with three coaxial cables with female BNC connectors for word clock, audio data left and audio data right inputs.

APOGEE Model PS-1000 is a rack mountable dual worldwide power source for the DA-1000E with selectable 100, 120, 220 and 240 Volts AC @ 50/60Hz providing dual 12 Volts DC regulated outputs @ 1500mA each. Unit is 1/3 rack size and matches style and finish of the DA-1000E. Power is distributed by two 15 pin HD sub connectors on rear panel.

APOGEE Model RM-1000 is a rack mountable carrier frame capable of holding up to three AD-500, DA-1000E or PS-1000 in a 1U EIA space.

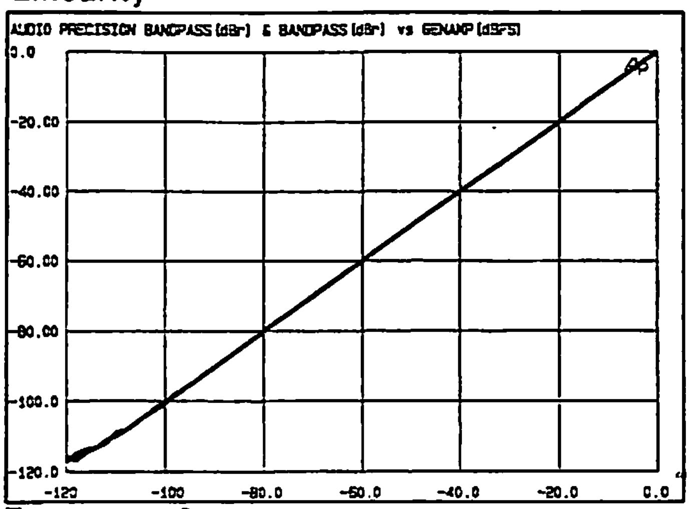
1) TYPICAL PERFORMANCE MEASUREMENTS

a) Total Harmonic Distortion Plus Noise



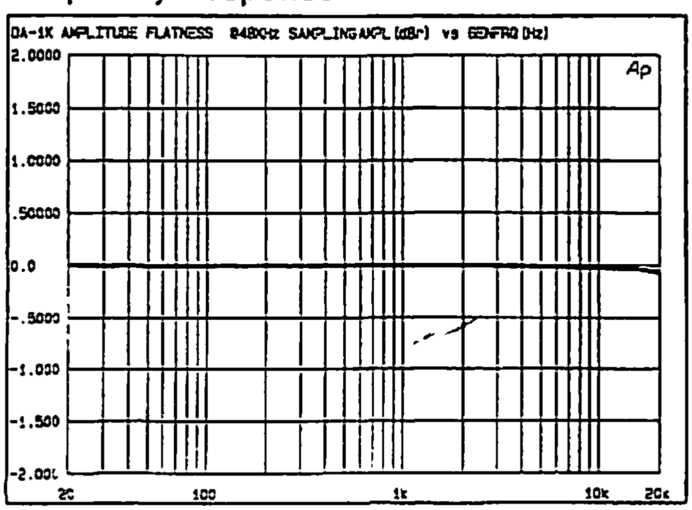
16 bit digital audio input at 48kHz Sample Rate showing Total Harmonic Distortion plus noise of one channel
bottom trace: 0 dB Full Scale
middle trace: -20 dB Full Scale
top trace: -40 dB Full Scale

b) Linearity



16 bit digital audio input at 48kHz Sample Rate showing linearity of both channels at 500 Hz signal input frequency

c) Frequency Response



16 bit digital audio input at 48kHz Sample Rate showing frequency response and flatness of one channel with input level of 0 dB Full Scale

B. UNPACKING

Your DA-1000E is packed in a foam lined shipping container. Be sure to save the container for any further shipment of the unit.

ACCESSORIES

1 x Operation Manual
1 x Warranty Card
1 x Male BNC to female RCA Adapter
1 x 3/32 Hex Wrench
1 x 10-32 x 7/16 Mounting screw for rack mount
1 x Pink handled PROM extraction tool
2 x Rubber feet
2 x 698 ohm miniature programming jumpers
10 x Miniature programming jumpers

NOTE: Power supplies and associated cables are shipped on separate order.

C. GETTING STARTED QUICKLY

Your DA-1000E is shipped from Apogee ready to go. It has spent at least 48 hours 'burning in'. This burn-in procedure involves powered operation at elevated temperatures to isolate units that would possibly fail due to infant mortality.

We recommend you read the entire manual before using your DA-1000E. If you are anxious to get started you will find the operation is intuitive. We suggest you take the following steps--

1- Connect the left and right analog XLR outputs on the rear panel to your equipment. The outputs can be seen more clearly if you consult figure 7. The left and right analog outputs polarity is set at the factory for pin 3 hot, unbalanced. If you need balanced output configuration or consumer output configuration consult section J2.

2- To connect the power, a male 15 pin power connector with pin 15 connected to +12 Volts DC and pin 14 connected to the +12 Volt Return is necessary. If you have purchased a TT-1200 from APOGEE, all that is needed is to plug the 15 pin connector to the rear panel (See figure 7) and plug in the AC. If you have purchased a PS-1000 from APOGEE, then connect appropriate cable from the PS-1000 to the DA-1000E. The PS-1000 then can be plugged into the AC. Consult section F1 for more information.

3- Connect the digital audio input to either the AES/EBU male XLR connector or the S/P DIF female BNC connector or the OPTICAL connector on rear panel (See figure 7).

4- Select the front panel POWER selector switch to ON.

5- Select the INPUT selector switch to the appropriate input being used. When AES is selected, the male XLR connector is actively accepting AES/EBU format. When SPDIF is selected, the SPDIF female BNC connector (using the supplied BNC to RCA adapter) or OPTICAL connector is actively accepting S/P DIF format. The SPDIF electrical input will override the OPTICAL input it both are active.

6- Select EMPHASIS to AUTO. (See section H3)

7- Select DELAY to OFF. (See section H4)

8 - Your DA-1000E should now be functioning. If you have any trouble with the SPDIF input see section L3b for explanation of AUX A/AUX B operation and Section N if you own a Panasonic 3700 or 3900 DAT machine.

D. INSTALLATION

Your DA-1000E is designed for free standing or rack mounting operation. It is important to allow for adequate ventilation as the DA-1000E is internally very densely packed with electronics. The DA-1000E typically generates 12 watts of heat and unless adequate ventilation is provided it will get hot to the touch. It is normal for the DA-1000E to run warm.

1) Free Standing Operation

When used in a free standing mode, make sure it is sitting on its rubber feet (spares provided in accessories kit) and the cooling slots on the bottom are not obstructed. Operate on a clear flat surface free of sheets of paper that may restrict natural cooling.

You can use APOGEE's table top power supply, batteries or provide your own power supply. See Section F for more details on power sources.

2) Rack Mount Operation

For rack mounted operation, check that the APOGEE RM-1000 rack mount tray is not tightly sandwiched between other items in the rack so as to restrict adequate ventilation.

If you are using APOGEE's PS-1000 rack mountable power supply along with your converter, we recommend the following layout. As viewed from the front panel, the PS-1000 should be mounted on the right side of the RM-1000 with the DA-1000E mounted in the center and the AD-500 mounted on the left hand side. With this arrangement the least amount of magnetic inductive coupling from the toroidal transformer in the PS-1000 is introduced into the converters. This could cause hum. For this reason AD-500 is farthest from the transformer, thereby not allowing any hum artifact to be transferred from the converter to the recording medium.

To mount the converter, use the 10-32 x 7/16 screw (provided in accessories kit). The converter has one large threaded hole in the center of the enclosure on the bottom. Line up the DA-1000E on top of the RM-1000 Rack Mount tray in the appropriate position. The alignment is provided by a milled lip on the front of the RM-1000 which holds the front panel of the converter in place. (Notice that the RM-1000 has nine holes in it. The DA-1000E would mount using holes number 2, 5 or 8. See figure 1.) Insert the screw in the proper hole number and tighten.

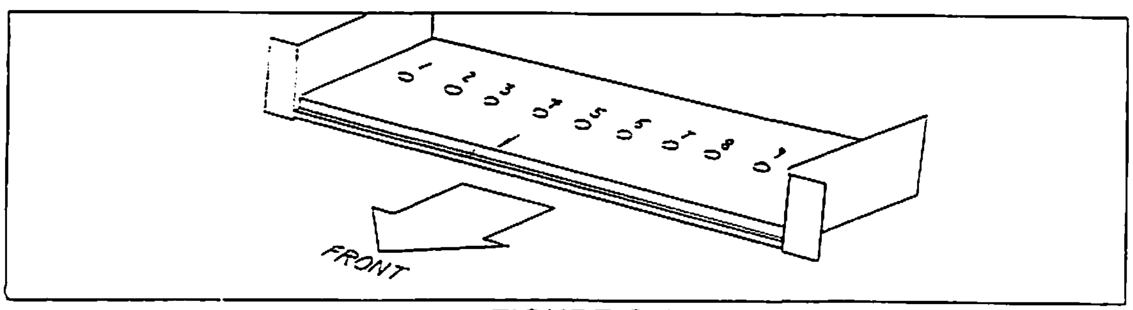


FIGURE ONE

E. OVERVIEW

1) Features

The Apogee DA-1000E is a professional reference standard digital to analog converter enclosed in a convenient, robust package that can be rack mounted, three to a 1 U size carrier or used individually in portable reference applications.

The DA-1000E incorporates the results of two years research into uncolored, faithful audio reproduction. The DA-1000E features a number of unique, proprietary Apogee developments.

For those interested in technical matters the DA-1000E features:

* Updatable and reprogrammable digital interfaces and digital audio processing delivers optimum sonic performance.
* Easy selection of different analog output formats; including balanced professional levels, unbalanced professional levels and unbalanced consumer levels; all with selection of pin 2 or 3 'hot'. Relay protection prevents any power up thumps and gives fault protection. Front panel calibration pots for output level adjustment.
* Latest D/A converter technology with 8X oversampled digital data driving two 20 bit dual Digital to Analog converters. These dual converters utilize the latest segmented techniques to eliminate the usual large transitions around zero, thus providing excellent low level performance in addition to the inherent high level performance.
* Apogee C768 Low Jitter Clock locks to any sampling rate from 32KHz to 54KHz for uncolored high frequency sound reproduction. Accurate timing regeneration without the use of narrow band crystals permits full sonic performance even in vari-speed operation.
* Apogee's new model 964-IV-63 current to voltage and filter module matches our latest filter advances with a unique proprietary circuit that converts high speed electrical currents from the DACs into the necessary voltage with none of the shortcomings of traditional op-amp approaches. This lightning fast circuit has no feedback yet still delivers distortion performance that rivals the measurement ability of our test equipment.
* Apogee's exclusive P818 ultra low noise DC/DC power supply with full synchronous operation. Specifically designed for combination analog and digital circuitry.
* The 964 filter and C768 low jitter clock are major factors in the DA-1000E sonic transparency.
* A half word digital delay is provided to correct timing of EIAJ(F1) format recordings. This is selectable to either the right or left channel.
* The DA-1000E is mounted in a rugged enclosure. The small size lends itself to portable applications. With Input supply voltage of +12 volts DC, many power options are available.
* Jumper selectable and EPROM programmable digital audio interface supports all popular (and some unpopular) digital audio inputs with the ability to enhance the performance with field upgradability.

2) What the professionals have to say about APOGEE products

"I have been listening to A/D and D/A converters since 1977. Each time a new converter came out, it sounded better than the one before it. Last week in Nashville, I compared three of the best converter packages available. The very best converter package should do absolutely nothing-- the playback of the digitally stored data should sound no better and no worse than the original source. All I want is a straight wire. The clear winner of the Nashville test was the Apogee converter package. I had put off buying my own converters for a couple years. After the Nashville test, I had no more excuses."

--Roger Nichols, Recording Engineer and Producer

"The DA-1000E lets me hear exactly what I'm putting on the CD Master. The clarity and lack of coloration is remarkable. There is no harshness, thinness or artificial warmth. The image is stable and defined. Each instrument is in its correct position without having its size changed or its edges blurred. Attacks and decays are accurate. Reverb and delays sound correct without drying up."

-- Ted Jensen, Sterling Sound; New York, NY

"In our tests, the Apogee AD-500 retained more of the ambience and definition than ever before and was discernable all the way to the end product, the CD disc. We feel that it is now possible to bring forth more of the information that was originally intended for the CD consumer."

-- Bernie Grundman, Grundman Mastering; L.A., CA

"I have lots of conversion equipment, but with the DA-1000E, it's as if I'm listening directly to the analog source."

-- Stephen Marcussen, Precision Mastering; L.A., CA.

"How do I get the highest quality analog to digital conversion without having to lug DAT machines all over the planet, and still maintain some sort of consistency...? Your marvelous, portable AD-500 and DA-1000E Digital Converters! The Apogees have made my life so much simpler."

-- Bob Clearmountain, Recording Engineer and Producer

"After listening to virtually all of the currently available D/A converters, the Apogee is by far the most accurate!"

-- Glenn Meadows, Mastering Engineer; Masterfonics, Nashville, TN.

"As you know, we have been very pleased with the DA-1000E. With the addition of the AD-500, we have added a new standard of musicality and sonic accuracy. When we compared the sound of the entire digital chain using the AD-500 and DA-1000E, the end result was the closest to the original source we have ever heard!"

--Steve Hall, Future Disc Systems; L.A., CA.

F. OPERATING VOLTAGES

1) AC Mains operation

Your DA-1000E has been designed to operate on a nominal 12 Volt DC power input to be free of AC mains influenced noise. This way the power supply is external to the sensitive analog and digital conversion circuitry.

We recommend you utilize APOGEE's table top TT1200 or rack mountable PS-1000. The TT1200 works on 110 VAC 60 Hz input. The PS-1000 will work from 100 to 240 VAC 50/60 Hz. Both units output the necessary 12 Volts DC at 1200 mA.

You can also provide your own power source, although we recommend use of linear power supplies and they are capable of providing 12 Volts DC at 1200mA. Less expensive wall mounted unregulated power adaptors may be used but we do not recommend them. (The typically large amounts of AC ripple in these units can compromise DA-1000E performance under low mains voltage conditions.) The actual input voltage may vary from 11.5V DC to 15V DC.

2) DC or bättery operation

The input voltage of the DA-1000E can vary from 11.5 to 15 Volt DC thus enabling portable power sources such as batteries. Some DA-1000E converters have been installed in automobiles and boats with excellent results.

Some important criteria must be followed to provide stable operation of the DA-1000E using battery power. The first is providing enough current to properly power the DA-1000E. Although the DA-1000E has internal electronic protection and protective fusing; as with any battery operated component, it is advisable to place a protective fuse on the positive side of the power.

G. GROUNDING

The DA-1000E can be operated in a wide variety of rack mount and free standing configurations. Grounding and ground loops are a concern in any audio environment. The DA-1000E has a very flexible grounding structure designed to enable optimum performance in both standalone and rack mount system approaches. There are three separate and isolated ground systems within the DA-1000E.

A. Chassis Ground - This ground is connected to pin 11 on the rear 15 pin HD sub connector and ties all mechanical components together to ground.

B. Battery Ground - This ground is tied to the incoming 12 volt power (can be battery). Ground on pin 14 of the rear 15 pin HD sub connector. The power supply pre-regulator and DC/DC converter are tied together.

C. Analog Ground - This ground is tied to the pin 1 pins of the XLR output connectors. This is the audio ground which also ties some of the digital circuitry such as the digital to analog converters.

If operating in a rack mount environment, it is often important to isolate the chassis ground from the analog ground to avoid ground loops from the various analog inputs through the chassis. Ground loops can also be caused through the AES output, pin 1 connection to the chassis ground and other devices via their ground connection to the AC ground.
Three grounding jumpers on the main board give control of the grounding connections.

Jumper JMP located on DC/DC Power supply P818 PSU board(square purple case w/ gold label)
ON connects battery ground to audio analog ground
OFF isolates battery ground from audio analog ground
(The DA-1000E is normally shipped with JMP in the OFF position)

Jumper JMP-901 located on the rear corner of the main board next to the 15 pin HD sub connector
ON connects chassis ground and 15 pin connector pin 11 to battery ground
OFF isolates chassis ground and 15 pin connector pin 11 from the battery ground
(The DA-1000E is normally shipped with JMP-901 in the OFF position)

Jumper JMP-900 located on the rear corner of the main board next to the 15 pin HD sub connector
ON connects chassis ground to audio analog ground
OFF isolates chassis ground from audio analog ground
(The DA-1000E is normally shipped with JMP-901 in the ON position)

If you need to change any of the above grounding configurations, consult section M1 for disassembly of DA-1000 and view figure 17 for jumper locations.

GROUNDING SIMPLIFIED

Chassis ground is isolated from battery ground. The audio analog ground is to the chassis ground. When the battery ground and the chassis ground are tied together via JMP901, the audio analog ground is isolated. See figure two below for a block diagram approach to grounding.

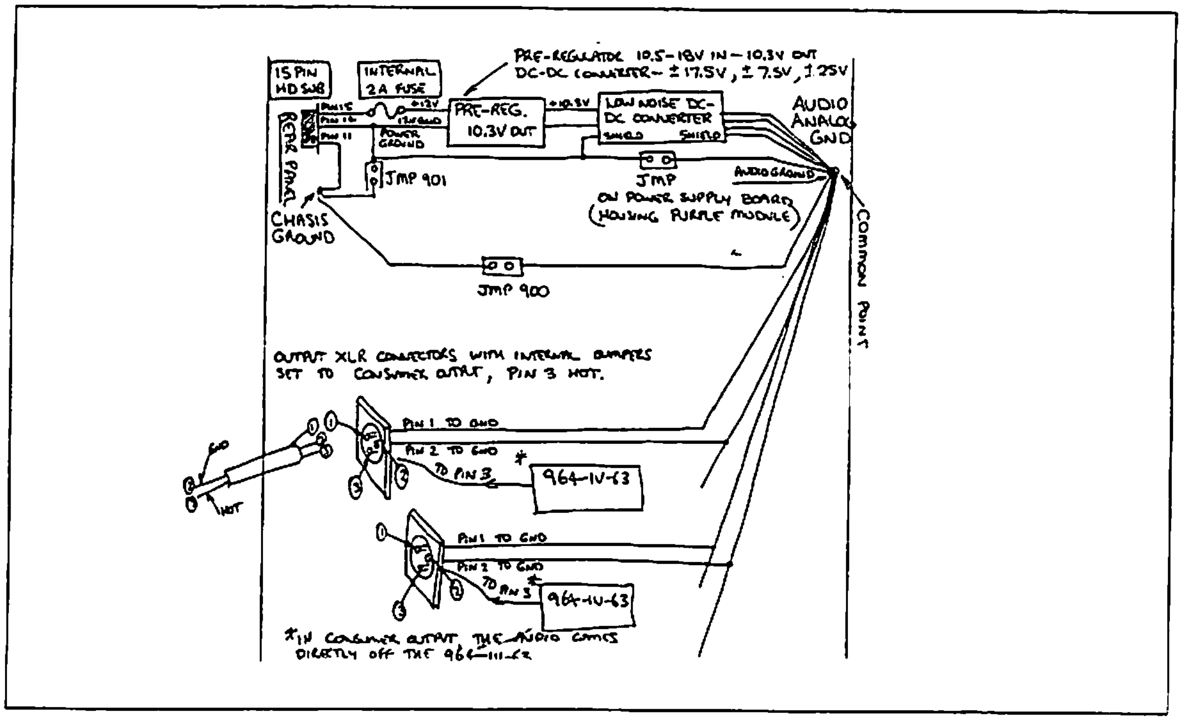


FIGURE TWO

H. LOCATION AND FUNCTIONS OF CONTROLS

1) Digital audio input selector and activity indicators.

Located on the left side of front panel is the INPUT selector. A three position switch selects the Digital Audio Input and two LED's indicate input signal activity. (See figure 3)

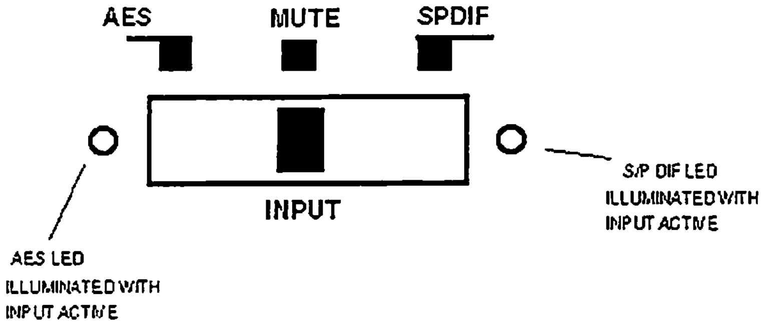


FIGURE THREE

AES position-
With the INPUT selector switch in the left most position, AES/EBU format is selected. An AES/EBU digital audio inputs is applied to the rear panel AES/EBU female XLR connector. (See Figure 7) The green AES indicator will illuminate whenever there is a signal present. The red MUTE and red ERROR lights will illuminate whenever the signal is not present and the AES input position is selected. (See section J3 for AES input impedance selection)

MUTE position-
With the INPUT selector switch in the center position, MUTE is selected. The MUTE LED indicator is located between the POWER selector and the EIAJ DELAY selector.(See figure 5) This selects the analog outputs to off by closing the output mute relays and loading the digital to analog convertors with digital 'audio black' (silence). Switching from either AES or SPDIF inputs to MUTE keeps the DA1000E locked to the last selected input. This removes any delay that may occur from having to reacquire the input.

SPDIF position-
With the INPUT selector switch in the right most position, S/P DIF format is selected. An S/P DIF digital audio input is applied to the rear panel S/P DIF OPTICAL input port or the S/P DIF electrical input connector(a female BNC or on older models this is a female RCA jack); located on the rear panel. (See Figure 7) *Additional electrical S/P DIF inputs are available by utilizing the AUX A/AUX B capability outline in section L3c.* The green SPDIF indicator will illuminate whenever there is a signal present. The red MUTE and red ERROR lights will illuminate whenever the signal is not present. Both the S/P DIF OPTICAL and electrical connections can be attached at the same time, but the electrical input will override the OPTICAL input.(ie- both inputs are connected and signals are being applied to both. The electrical input will be selected. When the unit generating the signal being applied to the S/P DIF electrical input is turned off or unplugged, then the OPTICAL input then becomes operational)

2) <u>Digital audio sample rate indication.</u>

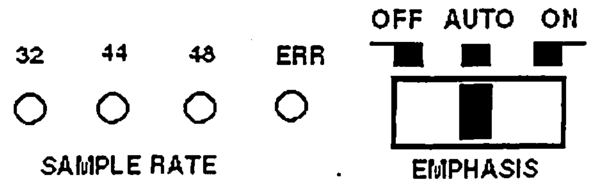


FIGURE FOUR

Three amber LED's and one red LED are located between the Input selector switch and the Emphasis selector switch.(See Figure 4) One of the three amber LED indicators will illuminate to indicate the digital audio input sampling frequency range of the digital audio signal that is selected, when it is present.. The red LED will illuminate to indicate an ERROR condition. An ERROR condition is any data which is non- recoverable. This would include noisy input lines.

3) <u>Emphasis auto/override control.</u>

Emphasis is often treated as a holdover from the early day of digital audio. Recording an audio signal with emphasis passes the analog signal through a special filter which boosts the high frequency sounds while leaving the low frequency sounds below 1KHz alone.

Audio signals in general tend to decay in level as the frequency is increased. this is especially true of classical music. Rock and roll is an exception and tends to have a lot of high frequency energy by way of direct inputs, hi-hats, cymbals, etc. By rolling off the high frequencies on playback we apply de-emphasis, which reverses the effects with a high frequency rolloff. The net result is an improved noise floor at higher frequencies.

Ten different people have ten different opinions on emphasis. In general, emphasis is a dying trend with its main use coming from the classical recording area. A number of recent model CD players produced in Japan don't support emphasis, resulting in lots of highs when playing back an emphasized program.

One of the main reasons for using emphasis was the F1 format which was the predecessor to DAT. The F1 (also known as EIAJ) processor took in analog signals and outputted digital audio in form of a video signal. This video signal was then recorded on inexpensive video recorders such as VHS and, at the time BETA. The F1 format applied pre-emphasis all the time. Although the F1 and it's relatives (Sony 501, 601, 701, etc.) were designed as consumer machines, they were extensively used as professional recorders which resulted in a lot of pre-emphasized masters.

Many of todays masters are made by way of DAT which in general does not offer the emphasis option.

The 50/15uS emphasis describes the shape of the pre-emphasis high frequency boost on recorc. The reciprocal de-emphasis high frequency cut is applied on playback. The DA-1000E only supports the 50/15uS standard of de-emphasis.

Emphasis on the DA-1000E is controlled by a three position selector switch. (See Figure 4)

OFF position (OFF override)-
With the EMPHASIS selector switch in the left most position, Emphasis is turned off. This function gives users control over incorrect emphasis information in the digital audio data stream, permitting override of the automatic emphasis selection function.

AUTO position (NORMAL position)-
With the EMPHASIS selector switch in the center position, Emphasis is in AUTO mode. In this mode, the DA-1000E will monitor the digital audio input in any format and check for data indicating that the material was recorded with emphasis. When emphasis on the digital audio data is present, the DA-1000E automatically switches into De-emphasis mode. This will de-emphasize the audio data. The amber EMPH LED will illuminate to indicate that de-emphasis is taking place. The EMPH LED is located between the EIAJ DELAY selector and the MUTE LED.(See figure 5) Whenever the EMPH LED is illuminated, the DA-1000E is de-emphasizing the digital audio input.

ON position(ON override)-
With the EMPHASIS selector switch in the right most position, Emphasis is in the ON mode. The Emphasis circuitry is switched in and the EMPH LED will illuminate. Any digital audio signal in any format will have de-emphasis applied. This function gives users control over incorrectly flagged emphasis indication in the digital audio data stream, permitting override of the automatic emphasis selection function.

4) <u>EIAJ DELAY Selector.</u>

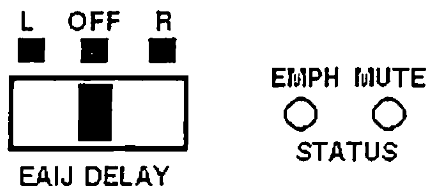


FIGURE FIVE

This is a three position selector switch. (See figure 5) This function corrects the inherent one-half word delay when playing back EIAJ format (F1) tapes .

L position-
With the EIAJ DELAY selector switch in the left most position. EIAJ Digital Delay of one-half word is selected to delay the left channel.

OFF position-
With the EIAJ DELAY selector switch in the center position. the EIAJ DELAY circuitry is completely switched out of the digital data. (See note below)

R position-
When the EIAJ DELAY selector switch in the right most position, EIAJ Digital Delay of one-half word is selected to delay the right channel.

(NOTE:This selector can be programmed to have additional functions. It can also be used to select additional S/P DIF and AES/EBU digital format signal inputs with the [AUX] capability and is further explained in section L3c on additional inputs.)

5) <u>Analog output level calibration adjustment</u>

Located between the MUTE LED indicator and the POWER selector switch are two multiturn screwdriver adjustable pots. (See Figure 6) These pots individually control the output level of both Left and Right analog audio signals which are available from the two male XLR connectors located on the back panel. (See Figure 7) Output level range depends on the position of the internal output selection configuration described in further detail in section J2.

6) Auto/manual power on select.

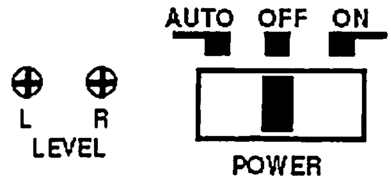


FIGURE SIX

Located on the right side of the front panel of the DA-1000E is the POWER selector switch. (See Figure 6)

AUTO position-
With the POWER selector switch in the left most position, the DA-1000E is in AUTO mode. In this mode, with no digital inputs, the DA-1000E uses very little current for low battery operating drain. The AUTO position permits you to operate the DA-1000E in conjunction with other equipment (ie, DAT recorder) for automatic power on selection. The DA-1000E will wait for a digital audio input signal to be applied to either the AES/EBU or electrical S/P DIF inputs before it goes into normal operation.

OFF position-
With the POWER selector switch in the center position, the DA-1000E is turned off.

ON position-
With the POWER selector switch in the right most position, the DA1000E is powered on.

J. LOCATION AND FUNCTION OF REAR PANEL CONNECTIONS

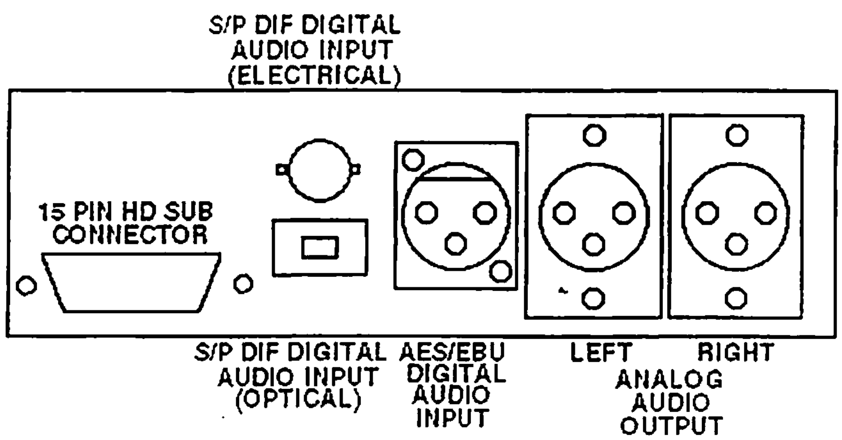


FIGURE SEVEN

The rear panel contains all input and output connections to the DA-1000E. (See Figure 7)

1) Left and Right Analog Audio Outputs

Two gold male XLR connectors are provided to access the analog audio outputs. We have equipped the DA-1000E with numerous grounding options and six different analog output pin configurations. This gives you the ability to cover all popular male XLR output connector configurations in use today. These include Pin 2 hot or Pin 3 hot at consumer output levels; Pin 2 hot or Pin 3 hot at unbalanced professional levels; and Pin 2 or pin 3 hot balanced at professional output levels.

In the two consumer level output positions (pin 2 hot, pin 3 hot), a full level digital input will deliver from +8dBu to +15.5dBu, adjustable with the front panel multiturn level controls. In the two professional balanced output positions (pin 2 hot, pin 3 hot), a full amplitude digital input delivers from +19dBu to +26.5dBu. In the two professional unbalanced output positions (pin 2 hot, pin 3 hot), a full amplitude digital delivers from +12.5dBu to +20dBu.

The selection between balanced and unbalanced is not automatic due to the sonic tradeoffs of automatic circuits. This requires you to make the output format selection using the simple six position plug inside.

CAUTION

It is important that you do not drive the balanced output into an unbalanced input as the output will be distorted. If there is a question as to whether the input is balanced or not, it is safer to use the unbalanced output positions.

2) Changing the analog audio output pin configuration

Your DA-1000E has been shipped with the XLR analog output connectors programmed for pin 3 hot, unbalanced.(This corresponds to Position Four as shown in figure 8) To change the output configuration, slightly loosen (two turns) the two bottom front panel hex screws with the supplied 3/32" hex wrench. Remove the top two screws on the front and the top two screws on the rear panel. (Do not remove the silver screw in the cover.) Remove the top cover by both hinging it over to the right, and by setting it upside down on the right side, as you look at the front panel.

Towards the rear of the unit just behind the two ribbed filters, you will see a "forest" of gold pins. (See figure 8) This is the output selection matrix. The eight position black housing where the wires from the rear XLR connectors terminate determines the analog output configuration. The most forward position towards the front panel is position number one. This is consumer output level, Pin 2 hot, Pin 3 Ground, Pin 1 Shield (ground).

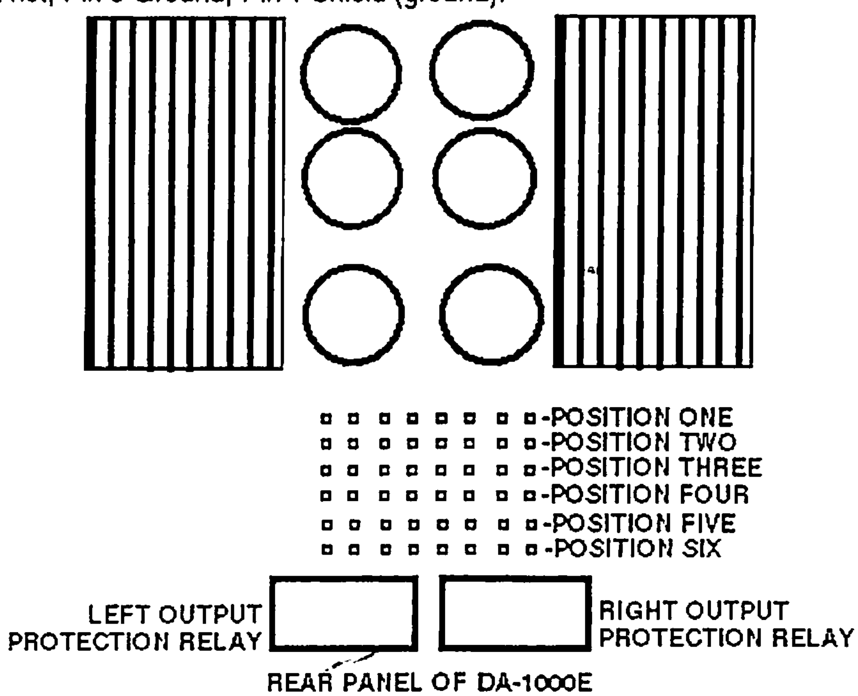


FIGURE EIGHT

The following chart identifies the position and the output configuration expected if used:

Position 1	CONSUMER LEVEL UNBALANCED PIN 2 HOT
Position 2	CONSUMER LEVEL UNBALANCED PIN 3 HOT
Position 3	PRO UNBALANCED PIN 2 HOT
Position 4	PRO UNBALANCED PIN 3 HOT
Position 5	PRO BALANCED PIN 2 HOT
Position 6	PRO BALANCED PIN 3 HOT

3) AES/EBU digital audio input

On the rear panel is a female XLR connector.(See figure 7) This connector is the input for AES/EBU digital audio. This is a transformer isolated at 110 ohms standard. There is the capability of changing this impedance to the old original AES/EBU standard of 220 ohms. This is achieved by removing the internal jumper on J104 pins 3 and 4. See figure 17 for location of J104. Refer to Section M on disassembly of the DA-1000E.
It is important to use the proper 110 ohm AES/EBU wire to ensure maximum performance. Available now from APOGEE is standard length prewired AES/EBU cables with high quality XLR connectors.

4) S/P DIF digital audio input - (electrical)

On the rear panel is a female 75 ohm female BNC connector. (See figure 7) This connector is the input for S/P DIF digital audio. This BNC connector is designed for use with the supplied BNC to RCA adapter.
It is important to use the proper 75 ohm SPDIF coaxial wire to ensure maximum performance. Available now from APOGEE is standard length prewired SPDIF cables with high quality BNC and RCA connectors.

5) OPTICAL digital audio input

On the rear panel an optical port. (See figure 7) This optical port will accept S/P DIF or AES/EBU digital formats that are transmitted optically. (Normally this is a consumer only format.)

6) 15 pin HD sub connector

The 15 pin HD sub connector is located on the rear panel. (See figure 7) The 15 pin HD sub connector provides a means of inputting the required supply voltage of nominal 12 volts DC at 1.0 amps. These power sources can be a 12 volt battery, an Apogee table top power supply(TT-1200/DA), or an Apogee rack mount dual power supply(PS-1000). The nominal 12 volts DC can vary from +11.5V DC to +15V DC with the ability to withstand short term peaks up to 30V DC. The power is applied to pin 15 (+) and pin 14 (return) of the 15 pin HD sub connector. Other digital inputs are also interfaced through this connector and will be further explained in section L3 on additional inputs.

K. A User's Guide to Understanding Digital Audio Interconnects

1) Digital Audio Without Making Your Eyes Glaze Over

You've probably read, or at least started to read many articles on digital audio. Like me, 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.......its 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 its 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 article will give you insight to deal with the peculiarities of digital interconnects without the usual technical smoke screen.

2) 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 accurately capture the smallest and biggest details in sounds. 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 around 25 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 is called the sample rate

A sample rate of 32,000 is used in digital broadcasting applications. Compact Disks use a 44,100 sample rate. you will often see these sample rates represented as KHz or Kilo Hertz (K - One Thousand; Hz - Cycles per second). A 44.1 sample rate is 44.1 KHz or 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 like 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 100,000 Hertz to faithfully reproduce 20,000 Hertz audio. A digital signal path for the same 20,000 hertz audio requires a frequency response of several million Hertz. 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.

3) 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. (see section H2)

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 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 only a very small percentage of recorders and work stations capable of handling or storing more.

4) 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 sample 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. Its 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 bar 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.

5) Its All In The Timing

A drummers timing can make the difference between good music and a memorable hit. Digital audio needs good timing to make it from one place to another with uncompromised sound quality. The timing in 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. (see section K6). 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.*

6) 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 its 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 drummers 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 moves up and down. The random click track variations around a perfectly steady tempo is 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 mains and external interference can all add dreaded 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 connections over high quality data cable will pick up less jitter than the same signal run through a bunch of microphone cable, XLR connectors and patch bays. A consumer coaxial wire connection is usually cleaner than the consumer 'TOSLINK' optical version, mainly due to 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 to A's, 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 to A are designed to ignore any jitter and the samples are correctly transmitted. Manufacturers can claim low jitter circuitry although its only a relative claim as at the moment there are no accepted standards for jitter measurement 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.

7) A Simple Technical Overview of Interconnect Formats Before AES/EBU

a) Sony SDIF Interface

The Sony SDIF interconnect is a good example of a basic digital connections between two digital audio boxes. SDIF format interconnects are found on the Sony 1610 and 1630 processors (used to generate CD masters on U-Matic video cassettes) as well as most 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 versus balanced for SDIF II. The 3324 and 3348 digital multi-track 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.

b) Mitsubishi (Melco) PD Interface

The Mitsubishi 2 track and multi-track interconnect formats are similar to the Sony SDIF format but not enough for direct connectability. 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 a 48 times multiple of 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 make errors in the audio when it is decoded.

The multi-track 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 have 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.

c) 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 lines 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 LSB's of the left and right samples. The stereo data is transmitted on only one lines 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.

d) JVC, NED, PCM-F1, Digidesign and Other Interconnects

These are some of the more obscure interconnects using multi-line interfaces. Some are proprietary, used to connect products from the same manufacturer and others are internal interconnects, sometimes adapted for connection to the outside world by another specialty manufacturer. You will probably never have to make a connection to another box with any of them so just be fulfilled in the knowledge they exist to complicate our lives.

8) 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 the different standards we lump together and call 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 cable to go from here to there. 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 application 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 multiline 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 4 more bits. With 32 bits available, there are 4 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 is redundant. The AES/EBU interconnect takes two bits of each subframe and calls them user data and channel status bits. To p[ack 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 intõ a string 192 bits long. The collection of 192 bits repeats every 230 time a second for a 44,100 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.

L. Firmware and Programs in the DA-1000E

1) The DA-1000E's Programmable Read Only Memory(PROM)

Your new DA-1000's functions and personality are controlled by a family of programs contained within a small plug-in chip called a PROM . This PROM enables the DA-1000E to automatically configure itself to the various different Digital Audio Input formats. Apogee has shipped units with PROM programs known as REV 3.15; REV 4.0B, REV 5.0, REV 5.0A, REV 5.1, REV 6.0 and REV 7.1. The serial number label located on the rear panel will indicate the version PROM installed in your unit. All units now shipped from the factory have the latest upgrade, REV 7.1. *NOTE: Older units with the PROM at a lower revision than 7.1 can be upgraded. Contact the factory for special instructions and update packages.* In concert with internal programmable jumpers, (J702, J703, J704, JP100 and JP200), the DA-1000E is capable of handling the following formats:

FORMAT TYPE	REV 3.15	REV 4.0B	REV 5.0	REV 5.0A	REV 5.1	REV 6.0	REV 7.1
AES/EBU	yes	yes	yes	yes	yes	yes	yes
S/P DIF	yes	yes	yes	yes	yes	yes	yes
SDIF(balanced/unbalanced)	yes	no	yes	yes	yes	yes	yes
Yamaha	yes	no	yes	yes	yes	yes	yes
JVC	yes	no	no	no	no	no	no
Mitsubishi Two Track	yes	no	no	no	no	no	no
Mitsubishi Multitrack	yes	no	no	no	no	no	no

TABLE ONE

2) Phase Reversal Mode Option

PROM versions REV 5.0A , REV 5.1, REV 6.0 and REV 7.1 have the capability of phase reversal. By adding a jumper to JP705, the outputs of the DA-1000E are phase reversed 180 degrees. This option works in all format modes.
It is also possible to add a switch connecting the two contacts of JP705 and running wires out through the vent holes on the top of the DA-1000E to an external switch thus enabling easy phase reversal switching. Experienced listening will reveal whether the material being monitored is in phase or reversed phase.
Refer to Section M2 for access to JP705. Solder two wires to both contacts of JP705 and connect the wires to a Single Pole Single Throw (SPST) switch. See figure 9.

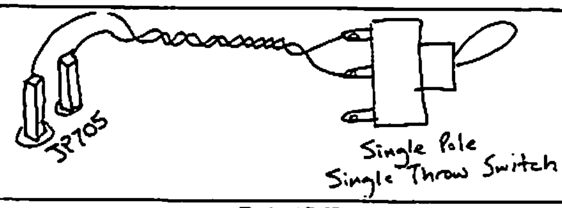


FIGURE NINE

3) Additional Inputs

In addition to controlling configurations, PROM REV 4.0B through REV 7.1 allows the capability of allowing multiple AES/EBU OR S/P DIF inputs. This gives the DA-1000E even more capability.

As shipped from the factory, AES/EBU and S/P DIF (electrical and optical) are functional. The DA-1000E has other inputs, located on the 15 pin HD sub-connector, which enable it to accept other types of inputs and other types of formats. The next few sections will inform you as to how to make the DA-1000E accept these formats and inputs.

a) Standard Mode

The DA-1000E is shipped from the factory in "Standard" mode. Connecting an AES/EBU or S/P DIF format signal to the appropriate connector on the back panel is all that is needed. Standard Mode also includes the ability to receive SDIF inputs. The jumpers are set up ready for SDIF <u>with the exception of one</u>. This is to eliminate the possibility of unwanted mutes when receiving AES/EBU or S/P DIF digital inputs. These mutes can be due to unterminated auxiliary inputs used for SDIF reception. (Please see section L3b for SDIF operation) Before making any changes to the jumpers, read section M on Disassembly.

The JP100 and JP200 jumper configurations for Standard mode are as shown below.

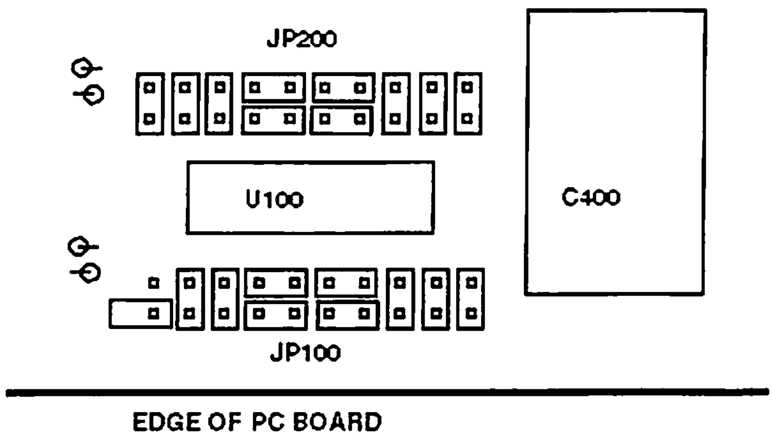


FIGURE TEN

JP702	OFF
JP703	OFF
JP704	ON

Refer to Figure 17 for locations of JP702, JP703 and JP704. Before accessing these jumpers, please refer to section M2.

b) Sony Digital Interface(SDIF) Operation Mode

With PROM REV 3.15, REV 5.0, REV 5.0A, REV 5.1, REV 6.0 and REV 7.1 this format can be received by the DA-1000E. The DA-1000E only needs to have one jumper changed internally and the proper cable to access the SDIF inputs. The cables can be purchased from APOGEE or you can follow the diagram below to put together a cable. APOGEE supplies the SDIF cable accessory as a TT1200/DA/SDIF (Table Top power supply with cabling) or the PS-1000/DA/SDIF (Cabling for PS-1000 Rack mount Power Supply). Contact your distributor or APOGEE for more information.

When SDIF input operation is required, a special cable assembly is needed. This uses three BNC connectors known as AUX A(word sync), AUX B(Channel 1-Left Data), and AUX C(Channel 2-Right Data).

Word Sync (also known as Word Clock) is a symmetrical square wave at the sampling frequency. For example, if the sampling rate is 44.1KHz, then the word sync is a square wave at 44.1KHz. Word Sync (Word Clock,WC) is often used to distribute timing to more than one digital processor to synchronize them at the correct sampling rate and phase.

Data Left is a serial representation of the left (or A) digital audio. The data rides in 32 bit cells for each individual digital audio sample. This repeating train of 32 cells contain a code representing the digital audio in addition to a flag used to indicate whether emphasis was used when converting from analog to digital. If the sampling rate was at 44.1KHz, then these trains of 32 cells pass by at 44,100 times per second.

Data Right (or B) is a serial representative of the right digital audio in the same format as described in the Data Left paragraph above.

On the special cable assembly there are three BNC connectors which go to the back panel 15 pin D-SUB connector. (See Figure 12) You will also need to change one of the internal Jumper configurations(JP100) to fully activate the SDIF mode, if your unit is already in standard configuration(as shipped from factory unless otherwise specified at time of sale). If you do not know what configuration your unit is in, then it is suggested that you verify the below settings to confirm SDIF mode operation.

NOTE: Whenever a digital audio input is present on the AUX inputs, it will override the AES/EBU, S/P DIF or Optical inputs and cause the DA1000E to reprogram itself to receive the AUX input, in this case SDIF. The AUX A input(word clock) automatically selects the AES/EBU override when active. Removing AUX A input or turning the power off to the AUX device(operating SDIF) causes the DA-1000E to reacquire the selected AES/EBU, S/P DIF or Optical inputs.

The JP200 and JP100 configurations for SDIF mode are as shown below.

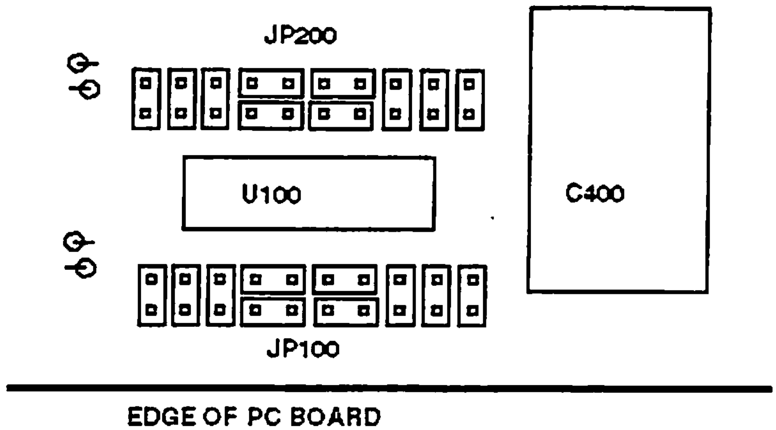


FIGURE ELEVEN

Refer to Figure 17 for locations of JP702, JP703 and JP704. Before accessing these jumpers, please refer to section M2.

JP702	OFF
JP703	OFF
JP704	ON

The diagram below shows the necessary wiring that is needed to operate in SDIF mode with the DA-1000E.

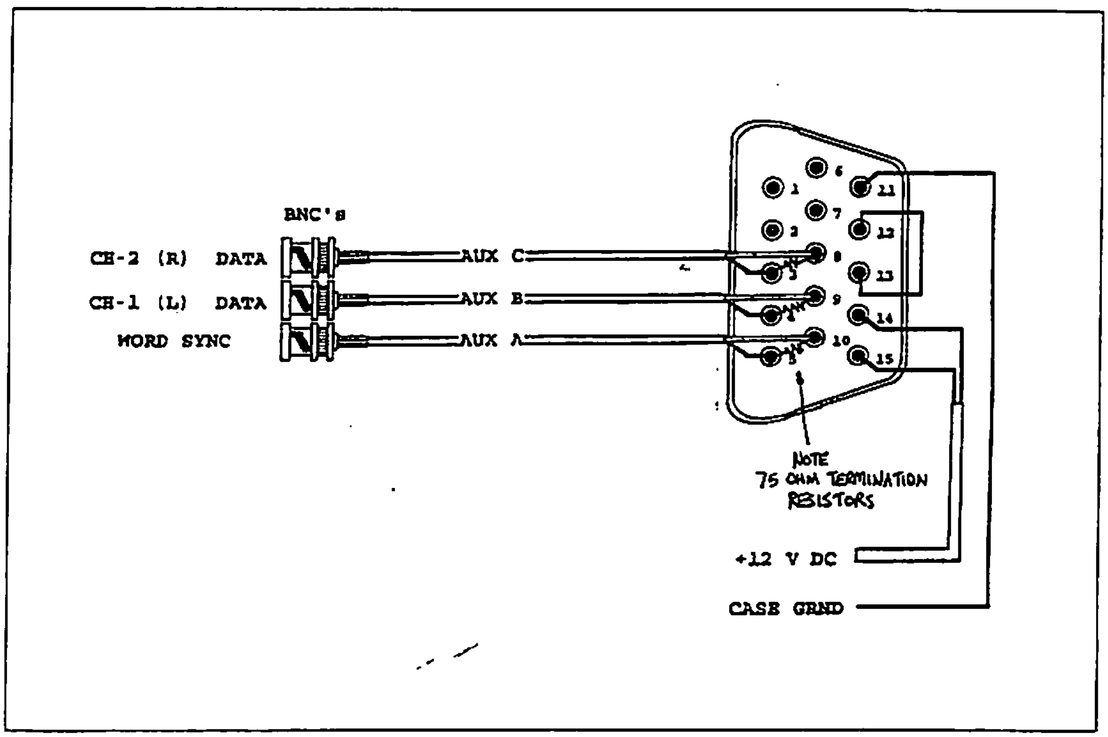


FIGURE TWELVE

c) AUX A/AUX B mode

With PROM REV 4.0B through REV 7.1, it is possible to add two more AES/EBU or S/P DIF digital audio inputs to your DA-1000E. These extra inputs are available if you do not use any of auxiliary inputs for interfaces such as SDIF, Yamaha, or Mitsubishi. The AUX A/AUX B mode is activated by setting the appropriate Jumpers and using the proper cable to access the inputs. The cables can be purchased from APOGEE or you can follow figure 14 to assemble your own cable. APOGEE supplies the AUX A/AUX B cable accessory as a TT-1200/DA (Table Top power supply with cabling) or the PS-1000/DA (Cabling for PS-1000 Rack mount Power Supply). Contact your distributor or APOGEE for more information

When AUX A or AUX A and AUX B input operation is required, a special cable assembly is needed. This uses BNC connectors known as AUX A and AUX B. These two BNC connectors on the cable assembly go to the back panel 15 pin D-SUB connector. (See Figure 14) You will also need to change the internal Jumper configurations if your unit is already in standard configuration(as shipped from factory unless otherwise specified at time of sale). If you do not know what configuration your unit is in, then it is suggested that you verify the below settings to confirm AUX A/AUX B mode operation. Before making any changes to the jumpers, read section M on Disassembly.

The AUX A and AUX B connectors do not use transformers as do the standard digital inputs, AES/EBU and S/P DIF (electrical or optical). This option thus makes it useful for interfacing to equipment that does not work optimally with the transformer inputs. When only one AUX input is needed, use the AUX A input. (This is because the AUX B connector will not work unless there is a signal present at the AUX A connector).

To select between the two AUX inputs: the jumpers JP702, JP703 and JP704 change the operation of the front panel selector switch, EIAJ DELAY. This selector switch now becomes the selector switch for AUX A and AUX B. With the switch to the left, AUX A is activated. With the switch to the right, AUX B is selected. *NOTE: AUX A must also be active for AUX B to work even if AUX A is not selected.* When the switch is in the middle position, AUX A and AUX B are de-activated and the standard inputs are activated. *NOTE: INPUT selector switch must be switched to SPDIF input.*

The JP200 and JP100 configurations for AUX A/AUX B mode are shown below.

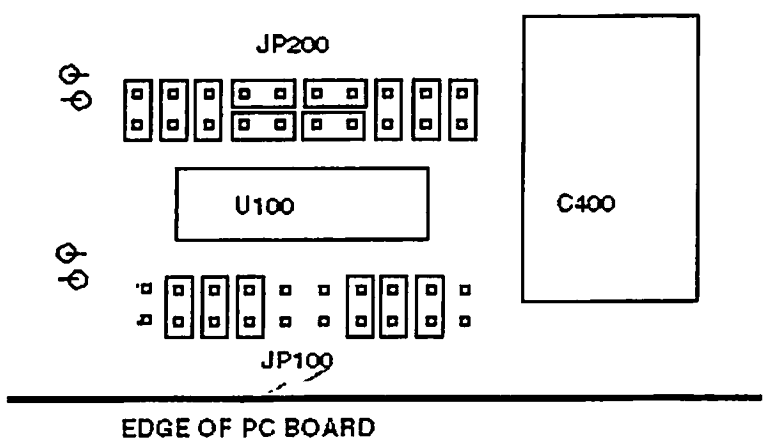


FIGURE THIRTEEN

Refer to Figure 17 for locations of JP702, JP703 and JP704. Before accessing these jumpers, please refer to section M2.

JP702	ON
JP703	OFF
JP704	OFF

For AES/EBU inputs, the terminating resistor should be 120 ohms and S/P DIF inputs should be 75 ohms. *NOTE: the diagram in FIGURE 14 shows 75 ohm resistors thereby being an AUX A/AUX B cable assembly for use in S/P DIF applications.* If you want to drive another piece of digital audio equipment with the same AES/EBU or S/P DIF line, the termination resistor can be omitted and the signal looped on to the next digital input. *(Note: This may not work in all situations due to reflections in the connectors and cables.)*

The diagram below shows the necessary wiring that is needed to operate in AUX A/AUX B mode with the DA-1000E.

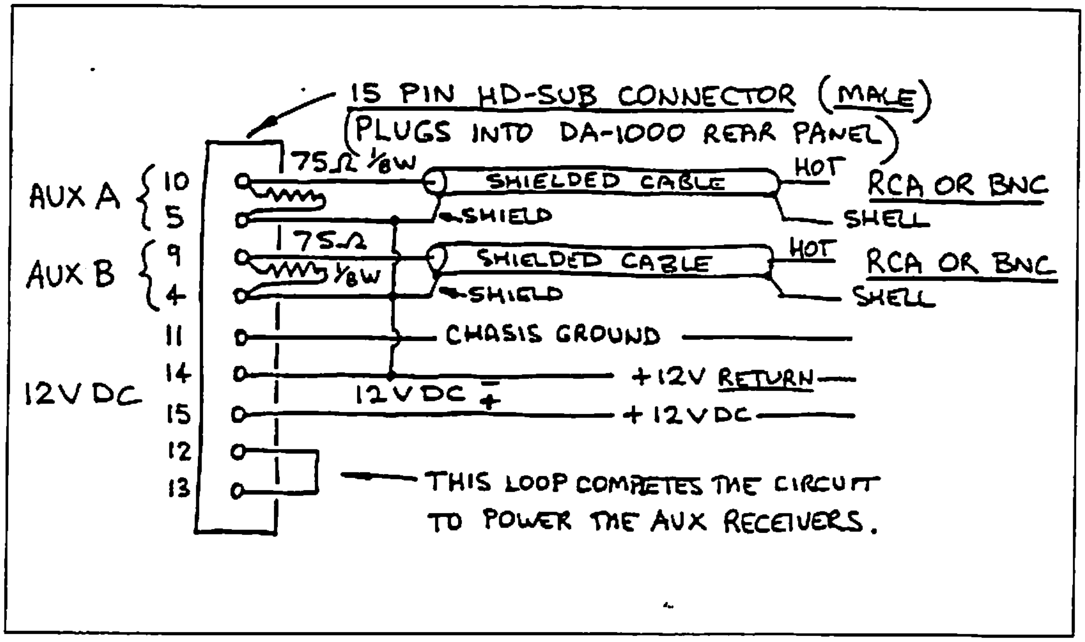


FIGURE FOURTEEN

d) Yamaha Operation Mode

With PROM REV 5.0 through REV 7.1 Yamaha format can be received by the DA-1000E. The DA-1000E only needs to have jumpers changed internally and the proper cable to access the Yamaha inputs. The cables can be special ordered from APOGEE or you can follow diagram 16 to put together a cable. APOGEE supplies the Yamaha cable accessory as a TT1200/DA/YAM (Table Top power supply with cabling) or the PS-1000/DA/YAM (Cabling for PS-1000 Rack mount Power Supply). Contact your distributor or APOGEE for more information.

When Yamaha input operation is required, a special cable assembly is needed. This uses an eight pin circular DIN connector in which two signals known as AUX A(word sync or word *clock) and* AUX B(Right and Left Data) are utilized.

Word Sync (also known as Word Clock) is a symmetrical square wave at the sampling frequency. For example, if the sampling rate is 44.1KHz, then the word sync is a square wave at 44.1KHz. Word Sync (Word Clock,WC) is often used to distribute timing to more than one digital processor to synchronize them at the correct sampling rate and phase.

Right and Left Data is a serial representation of the both the left (or A) and right (or B) digital audio. The data rides in 32 bit cells for each individual digital audio sample. This repeating train of 32 cells contain a code representing the digital audio in addition to a flag used to indicate whether emphasis was used when converting from analog to digital. If the sampling rate was at 44.1KHz, then these trains of 32 cells pass by at 44,100 times per second. In this format the least significant bit is at the front of the data cell as opposed to other format in which the most significant bit is first.

On the special cable assembly there is an eight pin Male DIN connector which goes to the back panel 15 pin D-SUB connector. (See Figure 16) You will also need to confirm the internal Jumper configurations settings for JP100 and JP200 to fully activate the Yamaha mode. (unless your unit is shipped from factory specified for Yamaha mode at time of sale). Before making any changes to the jumper configurations, read section K on Disassembly. If you do not know what configuration your unit is in, then it is suggested that you verify the below settings to confirm Yamaha mode operation.

NOTE: Whenever a digital audio input is present on the AUX (Yamaha) inputs, it will override the AES/EBU, S/P DIF or Optical inputs and cause the DA1000E to reprogram itself to receive the AUX input, in this case Yamaha mode. The AUX A input(word clock) automatically selects the AES/EBU override when active. Removing AUX A input or turning the power off to the AUX device (operating Yamaha mode) causes the DA-1000E to reacquire the selected AES/EBU, S/P DIF or Optical inputs.

The JP200 and JP100 configurations for Yamaha mode are as shown below.

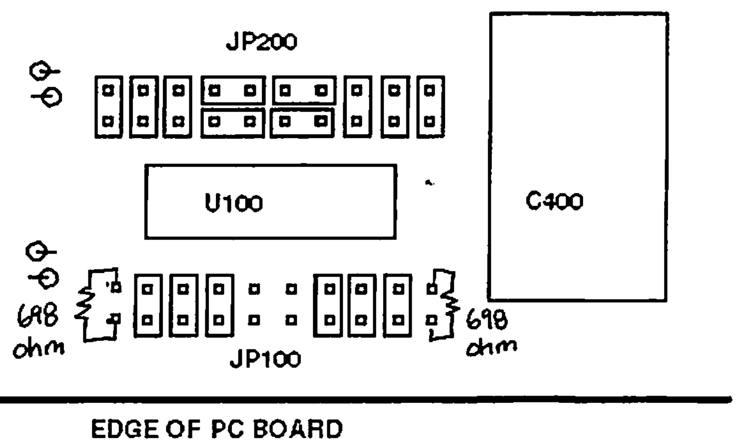


FIGURE FIFTEEN

Refer to Figure 17 for locations of JP702, JP703 and JP704. Before accessing these jumpers, please refer to section M2.

JP702	ON
JP703	OFF
JP704	ON

The diagram below shows the necessary wiring that is needed to operate in Yamaha mode with the DA-1000E.

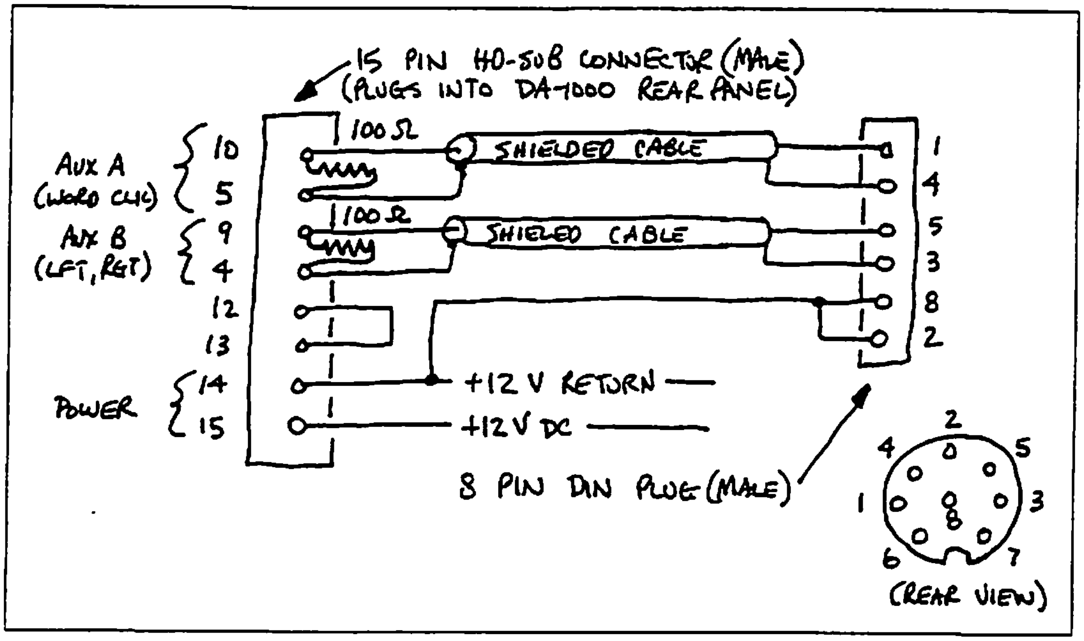


FIGURE SIXTEEN

4) Removing and Upgrading the PROM

One of the built-in features of your DA-1000E is the ability to be upgraded in the field. With just a change of the PROM will enable you to always keep the firmware at the latest revision. As APOGEE makes changes in the PROM for the DA-1000E, you will receive a free upgrade within the warranty period. It is very easy to change this part. The following instructions will guide you.

a. Refer to Section M on Disassembly to access the DA-1000E circuit board.

b. Check for the PROM that is already installed and make a note of which revision level it is at(The revision level is marked on top on PROM in white ink). In TABLE ONE of this manual lists the PROM revision level and the applicable digital audio formats that it supports. Figure 17 identifies location of PROM (U701).

c. Locate the PROM extraction tool (with pink handle) in the supplied accessories kit. Hook and pull out old PROM. (See Figure 17)

d. Insert the new PROM taking care to install it with the beveled side and arrow or dot facing the purple power supply. (See figure 17)

M. Disassembly of the DA-1000E

1) Enclosure Disassembly

Ensure that the power is disconnected from the DA-1000E.

Using the supplied 3/32" hex wrench (Allen key) loosen the two bottom front panel screws two turns. Then remove the two top front panel screws and the two top back panel screws. (Do not remove the silver screw in the top cover.)

Remove the top cover of DA-1000E by 'hinging' it over to the right and by setting it upside down on the right side as you look at the front panel. Do not disconnect wires from the component mounted on top cover.

Utilize figure 17 to locate the particular place on the PC board in which you need to change jumpers or change EPROM.

To reassemble the DA-1000E, follow the steps for disassembly in the reverse order.

2) Access to JP702, JP703, JP704 and JP705

See figure 17 for location of JP702, JP703, JP704 and JP705. Note that these jumpers are located under the small square circuit board which has a purple unit affixed to it. There are three retaining clips which hold this circuit board down. To release the circuit board use needle nose pliers to compress clip while applying upward pressure to circuit board. Once free, carefully swing circuit board over to one side. There are still wires connected to the small circuit board so be careful not to pull hard on the wires. There is enough slack in the wiring harness to allow flipping the circuit board on its back.

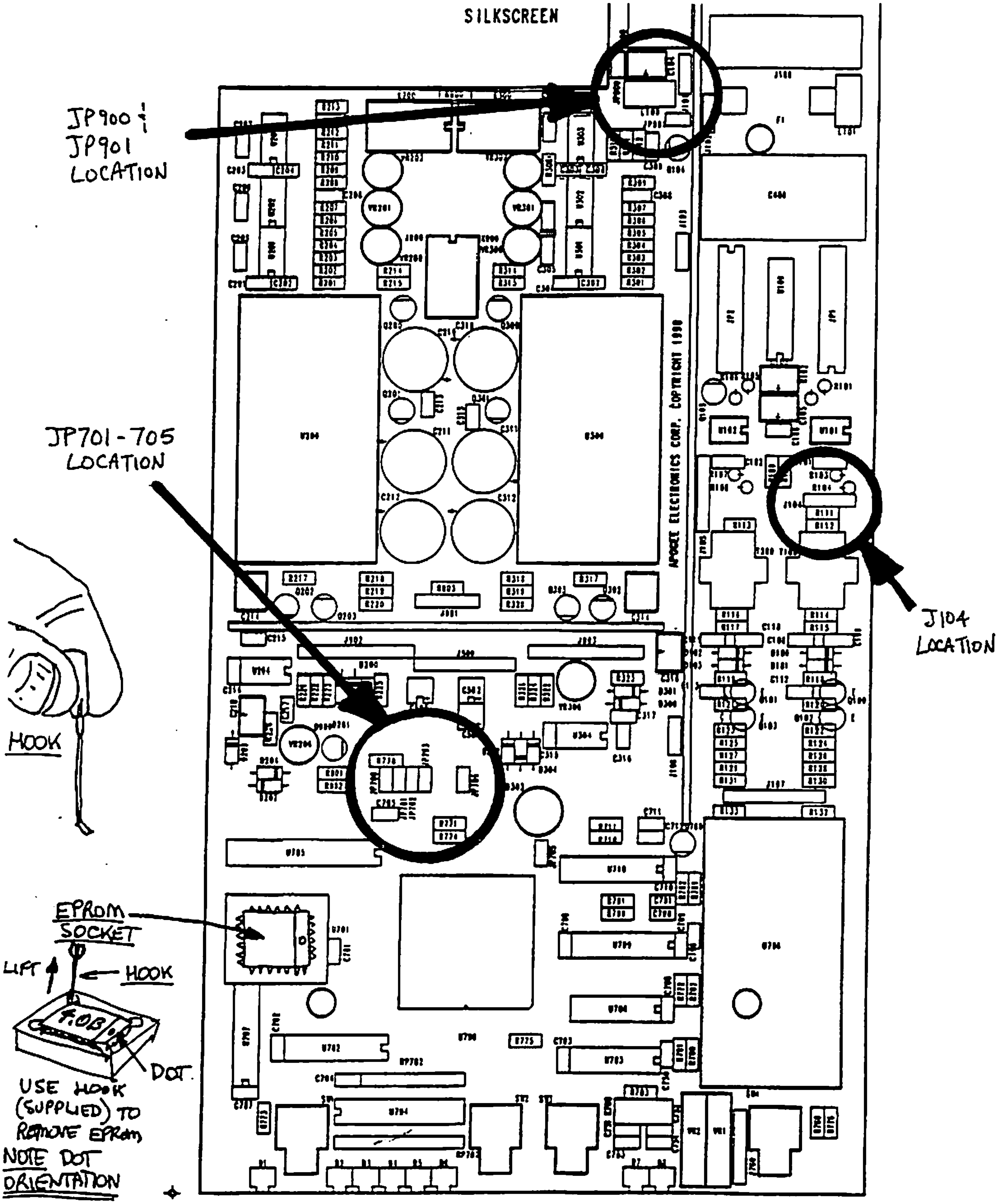


FIGURE SEVENTEEN

N. Using the DA-1000E with Panasonic DAT's SV3700 and SV3900

(This is a technical note to improve the S/P DIF output on the above DAT machines for all digital audio applications, not just the DA-1000E)

1) Probable Causes for poor S/P DIF operation by Panasonic DAT's

This report is from an analysis done by Bruce Jackson, Apogee Electronics on June 11, 1991.

Many owners of Panasonic SV3700 and SV3900 DAT machines have reported troubles locking digital audio gear to the Panasonic DAT's S/P DIF output. This analysis points out several deficiencies located in the S/P DIF output circuitry of the DAT machines. The problem is caused by four different situations:

1. The output drive is supplied by a 74HCU04 inverter IC, which is unable to drive the subsequent circuitry symmetrically.
2. Low frequency coupling is adversely affected by the coupling capacitor, which is too small.
3. The digital output transformer has an insufficient core to drive the long sync pulse embedded in the S/P DIF serial data stream.
4. The large inductor and capacitor installed to curb RF radiation (probably for FCC regulations) causes waveform distortion in the form of high frequency rolloff and phase delay.

The best remedies would be an improved design, but as there are a large number of Panasonic units in the field, we recommend the following corrective actions. They are listed in the order of effectiveness.

2) Corrective actions to improve S/P DIF operation on Panasonic DAT's

1. Replace the S/P DIF transformer with an improved design, such as the one available from APOGEE (Part Number RL-2560; contact APOGEE for cost) The transformer is not a direct replacement but can be modified to work by clipping the center pins and spreading the leads. This fix corrects specific symptoms related to units that do not lock or stay locked on frequency from the S/P DIF output.
2. Replace the 0.01uF coupling capacitor (C944) with a 0.1uF coupling capacitor (Part Number 0.1uF 0805 Cap; contact APOGEE for cost). This capacitor is a surface mount part, 0.1uF, 50 Volt, 0805 size, ceramic X7R or Z5U material capacitor.
3. Bypass the output filter (inductor L922) by soldering a jumper across the pads on the underside of the board. As an additional step remove C946, a surface mount 47pF ceramic filter capacitor. This only contributes a small improvement.

NOTE: Removing C944 and L922 will cause the S/P DIF output RCA to radiate increased RF energy when not connected. This may affect the FCC regulation compliance. This should not be a problem if an RCA cable is attached to the next piece of equipment in the chain. It is also not problem when the AES output is active because the Panasonic unit disengages the S/P DIF output when AES is activated.

Performing all three modifications delivers substantially improved S/P DIF waveforms. In our tests at APOGEE, a 500MHz oscilloscope observing a S/P DIF waveform showed good waveform fidelity with slightly reduced symmetry due to the 74HCU04 driver IC located on the Panasonic machines. The APOGEE DA-1000E which initially wouldn't lock to the

Panasonic, locked immediately with only modification number 1. A full implementation of all three modifications should satisfy <u>any</u> S/P DIF requirement.

3) <u>Instructions to preform suggested modifications to Panasonic DAT's.</u>

a) Replacing the S/P DIF output transformer

Remove the digital in/out terminal circuit board by removing the four screws from the 'AES/EBU' XLR connectors and the screw between the 'IEC TYPE II' RCA connectors located on the rear panel. Place an insulating surface over the circuitry inside the unit (piece of paper, etc.) and desolder the 5 pins of T901 on the solder side of the board. Remove the stock transformer as shown in Figure 18.

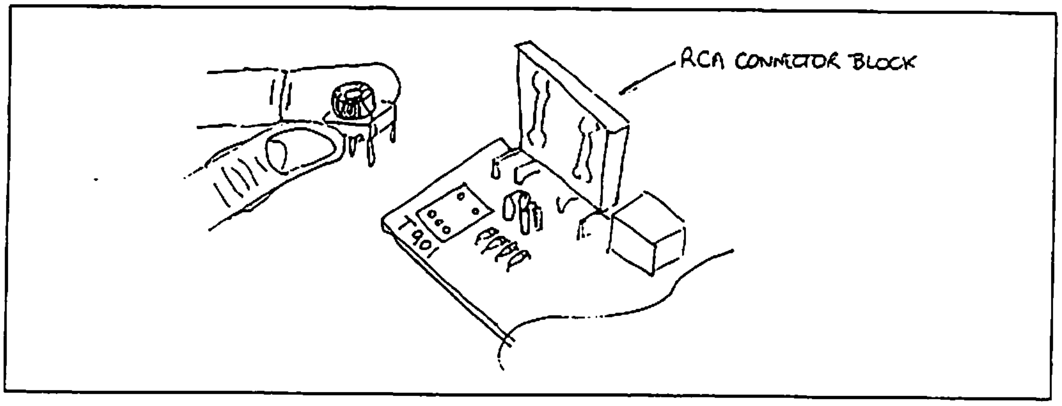


FIGURE EIGHTEEN

Take the APOGEE transformer (Part Number RL-2560) and clip two center pins and splay the other four pins to match the footprint of the original transformer unit removed. See figure 19.

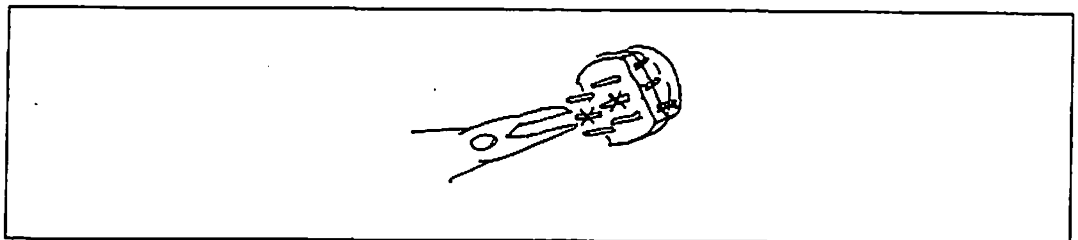


FIGURE NINETEEN

Insert the modified APOGEE transformer into the T901 position of the Panasonic board. Align the red dot on the transformer so that it is pointing towards the front panel. See figure 20.

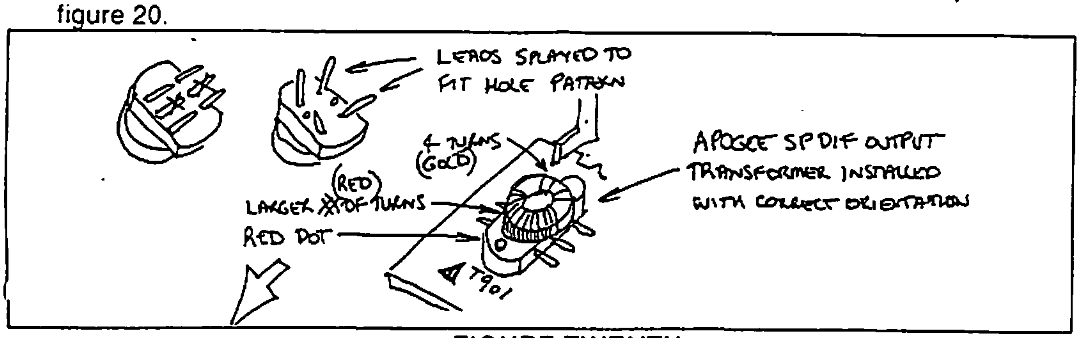


FIGURE TWENTY

b) Changing capacitor C944

Remove C944 using surface mount soldering tweezers or with solder wick. Replace with same size (0805 size) 0.1uF 50 Volt ceramic surface mount capacitor. See figure 19.

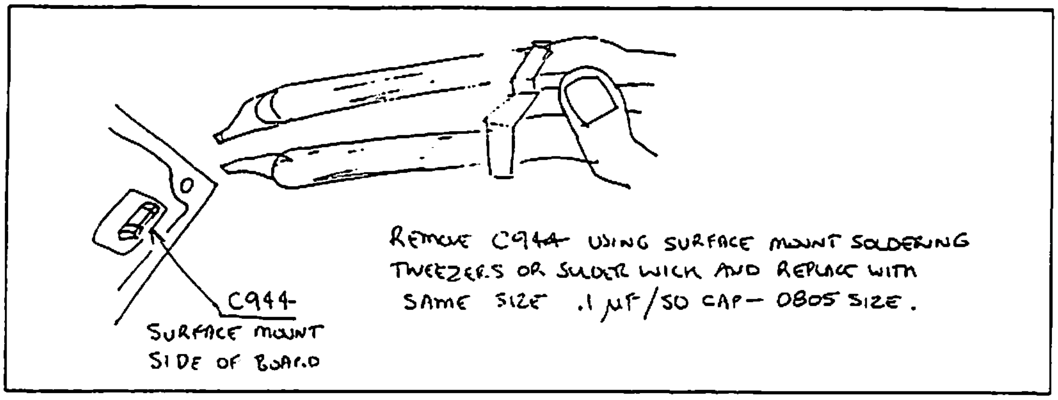


FIGURE TWENTY-ONE

c) Jumping RF inductor and optional removal of C946

On solder side of PC board remove C946. Place a jumper wire across L922 to defeat this output filter or replace L922 with smaller size (Panasonic part available from DigiKey). See figure 20.

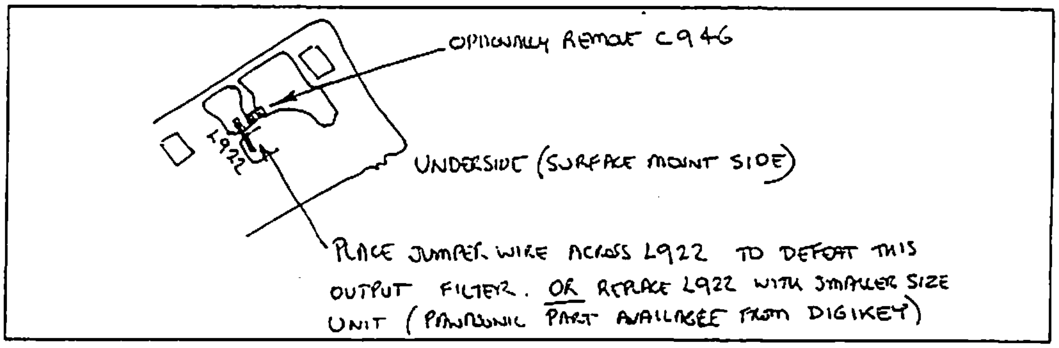


FIGURE TWENTY-TWO

P. MAINTENANCE

If the DA-1000E 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. The electrolytic capacitors used are of a high quality and will give many thousands of hours service.

The DA-1000E contains no serviceable components other than the PROM upgrades which can be installed by the user. Refer to qualified service personnel for repair or upgrade. Your warranty will be voided if you tamper with the internal components. If you should have any question with regard to the above, please contact APOGEE at (310) 915-1000 by phone or (310) 391-6262 by fax.

Q. WARRANTY

Be sure to return the enclosed warranty card. APOGEE will contact you with any manual updates. As enhancements and upgrades are developed, you will be contacted at the warranty registration address. Please address any inquiries to your dealer or directly to APOGEE:

APOGEE ELECTRONICS CORPORATION
3145 Donald Douglas Loop South
Santa Monica, California 90405
USA
TEL- (310) 915-1000
FAX- (310) 391-6262

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 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.

This warranty is void if APOGEE determines, in its sole business judgement, 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 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.

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

RMA-RETURN MATERIAL AUTHORIZATION

In the event your DA-1000E needs to be upgraded or repaired, it is necessary to contact APOGEE prior to shipping, and a RMA (Return Materials Authorization) number will be assigned. This number will serve as a reference for you and helps facilitate and expedite the return process. APOGEE requests that all domestic returns be sent via UPS, and all international returns shipped via Federal Express--unless otherwise authorized in advance.

Any shipment sent without a RMA number will not be accepted.

DA-1000E MANUAL REVISION D
DECEMBER 15, 1992
BRAD SANDERS

IF MISTAKES ARE FOUND, PLEASE CONTACT APOGEE ELECTRONICS.