

Model 3000S 96kHz Digital Optimizer

Operations Manual

Lavry Engineering, Inc.

<http://www.lavryengineering.com>

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Address questions or comments to: info@lavryengineering.com

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PART I: Introduction

Operation of the **Lavry Engineering Model 3000S Digital Optimizer** requires the use of five push button switches located on the front panel. The two-line liquid crystal display is organized for quick and intuitive operator interface. The unit features eleven screen displays.

Push Button Switches:

- ENTER** introduces and moves the cursor through screen positions that can be edited by the operator.
- CURSOR UP/DOWN** push buttons allow the user to choose a screen and to select the cursor position. The UP/DOWN buttons are also utilized when changing digits of a selected entry.
- GO** executes an edited screen.
- EXIT** removes the cursor and allows selection of other screens with the UP/DOWN buttons. Pressing EXIT again will cause the unit to go to the main screen.

Scroll Through Screens:

If the screen cursor is present, you need to remove it by pressing **EXIT**. When the screen cursor is not present, the **UP** and **DOWN** buttons serve to move from screen to screen.

Edit Screens

Press **ENTER** to introduce the screen cursor at the first editable location. Each additional pressing of **ENTER** will move the screen cursor to the next editable location. The settings at an editable location may be changed by the use of the UP/DOWN buttons.

Holding the **UP** or **DOWN** button will automatically advance the settings. This feature is particularly useful for increasing or decreasing numbers (such as test tone frequency, preset number, word length and amplitude).

The **GO** button is used to execute an action. Seven screens react to a change of settings immediately, and pressing **GO** is not necessary. Three screens require the operator to press **GO** for execution (this allows the operator to edit more than one setting before execution). Details are found in the following sections.

The following pages are organized to match the display screens. Some illustration screens may contain more lines of information than the display itself normally does. They are intended as a quick guide to possible settings.

Main Screen

IN:PRO 44	44100	50/15uS
OUT:CON 44	BYPASS MODE	

The top line consists of:

- IN:** Indicates input signal information
- PRO** or **CON** (professional or consumer format)
- 44** or **48** (input frequency indication embedded in the input format)
- 44100** input frequency measurement to 1 Hz
- CCITT** or **50/15uS**, **NO-EMPH** or **NOT-INDICATED** emphasis indication

The bottom line consists of:

- OUT:** Indicates output signal information
- PRO** or **CON** (professional or consumer format)
- 44** or **48** (output frequency indication embedded in the output format)
- BYPASS** output operating mode. Possible output frequency or sync modes:
 - **BYPASS MODE**
STRAIGHT-THROUGH MODE, OUTPUT FQ = INPUT FQ 40-100K
 - **44.1k BYPASS**
JITTER ATTENUATION MODE, OUTPUT FQ = INPUT FQ 44.1K +-100 PPM
 - **48.0k BYPASS**
JITTER ATTENUATION MODE, OUTPUT FQ = INPUT FQ 48.0K +-100 PPM
 - **44.1 kHz XTAL**
INTERNAL CRYSTAL CONTROLS OUTPUT FREQ., SAMPLE-RATE-CONVERSION
 - **48.0 kHz XTAL**
INTERNAL CRYSTAL CONTROLS OUTPUT FREQ., SAMPLE-RATE-CONVERSION
 - **AES LOCK 44.1**
44.1K +-100 PPM LOCKS OUTPUT FREQUENCY TO SYNC INPUT
 - **AES LOCK 48.0**
48.0K +-100 PPM LOCKS OUTPUT FREQUENCY TO SYNC INPUT
 - **AES WIDE LOCK**
40K-50K WIDE LOCKS OUTPUT FREQUENCY TO SYNC INPUT
 - **WC LOCK 44.1**
44.1K +-100 PPM LOCKS OUTPUT FREQUENCY TO SYNC WORD CLOCK INPUT
 - **WC LOCK 48.0**
48.0K +-100 PPM LOCKS OUTPUT FREQUENCY TO SYNC WORD CLOCK INPUT
 - **WC WIDE LOCK**
40K-50K WIDE LOCKS OUTPUT FREQUENCY TO SYNC WORD CLOCK INPUT
 - **88.2 kHz XTAL**
INTERNAL CRYSTAL CONTROLS OUTPUT FREQ., SAMPLE-RATE-CONVERSION
 - **96.0 kHz XTAL**
INTERNAL CRYSTAL CONTROLS OUTPUT FREQ., SAMPLE-RATE-CONVERSION
 - **24 Bit ENCODE**
ENCODES 24 BIT DATA INTO TWO HIGH-SPEED 16 BIT WORDS
 - **24 Bit DECODE**
DECODES 24 BIT DATA FROM TWO HIGH-SPEED 16 BIT WORDS
 - **2:1 DOWNSAMPL**
SYNCHRONOUS DOWNSAMPLING FROM 88-96K TO 44-48K

24 High Frequency Sample Rate Conversion

```
IN:PRO 44 44100 50/15uS
OUT:CON 44 96.0 kHz XTAL
```

Up/Down Sample Rate Conversion between high-speed modes (88.2 kHz and 96 kHz sample rates) and low speed modes (44.1 kHz and 48 kHz sample rates) is accomplished by the Model 3000S. The input is automatically sensed by the 3000S, and is displayed on the top line. The output frequency is selected by the mode select procedure above.

DC Removal, Boost/Cut, and Delay functions are disabled in high-speed output modes. Conversion between 96 kHz and 88.2 kHz is not possible.

24 Bit Encode and Decode

```
IN:PRO 44 44100 50/15uS
OUT:CON 44 24 BIT ENCODE
```

The 24 BIT ENCODE mode is used for encoding a 48 kHz or 44.1 kHz 24 bit stereo signal into a 96 kHz or 88.2 kHz 16 bit stereo signal. The 24 BIT DECODE mode is used to reconstruct the original 24 bit stereo signal. Both modes operate at a synchronous 2:1 or 1:2 data rate; the original bits are not modified in any way.

This mode allows high-sample rate 16 bit recorders, such as those made by Pioneer, to be used for wide-word recording at conventional sample rates.

2:1 Synchronous Downsampling

```
IN:PRO 44 44100 50/15uS
OUT:CON 44 2:1 DOWNSAMPL
```

A 2:1 synchronous downsampling mode has been provided. This mode has superior THD and dynamic range for down-sampling from the AD122-96. (-125 dB THD+N).

The mode is invoked by selecting **2:1 DOWNSAMPL** on the main screen for the output mode.

If the input signal is switched from high-sampling (88-96kHz) to low sampling (44.1-48kHz) the 3000S will, in this mode, pass the signal through 1:1 (similar to **BYPASS**). This allows quick selection of sample-rate for listening comparisons.

All dither modes and delay are active in this mode; DC Removal and Boost-Cut are not available.

Input Selection Screen

```

SELECT INPUT:
AES/EBU INPUT SELECTED

SELECT INPUT:
SPDIF INPUT SELECTED

```

This screen is used for selecting one of the two available inputs located at the rear panel: the XLR connector (for AES/EBU) or the RCA connector (for S/PDIF). Input Rate switching is automatic for low range of 40-50 kHz or high range of 88-96 kHz input sample rates.

Word Length / Copy Protection / Generation Screen^{In}

professional mode, this screen shows the indicated input word length, and allows the user to change the output coding word length indication, and for truncation (Truncation can be done with or without dither as will be shown later):

```

INPUT WORD LENGTH = 24
OUTPUT WORD LENGTH = 16

```

NOTE: reduced word length can be used for rounding. For example, 20 bits input can be rounded to 16 bits by setting the output to 16 bits.

```

IN: CATEGORY=CD C=1 L=0
OUT: CATEGORY=DAT C=1 L=0

```

In consumer mode, this screen contains information regarding category, C bit and L bit settings (pertaining to copy protection) and generation information.

Possible choices (X can be 0 or 1)

COPY PROHIBIT BIT C bit (C2):

Valid for all categories: 0 = Copy prohibit
1 = Copy permitted

GENERATION BIT L bit (C15):

CD category only: 0 = Original / commercially recorded data
1 = No indication / first generation or higher

DAT or PCM category: 0 = No indication / first generation or higher
1 = Original / commercially recorded data

GEN category only: 0 = No indication
1 = No indication

Mixed screens result from different modes for input and output:

```

IN: CATEGORY= CD C=1 L=0
OUTPUT WORD LENGTH = 16

```

Audio Control Screen

```
CUT/BOOST: ON          -20 DB
DC REMOVAL ON        +POLARITY
```

Cut/Boost: The signal level can be adjusted over a range of +60 dB to -60 dB in 1 dB steps. The function can be turned on and off without changing the amplitude change value.

DC offset removal: Removes any DC offset in the signal. DC removal should be activated prior to processing of audio, to allow for some settling time (less than 3 seconds) and to avoid a "click" sound. The algorithm continues to monitor and correct for slow varying offset variations.

Polarity: **+POLARITY** corresponds to analog input signal connection with "pin 2 hot". **-POLARITY** corresponds to analog input signal connection with "pin 3 hot"; **-POLARITY** inverts the phase of the signal by 180°.

Bar-Graph, Emphasis, and Display Screen

```
FINE:OFF 20 METER: -00 H
OUT: NO-EMPH DISPLAY:3
```

FINE:OFF 20 / BIT

This screen is used to select the BIT display mode and the FINE mode with its corresponding reference point from -10 dB to -20 dB. In FINE mode, the reference LED will be lit, and the channel level will be displayed in increments of 0.2 dB around that reference point. This is useful for setting precise levels. Indications are OFF, ON, and BIT. The BIT mode displays the active bits in the output stream, with the most significant bit displayed to the right (0 dB LED). This display shows active bits in the data.

METER: -00 (Meter Range)

Normal display showing peak and peak-hold indicates levels ranging from 0 to -55dB. While such a setting is useful in normal day to day operation, extending the display to indicate lower level activity allows monitoring low levels.

Meter Setting	Display range	Read Bar-Graph
METER: -00	0 dB to -55 dB	Read Bar-Graph panel indicator
METER: -30	-30 dB to -85 dB	Add -30 dB to panel indicator
METER: -60	-60 dB to -115 dB	Add -60 dB to panel indicator
METER: -90	-90 dB to -145 dB	Add -90 dB to panel indicator

H (or L)

LED bar-graph brightness may be adjusted.

DISPLAY: 3

Display contrast may be adjusted from 0 to 7.

Options Screen

```
CLIP= 1  OFF
CHAN STATUS LEFT
```

This screen allows the user to set or remove the following automatic features:

CHAN STATUS LEFT/RIGHT:

This selects whether the Channel Status bits are read from the left channel data (channel A default), or from the right channel data (channel B). Normal operation is to read from left channel data (channel A).

CLIP:

When set to ON, the MAXIMUM LED will turn on and stay on when output data hits full scale value for 1 to 6 successive samples (user-settable). The appropriate 0 dB bar-graph lamp (left channel, right) will start blinking. The MAXIMUM and corresponding lamps may be reset by pressing any of the push button switches. When CLIP is off, the MAXIMUM light will flicker when a single full-scale sample is encountered.

Acoustic Bit Correction Screen

```
DITHER: OFF
NOISE-SHAPING: OFF
```

This screen allows the user to set or remove Dither and Noise Shaping. Based upon customer feedback and further studies, V1.4 software offers High Pass or Flat Dither type and four noise-shaping curves. Our research taught us two interesting facts:

- a. Certain data compression schemes require the dither to have flat frequency response characteristics.
- b. Recording engineers' preference of dither type (high pass or flat) and noise shaping curves often depends on characteristics of the music involved.

Acoustic Bit Correction Mode Selection is as follows:

DITHER:

when set to DITHER ON, Model 3000S inserts truncation dither prior to the truncation operation. This feature is recommended when reducing word length.
Options: OFF, HPDF (High Pass Distribution Function) or FLAT

NOISE-SHAPING:

adds noise-shaping to optimize the noise-floor and the psychoacoustic effect of the dither. Four noise-shaping curves are available.
Options: NS1-NS4

88.2 kHz and 96 kHz output modes allow only HPDF dither and NS1 special noise-shaping curve optimised for wide bandwidth.

Available Combinations:

Dither	Noise Shaping	Comments
OFF	OFF	Setting dither to OFF also removes noise shaping
FLAT	OFF	Ideal for data compression applications
FLAT	NS1	Four Noise-shaping curves NS1-NS4
FLAT	NS2	
FLAT	NS3	
FLAT	NS4	
HPDF	OFF	High Pass Distribution Function
HPDF	NS1	
HPDF	NS2	
HPDF	NS3	
HPDF	NS4	

For a complete discussion of the uses of Acoustic Bit Correction, see Appendix I.

Store Preset Screen

STORE RAM PRESET:
(PRESS GO/EXIT)

The operator may save the present settings to the non-volatile RAM power up preset. To store settings in the preset, press GO. Press EXIT to abort the operation.

The word DONE appears after pressing GO, when the storage into non-volatile RAM is completed.

The factory default preset may be recalled by holding down the EXIT button while powering up the 3000S. The preset may then be modified and saved by following the procedure above.

Factory default Preset settings (these are stored as the power-up preset):

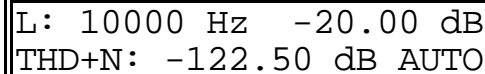
```

Test Tone Frequency:    1000 Hz
  Test Tone Level:      -20 dB
  Test Tone Channels:    L,R
  Boost/Cut Level:      0 dB
  LCD Contrast:         4
  Fine reference:        -20 dB
  Clip value:           1
  Emphasis:             No-Emphasis
Output Word Length:     24
  Dither:               OFF
  Noise-shaping:        OFF
  Operating Mode:       BYPASS
  Mode:                 PRO
  Input Select:         AES
  Polarity:             + (normal)
  DC Removal:          ON
  Peak-Hold:           OFF
Channel Status Read:    LEFT
    
```


The following are global non-programmable power-up conditions:

Boost/Cut:	OFF
Delay:	OFF
FINE Mode:	OFF
Meter Reference Level:	0 dB
LED Intensity:	Low
THD Channel:	Left
THD Mode:	AUTO

Signal Analysis Screen



```
L: 10000 Hz -20.00 dB
THD+N: -122.50 dB AUTO
```

The Signal Analysis screen enables measurement of a single sine-tone frequency, amplitude, and THD+N (total harmonic distortion plus noise) of a digital audio signal. The user may select **L** or **R** (Left or Right channel), and **AUTO** or **MANUAL** mode.

To change screen parameters press **ENTER** to introduce and move the cursor and **UP/DOWN CURSOR** buttons to select channel (L/R) or mode (AUTO/MANUAL).

Pressing the **GO** button activates the analysis routine. The Model 3000S starts the analysis by measuring and displaying the incoming tone frequency. The next measurement shows wide-band amplitude in dB full scale. The third measurement is THD+N. The unit will update THD+N results about every 1.5 seconds, depending on input sample rate..

AUTO mode enables the Model 3000S to lock to the incoming tone frequency. **MANUAL** mode sets the analysis frequency to the frequency programmed by the **TEST TONE SCREEN**. This allows the notch filter to be manually tuned by the user.

The Model 3000S's test tones may be used simultaneously with the THD+N measurement with input sample rates below 50 kHz. To do so, first go to the test tone screen and press **GO**. Then go to the THD+N screen and press **GO**. Tone will be output while the THD+N test is being done. This allows looping through external digital devices for testing. Tones are automatically output in full 24 bit word length. **NOTE:** connecting the Model 3000S's output to its own input can cause feedback. The Model 3000S should be set to an internal sample frequency (not **BYPASS**) on the main screen for loop-through tests.

Amplitude and THD+N are expressed in dB referenced to full scale. Display accuracy is 0.25 dB for amplitude from 0 dB to -120 dB and frequencies from 120 Hz to 20 kHz at input sample rates below 50 kHz.

High frequency analysis (input sample rate 88.2 or 96 kHz) is available only if 44.1 kHz or 48 kHz crystal operation is selected. Tone range is fro 500 Hz to 43000 Hz in this mode.

Test Tone Screen

```

FREQUENCY: 1000 Hz
LEVEL:    -20 dB  L+R

```

This screen allows the user to replace the audio with high quality test tones. A tone (sine wave) may be fed to the left (L,-) or right (-,R) channel or both (L+R). Tone frequency and amplitude are user programmable and changeable during operation.

In test tone mode, the Model-3000S checks the output frequency setting under the main screen. The test tone data rate will be locked to the input data frequency when output sample rate is set to BYPASS. The Model-3000S becomes a “stand alone” test tone generator (ignoring input data) when the output frequency is set to an internal crystal (44100, 48000, 88200 or 96000). In all cases, the test tone transmission includes all present data settings in display mode.

Setting Test Tones

Note: press and hold cursor button for faster increase or decrease in frequency or level.

Frequency settings: using the **CURSOR** buttons the tone frequency may be selected in 1 Hz increments between 1 Hz and 20000 Hz. 88.2 kHz sample rate mode is limited to 43000 Hz. 96 kHz sample rate mode is limited to 46000 Hz

Level settings: using the **CURSOR** buttons the tone amplitude may be selected in 1 dB increments between 0 dB (full scale) and -120 dB. An additional step below -120 dB, indicated by “OFF” on the display, outputs digital black (all zero bits). Lower level tones are offered for testing of higher resolution equipment. The digital black “OFF” setting may be useful in applications ranging from listening to system noise, detection of DC offsets and dither, etc.

Delay Screen

```

DELAY: ON          1.0 FRAMES
      1664 SAMPLES  48002 HZ

```

This screen allows the user to control a digital delay feature. Delay is programmable from a minimum value of 384 samples to 10 frames, in 0.1 frame increments. Frames are calculated at 30 fps. The output frequency as measured is indicated at the lower right of the screen. The output frequency is updated when the screen is entered.

The Delay function is not available at 88.2 kHz or 96 kHz outputs settings.

PART II: Tutorial

This tutorial is designed for the first time user of the Model 3000S. It requires that the default settings conform with the "factory default settings". To reset the unit to the factory default settings, press and hold the push button labeled EXIT while powering up the unit.

A SOURCE is defined as an output device generating the digital audio data stream, and a DESTINATION is defined as the device receiving the data.

For most applications, the Model 3000S input is connected to the output of a source. The Model 3000S acts as a "stand alone" source when used as a test tone generator with an internal crystal based oscillator setting the output sample rate.

The Model 3000S output always drives destination device inputs, at an output sample rate determined by the Main Screen Mode settings.

Before proceeding further, we recommend reading the introduction of the Operating Instructions in this manual. The following sections assume familiarity with operation of the push button switches.

Connections

Connect one or two source devices to the rear panel connectors marked AES/EBU INPUT XLR plug and/or S/PDIF INPUT RCA plug (these connections are not needed when the Model 3000S is operated as a "stand alone" test tone generator).

Connect the Model 3000S to its destination devices via the output plugs marked as AES/EBU OUTPUT, and S/PDIF OUTPUT. You may use either or both of the outputs simultaneously.

An external Sync source may be connected to the AES SYNC INPUT connector.

Starting Up

Apply AC power to the Model 3000S, and to your sources and destination devices. The Model 3000S displays its ROM version for 5 seconds (referring to its internal software version). The user may choose to bypass the 5 second wait by pressing any of the front panel buttons.

The first screen is the main screen. The top line of the LCD (liquid crystal display) shows data associated with the selected input connector. The choice of input selected is indicated by an LED located to the left of the LCD display (AES/EBU or S/PDIF LEDs). The SYNC LOCK LED indicates the lock condition of the sync input, if the selected mode requires sync. The RCV LOCK LED indicates that the MODEL 3000S has detected a digital audio signal at its input and locked to it.

Input Selection

Press the CURSOR UP button once to view the input selection screen. You may view the present input selection at the screen and the LED simultaneously. Press the ENTER button to introduce a screen cursor (on the word AES/EBU or S/PDIF). Press the CURSOR DOWN or CURSOR UP button several times to view the input selection process.

Monitoring Input Data

Press the EXIT button to remove the cursor. Press EXIT again to return to the main screen. (Pressing EXIT from a screen with no cursor quickly returns you to the main screen. You may choose to use the CURSOR UP/DOWN buttons to scroll through

the screens). You may now view incoming parameters associated with the incoming data from the selected input on the top display line indicated by "IN:". The first word (CNS or PRO) indicates Consumer or Professional data format, followed by a number 32, 44 or 48 indicating the frequency coding embedded in the data stream. The next five digit number displays the actual data rate measured by the internal counter of the Model 3000S. The last indication (on the right side of the top line) indicates the emphasis type coding.

Bar-Graph

View the BAR-GRAPH LEDs while your source is playing audio data. Observing the peak-hold action of the far right LED will help avoid clipping of the audio.

Press the CURSOR UP button to change screens. The screen starting with the word "FINE" is used for adjusting DISPLAY MODE, METER RANGE, LCD DISPLAY brightness and METER intensity.

Press ENTER to introduce a cursor for FINE:OFF/ON. Select mode using the CURSOR UP/DOWN buttons. With FINE mode ON, the number to the right (the next cursor position) sets the reference level in 1 dB steps from -10 to -20 dB. The LED corresponding to the reference level will light continuously on the bar-graph and the level will be displayed in 0.2 dB increments for accurate calibration.

Press ENTER a third time to move the cursor to METER position, then set desired meter range from -00 to -90. Pressing ENTER again, moves cursor to L or H for LED display brightness. Set the desired intensity using the CURSOR UP/DOWN buttons.

Press ENTER to move the cursor to DISPLAY position, then set for the desired contrast by using the CURSOR UP/DOWN buttons.

Press ENTER several times, observing how the cursor location moves on the screen. You may "park" the cursor at any location and edit the settings by using the CURSOR UP/DOWN buttons. Press EXIT to remove the cursor. This sequence is applicable to all of the screens.

Output Format And Sample Rate Conversion

Find the main screen either by scrolling to it or by pressing EXIT twice. Press the CURSOR UP (or CURSOR DOWN) button for several seconds to quickly scroll through the screens (This quick scrolling action will be particularly useful under test tone generator programming of frequency and amplitude). Press EXIT to return to the main screen.

Press ENTER to introduce a cursor. Set the output data format to CNS (consumer format).

Move the cursor to output frequency indicator and set it as desired.

Move the cursor to the next location. Scroll through the choices. BYPASS setting provides for the output data transfer rate to be synchronized with the input rate (direct transfer). The other choices change the sample rate of the output to an internal crystal operation or external sync operation, as well as the 24 BIT ENCODE and 24 BIT DECODE modes, for use with 16-bit 96 kHz digital recorders.

After setting all of the output parameters, press the GO button to activate. (You may press GO before or after removing the screen cursor with the EXIT button). If GO is not pressed within five seconds of changing a parameter, the parameters revert to their programmed settings.

Press EXIT to remove the cursor. Press the CURSOR DOWN once to observe further input and output settings associated with the CNS (consumer output format) selected in the communications screen. This screen is aimed at controlling COPY

PROTECTION AND GENERATION settings. Use the ENTER and CURSOR buttons to become familiar with these screen settings.

In PRO mode you can change the WORD LENGTH indication. Note that a change in word length indication is coupled with an actual change in the digital audio data word length. This can be used for truncation of a long incoming word (up to 24 bits) to a shorter word length. NI (word length not indicated) is interpreted as 20 bit output word length. Truncation may be done with or without the ACOUSTIC BIT CORRECTION as explained in Part I.

Options Screens

Control of the two OPTIONS screens is done in the same manner as previously stated (find the screen with the CURSOR buttons, ENTER to introduce and move a cursor, CURSOR UP/DOWN for setting selection and EXIT for removing a cursor). These screens allow the user to configure the Model 3000S operation as described below:

CLIP AUTO-MAX ON/OFF: When set to ON, the BAR-GRAPH display serves to warn the user when the audio has reached a peak (maximum or minimum code value). A peak condition will turn the MAX LED on and start the 0 dB LED blinking (for the corresponding channel). Pressing any button will turn these LEDs off until the next peak is detected.

DITHER ON/OFF: Use of Acoustic Bit Correction, the built-in noise shaper plus dither is recommended for truncating from a longer word to a shorter one. See Appendix I for more details of Acoustic Bit Correction.

READ LEFT/RIGHT CSBITS: When set to LEFT, the Model 3000S reads the digital audio information from the left channel (this is likely to cover most source material). The user may select the right channel.

Power-Up Preset

The Preset screen starts with "STORE RAM PRESET". You may store the present settings of the Model 3000S into the RAM preset (press GO) or exit the screen.

Test Tones

To generate a test tone, scroll to the test tone screen and set the frequency, amplitude and destination (left, right or both). To change the frequency or amplitude by large increments, hold down the CURSOR UP or CURSOR DOWN buttons. For small increments, press the CURSOR UP/DOWN repeatedly.

Press GO to start the tone. You may change settings while the tone is generated. To stop the tone press EXIT twice (returning to the communications screen).

Note: internal tone generation consists of 24 bits of resolution. The word length at the output is subject to truncation and dither settings. CNS word length is always 16 bits.

Using the Tone Generator and THD+N Features

Model 3000S is an all digital machine, capable of supplying and analyzing AES/EBU and SPDIF digital audio data. While aimed at digital in/digital out applications, the use of external reference A/D and D/A converters can expand usability to include analog domain measurements (accuracy will be limited to the performance of such devices).

Example A: evaluating a single digital in/digital out unit.

1. Connect model 3000S output to unit under test. Select output sample rate on main screen. Connect unit's output to Model 3000S input.
2. Set Model 3000S for appropriate output parameters: Highest tone accuracy is achieved when unit is set to output 24 bits (Professional format).
3. Select test tone screen, set for desired tone frequency and amplitude, and press the "GO" button.
4. Select THD+N screen and press the "GO" button.
5. Read results (tone frequency, amplitude and THD+N).

You may now adjust "unit under test" parameters while viewing THD+N. You may go back and forth between tone generation screen and THD+N screen for testing different amplitudes at the set frequency. Selecting a new frequency requires you to press GO to re-tune the notch.

Example B: optimizing multiple unit interconnections.

Clearly, the microphone preamplifier resides at the front end, and the format converter at the back end. Some units may be placed in various locations in the audio chain (which is better? digital equalizer followed by sample rate converter? or vice versa). The old "trial and error" method may work well for two units in series. The possibilities for connecting three units become cumbersome.

Evaluating each unit individually (see example A) and then connecting them in descending order of performance yields the best results (i.e., best unit first, driving second best, with lowest performer last in the chain).

PART III: Specifications

Sample Rate Conversion

Input Signal: Full scale (0 dB), 24 bits, no dither.
Equipment used: Audio Precision, System One - Dual Domain
 D-D THD+N test set to 6 samples, 500 msec delay, tuning = frequency

Output word length =24 bits

INPUT	OUTPUT	1 kHz tone	10 kHz tone	18 kHz tone
48 kHz	48 kHz :	-110 dB (.00032%)	-121 dB (.00009%)	-123 dB (.00007%)
48 kHz	44.1 kHz :	-105 dB (.00056%)	-112 dB (.00025%)	-106 dB (.00050%)
44.1 kHz	48 kHz :	-104 dB (.00063%)	-109 dB (.00035%)	-103 dB (.00071%)
44.1 kHz	44.1 kHz :	-111 dB (.00028%)	-121 dB (.00009%)	-121 dB (.00009%)

Output word length =20 bits

INPUT	OUTPUT	1 kHz tone	10 kHz tone	18 kHz tone
48 kHz	48 kHz :	-110 dB (.00032%)	-118 dB (.00013%)	-119 dB (.00011%)
48 kHz	44.1 kHz :	-105 dB (.00056%)	-112 dB (.00025%)	-106 dB (.00050%)
44.1 kHz	48 kHz :	-104 dB (.00063%)	-109 dB (.00035%)	-103 dB (.00071%)
44.1 kHz	44.1 kHz :	-111 dB (.00028%)	-118 dB (.00013%)	-118 dB (.00013%)

Output word length =18 bits

INPUT	OUTPUT	1 kHz tone	10 kHz tone	18 kHz tone
48 kHz	48 kHz :	-107 dB (.00045%)	-110 dB (.00032%)	-110 dB (.00032%)
48 kHz	44.1 kHz :	-104 dB (.00063%)	-108 dB (.00040%)	-105 dB (.00056%)
44.1 kHz	48 kHz :	-102 dB (.00079%)	-107 dB (.00045%)	-102 dB (.00079%)
44.1 kHz	44.1 kHz :	-106 dB (.00050%)	-110 dB (.00032%)	-119 dB (.00035%)

Output word length =16 bits

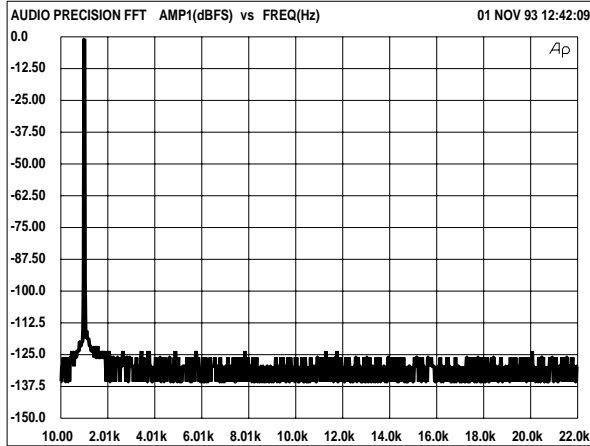
INPUT	OUTPUT	1 kHz tone	10 kHz tone	18 kHz tone
48 kHz	48 kHz :	-98.0 dB (.00126%)	-98.7 dB (.00100%)	-99.0 dB (.00112%)
48 kHz	44.1 kHz :	-97.5 dB (.00133%)	-98.6 dB (.00117%)	-98.1 dB (.00124%)
44.1 kHz	48 kHz :	-97.2 dB (.00138%)	-98.4 dB (.00120%)	-96.7 dB (.00146%)
44.1 kHz	44.1 kHz :	-98.1 dB (.00124%)	-98.8 dB (.00115%)	-97.8 dB (.00129%)

High Frequency Conversion Output word length =20 bits

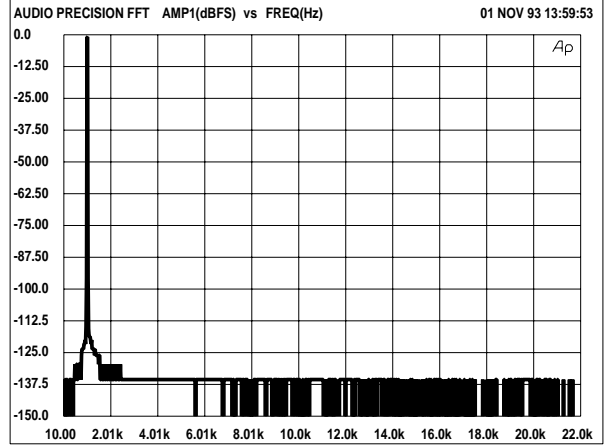
INPUT	OUTPUT	1 kHz tone	10 kHz tone	18 kHz tone
48 kHz	96 kHz :	-110 dB (.00032%)	-118 dB (.00013%)	-119 dB (.00011%)
96 kHz	48 kHz :	-111 dB (.00028%)	-112 dB (.00025%)	-112 dB (.00025%)
44.1 kHz	88.2 kHz :	-110 dB (.00032%)	-118 dB (.00013%)	-119 dB (.00011%)
88.2 kHz	44.1 kHz :	-111 dB (.00028%)	-112 dB (.00025%)	-112 dB (.00025%)

Note: 96 kHz to 44.1 kHz sample-rate-conversion can be accomplished in one step by the 3000S. However, it is strongly recommended that, for best quality, a 2:1 conversion be combined with a low-frequency conversion in two passes.

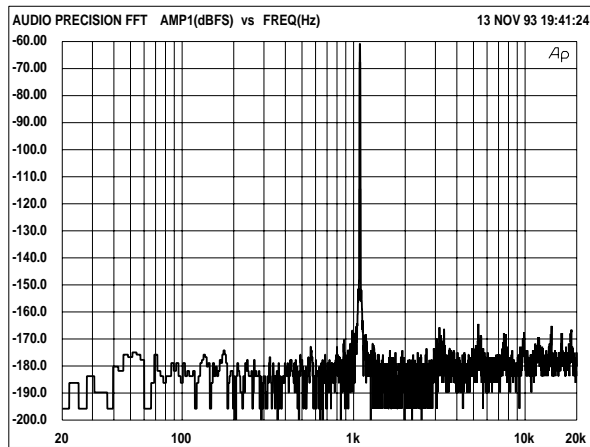
dB-3000S



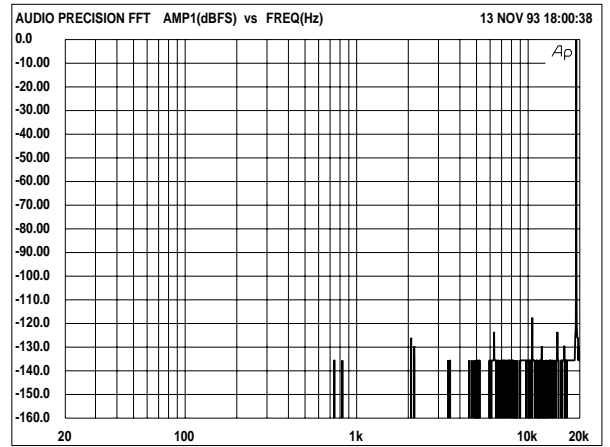
1 kHz tone 44.1k to 44.1k Conversion
0 dB, 16 bit word length



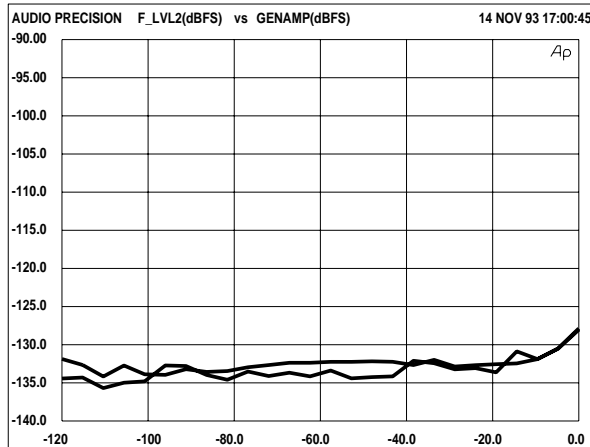
1 kHz tone, 44.1k to 44.1k Conversion
0 dB, 20 bit word length



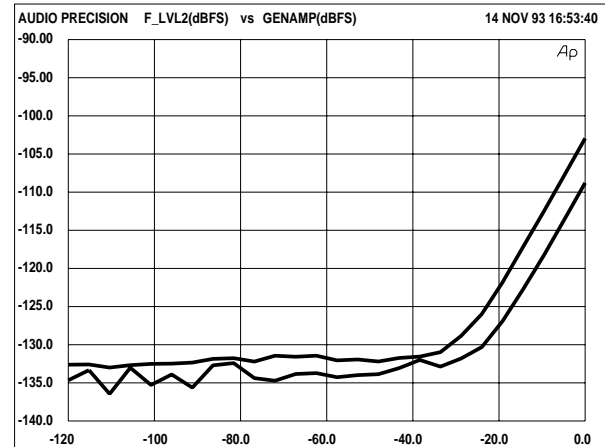
Dynamic Range
1 kHz tone, 48k to 44.1k Conversion
-60 dB, 24 bit word length



High Frequency Full Scale Performance
18 kHz tone, 48k to 44.1k Conversion
0 dB, 24 bit word length



THD+N vs.level- 44.1k to 44.1k Conversion
1 kHz and 19 kHz tones



THD+N vs.level- 48k to 44.1k Conversion
1 kHz and 19 kHz tones

Test Tone Generator

Output Signal: Full scale (0 dB), no dither.
Equipment used: Audio Precision, System One - Dual Domain
 D-D THD+N test set to 6 samples, 500 msec delay, tuning = frequency

Word length/Freq.	1 Hz tone	10 Hz tone	18 Hz tone
24 bits / 44.1 Hz	-122 dB (.00008%)	-124 dB (.00006%)	-130 dB (.00003%)
24 bits / 48 Hz	-122 dB (.00008%)	-124 dB (.00006%)	-130 dB (.00003%)
20 bits / 44.1 Hz	-119 dB (.00011%)	-120 dB (.00010%)	-122 dB (.00008%)
20 bits / 48 Hz	-119 dB (.00011%)	-120 dB (.00010%)	-121 dB (.00009%)
18 bits / 44.1 Hz	-110 dB (.00032%)	-110 dB (.00032%)	-110 dB (.00032%)
18 bits / 48 Hz	-110 dB (.00032%)	-110 dB (.00032%)	-109 dB (.00028%)
16 bits / 44.1 Hz	-98.1 dB (.00124%)	-98.9 dB (.00113%)	-98.5 dB (.00119%)
16 bits / 48 Hz	-98.3 dB (.00122%)	-98.9 dB (.00113%)	-99.0 dB (.00112%)

Inputs and Outputs

Signal	Connector	Impedance	Isolation
AES/EBU INPUT	XLR-3	110 Ohms	Transformer isolated
AES/EBU SYNC INPUT	XLR-3	110 Ohms	Transformer isolated
AES/EBU OUTPUT	XLR-3	110 Ohms	Transformer isolated
S/PDIF INPUT (IEC958 type II)	RCA	75 Ohms	Transformer isolated
S/PDIF OUTPUT (IEC958 type II)	RCA	75 Ohms	Transformer isolated

General specifications

Dimensions: 17" x 7 5/8" x 1 3/4" (standard 1U high 19 inch rack mount)
 Weight: 7 lb.
 AC Power: 90-264VAC, 47-63 Hz. Maximum power consumption 20

Watts.

Appendix I: Acoustic Bit Correction™

High quality digital audio sound often incorporates long words (many bits). Both high resolution A/D converters (18 bits or more) and signal processing equipment (digital mixers, equalizers, reverb and more) generate longer words than those supported by standards (such as the 16 bit CD format). A straightforward word length reduction severely deteriorates the musical fine detail of low level signals by introducing distortions and a noise floor that is modulated by the signal.

Proper dithering serves to eliminate the distortions and noise modulation effects. The desirability of such improvement is unquestionable but is coupled with increasing noise floor. Noise shaping is based on shifting the noise from high sensitivity frequency regions to less sensitive ones. That process is based on psychoacoustic research. A "stand alone" noise shaper does not improve distortions and noise modulation problems. Lavry Engineering's Acoustic Bit Correction incorporates both concepts to reduce distortion and noise modulation and shape the noise psychoacoustically.

Customer feedback and further studies taught us two interesting facts:

- Various data compression schemes require the dither to have flat frequency response characteristics.
- Recording engineers' preference of dither type (high pass or flat) and noise shaping curves often depends on characteristics of the music involved.

The Model 3000S offers High Pass or Flat Dither type and four noise-shaping curves.

Model 3000S allows the user to use dither with or without noise shaping. Operating noise shapers without dither is undesirable so the unit automatically shuts the noise shaper off when dither is off. Using dither (with or without noise shaping) requires user discretion. Ideal data transfer and processing should retain long word length. A single word length reduction (and therefore dithering and noise shaping) should take place last.

Figure 1 shows a -100 dB 1 kHz 24 bit tone truncated to 16 bits. The distortion is severe. Figure 2 shows the same tone with high-pass dither only. The distortion is gone, and the tone's level appears at a true -100 dB. Figure three shows the same tone with Acoustic Bit Correction applied. Notice what the addition of noise-shaping to the dither does to the noise floor in the ear's most sensitive mid-range area (approx. 2kHz to 8 kHz).

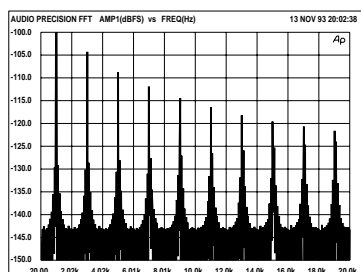


Fig. 1- truncation only

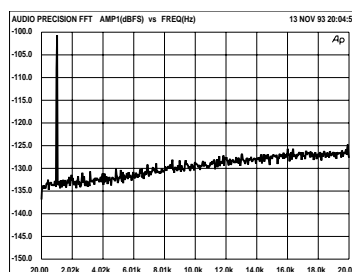


Fig. 2 - high-pass dither

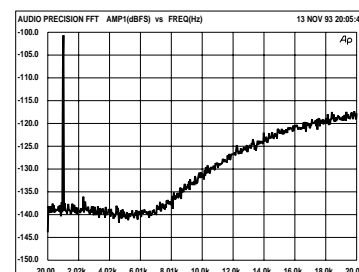


Fig. 3 - Acoustic Bit Correction
High Pass- NS2

A number of mathematical and psychoacoustic researchers introduced two concepts for sonic improvement: dithering and noise shaping. Credit is due to L. Fielder of Dolby Labs for providing the basis for various noise shaping curves. The reasons for triangle PDF dither have been explained by S. P. Lipshitz and J. Vanderkooy of the University of Waterloo. Additional publications by M. Gerzon, P. G. Craven, J. R.

Stuart and J. R. Wilson (all from the United Kingdom) and J. Goodwin (from the U.S.A.) shed light on this complex subject.

Dithered noise shaping technology has been incorporated into a handful of hardware devices. While all are based on the same concepts, some perform better than others. After simulating and listening to all available public domain algorithms, Lavry Engineering came to some conclusions in forming a basis for Acoustic Bit Correction. The principal conclusions are:

- a. The practice of greatly amplifying low level signals to determine triangular flat PDF (probability density function) dither reveals the effectiveness of distortion and noise modulation elimination. This practice yields misleading results when testing unflattened dithers and/or noise shapers. It conflicts directly with L. Fielder's findings showing completely different threshold detectability curves for quiet and loud levels. Noise shaping listening tests must be done at "reasonable" volume levels.
- b. Given the above requirement, our listening tests concluded a strong preference for "triangle high pass" dither (this dither is produced by simultaneously adding a new random number and subtracting the previous value). Such dither is frequency-shaped to carry more high frequency energy (the energy content at low frequencies is minimal).
- c. Listening tests revealed a preference for smoothly varying noise shaping curves. Peaks and notches seem to irritate the listener (admittedly while turning the volume up). In addition, despite the temptation to optimize the noise shaping curve to the average listener's hearing threshold, given a significant variation from listener to listener requires reasonable compromises in tailoring such a curve. In other words, smooth the curve.

The improvements offered by dither and noise shaping vary with source material and final word length. An A/B/X test at 16 bit level, requires a quiet environment and low level (loudness) audio. The listener must resist the temptation to turn the volume up to unreasonable levels. The practice of truncating to short word length (8-12 bits) should be avoided, although the Model 3000S performs this admirably. The ideal noise shaping curve may be irritating at loud levels. (Model 3000S's 8 bit noise shaping curves are based on a different curve).

Lavry Engineering's listening tests were based on test tones and repeating loops of quiet passages of various material (mostly classical music) with flat amplifier response. Listening to test tones was straightforward: we used the Model 3000S test tone generator mode switching the Acoustic Bit Correction on and off. The frequency and amplitude programmability was very useful.

Listening to music required 18 to 20 bit material. Distortion present at the 16 bit input word can not be removed by dither. The algorithm was functional (to a lesser degree) from a noise shaping standpoint, but distortion removal did not take place (the data distortion content due to the previous truncation was interpreted as signal). Acoustic Bit Correction is aimed at correcting truncation problems associated with the shortening of word length.

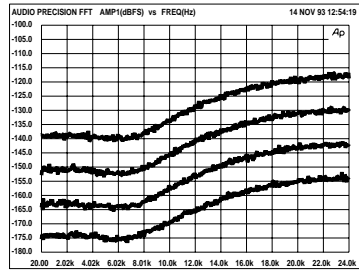


Fig. 4 - Acoustic Bit Correction
16 (top), 18, 20, and 22 bits
no input signal

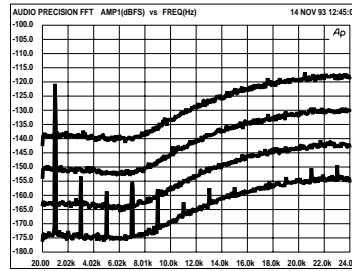


Fig. 5 - ABC HP-NS2
16 (top), 18, 20, and 22 bits
-120 dB 1 kHz sine tone

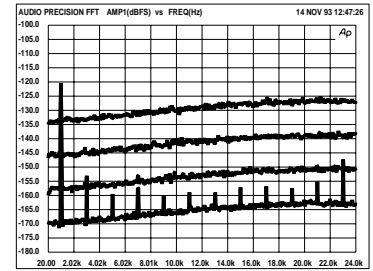


Fig. 6 - dither only
16 (top), 18, 20, and 22 bits
-120 dB 1 kHz sine tone

Acoustic Bit Correction may be used with words of lengths wider than 16 bits. Figure 4 shows the noise floor of Acoustic Bit Correction (High Pass, NS2) without a signal at 16, 18, 20, and 22 bit wordlengths. Note the curved noise-floor with lowest level in the ear's most sensitive mid-range region.

Figure 5 shows Acoustic Bit Correction (High Pass, NS2) applied to a 24 bit input, reducing to 16, 18, 20, and 22 bit output widths. Notice that there is no noise modulation present.

Figure 6 shows the effect of high-pass triangular dither only. The more coherent data in the dithered bits, the better the dithering process works. Note that dithering from 24 bits to 22 bits has a lower noise floor but more visible distortion components than dithering to a shorter word length, due to fewer bits in the portion of the word being truncated.

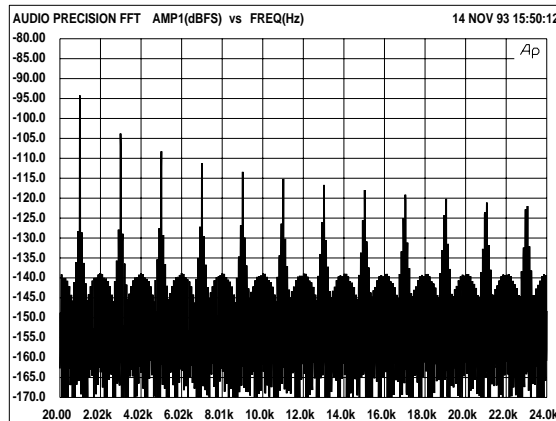


Fig. 7- 16 bit input signal -100 dB
truncation by previous processing

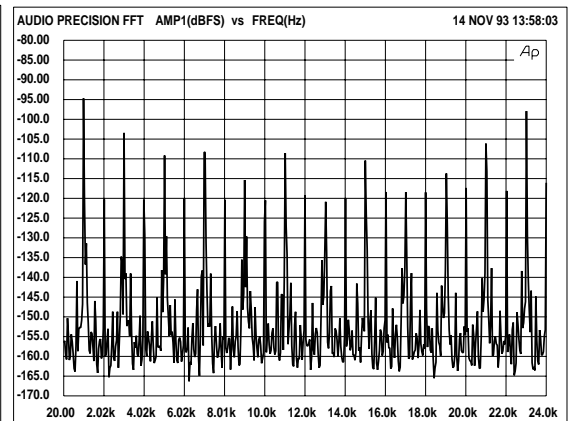


Fig. 8- 16 bit input signal -100 dB
additional processing and truncation

Should dither be applied to input signals of 16 bit word length? Figure 7 shows a -100 dB tone of 16 bit word width with no dither or noise shaping. Note the presence of all of the odd harmonics, created when the truncation process turned the tone into a 1 lsb square wave.

Figure 8 shows the result of applying additional processing to this signal, and truncating the result, thereby creating additional distortion components.

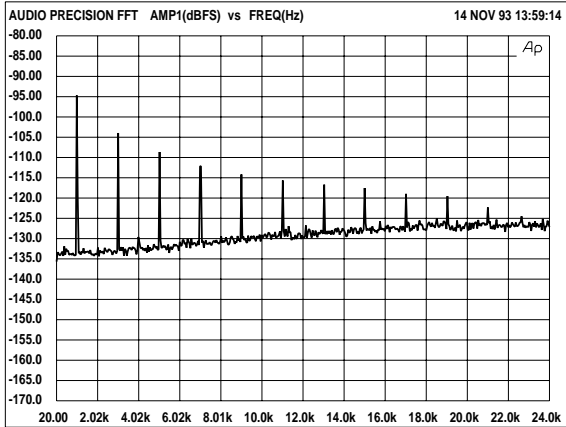


Fig. 9- 16 bit input signal -100 dB
high pass dither applied

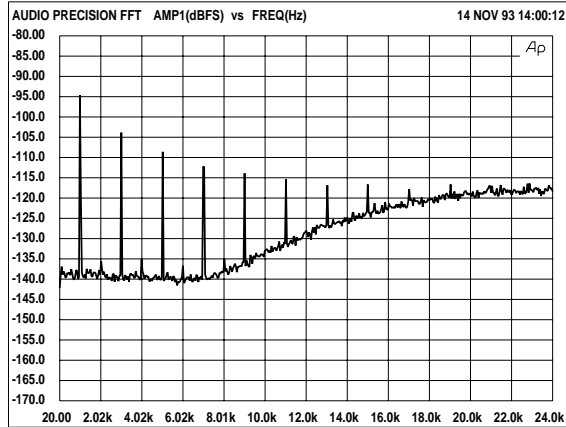


Fig. 10- 16 bit input signal -100 dB
Acoustic Bit Correction applied

Figure 9 shows the original 16 bit input signal processed with high-pass dither only. Notice that the additional distortion is gone.

Figure 10 shows Acoustic Bit Correction (dither and noise-shaping) applied to the same signal with an increase in the effective noise floor. Acoustic Bit Correction cannot remove truncation distortion in an incoming 16 bit signal, but it can avoid additional truncation distortion if additional processing, such as sample rate conversion, has been applied.

Noise-Shaping Curves

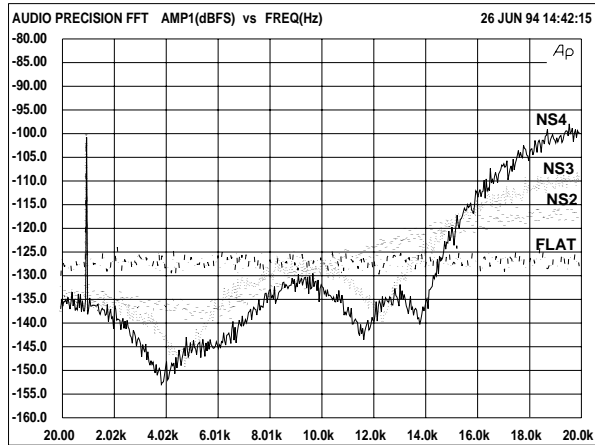


Fig. 11- 1kHz input signal -100 dB
Flat dither, with multiple noise-shapers shown

Figure 11 compares Flat dither with NS2, NS3, and NS4 noise-shaping curves. Note the increasing aggressiveness of the curves, with the pronounced dip at 4kHz and in the 12kHz region. These correspond to the most sensitive areas of the ear to noise.

Quantization noise is shifted from these areas to high frequencies, where the ear is less sensitive. Noise-shaping curves should be selected based upon listening tests of specific material to be dithered.

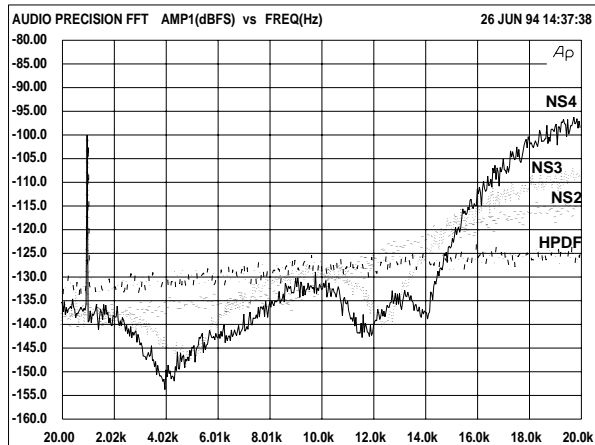


Fig. 12- 1kHz input signal -100 dB
HPDF dither, with multiple noise-shapers shown

Figure 12 compares High Pass dither with NS2, NS3, and NS4 noise-shaping curves. High Pass dither gives more noise shaping in the high-frequency band, while slightly lowering low frequency quantization.

Note: in Figures 11 and 12, the NS1 curve has been omitted for clarity. It lies below NS2 in both cases.

Appendix II: Testing Of Asynchronous Sample Rate Converters

The theoretically ideal sample rate converter is a device that converts the data transfer rate without changing the content of the material.

Listening tests:

Many devices are judged by their particular sonic quality. A recording engineer may prefer, for example, a small amount of distortion to add some "color" to the sound. The same engineer, testing a number of processing devices separately, may choose his building blocks to achieve a certain characteristic sound. Let us assume that the desired characteristic is based on some small amount of second order distortion which imparts a characteristically "warm" sound. A problem may arise when processing the sound through more than one unit. The "desired distortion" may be compounded beyond the desired level (in our example, the second order distortion may increase each time the sound is processed to an unacceptable end result).

Sample rate converters may serve to reduce excessive clock jitter. This improvement can take place for any sampling-rate ratio (including 1:1). Jittery incoming data introduces signal dependent noise and distortions (increasing with signal amplitude and frequency). Such jitter reduction yields noticeable sonic improvement, thus complicating the objectivity of listening tests. Some manufacturers' comparison tests inappropriately compare a high jitter input to a low jitter output. Further confusion is due to the fact that digital domain FFT tests do not adequately show the effects of input jitter.

Listening tests should be based on comparing the audio of a directly applied signal against the converted version of the same material. The greater the difference, the less ideal the converter.

The sample rate converter should receive a low jitter data source, and drive a reference grade D/A converter, a high quality power amplifier and top grade speakers, all matched to 0.1 dB.

"Blind" listening comparisons (A/B/X tests) by recording professionals yield the best unbiased results.

Measuring performance:

The most commonly used measurements are based on a standard FFT, THD plus Noise testing (in the digital domain) and phase linearity. Interpreting measurements performed on asynchronous sample rate converters is less straightforward. The asynchronous sample rate converter can not control the input and output rates (these rates are forced by the driving source and required destination devices). The converter is required to reconstruct the data content while accommodating receiver and transmitter clock rates.

The digital nature of the process introduces quantization effect into the conversion. The ratio does not change smoothly. It tracks the clock rate variations in a quantized fashion (small incremental jumps). Proper design requires that the quantized ratio changes fall below the ear sensitivity levels. Tracking the clock rates sets restrictions on the maximum tolerable ratio step size and the manner in which the ratio tracks the clocks rates. Let us focus on the two extreme cases for ratio tracking:

- a. Steady clocks: The converter is adjusting the ratio up and down by a small amount around the correct average ratio.
- b. Fast "varispeed": The adjustment accuracy is reduced with fast varispeed affecting the accuracy of ratio adjustment.

Technological limitations require careful consideration for optimizing both ratio step size and the tracking mechanism. Psychoacoustic considerations (listening tests) and practical limitations of varispeed should form the basis for proper performance criteria. Measuring asynchronous sample rate converters may reveal some of these compromises.

While the input and output of theoretical converters measure identically, real converters continuously track and adjust internal coefficients. Such ratio modulation appears on FFT measurements as a "widening of the main lobe" of a sinusoidal test tone. The amount of widening depends greatly on variables such as ratio step size, ratio tracking, FFT size and type of FFT window used.

Common digital domain measurements (FFT based measurements) do not show the effects of low levels of incoming jitter. A common indirect approach is based on measuring the THD plus noise reduction at the output of a reference DAC (driven by full scale high frequency tone). While such a measurement does not quantify jitter, it yields (in principle) the desired end result. Real world limitations of reference DAC performance set limits to such measurements (DAC performance is typically lower at high amplitudes and frequencies). Model 3000S utilizes a high Q (steep resonance) LC circuit for de-jittering incoming data. Further jitter reduction may be achieved with a 1:1 sample rate conversion, using a low jitter crystal clock oscillator for the output data clock.

Sample rate converter performance should be measured over the usable audio range. Poor performance at high frequencies cannot be dismissed as inaudible noise. A typical justification for high frequency performance degradation is based on the fact that real music contains less energy at this frequency range. This could allow, at most, a few dB reduction in THD+N. Unlike many unsampled analog circuits which tend to generate higher frequency distortions, sampling folds back distortion energy to lower frequencies, including the ear's most sensitive mid-range region.

When processing long words a desirable performance specification should exceed the limitations of a 16 bit word by a significant margin. This will ensure that overall performance is limited almost completely by word length bottleneck. (Truncation of a long word to 16 bits yields theoretical results of approximately 98 dB THD+N). The combination of long word format and good performance specifications is even more desirable when additional processing may take place. Premature truncation amounts to loss of detail.

Appendix III: Signal Analysis of THD+N

Introduction

Lavry Engineering's latest revision to the Model 3000S Sample-Rate Data Format Converter consists of THD+N (total harmonic distortion and noise) measurement capabilities. The new function expands the usefulness of this "digital audio all in one tool". The new feature offers the high performance and ease of use found in all the other features (data format conversion, sample rate conversion, Acoustic Bit Correction (TM), reference meter bridge and test tone generation). 24 bit tone generation and THD+N measurement accuracy in excess of 122 dB provide sufficient margin for the next generation of digital audio equipment.

The Model 3000S's test tone generation and Signal analysis are independent of each other. This flexibility allows performing either function separately or both simultaneously.

Distortion and noise

Real hardware generated tones contain both distortion and noise. Harmonic distortion is referred to as energy residing at multiples of the tone frequency. Noise is all other undesirable energy.

Some common sources of distortion:

- **Amplifiers:** non linear signal processing causes distortion. While mostly corrected for by use of negative feedback, amplifiers tend to degrade at higher frequencies.
- **Component imperfection:** distortions occur when component values depend on the signal. For example, capacitors tend to counteract a changing signal (dielectric absorption).
- **Digital truncation:** limitations of word length (number of bits) is a non linear process. Such distortions increase at lower signal levels.

By definition, noise is not all random (tones occurring at non harmonic frequencies are considered noise). Well known and understood is the critical noise requirement associated with signal amplification. Often overlooked is the accumulated noise due to connecting of many units in series.

Common sources of random noise:

- **Resistor noise** (flat frequency distribution) increases for larger value of resistance.
- **Semiconductor noise**, mostly flat frequency distribution. Increased noise levels at very low frequencies usually occurs below audible frequencies.
- **Capacitor noise**, inconsequential for higher values, has become a performance limiting factor with the introduction of very small capacitor values incorporated in modern semiconductors such as oversampling sigma delta converters.
- **Digital truncation**: limitations of word length (number of bits) in A/D converters, signal processors and more. The problem grows with increased amount of processing.

Common sources of non-random noise:

- **Coupling to analog signal path**: AC power line, RFI/EMI, coupling of digital signals to analog path, inadequate power supply rails and more.
- **Digital truncation**: limitations of word length (number of bits) in A/D converters, signal processors and more. The problem increases for low level signals.
- **Limit cycles**: cyclical patterns behavior in feedback based digital signal processing (such as sigma delta converters and IIR filter structures).

Listening tests and measurement of individual equipment in the audio chain does not guarantee optimum THD+N performance. Setup optimization is a very complex subject. Top recording engineers blend artistic considerations and engineering know-how into the process. The following discussion does not deal with artistic aspects. We acknowledge the great importance of artistry in music production, but are bound to limit our discussion to measurable and objective phenomena.

Optimizing THD+N (some engineering considerations):

Analog amplification: good signal to noise ratio requires "early" signal amplification, but with careful attention to tradeoffs between distortions and noise.

Analog attenuation: undesirable from noise standpoint, may be required to accommodate signal range limitations of various gear.

Jitter: clock jitter in A/D, D/A and Sample Rate Converters degrades THD+N. Reference D/A clock jitter in a studio may have little to do with the end product quality, but may make the monitoring process difficult. Sample rate converters perform best with low jitter on both incoming and outgoing clocks.

Configuring proper digital chain: whenever unit A may drive unit B or visa versa. A good "rule of thumb" is to have the better performer drive the lower performer. A quality digital device utilizing 24 bit words is limited to 16 bits when driven by a 16 bit device. The compounded outcome is that of "2 x 16 bit devices". Reversing the order allows the first process to retain its high accuracy, leaving a compounded outcome of one 16 bit device.

Measuring THD+N

The common method for measuring THD+N is based on feeding a "device under test" with a quality reference test tone and measuring the undesirable energy (THD+N) at its output. Lavry Engineering's Model 3000S provides the user with a reference test tone. The processed tone (or any other source) may be fed back to Model 3000S input. The input signal is filtered by a very sharp notch to separate the

desired signal component from the undesirable energy (THD+N). The undesirable energy (THD+N) is then displayed in dB (referenced to full scale).

The notch filter must be very deep and narrow. Notch depth assures that no energy at the fundamental frequency "leaks" to the meter. Narrow notch is required to leave noise and harmonics intact. How steep should the notch be? For theoretical tones one may strive for the steepest notch possible. Real applications require full attenuation over a slightly wider frequency range to accommodate possible small jitter of both sampling clock and the tone itself. Model 3000S provides sufficient attenuation over about .1% (allowing about 20 nsec jitter).

Additional 20 Hz high pass filtering ensures that very low frequency components (such as DC and other low frequency inaudible energy) does not alter the outcome.

Model 3000S begins its measurement by locking to the test tone frequency. There are 2 modes (user selectable) for achieving lock:

- **Auto mode:** Proper locking in the presence of noise and distortion requires "reasonable" signal to noise and distortion ratio. When using Auto mode we recommend starting the test with a large enough signal (full scale signals are ideal, but locking will take place at 40 dB over the noise and distortions floor). Pressing the "Go" button sets the notch in place, allowing THD+N measurements at any level.
- **Normal mode:** when using Model 3000S tone generator, the notch frequency follows the tone frequency settings. Normal mode frees the lock mechanism from any signal to noise restrictions. This mode requires the tone sample rate and incoming signal sample rate to match within +/- .1% of each other (testing Sample rate converters requires Auto mode).

Tone Generation

The Model 3000S retains all digital audio format and sample rate settings when generating a tone. The output tone may be sent to any AES/EBU or SPDIF device. Use of a quality reference D/A converter allows for listening for various effects (distortions, channel separation, room and speaker cabinet vibrations and more). The output of device(s) or setup under test may be applied back to Model 3000S inputs for the following reasons:

- Accurate level settings via the internal reference meter bridge
- Measurement of THD+N.
- Measurement of tone frequency and amplitude.

Regardless of the functions performed (sample rate conversion, Acoustic Bit Correction, metering, or any of the format conversion tasks) the unit provides for quick and easy insertion or substitution of digital audio signals with test tones. The unit resumes "normal operation" when exiting tone generation mode.

The test tone screen enables setting tone characteristics:

- Frequency: 1 Hz to 20 kHz in 1 Hz steps
- Amplitude: 0 dB (full scale) to -99 dB in 1 dB steps
- Destination: Right channel, Left channel or both

Test tone "quality":

Tones purity at maximum word length (22-24 bits) exceeds full scale performance of 122 dB for all frequencies.

Consumer data format dictates 16 bits. Professional mode allows generation of 16 to 24 bits. Note that while you may set the word length as desired, shorter words mean increased distortions and noise (119 dB for 20 bits, 110 dB for 18 bits and 98 dB for 16 bits). Distortion performance improves for lower amplitudes.

Transfer rate settings (main screen) are retained:

When main screen is set to "BYPASS", output data (audio data or tone) is transmitted synchronously to input data.

Selection of one of the internal crystal oscillators (44100, 48000, 88200, or 96000) provides stand-alone tone generation.