LAVRY

MODEL AD122-96 MX

MASTER ANALOG TO DIGITAL CONVERTER



OPERATIONS MANUAL

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MX PREFACE

The MX is our 5th generation gold series analog to digital converter.

The MX provides the same 127dB dynamic range (without A weighing) as the MkIII model, and consistently clean performance over the entire audio frequency range. High quality clocking circuitry provides extremely low jitter with both internal clocking and external lock. Precise matching of the two channels throughout the dynamic range preserves subtle details in the stereo image without degrading spatial location or detail of the lowest level signals.

The MX is a redesign of the MKIII, with the additional feature of increased ability to handle clipping of the audio signal without apparent loss of signal quality. In the quest to achieve the elevated levels common in pop music; Mastering engineers as well as Mix engineers can utilize clipping as a means of dynamic range control without the down-side of attack and release times associated with compressors and limiters. Although the MkIII model already had a reputation for "clean clipping," the MX model extends the useful range of clipping to even higher levels.

The MX model retains the Lavry exclusive Soft Saturation modes which increase the level below the saturation threshold by a selectable 3 or 6db. Audio signals above the threshold are processed by an emulation of analog tape saturation without artifacts associated with attack and release times. In combination with the improved handling of clipping level signals, the AD122-96MX offers more options for hotter recording levels than other converters.

Because the difference between the MkIII and MX models did not require changes to the converter's software, the MX model will display "Ver 2.1e" ©2000 dB tech, AD122-96MkII" in the initialization screen of the LCD display.

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Subject to the conditions set forth below, for one year after the original purchase date of the product, Lavry Engineering will repair the product free of charge in the United States in the event of a defect in materials or workmanship.

Lavry Engineering may exchange new or rebuilt parts for defective parts. Please call the factory for an RMA number prior to shipment. No product will be accepted for warranty service without a pre-issued RMA number.

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PART I: Introduction

The **Model AD122-96 MX Master Analog to Digital Converter** receives analog input signals at the left and right analog input channels. The converter can receive balanced or unbalanced analog input signals. Individual left and right analog gain adjustment can be made via the front panel adjustment pots. Fine tuning of reference levels can be achieved by setting the front panel bar graph display (LED's) to "Fine" mode (reference meter bridge). The digital output is available at the AES/EBU output. The AES/EBU XLR output is compatible with coaxial SPDIF using a simple adapter or adapter cable.

Front panel selection enables setting word length between 16 and 24 bits (NI is a 20 bit setting as per AES/EBU specifications). Redithering is available for output wordlengths of less than 24 bit using the available αcoustic βit Correction™ feature. Redithering to smaller words retains much of the acoustic information in the full word width output.

The converter can operate with internal or external clock synchronization. Available internal crystal frequencies are 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz. For external synchronization, a signal is applied to either the SYNC/WC XLR input or the Word Clock BNC input. The input types are digital audio (AES/EBU) or word clock signals. The desired frequency lock range settings are 44.1 kHz narrow lock range, 48 kHz narrow lock range, 40-50 kHz wide lock range, 88.2 kHz narrow lock range, 96 kHz narrow lock range, 88/44 kHz, or 96/48 kHz (conversion rate 2X sync rate).

Vernier Transfer Function Correction™:
The exceptional performance of Lavry Engineering's **AD122-96 MX** is due to design innovations in both analog and digital sections. The performance of the low-noise high-linearity converter is optimized by a set of proprietary algorithms utilizing a "digital vernier" technique which provides superior resolution and correction for changes in circuit parameters, including component drift due to temperature and aging.

Please note that because the difference between the MkIII and MX models did not require changes to the converter's software, the MX model will display "Ver 2.1e" ©2000 dB tech, AD122-96MkII" in the initialization screen of the LCD display.

Additional features:
Improved handling of input clipping: The MX model increases the maximum level that can be utilized with clipping as a form of dynamic range control, without audible degradation.

Test Tones: a fully programmable digital test tone generator for calibration.

Polarity Inversion: changes polarity of digitized audio signal.

Soft Saturation: emulates analog tape saturation for higher recording levels.

User-Defined Power-Up Default Setting

Stereophonic Bar Graph Display: features wide dynamic range; the bar graph can be extended by 30, 60 or 90dB for viewing the noise floor. The bar graph can also act as a reference meter bridge for precise input level calibration.

Clip Indication: when meter Peak Hold is enabled, the full scale "MAX" light can be set to hold until reset by the user.

Operating Instructions

Operation of the **Model AD122-96 MX Master Analog to Digital Converter** requires the use of five push button switches located on the front panel. The liquid crystal display is organized for a quick and intuitive operator interface. The unit features six screen displays.

Push Button Switches:

ENTER Introduces and moves the cursor through screen positions that can be edited by the operator.

CURSOR Push buttons allow the user to choose a screen and to select the cursor position. The **UP/DOWN** cursor buttons are also utilized when changing digits of a selected entry.

GO Initiates changes in Clock Source/Frequency selection, Test Tone, and Preset store.

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.

To 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/DOWN** buttons serve to move from screen to screen.

To Edit Screens:

Press **ENTER** to introduce the screen cursor at the first editable location. Each additional pressing of **ENTER** will move the 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** buttons will automatically advance the settings. This feature is particularly useful for increasing or decreasing numbers (such as test tone frequency and amplitude or word length).

Most operations react to a change of settings immediately without having to press **GO**. There are three settings requiring the use of the **GO** button:

- Starting the test tone generator.
- Storing the user power-up default settings.
- Changing Sample Frequency Source.

Main Screen

The main screen is used for selecting the A to D clock source and output word length parameters.

The top line controls the A to D clock source:



When setting for internal clock operation, choose sample rate and internal crystal mode:

44.1kHz INTERNAL CRYSTAL
48kHz INTERNAL CRYSTAL
88.2kHz INTERNAL CRYSTAL
96kHz INTERNAL CRYSTAL

When setting for external synchronization, the selections include:

- Sync. input connector: SYNC/WC INPUT XLR connector or the WORD CLOCK INPUT BNC connector.
- Digital audio or Word Clock: the XLR connector can accommodate both AES/EBU digital audio data and Word Clock. The BNC connector is for Word Clock only, as part of the IO-96 interface.
- Lock range: narrow lock range utilizing VCXO (internal voltage controlled crystal oscillators or wide lock range (internal low jitter VCO).

Possible choices for the SYNC/WC INPUT XLR connector are:

XLR AES LOCK 44.1	Phase synchronous
XLR AES LOCK 48.0	Phase synchronous
XLR AES WIDE LOCK	Phase synchronous
XLR AES LOCK 88.2	Phase synchronous
XLR AES LOCK 96.0	Phase synchronous
XLR AES LOCK 88/44	Phase synchronous to 44.1
XLR AES LOCK 96/48	Phase synchronous to 48

Possible choices for the WORD CLOCK INPUT BNC connector are:

BNC LOCK 44.1	Phase synchronous
BNC LOCK 48.0	Phase synchronous
BNC WIDE LOCK	Phase synchronous
BNC LOCK 88.2	Phase synchronous
BNC LOCK 96.0	Phase synchronous
BNC LOCK 88/44	Phase synchronous to 44.1
BNC LOCK 96/48	Phase synchronous to 48

The 88/44 and 96/48 settings are provide to synchronize high sample rate AD conversion (88.2 or 96 kHz) to half-speed external sync sources (44.1 or 48 kHz).

Word Length and αcoustic βit Correction™

The bottom line controls word length and αcoustic βit Correction™ parameters:

```
44.1kHz   INTERNAL CRYSTAL
20 BIT WL  ABC: HPDF      NS2
```

Setting Word Length:
The selection of word length (16 to 24 bits) sets the digital output word length. The **NI** setting (word length not indicated) is a 20 bits default setting as per AES/EBU specifications. The **Model AD122-96 MX** utilizes 22 bits for data (bits 23 and 24 contain little sonic information). We recommend a default word length setting of 22-24 bits. The front panel "20+" lamp indicates when the output contains 20 or more bits.

αcoustic βit Correction™ is designed for applications requiring the use of fewer bits. Simple truncation of lower order bits introduces sonic side effects (distortions and modulation of the noise floor by the signal). The αcoustic βit Correction™ feature provides a selection of dither and noise shaping algorithms. Dither overcomes distortions and noise modulation, but it also raises the noise floor. The noise shaping algorithms provide an effective means of retaining higher bit content in a lower bit format (it shifts the noise energy away from hearing sensitive frequency regions to less sensitive ones). See Appendix I for a complete discussion of αcoustic βit Correction™.

Dither choices are:

OFF	no dither
HPDF	high pass probability density function
FLAT	recommended for data compression applications

Noise shaping choices are:

NSOFF	no noise shaping
NS1, **NS2**, **NS3** or **NS4**	user selectable noise shaping curves

Test Tone Screen

To generate a test tone, scroll to the test tone screen and set the frequency, amplitude and destination (left, right or both). The frequency range is from 1Hz to 20kHz in 1 Hz steps. Amplitude range is 0dB (full scale) to -99dB in 1 dB steps, with an **OFF** setting for "digital black." To change the frequency or amplitude by large increments, hold down the **UP** or **DOWN** cursor buttons. For small increments, press the **UP** or **DOWN** cursor buttons repeatedly.

Press **GO** to start the tone (amplitude, frequency and destination settings may be changed during tone generation). To stop the tone, press **EXIT** twice (returning to the main screen).

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

Audio Control Screen

```
SOFT SATURATION  OFF       0
DC REMOVAL: ON    +POLARITY
```

The user may activate the soft saturation feature by altering the setting after **SOFT SATURATION** on the top line from "**OFF**" to "**+3dB**" or "**+6dB**"**.** Soft Saturation settings are not stored in RAM preset.

When activated, the audio signal is amplified by the dB amount displayed (3dB or 6dB boost), and signal peaks are compressed gently to avoid digital clipping. The feature emulates magnetic tape saturation. It is still possible to clip the signal if enough signal level is applied to the analog inputs.

DC offset removal:
The **Model AD122-96 MX** converts signals down to DC (0Hz). The total DC offset (due to DC offset within the audio signal and the offset of the converter itself) can be removed.

DC REMOVAL should be activated prior to processing of audio, to allow for some settling time (less then 3 seconds) and to avoid a "click" sound. The algorithm continues to monitor and correct for slow varying offset variations.

PLEASE NOTE: Removing the offset is necessary for normal operation when recording audio signals, so the lower line should always display **DC REMOVAL ON** during audio recording.

Polarity:
+POLARITY corresponds to analog input signal connection with "pin 2 non-inverting".
- POLARITY corresponds to analog input signal connection with "pin 3 non-inverting".

Note that polarity reversal works for both balanced and unbalanced analog input signals. The analog inputs (XLR connectors) are physically wired for pin 2 non-inverting polarity. In the AD122-96MXpolarity reversal is done digitally (after the conversion).

Data limiting settings:
The symbol **0** in the right corner of the top line, "**SOFT SATURATION OFF 0**" indicates that full-scale values are unchanged. The symbol **1** indicates that full-scale values are limited to 1 least-significant bit less than full scale. In this mode downstream devices will not show digital "overs." The MAX lamp indicates full-scale in either mode. This bit is stored in RAM preset. Default is "0."

Default Parameters Screen

The **Model AD122-96 MX** recalls its power-up default settings from a nonvolatile memory (RAM). To change the default settings scroll to the RAM PRESET screen:

```
STORE RAM PRESET:
(PRESS GO/EXIT)
```

Then press the GO button. The new default settings storage is done and the display changes to:

```
STORE RAM PRESET:
DONE...
```

Bar-Graph & Display Screen

```
FINE: OFF 20    METER: -00   0
PEAK/HOLD OFF    DISPLAY:3
```

Bar-Graph settings:
FINE: OFF sets the bar graph display (LED's) for normal audio monitoring. Setting to **FINE: ON 20** activates the reference meter bridge function, for fine-tuning of signal level to -20dB reference point. The reference point can be set to a level between -10dB and -20dB by changing the "20" to the desired reference level. The incoming signal is displayed in 0.2dB increments around the reference point. Fine-tuning of signal levels may be done via an external volume control or by tuning the GAIN adjustment pots located on the front panel (the adjustment requires a small screwdriver).

The normal setting, "**METER:-00**" is used for showing peak and peak-hold levels ranging from 0 to -55dB (from full scale). The display may be extended to indicate lower level activity such as noise and DC offset.

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

While this feature is useful for determining noise floor of the incoming signal, a few comments are in order:

1. The Bar Graph displays a sum of AC signal "riding" on a DC offset. Monitoring low-level signals (such as noise floor) requires prior setting of DC removal at the audio control screen **DC REMOVAL ON**.

2. Viewing the noise floor takes some getting used to. The display shows a true picture of peak and peak-hold information; the blinking of LED's is due to low level quantizations. Peak readings are at about 11dB higher than the rms noise for flat noise spectrum.

3. Measuring the converter noise floor is best done with specialized equipment (such as Lavry Engineering's Model LE3000S). To verify the proper converter noise floor via extended metering you must activate the DC removal (**DC REMOVAL ON**) and connect a low value resistor (0-1k Ohm) between pins 2 and 3 of the analog input XLR connector. Floating high impedance analog inputs are susceptible to noise pickup.

Peak/Hold indication:
When set to **ON**, the peak level is held until a front panel switch is depressed. In addition, the "MAX" LED will turn on and stay on when output data hits the full-scale value. The appropriate 0dB bar-graph lamp (left or right) will start blinking. The "MAX" and corresponding lamps may be reset by pressing any of the front panel switches. When PEAK/HOLD is **OFF**, the MAX lamp will flicker when a single full-scale sample is encountered.

Display contrast: The LCD display may be adjusted from 0 to 7.

The "0" in the upper right corner of the screen is no longer used for setting a parameter in the MX model. It can be changed to a "1," but it should normally be set to "0" as the default setting.

Warm-up and Calibration

The recommended warm up time is 15 minutes. The AD122-96MX feature self-calibration for consistent long-term accuracy. A complete calibration cycle (about 5 seconds in duration) is activated at power on. A calibration cycle may be initiated by pressing both the EXIT and GO buttons simultaneously.

The final stage of calibration occurs when audio signals in the range of -25 dB on the AD's meters are fed to the analog inputs. This signal can be music program that is consistent in level and varies just above and below -25, or a tone at -25 dB.

This final calibration needs to occur:

1.) After the unit is powered "On" and the initial calibration cycle is finished.
2.) Any time the clock frequency or clock source of the AD122-96MX is changed.
3.) If the AD122-96MX has been "idle" with the power "on" but no signal has been fed to its inputs for more than one hour.

To see the operation of the final calibration after changing the clock source, with no signal present at the analog inputs, switch on a tone at -25 dBFS feeding the analog inputs. For a few seconds after the tone is switched "on," the level displayed on the meters will change slightly until finally settling on "-25 dB."

Factory Defaults

The AD122-96MX can be reset to the factory defaults by holding down the **EXIT** button while turning "On" the AC power. Continue to hold down the EXIT button until the unit finishes initializing and the main screen appears:



The factory settings will not be stored unless you go to the Default Parameters screen and store them in the RAM Preset.

In some instances, when the AD122-96MX is not operating in a consistent manner, resetting to the factory presets may restore stable operation. It is advisable to go through the menus and write down your settings before resetting to the factory presets. After resetting, you can go back and change the settings to the ones you wrote down, and store them in the RAM presets.

Please Note: Soft Saturation settings are NOT stored in the RAM presets.

Software Version

The current software version is displayed for a brief time in the LCD display, immediately after power is applied. The current version is displayed as "Ver 2.1e" in the upper line of the LCD display.

Hardware Interconnections

Analog inputs are applied to the XLR connectors marked as LEFT ANALOG INPUT and RIGHT ANALOG INPUT. The input signals can be balanced or unbalanced. The converter power line signal rejection is very high (120dB common mode rejection) but use of high quality shielded audio cables is recommended. See the SPECIFICATIONS section on page 11 for absolute maximum input levels.

The XLR pin polarity with +POLARITY setting in the Audio Control Screen is:
Pin 2 non-inverting and Pin 3 inverting.

The digital output AES/EBU connector (XLR) is transformer coupled, allowing connection to both AES/EBU XLR and SPDIF/RCA inputs. A simple XLR to RCA adapter or adapter cable can be used to feed the vast majority of coaxial SPDIF inputs. Transformer adapters are not recommended.

SYNC\WC INPUT (XLR connector) is used for external synchronization to AES Sync source. *Do not connect external sync to both the SYNC IN XLR and BNC Word Clock input at the same time.*

The four BNC connectors provide channel one and channel two data, word clock out, and word clock in. They can be used for synchronizing additional converters to each other or an external TTL word clock. The Word Clock Input is NOT internally terminated.

Regarding Termination:
For simple systems where there is only one device feeding the AD122-96MX's Word Clock Input with a short BNC cable, termination is probably not necessary. If you encounter problems getting the AD122-96MX to lock properly, or there are other devices in the "chain;" termination may be necessary.

If the AD122-96MX is the only unit being fed the Word Clock signal, connect the cable using a BNC "T" adapter plugged into the "Word Clock Input." On the other side of the "T", place a 75 Ohm BNC terminator connector. If the Word Clock is chained to other devices using "T" connectors, be sure that there is a 75 Ohm terminator on the "T" of the LAST device in the chain and that none of the other devices have INTERNAL TERMINATION. Alternately, if there is only ONE device with internal termination, it can be wired at the "end of the chain" in place of the terminator. The important point is that there is only ONE termination and it is located at the end of the chain.

The same approach may be used to chain the Word Clock output of the AD122-96MX. In this case, there is no "T" on the AD122-96MX Word Clock Output. As described above, the "T" adapters should be placed on the Word Clock input of the devices the AD122-96MX is feeding, with the termination ONLY at the very end of the chain.

Maintenance

The **Model AD122-96 MX** is an auto calibrating converter requiring no periodic adjustments. The unit generates a significant amount of heat (25 watts maximum). The temperature rise is no cause for concern, but allowing for some airflow is always a benefit from a long-term reliability standpoint. It is especially important not to block the openings in the top cover of the unit.

The front panel is gold plated (24 karat gold). Use a soft cloth (and plain water, if necessary) to clean fingerprints.

Part II: Specifications

Analog Inputs:
The analog signal can be amplified by 0-12dB utilizing the front panel screw adjust trim pots. Full-scale signal with 0dB gain is 24dBu, (full-scale signal with 12dB gain is 12dBu).

Input impedance over 0-100 kHz: greater than 20k ohm (pin 2 to pin 3).

Absolute Maximum Input Level: +24dBu Balanced, +18dBu Unbalanced regardless of Gain setting.

Noise Floor at 44.1-48KHz sampling:
-130dB ±1dB from full scale A weighted
-126dB ±0.5dB from full scale unweighted
Noise Floor at 96KHz sampling:
-127dB ±1dB from full scale A weighted
-123dB ±1dB from full scale unweighted.

Notes:
1. Measuring noise floor with unconnected (opened) inputs may yield erroneous results. Unconnected high impedance inputs may pick up unwanted signals.

2. Care must be exercised when measuring the noise floor. The signal applied to the converter must be lower then the noise floor. Ordinary relays and switches may leak too much signal, and it may be necessary to move a test tone signal to frequency above Nyquist (approx. 30KHz).

Output Bits: 16 to 24 bits, user selectable.

Total Harmonic Distortion and Noise:
(Specified for 22-24 Bits output, balanced inputs)

0.00005% for signals lower than -40dBF
0.0004% for signals lower than -20dBF
0.0008% for signals lower than -6dBF

Channel Separation:
Greater than 120dB (0-20KHz)

Sync. Inputs (connect to only one input at any time):
XLR connector: AES/EBU levels into 110 Ohms for digital audio

BNC connector: TTL level word clock

A.C. Power:
90-264 VAC @ 47-63Hz , Fuses 0.75A Fast 250V Fuse Size 5mm x 20mm (or 0.25inch x 1.25 inch)
The power supply automatically adjusts to AC inputs in the range of 90 to 264 volts at line frequencies between 47 and 63 Hz. It is not necessary to change any settings. The voltage selector in the power input module on the back panel is wired to work properly in either position. Thus, either the “115V” or “230V” legend can be visible and the unit will work properly with any input in the rated operating range (the legend is visible through a small “window” and is white lettering on a red background).

Appendix I: αβC αcoustic βit Correction™

High quality digital audio sound often incorporates long words (many bits). Both high-resolution A/D converters (22 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.

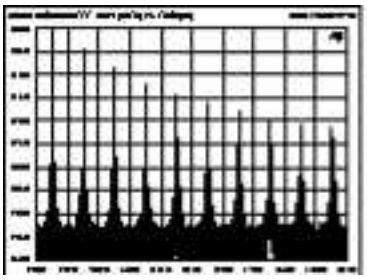
Proper dithering serves to eliminate the distortions and noise modulation effects. The desirability of such improvement is unquestionable but is coupled with increasing noise floor. Noise shaping is based on shifting the noise from high sensitivity frequency regions to less sensitive ones. That process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

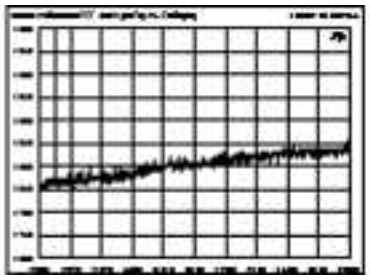
Customer feedback and further studies taught us two interesting facts:

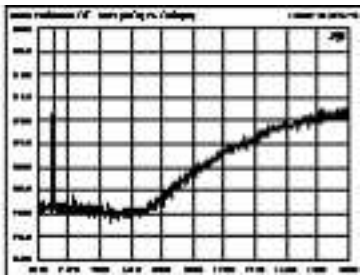
a. Various data compression schemes require the dither to have flat frequency response characteristics.

b. Recording engineers' preference of dither type (high pass or flat) and noise shaping curves often depends on characteristics of the music involved.

The **Model AD122-96 MX** offers High Pass or Flat Dither type and four noise-shaping curves. **Model AD122-96 MX** 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 αcoustic βit 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 BitCorrection High Pass- NS2

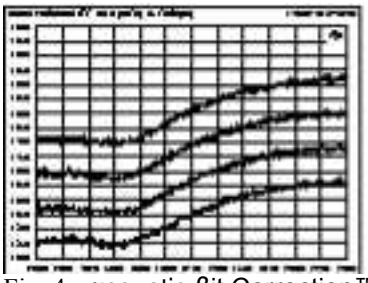
A number of mathematical and psychoacoustic researchers introduced two concepts for sonic improvement: dithering and noise shaping. Credit is due to L. Fielder of Dolby Labs for providing the basis for various noise shaping curves. The reasons for triangle PDF (probability density function)

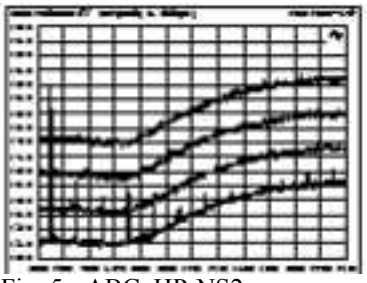
dither have been explained by S. P. Lipshitz and J. Vanderkooy of the University of Waterloo. Additional publications by M. Gerzon, P. G. Craven, J. R. Stuart and J. R. Wilson (all from the United Kingdom) and J. Goodwin (from the U.S.A.) shed light on this complex subject.
Dithered noise shaping technology 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 αcoustic βit Correction™. The principal conclusions are:

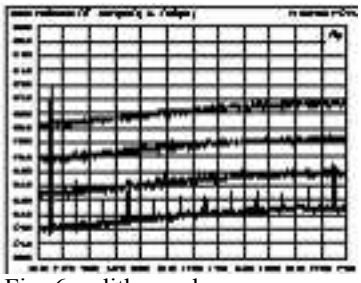
a. The practice of greatly amplifying low level signals to determine triangular flat PDF (probability density function) dither reveals the effectiveness of distortion and noise modulation elimination. This practice yields misleading results when testing unflattened dithers and/or noise shapers. It conflicts directly with L. Fielder's findings showing completely different threshold delectability 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. The ideal noise-shaping curve may be irritating at loud levels.
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 AD122-96 MX** test tone generator mode switching the αcoustic βit Correction™ on and off. The frequency and amplitude programmability was very useful.


Fig. 4 - αcoustic βit Correction™
16 (top), 18, 20, and 22 bits
no input signal


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


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

αcoustic βit Correction™ may be used with words of lengths wider than 16 bits. Figure 4 shows the noise floor of αcoustic βit 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 αcoustic βit 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.

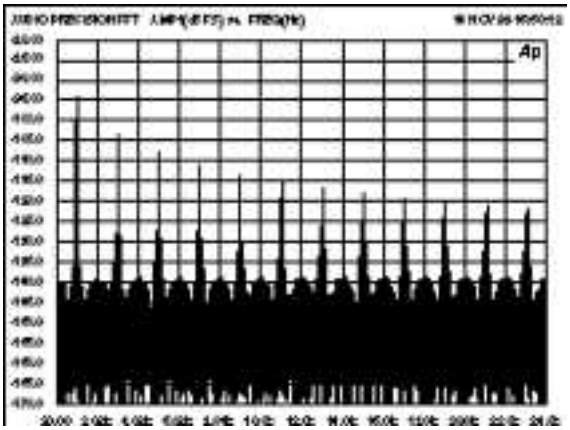


Fig. 7- 16 bit input signal -100 dB truncation by previous processing

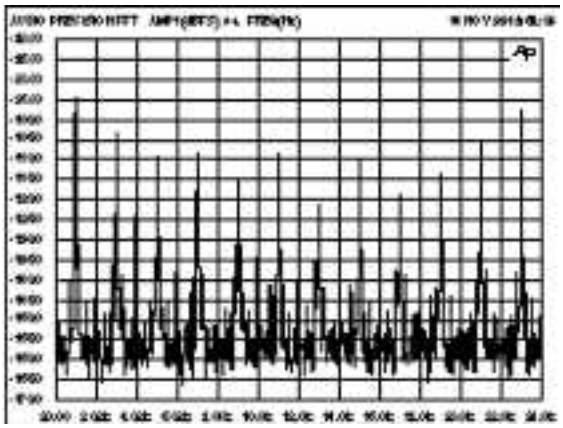


Fig. 8- 16 bit input signal -100 dB additional processing and truncation

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.

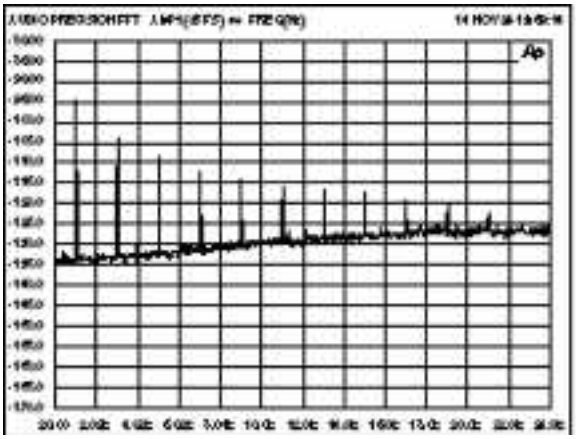


Fig. 9- 16 bit input signal -100 dB high pass dither applied

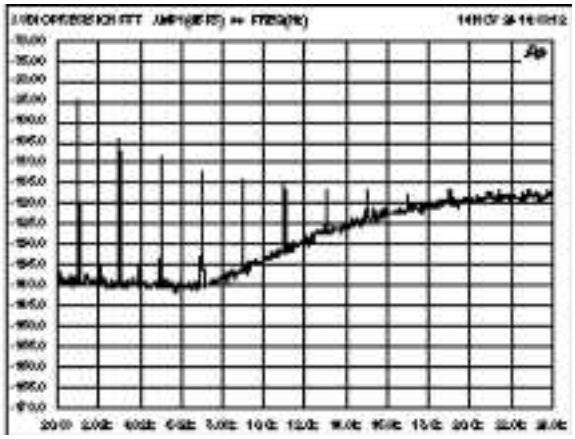


Fig. 10- 16 bit input signal -100 dB Acoustic Bit Correction applied

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 αcoustic βit Correction™ (dither and noise-shaping) applied to the same signal with an increase in the effective noise floor. αcoustic βit 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

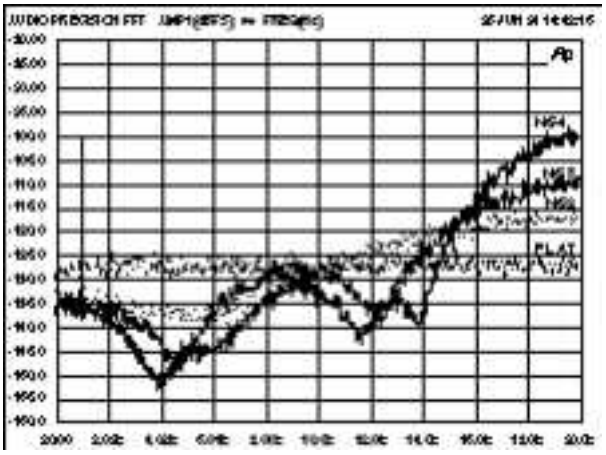


Fig. 11- 1kHz input signal -100 dB
Flat dither, with multiple noise-shapers shown

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.

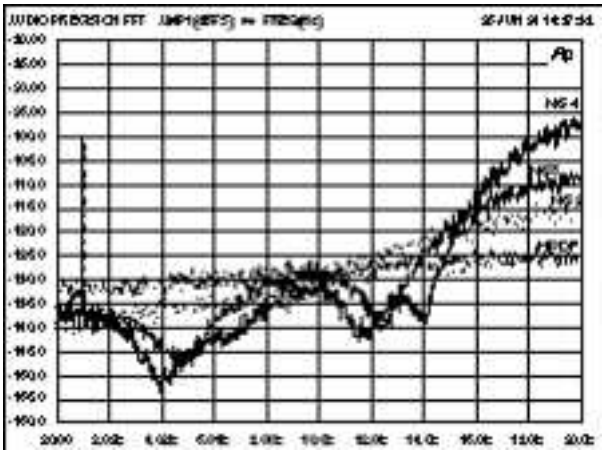


Fig. 12- 1kHz input signal -100 dB
HPDF dither, with multiple noise-shapers shown

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, NS1 curve has been omitted for clarity. It lies below NS2.

Appendix II: AD122-96 MX Input Considerations

Achieving 20 or more bits of resolution places a heavy demand on the linearity and noise of analog signals prior to the A/D conversion.

Linearity: Linearity problems (typically associated with larger signals) should be avoided to prevent overall signal distortions. Proper circuit design places non-linear devices inside a negative feedback path, thus allowing the transfer function to depend on a few linear components (such as resistor ratio). Closer examination reveals that the effectiveness of negative feedback is limited by the open loop gain of the amplifier (the available gain prior to the feedback action). Constructing a 10dB gain stage out of, say, a 50dB open loop amplifier provides 40dB of feedback; thus, circuit nonlinearities are corrected for by a factor of 100. A 1% nonlinearity reduces to a .01% level. The same stage, when set to a gain of 20dB, yields only 50-20=30dB of "linearizing power"; thus the same 1% circuit nonlinearity is corrected to 1/31.6 = .03%. The "obvious" solution of increasing circuit open loop gain is riddled with problems ranging from impulse response settling time to loss of bandwidth.

Noise: The quantization (A/D conversion) of analog noise causes loss of fine signal detail. The quantized noise introduces time varying signal dependent distortions. Such sample by sample deviation can not be compared to the seemingly "constant" analog background noise. The interaction between the noise and the quantization process effects the sonic detail at much higher amplitude than the noise level itself. The source material may have its own noise level, but the A/D manufacturer is in no position to decide which part of the material is signal and which part is noise. (Is it undesirable noise or is it the intentional recording of the wind blowing through the trees?) Thus, any signal must be reproduced accurately. All we can do, and all we should do amounts to avoiding additional A/D noise. A/D noise consists of two parts: analog noise (prior to quantization), and quantization noise. The quantization noise level is "fixed" by the converter itself. **Model AD122-96 MX** quantization noise relative full scale is about 126dBFS. Keeping complete system noise level at 126dBFS requires careful treatment of the analog path. Each circuit component (especially resistors and semiconductors) generates its own noise. The complex electron motion within the various materials adds up to a cumulative noise. Intentional amplification of signals yields undesirable amplification of noise. Once again, each dB of added gain takes away a dB from the available dynamic range.

AD122-96 MX Input Range: You may suspect by now that our aim is to provide you with an A/D converter and leave the problem of providing a "hot" signal at the hands of microphone and mic-preamplifier manufacturers. Like many A/D's, **Model AD122-96 MX** provides limited gain (0-12dB). There are 4 methods for analog input signal interconnection:

1. Balanced/ grounded shield
2. Balanced/ floating shield
3. Unbalanced/ grounded shield
4. Unbalanced/ floating shield.

Grounded shield: analog cable shield is connected pin 1 of the XLR at both ends of the cable driver end and A/D end.
Floating shield: analog cable shield is connected pin 1 of the XLR only at the driver end of cable. The shield is floating at the A/D side (not connected to pin 1).

Note: Much information is available regarding proper shielding. The well known book "Grounding And Shielding in Instrumentation" by Ralph Morrison provides an excellent explanation of the subject and points out clearly that the floating shield connection is the correct configuration. Engineers using the grounded configuration with 16-17 bit systems may wish to re-evaluate their interconnections for 20 bit systems.

1. Balanced input with floating shield: The unit handles up to 24dBu of peak signal with available 0-12 dB of front panel gain adjustment.

2. Balanced input with grounded shield: The unit handles up to 24dBu of peak signal with available 0-12 dB of front panel gain adjustment

3. Unbalanced input with floating shield: The unit handles up to 18dBu of peak signal with available 0-6 dB of front panel gain adjustment

4. Unbalanced input with grounded shield: The unit handles up to 18dBu of peak signal. The front panel gain range is 0-6 dB.

Comments: The first two configurations operate over the same range. Little attenuation is required in processing of a 24dBu signal (34.72V peak to peak).

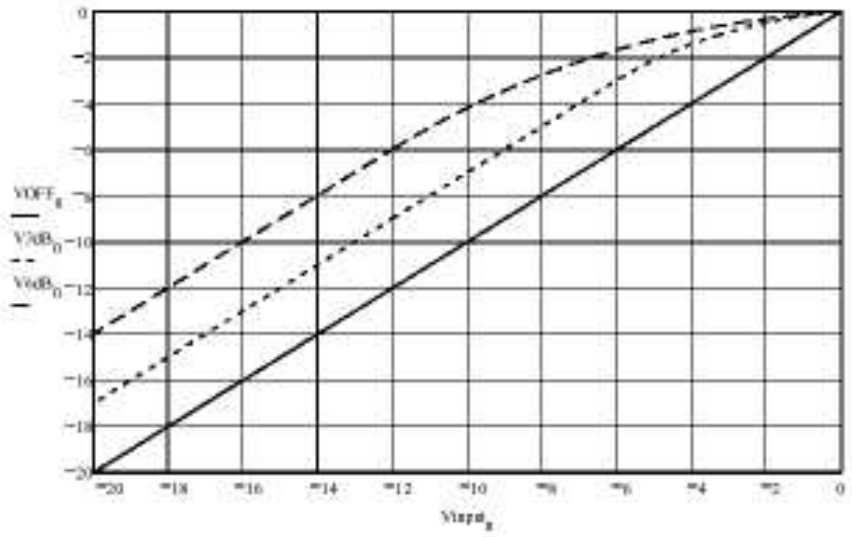
When applying unbalanced signals, use at least 6 dB of gain, using the gain adjustment pots located on the front panel.

Appendix III: AD122-96 MX Soft Saturation

The SOFT SATURATION feature provides a digital emulation of an ideal magnetic tape saturation characteristic. The three available settings are:

1. SOFT SATURATION OFF: yields normal converter operation (see lower curve VOFF).
2. SOFT SATURATION +3dB: provides a digital gain of 3dB for sample amplitudes below -6dBFS and digital compression for amplitudes greater then -6dBFS (see middle curve V3dB).
3. SOFT SATURATION +6dB: provides a digital gain of 6dB for sample amplitudes below -12dBFS and digital compression for amplitudes greater then -12dBFS (see upper curve 6dB).

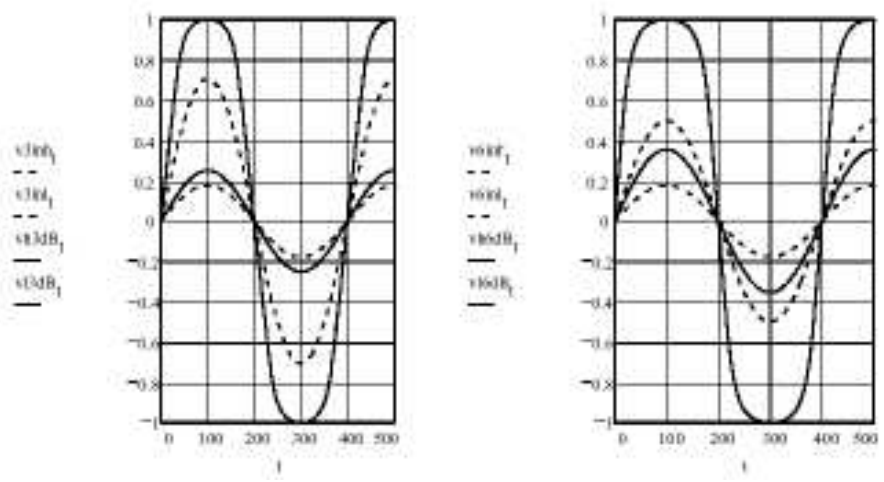
The graph below shows the three transfer functions. The plots show input versus output value between -20dBFS and 0dBFS.



The algorithm for soft saturation is fundamentally different from classical audio compression. A compression algorithm provides time varying gain as a function of signal power. The gain variations are slowed down (attack and decay envelopes) to minimize distortions (a sudden step in signal gain generates a time decaying distortion).

The soft saturation algorithm operates on a sample by sample basis. All sample values below a given threshold (-6dBFS or -12dBFS) are amplified by a constant gain (+3dB or +6dB) and all sample values above the threshold are reduced by a pre-assigned amount emulating ideal tape magnetization curve (exponential function). While signal peaks get "pushed down" (peak signal distortions), the dynamic performance for signals under the selected threshold stays intact.

Below are time domain plots for the +3dB setting (left plot) and +6dB setting (right plot).
Each plot shows the transformation of two input signals (dotted lines) to two output signals (solid lines). The settings provide different tradeoffs between gain and "peak signal distortion".



The soft saturation algorithm provides a "soft limiter" function (aimed at making "hotter recording"). The function provides several advantages over conventional analog limiters.

Analog limiters are based on physical components; thus, the attainable curves are restricted to available hardware, also resulting in less then ideal transition between the linear and the non-linear portions of the curve.

Digital processing is free of such restrictions. The curves go beyond simple gluing of two transfer functions at the transition point. Our experiments showed that the two factors affecting the sonic quality are the shape of the curve and the behavior around the transition point. Our choice of an exponential curve was based on the nature of the distortions (crudely speaking, less distortion energy at higher harmonics). The key to a smooth transition was to make the **slopes** of the linear and non linear portions are precisely equal to 22 bit accuracy (as always, such precision is much more apparent to a listener of 16 bit data then to a viewer of "9 bits graph".

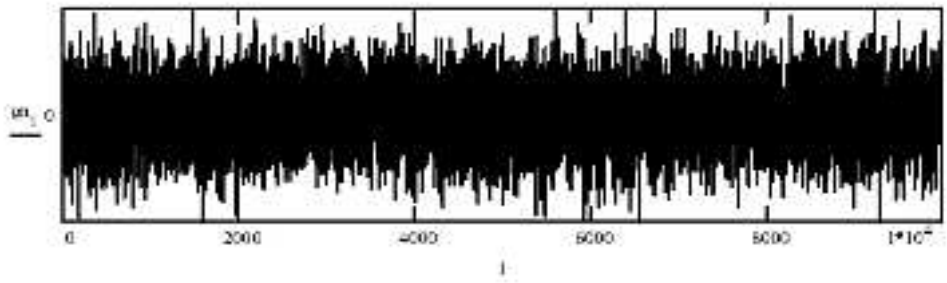
Unlike the "analog soft knee", the digital soft saturation is a precise transformation. The user may not drive the signal further and further into the saturation region (compromising between "hotter recording" and "acceptable distortions"). The +3dB setting provides perfect linearity to -6dBFS of the incoming signal (-3dBFS of the processed signal), and the +6dB setting provides perfect linearity to -12dBFS of the incoming signal (-6dBFS of the processed signal). It is easy to justify the usefulness of this function when the complete audio track is below these threshold levels. The compromise is restricted to the upper 6dB (or 12dB). The choice is left up to the user.

Note: The maximum light is operational during soft limit. It indicates that the signal has reached full scale, thus warning against overdriving into the clipping region.

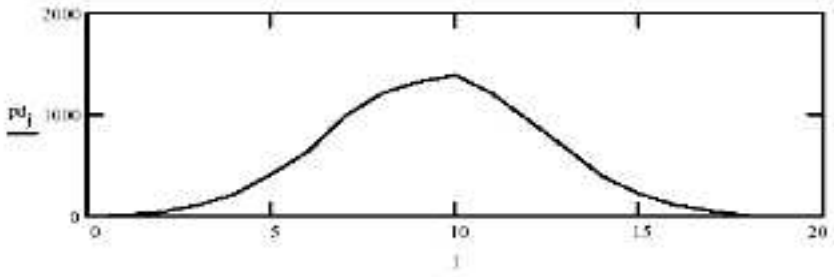
Appendix IV: Viewing Noise Floor on AD122-96 MX Display

The simplest form of random noise is "white noise". White noise is bounded between two limits. The probability of occurrence for any amplitude value between the two limits (boundaries) is equal; thus, it is also named uniform distribution noise. Analog noise is not white. It is mostly "Gaussian", with higher probability of occurrence for smaller amplitudes. The large amplitudes occur less frequently and the distribution follows the familiar "bell shaped curve".

Let us plot an "oscilloscope picture" sample of 10000 points of a Gaussian sequence:



To view the Gaussian data in its more familiar bell-shaped form, we use the following construction, computing a histogram by grouping the values in 20 bins of width .05:



Computing the mean and standard deviation yields close to perfect results (mean and stdev):

= mean( ) gn 0.001 = stdev( ) gn 1.001

Gaussian noise energy is measurable via a rms. (root mean square) meter. The proper measurement of converter noise should be done using a rms. instrument (such as Lavry Engineering's Model 3000, Audio Precision test system or others). However, the known bell shaped curve coupled with a peak reading offers a reasonable and quick estimate of the rms. value:
Step 1: Provide a low resistance path (0 to 500 Ohms) between pin 2 and pin three of the analog input connector(s).
Step 2: Set the DC FILTER ON (remove the DC offset to overcome its effects on the Bar-graph display)
Step 3: At the Display Control Screen, set to "METER: -90".
Step 4: Read the peak value on the LED Bar graph.
Step 5: The noise floor is at about 8dB below the reading
Example: The reading of peak is at -24dB. Subtracting 90dB yields -114dB of peak hold values. Subtracting 8 yields -114 - 8 = -122dB of noise floor.

Such measurement and calculation may aid in evaluating the noise of a driver device, or a series of devices. The distribution due to addition of Gaussian signals is still Gaussian. While the 8dB "translation factor" is true for Gaussian noise, it does not hold true for non-Gaussian noise. Such factor is 3dB for sine waves and 0dB for square waves. Higher than expected reading may indicate a pickup of non-random noise such as power line hum.

Explanation for the 8dB factor:

The Gaussian bell shape curve is unbounded. Theoretically speaking, very high amplitude may occur, but very high amplitudes occur very seldom (it has been said that waiting a few trillion years may produce a million volt spike). Practical observations show that most of the peaks occur within 3.5 times of the standard deviation (sometimes referred to as a crest factor of seven). The following calculation:

$20 \log (3.5\sqrt{2}/2)=7.871$

justifies adding about 7.81dB to the peak hold reading. Given a 1dB meter resolution, we round the correction factor to 8dB.