function, which is referenced to the 1-dB/step scale. Metering is fully digital and post conversion for absolute accuracy.

Power Supply Connection

The AD2402-96 is equipped with an industry standard 4-pin XLR power connector, and may be powered from a 12 V battery, or from an external DC power supply.

The input voltage should be between 11 and 18 VDC.

Pin 1 = Ground,

Pin 4 = +11 to +18 VDC at 0.8 A.

Pins 2 and 3 are not connected internally.

Chapter 2

Front Panel

Detailed Information on the AD2402-96 front panel.

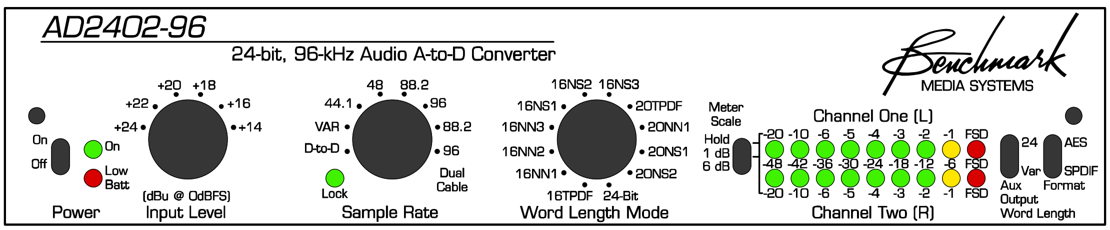

Figure 1. AD2402-96 Front Panel

"Power" Switch

The power switch is equipped with a short handle to prevent accidental shut off. If you are monitoring the output of the AD2402-96 with headphones or speakers, mute them before turning this switch on or off. Also, it is a good idea to turn this switch off before connecting or disconnecting the DC power.

"On" LED

The green "On" LED will light whenever the power switch is on <u>and</u> the input voltage is high enough for normal operation. The "On" LED will not light if the input voltage is less than 10.5 V. An illuminated "On" LED is your assurance that the input voltage is high enough for normal operation.

"Low Battery" LED

The AD2402-96 is equipped with a red "Low Batt" LED. The low-battery light is calibrated for use with 12-volt lead-acid "gel-cell" batteries but may be adjusted for use with other types of batteries. Lead-acid batteries should be charged shortly after the "Low Batt." LED turns on. Note that the AD2402-96 will continue to operate normally as long as the green "On" LED is illuminated. Typically, the AD2402-96 can operate for 1-hr with the red LED illuminated before the green LED will turn off. However, prolonged and/or frequent deep discharge cycles (operation with the red LED on) will shorten the lifetime of a lead-acid battery. Lead-acid batteries should be recharged after every use, and should be stored in a cool location, fully charged.

"Input Level" Switch

This rotary switch adjusts the sensitivity of the analog inputs. The numbers (from +24, down to +14) indicate the level (in dBu) at which a full-scale digital code will be reached. At "+24" the converter will clip when the analog input reaches or exceeds +24 dBu. At "+14", the converter will clip when the analog input reaches or exceeds +14 dBu.

The "Input Level" switch should be set so that the AD2402-96 will clip 3 to 6 dB *before* the microphone preamplifier. For example, if your microphone preamp has a maximum output level of +24 dBu, then set the switch to "+20". This will reserve 4 dB of headroom in the preamplifier. If too much headroom is allowed in the microphone preamplifier, noise from the preamp may reduce the dynamic range of your recording. If no headroom is provided, the microphone preamp may add distortion to your recording.

Set the "Input Level" switch to match the performance of your preamp. Do not use this switch to adjust levels during a recording. Instead, use the gain control on your microphone preamp. This switch is used specifically to calibrate the AD2402-96 to a line level source, mixing console or microphone pre-amp. It is NOT intended as a gain control for adjustment during a recording session. Set the switch in accordance with your pre-amp or mixer before beginning the recording session.

+24 dBu is a very hot level and is often only available at the output of high quality microphone preamplifiers. Some battery operated microphone preamps cannot achieve +24 dBu without clipping, and will require setting the AD2402-96 to a lower clip point (typically "+20" or "+18").

Use the "+14" setting for unbalanced inputs. Some consumer grade devices are not capable of achieving even +14 dBu levels without clipping. These low-level unbalanced outputs are not a good match for the performance of the AD2402-96 but can be connected through an external balancing interface amplifier if necessary.

All preamplifiers have an intrinsic noise floor that is set by basic physics. The contributing elements are the source impedance of the microphone, usually between 50 and 200 ohms, the noise of the first stage of the preamplifier, the noise of second fixed gain stage, and the noise of the output stage.

A preamplifiers dynamic range is defined by the difference between this intrinsic internal noise level and the peak output clip point of the preamp. If the clip point is reduced to say + 12 dBu, and nothing short of miraculous is done to reduce the second and third stage noise of the preamplifier, the dynamic range of the pre-amp will be limited to much less than the AD2402-96.

The only way to achieve a low noise level with the preamplifier is to use extremely low internal impedances and ultra low-noise operational amplifiers. Both of these required techniques are power hungry and incompatible with portable pre-amps where long battery life is an objective.

Therefore, avoid microphone preamps that cannot achieve at least +16 dBu out, without clipping. Even then the dynamic range of the preamp and not the AD2402-96 will, in all probability, be limiting the dynamic range of your high-quality 24-bit recordings.

"Lock" LED

The "LOCK" LED will turn on whenever the AD2402-96 is locked to a reliable digital audio reference of the correct frequency. The digital input/reference signal must be AES/EBU or SPDIF.

The AD2402-96 will lock at 1:1 or 2:1 ratios. For example, if the AD2402-96 is operating at 96 kHz, the converter will lock to either a 48 kHz or 96 kHz digital audio signal.

If the lock light is off, the AD2402-96 is operating as clock master (using the internal crystal reference).

Important Note:

An external reference is normally not required for 2-channel recording. The lock LED will *not* be on when the AD2402-96 is operating as a stand-alone (clock master) device. This is not an indicator of an error.

A digital audio input is required in the "Var" (variable sample rate) and "D-to-D" (digital to digital) modes. It is optional in all other modes.

When locking to an external reference, the "Lock" LED will flash for several seconds while the PLL is acquiring lock and will be lit once internal lock is achieved.

A continuously flashing "Lock" light indicates a lock error.

Possible causes of a lock error include:

- Reference sample frequency is incorrect or out of range.
- Reference signal is neither AES/EBU nor SPDIF.
- Reference signal is being received with errors.
- Reference is required ("Var" and "D-to-D" modes) but is not present.

Note: The AD2402-96 automatically mutes for the first 10 seconds after applying power. While muted, the "Lock" LED will flash nine times indicating that automatic calibration is in progress. This is not an indication of a lock error.

"Word Length Mode" Switch

The "Word Length Mode" switch is a twelve-position rotary switch. It selects 24, 20, or 16-bit word lengths at the main outputs. All 16 and 20-bit Word Length Reduction (WLR) modes are TPDF dithered prior to word length reduction.

There are seven word length reduction modes ("TPDF", "NN1", "NN2", and "NN3", "NS1", "NS2", and "NS3"). All forms of word length reduction raise the noise floor of a digital transmission system. TPDF dither is spectrally flat, it is not shaped, and will sound noisier than any of the "NN" or "NS" modes. "NS1", "NS2", and "NS3" are noise-shaping modes which are psycho-acoustically optimized to take advantage of the ear's low-level sensitivity curve. "NS1", "NS2", and "NS3" will sound 6, 12, and 18 dB quieter than "TPDF" respectively. "NS1", "NS2", and "NS3" can provide 17-bit, 18-bit, and 19-bit performance respectively at a 16-bit word length. At 20-bits, "NS1" and "NS2" provide 21-bit and 22-bit performance respectively. For the best performance, use the maximum word length that is compatible with your digital audio equipment. Maintain 24 or 20-bit word lengths as long as possible. If WLR to 16-bits is required, do so at the latest possible point and time. Avoid processing 16-bit signals. In general, NS2 will produce the quietest 20-bit signal, and NS3 will produce the quietest 16-bit signal.

Caution: Do not apply word length reduction when using the D-to-D function for dubbing 16-bit tapes. Use the "24" setting for all D-to-D functions unless you wish to simultaneously reduce the word length of the input digital audio signal. Use the "24" setting for 16-bit to 16-bit dubbing, 20-bit to 20-bit dubbing, and 24-bit to 24-bit dubbing. For more information see Table 1.

"Sample Rate" Switch

The "Sample Rate" Switch is a twelve-position rotary switch. This switch selects the sample clock frequency, the PLL (Phase Locked Loop) mode, dual or single cable interface modes, and digital-to-digital functions.

For typical A/D conversion applications, select either "44.1" or "48". If an external digital audio reference is connected to the "Digital Input", the AD2402-96 will automatically switch from internal to external clock, and the "Lock" LED will turn on. A feature unique to Benchmark converters is that all automatic transitions between external and internal clock are silent and will not interrupt the operation of the A/D converter.

For A/D conversion at non-standard frequencies, use the "Var" (varispeed) setting. In varispeed mode, the A/D sample rate is determined by the sample rate of the digital audio input signal. If no digital input is present, the converter will mute, and the digital outputs will default to 44.1 kHz. The "Var" mode supports any sample rate between 28 and 100 kHz but *does not* have the jitter immunity provided by the fixed frequency modes. Use fixed frequency mode whenever possible.

For A/D conversion at high sample-rates using "Single-Cable" interfaces, select either "88.2" or "96". These single-cable 88.2 and 96 kHz settings are labeled in white. These settings are compatible with high sample-rate recorders that are specifically designed to support single-cable interfaces.

For A/D conversion at high sample-rates using "Dual-Cable" interfaces, select either "88.2" "Dual-Cable" or "96" "Dual-Cable". These dual-cable 88.2 and 96 kHz settings are labeled in black. These settings allow high sample-rate recording on 44.1 and 48 kHz recorders. Channel 1 will be directed to the "Main" outputs. Channel 2 will be routed to the "Aux" outputs. Two tracks will be required for each channel.

> Caution: In "Dual Cable" mode, each digital output is dedicated to a single audio channel instead of a stereo pair. "Dual Cable" recordings require 88.2 and 96 kHz playback and editing equipment. Furthermore, "Dual Cable" recordings must always be dubbed and transferred from 44.1 and 48 kHz recorders in the digital domain. The "Dual Cable" digital output from the recorder must always be mixed, edited, processed, and converted to analog using 88.2 or 96 kHz equipment. Label all "Dual-Cable" recordings carefully! A 96 kHz "Dual Cable" recording may appear to play properly at the analog outputs of a 48 kHz machine, but these analog output signals will actually contain unwanted alias tones that cannot be removed with a filter.

For Digital to Digital processing, select "D-to-D". The "D-to-D" mode provides high-quality digital to digital processing functions that can be used individually or in combination.

"D-to-D" functions include; word length reduction, SPDIF to AES/EBU format conversion, AES/EBU to SPDIF format conversion, and SCMS copyright correction. In addition, the AD2402-96 can act as a 1-in, 4-out digital distribution amplifier.

The digital input will accept either consumer or professional format digital audio. The "Format" switch always determines the format of all four digital outputs.

Important Note: If the "Word Length" switch is set at "24", the "D-to-D" mode will alter the status bits but will not alter audio data bits. All other "Word Length" settings will add dither noise to the audio. **Use the "24" setting whenever you do not wish to alter the word length of the audio signal**. However, if you wish to reduce the word length of the audio data, use the "Word Length" switch to select the desired word length reduction function. See Table 1.

Table 1 - Word Length Mode Settings for D-toD Processing

Input Word Length	Output Word Length	Word Length Mode Switch Settings
16	16	24
18	16	16NS3, 16NS2, 16NS1, 16 NN3, 16NN2, 16NN1, or 16TPDF
18	18	24
20	16	16NS3, 16NS2, 16NS1, 16 NN3, 16NN2, 16NN1, or 16TPDF
20	20	24
24	16	16NS3, 16NS2, 16NS1, 16 NN3, 16NN2, 16NN1, or 16TPDF
24	20	20NS2, 20NS1, 20NN1, 20TPDF
24	24	24

"Meter Scale" Switch

The "Meter Scale" switch selects one of three digital meter functions. The down position sets the meter scale to 6-dB steps. The center position sets the meter scale to 1-dB steps. The up position sets the meter scale to 1-dB steps with peak hold. Moving the switch to the center position will clear the peak hold. Note that the bottom of the 1-dB scale is expanded and includes a -20 dBFS LED that can be used to set the input level relative to a 0-dB house reference. The "6 dB" setting makes it easy to verify that signals are present at the analog inputs. The "1 dB" and "Hold" settings permit accurate monitoring near full scale.

Digital LED Meters

Each conversion channel is equipped with a multi-function 9-segment LED meter. A **"Meter Scale"** switch selects either a 6 dB/step or 1 dB/step scale and controls the **peak hold function**. Metering is fully digital and post conversion for absolute accuracy.

Time constants are built into the meters so that all transient peaks can be observed easily. If a transient peak having a duration as short as one digital sample occurs, an LED will be illuminated, and will stay illuminated long enough to be observed by the human eye.

A peak indication mimics the action of the needle on a peak-reading analog meter, while the remaining LEDs will follow the instantaneous level of the audio.

The red "FSD" (**F**ull-**S**cale-**D**igital) LED indicates that a full-scale digital code has been reached and that digital clipping has occurred. Full-scale events as short as one digital sample, will light the "FSD" LED. Short single-sample digital clipping events are often audible, and all "FSD" events should be avoided.

> The AD2402-96 has a very large dynamic range (especially when operating at 24-bit output word lengths). It is wise to use some of this dynamic range to provide more headroom as insurance against clipping. Leave some extra headroom between your highest anticipated peak and the red "FSD" LED.

"Aux Output Word Length" Switch

This two-position toggle switch controls the word length of the auxiliary digital outputs. This switch allows simultaneous recording at 24-bits and 16-bits, or at 24-bits and 20-bits.

When the "Aux Out Word Length" switch is set to "24" the auxiliary outputs will always operate at 24-bits, while the "Word Length Mode" switch will control only the "Main" outputs. When the "Aux Out Word Length" switch is set to "Var", the "Word Length Mode" switch controls all outputs.

> The "Aux Output Word Length" switch has no function when the "Sample Rate" switch is set to one of the "Dual Cable" settings. In "Dual Cable" mode, the "Word Length Mode" switch controls the word length of all outputs.

"Format" Switch

The two-position "Format" switch controls the format of the digital outputs. The "AES" position sets the status bits to professional (AES-3) format. The "SPDIF" position set the status bits to consumer (IEC 60958-1) format.

> Many "professional" recorders can accept either professional or consumer formats. However, many "consumer" recorders cannot accept professional formats. In most cases, the "S/PDIF" setting will allow simultaneous output to both "professional" and "consumer" devices on both XLR and BNC or RCA (coax) cables.
>
> Consumer status bit formats do not currently have the capability of identifying 88.2 or 96 kHz sample rates. Consequently it is advisable to use the "AES" (professional) format setting for all 88.2 and 96 kHz recordings.

Chapter

3

Rear Panel

Detailed Information on the AD2402-96 Rear Panel.

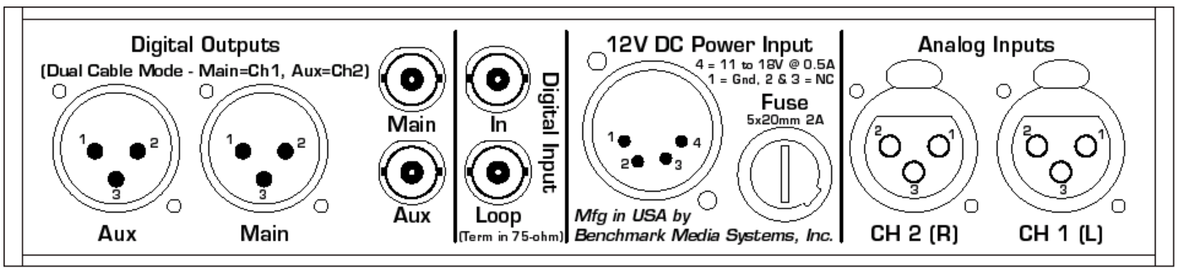

Figure 2. AD2402-96 Rear Panel

Analog Inputs

The analog inputs support balanced or unbalanced inputs. Input sensitivity is adjustable in 2 dB steps. Maximum sensitivity is +14 dBu at full scale (0 dBFS). Minimum sensitivity is +24 dBu at full scale (0 dBFS).

PLEASE NOTE: An internal absolute polarity inversion error was discovered some time after the creation of the AD2402-96/AD2K+. The following table is revised from the original and should be used instead of the original table.

Balanced Inputs: Pin 1 = Shield, Pin 2 = -Signal, and Pin 3 = +Signal.

Unbalanced Inputs: Pin 1 = Shield, Pin 2 = Signal Ground, Pin 3 = Signal.

Caution: Unbalanced inputs will not function properly if pin 2 is left unconnected.

DC Power Input

The AD2402-96 is equipped with an industry standard 4-pin XLR type power connector, and may be powered from a 12 V battery, or from an external DC power supply.

> The input voltage should be between 11 and 18 VDC.
>
> Pin 1 = Ground
>
> Pin 4 = +11 to +18 VDC at 0.8 A.
>
> Pins 2 and 3 are not connected internally.

Fuse

The AD2402-96 is equipped with a 2-Amp fuse. This fuse is designed to blow if the AD2402-96 is connected to improper voltages and/or in reverse polarity. It is important to use an exact replacement.

Digital Input Connectors

"In" Connector

This BNC connector provides a digital audio input to the AD2402-96. The digital audio input has two functions: It can provide a phase and frequency reference to the AD2402-96 sampling circuitry, or it can serve as an input for Digital–to-Digital processing functions.

> All digital inputs and outputs are transformer coupled. The transformers isolate the AD2402-96 from RF interference, provide protection against transients, and isolate the internal circuitry from the effects of ground loops. DC blocking capacitors are included to protect against damage from improper connections to phantom supplies, or digital microphone supplies.

> Use 110-ohm digital audio cable for XLR connections, and 75-ohm coax for BNC and RCA connections. Incorrect cable impedances may increase jitter, may cause

> data loss, and will reduce the maximum transmission distances. Standard analog audio cable should *not* be used for digital signals.

The Digital "In" connector accepts either professional or consumer status bit formats. The PLL automatically supports 1:1, and 2:1 lock ratios when a fixed sample rate is selected. In other words, a 48-kHz digital audio signal may be used as a reference when the converter is operating at 96 kHz. However, the varispeed mode ("VAR") requires a 1:1 lock ratio.

A digital audio input is required when:

- The converter will be operating in a variable speed mode ("VAR 1:1").
- More than two channels must be phase locked together.
- The converter is to be locked to an external studio reference.
- The digital-to-digital ("D-to-D") mode is selected.

If a reference is required, but no reference input is detected, the "LOCK" LED will flash rapidly. If a reference is present, but lock has not been achieved, the "LOCK" LED will flash slowly.

> Note: Using an external reference will not degrade or alter the low-jitter performance of the AD2402-96 converter at any of the fixed sample rates. The jitter performance of the fixed sample rate modes is maintained even when the reference has relatively high levels of jitter.
>
> However, the "VAR" variable speed mode has a wide frequency range that precludes the use of the final VCXO stage of the AD2402-96 PLL. If a variable speed mode is used, it is important to provide a low-jitter reference signal. Use the fixed sample rate settings whenever possible.

Digital-to-Digital processing functions include; Word Length Reduction, Pro to Consumer and Consumer to Pro channel-status conversion, SPDIF to AES/EBU and AES/EBU to SPDIF conversion, SCMS override, and 75 ohm unbalanced to 110 ohm balanced conversion. The D-to-D feature also allows the AD2402-96 to act as a 1 in 4 out digital distribution amplifier.

> In most 2-channel recording applications it is not necessary to connect a signal to the "Digital Input" connectors.

The digital input will only accept digital audio signals, and will automatically recognize either professional or consumer formats. It will not respond to non-audio signals such as word clock, super clock, or video.

"Loop" Connector

This BNC connector is provided to allow looping (or daisy chaining). The "In" and "Loop" connectors are directly connected to each other, and are transformer coupled to a high-impedance digital input circuit. The internal digital circuit does not place a significant load on the digital input connectors. This high-impedance design allows looping without degrading the quality of the digital input signal.

One digital audio reference signal can be looped through multiple AD2402-96 converters to provide phase accurate multi-track recording capability. Or, one digital-audio source can be looped through one or more AD2402-96 converters to provide a large digital-to-digital fan out for dubbing. If the loop feature is not being used, a 75-Ohm termination should be connected to the "Loop" connector (a 75-Ohm termination is supplied with the AD2402-96).

Digital Outputs

Four digital output jacks provide a unique set of features. Two outputs are 75 Ohm unbalanced at 1 Vpp, and two outputs are 110 Ohm balanced at 4 Vpp. All outputs can operate in either professional or consumer status formats. All outputs support 16, 20, or 24-bits, but the two "Aux" outputs can be set to operate at 24-bits while the "Main" outputs operate at either 16 20, or 24-bits. All outputs support stereo operation at sample rates up to 100 kHz on a single digital cable. In addition, a "Main" and "Aux" output pair can be used to record "dual-cable" 88.2 or 96 kHz.

The 75-ohm BNC connectors support SPDIF consumer format, as well as AES3-id and SMPTE 276M professional formats. The 75-ohm interfaces are well suited for long cable runs, and can easily achieve transmission distances of 1000 feet without cable EQ, and 3000 feet with cable EQ. AES3-id and SMPTE 276M use identical data formats and are fully inter-operable.

Chapter

4

Using the Meters

9-Segment LED Meters.

The AD2402-96 has a nine-segment LED meter for each audio channel. The meters are fully digital and respond to both positive and negative going peaks. Thresholds are determined by digital comparators, and therefore are exactly matched between channels. Dual time constants extend the on time of each LED so that peaks as short at only one sample can be displayed and measured accurately. The "FSD" clip indicator is accurate to one quantization level.

Using the Peak Hold Function

Moving the meter control switch all the way up enables the peak hold function. This sets the slow time constant to infinity. The highest peak will be held, all segments below the peak level will be controlled by the fast (7.8 msec) decay time constant.

Selecting Meter Range

The AD2402-96 has two meter ranges; one with 6-dB steps, and one with 1-dB steps. The 6-dB scale allows monitoring for signal presence, as well as coarse adjustment of levels. The 1-dB scale allows highly accurate adjustment of digital levels. In addition, the bottom two steps of both scales are expanded.

In either range, a light will not light until the appropriate threshold is reached. In other words, the -1 dBFS light will remain off until a digital code equal to or exceeding - 1 dBFS is encountered. Therefore, a signal at -1.01 dBFS will read -2 dBFS on the meter. This guarantees that the step between -2 and -1 is the same size as the step between -1 and FSD.

Meter Time-Constants

The meters on the AD2402-96 have instantaneous peak response. In other words, the amplitude of a single sample will read accurately on the meter. However, the response time of the human eye is much too slow to allow us to see an LED light up for one sample of a 44.1 kHz or 48 kHz clock. Therefore, the meters on the AD2402-96 incorporate a decay time constant which extends the on-time of all LEDs, and a second slower time constant which extends the on time of the highest LED triggered by a peak. Even the shortest transient can be observed.

Fast Decay Time-Constant

The fast time-constant is active in all meter modes. Its purpose is to compensate for the relatively slow response of the human eye. Here is how it works: If any segment of the LED meter turns on, it and all of the segments below it, are held on for 375 samples (or 7.8 msec). This 7.8 msec "on-time" is just long enough to make the LED clearly visible to the human eye. At the end of the 7.8-msec delay, the first time constant releases its control of the meter segments. If a higher peak should occur during the 7.8 msec interval, this new peak will be displayed, and the 7.8 msec timer will restart. Thus no peaks are ever missed, and all are visible.

Slow Decay Time-Constant

The slow time-constant is active whenever the peak hold function is off. After the first time constant releases control of the meter segments, a second 0.5-sec time constant will continue to keep the highest illuminated LED lit. All meter segments below this LED will continue to display peaks using the 7.8 msec time constant. At the end of 0.5 sec, the highest illuminated LED will shut off, and the LED below it will turn on. This will restart the 0.5-sec timer. Thus, peaks will decay at a rate of one-meter segment every 0.5 seconds. However, if at any time, an audio peak occurs which is higher than the one being held the new peak will be held and the timer will restart. Again, no peaks are lost, and all are visible.

Chapter

5

“FSD” Meters

What is “FSD”?

There are several different ways of labeling the clip LED on a digital meter. These include; 0 dBFS, Clip, Over, and FSD.

“FSD” stands for “**F**ull **S**cale **D**igital” and has a very specific and slightly different meaning than “0 dBFS”, or “Over”. It is important to understand the difference.

The “FSD” indicator will light whenever the minimum or maximum digital code of the converter is reached for a duration of one sample or more. No other digital codes will ever cause the “FSD” indicator to light.

How is “FSD” Different from “0 dBFS”?

Ideally there is no difference, but in practice, 0 dBFS has often been used to describe any digital code which is very close to full scale. “FSD” is used to describe only the minimum and maximum digital code.

An “FSD” meter must have knowledge of the digital word length in order to work properly. A 16-bit “FSD” code will not register “FSD” when feeding a “FSD” meter that is expecting a 20-bit word length. The reason for this is that the 16-bit word will have 4 trailing zeros appended to it (to make it a 20-bits word), and the 20-bit “FSD” meter will interpret this as a level which is 16 codes below full scale. Fortunately testing for non-changing trailing bits easily solves this problem.

How is “FSD” Different from “Over”?

An “Over” indicator (as specified in the Sony 1630 OVER standard) will only light if a minimum or maximum digital code is reached for three or more consecutive samples. One or two consecutive full-scale digital codes are not considered an “Over”. Some “Over” meters deviate from the Sony standard and allow the selection 4, 5, or 6 contiguous full scale codes before indicating an over.

The Fallacy of "Over" Indicators

"Over" indicators are based upon the assumption that a digital clip cannot be heard if it has a duration of three or less consecutive samples. Unfortunately "over" meters will ignore many audible overloads, and are poorly suited for 24-bit recording.

There is a high probability that a musical signal can clip over a duration of many samples without ever clipping on three successive samples. The reason for this is that the sensitivity of "Over" meters is a function of frequency. In fact, "over" meters are very insensitive to certain frequencies. For example, at a 44.1 kHz sample rate, a 10 kHz tone will not cause three consecutive full-scale codes until it is severely clipped. Given a complex musical signal, the reliability of an "over" meter can best be described as "hit or miss".

If you have an "over" meter, try this. Feed a 10 kHz tone into any A to D converter at a level equivalent to +3 dBFS. As shown in the graph below, peak voltages will reach almost 1.5 times clip level. This clipping will cause severe harmonics at 20 kHz, and distortion will exceed 14%. The resulting distortion will sound really nasty, but the "over" meter will not detect the clipping. A close examination of the graph will show that under these circumstances, 3 consecutive full-scale codes can never occur.

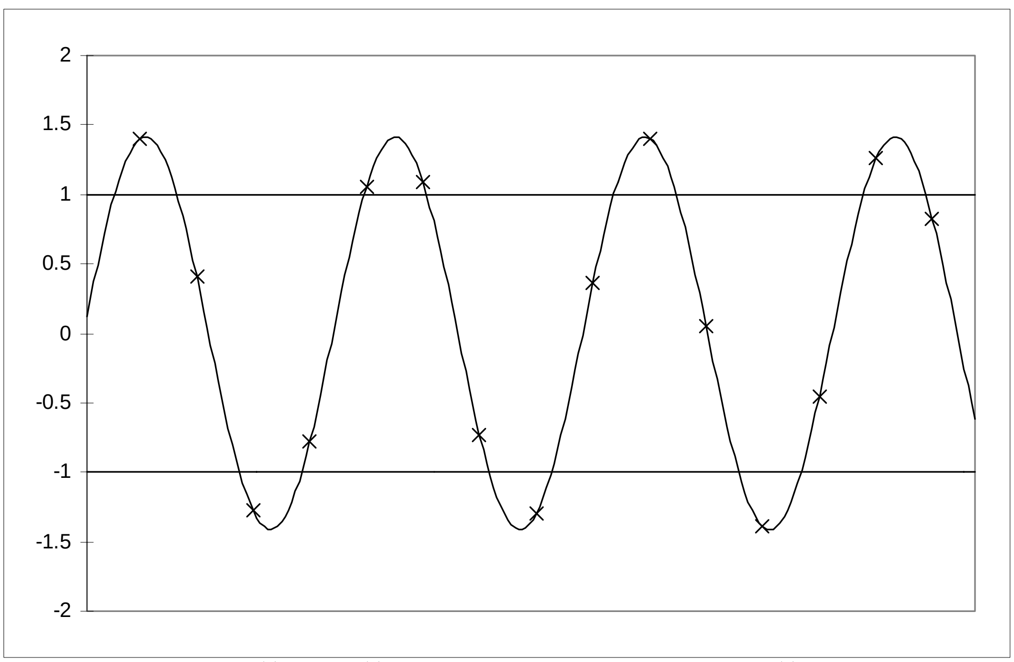

FIGURE 3. 10 kHz TONE AT +3 dBFS (1 AND -1 REPRESENT CLIP LEVELS)

Do "Over" Indicators Have a Place?

Over meters may be useful when creating very "hot" 16-bit masters. It is our contention that they should not be used for recording.

In spite of our best efforts to produce clean audio, there is always a demand for a CD that is as loud or louder than other CDs. The truth is, 16-bit masters that are created using "Over" meters may end up 3 dB hotter than masters that are created using FSD metering. What often happens, is that "Over" meters will allow clipping of the highest peaks (by about 3 dB). In most cases, this clipping is audible. But, the CD will sound louder and sometimes that is all that seems to matter.

A suggestion: Keep the original recording clean by using a FSD meter. Create a final 20 or 24-bit mix entirely with FSD meters. Then and only then, transfer to 16-bits (using an appropriate dither process), and adjust the levels using an "Over" meter. This way, if the clipping proves objectionable, you can still go back to the clean mix and repeat the transfer. Better yet, avoid the use of an "Over" meter entirely. Whenever you are being pressed to achieve maximum loudness, use one of the digital loudness processors.

Chapter 6

Design Philosophy

Our Goals and Design Methodology

The AD2402-96R was designed by John Siau, Allen H. Burdick, and Ralph Henry at Benchmark Media Systems, Inc. It is carefully engineered to reliably provide the highest possible audio transparency. We have not tried to add "warmth" or "color" to the audio. Instead, we have attempted to produce a piece of equipment that sounds as close to a piece of wire as possible.

Converters are often viewed as digital products, and are most often designed by digital hardware and software engineers without adequate attention to the analog sections. The importance of the analog circuitry is often overlooked, and the resulting defects easily go undetected. Unfortunately, given time, our ears usually detect these defects.

Our ears have a dynamic range of about 130 dB, and we can hear tones 20 to 30 dB below white noise. The presence of noise may allow audible defects to escape the scrutiny of many bench tests. We can hear multiple tones at various amplitudes simultaneously. Many audio measurements are only capable of measuring the tone having the highest amplitude. Again, bench tests can ignore critical information!

Because bench tests have often failed to detect audio defects, some have discounted their value and have relied solely on listening tests. We believe listening tests are a valuable tool for evaluating a finished product and for confirming the validity of a careful and thorough bench testing/development cycle. The key is; *we must employ test techniques that expose all audio defects*. Then these defects can be eliminated or reduced long before they could be detected through listening tests and/or field use.

Our experience shows that a careful and thorough bench testing/development program enables us to produce products that sound right the first time. It is important to note that specifications can be very misleading. Each spec by itself only paints a small picture of the total performance that can be expected. It is quite possible to select certain tests, test signals, frequencies, levels, and parameters that make a given product look good on a spec sheet. It is also our view that specifications should never be selected for marketing purposes. Product specifications should be comprehensive, and should include graphs and FFT plots. Bench testing must be viewed as a product development tool, and not as a marketing tool.

Appendix 1

Specifications

Sample rates =	44.1, 48, 88.2, 96 kHz, and variable
THD + N =	-107 dBFS (0.00033%) at -1 dBFS
Dynamic range =	117 dB, A weighted
Conversion Jitter =	9 Pico Sec., typical
Word length reduction =	7 settings for 16-bit output, 4 settings for 20-bit output
Outputs =	2 AES (XLR) and 2 S/PDIF (BNC) (Simultaneous 24-bit & 16-bit DAT outputs possible)
88.2 or 96 kHz =	single or dual-cable output
Analog inputs =	balanced XLR
Input level for 0dBFS =	+14 to +24 dBu - switch selectable
Metering =	9-segment, fully digital with "peak-hold"
Digital Input =	75-ohm loop-through (high Z)
D-to-D =	word length reduction/format conversion
D-to-D & Reference =	AES/EBU or S/PDIF format
Power =	+12 to +18 VDC, 4-pin XLR connector
Current drain =	800 mA max, Fuse = 5x20 mm, 2 A
Low Battery indicators =	11.5 V red on, 10.5 green off
Chassis =	8"W x 5"D x 1 3/4"H, weight 1.98 lb., 0.897 kg
R.F. Emissions =	complies with CE and FCC "B" requirements

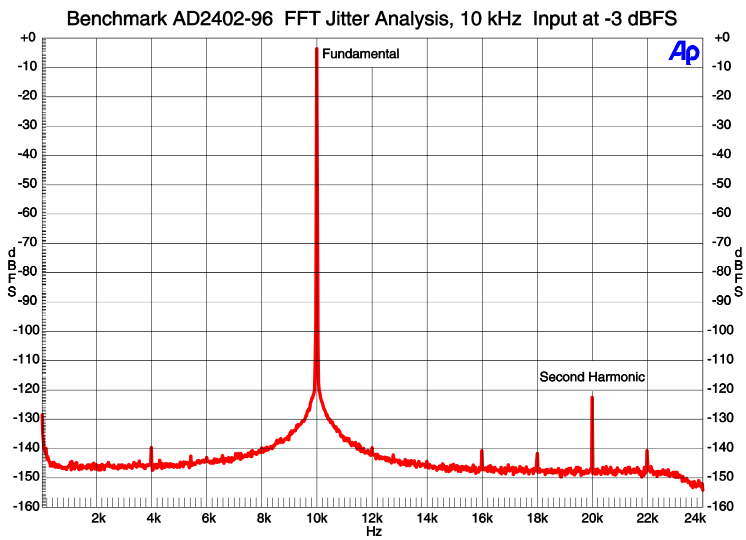

FIGURE 4. AD2402-96 JITTER ANALYSIS

Benchmark NN™ and NS™ Word Length Reduction System
Output Word Length = 16 Bits

Benchmark WLR Curve	Coefficient Table ID	Sample Rate (kHz)	Reference (16-BIT TPDF) Noise Power (dBFS)	Design Noise Floor (dBr)	Relative Unweighted Noise Power (dBr)	Relative F-Weighted Noise Power (dBr)	Unweighted Noise Power (dBFS)	F-Weighted Noise Power (dBFS)	Audibility when 0 dBFS= 105dBSPL (dB)	Audibility when 0 dBFS= 99dBSPL (dB)	Audibility when 0 dBFS= 97dBSPL (dB)
TPDF	TPDF	44.1	-93.3	NA	0.0	0.0	-93.3	-93.3	11.7	5.7	3.7
NN-1	U030718A	44.1	-93.3	NA	7.3	-3.1	-86.1	-96.5	8.5	2.5	0.5
NN-2	U051016A	44.1	-93.3	NA	10.1	-4.7	-83.3	-98.1	6.9	0.9	-1.1
NN-3	U071516A	44.1	-93.3	NA	15.9	-7.2	-77.4	-100.6	4.4	-1.6	-3.6
NS-1	F040406A	44.1	-93.3	-6	3.2	-8.2	-90.1	-101.5	3.5	-2.5	-4.5
NS-2	F111218A	44.1	-93.3	-18	12.2	-12.4	-81.1	-105.7	-0.7	-6.7	-8.7
NS-3	F141830A	44.1	-93.3	-30	18.7	-14.2	-74.6	-107.5	-2.5	-8.5	-10.5
TPDF	TPDF	48	-93.3	NA	0.0	0.0	-93.3	-93.3	11.7	5.7	3.7
NN-1	U061418B	48	-93.3	NA	13.8	-5.7	-79.5	-99.0	6.0	0.0	-2.0
NN-2	U060716B	48	-93.3	NA	7.1	-5.9	-86.2	-99.2	5.8	-0.2	-2.2
NN-3	U101916B	48	-93.3	NA	19.0	-9.8	-74.3	-103.1	1.9	-4.1	-6.1
NS-1	F040406B	48	-93.3	-6	3.1	-9.0	-90.3	-102.3	2.7	-3.3	-5.3
NS-2	F131218B	48	-93.3	-18	12.3	-14.5	-81.0	-107.8	-2.8	-8.8	-10.8
NS-3	F171930B	48	-93.3	-30	18.7	-16.8	-74.7	-110.1	-5.1	-11.1	-13.1
TPDF	TPDF	88.2	-93.3	NA	0.0	-3.0	-93.3	-96.3	8.7	2.7	0.7
NN-1	U080218C	88.2	-93.3	NA	2.0	-8.3	-91.3	-101.6	3.4	-2.6	-4.6
NN-2	U140618C	88.2	-93.3	NA	6.2	-14.0	-87.2	-107.3	-2.3	-8.3	-10.3
NN-3	U221318C	88.2	-93.3	NA	12.5	-21.6	-80.8	-114.9	-9.9	-15.9	-17.9
NS-1	F080306C	88.2	-93.3	-6	2.0	-14.3	-91.3	-107.6	-2.6	-8.6	-10.6
NS-2	F190718C	88.2	-93.3	-18	7.4	-24.1	-85.9	-117.4	-12.4	-18.4	-20.4
NS-3	F281230C	88.2	-93.3	-30	11.5	-29.8	-81.8	-123.1	-18.1	-24.1	-26.1
TPDF	TPDF	96	-93.3	NA	0.0	-3.0	-93.3	-96.3	8.7	2.7	0.7
NN-1	U100218D	96	-93.3	NA	1.9	-9.8	-91.4	-103.1	1.9	-4.1	-6.1
NN-2	U150518D	96	-93.3	NA	5.2	-14.5	-88.1	-107.8	-2.8	-8.8	-10.8
NN-3	U221018D	96	-93.3	NA	10.6	-22.3	-82.8	-115.6	-10.6	-16.6	-18.6
NS-1	F080306D	96	-93.3	-6	2.0	-15.1	-91.3	-108.4	-3.4	-9.4	-11.4
NS-2	F200718D	96	-93.3	-18	6.8	-25.1	-86.5	-118.4	-13.4	-19.4	-21.4
NS-3	F291130D	96	-93.3	-30	10.8	-31.3	-82.5	-124.6	-19.6	-25.6	-27.6

Audibility:					
	Audible after 1 pass	Inaudible after 1 pass	Inaudible after 2 passes	Inaudible after 4 passes	Inaudible after 8 passes

16-BIT NOISE BUILDUP VS NN™ AND NS™ VS NUMBER OF PASSES

Benchmark NN™ and NS™ Word Length Reduction System
Output Word Length = 20 Bits

Benchmark WLR Curve	Coefficient Table ID	Sample Rate (kHz)	Reference (20-BIT TPDF) Noise Power (dBFS)	Design Noise Floor (dBr)	Relative Unweighted Noise Power (dBr)	Relative F-Weighted Noise Power (dBr)	Unweighted Noise Power (dBFS)	F-Weighted Noise Power (dBFS)	Audibility when 0 dBFS = 105 dB SPL (dB)	Audibility when 0 dBFS = 99 dB SPL (dB)	Audibility when 0 dBFS = 97 dB SPL (dB)
TPDF	TPDF	44.1	-117.4	NA	0.0	0.0	-117.4	-117.4	-12.4	-18.4	-20.4
NN-1	U030718A	44.1	-117.4	NA	7.3	-3.1	-110.1	-120.5	-15.5	-21.5	-23.5
NN-2	U051016A	44.1	-117.4	NA	10.1	-4.7	-107.3	-122.1	-17.1	-23.1	-25.1
NN-3	U071516A	44.1	-117.4	NA	15.9	-7.2	-101.5	-124.6	-19.6	-25.6	-27.6
NS-1	F040406A	44.1	-117.4	-6	3.2	-8.2	-114.2	-125.6	-20.6	-26.6	-28.6
NS-2	F111218A	44.1	-117.4	-18	12.2	-12.4	-105.2	-129.8	-24.8	-30.8	-32.8
NS-3	F141830A	44.1	-117.4	-30	18.7	-14.2	-98.7	-131.6	-26.6	-32.6	-34.6
TPDF	TPDF	48	-117.4	NA	0.0	0.0	-117.4	-117.4	-12.4	-18.4	-20.4
NN-1	U061418B	48	-117.4	NA	13.8	-5.7	-103.6	-123.1	-18.1	-24.1	-26.1
NN-2	U060716B	48	-117.4	NA	7.1	-5.9	-110.3	-123.3	-18.3	-24.3	-26.3
NN-3	U101916B	48	-117.4	NA	19.0	-9.8	-98.4	-127.2	-22.2	-28.2	-30.2
NS-1	F040406B	48	-117.4	-6	3.1	-9.0	-114.3	-126.4	-21.4	-27.4	-29.4
NS-2	F131218B	48	-117.4	-18	12.3	-14.5	-105.1	-131.9	-26.9	-32.9	-34.9
NS-3	F171930B	48	-117.4	-30	18.7	-16.8	-98.8	-134.2	-29.2	-35.2	-37.2
TPDF	TPDF	88.2	-117.4	NA	0.0	-3.0	-117.4	-120.4	-15.4	-21.4	-23.4
NN-1	U080218C	88.2	-117.4	NA	2.0	-8.3	-115.4	-125.7	-20.7	-26.7	-28.7
NN-2	U140618C	88.2	-117.4	NA	6.2	-14.0	-111.3	-131.4	-26.4	-32.4	-34.4
NN-3	U221318C	88.2	-117.4	NA	12.5	-21.6	-104.9	-139.0	-34.0	-40.0	-42.0
NS-1	F080306C	88.2	-117.4	-6	2.0	-14.3	-115.4	-131.7	-26.7	-32.7	-34.7
NS-2	F190718C	88.2	-117.4	-18	7.4	-24.1	-110.0	-141.5	-36.5	-42.5	-44.5
NS-3	F281230C	88.2	-117.4	-30	11.5	-29.8	-105.9	-147.2	-42.2	-48.2	-50.2
TPDF	TPDF	96	-117.4	NA	0.0	-3.0	-117.4	-120.4	-15.4	-21.4	-23.4
NN-1	U100218D	96	-117.4	NA	1.9	-9.8	-115.5	-127.2	-22.2	-28.2	-30.2
NN-2	U150518D	96	-117.4	NA	5.2	-14.5	-112.2	-131.9	-26.9	-32.9	-34.9
NN-3	U221018D	96	-117.4	NA	10.6	-22.3	-106.8	-139.7	-34.7	-40.7	-42.7
NS-1	F080306D	96	-117.4	-6	2.0	-15.1	-115.4	-132.5	-27.5	-33.5	-35.5
NS-2	F200718D	96	-117.4	-18	6.8	-25.1	-110.6	-142.5	-37.5	-43.5	-45.5
NS-3	F291130D	96	-117.4	-30	10.8	-31.3	-106.6	-148.7	-43.7	-49.7	-51.7

Audibility:					
	Audible after 1 pass	Inaudible after 8 passes	Inaudible after 16 passes	Inaudible after 32 passes	Inaudible after 64 passes

20-BIT NOISE BUILDUP VS NN™ AND NS™ CURVES VS NUMBER OF PASSES