Revision 1

BENCHMARK MEDIA SYSTEMS, INC.

AD2004 - QUAD ANALOG TO DIGITAL CONVERTER

Operating Manual

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Operating Manual

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1

Quick Start Guide

For those of us who hate to read manuals first.

This guide will take you thorough a typical setup of the AD2004R and will direct you to other sections of this manual if your installation has special requirements.

Quick System Overview

The AD2004R is a four-channel 20-bit analog-to-digital converter specifically designed for applications requiring four or more perfectly phased A to D conversion channels. It has four balanced audio inputs on a 15-pin D-Sub connector. A 26-pin high density d-sub connector provides two digital audio outputs, a digital audio reference input, a loop through for the reference input, and a word clock output. All digital audio signals are available in both 75 ohm 1 Vpp AES3-id format and 110 ohm balanced AES/EBU format. Outputs are professional format, inputs can be either professional format or SPDIF consumer format. The digital audio reference input allows the AD2004 to be locked to an external digital audio reference signal.

How to Connect the Audio Interfaces

Analog Audio Inputs must be balanced. Factory preset input reference level is +4 dBu. Headroom is 20 dB. In other words, 0 dBFS (the full scale digital clip point of the converters) will be reached when the input levels reach +24 dBu. Internal jumpers can be changed to select other 0 dBFS clip points ranging from +16.5 dBu to +28 dBu (see page ?????). Pre-wired 110 ohm XLR cable sets are supplied as standard equipment with the AD2004R. If you have wish to make your own cables, or make use of the 75 ohm interfaces, see page ???? for a complete schematic.

Digital Audio Interfaces comply with AES/EBU and AES3-id. The 110 ohm interface is by far the most popular interconnect for professional audio gear. However, the 75 ohm AES3-id interface is rapidly gaining popularity in application which require long transmission distances. The AES3-id interface utilizes 75 ohm coax and can easily achieve transmission distances of 1000 feet without cable EQ, and 3000 feet with cable EQ.

The Digital Audio Reference Input accepts either professional or consumer status bit formats. Outputs are always professional format. The standard digital audio cable supplied with the converter has an XLR female for reference input, and an XLR male for reference loop through. Note that the reference input is not terminated internally. The "Input" and "Loop" connectors are hardwired together so that the reference signal can be looped through several converters. However, if the reference signal is not being looped to another device, it is necessary to add the 110 ohm XLR terminator supplied with the cable set. Failure to add the terminator may cause receive errors on the reference signal if cable lengths approach 1000 feet. **If you are using a digital audio reference to phase lock a single AD2004 in a slave mode, connect this reference to the "Input" connector, and connect the 110 ohm XLR terminator (supplied with every AD2004R) to the "Loop" connector. If you are not using a digital audio reference, store the XLR terminator on the "Loop" connector.** If you are installing multiple AD2004R converters, you can daisy chain a single digital audio reference through multiple AD2004R devices. Be sure to connect the 110 ohm terminator to the "Loop" connector of the last AD2004R on the daisy chain. For more information see page ????.

How to Set the Front Panel Switches

The "SYNC" Switch is a three-position toggle switch, which is located on the left side of the front panel. It is recessed to prevent accidental switching during a recording session or while on air. This switch selects the sample clock frequency (on AD2004R dual frequency models), and selects the PLL (Phase Locked Loop) mode (on all AD2004R models). **If you have a single frequency AD2004R set the sync switch all the way up. If you have a dual frequency AD2004R, set the sync switch up for 48 kHz, or in the center for 44.1 kHz.** In both cases, these switch settings will set the free running frequency of the converter, and will allow automatic PLL operation if a reference is applied to the digital audio input. For more information see page ????.

The "METER" Switch is a three-position toggle switch, which is located to the right of the sync switch. It selects one of three digital meter functions and may be switched at any time. The down position sets the meter scale to 6 dB steps. The center position sets the meter scale to 1 dB steps. The top position sets the meter scale to 1 dB steps and enables a peak hold function. Peak hold may be cleared by moving the switch to the center position. Note that the bottom of the 1 dB scale is expanded and includes a -20 dBFS LED which can be used to set the input level relative 0 dB house reference. **Start with the meter switch in the down position, as this will make it easier to verify that signals are present at the analog inputs.**

Understanding the Status LEDs

The "PWR" LED is located on the front panel between the "Sync" and "Meter" switches. It is driven from the +5 V digital supply and indicates that the unit is receiving power. To avoid accidental shut down, there is no power switch on the AD2004. The converter runs very cool and may be left on continuously, even in densely populated equipment racks.

The "LOCK" LED is located on the front panel above the "PWR" LED. This light will turn on whenever the AD2004R is locked to a reliable digital audio reference of the correct frequency. If the light is off, the AD2004R is operating as a master sync generator using an internal crystal reference. For more information see page ???

Note - single frequency versions only

When the "SYNC" switch is in the center ("INT") position, the AD2004R ignores any signal which may be present on the digital audio sync input, and the "LOCK" LED will remain off.

Power Supply Connection

Connect the power supply to the nine-pin D-sub connector on the rear panel, and apply power. The "PWR" LED will light, and the "LOCK" LED will flash 8 times indicating that automatic calibration is in progress. For power supply requirements, specifications, options, and pin assignments, see page ????

Chapter

2

Design Philosophy

Our Goals and Design Methodology

The AD2004R was designed by John Siau and Allen H. Burdick at Benchmark Media Systems, Inc. It is carefully engineered to reliably provide the highest possible audio transparency under both ideal and adverse operating conditions. One of our goals was to create a simple and reliable 20-bit converter with the highest performance available, while at the same time, achieving one of the lowest per-channel costs in the industry. We chose to control costs by limiting the number of features rather than limiting the performance. In addition we combined four channels into one package in order to reduce the per-channel system cost. We reduced the parts count, board size, and power consumption by creating a custom FPGA to control the digital side of the system.

Design Goals

- Highest level performance of any 20-bit design
- Low per-channel cost when compared to existing products
- Consistent and repeatable performance
- Low jitter even when locked to a high jitter reference signal
- No performance degradation when phase locked
- High density package
- Useful level meters
- Sample, frame, and block accurate phase locking of any number of channels
- Low cost, easy implementation of multi-channel systems

- RF and static discharge immunity

At Benchmark Media Systems:

> Reliability is: The ability to achieve and consistently maintain specified performance under a wide range of adverse operating conditions.

True performance is only achieved if "real world" results consistently match tests conducted in a controlled laboratory environment.

Design Methodology

- Computer aided circuit analysis
- RF design techniques
- Proper grounding techniques
- Extensive shielding and isolation of digital signals
- Extensive power supply isolation
- Extensive testing on AP System Two
- Simulation of adverse operating conditions
- Low level, and high level listening tests
- ESD and RF immunity testing
- FCC emissions testing

Chapter

3

System Overview

An Inside View of the AD2004R

The AD2004R has five major function blocks. In combination, these blocks allow the AD2004R to set new performance benchmarks. These outstanding benchmarks include; low-noise, low-distortion, low-jitter, and high-jitter immunity.

AD2004R Function Blocks:

- Analog Input Stage
- CS5390 Converters
- AES Input / Clock Recovery
- AES Outputs
- Digital System Control
- Digital Level Meters

The first four blocks are in the audio "critical path". As such, these blocks are designed to achieve maximum audio transparency. The fifth block controls the meters, the PLL, the digital I/O, and the user interface. While this fifth block is not in the audio "critical path" is certainly could cause interference with the "critical path" if it were inadequately isolated from the audio. Finally, the digital level meters are designed to efficiently convey useful signal level information.

Analog Input Stage

The analog input stage is designed to provide the ultimate in transparent and uncolored audio. To this end, the input stage is designed to provide frequency response which is flat to better than +/- 0.01 dB from 10 to 20 kHz. Frequency response extends well beyond 200 kHz. Phase non-linearity is less than ???? degrees from 20 Hz to 20 kHz.

A proprietary analog preprocessing circuit reduces the odd harmonic distortion that is typical of most analog-to-digital converters. The result is that the THD+N of the AD2004 is limited by white noise rather than by distortion products. A second and perhaps more important achievement is that IMD is reduced. This is important since IMD products are not harmonically related to the fundamental, and are therefor much more objectionable than harmonic distortion.

With the proliferation of digital audio equipment, computers, and wireless technology, our studios and recording venues are rapidly becoming high RF environments. For a number of years, Benchmark has been building Microphone Preamplifiers and Audio Distribution Amplifiers for use in broadcast facilities. Many of these facilities are located at or near transmission sites and must perform flawlessly in high RF fields. The AD2004 brings this technology to digital audio equipment. To prevent RF interference, the AD2004 has two stages of passive RF filtering prior to the active section of the input amplifier. The active section itself is wide-band and RF-stable. The result is freedom from the “unexplainable” grunge and distortion that can result when RF interference signals cause an input stage to oscillate or clip.

CS5390 Converters

AES Input / Clock Recovery

AES Outputs

The standard cable sets supplied with the AD2004R provide access to the 110 ohm balanced AES/EBU outputs. The 110 ohm interfaces are isolated with high quality shielded transformers. Transient voltage protection is provided by Schottkey diodes. The outputs will withstand direct hits from 8000 volt static discharge, accidental connection to phantom power, and short circuits. The AES/EBU outputs are best suited for transmission distances of less than 1000 feet. Specially designed 110 ohm digital audio cable should be used. It is absolutely essential to use properly terminated digital audio cable when transmission distances approach 1000 feet.

Two additional 1 Vpp, 75 ohm unbalanced, coaxial, AES3-id digital outputs are provided at the 26 pin D-sub connector. These outputs support long transmission distances; 1000 feet without cable EQ, and over 3000 feet with cable EQ. These 75 ohm digital outputs are rise time limited (per AES3-id-1995) to allow the distribution of digital audio using non-clamping NTSC or CCIR video distribution amplifiers. In addition, the outputs are designed with accurate 75 ohm source impedance which extends above the bandwidth of the output. This unique Benchmark design eliminates the possibility of high frequency standing waves in the coax cable. The are protected from transients using Schottkey diodes and the output filter itself.

The 110 ohm and 75 ohm outputs are available simultaneously and are isolated from each other. Thus 2 outputs are available for each channel pair. The 75 ohm output could be used to drive a long cable while the 100 ohm output could be used for local monitoring. Both are driven from the same active electronics, and therefore, the local monitor provides full confidence that the remote feed is active.

Digital System Control

Digital Level Meters

Chapter 4

Jumper Settings

Setting Analog Input Level Preset Jumpers.

The AD2004R has jumpers for selecting preset input levels, and for enabling the front panel level controls. Presets allow setting the 0 dBFS clip point at +28 dBu, +24 dBu, and +22 dBu. Use the preset levels whenever possible. The preset levels are matched to an accuracy of better than +/- 0.3% and are not subject to the possibility of potentiometer noise.

Table 1

0 dBFS	REFERENCE LEVEL (20 dB headroom)	A	B	C	Cable from Front Panel Pots
+16.5 dBu	-3.5 dBu	OFF	OFF	OFF	P#0 **
+22 dBu	+2 dBu	ON	OFF	OFF	P#0 **
+24 dBu	+4 dBu	OFF	ON	OFF	P#0 **
+28 dBu	+8 dBu	OFF	OFF	ON	P#0 **
Variable (+18 dBu to +28 dBu)	Variable (-2 dBu to +8 dBu)	OFF	OFF	ON	P#1 **

** # = channel number

Chapter

5

Using the Meters

9 Segment LED Meters.

The AD2004R has a nine segment LED meter for each of the four audio channels. 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 having a duration of only one sample can be displayed and measured accurately. The "FSD" clip indicator is accurate to one quantization level.

Meter Time Constants

The meters on the AD2004R 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 AD2004R 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. The result is that any event having a duration of only one audio sample is easily 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.

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 AD2004 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 the 1 dB scale are expanded 4 dB and 10 dB steps. ranges have expanded step sizes at the bottom end of the scale. More specifically, the 6 dB scale is expanded to the scale is expanded at the low end. 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.

What is “FSD”?

“FSD” stands for “Full Scale Digital” 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.

More specifically, AES3 and SPDIF digital audio transmission systems use twos-complement notation so that both positive and negative voltages can be represented.

In 20-bit twos complement hexadecimal MSB first notation, codes 7FFFF and 80000 will 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 were developed based upon two assumptions:

1) A digital clip cannot be heard if it has a duration of three or less consecutive samples.

2) Because a signal may reach the maximum or minimum digital code for one or two samples without exceeding it, and because it is important to preserve all available quantization levels (digital codes), it is necessary to assume that such an event does not necessarily indicate a clip.

While assumption 1 may be true for an isolated non-repetitive event, it does not hold true in the real world where we are recording music. We are not are not usually recording random noise but instead are recording complex combinations of musical tones. Individually, each musical tone is a highly repetitive waveform. When combined, these musical tones form a waveform that is at least somewhat repetitive and often very repetitive. In such a case, it is possible, and quite probable, that a clip which has a duration of one or two samples will be repeated a number of times. Such a clip may never produce three consecutive full-scale codes, but is often very audible.

The fallacy of assumption 2 is that there is virtually nothing to be gained by preserving 2 quantization levels, but much to be lost by reducing our ability to detect clipping. The "advantage" of "Over" detection is that the minimum and maximum digital codes can be used to carry the audio signal without generating a clip indication. Without "Over" detection, these two digital codes essentially become illegal because they will generate a clip indication whenever they are used. But how much of an advantage is there in preserving these two codes? In a 20-bit system, we have 1,048,576 unique digital codes (or quantization levels) to work with. If we reduce this by two, we still have 1,048,574 codes to work with! In other words, we reduce our headroom by only 0.000016 dB! Even in a 16-bit system, the headroom reduction is still only 0.00027 dB! Furthermore, if we consider the proability of reaching but not exceeding the minimum or maximum code, we discover that it is very unlikely that a clip has not occurred. Remember that the maximum code represents an infinite number of quantization levels above clip, and only one level below clip. Therefor, it makes far more sense to assume a clip has occurred whenever a minimum or maximum code occurs.

In summary, on "Over" meter can allow an additional 0.000016 dB of headroom before clip indication, but we lose our ability to detect every audible clip. In fact, severe and highly audible clipping can occur without ever generating three consecutive full-scale codes.

Still not a believer? Try this simple test: 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, clipping will cause severe harmonics at 20 kHz, distortion will exceed 14%, it will sound horrendous, but an "Over" (3 consecutive full scale codes) never occurs. **This test illustrates that "Over" indicators may ignore audible clipping!**

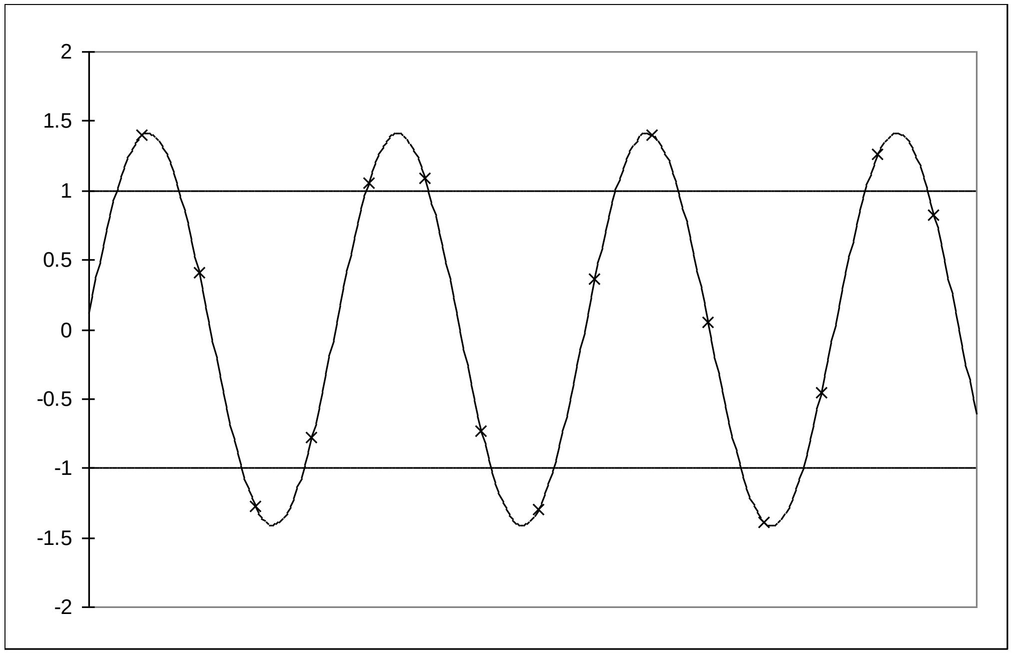

Figure 1 10 kHz Tone at +3 dBFS (1 and -1 represent clip levels)

Do “Over” Indicators Have a Place?

Perhaps

In spite of our best efforts to produce clean audio, there is always a demand for a CD that is louder. 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. But didn’t we just say that using an “Over” meter increases the headroom by only 0.00027 dB? Yes but, what really happens, is that the new recording often has 3 dB clipped off of the highest peaks. And yes this clipping can probably be heard. And yes the CD will sound louder. But sometimes that is all that seems to matter.

A suggestion: Keep the 20-bit originals 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 not being pressed to achieve maximum loudness.

Chapter 6

Jitter

What is it, and what does it do to my audio?

Jitter is often misunderstood, and can be difficult to measure. In certain circumstances, jitter can be benign, but in others, jitter may cause sonic artifacts. These artifacts may be far more serious than most people realize. It is extremely important to understand where and when jitter is a problem, how to measure it, and what can be done to prevent it.

What is Jitter?

Jitter is time-base error. More specifically it is a measure of how early or late a digital transition occurs. These digital transitions may be the rising and falling edges of system clocks or of digital audio signals. Jitter in conversion devices such as; Analog to Digital Converters (ADCs), Digital to Analog Converters (DACs), or Asynchronous Sample Rate Converters (ASRCs), will cause phase modulation of the audio signal. Jitter between two fully digital devices will not usually cause phase modulation but may cause bit errors if the jitter levels are unusually large. However, jitter at these digital-to-digital interfaces can often pass through a system to a conversion device were jitter can cause phase modulation.

What Causes Jitter?

Jitter can be caused by poor circuit design, bandwidth and noise limitations of digital transmission systems, electromechanical variations in record and playback devices, and even the physical spacing of the optical pits on a CD. The bandwidth limitations of the AES/EBU transmission system guarantee that jitter on the interface will exceed levels that are acceptable for 20-bit data conversion. This does not mean that the AES/EBU interface should be abandoned, it simply means that a jitter free clock must be recovered from the interface before the clock is sent a conversion device. Low-jitter clock recovery can be achieved using proper Phase Locked Loop (PLL) design techniques.

What is a PLL?

A PLL or Phase-Locked-Loop is the electrical equivalent of a mechanical flywheel.

Even if we were to use high bandwidth distribution of a 256X (11.2896 MHz) digital clock, it would be difficult to achieve jitter levels low enough for 20-bit data conversion without using a well designed PLL. However, very few PLL designs have achieved RMS jitter levels in the sub 100 psec range. To make matters worse, many ADC and DAC devices cannot achieve the necessary performance when operating in a master clock mode.

Is There a Cure?

Yes. The AD2004 has a very unique multi-stage PLL which sets new benchmarks for low jitter clock recovery. When locking to an AES/EBU reference having jitter as high as 5 nsec, the internal PLL will produce a phase locked clock having jitter of 12 to 16 psec RMS. With a 10 kHz , -1 dBFS test tone, the sum total of all jitter induced sideband energy in the digital audio output of the AD2004 is -123 dBFS or lower when locked to a moderately jittery AES/EBU reference. This is near the theoretical limit of non-dithered 20 bit audio. The AD2004 is a rare example of such a device. It has the ability to reduce jitter by 50 dB (a ratio of 316 : 1). It can lock to any signal having jitter as high as 5 nsec RMS (5 billionths of a second) and still produce a clock which has jitter below 16 psec (16 trillionths of a second) at the converter. However, as we shall see, it is only necessary to

Are there Different Types of Jitter?

Do "Jitter Killers" Really Work?

Can Compact Disks Contain Jitter?

When is Jitter Low Enough?

At first glance, it seems obvious that low jitter amplitude is important. However, as we look closer, we discover that the frequency (or spectral distribution) of the jitter is at least as important as amplitude. Next, as we look throughout the entire digital recording and playback system, we discover that certain parts of the digital chain are extremely sensitive to jitter while others appear to be nearly immune to jitter. But, upon further investigation, we start to discover that devices can interact with each other and cause "strange phenomena". These "phenomena" are really nothing more than design defects which cause sonic artifacts in the presence of jitter. More on this latter. First lets step out of the world of ones and zeros consider something which we can put our hands on:

The Tape Machine - A Jitter Analogy:

Jitter is the digital equivalent of wow and flutter. An analog to digital converter (ADC) is the digital equivalent of an analog tape machine in record mode. A digital to analog converter (DAC) is the digital equivalent of an analog tape machine in playback mode.

Analog and digital systems are both subject to time base errors: In analog recording, temporary tape speed variations will cause sections of a tape to pass a head a little too early or a little to late. In a digital ADC or DAC, jitter can cause a sample clock pulse to occur a little too early or a little too late. In either case, the results are the same: Audio signals will be temporarily shifted to slightly higher or lower frequencies until servos can make corrections. In analog recording, a large flywheel is attached to the capstan to reduce rapid fluctuations (flutter), and the capstan motor speed is controlled by a carefully designed servo which reduces slower variations (wow). Typically, the servo will compare the speed of the capstan to a crystal reference in order to accurately control the average speed of the capstan. If a second tape machine is slaved to the first machine, the servo in the second machine may use a time code pulse as a speed reference. Similarly, in high quality digital systems, a clock signal is either supplied directly from a crystal oscillator (master mode), or by phase locking a voltage controlled crystal oscillator (VCXO PLL) to an external clock reference (slave mode). In both analog and digital systems, time-base errors which occur during recording will create permanent frequency fluctuations in the audio signal. Time-base errors which occur during playback will have the same sonic effect as time-base errors which occur during record. The sonic artifacts caused by time-base errors in record and playback modes are additive. However, there is one very important difference. Playback time-base errors can be eliminated by fixing the playback device. Record time-base errors can only be fixed by repeating the recording session! This difference underscores the necessity of having low jitter in the ADC.

Now, lets extend the tape machine analogy to address interface jitter. he AD2004R has jumpers for selecting preset input levels, and for enabling the front panel level controls. Use the preset levels whenever possible. The preset levels are matched to an accuracy better than +/-

Chapter

7

Signal Interconnect

Detailed interface information..

The AD2004R has a 10-pin pluggable barrier strip for the analog audio inputs, a 6-pin Molex connector for the power supply input, and four BNC connectors for the digital audio interfaces.

Analog Audio Inputs

Analog Audio Inputs must be balanced. Factory preset input reference level is +4 dBu. Headroom is 20 dB. In other words, 0 dBFS (the full scale digital clip point of the converters) will be reached when the input levels reach +24 dBu. Internal jumpers can be changed to select other 0 dBFS clip points ranging from +16.5 dBu to +28 dBu (see Table 1 on page 12). Barrier strip connections are marked on the rear panel. Pre-wired XLR cable sets are available for the AD2004R.

Connector pin assignments:

1) **Channel 1 Negative**

2) **Channel 1 Positive**

3) **Chassis Ground**

4) **Channel 2 Negative**

5) **Channel 2 Positive**

6) **Channel 3 Negative**

7) **Channel 3 Positive**

8) **Chassis Ground**

9) **Channel 4 Negative**

10) Channel 4 Positive

Digital Audio Interfaces comply with AES3-id. They are 75 ohm, 1 Vpp, professional format coaxial interfaces on BNC connectors which provide outstanding performance when long transmission distances are required. If your other equipment uses a different interface, see page ????? for special wiring instructions.

The Digital Audio Reference Input accepts either AES3-id or consumer SPDIF formats. Two BNC connectors are provided ("Input" and "Loop"). The two BNC connectors are wired in parallel and are not terminated internally. **If you are using a digital audio reference to phase lock a single AD2004 in a slave mode, connect this reference to the "Input" connector, and connect a 75 ohm BNC terminator (supplied with every AD2004R) to the "Loop" connector. If you are not using a digital audio reference, store the BNC terminator on the "Loop" connector.** If you are installing multiple AD2004R converters, you can daisy chain a single digital audio reference through multiple AD2004R devices. Be sure to connect a 75 ohm terminator to the "Loop" connector of the last AD2004R on the daisy chain. For more information see page ????.

Chapter 8

Jumpers

Setting Analog Input Level Preset Jumpers.

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Chapter 9

Jumpers

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