Chapter 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 AD2408-96 and will direct you to other sections of this manual if your installation has special requirements.

Quick System Overview

The AD2408-96 is an eight-channel 24-bit analog-to-digital converter specifically designed for applications requiring eight or more perfectly phased, low jitter, low distortion, A to D conversion channels. The AD2408-96 has two fully independent four-channel converter modules that share a common digital-audio reference input. The AD2404-96 is a four-channel version of the AD2408-96, and uses only one of these converter modules. This manual applies to both products.

The converter operates over a sampling frequency range of 24 to 100 kHz, and provides 24, 20, and 16-bit word lengths. The 20 and 16-bit word lengths are re-dithered from 24-bits using selectable noise-shaping curves or TPDF dither. All noise-shaping curves are optimized for the selected sample rate, and for the number of bits being removed.

A digital-audio reference input jack allows the AD2408 to be locked to an external digital-audio reference signal. This jack also serves as the input for Digital-to-Digital processing.

Digital-to-Digital processing functions include Word Length Reduction (re-dithering), Sample Rate Conversion, and, conversion between "Single Cable" and "Dual Cable" interface modes.

All digital inputs and outputs support sample rates from 28 kHz to 100 kHz using the "Single Cable" interface mode (also know as "AES3 Multichannel Mode"). In this mode, a channel pair is transmitted on a single cable, at a frame rate equal to the sampling rate.

A "Dual Cable" interface mode (also known as "AES3 Single Channel Double Sampling Frequency Mode") allows 88.2 and 96 kHz recording on 44.1 and 48 kHz equipment. In this mode, two successive samples of a single high sample-rate channel are transmitted in place of a pair of low sample-rate channels. Two cables are required for two channels.

Each conversion channel has a fully digital multi-function 9-segment LED meter.

Power Supply Connection

The AD2404 is available with either an internal international power supply, or a pair of 9-pin D-sub connectors for connection to an external supply.

Internal Supply: Check voltage selector switch for proper AC input voltage. If voltage setting needs to be changed, remove power cord, open the access door, and then remove voltage selector cam. Do not rotate cam while it is inserted in the housing, this may cause damage to the voltage selector switch! Reinsert the cam with the proper voltage selection facing outward. Close the access door, insert the appropriate IEC terminated power cord, and then apply power.

External Supply: Connect the converter to the external DC supply using either nine-pin D-sub connector on the rear panel, then apply power. A second D-sub connector is provided for daisy chain wiring. For power supply requirements, specifications, options, and pin assignments, see page ????

After applying power, the "POWER" LED will light, and the "LOCK" LED will flash up to eight times indicating that automatic calibration is in progress.

Understanding the Status LEDs

The "POWER" LED is located at the far left of the front panel. It is driven from the +5 V digital supply and indicates that the 4-channel converter module is receiving power. To avoid accidental shut down, there is no power switch on the AD2404. The converter is designed for continuous duty.

The "PHASE LOCK (Slave)" LED is located on the front panel above the "POWER" LED. This light will turn on whenever the AD2404 is locked to a reliable digital audio reference of the correct frequency. If the light is off, the AD2404 is operating as a master sync generator using an internal crystal reference. A flashing light indicates an error condition. For more information see page ???

How to Connect the Audio Interfaces

Analog Audio Inputs may be balanced or unbalanced. 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. Make sure your signal source can achieve levels of at least 24 dBu without clipping. An optional "Variable Gain" motherboard is available. Converters equipped with this option have jumpers and gain trim pots which can be changed to select 0 dBFS clip points ranging from +4 dBu to +24 dBu (see page ?????).

Digital Audio Interfaces are available as either an XLR equipped AES3 ("R" option) or as a BNC equipped SMPTE 276M ("B" option). The "R" option provides one 110-ohm AES3 digital audio input, and four 110-ohm AES3 digital audio outputs. The "B" option provides one 75-ohm SMPTE 276M digital audio input, and eight 75-ohm SMPTE 276M digital audio outputs. All outputs use professional status bit formats.

> Note: All four-channel converters and all "B option" converters have two sets of digital outputs. These additional outputs provide some additional features. In most modes of operation, the primary outputs have adjustable word lengths, while the auxiliary outputs have fixed 24-bit word lengths. This allows simultaneous 24 and 16 bit recordings from the same converter. Also, if the "Dual Cable" modes are used, there are enough digital outputs to provide 8 channels of conversion. "Dual Cable" operation on eight-channel "R" option converters is supported but is limited to 4 channels. (see page ?????)

The 110-ohm AES3 interface is by far the most popular interconnect for professional digital audio equipment. However, the 75-ohm SMPTE 276M interface is rapidly gaining popularity in applications that require long transmission distances. The SMPTE 276M interface utilizes 75-ohm coax and can easily achieve transmission distances of 1000 feet without cable EQ, and 3000 feet with cable EQ. AES3 and SMPTE 276M use identical data formats. Note that SMPTE 276M is a formal standard that references AES3-id (an earlier information document issued by the AES). Consequently, SMPTE 276M interfaces are often called AES3-id. SMPTE 276M is slightly more specific than AES3-id, but the two interfaces are fully inter-operable. Use 110-ohm digital audio cable for AES3 connections, or 75-ohm coax for SMPTE 276M connections. Incorrect cable impedances may increase jitter, may cause data loss, and will reduce the maximum transmission distances. Standard analog audio cable should be avoided.

The Digital Audio Reference Input accepts either professional or consumer status bit formats. The PLL automatically supports 1:1, 2:1 and 1:2 frequency ratios when a fixed sample rate is selected. The PLL supports 1:1 or 2:1 frequency ratios when variable sample rates are selected. On "R" option converters, the reference input is a female XLR connector terminated with 110 ohms. On "B" option converters, the reference input is a BNC connector without an internal termination. This high impedance BNC input allows looping of a single reference input to multiple converter frames. The BNC reference input on "B" option converters must be externally terminated using the supplied BNC "T" connector and 75-ohm BNC terminator. (see page ?????)

A reference input is required when:

- The converter will be operating in a variable speed mode ("VAR 1:1", or "VAR 2:1").
- More than four channels must be phase locked together.
- The converter is to be locked to an external studio reference.
- Any "Digital-to-Digital" 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 AD2408 converter at any of the fixed sample rates. The jitter performance of the fixed sample rate modes is maintained even when the reference has very high levels of jitter. However, the "VAR 1:1" and "VAR 2:1" variable speed modes have wide frequency ranges that preclude the use of the final VCXO stage of the AD2408 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.

How to Set the Front Panel Switches

The "SAMPLE RATE" Switch is a twelve-position rotary switch, which is located on the left side of the front panel. 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 one of the fixed "Single Cable" sample rates ("44.1", "48", "88.2", or "96"). This will set the free running frequency of the converter, and will allow automatic PLL operation if a reference of the proper frequency is applied to the digital audio input. When a fixed sample rate is selected, the PLL automatically supports lock ratios of 1:1, 2:1, and 1:2. For example, 96kHz conversion can be locked to either a 48 or 96 kHz AES digital audio reference. Use the "VAR 1:1" or "VAR 2:1" for non-standard sampling rates. "VAR 1:1" locks in a 1:1 ratio to the reference sample rate. "VAR 2:1" locks in a 2:1 ratio to the reference sample rate. For more information see page ????.

> Note: Converters will enter a calibration mode whenever the "SAMPLE RATE" switch is rotated. Calibration is completed within 5 seconds.

The "WORD LENGTH" Switch is a twelve-position rotary switch, which is located to the right of the "SAMPLE RATE" 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 noise shaping.

There are four word length reduction modes ("TPDF", "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 it will sound noisier than the noise-shaped 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. 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. The other WLR settings are optimized for special circumstances. See page ???? for details.

> Note: The auxiliary outputs (available only on 4-channel "R", and 4 or 8-channel "B" converters) are always fixed at 24-bits. These outputs allow simultaneous 16 and 24-bit outputs from the same converter.

The "METER" Switch is a three-position toggle switch 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 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. 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.