ANALOG DEVICES

Circuit Note
CN-0219

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Devices Connected/Referenced	
ADAU1761	SigmaDSP® Stereo, Low Power, 96 kHz, 24-Bit Audio Codec with Integrated PLL
ADAV801/ ADAV803	Audio Codec for Recordable DVD (SPI/I²C Control Interface)

S/PDIF and I²S Interface for a SigmaDSP Codec Using the ADAV801/ADAV803 Audio Codec

EVALUATION AND DESIGN SUPPORT

Circuit Evaluation Boards

- **ADAU1761 Evaluation Board (EVAL-ADAU1761Z)**
- **USBi USB Interface Board (EVAL-ADUSB2EBZ) (Included with EVAL-ADAU1761Z Board)**
- **ADAV801 Evaluation Board (EVAL-ADAV801EBZ) or ADAV803 Evaluation Board (EVAL-ADAV803EBZ)**

Design and Integration Files

- **Schematics, Layout Files, Bill of Materials**

CIRCUIT FUNCTION AND BENEFITS

S/PDIF (Sony/Philips Digital Interface) is a high quality digital audio format that is commonly used in consumer electronics and is used to interconnect audio equipment. Many audio codecs/DSPs only support I²S as digital audio input/output, which is a problem when using these parts in circuits that need to support both S/PDIF or the AES (Audio Engineering Society) professional standard.

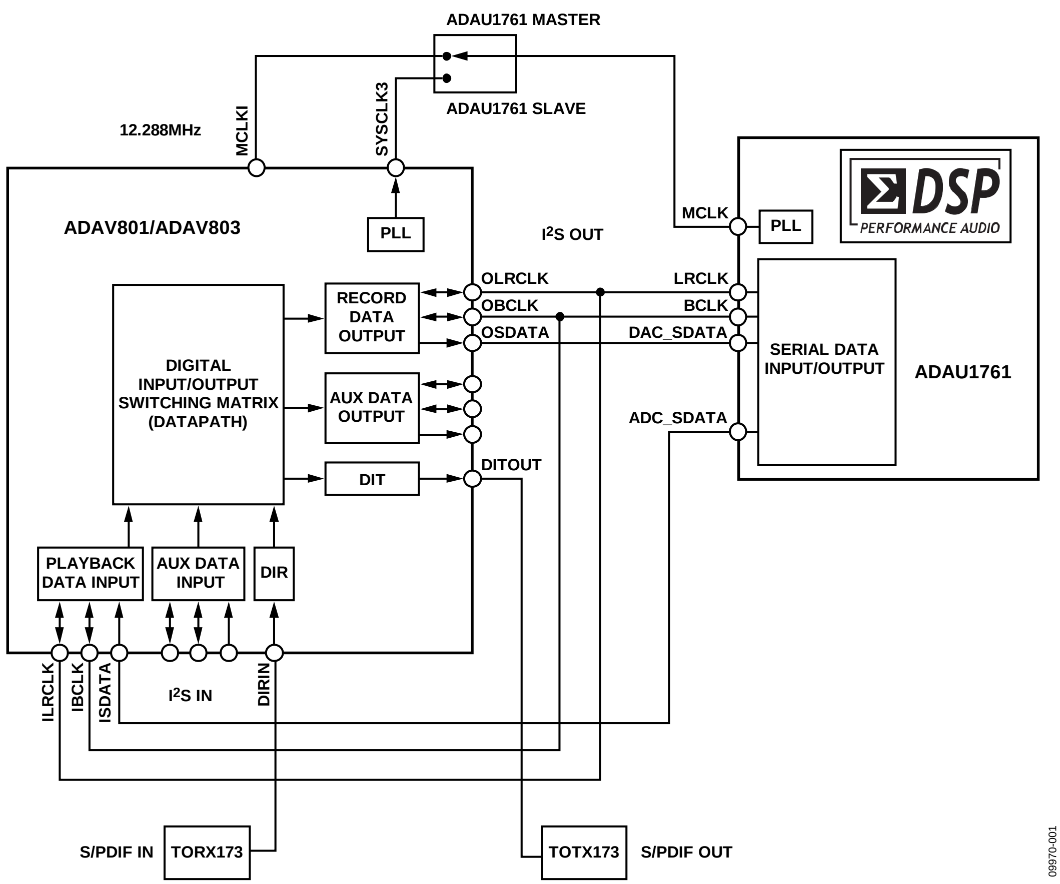

Figure 1. ADAV801/ADAV803 Connections for S/PDIF In/Out to ADAU1761 SigmaDSP (Simplified Schematic: Power Supply Decoupling and All Connections Not Shown)

Rev.0

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The circuit, in Figure 1, shows how to overcome this problem by connecting the ADAV801 or the ADAV803 audio codec to a SigmaDSP® device, such as the ADAU1761.

The audio input in S/PDIF format is converted to I^{2}S before processing by the ADAU1761, and the processed audio output in I^{2}S format is converted back to S/PDIF by the ADAV801/ ADAV803. The ADAV801/ ADAV803 has a flexible digital input/output routing matrix that allows it to process audio in either I^{2}S or S/PDIF format and output it in either format as a master or slave with the use of an onboard SRC (sample rate converter). The ADAV801/ ADAV803 support the consumer audio standard, and channel status data can be embedded in the audio stream by writing to the relevant registers in the ADAV801/ ADAV803. This is a useful feature for passing configuration information between devices. The ADAV801/ ADAV803 has a stereo DAC/ADC that can also be used to process audio as needed.

CIRCUIT DESCRIPTION

The ADAV801/ ADAV803 has two sets of input/output I^{2}S ports, either of which can be used. In the configuration shown in Figure 1, the playback port ILRCLK and record port OLRCLK pins are connected to the LRCLK pin of the ADAU1761. The IBCLK and OBCLK pins are connected to the BCLK pin of the ADAU1761. The ISDATA pin is connected to the ADC_SDATA pin of the ADAU1761, and the OSDATA is connected to the DAC_SDATA pin of the ADAU1761.

The S/PDIF input comes from the TORX173 fiber optic receiver module into the DIRIN pin and is then output to the ADAU1761 on the record port in I^{2}S format. Once the audio is processed by the ADAU1761 SigmaDSP® device it is output on the ADC_SDATA pin to the playback port of the ADAV801/ ADAV803 in I^{2}S format. It is then converted to S/PDIF format on the DITOUT pin and fed to the TOTX173 fiber optical transmitter module.

The circuit is powered from a 3.3 V AVDD supply. The master clock for the circuit is generated either by the ADAV801/ ADAV803 or by an external oscillator, depending on whether the ADAU1761 is to be configured as master or slave. In the case where the ADAU1761 is a slave, i.e. the BLCK and LRCLK are driven by the ADAV801/ ADAV803, the MCLK is 256× the recovered audio clock from the S/PDIF stream. It can also be configured to be 512× the recovered clock. This clock is accessed on the SYSCLK3 pin of the ADAV801/ ADAV803 and connected to the MCLK pin of the ADAU1761.

When the ADAU1761 is master, the MCLK is generated by an onboard oscillator and is supplied to the ADAV801/ADAV803 on the MCLKI pin. In this case, the ADAU1761 drives the LRCLK and BCLK lines, and the SRC on the ADAV801/ ADAV803 is used to synchronize the audio between the I^{2}S port and the S/PDIF port.

Register Settings

A complete design support documentation package for this circuit note can be found at www.analog.com/CN0219-DesignSupport. This includes register setting files for both master and slave configuration for the ADAV801/ADAV803 and ADAU1761. These register settings files can be loaded using the relevant evaluation board software.

COMMON VARIATIONS

This circuit can also be set up with any part that has a SigmaDSP processor core and requires an S/PDIF/AES audio interface, including the ADAU1401A, ADAU1701, and ADAU1781. Although not described in this circuit note, the above circuit can be modified to work with the AES audio format. Instead of optical connectors, XLR connectors would be used, and transformers would be required to convert from differential to single-ended signals and vice versa.

CIRCUIT EVALUATION AND TEST

This circuit is tested using the ADAV801/ ADAV803 (EVAL-ADAV801EBZ or EVAL-ADAV803EBZ) and ADAU1761 (EVAL-ADAU1761Z) evaluation boards. The necessary connections between the two boards and link configurations are contained in the design support documentation. Figure 2 shows the full test setup using both evaluation boards.

Equipment Needed

The ADAU1761 evaluation board is programmed using SigmaStudio thru a USBI board (EVAL-ADUSB2EBZ). The SigmaStudio GUI software requires a PC with the following: Windows® 7, Windows Vista, or Windows XP Professional or Home Edition with SP2, 128 MB of RAM (256 MB recommended), 50 MB of available hard disk space, 1024 × 768 screen resolution, and USB 1.1/2.0 data port. The ADAV801/ ADAV803 board is controlled using the printer port of a PC with its own software that can be downloaded from the ADI website. Two optical connectors are needed to connect the S/PDIF input/output to the ADAV801/ ADAV803 board. Eight single pin jumper cables are needed to make the necessary connections between the two evaluation boards.

Getting Started

From this point, follow the documentation for the EVAL-ADAU1761Z and EVAL-ADAV801/ EVAL-ADV803EBZ regarding software installation, setup, and operation of the system.

The SigmaStudio software is used to program and tune the registers and SigmaDSP core in the ADAU1761. SigmaStudio can be downloaded from www.analog.com/sigmastudio.

The software for the ADAV801/ ADAV803 can also be downloaded from the ADI website. Once the software is installed, the register setting files in the design documentation can be loaded to program both boards depending on whether you want the ADAU1761 device to be master or slave. The ADAU1761 SigmaStudio project has just a simple audio pass-thru with volume control for the purposes of testing the circuit of Figure 1.

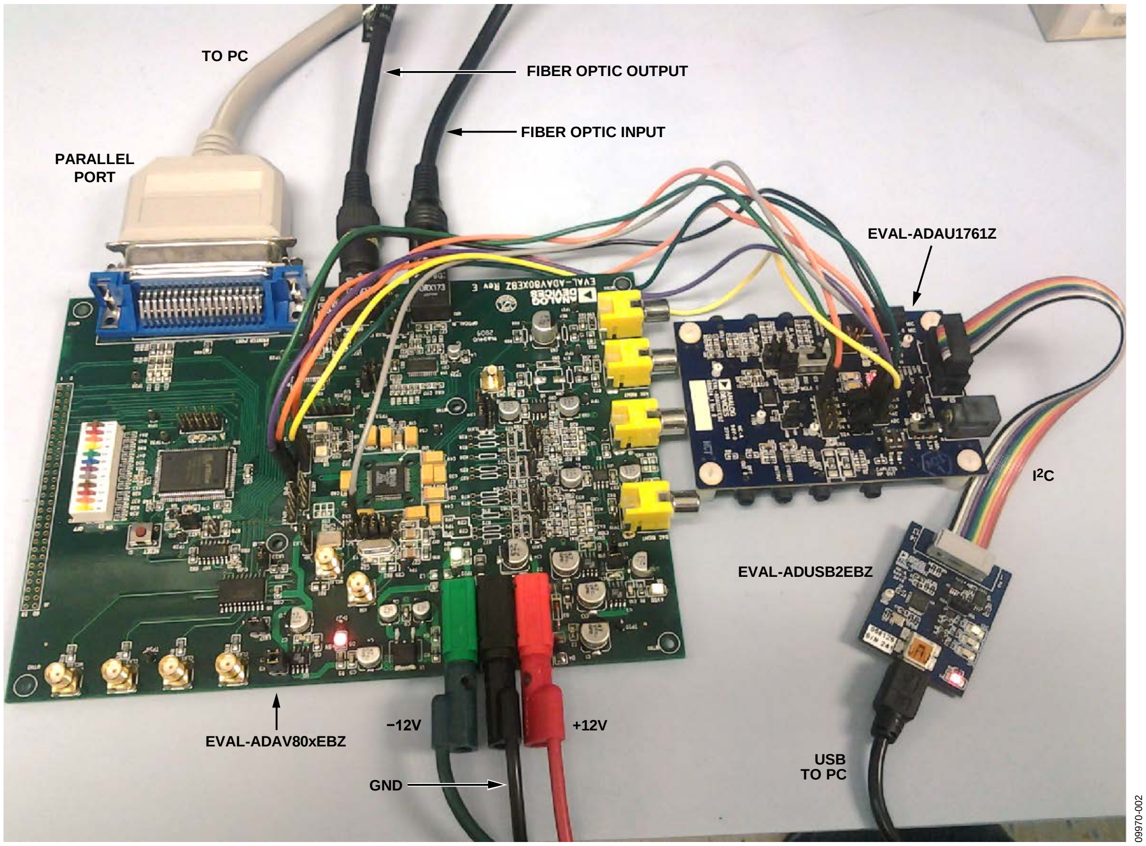

Figure 2. Test Setup for Connecting the ADAV801/ADAV803 Board to the ADAU1761 Board

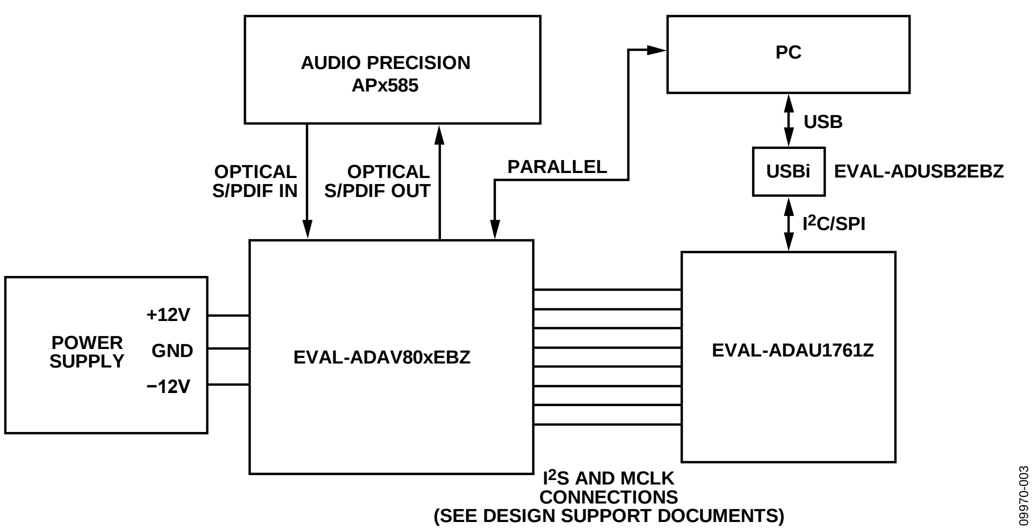

Figure 3. Functional Diagram of Test Setup

Setup and Test

An Audio Precision APx585 multichannel audio analyzer can be used to generate the S/PDIF input and capture the S/PDIF output. With the ADAU1761 as master and a full-scale 1 kHz input tone, the THD + N should be ~130 dB at the S/PDIF output. In slave mode, the THD + N should be ~142 dB, since there is no SRC needed to synchronize the S/PDIF stream to the ADAU1761 I^2S stream.

LEARN MORE

CN0219 Design Support Package: www.analog.com/CN0219-DesignSupport

Gildersleeve, Brett, *Using the EVAL-ADUSB2EBZ*, Application Note AN-1006, Analog Devices.

SigmaStudio™ Graphical Development Tool: www.analog.com/sigmastudio

Data Sheets and Evaluation Boards

ADAU1761 Data Sheet

ADAU1761 Evaluation Board

ADAV801 Data Sheet

ADAV803 Data Sheet

ADAV801/ADAV803 Evaluation Board and Software

REVISION HISTORY

10/11—Revision 0: Initial Version

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