Alesis ADAT
Proprietary Multichannel
Optical Digital Interface

ADDENDUM February 2001

2X Sample Rate (96kHz) Specification

Version 1.0 - February 3, 2001

BACKGROUND

The Alesis ADAT Multichannel Optical Digital interface, designed in 1990, has become an music industry standard for multichannel digital audio data interchange, primarily because of its ease of use, its low cost of implementation, its space-saving footprint, and because of the ubiquity of ADAT and ADAT-compatible products on the market.

The ADAT Optical format was designed to provide easy interconnection of multichannel audio sources (primarily ADATs, when the specification was developed), and is specified as a nominally 48kHz, 8-channel, 24-bit data path. Audio data is transmitted and received on separate optical paths; that is, to both transmit and receive ADAT optical format, a device must use two optical cables.

In recent years, particularly with the advent of the DVD standard, a need has developed in the music industry to record and transmit audio data at 96kHz sampling rates. Since the ADAT Optical is specified at a nominal 48kHz sample rate, it was deemed necessary to develop a method for implementation of 96kHz data transfers. This document describes the Alesis approved implementation of a 96kHz, 4-channel audio interface using existing ADAT optical hardware.

UNCHANGED FOR 96K IMPLEMENTATION

Both the transmitting and receiving hardware and the channel coding scheme specified in the original ADAT Multichannel Optical Digital Interface document remain the same in this 96kHz scheme; the rate at which the transmitters and receivers operate is maintained at a nominal 48kHz frame rate, with a 12.288MHz nominal NRZI bit rate. In addition , there are still 4 user bits transmitted per frame, as previously specified.

WHAT IS NEW

Two significant changes are involved in supporting 96kHz/4-channel mode: sample-splitting and user bit implementation.

User bit Implementation

The ADAT Multichannel Optical Digital Interface specifies 4 user bits per frame, labeled bits 0-3. Bit U0 is defined as Time Code, and U1 as MIDI data. This document specifies user bit U2 implementation as follows:

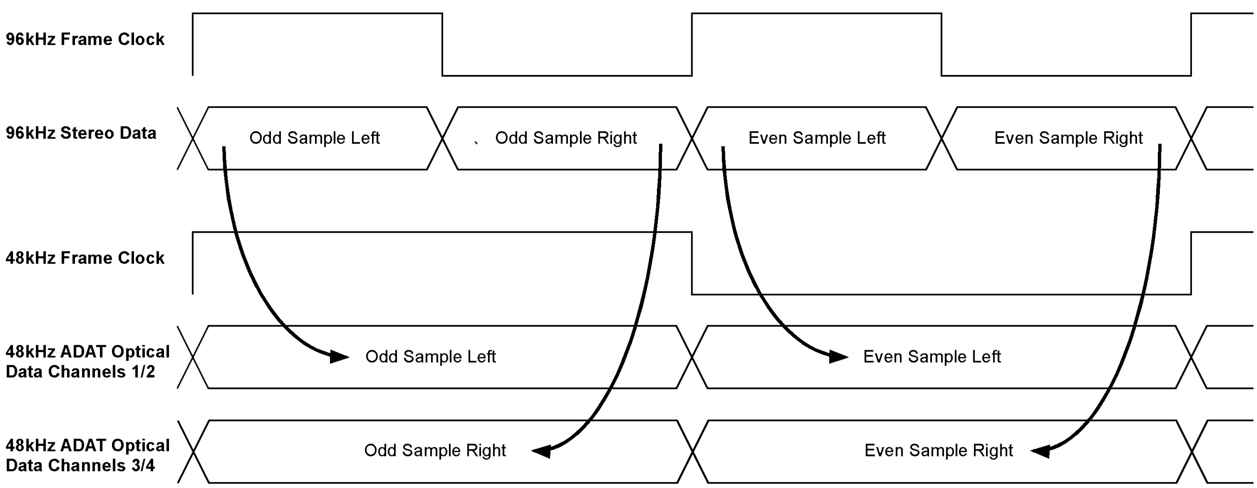
For 1X (48kHz) operation, U2 is to be held at logic low
For 2X (96kHz) operation, U2 is to be held at logic high

This is clearly an under-utilization of user bit U2, as it is capable of transmitting 48,000 bits per second of information. However, simply setting the bit high or low negates the need for a sample-synchronous receiver or transmitter implementation, and allows a receiving system to poll the user bit on a relatively infrequent basis to determine incoming sample frequency.

Sample-Splitting Implementation

Since a 96kHz recording produces exactly twice as many samples per unit time as does a 48kHz recording, it is convenient to use two audio channels operating at 48kHz to communicate one channel of 96kHz audio data. This means that samples from one audio data stream must be "split" across two audio data streams. Because the Alesis ADAT Multichannel Optical Digital Interface is an eight-channel, 48kHz audio data path, there are several options for sample splitting amongst audio pairs; The Alesis approved method is described below.

Samples of a single channel of 96kHz audio information are split amongst adjacent ADAT Optical channels, with odd samples (samples 1,3,5,7, etc.) transmitted on an "odd" ADAT Optical channel (1,3,5, or 7), and even samples (samples 2,4,6,8, etc.) transmitted on an "even" ADAT Optical channel (2,4,6, or 8). This is illustrated below:



This example shows stereo 96kHz audio data being mapped to four channels of ADAT Optical. The left and right data is shown interleaved, as it might be seen inside a device going to or from DACs or ADCs. Note that this drawing implies that the right channel samples be advanced in time; in fact, the sample-splitting process can only be accomplished by delaying the output (48kHz) samples at least one 96kHz sample period with respect to the input (96kHz) samples.

One small advantage to splitting the samples in this manner is that if a single channel (either odd or even samples) of the resultant "split" data is monitored using a 48kHz digital-to-analog conversion process, the audio will sound almost normal, because in essence a crude sample rate conversion has occurred from 96kHz to 48kHz. Note that signals above 22kHz that were originally recorded will alias back into the audio using this technique, potentially introducing strange artifacts into the signal being monitored.