Advantages (and Disadvantages) of Digital I/O

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Since dinosaurs ruled the earth, audio professionals have used analog connections between their gear. The advantage of an analog connection is that it is universal—other than level differences and incompatibilities between balanced and unbalanced inputs and outputs, there is very little that can prevent *some* sort of signal transfer between the originating and receiving devices. However, analog connections have their well-known weaknesses: they are prone to hum and noise pickup, and long connections can easily cause unacceptable high-frequency loss. In a complex plant with active devices in the signal path, levels must be carefully lined up to prevent amplifiers in the signal path from clipping. In a stereo or multi-channel facility, cable lengths and types must be matched to prevent relative phase shifts from appearing between channels. It is all too easy to accidentally swap left and right channels, mirroring the stereo image. Worse yet, reversing polarity of one channel causes cancellation of the monophonic sum signal. And, for you absolute-polarity mavens out there, it is even easier to invert absolute polarity so that speakers suck when they should be pushing.

Digital I/O solves most of these problems. In the common AES/EBU and SP/DIF standards, one cable carries two channels of audio. There can be no problems with relative phase shift between channels, amplitude unbalance, polarity swapping, or gain shifts to cause clipping. If you use AES/EBU properly, status bits tell you whether the signal is stereo or mono, whether it is pre-emphasized, and its sample rate. If you choose your cable carefully (110 ohm balanced cable is required), you can run AES/EBU several hundred feet with no signal loss. Using a balanced-to-unbalanced adapter with 75 ohm coax and BNC connectors can extend that distance to several thousand feet. And if you go too far, you'll know it—the receiver will simply fail to decode the signal. Subtle problems are quite unlikely.

One subtle problem that *can* arise in AES/EBU transmission is jitter in the clock recovered by the receiver. Jitter means that the period of each recovered clock cycle is not quite identical to the period of the cycle preceding and following it. If such an unstable clock is used to clock a digital-to-analog converter, the resulting frequency- or phase-modulation sidebands around the clock and its harmonics result in what amounts to modulation noise in the recovered audio. "One-bit" converters (like the MASH converter and its cousins) tend to be more sensitive to this than the more traditional multi-bit designs.

To a modest extent, the AES/EBU standard itself tends to cause jitter because the receiver must recover the clock from the bitstream itself. This is not entirely uniform when being modulated by a digital audio signal. However, if a low-pass filter is interposed between the AES/EBU transmitter and receiver, this can cause the position of the bit transitions to become much more uncertain, greatly increasing the probability that the receiver will recover a jittered clock. The cheap TOSLINK optical connections provided on consumer CD players are particularly prone to this behavior, but any long cable will roll off the higher frequencies (the AES/EBU signal has significant energy up to 3-4MHz) and cause similar problems.

Another important issue regarding digital signals in a practical studio environment is the question of how one mixes them. With analog, there was no problem; a couple of resistors would do quite nicely. However, digital is not nearly as forgiving because digital mixing must be synchronous. To mix two PCM digital signals, their sampling frequencies must be *identical.* The situation is completely analogous to video, where mixing two video signals requires that they be timed identically. In digital audio, requirements are not quite as stringent because a DSP mixer will accumulate the serial input bitstreams in a buffer memory until all of the bits in a given digital word are available. Only then are they added, so the video requirement of exact phase matching between inputs does not exist in the world of digital interconnect. However, if the digital words are appearing at the inputs at different *rates,* then the buffer memory will quickly overflow as it tries to line up the words from the different inputs. So identical sample rates are a must.

Fortunately, the last year has given us an answer to both the jitter and sample rate problems simultaneously with the advent of low-cost integrated asynchronous sample-rate-converter (SRC) chips from Analog Devices and Philips. Regardless of the sample rate at their inputs, these chips will emit a bitstream that is synchronous to the system clock that drives them. Fortuitously, they simultaneously remove jitter at their inputs, and the jitter at their outputs is essentially as low as the jitter of the system clock driving the chips. If you place an SRC at each input of a digital mixer (or any other device where jitter removal is required), interconnecting a digital studio becomes a plug-and-play operation. No house sync is required to ensure that various digital sources emit identical sample rates; an input can accommodate any digital signal meeting the standard for which the receiver is designed (such as AES/EBU).

Our experience with the Analog Devices chip shows that there is only one serious potential problem. Internally, asynchronous SRCs require large FIR filters to do their job. At a 32kHz output sample rate, the filter used in the Analog Devices chip is down about 0.5dB at 15kHz. (It's essentially flat to about 14.5kHz.) So multiple passes through SRCs outputting 32kHz may result in significant loss of frequency response at 15kHz. However, since 32kHz is generally used only in studio-transmitter links, this is not usually a significant problem. For the common studio sample rates of 48kHz and 44.1kHz, the new SRC chips are essentially transparent and will usher in a new generation of high-performance, easy-to-connect digital studio equipment.