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Digital Audio Connections and Synchronization

Many PreSonus products are equipped with standardized digital audio inputs and outputs, enabling them to transfer digital audio to and from an assortment of devices without conversion to and from the analog domain. Some PreSonus products also have dedicated, standardized connections for digital-audio synchronization signals. To help you understand these technologies, we've prepared a brief tutorial on digital connections and synchronization.

ADAT Optical

The Alesis ADAT modular digital mutitrack tape recorder allowed users to record eight tracks of digital audio simultaneously. The ADAT Optical interface protocol, commonly referred to as "ADAT Lightpipe," was developed to stream eight channels of 16-, 20-, or 24-bit digital audio at 44.1 kHz or 48 kHz, allowing multichannel digital transfers between ADAT digital recorders and other digital audio devices over a single fiber-optic cable. The ADAT Lightpipe format has been adopted by many audio manufacturers because it is a compact way to transfer multichannel digital audio data between devices.

FIG. 1: ADAT Lightpipe and S/PDIF optical use the Toslink connection system

ADAT Lightpipe uses the same type of optical cables and Toslink connectors as the S/PDIF two-channel optical digital audio protocol (discussed shortly). These cables can be purchased at your local recording-equipment store. Toslink is an optical-fiber connection system developed by Toshiba that uses a JIS F05 connector (see **Fig. 1**). The generic name for this standard is "EIAJ optical".

Current PreSonus products that offer ADAT Lightpipe I/O include the FireStudio™Lightpipe, FireStudio 26x26, and DigiMax™ FS. The DigiMax D8 offers ADAT Lightpipe output only.

S/MUX

"Sample Multiplexing" or S/MUX is used to transmit high-bandwidth digital audio using lower-bandwidth technology, such as ADAT Lightpipe. S/MUX works by joining two or more digital audio channels to represent a single higher-bandwidth channel. By using S/MUX technology, you can stream 4 channels of digital audio at 88.2 kHz or 96 kHz over the same Lightpipe connection originally designed to stream 8 channels of 44.1 kHz or 48 kHz audio.

The DigiMax FS and FireStudio 26x26 offer one set of S/MUX inputs and outputs. The FireStudio Lightpipe is equipped with two sets of S/MUX inputs and outputs, so it can record and play back 16 channels of audio at 88.2 kHz or 96 kHz or 32 channels at 44.1 or 48 kHz.

AES/EBU and S/PDIF

Developed by the Audio Engineering Society and the European Broadcasting Union, AES/EBU (officially known as AES3) is a 2-channel format that can carry audio signals at up to 192 kHz. AES/EBU employs a 3-pin XLR connector, which is the same connector used for most professional microphones. A single cable carries both channels of audio data. No current PreSonus products have AES/EBU digital connections but the now-discontinued DigiMax 96k was one of the first 8-channel mic preamps to offer them.

S/PDIF (Sony/Philips Digital Interface) was co-developed by Sony and Philips to transfer stereo digital audio. It is essentially a consumer version of AES/EBU, and as with AES/EBU, a single cable carries both channels of the stereo audio signal. Digital Audio Tape (DAT) machines were among the first devices to be equipped with this protocol but it has since become popular in consumer audio products such as DVD players, as well as in semi-pro and professional audio products.

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The most common connector used for S/PDIF is an RCA coaxial connector (see **Fig. 2**). While S/PDIF RCA coaxial uses the same connector as the analog RCA connection on consumer audio products, the cables are not the same, and these connections should not be confused. PreSonus products that offer S/PDIF RCA coaxial inputs and outputs include the FireStudio 26x26, FireStudio Mobile, and FireStudio Project. The StudioLive™ 16.4.2 and StudioLive 24.4.2 digital mixers have an S/PDIF RCA output, and the Central Station has an S/PDIF RCA input.

FIG. 2: RCA coaxial connectors are commonly used for S/PDIF digital audio.

S/PDIF also can be sent over Toslink optical connections, and these have become fairly common in semi-pro and professional audio gear. This is the same connector used for ADAT Lightpipe but the digital protocol is different. The Central Station offers an S/PDIF Toslink optical input, in addition to its S/PDIF RCA input; it is the only current PreSonus product that offers S/PDIF on Toslink.

TDIF

TDIF (Tascam Digital Interface) transmits and receives up to 8 channels of digital audio over a single cable and uses a 25-pin D-sub connector. Tascam introduced TIDF with the DA-88 modular digital multitrack recorder, and the format was later adopted by other companies. However, PreSonus products do not use TDIF.

The TDIF-1 specification has developed over the years. Early versions did not support self-clocking so a separate BNC word-clock connection was required. Later products added self-clocking, and the TDIF-1 version 2.0 specification added transfer of higher-resolution (up to 192 kHz) digital audio, albeit with reduced track counts.

Digital Clocking, Word Clock, and BNC

Digital clocking signals are used to synchronize digital audio signals flowing between devices in order to avoid data errors. Why is that necessary? Read on!

Analog audio is transferred through a cable as a continuous electrical waveform–it's not divided into discrete steps—and electricity travels through a straight wire at almost the speed of light. So when you route audio between analog devices, the signals arrive instantaneously, for practical purposes. Therefore, you don't have to synchronize analog audio when routing between devices.

Transferring digital audio is a very different matter. Computers and other digital devices operate one step at a time, which happens very quickly but it's not instantaneous, and digital signals are not inherently in perfect time. True, uncompressed digital audio plays at a fixed rate (the sampling frequency) but digital clocks are not perfect; their frequency can drift, and they almost always have at least some irregular errors known as *jitter*. Therefore, two devices, each following its own clock, are highly unlikely to stay in agreement about precisely when a sample starts and ends. The result is usually an artifact: a pop or glitch in the audio.

To avoid this problem, when transferring digital audio in real time among multiple digital devices, all digital devices need to follow a single master clock. That means the master clock has to send a signal that essentially says, "everyone start *at this moment* and follow me!" Even if the master clock's timing is imperfect, all of the slave devices will follow the timing errors exactly and will stay in sync with each other. Therefore, you won't get timing-related artifacts.

Three basic clocking systems are in common use in digital-audio systems. Each is used in various PreSonus gear.

In a *self-clocking system*, the clock signal is embedded in the audio stream. Samples are streaming in real time, with the clock marking the start time of each sample. The receiving device extracts the embedded clock signal from the digital audio. Self-clocking is implemented in every digital audio protocol (ADAT, S/PDIF, AES/EBU, etc.), and all PreSonus devices that offer ADAT Lightpipe or S/PDIF I/O employ self-clocking systems for those connections.

Distributed clocking systems are commonly used in professional environments with multiple digital devices because a clocking stream that does not contain audio will be cleaner than a clocking stream that does contain audio, as in a self-clocking system. In a distributed system, sync signals and audio signals are separate. The clocking signal in a distributed clocking system is called *word clock*. (A *word,* in this case, is a fixed-size group of digital bits.) A word-clock generator creates digital pulses that control the frequency of the internal oscillator of each device to avoid frequency drift. Word clock does not carry location information, the way time code does, nor does it carry audio information; it only determines the rate at which samples should occur.

Word clock is used for a few types of distributed-clocking systems but the most common in professional audio uses a BNC connector (see **Fig. 3**). In this system, word clock is sent over dedicated, shielded, coaxial cables with standard twist-lock, BNC-type connections on each end. The cables are rugged and can carry clock signals much farther than standard optical cable. BNC connections are made in several impedances but PreSonus devices require an impedance of 75Ω to achieve consistent sync. We'll discuss various ways of wiring word-clock devices in the section "Synchronizing Multiple Digital Audio Devices." PreSonus devices that offer BNC word-clock connections include the FireStudio 26x26, FireStudio Lightpipe, DigiMax FS, and DigiMax D8.

FIG. 3: Word clock is usually transmitted over BNC connectors.

FireWire and USB don't use an embedded clock for timing; instead, they use a process called *reclocking*, in which the audio stream is divided into packets of data that are transmitted sequentially. The receiving device buffers the signal, reassembles the sample stream, and then sends the audio signal from the buffer at the appropriate time, as determined by the device's sample rate. This ensures accurate timing when going from a FireWire or USB interface to a computer and back.

(Note that copying an audio file is not a real-time transfer, so audio sync is not an issue.)

Synchronizing Multiple Digital Audio Devices

Whether you are using BNC or another digital output to generate word clock, it is necessary to designate one device as the "master" word-clock device to which all other digital devices are synced, or "slaved." Many digital devices perform equally well as a master or a slave, although some devices must be one or the other. Synchronizing a device to a lesser-quality clock source is likely to degrade its performance, and not all word-clock generators are created equal. You generally need to determine which device has the best clock and to designate that device as the word-clock master. This is done with careful listening and A/B testing.

Once you've determined which device is to be your master clock, you will need to sync the remaining digital devices through series or parallel distribution or some combination thereof. Of course, if your digital-device chain only consists of one master and one slave, syncing the two is as simple as connecting a BNC word-clock cable from the output of your master device to the input of the device you are slaving.

When working with multiple slaved devices, the job gets a bit more complicated. Series distribution requires that your digital devices have both a BNC word-clock input and a BNC word-clock output. Parallel distribution uses a BNC T-connector attached to the BNC word-clock input of each slaved device. This allows the word-clock signal to be sent to that device and then sent on to another device. A BNC word-clock output on the slaved devices is not used for parallel word-clock distribution.

If the last device in the chain does not have a word-clock terminate switch, you will need to attach a BNC terminator plug to the other side of the T-connector. This helps to stabilize the word-clock sync and keep the word-clock signal clean. Both word-clock terminator plugs and BNC T-connectors can be purchased at most recording-supply retailers.

A third option for syncing your digital devices is to purchase a high-quality, dedicated word-clock generator. Many engineers believe that using dedicated word-clock generators enhances the performance of digital audio devices more effectively than series or parallel word-clock distribution. A dedicated word-clock generator and distribution amplifier exists for one purpose and one purpose only: to be a master clock.

Word-clock generators usually have one BNC word-clock input and multiple BNC word-clock outputs, and sometimes they have TDIF, S/PDIF, or ADAT outputs to make them compatible with as many types of digital devices as possible. Without a dedicated word-clock generator, it is necessary to split the master device's word-clock signal by daisy-chaining the slaved devices as described earlier. Because of this, many engineers feel that the resulting digital audio signals will be of a higher quality when a dedicated word-clock generator is used; in this way, all digital devices are receiving the same digital pulse from the same source at exactly the same time.

Whichever approach you use, it is always advisable to use quality BNC cables that are no longer than necessary for the job. As with audio cabling, it is good to keep word-clock cables separate from AC cable lines or other possible sources of interference.