AVIOM

Application Note

Managing Jitter, Wander, and Latency in Digital Audio Networks

Designers of professional-grade digital audio equipment face a number of challenges in their quest to reproduce and distribute truly transparent audio. Chief among these are jitter, wander, and latency. These challenges are magnified when the system needs to sync to external clock sources, support a range of sample rates, and network many devices together.

Jitter, wander, and latency affect not only the performance of an audio network, but the quality of the sound itself. If a digital audio network is to achieve true high fidelity and preserve the nuance and detail of every audio signal in the system, these issues must be addressed.

WHAT IS JITTER?

Think of jitter as the right thing happening at the wrong time. Imagine a train station where trains are scheduled to arrive every hour on the hour. Of course in the real world, the trains will almost never arrive at exactly 7:00 AM, 8:00 AM, etc. – one train will be a minute early, the next one a few minutes late, and so on. These small variations between the scheduled and actual arrival times over the course of a day are, in effect, jitter.

The same scenario applies in a digital audio system, except that instead of hours and minutes, the intervals are measured in units as small as picoseconds. In order to reconstruct an accurate waveform, a D/A converter depends on a clock signal that is "scheduled" to arrive at exact intervals. But as with our hypothetical train, there is no such thing as a "perfect" digital clock – there will always be some small variations. And while the variations in a train's arrival time are just a minor inconvenience, significant clock variations in digital audio can lead to real problems.

WHY DOES JITTER MATTER?

Simply put, jitter causes distortion and/or raises the noise floor. It can degrade the performance of an audio network, both in the digital domain and ultimately in the analog output of the devices in the system. In severe cases it can result in a total loss of sync.

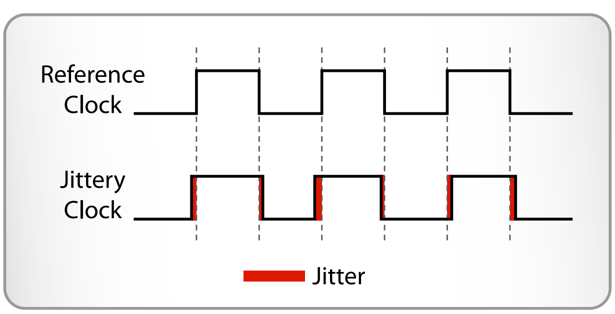

Figure 1. **A jittery clock compared to an ideal reference clock. The areas in red represent the jitter in the reconstructed clock.**

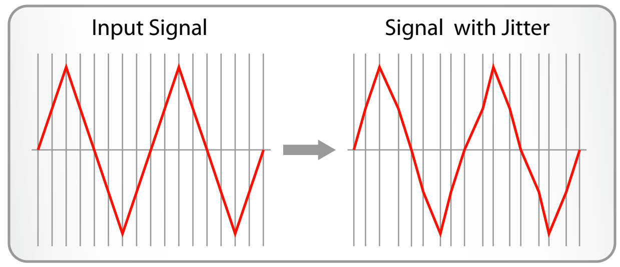

Figure 2. **Effects of jitter on digital audio. Variations in the incoming clock signal result in distortion to the output waveform.**

Because D/A and A/D converters rely on the reference clock to correctly interpret the digital data going in or out, an irregular digital clock actually changes the analog waveform. The effect is most noticeable in the high frequencies where the most subtle audio cues, such as depth and location information, reside. The result is a loss of detail, or a certain "harshness" about the sound.

Obviously the key to transparent audio reproduction is keeping these variations in clock frequency so small that audio introduced at one end of the system comes out the other end sounding exactly the same.

IT GETS WORSE

It's one thing to design a simple, closed system such as a CD player with ultra-low jitter, but quite a different challenge when many devices are networked together. In a digital audio network – particularly one with a serial, or daisy chain, topology – jitter can accumulate as the clock signal is passed from one device to the next. Great care must be taken in the design of such devices to "de-jitter" the incoming clock before sending it to internal converters for D/A conversion, and then passing it along to the next device in the chain.

Keeping jitter inaudibly low is particularly challenging in systems with variable sample rates, since low jitter and frequency variability are essentially at odds with one another. But, in the real world, different devices, even different audio networks, must be able to sync to each other, and to all sorts of clock sources, from Word Clock to SMPTE to an AES3 input.

Again it comes down to design. If an audio transport protocol is to achieve the highest possible audio fidelity while retaining the ability to sync to virtually any external device or clock source, it needs to be designed from the ground up for the purpose. Ideally, it should not use sample rate converters to accomplish digital I/O, as sample rate converters add cost, increase latency, and change (distort) the data.

Additionally, each device on a network must have the ability to accept any incoming clock signal, effectively de-jitter

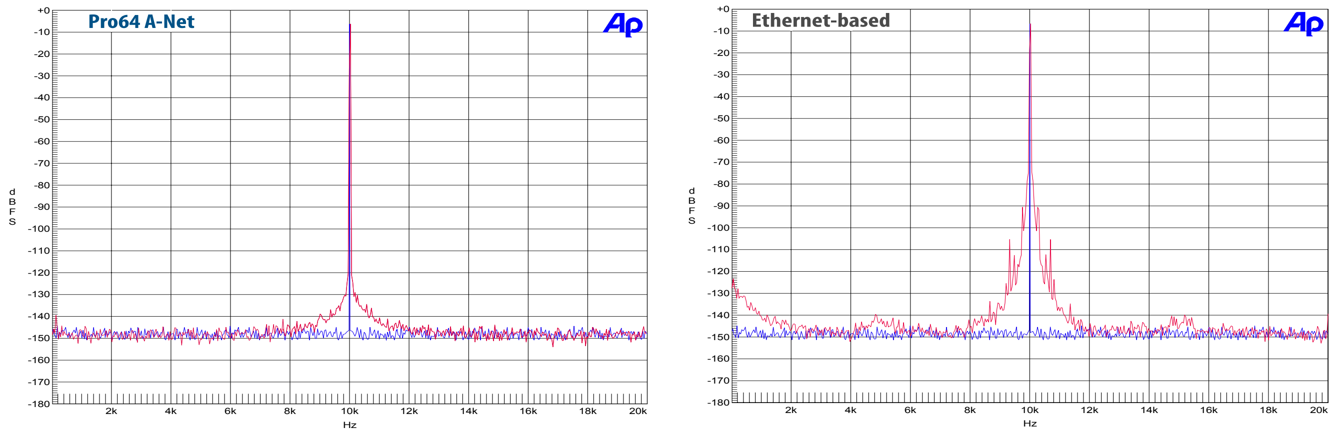

Figure 3. **Jitter artifacts in an Ethernet-based audio network (right) compared to a Pro64 A-Net network (left), both after four network "hops." FFT shows frequency components from 20Hz to 20kHz with a 10kHz input signal at -6 dBFS. Major deviations from the ideal response (shown in blue) indicate jitter in the regenerated master clock.**

it for local use, and then retransmit it to the next device in the chain without adding any additional jitter of its own. These capabilities are best implemented at the system level, with tight hardware and software integration, sophisticated jitter-limiting algorithms, and efficient use of network bandwidth.

WANDER

Wander is essentially low frequency jitter caused by the accumulation of jitter as clock information is retransmitted from device to device. To return for a moment to our hypothetical train station, if, over the course of the day, little delays, unscheduled stops, rail problems, etc., piled up until the 8:00 PM Express was forty-five minutes late, that would be analogous to *wander.*

Wander can appear any time digitally clocked devices are daisy chained. The jitter patterns in separate devices will be random, so as more devices are added to the chain, small errors can add up as the clock is passed on, soon becoming big errors (see **Figure 4**). One possible work-around would be to avoid daisy chains, but this would unnecessarily limit the flexibility of the network. If wander continues to accumulate, it can ultimately cause individual devices to lose sync with the network clock.

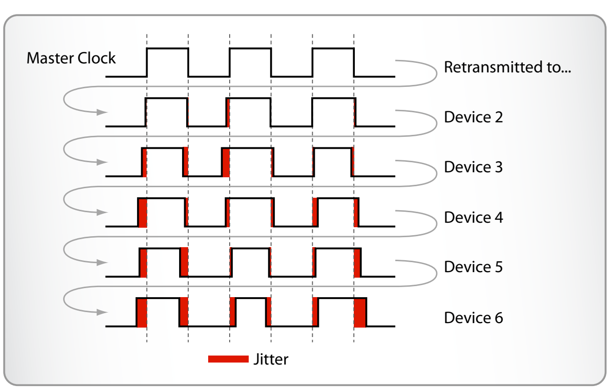

Figure 4. **Small variations in the clock frequency (jitter) can add up as the clock is retransmitted from one device to the next, resulting in jitter accumulation, or wander.**

LATENCY

Professional audio applications generally require systems with extremely low latency. Total system latency in a digital audio network is the time delay between the audio input at one device and the audio output at another. Latency is cumulative, with each additional device connected to the network adding its own delay.

Ultra-low latency is crucial if a digital audio system is to deliver the kind of performance that audio professionals expect—and what they *really* expect is for a digital audio path to sound and act like an analog cable (a very short and well shielded cable, at that!). Controlling jitter and wander lets a digital audio network *sound* like a cable; minimizing latency lets it *act* like one.

Moreover, since other components external to the network (such as digital mixing consoles, DSP processors, and A/D/A conversions) introduce their own delays, it is critical that the latency within the network be kept to an absolute minimum. Delays of more than a millisecond in the arrival time of the audio signal are just as unacceptable as the distortion and coloration caused by jitter. This is especially true in live sound and recording applications, where musicians are relying on instantaneous feedback to adjust the timing and intonation of their performances, and in the box-to-box time alignment of complex speaker systems. In professional audio, milliseconds matter.

A SYSTEM-LEVEL SOLUTION

In developing its Pro64® A-Net® audio networking protocol, Aviom took all these factors into account from the earliest design stage. Since Pro64 A-Net was built from the ground up for networking audio, it delivers the highest quality 24-bit audio, the widest range of sample rate variability, and the most flexible network topology available, all the while retaining sub-millisecond latency and lower levels of jitter and wander than any protocol based on older, less efficient, networking standards.