



Clocking, Jitter and the Digidesign 192 I/O Audio Interface

In this paper, we will discuss the nature of clocking and jitter in digital audio interfaces and in particular the Digidesign 192 I/O interface. The 192 I/O is a multi-channel digital audio interface with several clocking options that can be used in single- and multi-interface configurations. There is much talk about the use and possible benefit of using alternative clocking schemes with the system i.e. those other than "Internal" and "Loop Sync". Debate about what sounds better continues between those who prefer to use external clocks vs. those who prefer the internal clocking scheme. And while we can't resolve the sound quality issue in the context of this paper, by looking closely at the 192 I/O's clocking options users will gain a better understanding of their equipment and be less vulnerable to misinformation.

First, we'll look at some building blocks and background information (skip it if you're already familiar with basics), then examine the clocking schemes used within the 192 I/O internally and in systems with multiple interfaces. We'll also look at some other manufacturer's clocking products and their effect on the 192 I/O. Several FFTs illustrate that for some, "less is better" is not always the case with jitter in audio clocks. And although jitter in clocks is easy to represent graphically, its effect on the listening experience is less easily measured. There isn't a clear champion—that is left up to the individual listener—but by understanding the system better, the user should be able to listen without bias and that's our goal. So, let's get started.

BUILDING BLOCKS

By its very nature, digital audio relies on the ability of the system to take snapshots (samples) of audio waveforms at fixed intervals and play them back at the same fixed intervals. A stable clock provides the time reference for the system to record and reproduce the audio material accurately. Variation in the time base or lack of regularity of the sampling clock is referred to as "jitter". Jitter introduces distortion to sampled audio, but only in two critical phases of the audio sampling process—at the conversion from analog to digital and the conversion from digital back to analog. Jitter is irrelevant in **digital to digital** transmissions because samples are sent from one system to another in sequence with their clocking

information. So, as long as the audio is captured as jitter free as possible during the conversion process, and the final digital to analog conversion is done with a stable clock, no damage will have been done to the audio during any interim stage.

There are several types of clocks used in digital audio, each with their own characteristics. Because of this, you'll find clocks of different types used in different places in digital audio equipment. In addition to the clock type, other factors such as power supply regulation, transmission technique and component tolerances affect the design and resulting quality of the interface. The goal of this paper is to provide the reader with insights about clocking, jitter and synchronization that will help them get the most satisfying result from their Pro Tools HD system.

First, here's some background information.

WHAT IS A CLOCK?

Sample clocks, as used in digital audio systems, provide the timing intervals that determine when audio samples are recorded or played back. Unlike clocks that we use to determine the time of day, a sample clock's primary function is to set a steady pace, much like a very fast metronome. The timing intervals generated by the clock are used for several operations within a digital audio system, such as determining the rate at which audio is sampled, or determining when the

samples get transferred to memory buffers as they are moved through the digital system. All of these operations are dependent upon a stable system clock.

Clocks of different but related speeds are used in all digital audio systems. Most commercial analog to digital converters (ADCs) are the delta-sigma type. These converters oversample audio at a very high rate, then use digital signal processing techniques to produce the final output samples. At 44.1 kHz, for example, a typical delta-sigma converter's sampling clock runs at 256 times the output sample rate, or about 11.29 MHz. At 48 kHz, the clock runs at about 12.29 MHz. It is this high-speed oversampling sampling clock that determines the actual instant in time when the analog signal is converted, and it is the stability of this clock that determines the resulting audio fidelity.

It should be obvious that for a sampling system to function properly, an extremely stable clock is required—one that is regular and without variation in speed. Digital audio systems assume that the sampling process is perfect and there is normally no provision for correcting for audio that has been sampled at inconsistent intervals. If the interval between samples varies during the analog-to-digital conversion, those variations are embedded into the recording in the form of distortion and/or noise. Subsequent playback of the recording will reproduce the recorded distortions. If the playback clock

is also not perfect, additional distortion and noise will be added. However, as mentioned earlier, jitter is only problematic at the *conversion stages*, either analog-to-digital or digital-to-analog. Jitter in signals transmitted digitally from device to device is not a problem unless it is so great that it causes a transmission error.

To be clear, it is impossible to create a perfect clock or one that is completely free from jitter. Though some systems can claim very low amounts of jitter, all systems have “non-zero” jitter. The challenge in converter design is managing the jitter “budget” and to make the device as stable and sonically pleasing as possible.

THE NATURE OF SAMPLING

Clock signals are pulses and are represented electrically by square waves. A perfect sample clock looks something like Figure 1.

Each “On” edge is used to trigger a measurement (a sample) or reproduce a sample. In this simplified drawing, the triggers are perfectly spaced, the transitions from Off to On are instantaneous and there is no noise or distortion in the waveform. Real life sample clock waveforms are subject to many factors that change their shape so that they only partially resemble the perfect square wave.

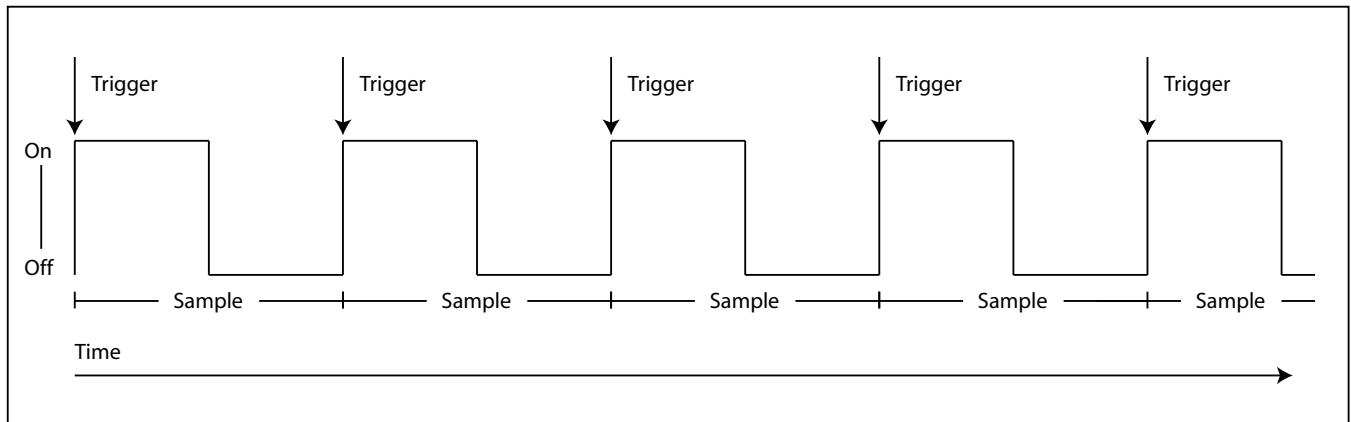


Figure 1: A Perfect Sampling Clock

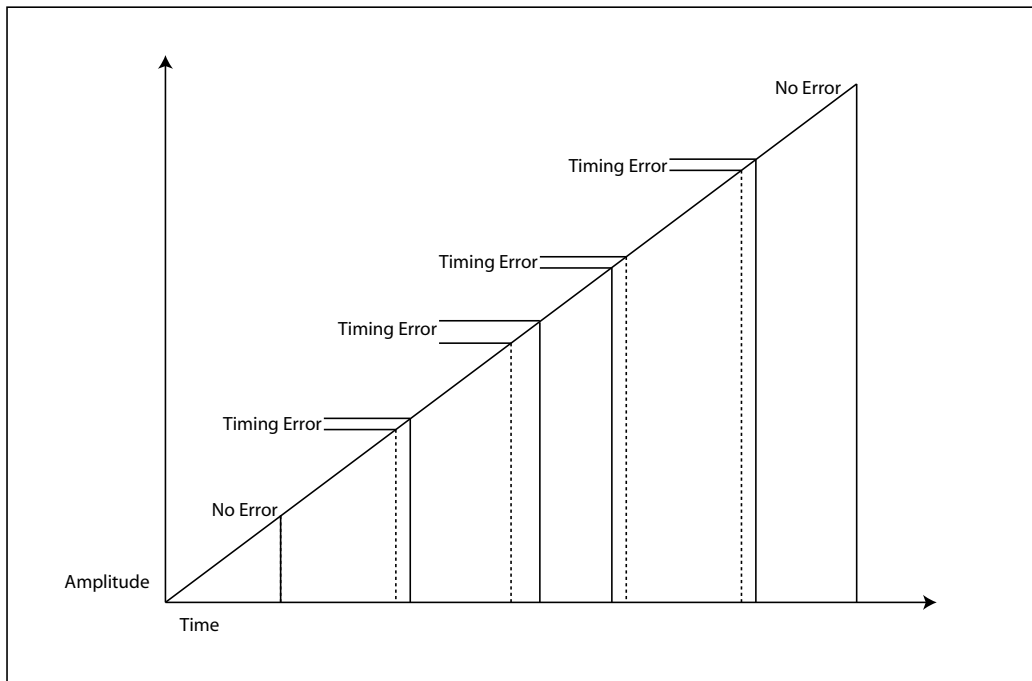


Figure 3: Sampled partial waveform

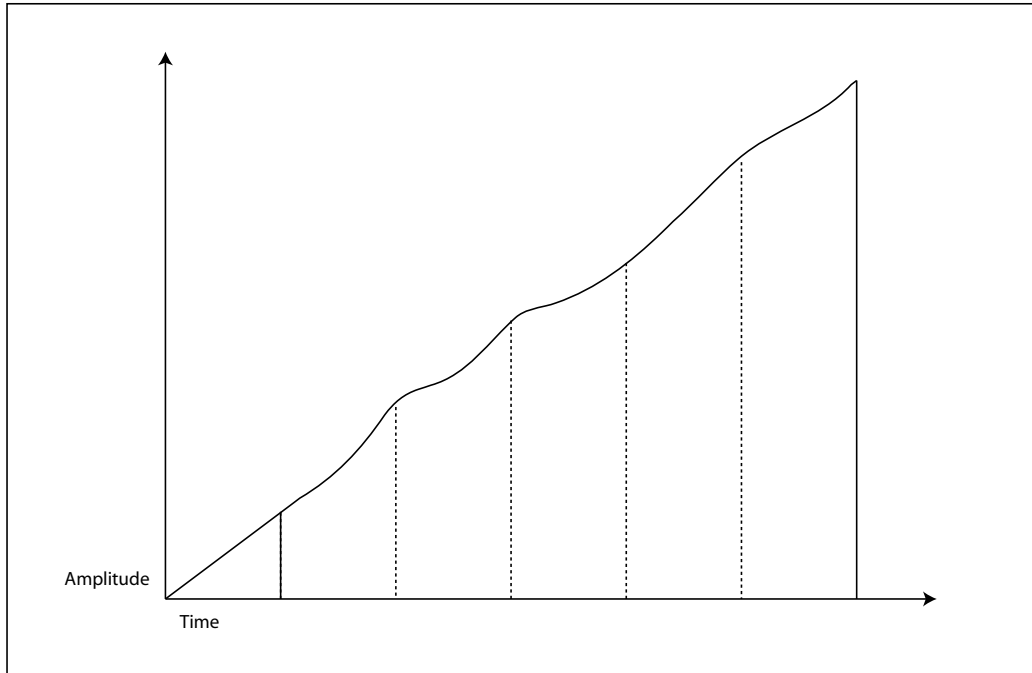


Figure 4: Reconstructed partial waveform sampled with jitter

Unlike random jitter, *periodic jitter* is that which occurs at regular intervals. Its effect on audio is to cause sidebands to occur at specific frequencies that are mathematically related to the fundamental frequency of the recorded signal. This type of distortion is called *frequency modulation* because the frequency of the jitter interacts with the source audio and the resulting noise is based upon that interaction. Since both the audio signal and jitter will vary over time (unless you're recording stable tones with fixed jitter) frequency modulated noise is somewhat unpredictable, but the relationship between the recorded audio and jitter can be clearly identified. For example, a 5 kHz signal recorded with 1 kHz jitter would produce sidebands at 4 kHz and 6 kHz in addition to the fundamental of 5 kHz. The amplitude of the sidebands is proportionally related to the amplitude of the recorded signal and jitter. FM distortion can produce sum and difference tones, which then can produce sum and difference tones of their own, very quickly creating complex distortion characteristics. Periodic jitter can be caused by the sample clock being modulated by a power supply or other sources which produce steady state tones.

The interaction between jitter and audio is complex and makes analyzing the audible result of jitter difficult. This is because some jitter artifacts are masked by the audio content, while others stand out and can be heard clearly at certain frequencies and levels. Listening tests on equipment with varying amounts of jitter have even resulted in a *positive* subjective reaction to those with large amounts of audible jitter artifacts, presumably because the jitter artifacts have a euphonic effect on the material. While a subjective discussion is not feasible in this forum, we will attempt to provide some information about clocking and jitter to help the reader better identify for themselves what they hear and how to obtain the best result from today's generation of equipment.

OSCILLATORS

There are several types of oscillators used to construct clocks used in digital audio interfaces and their characteristics make them better suited for different applications.

Crystal Oscillators or XOs

In these oscillators, a tiny fragment of quartz is excited with electricity to the point that it resonates at a frequency which is determined by the size and orientation of the crystal. Once the oscillator circuit stabilizes, crystals resonate at a very stable frequency and their accuracy is generally in the 100 ppm (parts per million) range. That means that a crystal with a nominal frequency of 80 MHz might resonate at anywhere between 79.99200 and 80.00800 MHz. Crystals are somewhat

susceptible to variations in temperature and are sometimes enclosed in an insulating capsule or even a tiny oven that maintains a stable ambient temperature. These devices are called *TCXOs* (*Temperature Compensated Crystal Oscillators*) or *OCXOs* (*Oven Controlled Crystal Oscillators*). Due to their stability, they are most commonly used as sampling clocks.

Voltage Controlled Crystal Oscillators or VCXOs

This is a variant of the XO where changes in a voltage applied to elements of the crystal oscillator circuit modify its resonant frequency. These oscillators have the benefit of the stability of the crystal design and the ability to "pull" the resonant frequency of the crystal up or down. One drawback of VCXOs is that they only operate over a very narrow frequency range. That property generally precludes them from use in circuits that need to slave to external clock sources whose frequency can vary widely.

They are therefore best suited for sampling clocks that are not required to operate over a wide frequency range.

Voltage Controlled Oscillators or VCOs

A VCO is an oscillator whose frequency can be adjusted by applying a suitable control voltage. They are differentiated from a VCXO in that they do not use a quartz crystal as the frequency determining network. VCOs are quite common in digital audio interfaces. They are capable of providing a range of frequencies by changing the voltage in a feedback circuit that modifies an oscillator. If properly designed, they can be used to make very good clocks.

The Digidesign 192 I/O uses a VCO slaved to an XO in order to generate its master internal clock.

Numerically Controlled Oscillators—NCOs

This type of oscillator is a circuit that uses a fixed-frequency oscillator, such as an XO, then uses a digital circuit to produce an output clock at the desired frequency. Lookup tables are used to generate the dividing coefficients instead of a feedback circuit. NCOs are extremely well suited for applications that require variable clocking rates, such as devices that slave to time code or video reference.

In some NCOs, a digital phase locked loop (PLL) which models the functions of a phase comparator and loop filter is also used. A lookup table and digital-to-analog converter (DAC) are then used to convert the NCO output into a sine wave. This type of NCO is referred to as a Direct Digital Synthesizer or DDS, and is commonly used in radios and telecommunications equipment.

The Digidesign SYNC I/O uses a DDS to produce its master internal clock.

CLOCKS

Sampling Clock

The most commonly used audio converters are the 256x Delta Sigma type. They take low resolution snapshots at 256 times the audio sample rate and then use on-chip digital interpolation filters to convert them into 24-bit audio words at the desired “base rate” (word clock). The base rate is the speed at which samples are processed through the system, and are the common rates of 44.1 kHz up to 192 kHz.

Word Clock

Word clock is a term used to describe the base rate of the digital system. The frequency of the word clock is the rate at which packets of audio or multi-bit samples are passed through the system.

PHASE LOCKED LOOPS

Another common building block in digital clocking systems is the *Phase Locked Loop* or *PLL*. The basic function of a PLL is to synchronize an oscillator’s output with its input. They are commonly used in for three functions in digital audio systems: *alignment*, *timing*, and in *clock synthesizers*.

Alignment, or clock recovery interface PLLs, are used in digital interface receiver clock recovery circuits to track whatever jitter is present on a digital interface in order to clock in data on that interface. Alignment PLLs are also used in jitter measurement circuits such as the “Meitner LIM-1” described in references at the end of this paper.

Timing or jitter attenuating PLLs are used to reshape the spectrum of jitter present in a clock source and help manage the jitter budget by shifting noise outside the audible band.

Synthesizer PLLs are used to generate clocks which are multiples of the input reference clocks. These are most commonly used for multiplying a wordclock rate up to a higher frequency for converters or digital signal processing circuitry. Synthesizers are also used for generating clocks which run at different ratios of speeds such as for locking audio to video equipment in “genlock.”

By utilizing a feedback circuit and low pass filters, the input and output of the PLL are compared to each other and the difference is used to speed up or slow down a VCO to match the incoming clock source. That source can be a crystal oscillator or reference clock from an external device. The primary design challenge with PLLs is in creating low pass filters that reshape the clock in such a way that stabilizes a clock’s pulse intervals and yet allows it to track frequency fluctuations at its input. PLLs are a critical component in synchronizing digital devices to video black burst, word clock, timecode or other timebase references.

The other design goal for low pass filters is to attenuate jitter on an incoming clock so that the receiver can produce a clean version of it. Entire books have been written on PLL and filter design and it is a very deep and involved subject. The point of their mention here is to alert the user to their presence and purpose in digital audio peripherals.

The PLL works by comparing the phase of an incoming clock with that of outgoing clock pulses. If the two are perfectly aligned and trigger pulses are synchronized, the PLL does nothing. If, however, there is a difference between the incoming and outgoing pulses, the PLL adjusts the oscillator to either speed up or slow down so as to remain synchronized with the incoming reference clock.

As the output of the phase comparator passes through a low-pass filter, the filter removes noise and stabilizes the overall circuit. The design of the low-pass filter is a balance between the speed at which the circuit responds to out-of-phase conditions i.e. how quickly phase coherency is restored, and the noise rejection of the system i.e. the clock stability/accuracy. The balance between these goals has a direct effect on the system’s ability to lock to and track external clock signals quickly. There is a real trade off between fast lock time and low jitter.

The basic building blocks common to all PLLs are shown in Figure 5 (below). The reference clock signal is applied to a phase comparator. The comparator compares the phase of the reference clock with the PLL’s output, and generates an error signal proportional to that phase difference. That error

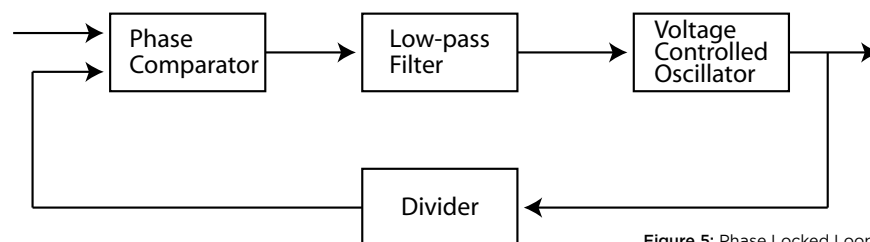


Figure 5: Phase Locked Loop

signal is then amplified and filtered by the low pass loop filter, and the result is applied to the VCO's control voltage input. An optional clock divider may be interposed between the VCO output and the phase detector input. This configuration will then generate an output clock whose frequency is a multiple of the reference frequency. This is very common in digital audio applications where a 256x oversampling clock needs to be generated from a word clock reference.

THE DIGIDESIGN HD CLOCKING SCHEME

All Digidesign audio interface peripherals in the HD system are modular, meaning they are designed to operate in single unit as well as in multi-unit configurations. Clocking in the system is also configurable, depending on the number of interfaces and desired application. Loop Sync is Digidesign's standard method for clocking multiple interfaces, however it is also possible to use an external clock and a "star" configuration to distribute the clock to interfaces individually. Slaving to external sources such as video reference and LTC also affects one's clocking scheme. So, let's take a look at these options.

CLOCK OPTIONS AND SELECTION

In a Digidesign 192 I/O, master reference clock signals can originate from several sources: the internal crystals, Loop Sync, External clock, or any one of the pairs of digital inputs on each of the card slots (Figure 6 below). Input sources are selected via software control from the Session Setup window or Hardware Setup dialog in Pro Tools. The software controls a switch that selects one of the available sources as the base frequency reference. This source is referred to as the Master

The Master reference clock is passed on to a frequency synthesizer which contains a very low jitter VCO and multiple clock dividers. The reference input to the synthesizer is always a 1x word clock. The synthesizer multiplies this 1x clock by 1024 to produce a stable sampling clock. A clock divider uses the 1024x clock to produce all of the internal timing signals, including a 1x word clock for use as the base reference frequency for the rest of the system.

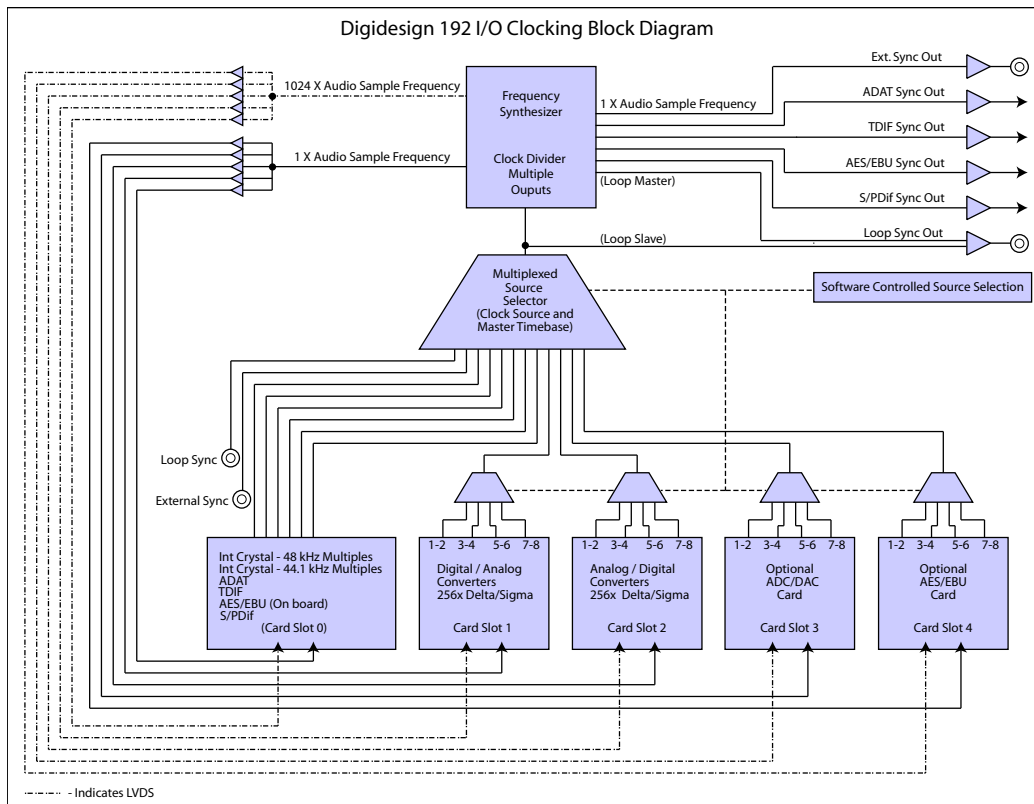


Figure 6: Digidesign 192 I/O Clock Distribution Scheme

The high speed sampling clock is delivered to the converter slots via a *low voltage differential* signal path (LVDS) in order to maintain its integrity and reject potential noise or electromagnetic interference (EMI). Each slot has a separately buffered feed for both sample clock and word clock which deliver the clocks to the slots with negligible degradation.

The 1x word clock is used to signify the start of each audio sample (word) and also synchronizes the output of the ADCs, DACs and digital interfaces and determines when to deliver PCM samples to the rest of the system.

We'll now follow the various clock selections through the system and discuss the results of clocking choices on the amount of jitter in the system.

INTERNAL

When Internal is chosen as the clock source for the system, one of the crystal oscillators (48 kHz shown in Figure 7) is selected as the master reference. When selected, this oscillator's output is passed on to the frequency synthesizer via the source selector. At the synthesizer, the word clock frequency of the oscillator is multiplied by 1024 to create the sample clock that drives the converters and is then divided back down

to provide the 1x word clock for PCM operations and external clock sources.

Internal clock provides the lowest possible jitter in a single interface system. The average jitter measured in a single interface clocked internally is approximately 63 picoseconds. The spectrum is relatively flat from 5 kHz on up. (See figures 8 and 9)

Pro Tools systems with a single interface that are not required to lock to external sources such as timecode or video reference have the lowest amount of jitter when set to Internal.

Note: The measurements cited throughout this paper were made using a Meitner LIM-1 and an Audio Precision 2700 series analyzer. The LIM-1 is a specialized device for measuring jitter that contains two loop filters outside of its PLL. One loop filter is extremely slow reacting (8 Hz) and is used to create a stable reference frequency. The secondary, fast loop filter (80 kHz) is used to track variations on incoming signals against the reference clock. The output is converted to voltages which are recorded on the Audio Precision as FFTs where timebase variations are shown as a function of voltage. 1 millivolt on the vertical axis indicates 100 picoseconds of jitter. So, 10 microvolts indicates 1 picosecond of jitter and so forth.

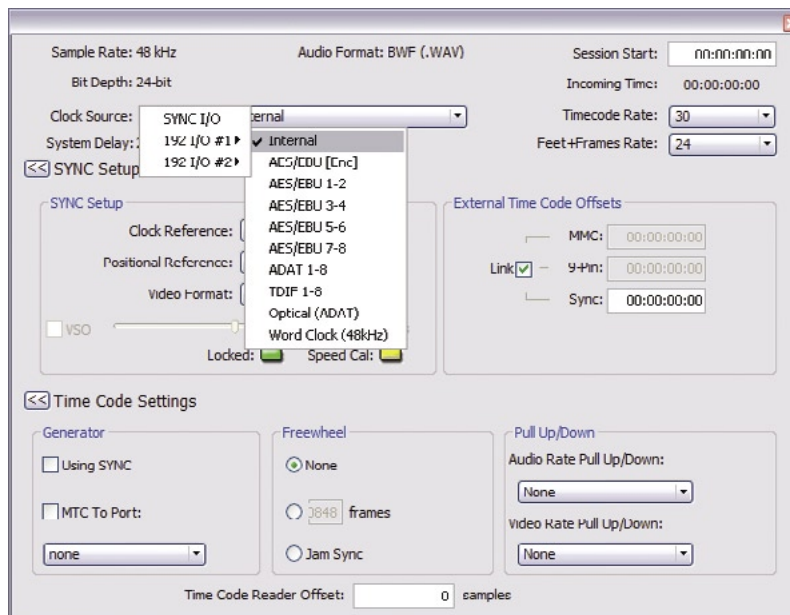


Figure 7: Pro Tools Session Setup Window, 192 I/O Internal

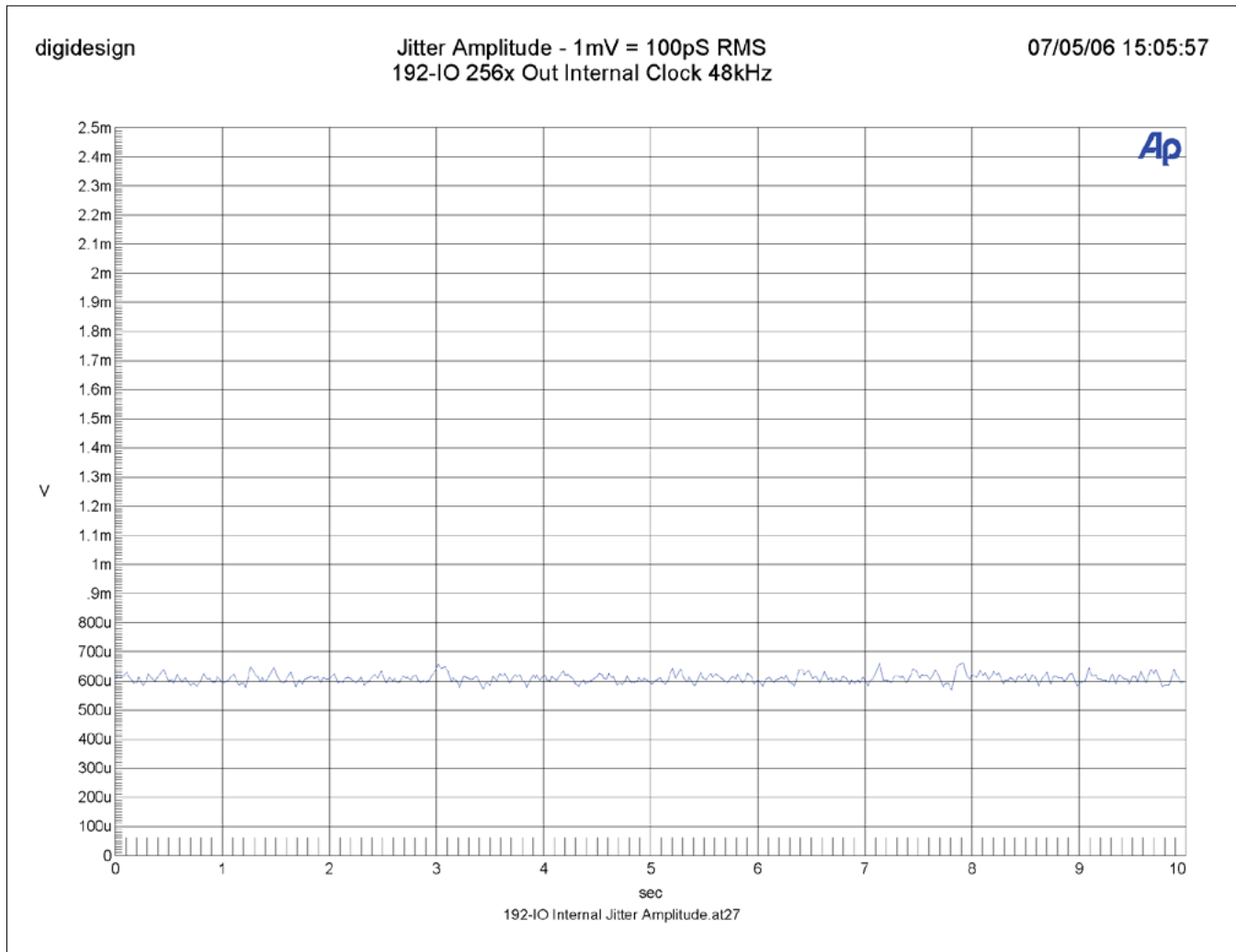


Figure 8: 192 I/O Jitter Amplitude—Internal Clock as Master

Two types of measurements are used throughout this paper—1) *jitter amplitude*, which is a timed test of average jitter amount and 2) *jitter spectrum*, which shows the amount of jitter across the frequency spectrum of 20 Hz to 20 kHz.

Figure 8 shows the jitter amplitude in a 192 I/O measured over 10 seconds. This is the average amount of jitter across all frequencies for that duration. The vertical scale ranges from 0 to 2.5 millivolts which translates to a range of 0 to 250 picoseconds. The average across the 10 second period hovers a little over 600 microvolts or approximately 63 picoseconds.

Figure 9 shows the jitter spectrum of a 192 I/O using internal clock as its master. As opposed to jitter amplitude measurements

which show average amount of jitter over time, spectrum analysis shows the jitter amount at given frequencies for a brief window of time. The amplitude axis in this figure shows a range of 0 to 1 millivolt (1000 microvolts) or 0 to 100 picoseconds. You can see that jitter between 4.5 kHz and 20 kHz is roughly 0.8 to 1 picosecond with an increase at frequencies below 4 kHz. (You'll start to recognize a pattern of higher amounts of low frequency jitter in many of the following figures.) Low frequency jitter components are believed to be less audible than higher frequencies and it is common to find higher amounts of low frequency jitter in many audio devices.

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Jitter Spectrum - 1mV = 100pS RMS
192-IO 256X Out Internal Clock 48kHz

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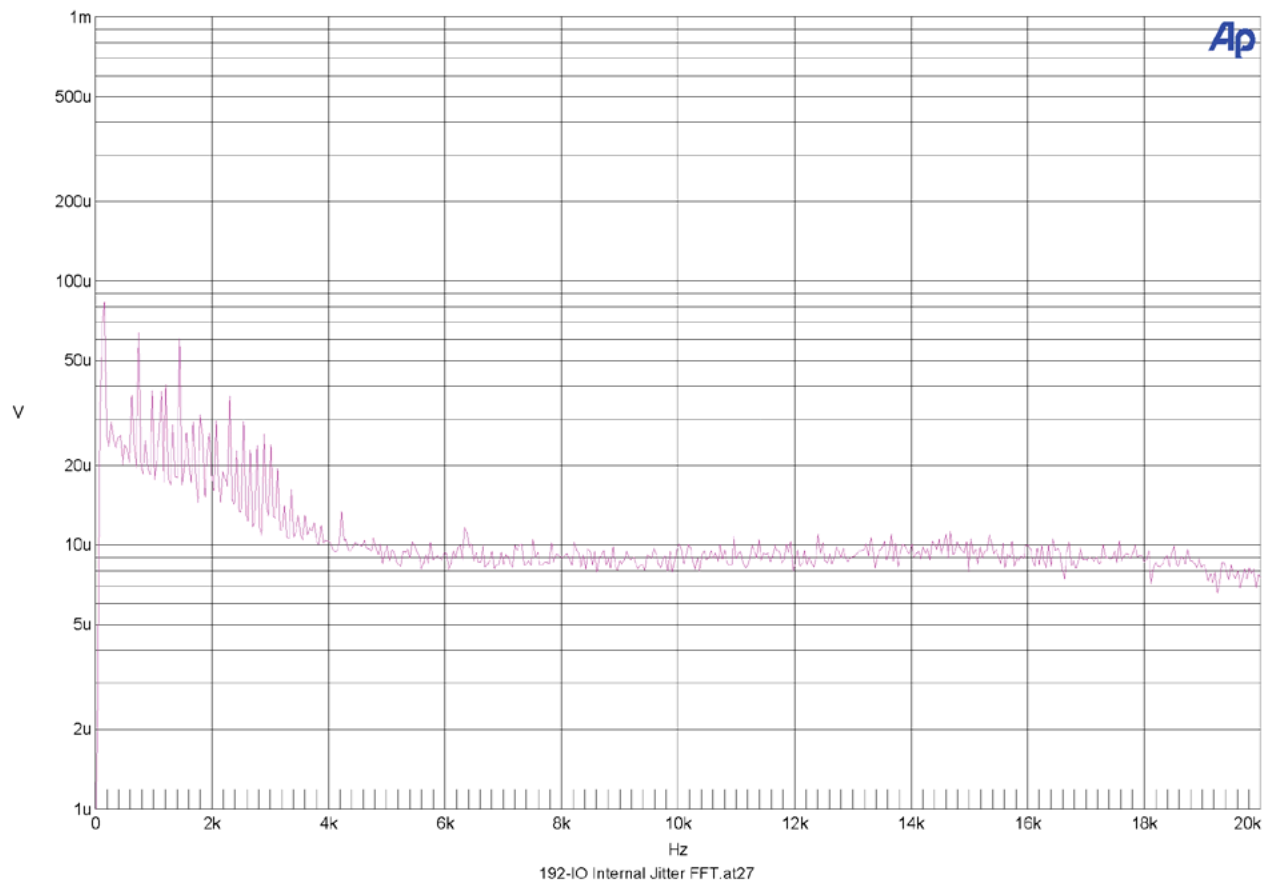


Figure 9: 192 I/O Jitter Spectrum—Internal Clock as Master

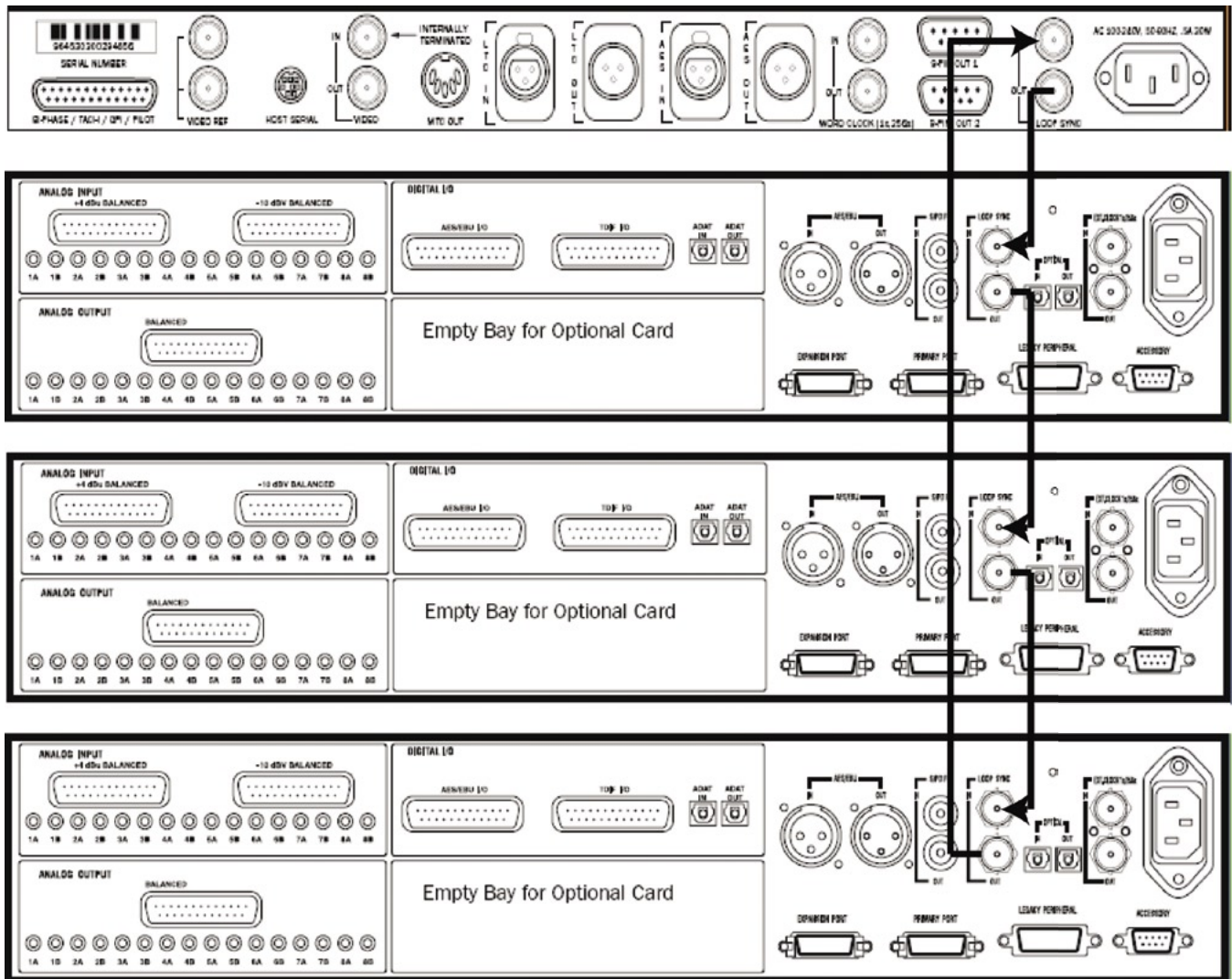


Figure 10: Loop Sync Connections

LOOP SYNC

Loop Sync is unique to the Pro Tools HD series interfaces and used to synchronize multiple modular interfaces. Its most distinct feature is that it enables any of the members in the system to become the Loop Master. The Loop Master is the HD peripheral whose clock is used for all other devices that have Loop Sync selected (Loop Slaves). The interface that is declared as the Loop Master issues a clock from the frequency synthesizer to its Loop Sync Out connector. All other units are Loop Slaves and 1x word clock is passed from interface to interface via the Loop Sync cables. At each Loop Slave

interface, the clock is duplicated at the source selector and a non-PLL'd duplicate is passed to its Loop Sync Out connector while the other duplicate is used as the Master reference for that unit. By maintaining non-PLL'd loops throughout the system, each member of the loop gets essentially the same clock as its input. And, because each unit utilizes its own synthesizer on what is nearly a first generation clock, overall system jitter is kept to a minimum.

The advantage of the Loop Sync system is that besides not requiring a clock distribution amplifier, any member of the loop can function as the master without having to re-patch

clock cables, so it is a very useful feature in systems that are required to lock to multiple digital sources. Additionally, because the entire system is under software control, the sample frequency and clock source follows the session. With Loop Sync in use, it is nearly impossible to mismatch the session's sample rate and the hardware's sample rate.

To the source selector, Loop Sync is simply another choice of input and is routed internally the same as all other sync sources. However, the amount of jitter in the system is increased slightly at the buffer amplifiers of each loop output, which contributes to an increase of jitter in systems with multiple interfaces.

SYNC I/O

In cases where the system is resolved to external sources such as video reference, word clock or LTC, a Digidesign SYNC I/O is often used as the Loop Master. By means of a frequency synthesizer as described earlier, SYNC I/O provides a very low-jitter clock even while locked to external sources and providing positional reference. Figure 11 shows the effect of a SYNC I/O Loop Master on a 192 Loop Slave.

The high frequency jitter is nearly identical and the increase in average amount is driven primarily by low frequencies where jitter reduction is less strong due to the need to track other devices. The average amount jitter shown here increases from 77 picoseconds to just under 180 picoseconds.

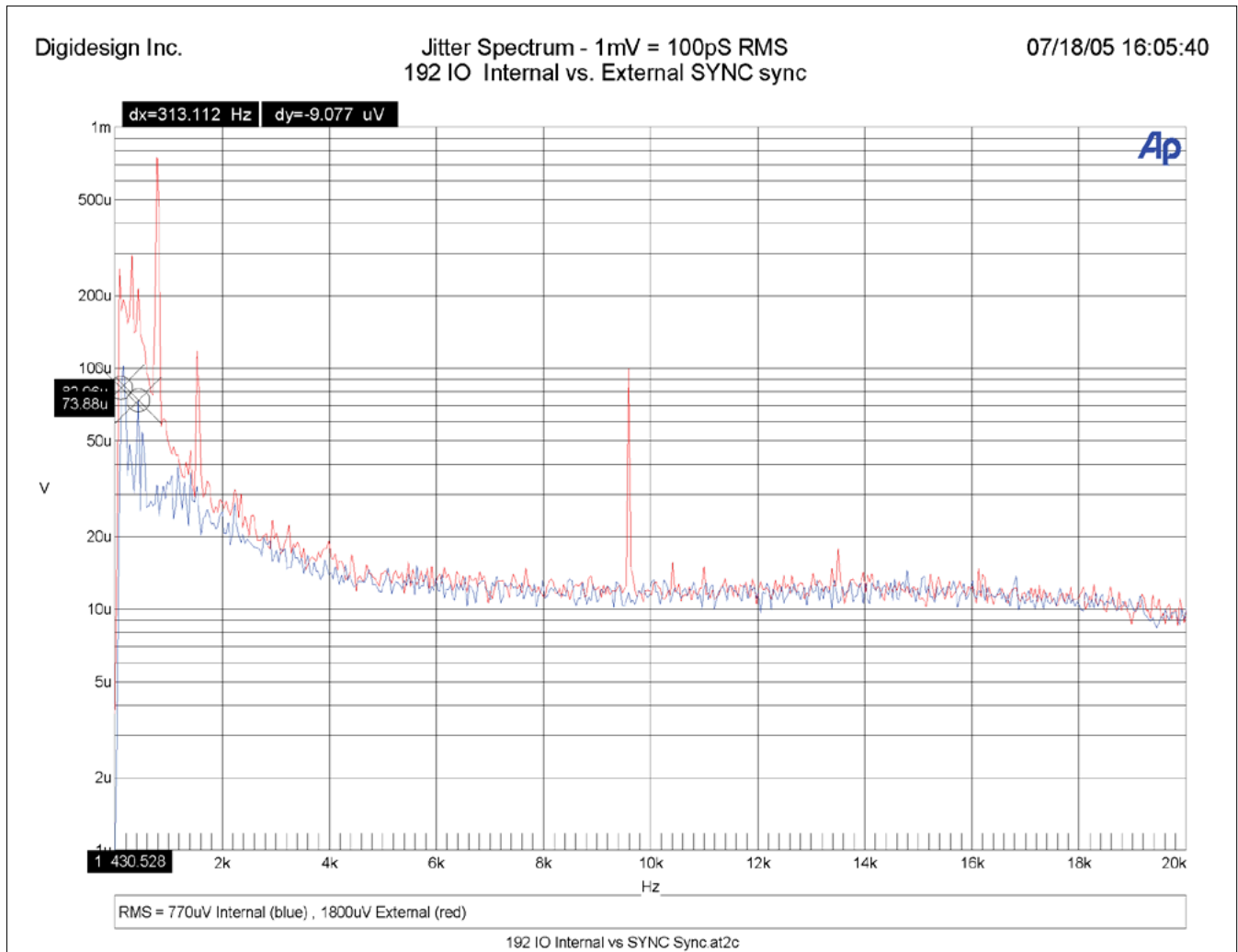


Figure 11: 192 Loop Slaved to SYNC I/O Loop Master while locked to video

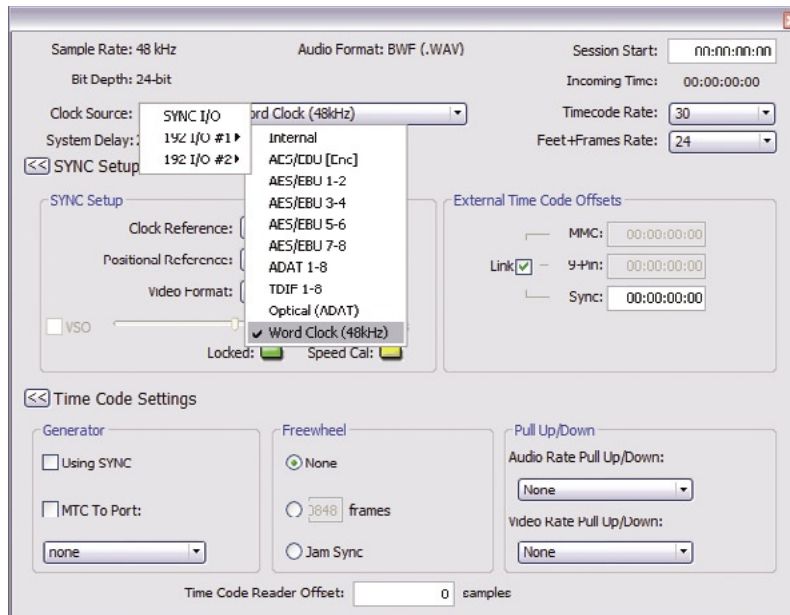


Figure 12: 192 I/O—External Clock Selected

EXTERNAL CLOCKING

When External clock or Word Clock is chosen as the clock source for the system, the clock signal is derived from the input connector on the rear panel of the 192 labeled “Ext Clock 1x/256x”. (The Loop Sync In connector can be used at 44.1 kHz and 48 kHz as long as “Internal” is selected as Clock source. At higher sample rates, the “Ext Clock 1x/256x” connector must be used.) As with Loop Sync, word clock is passed on to the clock synthesizer via the source selector.

As with internal sources, external clocks pass through the 192 I/O’s internal synthesizer and are modified by the spectrum of its PLL. All clock sources—internal and external—are handled identically. High frequency jitter is attenuated and shaped by the 192’s PLL loop filter such that above the filter’s corner frequency (approx. 5 kHz), the signature of the synthesizer will dominate the external clock. Low frequency jitter is attenuated less and will generally be higher than internal clock.

Some dedicated clocks use non-PLL’ed XO’s and can have lower amounts of jitter compared to the 192’s PLL’ed internal clock. But since all sources pass through the 192’s PLL, all clocks assume a similar spectrum above the loop filter corner frequency. Below the corner frequency less jitter attenuation occurs, so if the external clock has more low frequency jitter, it will contribute to higher jitter in the 192 in those frequencies. Figure 13 shows the jitter spectrum of a popular dedicated clock.

Though this clock has approximately the same average amount of jitter as a 192 on internal, this clock has a different spectrum with some large high frequency spikes and low frequency rise—hardly *revolutionary* as their marketing material would indicate.

When used to clock a 192 interface, the clock is ‘scrubbed’ by the 192’s PLL and the result is shown in Figure 14.

Figure 14 shows two traces indicating the 192 on internal sync and using Clock “A” as external master. The RMS average jitter increases slightly from 73 picoseconds when clocked internally to 76 picoseconds when clocked externally. The high frequency spectrum is quite similar—notice the high frequency spikes that were present in the raw output have been attenuated. Most of the change occurs in the lower frequencies where jitter attenuation is less strong.

It is important to note that most commercially available dedicated clocks use non-PLL’ed XO’s to produce low-jitter outputs. Since these are direct outputs from the crystals, they typically boast very low jitter specs which are quoted in marketing materials. However, when these devices are locked to video or word clock, they must engage a PLL which often has much worse characteristics. For example, Figure 15 shows a Clock “A’s” output when resolved to internal sync (crystal) and then to word clock. Internal sync shows an average of less than 70 picoseconds while externally synced to word clock, it increases to over 190 picoseconds.

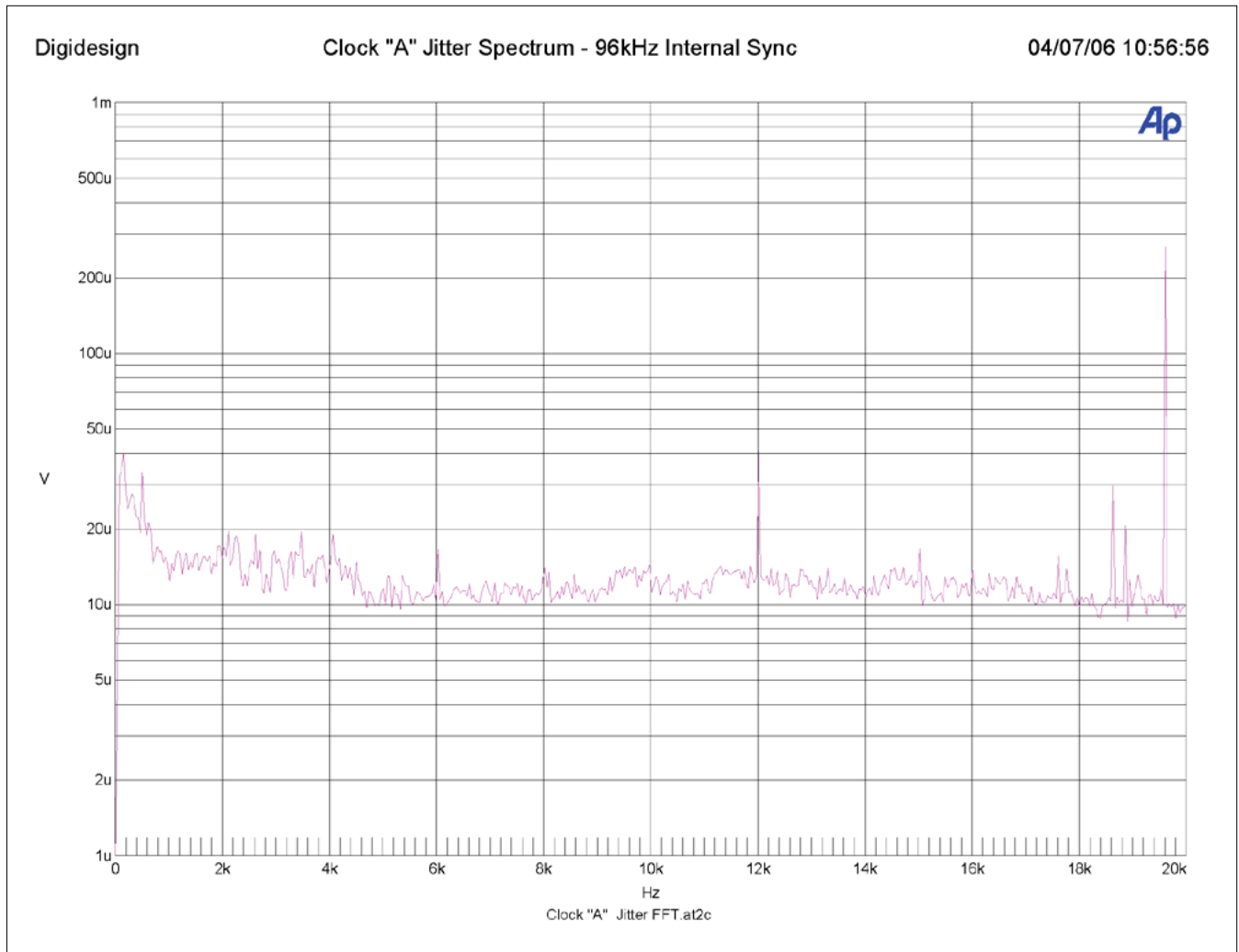


Figure 13: Clock "A" jitter spectrum

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Jitter Spectrum - 1mV = 100pS RMS
192 IO Internal vs. External Clock "A" Sync

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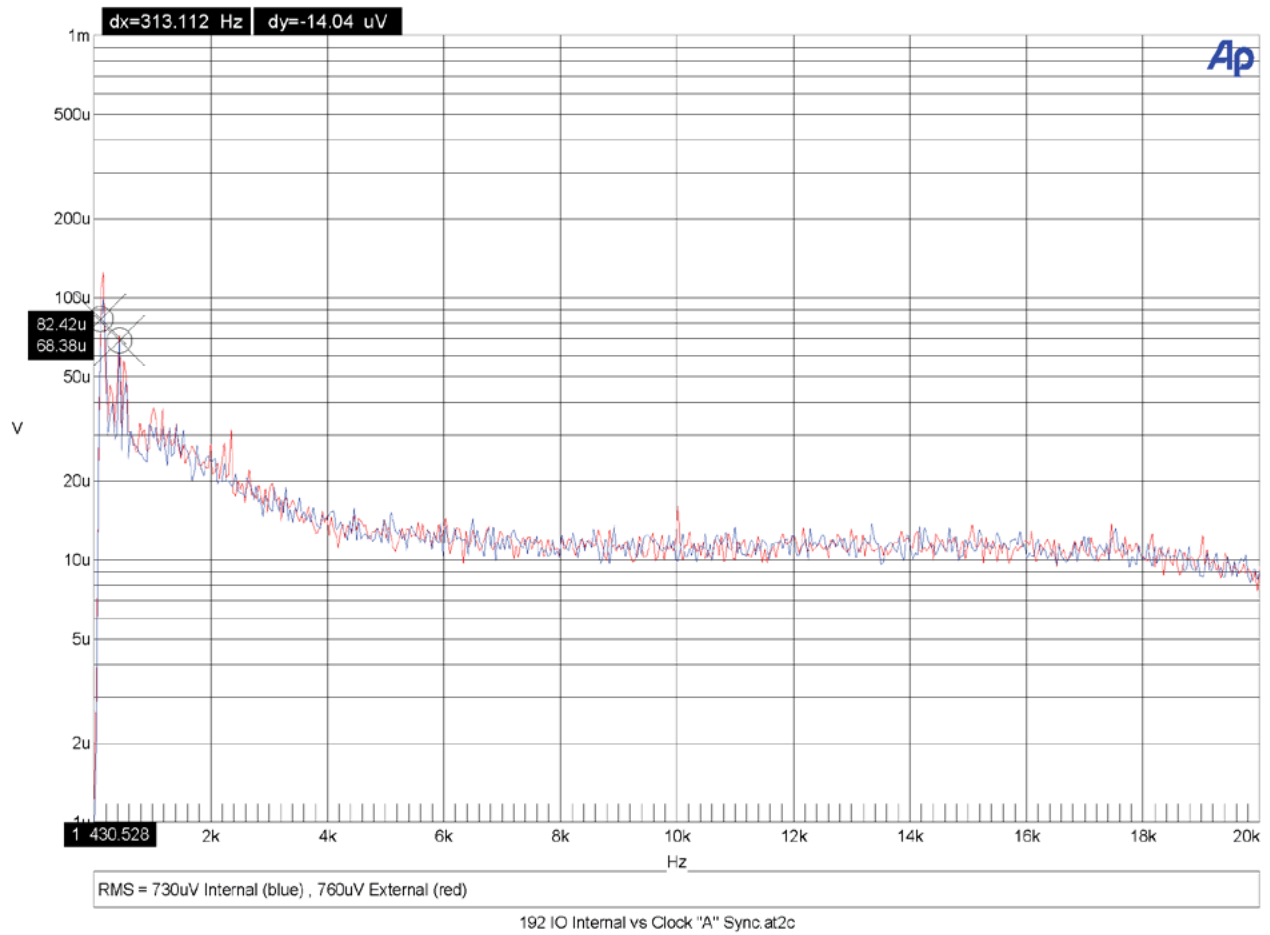


Figure 14: 192 I/O with Internal clock vs. External Clock "A"

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Jitter Amplitude - 1mV = 100pS RMS
Clock "A" Internal & External Clock 48kHz

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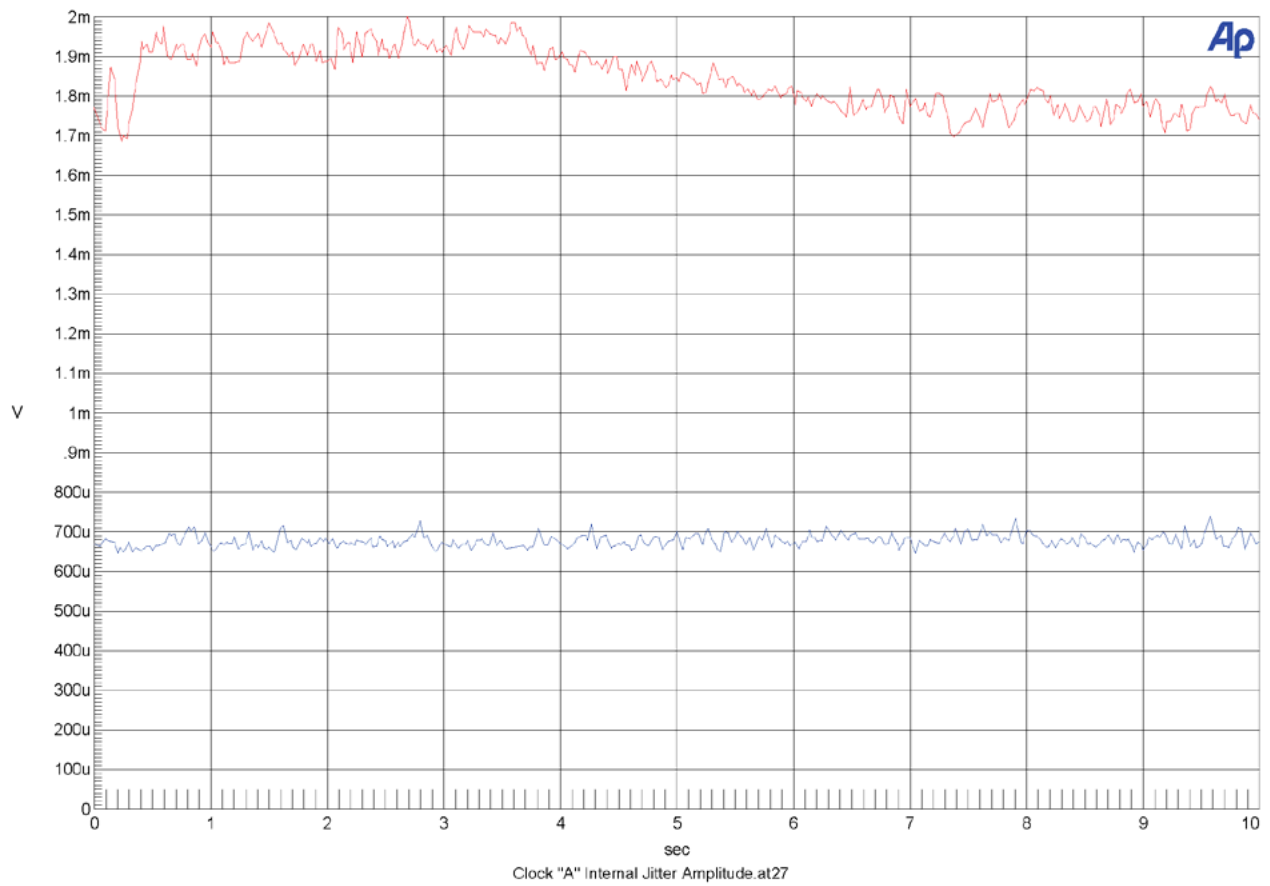
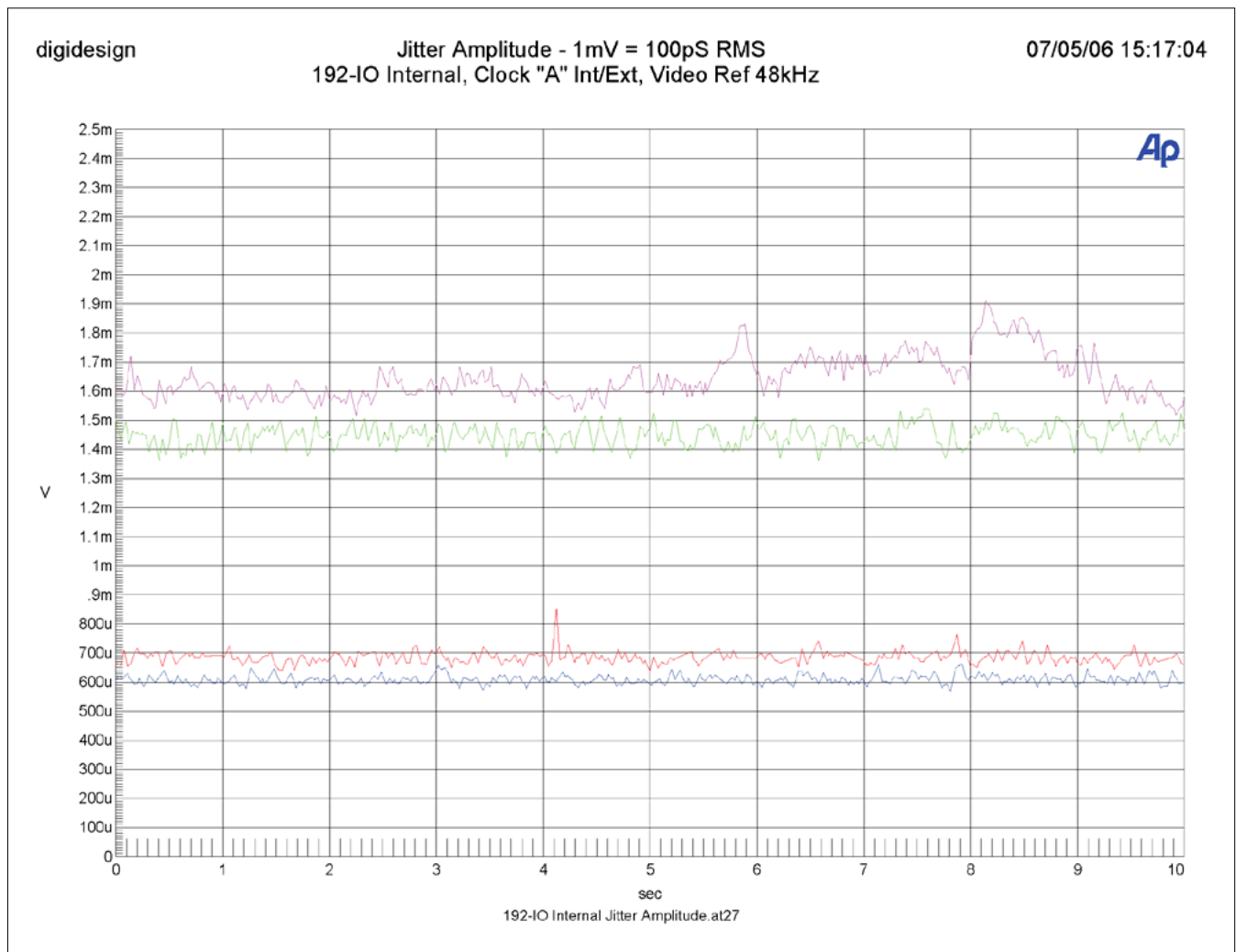


Figure 15: Clock "A" Internal vs. External sync

The increase in jitter is, of course, passed on to the rest of the system as is shown in Figure 16. The blue trace represents a 192 on internal sync—average amount of jitter is around 62 picoseconds. The red trace shows the 192 on external sync, using Clock “A” on internal sync—average jitter is roughly 68 picoseconds. The green trace shows the 192 on external sync with the same clock resolved to word clock—average jitter is now 145 picoseconds. And finally the purple trace shows the 192 on external sync with the Clock “A” locked to video—the average jitter is 190 picoseconds.

The 192's PLL is very effective in reducing the amount of high frequency jitter and increases in jitter amplitude occur predominantly in low frequencies. Figure 17 shows the spectrum of Clock “A” while locked to blackburst and Figure 18 shows the output of the 192 while referencing that clock. The amount of reduction in the high frequencies is quite dramatic and illustrates the quality of the 192's PLL design.



Blue= Internal
Red= External with clock on Internal
Green= External with clock locked to Word Clock
Purple= External with clock locked to Video

Figure 16: 192 I/O Average Jitter

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Jitter Spectrum - 1mV = 100pS RMS
Clock "A" Locked to Video Ref 48kHz

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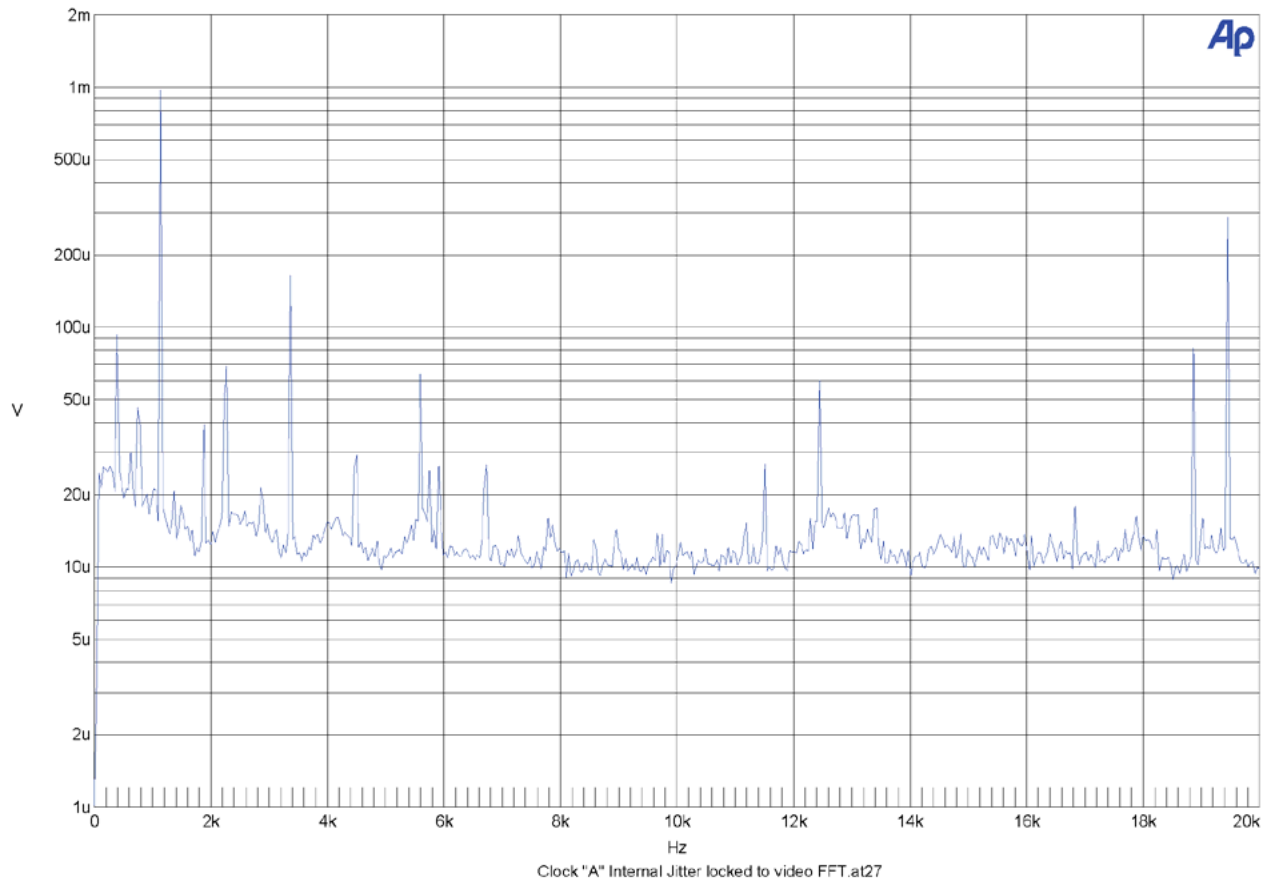


Figure 17: Clock "A" Internal Jitter—Locked to Blackburst

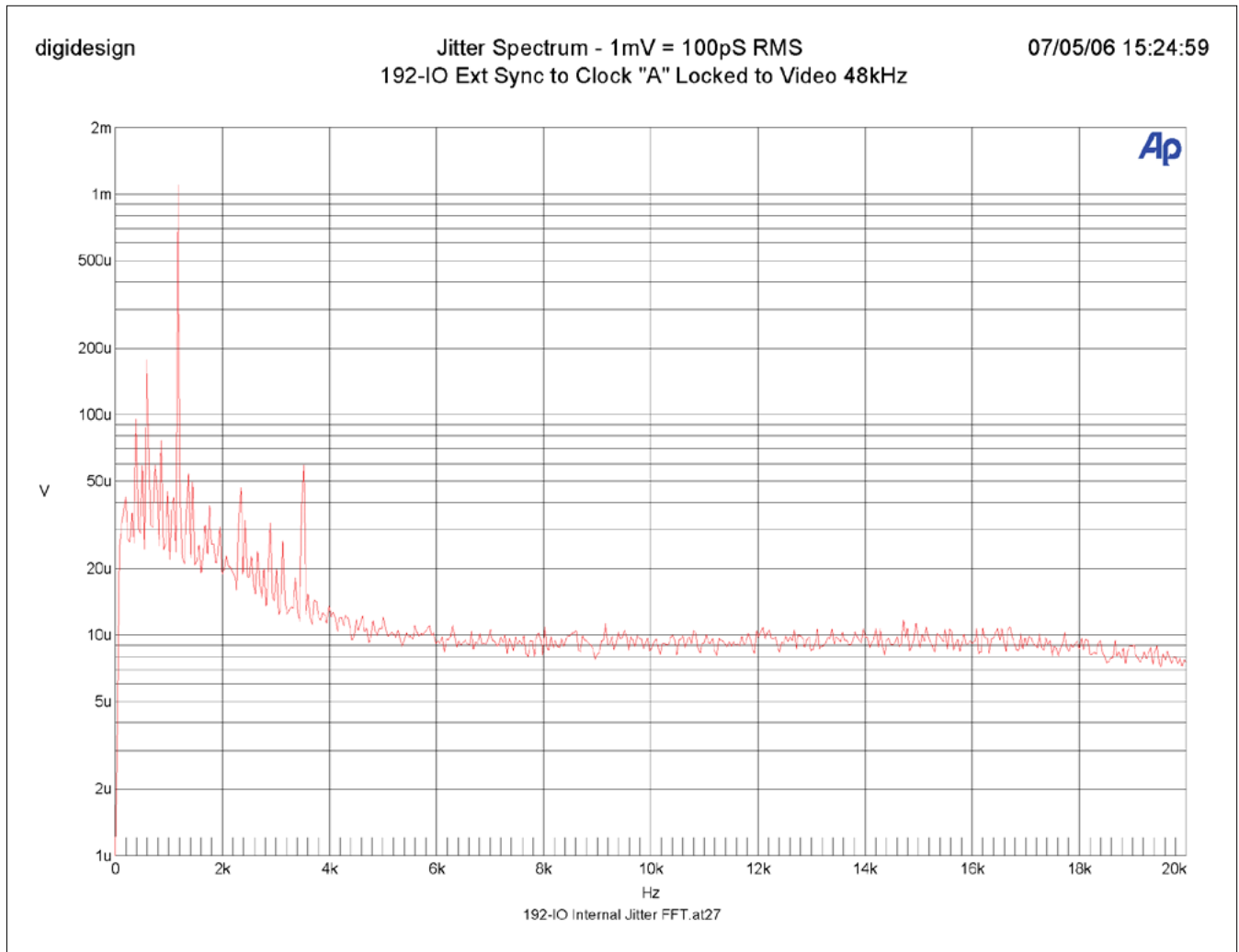


Figure 18: 192 I/O Clock referencing Clock "A" while locked to blackburst

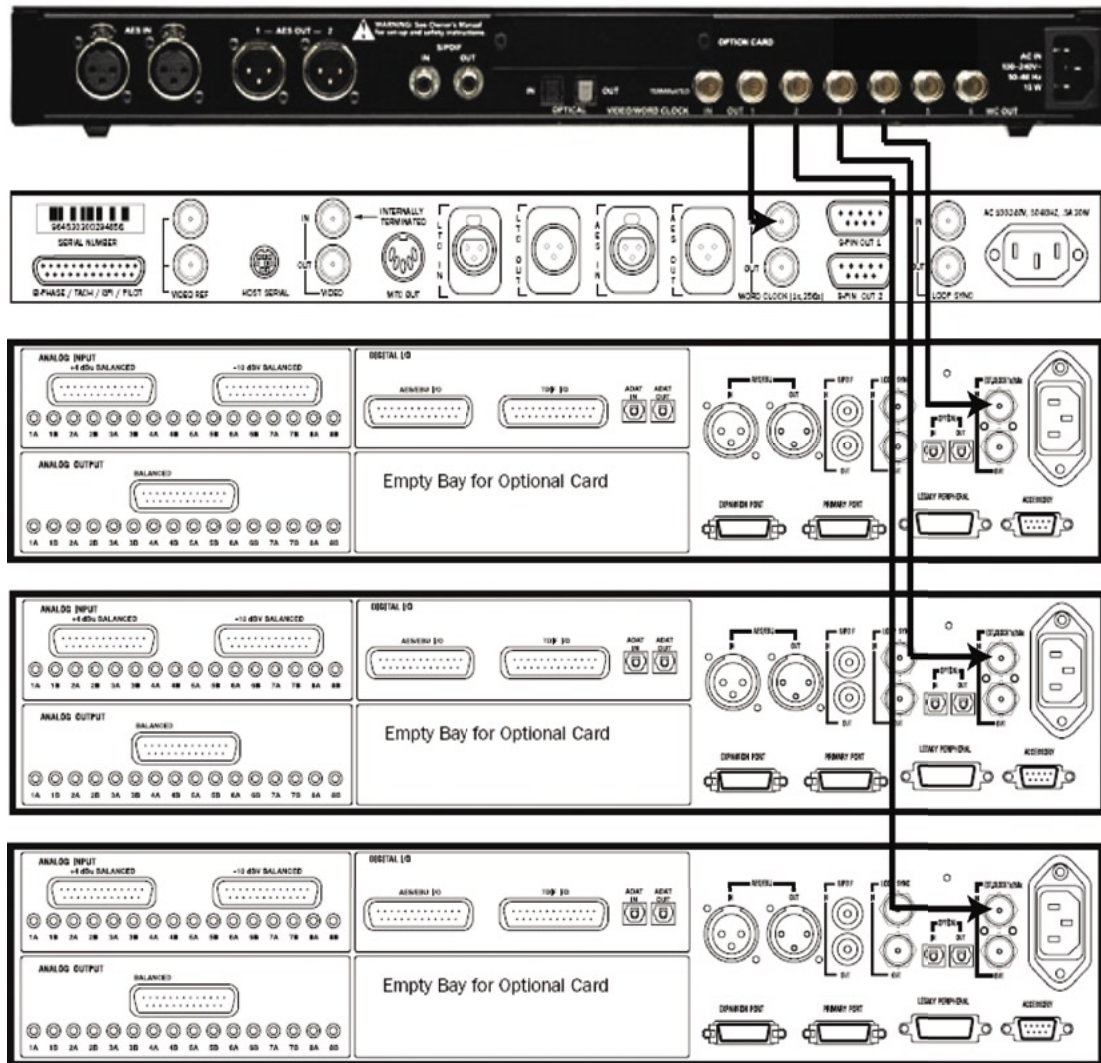


Figure 19: Distributed Clock Connection 1

DISTRIBUTED CLOCK

A popular method of clocking systems with multiple interfaces is to use an external clock with a distribution amplifier to send master clock signals to each interface individually.

In this topology, the clock arrives at each device directly from the DA so the cumulative effect of multiple source selectors and buffers is avoided. Although the amount of jitter in each interface is higher than if it were clocked internally, it is possible to reduce the amount of system-wide jitter by using a distributed clock topology. However, this is only the case

in systems with several interfaces and is completely dependent on the amount of jitter present in the master clock. The distributed clock system has the advantage of keeping jitter even across all interfaces, but is more prone to sample rate mismatches and configuration problems.

In this scheme, the user must manage the sample rate of the interfaces manually to ensure the session sample rate and interfaces are operating at the same frequency. If they do not match, the session will play back at the wrong pitch when played on a system with the correct clocking scheme. This scheme requires additional connections when locking

to video reference because though the clock may be resolved to blackburst or word clock, the system also requires positional information to be supplied from another source such as SYNC I/O or a SMPTE to MIDI interface.

When locking to external sources such as LTC or bi-phase, other connections are required. The SYNC I/O can be used to resolve the system and its word clock can be sent to the external clock's input to be 'scrubbed'. The system is resolved via the SYNC I/O and the externally generated clock is distributed to the interfaces via a star topology. (See Figure 20) The

benefit of this configuration is that the SYNC I/O interfaces directly with Pro Tools and sets the sample rate and time code type via the session settings.

As mentioned earlier however, externally syncing XO based devices has the potential to degrade their performance so one should consider this when locking to external sources. It would be best to consult the manufacturer on the performance of their devices when clocked externally vs. internally and follow their recommendation.

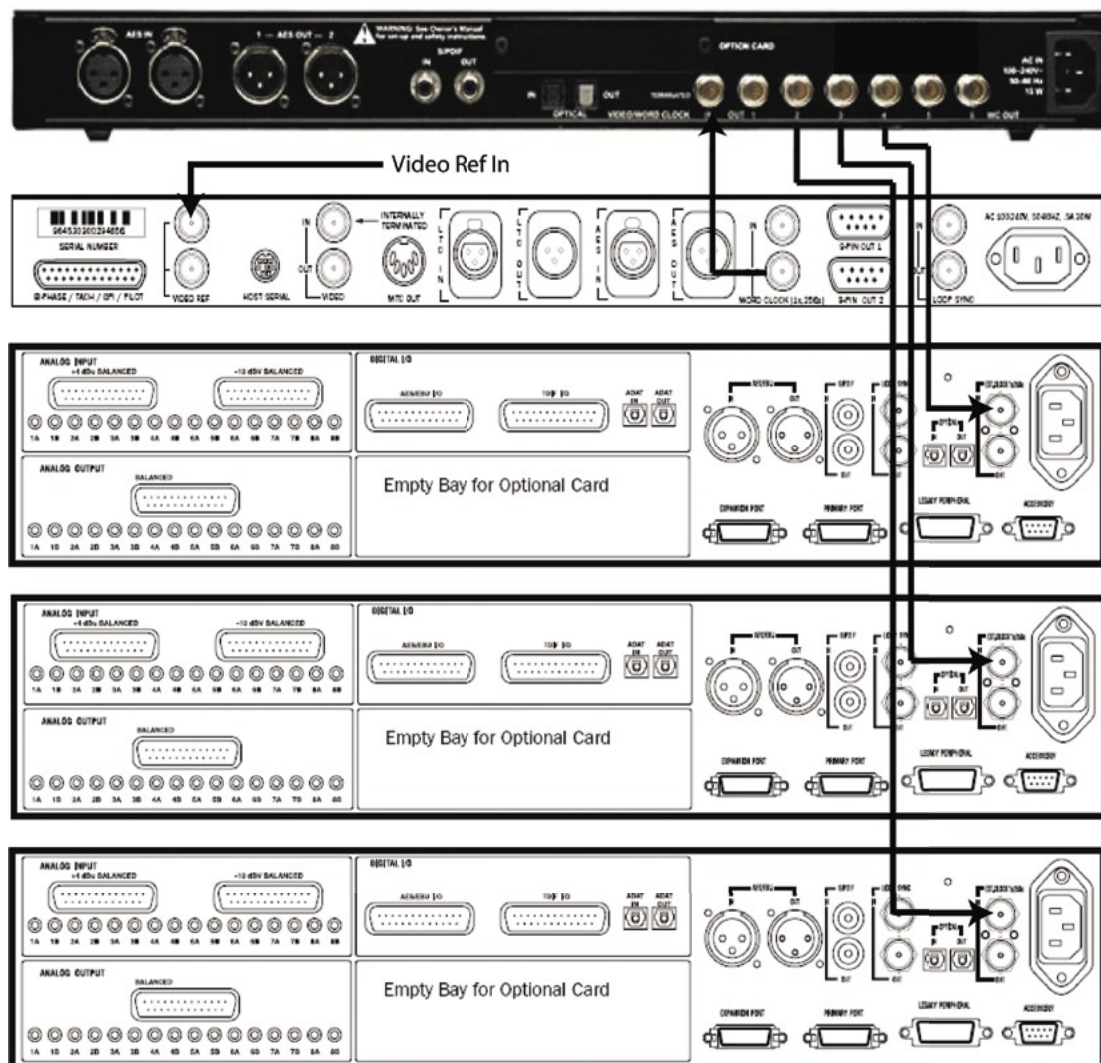


Figure 20: Distributed Clock with SYNC I/O as a "Resolver" and third party clock "Scrubber"

THE EFFECT OF JITTER ON AUDIO

Here begins the tricky part of the discussion... Though much research has been done and continues on the topic of jitter's effect on audio, universally accepted conclusions are difficult to arrive at because the subjective nature of hearing is hard to quantify. In the abstract, it would seem that a system with zero jitter would be the best sounding and most pleasing to listen to. However, many of today's most popular dedicated clocking devices have quite large amounts of jitter (compared to the 192 I/O), yet are considered by many to improve the sound of their Pro Tools systems! The following figures are

jitter measurements of popular 3rd party clock generators. These devices were measured on the same system used throughout this paper. Notice that each generator exhibits higher jitter amplitudes than the 192 I/O, yet many users prefer the sound of these clocks. It's very counterintuitive!

Figure 21 shows the jitter spectrum of a popular dedicated clock. Compared to the 192 I/O's internal clock, there are several spikes in the audible band as well as significant high frequency spikes. The adjectives most associated with the sound this device imparts on the audio are "warm", "punchy low end", "defined imaging", and "pleasing top end".

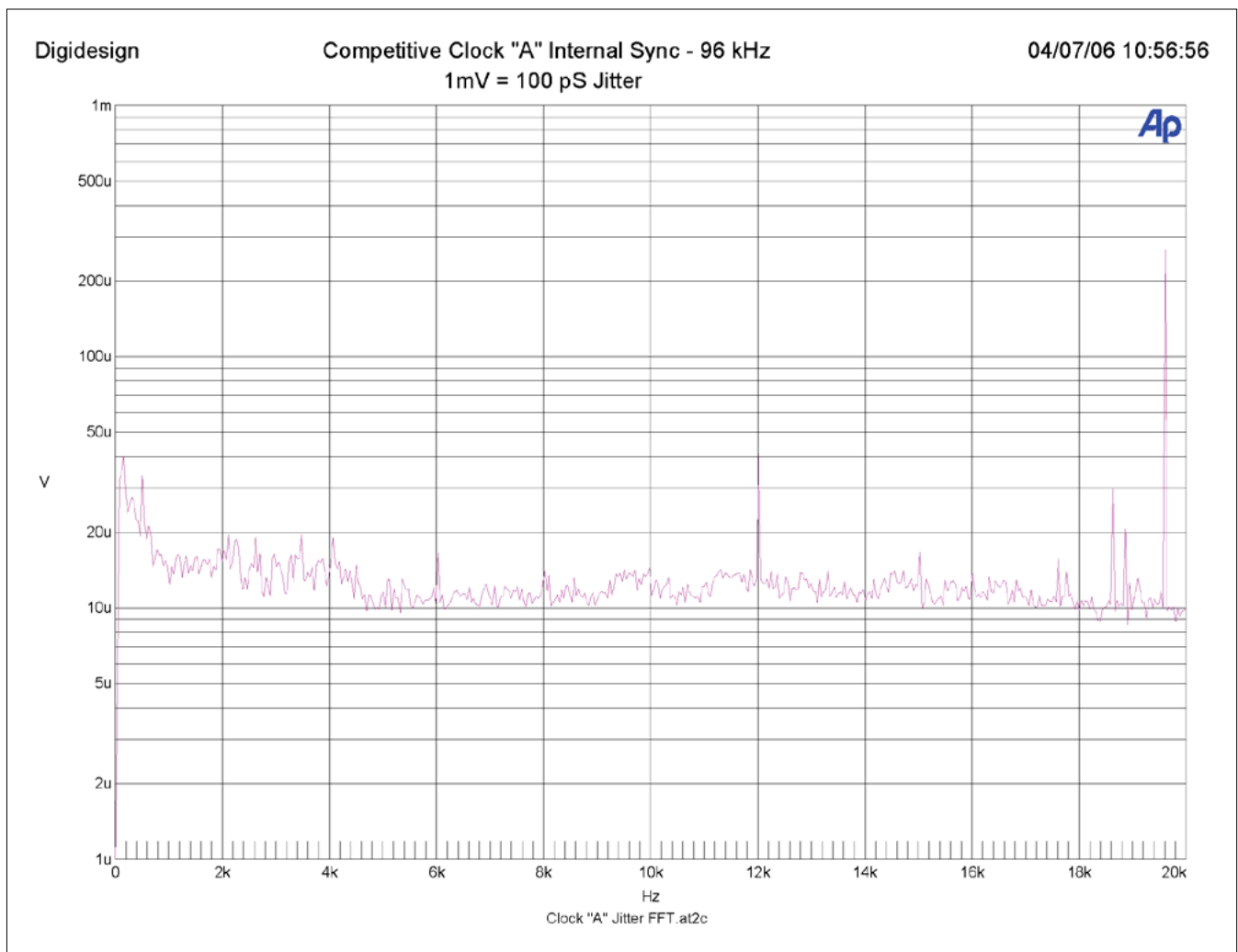


Figure 21: Dedicated Clock "A"

Figure 22 shows the jitter spectrum of another extremely popular clock. And again, anecdotal perception of this clock is that it provides a “clearer” image when used to clock a Pro Tools system. From measurements only, it has more than twice the amount of average jitter and at some frequencies 10 times more than a 192 I/O! Why, then, is it perceived to sound better?

Figure 23 shows the jitter spectrum of a third popular dedicated clock. Note the overall rise in the midrange frequencies and large spikes at 5 kHz, 17 kHz and 18.8 kHz. Again, the words used to describe the benefits of this clock are “wider image”, “more stable”, and “brighter without being harsh”.

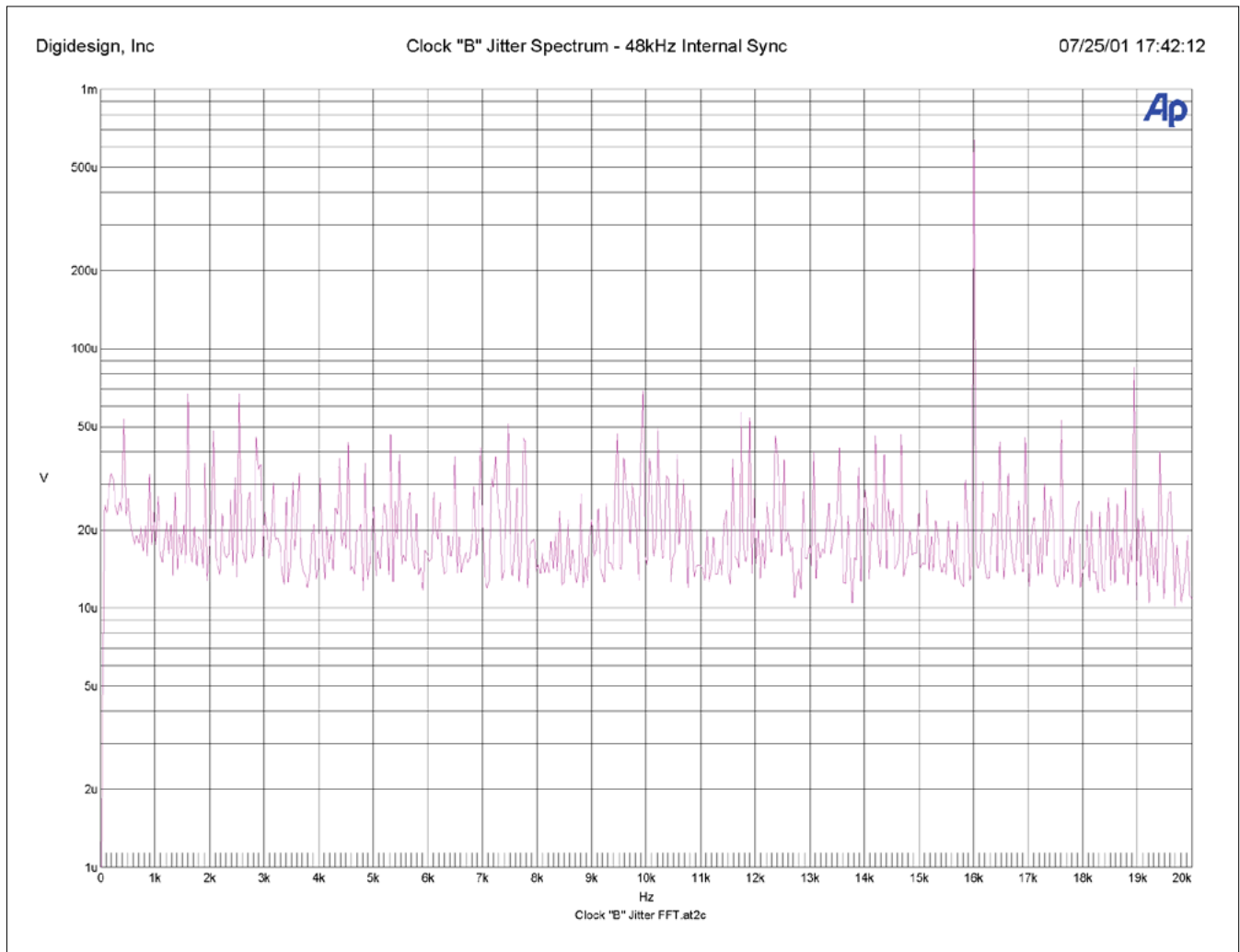


Figure 22: Dedicated clock “B”

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Clock "C" Jitter Spectrum - 48kHz Internal SyncCloc

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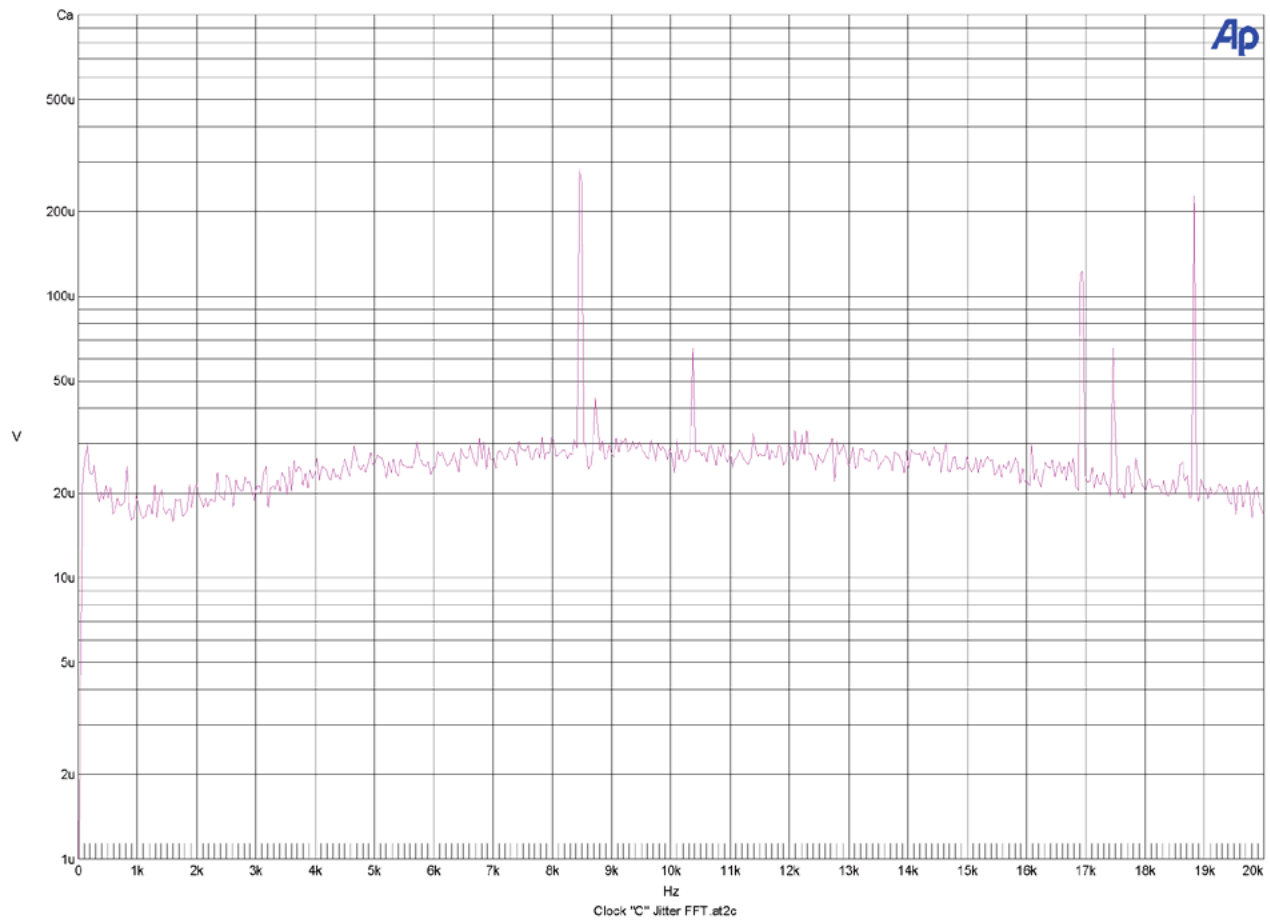


Figure 23: Dedicated Clock "C"

CONCLUSION

It's clear from these examples and from discussions with audio engineers from around the world that it is difficult if not impossible to come to a single conclusion when subjectively identifying the audible effects of jitter. While jitter is quantifiable and relatively easy to measure, how it affects complex waveforms such as music is much harder to quantify.

Many consider the 192 I/O to have the finest sounding clocking scheme while others prefer third party dedicated clocks and using the distributed technique in their systems—even though the effect of external clocks often produces higher amounts of jitter. On the bench, the 192 I/O compares favorably with all of its similarly priced competitors producing low jitter across the spectrum. In fact, it has far less jitter than many dedicated clocks and yet many prefer the sound of these clocks over the 192's internal clock.

Because of this apparent lack of consistency between theory and actual experience, more science and measurement is called for, particularly on the perceptual side. Controlled listening tests involving a cross section of participants using established standards need to be done in order to more fully understand the issue. Marketing materials from many manufacturers use anecdotal evidence and testimony from highly regarded individuals to promote their products and while that is powerful product endorsement, it needs to be recognized as such. Subjective reactions and uncontrolled listening tests should not be used as a substitute for science and it's clear that more study needs to be done to fully understand the audible effects of jitter on digital audio.

Please send your feedback on this article and requests for other discussion topics to techpapers@digidesign.com

The following papers may be of interest to those wanting more information on measurement and the audible effects of jitter.

1. *Eric Benjamin and Benjamin Gannon, 'Theoretical and Audible Effects of Jitter on Digital Audio Quality', Preprint 4826 of the 105th AES Convention, San Francisco, September 1998.*
2. *Julian Dunn, 'Jitter: Specification and Assessment in Digital Audio Equipment', AES Preprint 3361, October 1992.*
3. *Julian Dunn, Barry McKibben, Roger Taylor and Chris Travis, 'Towards Common Specifications for Digital Audio Interface Jitter', Preprint 3705, at the 95th AES Convention, New York, October 1993.*
4. *Julian Dunn, 'Jitter and Digital Audio Performance Measurements', Published in 'Managing the Bit Budget', the Proceedings of the AES UK Conference, London, 16-17 May 1994.*
5. *Edmund Meitner and Robert Gendron, 'Time Distortions Within Digital Audio Equipment Due to Integrated Circuit Logic Induced Modulation Products', Preprint 3105, AES 91st Convention, October 1991.*

Also of interest is Julian Dunn's website <http://www.nanophon.com/>



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