Understanding the Playback Engine

Have you ever noticed a delay or latency while you were recording but didn't know what to do about it? Or maybe a H/W buffer error popped up and you had no clue what was happening? Well allow me to introduce you to one of the most important set of preferences in Pro Tools, the Playback Engine. This week at the Pro Tools corner we will cover the Pro Tools playback engine and how you can effectively mange your resources in each stage of the production process.

The Basics

As real-time as DAW recording may seem, computers actually run in step time, calculation by calculation one in front of another. We get the sense of a real time environment only because the processor churns out these calculations so fast that they appear to be happening in real time. But because our needs change during the creative process (moving from recording, to editing, to mixing and so forth), the demands we put on our system change as well. When we are recording we want lower latencies or less delay, making the system feel more "real-time." When we are editing and mixing we want more power to run plug-ins and route tracks. In order to achieve the best of both worlds, low latency with plenty of power, the system ends up needing a little tweaking on the users end. This is where the playback engine comes in.

Essentially, the playback engine preferences in Pro Tools govern the way your computer crunches numbers when dealing with digital audio. Basically it gives the user control over the resources he or she wants to dedicate to Pro Tools and how system will best use those resources. Sounds complicated but it is actually quite simple and I am going to teach you an easy to remember rule you can follow when you are setting it up.

To open the Playback Engine choose Setup > Playback Engine

To start out, there are really two main preferences that you want to adjust in the playback engine, H/W Playback Buffer and CPU usage.

H/W Playback Buffer: This number is going to govern your systems latency (delay) as well as the overall processing load it will be able to handle. The value is set in samples and measures the time it takes to send audio round trip through the system. At a sample rate of 44,100 Khz (or 44,100 samples per second), a H/W buffer setting of 256 samples would yield a latency of roughly 6 ms. So from the time you strike a note on your guitar to the time you hear that note come back through your speakers around 6ms would have passed. The higher the H/W buffer, the more latency the system will express but the more time the system has to complete processing tasks.

Think of your computer as a number eating machine, a lower H/W buffer setting requires it to take smaller bites and process less overall data per cycle. A larger buffer setting allows the computer to take larger bites of data and handle more processing but at the expense of longer latency times.

Ideally we would want to set the H/W playback buffer as low as possible and leave it there, but in reality today's computers cannot handle running all of the real-time plug-ins we want to run with ultra low latency. Setting the HW buffer size low when you are recording will yield lower latency times but will reduce the overall DSP capabilities of your system. Setting the buffer higher when editing and mixing will allow your system to process more plug-ins overall and since you aren't recording, a higher latency of a 10-20 ms will go unnoticed.

The Rule

I wish I could give you a specific number to set your buffer to but I cant. Because every computer system is different and each individual's needs are different, no one setting is going to work for everyone. Some older computers may not be able to run at all with lower buffer settings, and will spit HW buffer errors with only a few plug-ins and tracks running. No matter what system you have if you follow this simple rule you should be able to maximize that capabilities of your system during each stage of your workflow:

To minimize latency when recording (both audio and midi), set the H/W playback buffer as low as it can go without getting errors.

To maximize your systems processing capabilities when editing and mixing, set the H/W playback buffer as high as it can go.

Note: If you are using an HD system, the H/W playback buffer only effects your RTAS processing capabilities and will not change the recording latency of your audio tracks (which is extremely low). Therefore, recording in audio streams through your interface will incur almost no latency, but playing RTAS virtual instruments will.

CPU usage limit: CPU usage limit is a value measured in percentage points that sets the overall processing cap for Pro Tools. This value acts sort of like a RPM limiter on your car; if you go over a certain RPM (redline) most cars have a governor that kicks in for your safety. If the CPU usage limit is set to 80% and the processing overhead in Pro Tools exceeds that limit, playback will stop and the system will display an error. If you are just using Pro Tools and no other software at the same time, you want to set this limit fairly high to maximize the processing abilities of your system. When you are using rewire applications like Reason or Live, you may get better results setting the CPU usage limit a bit lower to provide some head room for those other applications.

Number of Processors: If you have a dual or quad processor machine, you can tell Pro Tools to use more then one processor. I generally set this value to 2 on a dual processor machine.

New to 7.3 - Ignore Errors During Playback/Record:

A new preference in Pro Tools 7.3, this checkbox allows pro tools to ignore certain errors (like dropping samples of audio) in favor of keeping the system chugging. This can be a life saver when you are trying to squeeze the last ounce of juice out of your rig, but should generally be left off when recording audio since pops and click can occur.

DAE Playback Buffer:

The DAE playback buffer is one of those set it and forget it parameters that you don’t want to play with too much. Generally a setting of Level 2 (Default) is ideal for most systems. The DAE playback buffer determines the amount of memory (system ram) Pro Tools reserves for the play back buffer. Because a hard drive is still a mechanical mechanism a ram buffer is required to stream audio from your drives. Essentially Pro Tools pulls audio from the drive into the DAE playback buffer and then streams that audio from the faster, more consistent RAM.

If you find that you have to set this higher then Level 2, that may signify a problem with your machine or hard drives, such as a seriously fragmented drive or a drive that is to slow for Pro Tools to pull audio from. I recommend at least a 7200RPM drive for audio playback and DO NOT recommend that you use your operating system drive for audio. I find that I can run sessions of 50-60 tracks no problem from a single firewire 400 drive.

In Closing

At some point all systems will spit out a H/W buffer error as even the fastest processors can only handle so many plug-ins and tracks. But with a little playback engine know how you should be able to extend any systems capabilities at any stage of your workflow.

New Pro Tip: The Mbox's "Mix" Knob

Because the Mbox (original), Mbox 2 and Mbox Mini all connect via USB 1.1, the lowest possible latency you can achieve is 256 samples (roughly 6 ms). For some people this is just too much latency. Fortunately the Mbox line has a cool built in solution for combating this dreaded latency and it is actually part of your hardware.

On the front of the Mbox 2 there is a knob that reads "Mix" (on the original Mbox the knob reads "Input/Playback"). Turning the knob fully clockwise (to the right) gives you the output of your D/A converters as the signal plays back through the software and is the desired setting for mixing and editing. Turning the knob fully counter-clockwise (to the left) gives you the analog input of your source before it hits the A/D converters, effectively offering "zero latency" monitoring, since the signal you are hearing has yet to enter the digital realm. By having the knob somewhere in the middle you can effectively blend the two signals and help alleviate some of that inevitable latency.

With the knob somewhere in the middle you may hear a phasing or chorusing effect, this is because you are hearing both the live analog signal as well as the signal as it passing back through the D/A a few milliseconds later. To avoid this, simply mute the record enable track in the software. Unfortunately you will not be able to monitor any plug-ins, like guitar amp simulators, using this workflow since you are monitoring purely the dry analog signal before it hits the A/D.

If you want to get really picky, since you are technically recording to a signal that is delayed a few milliseconds in time, you can shift your newly recorded region back by the amount of samples set in your playback engine. So if you had a 256 sample latency (6ms) and you were recording in a guitar track, you would shift that guitar region back 256 samples to be perfectly in time with the rest of the session. You can easily use the Edit > Shift function to do this. Going to the trouble of shifting tracks is often not necessary on all instruments all the time, let your ears can tell you if something feels a little late.

Note: the "Mix" knob does not apply to the Mbox 2 Pro, since it connects via firewire and can achieve lower latencies.

If you have a Pro Tools related question that you would like me to take a shot at answering or have any feedback, please send it to brian@audioMIDI.com. I will do my best to select the questions that seem to stump the most people.

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