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# LATENCY WITH APHEX USB AUDIO INTERFACES

## LATENCY DEFINED

Latency is a term used with computer audio recording. Latency refers to the amount of time it takes for you to sing in to a microphone and hear your voice in the headphones coming back from the computer. If that amount of time is less than 10 milliseconds (ms) you're in good shape! If the latency is more than 20ms your brain will actually hear the sound from the computer to your headphones as an "echo." This makes it hard for any singer or musician to perform in time with the rhythm of the song.

## HOW WE'VE DEALT WITH IT IN THE PAST

In the early days of computer recording the computers simply didn't have the capability to provide less than 10ms of latency. So we monitored the input signal instead of the recorded signal. However this technique required a mixer and effects processor if you wanted to monitor with reverb. Overdubbing vocals was difficult because the reverb was only on the input signal and not on the dry track you were playing back and punching in on or there was a mismatch in the reverbs.

Later in the evolution of computer recording, makers of PCI, USB and Firewire audio interfaces started incorporating hardware DSP for monitor mixing software and effects with their products. These new features increased the cost of the product and created the potential for new software bugs. This technique offered better latency than monitoring from the DAW and still allowed you to monitor with effects like reverb. But overdubbing vocals was still difficult because the reverb was only on the input signal and not on the dry track you were playing back and punching in on. You also had to invest some time learning how to use the new monitor mixing software.

## HOW TO DEAL WITH IT NOW

Contemporary computers are now capable of letting us monitor through the DAW with less than 10ms of latency. We can even use plug-ins to monitor with reverb. And because we're monitoring the DAW the reverb is always present on the track so overdubbing is a breeze. You can even use your high-end reverb plug-ins and start building your mix during the overdubbing process. This is something you simply cannot do using monitor mixing software.

For modern workflows, the hardware DSP monitoring mixers now just get in the way and there is often no way to bypass them! So if your audio interface incorporates one, you're still required to learn how to use it, set it up to make your signal flow work properly and then go back to it to make changes.

**WORKING SMART**

Contemporary computers still have their limits. You can still overload your processors by using a large amount of tracks, plug-ins and virtual instruments. Here are some ways to stretch those limits:

1. Instead of inserting plug-ins on every track, use aux sends and reurns for effects whenever possible. This will put less strain on your CPU and make your system run more efficiently. For example, if you plan to use reverb on 16 of your tracks use an aux send from each of the 16 tracks to send to one instance of a reverb plug-in instead of inserting a reverb plug-in on all 16 tracks. All time-based effects should be used this way. Delay, chorus, flanger, etc. Inserts are primarily for dynamics processing like compressors and gates.

2. Your DAW's input sample buffer needs to be set as low as possible to get the best latency performance when you're recording audio or MIDI tracks. But when it's time to mix, you want to set the input sample buffer as high as possible. This takes a great deal of stress off of your CPU.

3. When you're dealing with a large session that requires you to record a lot of audio and MIDI tracks there are some things you can do to take the strain off of your CPU and keep your input buffer low for recording:

> A. Instead of leaving all of your virtual instruments on, record the output of the virtual instruments to audio tracks. Then make the virtual instrument tracks inactive. This will take a great deal of stress off of your CPU. You can activate the tracks anytime that MIDI editing is required. Once you're done recording your tracks, you can increase the input buffer to maximum to take the load off of your CPU for mixing.
>
> B. You can also simply make a stereo mix of the session, open the stereo file in a new session and record your new tracks while monitoring the stereo mix. Once you're done recording your new tracks you can import them in to the original session and raise the input buffer for mixing.

**THE NUMBERS**

Computer used for latency numbers**:**

Macbook Pro 7,1 (This is a 5-year-old computer. Current 2014 models have 4 cores instead of 2, twice the amount of RAM and Cache and are MUCH more efficient. Meaning you'll get even better performance then the numbers shown below.)

Mac OS X 10.6.8, 2.4GHz Intel Core 2 Duo, 4GB RAM 1067MHz DDR3, L2 Cache 3MB, Cores 2, Processors 1. Using Cubase 7.0.

**Microphone X latency reported by the driver to the DAW.**

| 32-sample buffer | 64-sample buffer | 128-sample buffer |
|---|---|---|
| Sample rate 44.1kHz | Sample rate 44.1kHz | Sample rate 44.1kHz |
| Input latency 3.2ms | Input latency 4ms | Input latency 5.4ms |
| Output latency 2.2ms | Output latency 3ms | Output latency 4.4ms |
| Sample rate 96kHz | Sample rate 96kHz | Sample rate 96kHz |
| Input latency 1.5ms | Input latency 3.2ms | Input latency 3.8ms |
| Output latency 1ms | Output latency 2.2ms | Output latency 2.8 |

**IN2 latency reported by the driver to the DAW.**

| 64-sample buffer | 128-sample buffer |
|---|---|
| Sample rate 44.1kHz | Sample rate 44.1kHz |
| Input latency 3.7ms | Input latency 5.2ms |
| Output latency 3.5ms | Output latency 4.9ms |
| Sample rate 96kHz | Sample rate 96kHz |
| Input latency 2.9ms | Input latency 3.6ms |
| Output latency 2.7ms | Output latency 3.3 |
| Sample rate 192kHz | Sample rate 192kHz |
| Input latency 2.6ms | Input latency 2.9ms |
| Output latency 2.3ms | Output latency 2.7ms |

**USB 500 RACK latency reported by the driver to the DAW.**

| 32-sample buffer | 64-sample buffer | 128-sample buffer |
|---|---|---|
| Sample rate 44.1kHz | Sample rate 44.1kHz | Sample rate 44.1kHz |
| Input latency 3ms | Input latency 3.7ms | Input latency 5.1ms |
| Output latency 2.8ms | Output latency 3.5ms | Output latency 4.9ms |
| Sample rate 96kHz | Sample rate 96kHz | Sample rate 96kHz |
| Input latency 2.6ms | Input latency 2.9ms | Input latency 3.6ms |
| Output latency 2.3ms | Output latency 2.7ms | Output latency 3.3ms |

These numbers do NOT include the latency from the A/D and D/A converters, which is approximately 1.5ms.

So the best round trip latency for the Microphone X with this computer would be 4ms.

The best round trip latency for the IN2 with this computer would be 6.4ms.

The best round trip latency for the USB 500 RACK with this computer would be 6.4ms.



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