How To Optimize Your Audio Interface Latency By Tweaking Your Audio Buffer Size

By
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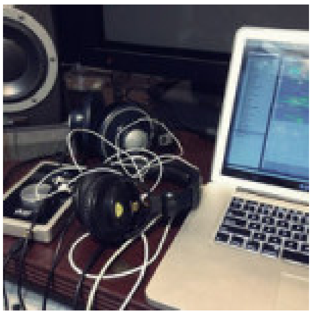
The Reason The Default Sound Card Is Not Suitable For Music Production



The most important reasons that the default sound card – the one you get when you buy a pc – can't help you record and mix are:

1) It doesn't have the necessary inputs we need to connect our mics (XLR Inputs) or the cables that our guitars and bass guitars need (TRS Inputs).

2) They have too much latency.

What Is Latency?

With simple words we can say that:

Latency is the delay that exists in the sound from the time that the sound "enters" our audio interface and our DAW until the time that "gets out" of our speakers.

For example, if we have 10ms latency and play the guitar then we will hear the guitar's sound in 10ms (milliseconds).

As you realized, the less ms the better.

We can easily record if the latency is up to 10ms but if it exceeds that limit then our ears start to anticipate it and it's too hard to record on time using the metronome.

For this reason, when you want to choose an audio interface, don't only pay attention to the input quantity but also keep in mind the latency speed it offers you.

Tweaking The Buffer Size

Buffers optimize our computer's audio playback with the help of the AD/DA Converters.

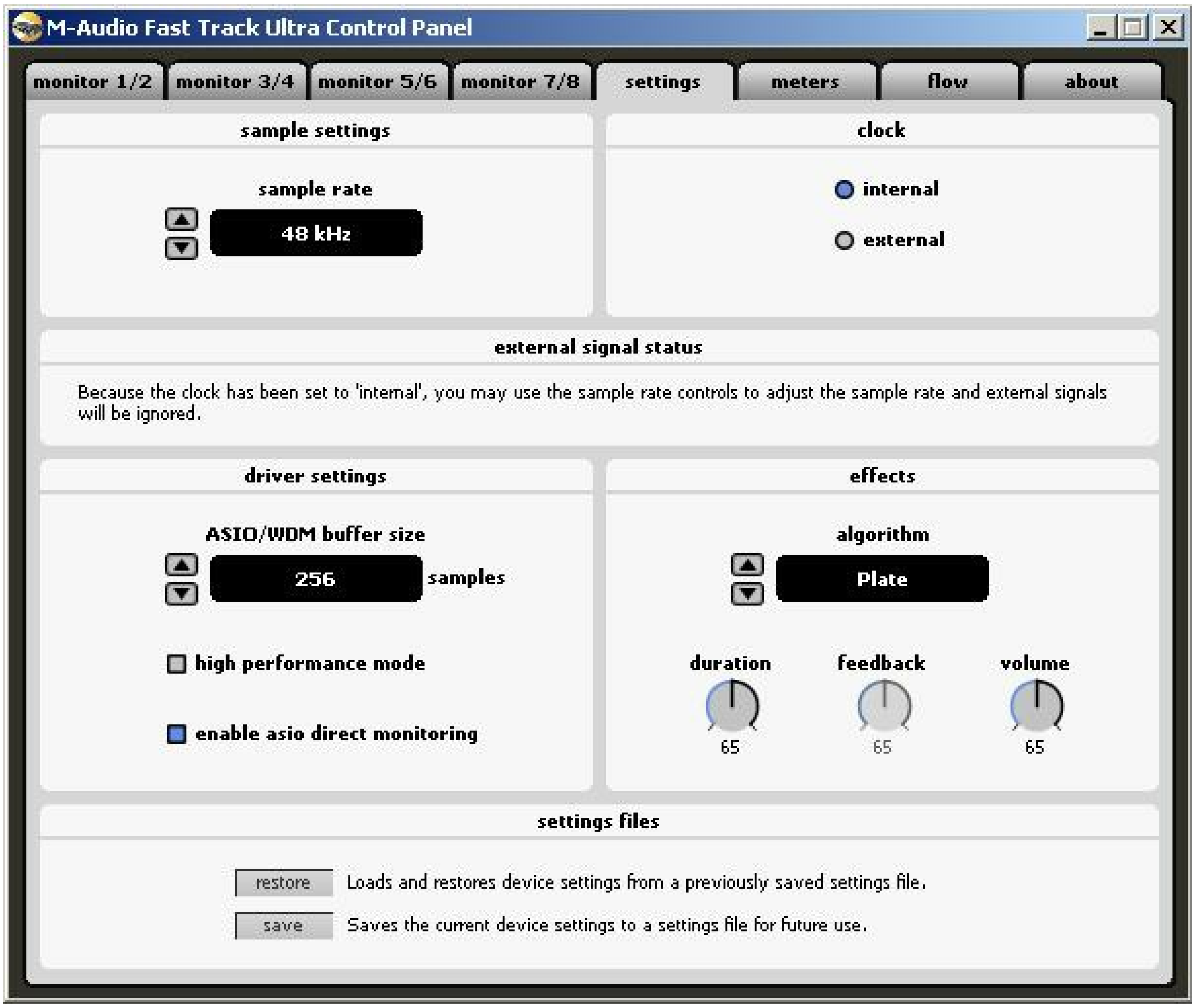
I don't want to confuse you with technical terms because you won't need any of this info unless you want to build your own sound card, so I'll keep it simple.

Buffers are measured in **samples**.

The most used samples are: **256 samples, 512 samples, 768 samples, 1024 samples** and **2048 samples**.

Important:

- The less the samples the less latency we have, but we need a strong pc cause we need more resources.
- The more the samples the more latency we have, but we free pc resources especially if our pc is not powerful enough.



What Buffer Size Should I Choose For My Computer?

It really depends on the drivers of your sound card and your computer's processing power, but here's a sure-fire way to find the optimum buffer setings for your pc or mac:

1. Begin with the highest buffer setting (1024 buffers is a good start). The more you reduce them the latency will drop too.
2. You'll reach a point where your computer will start producing some strange noises during playback (also known as **glitches**). This is the setting that your computer cannot handle.
3. Raise the buffers to previous setting right before your computer started to produce these glitches.

I have an M-Audio Firewire Solo at home and I have 8ms latency using 512 samples.
Pretty cool if you ask me!

An Extra Mini Tip

Like we said above, less samples need more cpu power, so the more VSTs the more pressure we put to our computers.

To solve this problem we can record with the least samples possible (so that we can record with the lowest latency possible) and THEN raise the buffers when mixing.

Sound is not affected with this "method", we are done with our recording stage, so we don't really need to have low latency.

Let’s “exchange” Latency with CPU power then for our mixing phase!

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