

## Zero Latency Monitoring Handbook

### What is "Z" monitoring?

"Zero" Latency or Direct Monitoring via ASIO 2.0 is somewhere between 2-5ms. So is that good enough when monitoring or overdubbing a live performs? Well it depends on the person. Vocals and Drums typically would consider anything under 6 ms good enough. Because of the typical distance an electric guitar player stands from their amp anything below 12 ms is good enough. Most piano players would say 10 ms is good enough.

**NOTE:** *Keyboard players playing a soft synth will hear a delay between their fingers on the key and the sound in the speaker approximately , equal to the ASIO "Output" buffer size as seen in Studio One.*


To put things in perspective, the human performance of even a tight musician to the playback will typically vary (plus or minus) a two digit number. Roughly speaking 1 ms = 1 foot. So for example, musicians 20 feet from each other will be dealing with 20 ms delays. So depending on the physical situation, the musician and instrument, "good enough latency" varies

"Z" Monitoring or Zero Latency is supported **as of Studio One 2.6+** on the following interfaces:

### Hardware Interfaces

- FireStudio Mobile, Project
- Audio Box 22VSL, 44VSL, 1818VSL, 192 Studio
- Any 3<sup>rd</sup> party ASIO 2.0 DM interface device (On Windows only)

### How do you know if you have Zero Monitoring Capabilities

- 1) You will see a "Z" below the hardware interface icon. 



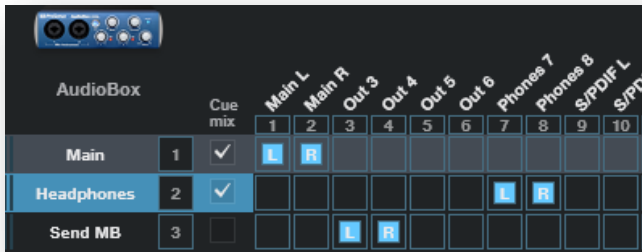
- 2) If you have a Cue mix set up for your main outs you will see a "Z" in your channels.
- 3) If you have a Cue mix set up for an alternative output you will see a "Z" on the cue mix.



## Basic Set Up Procedure:

### Create cue mixes

Create cue mixes for monitoring any "Z" mix.



#### 5.10.1 Creating a Cue Mix Output

In Studio One, it is possible to quickly and easily create multiple cue mixes. A cue mix is separate from the main mix and is usually provided to musicians for monitoring purposes during recording.

For instance, when recording vocals, the engineer and vocalist will probably want to hear different mixes. Most vocalists want to hear more of their vocals in the mix, possibly with some reverb to make it sound natural, while the engineer might focus on how the performance balances with the rest of the mix. Together, Studio One and a FireStudio™-series interface make this simple.

The first step in building a cue mix is to create another Output Channel. To do this, open the Song/Song Setup/Audio I/O Setup menu in a Song, switch to the Outputs tab, and add a new Stereo Output channel. Next, specify that this output is a cue-mix output by clicking on the channel's Cue Mix checkbox. You can create as many cue mixes as your audio interface has available stereo outputs.

#### 5.10.2 Mixing the Cue Mix

By default, the level and pan values are locked to the channel level and pan faders. This means that each Cue Mix will be identical to the main mix in the Console. Changing the level or pan in the Cue Mix object will unlock both settings, allowing independent control of level and pan for each channel in each Cue Mix. Thus, the level and pan for channels in a Cue Mix can be completely different from the related level and pan in the main mix. To completely remove any channel from a cue mix, simply deactivate the Cue Mix object for that channel.


## Enable Z

Click the "Z" On, either on the channel for main outputs or on the cue mix. You will hear the "Z" latency in your corresponding output when you do this.



## "Z" Monitoring

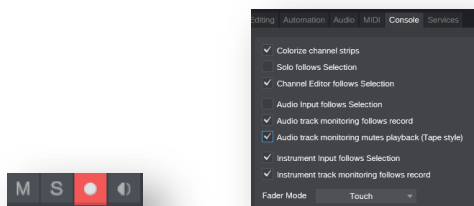
The procedure is for recording or auto punch in recording mode:

1. Record-enable AND Monitor-enable the new track(s). 
2. Verify the "Z" is set for you monitoring cue.
3. You will hear the source at a "Z" latency level. *(You are hearing the monitoring from the hardware)*
4. Record the track listening to the "Z" cue mix.
5. To hear the recorded track turn off "Z" or the Monitor button.

## "Z" Monitoring with (Tape Style)

If you are using the Tape Style monitoring, use this procedure:

1. Turn Tape Style monitoring on in the Options/Advance/Console
2. Record-enable the new track (Disable the Monitor button if you want to use auto punch in)
3. Verify the "Z" is set for you monitoring cue.
4. Record the track listening to the "Z" cue mix.
5. In Tape Style, because the Monitor button is off you will hear the recorded track on playback.



## Tape Style Monitoring vs. Non Tape Style Monitoring

	Before recording	During playback before/after punch-in	During Punch in Recording	Record from stop (non-punch)	Comment
<b>Tape Style On Mon = on</b>	Hear input See input on channel	Don't hear track See input on channel	Hear input when recording See input on channel	Hear input when recording See input on channel	Don't hear track See input on channel
<b>Tape Style On Mon = off</b>	Hear input See input on channel	Hear existing track Do not see input on channel	Hear input when recording See input on channel	Hear input when recording See input on channel	<i>Meant for <u>auto punch</u> in recording, but works for non-punch in as well (Un-arm recording to stop monitor)</i>
<b>Tape Style off Mon = on</b>	Hear input See input on channel	N/A	N/A	Hear input when recording See input on channel	Meant for <u>non punch</u> in recording. No need to hear previous recording (monitor=off to hear track)
<b>Tape Style off Mon = off</b>	Don't hear track	Hear track	Don't hear Recording	Don't hear Recording	Playback mode

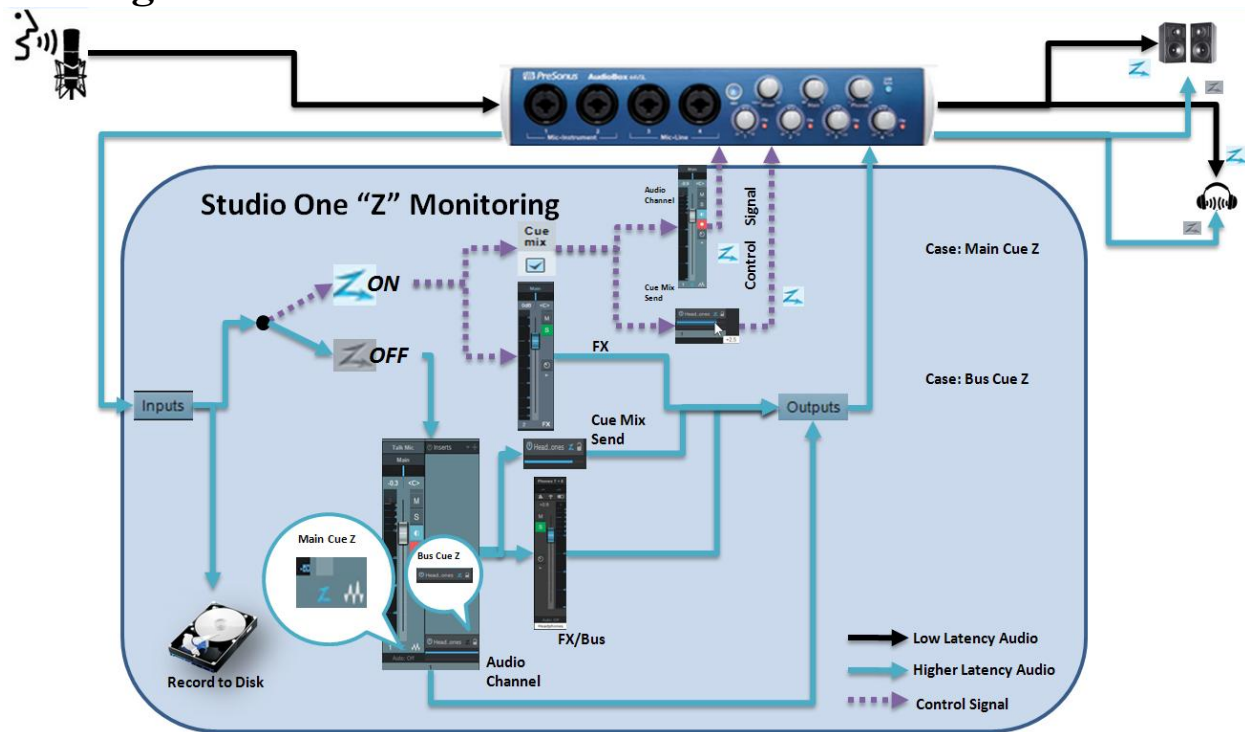
### Monitoring with Effects (optional)

If your audio device supports zero-latency hardware monitoring, use that as the primary monitor source, so that no delay is heard. In addition, you can use sends and FX channels in the Console, as you normally would in a mix, to build a better monitor sound. For instance, on the audio track to which you are recording, you could add a send to an FX channel with a reverb. You could then route the FX Channel output to a Sub Out Channel and back to your audio interface, where it can be mixed with the zero-latency dry signal.

When adding time-based effects, such as reverb or delay, you generally don't have to be concerned about plug-in delay and latency that could result from using software plug-ins on a live input source. A few milliseconds of processing delay on a reverb will probably not be audible.

## Signal Flow Reference

### “Z” Signal Paths



### Z On

- 1) Make sure the input track(s) are set to monitor ( except with Tape Style Monitor enabled).
- 2) In the case of recording, the audio signal follows the normal signal flow through the mixer to the hard disk.
- 3) In the case of monitoring, the audio follows the normal signal flow through the **mixer however the monitoring path is switch off on any “Z” enabled Audio Channel and instead sends a fader level control signals to the hardware interface (via ASIO 2.0) to adjust the appropriate Direct Monitor output level.**
- 4) **NOTE: The FX bus will send the FX signal also to the hardware interface but with a delay due to the ASIO buffer size. If you are using delayed based effects such as Reverb or Delay, this delay in the wet signal will probably not be noticed especially when mixed with the zero latency signal.**

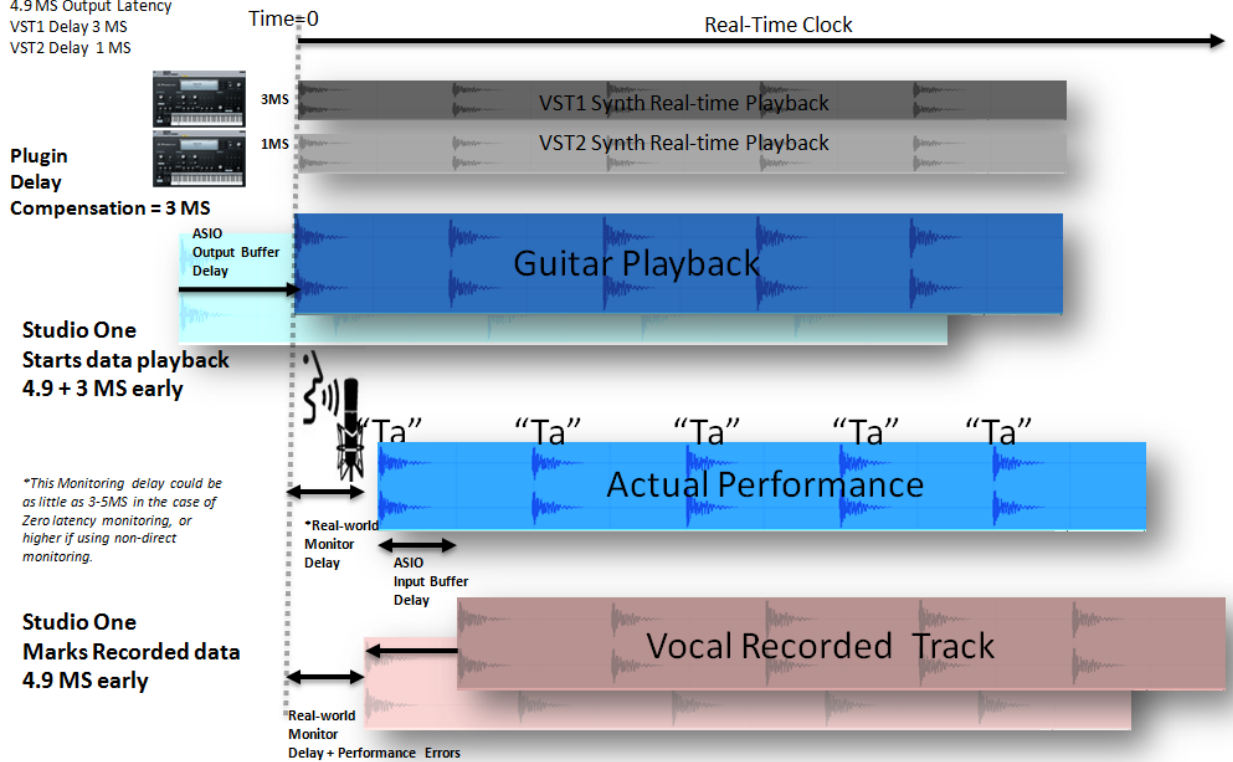
### “Z” Off

- 1) In the case of recording and or monitoring, all signal paths will follow normal routing and be delayed in the monitoring by the size of the ASIO buffer.

# How The Delays are Handled by Studio One

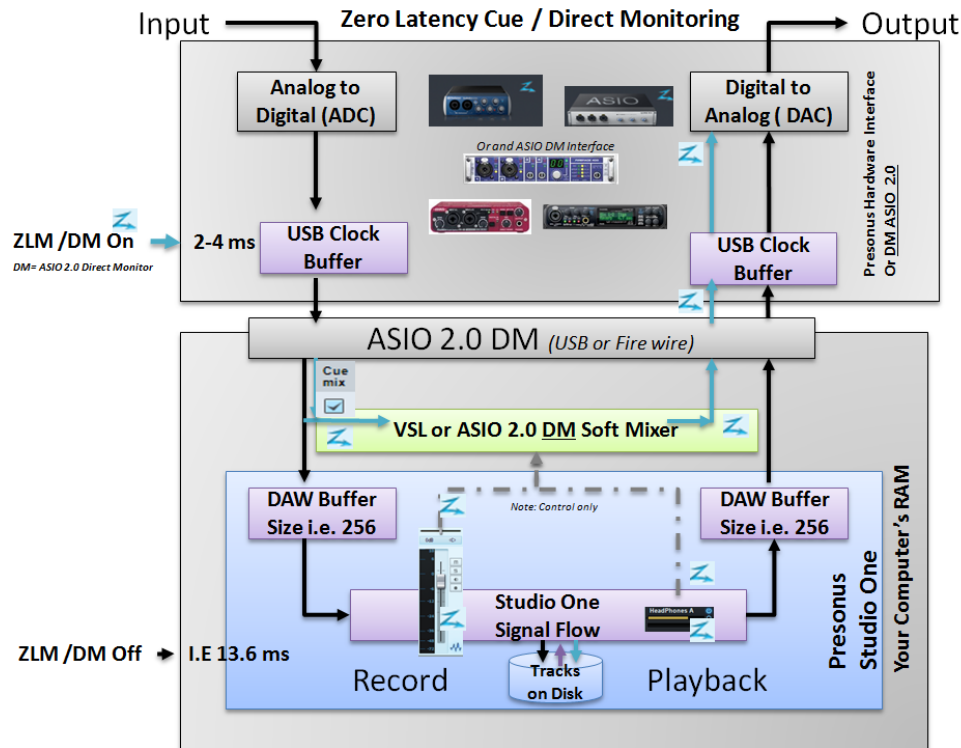
## ASIO Buffer + PDC Latency Example

Example:  
 ASIO Buffer 128 =  
 4.9 MS Input Latency  
 4.9 MS Output Latency  
 VST1 Delay 3 MS  
 VST2 Delay 1 MS



There are many different kinds of delays in the recording overdubbing process. Delays in playback or recording due to the ASIO buffer are handled by the Record/Play back engine. **So from the overdub musicians perspective the playback is always at Time Zero.** In the example above delays inherent to soft synths and or FX plugins and non delayed recorded tracks are handled by **Automatic Plugin Delay Compensation.** **Real-world overdub delays** are the relationship of the playback (which is at time zero) and your recorded input between 3-5 ms (zero latency) to a number much larger depending on your ASIO buffer size. For this reason, (if not using Zero latency), it is a good idea to keep your buffer sizes as low as possible when recording live input. *Tip: Many set their ASIO buffer size low for recording and higher for mixing.*

Technical References



**New in 2.6**  
**Direct**  
**Monitoring on**  
**any ASIO 2.0 DM**  
**Capable Device**

*Check the specification of your interface to see if it is capable of DM.*

Note: Exact signal flow for ZLM is only known by Presonus designers. This is for demonstration purposes to explain the affect of using ZLM to the user.

Reference

<http://www.presonus.com/news/articles/The-Truth-About-Digital-Audio-Latency>