Presonus Audiobox 1818VSL

USB 2 Audio Interface

SAM INGLIS

Presonus's new USB 2 interface promises tight integration with their Studio One software at a competitive price.

Although it's been clear for a while that the Firewire standard is on the way out, manufacturers have taken their time to develop multi-channel audio interfaces that use other protocols. It's only over the last year or so that a reasonably broad selection of multi-channel USB interfaces has become available, and they still tend to be noticeably more expensive than comparable Firewire boxes. Presonus are thus shaking up this market with their new Audiobox 1818VSL, which significantly undercuts similarly featured rivals.

In many respects, the Audiobox takes ideas that are familiar from Presonus's Firewire interfaces and ports them to the USB realm. For example, unlike most rivals, Presonus put all their mic sockets on the front panel — and the Audiobox's front panel is exactly the same as that of the existing Firestudio, except painted blue. There are eight of these sockets, and all of them are combination XLR/jack affairs. The jack sockets serve as high-impedance instrument inputs on inputs 1/2, and line inputs on the other six channels. They are joined by a single headphone socket with its own volume control, plus a master volume and associated stereo LED ladder meter. A single LED lights solid blue when things are hunky dory and flashes red when any sort of sync problem is encountered. At the extreme right is a stereo LED ladder meter that displays the levels reaching the main outputs, but input metering is restricted to red clip LEDs by each gain pot — there's not even a 'signal present' indication.

The rear panel is slightly more spartan than that of the Firestudio. There are eight line outputs plus an additional pair of master outs which mirror outputs 1/2, but with hardware volume control. Digital I/O consists of coaxial S/PDIF in and out, MIDI in and out, and optical ADAT in and out, along with a BNC word-clock output (but no input). Completing the picture are a single USB port, a socket for the external 18V DC power supply, and the on/off switch. (It seems odd that Presonus are convinced of the virtues of putting mic sockets on the front, but not power switches.) Totting up all the analogue and digital I/O gives a total of 18 channels either way, with the usual loss of four when the ADAT I/O is used at high sample rates.

Like most multi-channel interfaces, the Audiobox has a software utility to control the routing and monitor mixing that take place on its onboard DSP

Presonus Audiobox 1818VSL $449

PROS

- Sounds good, with decent preamps and a useful selection of I/O.
- Costs significantly less than most comparable multi-channel USB interfaces.
- Virtual Studio Live provides compression and EQ on every channel, plus reverb and delay.
- Slick integration with Presonus' Studio One DAW, and the promise of iPad control.

CONS

- Some other aspects of Virtual Studio Live are frustrating at present.
- Only one headphone socket — which can't be used to listen to the main VSL mix.
- Can't operate as a stand-alone A-D converter.
- Metering isn't great.

SUMMARY

The Audiobox 1818VSL is a good-sounding USB interface which comes with powerful DSP effects and a lower price tag than the competition, at the cost of limited headphone routing and a slightly raw Virtual Studio Live mixing utility.

Test Spec

- Audiobox VSL Windows version 1.1.4364 and 1.2.
- Scan PC with 3.4GHz quad-core Intel Core i7 CPU and 8GB RAM, running Windows 7 Home Premium 64-bit SP1.
- Tested with Presonus Studio One 2, Avid Pro Tools 10 and Steinberg Cubase 6.

chip. The notable feature of Presonus' Virtual Studio Live is that it replicates pretty closely the features and functionality of the company's Studio Live mixers; in essence, the Audiobox's DSP is a Studio Live mixer without the physical controls. This means that not only can your monitor mixes employ hardware-powered reverb and delay, but that every input and output channel has its own gate, compressor and equaliser too. Many manufacturers now offer some processing capability, but to have this much power on offer is unusual. What's more, an iPad app has just been released that will allow you to control VSL remotely from your tablet-based executive toy, which means that you will be able to go and stand with the band and adjust their monitor mix from the stage if you wish.

Virtual Reality

There are two schools of thought when it comes to designing software mixer utilities. On the one hand, manufacturers such as RME and Focusrite dispense with aux sends and so on, to provide what are in effect separate mixers for each output pair. Presonus, along with M-Audio and others, take the more traditional route of having only one mixer with one set of faders. The output from this mixer is hard-wired to the main (1/2) output pair, and if you want to route signals to any of the other outputs, you'll need to use the aux sends above the faders. This caused me some head-scratching to start with, because originally there was nothing visible to show that the headphone socket is hard-wired to output pair 7/8 rather than 1/2. Fortunately, a software update just before we went to press added a large headphone icon.

The outputs are always treated as stereo pairs, and each mixer channel thus has three aux sends which can be panned. If you want to route channel 1 to output 3 only, for example, you'd enable the first aux send and pan it hard left. These auxes are, thankfully, pre-fade but post channel insert, so monitor mixes created using the aux sends are unaffected by fader moves but very much affected by EQ, compression and gating.

The processing available on input channels within VSL is pretty much unparalleled at this sort of price. First in the queue are a polarity reversal button and a high-pass filter. I'd much rather this be implemented in the analogue domain to prevent unwanted low frequencies causing problems at the converter stage, but from the point of view of setting up a live monitor mix, the fact that it can be swept from 18Hz to 1.3kHz(!) is handy. These are followed by a simple noise gate, a fairly conventional compressor/limiter and a three-band equaliser. All of these are versatile enough to do their job in a live situation, or when setting up a monitor mix, and, more importantly, are easy to set up and use. Should you want to record the processed sources, each input channel has a Post button, which can be clicked to place them into the record path. A simple drag-and-drop browser system allows presets for channels and individual effects to easily be created, stored and manipulated.

Into The Blue

With the possible exception of RME's TotalMix FX, then, Virtual Studio Live offers perhaps the most comprehensive processing facilities available in any such utility. That said, the version of VSL I tested also had its share of niggles. For one thing, its meters have no numeric scale, and provide no warning at all when signals are clipping. They don't even tell you when your inputs are getting hot: the level indication remains blue all the way to the top rather than turning yellow or red. This, to my mind, is asking for trouble, especially on a unit with limited hardware metering.

The faders, meanwhile, default to the top of their travel, which represents the unity gain position (not that this is anywhere mentioned in the documentation), meaning that they can only be used to attenuate the input signal and not to boost it. Although this has the advantage of making it very difficult to run out of headroom within VSL, I personally would have liked the flexibility of being able to add gain to a source within the monitor path. There are times when gain-structure rectitude comes second to the need for more vocal in the monitors.

The routing also seems a little restrictive. Within VSL, there is simply no way to access any outputs other than the »

At the back is the power switch, a 12V DC power input, a USB 2 socket, a BNC word-clock output, ADAT and S/PDIF ports, MIDI I/O sockets, and, all on quarter-inch jacks, a pair of main outputs and eight individual line outputs.

Vital Statistics

- USB 2 audio & MIDI interface.
- Compatible with: Windows XP, Vista, 7, Mac OS 10.6 and later.
- Analogue inputs: eight, on 'combi' jack/XLRs (six mic/line, two mic/instrument).
- Built-in mic preamps: eight.
- Analogue outputs: eight, plus Main Outs duplicating outputs 1/2.
- Headphone output: one, duplicating outputs 7/8.
- Digital inputs: stereo coaxial S/PDIF, optical ADAT.
- Digital outputs: stereo coaxial S/PDIF, optical ADAT, word clock.
- MIDI In and Out.

Virtual Studio Live in action. The browser at the right stores channel presets, which can simply be dragged and dropped onto any mixer channel. What's not so obvious is that input 1 is clipping.

main output pair, except by using aux sends. This means, among other things, that there's no way to listen to your main VSL mix through the headphone socket. There's not even a 'copy faders to sends' option that would allow you to easily duplicate your mix on the appropriate auxes. The aux sends themselves are quite small, making setting up a headphone mix a little fiddly.

When I used the Audiobox to record a band playing live, another limitation became apparent. Solo is a global function within VSL, and is destructive, so it's impossible to solo a source in one output — such as the headphones — without soloing it on all the outputs. This makes it difficult to use headphones to track down a problem during a performance, although I suppose you could solo channels within whatever DAW you're recording to.

There are a few more minor irritations on offer, too. For instance, although the VSL window can be resized, and various bits of it can be hidden, there's no configuration in which you can see all 18 input channels at once without scrolling, no matter how large your monitor. Finally, when dealing with a large number of inputs, it would be lovely if it was possible to name them. It's much easier to locate a problem when you're presented with meaningful input names rather than 'ADAT 1' and so on.

Having presented what might look like quite a long list of criticisms, though, I hasten to add that all of them fall into the category of annoyances rather than fatal flaws. I used the Audiobox to record several fairly complex sessions, and although VSL frustrated me sometimes, none of its quirks presented a serious obstacle to the recording process. Also, while my own work consists mainly of location recording to tight deadlines, where I don't have time to set up compression and EQ in the monitor mix, I'm sure many other users will find that the Audiobox's impressive DSP effects outweigh any frustrations with the rest of VSL.

Presonus are also actively working to improve Virtual Studio Live, and told me that some of the points mentioned above are on their list for future updates.

Driving Miss DASIO

Another important software component of a product such as this is, of course, its ASIO and Core Audio drivers. Much criticism has been levelled at some manufacturers for the way in which their drivers either report lower latency values than they actually deliver, or hog CPU cycles at low buffer sizes, or both. On Windows, the Audiobox offers seven buffer sizes ranging from 32 to 2048 samples, but even on my new, purpose-built Windows 7 PC from Scan Computers, I couldn't persuade either Cubase 6, Studio One 2 or Pro Tools 10 to play back audio successfully at any of the buffer settings below 256. That said, the v1.2 update I received just prior to going to press incorporated a new driver version, which is claimed to offer better performance, but unfortunately there was not time to carry out my tests again.

I used Vin Curigliano's DAWbench test to measure the relative performance of Cubase and Pro Tools with the Audiobox at different buffer sizes, at 44.1kHz. At the 1024-sample setting, Cubase was capable of running 80 instances of Softube's Tube-Tech Classic Channel, a figure which fell to 56 at the 256-sample setting. By comparison, my own Focusrite Saffire Pro 40 achieved similar results at 1024 samples, but would only run 30 instances at 256. Pro Tools 10 behaved very oddly with both interfaces, achieving surprisingly good results at 256 samples and refusing to work at all at 1024!

Virtual Studio Live incorporates its own safety buffer, with Safe, Normal or Fastest settings which allegedly add 4, 2 or 1 ms of latency, respectively, on top of that determined by the ASIO buffer size. At a 44.1kHz sample rate and 256-sample buffer setting, Oblique Audio's Round Trip Latency utility measured an actual »

"Ultimately, though, the most important point about the Audiobox 1818VSL is that it provides a good-sounding and reliable way of getting multi-channel audio in and out of a computer."

The Audiobox's front-panel features, from left to right, grouped phantom-power buttons, two mic/instrument and six mic/line inputs on 'combi' sockets, eight gain controls with individual clip LEDs, master and headphone volume knobs, a headphone socket and, finally, stereo output level meters.

» round-trip latency of 686 samples (15.6ms) at the Fastest setting and 952 samples (21.6ms) at the Safe setting. This was less than 2ms more than the combined input and output latencies that were reported by Cubase, and since Cubase's measurements don't include converter delays, this would suggest that the Audiobox is being commendably honest in reporting back to its host. The only minor driver issue I encountered was that once or twice it set the sample rate to 96kHz when I hadn't asked it to, but this never happened at a point where it could pose a problem. In practice, I had no stability problems with the Audiobox, which proved reliable even during a long day's recording with nearly all its I/O in use.

I was, however, a little surprised to find that the Audiobox can't be used as a stand-alone A-D converter. Obviously, you wouldn't expect to be able to take full advantage of Virtual Studio Live without a computer to control it from, but I would at least have liked to be able to use the Audiobox as an ADAT expander for another interface.

Summing Up

From a hardware point of view, I liked the Audiobox, and at this price, I think Presonus can be forgiven for the external line-lump power supply and the almost non-existent input level metering. It's noticeably smaller and more portable than most comparable interfaces, yet its front-panel layout doesn't feel as cramped as some. Presonus' XMAX preamp is an established design that sounds good and offers a usefully wide gain range of -15 to +65 dB. Whereas most project-studio interfaces have continuous gain pots, these have a finely granulated feel, with about 40 steps across the entire gain range. I'm not sure if these are evenly spaced in terms of how much gain they apply, or consistent across channels, but they definitely help to make gain settings more repeatable and precise, and help you find a unity-gain position for line inputs where appropriate. Most audio interface preamps top out around 60dB gain, and the extra 5dB or so on offer here is useful if you work with ribbon or moving-coil mics.

I must nevertheless point out two gripes about the I/O. The smaller of these is that there are only six line inputs. A bigger deal is that there is only one headphone socket, and that subject to the routing limitations described earlier. Most interfaces with this sort of I/O count include two headphone sockets, and for good reason!

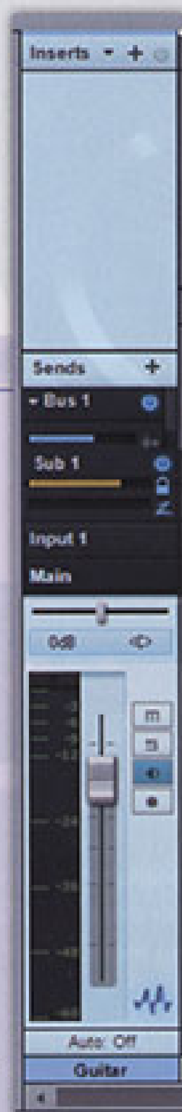
As described earlier, Presonus's USB driver seems stable and moderately efficient, while the quantity of DSP effects processing available for monitor mixes is unrivalled at this price. The promise of iPad control is also interesting, especially for use within a live-sound context. The Virtual Studio Live mixer utility itself currently has its share of frustrations, but none of these are fatal, and hopefully Presonus will address them in future updates.

Ultimately, the most important point about the Audiobox 1818VSL is that it provides a good-sounding and reliable way of getting multi-channel audio in and out of a computer — at an extremely competitive price. With a street price of $449, it's a direct rival for Firewire interfaces such as Focusrite's Saffire Pro 40 or M-Audio's Profire 2626, and substantially more affordable than its closest USB-capable rivals, Steinberg's UR824 and MOTU's 828 Mk3 — to the point where you could budget for an external headphone amp and still have change in your pocket. ■■■

The Audiobox & Studio One

Every Audiobox ships with Studio One Artist, a cut-down but nonetheless very usable version of Presonus's DAW software. I reviewed the full version of Studio One v2 back in December last year (www.soundonsound.com/sos/dec11/articles/studio-one-v2.htm), and if you're looking for a no-frills package that's intuitive and easy to learn, I think it's hard to beat. The main limitation in the Artist version is that third-party VST and Audio Units plug-ins are not supported.

The packaging states that Studio One is "perfectly integrated" with the Audiobox, a claim which refers to the way in which low-latency monitoring can be set up without leaving the DAW. This functionality was already implemented with Presonus's Firewire interfaces, but was only enabled for the Audiobox in the VSL update I received just before going to press. In keeping with the general streamlining of Studio One's design, it's nothing like as flexible or powerful as something like Cubase's Control Room, but a lot simpler and more immediate. In essence, it means that when you're working with a suitable interface, you can designate any output pair a Cue Output. Each audio track then sports a send to this cue output, and if you click the curly 'Z' (for 'zero latency') icon next to one of these sends, the input signal on that track will be routed directly to it within the Audiobox, rather than via Studio One's buffers. The upshot is that you can keep the buffer size as high as you like, yet monitor inputs with minimal latency: simple, but very effective.



Here, I've set up a Cue Send within Studio One v2. The crinkly 'Z' icon, when lit, allows signal to be passed to the Cue Send without passing through the Audiobox's input and output buffers, for low-latency monitoring.

$ $449.
T Presonus +1 225 216 7887.
E info@presonus.com
W www.presonus.com