Focusrite Scarlett 18i8 Review
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It's been just about two years since I reviewed the Focusrite Scarlett 8i6 and 18i6 USB audio interfaces. Focusrite has added several models to the Scarlett line since then on both ends of the scale, that scale being mostly a function of the number of hardware inputs and outputs. I'm personally pretty well set for studio recording, but in order to start using a computer for more of my field recording activities, I've been waiting for the right sized interface to come along. I want four mic inputs, packaged in a box that's the right size and shape to use as a platform for my little netbook computer. Is the Scarlett 18i8 just right? Let's see.

Overview

The 18i6 is in a tabletop format box, 2½" high, 9" wide and 7" deep, tipping the scales at a bit over 3½ pounds. The case is a sturdy Focusrite red aluminum extrusion. At 0.156" (5/32") thick, you could probably drive over it with a car and not hurt anything inside (but please don't try it). It doesn't quite pass the Mike Metlay Laptop Bag Test in that the front panel knobs protrude slightly, but the large monitor volume knob is secured with a nut, and the headphone volume and input gain knobs protrude through close fitting holes in the panel that supports them securely. There's no distressing knob wobble on this unit.

I usually like to look inside the box, not because I expect that mere mortals can repair these things, but rather so I can comment on the quality of the construction and the components. Try as I might, I was unable to separate the case from the circuit boards (there are two, one for the input level controls and their indicators and the other for the rest of the guts), so I didn't get my usual up close look at this one. Power requirements exceed what's available from a USB 2.0 port, so a

universal (100-240 V 50-60 Hz in) 12 V, 1250 mA switching type wall wart power supply with a coaxial plug is supplied. There's a power switch on the rear so it's not necessary to unplug it in order to turn it off.

Computer connection is via USB 2.0, with USB 3.0 port compatibility. The driver installation package includes an ASIO and WDM driver for Windows 7 and 8 (psssst . . . it works fine under XP SP3 too) with plug-and-play compatibility with Mac OSx 10.7 and above. Supported sample rates are 44.1, 48, 88.2 and 96 kHz. It's a class-compliant USB audio device under iOS, though I don't have an iGadget with which to test it.

In the interest of always using the latest version, no software disk is included in the package - there's a card with instructions for registering the product and downloading the required software. The Windows installation file includes both the ASIO and WDM driver and Mix Control, an on-screen control panel for the built-in DSP monitor mixer. Since the 18i8 uses the Apple Core Audio driver, Mix Control is installed as a separate stand-alone Mac application. There is not yet (possibly not ever) an iOS app version of Mix Control.

Optional downloads which are authorized with a license code that comes along with registration are The Scarlett Plug-in suite which includes a compressor, gate, equalizer, and reverb in RTAS, AU, and VST versions, a copy of Abelton Live Lite, the Novation Bass Station mono synth VSTi, and a gigabyte of royalty free samples from Loopmasters.

Gozintas and Gozoutas

Focusrite's model numbering system, at least for the Scarlett series, begins with the number of inputs streams and ends with the number of output streams, with an "i" in between. I say "streams" because that's how many digital audio signals are available to the computer. Some correspond to analog input and output jacks on the box, others correspond to digital inputs and outputs that require external hardware between them and a microphone or loudspeaker,

There are four Amphenol (Neutrik equivalent) combo XLR-TRS ¼" jacks on the front panel. Inputs 1 and 2 can serve as microphone, line level, or high impedance instrument level DI inputs. The XLR is always a mic level input. The ¼" jack can be switched between line level and instrument DI inputs via the Mix Control software. Inputs 3 and 4 are also combo XLRs but don't offer the instrument DI option for the ¼" jack. A 10 dB pad which functions on both the XLR and ¼" jacks can be individually engaged for each of these inputs. Each of the four front panel inputs has a corresponding front panel gain control.

The rear panel carries four more fixed gain line level inputs on ¼" jacks. All of the analog inputs, including when in the DI mode, are differential ("balanced").

Phantom power is available on all four mic inputs, switched in pairs (mic 1-2 and mic 3-4) by two front panel switches. Phantom voltage is within tolerance at 44.6 volts. The mic input impedance is 3.1 kΩ, on the high end of typical for modern general purpose mic preamps. The input impedance is unaffected by either phantom power or the 10 dB pad. Common mode rejection for the mic inputs is 72 dB, which is pretty good for a run-of-the-mill electronically balanced input.

At maximum gain, -42.5 dBu at the mic input yields a digital level of 0 dBFS. At minimum gain without the pad, the mic input sensitivity is +8.0 dBu for 0 dBFS. Digital clipping is reached before analog clipping, so whatever doesn't clip digitally will be clean.

The front panel line inputs on the combo jacks have an impedance of 42 kΩ, or half that if you're driving them from an unbalanced source. At maximum gain, 0 dBFS is reached with an input level of -20 dBu. At minimum gain, the line input will take a whopping +30 dBu to reach 0 dBFS. Clearly the gain control is in the right place in the signal path since it's actually possible to put that level in without the onset of analog clipping. And if you want to record the speaker output of your guitar amplifier, you can engage the 10 dB pad and get 10 dB more input headroom, probably enough to turn the amplifier up to 11 (a Spinal Tap reference). I hope y'all appreciate that I had to rewire my monitor amplifier in order to get a signal at that level for testing. My shop signal generator maxes out at +20 dBu.

The rear line level inputs hit 0 dBFS with 10.2 dBu going in. This a little on the hot side, given that a professional grade mic preamp can typically put out +20 dBu or more before clipping. It should work fine for keyboards or drum machines, or any device that has an output level control. The input impedance of the rear line level inputs is 17 kΩ when fed from a balanced source, but rather than the impedance being equal on both the tip and ring, the tip is 11 kΩ to ground and the ring is 6.7 kΩ. An unbalanced source feeding the rear panel line inputs sees the 11 kΩ input load. Common mode rejection of both the rear and front panel line inputs is 56 dB, showing that the apparently unbalanced input impedance at the rear jacks does no appreciable harm.

With no access to the mic preamp output, it's impossible to measure the gain. Whatever Focusrite says is probably correct, but what's relevant is the sensitivity, how many dBu it takes to reach the maximum record level, as measured above.

While the front panel inputs are still differential when switched to the Instrument DI mode, balanced instrument outputs are pretty rare, so I checked them out on the assumption that you'll be feeding them with from conventional unbalanced pickups. The input impedance measured from tip to ring with the sleeve grounded (what you'd have if you plugged in a common T-S guitar cable) is just a bit over 200 kΩ, which should work fine with any magnetic and most piezoelectric

pickups. At maximum gain, the input sensitivity for 0 dBFS is -42 dBu. At minimum gain, +7.5 dBu in gives a record level of 0 dBFS. While the 10 dB pad works with the DI inputs, with the pad engaged, the analog stage starts to clip at about +12 dBu. Slamming chords with an instrument that has really hot pickups might require that you turn the instrument volume down a bit. Or you might like the distortion. I didn't.

The six analog outputs are appear physically as two balanced outputs on rear panel mounted ¼" TRS jacks and two ¼" stereo headphone jacks on the front panel. The rear outputs are designated Outputs 1 and 2 and are nominally intended to feed control room monitor speakers. The headphone jacks are designated Outputs 3-4 and 5-6. Each headphone jack has its own stereo volume control while the large volume knob on the front panel controls the level to the two rear panel jacks. There's more to know about the output level controls and pairing (or not) of the signals going to the output jacks, and I'll get into that when I discuss the Mix Control software

The rear panel outputs are active balanced, however they don't complain when connected to an unbalanced input. Maximum output level into a balanced input is +15.8 dBu, or +9.8 dBu into an unbalanced input. This is hot enough to drive just about any powered speaker to full level. The headphone outputs are, of course, unbalanced, and can drive 130 milliwatts into 60Ω headphones, which seems to be the impedance for which they're optimized. The headphone amplifier can drive a load as low as 15Ω but with a low impedance load it starts clipping at around 90 mW. Either way, the headphone feed should be reasonably loud unless you're the drummer (or you've stood too close to one for too long). I had no problem getting adequate volume with my Sony 7506 phones, which are around 60 ohms.

Other rear panel I/O consists of a stereo S/PDIF input and output on RCA jacks, an 8-channel ADAT optical input on a Toslink connector, and MIDI input and output on traditional 5-pin DIN connectors. There's neither a word clock input nor output. Word clock synchronization options are the internal clock (which is what you'll use most of the time), ADAT, or S/PDIF. The fact that S/PDIF synchronization is an option suggests that this input doesn't re-clock the data stream to the internal clock and should be selected if you're using the S/PDIF input. If you're running a full house with both the ADAT and S/PDIF inputs, it will likely be necessary to synchronize those two devices externally, then select one or the other to synchronize the Scarlett's internal clock.

Technical Details and Specs

These days, distortion in devices like this is so low and frequency response so flat that it's hardly worth writing about, but understanding that these are specs

that people compare and sometimes agonize over when comparing similar units, here are some facts that you may find useful.

Frequency Response

At 44.1 kHz sample rate, the frequency response is within ±0.1 dB from 20 Hz to 20 kHz. At the low end, it's 0.5 dB down at 10 Hz. The high end drops off rapidly above 20 kHz as expected. It's 0.3 dB down at 22 kHz, and goes down like a rock above 22 kHz. At 96 kHz sample rate, frequency response below 20 kHz is identical to that at 44.1 kHz sample rate. It's 0.2 dB down at 25 kHz, 1 dB down at 33 kHz, and 2 dB down at 46 kHz.
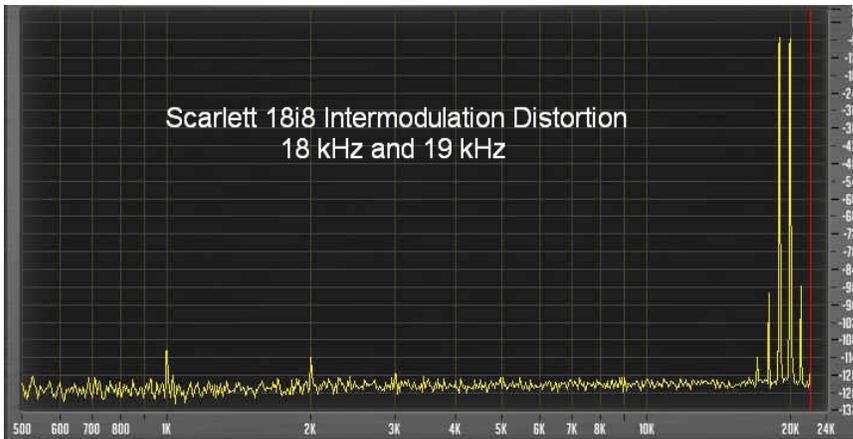
Distortion

THD+N (total harmonic distortion plus noise) from record to playback at 1 kHz measures .005%, which equates to -86 dB. It may be a little better than this since the residual distortion in my generator measures .004% (only 2 dB lower than without the Scarlett in the chain). This is the distortion figure that you'll see on a spec sheet. It's more interesting to look at what constitutes that distortion. Here's the spectrum of the playback of a 500 Hz sine wave recorded at -6 dBFS:
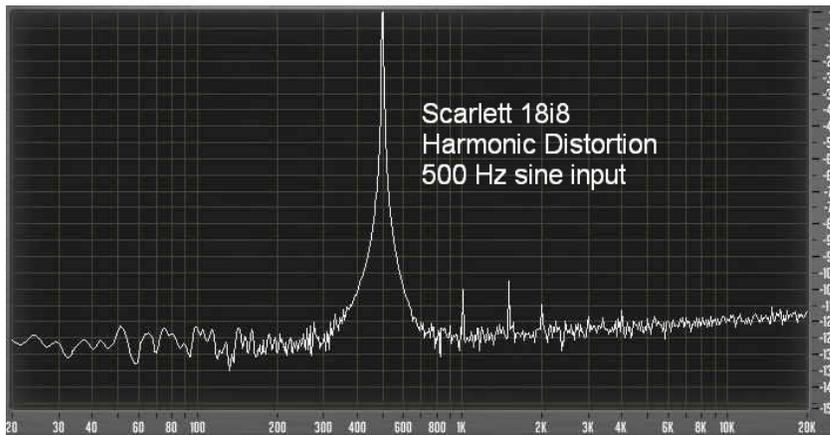
Note the spikes at 1, 1.5, and 2 kHz, the 2nd, 3rd, and 4th harmonics of the test frequency. Everything beyond the 4th harmonic is buried in the noise. Note that the 3rd harmonic, the largest of the three distortion components is about -100 dBFS (0.001%). This is very good performance. There aren't any wild frequencies there. In case you're wondering, you don't see the 60 Hz line frequency that appears in the noise spectrum because this was recorded at quite a bit less than full gain.

Intermodulation distortion (IMD) used to be more prevalent in A/D and D/A conversion than it is today, but it's worth a look. IMD is the set of unwanted frequencies created when two frequencies are mixed together in a device with some non-linearity. The two big spikes at the right end of the spectrum are the 19 and 20 kHz test signals. The distortion products are related to the 1 kHz difference between the two test frequencies. The spike at 1 kHz and its 2nd harmonic are about 105 and 108 dB respectively below the amplitude of the test
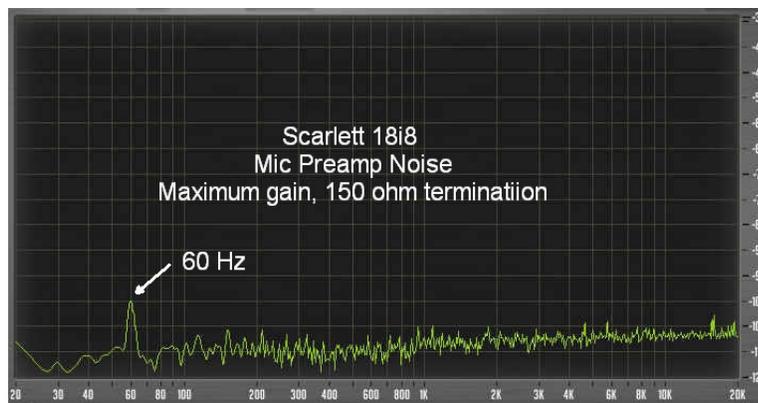
frequencies. These are the primary IMD products. The approximately -90 dBFS spikes at the high end of the spectrum are 1 kHz and 2 kHz below and above the two test frequencies. This is a pretty stringent test. With the worst distortion products being 90 dB down and nearly out of hearing range, total IMD is on the order of 0.006%, which is excellent performance.

Mic Preamp Noise

Quiescent noise of the mic preamps measured at 44.1 kHz sample rate and 24-bit resolution, at maximum gain and the mic input terminated with a 150 resistor (simulating a mic) is -86 dBFS RMS as calculated by Sound Forge. By examining the spectrum of the noise, we can see that it's mostly good, honest random noise, eyeball-averaging around -110 dBFS. The greatest contribution to that RMS value is the hump at 60 Hz, presumably AC line hum, peaking at around –102 dBFS.

Latency

When we speak of latency in the context of digital audio interfaces (or really, any digital device) we're almost always referring to how long it takes for a sound originating from a microphone or line level input source to arrive at our ears via headphones or speakers. With a stand-alone analog mixer or the equivalent in the form of a simple analog mixer built into the interface, the time between input and output is so short that only an on-line pedantic would argue against calling it zero. Essentially, it's at the speed of electricity (and light). However, the term "zero latency monitoring" tends to get thrown around with reckless abandon. My quip about it, unless I know the real delay time, is "for large values of zero."

The reason why we make a fuss over latency is that in the early days of DAWs and audio interfaces, the analog mic signal needed to get digitized, go into the computer, come out of the computer, and get converted back to analog before we could hear it. This is still the only way that many DAWs work. With fast computer hardware, a well designed driver for the interface, and some tweaking of the computer to minimize the number of other things it's doing besides getting your audio back to your ears, this can take as little as a few milliseconds, but without optimizing the system for low latency, it can easily be in the 10 to 20 millisecond range. That's long enough to hear as an echo, and in the 2-3 millisecond range, when recording vocals, the acoustic signal through your head and the delayed monitor signal combine at your ear to put some notches in the frequency response that you (but nobody else) hear due to what's known as comb filtering.

Generally if the delay is less than 1 millisecond, its effect isn't objectionable, so that's a good goal to try to achieve. It's nearly impossible with today's computer hardware and A/D and D/A converters, but incorporating a digital mixer based around dedicated DSP (digital signal processing) hardware as part of the interface can work wonders in reducing monitor latency. The Scarlett 18i8, and in fact all of the Scarletts that I've reviewed, exceed this <1 ms goal. At 44.1 kHz sample rate, the delay between input and output through the internal mixer is 0.6 ms. At 96 kHz sample rate, it's under 0.1 ms, 8 samples as I counted it. This is close enough to "zero latency monitoring" to keep me happy, though of course it's not really zero.

Delay when taking the computer route is considerably more. There's a setting for the ASIO buffer size (in milliseconds) that has a significant effect both on the latency and on achieving glitch-free recordings. The default setting is 10 ms. That's sufficiently long so that on just about any non-optimized system, you'll get a clean recording. However, forget about through-the-DAW monitoring at that setting. At the minimum ASIO buffer setting of 1 ms, I measured 7 ms of latency. With the buffer set for 4 ms, the latency increased to 22 ms. Doubling the sample rate shaves off only about 1 millisecond. Clearly there isn't a one-to-one correspondence here, and this is typical of most interfaces I've reviewed or used.

The lesson here, and I think this is nearly one of those EKTs ("everyone knows that") is to set the buffer latency so you get glitch-free recordings and playback (it can be as long as it gets for playback, which is sometimes necessary if you're mixing with a lot of plug-ins), and use the low latency input monitoring for tracking whenever you can. Obviously it's necessary to monitor the computer output when playing a virtual instrument, but one way to mitigate the delay is to take an analog sound from your keyboard that's close enough to the sound you really want, and monitor that rather than the virtual instrument sound.

Most DAW programs read the known delay associated with the ASIO driver (or measure it to be sure) and adjust the recorded track position so that overdubs come out in sync. I verified this for the 18i8 using both Reaper and Studio One and found that tracks recorded when monitoring the input through the DAW were synchronized to within about 0.5 milliseconds.

And this brings us to the DSP mixer and Mix Control, the software application that emulate the knobs and buttons of a hardware mixer.

Mix Control

Focusrite has been using this concept and program for quite some time now. It hasn't changed much since I've been reviewing Focusrite interfaces but it's pretty complete and, once you get the hang of it, it works quite well. In concept, it's an 18-input, 8-output mixer (18i8 – get it?) that can be configured in several ways. Here's an overall look at the Mix Control screen:
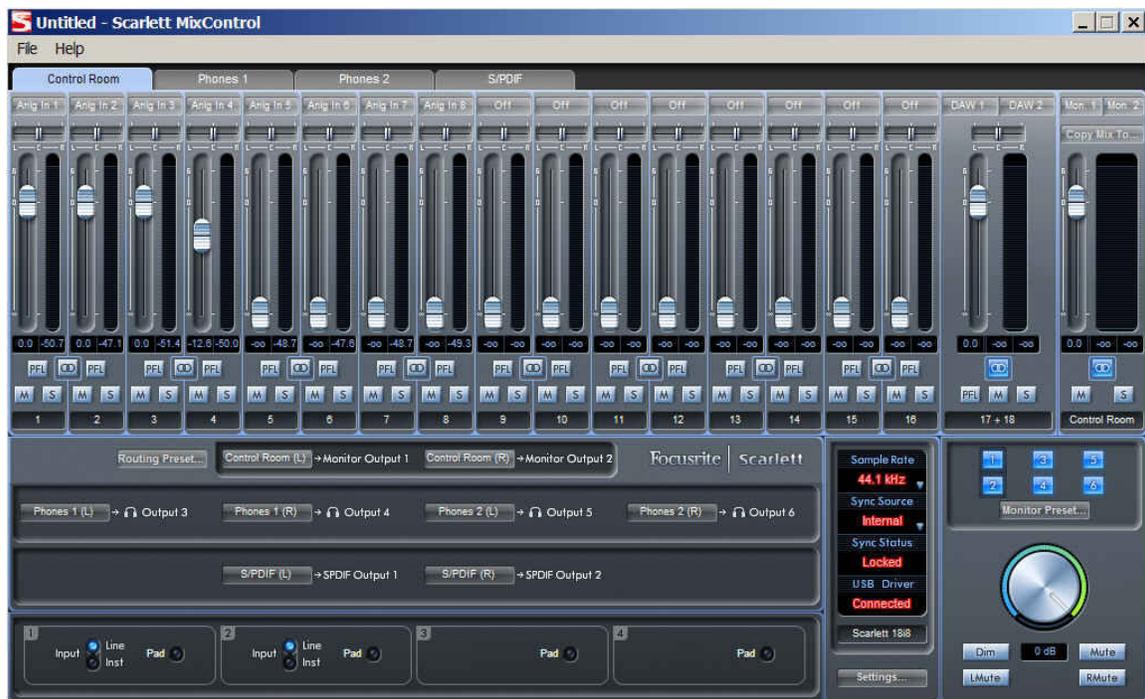
The setup shown above is configured as four stereo mixes, with the four tabs at the top selecting which mix is displayed. This is typical for a tracking session with one mix (the one shown here) going to the control room monitors through Main outputs 1 and 2, and two independent stereo headphone mixes routed to the two front panel headphone jacks. This accounts for all of the available analog outputs, so the fourth mix to the remaining outputs (S/PDIF) might be used to feed another headphone mix through a headphone amplifier that has an S/PDIF input such as the Aphex Headpod 4. For test purposes, I connected a stand-alone CD recorder to make reference CDs of a session.

Let's look at some other possibilities, options, and restrictions. Any input can be selected as the source for any of the eighteen mixer channels. The only rule here is that once a source has been assigned to a channel, it can't be used as the source for another channel. And like with a hardware mixer, the input that's assigned to Channel 1 on one mix is also the source for Channel 1 of all of the other mixes. Note the grayed-out sources in the screen shot to the right. Those have already been assigned to channels.
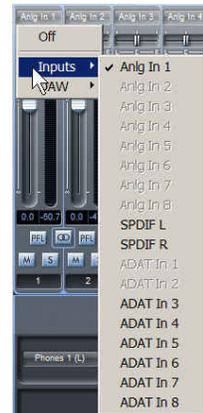
In addition to the eight analog inputs, eight ADAT and two S/PDIF digital streams, mixer channels can get their inputs from DAW streams. It's fairly conventional to set up a stereo mix of previously recorded tracks in the DAW's mixer and use them in a monitor mix as a reference for overdubs. However, up to 8 streams from the DAW are available as sources for Mix Control mixer sources. In Reaper, and I suspect in most DAWs, these appear as their physical names: Monitor1=DAW1, HP1(R)=DAW4, etc. Using DAW streams in the monitor mix offers the possibility of making custom headphone mixes of previously recorded tracks while the engineer in the control room can listen to the a balanced stereo mix that he's set up in the DAW to check playbacks.

This screen shot shows a headphone mix consisting of the analog inputs 1-8 plus six mono DAW outputs. The "main" DAW outputs (DAW 1 and DAW 2) also appear in this mix since we want to use them in the Control Room mix, but in the

Phones 2 mix in this example, their fader is pulled all the way down (or it could simply be muted.

Consecutive input pairs can be configured as a stereo channel with a single fader to adjust the level of both channels simultaneously.

The mix output can be either mono or stereo. We've been looking at four stereo mixes here, but we could have eight mono mixes, each with its own tab, or we could have some stereo (Control Room and Phones 1) and some mono mixes (Phones 2 and Phones 3) as shown above. The limitation is that there are only eight physical outputs so that's the maximum possible number of tabs (and mixes). The screen shot above shows a stereo mix going to the Phones 1 jack, while the Phones 2 mix has been split into two mono mixes which physically appear as the left and right outputs respectively at the Phones 2 jack. This will of course require some adapting or an outboard headphone amplifier to feed both ears, but it's a way to get more headphone feeds when you need them.

Mix Control will rename a mix based on its existing name when a stereo mix is split to two mono mixes, or the two mono mixes are combined to become a stereo mix, but you're free to name any mix as you choose simply by clicking on the mix name below its "master" fader and typing a new name. You could call them John and Marsha if those are the two players using those mixes.

But wait! There's more! A mixer output can be assigned to more than one physical output. If, for example, you want to have a headphone jack in the control room with the same mix as the player in the studio hears you can assign his headphone mix to both headphone jacks. Another useful feature is the ability to copy one mix to any other mix. This is a quick way of setting up two slightly different headphone mixes, or starting the headphone mixes with your control room mix and then making adjustments to suit the players.

The pull-down File menu allows a snapshot of all Mix Control settings to be saved to the host computer as a named file for later recall. Another option on the File menu is Save To Hardware. This allows the current setup to be stored in the interface itself. If the Scarlett is powered up without a computer connection, it will come to life with the saved-to-hardware mix. This is potentially useful should you want to use the 18i8 as a simple mixer for a live show. Your only control over the mix is through the input gain controls but it could work for a simple show.

The panel below the mixer section is the routing matrix. This is where you select the source for each physical output. We've been talking about assigning mixer outputs to physical outputs but there are some other choices. By clicking on a source button in this section, you can bypass the mixer and assign an input or a DAW stream directly to a physical output. This could be a quick way of allowing a player in the studio to hear a playback of your DAW mix.

Control Room Mix to Monitor 1

DAW Mix to Monitor 1
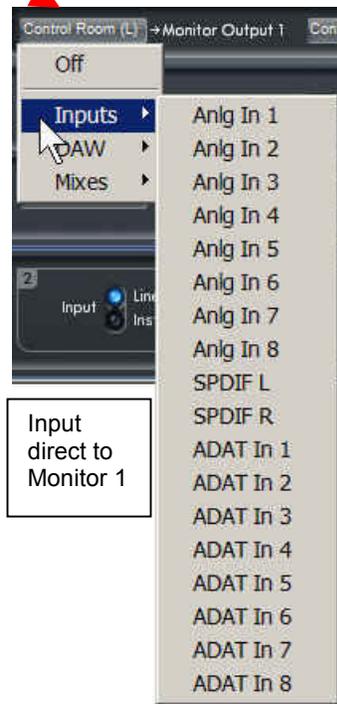
Input direct to Monitor 1

The Monitor Control section in the lower right corner serves a number of functions. The virtual rotary knob will be most useful to the user who has the interface box stashed out of the way and needs a way to control the monitor volume from the computer. The six buttons above it correspond to the six analog outputs and determine which outputs are under control of the knob. When all of the buttons are blue, the knob controls the volume of all analog outputs together. A typical application for this setup is 5.1 surround mixing. Since the physical knobs on the interface box are always active, they can be used to establish the relative levels of the main and surround speakers, then using the software knob to adjust the monitoring level.

Clicking on one or more of the numbered buttons turns it red and mutes that output. Shift-clicking a numbered button (they turn sort of gray) disables the Mix Control volume knob and sends the full signal level to all of the hardware volume controls. For studio use with the 18i8 within reach, I found this to be the most useful setting. The front panel knobs work as expected, and you won't wonder where your monitors went if you inadvertently change the software monitor control knob.

Clicking on the Mute, LMute and RMute buttons mutes all outputs, all left outputs, or all right outputs. The Dim button drops all active monitor levels by 18 dB. The S/PDIF output level is unaffected by anything in this Mix Control section.

The Monitor Preset button drops down a menu with three factory-configured settings of the monitor output control buttons. *Off* mutes all analog outputs. *Mono* mutes all outputs but Output 3 (Headphone 1 left), which you've previously configured as a mono mix. The intent here is that you have a single speaker for checking a mix in mono. This button would be more useful if it did a little extra housekeeping, replacing whatever was assigned to Output 3 with the left plus right sum of the main mix outputs. Due to the panning law that the mixer uses (-6 dB in the center) panning all channels of a stereo mix to the center isn't the same thing. *Stereo* mutes both headphone outputs, leaving only the main left and right outputs active. I expect that few users will find any of these presets useful, and unfortunately, they're not user-configurable.

Let me remind you of why it might be useful to have more inputs and outputs than just one or two mics and a set of speakers or headphones. It's not just about having enough mic inputs to record a full drum kit or the whole band playing live. In the hardware days, no studio was complete without at least a few outboard compressors, equalizers, or effect devices such as reverbs. With today's software plug-ins, you may think you no longer have need for outboard hardware, but the industry doesn't fade away that quickly. In the last several years, there's been a renaissance of old and new signal processing gear built both in traditional

rack mount boxes and in the "500 series" (VPR Alliance) form factor. It's amazing how quickly you can fill an 8-bay 3 rack space housing with mic preamps and compressors, and still want more. But in your console-less studio, you need a way to connect those devices.

During mixdown, you can use Mix Control to route a track playback to directly to one of the hardware outputs on the 18i8, patch that (with a real cable, of course) over to your 500-series rack where you've put together your magic vocal processing chain, then patch the processed signal back to one of the line inputs. Assign that line input to a new DAW track and record the hardware-processed track to be used as part of your in-the-box mix.

Another application is when a singer asks for some reverb in his headphones, when tracking, but you don't want to record the reverb. You could use a reverb plug-in on the track output, put the DAW in monitor mode, and monitor that, but then you'd have all the latency involved in making that round trip through the computer. You could send just the reverb output (as a DAW stream) back to the monitor mix to add to the direct vocal, but it may be delayed long enough to be distracting.

Hardware to the rescue! There are a lot of inexpensive reverb units (you might even have one collecting dust in a closet) that may not be what you want to use in your final mix but will work fine for monitoring while tracking. By assigning a Scarlett mic input directly to an output (or split the output of an outboard preamp), you can use that output to feed the reverb, connect the reverb output to one of the line inputs, and use that line input as a "reverb return" in your monitor mix. The direct mic input is always available to the DAW for recording, so you can record a dry track and process it as needed during mixdown. There's a lot of flexibility here if you keep your wits about you.

The "Save To Hardware" (18i8 internal memory) function was a little iffy. Perhaps I misunderstood how this feature is supposed to work, but my assumption is that once a mix has been saved to hardware, that's how it would be set up when powered up without a computer connected. Further, I expected that setup to be recalled until it was overwritten with a different setup regardless of how the interface had been used in between. At times when powering it up without the computer connected, it came up with the mix that I saved to the internal memory, and at other times it came up with a different mix, the last one that was active before shutting down. Focusrite Tech Support told me that closing Mix Control with the 18i8 still connected to the computer automatically saves the current mix to the hardware unit (this is not stated in the manual), and that if you want to save a mix to the hardware memory, the 18i8 should be disconnected before the mix is changed or Mix Control is closed. I followed that procedure (even wrote it out so I wouldn't forget a step) and found that this didn't work correctly every time I tried it. I'll have to leave this one as a puzzle.

The manual suggests two possible uses for this feature: using it as a simple preset mixer for a live gig, or moving the interface to a different computer and continuing the recording session elsewhere. Another, for whatever reason, is setting up multiple units with the same mix. As for using it at a gig, I'd need more confidence than I have now that it would work as I expected. You can indeed carry it to another computer and there's a fair chance that you can start tracking with the same monitor mix as where you left off, but as soon as you open Mix Control, the mix will switch to the one last used with that computer, or the default mix if this is a new installation.

To be sure you can recall a previously used mix when taking the 18i8 over to your buddy's house to do some more tracking, copy the mix file to a USB thumb drive and copy that to the Mix Control folder on your buddy's computer, then recall it from there. As far as using it "barefoot" for a live gig, well, this is a darn good reason for Focusrite to develop a mobile device app version of Mix Control. At least if it powers up with an unexpected mix, you can plug in your phone and adjust or retrieve a previously saved (to your phone's memory) mix.

Here I am, fourteen pages into the review, and I haven't really griped about anything yet, so let me start now. Eight outputs sounds like a lot, but when you consider that four of them are configured as two stereo headphone jacks and another two come out of a single RCA jack in S/PDIF format, you'll need to get creative if you want to make use of them to add outboard gear to your setup. The S/PDIF I/O is useful only if you have a device which mates with it (some hardware reverbs have digital I/O). Otherwise, you'll need an outboard D/A and A/D converter in order to use those two outputs and inputs for connecting analog gear. You can use the headphone jacks as line outputs, but you'll be giving up a headphone mix, and you'll need an adapter in order to plug into most outboard devices. This is one of the compromises that are part of what makes the 18i8 both compact and relatively inexpensive. Connectors and chassis space are costly and many users will never use them.

In Use

The first step is to install the software. After downloading and running the current Windows driver and Mix Control application installer from the Focusrite web site, when I attempted to open Mix Control, I was greeted with a message that the software was an outdated version for this unit. So much for downloading the latest version from the web site. A quick call to Focusrite Tech Support sent me to their beta software site to download a newer version that includes a firmware update that runs automatically if needed (or maybe always, just in case). This new installer brought the software and hardware onto the same page, Mix Control opened properly, and DAW programs recognized the interface and all of the inputs and outputs. I played back a previous project in Reaper to get the hang of the output designations and then connected some mics to try recording.

A wide assortment of dynamic and condenser mics sounded fine through the Mic inputs, and the instrument DI input was clean and quiet. A short multitrack project with everything close miked sounded excellent. There's no reason to worry about sound quality with the Scarlett 18i8, though having had favorable results when using other models in this series, I didn't expect any surprises here.

One thing that the Scarletts have in common is that the mic input sensitivity is fairly low. I measured and reported some numbers earlier in this review but those numbers really become meaningful after connecting mics and putting them in front of instruments or vocalists. I found that for acoustic guitar, banjo, and mandolin, with the exception of my Neumann KM84s, which are the hottest mics I own, if a mic is placed about a foot away where I prefer it, I had to turn the input gain up to near maximum in order to get peaks higher than about -10 dBFS. When speaking in a normal conversational voice eight inches from a Shure SM57, peaks rarely got above -18 dBFS. Yelling as loud as I could at that distance peaked at around -5 dBFS.

I like to be conservative and leave a reasonable amount of headroom, but unless I expect my narrator to start screaming, I'd like to run my recordings a little hotter than -18 dBFS. 24-bit recording has pretty much eliminated the equivalent of tape hiss that encouraged us to record as hot as we could, but when it's necessary to digitally boost a track by 10 dB or more to get it up to mix level, you'll be amplifying the mic preamp hiss as well. I'm always telling people that it's OK to turn the knobs up all the way if that's what makes it work, but I'd be happier with about 10 dB more gain, particularly when using my Beyer M-160 ribbon mics.

Compared with the 18i8's maximum input sensitivity of -42.5 dBu for 0 dBFS, Focusrite's Forte (admittedly, this is a 2 channel interface that's double the price of the 18i8) gets to full scale with only -62.5 dBu at the mic inputs, a full 20 dB more sensitive than the 18i8. The PreSonus 44VSL, comparable in price and I/O though without the ADAT inputs, clocks in at -57 dBu for 0 dBFS. I have no quibbles about the sound or quiescent noise levels, but it does take a pretty hot source in order to get a good recording level.

I consider metering to be really important for any recording device. Instead of the more conventional LED ladder with indicators at a few levels between "signal present" and "clip," the 18i8 uses a clever red and green illuminated ring surrounding the input gain controls. It's off at low levels, turns green when the digital level reaches -24 dBFS, and remains green until clipping (0 dBFS) is reached, at which point it turns red. The intensity is constant over the full "green level" range, so there's nothing telling you how close you are to clipping. With speech, the green constantly blinks on and off between words (it seems to have a response time of about 0.25 seconds), and with music it remains green almost constantly until

either the music stops or it flashes red, indicating clipping. It does have an orange state (red plus green, I assume), but that appears only very briefly during the transition from red to green when the level drops below clipping.

I'd like to see the green light come on at around -35 or –40 dBFS, have it switch to orange at -10 to -12 dBFS, and flash red 1 or 2 dB before clipping. A quick glance would tell you when you were in the ballpark. Perhaps a firmware update could take care of this. Although there's no scale on the Mix Control meters other than a single mark at -10 dBFS, those meters, or the meters in your DAW program, are far more useful for keeping track of the recording level than the front panel "oops" indicator.
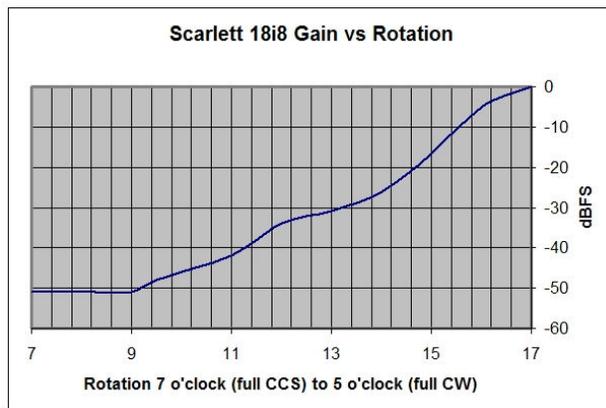
When it comes to setting the input level, the gain knobs much the same as many modern consoles and interfaces. There's practically no gain change at the bottom of the range, between 7 and 9 o'clock, but quite a bit of gain change for a small amount of rotation near the top of the range. The gain changes about 20 dB between 2 and 4 o'clock, then tapers off to about a 5 dB change between 4 and 5 o'clock. It's a little fussy to get a precise level setting, but it does encourage you to be conservative with your record levels.

When it comes down to actually recording music, however, with careful mic placement, it does a fine job. Recording a singing duo with two acoustic guitars, one miked and the other plugged into the DI input, was straightforward. The singers were working close enough to the vocal mics so that full gain wasn't required, the instruments sounded clean, and the pickup was quiet. A four piece string band with three singers needed more mics that the four inputs could accommodate, so I used my Mackie 800R to expand the 18i8 through the ADAT input.

The string band played without headphones. The duo used headphones, one headset plugged into each of the two front panel jacks. I copied my control room mix to the two headphone mixes, then made slight adjustments to each headphone mix to make the singers comfortable. Unless I gave up the control room speakers or added an outboard D/A converter to use the S/PDIF output, I was maxed out for outputs. Fortunately they didn't ask for reverb in their headphones. I did, however, duplicate the setup on the bench and verified that the necessary routing to patch a hardware reverb unit into the monitor mix was available, maintaining the near-zero latency monitoring with some reverb added.

Although the string band was recording in my living room, I pretended that I was doing it on location and used my little flea powered netbook computer (Atom 450 processor) running Reaper under Windows XP, recording the seven tracks for the band plus another stereo mic in the room. I set the ASIO buffer to 4 ms and recorded nine simultaneous tracks with no glitches. I keep reminding myself that while mixing, particularly with plug-ins, can take quite a bit of computer horsepower, tracking doesn't work the computer very hard. When I was reviewing the Scarlett 18i6 I ran the netbook recording 16 tracks for an hour without a single glitch. The 18i8 is sufficiently similar that I didn't attempt to find a limit (which would probably be different on your computer than mine anyway).

The ASIO buffer setting of 4 ms would have resulted in unacceptable latency had I been monitoring the track outputs, but the flexibility of the built-in monitor mixer made this unnecessary. Of course for playback and mixdown, the buffer can be as large as it needs to be. I just settled on 4 ms and only deviated from this setting throughout my use in order to see how it affected the monitor latency.

In the course of my Scarlett 8i6 and 18i6review, on a couple of occasions the USB connection to the computer was lost after operating for a few hours, requiring unplugging and re-plugging to bring the interface it back to life. I've had the 18i8 powered up and connected for days without it taking a nap. Perhaps the current driver is more stable than the version I had for the earlier units, but it may also be a result of a recent Windows tweak that I'll pass along in case you're having this problem with another USB interface (the PreSonus 44VSL also would

get lost). Open the Windows Device Manager and double-click on each USB Root Hub to bring up its Properties window. Select the Power Management tab, then un-check the box that says: "Allow the computer to turn off this device to save power." This option is checked by default in most installations. You can find a lot of solutions posted on the net, but this one, for this problem, sure makes sense to me.
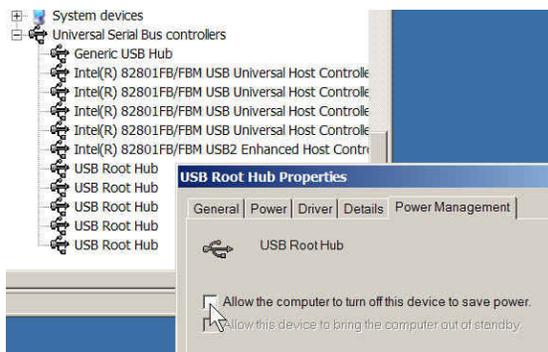
The Wrap Up

With a unit that has as many features as the Focusrite Scarlett 18i8, it may be difficult to decide if it's right for you. For some it will be not quite enough, for some it'll be what you want plus some features that you doubt you'll ever use. I like to think of it as a four input recording interface with benefits. The sound quality from input to output is excellent, and it should meet the recording needs of any solo performer or the most common duo format (2 vocals, 2 instruments). Having two headphone jacks with individual volume controls and the capability of

independent mixes is a bonus when recording a duo or group live. Having a built-in near-zero-latency monitoring mixer simplifies getting good headphone mixes, which is really important to get the best performance from the players in the studio. You'll use the four front panel input connectors most of the time, but those fourteen additional inputs can greatly expand the tracking when you need it, providing you have the gear with line level analog, ADAT optical, and S/PDIF outputs to connect to those inputs.

The four rear panel analog line inputs, in addition to being the obvious place to plug in synthesizers, are ideal for using outboard mic preamps, some of which you may already own and like. The rather low sensitivity of the built-in preamps, which may or may not be a problem for your applications, led me to connect an outboard preamp to the rear line inputs on several occasions. In addition to providing more gain than the Scarlett preamps, an alternate preamp can add a different color to some of your tracks or bring out the best in certain mics. Adding an 8-channel mic preamp with ADAT optical output will easily give you enough inputs to track a full rhythm section with multiple mics on the drums, or perhaps the full band.

The outputs all sound fine, and you can route anything you choose to any of them. However, should you want to use outboard signal processors during mixdown or tracking, although you have plenty of inputs available for signal returns, you may find yourself short of outputs to connect to the outboard processors. Each headphone jack will give you two mono or one stereo unbalanced output at a reasonable line level, but you'll almost certainly need some adapters to get from the headphone jack to the processor input. It's not a big deal, and that setup works fine, but it may not be obvious to a beginner. Squeezing another two TRS jacks on to the rear panel, even if they duplicated the audio streams going to the headphone jacks, would have been a nice touch.

While input level metering is no worse than the single clip LED commonly found on other interfaces in this price class, I wish they had done a better job of providing a more meaningful indication that you're in the right ballpark. As long as you're careful with your initial level setting and keep an occasional eye on the DAW or Mix Control meters when tracking, you can make an excellent sounding recording with the Scarlett 18i8.

I believe the Scarlett 18i8 would find the most utility in the home studio of a songwriter or performing duo. If you do voiceover or narration work, you might want to consider augmenting it with an outboard mic preamp or a pre-preamp such as the Cloudlifter from Cloud Microphones to avoid having to squeeze the last bit of gain out of the Scarlett's mic preamps. If you're thinking about expandability for handling future projects, you might want to consider stepping up to the Scarlett 18i20 for its ten analog outputs on TRS jacks, plus its ADAT output as well as input. Add an inexpensive A/D-D/A converter like the Behringer ADA-8000 and you'll have enough I/O for a full rack of outboard processors.

As for me, I got spoiled by the quality of the Rednet-based mic preamps in the Focusrite Forte. I'm going to wait for Focusrite to come up with a 4 mic input Forte in a tabletop box.

Quick Facts

Street price: $349

- Excellent sound
- Robust Construction
- Four mic inputs
- Two independent headphone outputs
- Expandable to up to 14 additional inputs for big jobs
- Really near-zero latency monitoring
- Flexible routing between inputs, DAW streams and outputs with Mix Control software

- Mic gain fairly low, may be inadequate for distant miking or low output mics
- Input level indicators aren't very informative

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