ALVA Nanoface Review

Mike Rivers - ©2012

“Reduce to the max.” That’s the Nanoface philosophy, and reduce to the max they did. What’s this all about? Well, it becomes apparent when you look at the Nanoface. It’s likely the prettiest device on your desk, really streamlined. There are a few bells and whistles, but they’re all controlled by a single knob. The Nanoface is a no frills USB audio interface that’s intended to do its job and get out of your way. If you want EQ, effects, custom monitor mixes, and signal processing, well, that’s what your DAW program is for.

ALVA may be a new name to you, but if you follow the well known RME line of studio interfacing products, you’ll recognize the functional design as being mighty similar to RME’s Babyface multi-channel desktop audio interface, and indeed there’s some kinship, though not a lot. ALVA’s primary business is making cables, adapters, and rack accessories. A couple of years ago, RME brought them in as a partner as a source of cables for their interfaces, but that’s really as far as the connection goes. The Nanoface isn’t a scaled-down Babyface. It wasn’t designed by RME and employs none of RME’s unique circuitry or software. It’s a different product with a different design specification that mimics (I assume with RME’s blessing) the RME Babyface look and feel, while taking advantage of off-the-shelf components and software..

From The Top

The Nanoface is indeed tiny, a 4 by 6 inch case barely ¾” thick. Weighing in at about 7 ounces, it’s just the ticket for tucking into your laptop computer case when you’re on the road. The case is all plastic, but doggone, it just looks and feels so nice you almost can forget that it’s not burnished steel or aluminum.

Most of the connections are through a breakout squid that attaches to the rear of the box with a 15-pin D-subminiature connector (a DE-15, the same as what’s commonly used for a VGA monitor). The right side of the box has two ¼” phone jacks, one is for headphones, the other is a high impedance unbalanced input for direct connection to an instrument pickup. The rear, in addition to the breakout connector, has a USB connector and a pair of TOSLINK sockets for S/PDIF optical digital input and output.

And now for my first beef: The supplied breakout cable assembly is just 12 inches long, and that includes 2 inches for the chassis connector. There's a 1 inch long ferrite clamp-on EMI choke another inch toward the breakout end from the multipin connector, leaving just five or six inches of flexible cable between the chassis end and the I/O connectors – two female XLRs, four female RCA jacks, and three DIN MIDI connectors. Once you get a few things hooked up, you have quite a pile of connectors too close to where you're most likely to be working. Optional 1 and 3 meter extensions are available, but I think we'd all be happier if the breakout cable was another foot or half meter longer to allow the cables to be dressed more neatly out of the way. That seems like a simple enough request, but to the company, it's more than a couple of feet more copper, they'd have to redesign the retail box to make room for a longer cable! It's a tight fit.

The Nanoface takes its power from the computer's USB port - there's no provision for an external power supply. The supplied USB cable is a "double header" which can connect to a second USB port if additional current is necessary, though the manual states that it's just in case, and that a single port on all the computers with which they've tested the Nanoface provides ample current even when using phantom powered mics.

Inputs and Outputs

This little box packs plenty of I/O.

There are two balanced XLR mic inputs with each input having its own phantom power switch. Input impedance is 6.1 kΩ which seems to be keeping with the modern trend of higher impedance mic inputs than the 2 to 2.5kΩ that was common for many years. Mic input sensitivity is around 10 dB less than what I've been seeing on other interfaces that I've reviewed recently.

At maximum gain, an input level of -39.5 dBu produces 0 dBFS digital output. This is no problem for a rock vocalist, but in practical terms, for spoken word using a Shure SM-57, you'll probably hit peaks no higher than around –15 dBFS with the gain turned up to maximum. Quiescent noise at full gain is a very respectable –76 dBFS RMS, so you shouldn't feel reluctant to crank it if your source or mic calls for full gain to achieve a respectable record level. On the other end of the scale, at minimum gain, the inputs will take +9 dBu before clipping. Common mode rejection is about 63 dB.

The mic preamps follow the contemporary design philosophy that with today's ICs, it's not difficult to make a clean and quiet preamp, so that's what you get. There's no attempt to emulate or create the distortion products present in colored designs of yore. Flavor the clean signal to taste should you want to record anything other than what comes out of the microphone. My only editorial comment about the mic inputs is that I would like to have more gain available as long as it didn't compromise the low noise and distortion characteristics.

The right XLR (Input 2) can be swapped with the ¼" instrument input jack which presents an input impedance of a bit over 1 megohm, It'll take +12 dBu before clipping, and at maximum gain, -30 dBu produces full scale digital level. This is a very quiet DI input. Buzz is imperceptible with the gain set for peaks of about -6 dBFS, and with the strings muted, adding 40 dB of gain to the recording only produced some audible hiss.

Line inputs and outputs are unbalanced RCA jacks. They're what I'd call "near consumer level." Input impedance is 31 kΩ with clipping occurring at +10 dBu at minimum gain. At maximum gain, -17 dBu gets you to 0 dBFS. With the playback level set to maximum, the line outputs put out +6 dBu when playing a full scale recording This is a bit on the wimpy side, but it should be sufficient to drive most powered speakers to a good listening volume.

The gozinta-gozouta count is six by six, You have the two XLRs (Inputs 1-2), two RCA inputs (3-4) and S/PDIF digital (5-6). The instrument DI jack doesn't really count as another input since it replaces Input 2 when it's selected. Outputs are the RCA jacks (Line Out 1-2), the headphone jack (3-4), and S/PDIF (5-6). All are available to the DAW through the driver.

In addition to those six audio inputs and outputs, there's one MIDI input and two MIDI outputs. I use MIDI very little these days (just mics here, no virtual instruments), but it seems to me that they got this backwards. Two MIDI inputs and one output seems like it would be a more useful arrangement. Two inputs could be used to connect a black-and-white keyboard and a DAW control surface, or perhaps or an alternate musical controller like a drum pad set. The single output could feed that sound module or keyboard synth that you can't live without (or for minimum monitoring latency when recording a virtual track), to control an external signal processor, or to send MIDI time code to a hardware sequencer.

The four analog output streams provide some useful monitoring options. Default routing (and as the connectors are labeled), is with the RCA output jacks fed from DAW Outputs 1-2 and typically connected to the studio monitor spekers. The headphone jack is fed from DAW Outputs 3-4. The volume of each of the two output pairs can be controlled independently with the big knob. The digital Outputs 5-6 are available directly from the DAW. With some juggling and a handful of adapters, you can play back a 5.1 surround mix.

You can switch Outputs 1-2 so that they appear both at the RCA jacks and the headphone jack while retaining independent volume of each of the output pairs. For simple recording projects, this is probably the operating mode that you'll prefer. However, for overdubbing in the studio, rather than listening to a balanced mix, the player may want to hear something exaggerated or dropped out of the mix in order to play or sing better. By taking advantage of a DAW's auxiliary busing, you can create and route separate mixes for the player and the engineer.

The headphone jack delivers maximum output power of about 20 mW into a 100Ω load, but even when loaded with 15Ω, THD remains below 0.05%. With my 60Ω Fostex T20 phones, volume is adequate but not ear splitting. Drummers who are a bit hard of hearing may want an outboard headphone amplifier.

That Big Knob

The entire user interface for the Nanoface is contained in a single encoder knob/pushbutton, two LED columns which do double duty as input level meters and indicating the control setting when adjusting a level, and a column of LEDs labeled with the current function of the knob. Pressing the knob ("clicking") cycles through the set of control functions. It requires a pretty hard press, no doubt to avoid changing the function when you intend to make an adjustment.

The operating concept is simple. If you want to adjust the monitor volume, click the knob until the ***Out*** LED is illuminated. Turning the knob then controls the level for Outputs 1-2. Similarly, if you want to adjust the gain of the mic inputs, click the knob until the ***In*** LED illuminates and the knob becomes the Input 1-2 gain control. To adjust the headphone volume, click down to the ***phones*** LED and turn the knob, or .

There are several secondary functions which take a bit of manual dexterity and a good memory or frequent referral to the manual or the cheat sheet card included in the package. For example, as a default, turning the encoder adjusts the gain of the selected pair of inputs together. If you want to adjust only the left channel gain, press and hold the knob and turn it a notch to the left. When you release the button, turning the knob adjusts Input 1 only. The same trick works if you want to adjust the gain of the right channel.

But here's the rub: When you set the control to adjust a single channel, it's up to you to remember that you're in that mode. Not until you turn the knob will you see that only the left channel LEDs change. Also, the press-hold-turn toward the channel you want to adjust isn't a straightforward left-right toggle operation. It you're in the left adjustment mode, the first hold-turn-right operation puts you back into the "both channels" mode, then you need to repeat the switching operation in order to adjust the right channel. From the mode where you're adjusting the right channel only, hold-turn-left gets you back to two-channel control. It can get tedious, and I goofed a few times in use and had to re-adjust a gain that I had previously set.

When you're in the mode where the knob adjusts both channels, if you've offset the two channel gains the gain offset is maintained when adjusting both channels together. This is a useful feature, but there's a potential gotcha. If you use the knob to fade out at the end of a take, when both channels reach zero, it forgets that there was a gain difference between the channels. When you bring the gain back up for the next take, the channel gains will no longer be offset, but will be equal. A fadeout is something that you do intentionally, but should you raise the gain to the point where both channels are maximum, they'll also stick together and no longer be offset when you turn it down.

A couple more things to note. First, the gain steps are fairly large, around 5 dB. With both channels tracking together, I measured a 0.5 dB difference between the channel gains, so if you're a stickler for gain matching, you'll never get them exact. Second, while the Line inputs 3-4 are adjustable, they can't be adjusted individually. Neither of these are deal breakers, but DAWs and their metering to tenths or hundredths of a dB tend to make us fussier than we have a right to be.

All settings are retained with the power off, which could be a blessing or a curse. It's handy if you're going right back to doing what you'd been doing but if you're starting a new project, you may not know exactly how it's going to operate until you try to adjust something. Holding the button down when booting the computer or connecting the USB cable restores the default settings, so it's not difficult to get back to a reference point before starting a new session as long as you remember to do so..

Drivers and Latency

ALVA's partner, RME, is noted for their robust and low latency drivers. Unfortunately, you don't get an RME driver and monitor mix software control panel with the Nanoface, you get a generic ASIO and WDM USB 2.0 audio driver from Ploytec (or as they write it in their logo, πλoytec), a company that supplies drivers for a number of familiar audio interfaces from TASCAM, Alesis, ESI, Akai, Sound Devices, and many more. All of those products have a good reputation, I have no quibbles with the Nanoface here. Windows Installation was a snap.

What's notable over on the Mac side of the universe is that the Nanoface doesn't use the Apple Core Audio driver, but rather has its own driver package that must be installed. The Mac version is for OS-X 10.5 and newer. Apparently there's a compatibility problem with the USB3 interface on new Macs for which there's a temporary work-around for both Lion and Mountain Lion which involves replacing the Mac USB3 driver with the legacy USB2 driver. Neither the problem nor the hack should be considered permanent.

The control panel for ASIO driver latency settings is accessed from the DAW application program. There are six latency settings that have descriptive names but odd numbers of samples, reported as follows:.

Nomenclature	Latency 44.1 kHz		Latency 96 kHz	
	Input	Output	Input	Output
High Speed	115 / 2	282 / 7	115 / 1	557 / 5
Rapid	175 / 4	391 / 8	179 / 1`	781 / 8
Fast	243 / 5	639 / 14	243 / 2	1101 / 11
Normal	307 / 6	1016 / 23	307 / 3	1613 / 16
Relaxed Normal	565 / 12	1642 / 37	563 / 5	2317 / 24
Relaxed	2455 / 25	1075 / 24	2765 / 28	1075 / 11

Things are a little speedier at 2x sample rate, but if you judge an interface by its latency numbers, you'll probably be disappointed. You can't set it for 32 or 64 samples and brag about it on your favorite forum.

What's interesting is that the actual measured latency between analog input and output is exactly the sum of the displayed input and output latency times. With Reaper using the driver-reported driver latency for delay compensation, I recorded the playback of one track on to another track via a patch from Output 1 to Input 2, and coincidence of the two tracks was within 3 samples. I was impressed! So whether or not you like the absolute latency numbers, this driver doesn't lie or mislead. If your DAW has latency compensation when recording, it's bound to work fine.

In addition to the moderately low latency driver, a direct analog path from inputs to the headphone output is available. Double tap on the button with the Output mode selected and inputs 1-2 as well as the playback from the computer are summed to mono at the headphone jackm giving you real zero latency input monitoring. This mode is particularly advantageous when recording a virtual instrument track. If your MIDI controller also has a sound generator built in, even if it's not exactly the sound you want to use in your final mix, by feeding its analog output to one of the analog inputs, you won't be distracted by the typical delay getting MIDI through the driver, playing the sample, and getting it out the D/A converter. When using this mode, of course it's necessary to turn off input monitoring for the DAW track you're recording in order to avoid monitoring the source twice, once through the direct path and again through the driver and audio hardware.

Should you be inclined to fool around, I found that the Nanoface works with the ASIO4ALL driver, which gives you a greater (at least in numbers) range of buffer sizes. I was able to get it to work without glitching using my antiquated Pentium 4 computer running Windows XP with the ASIO4ALL buffer set to 128 samples, but I didn't measure the actual latency with this setup. I'll leave that as an exercise for the latency impaired tweakers.

Pro Tools users need to be cognizant of the buffer settings as that program only wants to see buffer sizes in increments of 64 samples. Since the buffer sizes that the ALVA control panel application reports aren't anywhere close to those incremental sizes, I experimented a bit to see what worked and what didn't. My experiments may deserve a place in the Journal of Irreproducible Results. The first time I tried to change the buffer size from within Pro Tools, The expected control panel with the odd numbers of samples and descriptive text popped up. After switching the ASIO buffer from Normal to Fast, Pro Tools acknowledged the change by telling me that the program needed to be restarted. Upon restart, it told me that it couldn't operate with the selected buffer size and shut down.

I tried a restart, and got the same result. I wasn't able to get Pro Tools running with the Nanoface again until I opened another program, accessed the buffer settings from that program, and changed it back to Normal. Then Pro Tools was happy. On another try, it accepted the change, asked for a restart, didn't like it, but came up again after a second restart, this time showing Fast in the buffer selection window as it should. Once when I tried to change the buffer size in Pro Tools, instead of the expected ALVA control panel, I got one that had only a pull-down buffer size window, with 256 samples being the only choice. I haven't encountered that again. I'm willing to put this down as Pro Tools inconsistencies which, as a new Pro Tools user, I'm finding many. But I'd feel irresponsible if I didn't tell you that using the Nanoface in Pro Tools wasn't entirely smooth sailing. If you set it up and leave it alone, it'll be fine. Once it's working, it keeps working with no surprises.

In Use

Now that I've described most of my annoyances with the Nanoface, I'll tell you that for routine recording and playback, it works just fine. The input gain and output levels are a bit lower than I'd like, but it was always usable, recordings were quiet, and playback was clean. I recorded through a wide variety of condenser and dynamic mics and heard what I expected, however I really would like to have more gain to use with my older ribbon mics. It's necessary to spend some time getting familiar with how the controls work, and still I needed to refer to the manual more often than I normally would. Once when I connected it, it came up with the ***Output*** LED on as usual, but the ***Input*** LED was blinking as well. I must have searched the manual four or five times until I saw that state illustrated on the page that explains the analog input monitoring. I must have inadvertently double pressed the button when it was switched to Input, putting it into that mode without realizing it. No harm done, it was just puzzling, particularly since I don't like to see blinking lights.

The photo of the top panel a couple of pages back is nice and clear, but in dim light, I found it difficult to read the captions for the mode indicator LEDs and hadn't yet memorized them by position. To avoid confusion, I cheated, putting a strip of white tape next to the center column of LEDs and writing labels on it that were easier to read. They don't make these things for old people, I guess. It's not the first piece I've had in the past couple of years with panel legends which have been a bit difficult for me to read.

There are no controls for the digital and MIDI I/O. I just plugged something in and verified that they work, which they do.

I was really wishing for a breakout cable extension to get the bulky connector bundle out of the way. It crowded the desk surface, and when I let the connector bundle dangle over the edge, the weight of all those connectors, which is greater than the weight of the interface itself, tended to drag the Nanoface off the table. Since it's fairly new in the market, perhaps they'll take my suggestion and supply a longer squid in future production runs, but, darn it, it won't just add the cost of another foot of copper, they'll have to redesign the carton since the existing breakout cable just fits. Oh, well.

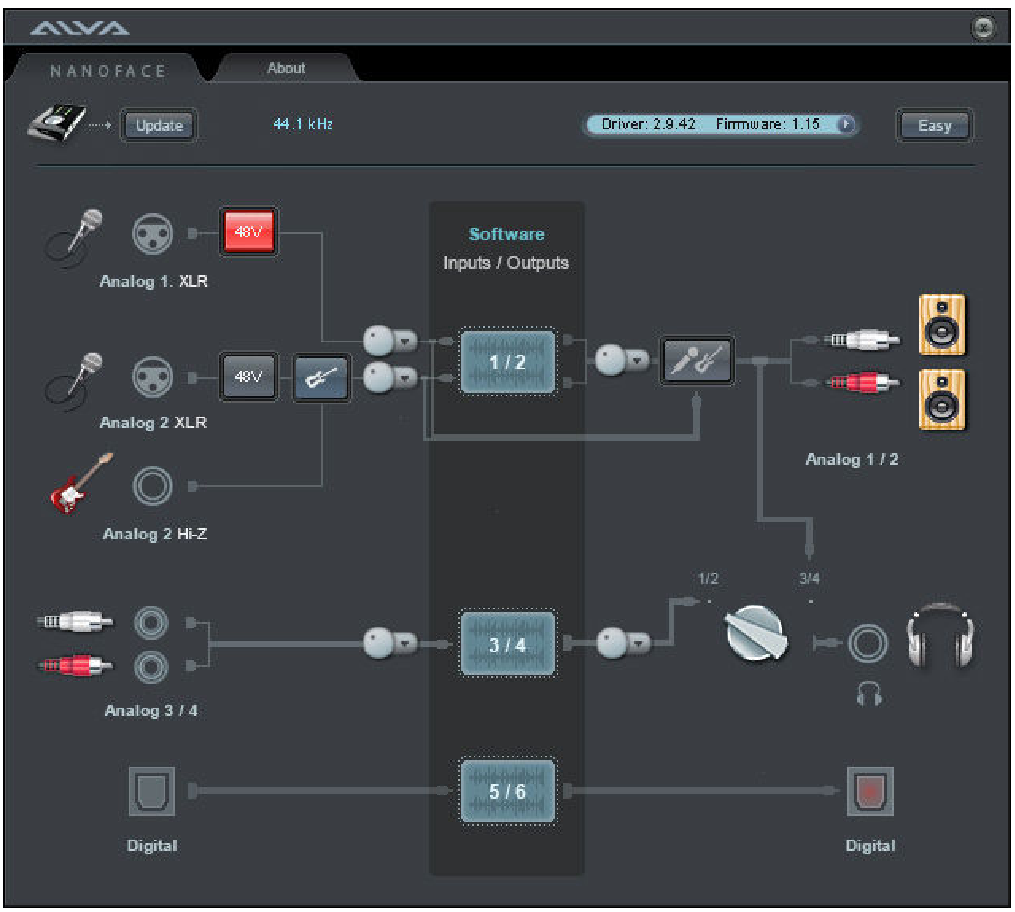
Help Is On The Way

I could continue to rant and complain about the lack of real knobs and indicators, though what bugs me the most about the Nanoface's human interface is that I can't always tell what the knob will do until I turn it. A couple more indicator lights and maybe a button or two would make it all better, but it's the knobs and switches that are often the first to go when designing a product for low cost. While I'm sure the designer had good intentions to streamline the user interface for the novice, it's made operation more difficult for those who have moved beyond point-and-shoot.

When I sent this review to the company for fact checking (it's the polite thing to do) I got a note back from Brian McCall of Synthax, the US distributor, who clued me in to an upcoming software control panel for the Nanoface. Apparently I'm not the only one who finds the simple-to-a-fault user interface a bit difficult to deal with. While the application wasn't ready yet, he shared a couple of screen shots with me and invited me to include them with the review. The application offers both an easy mode and an expert mode. The easy mode allows phantom power switching for inputs 1-2, selects whether the headphones are fed from outputs 1-2 (the normal monitor outputs) or from outputs 3-4 for an indepenent headphone mix. There's also a button for direct input monitoring. .

Easy Mode

The Expert mode offers more controls for input and output, essentially as a block diagram of the interface with all the switch options.



Expert Mode

Synthax was gracious enough to let me hang on to the Nanoface until the control panel is ready and I'll update this review when I've had time to check it out. Stay tuned.

Summary

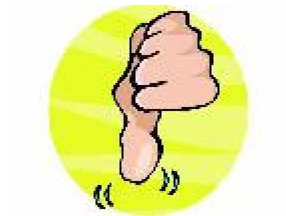
At a list price of $299 (at the moment, US street prices range from $249-$289), The Nanoface has some strong competition. There isn't really a substantial difference in sound quality among interfaces in this price range, it's the features and quirks that set them apart. Sometimes the choices are tough. The Focusrite Scarlett 8i6 and PreSonus AudioBox 44VSL which I've reviewed offer a similar feature set to that of the Nanoface in a larger box but with real knobs to control whatever you need to do. To me, a full set of controls rather than the single knob easily outweighs the size and weight advantage of the Nanoface, but what's important to me may or may not be important to you.

Newcomers to digital audio production should take note, too, that the only software included with the Nanoface is the driver, though I expect that when the control panel is ready, that will come as part of the driver installation program. There's no bundled recording software, not even a "lite" version of one of the big-time programs to get you started. It works fine with Audacity, a free cross-platform recording and editing program, and Reaper, an inexpensive full featured DAW, but you need to acquire the software yourself.

If you're looking for portability and you don't mind the user interface that I find quirky, the ALVA Nanoface is definitely worth a hard look. Additional features such as dedicated line inputs, S/PDIF I/O, and the second pair of outputs capable of delivering an independent cue mix are really more at home in a fixed location than in a hotel room, so that could be a bonus or a "not needed, thanks anyway." But there's certainly no question about the quality and performance. It's as good as any in its price range.

Quick Product Points

- Elegant design, would look great on your coffee table
- Very compact, travels easily
- Sounds fine
- Particularly good instrument DI input
- Stereo Digital I/O through optical S/PDIF



- Not enough knobs and switches, too much button pressing and too much thinking to get around on it (though this should be remedied when the software control panel is delivered)
- Short breakout cable gets in the way
- Mic input gain is fairly low, may not be suitable for some mics or when recording quiet sources

List Price: $299 (USA)

For further info:
http://www.alva-audio.de/nanoface/en_index.php