

HUGH ROBJOHN'S

Alongside a variety of highly regarded audio interfaces and mic preamps, RME manufacture a small number of less well-known, but very high-quality audio and digital-format converters. For example, I reviewed their ADI-2 two-channel converter and ADI-4DD digital interface in *SOS* May 2005 (<http://sosm.ag/rme-adi2-adi4>). I found the latter so useful that I bought the review unit. That I still use it regularly today is a testament to the quality and usefulness of these compact units, which remain in production over a decade later. The cutting edge of converter technology has improved quite significantly in some areas since then, though, and so RME have recently supplemented the range with a brand new 'anniversary' model: the ADI-2 Pro.

Overview

This is a radically enhanced and upgraded product in comparison to its namesake, and it is claimed to be the most versatile converter on the market, with more features than any other device of similar size. The thick A5-sized, spiral-bound User Handbook goes a long way to supporting that assertion all on its own; the English-language section spans a whopping 88 pages with small type and some colour diagrams!

Perhaps the most obvious advance over the older ADI-2 is the addition of a USB 2.0 interface, and the unit also has

RME ADI-2 Pro

A-D/D-A Converter

This modest-looking box is among the best-performing two-way converters available, and boasts a huge array of thoughtful features.

a much sleeker and classier style. The latter is no doubt intended to improve its domestic acceptability as, unusually for RME, this new ADI-2 Pro model is also aimed quite deliberately at both the professional studio user and the technologically savvy hi-fi enthusiast. Another aspect that helps to set it apart from its stablemates is an automatic setup scheme which configures the unit's parameters appropriately, based on which input and output connections are detected, making its basic operation very simple and straightforward indeed.

At its heart, the ADI-2 Pro is a simple two-channel, mastering-quality A-D and D-A converter, and with AKM's 'velvet sound' converter technology, it boasts a dynamic range greater than 120dB, 32-bit processing, and sample rates up to 768kHz. It also supports several DSD (one-bit) formats. Usefully, it also features a second, identical, two-channel D-A section to provide a total of four analogue outputs, this secondary pair being dedicated to an additional independent headphone socket (which can also be used as an extra two-channel unbalanced line output, if required). So it's really a two-in, four-out device as far as the analogue connectivity is concerned.

Mechanically, the unit is half-rack width and 1U high, with a black steel case incorporating side vents — they're needed, as it does run a little warm in use. The unit is powered from a ground-free (double-insulated) universal (100-240 Volts AC) line-lump power supply, with a bayonet-locking coaxial plug, which connects at the rear.

The front panel is silver, with an inlaid dark-blue section around the controls and the small colour screen, with backlit legends for the four control buttons. There are also three rotary encoders, all with push-button actions, and featuring elegant, black-anodised, aluminium knobs. The largest knob, which serves

as the primary digital volume control, features an illuminated ring and adjusts levels in variable increments based on the rotation speed. This makes it very easy to quickly turn things up or down, but also to adjust the output level with fine precision when needed.

Digital connectivity includes AES3 (XLR) and S/PDIF (RCA phono) in and out, connected via a nine-way D-sub socket (a suitable breakout cable is supplied, although the leads are only about six inches long). There's also a pair of optical JIS F-05 light-pipe connectors, and the input auto-detects S/PDIF or ADAT formats, while the output format is switchable in the operating software. However, since this is a two-channel converter, only ADAT channels 1/2 can be accessed (although the appropriate

RME ADI-2 Pro £1349

PROS

- Absolutely superb technical performance and elegant physical design.
- Versatile I/O configurations with an auto-setup feature.
- Stereo or multichannel USB modes.
- Comprehensive DSP facilities.
- Balanced headphone mode.
- Firmware development brings enhanced features.

CONS

- None.

SUMMARY

A surprisingly powerful and cost-effective (if not 'cheap') two-channel A-D/D-A converter, preamp and headphone amplifier, with stunning technical performance, USB interfacing, and comprehensive DSP facilities.



S/MUX formats are employed for high sample rates). If required, though, all eight ADAT channels can be passed through intact to the ADAT output, if the unit is configured in its 'Digital Thru' mode.

As you'd expect, all of the different digital output formats always carry the same source signal, so digital format conversion is possible between the different flavours of I/O. A sample-rate converter is also available and can be deployed for the AES or S/PDIF input, while the internal clocking system is courtesy of RME's latest generation of SteadyClock 'jitter-busting' technology, which boasts 20dB more jitter reduction than the previous incarnation (taking the jitter attenuation to over 50dB).

All of the digital inputs and outputs support sample rates up to 192kHz, and can be routed to or from the class-compliant USB 2.0 interface for computer recording and playback. For those with a penchant for insanely high sample rates, the eight- and 16-times sample-rate options (352.8-768 kHz) are supported by the analogue converters, and thus available for the signal paths between the analogue I/O and USB interface.

Talking of the USB interface, while the unit is fully class-compliant for Mac OS

and iOS, RME's standard MADIFace-series WDM/ASIO drivers are required for use with Windows. In both cases, the USB interface defaults to providing stereo in and out of the computer. However, it can alternatively be configured as a six-input and eight-output USB interface, making use of the digital I/O to access the extra channels. In this multi-channel mode, the stereo analogue, AES3, and S/PDIF inputs are used to provide the six separate input signal paths into the USB interface, while the outputs employ the same six connections supplemented with the second stereo analogue headphone output, to make eight separate output paths. Requiring a combination of analogue and digital interfacing, this obviously isn't the world's most convenient multi-channel interface, but the feature could nonetheless be useful in some situations.

Two analogue inputs to the A-D converter are catered for with rear-panel 'combi' XLR sockets accommodating either balanced or unbalanced line-level signals, while the primary D-A converter provides two analogue outputs on the back panel via both balanced XLRs and unbalanced quarter-inch TS jack sockets. These output signals are also duplicated by the front-panel headphone 1-2 output. The secondary D-A converter's outputs

are only available via the separate front-panel headphone 3-4 output.

In RME's usual way, the maximum analogue signal levels accepted and reproduced by the line inputs and outputs can be configured to one of four preset options (+4, +13, +19, or +24dBu). This useful facility is further enhanced by a digital trim option, allowing a further 6dB adjustment range in 0.5dB steps. The result is the ability to align the unit's 0dBFS digital peak signal level to an analogue signal level anywhere between -2 and +24 dBu — system gain-structure matching couldn't be easier!

Headphone Amp

The two stereo headphone outputs are independently controllable for volume and balance, and while the 1-2 phones output always carries the same source as the main analogue outputs, the second (3-4) headphone source can be separately selected from any of the ADI-2 Pro's inputs. RME have equipped both headphone outputs with their new 'Extreme Power' amplifier technology, which is essentially a high-current driver with selectable Standard and Hi-power modes, providing high-quality and powerful output signals almost regardless of the headphone impedance. Usefully, the technology also serves as a superb

»





The optical input automatically detects the presence of a S/PDIF or ADAT signal, and can accept either (though only two ADAT channels are usable).

» line driver, too, if additional analogue outputs are required. The output level for 3-4 is selected in the configuration software, but for 1-2 it follows the analogue output reference level setting, operating in Standard mode for the +4 and +13 dBu presets, and Hi-power in the +19 and +24 dBu settings.

As the headphone amps are so powerful, RME have incorporated a thoughtful protection system: when headphones are plugged in, the volume automatically resets to a low level and fades back up to the previously used listening level slowly, affording the user time to adjust the setting as necessary — without the shock of getting blasted by a previous loud setting. The two headphone outputs also operate completely independently, with separate memories of the previous volume settings. This is very convenient if, for example, you need to switch regularly between sensitive closed-back headphones for tracking, and less sensitive open-backed headphones for mixing. Just plugging into the appropriate socket will automatically deliver the preset monitoring level. This feature is also very handy when comparing and evaluating different headphones, too.

Interestingly, the two headphone outputs can be configured to operate together as a single balanced stereo headphone output, with the 3-4 socket providing the balanced left output and the 1-2 socket the balanced right output.

A suitable external wiring adaptor is necessary to connect properly-wired balanced headphones to take advantage of this feature, but balanced headphones are all the rage in hi-fi circles, and the idea has some technical merit! In this balanced mode, the Extreme Power amplifiers are capable of delivering almost 20V (+28dBu) into high-impedance phones, and 260mA into low-impedance (32Ω) phones... which is over 2W and quite a lot, even with the most insensitive phones. Even in the low-power mode, I found the headphone amp to be plenty powerful enough with typical studio headphones (AKG K702 and Sennheiser HD650).

It's also worth noting that even if you don't own any balanced headphones, this operating mode is still useful, since it inherently provides a duplicate balanced stereo line-level output on the front panel for channels 1-2.

DSD

In terms of the core converter technology, RME are using the latest 32-bit, 768kHz PCM/DSD converters from AKM's premium 'Verita' chips: a single AK5574 ADC and two AK4490 DACs. In addition to standard PCM conversion, these converters also support DSD signals using the 'DSD over PCM' or 'DoP' format, in which DSD data is packaged to look like PCM for easy transmission over standard digital interfaces. (If you want more information about DoP, try here: <http://dsd-guide.com/dop-open-standard>).

The original DSD format operated at 64 times the CD sample rate, at 2.8MHz (also known as DSD64) but, despite all the marketing claims, this rate is too low and in the following years we've seen updated DSD formats operating at much higher sample rates. Consequently, the ADI-2 Pro supports the 'enhanced' DSD rates of 5.6MHz (DSD128) and 11.2MHz (DSD256). DSD content can also be passed over USB interface via ASIO (either as DoP again, or, with the very latest driver version, the ASIO native format).

One of the intrinsic difficulties with handling DSD audio is applying any form of digital processing, and that's something of an issue in a device like the ADI-2 Pro, which is a digital processor. As a result, there are two DSD operating modes. Standard DSD operations involve converting the one-bit signal internally to a standard PCM format at appropriately high sample rates (DSD64 is processed at 192kHz, while DSD128 is converted to 384kHz, and DSD256 to 768kHz). In that converted-PCM condition digital volume control is provided directly within the DAC, but the normal DSP features remain unavailable. But for absolute purists that want their DSD signals to stay as a one-bit format, there's also a DSD-Direct mode. This converts the DSD signal directly to/from analogue without any interim PCM stages. The inherent disadvantage of this mode, though, is that the DSP and volume control functions are all disabled, along with the headphone outputs (which would »

» become unsafe without a volume control). In effect, the ADI-2 Pro becomes a straight converter, and the analogue input/output levels can only be adjusted in steps by changing the preset operating levels.

DSP

I've already mentioned the digital volume controls, used for all analogue outputs, but the signal processing features go much further than that, courtesy of a 2.17 GigaFLOPS DSP chip working in concert with an FPGA (which handles the signal routing and mixing functions). Amongst the functions this DSP capability offers are peak metering of inputs and outputs, a 30-band spectrum analyser tool, and both simple ± 6 dB bass/treble EQ, and five-band fully-parametric equalisation on all analogue inputs and outputs. The simple bass/treble EQ is intended for overall tonal balance purposes, while the parametric EQ (which can be used as stereo or with unlinked channels) is intended for room correction and similar applications — there are 20 memories or user settings available. Other DSP features include an elegant adjustable Loudness Compensation system, polarity inversion (both overall and per channel), mono summation (providing mono either to both outputs, or mono to the left output only), Mid-Sides processing on the outputs (but not the inputs, strangely), and headphone cross-feed processing based on the 'Bauer binaural' concept — which is intended to give a more loudspeaker-like presentation via headphones.

These signal-processing facilities are all mostly very familiar, but the Loudness Compensation mode warrants further comment, as it's much more usable and beneficial than most incarnations. This system is intended to compensate for the natural reduction in hearing sensitivity to both low- and high-frequency elements at low listening levels. When enabled, the user can set the maximum amounts of boost to the bass and treble regions (up to 10dB), as well as a volume level threshold (anywhere from -20 to -80dB). Below this threshold the maximum permitted boost is applied to the high- and low-frequency regions, but

as the volume is turned up the boost is gradually reduced back to zero. By the time the volume is 20dB louder than the threshold value the frequency response is completely flat again. If set up carefully, this facility makes low-level listening far more balanced and natural, and yet melts away seamlessly when the volume is turned up for more critical listening.

Headphone cross-feed systems are quite common, and aim to reproduce the acoustic crosstalk that happens naturally with loudspeaker monitoring, where both ears hear both speakers, but with some delay and spectral changes. The Bauer binaural cross-feed filtering employed here essentially introduces a short delay at low frequencies (around 200 μ s) below 700Hz in the feed to the opposite channel, as well as something like a 4.5dB roll-off above 700Hz. Four

configured with either a 'Sharp' or 'Slow' response. The Slow option imposes an earlier and gentler roll-off than the more familiar Sharp mode. However, the Slow filters are typically around -1dB at 15kHz and -5dB at 20kHz, so may be perceived as slightly duller or less bright sounding. In the standard (linear-phase) mode, the Slow filter option also exhibits more aliasing artifacts with some material.

The chip manufacturers, AKM, offer a fifth and final filter option called 'Super-slow', although RME has renamed it 'NOS' which is short for 'non-oversampling'. This mode appears not to employ a conventional oversampled digital filter topology at all, instead replicating the earliest D-A converter designs of the 1980s! This form has a very clean impulse response, but suffers rather more out-of-band noise and may reveal increased aliasing artifacts with some material.

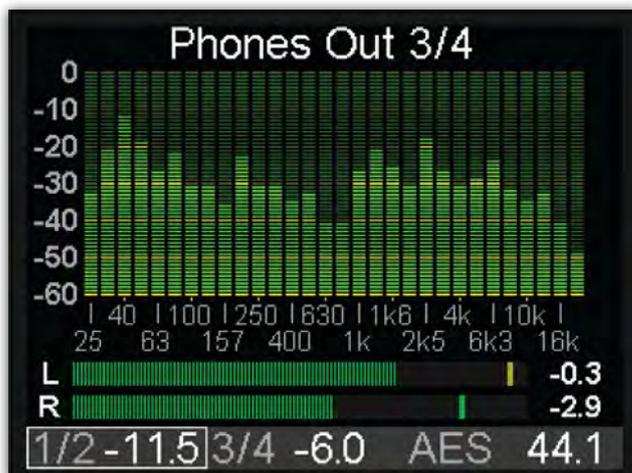
In Use

When the review model arrived it was running with v69 firmware but, when downloading the USB and ASIO drivers for my Windows 7 OS, I noticed that an update (v70) was already available. Installing this was as simple as downloading and running the auto-updating file with the ADI-2 Pro connected via USB. The v70 release notes informed me that alongside some minor bug fixes, inactive menu options

were now greyed out, and the Loudness Compensation mode's lowest threshold value was reduced to -80dB (from -40dB previously), usefully expanding its range of operation by those with particularly sensitive headphones/monitors.

Overall, the ADI-2 Pro's operation is well thought-out, with most functions being selected and adjusted using the three rotary encoders in an obvious and logical way. I found my way around most things without needing to read the manual — although once I *had* read it, I discovered capabilities that I hadn't even realised were on offer!

I think there's still a little room for some further refinements in one or two places, though. For example, when the headphone volume window is displayed, the top small encoder controls the volume and the lower one the balance, but if



The front-panel screen offers a clear view of the various on-board facilities. Pictured here is a frequency analyser.

pre-defined settings are provided, and I found the first two modes gave the most convincing effect, reducing the perceived stereo image width on sounds panned hard left/right, and creating the impression of moving a little further back from the source. (More information can be found on the Bauer system at <http://bs2b.sourceforge.net>.)

Although not strictly a DSP facility, the A-D and D-A converter's anti-alias/reconstruction filtering parameters can be adjusted, too. There are two 'short-delay' options, which are both minimum-phase types (with no impulse response pre-ringing), and two standard linear-phase types (with symmetrical impulse responses and a longer converter latency). Each filter type can be further

Alternatives

I can't think of another two-channel A-D/D-A converter that can match the **ADI-2 Pro's** versatility, and few can match its technical performance. Perhaps the most obvious competition is the **Lynx Hilo USB**, which has similar high-end performance and an excellent headphone amp, but lacks most of the signal processing of the RME and costs considerably more. **Prism Sound's Lyra 2** is a premium USB interface/converter, though this also lacks the DSP and costs more than the RME. Another strong contender is **Antelope Audio's Eclipse 384**, which outperforms the RME slightly in its technical specs, but lacks the DSP facilities and costs almost twice as much.

» the balance is offset and you want to re-centre it, the lower encoder knob has to be turned carefully to find the <C> position. How much nicer would it be if a quick double-click re-centred it instantly — surely something that would be easy to implement in a future firmware update? A similar approach could be implemented to zero the EQ settings — indeed, RME told me just before we went to press that they're considering ways to implement something like this.

One of the clever aspects of the ADI-2 Pro is its four distinct operating modes, the first three of which can be selected automatically, if preferred, depending on whether and which digital inputs are detected. The 'Preamp' mode routes the analogue inputs to the analogue outputs via the A-D and D-A converters, and this configuration can be selected automatically if no digital input or USB connection is detected.

The 'A-D/D-A' mode routes the analogue inputs to all digital outputs, and a digital input to the analogue outputs. This set-up can be selected automatically when a digital input is detected, and that input also becomes the D-A and clock source. (If more than one digital input is present the user must select the required one.) The third option is the 'USB' mode, selected automatically if a USB connection is detected, and it takes priority over the other two auto-detect modes. In the two-channel USB configuration the analogue inputs are routed to the USB, and the USB return appears on all outputs. The multi-channel USB configuration provides six inputs and eight outputs using the full gamut of connectivity, as previously described.

A fourth configuration is also available, but must be selected manually. This 'Digital Thru' mode connects a selected digital input through to all of the digital

and the primary analogue outputs. In all four modes, the 3-4 (headphone) output is always controlled manually, with the signal source being selected from any of the physical inputs and the USB stereo output, when available. The only caveat here is that in the multi-channel USB mode the headphones can audition outputs 1-2 or 3-4, but not 5-6 or 7-8, which seems an unnecessarily restrictive limitation.

Bench Tests

Performing my usual suite of bench tests with an Audio Precision analyser, I measured AES17 dynamic range figures of 124dB for the A-D stage, and 121dB for the D-A section (both A-weighted figures). This A-D figure tops the leader-board of the converters I've measured to date, beating the Lavry AD11 by a single decibel. The D-A performance places it equal fifth in my league, matching Antelope's Eclipse 384 and sitting 4dB below Benchmark's DAC2 HGC and Universal Audio's 2192, and 5dB below the Merging Technologies HAPI. (The current board leader is Apogee's Symphony, at an astonishing 129dB A-weighted.) This is particularly impressive when you consider that all these units cost significantly more than the ADI-2 Pro (although, to be fair, some of them offer more channels of conversion).

These dynamic range figures far exceed the practical dynamic range requirements of real-life recording and playback situations and may therefore appear largely irrelevant. However, the reality is that they provide a useful insight into the product's overall design and construction, because to achieve such high performance requires exquisite attention to detail in every aspect of the power supplies, analogue grounding, clocking, circuit-board layouts, and much more besides. In other words, they provide a strong indication of the overall design and build quality — and in this case it is exemplary!

Further bench tests simply confirm how technically impressive this product is. The low-frequency roll-off for the A-D is -3dB at 1Hz (below the capability of my Audio Precision system) and the D-A converter is DC-coupled. I obtained signal-to-noise measurements of -103.5dB A-weighted for the A-D (ref +4dBu) and -99.5dB A-weighted for the D-A (ref -20dBFS). Crosstalk at 10kHz between channels is better than 94dB for the A-D and 106dB

for the D-A. The most impressive numbers come with the THD+N measurements, which came out at 0.0005 percent ref 0dBFS (and 0.0013 percent ref -20dBFS) for the D-A section, and 0.0002 percent ref +24dBu (and 0.001 percent ref 0dBu) for the A-D stage. In short, every test I performed was unreservedly superb!

Conclusion

This is clearly quite a beast of a converter, both in terms of its technical prowess and its operational flexibility. I've really only outlined here what the ADI-2 Pro can do (and how well it does it!), but the fine detail of all the options and facilities is most impressive, and well worth further exploration on RME's web site. Working through the comprehensive manual revealed extra features and functions that impressed me on almost every page, and highlighted a genuinely inspiring level of attention to detail — which it absolutely should at this kind of price, of course.

I am also greatly encouraged by the useful enhancements that RME are still introducing through firmware updates, no doubt in response to early user feedback, to make an excellent product even better — witness the expanded range of the Loudness mode, for example. Although it feels churlish to be critical with what's already such a highly impressive product, I would like to suggest an M-S decoder option on the inputs (to complement the output encoder), the ability to monitor all eight outputs in the USB multichannel mode, and perhaps some short-cut actions to reset some of the DSP options. (RME are thinking about making the top three buttons reassignable to allow something like this).

However, in the grand scheme of things, the ADI-2 Pro is proof of that old adage, great things come in small packages! And while the asking price is not inconsiderable, in comparison with other two-channel A-D/D-A converters RME's latest offering represents excellent value for money — the asking price is not insignificant, but it delivers similar technical performance to units costing significantly more, while offering greater flexibility and a useful suite of high-quality digital processing. ■■■

£ £1349 including VAT.
T Synthax UK +44 (0)1727 821 870
E info@synthax.co.uk
W www.synthax.co.uk
W www.rme-audio.com
W www.audioag.com

Mix with the best!



"Besides the excellent interviews and fascinating, in-depth recording and mixing articles, I can always depend on Sound On Sound for complete, unbiased reviews of the latest pro-audio gear."

Bob Clearmountain, engineer, producer and mixer, Grammy Award winner (Bruce Springsteen, The Rolling Stones, Paul McCartney, INXS)



"As a professional I admire Sound On Sound as one of the most trusted and credible sources of inspiration and information."

Jack Joseph Puig, mixer, producer, Grammy Award winner (Rolling Stones, U2, Mary J Blige, Black Eyed Peas)

SOUND ON SOUND

The World's Best Recording Technology Magazine



This article was originally published in Sound On Sound magazine, April 2017 edition



Available on the App Store



follow us on Twitter



find us on Facebook



go to the SOS YouTube channel



visit the SOS forum

Subscribe and Save Money!

Visit our subscriptions page at www.soundonsound.com/subscribe for more information on the Sound On Sound App go to: www.soundonsound.com/app

Sound On Sound, Media House, Trafalgar Way, Bar Hill, Cambridge, CB23 8SQ, United Kingdom
Email: subscribe@soundonsound.com Tel: +44 (0) 1954 789888 Fax: +44 (0) 1954 789895

All contents copyright © SOS Publications Group and/or its licensors, 1985-2017. All rights reserved.

The contents of this article are subject to worldwide copyright protection and reproduction in whole or part, whether mechanical or electronic, is expressly forbidden without the prior written consent of the Publishers. Great care has been taken to ensure accuracy in the preparation of this article but neither Sound On Sound Limited nor the publishers can be held responsible for its contents. The views expressed are those of the contributors and not necessarily those of the publishers.