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Compact converter

RME ADI-2 Pro dual-channel A/D D/A converter and headphone amplifier

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This year, RME celebrate their 20th anniversary. For this reason, the team headed by chief developer Matthias Carstens turned to a special project, fundamentally re-engineering their two-channel A/D and D/A converter ADI-2 into a new version named ADI-2 Pro, which is available now. The two devices technically have nothing in common, resembling each other only in name, housing format and channel number. Under the hood, a kind of technology statement is hidden, which RME see as their own reference. It's supposed to be a statement saying 'here we stand after 20 years', so to speak. Alongside aiming at studio users with the ADI-2 Pro, RME also intend to open a door to the Hi-Fi world, where they try to excel with a favorable price and audiophile delicacies - in addition to the sound quality. Of course, whether this attempt will succeed is for Hi-Fi customers to decide. Typically, such developments in our field are closely monitored from their scene and eventually find their way into the respective discussions. We will take a detailed look at the device, but mostly based on studio technology relevance.

While the predecessor was certainly quite successful in the market, it never really acquired a special reputation for its sound. It was considered a solid working animal, which it certainly was. In this respect, the decision to take over the old name and to add the ending 'Pro' might perhaps even be a small obstacle. Or better, an even greater challenge to definitely convince with sound quality. According to the manufacturer, particularly high-quality components were used for the ADI-2 Pro, along the lines of the motto 'money is no object'. Though the final result remains quite affordable - RME simply can't let go of that habit.

Hardware Overview

The device primarily provides the three major components of analogue-to-digital converters (A/D), digital-to-analogue converters (D/A) and headphone amplifiers. This is, however, much too unspecific, because these three functions are accompanied by an extensive arsenal of additional functionality. The analogue inputs and outputs use balanced XLR sockets. For the inputs, a combo-socket is used, which also allows the connection via TRS plugs. The outputs add separate TS jacks (unbalanced). TS and XLR can be used simultaneously, but will be fed by the same output signal. AES3 and S/PDIF connections are available via a breakout cable from a D-sub 9-pin socket. A pair of optical connections for TOSLINK (S/PDIF) and ADAT (channels 1 and 2 only) and the USB-B socket (version 2.0) complete the data connections. Power is provided with an external plug-in power supply featuring a separate supply chord, which in practice is significantly more convenient than a classic 'wall wart', as the latter tends to block adjacent sockets in a multiplug outlet. The low-voltage plug can be locked by turning it inside the socket. Alternatively, a different PSU or a battery supply can be used. On the front panel there is a power button, two headphone jacks, which are described in more detail below, and a color-contrasting display section. The color TFT display of the IPS type (In-Plane-Switching) is large and clear and can be read from any viewing angle. Four labeled function buttons and three rotary encoders with center-push function are available for operation.

Headphone Amplifier

The completely newly developed headphone amplifiers are one central element of the ADI-2 Pro. The manufacturer has named the concept "Extreme Power" and the operating manual raves about the advantages and features quite extensively and with legitimate pride. The core of the system is made up of two output drivers, which can also be shared when needed to provide balanced headphone signals. In this mode, the entire analogue signal path, including the output of the converter chip, switches to balanced operation. This results directly in a lower loading of the drivers, because at equal volume, both channels have to provide 6 dB less level. The dynamic range also increases, because the uncorrelated noise sums up to only 3 dB. The individual output stages are designed to drive low impedance down to 16 ohms or even less reliably, with low distortion and perfect sound even at high volumes. A current limiter prevents damage with combinations of low load impedance and high output levels. The contacts of the two jack sockets are equipped with additional sensors which recognize the plugging and removal of the headphone, and can derive system functions from it. These are, for

example, warning messages when the level is very high, but also direct audio functions, such as the slow fade-in of the signal when headphones are plugged in. This is useful in order for the user to quickly remove the headphones from the ear before the signal is becoming too loud. The sensors are also used as switches. For example, when an overload has been detected, the plug will have to be pulled and reconnected after a moment to reset the system. The amplifiers of the two outputs PH 1/2 and PH 3/4 are functionally identical, but differ in the audio source. While PH 3/4 is a separate sink, which can also be supplied by a separate input, PH 1/2 always reflects the signal which is applied to the analogue main output. For most applications, however, this is not a problem, since, of course, individual settings can still be made. Many users will use all three outputs with the same audio signal anyway. If only one headphone is used and plugged into PH 1/2, the system indicates that one should use PH 3/4 instead or activate the option for two headphones in the setup. Very convenient!

Overview of software and functions

The ADI-2 Pro connects to a computer by USB 2.0. Given the limited number of channels, this is quite sufficient, and RME have shown time and time again how many channels they can transmit through this bottleneck, even at low latency. Available drivers include both Windows ASIO and WDM/WASAPI, while for the Mac the native MacOS system driver can be used. According to the manufacturer, the ASIO driver is based on the MADiface series driver, so one can expect the same performance at low latencies. In addition, the device also operates in Class Compliant mode and therefore directly works with iOS devices. A Texas Instruments DSP was integrated for digital signal processing. The functions rendered in the DSP are intended both for the use as a monitoring converter and for channel conversion in production mode (for example with an analogue chain in mastering). What's more, the manufacturer has integrated features that encourage experimentation and, in some cases, make practical tools for RME's own development department. All functions outside the pure conversion are implemented within the digital domain. There is also, for example, no analogue instance with which the level could be set. This has many advantages, but also some minor disadvantages, as we will see in the DSD section. Most of the functions are available at all PCM sampling rates, up to 768 kHz (albeit no longer simultaneously at this point), however they can not be used in DSD mode. As long as the unit operates in PCM mode, the sound of the converters can be varied by switching between two different anti-aliasing filters on the A/D side (see measurements section) and even five reconstruction filters on the D/A converter. All filter types have their individual advantages and disadvantages. The user will have to find the best compromise for parameters like slope steepness, total latency, ripple in the passband and frequency response - but can also simply use his ears to determine which filter is closest to his sonic preferences. For technicians, switching is extremely lucrative, as the filters can be used to optimize various requirements. As an example, during recording a low latency filter would be selected, while this criterion hardly matters for mastering, postproduction, or Hi-Fi listening.

Filter, loudness and equalizer

The ADI-2 Pro provides three separate instances by which the spectral quality can be influenced. First, the main output and the headphone outputs can be adapted independently of each other with a simple shelving filter for bass and treble. This comes in handy for spontaneous compensation of sound differences in recordings, or to make an adjustment according to taste. Gain is limited to +/- 6 dB per filter. If finer adjustments are required, such as room or loudspeaker correction, a fully parametric five-band equalizer with access to frequency, quality and gain is available, of which the system provides six independent mono instances. They are located in the two input and four output channels and can be used both in stereo as well as individually per channel, allowing independent equalization on both stereo channels. All five bands can be tuned up to 20 kHz, but they have different lower frequency limits (20 Hz, 20 Hz, 150 Hz, 200 Hz, 200 Hz). The graphic implementation is well done, showing each band with a colored section on the frequency response curve. The resulting small rainbow always shows the order of the filters by colors, while the parameters of the selected band are also marked with the appropriate color. This is very neat, despite the small display. Bands 2, 3 and 4 can only function as bell filters, while bands 1 and 5 offer the three types of bell (peak), shelf and resonant high and low pass filters. The third filter function in the ADI-2 Pro is the loudness filter. It is psycho-acoustically based on the standard volume curves and compensates the decreasing sensitivity of the hearing with regard to low and high frequencies at low volumes. A fairly unsuccessful version of such compensation hides behind the loudness button on many Hi-Fi amplifiers. Of course, RME have come up with a much more sophisticated approach, allowing the user to adjust the maximum boost independently for bass and treble, and the level at which this takes place. One would select the lowest level typically used for listening, and set this as reference point. If the volume is now turned up, the filtering is constantly reduced over a range of 20 dB. Volume settings 20 dB above the reference value

will no longer be filtered. A small effort with great effect, because this way the 'Loudness butto' finally works.

One has to keep in mind that each filter action reduces available headroom. If, for example, bass level is raised, the overall output level will need to be compensated/reduced manually. This is especially important if the converter feeds a signal chain for subsequent processing and therefore provides full signal level. What may appear logical to the experienced Studio user may become a pitfall for the home listener, although in most cases the output level will be reduced anyway. The built-in metering provides adequate help for level adjustment.

Metering

In order to make the best use of the space on the small display, the level instrument for the outputs is displayed horizontally. It provides a good overview of the level and can help, for example, in the detection of overloads due to filter operations, because measurement takes place after the level controller. The display has a special feature for the two headphone outputs. If the load at the output is less than 32 ohms, the maximum output level of the amplifiers decreases. In this case, the level display automatically scales and resets the red overload area to the correct position. In addition to the two instruments, there is a peak level hold display as a numerical value. These are sample-peak-based displays; there is no interpolation of any inter-sample peaks (true peak). At sampling rates above 48 kHz, the display is limited to 40 kHz audio bandwidth. Above the level meter, there is a spectrum analyzer with a resolution of 30 bands across a frequency range from 0 Hz to 20 kHz. It is based on the same technology as the in-house metering tool Digicheck and works with a filter bank, instead of the nowadays more commonly used FFT. This has various advantages. On the one hand, there is an aurally compensated distribution of the sound energy across the bands. On the other hand, a high-resolution analysis can be offered with comparatively low system load and latency. The even larger resolution of an FFT could not be displayed in the small display anyway. Although there are no configuration options, one feels immediately informed as a user with Digicheck experience. One analyzer each is available for the main and headphone output as well as the analogue input. Additionally, there are two other pages with signal related information. On a global level map, the levels of the two analogue and digital outputs (AES and S/PDIF) as well as the analogue, AES3 and S/PDIF inputs can be monitored. The USB source is excluded here, so it can only be checked indirectly via the output. A state overview page provides information about the applied signals, sampling rates, status data, the sample rate converter, and the current clock source.

DSD and DoP

The DSD (Direct Stream Digital) format was launched many years ago as a base for Sony's Super Audio CD (SACD) and failed splendidly. Whatever one might think of the technology as such, it has experienced a small renaissance as a digital consumer format without optical data medium in recent years. As a result, many high-class converters are now equipped with DSD capability. The ADI-2 Pro is no exception. However, not all DSD is alike, because there are different versions of the format. The two most widely used ones are DSD64 at approximately 2.8 MHz sampling rate (64x 44.1 kHz) and DSD128 at around 5.6 MHz. Moreover, there are formats such as DSD256 and the corresponding multiples to 48 kHz, but these are rarely used. The signal is no longer transferred over the now rare SDIF-2 interface, but as a so-called DSD over PCM (DoP) transport. The data is not converted here, unlike in the case of DXD, but only gets transmitted in data packets. The individual bits of the DSD bit stream are stored in each PCM sample. Specifically, the lowest 16 of the available 24 bits are used, while the upper eight contain a header for identification. At 192 kHz sampling rate, this results in over 3 million bits per second and channel into which the DSD64 signal fits well with its 2.8 million bits per second and channel. For the transfer of DSD128, DoP uses the 384 kHz PCM format to provide twice the data capacity. The transmission can take place across S/PDIF, AES3 or USB connection in principle. The sending device obviously needs to support DoP and interface connections may also limit the sampling rate. On Windows computers, there is also a WDM driver limitation at 384 kHz. If more is needed, one will have to use ASIO. RME seems to see some potential in DSD, because it was already announced to us that from spring 2017, there will be native DSD support for ASIO, in which DoP is no longer necessary. Nonetheless, DSD does have a few distinct disadvantages. One is that it can not be processed by normal digital systems. This includes the simplest processes such as volume changes. As a result, a DSD to PCM conversion is required for the ADI-2 Pro to be able to adjust the volume in the digital domain. The listener therefore does not get a 'real' DSD signal. Alternatively, the converter offers a DSD Direct mode, in which the DSD signal is converted to analogue without prior conversion to PCM, and therefore without the possibility of volume adjustment. Such a 'PCM bypass' is only offered by very few converter chips, as most chips offer a digital level adjustment and therefore

'secretly' and always convert to PCM. In DSD Direct mode, the user must add an analogue level control behind the ADI-2 Pro. The headphone output 3/4 remains operational with DSD Direct, because here the signal is always converted to PCM, which can be controlled by the DSP. With a lot of effort (DSD formats are not easy to handle from a PC) we were able to make the ADI-2 Pro receive and play back a DSD signal from the Foobar2000 Mediaplayer. In this quick test audio quality was on par with PCM playback.

Operation

The driver of the ADI-2 Pro is based on the MADiface, but does not offer the additional capabilities of the TotalMix software, because the mixer is simply not available here. An exception to this is the metering software Digicheck, which works perfectly well with the ADI-2 Pro. All settings beyond the sampling rate and buffer size of the driver are made via the menu on the device front. Menu operation appears cryptic at first, but one gets used to it quickly. The large rotary encoder sets the volume for the two stereo outputs, which are selected by pressing the rotary encoder. The four labeled menu keys select multiple sub-levels when pressed repeatedly. The upper rotary encoder is used to navigate, the lower one to adjust the selected value. The manual provides an illustration of the menu structure that one should familiarize with to be able to use the device as more than just a 'simple' converter. The menu itself also helps with navigation, as a 1 or 2 is always displayed on the edge when the corresponding rotary encoder takes over a parameter.

Measurements

If you've been following our measurements section in the past, you will have noticed that we had to revise a point from the test report of the RME UFX+ in the last edition, as the result of a legitimate criticism by RME's mastermind Matthias Carstens, which turned into a very exciting discussion about measurements and conclusions. Among other things, it has led to a decision to increase the effort taken in measurements of headphone outputs in the future in order to provide even more meaningful information. This, however, provided that the headphone output of the test device in question is a central feature and not only an 'accessory'. With the ADI-2 Pro this is definitely the case and Matthias Carstens told us in detail about the development effort which went into the device. But more on that later, because the measurements section starts with the D/A converter. Our Audio Precision APx555 Analyzer was connected to the ADI-2 Pro with balanced analogue and optical S/PDIF connections. This has the advantage of electrical isolation on the digital side. The AK4490 from AKM's premium "Verita" series is used as D/A converter chip. The reference level was set to +24 dBu, the measured level was 24.02 dBu. Diagram 1 shows the amplitude and phase frequency responses of the D/A converter at 48 kHz and 96 kHz sampling rate. For these measurements, the reconstruction filter was set to SD Sharp, the default setting. Diagram 2 shows the available filters SD Sharp (blue), SD Slow (red) and NOS (turquoise). The two versions Sharp and Slow are not shown, but equal in amplitude frequency response to the respective SD variants (Short Delay). Only the phase remains constant through to the end, since these are linear phase filters. The D/A converter noise is -93.3 dB RMS (20 Hz to 20 kHz) unweighted. The quasi-peak values remain at an appropriate distance with -82.5 dBu, and the view of the noise spectrum in Diagram 3 confirms a flat spectrum without interferences. This results in a very good dynamic range of 117.2 dB at the output. The distortion behavior of the D/A converter is just as perfect. The measured THD+N at -1 dBFS level is an excellent 0.0003% and falls further below at lower levels. Diagram 4 shows THD+N over the level and Diagram 5 allows a detailed view of the topmost 20 dB. Measurements continue with the headphone amplifier, which we have traditionally tested with a load of 30 Ohms. The maximum output level in the low power mode is +6.94 dBu, which results in a power of 100 mWatt. In high power mode, we could get up to +18 dBu. This results in an impressive output power of 1.26 Watts. We could probably achieve a little bit more, but then you are so close to the current limiter that we had to keep some safety distance. According to the manufacturer, 1.5 Watts at 30 Ohms are possible, thus a decibel more output level. That's loud enough anyway. At the mentioned +18 dBu in the high power mode, the headphone amplifier's noise is at -95.2 dBu RMS unweighted (20 Hz to 20 kHz), resulting in an excellent dynamic range of 113.2 dB. At low power, the noise level is at -108.9 dBu RMS unweighted, resulting in a dynamic range of 115.9 dB. Diagram 6 shows that the noise spectrum is absolutely clean. The distortion at full output level is almost ideal. At 30 Ohm load and +7 dBu THD+N is 0.00034%, and at +18 dBu THD+N is only 0.00029%! Diagram 7 shows the distribution of the THD+N over the level in high power mode. These are all outstanding values with which only a few manufacturers can compete. Being in excitement over these great measurements, we almost forgot about to look at the frequency responses. Diagram 8 shows the amplitude and phase frequency characteristics in low power mode at +4 dBu. At last the A/D converter is measured, starting with the amplitude and phase frequency

response at 48 kHz sampling rate, shown in Diagram 9. The two anti-aliasing filters Sharp (blue) and Slow (red) are shown. While Sharp has a relatively pronounced ripple in the passband, Slow's treble loss already starts in the audio band. When switching to 96 kHz, as shown in Diagram 10, the filtering is removed outside the audible range with the Slow filters. The filter curves are part of the AKM AK5547 converter chip. Matthias Carstens told us that they were also not overly happy with the amount of ripple, but decided to live with it, since the converter is still so good. All further measurements were carried out at 48 kHz and with the Sharp filter. The maximum input level is set to +24 dBu and measured at +24.01 dBu. The noise level unweighted (20 Hz to 20 kHz) is at -121.5 dB RMS, thus casting down all other converters we have measured so far from the throne. Its noise figure is 3 dB lower than that of the Merging Hapi. The noise spectrum is perfect and can be seen in Diagram 11. Partly the low noise floor is a result of using the four-channel chip as a stereo pair. Thus, two inputs per chip are used per audio input, which increases dynamic range by 3 dB. But it's not yet time to put away the superlatives, because the distortion figures are equally spectacular. THD+N is under 0.00022% at full level, thus undercutting the A/D converters of the Crookwood M4, which tested best a few months ago. Diagram 12 shows its equally perfect course over the level. The crosstalk between the two inputs falls below the -110 dB mark above 1 kHz and is documented in Diagram 13. The measurement of the CMRR common mode rejection in Diagram 14 completes the measurements. All in all we can and want to issue an outstanding testimony to the technical data of the ADI-2 Pro. At many important points, the converter takes the reference position, especially with the low noise of the A/D converter. From a measurement viewpoint alone, this is an outstanding object of engineering art.

Listening and hands-on

I was fortunate enough to have the ADI-2 Pro spend quite some time with me, providing extensive opportunity to familiarize myself with it. The device was used both in studio operation and as a pure listening converter, for example while writing for the Studio Magazin, because I mostly sit at my mastering workplace when I do so. The D/A conversion was compared directly with an Antelope Zodiac Gold, which otherwise takes over the casual listening task and also serves as an analogue level adjuster for the DAW monitoring path (i.e. the mastering path) with Merging's Hapi. In a direct comparison, ADI-2 Pro was able to assert itself primarily in the imaging of the signal. While both converter's spectrum is tidy and clean, without any emphasis, the signal from the RME ADI-2 Pro appears even more detached from the speakers, more into the room, with plenty of fine detail of individual instruments. The differences are not huge, we are talking about two very high-quality products, but they are clear. After long listening, I felt I could tell a difference between the various reconstruction filters. The only drawback here is that the last filter that uses lower oversampling (similar to a non-over-sampling filter NOS), causes a loud click when switching. Since the quality of an A/D converter can not be checked without a D/A conversion, I decided to compare a direct path from the Hapi with a complete conversion through the ADI-2 Pro. The result is astonishing, because the differences are really very small. Its result was a minimal change of the stereo width, also depending on the selected filter, however, no loss in resolution and no coloration. Depending on the musical content, I was sometimes unable to distinguish between the two signal paths. The ADI-2 Pro ran at 48 kHz. Of course, it would have been even better here to have an analogue original source, but unfortunately my Telefunken tape machine is currently in maintenance. All in all, we can attest to the excellent neutrality of the converters in the usual PCM operating modes at 44.1 kHz to 96 kHz. Higher sampling rates and DSD were not tested intensively enough for adequate judgment. This is due to the fact that whenever possible, I try to integrate test equipment into real work situations. However, higher sampling rates do not play a practical role in my studio. What really excited me was the headphone amplifier. Using my test headphones Audeze LCD-2, the achievable resolution can only be referred to as acoustic magnifying glass in certain cases. One has to have a real desire to know what is going on in the music to appreciate this kind of detail. For me, fortunately, this is part of my professional life as a mastering engineer. The sharpness of the image is particularly evident when two detailed signals are superimposed. A good example is "In My Life" by Johnny Cash, where Cash's voice is very contoured in the middle and the guitar behind it must compete. The ADI-2 Pro manages to reproduce the two signals neatly despite this struggle, so that one can still follow all the details. It also managed to reveal previously undetected editing errors in the vocal track of my idol. This amount of detail, incidentally, is only partly due to the converter, because we connected a high-quality analogue headphone amplifier to the line output for comparison. The competitor delivers a similarly high sound quality, but has a certain amount of warmth, which sounds very nice, but is not part of the original signal according to my listening experience. The ADI-2 Pro is equipped with a mastering grade headphone amplifier in the best sense. If the converter is used for music enjoyment, then the loudness function is a real enrichment for later evening hours when you want to relax and still listen appropriately. The loudness setup is quickly done. Set the minimum volume level, set it as the reference point, and set the

maximum boost. To my taste a value of 4 or 5 dB is appropriate, but this is of course highly individual and depends on distinct volume perception. You get used to the change very quickly and you will miss the adjustment when hearing linear again. Theoretically, the ADI-2 Pro is equipped with most essential monitoring functions. As it were, operation is designed a bit too much around the menus for hectic everyday life. A high-quality monitoring controller with direct access buttons is still good to have. The 'intelligence of the system' is great, because most of the basic functions, such as the source and clock selection, have an automatic mode and thus almost immediately deliver the desired result. Plug in and go is actually cast into software here. Manual adjustments are only required for more complicated setups and can be adapted in the menu. The only criticism of the device concerns the operating philosophy of the menu in combination with the encoder. Even switching the volume encoder between headphone and main output can sometimes take a moment. The menu structure, even if one has learned it and becomes familiar with navigating it after a few days, can slow down the pace of work here and there. But honestly, that's lamenting at the highest level.

Conclusion

But does that change the bottom line? Absolutely not, because the quality the ADI-2 Pro offers is outstanding and the use as an A/D and D/A converter with excellent headphone amplifier alone justifies the price quoted by German distributor Synthax, which is 1599 Euros, and that is including VAT. The technical quality of the device is beyond doubt - who would even begin to criticize the price! I believe we have not written this much about a small unit with seemingly limited functionality, but effort where effort is due, and we still were not able to look at all the features in detail. Sample rate converter, headphone crossfeed function, stereo width adjustment ... there is still a lot more to be found in this little box! With the ADI-2 Pro, RME re-enter a league where the company was likely underestimated in the past. It will not be easy to establish themselves in the Hi-Fi scene between often dramatically more expensive units with spectacular front panels and tenfold weight. But even studio people are susceptible to a bit of 'bling bling' and RME's 'little one' does not really provide that. It does convince, and I must reiterate that, with outstanding technical quality and excellent sound that can compete with the big converter references and even surpasses them quite effortlessly in a number of aspects. It convinces with a headphone amplifier that elicits detail from some headphones that you never suspected in the signal. Anyone who will employ such criteria as a part of their choice in audio tools will now have a new, hot candidate to consider, which leaves the majority of the competition behind no later than at the point of checkout. Fantastic work!

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Chart 1: amplitude (solid) and phase response (dashed line) of the D/A converter at 48 kHz (red) and 96 kHz (blue) sampling rate

Chart 2 : The reconstruction filter SD Sharp (blue), SD Slow (red) and NOS (turquoise)

Chart 3 : Flawless noise spectrum of the D/A converter at the line output

Chart 4 : THD+N ratio over output level of the D/A converter

Chart 5 : THD+N ratio over the top 20 dB level of the D/A converter

Chart 6 : Flawless noise spectrum of the headphone amplifier in low power mode

Chart 7 : THD+N ratio over the output level of the headphone amplifier in high power mode

Chart 8 : amplitude (solid) and phase response (dashed line) of the headphone amplifier in the low power mode at +4 dBu

Chart 9 : amplitude (solid) and phase response (dashed line) of the anti-aliasing filter of the A/D converter at 48kHz, Slow (red) and Sharp (blue)

Chart 10 : amplitude (solid) and phase response (dashed line) of the anti-aliasing filter Slow at 48 kHz (blue) and 96 kHz (red) sampling rate

Chart 11 : noise spectrum of the A/D converter

Diagram 12 : THD+N ratio over the input level of the A/D converter

Chart 13 : crosstalk between the two channels of stereo input

Chart 14: Common-mode rejection of the analogue input

RME_ADI-2Pro_FK-5_DisplayAnalyzer

Figure 1: Analyzer and Peak Meter for the selected input or output

RME_ADI-2Pro_FK-7_DisplLevel

Figure 2: On the level overview page all inputs and outputs can be monitored (except USB)

Figure 3: This view provides the most important information about the level, filters and level adjustments to the outputs

Figure 4: The basic settings of the headphone outputs also allow switching between low and high power operation

Figure 5: The graph view of the five-band equalizer

Figure 6: In disorders or dangerous condition changes, the device responds with such warnings

Note: to view all the pictures/graphs/measurements please download the original (German) pdf here:

http://www.rme-audio.de/download/StudioMagazin_ADI2Pro_d.pdf